## CONTENTS

### CHAPTER 1

**Cisco Unified CME Commands: A**  
- accept 3  
- access-digit 5  
- addons 6  
- address (voice emergency response location) 7  
- addons 8  
- after-hour exempt 9  
- after-hour login http 11  
- after-hours block pattern 13  
- after-hours date 16  
- after-hours day 18  
- after-hours override-code 20  
- after-hours pstn-prefix 22  
- allow watch 24  
- anonymous block 26  
- application (telephony-service) 27  
- application (voice register global) 28  
- application (voice register pool) 30  
- apply-config 32  
- ata-ivr-pwd 33  
- attempted-registrations size 34  
- attendant-console 36  
- audible-tone 37  
- authen-method 38  
- authenticate (voice register global) 40  
- authentication credential 42
<table>
<thead>
<tr>
<th>Call-Waiting</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-waiting (voice register pool)</td>
<td>168</td>
</tr>
<tr>
<td>call-waiting beep</td>
<td>169</td>
</tr>
<tr>
<td>call-waiting ring</td>
<td>171</td>
</tr>
<tr>
<td>Camera</td>
<td>173</td>
</tr>
<tr>
<td>Capf-auth-str</td>
<td>175</td>
</tr>
<tr>
<td>Capf-server</td>
<td>177</td>
</tr>
<tr>
<td>Cert-enroll-trustpoint</td>
<td>178</td>
</tr>
<tr>
<td>Clear Cti Session</td>
<td>179</td>
</tr>
<tr>
<td>Clear Telephony-Service Conference Hardware</td>
<td>180</td>
</tr>
<tr>
<td>Clear Telephony-Service Ephone-attempted-registrations</td>
<td>181</td>
</tr>
<tr>
<td>Clear Telephony-Service Xml-event-log</td>
<td>182</td>
</tr>
<tr>
<td>Clear Voice Fac Statistics</td>
<td>183</td>
</tr>
<tr>
<td>Clear Voice LpCor Statistics</td>
<td>184</td>
</tr>
<tr>
<td>Clear Voice Moh-Group Statistics</td>
<td>185</td>
</tr>
<tr>
<td>Clear Voice Register Attempted-registrations</td>
<td>186</td>
</tr>
<tr>
<td>Cnf-File</td>
<td>187</td>
</tr>
<tr>
<td>Cnf-File Location</td>
<td>189</td>
</tr>
<tr>
<td>Codec (Ephone)</td>
<td>191</td>
</tr>
<tr>
<td>Codec (Telephony-Service)</td>
<td>194</td>
</tr>
<tr>
<td>Conference (Ephone-Dn)</td>
<td>195</td>
</tr>
<tr>
<td>Conference (Voice Register Template)</td>
<td>197</td>
</tr>
<tr>
<td>Conference Add-mode</td>
<td>198</td>
</tr>
<tr>
<td>Conference Add-mode (Voice Register)</td>
<td>199</td>
</tr>
<tr>
<td>Conference Admin</td>
<td>200</td>
</tr>
<tr>
<td>Conference Admin (Voice Register)</td>
<td>202</td>
</tr>
<tr>
<td>Conference Drop-mode</td>
<td>203</td>
</tr>
<tr>
<td>Conference Drop-mode (Voice Register)</td>
<td>205</td>
</tr>
<tr>
<td>Conference Hardware</td>
<td>207</td>
</tr>
<tr>
<td>Conference Hardware (Voice Register Global)</td>
<td>209</td>
</tr>
<tr>
<td>Conference Max-length</td>
<td>210</td>
</tr>
<tr>
<td>Conference-pattern blocked</td>
<td>211</td>
</tr>
<tr>
<td>Conference Transfer-pattern</td>
<td>212</td>
</tr>
<tr>
<td>Cor (Ephone-Dn)</td>
<td>213</td>
</tr>
<tr>
<td>Cor (Voice Register)</td>
<td>214</td>
</tr>
</tbody>
</table>
corlist 217
create cnf-files 219
create cnf-files (voice-gateway) 220
create profile (voice register global) 221
credentials 222
cti csta mode basic 224
cti message device-id suppress-conversion 225
cti notify 226
citi watch 228
cti-aware 230
ctl-client 231
ctl-service admin 232

CHAPTER 4
Cisco Unified CME Commands: D 233
date-format (telephony-service) 235
date-format (voice register global) 236
debug callmonitor 237
debug capf-server 240
debug ech323 video 242
debug credentials 244
debug cti 246
debug ctl-client 248
debug ephone alarm 249
debug ephone blf 251
debug ephone ccm-compatible 253
debug ephone detail 255
debug ephone error 258
debug ephone extension-assigner 260
debug ephone hfs 262
debug ephone keepalive 264
debug ephone loopback 266
debug ephone lpcor 271
debug ephone message 272
debug ephone mlpp 274
group phone 448

group (voice register global) 450

group (voice register pool) 451

---

CHAPTER 8

Cisco Unified CME Commands: H 453

headset auto-answer line 454

hfs enable 456

hfs home-path 458

hlog-block (voice hunt-group) 460

hold-alert 461

hold-alert (voice register global) 464

hops 465

hops (voice hunt-group) 467

host-id-check 468

hunt-group report url 470

hunt-group statistics write-v2 471

hunt-group logout 473

hunt-group report delay hours 476

hunt-group report every hours 478

hunt-group statistics write-all 480

huntstop (ephone-dn and ephone-dn-template) 483

huntstop (voice register dn) 487

---

CHAPTER 9

Cisco Unified CME Commands: I 489

ica 490

id (voice register pool) 491

import certificate 493

index (lpcor ip-phone) 494

index (lpcor ip-trunk) 496

intercom (ephone-dn) 498

intercom (voice register dn) 501

internal-call 503

ip address trusted authenticate 504

ip address trusted call-block cause 505
Cisco Unified Communications Manager Express Command Reference

Contents

CHAPTER 10

Cisco Unified CME Commands: K 515
  keepalive (ephone and ephone-template) 516
  keepalive (telephony-service) 518
  keepalive (voice register global) 519
  keepalive (voice register session-server) 520
  keepalive (vpn-profile) 521
  keep-conference 522
  keep-conference (voice register) 525
  keygen-retry 527
  keypad-normalize 528
  keyphone 529

CHAPTER 11

Cisco Unified CME Commands: L 531
  label 532
  label (voice register dn) 533
  list (ephone-hunt) 534
  list (voice hunt-group) 537
  live-record 539
  load (telephony-service) 540
  load (voice register global) 544
  load-cfg-file 547
  loc2 548
  location (voice emergency response zone) 549
  log password 551
  log table 552
  logging (voice emergency response settings) 553
  login (telephony-service) 555
  logo (voice register global) 557
  logout-profile 558
<table>
<thead>
<tr>
<th>Command</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>moh (telephony-service)</td>
<td>620</td>
</tr>
<tr>
<td>moh (voice moh-group)</td>
<td>622</td>
</tr>
<tr>
<td>moh-file-buffer</td>
<td>623</td>
</tr>
<tr>
<td>moh-group (ephone-dn)</td>
<td>625</td>
</tr>
<tr>
<td>mtp</td>
<td>626</td>
</tr>
<tr>
<td>mtu (vpn-profile)</td>
<td>628</td>
</tr>
<tr>
<td>multicast moh</td>
<td>629</td>
</tr>
<tr>
<td>mwi (ephone-dn and ephone-dn-template)</td>
<td>631</td>
</tr>
<tr>
<td>mwi (voice register dn)</td>
<td>633</td>
</tr>
<tr>
<td>mwi expires</td>
<td>634</td>
</tr>
<tr>
<td>mwi prefix</td>
<td>635</td>
</tr>
<tr>
<td>mwi qsig</td>
<td>637</td>
</tr>
<tr>
<td>mwi reg-e164</td>
<td>639</td>
</tr>
<tr>
<td>mwi relay</td>
<td>640</td>
</tr>
<tr>
<td>mwi sip</td>
<td>641</td>
</tr>
<tr>
<td>mwi sip-server</td>
<td>643</td>
</tr>
<tr>
<td>mwi stutter (voice register global)</td>
<td>645</td>
</tr>
<tr>
<td>mwi-line</td>
<td>646</td>
</tr>
<tr>
<td>mwi-type</td>
<td>648</td>
</tr>
</tbody>
</table>

**CHAPTER 13**

**Cisco Unified CME Commands: N**

<table>
<thead>
<tr>
<th>Command</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>name (ephone-dn)</td>
<td>652</td>
</tr>
<tr>
<td>name (ephone-hunt)</td>
<td>654</td>
</tr>
<tr>
<td>name (voice emergency response location)</td>
<td>656</td>
</tr>
<tr>
<td>name (voice hunt-group)</td>
<td>657</td>
</tr>
<tr>
<td>name (voice register dn)</td>
<td>659</td>
</tr>
<tr>
<td>network-locale (ephone-template)</td>
<td>660</td>
</tr>
<tr>
<td>network-locale (telephony-service)</td>
<td>662</td>
</tr>
<tr>
<td>network-locale (voice-gateway)</td>
<td>667</td>
</tr>
<tr>
<td>night-service bell</td>
<td>669</td>
</tr>
<tr>
<td>night-service bell (ephone-dn)</td>
<td>671</td>
</tr>
<tr>
<td>night-service code</td>
<td>673</td>
</tr>
<tr>
<td>night-service date</td>
<td>675</td>
</tr>
<tr>
<td>night-service day</td>
<td>677</td>
</tr>
</tbody>
</table>
night-service everyday  679
night-service weekday  681
night-service weekend  683
no-reg  685
no-reg (voice register dn)  687
nte-end-digit-delay  688
ntp-server  690
number (ephone-dn)  691
night-service bell (voice register dn)  694
night-service bell (voice register pool)  696
night-service bell (voice register template)  698
number (voice register dn)  700
number (voice register pool)  702
number (voice user-profile and voice logout-profile)  704
num-buttons  708
num-line  710

CHAPTER 14

Cisco Unified CME Commands: O  711
olsontimezone  712
olsontimezone  714
overlap-signal  716
overwrite-dyn-stats (voice hunt-group)  719

CHAPTER 15

Cisco Unified CME Commands: P  721
paging  724
paging group  727
paging-dn  731
paging-dn (voice register)  734
param  736
param aa-hunt  739
param aa-pilot  741
param call-retry-timer  743
param co-did-max  745
param co-did-min  747
pattern trunk-to-ext no-answer  821
phone-display  823
phone-mode only  824
phone-key-size  825
phoneload  826
phoneload-support  827
phone-redirect-limit (voice register global)  828
phone-ui park-list  829
phone-ui speeddial-fastdial  830
phone-ui voice-hunt-groups  831
pickup-call any-group  832
pickup-group  833
pilot  835
pilot (voice hunt-group)  837
pin  839
pin (voice logout-profile and voice user-profile)  841
pin (voice register pool)  843
port (CAPF-server)  844
preemption reserve timer  845
preemption tone timer (voice MLPP)  846
preemption trunkgroup  847
preemption user  848
preference (ephone-dn)  849
preference (ephone-hunt)  851
preference (voice hunt-group)  853
preference (voice register dn)  855
preference (voice register pool)  857
presence  859
presence call-list  861
presence enable  863
present-call  864
present-call (voice hunt-group)  866
privacy (ephone)  867
privacy (telephony-service)  869
privacy (voice register global) 871
privacy (voice register pool) 873
privacy-button 875
privacy-button (voice register pool) 877
privacy-on-hold 879
privacy-on-hold (voice register global) 880
protocol mode 881
protocol-mode (telephony-service) 883
provision-tag 885

CHAPTER 16
Cisco Unified CME Commands: R 887
refer target dial-peer 888
refer-ood enable 889
reference-pooltype 890
regenerate (ctl-client) 891
register-id 892
registrar server (SIP) 893
reset (ephone) 895
reset (telephony-service) 896
reset (voice logout-profile and voice user-profile) 899
reset (voice register global) 900
reset (voice register pool) 901
reset (voice-gateway) 902
reset tapi 903
restart (ephone) 904
restart (telephony-service) 905
restart (voice register) 907
restart (voice-gateway) 909
ring (ephone-dn) 910
route-code 912
rule (voice translation-rule) 913

CHAPTER 17
Cisco Unified CME Commands: S1 917
sast1 trustpoint 919
show ephone-dn loopback 1044
show ephone-dn paging 1046
show ephone-dn park 1049
show ephone-dn statistics 1050
show ephone-dn summary 1052
show ephone-dn whisper 1054
show ephone-hunt 1056
show ephone-hunt statistics 1063
show fb-its-log 1068
show ip address trusted list 1070
show presence global 1071
show presence subscription 1073
show sdspfarm 1077
show shared-line 1083
show telephony-service admin 1085
show telephony-service all 1087
show telephony-service bulk-speed-dial 1091
show telephony-service conference hardware 1093
show telephony-service directory-entry 1097
show telephony-service ephone 1098
show telephony-service ephone-dn 1101
show telephony-service ephone-dn-template 1103
show telephony-service ephone-template 1104
show telephony-service fac 1107
show telephony-service security-info 1108
show telephony-service tftp-bindings 1109
show telephony-service voice-port 1110
show voice emergency 1112
show voice emergency addresses 1113
show voice emergency all 1114
show voice emergency callers 1116
show voice emergency zone 1117
show voice fac statistics 1118
show voice hunt-group 1119
show voice hunt-group statistics 1124
show voice register all 1128
show voice register credential 1139
show voice register dial-peers 1141
show voice register dialplan 1143
show voice register dn 1145
show voice register global 1148
show voice register hfs 1152
show voice register pool 1153
show voice register pool after-hour-exempt 1161
show voice register pool attempted-registrations 1163
show voice register pool cfa 1165
show voice register pool connected 1167
show voice register pool ip 1170
show voice register pool mac 1172
show voice register pool on-hold 1174
show voice register pool phone-load 1177
show voice register pool registered 1178
show voice register pool remote 1184
show voice register pool ringing 1186
show voice register pool telephone-number 1188
show voice register pool type 1190
show voice register pool type summary 1193
show voice register pool unregistered 1194
show voice register profile 1196
show voice register session-server 1198
show voice register statistics 1200
show voice register template 1204
show voice register tftp-bind 1208
shutdown(telephony-service) 1210
sip-prefix 1211
snr 1212
snr (voice register dn) 1214
snr answer-too-soon 1216
snr answer-too-soon (voice register dn) 1217
snr calling-number local 1218
snr calling-number local (voice register dn) 1219
snr mode 1220
snr ring-stop 1221
snr ring-stop (voice register dn) 1222
softkeys alerting 1223
softkeys connected (voice register template) 1225
softkeys connected 1227
softkeys hold 1230
softkeys idle 1232
softkeys idle (voice register template) 1235
softkeys personal-conf-user (voice register template) 1237
softkeys remote-in-use 1239
softkeys remote-in-use (voice register template) 1240
softkeys ringin (voice register template) 1242
softkeys ringing 1244
softkeys seized 1246
softkeys seized (voice register template) 1248
source-addr 1250
source-address (voice register global) 1251
speed-dial 1253
speed-dial (voice logout-profile and voice user-profile) 1256
speed-dial (voice register pool) 1258
srst dn line-mode 1260
srst dn template 1262
srst ephone description 1263
srst ephone template 1264
srst mode auto-provision 1265
standby username password 1267
statistics collect 1268
statistics collect (voice hunt-group) 1270
subnet 1271
system message 1272
CHAPTER 19  Cisco Unified CME Commands: T  1273
  telephony-service  1275
telnet-support  1279
  template (auto-register)  1280
template (voice register pool)  1282
tftp-path (voice register global)  1283
tftp-server-credentials trustpoint  1284
time-format  1285
time-format (voice register global)  1286
timeout (ephone-hunt)  1287
timeout (voice hunt-group)  1289
timeouts busy  1290
timeouts interdigit (telephony-service)  1291
timeouts interdigit (voice register global)  1292
timeouts night-service-bell  1293
timeouts ringing (telephony-service)  1295
timeouts transfer-recall  1296
timeouts transfer-recall (voice register global)  1298
timeouts transfer-recall (voice register dn)  1300
time-webedit (telephony-service)  1302
time-zone  1303
timezone (voice register global)  1306
transfer max-length  1309
transfer-attended (voice register template)  1310
transfer-blind (voice register template)  1311
transfer-digit-collect  1312
transfer-mode  1314
transfer-park blocked  1316
transfer-pattern (telephony-service)  1318
transfer-pattern blocked  1320
transfer-system  1322
translate (ephone-dn)  1325
translate callback-number  1327
<table>
<thead>
<tr>
<th>Command Type</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>translate-outgoing (voice register pool)</td>
<td>1329</td>
</tr>
<tr>
<td>translation-profile</td>
<td>1331</td>
</tr>
<tr>
<td>translation-profile incoming</td>
<td>1333</td>
</tr>
<tr>
<td>transport (voice register pool-type)</td>
<td>1334</td>
</tr>
<tr>
<td>trunk</td>
<td>1335</td>
</tr>
<tr>
<td>trustpoint (credentials)</td>
<td>1338</td>
</tr>
<tr>
<td>trustpoint-label</td>
<td>1340</td>
</tr>
<tr>
<td>type</td>
<td>1341</td>
</tr>
<tr>
<td>type (voice register dialplan)</td>
<td>1346</td>
</tr>
<tr>
<td>type (voice register pool)</td>
<td>1348</td>
</tr>
<tr>
<td>type (voice-gateway)</td>
<td>1354</td>
</tr>
</tbody>
</table>

**CHAPTER 20**

**Cisco Unified CME Commands: U**

<table>
<thead>
<tr>
<th>Command Type</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>upa</td>
<td>1355</td>
</tr>
<tr>
<td>upgrade (voice register global)</td>
<td>1356</td>
</tr>
<tr>
<td>url (telephony-service)</td>
<td>1357</td>
</tr>
<tr>
<td>url (voice register global)</td>
<td>1359</td>
</tr>
<tr>
<td>url (voice register template)</td>
<td>1362</td>
</tr>
<tr>
<td>url (voice register template)</td>
<td>1364</td>
</tr>
<tr>
<td>url authentication</td>
<td>1366</td>
</tr>
<tr>
<td>url idle</td>
<td>1368</td>
</tr>
<tr>
<td>url services (ephone-template)</td>
<td>1369</td>
</tr>
<tr>
<td>url-button</td>
<td>1371</td>
</tr>
<tr>
<td>url-button (voice-register-template)</td>
<td>1373</td>
</tr>
<tr>
<td>user (voice logout-profile)</td>
<td>1374</td>
</tr>
<tr>
<td>user (voice user-profile)</td>
<td>1376</td>
</tr>
<tr>
<td>user-locale (ephone-template)</td>
<td>1378</td>
</tr>
<tr>
<td>user-locale (telephony-service)</td>
<td>1380</td>
</tr>
<tr>
<td>user-locale (voice register)</td>
<td>1386</td>
</tr>
<tr>
<td>username (ephone)</td>
<td>1389</td>
</tr>
<tr>
<td>username (voice register pool)</td>
<td>1391</td>
</tr>
<tr>
<td>utf8</td>
<td>1393</td>
</tr>
</tbody>
</table>

**CHAPTER 21**

**Cisco Unified CME Commands: V**

<table>
<thead>
<tr>
<th>Command Type</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>vad (voice register pool)</td>
<td>1395</td>
</tr>
<tr>
<td></td>
<td>1397</td>
</tr>
</tbody>
</table>
vad (voice register template) 1398
vca 1399
video 1401
video (ephone) 1403
video (telephony-service) 1404
video screening (voice service sip) 1405
video-bitrate (ephone) 1406
vm-device-id (ephone) 1407
vm-integration 1408
voice class mlpp 1410
voice emergency response location 1411
voice emergency response settings 1413
voice emergency response zone 1415
voice hunt-group 1416
voice-hunt-groups login 1419
voice lpco call-block cause 1421
voice lpco custom 1425
voice lpco enable 1426
voice lpco ip-phone mobility 1427
voice lpco ip-phone subnet 1428
voice lpco ip-trunk subnet incoming 1430
voice lpco policy 1431
voice mlpp 1433
voice moh-group 1434
voice register dialplan 1435
voice register dn 1437
voice register global 1439
voice register pool 1441
voice register pool-type 1443
voice register session-server 1446
voice register template 1448
voice user-profile 1449
voice-class codec (voice register pool) 1451
voice-class mlpp (dial peer) 1453
Cisco Unified CME Commands: A

- accept, on page 3
- access-digit, on page 5
- addons, on page 6
- address (voice emergency response location), on page 7
- addons, on page 8
- after-hour exempt, on page 9
- after-hour login http, on page 11
- after-hours block pattern, on page 13
- after-hours date, on page 16
- after-hours day, on page 18
- after-hours override-code, on page 20
- after-hours pstn-prefix, on page 22
- allow watch, on page 24
- anonymous block, on page 26
- application (telephony-service), on page 27
- application (voice register global), on page 28
- application (voice register pool), on page 30
- apply-config, on page 32
- ata-ivr-pwd, on page 33
- attempted-registrations size, on page 34
- attendant-console, on page 36
- audible-tone, on page 37
- authen-method, on page 38
- authenticate (voice register global), on page 40
- authentication credential, on page 42
- auto assign, on page 44
- auto-assign (auto-register), on page 49
- auto logout, on page 51
- auto logout (voice hunt-group), on page 55
- auto-answer, on page 57
- auto-line, on page 58
- auto-network-detect, on page 60
- auto-register, on page 62
• auto-reg-ephone, on page 64
accept

To allow a logical partitioning class of restriction (LPCOR) policy to accept calls associated with another resource-group, use the `accept` command in LPCOR policy configuration mode. To reject calls associated with a resource group, use the `no` form of this command.

`accept lpcor-group [fac]`

`no accept lpcor-group`

**Syntax Description**

<table>
<thead>
<tr>
<th>lpcor-group</th>
<th>Name of the LPCOR resource group.</th>
</tr>
</thead>
<tbody>
<tr>
<td>fac</td>
<td>Enables forced authorization code for calls from this resource group.</td>
</tr>
</tbody>
</table>

**Command Default**

Calls from other resource groups are rejected.

**Command Modes**

LPCOR policy configuration (cfg-lpcor-policy)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was modified. The fac keyword was added to the accept command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to create a LPCOR policy by specifying the other resource groups from which this resource group can accept calls. If a resource group is not explicitly set to accept with this command, calls associated with that resource-group policy are rejected. You can create one LPCOR policy for each resource group.

If you create a LPCOR policy using the `voice lpcor policy` command and do not explicitly accept any other resource groups by using the `accept` command, that policy blocks all incoming calls associated with any LPCOR resource group other than its own. The fac keyword in the accept command restricts the caller from routing to a destination LPCOR group without entering a valid authorization code.

**Examples**

The following example shows the LPCOR policy for the resource group named sccp_phone_local. It accepts calls from the resource groups analog_phone_local and sip_phone_local but rejects calls from the group named analog_phone_remote because it is not included in the policy.

```plaintext
voice lpcor policy sccp_phone_local
accept analog_phone_local
accept sip_phone_local
```

The following example shows that sccp_phone_local blocks calls that are associated with any other LPCOR policy because its policy does not accept other resource groups.

```plaintext
voice lpcor policy sccp_phone_local
```
The following example shows that the policy local_phone is configured to not accept any calls associated with itself. SIP phone 1 and SCCP phone 2 both belong to the local_phone resource group and its policy prevents them from accepting calls from each other.

voice register pool 1
  lpcor type local
  lpcor incoming local_phone
  lpcor outgoing local_phone
  id mac 0021.A02D.B360
  type 7960
  number 1 dn 1
!
voice lpcor custom
  group 1 local_phone
  group 2 remote_phone
  group 3 analog_phone
!
voice lpcor policy local_phone
  no accept local_phone
  accept analog_phone
!
ephone 2
  lpcor type local
  lpcor incoming local_phone
  lpcor outgoing local_phone
  mac-address 0021.A02D.B580
  type 7960
  button 1:10

The following example shows that the authorization code is required by callers who belong to the LocalUser group and RemoteUser group.

!
voice lpcor policy PSTNTrunk
  service fac
  accept Manager
  accept LocalUser fac
  accept RemoteUser fac
  no accept PSTNTrunk
  no accept IPTrunk

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice lpcor policy</td>
<td>Displays the LPCOR policy for the specified resource group.</td>
<td></td>
</tr>
<tr>
<td>voice lpcor custom</td>
<td>Defines the LPCOR resource groups on the Cisco Unified CME router.</td>
<td></td>
</tr>
<tr>
<td>voice lpcor policy</td>
<td>Creates a LPCOR policy for a resource group.</td>
<td></td>
</tr>
</tbody>
</table>
access-digit

To define the access digit that phone users dial to request a precedence call, use the `access-digit` command in voice MLPP configuration mode. To reset to the default, use the `no` form of this command.

```
access-digit digit
no access-digit
```

**Syntax Description**

<table>
<thead>
<tr>
<th>digit</th>
<th>Single-digit number users dial. Range: 0 to 9. Default: 0.</th>
</tr>
</thead>
</table>

**Command Default**

Access digit is 0.

**Command Modes**

Voice MLPP configuration (config-voice-mlpp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command defines the MLPP access digit that a user must dial when making a precedence call. Phone users request a precedence call by dialing the prefix NP, where N is the preconfigured MLPP access digit and P is the requested precedence level, followed by the phone number.

**Note**

Your domain type must support the access digit that you select. For example, the valid range for the DSN is 2 to 9.

**Examples**

The following example shows the MLPP access digit set to 6:

```
Router(config) # voice mlpp
Router(config-voice-mlpp)# access-digit 6
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>mlpp preemption</td>
<td>Enables preemption capability on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td>preemption trunkgroup</td>
<td>Enables preemption capability on a trunk group.</td>
</tr>
<tr>
<td>preemption user</td>
<td>Enables preemption capability for all supported phones.</td>
</tr>
</tbody>
</table>
addons

To define the maximum number of add-on modules supported by the new Cisco Unified SIP IP phone on Cisco Unified CME, use the `addons` command in voice register pool mode. To remove the add-on modules, use the no form of this command.

```plaintext
addons max-addons
no addons max-addons
```

**Syntax Description**

- `max-addons`: Defines the maximum number of addon modules that can be configured while defining the pool for the phone. Range is 1 to 3.

**Command Default**

The default value of the addons is 0. When the `reference-pooltype` command is configured, the add-on module value of the reference phone is inherited.

**Command Modes**

Voice Register Pool-type Configuration (config-register-pool-type)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(3)M</td>
<td>Cisco SIP CME 10.0</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to define the maximum number of addon modules for a Cisco Unified SIP IP phone on Cisco Unified CME. When you use the no form of this command, the inherited properties of the reference phone take precedence over the default value.

**Examples**

The following example shows how to enter voice register pool configuration mode and define the maximum number of addon modules for a Cisco Unified SIP IP phone on Cisco Unified CME:

```plaintext
Router(config)# voice register pool 1
Router(config-register-pool-type)# type 9900 addon 1 CKEM 2 CKEM 3 CKEM
Router(config-register-pool-type)# id mac 1234.4567.7891
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register pool-type</td>
<td>Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.</td>
</tr>
<tr>
<td>type</td>
<td>Defines a phone type for a SIP phone.</td>
</tr>
</tbody>
</table>
address (voice emergency response location)

To define the civic address for an ERL that is used for the ALI database upload, use the `address` command in voice emergency response location mode. To remove this definition, use the `no` form of the command. This command is optional.

```
address string
no address
```

**Syntax Description**

| string | String (1-247 characters) used to identify an ERL’s civic address. |

**Command Default**
The civic address is not defined.

**Command Modes**
Voice emergency response location configuration (cfg-emrgncy-resp-location)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to create a comma separated text entry of the ERL’s civic address. The address information must be entered to conform with the NENA-2 Data Record specifications or the recommendations by the service provider.

**Examples**

In this example, a civic address is displayed for ERL 60.

```
voice emergency response location 60
subnet 1 209.165.200.224 255.255.0.0
elin 1 4085550100
name Cookies and More Incorporated,
address 1,408,5550100,,11902,,Main Street,Emerald City,CA,Idina Menzel,1,,,,,
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>elin</td>
<td>Specifies a PSTN number that will replace the caller’s extension.</td>
</tr>
<tr>
<td>name</td>
<td>Specifies a string (up to 30 characters) used internally to identify or describe the emergency response location.</td>
</tr>
<tr>
<td>subnet</td>
<td>Defines which IP phones are part of this ERL.</td>
</tr>
</tbody>
</table>
addons

To define the maximum number of add-on modules supported by the new Cisco Unified SIP IP phone on Cisco Unified CME, use the `addons` command in voice register pool mode. To remove the add-on modules, use the no form of this command.

```
addons max-addons
no addons max-addons
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>addons</code></td>
<td>Defines the maximum number of addon modules that can be configured while defining the pool for the phone. Range is 1 to 3.</td>
</tr>
</tbody>
</table>

### Command Default

The default value of the addons is 0. When the `reference-pooltype` command is configured, the add-on module value of the reference phone is inherited.

### Command Modes

Voice Register Pool-type Configuration (config-register-pool-type)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(3)M</td>
<td>Cisco SIP CME 10.0</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use this command to define the maximum number of addon modules for a Cisco Unified SIP IP phone on Cisco Unified CME. When you use the no form of this command, the inherited properties of the reference phone takes precedence over the default value.

### Examples

The following example shows how to enter voice register pool configuration mode and define the maximum number of addon modules for a Cisco Unified SIP IP phone on Cisco Unified CME:

```
Router(config)# voice register pool 1
Router(config-register-pool-type)# type 9900 addon 1 CKEM 2 CKEM 3 CKEM
Router(config-register-pool-type)# id mac 1234.4567.7891
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register pool-type</td>
<td>Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.</td>
</tr>
<tr>
<td>type</td>
<td>Defines a phone type for a SIP phone.</td>
</tr>
</tbody>
</table>
after-hour exempt

To specify that an individual IP phone in Cisco Unified CME does not have any of its outgoing calls blocked even though after-hour call blocking has been enabled, use the `after-hour exempt` command in ephone or ephone-template configuration mode. To remove the exemption, use the `no` form of this command.

`after-hour exempt`

`no after-hour exempt`

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
The SCCP phone is not exempt from call blocking.

**Command Modes**
Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in the ephone-template configuration mode was integrated into Cisco IOS 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to exempt an individual SCCP phone from call blocking and enable the phone user to place outgoing calls regardless of whether the outgoing called number matches the defined pattern of digits during the call blocking periods.

By default, all IP phones in a Cisco Unified CME system are subject to call blocking if the Call Blocking feature is configured.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

**Examples**
The following example shows how to configure this phone so that outgoing calls are not blocked:

```
Router(config)# ephone 23
Router(config-ephone)# mac 00e0.8646.9242
Router(config-ephone)# button 1:33
Router(config-ephone)# after-hour exempt
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>after-hours block pattern</code></td>
<td>Defines a pattern of digits for blocking outgoing calls from IP phones.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td><code>after-hours date</code></td>
<td>Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
<tr>
<td><code>after-hours day</code></td>
<td>Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
</tbody>
</table>
after-hour login http

To unblock an individual IP phone in Cisco Unified CME that is configured for after-hour call blocking, use the **after-hour login http** command in ephone, telephony-service or ephone-template configuration mode. To disable after-hour login http feature, use the no form of the command.

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
The after-hour login http feature is not enabled.

**Command Modes**
Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)
Telephony-service configuration (config-telephony)

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to log in to a phone to unblock the after hour block and enable the phone user to place outgoing calls regardless of whether the outgoing called number matches the defined pattern of digits during the call blocking periods.

When you configure after-hours login http command, you will experience slightly different login behavior compare to the current one. This difference is because the after hours login mechanism is enhanced due to some UI limitation in the current model. By default, after-hours login http is not applied, which mean user will be using the existing after hours login mechanism.

By default, all IP phones in a Cisco Unified CME system are subject to call blocking if the Call Blocking feature is configured.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

**Examples**
The following example shows how to configure this phone with pin login so that outgoing calls are not blocked:

```
Router(config)# ephone 6
Router(config-ephone)# mac 00e0.8646.242
Router(config-ephone)# button 1:33
Router(config-ephone)# Pin 123
Router(config-ephone)# after-hour login http
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>after-hours block pattern</td>
<td>Defines a pattern of digits for blocking outgoing calls from IP phones.</td>
</tr>
<tr>
<td>after-hours date</td>
<td>Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
<tr>
<td>after-hours day</td>
<td>Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
</tbody>
</table>
**after-hours block pattern**

To define a pattern of outgoing digits for Call Blocking from IP phones, use the **after-hours block pattern** command in telephony-service or ephone-template configuration mode. To delete a call-blocking pattern, use the **no** form of this command.

**Syntax Description**

<table>
<thead>
<tr>
<th>syntax</th>
<th>description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>pattern-tag</strong></td>
<td>Identifier for a call-blocking pattern. Up to 100 call-blocking patterns can be defined in separate commands.</td>
</tr>
<tr>
<td><strong>pattern</strong></td>
<td>Outgoing call digits to be matched for blocking, including specific digit patterns and general regular expressions.</td>
</tr>
<tr>
<td><strong>7-24</strong></td>
<td>(Optional) If the 7-24 keyword is specified, the pattern is always blocked, 7 days a week, 24 hours a day. If the 7-24 keyword is not specified, the pattern is blocked during the days and dates that are defined with the <strong>after-hours day</strong> and <strong>after-hours date</strong> commands.</td>
</tr>
</tbody>
</table>

**Command Default**
No pattern is defined.

**Command Modes**
Ephone-template (config-ephone-template)
Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>Support for this command was extended to all SCCP, H.323, SIP, and POTS calls that go through the Cisco Unified CME router, including all incoming calls to the router, except calls from an exempt phone.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was added to ephone-template configuration mode.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was added to ephone-template configuration mode.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
<tr>
<td>15.3(2)T</td>
<td>Cisco Unified CME 9.5</td>
<td>This command was modified to include regular expressions as a value for the <strong>pattern</strong> argument.</td>
</tr>
</tbody>
</table>
Call Blocking on IP phones is defined in the following way. First, one or more patterns of outgoing digits (0-9) are defined using the `after-hours block pattern` command. Next, one or more time periods during which calls that match those patterns are to be blocked are defined using the `after-hours date` or `after-hours day` command or both. By default, all IP phones in a Cisco Unified CME system are restricted during the specified time if at least one pattern and at least one time period are defined.

Before Cisco CME 3.4, Call Blocking is supported on IP phones and on analog phones connected to SCCP-controlled analog telephone adaptors (Cisco ATA) or SCCP-controlled foreign exchange station (FXS) ports. In Cisco CME 3.4 and later, the call-blocking configuration applies to all SCCP, H.323, SIP and POTS calls that go through the Cisco Unified CME router. All incoming calls to the router, except calls from an exempt phone, are also checked against the after-hours configuration.

Individual phones can be exempted from call blocking using the `after-hour exempt` or the `after-hours override-code` command.

Blocked calls return a fast-busy tone to the user for approximately 10 seconds before the call is terminated and the line is returned to on-hook status.

In Cisco Unified CME 9.5 and Cisco Unified SRST 9.5, support for after-hours pattern blocking is extended to regular expression patterns for dial plans on Cisco Unified SIP and Cisco Unified SCCP IP phones. With this support, users can add a combination of fixed dial plans and regular expression-based dial plans.

When a call is initiated after hours, the dialed number is matched against a combination of dial plans. If a match is found, the call is blocked.

To enable regular expression patterns to be included when configuring after-hours pattern blocking, the `after-hours block pattern` command is modified to include regular expressions as a value for the `pattern` argument.

```
Note

The maximum length of a regular expression pattern is 32 for both Cisco Unified SIP and Cisco Unified SCCP IP phones.

For a summary of the basic Cisco IOS regular expression characters and their functions, see the Cisco Regular Expression Pattern Matching Characters section of Terminal Services Configuration Guide.
```

The following example defines pattern 1, which blocks international calls after hours for a Cisco Unified CME system that requires dialing 9 for external calls:

```
Router(config)# telephony-service
Router(config-telephony)# after-hours block pattern 1 9011
```

The following example shows how to configure several after-hours block patterns of regular expressions:

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# telephony-service
Router(config-telephony)# after-hours block pattern 1 ?
WORD Specific block pattern or a regular expression for after-hour block pattern
Router(config-telephony)# after-hours block pattern 1 1234
Router(config-telephony)# after-hours block pattern 2 .T
Router(config-telephony)# after-hours block pattern 3 987654([1-3])+```
Router(config-telephony)# after-hours block pattern 4 98765432[1-9]
Router(config-telephony)# after-hours block pattern 5 98765(432|422|456)

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>after-hour exempt</td>
<td>Specifies that an IP phone does not have any of its outgoing calls blocked although call blocking is defined.</td>
</tr>
<tr>
<td></td>
<td>after-hours date</td>
<td>Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
<tr>
<td></td>
<td>after-hours day</td>
<td>Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
<tr>
<td></td>
<td>after-hours override-code</td>
<td>Specifies that call blocking on an IP phone can be overridden by entering a defined code.</td>
</tr>
<tr>
<td></td>
<td>after-hours pstn-prefix</td>
<td>Specifies that trunk lines on an IP phone are blocked similarly to that configured for nonPSTN lines.</td>
</tr>
<tr>
<td></td>
<td>ephone-template (ephone)</td>
<td>Applies template to a SCCP phone.</td>
</tr>
</tbody>
</table>
after-hours date

To define a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones, use the **after-hours date** command in ephone-template or telephony-service configuration mode. To delete a defined time period, use the **no** form of this command.

**after-hours date** month date start-time stop-time

**no after-hours date** month date

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>month</strong></td>
<td>Abbreviated month. The following abbreviations for month are valid: <strong>jan, feb, mar, apr, may, jun, jul, aug, sep, oct, nov, dec.</strong></td>
</tr>
<tr>
<td><strong>date</strong></td>
<td>Date of the month. Range is from 1 to 31.</td>
</tr>
<tr>
<td><strong>start-time</strong></td>
<td>Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. The stop time that is entered will be the next available time that follows the start time. The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, calls are blocked for the entire 24-hour period on the specified date.</td>
</tr>
</tbody>
</table>

### Command Default

No time period based on date is defined for call blocking.

### Command Modes

- Ephone-template configuration (config-ephone-temp)
- Telephony-service configuration (config-ephone)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was added to ephone-template configuration mode.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was added to ephone-template configuration mode.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use this command to define call blocking that recurs annually on the date specified in the command. Call blocking on IP phones is defined as follows:

- First, one or more patterns of outgoing digits (0-9) are defined using the **after-hours block pattern** command.
• Next, one or more time periods during which calls that match those patterns are to be blocked are defined using the **after-hours date** or **after-hours day** command or both.

By default, all IP phones in a Cisco Unified CME system are restricted during the specified time if at least one pattern and at least one time period are defined. Individual IP phones can be exempted from call blocking using the **after-hour exempt** or **after-hours override-code** commands.

### Examples

The following example defines January 1 as an entire day on which calls that match the pattern specified in the **after-hours block pattern** command are blocked:

```
Router(config)# telephony-service
Router(config-telephony)# after-hours date jan 1 00:00 00:00
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>after-hour exempt</strong></td>
<td>Specifies that an IP phone does not have any of its outgoing calls blocked although call blocking is defined.</td>
</tr>
<tr>
<td><strong>after-hours block pattern</strong></td>
<td>Defines a pattern of digits (0-9) for blocking outgoing calls from IP phones.</td>
</tr>
<tr>
<td><strong>after-hours day</strong></td>
<td>Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
<tr>
<td><strong>after-hours override-code</strong></td>
<td>Specifies that call blocking on an IP phone can be overridden by entering a defined set of digits (0-9).</td>
</tr>
<tr>
<td><strong>after-hours pstn-prefix</strong></td>
<td>Specifies that trunk lines on an IP phone are blocked similarly to that configured for nonPSTN lines.</td>
</tr>
<tr>
<td><strong>ephone-template (ephone)</strong></td>
<td>Applies template to SCCP phone.</td>
</tr>
</tbody>
</table>
after-hours day

To define a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones, use the **after-hours day** command in ephone-template or telephony-service configuration mode. To delete a defined time period, use the **no** form of this command.

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>day</strong></td>
<td>Abbreviated day of the week. The following abbreviations for day of the week are valid: sun, mon, tue, wed, thu, fri, sat.</td>
</tr>
<tr>
<td><strong>start-time</strong> start-time</td>
<td>Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. The stop time that is entered will be the next available time that follows the start time. The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, calls are blocked for the entire 24-hour period on the specified day.</td>
</tr>
</tbody>
</table>

**Command Default**

No time period based on day of the week is defined for call blocking.

**Command Modes**

Ephone-template configuration (config-ephone-template)

Telephony-service configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was added to ephone-template configuration mode.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was added to ephone-template configuration mode.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to define call blocking during the hours between the start time and stop time on the day of the week that is specified in this command. This time period recurs weekly unless it is removed using the **no** form of this command.

Call blocking on IP phones is defined as follows:

- First, one or more patterns of outgoing digits (0-9) are defined using the **after-hours block pattern** command.
• Next, one or more time periods during which calls that match those patterns are to be blocked are defined using the `after-hours date` or `after-hours day` command or both.

By default, all IP phones in a Cisco Unified CME system are restricted during the specified time if at least one pattern and at least one time period are defined. Individual phones can be exempted from call blocking using the `after-hour exempt` or `after-hours override-code` commands.

**Examples**

The following example defines the period from Monday night at 7 p.m. to Tuesday morning at 7 a.m. as an after-hours call-blocking period:

```
Router(config)# telephony-service
Router(config-telephony)# after-hours day mon 19:00 07:00
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>after-hour exempt</code></td>
<td>Specifies that an IP phone does not have any of its outgoing calls blocked although call blocking is defined.</td>
</tr>
<tr>
<td><code>after-hours block pattern</code></td>
<td>Defines a pattern of digits (0-9) for blocking outgoing calls from IP phones.</td>
</tr>
<tr>
<td><code>after-hours date</code></td>
<td>Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
<tr>
<td><code>after-hours override-code</code></td>
<td>Specifies that call blocking on an IP phone can be overridden by entering a defined set of digits (0-9).</td>
</tr>
<tr>
<td><code>after-hours pstn-prefix</code></td>
<td>Specifies that trunk lines on an IP phone are blocked similarly to that configured for nonPSTN lines.</td>
</tr>
<tr>
<td><code>ephone-template (ephone)</code></td>
<td>Applies template to SCCP phone.</td>
</tr>
</tbody>
</table>
after-hours override-code

To specify that a defined blocking pattern can be overridden, use the **after-hours override-code** command in ephone-template or telephony-service configuration mode. To remove the exemption, use the **no** form of this command.

```
after-hours override-code pattern
no after-hours override-code pattern
```

**Syntax Description**

| **pattern** | Specifies the pattern of digits (0-9) that must be dialed by the phone user to override the call blocking rules. The override code is provided to the phone user by the system administrator. |

**Command Default**

No override is defined.

**Command Modes**

- Ephone-template configuration (config-ephone-template)
- Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th><strong>Cisco IOS Release</strong></th>
<th><strong>Cisco Product</strong></th>
<th><strong>Modification</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was added to ephone-template configuration mode.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to allow a phone user to override call blocking rules and enable the phone user to place outgoing calls regardless of whether the outgoing called number matches the defined pattern of digits during the call blocking periods.

By default, all IP phones in a Cisco Unified CME system are subject to call blocking if the Call Blocking feature is configured. By entering the override code as defined by the system administrator, the phone user can override all call blocking rules.

The **after-hours override-code** command, configured by either ephone-template or telephony-service, overrides any global telephony-service call block configuration. If the **after-hour exempt** command is configured, it has priority over the **after-hours override-code** command.

**Examples**

The following example defines a blocking pattern using telephony-service configuration which can be overridden using the override code of 1234:

```
Router(config)# telephony-service
Router(config-telephony)# after-hours block pattern 1 91900
Router(config-telephony)# after-hours day mon 19:00 07:00
Router(config-telephony)# after-hours date Jan 25 00:00 07:00
Router(config-telephony)# after-hours override-code 1234
```
**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>after-hour exempt</strong></td>
<td>Specifies that an IP phone does not have any of its outgoing calls blocked although call blocking is defined.</td>
</tr>
<tr>
<td><strong>after-hours block pattern</strong></td>
<td>Defines a pattern of digits (0-9) for blocking outgoing calls from IP phones.</td>
</tr>
<tr>
<td><strong>after-hours date</strong></td>
<td>Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
<tr>
<td><strong>after-hours day</strong></td>
<td>Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
<tr>
<td><strong>after-hours pstn-prefix</strong></td>
<td>Specifies that trunk lines on an IP phone are blocked similarly to that configured for non PSTN lines.</td>
</tr>
<tr>
<td><strong>ephone-template (ephone)</strong></td>
<td>Applies a template to an ephone.</td>
</tr>
</tbody>
</table>
after-hours pstn-prefix

To specify that all configured blocking patterns apply to trunk or PSTN lines, use the `after-hours pstn-prefix` command in telephony-service configuration mode. To delete call blocking configuration for PSTN lines, use the `no` form of this command.

```
after-hours pstn-prefix  tag  pattern
no after-hours pstn-prefix  tag  pattern
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>tag</code></td>
<td>Identifier for a PSTN call-blocking pattern. Up to 4 call-blocking patterns can be defined in separate commands.</td>
</tr>
<tr>
<td><code>pattern</code></td>
<td>Outgoing call digits (0-9) to be matched for PSTN blocking.</td>
</tr>
</tbody>
</table>

**Command Default**

No pattern is defined.

**Command Modes**

Telephony-service configuration

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was added to ephone-template configuration mode.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `after-hours pstn-prefix` command to indicate that the after-hours call blocking patterns are configured to include one or more PSTN access digits that are normally dialed by phone users using regular ephone-dn lines. For example, the patterns are configured with a leading digit 9 to correspond to the conventional “dial 9 for outside line.” The `after-hours pstn-prefix` command instructs the system to skip over the PSTN prefix digits (in the blocking patterns) for calls that are dialed directly to the PSTN on ephone-dns that are configured using the trunk feature. These lines do not require the user to dial a PSTN access code (for example, 9) because they are configured to directly connect to the PSTN FXO ports. For example, a user of a regular ephone-dn would dial 9-1-900-xxx-xxxx, whereas a user on a trunk ephone-dn would omit the leading 9 and dial only 1-900-xxx-xxxx. The blocking pattern would be configured as 91900 to restrict calls on regular ephone-dn lines, and this pattern would be interpreted as 1900 on ephone-dns configured using the trunk feature. If you do not specify the `after-hours pstn-prefix` command, then no blocking is performed on calls dialed on trunk ephone-dn lines.

Call blocking on IP phones is defined as follows:

- First, one or more patterns of outgoing digits (0-9) are defined using the `after-hours block pattern` command.
- Next, one or more time periods during which calls that match those patterns are to be blocked are defined using the `after-hours date`, the `after-hours day`, or both commands.
By default, all IP phones in a Cisco Unified CME system are restricted during the specified time if at least one pattern and at least one time period are defined.

Blocked calls return a fast-busy tone to the user for approximately 10 seconds before the call is terminated and the line is returned to on-hook status.

Examples

The following example defines a blocking pattern using telephony-service configuration which is applied to a PSTN line:

Router(config)# telephony-service
Router(config-telephony)# after-hours block pattern 1 91900
Router(config-telephony)# after-hours pstn-prefix 1 9
Router(config-telephony)# after-hours day mon 19:00 07:00
Router(config-telephony)# after-hours date Jan 25 00:00 07:00

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>after-hour exempt</td>
<td>Specifies that an IP phone does not have any of its outgoing calls blocked although call blocking is defined.</td>
</tr>
<tr>
<td>after-hours block pattern</td>
<td>Defines a pattern of digits (0-9) for blocking outgoing calls from IP phones.</td>
</tr>
<tr>
<td>after-hours date</td>
<td>Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
<tr>
<td>after-hours day</td>
<td>Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
<tr>
<td>after-hours override-code</td>
<td>Specifies that call blocking on an IP phone can be overridden by entering a defined series of digits (0-9).</td>
</tr>
</tbody>
</table>
allow watch

To allow a directory number on a phone registered to Cisco Unified CME to be watched in a presence service, use the allow watch command in ephone-dn, ephone-dn-template, or voice register dn configuration mode. To reset to the default condition, use the no form of this command.

allow watch
no allow watch

Syntax Description

This command has no arguments or keywords.

Command Default

Watching of the phone line is disabled.

Command Modes

Ephone-dn configuration (config-ephone)
Ephone-dn-template configuration (config-ephone-dn-template)
Voice register dn configuration (config-register-dn)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

This command controls whether a phone line associated with a directory number can be watched as part of a presence service. The directory number is enabled as a presence entity that can be watched by internal and external watchers. Presence service must be enabled on Cisco Unified CME. Another phone, acting as a watcher, can monitor the status of this phone line when the blf-speed-dial or presence call-list command is enabled for that phone.

If you use an ephone-dn template to apply a command to a directory number and you also use the same command in ephone-dn configuration mode, the value that you set in ephone-dn configuration mode has priority over the ephone-dn template configuration.

Examples

The following example shows that the extension associated with voice register dn 2 can be watched by the phone associated with voice register pool 1:

```
Router(config)# voice register dn 2
Router(config-register-dn)# number 2102
Router(config-register-dn)# allow watch
Router(config)# voice register pool 1
Router(config-register-pool)# id mac 0015.6247.EF90
Router(config-register-pool)# type 7971
Router(config-register-pool)# number 1 dn 2
Router(config-register-pool)# blf-speed-dial 1 2102 label 2102
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>blf-speed-dial</td>
<td>Enables Busy Lamp Field (BLF) monitoring for a speed-dial number on a phone registered to Cisco Unified CME.</td>
</tr>
<tr>
<td></td>
<td>presence</td>
<td>Enables presence service and enters presence configuration mode.</td>
</tr>
<tr>
<td></td>
<td>presence call-list</td>
<td>Enables BLF monitoring for call lists and directories on phones registered to Cisco Unified CME.</td>
</tr>
<tr>
<td></td>
<td>presence enable</td>
<td>Allows the router to accept incoming presence requests.</td>
</tr>
<tr>
<td></td>
<td>show presence global</td>
<td>Displays configuration information about the presence service.</td>
</tr>
<tr>
<td></td>
<td>show presence subscription</td>
<td>Displays information about active presence subscriptions.</td>
</tr>
</tbody>
</table>
anonymous block

To enable anonymous call blocking in a SIP phone template, use the `anonymous block` command in voice register template configuration mode. To return to the default, use the `no` form of this command.

```plaintext
anonymous block
no anonymous block
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Disabled

**Command Modes**
Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command blocks incoming calls in which the caller is not identified. To apply a template to a SIP phone, use the `template` command in voice register pool configuration mode.

**Examples**
The following example shows how to set anonymous call blocking in template 1:

```plaintext
Router(config)# voice register template 1
Router(config-register-temp)# anonymous block
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>caller-id block (voice register template)</code></td>
<td>Enables caller-ID blocking for outbound calls from a SIP phone.</td>
</tr>
<tr>
<td><code>template (voice register pool)</code></td>
<td>Applies a template to a SIP phone.</td>
</tr>
</tbody>
</table>
application (telephony-service)

To select a session-level application for all extensions (ephone-dns) in a Cisco Unified CME, use the `application` command in telephony-service configuration mode. To disable use of an application for all extensions, use the `no` form of this command.

```
application application-name
no application
```

Syntax Description

| `application-name` | Interactive voice response (IVR) application name. |

Command Default

No application is selected for all extensions.

Command Modes

Telephony-service configuration (config-telephony)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

Use this command to assign a Tool Command Language (Tcl) IVR application to all extensions served by the CME router.

Use the `show call application voice summary` command to display a list of applications.

Examples

The following example sets the IVR application for all phones:

```
Router(config)# telephony-service
Router(config-telephony)# application TCL IVR
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>show call application voice summary</code></td>
<td>Displays information about voice applications.</td>
</tr>
</tbody>
</table>
application (voice register global)

To select the session-level application for all dial peers associated with Session Initiation Protocol (SIP) phones, use the application command in voice register global configuration mode. To disable use of the application, use the no form of this command.

```
application application-name
no application
```

**Syntax Description**

- `application-name`: Interactive voice response (IVR) application name.

**Command Default**

Default application on router

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>Cisco SIP SRST 3.4</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

During Cisco Unified CallManager Express (Cisco Unified CME) or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) registration, a dial peer is created for each SIP phone and that dial peer includes the default session application. The application command allows you to change the default application for all dial peers associated with the Cisco SIP IP phones, if desired. The applied application (or TCL IVR script) must support call redirection. Use the show call application voice summary command to display a list of applications.

The application command in voice register pool configuration mode takes precedence over this command in voice register global configuration mode.

**Note**

Configure the id (voice register pool) command before any other voice register pool commands, including the application command. The id command identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.

**Examples**

The following example shows how to set the Tcl IVR application globally for all SIP phones:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# application sipapp2
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application (dial-peer)</td>
<td>Enables a specific application on a dial peer.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>----------------------------------------------------------------------------</td>
</tr>
<tr>
<td><strong>application (voice register pool)</strong></td>
<td>Selects the session-level application for the dial peer associated an individual SIP phone in a Cisco Unified CME environment or for a group of phones in a Cisco Unified SIP SRST environment.</td>
</tr>
<tr>
<td><strong>id (voice register pool)</strong></td>
<td>Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.</td>
</tr>
<tr>
<td><strong>mode (voice register global)</strong></td>
<td>Enables the mode for provisioning SIP phones in a Cisco Unified CME system.</td>
</tr>
<tr>
<td><strong>show call application voice summary</strong></td>
<td>Displays information about voice applications.</td>
</tr>
<tr>
<td><strong>show dial-peer voice</strong></td>
<td>Displays information for dial peers.</td>
</tr>
<tr>
<td><strong>voice register pool</strong></td>
<td>Enters voice register pool configuration mode for SIP phones.</td>
</tr>
</tbody>
</table>
application (voice register pool)

To select the session-level application for the dial peer associated with an individual Session Initiation Protocol (SIP) phone in a Cisco Unified CallManager Express (Cisco Unified CME) environment or for a group of phones in a Cisco Unified SIP Survivable Remote Site Telephony (SRST) environment, use the application command in voice register pool configuration mode. To disable use of the application, use the no form of this command.

```
application application-name
no application
```

**Syntax Description**

- **application-name**: Name of the selected interactive voice response (IVR) application name.

**Command Default**

Default application on router

**Command Modes**

Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was added to Cisco CME.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

During Cisco Unified CME or Cisco Unified SIP SRST registration, a dial peer is created for each SIP phone and that dial peer includes the default session application. The application command allows you to change the default application for all dial peers associated with the Cisco SIP IP phones, if desired. The applied application (or TCL IVR script) must support call redirection. Use the show call application voice summary command to display a list of applications.

The application command in voice register pool configuration mode takes precedence over this command in voice register global configuration mode.

**Note**

Configure the id (voice register pool) command before any other voice register pool commands, including the application command. The id command identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.

**Examples**

The following example shows how to set the IVR application for the SIP phone specified by the voice register pool command:

```
Router(config)# voice register pool 1
Router(config-register-pool) application sipapp2
```
The following partial sample output from the `show running-config` command shows that voice register pool 1 has been set up to use the SIP.app application:

```
voice register pool 1
id network 172.16.0.0 mask 255.255.0.0
application SIP.app
voice-class codec 1
```

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>application (dial-peer)</strong></td>
<td>Enables a specific application on a dial peer.</td>
<td></td>
</tr>
<tr>
<td><strong>application (voice register global)</strong></td>
<td>Selects the session-level application for all dial peers associated with SIP phones.</td>
<td></td>
</tr>
<tr>
<td><strong>id (voice register pool)</strong></td>
<td>Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.</td>
<td></td>
</tr>
<tr>
<td><strong>mode (voice register global)</strong></td>
<td>Enables the mode for provisioning SIP phones in a Cisco Unified CME system.</td>
<td></td>
</tr>
<tr>
<td><strong>show call application voice summary</strong></td>
<td>Displays information about voice applications.</td>
<td></td>
</tr>
<tr>
<td><strong>show dial-peer voice</strong></td>
<td>Displays information for dial peers.</td>
<td></td>
</tr>
</tbody>
</table>
apply-config

To dynamically apply the phone configuration on Cisco Unified SIP IP phones 8961, 9951, and 9971, without restarting the phones, use the apply-config command in voice register global and voice register pool configuration modes.

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Apply-config is not enabled by default.

**Command Modes**

Voice register global
Voice register pool

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(4)M</td>
<td>Cisco Unified CME 8.6</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to dynamically apply the phone configuration on Cisco Unified SIP IP phones 8961, 9951, and 9971. Once you configure the apply-config command, you are not required to restart the phone. The phone restarts by itself or dynamically applies the changes to the phone configuration without restarting.

**Examples**

The following example shows the apply-config command configured in voice register pool 5:

Router# configure terminal
Router(config)# voice register pool 5
Router(config-register-pool)# apply-config

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>camera</td>
<td>Enables USB camera capability on Cisco Unified IP Phones 9951 and 9971</td>
</tr>
<tr>
<td>video</td>
<td>Enables video capability on Cisco Unified SIP IP Phones 9951 and 9971</td>
</tr>
</tbody>
</table>
ata-ivr-pwd

To define a password to access interactive voice response (IVR) and change the default phone settings on Cisco Analog Telephone Adaptors, use the ata-ivr-pwd command in voice register pool configuration mode. To return to the default, use the no form of the command.

**ata-ivr-pwd** [0|6] *password*
**no ata-ivr-pwd**

**Syntax Description**

- **password**: Four-digit string to be used as password to access IVR. Password string must contain numbers 0 to 9.
  The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

**Command Default**

No valid password is set.

**Command Modes**

Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>Unified CME 9.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a</td>
<td>Unified CME 12.6</td>
<td>The command was enhanced for password encryption. Password policy is not enforced, as the command supports a four-digit password.</td>
</tr>
</tbody>
</table>

**Examples**

The following example shows how 1234 is defined as the password to access IVR on Cisco ATA-187:

```plaintext
voice service voip
allow-connections sip to sip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
voice register pool 11
ata-ivr-pwd 1234
id mac 93FE.12D8.2301
session-transport tcp
type ATA-187
number 1 dn 33
username ata112 password cisco
codec g711ulaw
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode and creates a pool configuration for Cisco Unified SIP IP phones in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
attempted-registrations size

To set the size of the table that shows a number of attempted-registrations, use the attempted-registrations command in voice register global mode. To set the size of attempted-registrations table to its default value, use the no form of this command.

```
attempted-registrations size size
no attempted-registrations size
```

**Syntax Description**

- **size**: Number of entries in attempted registrations table. Size range from 0 to 50.

**Command Default**

The default size for attempted registration table is 10.

**Command Modes**

voice register global

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td></td>
<td>Cisco Unified SRST 8.1</td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to define the size of the table that stores information related to SIP phones that attempt to register with Cisco Unified CME or Cisco Unified SRST and fail. The default size of an attempted registration table is 10. The minimum size of attempted registration table is 0. Use the attempted-registration size 0 when you do not wish to store any information about phones attempting to register with the Cisco Unified CME or Cisco Unified SRST and fail. The maximum size of attempted registration table is 50.

When the current number of entries in the table is more than the new size that is being configured, system prompts the user for the following confirmation, “This will remove x old entries from the table. Proceed? Yes/No?”. The default user confirmation is “No”. Where “x” represents the number of entries that will be deleted. The old entries are classified on basis of the time-stamp of the latest register attempt made by the phone.

During rollback, the user confirmation is not sought and the target configuration is applied. If the current number of entries in the table is more than the default value of the table size, then entries in excess of the default table size are cleared before reverting to the target table size.

For example, if the configured table size is 40 and there are currently 35 entries in the table, any change in the size of the attempted registration table during rollback removes 25 oldest entries leaving only the default (10) entries before making the table size equal to the size in target configuration.

**Examples**

The following example shows attempted-registrations size:

```
Router# conf t
Router(config)# voice register global
Router(config-register-global)# attempted-registrations size 15
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>clear voice register attempted-registrations</td>
<td>Allows to delete entries in attempted-registration table.</td>
</tr>
<tr>
<td>show voice register attempted-registrations</td>
<td>Displays details of phones that attempted to register and failed.</td>
</tr>
</tbody>
</table>
attendant-console

To specify the phone number of the MLPP attendant-console service, use the `attendant-console` command in voice MLPP configuration mode. To revert to the default, use the `no` form of this command.

`attendant-console number redirect-timer seconds`

`no attendant-console`

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>number</code></td>
<td>Pilot number of the MLPP attendant-console service, such as the Cisco Unified CME basic automatic call distribution (B-ACD) and auto-attendant (AA) service.</td>
</tr>
<tr>
<td><code>seconds</code></td>
<td>Number of seconds that a call rings before being redirected to the attendant-console service. Range: 10 to 60.</td>
</tr>
</tbody>
</table>

**Command Default**

MLPP call is not diverted to an attendant-console service.

**Command Modes**

Voice MLPP configuration (config-voice-mlpp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>—</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables Cisco Unified CME to divert all unanswered precedence calls above Routine to the specified target number after the specified period of time. This target directory number typically specifies the pilot number of the attendant-console service that is used as a destination of last resort for forwarded MLPP calls.

**Examples**

The following example shows that any MLPP call that is not answered after 30 seconds is redirected to extension 81005, which is the extension of the BACD queue.

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# attendant-console 81005 redirect-timer 30
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>access-digit</td>
<td>Defines the access digit that phone users dial to request a precedence call.</td>
</tr>
<tr>
<td>mlpp preemption</td>
<td>Enables preemption capability on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td>service</td>
<td>Associates a dial peer with an auto-attendant (AA) service.</td>
</tr>
</tbody>
</table>
audible-tone

To configure audible tones to indicate successful join or unjoin and login or logout from any hunt group, use the **audible-tone** command in ephone or ephone-template configuration mode. To revert to the default behavior of not playing any audible tone, use the **no** form of this command.

**Syntax Description**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>audible-tone</td>
<td>This command has no arguments or keywords.</td>
</tr>
<tr>
<td>no audible-tone</td>
<td>By default, this feature is disabled.</td>
</tr>
</tbody>
</table>

**Command Modes**

- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the **audible-tone** command to set an audible tone to confirm successful join or unjoin and log in or log out from specific hunt groups.

**Example**

The following example shows how to configure audible tone in ephone configuration mode:

```
Router(config)# ephone 1
Router(config-ephone)# audible-tone
```

The following example shows how to configure an audible tone in ephone-template configuration mode:

```
Router(config)# ephone-template 1
Router(config-ephone-template)# audible-tone
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-hunt group</td>
<td>Configures an ephone hunt group in Cisco Unified CME.</td>
</tr>
<tr>
<td>voice-hunt group</td>
<td>Configures a voice hunt group in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
authen-method

To define authentication method for a vpn-profile, use the authen-method command in vpn-profile configuration mode. To disable the authentication method, use the no form of this command.

authen-method [both|none|password]

no authen-method

Syntax Description

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>both</td>
<td></td>
<td>Requires both user id and password to authenticate.</td>
</tr>
<tr>
<td>password</td>
<td></td>
<td>Requires only password to authenticate.</td>
</tr>
<tr>
<td>none</td>
<td></td>
<td>Does not allow authentication.</td>
</tr>
</tbody>
</table>

Command Default

Both User ID and Password are required for authentication.

Command Modes

Voice service voip (cfg-lpcor-policy)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

Usage Guidelines

Use this command to define authentication method for a vpn-profile. You can define an authen-method with both user id and password, or you can define an authen-method with just password. You can choose to not allow any authentication method by configuring authen-method none.

Examples

The following example shows the authen-method both defined for vpn-profile 2:

```
Router# show run
!
!
!
voice service voip
  ip address trusted list
    ipv4 20.20.20.1
  vpn-group 1
    vpn-gateway 1 https://9.10.60.254/SSLVPINphone
    vpn-trustpoint 1 trustpoint cme_cert root
    vpn-hash-algorithm sha-1
    vpn-profile 1
      keepalive 50
      auto-network-detect enable
      host-id-check disable
      mtu 1300
    vpn-profile 2
      authen-method both
      password-persistent enable
      host-id-check enable
      vpn-profile 4
      fail-connect-time 50
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>vpn-profile</td>
<td>Defines a VPN-profile.</td>
</tr>
</tbody>
</table>
authenticate (voice register global)

To define the authenticate mode for SIP phones in a Cisco Unified CME or Cisco Unified SRST system, use the `authenticate` command in voice register global configuration mode. To return to the default, use the `no` form of this command.

**Cisco IOS Release 12.4(11)XJ and Later Releases**

```
authenticate {credential tag location|ood-refer|presence|realm string|register}
no authenticate {credential tag location|ood-refer|presence|realm string|register}
```

**Cisco IOS Release 12.4(4)T**

```
authenticate [all] [realm string]
no authenticate [all] [realm string]
```

### Syntax Description

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>credential</td>
<td>Number that identifies the credential file to use for out-of-dialog REFER (OOD-R) or presence authentication. Range: 1 to 5.</td>
</tr>
<tr>
<td>tag</td>
<td>Name and location of the credential file in URL format. Valid storage locations are TFTP, HTTP, and flash memory.</td>
</tr>
<tr>
<td>location</td>
<td>Incoming OOD-R requests are authenticated using RFC 2617-based digest authentication.</td>
</tr>
<tr>
<td>ood-refer</td>
<td>Incoming presence subscription requests from an external presence server are authenticated.</td>
</tr>
<tr>
<td>presence</td>
<td>Realm parameter for challenge and response as specified in RFC 2617 is authenticated.</td>
</tr>
<tr>
<td>realm string</td>
<td>All incoming registration requests are challenged and authenticated. Valid for Cisco Unified CME only.</td>
</tr>
</tbody>
</table>

### Command Default

Authenticate mode is disabled.

### Command Modes

Voice register global configuration (config-register-global)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>The <code>credential</code>, <code>ood-refer</code>, <code>presence</code>, and <code>register</code> keywords were added. The <code>register</code> keyword replaced the <code>all</code> keyword.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>The modifications to this command were integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

The `credential` keyword allows OOD-R and presence service to use credential files for authentication. Up to five text files containing username and password pairs can be defined and loaded into the system. The contents of these five files are mutually exclusive; the username and password pairs must be unique across all the files. For Cisco Unified CME, the username and password pairs cannot be the same ones defined for SCCP or SIP phones with the `username` command.
The **ood-refer** keyword specifies that any OOD-R request that passes authentication is authorized to setup calls between referee and refer-target if OOD-R is enabled with the **refer-ood enable** command.

The **presence** keyword enables digest authentication for external watchers. Credentials are verified against a credential file stored in flash. This applies to both OOD-R and presence. The default is to authenticate all SUBSCRIBE requests from external watchers. An external watcher that passes authentication is authorized to subscribe to presence service for all lines allowed to be watched.

The **register** keyword enables authentication for registration requests in which the MAC address of the SIP phone cannot be identified by using other methods. All incoming register requests are challenged and authenticated. The **realm** keyword with the **string** argument specifies the character string to be included in the challenge.

### Examples

The following example shows that all registration requests from SIP phones in a Cisco Unified CME system must be authenticated:

```plaintext
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# authenticate register
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>credential load</strong></td>
<td>Reloads a credential file into flash memory.</td>
</tr>
<tr>
<td><strong>mode cme</strong></td>
<td>Enables the mode for provisioning SIP phones in a Cisco Unified CME system.</td>
</tr>
<tr>
<td><strong>presence-enable</strong></td>
<td>Allows incoming presence subscribe requests from SIP trunks.</td>
</tr>
<tr>
<td><strong>refer-ood enable</strong></td>
<td>Enables OOD-R processing.</td>
</tr>
<tr>
<td><strong>username (ephone)</strong></td>
<td>Defines a username and password for SCCP phones.</td>
</tr>
<tr>
<td><strong>username (voice register pool)</strong></td>
<td>Defines a username and password for authenticating SIP phones.</td>
</tr>
</tbody>
</table>
authentication credential

To create an entry for an application’s credential in the database used by the Cisco Unified CME authentication server, use the `authentication credential` command in telephony-service configuration mode. To remove the credential, use the `no` form of this command.

```
authentication credential application-name password
no authentication credential application-name password
```

**Syntax Description**

- `application-name`: String sent by application to identify itself to the server. Length of string: 1 to 15 characters.
- `password`: String sent by application to identify itself to the server. Length of string: 1 to 15 characters. The 0 in the parameter `[0|6]` mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

**Command Default**

The credential is not stored in the database.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command was enhanced for password encryption, based on Unified CME password policy.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command stores a credential in the database used by the Cisco Unified CME authentication server. The authentication server uses this data to authenticate and authorize HTTP requests from IP phones in Cisco Unified CME.

Up to eight credentials can be stored in the database for the Cisco Unified CME authentication server.

For applications other than Extension Mobility, the credential must be created in the application.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters `[0|6]`. This in accordance with Unified CME Password Policy. The 0 in the parameter `[0|6]` mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

**Note**

This command is not required for authorizing requests from Extension Mobility phones in Cisco Unified CME.
The following example shows how to configure IP phones in Cisco Unified CME to request authorization from the internal authentication server. When the IP phone receives a command from the application, the phone requests authorization from the Cisco Unified CME authentication server to execute that command. The authorization request from the phone includes the specified credential. The authentication server compares the credential in its database to the one in the request, and authorizes or rejects the request based on the results.

Router(config)# telephony-service
Router(config-telephony)# authentication credential att psswrd
Router(config-telephony)# url authentication http://192.0.2.0/CMCIP/authenticate.asp att psswrd
Router(config-telephony)#

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>url authentication</strong></td>
<td>Specifies authentication server and credential to be used by an application.</td>
</tr>
</tbody>
</table>
auto assign

To automatically assign an already defined telephone or extension number to button 1 of Cisco Unified IP phones as they register for service with a Cisco Unified CME router, use the **auto assign command** in telephony-service configuration mode. To return to the default of not automatically assigning dn-tags, use the **no** form of this command.

```
auto assign dn-tag to dn-tag [type phone-type] [cfw extension-number timeout seconds]
no auto assign dn-tag to dn-tag [type phone-type] [cfw extension-number timeout seconds]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>dn-tag to dn-tag</code></td>
<td>Range of ephone-dn tags for already configured ephone-dns, from which a tag is assigned to the ephone being created. The maximum number of directory numbers supported is version and platform dependent. Type <code>?</code> to display the value.</td>
</tr>
<tr>
<td><code>type phone-type</code></td>
<td>(Optional) Type of Cisco Unified IP phone to which to restrict automatic assignment of ephone-dn tags. Valid entries are the following:</td>
</tr>
<tr>
<td></td>
<td>• 12SP—12SP+ and 30VIP phones.</td>
</tr>
<tr>
<td></td>
<td>• 7902—Cisco Unified IP Phone 7902G.</td>
</tr>
<tr>
<td></td>
<td>• 7905—Cisco Unified IP Phone 7905G.</td>
</tr>
<tr>
<td></td>
<td>• 7906—Cisco Unified IP Phone 7906G.</td>
</tr>
<tr>
<td></td>
<td>• 7910—Cisco Unified IP Phone 7910 and 7910G.</td>
</tr>
<tr>
<td></td>
<td>• 7911—Cisco Unified IP Phone 7911G.</td>
</tr>
<tr>
<td></td>
<td>• 7912—Cisco Unified IP Phone 7912G.</td>
</tr>
<tr>
<td></td>
<td>• 7920—Cisco Unified Wireless IP Phone 7920.</td>
</tr>
<tr>
<td></td>
<td>• 7921—Cisco Unified Wireless IP Phone 7921.</td>
</tr>
<tr>
<td></td>
<td>• 7931—Cisco Unified Wireless IP Phone 7931G.</td>
</tr>
<tr>
<td></td>
<td>• 7935—Cisco Unified IP Conference Station 7935.</td>
</tr>
<tr>
<td></td>
<td>• 7936—Cisco Unified IP Conference Station 7936.</td>
</tr>
<tr>
<td></td>
<td>• 7937—Cisco Unified IP Conference Station 7937</td>
</tr>
<tr>
<td></td>
<td>• 7940—Cisco Unified IP Phones 7940 and 7940G.</td>
</tr>
<tr>
<td></td>
<td>• 7941—Cisco Unified IP Phone 7941G.</td>
</tr>
<tr>
<td></td>
<td>• 7941GE—Cisco Unified IP Phone 7941G-GE.</td>
</tr>
<tr>
<td></td>
<td>• 7942—Cisco Unified IP Phone 7942.</td>
</tr>
<tr>
<td></td>
<td>• 7945—Cisco Unified IP Phone 7945.</td>
</tr>
</tbody>
</table>
**type phone-type**

- **7960**—Cisco Unified IP Phones 7960 and 7960G.
- **7961**—Cisco Unified IP Phone 7961G.
- **7961GE**—Cisco Unified IP Phone 7961G-GE.
- **7962**—Cisco Unified IP Phone 7962.
- **7965**—Cisco Unified IP Phone 7965.
- **7970**—Cisco Unified IP Phone 7970G.
- **7971**—Cisco Unified IP Phone 7971G-GE.
- **7975**—Cisco Unified IP Phone 7975.
- **7985**—Cisco Unified IP Phone 7985.
- **CIPC**—Cisco IP Communicator.
- **all**—All ephone types.
- **anl**—Analog gateway.
- **ata**—Cisco ATA-186 or Cisco ATA-188.
- **bri**—SCCP gateway.
- **vge-phone**—vg248 phone emulation for analog phone.

**Note** You can also add a new phone type to your configuration by using the **ephone-type** command.

**cfw** (Optional) Automatically assigned ephone-dns are provisioned for call-forward busy and no-answer to the specified extension number.

**extension-number** (Optional) Extension number to which calls are to be forwarded on busy and no-answer conditions.

**timeout seconds** (Optional; required if the cfw keyword is used) Amount of time, in seconds, to wait when a call is not being answered before forwarding it. Range: 3 to 60000.

**Command Default**

Ephone-dn tags are not automatically assigned to registering Cisco Unified IP phones.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.3(7)T</td>
<td>Cisco CME 3.1</td>
<td>The 7920 and 7936 keywords were added.</td>
</tr>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>The 7970 keyword was added.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>The 7971 keyword was added, and this command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The 7941, 7941GE, 7961, and 7961GE keywords were added.</td>
</tr>
<tr>
<td>Cisco IOS Release</td>
<td>Cisco Product</td>
<td>Modification</td>
</tr>
<tr>
<td>------------------</td>
<td>--------------</td>
<td>--------------</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(6)XE</td>
<td>Cisco Unified CME 4.0(2)</td>
<td>The <strong>7931</strong> keyword was added.</td>
</tr>
<tr>
<td>12.4(4)XC4</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>The <strong>7931</strong> keyword was added.</td>
</tr>
<tr>
<td>12.4(11)T</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>This command was integrated into Cisco IOS Release 12.4(11)T.</td>
</tr>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>The <strong>7921</strong> and <strong>7985</strong> keywords were introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>12.4(15)T1</td>
<td>Cisco Unified CME 4.1(1)</td>
<td>The <strong>7942, 7945, 7962, 7965</strong>, and <strong>7975</strong> keywords were introduced.</td>
</tr>
<tr>
<td>12.4(11)XW3</td>
<td>Cisco Unified CME 4.2</td>
<td>The <strong>7942, 7945, 7962, 7965</strong>, and <strong>7975</strong> keywords were introduced.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>The <strong>7942, 7945, 7962, 7965</strong>, and <strong>7975</strong> keywords were introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>The <strong>7942, 7945, 7962, 7965</strong>, and <strong>7975</strong> keywords were introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>The <strong>7937</strong> keyword was introduced and this command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to create an ephone configuration for a Cisco Unified IP phone whose MAC address is not explicitly configured as it registers in Cisco Unified CME. The system-created ephone configuration includes the MAC address of the Cisco Unified IP phone being registered and an already-defined available ephone-dn assigned to button 1 of this phone.

The **auto-reg-ephone** command must be enabled (default) to use this command. If the auto registration feature is disabled, a Cisco Unified IP phone whose MAC address is not explicitly configured cannot register in Cisco Unified CME.

Before using this command, configure the ephone-dn tags to be assigned and define at least one primary number for each dn-tag.

All ephone-dns in a specified range should be of the same type, either single-line or dual-line.

Ephone-dn tags to be assigned must belong to normal ephone-dns and cannot belong to paging ephone-dns, intercom ephone-dns, music-on-hold (MOH) ephone-dns, or message-waiting-indication (MWI) ephone-dns.

The **auto assign** command cannot create shared lines.

If an insufficient number of dn-tags is available, some ephone configurations will not include a telephone or extension number.

Use multiple **auto assign** commands to assign discontinuous ranges of ephone-dn tags and to support multiple types of IP phones. Overlapping ranges of dn-tags may be assigned so that they map to more than one type.
of phone. If no type is specified, the values in the range are assigned to phones of any type, and if a specific range is assigned for a specific phone type, the available ephone-dn tag in that range are used first.

If the phone being registered is connected to a Cisco VG200 series analog phone gateway, configuring the auto assign command will automatically create one ephone configuration for each configured port, as the port registers with the Cisco Unified CME router. To ensure that the tag-to-port assignment will match the numbering order of the physical ports; for example, dn-tags 1 to 24 assigned to ports 1 to 24 of a Cisco VG224 analog phone gateway, in that order, we recommend that the Cisco Unified CME system be up, running, and configured before you boot the analog phone gateway.

The auto assign command cannot be used for the Cisco Unified IP Phone 7914 Expansion Module. Phones with one or more expansion modules must be configured manually.

After using this command, reboot the phone for which an ephone is to be configured.

This command is also used by the Cisco Unified CME setup tool to automatically assign ephone-dns after the tool has gathered information about the setup from the user. When lines are assigned by the Cisco Unified CME setup tool in keyswitch mode with two ephone-dn entries created for each individual extension number, the automatic assignment mechanism assigns both ephone-dn entries to an individual ephone associated with an IP phone.

**Note**

Care should be taken when using the auto assign command because this command grants telephony service to any IP phone that attempts to register. If you use the auto assign command, ensure that your network is secure from unauthorized access by unknown IP phones.

**Examples**

The following examples show how to configure the Auto Assign feature, including prerequisite commands for configuring the auto assign command.

The following example shows how to enter the ephone-dn configuration and create ephone-dns configurations, tags 1-4, each having a single primary number:

```sh
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 2000
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 3000
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 3
Router(config-ephone-dn)# number 4000
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 4
Router(config-ephone-dn)# number 4001
Router(config-ephone-dn)# exit
```

The following example shows how to designate ephone-dn tags 1 to 4 for automatic assignment to any type of IP phone and then perform a fast reboot of all phones:

```sh
Router(config)# telephony-service
Router(config-telephony)# auto assign 1 to 4
Router(config-telephony)# restart all
```
The following example is the partial output from the `show ephone registered` command listing four registered IP phones, to which ephone-dn tags 1 to 4 have been automatically assigned, after the phones were booted:

```
Router# show ephone registered
  mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
  IP:10.0.0.2 51671 Telecaster 7940 keepalive 28 max_line 2
  button 1: dn 1 number 2000
ephone-2 Mac:0030.94C3.F43A TCP socket:[1] activeLine:0 REGISTERED
  mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
  IP:10.0.0.3 50094 Telecaster 7960 keepalive 28 max_line 6
  button 1: dn 2 number 3000
ephone-3 Mac:0003.6B40.99DA TCP socket:[3] activeLine:0 REGISTERED
  mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
  IP:10.0.0.200 51879 Telecaster 7960 keepalive 28 max_line 6
  button 1: dn 3 number 4000
  mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
  IP:10.0.0.012 51879 Telecaster 7960 keepalive 28 max_line 6
  button 1: dn 4 number 4001
```

The following example shows how to designate ephone-dn tags 1 to 12 for automatic assignment to Cisco Unified IP Phone 7910Gs only and ephone-dn tags 13 to 20 for automatic assignment to a Cisco Unified IP Phones 7960 and 7960Gs only, with call forwarding to extension 5001 on busy or after 30 seconds of ringing with no answer:

```
Router(config)# telephony-service
Router(config-telephony)# auto assign 1 to 12 type 7910
Router(config-telephony)# auto assign 13 to 20 type 7960 cfw 5001 timeout 30
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>auto-reg-ephone</code></td>
<td>Enables registration of Cisco Unified IP phones for which MAC addresses are not explicitly configured.</td>
</tr>
<tr>
<td><code>number</code></td>
<td>Associates a telephone or extension number with an ephone-dn.</td>
</tr>
<tr>
<td><code>restart (ephone)</code></td>
<td>Performs a fast reboot of a single phone associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td><code>restart (telephony-service)</code></td>
<td>Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td><code>show ephone</code></td>
<td>Displays statistical information about registered Cisco Unified IP phones.</td>
</tr>
<tr>
<td><code>show ephone registered</code></td>
<td>Displays the status of registered phones.</td>
</tr>
</tbody>
</table>
auto-assign (auto-register)

To configure the mandatory DN range for automatic registration of SIP phones with the Cisco Unified CME system, use the auto-assign command in voice auto register configuration mode. This command is a sub-mode CLI of the command auto-register. To disable configuring DN range for auto registration of SIP phones, use the no form of this command.

auto-assign  First DN number  to  Last DN number
no  auto-assign

Syntax Description

| auto-assign First DN number to Last DN number | The mandatory range of directory numbers configured for phones auto registering on Unified CME. Range: 1 to 4294967295. |

Command Default

By default, this command is disabled.

Command Modes

voice auto register configuration (config-voice-auto-register)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.6(3)M</td>
<td>Cisco Unified CME 11.5</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>16.3.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Usage Guidelines

This command is a sub-mode option of the command auto-register. The command enables the administrator to configure the DN range for the SIP phones auto registering on Unified CME. For SIP phones to successfully auto register on Unified CME, it is mandatory that the DN range is defined. The assigned value of First DN number should be greater than zero.

Examples

The following example shows how to configure DN range for auto registration of SIP phones:

```
Router(config)#voice register global
Router(config-register-global)#auto-register
Router(config-voice-auto-register)# ?

VOICE auto register configuration commands:
auto-assign  Define DN range for auto assignment
default     Set a command to its defaults
exit        Exit from voice register group configuration mode
no          Negate a command or set its defaults
password    Default password for auto-register phones
service-enable  Enable SIP phone Auto-Registration
template    Default template for auto-register phones

Router(config-voice-auto-register)#auto-assign 1 to 10
```

## Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>service-enable (auto-register)</code></td>
<td>Temporarily disables the auto registration process, but retains the password and DN range configurations. Once auto-register command is entered, the service is enabled by default.</td>
</tr>
<tr>
<td><code>password (auto-register)</code></td>
<td>Configures the mandatory password that administrator sets for auto registration of SIP phones on Unified CME.</td>
</tr>
<tr>
<td><code>auto-register</code></td>
<td>Enables automatic registration of SIP phones with the Cisco Unified CME system.</td>
</tr>
<tr>
<td><code>template (auto-register)</code></td>
<td>Creates a basic configuration template that supports all the configurations available on the voice register template.</td>
</tr>
<tr>
<td><code>auto-reg-ephone</code></td>
<td>Enables automatic registration of ephones with the Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
auto logout

To enable the automatic change of an ephone hunt group agent’s ephone-dn to not-ready status after a specified number of hunt-group calls are not answered, use the auto logout command in ephone-hunt configuration mode. To disable automatic logout, use the no form of this command.

auto logout [number-of-calls] [{dynamic|static}]
no auto logout [number-of-calls] [{dynamic|static}]

Syntax Description

| number-of-calls | (Optional) Number of unanswered hunt-group calls to an ephone-dn before the ephone-dn is automatically changed to not-ready status. Range is from 1 to 20. Default is 1. |
| dynamic | (Optional) Specifies that this command applies only to dynamic hunt group members (those who are specified by an asterisk (*) wildcard in the hunt group configuration). If neither the dynamic nor static keyword is used, automatic logout applies to both dynamic and static hunt group members. |
| static | (Optional) Specifies that this command applies only to static hunt group members (those whose extension numbers are explicitly named in the hunt group configuration). If neither the dynamic nor static keyword is used, automatic logout applies to both dynamic and static hunt group members. |

Command Default

Automatic change of agent status to not-ready is disabled.

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The number-of-calls argument and the dynamic and static keywords were added. The criterion for this command was changed from exceeding the timeout command limit to exceeding the number of calls specified in this command.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>The modifications made to this command were integrated into Cisco IOS 12.4(9)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

This command is valid only for the following Cisco IP phones:

- Cisco Unified IP Phone 7905G
- Cisco Unified IP Phone 7912G
- Cisco Unified IP Phones 7940 and 7940G
- Cisco Unified IP Phones 7960 and 7960G
This command is used with the Automatic Agent Status Not-Ready feature for ephone hunt groups, which automatically puts an agent’s phone in not-ready status when it exceeds a specified limit. The limit at which the Automatic Agent Status Not-Ready feature is triggered depends on the Cisco CME version that you are using, as follows:

- Cisco CME 3.3 and earlier versions—Automatic Agent Status Not-Ready is invoked when an ephone-hunt group call rings longer on a member ephone-dn than the period of time configured in the timeout command in ephone-hunt configuration mode.
- Cisco Unified CME 4.0 and later versions—Automatic Agent Status Not-Ready is invoked when the specified number of ephone-hunt group calls is unanswered by an agent. The default is one call if the number of calls is not explicitly specified.

When Automatic Agent Status Not-Ready is specified for an ephone hunt group and it is triggered because an ephone-dn member does not answer a specified number of ephone hunt group calls, the following actions take place:

- If the hunt-group logout HLog command has been used, the agent is placed in not-ready status. The agent’s phone will not receive further hunt-group calls but will receive calls that directly dial the phone’s extension numbers. An agent in not-ready status can return to ready status by pressing the HLog soft key or by using the HLog feature access code (FAC).
- If the hunt-group logout HLog command has not been used or if the hunt-group logout DND command has been used, the phone on which the ephone-dn appears is placed into Do Not Disturb (DND) mode, in which the ephone-dn does not receive any calls at all, including hunt-group calls. The red lamp on the phone lights to indicate DND status. An agent in DND mode can return to ready status by pressing the DND soft key or by using the DND FAC.
- When an agent returns to ready status, the ephone hunt group resumes sending calls to the agent’s ephone-dn.

Note

When an agent who is a dynamic member of a hunt group is in not-ready status, the agent’s slot in the ephone hunt group is not relinquished. It remains reserved by the agent until the agent leaves the group.

You can use the auto logout command with any number of ephone hunt groups, but any ephone-dn to which the auto logout command applies must belong to only one ephone. Automatic Agent Status Not-Ready is not supported on shared lines.

Examples

This section provides the following examples:

- Cisco CME 3.3 and Earlier Versions
- Cisco Unified CME 4.0 and Later Versions

Cisco CME 3.3 and Earlier Versions

In the following example, ephone hunt group 1 is configured to permit automatic logout. If hunt group calls that are presented to 1001 and 1002 are unanswered (that is, if they ring longer than 40 seconds each), ephone 1 and ephone 2 are automatically put into DND mode. All unanswered calls are sent to voice mail (5000).

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1001
Router(config)# ephone-dn 2
```
Cisco Unified CME 4.0 and Later Versions

In the following example, Automatic Agent Status Not-Ready is limited to dynamic hunt group members who do not answer two consecutive ephone hunt group calls. Ephone-dn 33, extension 1003, has dynamically joined ephone-hunt group 1. Ephone 3 will be put into DND mode if extension 1003 does not answer two consecutive hunt group calls. Ephones 1 and 2 will not be put into DND if they do not answer hunt-group calls, because the **auto logout** command applies only to dynamic hunt-group agents.

In the following example, Automatic Agent Status Not-Ready cannot be used because all of the ephone-dns are shared.
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td><strong>hunt-group logout</strong></td>
<td>Enables separate handling of DND and HLog functionality for hunt-group agents and the display of an HLog soft key on phones.</td>
</tr>
<tr>
<td></td>
<td><strong>timeout</strong></td>
<td>Defines the number of seconds after which a call that is not answered is redirected to the next number in a Cisco Unified CME ephone-hunt-group list.</td>
</tr>
</tbody>
</table>
auto logout (voice hunt-group)

To enable the automatic change of a voice hunt group agent’s voice register dn or ephone-dn to not-ready status after a specified number of hunt-group calls are not answered, use the auto logout command in voice hunt group configuration mode. To disable automatic logout, use the no form of this command.

**Syntax Description**

| number-of-calls | Number of unanswered hunt-group calls to a voice register dn or ephone-dn before the DN is automatically changed to not-ready status. Range is 1 to 20. Default is 1. |

**Command Default**

Automatic change of agent status to not-ready is disabled.

**Command Modes**

voice hunt-group configuration (config-voice-hunt)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IOS XE Everest 16.4.1</td>
<td>Cisco Unified CME 11.6</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.6(3)M1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Automatic Agent Status Not-Ready feature for voice hunt groups, which automatically puts an agent’s phone in not-ready status when it exceeds a specified limit. The limit at which the Automatic Agent Status Not-Ready feature is triggered has to be specified from the range of 1 to 20. If no value is defined, the default value of 1 is applied. When an agent returns to ready status, the voice hunt group resumes sending calls to the agent’s DN.

When Automatic Agent Status Not-Ready is specified for a voice hunt group and it is triggered because a DN member does not answer a specified number of voice hunt group calls, the following actions take place:

- If the hunt-group logout HLog command is configured, then the DN of that hunt group switch to not-ready state when the number of successive unanswered hunt group calls specified under auto logout command is matched. When hunt-group logout HLog command is configured, phone level logout happens for SIP phones. However, SCCP phones log out at line level. An agent in not-ready status can return to ready status by pressing the HLog soft key, HLog feature access code (FAC), or Feature Button.

- If the hunt-group logout DND command is configured, then phone switches to DND mode and logs out the member when the number of successive unanswered hunt group calls specified under auto logout command is matched. For hunt-group logout DND command, both SIP and SCCP phones log out at phone level. An agent in not-ready status can return to ready status by pressing the DND softkey.

**Cisco Unified CME 11.6 and Later Versions**

In the following example, voice hunt-group 1 is configured to permit automatic logout. If hunt group calls that are presented to 1001, 1002, 1003, and 1004 are unanswered (that is, if they ring longer than 40 seconds each), voice register pool 1, voice register pool 2, ephone 1, and ephone 2 are automatically logged out. All unanswered calls are sent to DN 5000.
auto logout (voice hunt-group)

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>auto logout</td>
<td>Enables automatic change of an ephone hunt group agent’s ephone-dn to not-ready status after a specified number of hunt-group calls are not answered.</td>
</tr>
</tbody>
</table>
auto-answer

To enable the intercom auto-answer feature on a SIP phone extension, use the auto-answer command in voice register dn configuration mode. To return to the default, use the no form of this command.

```
auto-answer
no auto-answer
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Disabled

**Command Modes**

Voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command creates an IP phone line connection that resembles a private line, automatic ring-down (PLAR). The auto-answer causes an extension (directory number) to operate in auto-dial fashion for outbound calls and auto answer-with-mute for inbound calls. If an extension is configured for intercom operation, it can be associated with one Cisco IP phone only.

Any caller can dial an intercom extension, and a call to an intercom extension that is originated by a nonintercom caller triggers an automatic answer exactly like a legitimate intercom call. To prevent nonintercom originators from manually dialing an intercom destination, you can use alphabetic characters when you assign numbers to intercom extensions by using the number command. These characters cannot be dialed from a normal phone but can be dialed by preprogrammed intercom extensions when calls are made by the router.

Use the reset command to reset an individual SIP phone after you make changes to an extension for a SIP phone in Cisco CME.

**Examples**

The following example shows how to set the auto-answer feature on SIP phone directory number 1:

```
Router(config)# voice register dn 1
Router(config-register-dn) number A5001
Router(config-register-dn) auto-answer
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>number (voice register dn)</td>
<td>Associates a telephone or extension number with a directory number.</td>
</tr>
<tr>
<td>reset (voice register global)</td>
<td>Performs a complete reboot of all SIP phones associated with a Cisco CME router.</td>
</tr>
<tr>
<td>reset (voice register pool)</td>
<td>Performs a complete reboot of a single SIP phone associated with a Cisco CME router.</td>
</tr>
</tbody>
</table>
auto-line

To enable automatic line selection on an IP phone in a Cisco CallManager Express (Cisco CME) system, use the `auto-line` command in ephone configuration mode. To disable automatic line selection, use the `no` form of this command.

```
auto-line [{button-number |answer-incoming}|incoming}]
no auto-line
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Argument</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>button-number</td>
<td>(Optional) Selects the line associated with the specified button when the handset is lifted.</td>
</tr>
<tr>
<td>answer-incoming</td>
<td>(Optional) Enables automatic line selection for incoming calls on the line associated with the <code>button-number</code> argument.</td>
</tr>
<tr>
<td>incoming</td>
<td>(Optional) Enables automatic line selection for incoming calls only.</td>
</tr>
</tbody>
</table>

**Command Default**

Automatic line selection is enabled.

**Command Modes**

Ephone configuration (config ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.3(7)T</td>
<td>Cisco CME 3.1</td>
<td>The <code>button-number</code> argument was added.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The <code>answer-incoming</code> keyword was added.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>The <code>answer-incoming</code> keyword was integrated into Cisco IOS 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `auto-line` command with no keyword or argument enables automatic line selection on the specified ephone. Picking up a handset answers the first ringing line or, if no line is ringing, selects the first idle line. This is also the default behavior if this command is not used.

Use the `auto-line incoming` command enables automatic line selection for incoming calls only. Picking up the handset answers the first ringing line and, if no line is ringing, does not select an idle line for an outgoing call. Pressing a line button selects a line for an outgoing call.

Use the `auto-line` command with the `button-number` argument specifies the line that will automatically be selected when the handset is picked up to make an outgoing call. If a button number is specified and the line associated with that button is unavailable (because it is a shared line in use on another phone), no dial tone is heard when the handset is lifted. You must press an available line button to make an outgoing call. Incoming calls must be answered by pressing the Answer soft key or pressing the ringing line button.

Use the `answer-incoming` keyword with the `button-number` argument enables automatic line selection for incoming calls on the specified button. Picking up the handset answers the incoming call on the line button associated with the `button-number` argument.
Use the `no auto-line` command disables automatic line selection on the ephone that is being configured. Pressing the Answer soft key answers the first ringing line, and pressing a line button selects a line for an outgoing call. Picking up the handset does not answer calls or provide dial tone.

### Examples

The following example shows how to disable automatic line selection. The phone user must use the Answer soft key or press a line button to answer calls, or the phone user must press a line button to initiate outgoing calls.

```
Router(config)# ephone 23
Router(config-ephone)# no auto-line
```

The following example shows how to enable automatic line selection for incoming calls only. The phone user picks up the handset to answer the first ringing line. To make outgoing calls, the phone user must press a line button.

```
Router(config)# ephone 24
Router(config-ephone)# auto-line incoming
```

The following example shows how to enable the automatic selection of line button 3 for outgoing calls when the handset is lifted. There is no automatic answering of incoming calls; the user presses the Answer soft key or presses a line button to answer a call.

```
Router(config)# ephone 26
Router(config-ephone)# auto-line 3
```

The following example shows how to enable the automatic selection of line button 3 when the handset is lifted to answer incoming calls or to make outgoing calls.

```
Router(config)# ephone 26
Router(config-ephone)# auto-line 3 answer-incoming
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone</td>
<td>Enters ephone configuration mode.</td>
</tr>
</tbody>
</table>
auto-network-detect

To enable phones to automatically detect whether they are inside the corporate network or not, use the auto-network-detect command in vpn-profile configuration mode.

\[
\text{auto-network-detect} \quad \{\text{enabledisable}\}
\]

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>enable</th>
<th>Enables auto-network detection option for a vpn-profile.</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>disable</td>
<td>Disables automatic network detection option for a vpn-profile.</td>
</tr>
</tbody>
</table>

**Command Default**

Auto-network-detect is disabled.

**Command Modes**

Vpn-profile configuration (conf-vpn-profile)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to configure automatic network detection parameter in phones. The auto-network-detect command enables phones to automatically detect whether they are inside the corporate network or not. When the auto-network detection is enabled, the phone detects the corporate network and does not require a VPN connection to start functioning. Automatic network detection is disabled by default.

**Examples**

The following example shows auto-network-detect enabled for vpn-profile 1:

```
Router# show run
!
!
voice service voip
  ip address trusted list
    ipv4 20.20.20.1
  vpn-group 1
    vpn-gateway 1 https://9.10.60.254/SSLVPNphone
    vpn-trustpoint 1 trustpoint cme_cert root
    vpn-hash-algorithm sha-1
  vpn-profile 1
    keepalive 50
    auto-network-detect enable
    host-id-check disable
  vpn-profile 2
    mtu 1300
    password-persistent enable
    host-id-check enable
  vpn-profile 4
    fail-connect-time 50
    sip
    !
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>vpn-profile</td>
<td>Defines a VPN-profile.</td>
</tr>
</tbody>
</table>
To enable automatic registration of SIP phones with the Cisco Unified CME system, use the `auto-register` command in voice register global configuration mode. This command is the parent CLI, and is used to enter into the auto registration configuration mode. To disable automatic registration of SIP phones, use the no form of this command.

```
auto-register
no auto-register
```

**Syntax Description**

- `auto-register` Enters auto registration mode for SIP phones registering on Unified CME.

**Command Default**

By default, this command is enabled.

**Command Modes**

voice register global configuration (config-voice-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.6(3)M</td>
<td>Cisco Unified CME 11.5</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>16.3.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is enabled by default and allows automatic registration of SIP phones on Unified CME, provided the administrator configures the password and DN range using the relevant sub-mode CLI options. It is mandatory that the password is configured before assigning the DN range. The command `service-enable` is enabled by default when `service-enable` is enabled.

The no form of this command disables the auto registration of phones, and removes the password and DN range configurations. To disable auto registration temporarily without losing the configurations such as password and DN range, use the sub-mode CLI option, `no service-enable`.

**Examples**

The following example shows how to temporarily disable auto registration using the no form of the sub-mode option, service-enable:

```
Router(config)#voice register global
Router(config-register-global)#auto-register
Router(config-voice-auto-register)#
```

VOICE auto register configuration commands:
- `auto-assign` Define DN range for auto assignment
- `default` Set a command to its defaults
- `exit` Exit from voice register group configuration mode
- `no` Negate a command or set its defaults
- `password` Default password for auto-register phones
- `service-enable` Enable SIP phone Auto-Registration
- `template` Default template for auto-register phones
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>service-enable (auto-register)</td>
<td>Temporarily disables the auto registration process, but retains the password and DN range configurations. Once auto-register command is entered, the service is enabled by default.</td>
</tr>
<tr>
<td>password (auto-register)</td>
<td>Configures the mandatory password that administrator sets for auto registration of SIP phones on Unified CME.</td>
</tr>
<tr>
<td>auto-assign (auto-register)</td>
<td>Configures the mandatory range of directory numbers for phones auto registering on Unified CME.</td>
</tr>
<tr>
<td>template (auto-register)</td>
<td>Creates a basic configuration template that supports all the configurations available on the voice register template.</td>
</tr>
<tr>
<td>auto-reg-ep hone</td>
<td>Enables automatic registration of ephones with the Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
auto-reg-ephone

To enable automatic registration of ephones with the Cisco Unified CME system, use the `auto-reg-ephone` command in telephony-service configuration mode. To disable automatic registration, use the `no` form of this command.

```
auto-reg-ephone
no auto-reg-ephone
```

**Syntax Description**

This command has no keywords or arguments.

**Command Default**

Automatic registration is enabled.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is enabled by default and allows automatic registration, in which Cisco Unified CME allocates an ephone slot to any ephone that connects to it, regardless of whether the ephone appears in the configuration or not.

The `no` form of this command blocks the automatic registration of ephones whose MAC addresses are not explicitly listed in the configuration. When automatic registration is blocked, Cisco Unified CME records the MAC addresses of phones that attempt to register but cannot because they are blocked.

Use the `show ephone attempted-registrations` command to view the list of phones that have attempted to register but have been blocked. Use the `clear telephony-service ephone-attempted-registrations` command to clear the list of phones that have attempted to register but have been blocked.

**Examples**

The following example disables automatic registration of ephones that are not listed in the configuration:

```
Router(config)# telephony-service
Router(config-telephony)# no auto-reg-ephone
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>clear telephony-service ephone-attempted-registrations</code></td>
<td>Empties the log of ephones that unsuccessfully attempt to register with Cisco Unified CME.</td>
</tr>
<tr>
<td><code>show ephone attempted-registrations</code></td>
<td>Displays the log of ephones that unsuccessfully attempt to register with Cisco Unified CME.</td>
</tr>
</tbody>
</table>
Cisco Unified CME Commands: B

- b2bua, on page 66
- background save interval, on page 68
- bandwidth video tias-modifier, on page 69
- blf-speed-dial, on page 71
- bnea, on page 73
- bpa, on page 74
- bulk, on page 76
- bulk-speed-dial prefix, on page 78
- busy-trigger-per-button, on page 80
- busy-trigger-per-button (voice register pool), on page 82
- button, on page 83
- button-layout (voice register template), on page 90
- button-layout, on page 92
**b2bua**

To configure a dial peer associated with an individual Session Initiation Protocol (SIP) phone in Cisco Unified CME or a group of phones in a Cisco Unified SIP Survivable Remote Site Telephony (SRST) environment to point to Cisco Unity Express, use the `b2bua` command in dial-peer configuration mode. To disable B2BUA call flow on the dial peer, use the `no` form of this command.

```
b2bua
no b2bua
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

B2BUA callflow is disabled.

**Command Modes**

Dial-peer configuration (config-dial-peer)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `b2bua` command to set the Cisco Unified CME source address as the 302 redirect contact address for all calls forwarded to Cisco Unity Express.

```
Note
```

Use the `b2bua` command to configure Cisco SIP SRST 3.4 only after using the `allow-connections` command to enable B2BUA call flow on the SRST gateway.

**Examples**

The following example shows `b2bua` included in the configuration for voice dial peer 1:

```
dial-peer voice 1 voip
destination-pattern 4...
session target ipv4:10.5.49.80
session protocol sipv2
dtmf-relay sip-notify
b2bua
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>allow-connections</code></td>
<td>Enables calls between SIP endpoints in a VoIP network.</td>
</tr>
<tr>
<td><code>dial-peer voice</code></td>
<td>Defines a dial peer and enters dial-peer configuration mode.</td>
</tr>
<tr>
<td><code>mode (voice register global)</code></td>
<td>Enables the mode for provisioning SIP phones in a Cisco Unified CME system.</td>
</tr>
<tr>
<td><code>show dial-peer voice</code></td>
<td>Displays information for dial peers.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>source-address (voice register global)</td>
<td>Identifies the IP address and port through which SIP phones communicate with a Cisco Unified CME router.</td>
</tr>
<tr>
<td>voice register global</td>
<td>Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco Unified CME or Cisco Unified SIP SRST environment.</td>
</tr>
</tbody>
</table>
background save interval

To set the interval of the background save process, use the background save interval command in telephony-service configuration mode.

**background save interval interval minutes**

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>interval minutes</th>
<th>Interval value in minutes. Range: 1 to 1440. Must be in increments of 10.</th>
</tr>
</thead>
</table>

**Command Default**
The default interval is 10 minutes.

**Command Modes**
Telephony-service configuration mode

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(4)M</td>
<td>Cisco Unified CME 8.6</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to define the background saving interval. The configured interval value should be in increments of 10 minutes. If 0 is configured as the interval, no backup will be created. The default interval is 10 minutes.

**Examples**
The following example shows background save interval command configured under telephony-service configuration:

```plaintext
(config-telephony)#background
(config-telephony)#background save
(config-telephony)#background save interval
(config-telephony)#background save interval 20
(config-telephony)#background save interval 20 minutes
```
**bandwidth video tias-modifier**

To set the maximum video bandwidth bytes per second (bps) for SIP IP phones, use the `bandwidth video tias-modifier` command in voice register global configuration mode. To reset the maximum video bandwidth for SIP phones, use the `no` form of this command.

`bandwidth video tias-modifier bandwidth value [negotiate end-to-end]`

`no bandwidth video tias-modifier`

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>bandwidth value</code></td>
<td>Bandwidth value in bps. Range: 1 to 99999999.</td>
</tr>
<tr>
<td><code>negotiate end-to-end</code></td>
<td>Negotiate the minimum SIP-line video bandwidth in SDP end-to-end.</td>
</tr>
</tbody>
</table>

**Command Default**

No default bandwidth is set.

**Command Modes**

Voice register global

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(4)M</td>
<td>Cisco Unified CME 8.6</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to set the maximum video bandwidth for SIP IP phones. Video calls require much higher bandwidth usage than audio only calls. When there is a limitation of resources, video call bandwidth control becomes very crucial for the system. Using the bandwidth video tias-modifier command, video calls on Cisco Unified IP Phones 9951 and 9971 can use up to 1Mbps for VGA quality video.

**Examples**

The following example shows `bandwidth video tias-modifier` command configured under voice register global:

```
Router#show run
!
!
voice service voip
  allow-connections sip to sip
!
!
voice register global
  mode cme
  source-address 10.100.109.10 port 5060
  bandwidth video tias-modifier 256 negotiate end-to-end
  max-dn 200
  max-pool 42
  create profile sync 0004625832149157
!
voice register pool 1
  id mac 1111.1111.1111
  camera
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>video</code></td>
<td>Enables video capability on Cisco Unified SIP IP Phones 9951 and 9971.</td>
</tr>
</tbody>
</table>
To enable Busy Lamp Field (BLF) monitoring for a speed-dial number on a phone registered to Cisco Unified CME, use the `blf-speed-dial` command in ephone or voice register pool configuration mode. To disable BLF monitoring for speed-dial, use the `no` form of this command.

```
blf-speed-dial tag number label string [device]
no blf-speed-dial tag
```

### Syntax Description

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>tag</code></td>
<td>Number that identifies the speed-dial index. Range is 1 to 75 (Skinny Client Control Protocol, SCCP); 1 to 113 (Session Initiation Protocol, SIP).</td>
</tr>
<tr>
<td><code>number</code></td>
<td>Telephone number to speed dial.</td>
</tr>
<tr>
<td><code>label</code></td>
<td>Alphanumeric label that identifies the speed-dial button. The string can contain up to 30 characters.</td>
</tr>
<tr>
<td><code>string</code></td>
<td>(Optional) Enables phone-based monitoring.</td>
</tr>
<tr>
<td><code>device</code></td>
<td></td>
</tr>
</tbody>
</table>

### Command Default

BLF monitoring is disabled.

### Command Modes

Ephone configuration (config-ephone)

Voice register pool configuration (config-register-pool)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was modified to add the <code>device</code> keyword.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
<tr>
<td>15.2(4)M</td>
<td>Cisco Unified CME 9.1</td>
<td>This command was modified to increase the BLF speed-dial index for Cisco Unified SIP phones to 113.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command enables a phone to monitor the status of a line associated with a speed-dial button. The `device` keyword enables BLF monitoring of the phone for which the watched directory number is the primary line. This allows watchers to monitor whether a user is on the phone, not just on an individual line on the phone.

The directory number associated with the speed-dial number must have presence enabled with the `allow watch` command. For device-level monitoring, all directory numbers associated with the monitored phone require the `allow watch` command. If any of the directory numbers is missing this configuration, the device status reported to the watcher could be inconsistent.

After using the `blf-speed-dial` command for Cisco Unified SIP IP phones, you must generate a new configuration profile using the `create profile` command and then restart the phones with the `restart` command.
For information on the BLF status indicators that display on specific types of phones in Cisco Unified CME, see the Cisco Unified IP Phone documentation for your phone model.

Examples

The following example shows BLF speed-dial monitoring enabled on phone 1 for individual directory numbers. The line status of extensions 51212 and 51214 displays on phone 1 show that presence is enabled for those directory numbers.

Router(config)# ephone 1
Router(config-ephone)# blf-speed-dial 1 51212 label sales
Router(config-ephone)# blf-speed-dial 2 51214 label payroll

Router(config)# voice register pool 1
Router(config-register-pool)# blf-speed-dial 1 51212 label sales
Router(config-register-pool)# blf-speed-dial 2 51214 label payroll

The following example shows phone-based BLF speed-dial monitoring enabled on phone 2. The line status of all extensions on the phone for which 51212 is the primary number display shows that presence is enabled for those directory numbers.

Router(config)# ephone 2
Router(config-ephone)# blf-speed-dial 1 51212 label sales device

Router(config)# voice register pool 2
Router(config-register-pool)# blf-speed-dial 1 51212 label sales device

The following example shows BLF speed-dial monitoring enabled on key 13 of phone 3:

Router(config)# voice register pool 3
Router(config-register-pool)# blf-speed-dial 13 51212 label sales device

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow watch</td>
<td>Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.</td>
</tr>
<tr>
<td>create profile</td>
<td>Generates the configuration profile files required for Cisco Unified SIP IP phones.</td>
</tr>
<tr>
<td>presence</td>
<td>Enables presence service and enters presence configuration mode.</td>
</tr>
<tr>
<td>presence call-list</td>
<td>Enables BLF monitoring for call lists and directories on phones registered to a Cisco Unified CME router.</td>
</tr>
<tr>
<td>restart (voice register)</td>
<td>Performs a fast restart of one or all Cisco Unified SIP IP phones associated with a Cisco Unified CME router.</td>
</tr>
</tbody>
</table>
**bnea**

To specify the audio file used for the busy station not equipped for preemption announcement, use the `bnea` command in voice MLPP configuration mode. To disable use of this audio file, use the `no` form of this command.

```
bnea audio-url
no bnea
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>audio-url</code></td>
<td>Location of the announcement audio file in URL format. Valid storage locations are TFTP, FTP, HTTP, and flash memory.</td>
</tr>
</tbody>
</table>

**Command Default**

No announcement is played.

**Command Modes**

Voice MLPP configuration (config-voice-mlpp)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Products</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies the G.711 a-law or u-law 8-KHz encoded audio file (.wav or .au format) for the announcement that is played to the caller when the dialed number is not preemptable.

The `mlpp indication` command must be enabled (default) for a phone to play preemption announcements.

This command is not supported by Cisco IOS help. If you type `?`, Cisco IOS help does not display a list of valid entries.

**Examples**

The following example shows the busy station not equipped for preemption announcement is set to the file named `bnea.au` located in flash:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# bnea flash:bnea.au
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>access-digit</td>
<td>Defines the access digit that phone users dial to request a precedence call.</td>
</tr>
<tr>
<td>bpa</td>
<td>Specifies the audio file used for the blocked precedence announcement.</td>
</tr>
<tr>
<td>mlpp indication</td>
<td>Enables MLPP indication on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td>mlpp preemption</td>
<td>Enables preemption capability on an SCCP phone or analog FXS port.</td>
</tr>
</tbody>
</table>
To specify the audio file used for the blocked precedence announcement, use the `bpa` command in voice MLPP configuration mode. To disable use of this audio file, use the `no` form of this command.

`bpa audio-url`  
`no bpa`

**Syntax Description**  
| `audio-url` | Location of the announcement audio file in URL format. Valid storage locations are TFTP, FTP, HTTP, and flash memory. |

**Command Default**  
No announcement is played.

**Command Modes**  
Voice MLPP configuration (config-voice-mlpp)

**Command History**  
<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Products</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**  
This command specifies the G.711 a-law or u-law 8-KHz encoded audio file (.wav or .au format) for the announcement that is played to the caller in the following situations:

- Destination party for the precedence call is off hook.
- Destination party is busy with a precedence call of an equal or higher precedence and the destination party does not have Call Waiting or Call Forward configured, and does not have an attendant-console service configured.

The `mlpp indication` command must be enabled (default) for a phone to play precedence announcements.

This command is not supported by Cisco IOS help. If you type `?`, Cisco IOS help does not display a list of valid entries.

**Examples**  
The following example shows the blocked precedence announcement is set to the file named bpa.au located in flash:

```bash
Router(config)# voice mlpp
Router(config-voice-mlpp)# bpa flash:bpa.au
```

**Related Commands**  
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>attendant-console</code></td>
<td>Specifies the phone number of the MLPP attendant-console service.</td>
</tr>
<tr>
<td><code>bnea</code></td>
<td>Specifies the audio file used for the busy station not equipped for preemption announcement.</td>
</tr>
<tr>
<td><code>mlpp indication</code></td>
<td>Enables MLPP indication on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>--------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>mlpp preemption</td>
<td>Enables preemption capability on an SCCP phone or analog FXS port.</td>
</tr>
</tbody>
</table>
bulk

To set bulk registration for E.164 numbers that will register with SIP proxy server, use the `bulk` command in voice register global configuration mode. To disable bulk registration, use the `no` form of this command.

```
bulk  number-pattern
no bulk
```

**Syntax Description**

| number-pattern | A sequence of digits including wild card character. |

**Command Default**

Bulk registration is disabled.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command allows you to configure bulk registration for registering a block of phone numbers with an external registrar so that calls can be routed to Cisco CME from the SIP network.

Numbers that match the number pattern defined by using the `bulk` command register with the external registrar. The block of numbers that is registered can include any phone that is attached to Cisco CME using SIP or SCCP, or any analog phone that is directly attached to a Cisco router FXS port.

A number can contain one or more periods (.) as wildcard characters that will match any dialed number in that position. For example, 51.. rings when 5100 is dialed, when 5101 is dialed, and so forth.

The external registrar is configured by using the `registrar server` command under the SIP user-agent configuration mode.

**Examples**

The following example shows how to specify that numbers matching 1235 and any other dialed number in the next four positions, be routed to the Cisco CME from the SIP network.

```
Router(config)# voice register global
Router(conf-register-global)# mode cme
Router(conf-register-global)# bulk 1235...
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>mode (voice register global)</code></td>
<td>Enables the mode for provisioning SIP phones in a Cisco CallManager Express (Cisco CME) system.</td>
</tr>
<tr>
<td><code>no reg (voice register dn)</code></td>
<td>Specifies that a directory number in a SIP Cisco CallManager Express (Cisco CME) system not register with an external proxy server</td>
</tr>
<tr>
<td><code>no reg (voice hunt-group)</code></td>
<td>Specifies that a pilot number for a voice hunt group not register with an external proxy server</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>---------</td>
<td>------------------</td>
</tr>
<tr>
<td>registrar</td>
<td>Enables SIP registrar functionality.</td>
</tr>
</tbody>
</table>
bulk-speed-dial prefix

To set the prefix code that phone users dial to access speed-dial numbers from a global bulk speed-dial list, use the bulk-speed-dial prefix command in telephony-service configuration mode. To return the prefix code to the default, use the no form of this command.

```
bulk-speed-dial prefix prefix-code
no bulk-speed-dial-prefix
```

### Syntax Description

| prefix-code | One to four-character access code for speed dial. Default is #. |

### Command Default

The default prefix code (number sign [#]) is used.

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS 12.4(9)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command changes the prefix code that a phone user must dial to access speed-dial numbers from a speed-dial list that is enabled using the bulk-speed-dial list command in telephony-service configuration mode. The default prefix is # (number sign).

If a bulk speed-dial list is enabled using this command in telephony-service configuration mode and is also enable using this command in ephone configuration mode, the list enabled in ephone configuration mode takes precedence over the list at the global level for a given prefix. However, if the prefix used at the global level is different than the prefix used at the phone level, the lists are treated as separate lists - each list being associated with a different prefix, and at the phone level, you can access both lists.

Use the show telephony-service bulk-speed-dial to display information about bulk speed-dial lists that are configured in Cisco Unified CME.

### Examples

The following example changes the default bulk speed-dial prefix to #7 and enables global bulk speed-dial list number 6 for all phones. It also enables a personal bulk speed-dial list for ephone 2. In this example, ephone 2 can access all of the numbers in both lists because each list is assigned a different prefix (# and #7).

```
telephony-service
  bulk-speed-dial list 6 flash:sd_dept_01_1_87.txt
  bulk-speed-dial prefix #7
  ephone-dn 3
    number 2555
  ephone-dn 4
    number 2557
  ephone 2
    button 1:3 2:4
  bulk-speed-dial list 7 flash:lmi_sd_list_08_24_95.csv
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>bulk-speed-dial list</td>
<td>Enables a bulk speed-dial list.</td>
</tr>
<tr>
<td>show telephony-service bulk-speed-dial</td>
<td>Displays information about bulk speed-dial lists that are configured in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
### busy-trigger-per-button

To set the maximum number of calls allowed on an octo-line directory number before activating Call Forward Busy or a busy tone, use the `busy-trigger-per-button` command in ephone or ephone-template configuration mode. To reset to the default, use the `no` form of this command.

**Syntax Description**

```
number-of-calls
```

Maximum number of calls. Range: 1 to 8. Default: 0 (disabled).

**Command Default**

Disabled (busy trigger is 0).

**Command Modes**

- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command limits the calls to an octo-line on the specified phone by triggering Call Forward Busy or a busy tone. After the number of active calls, incoming and outgoing, on an octo-line directory number reaches the limit set with this command, the next incoming call to the directory number is forwarded to the Call Forward Busy destination. If Call Forward Busy is not configured, Cisco Unified CME rejects the call and plays a busy tone.

This command applies to each octo-line directory number on the phone.

If a directory number is shared among different phones, the busy trigger is initiated after the number of existing calls exceeds the limit set on any of the phones that share the directory number.

This command must be set to a value that is less than or equal to the value set with the `max-calls-per-button` command.

If you use an ephone template to apply a command to an ephone and you also use the same command in ephone configuration mode for the same ephone, the value that you set in ephone configuration mode has priority.

**Examples**

The following example shows that after an octo-line on ephone 1 receives four calls, the fifth incoming call triggers Call Forward Busy or a busy tone.

```
Router(config)#
ephone 1
Router(config-ephone)# busy-trigger-per-button 4
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>call-forward busy</strong></td>
<td>Enables call forwarding so that incoming calls to a busy extension are forwarded to another extension.</td>
</tr>
<tr>
<td><strong>ephone-dn</strong></td>
<td>Configures a directory number for SCCP phones.</td>
</tr>
<tr>
<td><strong>max-calls-per-button</strong></td>
<td>Sets the maximum number of calls allowed on an octo-line directory number on an SCCP phone.</td>
</tr>
</tbody>
</table>
busy-trigger-per-button (voice register pool)

To set the maximum number of calls allowed on a SIP directory number before activating Call Forward Busy or a busy tone, use the **busy-trigger-per-button** command in voice register pool configuration mode. To reset to the default, use the **no** form of this command.

**busy-trigger-per-button number**
**no busy-trigger-per-button**

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>number</th>
<th>Maximum number of calls. Range: 1 to 50.</th>
</tr>
</thead>
</table>

**Command Default**

**no busy-trigger-per-button**

**Command Modes**

Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command limits the number of calls to each directory number on the specified phone by triggering Call Forward Busy or a busy tone. After the number of active calls, both incoming and outgoing, reaches the number of calls set with this command, Cisco Unified CME forwards the next incoming call to the Call Forward Busy destination. Cisco Unified CME rejects the call and plays a busy tone if Call Forward Busy is not configured.

If a directory number is shared among different phones, the busy trigger is initiated after the number of existing calls exceeds the limit set on all of the phones that share the directory number.

This command must be set to a value that is less than or equal to the value set with the **max-calls-per-button** command.

**Examples**

The following example shows that after a shared-line on phone 1 receives four calls, the fifth incoming call triggers Call Forward Busy or a busy tone.

```
Router(config)#
voice register pool 1
Router(config-register-pool)# busy-trigger-per-button 4
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>huntstop (voice register dn)</td>
<td>Disables call hunting behavior for a directory number on a SIP phone.</td>
</tr>
<tr>
<td>max-calls-per-button</td>
<td>Sets the maximum number of calls allowed on an octo-line directory number on an SCCP phone.</td>
</tr>
<tr>
<td>shared-line</td>
<td>Creates a directory number to be shared by multiple SIP phones.</td>
</tr>
</tbody>
</table>
button

To associate ephone-dns with individual buttons on a Cisco Unified IP phone and to specify line type or ring behavior, use the **button** command in ephone configuration mode. To remove an ephone-dn association from a button, use the **no** form of this command.

```
button  button-number {separator} dn-tag [,dn-tag...] [button-number{x}overlay-button-number] [button-number...]
no button  button-number {separator} dn-tag [,dn-tag...] [button-number{x}overlay-button-number] [button-number...]
```

**Syntax Description**

| **button-number** | Number of a line button on a Cisco Unified IP phone that is to be associated with an extension (ephone-dn).
| **separator** | Single character that denotes the characteristics to be associated with this phone button. Valid entries are as follows:

- **:** (colon)—Normal ring. For incoming calls on this extension, the phone produces audible ringing, a flashing icon in the phone display, and a flashing red light on the handset. On the Cisco IP Phone 7914 Expansion Module, a flashing yellow light also accompanies incoming calls.
- **b**—Beep but no ring. Audible ring is suppressed for incoming calls, but call-waiting beeps are allowed. Visible cues are the same as those described for a normal ring.
- **c**—Call waiting. Provides call waiting for secondary calls to an overlaid ephone-dn. See also the **o** keyword.
- **f**—Feature ring. Differentiates incoming calls on a special line from incoming calls on other lines on the phone. The feature-ring cadence is a triple pulse, as opposed to a single pulse for normal internal calls and a double pulse for normal external calls.
- **m**—Monitor mode for a shared line. Visible line status indicates whether the line is in-use or not. Monitored lines cannot be used on this phone for incoming or outgoing calls.
- **o**—Overlay line. Multiple ephone-dns share a single button, up to a maximum of 25 on a button. See also the **c** keyword.
- **s**—Silent ring. Audible ring and call-waiting beep are suppressed for incoming calls. The only visible cue is a flashing (< icon in the phone display.

**Note**

- In Cisco IOS Release 12.4(4)XC and later releases, the silent ringing behavior is overridden during active night-service periods. Silent ringing does not apply during designated night-service periods when the **s** keyword is used.
- **w**—Watch mode for all lines on the phone for which this directory number is the primary line. Visible line status indicates whether watched phone is idle or not.

The maximum number of button–ephone-dn pairs is determined by the phone type.

**Note**

The Cisco Unified IP Phone 7910G has only one physical line button, but you can assign it two button–ephone-dn pairs.
The **button** command assigns telephone extensions to Cisco Unified IP phones by associating a button number with one or more directory numbers (ephone-dns).
After adding or changing a phone button configuration using this command, you must perform a quick reboot of the phone using the `restart` command.

**Note** Telephone services such as call waiting and three-party conferences require a minimum of two phone lines (ephone-dns defined with the `ephone-dn` command) to be available and configured on a Cisco IP phone.

The Cisco Unified IP Phone 7910G has only one physical line button. To support call waiting and three-party conferences on a Cisco Unified IP Phone 7910G, a second (hidden) line is required. This line cannot be selected directly using a line button. You can access the second line when you press the Conference button. You can also support multiple-call services using the `ephone-dn dual-line` configuration option.

**Feature Ring (f)**

A feature ring is a third type of ring cadence, in addition to the internal call and external call ring cadences. For example, an internal call in the United States rings for 2 seconds on and 4 seconds off (single-pulse ring), and an external call rings for 0.4 seconds on, 0.2 seconds off, 0.4 seconds on, and 0.2 seconds off (double-pulse ring). A feature ring is a triple-pulse ring. The purpose of associating a feature ring with a line button is to be able to identify from a distance a special line that is ringing on a multiline phone.

**Monitor Mode (m)**

A line button set in monitor mode on one phone displays visual line status for a line that also appears on another phone. When monitor mode is set for a button with a shared line, the line status indicates that the shared line is either idle or in use. The line and line button are available in monitor mode for visual status only. Calls cannot be made or received using a line button that has been set in monitor mode. Incoming calls on a line button that is in monitor mode do not ring and do not display caller ID or call-waiting caller ID.

Monitor mode is intended for use only in the context of shared lines so that a receptionist can visually monitor the in-use status of several users' phone extensions (for example, as a busy-lamp field). To monitor all lines on an individual phone so that a receptionist can visually monitor the in-use status of that phone, see the Watch Mode (w) section.

The line button for a monitored line can also be used as a direct-station-select for a call transfer when the monitored line is in an idle state. In this case, the receptionist who transfers a call from a normal line can press the Transfer button and then press the line button of the monitored line, causing the call to be transferred to the phone number of the monitored line.

**Overlay (o)**

Overlay lines are ephone-dns that share a single button on a multibutton phone. When more than one incoming call arrives on lines that are set on a single button, the line (ephone-dn) that is the leftmost in the `button` command list is the primary line and is given the highest priority. If this call is answered by another phone or if the caller hangs up, the phone selects the next line in its overlay set to present as the ringing call. The caller ID display updates to show the caller ID for the currently presented call.

Ephone-dns that are part of an overlay set can be single-line ephone-dns or dual-line ephone-dns, but the set must contain either all single-line ephone-dns or all dual-line ephone-dns, and not a mixture of the two.

The primary ephone-dn on each phone in a shared-line overlay set should be unique to the phone being configured to guarantee that there is a line available for outgoing calls, and to ensure that the phone user can obtain dial-tone even when there are no idle lines available in the rest of the shared-line overlay set. Use a unique ephone-dn in this manner to provide for a unique calling party identity on outbound calls made by the phone so that the called user can see which specific phone is calling.
The name of the first ephone-dn in the overlay set is not displayed because it is the default ephone-dn for calls to the phone, and the name or number is permanently displayed next to the phone’s button. For example, if there are ten ephone-dns in an overlay set, only the last nine ephone-dns are displayed when calls are made to them.

**Overlay Ephone-dns with Call Waiting (c)**

The configuration for the overlaid ephone-dns with call waiting (keyword `c`) and without call waiting (keyword `o`) is the same.

Ephone-dns accept call interruptions, such as call waiting, by default. For call waiting to work, the default must be active. To ensure that the default is active, remove the `no call-waiting beep accept` command from the configurations of ephone-dns for which you want to use call waiting.

In Cisco Unified CME 4.0(3), the Cisco Unified IP Phone 7931G cannot support overlays that contain ephone-dn configured for dual-line mode.

---

**Note**

In general, all the ephone-dns within an overlay must be of the same type (dual-line or single line mode).

**Silent Ring (s)**

You can configure silent ring on any type of phone. However, you typically set silent ring only on buttons of a phone with multiple lines, such as a Cisco Unified IP Phone 7940, Cisco Unified IP Phones 7960 and 7960G, or a Cisco Unified IP Phone 7914 Expansion Module. The only visible cue is a flashing `(<` icon in the phone display.

If you configure a button to have a silent ring, you will not hear a call-waiting beep or call-waiting ring regardless of whether the ephone-dn associated with the button is configured to generate a call-waiting beep or call-waiting ring.

In Cisco IOS Release 12.4(4)XC and later releases, the silent ringing behavior is overridden during active night-service periods. Silent ringing does not apply during designated night-service periods when the `s` keyword is used.

**Watch Mode (w)**

A line button that is configured for watch mode on one phone provides visual line status for all lines on another phone (watched phone) for which the watched directory number is the primary line. Watched mode allows a phone user, such as a receptionist, to visually monitor the in-use status of an individual phone. The line and line button on the watching phone are available in watch mode for visual status only. Calls cannot be made or received using a line button that has been set in watch mode. Incoming calls on a line button that is in watch mode do not ring and do not display caller ID or call-waiting caller ID.

If any of the following conditions are true, the status of the line button in watch mode is that the watched phone is in-use:

- Watched phone is off-hook
- Watched phone is not registered
- Watched phone is in the do-not-disturb (DND) mode
- Watched directory number is not idle

If the watched directory number is a shared line and the shared line is not idle on any phone with which it is associated, then in the context of watch mode, the status of the line button indicates that the watched phone is in use.
For best results in terms of monitoring the status of an individual phone based on a watched directory number, the directory number to be configured for watch mode should not be a shared line. To monitor a shared line so that a receptionist can visually monitor the in-use status of several users’ phone extensions, see the Monitor Mode (m) section.

If the watched directory number is associated with several phones, then the watched phone is the one on which the watched directory number is on button 1 or the one on which the watched directory number is on the button that is configured by using the **auto-line** command, with auto-line having priority.

If more than one phone meets the criteria for primary line as described above, then the watched phone is the first phone that that meets the criteria. Typically, that is the phone with the lowest ephone tag value. However, if the watched directory number is configured on button 1 of ephone 1 and the same directory number is also configured on button 3 with “auto-line 3” of ephone 24, then ephone 24 is the watched phone because the auto-line configuration has priority.

The line button for a watched phone can also be used as a direct-station-select for a call transfer when the watched phone is idle. In this case, the phone user who transfers a call from a normal line can press the Transfer button and then press the line button of the watched directory number, causing the call to be transferred to the phone number associated with the watched directory number.

**Expansion Buttons for Overlay Ephone-dn (s)**

This feature works to expand coverage for an overlay button that has been configured using the **o** separator in the **button** command. Overlay buttons with call waiting that use the **c** separator in the **button** command are not eligible for overlay rollover.

**Examples**

The following example assigns four button numbers on the phone to ephone-dn tags. Button 4 is configured for a silent ring:

```plaintext
ephone-dn 1  
  number 233

ephone-dn 4  
  number 234

ephone-dn 16  
  number 235

ephone-dn 19  
  number 236

ephone 1
  button 1:1 2:4 3:16 4s19
```

The following example shows three phones that each have three instances of extension number 1001 overlaid onto a single button, which allows three simultaneous calls to extension 1001. The first call arrives on ephone-dn 1 and rings button 1 on all three phones. The call is answered on ephone 10. A second call for 1001 hunts onto ephone-dn 2 and rings on the remaining two ephones, ephones 11 and 12, and is answered by ephone 12. A third call to 1001 hunts onto ephone-dn 3 and rings on ephone 12, where it is answered. This configuration creates a three-way shared line across three IP phones and can handle three simultaneous calls to the same telephone number. Note that if ephone 12 is busy, the third call will go to voice mail (7000). Note also that if you want to configure call waiting, you can use the same configuration, except use the **c** keyword instead of the **o** keyword. Ephone 10 uses call waiting.

```plaintext
ephone-dn 1  
  number 1001  
  no huntstop  
!  
ephone-dn 2
```
number 1001
no huntstop
preference 1
!
ephone-dn 3
number 1001
preference 2
call-forward busy 7000
!
! The next ephone configuration includes the first instance of shared line 1001.
ephone 10
mac-address 1111.2222.3333
button 1c1,2,3
!
! The next ephone configuration includes the second instance of shared line 1001.
ephone 11
mac-address 1111.2222.4444
button 1o1,2,3
!
! The next ephone configuration includes the third instance of shared line 1001.
ephone 12
mac-address 1111.2222.5555
button 1o1,2,3
!
! The next ephone configuration includes (unique) ephone-dn 1 as the primary line in a simple shared-line overlay configuration. The no huntstop command is configured for all the ephone-dns except ephone-dn 12, the last one in the overlay set. Because the ephone-dns are dual-line dns, the huntstop-channel command is also configured to ensure that the second channel remains free for outgoing calls and for conferencing.

ephone-dn 1 dual-line
number 101
huntstop-channel
!
ephone-dn 2 dual-line
number 102
huntstop-channel
!
ephone-dn 10 dual-line
number 201
no huntstop
huntstop-channel
!
ephone-dn 11 dual-line
number 201
no huntstop
huntstop-channel
!
ephone-dn 12 dual-line
number 201
huntstop-channel
!
! The next ephone configuration includes (unique) ephone-dn 1 as the primary line in a shared-line overlay
ephone 1
mac-address 1111.1111.1111
button 1o1,10,11,12
!
! The next ephone configuration includes (unique) ephone-dn 2 as the primary line in another shared-line overlay
ephone 2
mac-address 2222.2222.2222
button 1o2,10,11,12
Shared-line overlays can be constructed using the “button o” or “button c” formats, depending on whether call-waiting is desired. The following example shows an ephone configuration that enables call waiting (c) in a shared-line overlay:

```plaintext
ephone 1
  mac-address 1111.1111.1111
  button 1c1,10,11,12
!
ephone 2
  mac-address 2222.2222.2222
  button 1c2,10,11,12
```

The following example configures a “3x3” shared-line setup for three ephones and nine shared lines (ephone-dns 20 through 28). Each ephone has a unique ephone-dn for each of its three buttons (ephone-dns 1 to 3, ephone-dns 4 to 6, and ephone-dns 7 to 9). The remaining ephone-dns are shared among the three phones. Three phones with three buttons each can take nine calls. The overflow buttons provide the ability for an incoming call to ring on the first available button on each phone.

```plaintext
ephone 1
  button 1o1,2,3,20,21,22,23,24,25,26,27,28 2x1 3x1
ephone 2
  button 1o4,5,6,20,21,22,23,24,25,26,27,28 2x1 3x1
ephone 3
  button 1o7,8,9,20,21,22,23,24,25,26,27,28 2x1 3x1
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-waiting beep</td>
<td>Allows phone buttons to accept or generate call-waiting beeps.</td>
</tr>
<tr>
<td>restart (ephone)</td>
<td>Performs a fast reboot of a single phone associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td>restart (telephony-service)</td>
<td>Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td>show ephone</td>
<td>Displays information about ephones and the corresponding Cisco Unified IP phones.</td>
</tr>
<tr>
<td>show ephone overlay</td>
<td>Displays the configuration and current status of registered overlay ephone-dns.</td>
</tr>
</tbody>
</table>
button-layout (voice register template)

To organize the order of the display of all buttons including line, speed dial, blf speed dial, feature buttons, and url buttons on a Cisco Unified SIP IP phone, use the `button-layout` command in voice register template configuration mode. To disable the feature button set and change the action of the buttons on IP phones, use the `no` form of this command.

```
button-layout [button-string] [button-type]
no button-layout
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>button-string</code></td>
<td>(Optional) Specifies a comma-separated list of physical button number or ranges of button numbers.</td>
</tr>
<tr>
<td><code>button-type</code></td>
<td>(Optional) Specifies one of the following button types: Line, Speed-Dial, BLF-Speed-Dial, Feature, URL.</td>
</tr>
</tbody>
</table>

**Command Default**

No fixed set of line or feature buttons are defined.

**Command Modes**

Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the button-layout command to assign physical button numbers or ranges of numbers with button types such as line, feature, url, speed-dial, and blf-speed-dial. After creating a voice register template and applying the template to the voice register pool you can assign the button-layout configuration to a Cisco Unified IP Phone.

**Note**

The first button needs to be the line button so that the phone can complete provisioning.

**Examples**

The following example shows button-layout configured on voice register template 2 and voice register template 5.

```
Router# show voice register template all
!
voice register dn 65
   number 3065
   name SIP-7965
   label SIP3065
!
voice register template 5
   button-layout 1 line
   button-layout 2,5 speed-dial
   button-layout 3,6 blf-speed-dial
   button-layout 4,7,9 feature-button
   button-layout 8,11 url-button
!
```

Cisco Unified Communications Manager Express Command Reference
voice register template  2
button-layout  1,5  line
button-layout  4  speed-dial
button-layout  3,6  blf-speed-dial
button-layout  7,9  feature-button
button-layout  8,10-11  url-button

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>ephone-template (ephone)</td>
<td>Applies template to an ephone.</td>
</tr>
<tr>
<td></td>
<td>show voice register template</td>
<td>Displays all configuration information associated with a SIP phone template.</td>
</tr>
</tbody>
</table>
To configure a fixed set of line or feature buttons in an ephone-template which can then be applied to a supported IP phone in Cisco Unified CME, use the `button-layout` set command in ephone-template configuration mode. To disable the feature buttons set and change the action of the buttons on IP phones, use the `no` form of this command.

```
button-layout [{phone-type {1|2}|button-string|button-type}]
no button-layout
```

### Syntax Description

<table>
<thead>
<tr>
<th><code>phone-type</code></th>
<th>Type of IP phone. The following choices are valid:</th>
</tr>
</thead>
<tbody>
<tr>
<td>7931</td>
<td>Cisco Unified IP Phone 7931.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><code>1</code></th>
<th>Number of fixed line or feature set containing the following buttons:</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>• Button 24—Menu.</td>
</tr>
<tr>
<td></td>
<td>• Button 23—Headset.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><code>2</code></th>
<th>Number of fixed line or feature set containing the following buttons:</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>• Button 24—Menu.</td>
</tr>
<tr>
<td></td>
<td>• Button 23—Headset.</td>
</tr>
<tr>
<td></td>
<td>• Button 22—Directories.</td>
</tr>
<tr>
<td></td>
<td>• Button 21—Messages.</td>
</tr>
</tbody>
</table>

| `button-string` | (Optional) Specifies a comma separated list of physical button number or ranges of button numbers. |

| `button-type` | (Optional) Specifies one of the following button types: Line, Speed-Dial, BLF-Speed-Dial, Feature, URL |

### Command Default

No fixed set of line or feature buttons are defined.

### Command Modes

Ephone-template configuration (config-ephone-template)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(6)XE</td>
<td>Cisco Unified CME 4.0(2)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC4</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)T</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>This command was integrated into Cisco IOS Release 12.4(11)T.</td>
</tr>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was modified. Button String and Button Type arguments were added.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use this command to configure either Set 1 or Set 2 in an ephone-template which can then be applied to an individual Cisco Unified IP Phone 7931G in Cisco Unified CME.
After a template has been created, you can apply it to an ephone using the `ephone-template` command in ephone configuration mode. You cannot apply more than one ephone template to an ephone.

To view your ephone-template configurations, use the `show telephony-service ephone-template` command.

In Cisco Unified CME 8.5 and later versions, the `button-layout` command allows you to assign physical button numbers or ranges of numbers with button types such as Line, Feature, URL, Speed-Dial, BLF-Speed-Dial. After creating an ephone-template you can apply the `button-layout` configuration to a Cisco Unified IP Phone.

### Examples

1. The following example shows how to create ephone-template 12, containing set 2 feature buttons, and apply the template to ephone 36.

   ```
   Router(config)# ephone-template 12
   Router(config-ephone-template)# button-layout set 2
   Router(config-ephone-template)# exit
   Router(config)# ephone 36
   Router(config-ephone)# ephone-template 12
   Router(config-ephone)# exit
   Router(config)# telephony-service
   Router(config-telephony)# create cnf-files
   ```

1. The following example shows ephone-template 10, containing line button, speed-dial button, blf-speed-dial button, feature button, and url button.

   ```
   Router# show telephony-service ephone-template
   ephone-template 10
   button-layout 1 line
   button-layout 2,5 speed-dial
   button-layout 3,6 blf-speed-dial
   button-layout 4,7,9 feature
   button-layout 8,11 url
   ```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ephone-template (ephone)</code></td>
<td>Applies template to an ephone.</td>
</tr>
<tr>
<td><code>show telephony-service ephone-template</code></td>
<td>Displays ephone-template configurations.</td>
</tr>
</tbody>
</table>
Cisco Unified CME Commands: C

- call application voice aa-hunt, on page 98
- call application voice aa-name, on page 100
- call application voice aa-pilot, on page 101
- call application voice call-retry-timer, on page 103
- call application voice dial-by-extension-option, on page 105
- call application voice drop-through-option, on page 107
- call application voice drop-through-prompt, on page 108
- call application voice handoff-string, on page 109
- call application voice max-extension-length, on page 110
- call application voice max-time-call-retry, on page 111
- call application voice max-time-vm-retry, on page 113
- call application voice number-of-hunt-grps, on page 114
- call application voice queue-len, on page 116
- call application voice queue-manager-debugs, on page 118
- call application voice second-greeting-time, on page 120
- call application voice service-name, on page 122
- call application voice voice-mail, on page 123
- call application voice welcome-prompt, on page 124
- callback (voice emergency response settings), on page 126
- caller-id, on page 128
- caller-id block (ephone-dn and ephone-dn-template), on page 130
- caller-id block (voice register template), on page 132
- caller-id block code (telephony-service), on page 133
- call-feature-uri, on page 134
- call-forward, on page 136
- call-forward (voice register), on page 137
- call-forward all, on page 138
- call-forward b2bua all, on page 140
- call-forward b2bua busy, on page 142
- call-forward b2bua mailbox, on page 144
- call-forward b2bua night-service, on page 146
- call-forward b2bua noan, on page 147
- call-forward b2bua unreachable, on page 149
• call-forward busy, on page 151
• call-forward max-length, on page 154
• call-forward night-service, on page 156
• call-forward noan, on page 158
• call-forward pattern, on page 161
• calling-number local, on page 163
• calling-number local (voice register global), on page 165
• callqueue-display, on page 166
• call-park system, on page 167
• call-waiting (voice register pool), on page 168
• call-waiting beep, on page 169
• call-waiting ring, on page 171
• camera, on page 173
• capf-auth-str, on page 175
• capf-server, on page 177
• cert-enroll-trustpoint, on page 178
• clear cti session, on page 179
• clear telephony-service conference hardware number, on page 180
• clear telephony-service ephone-attempted-registrations, on page 181
• clear telephony-service xml-event-log, on page 182
• clear voice fac statistics, on page 183
• clear voice lpcor statistics, on page 184
• clear voice moh-group statistics, on page 185
• clear voice register attempted-registrations, on page 186
• cnf-file, on page 187
• cnf-file location, on page 189
• codec (ephone), on page 191
• codec (telephony-service), on page 194
• conference (ephone-dn), on page 195
• conference (voice register template), on page 197
• conference add-mode, on page 198
• conference add-mode (voice register), on page 199
• conference admin, on page 200
• conference admin (voice register), on page 202
• conference drop-mode, on page 203
• conference drop-mode (voice register), on page 205
• conference hardware, on page 207
• conference hardware (voice register global), on page 209
• conference max-length, on page 210
• conference-pattern blocked, on page 211
• conference transfer-pattern, on page 212
• cor (ephone-dn), on page 213
• cor (voice register), on page 214
• corlist, on page 217
• create cnf-files, on page 219
• create cnf-files (voice-gateway), on page 220
• create profile (voice register global), on page 221
• credentials, on page 222
• cti csta mode basic, on page 224
• cti message device-id suppress-conversion, on page 225
• cti notify, on page 226
• cti watch, on page 228
• cti-aware, on page 230
• ctl-client, on page 231
• ctl-service admin, on page 232
call application voice aa-hunt

Effective with Cisco IOS Release 12.3(14)T and later, the call application voice aa-hunt command is replaced by the param aa-hunt command. See the param aa-hunt command for more information.

To declare a Cisco Unified CME basic automatic call distribution (B-ACD) menu number and associate it with the pilot number of an ephone hunt group, use the call application voice aa-hunt command in global configuration mode. To remove the menu number and the ephone hunt group pilot number, use the no form of this command.

call application voice application-name aa-hunt menu-number pilot-number
no call application voice application-name aa-hunt menu-number pilot-number

Syntax Description

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>menu-number</td>
<td>Number that callers must dial to reach the pilot number of an ephone hunt group. The range is from 1 to 10. The default is 1.</td>
</tr>
<tr>
<td>application-name</td>
<td>Application name given to the call queue script in the call application voice command.</td>
</tr>
<tr>
<td>pilot-number</td>
<td>Ephone hunt group pilot number.</td>
</tr>
</tbody>
</table>

Command Default
Cisco CME B-ACD menu number 1 is configured, but it has no pilot number.

Command Modes
Global configuration (config)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.2.2</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was replaced with the param aa-hunt command.</td>
</tr>
</tbody>
</table>

Usage Guidelines

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD call queue scripts. Up to three menu options are allowed per call queue script. You can use any of the allowable numbers in any order.

The call application voice aa-hunt command allows each of the menu options to be associated with the pilot number of an ephone hunt group. The menu options are announced by the en_bacd_options_menu.au audio file, which you can rerecord. When a caller presses a number, the call will go to the pilot number of an ephone hunt group so it can be transferred to one of the ephone hunt group's ephone-dns. It will not go to any other ephone hunt group. The order in which ephone-dns are selected depends on the ephone hunt group's search method, which is configured with the ephone-hunt command, and whether an ephone-dn is busy or not.

If only one menu option is configured, callers will hear a greeting and be transferred directly to the pilot number of the corresponding ephone hunt group. They do not have to enter a number.

The highest aa-hunt number will automatically be set to zero (0) for the operator. In the following example, aa-hunt8 supports the menu option of 0 and 8.

call application voice queue aa-hunt1 1111
If a phone user presses 0 or 8, their call be sent to pilot number 3333.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

### Examples

The following example associates three menu numbers with three pilot numbers of three ephone hunt groups. Pilot number 1111 is for ephone hunt group 1 (sales); 2222 is for ephone hunt group 2 (customer service); and 3333 is for ephone hunt group 3 (operator). If sales is selected from the AA menu, the call will be transferred to 1111 and sent to ephone hunt group 1’s available ephone-dns (2001, 2002, 2003, 2004, 2005, 2006).

```
Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002, 1003, 1004, 1005, 1006, 1007, 1008, 1009, 1010
Router(config)# ephone-hunt 2 peer
Router(config-ephone-hunt)# pilot 2222
Router(config)# ephone-hunt 3 peer
Router(config-ephone-hunt)# pilot 3333
Router(config-ephone-hunt)# list 3001, 3002, 3003, 3004
Router(config)# call application voice queue flash:app-b-acd-x.x.x.x.tcl
Router(config)# call application voice queue aa-hunt1 1111
Router(config)# call application voice queue aa-hunt2 2222
Router(config)# call application voice queue aa-hunt3 3333
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>call application voice</code></td>
<td>Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.</td>
</tr>
<tr>
<td><code>call application voice aa-pilot</code></td>
<td>Associates an ephone hunt group with the Cisco CME basic service’s AA script by declaring the group’s pilot number.</td>
</tr>
<tr>
<td><code>call application voice welcome-prompt</code></td>
<td>Assigns an audio file that is used by a Cisco CME B-ACD AA script for the welcome greeting.</td>
</tr>
<tr>
<td><code>ephone-dn</code></td>
<td>Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.</td>
</tr>
<tr>
<td><code>ephone-hunt</code></td>
<td>Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco CME system.</td>
</tr>
<tr>
<td><code>pilot</code></td>
<td>Defines the ephone-dn that callers dial to reach a Cisco CME ephone hunt group.</td>
</tr>
</tbody>
</table>
call application voice aa-name

Effective with Cisco IOS Release 12.3(14)T and later, the call application voice aa-name command is not available in Cisco IOS software.

To associate the queue script for Cisco Unified CME basic automatic call distribution (B-ACD) with the Cisco Unified CME B-ACD auto-attendant (AA) script, use the call application voice aa-name command in global configuration mode. To remove the queue script and AA script association, use the no form of this command.

```
call application voice application-name aa-name aa-script-name
no call application voice application-name aa-name aa-script-name
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application-name</td>
<td>Application name given to the call queue script in the call application voice command.</td>
</tr>
<tr>
<td>aa-script-name</td>
<td>Application name given to the AA script in the call application voice command.</td>
</tr>
</tbody>
</table>

**Command Default**

No call queue script and AA script association is configured.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.2.2</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was replaced with the param aa-name command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD call queue scripts. Only one AA script can be associated with one call queue script.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

**Examples**

The following example associates a call queue script with an AA script:

```
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice queue flash:app-b-acd-x.x.x.x.tcl
Router(config)# call application voice queue aa-name aa
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call application voice</td>
<td>Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.</td>
</tr>
<tr>
<td>call application voice service-name</td>
<td>Associates a Cisco CME B-ACD AA script with a Cisco Unified CME B-ACD call queue script.</td>
</tr>
</tbody>
</table>
call application voice aa-pilot

Effective with Cisco IOS Release 12.3(14)T and later, the call application voice aa-pilot command is replaced by the param aa-pilot command. See the param aa-pilot command for more information.

To assign a pilot number to the Cisco Unified CME basic automatic call distribution (B-ACD) service, use the call application voice aa-pilot command in global configuration mode. To remove the Cisco Unified CME B-ACD pilot number, use the no form of this command.

`call application voice application-name aa-pilot pilot-number`

`no call application voice application-name aa-pilot pilot-number`

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>application-name</strong></td>
<td>Application name given to the auto-attendant (AA) script in the call application voice command.</td>
</tr>
<tr>
<td><strong>pilot-number</strong></td>
<td>Pilot number for Cisco CME B-ACD.</td>
</tr>
</tbody>
</table>

**Command Default**

No Cisco Unified CME B-ACD pilot number is configured.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
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<td>This command was introduced.</td>
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<td>Cisco CME 3.2.2</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was replaced by the param aa-pilot command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD AA scripts. Only one pilot number can be used for each Cisco Unified CME B-ACD service, and the voice ports handling AA must have dial peers that will send calls to the pilot number.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

**Examples**

The following example assigns 8005550100 as the pilot number to the Cisco Unified CME B-ACD service. Included in this example is the dial-peer configuration for the pilot number.

Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa aa-pilot 8005550100
Router(config)# dial-peer voice 1000 pots
Router(config)# incoming pilot number 8005550100
Router(config)# application aa
Router(config)# direct-inward-dial
Router(config)# port 1/0:23
Router(config)# forward digits-all
Router(config)# call application voice aa aa-pilot 80055501
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>call application voice</strong></td>
<td>Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.</td>
</tr>
<tr>
<td><strong>dial-peer voice</strong></td>
<td>Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode.</td>
</tr>
<tr>
<td><strong>ephone-dn</strong></td>
<td>Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.</td>
</tr>
<tr>
<td><strong>ephone-hunt</strong></td>
<td>Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
call application voice call-retry-timer

Effective with Cisco IOS Release 12.3(14)T and later, the call application call-retry-timer command is replaced by the param call-retry-timer command. See the param call-retry-timer command for more information.

To assign the length of time that calls to Cisco Unified CME basic automatic call distribution (B-ACD) must wait before attempting to transfer to an ephone hunt group pilot number, use the call application voice call-retry-timer command in global configuration mode. To remove the retry time, use the no form of this command.

call application voice application-name call-retry-timer seconds
no call application voice application-name call-retry-timer seconds

Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application-name</td>
<td>Application name given to the auto-attendant (AA) script in the call application voice command.</td>
</tr>
<tr>
<td>seconds</td>
<td>Number of seconds that a call must wait before attempting to transfer an ephone hunt pilot number or voice-mail pilot number. The range is from 5 to 30 seconds. The default is 15 seconds.</td>
</tr>
</tbody>
</table>

Command Default

The retry interval is 15 seconds.

Command Modes

Global configuration (config)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
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<td>12.3(14)T</td>
<td>Cisco CME 3.2.2</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was replaced by the param call-retry-timer command</td>
</tr>
</tbody>
</table>

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD AA scripts. The following sequence of events might occur:

- An outside call comes into a system configured with Cisco CME B-ACD.
- A menu option is selected that attempts to transfer the call to an ephone hunt group pilot number.
- All of the ephone hunt group’s ephone-dns are busy.

In that case, the call will wait in a queue for the period of time set by the call application voice call-retry-timer command and retry to the pilot number.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

Examples

The following example shows a configuration that allows outside calls to Cisco CME B-ACD to retry an ephone hunt group pilot number every 30 seconds. The example shows the configuration for one ephone hunt group, which is presented by Cisco CME B-ACD menu as the sales department and uses a simple configuration. If a caller selects the sales menu option (ephone-hunt 1) and all of
the ephone-dns configured in the `list` command (1001, 1002, 1003, 1004) are busy, the call will wait 30 seconds and then retry the pilot number (1111) until either an ephone-dn becomes available or a configured amount of time has elapsed (see the `call application voice max-time-call-retry` command).

Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002, 1003, 1004
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa call-retry-timer 30

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-dn</td>
<td>Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.</td>
</tr>
<tr>
<td>ephone-hunt</td>
<td>Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>call application voice</td>
<td>Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.</td>
</tr>
<tr>
<td>call application voice aa-hunt</td>
<td>Declares a Cisco Unified CME B-ACD menu number and associates it with the pilot number of an ephone hunt group.</td>
</tr>
<tr>
<td>call application voice aa-pilot</td>
<td>Associates an ephone hunt group with the Cisco Unified CME basic service’s AA script by declaring the group’s pilot number.</td>
</tr>
<tr>
<td>call application voice max-time-call-retry</td>
<td>Assigns the maximum length of time for which calls to Cisco Unified CME B-ACD can stay in a call queue.</td>
</tr>
</tbody>
</table>
call application voice dial-by-extension-option

Effective with Cisco IOS Release 12.3(14)T and later, the `call application voice dial-by-extension-option` command is replaced by the `param dial-by-extension-option` command. See the `param dial-by-extension-option` command for more information.

To enable direct extension access and set the access number for Cisco Unified CME basic automatic call distribution (B-ACD), use the `call application voice dial-by-extension-option` command in global configuration mode. To disable direct dial extension access and remove the access number, use the `no` form of this command.

```
call application voice application-name dial-by-extension number
no call application voice application-name dial-by-extension number
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>application-name</code></td>
<td>Application name given to the auto-attendant (AA) script in the <code>call application voice</code> command.</td>
</tr>
<tr>
<td><code>number</code></td>
<td>The single digit that callers press to be able to enter an extension number from the AA menu. The range is from 1 to 10. There is no default.</td>
</tr>
</tbody>
</table>

**Command Default**

Direct dial access is disabled. No access number is configured.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.2.2</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was replaced by the <code>param dial-by-extension-option</code> command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD AA scripts. It enables the `en_bacd_enter_dest.au` audio file. The default announcement says, “Please enter the extension number you want to reach.” The `call application voice dial-by-extension-option` command also allows for the configuration of the number that callers must press before they can enter the extension number that they want to call.

Callers who select the extension access option can then dial any extension. If they dial an ephone hunt group ephone-dn or pilot number, their call will not be sent to the ephone hunt-group call queue.

**Examples**

The following example configures Cisco CME B-ACD to include an option that allows callers to press the number 4 so they can dial an extension number.

```
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa dial-by-extension 4
```
call application voice dial-by-extension-option

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call application voice</td>
<td>Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.</td>
</tr>
</tbody>
</table>
**call application voice drop-through-option**

Cisco IOS Release 12.3(14)T and later releases support Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tcl scripts version 2.1.0.0 and greater. In these releases, the `call application voice drop-through-option` command has been replaced by the `param drop-through-option` command.
call application voice drop-through-prompt

Cisco IOS Release 12.3(14)T and later releases support Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tel scripts version 2.1.0.0 and greater. In these releases, the `call application voice drop-through-prompt` command has been replaced by the `param drop-through-prompt` command.
call application voice handoff-string

Cisco IOS Release 12.3(14)T and later releases support Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tcl scripts version 2.1.0.0 and greater. In these releases, the call application voice handoff-string command has been replaced by the param handoff-string command.
call application voice max-extension-length

Cisco IOS Release 12.3(14)T and later releases support Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tel scripts version 2.1.0.0 and greater. In these releases, the call application voice max-extension-length command has been replaced by the param max-extension-length command.
call application voice max-time-call-retry

Effective with Cisco IOS Release 12.3(14)T and later, the call application voice max-time-call-retry command is replaced by the param max-time-call-retry command. See the param max-time-call-retry command for more information.

To assign the maximum length of time for which calls to Cisco Unified CME basic automatic call distribution (B-ACD) can stay in a call queue, use the call application voice max-time-call-retry command in global configuration mode. To remove the maximum length of time, use the no form of this command.

Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>application-name</strong></td>
<td>Application name given to the auto attendant (AA) script in the call application voice command.</td>
</tr>
<tr>
<td><strong>seconds</strong></td>
<td>Maximum length of time that the Cisco Unified CME B-ACD AA script can keep redialing an ephone hunt group pilot number. The range is from 0 to 3600 seconds. The default is 600 seconds.</td>
</tr>
</tbody>
</table>

Command Default

The default maximum length of time that calls can stay in a call queue is 600 seconds.

Command Modes

Global configuration (config)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
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</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.2.2</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was replaced by the param max-time-call-retry command.</td>
</tr>
</tbody>
</table>

Usage Guidelines

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD AA scripts. The call application voice max-time-retry command allows you set a time limit for the redialing of pilot numbers under the following circumstances:

- An outside call comes into a system configured with Cisco Unified CME B-ACD.
- A menu option is selected that transfers the call to an ephone hunt-group pilot number.
- All of the ephone hunt group’s ephone-dns are busy.
- The call is sent to a queue and tries the pilot number at intervals of time set by the call application voice call-retry-timer command.

When the time period set by the call application voice max-call-retry command expires, one of the following two events will occur:

- If a voice-mail pilot number has been configured in Cisco Unified CME and mail boxes for hunt group pilot numbers have been configured in a voice-mail application, calls will be transferred to voice mail.
• If voicemail has not been configured, a default message will be played that says, “We are unable to take your call at this time. Please try again at a later time. Thank you for calling.”

### Examples

In the following example, the length of time for which calls can try to reach ephone hunt group 1 and ephone hunt group 2 is 90 seconds. If a caller selects the AA menu option for either hunt group and all of its ephone-dns configured in the `list` command are busy, the call will keep retrying the ephone hunt group’s pilot number until one of the ephone-dns is available or 90 seconds has elapsed. When 90 seconds elapses, the call will go to voice mail.

```sh
Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002, 1003, 1004
Router(config)# ephone-hunt 2 peer
Router(config-ephone-hunt)# pilot 2222
Router(config)# call application voice aa flash:app-b-acd-AA-x.x.x.x.tcl
Router(config)# call application voice aa max-call-retry-timer 90
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>call application voice</code></td>
<td>Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.</td>
</tr>
<tr>
<td><code>call application voice call-retry-timer</code></td>
<td>Assigns the length of time that calls to Cisco Unified CME B-ACD must wait before attempting to transfer to an ephone hunt-group pilot number or to voice mail.</td>
</tr>
<tr>
<td><code>call application voice max-time-vm-retry</code></td>
<td>Assigns the maximum number of times that calls to Cisco Unified CME B-ACD can attempt to reach voice mail.</td>
</tr>
<tr>
<td><code>ephone-dn</code></td>
<td>Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.</td>
</tr>
<tr>
<td><code>ephone-hunt</code></td>
<td>Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
call application voice max-time-vm-retry

Cisco IOS Release 12.3(14)T and later releases support Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tcl scripts version 2.1.0.0 and greater. In these releases, the call application voice max-time-vm-retry command has been replaced by the param max-time-vm-retry command.
**call application voice number-of-hunt-grps**

Effective with Cisco IOS Release 12.3(14)T and later, the `call application voice number-of-hunt-grps` command is replaced by the `param number-of-hunt-grps` command. See the `param number-of-hunt-grps` command for more information.

To declare the maximum number of ephone hunt-group menus supported by Cisco Unified CME basic automatic call distribution (B-ACD), use the `call application voice number-of-hunt-grps` command in global configuration mode. To remove the maximum number of ephone hunt-group menus supported by Cisco CME B-ACD, use the `no` form of this command.

```
call application voice application-name number-of-hunt-grps number
no call application voice application-name number-of-hunt-grps number
```

**Syntax Description**

- **application-name** Application name given to the auto-attendant (AA) script in the `call application voice` command.
- **number** Number of hunt groups used by the Cisco Unified CME B-ACD AA script and call queue script. The range is from 1 to 3. The default is 3.

**Command Default**

Three ephone hunt-group menus are supported by Cisco CME B-ACD.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
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<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was replaced by the <code>param number-of-hunt-grps</code> command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD AA scripts. The `number` argument declares the number of ephone hunt groups only. The menu option for direct extension access (see the `call application voice dial-by-extension-option` command) is not included.

**Examples**

The following example configures a Cisco Unified CME B-ACD call queue script to use three ephone hunt groups and one direct extension access number, making the `number` argument in the `call application voice number-of-hunt-grps` equal to 3. The `ephone-hunt` command is used to configure the three ephone hunt groups. The `call application voice dial-by-extension-option` command is used to enable direct extension access and set the access number to 1.

```
Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002, 1003, 1004, 1005, 1006, 1007, 1008, 1009, 1010
Router(config)# ephone-hunt 2 peer
```
Router(config-ephone-hunt)# pilot 2222
Router(config-ephone-hunt)# final 9000
Router(config)# ephone-hunt 3 peer
Router(config-ephone-hunt)# pilot 3333
Router(config-ephone-hunt)# list 3001, 3002, 3003, 3004
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa dial-by-extension 1
Router(config)# call application voice aa number-of-hunt-grps 3

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>call application voice</td>
<td>Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.</td>
</tr>
<tr>
<td></td>
<td>call application voice dial-by-extension-option</td>
<td>Enables direct extension access and sets the access number for Cisco CME B-ACD.</td>
</tr>
<tr>
<td></td>
<td>ephone-hunt</td>
<td>Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco CME system.</td>
</tr>
</tbody>
</table>
call application voice queue-len

Effective with Cisco IOS Release 12.3(14)T and later, the `call application queue-len` command is replaced by the `param queue-len` command. See the `param queue-len` command for more information.

To set the maximum number of calls allowed for each ephone hunt group’s call queue that is used by Cisco Unified CME basic automatic call distribution (B-ACD), use the `call application voice queue-len command` in global configuration mode. To remove the queue-length setting, use the `no` form of this command.

```
call application voice application-name queue-len number
no call application voice application-name queue-len number
```

**Command Default**

<table>
<thead>
<tr>
<th>application-name</th>
<th>Application name given to the call queue script in the <code>call application voice command</code>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>number</td>
<td>Number of calls that can be waiting in each ephone hunt group’s queue. The range is dependent on your hardware configuration. The range is from 1 to 30. The default is 10.</td>
</tr>
</tbody>
</table>

**Command Default**

Thirty calls are allowed in each call queue.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.2.2</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was replaced by the <code>param queue-len</code> command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD call queue scripts. The following sequence of events might occur:

- An outside call comes into a system configured with Cisco Unified CME B-ACD.
- A menu option is selected that transfers the call to an ephone hunt-group pilot number.
- All of the ephone hunt group’s ephone-dns are busy.

In that case, the call will be sent to a queue for that individual hunt group. The number of calls that each ephone hunt group can hold in its queue is configured by the `call application voice queue-len` command.

In the following configuration example, ephone hunt group 1 supports two ephone-dns; ephone hunt group 2 supports three ephone-dns; and the queue length is 10 for both ephone hunt groups:

```
ephone-hunt 1 peer
   pilot 1111
   list 1001, 1002

ephone-hunt 2 peer
   pilot 2222

call application voice queue flash:app-b-acd-x.x.x.x.tcl

call application voice callqueuescriptfilename queue-len 10
```
If ephone hunt group 1’s ephone-dns are busy, ten more calls can be made to ephone hunt group 1. During that time, the calls in the queue would periodically retry the pilot numbers (**call application voice max-time-retry-timer** command) and receive secondary greetings (**call application voice second-greeting-time** command). If none of the calls has hung up or connected to an ephone-dn, the eleventh caller would hear the en_bacd_disconnect.au message and a busy signal. The default message is, “We are unable to take your call at this time. Please try again at a later time. Thank you for calling.” Includes a four-second pause after the message.

For ephone hunt group 2, three calls can be connected to ephone-dns 2001, 2002, and 2003, and ten calls can be waiting in ephone hunt group 2’s queue. If the status remains unchanged, the fourteenth caller hears the disconnect message and a busy signal. But if one of the earlier calls disconnects (either by leaving the queue or by ending a call), the fourteenth call enters the queue.

The maximum number of calls allowed in the queues of ephone hunt groups must be based on the number of ports or trunks available. For example, if you had 20 foreign exchange office (FXO) ports and two ephone hunt groups, you could configure a maximum of ten calls per ephone hunt-group queue with the **call application voice queue-len 10** command. You could use the same configuration if you had a single T1 trunk, which supports 23 channels.

### Examples

The following example configures a Cisco Unified CME B-ACD call queue script to allow a maximum of 12 calls to wait in each ephone hunt group’s calling queue for ephone-dns to become available:

```
Router(config)# call application voice queue flash:app-b-acd-x.x.x.x.tcl
Router(config)# call application voice queue queue-len 12
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call application voice</td>
<td>Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.</td>
</tr>
<tr>
<td>call application voice</td>
<td>Assigns the length of time that calls to Cisco CME B-ACD must wait before attempting to transfer to an ephone hunt-group pilot number or to voice mail.</td>
</tr>
<tr>
<td>call-retry-timer</td>
<td>Assigns the length of time that calls to Cisco CME B-ACD must wait before attempting to transfer to an ephone hunt-group pilot number or to voice mail.</td>
</tr>
<tr>
<td>ephone-dn</td>
<td>Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.</td>
</tr>
<tr>
<td>ephone-hunt</td>
<td>Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco CME system.</td>
</tr>
</tbody>
</table>
call application voice queue-manager-debugs

Effective with Cisco IOS Release 12.3(14)T and later, the call application queue-manager-debugs command is replaced by the param queue-manager-debugs command. See the param aa-hunt command for more information.

To enable or disable the collection of call queue debug information from Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD), use the call application voice queue-manager-debugs command in global configuration mode. To remove the current setting, use the no form of this command with the keyword that was used in the previous occurrence of the call application voice queue-manager-debugs command.

call application voice application-name queue-manager-debugs [{0|1}]
no call application voice application-name queue-manager-debugs [{0|1}]

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application-name</td>
<td>Application name given to the call queue script in the call application voice command.</td>
</tr>
<tr>
<td>0</td>
<td>Enables debugging.</td>
</tr>
<tr>
<td>1</td>
<td>Enables debugging.</td>
</tr>
</tbody>
</table>

**Command Default**
The collection of call queue debug information from Cisco CME B-ACD is disabled.

**Command Modes**
Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.2.2</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was replaced by the param queue-manager-debugs command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD call queue scripts. It enables the collection of data regarding call queue activity. It is used in conjunction with the debug voip ivr script command. Both commands must be enabled at the same time.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

**Examples**
The following example configures a Cisco CME B-ACD call queue script to enable debugging for the collection of data for the debug voip ivr script command:

```
Router(config)# call application voice queue flash:app-b-acd-x.x.x.x.tcl
Router(config)# call application voice queue queue-manager-debugs 1
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>call application voice</code></td>
<td>Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.</td>
</tr>
<tr>
<td><code>debug voip ivr script</code></td>
<td>Display debugging messages for IVR scripts.</td>
</tr>
</tbody>
</table>
call application voice second-greeting-time

Effective with Cisco IOS Release 12.3(14)T and later, the call application voice second-greeting-time command is replaced by the param second-greeting-time command. See the param second-greeting-time command for more information.

To set the delay before the second greeting is played after a caller joins a Cisco Unified CME basic automatic call distribution (B-ACD) calling queue and set the interval of time at which the second-greeting message is repeated, use the call application voice second-greeting-time command in global configuration mode. To remove the second-greeting time, use the no form of this command.

Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application-name</td>
<td>Application name given to the auto-attendant (AA) script in the call application voice command.</td>
</tr>
<tr>
<td>seconds</td>
<td>Amount of time that second-greeting message must wait before it can be played. The range is from 30 to 120 seconds. The default is 60 seconds.</td>
</tr>
</tbody>
</table>

Command Default: The second-greeting delay time is 60 seconds.

Command Modes: Global configuration (config)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.2.2</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was replaced by the param second-greeting-time command.</td>
</tr>
</tbody>
</table>

Usage Guidelines

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD AA scripts. A second greeting is an audio message of up to 15 seconds in length. The default announcement is, “All agents are currently busy assisting other customers. Continue to hold for assistance. Someone will be with you shortly.” The second-greeting message is only presented to callers waiting in a CME B-ACD call queue.

The second-greeting time is clocked when the second-greeting message begins, not after it ends. For example, if the second greeting were 15 seconds in length and the configured second-greeting time were 70 seconds, the greeting would begin every 70 seconds, not 85 seconds as if to allow for the 15-second message.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

Examples

The following example configures a Cisco Unified CME B-ACD AA script to allow a second-greeting message to be repeated every 50 seconds as long as a call is in a call queue.
```markdown
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.tcl
Router(config)# call application voice AAscriptfilename second-greeting-time 50
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call application voice</td>
<td>Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.</td>
</tr>
<tr>
<td>ephone-dn</td>
<td>Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.</td>
</tr>
<tr>
<td>ephone-hunt</td>
<td>Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco CME system.</td>
</tr>
</tbody>
</table>
call application voice service-name

Cisco IOS Release 12.3(14)T and later releases support Cisco CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tcl scripts version 2.1.0.0 and greater. In these releases, the call application voice service-name command has been replaced by the param service-name command.
call application voice voice-mail

Effective with Cisco IOS Release 12.3(14)T and later, the call application voice voice-mail command is replaced by the param voice-mail command. See the param voice-mail command for more information.

To assign a pilot number for the Cisco Unified CME basic automatic call distribution (B-ACD) service’s voicemail, use the call application voice voice-mail command in global configuration mode. To remove the voice-mail pilot number, use the no form of the command.

```
call application voice application-name voice-mail number
no call application voice application-name voice-mail number
```

### Syntax Description

<table>
<thead>
<tr>
<th><strong>application-name</strong></th>
<th>Application name given to the auto attendant (AA) script in the call application voice command.</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>number</strong></td>
<td>Pilot number of the voice mail to which calls to Cisco CME B-ACD will be transferred.</td>
</tr>
</tbody>
</table>

### Command Default

No voice-mail pilot number is configured for Cisco Unified CME B-ACD.

### Command Modes

Global configuration (config)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.2.2</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was replaced by the param voice-mail command.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD AA scripts. Only one pilot number is allowed per Cisco CME B-ACD service. Calls to the service will be sent to this voice mail number.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

### Examples

The following example configures a Cisco Unified CME B-ACD voice-mail pilot number as 5000.

```
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa voice-mail 5000
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call application voice</td>
<td>Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.</td>
</tr>
</tbody>
</table>
call application voice welcome-prompt

effective with Cisco IOS Release 12.3(14)T and later, the call application voice welcome-prompt command is replaced by the param welcome-prompt command. See the param welcome-prompt command for more information.

To assign an audio file that is used by the Cisco Unified CME basic automatic call distribution (B-ACD) auto-attendant (AA) script for the welcome greeting, use the call application welcome-prompt command in global configuration mode. To remove the audio file assignment, use the no form of this command.

call application voice application-name welcome-prompt _ audio-filename
no call application voice application-name welcome-prompt _ audio-filename

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application-name</td>
<td>Application name given to the AA script in the call application voice command.</td>
</tr>
<tr>
<td>_audio-filename</td>
<td>Filename of the welcome greeting to be played when callers first reach the Cisco Unified CME B-ACD, preceded by the underscore (_) character. The filename must not have a language code prefix, such as “en,” for English.</td>
</tr>
</tbody>
</table>

**Command Default**
The welcome audio file downloaded with Cisco Unified CME B-ACD is used for the welcome prompt.

**Command Modes**
Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.2.2</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was replaced by the param welcome-prompt command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD AA scripts. The welcome greeting is the initial AA response to a caller. The default audio file used is en_bacd_welcome.au, which is is downloaded with Cisco CME B-ACD and announces, “Thank you for calling,” and includes a two-second pause after the message.

The filename must be preceded by an underscore (_) character. In addition, it must not contain a language-code prefix, such as “en” for English. For example, for en_bacd_welcome.au, you must configure welcome-prompt _bacd_welcome.au instead of welcome-prompt _en_bacd_welcome.au.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

**Examples**
The following example sets filename en_welcome.au as the welcome greeting for Cisco Unified CME B-ACD:

Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa welcome-prompt _bacd_welcome_2.au

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call application voice</td>
<td>Defines a name for a voice application and specifies the location of the Tel or VoiceXML document to load for this application.</td>
</tr>
<tr>
<td>call application voice aa-name</td>
<td>Associates a Cisco CME B-ACD call queue script with a Cisco Unified CME B-ACD AA script.</td>
</tr>
<tr>
<td>call application voice service-name</td>
<td>Associates a Cisco CME B-ACD AA script with a Cisco Unified CME B-ACD call queue script.</td>
</tr>
</tbody>
</table>
callback (voice emergency response settings)

To route an E911 callback to another number (for example, the company operator) if the callback cannot find the last 911 caller associated to the ELIN, use the **callback** command in voice emergency response settings configuration mode. This command is optional.

```
callback  number
no  callback
```

**Syntax Description**
- **number**: Identifier of the E.164 default number to contact if a 911 callback fails.

**Command Default**
A callback number is not defined.

**Command Modes**
Voice emergency response settings configuration (cfg-emrgncy-resp-settings)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to specify the default number to contact if a 911 callback cannot find the last 911 caller associated with an ELIN. If no default callback number is configured, and the expiry time is exceeded, the 911 operator may hear a reorder tone or be incorrectly routed.

**Examples**
In this example, the ELIN (4085550101) defined in the voice emergency response settings configuration is used if the 911 caller’s IP phone address does not match any of the voice emergency response locations. After the 911 call is placed to the PSAP, the PSAP has 120 minutes to call back 408-555-0101 to reach the 911 caller. If the call history has expired (after 120 minutes), any callback is routed to extension 7500.

```
voice  emergency  response  settings
callback  7500
elin  4085550101
expiry  120
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>elin</td>
<td>E.164 number used as the default ELIN if no matching ERL to the 911 caller’s IP phone address is found.</td>
</tr>
<tr>
<td>expiry</td>
<td>Number of minutes a 911 call is associated to an ELIN in case of a callback from the 911 operator.</td>
</tr>
<tr>
<td>logging</td>
<td>Syslog informational message printed to the console every time an emergency call is made.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>voice emergency response settings</td>
<td>Creates a tag for identifying settings for E911 behavior.</td>
</tr>
</tbody>
</table>
caller-id

To specify whether to pass the local caller ID or the original caller ID with calls from an extension in Cisco Unified CME that is using loopback, use the **callerid command** in ephone-dn configuration mode. To return to the default, use the **no** form of this command.

```
caller-id {local|passthrough}
no caller-id {local|passthrough}
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>local</td>
<td>Local caller ID for redirected calls.</td>
</tr>
<tr>
<td>passthrough</td>
<td>Original caller ID. Default.</td>
</tr>
</tbody>
</table>

### Command Default

Default is **passthrough**.

### Command Modes

Ephone-dn configuration (config-ephone)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ3</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command is valid only for ephone-dns that are being used for loopback.

This command with the **local** keyword is applied as follows:

- For transferred calls, caller ID is provided by the original caller-ID information source, such as from a separate loopback-dn that handles inbound calls or from a public switched telephone network interface.
- For forwarded calls, caller ID is provided by the original caller-ID information source or, for local IP phones, is extracted from the redirected information associated with the call.

This command with the **passthrough** keyword is applied as follows:

- For transferred calls, the caller ID is provided by the original caller-ID information that is obtained from the inbound side of the loopback-dn.
- For forwarded calls, the caller ID is provided by the original caller-ID information of the incoming call.

### Examples

The following example selects local caller ID for redirected calls:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# loopback-dn 15 forward 4
Router(config-ephone-dn)# caller-id local
Router(config-ephone-dn)# no huntstop
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>loopback-dn</strong></td>
<td>Creates a virtual loopback voice port (loopback-dn) to establish a demarcation point for VoIP voice calls and supplementary services.</td>
</tr>
</tbody>
</table>
**caller-id block (ephone-dn and ephone-dn-template)**

To specify caller-ID blocking for outbound calls from a specific extension, use the `caller-id block` command in ephone-dn or ephone-dn-template configuration mode. To disable caller-ID blocking for outbound calls, use the `no` form of this command.

```
caller-id block
no caller-id block
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Caller-ID display is not blocked on calls originating from a Cisco Unified IP phone.

**Command Modes**

- Ephone-dn configuration
- Ephone-dn-template configuration

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-dn-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command sets caller-ID blocking for outbound calls originating from a specific extension (ephone-dn). This command requests the far-end gateway device to block the display of the calling party information for calls received from the ephone-dn that is being configured. This command does not affect the ephone-dn calling party information display for inbound calls received by the ephone-dn.

If you want caller-ID name or number to be available on local calls but not on external calls, use the `clid strip name` command or the `clid strip` command in dial-peer configuration mode to remove caller-ID name or number from calls to VoIP. In this case, do not also use the `caller-id block` command, which blocks caller-ID information on all calls.

**Note**

This command is not valid for ephone-dns that are being used for loopback.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

**Examples**

The following example shows how to set caller-ID blocking for the directory number 5001:

```
Router(config)# ephone-dn 1
```
The following example uses an ephone-dn template to set caller-ID blocking for the directory number 5001:

```
Router(config) # ephone-dn-template 5
Router(config-ephone-dn-template) # caller-id block
Router(config-ephone-dn-template) # exit
Router(config) # ephone-dn 1
Router(config-ephone-dn) # number 5001
Router(config-ephone-dn) # ephone-dn-template 5
```

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>clid strip</td>
<td>Prevents display of caller-ID number on calls to VoIP.</td>
</tr>
<tr>
<td>clid strip name</td>
<td>Prevents display of caller-ID name on calls to VoIP.</td>
</tr>
<tr>
<td>ephone-dn-template (ephone-dn)</td>
<td>Applies ephone-dn template to an ephone dn.</td>
</tr>
</tbody>
</table>
caller-id block (voice register template)

**Note**
Effective with Cisco IOS Release 12.4(11)XJ, the caller-id block (voice register template) command is not available in Cisco IOS software.

To enable caller-ID blocking for outbound calls from a specific SIP phone, use the **caller-id block** command in voice register template configuration mode. To disable caller-ID blocking, use the **no** form of this command.

caller-id block  
no caller-id block

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Caller ID blocking is disabled.

**Command Modes**
Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was removed.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was removed in Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command sets caller-ID blocking for outbound calls originating from any SIP phone that uses the specified template. This command requests the far-end gateway device to block the display of the calling party information for calls received from the specified SIP phone. This command does not affect the calling party information displayed for inbound calls received by the SIP phone. To apply a template to a SIP phone, use the **template** command in voice register pool configuration mode.

**Examples**
The following example shows how to enable caller-ID blocking in template 1:

Router(config)# voice register template 1  
Router(config-register-temp)# caller-id block

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>anonymous block (voice register template)</td>
<td>Enables anonymous call blocking in a SIP phone template.</td>
</tr>
<tr>
<td>template (voice register pool)</td>
<td>Applies a template to a SIP phone.</td>
</tr>
<tr>
<td>voice register template</td>
<td>Enters voice register template configuration mode and defines a template of common parameters for SIP phones.</td>
</tr>
</tbody>
</table>
**caller-id block code (telephony-service)**

To set a code for a user to dial to block the display of caller ID on selected outgoing calls from Cisco IP phones, use the `caller-id block code` command in telephony-service configuration mode. To remove the code, use the `no` form of this command.

```
caller-id block code  code-string
no caller-id block code
```

### Syntax Description

| code-string | Character string to dial to enable blocking of caller ID display on selected outgoing calls. The first character must be an asterisk (*) and the remaining characters must be digits. The string can contain a maximum of 16 characters. |

### Command Default

No caller-ID blocking code is defined.

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Once the caller-ID blocking code has been defined using this command, phone users should enter the caller-ID blocking code before dialing any call on which they want their caller ID not to display.

### Examples

The following example sets a caller-ID blocking code of *4321:

```
Router(config)# telephony-service
Router(config-telephony)# caller-id block code *4321
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>telephony-service</td>
<td>Enters telephony-service configuration mode.</td>
</tr>
</tbody>
</table>
call-feature-uri

To specify the uniform resource identifier (URI) for soft keys on SIP phones registered to a Cisco Unified CME router, use the `call-feature-uri` command in voice register global configuration mode. To remove a URI association, use the `no` form of this command.

```
call-feature-uri {cfwdall|gpickup|pickup} service-uri
no call-feature-uri cfwdall {cfwdall|gpickup|pickup}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>cfwdall</code></td>
<td>Call Forward All (CfwdAll) soft key.</td>
</tr>
<tr>
<td><code>gpickup</code></td>
<td>Group Pickup (GPickUp) soft key.</td>
</tr>
<tr>
<td><code>pickup</code></td>
<td>Local Pickup (PickUp) soft key.</td>
</tr>
<tr>
<td><code>service-uri</code></td>
<td>URI that is requested when the specified soft key is pressed.</td>
</tr>
</tbody>
</table>

**Command Default**

No URI is associated with the soft key.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>The <code>gpickup</code> and <code>pickup</code> keywords were added.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command has been integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command updates the service URI for soft keys in the configuration file that is downloaded from the Cisco Unified CME router to the SIP phones during phone registration.

For Call Forward All, this URI and the call forward number is sent to Cisco Unified CME when a user enables Call Forward All from the phone using the CfwdAll soft key.

After you configure this command, restart the phone by using the `reset` or `restart` command.

This command is not supported on the Cisco Unified IP Phone 7905, 7912, 7940, or 7960.

**Examples**

The following example shows how to specify the URI for the call forward all soft key:

```
Router(config)# voice register global
Router(config-register-global)# call-feature-uri cfwdall http://10.10.10.11/cfwdall
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>call-forward b2bua all</td>
<td>Enables call forwarding for a SIP back-to-back user agent (B2BUA) so that all incoming calls are forwarded to another extension.</td>
</tr>
<tr>
<td></td>
<td>reset (voice register global)</td>
<td>Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td></td>
<td>reset (voice register pool)</td>
<td>Performs a complete reboot of one phone associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td></td>
<td>restart (voice register)</td>
<td>Performs a fast restart of one or all SIP phones associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td></td>
<td>service directed-pickup</td>
<td>Enables Directed Call Pickup and modifies the function of the PickUp and GPickUp soft keys.</td>
</tr>
</tbody>
</table>
call-forward

To globally apply dialplan-pattern expansion to redirecting numbers for extension numbers associated with SCCP IP phones in Cisco Unified CME, use the `call-forward system` command in telephony-service configuration mode. To disable the `call-forward system` command, use the `no` form of this command.

```
call-forward system redirecting-expanded
do call-forward system redirecting-expanded
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>system</td>
<td>Call forward system parameter.</td>
</tr>
<tr>
<td>redirecting-expanded</td>
<td>Expand redirecting extensions to an E.164 number.</td>
</tr>
</tbody>
</table>

**Command Default**
The redirecting number is not expanded.

**Command Modes**
Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to apply dialplan-pattern expansion on a per-system basis to individual nonSIP redirecting numbers, including original called and last reroute numbers, in a Cisco Unified CME system.

When A calls B, and B forwards the call to C; B is the original called number and the last reroute number. If C then forwards or transfers the call to another number, C becomes the original called number and the last reroute number. The dial-plan pattern expansion is applied to both redirecting numbers. Once the number is expanded, it remains expanded during the entire call instance.

The dial-plan pattern to be matched must be configured using the `dialplan-pattern` command.

**Examples**

The following example shows how to create a dialplan-pattern for expanding calling numbers to an E.164 number and to also apply the expansion globally to redirecting numbers.

```
Router(config)# voice register global
Router(config-register-global)# dialplan-pattern 1 5105550... extension-length 5
Router(config-register-global)# call-forward system redirecting-expanded
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dialplan-pattern</td>
<td>Create global prefix for expanding extension numbers of forward-to and transfer-to targets.</td>
</tr>
<tr>
<td>show telephony-service dial-peer</td>
<td>Displays dial peer information for extensions in a Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
call-forward (voice register)

To globally apply dialplan-pattern expansion to redirecting numbers for extension numbers associated with SIP IP phones in Cisco Unified CME, use the **call-forward system** command in voice register global configuration mode. To disable the **call-forward system** command, use the **no** form of this command.

**call-forward system redirecting-expanded**  
**no call-forward system redirecting-expanded**

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>system</strong></td>
<td>Call forward system parameter.</td>
</tr>
<tr>
<td><strong>redirecting-expanded</strong></td>
<td>Redirecting extension is to be expanded to an E.164 number.</td>
</tr>
</tbody>
</table>

**Command Default**

The redirecting number is not expanded.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9).</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to apply dialplan-pattern expansion on a per-system basis to individual SIP redirecting numbers, including original called and last reroute numbers, in Cisco Unified CME.

When A calls B, and B forwards the call to C; B is the original called number and the last reroute number. If C then forwards or transfers the call to another number, C becomes the original called number and the last reroute number. The dial-plan pattern expansion is applied to both redirecting numbers. Once the number is expanded, it remains expanded during the entire call instance.

This command supports call forward using B2BUA only.

The dial-plan pattern to be matched must be configured using the **dialplan-pattern** command.

**Examples**

The following example shows how to create a dialplan-pattern for expanding calling numbers of SIP phones to an E.164 number and to also apply the expansion globally to SIP redirecting numbers.

```
Router(config)# voice register global
Router(config-register-global)# dialplan-pattern 1 5105550... extension-length 5
Router(config-register-global)# call-forward system redirecting-expanded
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>dialplan-pattern (voice register)</strong></td>
<td>Create global prefix for expanding extension numbers of forward-to and transfer-to targets if the target is an extension on a SIP phone.</td>
</tr>
<tr>
<td><strong>show voice register dial-peer</strong></td>
<td>Displays dial peer information for extensions in a Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
**call-forward all**

To configure call forwarding so that all incoming calls to a directory number are forwarded to another directory number, use the **call-forward all command** in ephone-dn or ephone-dn-template configuration mode. To disable call forwarding, use the **no** form of this command.

```
call-forward all directory-number
no call-forward all
```

**Syntax Description**

<table>
<thead>
<tr>
<th><strong>directory-number</strong></th>
<th>Directory number to which calls are forwarded. Represents a fully qualified E.164 number.</th>
</tr>
</thead>
</table>

**Command Default**

Call forwarding for all calls is not set.

**Command Modes**

- Ephone-dn configuration (config-ephone-dn)
- Ephone-dn-template configuration (config-ephone-dn-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-dn-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The call forwarding mechanism applies to the individual directory number and cannot be configured for individual Cisco Unified IP phones.

**Note**

The **callforward all** command takes precedence over the **call-forward busy** and **call-forward noan** commands.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

**Examples**

The following example shows how to set call forwarding of all calls on directory number 5001 to directory number 5005. All incoming calls destined for extension 5001 are forwarded to another Cisco IP phone with the extension number 5005:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# call-forward all 5005
```
The following example uses an ephone-dn template to forward all calls for extension 5001 to extension 5005.

Router(config)# ephone-dn-template 3
Router(config-ephone-dn-template)# call-forward all 5005
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# ephone-dn-template 3

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>callforward busy</td>
<td>Configures call forwarding to another number when a Cisco Unified IP phone is busy.</td>
</tr>
<tr>
<td>callforward noan</td>
<td>Configures call forwarding to another number when no answer is received from a Cisco Unified IP phone.</td>
</tr>
<tr>
<td>ephonedn-template (ephone-dn)</td>
<td>Applies template to ephone-dn.</td>
</tr>
</tbody>
</table>
call-forward b2bua all

To enable call forwarding for a Session Initiation Protocol (SIP) back-to-back user agent (B2BUA) so that all incoming calls are forwarded to another extension, use the `call-forward b2bua all` command in voice register dn or voice register pool configuration mode. To disable call forwarding, use the `no` form of this command.

```plaintext
call-forward b2bua all directory-number
no call-forward b2bua all
```

**Syntax Description**

- `directory-number`: Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.

**Command Default**

Feature is disabled.

**Command Modes**

- Voice register dn configuration (config-register-dn)
- Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was removed from voice register pool configuration mode for Cisco Unified CME only.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>Command with modifications was integrated into Cisco IOS release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command in voice register dn configuration mode applies the call forward mechanism to an individual SIP extension in Cisco Unified CME or Cisco Unified SIP SRST. This command in voice register pool configuration mode is for Cisco Unified SIP SRST only and applies to SIP IP phones on which the extension appears.

If this command is configured in both the voice register dn and voice register pool configuration modes, the configuration under voice register dn takes precedence.

We recommend that you do not use this command to configure a SIP extension or SIP IP phone that is a member of a hunt group. If this command is configured for a member of a hunt group, remove the phone from any hunt group to which it is assigned to avoid forwarding calls to all phones in the hunt group.

The `call-forward b2bua all` command takes precedence over the `call-forward b2bua busy` and `call-forward b2bua noan` commands.

**Note**

This command in voice register dn configuration mode is not commonly used for Cisco Unified SIP SRST.
Cisco Unified CME and Cisco Unified SIP SRST

The following example shows how to forward all incoming calls to extension 5001 on directory number 4, to extension 5005.

Router(config)# voice register dn 4  
Router(config-register-dn)# number 5001  
Router(config-register-dn)# call-forward b2bua all 5005

Cisco Unified SIP SRST

The following example shows how to forward all incoming calls for extension 5001 on pool number 4, to extension 5005.

Router(config)# voice register pool 4  
Router(config-register-pool)# number 5001  
Router(config-register-pool)# call-forward b2bua all 5005

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-forward b2bua busy</td>
<td>Enables call forwarding for a SIP B2BUA so that incoming calls to a busy extension are forwarded to another extension.</td>
</tr>
<tr>
<td>call-forward b2bua mailbox</td>
<td>Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.</td>
</tr>
<tr>
<td>call-forward b2bua noan</td>
<td>Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that does not answer after a configured amount of time are forwarded to another extension.</td>
</tr>
<tr>
<td>call-waiting (voice register pool)</td>
<td>Enables call waiting on a SIP phone.</td>
</tr>
</tbody>
</table>
call-forward b2bua busy

To enable call forwarding for a Session Initiation Protocol (SIP) back-to-back user agent (B2BUA) so that incoming calls to a busy extension are forwarded to another extension, use the `callforward b2bua busy` command in voice register dn or voice register pool configuration mode. To disable call forwarding, use the `no` form of this command.

```
call-forward b2bua busy directory-number
no call-forward b2bua busy
```

**Syntax Description**

<table>
<thead>
<tr>
<th><code>directory-number</code></th>
<th>Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.</th>
</tr>
</thead>
</table>

**Command Default**

Feature is disabled.

**Command Modes**

- Voice register dn configuration (config-register-dn)
- Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was removed from voice register pool configuration mode for Cisco Unified CME only.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command with modifications was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command in voice register dn configuration mode applies the call forward mechanism to a individual SIP extension in Cisco Unified CME or Cisco Unified SIP SRST that is off-hook. This command in voice register pool configuration mode is for Cisco Unified SIP SRST only and applies to SIP IP phones on which the extension appears.

In Cisco Unified CME, call forward busy is also invoked when a call arrives for a destination that is configured but unregistered. A destination is considered to be configured if its number is listed under the voice register dn configuration.

If this command is configured in both voice register dn and voice register pool configuration modes, the configuration under voice register dn takes precedence.

We recommend that you do not use this command to configure a SIP extension or SIP IP phone that is a member of a hunt group. If this command is configured for a member of a hunt group, remove the phone from any hunt group to which it is assigned to avoid forwarding calls to all phones in the hunt group.

The `call-forward b2bua all` command takes precedence over the `call-forward b2bua busy` and `call-forward b2bua noan` commands.
This command in voice register dn configuration mode is not commonly used for Cisco Unified SIP SRST.

**Cisco Unified CME and Cisco Unified SIP SRST**

The following example shows how to forward all incoming calls to extension 5001 on directory number 4 to extension 5005 when extension 5001 is busy.

```
Router(config)# voice register dn 4
Router(config-register-dn)# number 5001
Router(config-register-dn)# call-forward b2bua busy 5005
```

**Cisco Unified SIP SRST**

The following example shows how to forward calls from extension 5001 in pool 4 to extension 5005 when extension 5001 is busy.

```
Router(config)# voice register pool 4
Router(config-register-pool)# number 5001
Router(config-register-pool)# call-forward b2bua busy 5005
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>call-forward b2bua all</code></td>
<td>Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.</td>
</tr>
<tr>
<td><code>call-forward b2bua mailbox</code></td>
<td>Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.</td>
</tr>
<tr>
<td><code>call-forward b2bua noan</code></td>
<td>Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that does not answer after a configured amount of time are forwarded to another extension.</td>
</tr>
<tr>
<td><code>call-waiting (voice register pool)</code></td>
<td>Enables call waiting on a SIP phone.</td>
</tr>
</tbody>
</table>
call-forward b2bua mailbox

To control the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange, use the `call-forward b2bua mailbox` command in voice register dn or voice register pool configuration mode. To disable call forwarding, use the `no` form of this command.

```
call-forward b2bua mailbox directory-number
no call-forward b2bua mailbox
```

### Syntax Description

<table>
<thead>
<tr>
<th><strong>Syntax</strong></th>
<th><strong>Description</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><code>directory-number</code></td>
<td>Telephone number to which calls are forwarded when the forwarded destination is busy or does not answer. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.</td>
</tr>
</tbody>
</table>

### Command Default

Feature is disabled.

### Command Modes

- Voice register dn configuration (config-register-dn)
- Voice register pool configuration (config-register-pool)

### Command History

<table>
<thead>
<tr>
<th><strong>Cisco IOS Release</strong></th>
<th><strong>Cisco Product</strong></th>
<th><strong>Modification</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was removed from voice register pool configuration mode for Cisco Unified CME only.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command with modifications was integrated into Cisco IOS Release 12.4(15)T</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command is used to denote the voice-mail box to use at the end of a chain of call forwards to busy or no answer destinations. It can be used to forward calls to a voice-mail box that has a different number than the forwarding extension, such as a shared voice-mail box.

This command in voice register dn configuration mode applies the call forward mechanism to a individual SIP extension in Cisco Unified CME or Cisco Unified SIP SRST. This command in voice register pool configuration mode is for Cisco Unified SIP SRST only and applies to SIP IP phones on which the extension appears.

If this command is configured in both the voice register dn and voice register pool configuration modes, the configuration under voice register dn takes precedence.

We recommend that you do not use this command to configure a SIP extension or SIP IP phone that is a member of a hunt group. If this command is configured for a member of a hunt group, remove the phone from any hunt group to which it is assigned to avoid forwarding calls to all phones in the hunt group.

This command is used in conjunction with the `call-forward b2bua all`, `call-forward b2bua busy`, and `call-forward b2bua noan` commands.
This command in voice register dn configuration mode is not commonly used for Cisco Unified SIP SRST.

### Cisco Unified CME and Cisco Unified SIP SRST

The following example shows how to forward all incoming calls to extension 5005 if an incoming call is forwarded to extension 5001, and extension 5001 is busy or does not answer.

```
Router(config)# voice register dn 4
Router(config-register-dn)# number 5001
Router(config-register-dn)# call-forward b2bua mailbox 5005
```

### Cisco Unified SIP SRST

The following example shows how to forward calls to extension 5005 if an incoming call is forwarded to extension 5001 on pool number 4, and extension 5001 is busy or does not answer.

```
Router(config)# voice register pool 4
Router(config-register-pool)# number 5001
Router(config-register-pool)# call-forward b2bua mailbox 5005
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-forward b2bua all</td>
<td>Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.</td>
</tr>
<tr>
<td>call-forward b2bua busy</td>
<td>Enables call forwarding for a SIP B2BUA so that incoming calls to a busy extension are forwarded to another extension.</td>
</tr>
<tr>
<td>call-forward b2bua noan</td>
<td>Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that does not answer after a configured amount of time are forwarded to another extension.</td>
</tr>
<tr>
<td>call-forward b2bua unreachable</td>
<td>Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that is not registered in Cisco Unified CME are forwarded to another extension.</td>
</tr>
<tr>
<td>call-waiting (voice register pool)</td>
<td>Enables call waiting on a SIP phone.</td>
</tr>
<tr>
<td>number (voice register dn)</td>
<td>Associates an extension number with a voice register dn.</td>
</tr>
<tr>
<td>voice register dn</td>
<td>Enters voice register dn configuration mode to define an extension for a SIP phone line.</td>
</tr>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode for SIP phones.</td>
</tr>
</tbody>
</table>
call-forward b2bua night-service

To automatically forward calls to another number during night-service hours, use the call-forward b2bua night-service command in voice register dn configuration mode. To remove the code, use the no form of this command.

```plaintext
call-forward b2bua night-service target-number
no call-forward b2bua night-service
```

**Syntax Description**
- `target-number` Phone number to which calls are forwarded.

**Command Default**
Calls are not forwarded during night-service hours.

**Command Modes**
Voice register dn configuration: (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.6(3)M</td>
<td>Cisco Unified CME 11.5</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>16.3.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**
You need to configure the night-service bell command under voice register dn. Night-service hours are defined using the night-service date and night-service day commands.

A voice register dn can have all other types of call forwarding defined at the same time: all-calls, no-answer, busy, and night-service. Each type of call forwarding can have a different forwarding destination defined in its target-number argument. If more than one type of call forwarding is in effect (is active) at one time, the precedence order for evaluating the different types is as follows:

- call forward night-service (only during night service hours)
- call forward all
- call forward busy and call forward no answer

**Examples**
The following example defines a call forward night-service configuration under voice register dn:

```plaintext
Router(config)# voice register dn tag
Router(config-register-dn)# call-forward b2bua night-service
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>night-service date</td>
</tr>
<tr>
<td>night-service day</td>
</tr>
</tbody>
</table>
call-forward b2bua noan

To enable call forwarding for a Session Initiation Protocol (SIP) back-to-back user agent (B2BUA) so that incoming calls to an extension that does not answer after a configured amount of time are forwarded to another extension, use the call-forward b2bua noan command in voice register dn or voice register pool configuration mode. To disable call forwarding, use the no form of this command.

call-forward b2bua noan directory-number timeout seconds  
no call-forward b2bua noan

Syntax Description

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>directory-number</td>
<td>Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.</td>
</tr>
<tr>
<td>timeout seconds</td>
<td>Number of seconds that a call can ring with no answer before the call is forwarded to another extension. Range is 3 to 60000. Default is 20.</td>
</tr>
</tbody>
</table>

Command Default

Feature is disabled.

Command Modes

Voice register dn configuration (config-register-dn)  
Voice register pool configuration (config-register-pool)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was removed from voice register pool configuration mode for Cisco Unified CME only.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command with modifications was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

This command in voice register dn configuration mode applies the call forward mechanism to an individual SIP extension in Cisco Unified CME or Cisco Unified SIP SRST that remains unanswered after a specified length of time. This command in voice register pool configuration mode is for Cisco Unified SIP SRST only and applies to SIP IP phones on which the extension appears.

If this command is configured in both the voice register dn and voice register pool configuration modes, the configuration under voice register dn takes precedence.

We recommend that you do not use this command to configure a SIP extension or SIP IP phone that is a member of a hunt group. If this command is configured for a member of a hunt group, remove the phone from any hunt group to which it is assigned to avoid forwarding calls to all phones in the hunt group.

The call-forward b2bua all command takes precedence over the call-forward b2bua busy and call-forward b2bua noan commands.

Note

This command in voice register dn configuration mode is not commonly used for Cisco Unified SIP SRST.
Cisco Unified CME and Cisco Unified SIP SRST

The following example shows how to forward calls to extension 5005 when extension 5001 is unanswered. The timeout before the call is forwarded to extension 5005 is 10 seconds.

Router(config)# voice register pool 4
Router(config-register-pool)# number 5001
Router(config-register-pool)# call-forward b2bua noan 5005 timeout 10

Cisco Unified SIP SRST

The following example shows how to forward calls to extension 5005 when extension 5001 on pool number 4 is unanswered. The timeout before the call is forwarded to extension 5005 is 10 seconds.

Router(config)# voice register pool 4
Router(config-register-pool)# number 5001
Router(config-register-pool)# call-forward b2bua noan 5005 timeout 10

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-forward b2bua all</td>
<td>Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.</td>
</tr>
<tr>
<td>call-forward b2bua busy</td>
<td>Enables call forwarding for a SIP B2BUA so that incoming calls to a busy extension are forwarded to another extension.</td>
</tr>
<tr>
<td>call-forward b2bua mailbox</td>
<td>Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.</td>
</tr>
<tr>
<td>call-waiting (voice register pool)</td>
<td>Enables call waiting on a SIP phone.</td>
</tr>
</tbody>
</table>
call-forward b2bua unreachable

Note
Effective with Cisco IOS Release 12.4(11)XJ, the call-forward b2bua unreachable command is not available in Cisco IOS software.

To forward calls to a phone that is not registered to Cisco Unified CME, use the call-forward b2bua unreachable command in voice register dn or voice register pool configuration mode. To disable call forwarding, use the no form of this command.

**Syntax Description**

```
directory-number
```
Telephone number to which calls are forwarded. Represents a fully qualified E.164 number.

**Command Default**
Feature is disabled

**Command Modes**
Voice register dn configuration (config-register-dn)
Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was removed.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was removed in Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Call forward unreachable is triggered when a call arrives for a destination that is configured but unregistered with Cisco CME. A destination is considered to be configured if its number is listed under the voice register pool or voice register dn configurations.

If call forward unreachable is not configured for a pool or directory number (DN) register, any calls that match the numbers in that pool or DN register will use call forward busy instead.

We recommend that you do not use this command with hunt groups. If the command is used, consider removing the phone from any assigned hunt groups, unless you want to forward calls to all phones in the hunt group.

**Examples**
The following example shows how to forward calls to extension 5005 when extension 5001 on directory number 4 is unreachable, either because it is unplugged or the network between the Cisco router and the extension is nonfunctional. The timeout before the call is forwarded to extension 5005 is 10 seconds.

```
Router(config)# voice register pool 4
Router(config-register-dn)# number 5001
Router(config-register-dn)# call-forward b2bua unreachable 5005 timeout 10
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-forward b2bua all (voice register dn and voice register pool)</td>
<td>Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.</td>
</tr>
<tr>
<td>call-forward b2bua busy (voice register dn and voice register pool)</td>
<td>Enables call forwarding for a SIP B2BUA so that incoming calls to a busy extension are forwarded to another extension.</td>
</tr>
<tr>
<td>call-forward b2bua mailbox (voice register dn and voice register pool)</td>
<td>Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.</td>
</tr>
<tr>
<td>call-forward b2bua noan (voice register dn and voice register pool)</td>
<td>Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that does not answer after a configured amount of time are forwarded to another extension.</td>
</tr>
<tr>
<td>call-waiting (voice register pool)</td>
<td>Enables call waiting on a SIP phone.</td>
</tr>
<tr>
<td>number (voice register dn)</td>
<td>Associates an extension number with a voice register dn.</td>
</tr>
</tbody>
</table>
call-forward busy

To configure call forwarding so that incoming calls to a busy extension (ephone-dn) are forwarded to another extension, use the **call-forward busy** command in ephone-dn or ephone-dn-template configuration mode. To disable call forwarding, use the **no** form of this command.

```
call-forward busy  target-number  [{primary|secondary}]  [dialplan-pattern]
no  call-forward busy
```

### Syntax Description

<table>
<thead>
<tr>
<th>Argument</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>target-number</td>
<td>Phone number to which calls are forwarded.</td>
</tr>
<tr>
<td>primary</td>
<td>(Optional) Call forwarding is selectively applied only to the dial peer created for the primary number for this ephone-dn.</td>
</tr>
<tr>
<td>secondary</td>
<td>(Optional) Call forwarding is selectively applied only to the dial peer created for the secondary number for this ephone-dn.</td>
</tr>
<tr>
<td>dialplan-pattern</td>
<td>(Optional) Call forwarding is selectively applied only to dial peers created for this ephone-dn by the dial-plan pattern.</td>
</tr>
</tbody>
</table>

### Command Default

Call forwarding for a busy extension is not enabled.

### Command Modes

- Ephone-dn configuration (config-dn-ephone)
- Ephone-dn-template configuration (config-ephone-dn-template)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The <strong>primary</strong>, <strong>secondary</strong>, and <strong>dialplan-pattern</strong> keywords were added, and this command was made available in ephone-dn-template configuration mode.</td>
</tr>
<tr>
<td>12.4(11)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command with the <strong>primary</strong>, <strong>secondary</strong>, and <strong>dialplan-pattern</strong> keywords added, and this command in ephone-dn-template configuration mode was integrated into Cisco IOS 12.4(11)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

The call forwarding mechanism is applied to an individual extension (ephone-dn) and is not applied to the phone on which the extension appears.

Normally, call forwarding is applied to all dial peers that are created by the ephone-dn. An ephone-dn can create up to four dial peers:

- A dial peer for the primary number
- A dial peer for the secondary number
- A dial peer for the primary number as expanded by the **dialplan-pattern** command
- A dial peer for the secondary number as expanded by the **dialplan-pattern** command
The **primary**, **secondary**, and **dialplan-pattern** keywords allow you to apply call forwarding selectively to one or more dial peers based on the exact called number that was used to route the call to the ephone-dn. If none of the optional keywords is used, call forwarding applies to all dial-peers that are defined for the ephone-dn.

An ephone-dn can have all four types of call forwarding defined at the same time: all-calls, no-answer, busy, and night-service. Each type of call forwarding can have a different forwarding target defined in its `target-number` argument. If more than one type of call forwarding is in effect (is active) at one time, the precedence order for evaluating the different types is as follows:

1. call forward night service
2. call forward all
3. call forward busy and call forward no answer

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

### Examples

The following example forwards all calls for the ephone-dn 2345 when it is busy.

```
Router(config)# ephone-dn 236
Router(config-ephone-dn)# number 2345
Router(config-ephone-dn)# call-forward
    busy 2000
```

The following example uses an ephone-dn template to forward calls for extension 2345 when it is busy.

```
Router(config)# ephone-dn-template 6
Router(config-ephone-dn-template)# call-forward
    busy 2000
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 236
Router(config-ephone-dn)# number 2345
Router(config-ephone-dn)# ephone-dn-template 6
```

The following example creates a dial-plan pattern to expand extension numbers into E.164 numbers. It then sets call forwarding of incoming calls to directory number 5005. In this example, call forwarding on busy is applied only when callers dial the primary number for this ephone-dn, 5001.

```
Router(config)# telephony-service
Router(config-telephony)# dialplan-pattern 1 40855501.. extension-length 4 extension-pattern 50..
Router(config-telephony)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001 secondary 5002
Router(config-ephone-dn)# call-forward
    busy 5005 primary
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>call-forward all</code></td>
<td>Configures call forwarding for all incoming calls to an ephone-dn.</td>
</tr>
</tbody>
</table>
## Command Reference

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-forward night-service</td>
<td>Configures call forwarding for all incoming calls to an ephone-dn during the hours defined for night service.</td>
</tr>
<tr>
<td>callforward noan</td>
<td>Configures call forwarding to another number when no answer is received from an ephone-dn.</td>
</tr>
<tr>
<td>ephone-dn-template (ephone-dn)</td>
<td>Applies template to ephone-dn.</td>
</tr>
</tbody>
</table>
call-forward max-length

To restrict the number of digits that can be entered using the CfwdALL soft key on an IP phone, use the **call-forward max-length command** in ephone-dn or ephone-dn-template configuration mode. To remove a restriction on the number of digits that can be entered, use the **no** form of this command.

```
call-forward max-length length
no call-forward max-length
```

**Syntax Description**
- `length`: Number of digits that can be entered using the CfwdAll soft key on an IP phone.

**Command Default**
There is no restriction on the number of digits that can be entered.

**Command Modes**
- Ephone-dn configuration (config-dn-ephone)
- Ephone-dn-template configuration (config-ephone-dn-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(7)T</td>
<td>Cisco CME 3.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-dn-template configuration mode.</td>
</tr>
<tr>
<td>12.4(11)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(11)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command can be used to prevent a phone user from using the CfwdALL soft key on an IP phone to forward calls to numbers that will incur toll charges when they receive forwarded calls.

If the `length` argument is set to 0, the CfwdALL soft key is completely disabled. If the ephone-dn associated with the first line button has an active call forward number when this command is used to set the `length` argument to 0, the CfwdALL soft key will be disabled after the next phone restart.

The restriction created by this command does not apply to destinations that are entered using the Cisco IOS command-line interface (CLI) or the Cisco Unified CME GUI.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

**Examples**
The following example restricts the number of digits that a phone user can enter using the CfwdALL soft key to four. In this example, extensions in the phone user’s Cisco Unified CME system have four digits, so that means the user can use the IP phone to forward all calls to any extension in the system, but not to any number outside the system.

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# call-forward max-length 4
```
The following example uses an ephone-dn-template to restrict the number of digits that a phone user can enter using the CfwdALL soft key to four.

```
Router(config)# ephone-dn-template 4
Router(config-ephone-dn-template)# call-forward max-length 4
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# ephone-dn-template 4
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>call-forward all</strong></td>
<td>Configures call forwarding for all incoming calls on one of the lines of a Cisco Unified IP phone.</td>
</tr>
<tr>
<td><strong>ephone-dn-template (ephone-dn)</strong></td>
<td>Applies an ephone-dn template to an ephone-dn.</td>
</tr>
</tbody>
</table>
call-forward night-service

To automatically forward calls to another number during night-service hours, use the call-forward night-service command in ephone-dn or ephone-dn-template configuration mode. To disable automatic call forwarding during night service, use the no form of this command.

```
call-forward night-service target-number
no call-forward night-service
```

Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>target-number</td>
<td>Phone number to which calls are forwarded.</td>
</tr>
</tbody>
</table>

Command Default

Calls are not forwarded during night-service hours.

Command Modes

Ephone-dn configuration (config-dn-ephone)  
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(11)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

You must also configure the night-service bell command for this ephone-dn.

Night-service hours are defined using the night-service date and night-service day commands.

An ephone-dn can have all four types of call forwarding defined at the same time: all-calls, no-answer, busy, and night-service. Each type of call forwarding can have a different forwarding destination defined in its target-number argument. If more than one type of call forwarding is in effect (is active) at one time, the precedence order for evaluating the different types is as follows:

1. call forward night-service
2. call forward all
3. call forward busy and call forward no answer

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example establishes night-service hours from 1 p.m. Saturday until 8 a.m. Monday. During that time, calls to extension 1000 (ephone-dn 1) are forwarded to extension 2346. Note that the night-service bell command has also been used for ephone-dn 1.

```
telephony-service
night-service day sat 13:00 12:00
night-service day sun 12:00 08:00
night-service code *1234
!
ephone-dn 1
number 1000
```
```plaintext
call-forward night-service

ephone-dn 2
number 2346
ephone 12
button 1:1
ephone 13
button 1:2

The following example uses an ephone-dn template to apply call forwarding for extension 2876 during the night service hours established in the previous example.
ephone-dn-template 2
call-forward night-service 2346
ephone-dn 25
number 2876
ephone-dn-template 2
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>callforward all</td>
<td>Configures call forwarding for all incoming calls to an ephone-dn.</td>
</tr>
<tr>
<td>callforward busy</td>
<td>Configures call forwarding to another number when an ephone-dn is busy.</td>
</tr>
<tr>
<td>callforward noan</td>
<td>Configures call forwarding to another number when no answer is received from an ephone-dn.</td>
</tr>
<tr>
<td>night-service bell (ephone-dn)</td>
<td>Marks an ephone-dn for night-service treatment.</td>
</tr>
<tr>
<td>night-service date</td>
<td>Defines a recurring time period associated with a month and day during which night service is active.</td>
</tr>
<tr>
<td>night-service day</td>
<td>Defines a recurring time period associated with a day of the week during which night service is active.</td>
</tr>
</tbody>
</table>
call-forward noan

To configure call forwarding so that incoming calls to an extension (ephone-dn) that does not answer are forwarded to another number, use the `callforward noan` command in ephone-dn or ephone-dn-template configuration mode. To disable call forwarding, use the `no` form of this command.

```
call-forward noan target-number timeout seconds [{primary|secondary}] [dialplan-pattern]
no call-forward noan
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>target-number</code></td>
<td>Phone number to which calls are forwarded.</td>
</tr>
<tr>
<td><code>timeout seconds</code></td>
<td>Sets the duration that a call can ring with no answer before the call is forwarded to the target number. Range is from 3 to 60000. There is no default value.</td>
</tr>
<tr>
<td><code>primary</code></td>
<td>(Optional) Call forwarding is selectively applied only to the dial peer created for the primary number for this ephone-dn.</td>
</tr>
<tr>
<td><code>secondary</code></td>
<td>(Optional) Call forwarding is selectively applied only to the dial peer created for the secondary number for this ephone-dn.</td>
</tr>
<tr>
<td><code>dialplan-pattern</code></td>
<td>(Optional) Call forwarding is selectively applied only to dial peers created for this ephone-dn by the dial-plan pattern.</td>
</tr>
</tbody>
</table>

**Command Default**

Call forwarding for an extension that does not answer is not enabled.

**Command Modes**

Ephone-dn configuration (config-dn-ephone)
Ephone-dn-template configuration (config-ephone-dn-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The primary, secondary, and dialplan-pattern keywords were added, and this command was made available in ephone-dn-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command with modifications was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The call forwarding mechanism is applied to an individual extension (ephone-dn) and is not applied to the phone on which the extension appears.

Normally, call forwarding is applied to all dial peers that are created by the ephone-dn. An ephone-dn can create up to four dial peers:

- A dial peer for the primary number
- A dial peer for the secondary number
- A dial peer for the primary number as expanded by the `dialplan-pattern` command
A dial peer for the secondary number as expanded by the **dialplan-pattern** command

The **primary**, **secondary**, and **dialplan-pattern** keywords allow you to apply call forwarding selectively to one or more dial peers based on the exact called number that was used to route the call to the ephone-dn. If none of the optional keywords is used, call forwarding applies to all dial-peers that are defined for the ephone-dn.

An ephone-dn can have all four types of call forwarding defined at the same time: all-calls, no-answer, busy, and night-service. Each type of call forwarding can have a different forwarding target defined in its **target-number** argument. If more than one type of call forwarding is in effect (is active) at one time, the precedence order for evaluating the different types is as follows:

1. call forward night service
2. call forward all
3. call forward busy and call forward no answer

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

### Examples

The following example forwards calls for the ephone-dn 2345 when it does not answer.

```bash
Router(config) # ephone-dn 236
Router(config-ephone-dn) # number 2345
Router(config-ephone-dn) # call-forward busy 2000
```

The following example uses an ephone-dn-template to forward calls for the ephone-dn 2345 when it does not answer.

```bash
Router(config) # ephone-dn-template 8
Router(config-ephone-dn-template) # call-forward busy 2000
Router(config-ephone-dn-template) # exit
Router(config) # ephone-dn 236
Router(config-ephone-dn) # number 2345
Router(config-ephone-dn) # ephone-dn-template 8
```

The following example creates a dial-plan pattern to expand extension numbers into E.164 numbers. It then sets call forwarding of incoming calls to directory number 5005. In this example, call forwarding on no answer is applied only when callers dial the primary number for this ephone-dn, 5001.

```bash
Router(config) # telephony-service
Router(config-telephony) # dialplan-pattern 1 40855501.. extension-length 4 extension-pattern 50..
Router(config-telephony) # exit
Router(config) # ephone-dn 1
Router(config-ephone-dn) # number 5001 secondary 5002
Router(config-ephone-dn) # call-forward noan 5005 primary
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-forward all</td>
<td>Configures call forwarding for all incoming calls for an ephone-dn.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>callforward busy</td>
<td>Configures call forwarding to another number when an ephone-dn is busy.</td>
</tr>
<tr>
<td>call-forward night-service</td>
<td>Configures call forwarding for all incoming calls to an ephone-dn during the hours defined for night service.</td>
</tr>
<tr>
<td>ephone-dn-template (ephone-dn)</td>
<td>Applies an ephone-dn-template to an ephone-dn.</td>
</tr>
</tbody>
</table>
**call-forward pattern**

To specify a pattern for calling-party numbers that are able to support the ITU-T H.450.3 standard for call forwarding, use the `call-forward pattern` command in telephony-service configuration mode. To remove the pattern, use the `no` form of this command.

```
call-forward pattern pattern
no call-forward pattern pattern
```

### Syntax Description

| `pattern` | String that consists of one or more digits and wildcard markers or dots (.) to define a specific pattern. Calling parties that match a defined pattern use the H.450.3 standard if they are forwarded. A pattern of `.T` specifies the H.450.3 forwarding standard for all incoming calls. |

### Command Default

No call-forward pattern is defined.

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)YT</td>
<td>Cisco CME 2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco CME 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use this command with Cisco IOS Telephony Services (ITS) V2.1, Cisco CallManager Express 3.0, or a later version.

When H.450.3 call forwarding is selected, the router must be configured with a Tool Command Language (Tcl) script that supports the H.450.3 protocol. The Tcl script is loaded on the router by using the `call application voice` command.

The pattern match in this command is against the phone number of the calling party. When an extension number has forwarded its calls and an incoming call is received for that number, the router sends an H.450.3 response back to the original calling party to request that the call be placed again using the forward-to destination.

Calling numbers that do not match the patterns defined using this command are forwarded using Cisco-proprietary call forwarding for backward compatibility.

### Examples

The following example specifies that all 4-digit directory numbers that begin with 4 should use the H.450.3 standard whenever they are forwarded:

```
Router(config)# telephony-service
Router(config-telephony)# call-forward pattern 4...
```

The following example forwards all calls that support the H.450.3 standard:

```
Router(config)# telephony-service
Router(config-telephony)# call-forward pattern .T
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>call application voice</strong></td>
<td>Defines an application, indicates the location of the corresponding Tcl files that implement the application, and loads the selected Tcl script.</td>
</tr>
</tbody>
</table>
calling-number local

To replace a calling-party number and name with the forwarding-party number and name (the local number and name) in calls forwarded using local hairpin call routing, use the calling-number local command in telephony-service configuration mode. To reset to the default, use the no form of this command.

```
calling-number local [secondary]
no calling-number local
```

**Syntax Description**

| secondary | (Optional) Uses the secondary number associated with the forwarding party instead of the primary number. The primary number is the default if this keyword is not used. |

**Command Default**

Calling-party numbers and names are used in forwarded calls.

**Command Modes**

Telephony-service configuration

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ3</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.3(15)ZJ4</td>
<td>Cisco CME 3.0</td>
<td>The secondary keyword was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>Support was added to the default IOS voice application framework and dependency on the TCL script was removed.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

In Cisco CME 3.2 and earlier versions, this command is used with the Tool Command Language (Tcl) script app-h450-transfer.2.0.0.7 or a later version.

In Cisco CME 3.3 and later versions, this command can be used without the TCL script because the functionality is integrated into the default IOS voice application framework.

If the ephone-dn used by a forwarding party has a secondary number in addition to its primary number and neither number is registered with the gatekeeper, the primary number is the number that appears as the calling number on hairpin-forwarded calls when the calling-number local command is used. If only one of the numbers is registered with the gatekeeper, the registered number is the number that appears as the calling number. If both numbers are registered with the gatekeeper, the primary number is the number that appears as the calling number.

If the ephone-dn used by a forwarding party has a secondary number in addition to its primary number and the calling-number local secondary command is used, the secondary number is the number that appears as the calling number on hairpin-forwarded calls if both numbers are registered with the gatekeeper or if both numbers are not registered. If only one number is configured to register with the gatekeeper, the number that is registered appears as the calling number.

**Examples**

The following example specifies use of the name and number of the local forwarding party in hairpin-forwarded calls:

```
Router(config)# telephony-service
```
Router(config-telephony)# calling-number local

The following examples demonstrate the use of the **calling-number local** command without the **secondary** keyword.

- The calling number for hairpin calls forwarded from ephone-dn 1 is 1234 in the following example:

```
calling-number local
  ephone-dn 1
  number 1234 secondary 4321 no-reg
```

- The calling number for hairpin calls forwarded from ephone-dn 1 is 4321 in the following example:

```
calling-number local
  ephone-dn 1
  number 1234 secondary 4321 no-reg primary
```

- The calling number for hairpin calls forwarded from ephone-dn 1 is 1234 in the following example:

```
calling-number local
  ephone-dn 1
  number 1234 secondary 4321 no-reg both
```

or

```
number 1234 secondary 4321
```

The following examples demonstrate the use of the **calling-number local secondary** command.

- The calling number for hairpin calls forwarded from ephone-dn 1 is 1234 in the following example:

```
calling-number local secondary
  ephone-dn 1
  number 1234 secondary 4321 no-reg
```

- The calling number for hairpin calls forwarded from ephone-dn 1 is 4321 in the following example:

```
calling-number local secondary
  ephone-dn 1
  number 1234 secondary 4321 no-reg primary
```

- The calling number for hairpin calls forwarded from ephone-dn 1 is 4321 in the following example:

```
calling-number local secondary
  ephone-dn 1
  number 1234 secondary 4321 no-reg both
```

or

```
number 1234 secondary 4321
```
calling-number local (voice register global)

To replace a calling-party number and name with the forwarding-party number and name (the local number and name) in calls forwarded using local hairpin call routing, use the `calling-number local` command in voice register global configuration mode. To reset to the default, use the `no` form of this command.

```plaintext
calling-number local
no calling-number local
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Calling-party numbers and names are used in forwarded calls. The command is disabled by default.

**Command Modes**
Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IOS XE Everest 16.6.1</td>
<td>Unified CME 12.0</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use the CLI Command `calling-number local` in voice register global configuration mode so that the number and name of the forwarding party appears as the calling number on hairpin-forwarded calls. Once `calling-number local` is configured under voice register global, the calls forwarded from local SIP phones will have the calling-number and name of the last forwarded party.

**Examples**
The following example specifies use of the name and number of the local forwarding party in hairpin-forwarded calls:

```plaintext
Router(config)# voice register global
Router(config-register-global)# calling-number local
```

The following examples demonstrate the use of the `calling-number local` command.

- The calling number for hairpin calls forwarded from voice register dn 1 is 1234 in the following example:

```plaintext
voice register global
calling-number local
...
voice register dn 1
name Phone 1
number 1234
```
callqueue-display

To configure call waiting notification display on the agent phone as continuous, periodic, or off, use the `callqueue display` command. To set the call waiting notification display to the default state of periodic (for voice hunt group) and continuous (for ephone hunt group), use the `default` form of this command.

```
callqueue display [{continuous | periodic | off}]
default callqueue display
```

**Command Default**

Call waiting notification is set to periodic for phones in voice hunt group, and to continuous for phones in ephone hunt group. The no form of this command also sets the call waiting display to default state.

**Command Modes**

- Ephone-hunt configuration (config-ephone-hunt)
- Voice hunt group configuration (config-voice-hunt-group)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.6(3)M</td>
<td>Cisco Unified CME 11.5</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>16.3.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The callqueue display command is valid for both voice hunt group as well as ephone hunt group.

**Examples**

The following example shows how to set call waiting notification display to periodic in an ephone hunt group:

```
Router(config)# ephone-hunt 1
Router(config-ephone-hunt)# call
Router(config-ephone-hunt)# callqueue
Router(config-ephone-hunt)# callqueue display
Router(config-ephone-hunt)# callqueue display periodic
```

The following example shows how to set call waiting notification display to continuous in a voice hunt group:

```
Router(config)# voice hunt-group 1
Router(config-voice-hunt-group)# callqueue display
Router(config-voice-hunt-group)# callqueue display continuous
```
call-park system

To define system parameters for the Call Park feature, use the `call-park system` command in telephony-service configuration mode. To reset to the default, use the `no` form of this command.

```
call-park system {application|redirect}
no call-park system {application|redirect}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>application</code></td>
<td>Enables Call Park and Directed Call Park for SCCP and SIP phones.</td>
</tr>
<tr>
<td><code>redirect</code></td>
<td>H.323 and SIP calls use H.450 or the SIP Refer method of call forwarding or transfer to park calls and to pick up calls from park.</td>
</tr>
</tbody>
</table>

**Command Default**

H.323 and SIP calls use hairpin call forwarding or transfer to park calls and to pick up calls from park.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>The <code>application</code> keyword and support for SIP phones was added.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command has been integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `application` keyword selects the enhanced Call Park method supported in Cisco Unified CME 7.1 and later versions for SCCP and SIP phones.

**Examples**

The following example specifies that H.323 and SIP calls will use the H.450 or SIP Refer method of call forwarding or transfer to park calls and pick up calls from park:

```
Router(config)# telephony-service
Router(config-telephony)# call-park system redirect
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>park reservation-group</code></td>
<td>Assigns a call-park reservation group to a phone.</td>
</tr>
<tr>
<td><code>park-slot</code></td>
<td>Creates a floating extension at which calls can be temporarily parked.</td>
</tr>
</tbody>
</table>
call-waiting (voice register pool)

To enable call-waiting option on a SIP phone, use the `call-waiting` command in voice register pool configuration mode. To disable call waiting, use the `no` form of this command.

```
call-waiting
no call-waiting
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Feature is enabled.

**Command Modes**

Voice register pool configuration (call-waiting)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The call waiting feature is enabled by default on SIP phones. To disable call waiting, use the `no call-waiting` command.

**Examples**

The following example shows how to disable call waiting on SIP phone 1:

```
Router(config)# voice register pool 1
Router(config-register-pool)# no call-waiting
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode for SIP phones.</td>
</tr>
</tbody>
</table>
call-waiting beep

To allow call-waiting beeps to be accepted by or generated from an ephone-dn, use the **call-waiting beep** command in ephone-dn or ephone-dn-template configuration mode. To disable the acceptance and generation of call-waiting beeps by an ephone-dn, use the **no** form of this command.

```plaintext
no call-waiting beep [{accept|generate}]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>accept</th>
<th>(Optional) Allows call-waiting beeps to be accepted by an ephone-dn.</th>
</tr>
</thead>
<tbody>
<tr>
<td>generate</td>
<td>(Optional) Allows call-waiting beeps to be generated by an ephone-dn.</td>
</tr>
</tbody>
</table>

**Command Default**

Call-waiting beeps are accepted and generated.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco CME 3.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-dn-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The **call-waiting beep** command must be used with the **ephone-dn** command. The **call-waiting beep** command is used like a toggle and can be switched on and off for each ephone-dn.

A beep can be heard only if both sending and receiving ephone-dns are configured to accept call-waiting beeps.

To display how call-waiting beeps are configured, use the **show running-config** command in the privileged EXEC configuration mode. If the **no call-waiting beep generate** and **no call-waiting beep accept** commands are configured, the **show running-config** output will display the **no call-waiting beep** command.

If you configure a button to have a silent ring using the **s** option of the **button** command, you will not hear a call-waiting beep regardless of whether the ephone-dn associated with the button is configured to generate a call-waiting beep.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

**Examples**

The following example configures ephone-dn 1 and ephone-dn 2 not to accept and not to generate call-waiting beeps:

```plaintext
Router(config)# ephone-dn 1
```
Router(config-ephone-dn)# number 2588
Router(config-ephone-dn)# no call-waiting beep accept
Router(config-ephone-dn)# no call-waiting beep generate
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 2589
Router(config-ephone-dn)# no call-waiting beep accept
Router(config-ephone-dn)# no call-waiting beep generate
Router(config-ephone-dn)# exit

The following example uses an ephone-dn template to set the same attributes as in the previous example:

Router(config)# ephone-dn-template 5
Router(config-ephone-dn-template)# no call-waiting beep accept
Router(config-ephone-dn-template)# no call-waiting beep generate
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 2588
Router(config-ephone-dn)# ephone-dn-template 5
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 2589
Router(config-ephone-dn)# ephone-dn-template 5
Router(config-ephone-dn)# exit

The following example configures ephone-dn 1 and ephone-dn 2 to switch back to accept call-waiting beeps. Ephone-dn 1 and ephone-dn 2 now accept but do not generate call-waiting beeps.

Router(config)# ephone-dn 1
Router(config-ephone-dn)# call-waiting beep accept
Router(config)# ephone-dn 2
Router(config-ephone-dn)# call-waiting beep accept

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>show running-config</code></td>
<td>Displays the contents of the currently running configuration file or the configuration for a specific interface, or map class information.</td>
</tr>
<tr>
<td><code>ephone-dn-template (ephone-dn)</code></td>
<td>Applies a template to an ephone-dn.</td>
</tr>
</tbody>
</table>
call-waiting ring

To allow an ephone-dn to use a ring sound for call-waiting notification, use the call-waiting ring command in ephone-dn or ephone-dn-template configuration mode. To disable the ring notification, use the no form of this command.

```
call-waiting ring
no call-waiting ring
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

The ephone-dn accepts call waiting and uses beeps for notification.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-dn-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

To use a ring sound for call-waiting notification on an ephone-dn, you must ensure that the ephone-dn will accept secondary calls while it is connected to another line. The acceptance of call waiting is the default ephone-dn behavior. However, the no call-waiting beep accept command can change this default so an ephone-dn does not accept call waiting. This command must be removed for ringing notification to work.

The call-waiting ring command will automatically disable a call-waiting beep configuration.

If you configure a button to have a silent ring using the s option of the button command, you will not hear a call-waiting ring regardless of whether the ephone-dn associated with the button is configured to generate a call-waiting ring.

**Note**

The call-waiting ring option cannot be used on the Cisco Unified IP Phone 7902, Cisco Unified IP Phone 7905, or Cisco Unified IP Phone 7912. Do not use the call-waiting ring command for ephone-dns associated with these types of phones.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.
The following example configures ephone-dn 1 and ephone-dn 2 to use ringing for their call-waiting notification:

Router(config)# ephone-dn 1
Router(config-ephone-dn)# call-waiting ring
Router(config)# ephone-dn 2
Router(config-ephone-dn)# no call-waiting ring

The following example uses an ephone-dn template to set the same attributes as in the previous example:

Router(config)# ephone-dn-template 9
Router(config-ephone-dn-template)# call-waiting ring
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn-template 10
Router(config-ephone-dn-template)# no call-waiting ring
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# ephone-dn-template 9
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 2
Router(config-ephone-dn)# ephone-dn-template 10
Router(config-ephone-dn)# exit

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-waiting beep</td>
<td>Allows call-waiting beeps to be accepted by or generated from an ephone-dn.</td>
</tr>
<tr>
<td>ephone-dn-template (ephone-dn)</td>
<td>Applies template to ephone-dn.</td>
</tr>
</tbody>
</table>
camera

To enable USB camera capability on Cisco Unified IP Phones 9951 and 9971, use the `camera` command in voice register global, voice register template, and voice register pool configuration modes. To disable video capabilities on Cisco Unified IP Phones 9951 and 9971, use the `no` form of this command.

```
camera
no
camera
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
USB camera capability is disabled on Cisco Unified IP Phones 9951 and 9971.

**Command Modes**
Voice register global
Voice register template
Voice register pool

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(4)M</td>
<td>Cisco Unified CME 8.6</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to enable USB camera capability on Cisco Unified IP Phones 9951 and 9971. You need to create profile and apply-config or restart to the phone to enable the video capability on phones.

**Examples**
The following example shows camera command configured in voice register global:

```
Router#show run
!
!
voice service voip
  allow-connections sip to sip
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  !
  !
voice register global
  mode cme
  bandwidth video tias-modifier 244 negotiate end-to-end
  max-pool 10
  camera
  voice register template 10
  !
  !
```

The following example shows video and camera commands configured under voice register pool 5, you can also configure both camera and video commands under voice register template:

```
Router#show run
!
!
voice service voip
  allow-connections sip to sip
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  !
  !
```
voice register global
   mode cme
   bandwidth video tias-modifier 244 negotiate end-to-end
   max-pool 10
   voice register pool 1
      id mac 1111.1111.1111
   voice register pool 4
   voice register pool 5
      logout-profile 58
      id mac 0009.A3D4.1234
      camera

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>apply-config</td>
<td>Allows to dynamically apply the phone configuration on Cisco Unified SIP IP phones 8961, 9951, and 9971,</td>
</tr>
</tbody>
</table>
capf-auth-str

To define a string of digits that a user enters at the phone for CAPF authentication, use the `capf-auth-str` command in ephone configuration mode. To return to the default, use the `no` form of this command.

```
capf-auth-str  digit-string
no  capf-auth-str
```

**Syntax Description**

| digit-string | String of digits that a phone user enters at the phone for CAPF authentication. |

**Command Default**

No authentication string exists for the phone.

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication to create or remove an authentication string (Personal Identification Number or PIN) for the specified secure ephone. Use this command if the `auth-string` keyword is specified in the `auth-mode` command. Once you specify a CAPF authentication string, it becomes part of the ephone configuration. This value can also be set globally or per ephone using the `auth-string` command in CAPF configuration mode.

Use the `show capf-server auth-str` command to display configured authentication strings.

When a phone is configured for a certificate upgrade that requires auth-string authentication, the CAPF initiation needs to be performed manually by the phone user using the following steps:

1. Press the Settings button.
2. If the configuration is locked, press **## (asterisk, asterisk, pound sign) to unlock it.
3. Scroll down the menu and select Security Configuration.
4. Scroll down the next menu to LSC and press the Update soft key.
5. When prompted for the authentication string, enter the string provided by the system administrator.

**Examples**

The following example specifies the type of authentication for ephone 392 is an authentication string that is entered from the phone, and then defines the string as 38593.

```
ephone 392
button 1:23 2:24 3:25
device-security-mode authenticated
cert-oper upgrade auth-mode auth-string
capf-auto-str 38593
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>auth-mode</td>
<td>Specifies the type of authentication to use during CAPF sessions.</td>
</tr>
<tr>
<td>auth-string</td>
<td>Generates or removes authentication strings for one or all secure ephones.</td>
</tr>
<tr>
<td>show capf-server</td>
<td>Displays configuration and session information for the CAPF server.</td>
</tr>
</tbody>
</table>
capf-server

To enter CAPF-server configuration mode to set CAPF server parameters, use the `capf-server` command in global configuration mode. To remove the CAPF server configuration, use the `no` form of this command.

```
capf-server
no  capf-server
```

**Syntax Description**
This command has no keywords or arguments.

**Command Default**
No CAPF server configuration is present.

**Command Modes**
Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command is used with Cisco Unified CME phone authentication.

**Examples**
The following example sets parameters for the CAPF server:

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```
cert-enroll-trustpoint

To enroll the CAPF with the CA or RA, use the `cert-enroll-trustpoint` command in CAPF-server configuration mode. To remove an enrollment, use the `no` form of this command.

```
cert-enroll-trustpoint ca-label password {0|1} password-string
no cert-enroll-trustpoint
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ca-label</code></td>
<td>PKI trustpoint label for the CA or for the RA if an RA is being used.</td>
</tr>
<tr>
<td><code>password</code></td>
<td>Values that follow apply to the password.</td>
</tr>
<tr>
<td>`0</td>
<td>1`</td>
</tr>
<tr>
<td></td>
<td>- 0 — Encrypted.</td>
</tr>
<tr>
<td></td>
<td>- 1 — Clear text.</td>
</tr>
<tr>
<td><code>password-string</code></td>
<td>Alphanumeric challenge password that is required for certificate enrollment.</td>
</tr>
</tbody>
</table>

**Command Default**

The CAPF server is not enrolled with the CA or RA.

**Command Modes**

CAPF-server configuration (config-capf-server)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication.

**Examples**

The following example specifies that the CAPF server should enroll with the trustpoint named server12 (the CA) using the password x8oWiet, which should be encrypted:

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```
clear cti session

To tear down the connection between a CSTA client application and Cisco Unified CME, use the **clear cti session** command in privileged EXEC configuration mode.

```
clear cti session session_id
```

**Syntax Description**

| session_id | Unique numeric identifier for the session. String length is 1 to 10 characters. String value is 1 to 2147483647. |

**Command Default**

The CTI session between the application and the router is active.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command gracefully disassociates the connection between a CSTA application and Cisco Unified CME. Use this command to direct Cisco Unified CME to send a SIP BYE for the CSTA call to the application and to clean up the session internally. This command does not reset the IP phone.

To disassociate the connection without using this command, you must reboot the router or the CSTA client application.

This command has a **no** form, but the **no** form has no effect.

To determine the ID for an active CTI session, use the **show cti session** command.

**Examples**

The following example shows how to tear down session 10133 between a CSTA client application and Cisco Unified CME:

```
Router# clear cti session 10133
Router#
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>show cti session</strong></td>
<td>Displays active CTI sessions.</td>
</tr>
</tbody>
</table>
clear telephony-service conference hardware number

To drop all conference parties and clear the conference call, use the `clear telephony-service conference hardware number` command in privileged EXEC mode.

```
clear telephony-service conference hardware number number
```

**Syntax Description**

- `number`: Conference telephone or extension number.

**Command Default**

The conference call continues with all current parties.

**Command Modes**

- Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `show telephony-service conference hardware` command to display the active hardware conferences. Use the `clear telephony-service conference hardware number` command to clear the desired conference.

**Examples**

The following example clears the conference number 1111 and drops all its parties:

```
Router# clear telephony-service conference hardware number 1111
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>show telephony-service conference hardware</code></td>
<td>Displays information about hardware conferences in a Cisco CME system.</td>
</tr>
</tbody>
</table>
clear telephony-service ephone-attempted-registrations

To empty the log of ephones that unsuccessfully attempt to register with Cisco Unified CME, use the `clear telephony-service ephone-attempted-registrations` command in privileged EXEC configuration mode.

**Syntax Description**
This command has no keywords or arguments.

**Command Default**
The log continues to accumulate attempted ephone registrations.

**Command Modes**
Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
The `no auto-reg-ephone` command blocks the automatic registration of ephones whose MAC addresses are not explicitly listed in the configuration. When automatic registration is blocked, Cisco Unified CME records the MAC addresses of phones that attempt to register but cannot because they are blocked.

Use the `show ephone attempted-registrations` command to view the list of phones that have attempted to register but have been blocked. The `clear telephony-service ephone-attempted-registrations` command clears the list.

**Examples**
The following example clears the attempted-registrations log.

```
Router# clear telephony-service ephone-attempted-registrations
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>auto-reg-ephone</code></td>
<td>Enables automatic registration of ephones with Cisco Unified CME.</td>
</tr>
<tr>
<td><code>show ephone attempted-registrations</code></td>
<td>Displays the log of ephones that unsuccessfully attempt to register with Cisco CME.</td>
</tr>
</tbody>
</table>
clear telephony-service xml-event-log

To clear the event table used for the Cisco Unified CME XML application, use the `clear telephony-service xml-event-log` command in privileged EXEC mode.

**Syntax Description**
This command has no keywords or arguments.

**Command Default**
The XML event table is not cleared.

**Command Modes**
Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
The `show fb-its-log` command displays the contents of the XML event table.

**Examples**
The following example clears the entries from the XML event table:

```
Router# clear telephony-service xml-event-log
```
clear voice fac statistics

To clear the voice FAC statistics information, use the clear voice fac statistics command in user EXEC or privileged EXEC mode.

```
clear voice fac statistics
```

Syntax Description
This command has no arguments or keywords.

Command Default
No default behavior or value.

Command Modes
Privileged EXEC.

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

Usage Guidelines
Use this command to clear the voice Forced Authentication Code (FAC) statistics information collected by the system.

Router #clear voice fac statistics

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice fac statistics</td>
<td>Displays details of phones that attempted to register and failed.</td>
</tr>
</tbody>
</table>
clear voice lpcor statistics

To clear all logical partitioning class of restriction (LPCOR) statistics that are displayed when the `show voice lpcor statistics` command is used, use the `clear voice lpcor statistics` command in privileged EXEC mode.

Syntax Description
This command has no arguments or keywords.

Command Default
Statistics continue to increment.

Command Modes
Privileged EXEC (#)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines
This command resets all LPCOR failed-call statistics to 0. Use the `show voice lpcor statistics` command to display the current statistics.

Examples
The following example resets the LPCOR statistics:

```
Router# clear voice lpcor statistics
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice lpcor statistics</td>
<td>Displays information about LPCOR policies and calls.</td>
</tr>
<tr>
<td>voice lpcor policy</td>
<td>Creates a LPCOR policy for a resource group.</td>
</tr>
</tbody>
</table>
clear voice moh-group statistics

To clear the display of MOH subsystem statistics information and reset the packet counters, use the **clear voice moh-group statistics** command in privileged EXEC mode.

### Syntax Description
This command has no arguments or keywords

### Command Modes
Privileged EXEC (#)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0 Cisco Unified SRST 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0 Cisco Unified SRST 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use this command to clear the display of MOH subsystem statistics information displayed by the show voice moh-group statistics command.

We recommend that the clear voice moh-group statistics should be used once every two years to reset the packet counters. Each packet counter is of 32 bit size limit and the largest count a packet counter can hold is 4294967296 intervals. This means that with 20 milliseconds packet interval (for G.711), the counters will restart from 0 any time after 2.72 years (2 years and 8 months).

### Examples

```plaintext
Router# clear voice moh-group statistics
All moh group stats are cleared
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice moh-group statistics</td>
<td>Displays the MOH subsystem statistics information</td>
</tr>
<tr>
<td>show voice moh-group</td>
<td>Displays the MOH groups configured</td>
</tr>
</tbody>
</table>
clear voice register attempted-registrations

To clear the attempted-registrations, use the clear voice register attempted-registrations command in voice register global mode.

```
clear voice register attempted registrations [{ip ip-address|mac H.H.H}]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ip</td>
<td>(Optional) IP address of the SIP phone attempting to register.</td>
</tr>
<tr>
<td>ip-address</td>
<td></td>
</tr>
<tr>
<td>mac</td>
<td>(Optional) MAC address of the SIP phone attempting to register.</td>
</tr>
<tr>
<td>H.H.H</td>
<td></td>
</tr>
</tbody>
</table>

**Command Default**

The attempted-registration entries are not cleared.

**Command Modes**

Privileged EXEC.

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to delete the entries in the attempted-registration table. The clear voice register attempted-registrations command does not alter the table size, but clears the existing entries. A user confirmation is sought before the cleanup is done.

The primary key to recognize the SIP phones that fail to register is through their MAC address (hardware address) and the secondary key is the IP address. You can clear the attempted registration entry for a specific phone that failed to register by providing its IP address or MAC address and create more space for new attempted registration entries in the attempted-registrations table. When no options (IP or MAC) are selected, all the entries are removed. A user confirmation is sought in such a case, before clearing the attempted-registrations table.

The ip keyword allows you to delete entries corresponding to a specific IP address. Similarly, the mac keyword allows you to clear the entries related to a specific MAC address. User confirmation is not sought if ip or mac option is used.

**Examples**

```
Router # clear voice regis attempted-registrations
This will clear all the entries. Proceed? Yes/No? [no]: Yes
```

```
Router# clear voice register attempted-registrations ?
  ip   IP Address of the phone
  mac  MAC Address of the phone
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>attempted-registrations size</td>
<td>Allows to set the size of the attempted-registrations table.</td>
</tr>
<tr>
<td>show voice register attempted-registration</td>
<td>Displays details of phones that attempted to register and failed.</td>
</tr>
</tbody>
</table>
cnf-file

To specify the generation of different phone configuration files by type of phone or by individual phone, use the `cnf-file` command in telephony-service configuration mode. To return to the default, use the `no` form of this command.

```
cnf-file {perphonetype|perphone}
no cnf-file {perphonetype|perphone}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>perphonetype</code></td>
<td>A separate configuration file is generated for each type of phone.</td>
</tr>
<tr>
<td><code>perphone</code></td>
<td>A separate configuration file is generated for each phone.</td>
</tr>
</tbody>
</table>

**Command Default**

A single configuration file is used for all phones.

**Command Modes**

Telephony-service (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Configuration files can be applied in the following ways:

- **Per system**—All phones use a single configuration file. This is the default behavior for Cisco Unified CME and does not require you to configure this command. The default user and network locale in a single configuration file are applied to all phones in the Cisco Unified CME system. Alternative and user-defined user and network locales are not supported. To use the per-system option after configuring this command, use the `no cnf-file` command to reset the option to default behavior.

- **Per phone type**—Creates separate configuration files for each phone type. For example, all Cisco Unified IP Phone 7960s use XMLDefault7960.cnf.xml, and all Cisco Unified IP Phone 7905s use XMLDefault7905.cnf.xml. All phones of the same type use the same configuration file which is generated using the default user and network locale. This option is not supported if the `cnf-file location` is configured for system.

- **Per phone**—Creates a separate configuration file for each phone, by MAC address. For example, a Cisco Unified IP Phone 7960 with the MAC address 123.456.789 creates the per-phone configuration file SEP123456789.cnf.xml. The configuration file for a phone is generated with the default user and network locale unless a different user and network locale is applied to the phone using an ephone template. This option is not supported if the location option is system.

To reset the type of configuration file to the default, use the `no` form of this command and the keyword that you previously used to set the type.

This feature is supported only on the following phones:

- Cisco Unified IP Phones 7940 and 7940G
- Cisco Unified IP Phones 7960 and 7960G
- Cisco Unified IP Phone 7970G
- Cisco Unified IP Phone 7971G-GE
Examples

The following example selects flash memory as the configuration file storage location and per-phone as the type of configuration files that the system should generate.

telephony-service
cnf-file location flash:
cnf-file perphone

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>cnf-file location</td>
<td>Specifies a storage location for XML configuration files.</td>
</tr>
<tr>
<td>create cnf</td>
<td>Generates the XML configuration files used for provisioning SCCP phones.</td>
</tr>
</tbody>
</table>
cnf-file location

To specify a storage location for phone configuration files, use the `cnf-file location` command in telephony-service configuration mode. To return to the default, use the `no` form of this command.

```
cnf-file location {flash: | slot0: | tftp|tftp-url}
no cnf-file location {flash: | slot0: | tftp|tftp-url}
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>flash:</th>
<th>Router flash memory.</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>slot0:</td>
<td>Router slot 0 memory.</td>
</tr>
<tr>
<td></td>
<td>tftp</td>
<td>External TFTP server at the specified URL.</td>
</tr>
<tr>
<td></td>
<td>tftp-url</td>
<td></td>
</tr>
</tbody>
</table>

**Command Default**

A single phone configuration file is stored in system memory and is used by all phones.

**Command Modes**

Telephony-service configuration

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

TFTP does not support file deletion. When configuration files are updated, they overwrite any existing configuration files with the same name. If you change the configuration file location, files are not deleted from the TFTP server.

You can specify any of the following locations in which to store configuration files:

- **System**—This is the default. When the system is the storage location, there is only one default configuration file and it is used for all phones in the system. All phones, therefore, use the same user locale and network locale. User-defined user and network locales are not supported. To use the system location, do not use this command to specify a location other than system or use the no version of this command to reset the option from a previous, different location.

If VRF Support on Cisco Unified CME is configured and the `cnf-file location` command is configured for `system:`, the configuration file for an ephone in a VRF group is created in `system:/its/vrf<group-tag>/`. The vrf group directory is created and appended to the TFTP path automatically. No action is required on your part. The location for locale files is not affected. Locale files are created in `system:/its/`.

- **Flash or slot 0**—When flash or slot 0 memory on the router is the storage location, you can create additional configuration files that can be applied per phone type or per individual phone. Up to five user-defined user and network locales can be used in these configuration files. To store configuration files in flash or slot 0, use the `cnf-file location flash:` or `cnf-file location slot0:` command. The generation of configuration files on flash or slot 0 can take up to a minute, depending on the number of files that are being generated.

If VRF Support on Cisco Unified CME is configured and the `cnf-file location` command is configured as `flash:` or `slot0:`, the per phone or per phone type file for an ephone in a VRF group is named...
flash:/its/vrf<group-tag>_<filename> or slot0:/its/vrf<group-tag>_filename>. The vrf group directory is created and appended to the TFTP path automatically. No action is required on your part. The location for locale files is not affected. Locale files are created in flash:/its/ or in slot0:/its.

**Note**

When the storage location chosen is flash and the file system type on this device is Class B(LEFS), make sure to check free space on the device periodically and use the `squeeze` command to free the space used up by deleted files. Unless you use the `squeeze` command, the space used by the moved or deleted configuration files cannot be used by other files.

- **TFTP**—When an external TFTP server is the storage location, you can create additional configuration files that can be applied per phone type or per individual phone. Up to five user-defined user and network locales can be used in these configuration files. To store configuration files on an external TFTP server, use the `cnf-file location tftp url` command.

**Examples**

The following example selects flash memory as the configuration file storage location and per-phone as the type of configuration files that the system should generate.

```plaintext
telephony-service
cnf-file location flash:
cnf-file perphone
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>cnf-file</td>
<td>Specifies the use of different phone configuration files by type of phone or by individual phone.</td>
</tr>
<tr>
<td>create cnf</td>
<td>Generates the XML configuration files used for provisioning SCCP phones.</td>
</tr>
</tbody>
</table>
codec (ephone)

To select a preferred codec for Cisco Unified CME to use when configuring calls for a phone, use the `codec` command in ephone or ephone-template configuration mode. To return to the command default, use the `no codec` form of this command.

```
codec {g711ulaw|g722r64|g729r8 [dspfarm-assist]|ilbc}
no codec
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>g711ulaw</code></td>
<td>Preferred codec: G.711 micro-law 64K bps.</td>
</tr>
<tr>
<td><code>g722r64</code></td>
<td>Preferred codec: G.722-64K bps.</td>
</tr>
<tr>
<td><code>g729r8</code></td>
<td>Preferred codec: G.729-8K bps.</td>
</tr>
<tr>
<td><code>dspfarm-assist</code></td>
<td>(Optional) DSP-farm resources are used for transcoding the segment between the phone and the Cisco Unified CME router if G.711 is negotiated for the call.</td>
</tr>
<tr>
<td><code>ilbc</code></td>
<td>Preferred codec: iLBC 20ms.</td>
</tr>
</tbody>
</table>

**Command Default**
G.711 micro-law is the preferred codec.

**Command Modes**
Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>The <code>g722r64</code> and <code>ilbc</code> keywords were added.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command can be used to help save network bandwidth for a remote IP phone.

For calls to phones that are not in the same Cisco Unified CME system (such as VoIP calls), the codec is negotiated based on the protocol that is used for the call (such as H.323). The Cisco Unified CME system plays no part in the negotiation.

The G.722-64K codec is supported on some varieties of phone models. Check your phone documentation to make sure the phone supports the G.722-64K codecs.

The telephone’s firmware version must support the specified codec. If a codec is specified by using this command and a phone does not support the preferred codec, then the phone will use the global codec as specified by using the `codec` command in telephony-service configuration mode or if the global codec is not supported, G.711 micro-law.
For calls to other phones in the same Cisco Unified CME system, an IP phone that is configured to use G.729 will always have its calls set up to use G.729. If the phone participates in a call on a line that is shared with a phone that is configured for G.729 or is paged together with another phone that is configured for G.729, it must use G.729.

When you use the `codec` command without the `dspfarm-assist` keyword, you affect only calls between two phones on the Cisco Unified CME router (such as between an IP phone and another IP phone or between an IP phone and a FXS analog phone). The command has no effect on a call directed through a VoIP dial peer unless you use the `dspfarm-assist` keyword.

When you use the `g729r8` keyword to select the G.729r8 codec for the RTP segment between the IP phone and the Cisco Unified CME router and you also use the `dspfarm-assist` keyword, the router attempts to use DSP-farm resources in the following way: If the IP phone is in a VoIP call (H.323 or SIP) or a Cisco Unified CME conference in which the codec must be set to G.711, the router uses configured DSP-farm resources to attempt to return the segment between the phone and the Cisco Unified CME router to G.729. Adequate DSP resources must be appropriately configured separately.

If the `dspfarm-assist` keyword is configured for a phone and a DSP resource is not available when needed for transcoding, a phone registered to the local Cisco Unified CME router will use G.711 instead of G.729r8. This is not true for non-SCCP call legs; if no DSP resource is available for the transcoding required for a conference, for example, the conference will not be created.

It is recommended that the `dspfarm-assist` keyword be used sparingly and only when absolutely required for bandwidth savings or when you know the phone will have few calls that require a G.711 codec.

You should consider your options carefully when deciding to use the `dspfarm-assist` keyword with the `codec` command. The benefit is that it allows calls to use the G.729r8 codec on the call leg between the IP phone and the Cisco Unified CME router, which saves network bandwidth. The disadvantage is that for situations requiring G.711 codecs, such as conferencing and Cisco Unity Express, DSP resources that can be scarce will be used to transcode the call, and delay will be introduced while voice is shuttled to and from the DSP. In addition, the overuse of this feature can mask configuration errors in the codec selection mechanisms involving dial peers and codec lists.

For information about configuring DSP-farm resources, see the Cisco Unified CME Administrator Guide.

---

**Note**

The `dspfarm-assist` keyword is ignored if the SCCP endpoint type is ATA, VG224, or VG248.

This command can also be configured in ephone-template configuration mode. If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

**Examples**

The following example selects the G.729 codec with DSP farm assist for calls that are being configured for ephone 25:

```bash
ephone 25
button 1:37
codec g729r8 dspfarm-assist
```

The following example uses ephone template 1 to apply the G.729 codec preference to ephone 25:

```bash
ephone-template 1
codec g729r8
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dspfarm (dspfarm)</td>
<td>Enables digital-signal-processor (DSP) farm service</td>
</tr>
<tr>
<td>dsp services dspfarm</td>
<td>Specifies the NM-HDV or NM-HDV-FARM on which DSP-farm services are to be enabled.</td>
</tr>
<tr>
<td>dspfarm transcoder maximum sessions</td>
<td>Specifies the maximum number of transcoding sessions to be supported by a DSP farm.</td>
</tr>
<tr>
<td>show dspfarm</td>
<td>Displays summary information about DSP resources.</td>
</tr>
</tbody>
</table>
codec (telephony-service)

To select a default codec for SCCP IP phones in Cisco Unified CME, use the codec command in telephony-service configuration mode. To disable the codec, use the no form of this command.

```
codec {g711ulaw|g722r64}
no codec
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>g711-ulaw</td>
<td>Preferred codec: G.711 micro-law.</td>
</tr>
<tr>
<td>g722-64k</td>
<td>Preferred codec: G.722 64K bps.</td>
</tr>
</tbody>
</table>

**Command Default**
The default is G.711 micro-law.

**Command Modes**
Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command can be used to help save network bandwidth for a remote IP phone.

The G.722-64K codec is supported on certain phones only, such as the Cisco Unified IP Phone 7906G, 7911G, 7941G-GE, 7942G, 7945G, 7961G-GE, 7962G, 7965G, and 7975G. Check your phone documentation to make sure your phones support the G.722-64K codec.

The telephone firmware version on a Cisco Unified IP phone must support the specified codec. If this command is configured and a phone does not support the specified codec, the default codec for that phone is G.711 micro-law.

**Examples**
The following example shows how to configure a G.722-64K codec in telephony-service configuration mode:

```
Router(config)# telephony-service
Router(config-telephony)# codec g722r64
Router(config-telephony)# service phone g722CodecSupport 2
Router(config-telephony)#
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>service</td>
<td>Modifies VendorConfig parameters in configuration files for IP phones.</td>
</tr>
<tr>
<td>phone</td>
<td></td>
</tr>
</tbody>
</table>
**conference (ephone-dn)**

To configure a conference associated with a directory number, use the `conference` command in ephone-dn configuration mode. To disable a conference associated with a directory number, use the `no` form of this command.

```
conference {ad-hoc [video]|meetme [video] [homogeneous]}[unlocked]
no conference {ad-hoc [video]|meetme [video] [homogeneous]} unlocked
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ad-hoc</td>
<td>Configures ad hoc conferences.</td>
</tr>
<tr>
<td>video</td>
<td>(Optional) Configures video conferences.</td>
</tr>
<tr>
<td>meetme</td>
<td>Configures meet-me conferences.</td>
</tr>
<tr>
<td>homogeneous</td>
<td>(Optional) Enables a homogeneous video conference in which all participants use the same video format. <strong>Note</strong> The video keyword must be specified in the command.</td>
</tr>
<tr>
<td>unlocked</td>
<td>Unlocks the meet-me conference bridge.</td>
</tr>
</tbody>
</table>

**Command Default**

No conference is associated with the directory number.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was modified. The command output was enhanced to display the unlocked meet-me conference setting.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
<tr>
<td>15.1(4)M</td>
<td>Cisco Unified CME 8.6</td>
<td>This command was modified to configure video conferences.</td>
</tr>
<tr>
<td>Cisco IOS XE Everest 16.5.1b</td>
<td>Unified CME 11.7</td>
<td>This command was integrated into Cisco IOS XE Everest 16.5.1 Release to support Cisco 4000 Series Integrated Services Router.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Ad hoc conferences are those that begin as a call between the conference creator and another party. The creator then calls other parties and adds them to the original call creating a conference.
Meet-me conferences have a designated meet-me telephone or extension number that all parties call to join the conference. The conference creator initiates the meet-me conference by pressing the MeetMe softkey, then dialing the meet-me number. Other parties join the conference by dialing the meet-me number. Homogenous video conferences only applies to meet-me conferences.

An unlocked meet-me conference allows the user to unlock the meet-me conference bridge. All DN tags with the same number should be configured with the unlocked option. Unlocking the meet-me conference bridge can allow unrestricted and uncontrolled access for the external callers. This feature is support only for meet-me conferences.

When you unlock meet-me conference bridge in Cisco Unified CME, the user can initiate a meet-me conference without pressing the MeetMe softkey, which would allow the external callers to initiate a meet-me conference.

Note

To configure an unlocked meet-me conference, all ephone-dn tags associated with the same number should have the unlocked option configured. If some of the ephone-dn tags do not have the unlocked option configured, the unlocked meet-me conference may not work properly.

Use the `ephone-dn` command to configure enough extensions for your conference needs. Each extension can handle two conference parties if the `dual-line` keyword is used with the `ephone-dn` command, as shown in the following example. Use the `show ephone-dn` command to display phone information for the extension.

Examples

The following example configures extension 9001 as a four-party meet-me conference number.

```
Router(config)# ephone-dn 1 dual-line
Router(config-ephone-dn)# number 9001
Router(config-ephone-dn)# conference meetme
Router(config-ephone-dn)# no huntstop
Router(config)# ephone-dn 2 dual-line
Router(config-ephone-dn)# number 9001
Router(config-ephone-dn)# conference meetme
Router(config-ephone-dn)# preference 1
```

You must configure additional directory numbers to add more parties to the conference.

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td><code>show ephone-dn</code></td>
<td>Displays phone information for specified dn-tag or for all dn-tags.</td>
</tr>
</tbody>
</table>
conference (voice register template)

To enable the soft key for conference in a SIP phone template, use the conference command in voice register template configuration mode. To disable the soft key, use the no form of this command.

```bash
conference
no conference
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Soft key for conference is enabled.

**Command Modes**

Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables a soft key for conference in the specified template which can then be applied to SIP phones. The conference soft key is enabled by default. To disable the conference soft key, use the no conference command. To apply a template to a SIP phone, use the template command in voice register pool configuration mode.

**Examples**

The following example shows how to disable the conference soft key in template 1:

```bash
Router(config)# voice register template 1
Router(config-register-temp)# no conference
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>template (voice register pool)</td>
<td>Applies a template to a SIP phone.</td>
</tr>
<tr>
<td>transfer-attended (voice register template)</td>
<td>Enables a soft key for attended transfer in a SIP phone template.</td>
</tr>
<tr>
<td>transfer-blind (voice register template)</td>
<td>Enables a soft key for blind transfer in a SIP phone template.</td>
</tr>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode for SIP phones.</td>
</tr>
</tbody>
</table>
conference add-mode

To configure the mode for adding parties to ad hoc hardware conferences, use the conference add-mode command in ephone or ephone-template configuration mode. To return to the default, use the no form of this command.

```bash
conference add-mode [creator]
no conference add-mode [creator]
```

**Syntax Description**
- `creator`: Specifies that only the creator can add parties.

**Command Default**
Any party can add other parties provided the creator remains in the conference.

**Command Modes**
- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Everest 16.5.1b</td>
<td>Unified CME 11.7</td>
<td>Support for this command was introduced on the Cisco 4000 Series Integrated Services Routers.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

For more control of conference participation, use this command to specify that only the creator can add new parties. This configuration ensures that no one can add parties to the conference without the creator’s knowledge.

Use this command to configure an ephone directly in ephone configuration mode, or use it to configure an ephone template in ephone-template configuration mode. Use the ephone-template command in ephone configuration mode to apply the ephone template to one or more ephones. Use the show telephony-service ephone command to display the add and drop modes for the ephone. Use the show telephony-service ephone-template command to display the ephone template.

**Examples**

The following example configures ad hoc hardware conferences so that only the creator can add participants.

```bash
Router(config)# ephone 1
Router(config-ephone)# conference add-mode creator
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone.</td>
</tr>
<tr>
<td>show telephony-service ephone</td>
<td>Displays configuration for the Cisco IP phones.</td>
</tr>
<tr>
<td>show telephony-service ephone-template</td>
<td>Displays the contents of ephone-templates.</td>
</tr>
</tbody>
</table>
conference add-mode (voice register)

To configure the mode for adding participants to ad-hoc hardware conferences on Cisco Unified SIP IP phones, use the `conference add-mode` command in voice register pool or voice register template configuration mode. To return to the default, use the `no` form of this command.

```
conference add-mode [creator]
no conference add-mode
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>creator</td>
<td>(Optional) Specifies that only the conference creator can add participants to an ad-hoc hardware conference.</td>
</tr>
</tbody>
</table>

**Command Default**
The conference creator or any of the participants can add a new participant.

**Command Modes**
Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>Cisco IOS XE Everest 16.5.1b</td>
<td>Support for this command was introduced on the Cisco 4000 Series Integrated Services Routers.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use the `conference add-mode creator` command to specify that only the conference creator can add new participants. This configuration ensures that no one can add participants to the hardware conference without the creator’s knowledge.

**Examples**
The following example shows how to configure the mode so that only the conference creator can add new participants to a hardware conference on voice register pool 10:

```
Router(config)# voice register pool 10
Router(config-register-pool)# conference add-mode creator
```

The following example shows how to configure the mode so that only the conference creator can add new participants to a hardware conference on template 1:

```
Router(config)# voice register template 1
Router(config-register-temp)# conference add-mode creator
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode and creates a pool configuration for Cisco Unified SIP IP phones in Cisco Unified CME.</td>
</tr>
<tr>
<td>voice register template</td>
<td>Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones.</td>
</tr>
</tbody>
</table>
conference admin

To configure the ephone as the ad hoc and meet-me hardware conference administrator, use the conference admin command in ephone or ephone-template configuration mode. To return to the defaults, use the no form of this command.

```
conference admin
no conference admin
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
This ephone is not the ad hoc and meet-me hardware conference administrator.

**Command Modes**
Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to configure an ad hoc and meet-me hardware conference administrator. The administrator can:

- Dial in to any conference directly through the conference number
- Use the ConfList soft key to list conference parties
- Remove any party from any conference

The administrator can control the use of conference bridges by enforcing time limits and making sure conference bridges are available for scheduled meetings.

Use this command to configure an ephone directly in ephone configuration mode, or use it to configure an ephone template in ephone-template configuration mode. Use the ephone-template command in ephone configuration mode to apply the ephone template to one or more ephones. Use the show telephony-service ephone command to display the add and drop modes for the ephone. Use the show telephony-service ephone-template command to display the ephone template.

**Examples**

The following example configures ephone 1 as the ad hoc and meet-me hardware conference administrator.

```
Router(config)# ephone 1
Router(config-ephone)# conference admin
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone.</td>
</tr>
<tr>
<td>show telephony-service ephone</td>
<td>Displays configuration for the Cisco IP phones.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------------------------------</td>
<td>---------------------------------------</td>
</tr>
<tr>
<td><code>show telephony-service ephone-template</code></td>
<td>Displays the contents of ephone-templates.</td>
</tr>
</tbody>
</table>
conference admin (voice register)

To assign a Cisco Unified SIP IP phone as the ad-hoc or meet-me hardware conference administrator, use the `conference admin` command in voice register pool or voice register template configuration mode. To return to the default, use the `no` form of this command.

```
conference admin
no conference admin
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

The Cisco Unified SIP IP phone is not the conference administrator.

**Command Modes**

Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `conference admin` command to assign an ad-hoc or meet-me hardware conference administrator. The administrator can:

- Dial in to any conference directly through the conference number.
- Use the Conflist soft key to list conference participants.
- Remove any participant from any conference.

The administrator can control the use of conference bridges by enforcing time limits and making sure conference bridges are available for scheduled meetings.

**Examples**

The following example shows how to configure voice register pool 25 as the conference administrator:

```
Router(config)# voice register pool 25
Router(config-register-pool)# conference admin
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode and creates a pool configuration for Cisco Unified SIP IP phones in Cisco Unified CME.</td>
</tr>
<tr>
<td>voice register template</td>
<td>Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones.</td>
</tr>
</tbody>
</table>
### conference drop-mode

To configure the mode for terminating ad hoc hardware conferences when parties drop out, use the `conference drop-mode` command in ephone or ephone-template configuration mode. To return to the default, use the `no` form of this command.

#### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>`conference drop-mode [{creator</td>
<td>local}]`</td>
</tr>
<tr>
<td>`no conference drop-mode [{creator</td>
<td>local}]`</td>
</tr>
</tbody>
</table>

#### Command Default

The conference is not dropped, regardless of whether the creator hangs up, provided three parties remain in the conference.

#### Command Modes

- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)

#### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T</td>
</tr>
<tr>
<td>Cisco IOS XE Everest 16.5.1b</td>
<td>Unified CME 11.7</td>
<td>Support for this command was introduced on the Cisco 4000 Series Integrated Services Routers.</td>
</tr>
</tbody>
</table>

#### Usage Guidelines

For more control of conference participation, use this command to specify that the conference drops when the creator hangs up (see the example). This configuration ensures that the conference cannot continue without the creator’s presence.

Use this command to configure an ephone directly in ephone configuration mode, or use it to configure an ephone template in ephone-template configuration mode. Use the `ephone-template` command in ephone configuration mode to apply the ephone template to one or more ephones. Use the `show telephony-service ephone` command to display the add and drop modes for the ephone. Use the `show telephony-service ephone-template` command to display the ephone template.

#### Examples

The following example configures ad hoc hardware conferences so that only the creator can add participants and the active conference terminates when the creator hangs up.

```
Router(config)# ephone 1
Router(config-ephone)# conference drop-mode creator
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone.</td>
</tr>
<tr>
<td>show telephony-service ephone</td>
<td>Displays configuration for the Cisco IP phones.</td>
</tr>
<tr>
<td>show telephony-service ephone-template</td>
<td>Displays the contents of ephone-templates.</td>
</tr>
</tbody>
</table>
conference drop-mode (voice register)

To specify who can terminate an active ad-hoc hardware conference by hanging up, use the `conference drop-mode` command in voice register pool or voice register template configuration mode. To return to the default, use the `no` form of this command.

```
conference drop-mode {creator|local}
no conference drop-mode
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>creator</td>
<td>Terminates the active conference when the conference creator hangs up.</td>
</tr>
<tr>
<td>local</td>
<td>Terminates the active conference when the last local participant hangs up or drops out of the conference.</td>
</tr>
</tbody>
</table>

**Command Default**

An active conference is never dropped.

**Command Modes**

Voice register pool configuration (config-register-pool)

Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>Cisco IOS XE Everest 16.5.1b</td>
<td>Support for this command was introduced on the Cisco 4000 Series Integrated Services Routers.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `conference drop-mode creator` command to specify that an active hardware conference is terminated when the creator hangs up. This configuration ensures that the hardware conference cannot continue without the creator’s presence.

**Examples**

The following example shows how to configure an active conference so that it is terminated when the conference creator hangs up:

```
Router(config)# voice register pool 60
Router(config-register-pool)# conference drop-mode creator
```

The following example shows how to configure an active conference so that it is terminated when the last local participant hangs up or drops out of the conference:

```
Router(config)# voice register template 7
Router(config-register-temp)# conference drop-mode local
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode and creates a pool configuration for Cisco Unified SIP IP phones in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
### conference drop-mode (voice register)

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register template</td>
<td>Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones.</td>
</tr>
</tbody>
</table>
conference hardware

To configure a Cisco Unified CallManager Express system for hardware conferencing only, use the conference hardware command in telephony-service configuration mode. To return to the default three-party software conferencing, use the no form of this command.

```
conference hardware
no conference hardware
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Three-party ad hoc software conferencing.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Release 12.4(15)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Everest</td>
<td>Unified CME 11.7</td>
<td>This command was integrated into Cisco IOS XE</td>
</tr>
<tr>
<td>16.5.1b</td>
<td></td>
<td>Everest 16.5.1 Release to support Cisco 4000</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Series Integrated Services Router.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Software conferencing allows a maximum of three parties in a conference. Use this command to take advantage of DSP farm resources for hardware conferencing that allows ad hoc conferences with more than three parties.

If you need ad hoc hardware conferences, you must use this command to configure DSP farm hardware conferencing. You can configure other conferencing features using the conference-join custom-cptone, conference-leave custom-cptone, and maximum conference-participants commands in DSP farm profile configuration mode. Use the show dspfarm profile command to display the DSP farm profile.

**Examples**

The following example configures hardware conferencing as the default for ad hoc conferences on this Cisco Unified CallManager Express system:

```
Router(config)# telephony-service
Router(config-telephony)# conference hardware
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>conference-join custom-cptone</td>
<td>Associates a custom call-progress tone to indicate joining a conference with a DSP farm profile.</td>
</tr>
<tr>
<td>conference-leave custom-cptone</td>
<td>Associates a custom call-progress tone to indicate leaving a conference with a DSP farm profile.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>maximum conference-participants</td>
<td>Configures the maximum number of conference participants allowed in each conference.</td>
</tr>
<tr>
<td>show dspfarm profile</td>
<td>Displays configured DSP farm profile information.</td>
</tr>
</tbody>
</table>
conference hardware (voice register global)

To configure Cisco Unified Communications Manager Express (Cisco Unified CME) DSPFarm hardware-based ad-hoc conferencing, use the `conference hardware` command in voice register global configuration mode. To return to the default, use the `no` form of this command.

```
conference hardware
no conference hardware
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Cisco Unified SIP IP phone local conference is enabled.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>Cisco Unified CME 9.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>Cisco IOS XE Everest 16.5.1b</td>
<td>Unified CME 11.7</td>
<td>This command was integrated into Cisco IOS XE Everest 16.5.1 Release yto support Cisco 4000 Series Integrated Services Router.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `conference hardware` command in voice register global configuration mode to take advantage of DSPfarm resources that allow ad-hoc hardware conferences with more than three parties.

Enable hardware conferencing in telephony-service configuration mode before configuring hardware conferencing in voice register global configuration mode. Otherwise, the configuration of hardware conferencing in voice register global configuration mode will be rejected.

If you apply any changes to the configuration of the hardware conference, you must use the `create profile` command in voice global configuration mode to regenerate the configuration profile files required for Cisco Unified SIP IP phones. Then, reboot the phone.

**Examples**

The following example shows how to configure Cisco Unified CME DSPFarm hardware-based ad-hoc conferencing:

```
Router(config)# telephony-service
Router(config-telephony)# conference hardware

Router(config)# voice register global
Router(config-register-global)# conference hardware
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>conference hardware</td>
<td>Configures a Cisco Unified CME system for hardware conferencing only in telephony-service configuration mode.</td>
</tr>
</tbody>
</table>
conference max-length

To allow conferencing, only if the number of dialed digits are within the max-length limit, use the `conference max-length` command. To remove the configuration, use the `no` form of this command.

```
conference max-length <value>
no conference max-length
```

**Syntax Description**

- **value**: Maximum number of digits that can be dialed. The range is from 3 to 16.

**Command Default**

By default, no value is defined for conferencing.

**Command Modes**

- Ephone (config-ephone)
- Ephone-template ephone (config-ephone-template)
- Voice register pool (config-register-pool)
- Voice register template(config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `conference max-length` command to configure, the Cisco Unified CME to allow conferencing, only if the dialed digits are within the maximum length limit.

**Example**

The following example shows how to configure the maximum length of 8 digits that can be dialed to make a conference call:

```
Router(config)# ephone 1
Router(config-ephone)# conference max-length 8
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>conference-pattern blocked</td>
<td>Blocks extensions on an ephone or a voice register pool from making conference calls.</td>
</tr>
<tr>
<td>conference transfer-pattern</td>
<td>Apply transfer-pattern configuration for conference cases.</td>
</tr>
<tr>
<td>transfer max-length</td>
<td>Allows transfer of calls to phones, where the number of dialed digits are less than the maximum length configured.</td>
</tr>
<tr>
<td>transfer-pattern (telephony-service)</td>
<td>Allows the transfer of calls to phones outside the Cisco Unified CME network.</td>
</tr>
</tbody>
</table>
conference-pattern blocked

To prevent extensions on an ephone or a voice register pool from initiating a conference to external numbers, use the `conference-pattern blocked` command. Note that the `conference-pattern blocked` command does not impact call transfer functions. To remove the configuration, use the `no` form of this command.

```
conference-pattern blocked
no conference-pattern blocked
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

No default values are defined.

**Command Modes**

- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)
- Voice register pool configuration (config-register-pool)
- Voice register template configuration (config-register-temp)

**Command History**

- **Cisco IOS Release**: 15.4(3)M
  - **Modification**: This command was introduced.

**Usage Guidelines**

Use the `conference-pattern blocked` command to prevent specific extensions from making conference calls to patterns generally allowed through the `transfer-pattern` command.

**Example**

The following example shows how to prevent extensions from making conference calls using the `conference-pattern blocked` command:

```
Router(config)# ephone 1
Router(config-ephone-template)# conference-pattern blocked
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>conference max-length</td>
<td>Allows conferences to numbers where dialed digits are within the configured maximum length value.</td>
</tr>
<tr>
<td>conference transfer-pattern</td>
<td>Apply transfer-pattern configuration for conference cases.</td>
</tr>
<tr>
<td>transfer-pattern blocked</td>
<td>Blocks individual phones from transferring calls to nonlocal numbers that have been globally enabled for transfer.</td>
</tr>
<tr>
<td>transfer-pattern (telephony-service)</td>
<td>Allows the transfer of calls to phones outside the Cisco Unified CME network.</td>
</tr>
</tbody>
</table>
conference transfer-pattern

To configure a Cisco Unified CallManager Express system to apply transfer-pattern <pattern> to the conference call using conference softkey or feature button, use the `conference transfer-pattern` command in telephony-service configuration mode. To return to the default, use the `no conference transfer-pattern` command.

```
congference transfer-pattern
no conference transfer-pattern
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Transfer-pattern <pattern> does not apply to call conferencing.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(4)M</td>
<td>Cisco Unified CME 9.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

There is no check for the conference numbers for call conferencing. Use this command to apply transfer-pattern for call conferencing.

**Examples**

The following example enables transfer-pattern to be applied for conference parties:

```
Router(config)# telephony-service
Router(config-telephony)# conference transfer-pattern
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>telephony-service</td>
<td>Enters telephony-service configuration mode.</td>
</tr>
</tbody>
</table>
cor (ephone-dn)

This command is now documented as the corlist command. For complete command information, see the corlist command page.
cor (voice register)

To configure a class of restriction (COR) on the VoIP dial peers associated with directory numbers, use the `cor` command in voice register pool or voice register template configuration mode. To disable a COR associated with directory numbers, use the `no` form of this command.

```
cor {incoming | outgoing} cor-list-name {cor-list-number starting-number [- ending-number] | default} 
no cor {incoming | outgoing} cor-list-name {cor-list-number starting-number [- ending-number] | default}
```

**Syntax Description**

- `cor-list-name`: COR list name.
- `cor-list-number`: COR list identifier.
- `starting-number`: Start of a directory number range, if an ending number is included. Can also be a standalone number.
- `-`: (Optional) Indicator that a full range is configured.
- `ending-number`: (Optional) End of a directory number range.
- `default`: Instructs the COR list to assume behavior according to a predefined default COR list.

**Command Default**

COR is not configured on VoIP dial peers.

**Command Modes**

- Voice register pool configuration (config-register-pool)
- Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was added to Cisco CallManager Express (Cisco CME).</td>
</tr>
<tr>
<td>Cisco IOS XE Fuji 16.7.1</td>
<td>Unified CME 12.1</td>
<td>This command was supported in voice register template configuration mode.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `cor` command sets the dial-peer COR parameter for dynamically created VoIP dial peers. A list-based mechanism assigns COR parameters to specific set of number ranges. The COR functionality provides the ability to deny certain call attempts on the basis of the incoming and outgoing class of restrictions provisioned on the dial peers. This functionality provides flexibility in network design, allows users to block calls (for example, calls to 900 numbers), and applies different restrictions to call attempts from different originators.
COR specifies which incoming dial peer can use which outgoing dial peer to make a call. Each dial peer can be provisioned with an incoming and an outgoing COR list.

A default COR is assigned to the directory numbers that do not match any COR list number or number range. During Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) registration, a dial peer is created and that dial peer includes a default COR value. The `cor` command allows you to change the automatically selected default.

In dial-peer configuration mode, build your COR list and add members. Then in voice register pool configuration mode, use the `cor` command to apply the name of the dial-peer COR list.

If the `cor` command is configured under voice register template and voice register pool configuration modes, precedence is for the COR configuration under voice register pool configuration mode.

You can have up to four COR lists for the Cisco Unified SIP SRST configuration, comprised of incoming or outgoing dial peers. The first four COR lists are applied to a range of phone numbers. The phone numbers that do not have a COR list configuration are assigned to the default COR list, providing that a default COR list has been defined.

Configure the `id` (voice register pool) command before any other voice register pool commands, including the `cor` command. The `id` command identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.

**Examples**

The following is sample output from the `show running-config` command:

```
.. voice register pool 1
  id mac 0030.94C2.A22A
  preference 5
  cor incoming call 91 1 91011
  translate-outgoing called 1
  proxy 10.2.161.187 preference 1 monitor probe icmp-ping
  alias 1 94... to 91011 preference 8
  voice-class codec 1
  .
  .
  .
  .
  dial-peer cor custom
  name 95
  name 94
  name 91
  !
  dial-peer cor list call 91
  member 91
  !
  dial-peer voice 91500 pots
corlist incoming call 91
corlist outgoing call 91
destination-pattern 91500
  port 1/0/0
  .
  .
  .
```

The following is a sample output of the `show running-config` for COR configured under voice register template configuration mode.
.. 
voice register template 1
  id mac 0030.94C2.A22A
  preference 5
  cor incoming call91 1 91011
  translate-outgoing called 1
  proxy 10.2.161.187 preference 1 monitor probe icmp-ping
  alias 1 94... to 91011 preference 8
  voice-class codec 1
  .
  .
  .
dial-peer cor custom
  name 95
  name 94
  name 91
  .
dial-peer cor list call91
  member 91
  .
dial-peer voice 91500 pots
  corlist incoming call91
  corlist outgoing call91
  destination-pattern 91500
  port 1/0/0
  .

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dial-peer cor custom</td>
<td>Specifies that named CORs apply to dial peers.</td>
</tr>
<tr>
<td>dial-peer cor list</td>
<td>Defines a COR list name.</td>
</tr>
<tr>
<td>id (voice register pool)</td>
<td>Explicitly identifies a locally available individual Cisco SIP IP phone, or</td>
</tr>
<tr>
<td></td>
<td>when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.</td>
</tr>
<tr>
<td>member (dial-peer cor list)</td>
<td>Adds a member to a dial-peer COR list.</td>
</tr>
<tr>
<td>name (dial-peer custom cor)</td>
<td>Provides a name for a custom COR.</td>
</tr>
<tr>
<td>show dial-peer voice</td>
<td>Displays information for voice dial peers.</td>
</tr>
</tbody>
</table>
corlist

This command was previously documented as the cor command.

To apply a class of restriction (COR) to the dial peers associated with a Cisco CME extension (ephone-dn), use the corlist command in ephone-dn configuration mode. To disable the COR associated with an extension, use the no form of this command.

```
corlist {incoming|outgoing} corlist-name
no corlist {incoming|outgoing}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>incoming</th>
<th>Specifies a COR list to be used by incoming dial peers.</th>
</tr>
</thead>
<tbody>
<tr>
<td>outgoing</td>
<td>Specifies a COR list to be used by outgoing dial peers.</td>
</tr>
<tr>
<td>corlist-name</td>
<td>COR list name.</td>
</tr>
</tbody>
</table>

**Command Default**

No COR is used by the dial peers associated with the extension that is being configured.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.</td>
</tr>
<tr>
<td>12.2(8)T1</td>
<td>Cisco ITS 2.0</td>
<td>This command was implemented on the Cisco 2600-XM and Cisco 2691.</td>
</tr>
<tr>
<td>12.2(11)T</td>
<td>Cisco ITS 2.01</td>
<td>This command was implemented on the Cisco 1760.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

COR is used to specify which incoming ephone-dn dial peer can use which outgoing ephone-dn dial peer to make a call. COR denies certain call attempts on the basis of the incoming and outgoing class of restrictions that have been provisioned on the dial peers. This functionality provides flexibility in network design, allows administrators to block calls (for example, calls to 900 numbers), and applies different restrictions to call attempts from different originators.

Each dial peer can be provisioned with an incoming and an outgoing COR list.

The corlist incoming and corlist outgoing commands in dial-peer configuration mode perform these functions for dial peers that are not associated with ephone-dns. The dial-peer cor list and member commands define the sets of capabilities, or COR lists, that are referred to in the corlist commands.

**Examples**

The following example shows how to set a COR parameter for incoming calls to dial peers associated with the extension that has the dn-tag 1:

```
Router(config)# ephone-dn 1
```
Router(config-ephone-dn)# corlist incoming
corlist1

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>corlist incoming</td>
<td>Specifies the COR list to be used when a specified dial peer acts as the incoming dial peer.</td>
</tr>
<tr>
<td>corlist outgoing</td>
<td>Specifies the COR list to be used by an outgoing dial peer.</td>
</tr>
<tr>
<td>dial-peer corlist</td>
<td>Defines a COR list name.</td>
</tr>
</tbody>
</table>
create cnf-files

To build the eXtensible Markup Language (XML) configuration files that are required for IP phones in Cisco Unified CME, use the create cnf-files command in telephony-service configuration mode. To remove the configuration files and disable the automatic generation of configuration files, use the no form of this command.

create cnf-files
no create cnf-files

Syntax Description
This command has no arguments or keywords.

Command Default
Required XML configuration files are not built.

Command Modes
Telephony-service configuration (config-telephony)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was modified to interact with the cnf-file command and the cnf-file location command.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>Modifications to this command for interacting with the cnf-file command and the cnf-file location command were integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines
Use this command to generate the XML configuration files used for provisioning SCCP phones and write the files to the location specified with the cnf-file location command.

Examples
The following example builds the necessary XML configuration files on the Cisco Unified CME router:

```
Router(config)# telephony-service
Router(config-telephony)# create cnf-files
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>cnf-file</td>
<td>Specifies the type of configuration file to be created.</td>
</tr>
<tr>
<td>cnf-file location</td>
<td>Specifies a storage location for phone configuration files</td>
</tr>
</tbody>
</table>
create cnf-files (voice-gateway)

To generate the eXensible Markup Language (XML) configuration files that are required to autoconfigure the Cisco voice gateway, use the `create cnf-files` command in voice-gateway configuration mode. To disable the generating of configuration files, use the `no` form of this command.

```
create cnf-files
no create cnf-files
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Required XML configuration files are not built.

**Command Modes**

Voice-gateway configuration (config-voice-gateway)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Cisco Unified CME writes the XML files generated by this command to the location specified with the `cnf-file location` command, or to the default location in system:/its/. The voice gateway downloads its configuration file from Cisco Unified CME when you run the autoconfiguration process on the voice gateway.

**Examples**

The following example shows that the gateway configuration files are generated by Cisco Unified CME:

```
voice-gateway system 1
network-locale FR
type VG224
mac-address 001F.A30F.8331
voice-port 0-23
create cnf-files
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>cnf-file location</td>
<td>Specifies a storage location for phone configuration files.</td>
</tr>
<tr>
<td>reset (voice-gateway)</td>
<td>Performs a complete reboot of all analog phones associated with the voice gateway and registered to Cisco Unified CME.</td>
</tr>
<tr>
<td>restart (voice-gateway)</td>
<td>Performs a fast restart of all analog phones associated with the voice gateway and registered to Cisco Unified CME.</td>
</tr>
</tbody>
</table>
create profile (voice register global)

To generate the configuration profile files required for SIP phones, use the `create profile` command in voice register global configuration mode. To return to the default, use the `no` form of this command.

```
create profile
no create profile
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Configuration files are not generated.

**Command Modes**
Voice register global configuration (config-register-global)

**Command History**
```
Cisco IOS Release  
Modification
12.4(4)T  
Cisco CME 3.4  
This command was introduced.
```

**Usage Guidelines**
This command generates the configuration files used for provisioning SIP phones and writes the files to the location specified with the `tftp-path` command.

**Examples**
The following example shows how to create the configuration profile:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# create profile
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>file text (voice register global)</td>
<td>Generates ASCII text files for SIP phones.</td>
</tr>
<tr>
<td>mode (voice register global)</td>
<td>Enables the mode for provisioning SIP phones in a Cisco CallManager Express (Cisco CME) system.</td>
</tr>
<tr>
<td>reset (voice register global)</td>
<td>Performs a complete reboot of all SIP phones associated with a Cisco CME router.</td>
</tr>
<tr>
<td>source-address (voice register global)</td>
<td>Identifies the IP address and port through which SIP phones communicate with a Cisco CME router.</td>
</tr>
<tr>
<td>tftp-path (voice register global)</td>
<td>Specifies the directory to which the provisioning file for SIP phones in a Cisco CallManager Express (Cisco CME) system will be written.</td>
</tr>
</tbody>
</table>
credentials

To enter credentials configuration mode to configure a certificate for a Cisco Unified CME CTL provider or for Cisco Unified SRST router communication to Cisco Unified CallManager, use the credentials command in global configuration mode. To set all commands in credentials configuration mode to the default of nonsecure, use the no form of this command.

credentials
no credentials

Syntax Description
This command has no arguments or keywords.

Command Default
Credentials are not provided.

Command Modes
Global configuration (config)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco SRST 3.3</td>
<td>This command was introduced for Cisco SRST.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced for Cisco Unified CME.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command for Cisco Unified CME was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Fuji 16.7.1 Release</td>
<td>Unified SRST 12.1</td>
<td>This command was introduced for Unified SRST support on Cisco 4000 Series Integrated Services Router.</td>
</tr>
</tbody>
</table>

Usage Guidelines
This command is used to configure credentials service for Cisco Unified CME and Cisco Unified SRST.

Cisco Unified CME

This command is used with Cisco Unified CME phone authentication to configure a CTL provider on each Cisco Unified CME router on which the CTL client is not running. That is, if there is a primary and a secondary Cisco Unified CME router and the CTL client is running on the primary router, a CTL provider must be configured on the secondary router, and vice versa. If the CTL client is running on a router that is not a Cisco Unified CME router, CTL providers must be configured on all Cisco Unified CME routers.

Credentials service for Cisco Unified CME runs on default port 2444.

Cisco Unified SRST

The credential server provides certificates to any device that requests a certificate. The credentials server does not request any data from a client; thus no authentication is necessary. When the client, Cisco Unified CallManager, requests a certificate, the credentials server provides the certificate. Cisco Unified CallManager exports the certificate to the phone, and the Cisco Unified IP phone holds the SRST router certificate in its configuration file. The device certificate for secure SRST routers is placed in the configuration file of the Cisco Unified IP phone because the entry limit in the certificate trust list (CTL) of Cisco Unified CallManager is 32.
Credentials service for SRST runs on default port 2445. Cisco Unified CallManager connects to port 2445 on the secure SRST router and retrieves the secure SRST device certificate during the TLS handshake.

Activate this command on all SRST routers.

**Caution**

For security reasons, credentials service should be deactivated on all SRST routers after provisioning to Cisco Unified CallManager is completed.

### Examples

**Cisco Unified CME**

The following example configures a CTL provider on the Cisco Unified CME router with the IP address 172.19.245.1. CTL providers must be configured on all Cisco Unified CME routers on which the CTL client is not running.

```bash
Router(config)# credentials
Router(config-credentials)# ip source-address 172.19.245.1 port 2444
Router(config-credentials)# trustpoint cmeca
Router(config-credentials)# ctl-service admin user4 secret 0 c89L8o
```

**Cisco Unified SRST**

The following example enters credentials configuration mode and sets the IP source address and the trustpoint:

```bash
Router(config)# credentials
Router(config-credentials)# ip source-address 10.6.21.4 port 2445
Router(config-credentials)# trustpoint srstca
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ctl-service admin</code></td>
<td>Specifies a user name and password to authenticate the CTL client during the CTL protocol.</td>
</tr>
<tr>
<td><code>debug credentials</code></td>
<td>Sets debugging on the credentials service that runs between a Cisco Unified CME CTL provider the CTL client or between an SRST router and Cisco Unified CallManager.</td>
</tr>
<tr>
<td><code>ip source-address (credentials)</code></td>
<td>Enables the Cisco Unified CME or SRST router to receive messages through the specified IP address and port.</td>
</tr>
<tr>
<td><code>show credentials</code></td>
<td>Displays the credentials settings on a Cisco Unified CME or SRST router.</td>
</tr>
<tr>
<td><code>trustpoint (credentials)</code></td>
<td>Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with an SRST router certificate.</td>
</tr>
</tbody>
</table>
cti csta mode basic

To set the CTI interface in Cisco Unified CME into basic mode, use the **cti csta mode basic** command in voice-service configuration mode. To return to default, use the **no** form of this command.

**cti csta mode basic**

**no cti csta mode basic**

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

CTI interface is in advanced mode.

**Command Modes**

Voice-service configuration (config-voi-serv)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>This command was deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command supresses all enhanced extensions/features, such as shared line and shared media, in a CTI message from Cisco Unified CME.

This command is required if the computer-based CSTA client application that is interacting with Cisco Unified CME is a Microsoft Office Communicator (MOC) client.

**Examples**

The following example shows a voice-service configuration with this command enabled:

```
!
voice service voip
no cti shutdown
cti csta mode basic
!
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>cti shutdown</td>
<td>Disables CTI integration.</td>
</tr>
</tbody>
</table>
cti message device-id suppress-conversion

To suppress the conversion or promotion of all extension numbers except the primary number in a CTI message, use the **cti message device-id suppress-conversion** command in voice-service configuration mode. To return to default, use the **no** form of this command.

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

All SCCP extension numbers are converted or promoted in CTI messages.

**Command Modes**

Voice-service configuration (config-voi-serv)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
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</tr>
</thead>
<tbody>
<tr>
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<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>This command was deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies that only the requested (primary) extension number is converted or promoted in the outgoing CTI message when an expanded number is presented in a RequestSystemStatus from a CSTA client application. Use this command to suppress the conversion or promotion of all secondary numbers in a CTI message.

By default, Cisco Unified CME converts or promotes all SCCP primary and secondary extension numbers when reporting events.

**Examples**

The following example shows the voice-service configuration with this command enabled:

```
! voice service voip
no cti shutdown
cti csta mode basic
cti message device-id suppress-conversion
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>cti shutdown</td>
<td>Disables CTI integration.</td>
</tr>
</tbody>
</table>
**cti notify**

To force an ephone-dn into a constant “up” state, use the **cti notify** command in ephone-dn or ephone-dn-template configuration mode. To return to default, use the **no** form of this command.

```
cti notify
no cti notify
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Cisco Unified CME cannot send notifications to the ephone-dn because a CTI session cannot be established.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
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</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
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<tr>
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<td>Unified CME 12.6</td>
<td>This command was deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command forces an ephone-dn into a constant “up” state.

Use this command to permit a CTI session to be established with a directory number that is not associated with a physical device, allowing Cisco Unified CME to send notifications to the directory number. If a directory number is not associated to an ephone configuration that includes the button command, a static fwd is applied to the directory number and all incoming calls are forwarded to another directory number.

If you use an ephone-dn template to apply this command to a directory number and you also use this command in ephone-dn configuration mode for the same directory number, the value that you set in ephone-dn configuration mode has priority.

**Examples**

The following example shows the configuration for ephone-dn 4 including this command. A CTI session can be established for this directory number (204) even though the number is not associated with an ephone configuration because this directory number is always “up.”

```
! ephone-dn 4
   number 204
   cti notify
   cti watch
!
! ephone 1
   mac-address 001E.4A34.A35F
   type 7961
   button 1:1
```
The following example shows how to create the same configuration for ephone-dn 4 using this command in ephone-dn template configuration mode and then applying the template to the directory number:

```
ephone-dn-template 15
cti notify
cti watch
ephone-dn 4
number 204
ephone-dn-template 15
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>ephone-dn-template (ephone-dn)</strong></td>
<td>Applies an ephone-dn template to an ephone-dn.</td>
</tr>
</tbody>
</table>
To allow a CSTA client application to monitor and control a directory number in Cisco Unified CME, use the `cti watch` command in ephone-dn or ephone-dn-template configuration mode. To return to default, use the `no` form of this command.

```plaintext
cti watch
no cti watch
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
A CSTA client application cannot use the CTI interface to monitor and control an ephone-dn in Cisco Unified CME.

**Command Modes**
- Ephone-dn configuration (config-ephone-dn)
- Ephone-dn-template configuration (config-ephone-dn-template)

**Command History**

<table>
<thead>
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<tr>
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<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>This command was deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command enables a CSTA client application to monitor and control a directory number in Cisco Unified CME.

If you use an ephone-dn template to apply this command to a directory number and you also use this command in ephone-dn configuration mode for the same directory number, the value that you set in ephone-dn configuration mode has priority.

**Examples**
The following example shows the configuration for ephone-dn 4 with this command configured. The CSTA application can monitor and control the directory number (204).

```
!  ephone-dn 4
    number 204
    cti notify
    cti watch
```

The following example shows how to create the same configuration for ephone-dn 4 using this command in ephone-dn template configuration mode and applying the template to the directory number:

```
ephone-dn-template 15
    cti notify
```
cti watch
ephone-dn 4
number 204
ephone-dn-template 15

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-dn-template (ephone-dn)</td>
<td>Applies an ephone-dn template to an ephone-dn</td>
</tr>
</tbody>
</table>
The `cti-aware` command is used to bind a session to the CTI subsystem. It is used in voice session-server configuration mode. To return to default, use the `no` form of this command:

```
cti-aware
no cti-aware
```

**Syntax Description**

This command has no keywords or arguments.

**Command Default**

CTI-register heartbeat continues even after the CTI session is shutdown.

**Command Modes**

Voice session-server configuration (config-register-fs)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
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</tr>
<tr>
<td>Cisco IOS XE Gibraltar</td>
<td>Unified CME 12.6</td>
<td>This command was deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
<tr>
<td>16.11.1a Release</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command causes the CSTA SIP keepalive response to stop if the CTI session between Cisco Unified CME and the CSTA client application expires or is down for any reason. By default, the CSTA SIP keepalive response continues even after the CTI session expires and the CSTA client application is unaware that the CTI session is not operational.

**Examples**

The following partial output shows the configuration for a session manager for a CSTA client application in which this command is configured:

```
router# show running-configuration

voice register session-server 1
  register-id appl
  keepalive 360
  cti-aware
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>keepalive (voice register session-server)</td>
<td>Duration for registration after which the registration expires unless the feature server or application reregisters before the registration expiry.</td>
</tr>
<tr>
<td>register-id</td>
<td>Creates an ID for explicitly identifying an external feature server or application during Register requests</td>
</tr>
</tbody>
</table>
ctl-client

To enter CTL-client configuration mode to set parameters for the CTL client, use the `ctl-client` command in global configuration mode. To return to the default, use the `no` form of this command.

```
ctl-client
no  ctl-client
```

**Syntax Description**

This command has no keywords or arguments.

**Command Default**

No CTL-client parameters are set.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication.

**Examples**

The following example defines server IP addresses and trustpoints for the CAPF server, the Cisco Unified CME router, and the TFTP server, as well as trustpoints for SAST1 and SAST2. It also specifies that a new CTL file should be generated.

```
Router(config)#  ctl-client
Router(config-ctl-client)#  server capf 10.2.2.2 trustpoint capftrust
Router(config-ctl-client)#  server cme 10.2.2.3 trustpoint cmetp
Router(config-ctl-client)#  server tftp 10.2.2.4 trustpoint tftptp
Router(config-ctl-client)#  sast1 trustpoint sast1tp
Router(config-ctl-client)#  sast2 trustpoint sast2tp
Router(config-ctl-client)#  regenerate
```
ctl-service admin

To specify a user name and password to authenticate the client during the CTL protocol, use the `ctl-service admin` command in credentials configuration mode. To return to the default, use the `no` form of this command.

```
ctl-service admin username secret {0|1} password-string
no ctl-service admin
```

### Syntax Description

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>username</code></td>
<td>Defines the name that will be used to authenticate the client.</td>
</tr>
<tr>
<td>`secret {0</td>
<td>1}`</td>
</tr>
<tr>
<td></td>
<td>• 0 — Not encrypted.</td>
</tr>
<tr>
<td></td>
<td>• 1 — Encrypted using Message Digest 5 (MD5).</td>
</tr>
<tr>
<td><code>password-string</code></td>
<td>Character string for login authentication</td>
</tr>
</tbody>
</table>

### Command Default

No user name or password is defined for authentication.

### Command Modes

Credentials configuration (config-credentials)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command is used with Cisco Unified CME phone authentication to define a user who will be used to authenticate the CTL client with a CTL provider.

### Examples

The following example creates a CTL provider on a Cisco Unified CME router that is not running the CTL client.

```
Router(config)# credentials
Router(config-credentials)# ip source-address 172.19.245.1 port 2444
Router(config-credentials)# trustpoint ctlpv
Router(config-credentials)# ctl-service admin user4 secret 0 c89L8o
```

### Command

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>debug credentials</code></td>
<td>Sets debugging on the credentials service that runs between a Cisco Unified CME CTL provider and the CTL client or between an SRST router and Cisco Unified CallManager.</td>
</tr>
<tr>
<td><code>show credentials</code></td>
<td>Displays the credentials settings on a Cisco Unified CME or SRST router.</td>
</tr>
<tr>
<td><code>trustpoint (credentials)</code></td>
<td>Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with an SRST router certificate.</td>
</tr>
</tbody>
</table>
Cisco Unified CME Commands: D

- date-format (telephony-service), on page 235
- date-format (voice register global), on page 236
- debug callmonitor, on page 237
- debug capf-server, on page 240
- debug cch323 video, on page 242
- debug credentials, on page 244
- debug cti, on page 246
- debug cti-client, on page 248
- debug ephone alarm, on page 249
- debug ephone blf, on page 251
- debug ephone ccm-compatible, on page 253
- debug ephone detail, on page 255
- debug ephone error, on page 258
- debug ephone extension-assigner, on page 260
- debug ephone hfs, on page 262
- debug ephone keepalive, on page 264
- debug ephone loopback, on page 266
- debug ephone lpcor, on page 271
- debug ephone message, on page 272
- debug ephone mlpp, on page 274
- debug ephone moh, on page 276
- debug ephone mwi, on page 278
- debug ephone paging, on page 280
- debug ephone pak, on page 282
- debug ephone qov, on page 284
- debug ephone raw, on page 286
- debug ephone register, on page 288
- debug ephone sccp-state, on page 290
- debug ephone shared-line-mixed, on page 291
- debug ephone state, on page 294
- debug ephone statistics, on page 296
- debug ephone video, on page 298
- debug ephone vm-integration, on page 300
• debug ephone whisper-intercom, on page 302
• debug mwi relay errors, on page 304
• debug mwi relay events, on page 305
• debug shared-line, on page 306
• debug voice register errors, on page 309
• debug voice register events, on page 311
• default (voice hunt-group), on page 315
• description (ephone), on page 316
• description (ephone-dn and ephone-dn-template), on page 317
• description (ephone-hunt), on page 319
• description (voice hunt-group), on page 320
• description (voice moh-group), on page 321
• description (voice register pool), on page 322
• description (voice register pool-type), on page 323
• device-id (ephone-type), on page 324
• device-name, on page 326
• device-security-mode, on page 327
• device-type, on page 329
• dial-peer no-match isdn disconnect-cause, on page 331
• dialplan, on page 332
• dialplan-pattern, on page 334
• dialplan-pattern (call-manager-fallback), on page 338
• dialplan-pattern (voice register), on page 341
• digit collect kpml, on page 344
• direct-inward-dial isdn, on page 345
• directory, on page 347
• directory entry, on page 348
• display-logout, on page 350
• dnd (voice register pool), on page 351
• dnd feature-ring, on page 352
• dnd-control (voice register template), on page 354
• dn-webedit, on page 355
• dst (voice register global), on page 356
• dst auto-adjust (voice register global), on page 358
• dtmf-relay (voice register pool), on page 359
To set the date display format on the Cisco IP phones in a Cisco CallManager Express (Cisco CME) system, use the dateformat command in telephony-service configuration mode. To display the date in the default format, use the no form of this command.

```
date-format {dd-mm-yy|mm-dd-yy|yy-dd-mm|yy-mm-dd}
no date-format
```

### Syntax Description

<table>
<thead>
<tr>
<th>dd-mm-yy</th>
<th>mm-dd-yy</th>
<th>yy-dd-mm</th>
<th>yy-mm-dd</th>
</tr>
</thead>
<tbody>
<tr>
<td>Format in which dates are displayed on the IP phone:</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>• dd—Two-digit day.</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>• mm—Two-digit month.</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>• yy—Two-digit year.</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Command Default

Default is `mm-dd-yy`.

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

### Examples

The following example sets the date format to day, month, and year, so that December 17, 2004 is represented as `17-12-04`.

```
Router(config)# telephony-service
Router(config-telephony)# date-format dd-mm-yy
```
date-format (voice register global)

To set the date display format on SIP phones directly connected in Cisco Unified CME, use the `date-format` command in voice register global configuration mode. To display the date in the default format, use the `no` form of this command.

```
date-format {dd-mm-yy|mm-dd-yy|yy-dd-mm|yy-mm-dd}
no date-format
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Format</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>d/m/y</td>
<td>Two-digit date of the month</td>
</tr>
<tr>
<td>m/d/y</td>
<td>Two-digit month</td>
</tr>
<tr>
<td>y/d/m</td>
<td>Two-digit year</td>
</tr>
<tr>
<td>yy/m-d</td>
<td>Four-digit year</td>
</tr>
</tbody>
</table>

**Command Default**

Date is displayed as m/d/y.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Examples**

The following example shows how to set the date format so that a date such as December 3, 2007 is represented as 2007-12-03. By using the default configuration, this same date appears as 12/03/07.

```
Router(config)# voice register global
Router(config-register-global)# date-format yy-m-d
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>dst auto-adjust (voice register global)</code></td>
<td>Enables automatic adjustment of daylight saving time on SIP phones.</td>
</tr>
<tr>
<td><code>time-format (voice register global)</code></td>
<td>Selects a 12-hour clock or a 24-hour clock for the time display format on SIP phones in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
debug callmonitor

To collect and display debugging traces for call monitor, use the **debug callmonitor** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

**debug callmonitor**  {**all**|**core**|**detail**|**errors**|**events**|**hwconf**|**info**|**xml**}

**no debug command**  {**all**|**core**|**detail**|**errors**|**events**|**hwconf**|**info**|**xml**}

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>all</td>
<td>All call-monitor debugging traces.</td>
</tr>
<tr>
<td>core</td>
<td>Core information debugging traces.</td>
</tr>
<tr>
<td>detail</td>
<td>Detailed debugging traces.</td>
</tr>
<tr>
<td>errors</td>
<td>Call-monitor error debugging traces.</td>
</tr>
<tr>
<td>events</td>
<td>Call-monitor event debugging traces.</td>
</tr>
<tr>
<td>hwconf</td>
<td>Debugging traces related to hardware configuration.</td>
</tr>
<tr>
<td>info</td>
<td>Call-monitor information debugging traces.</td>
</tr>
<tr>
<td>xml</td>
<td>Call-monitor XML encoding debugging traces.</td>
</tr>
</tbody>
</table>

### Command Default

There is no default for this command.

### Command Modes

Privileged EXEC

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XW2</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Examples

The following example is partial output from this command:

```
Router# debug callmonitor all
Syslog logging: enabled (11 messages dropped, 2 messages rate-limited, 0 flushes, 0 overruns, xml disabled, filtering disabled)
No Active Message Discriminator.
No Inactive Message Discriminator.
Console logging: disabled
Monitor logging: level debugging, 0 messages logged, xml disabled, filtering disabled
Buffer logging: level debugging, 444378 messages logged, xml disabled, filtering disabled
Logging Exception size (4096 bytes)
Count and timestamp logging messages: disabled
Persistent logging: disabled
Trap logging: level informational, 461 message lines logged
Log Buffer (1000000 bytes):
Jun 4 22:30:24.222: //CMM/INFO:
Jun 4 22:30:24.222: //CMM/INFO:
```

Cisco Unified Communications Manager Express Command Reference 237
Jun 4 22:30:24.222: //CMM/INFO: target_node 0
Jun 4 22:30:24.222: //CMM/INFO: CallEntry 709C3FB8
Jun 4 22:30:24.222: //CMM/INFO:increase_gcid_ref_count 99685 0
Jun 4 22:30:24.222: //CMM/INFO:count = 1
Jun 4 22:30:24.222: //CMM/INFO:insert_ssptrs_to_gcid for line_info 66AF3720 (dn 4016),
GCID 05591A85-122211DC-8645A1CA-4B604A7A
Jun 4 22:30:24.222: //CMM/INFO:target_node 0
Jun 4 22:30:24.222: //CMM/INFO:CallEntry 709C3FB8
Jun 4 22:30:24.222: //CMM/INFO:* cmm_crs_proc_tr_call_orig
Jun 4 22:30:24.222: //CMM/INFO:CallEntry 709C3FB8
Jun 4 22:30:24.222: //CMM/INFO: * cmm_crs_proc_tr_call_orig
Jun 4 22:30:24.222: //CMM/INFO: CallEntry 709C3FB8
Jun 4 22:30:24.222: //CMM/INFO: * cmm_crs_proc_tr_call_orig
Jun 4 22:30:24.222: //CMM/INFO:orig --> callID 99685, line_info 66AF3720,
call_inst 65AF384, gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun 4 22:30:24.222: //CMM/INFO: cmm_send_dialog_notify sub_info 0
Jun 4 22:30:24.222: //CMM/INFO: - Type 0
Jun 4 22:30:24.222: //CMM/INFO:originateFilter...
Jun 4 22:30:24.222: //CMM/INFO: cmm_send_dialog_notify sub_info 0
Jun 4 22:30:24.222: ss_ptr list :=
Jun 4 22:30:24.222: //CMM/INFO: - Type 0
00000000-00000000-00000000-00000000
[4016, , 4016]
Jun 4 22:30:24.222: //CMM/DETAIL:called number is not specified. [4016, , 4016]
Jun 4 22:30:24.222: //CMM/DETAIL:originateFilter[callid=99685, pdn=16, pchan=1] is not set, [4016, , 4016] is not filtered
Jun 4 22:30:25.670: //CMM/INFO:
Jun 4 22:30:25.670: //CMM/INFO:
Jun 4 22:30:25.670: //CMM/INFO: cmm_notify_trigger() 14, callID 99686, 8101, 1902058375, 0
.
.

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>callmonitor</td>
<td>Enable call monitoring messaging functionality on a SIP endpoint in a VoIP network.</td>
</tr>
<tr>
<td>gcid</td>
<td>Enable Global Call ID (Gcid) for every call on an outbound leg of a VoIP dial peer for a SIP endpoint.</td>
</tr>
</tbody>
</table>
**debug capf-server**

To collect debug information about the CAPF server, use the `debug capf-server` command in privileged EXEC mode. To disable collection of debug information, use the `no` form of this command.

```
default capf-server {all|error|events|messages}
no debug capf-server
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Command</th>
</tr>
</thead>
<tbody>
<tr>
<td>all</td>
<td>Collect all CAPF information available.</td>
</tr>
<tr>
<td>error</td>
<td>Collect only information about CAPF errors.</td>
</tr>
<tr>
<td>events</td>
<td>Collect only information about CAPF status events.</td>
</tr>
<tr>
<td>messages</td>
<td>Collect only CAPF system messages.</td>
</tr>
</tbody>
</table>

**Command Default**

Collection of CAPF debug information is disabled.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CallManager Express phone authentication.

**Examples**

The following example shows debug messages for the CAPF server.

```
Router# debug capf-server all
001891: Jul 21 18:17:07.014: %IPPHONE-6-UNREGISTER_NORMAL: ephone-1:SEP000E325C9A43
IP:10.10.10.194 So
cicket:3 DeviceType:Phone has unregistered normally.
001892: Jul 21 18:17:20.495: New Connection from phone, socket 1
001893: Jul 21 18:17:20.499: Created New Handshake Process
001894: Jul 21 18:17:20.499: SSL Handshake Error -6983
001896: Jul 21 18:17:22.555: SSL Handshake Successful
001897: Jul 21 18:17:22.555: ephone_capf_send_auth_req:
001898: Jul 21 18:17:22.555: ephone_capf_ssl_write: 12 bytes
001899: Jul 21 18:17:22.711: ephone_capf_ssl_read: Read 35 bytes
001900: Jul 21 18:17:22.711: ephone_capf_handle_phone_msg: msgtype 2
001902: Jul 21 18:17:22.711: ephone_capf_send_delete_cert_req_msg: SEP000E325C9A43
001903: Jul 21 18:17:22.711: ephone_capf_ssl_write: 8 bytes
001904: Jul 21 18:17:23.891: ephone_capf_ssl_read: Read 12 bytes
001905: Jul 21 18:17:23.891: ephone_capf_handle_phone_msg: msgtype 14
001906: Jul 21 18:17:23.891: certificate delete successful for SEP000E325C9A43
001907: Jul 21 18:17:24.695: ephone_capf_release_session: SEP000E325C9A43
001908: Jul 21 18:17:24.695: ephone_capf_send_end_session_msg: SEP000E325C9A43
001909: Jul 21 18:17:24.695: ephone_capf_ssl_write: 12 bytes
001910: Jul 21 18:17:25.095: %IPPHONE-6-REG_ALARM: 22: Name=SEP000E325C9A43 Load=7.2(2.0)
```
001911: Jul 21 18:17:25.099: %IPPHONE-6-REGISTER: ephone-1:SEP000E325C9A43 IP:10.10.10.194 Socket:2 DeviceType:Phone has registered.

001912: Jul 21 18:18:05.171: %IPPHONE-6-UNREGISTER_NORMAL: ephone-1:SEP000E325C9A43 IP:1.1.1.127 Socket:2 DeviceType:Phone has unregistered normally.

001913: Jul 21 18:18:18.288: New Connection from phone, socket 1

001914: Jul 21 18:18:18.288: Created New Handshake Process

001915: Jul 21 18:18:18.292: SSL Handshake Error -6983

001916: Jul 21 18:18:19.292: SSL Handshake Error -6983

001917: Jul 21 18:18:20.348: SSL Handshake Successful

001918: Jul 21 18:18:20.348: ephone_capf_send_auth_req:

001919: Jul 21 18:18:20.348: ephone_capf_ssl_write: 12 bytes^Z

001920: Jul 21 18:18:20.492: ephone_capf_ssl_read: Read 35 bytes


001923: Jul 21 18:18:20.492: ephone_capf_send_PhKeyGenReq_msg: SEP000E325C9A43 KeySize 1024

001924: Jul 21 18:18:20.492: ephone_capf_ssl_write: 13 bytes

001925: Jul 21 18:18:20.540: ephone_capf_ssl_read: Read 8 bytes

001926: Jul 21 18:18:20.540: ephone_capf_handle_phone_msg: msgtype 17

001927: Jul 21 18:18:20.540: ephone_capf_process_req_in_progress: SEP000E325C9A43 delay 0sh

001928: Jul 21 18:18:21.924: %SYS-5-CONFIG_I: Configured from console by user1 on console
**debug cch323 video**

To provide debugging output for video components within the H.323 subsystem, use the `debug cch323 video` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

```
debug cch323 video
no debug cch323 video
```

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to enable a debugging trace for the video component in an H.323 network.

**Examples**

**Originating Gateway Example**

The following is sample output of the debugging log for an originating Cisco Unified CallManager Express (Cisco Unified CME) gateway after the `debug cch323 video` command was enabled:

```
Router# show log
Syslog logging: enabled (11 messages dropped, 487 messages rate-limited, 0 flushes, 0 overruns, xml disabled, filtering disabled)  
  Console logging: disabled  
  Monitor logging: level debugging, 0 messages logged, xml disabled, filtering disabled  
  Buffer logging: level debugging, 1144 messages logged, xml disabled, filtering disabled  
  Logging Exception size (4096 bytes)  
  Count and timestamp logging messages: disabled  
  Trap logging: level informational, 1084 message lines logged  
  Log Buffer (6000000 bytes):  
  Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_get_peer_info: Entry  
  Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_get_peer_info: Have peer  
  Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_set_pref_codec_list: First preferred codec(bytes)=16(20)  
  Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_get_peer_info: Flow Mode set to FLOW_THROUGH  
  Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_set_caps_chn_info: No peer leg setup params  
  Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_set_h323_control_options_outgoing: h245 sm mode = 8463  
  Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_set_h323_control_options_outgoing: h323_ctl=0x20  
  Jun 13 09:19:42.010: //103030/C7838B198002/H323/cch323_rotary_validate: No peer_ccb available
```
Terminating Gateway Example

The following is sample output of the debugging log for a terminating Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) gateway after the **debug cch323 video** command was enabled:

```
Router# show log
Syslog logging: enabled (11 messages dropped, 466 messages rate-limited, 0 flushes, 0 overruns, xml disabled, filtering disabled)
Console logging: disabled
Monitor logging: level debugging, 829 messages logged, xml disabled, filtering disabled
Buffer logging: level debugging, 829 messages logged, xml disabled, filtering disabled
Logging Exception size (4096 bytes)
Count and timestamp logging messages: disabled
Trap logging: level informational, 771 message lines logged
Log Buffer (200000 bytes):
Jun 13 09:19:42.011: //103034/C7838B198002/H323/setup_ind: Receive bearer cap infoXRate 24, rateMult 12
Jun 13 09:19:42.011: //103034/C7838B198002/H323/cch323_set_h245_state_mc_mode_incoming: h245 state m/c mode=0x10F, h323_ctl=0x2F
Jun 13 09:19:42.015: //-1/xxxxxxxxxxxx/H323/cch245_event_handler: callID=103034
Jun 13 09:19:42.019: //-1/xxxxxxxxxxxx/H323/cch245_event_handler:Event CC_EV_H245_SET_MODE: data ptr=0x465D5760
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_set_mode: callID=103034, flow Mode=1 spi_mode=0x6
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_do_call_proceeding: set_mode NOT called yet...saved deferred CALL_PROC
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_h245_connection_sm: state=0, event=0, ccb=4461B518, listen state=0
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_process_set_mode: Setting inbound leg mode flags to 0x10F, flow mode to FLOW THROUGH
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_process_set_mode: Sending deferred CALL_PROC
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_do_call_proceeding: set_mode called so we can proceed with CALLPROC
Jun 13 09:19:42.027: //103034/C7838B198002/H323/cch323_h245_connection_sm: state=1, event=2, ccb=4461B518, listen state=1
Jun 13 09:19:42.027: //103034/C7838B198002/H323/cch323_send_cap_request: Setting mode to VIDEO MODE
Jun 13 09:19:42.031: //103034/C7838B198002/H323/cch323_h245_cap_ind: Masks au=0xC data=0x2 uinp=0x32
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>debug ephone video</code></td>
<td>Sets video debugging for the Cisco Unified IP phone.</td>
</tr>
<tr>
<td><code>show call active video</code></td>
<td>Displays call information for SCCP video calls in progress.</td>
</tr>
<tr>
<td><code>show call history video</code></td>
<td>Displays call history information for SCCP video calls.</td>
</tr>
<tr>
<td><code>show debugging</code></td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
</tbody>
</table>
debug credentials

To set debugging on the credentials service that runs between the Cisco Unified CME CTL provider and CTL client or between the Cisco Unified SRST router and Cisco Unified CallManager, use the **debug credentials** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

```
debug credentials
no debug credentials
```

### Syntax Description

This command has no arguments or keywords.

### Command Modes

Privileged EXEC

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>This command was introduced for Cisco Unified SRST.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>This command was introduced for Cisco Unified CME.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T for Cisco Unified CME.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

**Cisco Unified CME**

Use this command with Cisco Unified CME phone authentication to monitor a CTL provider as it provides credentials to the CTL client.

**Cisco Unified SRST**

Use this command to monitor Cisco Unified CallManager while it requests certificates from the Cisco Unified SRST router. It sets debugging on the credentials service that runs between the SRST router and Cisco Unified CallManager.

### Examples

**Cisco Unified CME**

The following sample output displays the CTL provider establishing a TLS session with the CTL client and providing all the relevant credentials for the services that are running on this router to the CTL client.

```
Router# debug credentials

Credentials server debugging is enabled
May 25 12:08:17.944: Credentials service: Start TLS Handshake 1 10.5.43.174 4374
May 25 12:08:17.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:18.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:19.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:20.964: Credentials service: TLS Handshake completes
```
Cisco Unified SRST

The following is sample output showing the credentials service that runs between the Cisco Unified SRST router and Cisco Unified CallManager. The credentials service provides Cisco Unified CallManager with the certificate from the SRST router.

Router# debug credentials
Credentials server debugging is enabled
Router#
May 25 12:08:17.944: Credentials service: Start TLS Handshake 1 10.5.43.174 4374
May 25 12:08:17.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:18.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:19.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:20.964: Credentials service: TLS Handshake completes

The below table describes the significant fields shown in the display.

**Table 1: debug credentials Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Start TLS Handshake 1 10.5.43.174 4374</td>
<td>Indicates the beginning of the TLS handshake between the secure Cisco Unified SRST router and Cisco Unified CallManager. In this example, 1 indicates the socket, 10.5.43.174 is the IP address, and 4374 is the port of Cisco Unified CallManager.</td>
</tr>
<tr>
<td>TLS Handshake returns OPSSLReadWouldBlockErr</td>
<td>Indicates that the handshake is in process.</td>
</tr>
<tr>
<td>TLS Handshake completes</td>
<td>Indicates that the TLS handshake has finished and that the Cisco Unified CallManager has received the secure Cisco Unified SRST device certificate.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>credentials</td>
<td>Enters credentials configuration mode to configure a Cisco Unified CME CTL provider certificate or a Cisco Unified SRST router certificate.</td>
</tr>
<tr>
<td>ctl-service admin</td>
<td>Specifies a user name and password to authenticate the CTL client during the CTL protocol.</td>
</tr>
<tr>
<td>ip source-address (credentials)</td>
<td>Enables the Cisco Unified CME or SRST router to receive messages through the specified IP address and port.</td>
</tr>
<tr>
<td>show credentials</td>
<td>Displays the credentials settings on a Cisco Unified CME or SRST router.</td>
</tr>
<tr>
<td>show debugging</td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
<tr>
<td>trustpoint (credentials)</td>
<td>Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with a Cisco Unified SRST router certificate.</td>
</tr>
</tbody>
</table>
debug cti

To enable debugging on the CTI interface in Cisco Unified CME, use the `debug cti` command in privileged EXEC mode. To disable debugging, use the `no` form of this command.

```
debug cti {all|callcontrol|core|dmgr|lm|protoif|session|xml}
no debug cti {all|callcontrol|core|dmgr|lm|protoif|session|xml}
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>all</td>
<td>All CTI debugging traces.</td>
</tr>
<tr>
<td>callcontrol</td>
<td>CTI call control debugging traces.</td>
</tr>
<tr>
<td>core</td>
<td>Basic call debugging traces.</td>
</tr>
<tr>
<td>dmgr</td>
<td>CTI device manager debugging traces.</td>
</tr>
<tr>
<td>lm</td>
<td>CTI line monitoring debugging traces.</td>
</tr>
<tr>
<td>protoif</td>
<td>CTI protocol interface debugging traces.</td>
</tr>
<tr>
<td>session</td>
<td>CTI session debugging traces.</td>
</tr>
<tr>
<td>xml</td>
<td>CTI xml debugging traces.</td>
</tr>
</tbody>
</table>

### Command Default

Debugging on the CTI interface is disabled.

### Command Modes

Privileged EXEC (#)

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command sets debugging for the CTI interface in Cisco Unified CME.

### Examples

The following partial output from the `debug cti core` command shows the events from the time a call is placed to when the connection is established:

```
Router# debug cti core
Core CTI debug flags are on
.
.
Jun 17 23:12:09.885: //CTI/PI:pi_parse_service event 0
.
```
Jun 17 23:12:09.889: //CTI/CC:Fsm_Idle_MakeCall calling 201, called 204
Jun 17 23:12:09.889: //CTI/DMGR:
Jun 17 23:12:09.889: MakeCall event sent to Device Manager.callID 47964, Mac:0019E83B211D, CallingNum:201, CalledNum:204
Jun 17 23:12:09.889: //CTI/DMGR:
Jun 17 23:12:09.889: MakeCall event sent to skinny server.Mac:0019E83B211D, CallingNum:201, CalledNum:204
Jun 17 23:12:09.893: //CTI/CM:-- trigger 1, callID 59291, dn 201, reason 0, result 0
Jun 17 23:12:09.893: //CTI/CM:  line_info 87674E4C, dn 201
Jun 17 23:12:09.893: //CTI/CM: * cmm_crs_proc_tr_call_orig
Jun 17 23:12:09.893: //CTI/CM:increase_gcid_ref_count 59291 0
Jun 17 23:12:09.893: //CTI/CM:create_gcidinfo_node 59291
Jun 17 23:12:09.893: === EVENT EV_ORIGINATED
Jun 17 23:12:09.893: 201 --> . cause normal
.
.
Jun 17 23:12:19.221: //CTI/SM:sm_handle_cc_service event 77
Jun 17 23:12:19.221: //CTI/SM: to return 87396444

UC520#

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show debugging</td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
</tbody>
</table>
debug ctl-client

To collect debug information about the CTL client, use the **debug ctl-client** command in privileged EXEC configuration mode. To disable collection of debug information, use the **no** form of this command.

```
debug ctl-client
no debug ctl-client
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Collection of CTL client debug information is disabled.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication.

**Examples**

The following example shows debug messages for the CTL client:

```
Router# debug ctl-client

```
debug ephone alarm

To set SkinnyStation alarm messages debugging for the Cisco IP phone, use the `debug ephone alarm` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

```
default ipaddress [mac-address mac-address]
no default ipaddress [mac-address mac-address]
```

**Syntax Description**

| mac-address | (Optional) Defines the MAC address of the Cisco IP phone. |
| mac-address | (Optional) Specifies the MAC address of the Cisco IP phone. |

**Command Default**

No default behavior or values

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series Integrated Access Devices (IADs).</td>
</tr>
<tr>
<td>12.2(8)T1</td>
<td>This command was implemented on the Cisco 3725 and Cisco 3745 routers.</td>
</tr>
<tr>
<td>12.2(8)T1</td>
<td>This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.</td>
</tr>
<tr>
<td>12.2(11)T</td>
<td>This command was implemented on the Cisco 1760 routers.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `debug ephone alarm` command shows all the SkinnyStation alarm messages sent by the Cisco IP phone. Under normal circumstances, this message is sent by the Cisco IP phone just before it registers, and the message has the severity level for the alarm set to “Informational” and contains the reason for the phone reboot or re-register. This type of message is entirely benign and does not indicate an error condition.

If the `mac-address` keyword is not used, the debug ephone alarm command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the `mac-address` keyword with the `no` form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the `show ephone` command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

**Examples**

The following example shows a SkinnyStation alarm message that is sent before the Cisco IP phone registers:

```
Router# debug ephone alarm
phone keypad reset
CM-closed-TCP
CM-bad-state
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug ephone detail</td>
<td>Sets detail debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone error</td>
<td>Sets error debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone keepalive</td>
<td>Sets keepalive debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone loopback</td>
<td>Sets MWI debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone pak</td>
<td>Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.</td>
</tr>
<tr>
<td>debug ephone raw</td>
<td>Provides raw low-level protocol debugging display for all SCCP messages.</td>
</tr>
<tr>
<td>debug ephone register</td>
<td>Sets registration debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone state</td>
<td>Sets state debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone statistics</td>
<td>Sets statistics debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>show debugging</td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
</tbody>
</table>
**debug ephone blf**

To display debugging information for Busy Lamp Field (BLF) presence features, use the `debug ephone blf` command in privileged EXEC mode. To disable debugging, use the `no` form of this command.

```
debug ephone blf [mac-address mac-address]
no debug ephone blf [mac-address mac-address]
```

**Syntax Description**

- `mac-address mac-address` (Optional) Specifies the MAC address of a specific IP phone.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>
| 12.4(15)T | This command was integrated into Cisco IOS Release 12.4(15)T.

**Usage Guidelines**

Use this command for troubleshooting BLF speed-dial and BLF call-list features for phones in a presence service.

**Examples**

The following is sample output from the `debug ephone blf` command.

```
Router# debug ephone blf
EPHONE BLF debugging is enabled
-Sep 4 07:18:26.307: skinny_asnl_callback: subID 16 type 4
-Sep 4 07:18:26.307: ASNL_RESP_NOTIFY_INDICATION
-Sep 4 07:18:26.307: ephone-1[1]:ASNL notify indication message, feature index 4, subID [16]
-Sep 4 07:18:26.307: ephone-1[1]:line status 6, subID [16]
-Sep 4 07:18:26.307: ephone-1[1]:StationFeatureStatV2Message sent, status 2
-Sep 4 07:18:26.307: skinny_asnl_callback: subID 23 type 4
-Sep 4 07:18:26.307: ASNL_RESP_NOTIFY_INDICATION
-Sep 4 07:18:26.311: ephone-2[2]:line status 6, subID [23]
-Sep 4 07:18:26.311: ephone-2[2]:StationFeatureStatV2Message sent, status 2
-Sep 4 07:18:28.951: skinny_asnl_callback: subID 16 type 4
-Sep 4 07:18:28.951: ASNL_RESP_NOTIFY_INDICATION
-Sep 4 07:18:28.951: ephone-1[1]:ASNL notify indication message, feature index 4, subID [16]
-Sep 4 07:18:28.951: ephone-1[1]:line status 1, subID [16]
-Sep 4 07:18:28.951: ephone-1[1]:StationFeatureStatV2Message sent, status 1
-Sep 4 07:18:28.951: skinny_asnl_callback: subID 23 type 4
-Sep 4 07:18:28.951: ASNL_RESP_NOTIFY_INDICATION
-Sep 4 07:18:28.951: ephone-2[2]:line status 1, subID [23]
-Sep 4 07:18:28.951: ephone-2[2]:StationFeatureStatV2Message sent, status 1
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>blf-speed-dial</td>
<td>Enables BLF monitoring for a speed-dial number on a phone registered to Cisco Unified CME.</td>
</tr>
<tr>
<td>presence call-list</td>
<td>Enables BLF monitoring for call lists and directories on phones registered to a Cisco Unified CME router.</td>
</tr>
<tr>
<td>show presence global</td>
<td>Displays configuration information about the presence service.</td>
</tr>
<tr>
<td>show presence subscription</td>
<td>Displays information about active presence subscriptions.</td>
</tr>
</tbody>
</table>
debug ephone ccm-compatible

To display Cisco CallManager notification updates for calls between Cisco CallManager and Cisco CallManager Express, use the `debug ephone ccm-compatible` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

```
debug ephone ccm-compatible [mac-address mac-address]
no debug ephone ccm-compatible [mac-address mac-address]
```

**Syntax Description**

- `mac-address mac-address` (Optional) Specifies the MAC address of a Cisco IP phone for debugging.

**Command Modes**

- Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(7)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command displays call flow notification information for all calls between Cisco CallManager and Cisco CallManager Express, but it is most useful for filtering out specific information for transfer and forward cases. For basic call information, use the `debug ephone state` command.

If you do not specify the `mac-address` keyword, the `debug ephone ccm-compatible` command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the `no` form of this command with the `mac-address` keyword.

Debugging can be enabled or disabled on any number of Cisco IP phones. Cisco IP phones that have debugging enabled are listed in the debug field of the `show ephone` command output. When debugging is enabled for a Cisco IP phone, debug output is displayed for all phone extensions (virtual voice ports) associated with that phone.

**Examples**

The following sample output displays call flow notifications between Cisco CallManager and Cisco CallManager Express:

```
Router# debug ephone ccm-compatible
*May 1 04:30:02.650:ephone-2[2]:DtAlertingTone/DtHoldTone - mediaActive reset during CONNECT
*May 1 04:30:02.654:ephone-2[2]:DtHoldTone - force media STOP state
*May 1 04:30:02.654://93/xxxxxxxxxxxx/CCAPI/ccCallNotify:(callID=0x5D,nData->
  bitmask=0x00000007)
  *May 1 04:30:02.654://93/xxxxxxxxxxxx/VTSP:(50/0/3):-1:0:5/vtsp_process_event: 
    vtsp:[50/0/3 (93), S_CONNECT, E_CC_SERVICE_MSG]
*May 1 04:30:02.654://93/xxxxxxxxxxxx/VTSP:(50/0/3):-1:0:5/act_service_msg_down:
*May 1 04:30:02.658:dn_callerid_update DN 3 number= 12009 name= CCM7960 in state CONNECTED
*May 1 04:30:02.658:dn_callerid_update (incoming) DN 3 info updated to
  calling= 12009 called= 13003 origCalled= 
  callingName= CCM7960, calledName= , redirectedTo =
*May 1 04:30:02.658:ephone-2[2][SEP003094C2999A]:refreshDisplayLine for line 1
  DN 3 chan 1
*May 1 04:30:03.318:ephone-2[2]:DisplayCallInfo incoming call
*May 1 04:30:03.318:ephone-2[2]:Call Info DN 3 line 1 ref 24 called 13003 calling 12009
  origCalled 13003 calltype 1
*May 1 04:30:03.318:ephone-2[2]:Original Called Name UUT4PH3
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>debug ephone state</code></td>
<td>Displays call state information.</td>
</tr>
<tr>
<td><code>show debugging</code></td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
<tr>
<td><code>show ephone</code></td>
<td>Displays information about registered Cisco IP phones.</td>
</tr>
</tbody>
</table>
debug ephone detail

To set detail debugging for the Cisco IP phone, use the `debug ephone detail` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

```
debug ephone detail [mac-address mac-address]
no debug ephone detail [mac-address mac-address]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>mac-address</td>
<td>(Optional) Defines the MAC address of the Cisco IP phone.</td>
</tr>
<tr>
<td>mac-address</td>
<td>(Optional) Specifies the MAC address of the Cisco IP phone.</td>
</tr>
</tbody>
</table>

**Command Default**

No default behavior or values

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YG</td>
<td>This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series Integrated Access Devices (IADs).</td>
</tr>
<tr>
<td>12.2(2)XT</td>
<td>This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.</td>
</tr>
<tr>
<td>12.2(8)T1</td>
<td>This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.</td>
</tr>
<tr>
<td>12.2(11)T</td>
<td>This command was implemented on the Cisco 1760 routers.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `debug ephone detail` command includes the error and state levels.

If the `mac-address` keyword is not used, the debug ephone detail command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the `mac-address` keyword with the `no` form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the `show ephone` command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

**Examples**

The following is sample output of detail debugging of the Cisco IP phone with MAC address 0030.94c3.8724. The sample is an excerpt of some of the activities that takes place during call setup, connected state, active call, and the call being disconnected.

```
Router# debug ephone detail mac-address 0030.94c3.8724
Ephone detail debugging is enabled
1d04h: ephone-1[1]:OFFHOOK
.
1d04h: Skinny Call State change for DN 1 SIEZE
```
1d04h: SkinnyGetCallState for DN 1 CONNECTED

1d04h: ephone-1[1]:SetCallState line 3 DN 2 TsOnHook

1d04h: ephone-1[1]:SetLineLamp 3 to OFF

1d04h: ephone-1[1]:SetCallState line 1 DN 1 TsOnHook

1d04h: ephone-1[1]:Clean Up Speakerphone state
1d04h: ephone-1[1]:SpeakerPhoneOnHook
1d04h: ephone-1[1]:Clean up activeline 1
1d04h: ephone-1[1]:StopTone sent to ephone
1d04h: ephone-1[1]:Clean Up phone ofhook state
1d04h: SkinnyGetCallState for DN 1 IDLE
1d04h: called DN -1, calling DN -1 phone -1
1d04h: ephone-1[1]:SetLineLamp 1 to OFF
1d04h: UnBinding ephone-1 from DN 1
1d04h: UnBinding called DN 2 from DN 1
1d04h: ephone-1[1]:ONHOOK
1d04h: ephone-1[1]:SpeakerPhoneOnHook
1d04h: ephone-1[1]:ONHOOK NO activeline

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug ephone alarm</td>
<td>Sets SkinnyStation alarm messages debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone error</td>
<td>Sets error debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone keepalive</td>
<td>Sets keepalive debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone loopback</td>
<td>Sets MWI debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone pak</td>
<td>Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.</td>
</tr>
<tr>
<td>debug ephone raw</td>
<td>Provides raw low-level protocol debugging display for all SCCP messages.</td>
</tr>
<tr>
<td>debug ephone register</td>
<td>Sets registration debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone state</td>
<td>Sets state debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone statistics</td>
<td>Sets statistics debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>show debugging</td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
</tbody>
</table>
**debug ephone error**

To set error debugging for the Cisco IP phone, use the `debug ephone error` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

```
default ephone error [mac-address mac-address]
no debug ephone error [mac-address mac-address]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>mac-address</code></td>
<td>(Optional) Defines the MAC address of the Cisco IP phone.</td>
</tr>
<tr>
<td><code>mac-address</code></td>
<td>(Optional) Specifies the MAC address of the Cisco IP phone.</td>
</tr>
</tbody>
</table>

**Command Default**

No default behavior or values

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series Integrated Access Devices (IADs).</td>
</tr>
<tr>
<td>12.2(2)XT</td>
<td>This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.</td>
</tr>
<tr>
<td>12.2(8)T1</td>
<td>This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.</td>
</tr>
<tr>
<td>12.2(11)T</td>
<td>This command was implemented on the Cisco 1760 routers.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `debug ephone error` command cancels debugging at the detail and state level.

If the `mac-address` keyword is not used, the debug ephone error command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the `mac-address` keyword with the `no` form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the `show ephone` command and look at the `debug` field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

**Examples**

The following is sample output of error debugging for the Cisco IP phone with MAC address 0030.94c3.8724:

```
Router# debug ephone error mac-address 0030.94c3.8724
EPHONE error debugging is enabled
socket [2] send ERROR 11
Skinny Socket [2] retry failure
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug ephone alarm</td>
<td>Sets SkinnyStation alarm messages debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone detail</td>
<td>Sets detail debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone keepalive</td>
<td>Sets keepalive debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone loopback</td>
<td>Sets MWI debugging for the Cisco IP phone.</td>
</tr>
<tr>
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<td>Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.</td>
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<tr>
<td>debug ephone raw</td>
<td>Provides raw low-level protocol debugging display for all SCCP messages.</td>
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<td>debug ephone register</td>
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</tr>
<tr>
<td>debug ephone state</td>
<td>Sets state debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone statistics</td>
<td>Sets statistics debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>show debugging</td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
</tbody>
</table>
**debug ephone extension-assigner**

To display status messages produced by the extension assigner application, use the `debug ephone extension-assigner` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

```
debug ephone extension-assigner
no debug ephone extension-assigner
```

### Syntax Description

This command has no arguments or keywords.

### Command Default

Debug ephone extension-assigner is disabled.

### Command Modes

Privileged EXEC

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC4</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command displays status messages produced by the extension assigner application, including messages related to the functions performed by the following Tcl commands:

- **phone query**—Verifies whether the ephone tag has been assigned a MAC address.
- **phone assign**—Binds the MAC address from the caller’s phone to a preexisting ephone template.
- **phone unassign**—Removes the MAC address from the ephone tag.

Before using this command, you must load the Tcl script for the extension assigner application.

### Examples

The following is sample output of extension assigner debugging as the extension assigner application queries phones for their status and issues commands to assign or unassign extension numbers.

```
*Jun 9 19:08:10.627: ephone_query: inCallID=47, tag=4, ephone_tag=4
*Jun 9 19:08:10.627: ephone_query: inCallID=47, tag=4, ephone_tag=4
*Jun 9 19:08:10.627: extAssigner_IsEphoneMacPreset: ephone_ptr->mac_addr_str = 000B46BDE075, MAC_EXT_RESERVED_VALUE = 02EAEAEA0000
*Jun 9 19:08:10.627: SkinnyGetActivePhoneIndexFromCallid: callID = 47
*Jun 9 19:08:10.627: SkinnyGetActivePhoneIndexFromCallid: vdbPtr->physical_interface_type (26); CV_VOICE_EFXS (26)
*Jun 9 19:08:10.627: SkinnyGetActivePhoneIndexFromCallid: vdbPtr->type (6); CC_IF_TELEPHONY (6)
*Jun 9 19:08:10.627: SkinnyGetActivePhoneIndexFromCallid: http->sig_type (26); CV_VOICE_EFXS (26)
*Jun 9 19:08:10.627: SkinnyGetActivePhoneIndexFromCallid: dn = 4, chan = 1
*Jun 9 19:08:10.627: ephone_query: EXTASSIGNER_RC_SLOT_ASSIGNED_TO_CALLING_PHONE
*Jun 9 19:08:22.763: ephone_unassign: inCallID=47, tag=4, ephone_tag=4
```
debug ephone state
Sets state debugging for Cisco IP phones.

debug voip application script
Displays status messages produced by voice over IP application scripts.
### debug ephone hfs

To collect and display debugging information on the download of IP phone configuration and firmware files using the HTTP File-Fetch Server (HFS) service in a Cisco Unified CME system, use the `debug ephone hfs` command in privileged EXEC mode. To disable collection of debug information, use the `no` form of this command.

[no] debug ephone hfs

#### Syntax Description

This command has no arguments or keywords.

#### Command Default

There are no debug logs on the console or buffer.

#### Command Modes

Privileged EXEC (#)

#### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(1)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

#### Usage Guidelines

Use the `debug ephone hfs` command to troubleshoot an attempt to download Cisco Unified SIP IP phone configuration and firmware files using the HFS service.

#### Examples

The following sample display shows a successful file fetch:

```
Router# debug ephone hfs
Jan 5 01:29:00.829: ephone_hfs_util_urlhook:URL Context --->
  svr_port=6970
  rem_port=63881
  is_ssl=0
  req_method=1
  url=/softkeyDefault.xml
Jan 5 01:29:00.833: ephone_hfs_util_urlhook:Found the binding, fn[softkeyDefault.xml], path[/system:/ephone/sipphone/softkeyDefault.xml]
Jan 5 01:29:00.833: ephone_hfs_util_get_action:Get HTTP-url[/softkeyDefault.xml], fetch_path[/system:/ephone/sipphone/softkeyDefault.xml], fetch_from_home[0]
Jan 5 01:29:00.853: HFS SUCCESS !!! fn=/system:/ephone/sipphone/softkeyDefault.xml size=4376 upload-time(s.ms)=0.016
```

The following sample display shows an unsuccessful file fetch, where the file is not found:

```
Router# debug ephone hfs
  svr_port=6970
  rem_port=63890
  is_ssl=0
  req_method=1
  url=/softkeyDefault2.xml
Jan 5 01:43:16.561: ephone_hfs_util_urlhook:File not found
```

The table describes the significant fields shown in the display.
### Table 2: debug ephone hfs Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>svr_port</td>
<td>Cisco Unified CME port where the request is sent by the remote Cisco Unified SIP IP phone.</td>
</tr>
<tr>
<td>rem_port</td>
<td>Remote port of the Cisco Unified SIP IP phone. The request originates from this port.</td>
</tr>
<tr>
<td>is_ssl</td>
<td>Indicates if a secure HTTP connection is established using the Secure Sockets Layer (SSL) method.</td>
</tr>
<tr>
<td>req_method</td>
<td>Indicates the type of HTTP request message. A value of 1 is equivalent to HTTP-GET while a value of 2 is equivalent to HTTP-POST.</td>
</tr>
<tr>
<td>url</td>
<td>Location of the file to be downloaded.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>hfs enable</td>
<td>Enables the HFS download service on an IP Phone in a Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
**debug ephone keepalive**

To set keepalive debugging for the Cisco IP phone, use the **debug ephone keepalive** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

```plaintext
debug ephone keepalive [mac-address mac-address]
no debug ephone keepalive [mac-address mac-address]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>mac-address</strong></td>
<td>(Optional) Defines the MAC address of the Cisco IP phone.</td>
</tr>
<tr>
<td><strong>mac-address</strong></td>
<td>(Optional) Specifies the MAC address of the Cisco IP phone.</td>
</tr>
</tbody>
</table>

**Command Default**

No default behavior or values

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series Integrated Access Devices (IADs).</td>
</tr>
<tr>
<td>12.2(2)XT</td>
<td>This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>This command was implemented on the Cisco 3725 and Cisco 3745 routers.</td>
</tr>
<tr>
<td>12.2(8)T1</td>
<td>This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.</td>
</tr>
<tr>
<td>12.2(11)T</td>
<td>This command was implemented on the Cisco 1760 routers.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The **debug ephone keepalive** command sets keepalive debugging.

If the **mac-address** keyword is not used, the debug ephone keepalive command debugging all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

**Examples**

The following is sample output of the keepalive status for the Cisco IP phone with MAC address 0030.94C3.E1A8:

```plaintext
Router# debug ephone keepalive mac-address 0030.94C3.E1A8
EPHONE keepalive debugging is enabled for phone 0030.94C3.E1A8
1d05h: ephone-1 Set interface FastEthernet0/0 ETHERNET
1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: ephone-1 Set interface FastEthernet0/0 ETHERNET
1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: Skinny Checking for stale sockets
1d05h: ephone-1 Set interface FastEthernet0/0 ETHERNET
```
debug ephone keepalive

1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: ephone-1 Set interface FastEthernet0/0 ETHERNET
1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: Skinny active socket list (3/96): 1 2 4

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug ephone alarm</td>
<td>Sets SkinnyStation alarm messages debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone detail</td>
<td>Sets detail debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone error</td>
<td>Sets error debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone loopback</td>
<td>Sets MWI debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone pak</td>
<td>Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.</td>
</tr>
<tr>
<td>debug ephone raw</td>
<td>Provides raw low-level protocol debugging display for all SCCP messages.</td>
</tr>
<tr>
<td>debug ephone register</td>
<td>Sets registration debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone state</td>
<td>Sets state debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone statistics</td>
<td>Sets statistics debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>show debugging</td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
</tbody>
</table>
debug ephone loopback

To set debugging for loopback calls, use the `debug ephone loopback` command in privileged EXEC mode. To disable debugging, use the `no` form of this command.

`debug ephone loopback [mac-address mac-address]`
`no debug ephone loopback [mac-address mac-address]`

**Syntax Description**

| mac-address mac-address | (Optional) Specifies the MAC address of a Cisco IP phone for debugging. |

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>This command was introduced for Cisco IOS Telephony Services (now known as Cisco CallManager Express) Version 2.0 on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.</td>
</tr>
<tr>
<td>12.2(8)T1</td>
<td>This command was implemented on the Cisco 2600-XM and Cisco 2691.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `debug ephone loopback` command sets debugging for incoming and outgoing calls on all loopback-dn pairs or on the single loopback-dn pair that is associated with the IP phone that has the MAC address specified in this command.

If you enable the `debug ephone loopback` command and the `debug ephone pak` command at the same time, the output displays packet debug output for the voice packets that are passing through the loopback-dn pair.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the `show ephone` command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with that Cisco IP phone.

**Examples**

The following example contains two excerpts of output for a call that is routed through a loopback. The first excerpt is output from the `show running-config` command and displays the loopback configuration used for this example. The second excerpt is output from the `debug ephone loopback` command.

```
Router# show running-config
.
.
ephone-dn 14
  number 1514
  !
  ephone-dn 42
```
A loopback call is started. An incoming call to 1911514 (ephone-dn 43) uses the loopback pair of ephone-dns to become an outgoing call to extension 1514. The number in the outgoing call has only four digits because the `loopback-dn` command specifies forwarding of four digits. The outgoing call uses ephone-dn 42, which is also specified in the `loopback-dn` command under ephone-dn 43. When the extension at 1514 rings, the following debug output is displayed:

Router# debug ephone loopback
Mar 7 00:57:25.376:Pass processed call info to special DN 43 chan 1
Mar 7 00:57:25.376:SkinnySetCallInfoLoopback DN 43 state IDLE to DN 42 state IDLE
Mar 7 00:57:25.376:Called Number = 1911514 Called Name =
Mar 7 00:57:25.376:Calling Number = 8101 Calling Name =
Copy Caller-ID info from Loopback DN 43 to DN 42
Mar 7 00:57:25.376:DN 43 Forward 1514
Mar 7 00:57:25.376:PredictTarget match 1514 DN 14 is idle
Mar 7 00:57:25.380:SkinnyUpdateLoopbackState DN 43 state RINGING calledDn -1
Mar 7 00:57:25.380:Loopback DN 42 state IDLE
Mar 7 00:57:25.380:Loopback DN 43 calledDN -1 callingDN -1 G711Ulaw64k
Mar 7 00:57:25.380:SkinnyUpdateLoopbackState DN 43 to DN 42 signal OFFHOOK
Mar 7 00:57:25.380:SetDnCodec Loopback DN 43 to DN 42 signal OFFHOOK
codec 4:G711Ulaw64k vad 0 size 160
Mar 7 00:57:25.380:SkinnyDnToneLoopback DN 42 state ALERTING to DN 43 state RINGING
orig Called Number -
Copy Caller-ID info from Loopback DN 43 to DN 42
Mar 7 00:57:25.380:DN 43 Forward 1514
orig Calling Number -
orig Called Number -
Copy Caller-ID info from Loopback DN 43 to DN 42
Mar 7 00:57:25.380:DN 43 Forward 1514
When extension 1514 answers the call, the following debug output is displayed:

Mar 7 00:57:32.158:SkinnyDnToneLoopback DN 42 state ALERTING to DN 43 state RINGING
When the called extension, 1514, goes back on-hook, the following debug output is displayed:

```
```

The below table describes the significant fields shown in the display.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Called Number</td>
<td>Original called number as presented to the incoming side of the loopback-dn.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Forward</td>
<td>Outgoing number that is expected to be dialed by the outgoing side of the loopback-dn pair.</td>
</tr>
<tr>
<td>PredictTarget Match</td>
<td>Extension (ephone-dn) that is anticipated by the loopback-dn to be the far-end termination for the call.</td>
</tr>
<tr>
<td>signal OFFHOOK</td>
<td>Indicates that the outgoing side of the loopback-dn pair is going off-hook prior to placing the outbound call leg.</td>
</tr>
<tr>
<td>Dial</td>
<td>Outbound side of the loopback-dn that is actually dialing the outbound call leg.</td>
</tr>
<tr>
<td>deferred alerting</td>
<td>Indicates that the alerting, or ringing, tone is returning to the original inbound call leg in response to the far-end ephone-dn state.</td>
</tr>
<tr>
<td>DN chain</td>
<td>Chain of ephone-dns that has been detected, starting from the far-end that terminates the call. Each entry in the chain indicates an ephone-dn tag and channel number. Entries appear in the following order, from left to right:</td>
</tr>
<tr>
<td></td>
<td>- Ephone-dn tag and channel of the far-end call terminator (in this example, ephone-dn 14 is extension 1514).</td>
</tr>
<tr>
<td></td>
<td>- other—Ephone-dn tag of the outgoing side of the loopback.</td>
</tr>
<tr>
<td></td>
<td>- lb—Ephone-dn tag of the incoming side of the loopback.</td>
</tr>
<tr>
<td></td>
<td>- far—Ephone-dn tag and channel of the far-end call originator, or -1 for a nonlocal number.</td>
</tr>
<tr>
<td></td>
<td>- final—Ephone-dn tag for the originator of the call on the incoming side of the loopback. If the originator is not a local ephone-dn, this is set to -1. This number represents the final ephone-dn tag in the chain, looking toward the originator.</td>
</tr>
<tr>
<td>codec match</td>
<td>Indicates that there is no codec conflict between the two calls on either side of the loopback-dn.</td>
</tr>
<tr>
<td>GetDnAddrInfo</td>
<td>IP address of the IP phone at the final destination extension (ephone-dn), after resolving the chain of ephone-dns involved.</td>
</tr>
<tr>
<td>disc_reason</td>
<td>Disconnect cause code, in decimal. These are normal CC_CAUSE code values that are also used in call control API debugging. Common cause codes include the following:</td>
</tr>
<tr>
<td></td>
<td>- 16—Normal disconnect.</td>
</tr>
<tr>
<td></td>
<td>- 17—User busy.</td>
</tr>
<tr>
<td></td>
<td>- 19—No answer.</td>
</tr>
<tr>
<td></td>
<td>- 28—Invalid number.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug ephone pak</td>
<td>Provides voice packet level debugging.</td>
</tr>
<tr>
<td>loopback-dn</td>
<td>Configures loopback-dn virtual loopback voice ports used to establish demarcation points for VoIP voice calls and supplementary services.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>show ephone</td>
<td>Displays information about registered Cisco IP phones.</td>
</tr>
<tr>
<td>show ephone-dn loopback</td>
<td>Displays information for ephone-dns that have been set up for loopback calls.</td>
</tr>
</tbody>
</table>
debug ephone lpcor

To display debugging information for calls using the logical partitioning class of restriction (LPCOR) feature, use the debug ephone lpcor command in privileged EXEC mode. To disable debugging, use the no form of this command.

debug ephone lpcor [mac-address mac-address]
no debug ephone lpcor [mac-address mac-address]

Syntax Description

| mac-address mac-address | (Optional) Specifies the MAC address of a specific IP phone. |

Command Modes

Privileged EXEC (#)

Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(XA)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(T)</td>
<td>This command was integrated into Cisco IOS Release 15.1(T).</td>
</tr>
</tbody>
</table>

Usage Guidelines

Use this command for troubleshooting LPCOR calls to phones in a Cisco Unified CME system.

If the mac-address keyword is not used, this command debugs all phones that are registered to the Cisco Unified CME router. You can disable debugging for specific phones by using the mac-address keyword with the no form of this command.

Examples

The following is sample output from the debug ephone lpcor command for a call between ephone-1 and ephone-2 that was blocked by LPCOR policy validation:

Router# debug ephone lpcor
*Jun 24 11:23:45.599: ephone-1[0/3][SEP003094C25F38]:ephone_get_lpcor_index: dir 0
*Jun 24 11:23:46.603: ephone-2[1/2][SEP0021A02DB62A]:ephone_get_lpcor_index: dir 1

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug voip application lpcor</td>
<td>Enables debugging of the LPCOR application system.</td>
</tr>
<tr>
<td>debug voip lpcor</td>
<td>Displays debugging information for the LPCOR feature.</td>
</tr>
<tr>
<td>lpcor incoming</td>
<td>Associates an incoming call with a LPCOR resource-group policy.</td>
</tr>
<tr>
<td>lpcor outgoing</td>
<td>Associates an outgoing call with a LPCOR resource-group policy.</td>
</tr>
<tr>
<td>show ephone</td>
<td>Displays information about phones registered to Cisco Unified CME.</td>
</tr>
<tr>
<td>show voice lpcor policy</td>
<td>Displays the LPCOR policy for the specified resource group.</td>
</tr>
</tbody>
</table>
debug ephone message

To enable message tracing between ephones, use the `debug ephone message` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

`debug ephone message [detail]`
`no debug ephone message`

**Syntax Description**

| detail | (Optional) Displays signaling connection control protocol (SCCP) messages sent and received between ephones in the Cisco Unified CallManager Express (Cisco Unified CME) system. |

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `debug ephone message` command enables message tracing between ephones.

The debug ephone command debugs all ephones associated with a Cisco Unified CME router.

You can enable or disable debugging on any number of ephones. To see the ephones that have debugging enabled, enter the `show ephone` command and look at the debug field in the output. When debugging is enabled for an ephone, the debug output is displayed for the directory numbers associated with the ephone.

**Examples**

The following is sample output for the `debug ephone message` command for ephones:

```
Router# debug ephone message
EPHONE skinny message debugging is enabled
*Jul 17 12:12:54.883: Received message from phone 7, SkinnyMessageID = StationKe epAliveMessageID
*Jul 17 12:12:54.883: Sending message to phone 7, SkinnyMessageID = StationKe epAliveAckMessageID
```

The following command disables ephone message debugging:

```
Router# no debug ephone message
EPHONE skinny message debugging is disabled
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug ephone alarm</td>
<td>Sets SkinnyStation alarm messages debugging for the ephone.</td>
</tr>
<tr>
<td>debug ephone detail</td>
<td>Sets detail debugging for the ephone.</td>
</tr>
<tr>
<td>debug ephone error</td>
<td>Sets error debugging for the ephone.</td>
</tr>
<tr>
<td>debug ephone mwi</td>
<td>Sets MWI debugging for the ephone.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>debug ephone pak</td>
<td>Provides voice packet level debugging and displays the contents of one voice packet in every 1024 voice packets.</td>
</tr>
<tr>
<td>debug ephone raw</td>
<td>Provides raw low-level protocol debugging display for all SCCP messages.</td>
</tr>
<tr>
<td>debug ephone register</td>
<td>Sets registration debugging for the ephone.</td>
</tr>
<tr>
<td>debug ephone state</td>
<td>Sets state debugging for the ephone.</td>
</tr>
<tr>
<td>debug ephone statistics</td>
<td>Sets statistics debugging for the ephone.</td>
</tr>
<tr>
<td>debug ephone video</td>
<td>Sets video debugging for the ephone.</td>
</tr>
<tr>
<td>show debugging</td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
<tr>
<td>show ephone</td>
<td>Displays information about ephones.</td>
</tr>
</tbody>
</table>
debug ephone mlpp

To display debugging information for Multilevel Precedence and Preemption (MLPP) calls to phones in a Cisco Unified CME system, use the `debug ephone mlpp` command in privileged EXEC mode. To disable debugging, use the `no` form of this command.

```
display debugging information for Multilevel Precedence and Preemption (MLPP) calls to phones in a Cisco Unified CME system, use the `debug ephone mlpp` command in privileged EXEC mode. To disable debugging, use the `no` form of this command.
```

**Syntax Description**

- `mac-address mac-address` (Optional) Specifies the MAC address of a specific IP phone.

**Command Modes**

- Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>
| 12.4(24)T   | This command was integrated into Cisco IOS Release 12.4(24)T.

**Usage Guidelines**

Use this command to troubleshoot calls that use the MLPP service.

**Examples**

The following is sample output from the `debug ephone mlpp` command. This example shows output for the following call scenario:

- Ephone 1 is connected to ephone 3 (nonMLPP call).
- Ephone 4 makes an MLPP call to ephone 3. Preemption tone is played to both ephone 1 and 3.
- Ephone 3 is disconnected after the preemption tone timeout and precedence ringing.
- Ephone 3 answers the MLPP call and is connected to ephone 4.

```
Router# debug ephone mlpp
Sep 5 14:23:00.499: ephone-4[3/3][SEP001AE2BC3EE7]:indication=1
Sep 5 14:23:00.499: ephone-4[3/3][SEP001AE2BC3EE7]:max precedence=0
Sep 5 14:23:02.299: ephone-4[3/3][SEP001AE2BC3EE7]:mlpp_ephone_display_update callID=294
Sep 5 14:23:02.299: ephone-4[3/3][SEP001AE2BC3EE7]:indication=1
Sep 5 14:23:02.299: ephone-4[3/3][SEP001AE2BC3EE7]:mlpp precedence=4, domain=0
Sep 5 14:23:02.303: ephone-3[2/1][SEP001B54BA0D64]:preemption=1
Sep 5 14:23:02.303: ephone-3[2/1][SEP001B54BA0D64]:preemption=1
Sep 5 14:23:02.303: mlpp_ephone_find_call: preempt_http->mlpp_preemt_cid=294
Sep 5 14:23:02.303: //294/K6B5C03A8141/VOIP-MLPP/voice_mlpp_get_preemptInfo: mlpp_ephone_find_call is successful
Sep 5 14:23:02.303: ephone-4[3/3][SEP001AE2BC3EE7]:indication=1
Sep 5 14:23:02.303: ephone-4[3/3][SEP001AE2BC3EE7]:mlpp precedence=4, domain=0
Sep 5 14:23:02.303: ephone-4[3/3][SEP001AE2BC3EE7]:indication=1
Sep 5 14:23:02.303: ephone-4[3/3][SEP001AE2BC3EE7]:mlpp precedence=4, domain=0
Sep 5 14:23:02.303: ephone-6[5/6][SEP0018187F49FD]:indication=1
Sep 5 14:23:02.303: ephone-6[5/6][SEP0018187F49FD]:mlpp precedence=4, domain=0
Sep 5 14:23:02.303: ephone-4[3/3][SEP001AE2BC3EE7]:indication=1
Sep 5 14:23:02.307: ephone-1[0/2][SEP0014A9818797]:indication=1
Sep 5 14:23:02.307: ephone-3[2/1][SEP001B54BA0D64]:indication=1
Sep 5 14:23:02.307: ephone-4[3/3][SEP001AE2BC3EE7]:indication=1
Sep 5 14:23:02.307: ephone-1[0/2][SEP0014A9818797]:indication=1
Sep 5 14:23:02.307: ephone-1[0/2][SEP0014A9818797]:DtPreemptionTone
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>debug voice mlpp</strong></td>
<td>Displays debugging information for MLPP service.</td>
</tr>
<tr>
<td><strong>mlpp indication</strong></td>
<td>Enables MLPP indication on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td><strong>mlpp max-precedence</strong></td>
<td>Sets the maximum precedence (priority) level that a phone user can specify when making an MLPP call.</td>
</tr>
<tr>
<td><strong>mlpp preemption</strong></td>
<td>Enables preemption capability on an SCCP phone or analog FXS port.</td>
</tr>
</tbody>
</table>
debug ephone moh

To set debugging for music on hold (MOH), use the `debug ephone moh` command in privileged EXEC mode. To disable debugging, use the `no` form of this command.

```
display debug ephone moh [mac-address mac-address]
no display debug ephone moh [mac-address mac-address]
```

**Syntax Description**
- `mac-address mac-address` (Optional) Specifies the MAC address of a Cisco IP phone for debugging.

**Command Modes**
- Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>This command was introduced for Cisco IOS Telephony Services (now known as Cisco CallManager Express) Version 2.0 and Cisco Survivable Remote Site Telephony (SRST) Version 2.0 on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.</td>
</tr>
<tr>
<td>12.2(8)T1</td>
<td>This command was implemented on the Cisco 2600-XM and Cisco 2691.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Always use the `no moh` command before modifying or replacing the MOH file in Flash memory.

When a configuration using the `multicast moh` command is used and the `debug ephone moh` command is enabled, if you delete or modify the MOH file in the router's Flash memory, the debug output can be excessive and can flood the console. The multicast MOH configuration should be removed before using the `no moh` command when the `debug ephone moh` command is enabled.

**Examples**

The following sample output shows MOH activity prior to the first MOH session. Note that if you enable multicast MOH, that counts as the first session.

```
Router# debug ephone moh
Mar 7 00:52:33.817:MOH AU file
Mar 7 00:52:33.817:skinny_open_moh_play set type to 3
Mar 7 00:52:33.825: 2E73 6E64 0000 0018 0007 3CCA 0000 0001
Mar 7 00:52:33.825: 0000 1F40 0000 0001 FFFF FFFF FFFF FFFF
Mar 7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar 7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar 7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar 7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar 7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar 7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar 7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar 7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar 7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar 7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar 7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar 7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
```

Cisco Unified Communications Manager Express Command Reference

276
The below table describes the significant fields shown in the display.

**Table 4: debug ephone moh Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>type</td>
<td>0—invalid 1—raw file 2—wave format file (.wav) 3—AU format (.au) 4—live feed</td>
</tr>
<tr>
<td>AU file processing Found .snd</td>
<td>A .snd header was located in the AU file.</td>
</tr>
<tr>
<td>AU file data start at, end at</td>
<td>Data start and end file offset within the MOH file, as indicated by the file header.</td>
</tr>
<tr>
<td>read file header type</td>
<td>File format found (AU, WAVE, or RAW).</td>
</tr>
<tr>
<td>pre-read block, write-offset</td>
<td>Location in the internal MOH buffer to which data is being written, and location from which that data was read in the file.</td>
</tr>
<tr>
<td>play-offset, write-offset</td>
<td>Indicates the relative positioning of MOH file read-ahead buffering. Data is normally written from a Flash file into the internal circular buffer, ahead of the location from which data is being played or output.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>moh (telephony-service)</td>
<td>Generates an audio stream from a file for MOH in a Cisco CME system.</td>
</tr>
<tr>
<td>multicast moh</td>
<td>Uses the MOH audio stream as a multicast source in a Cisco CME system.</td>
</tr>
</tbody>
</table>
debug ephone mwi

To set message waiting indication (MWI) debugging for the Cisco IOS Telephony Service router, use the `debug ephone mwi` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

```
display ephone mwi
no display ephone mwi
```

### Syntax Description
This command has no arguments or keywords.

### Command Default
No default behavior or values

### Command Modes
Privileged EXEC

### Command History
<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series Integrated Access Devices (IADs).</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>This command was implemented on the Cisco 3725 and Cisco 3745 routers.</td>
</tr>
<tr>
<td>12.2(8)T1</td>
<td>This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.</td>
</tr>
<tr>
<td>12.2(11)T</td>
<td>This command was implemented on the Cisco 1760 routers.</td>
</tr>
</tbody>
</table>

### Usage Guidelines
The `debug ephone mwi` command sets message waiting indication debugging for the Cisco IOS Telephony Service router. Because the MWI protocol activity is not specific to any individual Cisco IP phone, setting the MAC address keyword qualifier for this command is not useful.

**Note**
Unlike the other related `debug ephone` commands, the `mac-address` keyword does not help debug a particular Cisco IP phone.

### Examples
The following is sample output of the message waiting indication status for the Cisco IOS Telephony Service router:

```
Router# debug ephone mwi
```

### Related Commands
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>debug ephone alarm</code></td>
<td>Sets SkinnyStation alarm messages debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td><code>debug ephone detail</code></td>
<td>Sets detail debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td><code>debug ephone error</code></td>
<td>Sets error debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>debug ephone keepalive</td>
<td>Sets keepalive debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone pak</td>
<td>Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.</td>
</tr>
<tr>
<td>debug ephone raw</td>
<td>Provides raw low-level protocol debugging display for all SCCP messages.</td>
</tr>
<tr>
<td>debug ephone register</td>
<td>Sets registration debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone state</td>
<td>Sets state debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone statistics</td>
<td>Sets statistics debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>show debugging</td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
</tbody>
</table>
debug ephone paging

To collect debugging information on paging for both Cisco Unified SIP IP and Cisco Unified SCCP IP phones, use the `debug ephone paging` command in privileged EXEC mode. To disable debugging, use the `no` form of this command.

```
[no] debug ephone paging
```

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Examples**

The following example shows debug messages from the `debug ephone paging` command:

```
*Dec 7 21:53:42.519: Paging-dn 250 sccp count=1 sip count=2
*Dec 7 21:53:42.527: SkinnyBuildPagingList for DN 250
*Dec 7 21:53:42.527: SkinnySetPagingList added DN 251 to list for DN 250
*Dec 7 21:53:42.527: SkinnySetPagingList added DN 252 to list for DN 250
*Dec 7 21:53:42.527: Paging Group List: 251 252 0 0 0 0 0 0 0 0
*Dec 7 21:53:42.527: SkinnySetupPagingDnMulticast 239.1.1.0 20480 for DN 250
*Dec 7 21:53:42.527: Found paging DN 250 on ephone-2
*Dec 7 21:53:42.527: Added interface GigabitEthernet0/0 to multicast list for DN 250
*Dec 7 21:53:42.527: SkinnyStartPagingPhone 1 for DN 250 with multicast
*Dec 7 21:53:42.527: Found paging DN 250 on pool 1[40001] is_paging=FALSE
*Dec 7 21:53:42.527: SipPagingPhoneReq for pool 1[40001] with multicast start
*Dec 7 21:53:42.527: Found paging DN 250 on pool 2[40003] is_paging=FALSE
*Dec 7 21:53:42.527: SipPagingPhoneReq for pool 2[40003] with multicast start
*Dec 7 21:53:42.531: SkinnyBuildPagingList DN 250 for 1 targets
*Dec 7 21:53:42.531: SkinnyStartPagingMedia for 1 targets for DN 250
*Dec 7 21:53:57.471: SkinnyStopPagingPhone 1 for DN 250 with multicast
*Dec 7 21:53:57.471: SipPagingPhoneReq for pool 1[40001] with multicast stop
*Dec 7 21:53:57.471: SipPagingPhoneReq for pool 2[40003] with multicast stop
```

The following example shows another set of debug messages from the `debug ephone paging` command:

```
*Oct 27 22:39:32.543: Paging-dn 251 sccp count 1 sip count 1
*Oct 27 22:39:32.551: Added interface GigabitEthernet0/0 to multicast list for DN 251
*Oct 27 22:39:32.551: SkinnyStartPagingPhone for DN 251 with multicast
*Oct 27 22:39:32.551: SkinnyBuildPagingList DN 251 for 1 targets
*Oct 27 22:39:32.551: SkinnyStopPagingPhone for DN 251 with multicast
*Oct 27 22:39:38.331: SkinnyStopPagingMedia for 1 targets for DN 251
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>paging-dn</td>
<td>Creates a paging extension to receive audio pages on a Cisco Unified IP phone in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>paging-dn (voice register)</td>
<td>Registers a Cisco Unified SIP IP phone to an ephone-dn paging directory number.</td>
</tr>
</tbody>
</table>
debug ephone pak

To provide voice packet level debugging and to print the contents of one voice packet in every 1024 voice packets, use the `debug ephone pak` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

```
default ephone pak [mac-address mac-address]
no debug ephone pak [mac-address mac-address]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>mac-address</th>
<th>(Optional) Defines the MAC address of the Cisco IP phone.</th>
</tr>
</thead>
<tbody>
<tr>
<td>mac-address</td>
<td>(Optional) Specifies the MAC address of the Cisco IP phone.</td>
</tr>
</tbody>
</table>

**Command Default**

No default behavior or values

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series Integrated Access Devices (IADs).</td>
</tr>
<tr>
<td>12.2(2)XT</td>
<td>This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.</td>
</tr>
<tr>
<td>12.2(8)T1</td>
<td>This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.</td>
</tr>
<tr>
<td>12.2(11)T</td>
<td>This command was implemented on the Cisco 1760 routers.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `debug ephone pak` command provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.

If the `mac-address` keyword is not used, the debug ephone pak command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the `mac-address` keyword with the `no` form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the `show ephone` command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

**Examples**

The following is sample output of packet debugging for the Cisco IP phone with MAC address 0030.94c3.8724:

```
Router# debug ephone pak mac-address 0030.94c3.8724
EPHONE packet debugging is enabled for phone 0030.94c3.8724
01:29:14: ***ph_xmit_ephone DN 3 tx_pkts 5770 dest=10.2.1.1 orig len=32
pakcopy=0 discards 27 ip_enctype 0 0 last discard: unsupported payload type
```
Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug ephone alarm</td>
<td>Sets SkinnyStation alarm messages debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone detail</td>
<td>Sets detail debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone error</td>
<td>Sets error debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone keepalive</td>
<td>Sets keepalive debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone loopback</td>
<td>Sets MWI debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone raw</td>
<td>Provides raw low-level protocol debugging display for all SCCP messages.</td>
</tr>
<tr>
<td>debug ephone register</td>
<td>Sets registration debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone state</td>
<td>Sets state debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone statistics</td>
<td>Sets statistics debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>show debugging</td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
</tbody>
</table>
debug ephone qov

To display quality of voice (QOV) statistics for calls when preset limits are exceeded, use the `debug ephone qov` command in privileged EXEC mode. To disable debugging, use the `no` form of this command.

```
debeg ephone qov [mac-address mac-address]  
no debug ephone qov [mac-address mac-address]
```

**Syntax Description**

- `mac-address mac-address` (Optional) Specifies the MAC address of a Cisco IP phone for debugging.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ2</td>
<td>This command was introduced for Cisco CallManager Express 3.0 and Cisco Survivable Remote Site Telephony (SRST) Version 3.0.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Once enabled, the `debug ephone qov` command produces output only when the QOV statistics reported by phones exceed preset limits. Phones are polled every few seconds for QOV statistics on VoIP calls only, not on local PSTN calls. An output report is produced when limits are surpassed for either or both of the following:

- **Lost packets**—A report is triggered when two adjacent QOV samples show an increase of four or more lost packets between samples. The report is triggered by an increase of lost packets in a short period of time, not by the total number of lost packets.
- **Jitter and latency**—A report is triggered when either jitter or latency exceeds 100 milliseconds.

To receive a QOV report at the end of each call regardless of whether the QOV limits have been exceeded, enable the `debug ephone alarm` command in addition to the `debug ephone qov` command.

The `debug ephone statistics` command displays the raw statistics that are polled from phones and used to generate QOV reports.

**Examples**

The following sample output describes QOV statistics for a call on ephone 5:

```
Router# debug ephone qov
Mar 7 00:54:57.329:ephone-5[7]:QOV DN 14 chan 1 (1514) ref 4 called=1514 calling=8101
Mar 7 00:54:57.329:ephone-5[7][SEP000DBB0EF37D]:Lost 91 Jitter 0 Latency 0
Mar 7 00:54:57.329:ephone-5[7][SEP000DBB0EF37D]:previous Lost 0 Jitter 0 Latency 0
Mar 7 00:54:57.329:ephone-5[7][SEP000DBB0EF37D]:Router sent 1153 pkts, current phone got 1141 received by all (shared) phones 0
Mar 7 00:54:57.329:ephone-5[7]:worst jitter 0 worst latency 0
Mar 7 00:54:57.329:ephone-5[7]:Current phone sent 1233 packets
Mar 7 00:54:57.329:ephone-5[7]:Signal Level to phone 3408 (-15 dB) peak 3516 (-15 dB)
```

The below table describes the significant fields shown in the display.
### Table 5: debug ephone qov Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Lost</td>
<td>Number of lost packets reported by the IP phone.</td>
</tr>
<tr>
<td>Jitter, Latency</td>
<td>The most recent jitter and latency parameters reported by the IP phone.</td>
</tr>
<tr>
<td>previous Lost, Jitter, Latency</td>
<td>Values from the previous QOV statistics report that were used as the comparison points against which the current statistics triggered generation of the current report.</td>
</tr>
<tr>
<td>Router sent pkts</td>
<td>Number of packets sent by the router to the IP phone. This number is the total for the entire call, even if the call is moved from one phone to another during a call, which can happen with shared lines.</td>
</tr>
<tr>
<td>current phone got</td>
<td>Number of packets received by the phone currently terminating the call. This number is the total for the entire call, even if the call is moved from one phone to another during a call, which can happen with shared lines.</td>
</tr>
<tr>
<td>worst jitter, worst latency</td>
<td>Highest value reported by the phone during the call.</td>
</tr>
<tr>
<td>Current phone sent packets</td>
<td>Number of packets that the current phone claims it sent during the call.</td>
</tr>
<tr>
<td>Signal Level to phone</td>
<td>Signal level seen in G.711 voice packet data prior to the sending of the most recent voice packet to the phone. The first number is the raw sample value, converted from G.711 to 16-bit linear format and left-justified. The number in parentheses is the value in decibels (dB), assuming that 32,767 is about +3 dB.</td>
</tr>
</tbody>
</table>

**Note**: This value is meaningful only if the call uses a G.711 codec.

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug ephone alarm</td>
<td>Displays alarm messages for IP phones.</td>
</tr>
<tr>
<td>debug ephone statistics</td>
<td>Displays call statistics for IP phones.</td>
</tr>
</tbody>
</table>
debug ephone raw

To provide raw low-level protocol debugging display for all Skinny Client Control Protocol (SCCP) messages, use the `debug ephone raw` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

```
debug ephone raw [mac-address mac-address]
no debug ephone raw [mac-address mac-address]
```

**Syntax Description**

<table>
<thead>
<tr>
<th><code>mac-address</code></th>
<th>(Optional) Defines the MAC address of the Cisco IP phone.</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>mac-address</code></td>
<td>(Optional) Specifies the MAC address of the Cisco IP phone.</td>
</tr>
</tbody>
</table>

**Command Default**

No default behavior or values

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series Integrated Access Devices (IADs).</td>
</tr>
<tr>
<td>12.2(2)XT</td>
<td>This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>This command was implemented on the Cisco 3725 and Cisco 3745 routers.</td>
</tr>
<tr>
<td>12.2(8)T1</td>
<td>This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.</td>
</tr>
<tr>
<td>12.2(11)T</td>
<td>This command was implemented on the Cisco 1760 routers.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `debug ephone raw` command provides raw low-level protocol debug display for all SCCP messages. The debug display provides byte level display of Skinny TCP socket messages.

If the `mac-address` keyword is not used, the `debug ephone raw` command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the `mac-address` keyword with the `no` form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the `show ephone` command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

**Examples**

The following is sample output of raw protocol debugging for the Cisco IP phone with MAC address `0030.94c3.E1A8`:

```
Router# debug ephone raw mac-address 0030.94c3.E1A8
EPHONE raw protocol debugging is enabled for phone 0030.94C3.E1A8
1d05h: skinny socket received 4 bytes on socket [1]
  0 0 0 0
1d05h: 1d05h: SkinnyMessageID = 0
```
1d05h: skinny send 4 bytes 4 0 0 0 0 0 0 0 1 0 0
1d05h: socket [1] sent 12 bytes OK (incl hdr) for ephone-(1)
1d06h: skinny socket received 4 bytes on socket [1] 0 0 0 0
1d06h:
1d06h: SkinnyMessageID = 0
1d06h: skinny send 4 bytes 4 0 0 0 0 0 0 0 1 0 0
1d06h: socket [1] sent 12 bytes OK (incl hdr) for ephone-(1)

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug ephone alarm</td>
<td>Sets SkinnyStation alarm messages debugging for the Cisco IP phone.</td>
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</tr>
<tr>
<td>debug ephone register</td>
<td>Sets registration debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone state</td>
<td>Sets state debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone statistics</td>
<td>Sets statistics debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>show debugging</td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
</tbody>
</table>
**debug ephone register**

To set registration debugging for the Cisco IP phone, use the `debug ephone register` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

```
debug ephone register [mac-address mac-address]
no debug ephone register [mac-address mac-address]
```

**Syntax Description**

- `mac-address` (Optional) Defines the MAC address of the Cisco IP phone.
- `mac-address` (Optional) Specifies the MAC address of the Cisco IP phone.

**Command Default**

No default behavior or values

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
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<tr>
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<tr>
<td>12.2(8)T1</td>
<td>This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.</td>
</tr>
<tr>
<td>12.2(11)T</td>
<td>This command was implemented on the Cisco 1760 routers.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `debug ephone register` command sets registration debugging for the Cisco IP phones.

If the `mac-address` keyword is not used, the debug ephone register command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the `mac-address` keyword with the `no` form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the `show ephone` command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

**Examples**

The following is sample output of registration debugging for the Cisco IP phone with MAC address 0030.94c3.8724:

```
Router# debug ephone register mac-address 0030.94c3.8724
Ephone registration debugging is enabled
1d06h: New Skinny socket accepted [1] (2 active)
1d06h: sin_family 2, sin_port 50778, in_addr 10.1.0.21
1d06h: skinny_add_socket 1 10.1.0.21 50778
1d06h: ephone-(1)[1] StationRegisterMessage (2/3/12) from 10.1.0.21
1d06h: ephone-(1)[1] Register StationIdentifier DeviceName SEP003094C3E1A8
```
1d06h: ephone-(1)[1] StationIdentifier Instance 1  deviceType 7
1d06h: ephone-1[-1]:stationIpAddr 10.1.0.21
1d06h: ephone-1[-1]:maxStreams 0
1d06h: ephone-(1) Allow any Skinny Server IP address 10.1.0.6
.
.
1d06h: ephone-1[1]:RegisterAck sent to ephone 1: keepalive period 30
.

### Related Commands

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<td>debug ephone state</td>
<td>Sets state debugging for the Cisco IP phone.</td>
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<td>debug ephone statistics</td>
<td>Sets statistics debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>show debugging</td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
</tbody>
</table>
**debug ephone sccp-state**

To set debugging for the SCCP call state, use the `debug ephone sccp-state` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

```
debug ephone sccp-state [mac-address mac-address]
no debug ephone sccp-state [mac-address mac-address]
```

**Syntax Description**
- `mac-address mac-address` (Optional) Specifies the MAC address of a phone.

**Command Default**
Debugging is not enabled for SCCP state.

**Command Modes**
Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command is used with Cisco Unified CallManager Express (Cisco Unified CME).

This command outputs only the debug messages that correspond to SCCP messages sent to IP phones to indicate the SCCP phone call state, such as RingIn, OffHook, Connected, and OnHook. These debug messages are also included in the output for the `debug ephone detail` command among other information.

**Examples**

The following example sets SCCP state debugging for one Cisco Unified CME phone with the MAC address of 678B.AEF9.DAB5.

```
Router# debug ephone sccp-state mac-address 678B.AEF9.DAB5
EPHONE SCCP state message debugging is enabled
for ephones 000B.BEF9.DFB5
*Mar 8 06:38:45.863: %ISDN-6-CONNECT: Interface Serial2/0/0:22 is now connected to 4085254871 unknown
*Mar 8 06:38:50.487: ephone-2[13]:SetCallState line 4 DN 60(60) chan 1 ref 100 TsRingIn
*Mar 8 06:38:52.399: ephone-2[13]:SetCallState line 4 DN 60(-1) chan 1 ref 100 TsOffHook
*Mar 8 06:38:52.399: ephone-2[13]:SetCallState line 4 DN 60(-1) chan 1 ref 100 TsConnected

*Mar 8 06:38:58.415: %ISDN-6-CONNECT: Interface Serial2/0/0:22 is now connected to 4085254871 unknown
*Mar 8 06:38:59.963: ephone-2[13]:SetCallState line 4 DN 60(-1) chan 1 ref 100 TsOnHook
*Mar 8 06:38:59.975: %ISDN-6-DISCONNECT: Interface Serial2/0/0:22 disconnected from 4085254871, call lasted 7 seconds
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug ephone detail</td>
<td>Sets detail debugging for one or all Cisco Unified IP phones.</td>
</tr>
</tbody>
</table>
**debug ephone shared-line-mixed**

To display debugging information about mixed shared lines, use the `debug ephone shared-line-mixed` command in privileged EXEC mode. To disable debugging messages, use the `no` form of this command.

```plaintext
[no] debug ephone shared-line-mixed {all|errors|events|info}
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>all</td>
<td>Displays all mixed-line debugging messages.</td>
</tr>
<tr>
<td>errors</td>
<td>Displays mixed-line error messages.</td>
</tr>
<tr>
<td>events</td>
<td>Displays mixed-line event messages.</td>
</tr>
<tr>
<td>info</td>
<td>Displays general information about mixed shared lines.</td>
</tr>
</tbody>
</table>

### Command Modes

Privileged EXEC (#)

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use the `debug ephone shared-line-mixed` command to show the debugging messages for Cisco Unified SCCP IP phone users in the SCCP layer of a mixed shared line.

### Examples

The following is a sample output from the `debug ephone shared-line-mixed` command for an outgoing call:

```
Router# debug ephone shared-line-mixed
Mar 9 20:16:37.571: skinny_notify_shrl_state_change: shrl event 1 sccp_id 0 peer_tag 20014 callid 53 incoming 0
Mar 9 20:16:37.571: skinny_shrl_get_call_state: dn 14, chan 1 call state 0
Mar 9 20:16:37.571: skinny_shrl_reserve_idle_chan: reserve dn 14, chan 1
Mar 9 20:16:37.571: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 1
Mar 9 20:16:37.583: skinny_process_shrl_event: event type 1 callid 53 dn 14 chan 1
Mar 9 20:16:37.583: skinny_process_shrl_callproc: dn 14, chan 1, callid 53
Router#
Router#
Mar 9 20:16:45.151: skinny_notify_shrl_state_change: shrl event 2 sccp_id 112 peer_tag 20014 callid 53 incoming 0
Mar 9 20:16:45.151: skinny_shrl_get_call_state: dn 14, chan 1 call state 0
Mar 9 20:16:45.155: skinny_process_shrl_event: event type 2 callid 53 dn 14 chan 1
Mar 9 20:16:45.155: skinny_update_shrl_remote: incoming 0, remote_number 2509, remote_name 2509
Router#
Router#
Mar 9 20:16:57.775: skinny_notify_shrl_state_change: shrl event 3 sccp_id 112 peer_tag 20014 callid 53 incoming 0
Mar 9 20:16:57.779: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 3
Mar 9 20:16:57.779: skinny_process_shrl_event: event type 4 callid 53 dn 14 chan 1
```
The following is a sample output from the `debug ephone shared-line-mixed` command for an incoming call with hold and resume:

```
Router# debug ephone shared-line-mixed
Mar 9 20:17:19.143: skinny_notify_shrl_state_change: shrl event 2 sccp_id 112 peer_tag 20041 callid 57 incoming 1
Mar 9 20:17:19.143: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 2
Mar 9 20:17:19.147: skinny_process_shrl_event: event type 2 callid 57 dn 14 chan 1
Mar 9 20:17:19.147: skinny_update_shrl_remote: incoming 1, remote_number 2509, remote_name 2509
Mar 9 20:17:19.155: skinny_set_shrl_remote_connect: dn 14, chan 1
Mar 9 20:17:19.159: skinny_process_shrl_event: event type 3 callid 0 dn 14 chan 1
Router#
Mar 9 20:17:24.347: skinny_notify_shrl_state_change: shrl event 4 sccp_id 112 peer_tag 20041 callid 57 incoming 0
Mar 9 20:17:24.347: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 4
Mar 9 20:17:24.347: skinny_process_shrl_event: event type 5 callid 57 dn 14 chan 1
Mar 9 20:17:28.307: skinny_shrl_resume_non_active_line: ref 5 line 4
Mar 9 20:17:28.319: skinny_shrl_resume_non_active_line: fake redial to 2509
Mar 9 20:17:29.127: skinny_shrl_get_privacy: dn 14, chan 1 phone 2 privacy 0
Mar 9 20:17:29.135: skinny_notify_shrl_state_change: shrl event 3 sccp_id 112 peer_tag 20041 callid 57 incoming 0
Mar 9 20:17:29.135: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 3
Mar 9 20:17:29.135: skinny_shrl_set_resume_info: dn 14, chan 1
Mar 9 20:17:29.155: skinny_process_shrl_event: event type 4 callid 57 dn 14 chan 1
Router#
Mar 9 20:17:42.407: skinny_notify_shrl_hold_or_resume_request: dn 14, chan 1, hold 1
Mar 9 20:17:42.411: skinny_shrl_get_privacy: dn 14, chan 1 phone 2 privacy 0
Router#
Mar 9 20:17:46.979: skinny_notify_shrl_state_change: shrl event 1 sccp_id 112 peer_tag 20041 callid 64 incoming 0
Mar 9 20:17:46.979: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 1
Mar 9 20:17:46.983: skinny_shrl_get_privacy: dn 14, chan 1 phone 2 privacy 0
Mar 9 20:17:46.987: skinny_notify_shrl_state_change: shrl event 2 sccp_id 112 peer_tag 20041 callid 64 incoming 0
Mar 9 20:17:46.987: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 2
Mar 9 20:17:46.987: skinny_process_shrl_event: event type 1 callid 64 dn 14 chan 1
Mar 9 20:17:46.999: skinny_set_shrl_remote_connect: dn 14, chan 1
Mar 9 20:17:46.999: skinny_set_shrl_remote_connect: dn 14, chan 1
Mar 9 20:17:47.007: skinny_process_shrl_event: event type 3 callid 0 dn 14 chan 1
Mar 9 20:17:47.007: skinny_update_shrl_call_state: dn 14, chan 1, call state 13
Mar 9 20:17:47.007: skinny_process_shrl_event: event type 3 callid 0 dn 14 chan 1
Router#
Mar 9 20:17:53.795: skinny_notify_shrl_state_change: shrl event 3 sccp_id 112 peer_tag 20041 callid 64 incoming 0
Mar 9 20:17:53.795: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 3
Mar 9 20:17:53.795: skinny_process_shrl_event: event type 4 callid 64 dn 14 chan 1
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>shared-line</code></td>
<td>Creates a directory number to be shared by multiple Cisco Unified SIP IP phones.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>shared-line sip</td>
<td>Adds an ephone-dn as a member of a shared directory number in the database of the Shared-Line Service Module for a mixed shared line between Cisco Unified SIP IP phones and Cisco Unified SCCP IP phones.</td>
</tr>
<tr>
<td>show shared-line</td>
<td>Displays information about active calls using SIP shared lines.</td>
</tr>
</tbody>
</table>
debug ephone state

To set state debugging for the Cisco IP phone, use the `debug ephone state` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

```
diagnostic ephone state [mac-address mac-address]
no diagnostic ephone state [mac-address mac-address]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>mac-address</code></td>
<td>(Optional) Defines the MAC address of the Cisco IP phone.</td>
</tr>
<tr>
<td><code>mac-address</code></td>
<td>(Optional) Specifies the MAC address of the Cisco IP phone.</td>
</tr>
</tbody>
</table>

**Command Default**

No default behavior or values

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
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<tr>
<td>12.1(5)YD</td>
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<tr>
<td>12.2(11)T</td>
<td>This command was implemented on Cisco 1760 routers.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `debug ephone state` command sets state debugging for the Cisco IP phones.

If the `mac-address` keyword is not used, the debug ephone state command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the `mac-address` keyword with the `no` form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the `show ephone` command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

**Examples**

The following is sample output of state debugging for the Cisco IP phone with MAC address 0030.94c3.E1A8:

```
Router# debug ephone state mac-address 0030.94c3.E1A8
EPHONE state debugging is enabled for phone 0030.94c3.E1A8
1d06h: ephone-1[1]:OFFHOOK
1d06h: ephone-1[1]:SIEZE on activeline 0
1d06h: ephone-1[1]:SetCallState line 1 DN 1 TsOffHook
1d06h: ephone-1[1]:Skinny-to-Skinny call DN 1 to DN 2 instance 1
1d06h: ephone-1[1]:SetCallState line 1 DN 1 TsRingOut
```
1d06h: ephone-1[1]: Call Info DN 1 line 1 ref 158 called 5002 calling 5001
1d06h: ephone-1[1]: Jane calling
1d06h: ephone-1[1]: Jill
1d06h: ephone-1[1]: SetCallState line 3 DN 2 TsRingIn
1d06h: ephone-1[1]: Call Info DN 2 line 3 ref 159 called 5002 calling 5001
1d06h: ephone-1[1]: Jane calling
1d06h: ephone-1[1]: Jill
1d06h: ephone-1[1]: SetCallState line 3 DN 2 TsCallRemoteMultiline
1d06h: ephone-1[1]: SetCallState line 1 DN 1 TsConnected
1d06h: ephone-1[1]: OpenReceive DN 1 codec 4:G711Ulaw64k duration 10 ms bytes 80
1d06h: ephone-1[1]: OpenReceiveChannelAck 1.2.172.21 port=24010
1d06h: ephone-1[1]: StartMedia 1.2.172.22 port=24612
1d06h: DN 1 codec 4:G711Ulaw64k duration 10 ms bytes 80
1d06h: ephone-1[1]: CloseReceive
1d06h: ephone-1[1]: SetCallState line 3 DN 2 TsOnHook
1d06h: ephone-1[1]: SetCallState line 1 DN 1 TsOnHook
1d06h: ephone-1[1]: SpeakerPhoneOnHook
1d06h: ephone-1[1]: ONHOOK
1d06h: ephone-1[1]: SpeakerPhoneOnHook
1d06h: SkinnyReportDnState DN 1 ONHOOK

### Related Commands

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<td>Provides raw low-level protocol debugging display for all SCCP messages.</td>
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<tr>
<td>debug ephone</td>
<td>Sets statistics debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>show debugging</td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
</tbody>
</table>
debug ephone statistics

To set call statistics debugging for the Cisco IP phone, use the `debug ephone statistics` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

```
debug ephone statistics [mac-address mac-address]
no debug ephone statistics [mac-address mac-address]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>mac-address</th>
<th>(Optional) Defines the MAC address of the Cisco IP phone.</th>
</tr>
</thead>
<tbody>
<tr>
<td>mac-address</td>
<td>(Optional) Specifies the MAC address of the Cisco IP phone.</td>
</tr>
</tbody>
</table>

**Command Default**

No default behavior or values

**Command Modes**

Privileged EXEC

**Command History**

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<td>This command was implemented on the Cisco 1760 routers.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `debug ephone statistics` command provides a debug monitor display of the periodic messages from the Cisco IP phone to the router. These include transmit-and-receive packet counts and an estimate of drop packets. The call statistics can also be displayed for live calls using the `show ephone` command.

If the `mac-address` keyword is not used, the debug ephone statistics command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the `mac-address` keyword with the `no` form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the `show ephone` command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

**Examples**

The following is sample output of statistics debugging for the Cisco IP phone with MAC address 0030.94C3.E1A8:

```
Router# debug ephone statistics mac-address 0030.94C3.E1A8
EPHONE statistics debugging is enabled for phone 0030.94C3.E1A8
1d06h: Clear Call Stats for DN 1 call ref 162
1d06h: Clear Call Stats for DN 1 call ref 162
```
Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug ephone alarm</td>
<td>Sets SkinnyStation alarm messages debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone detail</td>
<td>Sets detail debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone error</td>
<td>Sets error debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone keepalive</td>
<td>Sets keepalive debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone loopback</td>
<td>Sets MWI debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone pak</td>
<td>Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.</td>
</tr>
<tr>
<td>debug ephone raw</td>
<td>Provides raw low-level protocol debugging display for all SCCP messages.</td>
</tr>
<tr>
<td>debug ephone register</td>
<td>Sets registration debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>debug ephone state</td>
<td>Sets state debugging for the Cisco IP phone.</td>
</tr>
<tr>
<td>show debugging</td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
</tbody>
</table>
debug ephone video

To set video debugging for ephones, use the `debug ephone video` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

```
debug ephone video
no debug ephone video
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Debugging is disabled for ephone video.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `debug ephone video` command sets ephone video traces, which provide information about different video states for the call, including video capabilities selection, start, and stop.

The debug ephone command debugs all ephones that are registered to the Cisco Unified CallManager Express (Cisco Unified CME) system.

You can enable or disable debugging on any number of ephones. To see the ephones that have debugging enabled, enter the `show ephone` command and look at the debug field in the output. When debugging is enabled for a ephone, the debug output is displayed for the directory numbers associated with the ephone.

**Examples**

The following is sample output for the `debug ephone video` command for ephones:

```
Router# debug ephone video
*Mar 13 16:10:02.703: SkinnyVideoCodecMatch_Caps2Caps: match capability: tx_idxcap = 4, tx_idxpref = 3, rx_idxcap = 0, rx_idxpref = 0, videoBitRate = 7040 tx_mpi = 1
*Mai 13 16:10:04.711: ephone-19[1][SEPFFFA00000019]:checkToOpenMultiMedia: dn=19, chan=1
*Mai 13 16:10:04.711: ephone-19[1]:skinnyDP[19].s2s = 0
*Mai 13 16:10:04.711: ephone-19[1]:s2s is not set - hence not video capable
*Mai 13 16:10:04.719: ephone-19[1][SEPFFFA00000019]:SkinnyStartMultiMediaTransmission: chan 1 dn 19
*Mai 13 16:10:04.723: ephone-19[1]:Accept OLC and open multimedia channel
*Mai 13 16:10:04.723: ephone-19[1][SEPFFFA00000019]:SkinnyOpenMultiMediaReceiveChannel: dn 19 chan 1
*Mai 13 16:10:04.967: ephone-19[1][SEPFFFA00000019]:fStationOpenReceiveChannelAckMessage: MEDIA_DN 19 MEDIA_CHAN 1
*Mai 13 16:10:04.967: ephone-19[1]:fStationOpenMultiMediaReceiveChannelAckMessage:
*Mai 13 16:10:04.967: ephone-19[1]:Other_dn == -1
sk7345-2#
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug ephone alarm</td>
<td>Sets SkinnyStation alarm messages debugging for the ephone.</td>
</tr>
<tr>
<td>debug ephone detail</td>
<td>Sets detail debugging for the ephone.</td>
</tr>
<tr>
<td>debug ephone error</td>
<td>Sets error debugging for the ephone.</td>
</tr>
<tr>
<td>debug ephone message</td>
<td>Sets message debugging for the ephone.</td>
</tr>
<tr>
<td>debug ephone mwi</td>
<td>Sets MWI debugging for the ephone.</td>
</tr>
<tr>
<td>debug ephone pak</td>
<td>Provides voice packet level debugging and displays the contents of one voice packet in every 1024 voice packets.</td>
</tr>
<tr>
<td>debug ephone raw</td>
<td>Provides raw low-level protocol debugging display for all SCCP messages.</td>
</tr>
<tr>
<td>debug ephone register</td>
<td>Sets registration debugging for the ephone.</td>
</tr>
<tr>
<td>debug ephone state</td>
<td>Sets state debugging for the ephone.</td>
</tr>
<tr>
<td>debug ephone statistics</td>
<td>Sets statistics debugging for the ephone.</td>
</tr>
<tr>
<td>show debugging</td>
<td>Displays information about the types of debugging that are enabled for your router.</td>
</tr>
<tr>
<td>show ephone</td>
<td>Displays information about registered ephones.</td>
</tr>
</tbody>
</table>
debug ephone vm-integration

To display pattern manipulation information used for integration with voice-mail applications, use the debug ephone vm-integration command in privileged EXEC mode. To disable debugging output, use the no form of this command.

```
debug ephone vm-integration [mac-address mac-address]
no debug ephone vm-integration [mac-address mac-address]
```

**Syntax Description**
- `mac-address mac-address` (Optional) Specifies the MAC address of a Cisco IP phone for debugging.

**Command Modes**
- Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(7)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command displays the voice-mail integration patterns that were created using the pattern commands in vm-integration configuration mode. The patterns are used to forward calls to a voice-mail number that is set with the voicemail command.

If you do not specify the `mac-address` keyword, the `debug ephone vm-integration` command debugs all Cisco IP phones that are registered to the router. To remove debugging for Cisco IP phones, enter the no form of this command with the `mac-address` keyword.

**Examples**

The following sample output shows information for the vm-integration tokens that have been defined:

```
Router# debug ephone vm-integration
*Jul 23 15:38:03.294:ephone-3[3]:StimulusMessage 15 (1) From ephone 2
*Jul 23 15:38:03.294:ephone-3[3]:Voicemail access number pattern check
*Jul 23 15:38:03.294:SkinnyGetCallState for DN 3 chan 1 IDLE
*Jul 23 15:38:03.294:called DN -1 chan 1, calling DN -1 chan 1 phone -1 s2s:0
*Jul 23 15:38:03.294:dn number for dn 3 is 19003
*Jul 23 15:38:03.294:Updated number for token 1 is 19003
*Jul 23 15:38:03.294:CDN number for dn 3 is
*Jul 23 15:38:03.294:Updated number for token 2 is
*Jul 23 15:38:03.294:Updated number for token 0 is
*Jul 23 15:38:03.294:Update is 219003*
*Jul 23 15:38:03.294:New Voicemail number is 19101219003*
```

The below table describes the significant fields shown in the display.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>token 0</td>
<td>First token that was defined in the pattern.</td>
</tr>
<tr>
<td>token 1</td>
<td>Second token that was defined in the pattern.</td>
</tr>
<tr>
<td>token 2</td>
<td>Third token that was defined in the pattern.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>pattern direct</td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when a user presses the Messages button on a phone.</td>
</tr>
<tr>
<td>pattern ext-to-ext busy</td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension reaches a busy extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td>pattern ext-to-ext no-answer</td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td>pattern trunk-to-ext busy</td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td>pattern trunk-to-ext no-answer</td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td>vm-integration</td>
<td>Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and analog voice-mail systems.</td>
</tr>
<tr>
<td>voicemail</td>
<td>Defines the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed.</td>
</tr>
</tbody>
</table>
debug ephone whisper-intercom

To display debugging messages for the Whisper Intercom feature, use the **debug ephone whisper-intercom** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

```
debug ephone whisper-intercom
no debug ephone whisper-intercom
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Debugging for Whisper Intercom is disabled.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command displays debugging information about the Whisper Intercom feature configured on a directory number of a SCCP phone.

**Examples**

The following example displays output from the **debug ephone whisper-intercom** command:

```
Router# debug ephone whisper-intercom
ephone-1[0] Mac:1111.C1C1.0001 TCP socket:[8] activeLine:0 whisperLine:2 REGISTERED in SCCP ver 12/12 max_streams=3
mediaActive:0 whisper_mediaActive:0 startMedia:1 offhook:1 ringing:0 reset:0 reset_sent:0 pacing 0 debug:0 caps:5
IP:10.6.2.185 9237 7970 keepalive 16 max_line 8
button 1: dn 1 number 2001 CH1 IDLE CH2 IDLE
button 2: dn 161 number 6001 auto dial 6002 CH1 WHISPER
Preferred Codec: g711ulaw
Active Call on DN 161 chan 1:6001 0.0.0.0 0 to 10.6.2.185 9280 via 10.6.2.185 G711Ulaw64k 160 bytes no vad
Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Jitter 0 Latency 0 callingDn -1 calledDn 162

mediaActive:0 whisper_mediaActive:1 startMedia:0 offhook:1 ringing:0 reset:0 reset_sent:0 pacing 0 debug:0 caps:5
IP:10.6.2.185 9240 7970 keepalive 16 max_line 8
button 1: dn 2 number 2002 CH1 IDLE CH2 IDLE
button 2: dn 162 number 6002 auto dial 6001 CH1 WHISPER
Preferred Codec: g711ulaw
Active Call on DN 162 chan 1:6002 10.6.2.185 9280 to 10.6.2.254 2000 via 10.6.2.185 G711Ulaw64k 160 bytes no vad
Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Jitter 0 Latency 0 callingDn 161 calledDn -1
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>show ephone-dn whisper</strong></td>
<td>Displays information about whisper intercom ephone-dns that have been created in Cisco Unified CME.</td>
</tr>
<tr>
<td><strong>whisper-intercom</strong></td>
<td>Enables the Whisper Intercom feature on a directory number.</td>
</tr>
</tbody>
</table>
debug mwi relay errors

To debug message waiting indication (MWI) relay errors, use the `debug mwi relay errors` command in privileged EXEC mode. To disable debugging output, use the `no` form of this command.

```
debug mwi relay errors
no debug mwi relay errors
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

No default behavior or values

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series Integrated Access Devices (IADs).</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.</td>
</tr>
<tr>
<td>12.2(8)T1</td>
<td>This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.</td>
</tr>
<tr>
<td>12.2(11)T</td>
<td>This command was implemented on the Cisco 1760 routers.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `debug mwi relay errors` command provides a debug monitor display of any error messages, when MWI Relay Server (Cisco IOS Telephony Server) is trying to do MWI Relay to extensions on remote Cisco IOS Telephony Service (ITS).

**Examples**

The following examples show errors when MWI Relay Server tries to do an MWI Relay to extension 7004, but location of 7004 is not known to the MWI Relay Server:

```
Router# debug mwi relay errors
mwi-relay error info debugging is on
01:46:48: MWI-APP: mwi_notify_status: No ClientID (7004) registered
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>debug ephone mwi</code></td>
<td>Sets MWI debugging for the Cisco IOS Telephony Service router.</td>
</tr>
<tr>
<td><code>debug mwi relay events</code></td>
<td>Sets MWI relay events debugging for the Cisco IOS Telephony Service router.</td>
</tr>
</tbody>
</table>
debug mwi relay events

To set message waiting indication (MWI) relay events debugging, use the **debug mwi relay events** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

```
debug mwi relay events
no debug mwi relay events
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

No default behavior or values

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series Integrated Access Devices (IADs).</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.</td>
</tr>
<tr>
<td>12.2(8)T1</td>
<td>This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.</td>
</tr>
<tr>
<td>12.2(11)T</td>
<td>This command was implemented on the Cisco 1760 routers.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The **debug mwi relay events** command provides a debug monitor display of events, when MWI Relay Server (Cisco IOS Telephony Server) is trying to do MWI Relay to extensions on remote Cisco IOS Telephony Services (ITS).

**Examples**

The following debugging messages are shown when the MWI Relay server tries to send MWI Information to remote client 7001 and the location of 7001 is known by the MWI Relay Server:

```
Router# debug mwi relay events
mwi-relay events info debugging is on
01:45:34: mwi_notify_status: Queued event for mwi_app_queue
01:45:34: MWI-APP: mwi_app_process_event:
01:45:34: MWI-APP: mwi_app_process_event: MWI Event for ClientID(7001)@(1.8.17.22)
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug ephone mwi</td>
<td>Sets MWI debugging for the Cisco IOS Telephony Service router.</td>
</tr>
<tr>
<td>debug mwi relay errors</td>
<td>Sets MWI relay errors debugging for the Cisco IOS Telephony Service router.</td>
</tr>
</tbody>
</table>
debug shared-line

To display debugging information about SIP shared lines, use the **debug shared-line** command in privileged EXEC mode. To disable debugging messages, use the **no** form of this command.

```
debug shared-line {all|errors|events|info}
no debug shared-line {all|errors|events|info}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>all</code></td>
<td>Displays all shared-line debugging messages.</td>
</tr>
<tr>
<td><code>errors</code></td>
<td>Displays shared-line error messages.</td>
</tr>
<tr>
<td><code>events</code></td>
<td>Displays shared-line event messages.</td>
</tr>
<tr>
<td><code>info</code></td>
<td>Displays general information about shared lines.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC (`#`)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following example shows output from the **debug shared-line all** command:

```
Router# debug shared-line all

```

Cisco Unified CME Commands: D

---

306

Cisco Unified Communications Manager Express Command Reference
Aug 21 21:57:01.689: %IPPHONE-6-REG_ALARM: 24: Name=SEP00141C48E126 Load=8.0(5.0)
Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_find_peer_by_ipaddr:Trying to match peer for member 20141@15.6.0.2
Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_find_peer_by_ipaddr:Matched member found: 20141@15.6.0.2
Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_find_peer_by_ipaddr:Trying to match peer for member 20141@15.6.0.2
Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_send_dialog_notify:Sending NOTIFY to remote user: 20143@15.10.0.1 privacy OFF
Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_send_dialog_notify:Dialog msg: dir: 1, orient: 2, local_tag: 2ed5b927-6ad6, remote_tag: 89DCF0-139B, local_uri: 20141@15.6.0.2, remote_uri: 20143@15.10.0.1
Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_process_connect:Updated callinfo for callid: 5401, member: '20141@15.6.0.2', peer-tag: 40002
Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_send_dialog_notify:Sending NOTIFY to remote user: 20141@15.6.0.2 about state 3 on incoming call from 20141@15.6.0.2 privacy OFF
Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_send_dialog_notify:Dialog msg: dir: 1, orient: 2, local_tag: 2ed5b927-6ad6, remote_tag: 89DCF0-139B, local_uri: 20141@15.6.0.2, remote_uri: 20143@15.10.0.1
Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_send_dialog_notify:Sending NOTIFY to remote user: 20143@15.10.0.1 privacy OFF
Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>shared-line</td>
<td>Creates a directory number to be shared by multiple SIP phones.</td>
</tr>
<tr>
<td>show shared-line</td>
<td>Displays information about active calls using SIP shared lines.</td>
</tr>
</tbody>
</table>
debug voice register errors

To display debug information on voice register module errors during registration in a Cisco Unified CallManager Express (Cisco Unified CME) or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) environment, use the `debug voice register errors` command in privileged EXEC mode. To disable debugging, use the `no` form of the command.

```yaml
debug voice register errors
no debug voice register errors
```

**Syntax Description**

This command has no arguments or keywords

**Command Default**

Disabled

**Command Modes**

Privileged EXEC mode

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>This command was introduced for Cisco SIP SRST 3.0</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T for Cisco SIP SRST 3.0.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>This command was added to Cisco Unified CME 3.4 and Cisco SIP SRST 3.4.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Registration errors include failure to match pools or any internal errors that happen during registration.

**Examples**

**Cisco Unified CME**

The following is sample output for this command for a registration request with authentication enabled:

```
*May 6 18:07:26.971: VOICE_REG_POOL: Register request for (4901) from (10.5.49.83)
*May 6 18:07:26.971: VOICE_REG_POOL: key(9499C07A000036A3) added to nonce table
*May 6 18:07:26.975: VOICE_REG_POOL: Contact doesn't match any pools
...
```

If there are no voice register pools configured for a particular registration request, the message “Contact doesn’t match any pools” is displayed.

When authentication is enabled and if the phone requesting registration cannot be authenticated, the message “Registration Authorization failed with authorization header” is displayed.

**Cisco Unified SIP SRST**

The following is sample output from this command:
If there are no voice register pools configured for a particular registration request, the message “Contact doesn’t match any pools” is displayed.

If the **max registrations** command is configured, when registration requests reach the maximum limit, the “Maximum registration threshold for pool (x) hit” message is displayed for the particular pool.

The below table describes the significant fields shown in the display.

**Table 7: debug voice register errors Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Contact (doesn’t match any pools)</td>
<td>Contact refers to the location of the SIP devices and the IP address.</td>
</tr>
<tr>
<td>key (MAC address)</td>
<td>Unique MAC address of a locally available individual SIP phone used to support a degree of authentication in Cisco Unified CME.</td>
</tr>
<tr>
<td>Register request for (telephone number)</td>
<td>The unique key for each registration is the telephone number.</td>
</tr>
<tr>
<td>from (IP address)</td>
<td></td>
</tr>
<tr>
<td>Registration Authorization (failed with</td>
<td>Registration Authorization message is displayed when <strong>authenticate</strong> command is configured in Cisco Unified CME.</td>
</tr>
<tr>
<td>authorization header)</td>
<td></td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>debug voice register events</strong></td>
<td>Displays debug information on voice register module events during SIP phone registrations in a Cisco Unified CME or Cisco Unified SIP SRST environment.</td>
</tr>
</tbody>
</table>
debug voice register events

To display debug information on voice register module events during Session Initiation Protocol (SIP) phone registrations in a Cisco Unified CallManager Express (Cisco Unified CME) or Cisco Unified SIP Survivable Remote Site Telephony (SRST) environment, use the `debug voice register events` command in privileged EXEC mode. To disable debugging, use the `no` form of this command.

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Disabled

**Command Modes**
Privileged EXEC mode

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>This command was introduced for Cisco SIP SRST 3.0</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T for Cisco SIP SRST 3.0.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>This command was added to Cisco CME 3.4 and Cisco SIP SRST 3.4.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Using the `debug voice register events` command should suffice to view registration activity. Registration activity includes matching of pools, registration creation, and automatic creation of dial peers. For more details and error conditions, you can use the `debug voice register errors` command.

**Cisco Unified CME**

The following example shows output from this command:

```
*May 6 18:07:27.223: VOICE_REG_POOL: Register request for (4901) from (1.5.49.83)
*May 6 18:07:27.223: VOICE_REG_POOL: Contact matches pool 1 number list 1
*May 6 18:07:27.223: VOICE_REG_POOL: key(4901) contact(10.5.49.83) add to contact table
*May 6 18:07:27.223: VOICE_REG_POOL: No entry for (4901) found in contact table
*May 6 18:07:27.223: VOICE_REG_POOL: key(4901) contact(10.5.49.83) added to contact table
*VOICE_REG_POOL pool->tag(1), dn->tag(1), submask(1)
*May 6 18:07:27.223: VOICE_REG_POOL: Created dial-peer entry of type 0
*May 6 18:07:27.223: VOICE_REG_POOL: Registration successful for 4901, registration id is 2
...
```

The phone number 4901 associated with voice register pool 1, voice register dn 1, registered successfully. A dynamic normal (type 0) VoIP dial peer has been created for entry 4901. The dial peer can be verified using the `show voice register dial-peers` and `show sip-ua status registrar` commands.

**Cisco Unified SIP SRST**

The following is sample output from this command:

```
Router# debug voice register events
```
The phone number 91011 registered successfully, and type 1 is reported in the debug, which means that there is a preexisting VoIP dial peer.

A dynamic VoIP dial peer has been created for entry 91021. The dial peer can be verified using the show voice register dial-peers and show sip-ua status registrar commands.
The below table describes the significant fields shown in the display.

**Table 8: debug voice register events Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Contact</td>
<td>Indicates the location of the SIP devices and may indicate the IP address.</td>
</tr>
<tr>
<td>contact table</td>
<td>The table that maintains the location of the SIP devices.</td>
</tr>
<tr>
<td>key</td>
<td>The phone number is used as the unique key to maintain registrations of SIP devices.</td>
</tr>
<tr>
<td>multiple contact</td>
<td>More than one registration matches the same phone number.</td>
</tr>
<tr>
<td>no entry</td>
<td>The incoming registration was not found.</td>
</tr>
<tr>
<td>type 0</td>
<td>Normal dial peer.</td>
</tr>
<tr>
<td>type 1</td>
<td>Existing normal dial peer.</td>
</tr>
<tr>
<td>type 2</td>
<td>Proxy dial peer.</td>
</tr>
<tr>
<td>type 3</td>
<td>Existing proxy dial peer.</td>
</tr>
<tr>
<td>type 4</td>
<td>Dial-plan dial peer.</td>
</tr>
<tr>
<td>type 5</td>
<td>Existing dial-plan dial peer.</td>
</tr>
<tr>
<td>type 6</td>
<td>Alias dial peer.</td>
</tr>
<tr>
<td>type 7</td>
<td>Existing alias dial peer.</td>
</tr>
<tr>
<td>un-registration successful</td>
<td>The incoming unregister was successful.</td>
</tr>
<tr>
<td>Register request/registration id number</td>
<td>The internal unique number for each registration; useful for debugging particular registrations.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>debug voice register errors</strong></td>
<td>Displays debug information on voice register module errors during registration in a Cisco Unified CME or Cisco Unified SIP SRST environment.</td>
</tr>
<tr>
<td><strong>show sip-ua status registrar</strong></td>
<td>Displays all the SIP endpoints that are currently registered with the contact address.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>show voice register dial-peers</td>
<td>Displays details of Cisco Unified SIP SRST configuration and of all dynamically created VoIP dial peers.</td>
</tr>
</tbody>
</table>
default (voice hunt-group)

To set a command to its defaults values, use the **default** command in voice hunt-group configuration mode.

**default  default-value**

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th></th>
</tr>
</thead>
</table>
| **default-value**   | One of the voice hunt group configuration commands. Valid choices are as follows:  
|                     | • hops (Peer or longest-idle voice hunt group only)  
|                     | • preference  
|                     | • timeout |

**Command Default**

There are no default behaviors or values.

**Command Modes**

Voice hunt-group configuration (config-voi-hunt-group)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to configure the default value for a voice hunt group command.

The default command instructs the voice hunt group to use the default value of the specified command whenever the hunt group is called. This has the same effect as using the no form of the specified command, but the default command clearly specifies which commands are using their default values.

To use the default values for more than one command, enter each command on a separate line.

**Examples**

The following example shows how to set the default values for two separate voice hunt-group commands:

```
Router(config)# voice hunt-group 4  
peer  
Router(config-voi-hunt-group)# default hops  
Router(config-voi-hunt-group)# default timeout
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice hunt-group</td>
<td>Defines a hunt group for SIP phones in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
description (ephone)

To provide ephone descriptions for network management systems using an eXtensible Markup Language (XML) query, use the `description` command in ephone configuration mode. To remove a description, use the `no` form of this command.

```
description  string
no  application
```

**Syntax Description**
- `string` Allows for a maximum of 128 characters, including spaces. There are no character restrictions.

**Command Default**
No ephone description is configured.

**Command Modes**
Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
The descriptions configured with this command will appear neither on phone displays nor in show command output. Instead, they are sent to network management systems, such as CiscoView. Network management systems obtain `description` command data by sending an XML ISgetDevice request to a Cisco CME system. Cisco CME responds by sending ISDevDesc field data to the network management system, which uses the data to perform such tasks as printing descriptions on screen.

**Examples**
The following example provides a description for ephone 1:

```
Router(config)# ephone 1
Router(config-ephone)# description S/N:SK09456PPH3, Location:SJ21- 2nd Floor E5-9, User: Smith, John
```
**description (ephone-dn and ephone-dn-template)**

To display a custom text-string description in the header bar of all supported Cisco Unified IP phones, use the `description` command in ephone-dn or ephone-dn-template configuration mode. To return to the default, use the `no` form of this command.

```
description string
no description
```

**Syntax Description**

<table>
<thead>
<tr>
<th>string</th>
<th>Alphanumeric characters to be displayed in the header bar of the phone display. If spaces appear in the string, enclose the string in quotation marks. The maximum string length is 40 characters.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Note</td>
<td>Display behavior depends on phone firmware version.</td>
</tr>
</tbody>
</table>

**Command Default**

The extension number of the first line on the phone appears in the header bar.

**Command Modes**

Ephone-dn configuration (config-ephone)
Ephone-dn-template configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)T</td>
<td>Cisco ITS 2.0.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1</td>
<td>The number of characters in the string was modified.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-dn-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-dn-template configuration mode was integrated into Cisco IOS 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command under the ephone-dn that is associated with the first line button on a Cisco Unified IP phone. This command is typically used to display the entire E.164 telephone number associated with the first line button in the header bar rather than just the extension number, which is the default.

This command is supported by the following IP phones:

- Cisco Unified IP Phone 7940 and 7940G
- Cisco Unified IP Phones 7960 and 7960G
- Cisco Unified IP Phone 7970
- Cisco Unified IP Phone 7971

For Cisco Unified IP Phone 7940s and 7940Gs or Cisco Unified IP Phone 7960s and 7960Gs, the `string` is truncated to 14 characters if the text string is greater than 14 characters.

For Cisco Unified IP Phone 797x, all characters in the `string` appear alternately with time and date, each for 5 seconds.
If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

**Examples**

The following example shows how to define a header bar display for a phone on which the first line button is the extension number 50155:

```
Router(config)# ephone-dn 4
Router(config-ephone-dn)# number 50155
Router(config-ephone-dn)# description
  888-555-0155
```

The following example shows how to use an ephone-dn template to define a header bar display for a phone on which the first line button is the extension number 50155:

```
Router(config)# ephone-dn-template 3
Router(config-ephone-dn-template)# description
  “888 555-0155”
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 4
Router(config-ephone-dn)# number 50155
Router(config-ephone-dn)# ephone-dn-template 3
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>number</td>
<td>Configures a valid number for a Cisco Unified IP phone.</td>
</tr>
</tbody>
</table>
description (ephone-hunt)

To create a label for an ephone hunt group, use the `description` command in ephone-hunt configuration mode. To return this value to the default, use the `no` form of this command.

```
description string
no description
```

**Syntax Description**

| string | Character string that identifies a hunt group. |

**Command Default**

No description exists for the ephone hunt group.

**Command Modes**

Ephone-hunt configuration (config-ephone-hunt)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command creates a label to identify the ephone-hunt group. This label helps make the configuration more readable.

**Examples**

The following example shows how to identify a hunt group for technical support agents.

```
ephone-hunt 3 peer
pilot 4200
list 1001, 1002, 1003
description Tech Support Hunt Group
hops 3
timeout 7, 10, 15
max-timeout 25
final 4500
```
description (voice hunt-group)

To specify a description for a voice hunt group, use the **description** command in voice hunt-group configuration mode. To remove the description, use the **no** form of this command.

```
description  description
no description  description
```

**Syntax Description**

| description | Specific description of the hunt group. |

**Command Default**

No description for the hunt group.

**Command Modes**

Voice hunt-group configuration (config-voice-hunt-group)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Examples**

The following example shows how to specify a description for voice hunt-group 12 using the **description** command and presents the description in the output of the **do show run** command:

```
Router(config)# voice hunt-group 12
Router (config-voice-hunt-group)# description ?
    LINE description for this hunt group
Router (config-voice-hunt-group)# description specific huntgroup description
Router (config-voice-hunt-group)# do show run | sec voice hunt-group
voice hunt-group 12 parallel
timeout 0
description specific huntgroup description
```

**Command Description**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice</td>
<td>Enters voice hunt-group configuration mode to create a hunt group for phones in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>hunt-group</td>
<td></td>
</tr>
</tbody>
</table>


description (voice moh-group)

To display a brief description specific to a MOH group, use the `description` command in voice moh-group configuration mode. To remove the description, use the `no` form of this command.

```
description string
no description
```

**Syntax Description**

- `string`: An alphanumeric string to add a brief description specific to a MOH group. Maximum length: 80 characters including spaces.

**Command Default**

No MOH group description is configured.

**Command Modes**

Voice moh-group configuration (config-voice-moh-group)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0 Cisco Unified SRST 8.0</td>
<td>This command was introduced</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0 Cisco Unified SRST 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(T).</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command allows you to type a brief text describing a specific voice-moh-group. You can use maximum 80 characters, including spaces to describe a MOH group.

**Examples**

The following example provides a description for voice-moh-group1:

```
Router(config)#
Router(config-voice-moh-group)#
Router(config-voice-moh-group) description this is a moh group for sales
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice-moh-group</td>
<td>Enter voice-moh-group configuration mode.</td>
</tr>
<tr>
<td>moh</td>
<td>Enables music on hold from a flash audio feed</td>
</tr>
<tr>
<td>multicast moh</td>
<td>Enables multicast of the music-on-hold audio stream.</td>
</tr>
<tr>
<td>extension-range</td>
<td>Specifies the extension range for clients calling a voice-moh-group.</td>
</tr>
</tbody>
</table>
description (voice register pool)

To display a custom description in the header bar of Cisco IP Phone 7940 and 7940G or a Cisco IP Phone 7960 and 7960G, use the `description` command in voice register pool configuration mode. To return to the default, use the `no` form of this command.

```
description string
no description
```

**Syntax Description**
- `string`: Allows for a maximum of 128 characters, including spaces. There are no character restrictions.

**Command Default**
The extension number of the first line on the phone appears in the header bar.

**Command Modes**
- Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to display a customized description in the header bar of a SIP phone instead of the extension number, which is the default. For example, you can display the entire E.164 telephone number associated with the first phone line.

String is truncated to 14 characters in the display of the Cisco IP Phone 7940, Cisco IP Phone 7940G, Cisco IP Phone 7960, and Cisco IP Phone 7960G.

**Examples**
The following example shows how to define a header bar display for a SIP phone on which the extension number is 50155:

```
Router(config)# voice register pool 4
Router(config-register-pool)# number 1 50155
Router(config-register-pool)# description 888-555-0155
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>number (voice register pool)</code></td>
<td>Configures a valid number for a SIP phone.</td>
</tr>
</tbody>
</table>
**description (voice register pool-type)**

To specify the description string for a new phone model, use the `description` command in `voice register pool-type` mode. To remove the description string, use the `no` form of this command.

```
description  description
no description  description
```

**Syntax Description**
- `description string` Specifies description of the phone model.

**Command Default**
Description for the phone model is not defined. When the `reference-pooltype` command is configured, the description of the reference phone is inherited.

**Command Modes**
Voice Register Pool-Type Configuration (config-register-pool-type)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(3)M</td>
<td>Cisco SIP CME 10.0</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to specify the description string for a new phone model. When you use the `no` form of this command, the inherited properties of the reference phone takes precedence over the default value.

**Example**
The following example shows how to specify the description string for a phone model using the `description` command:

```
Router(config)# voice register pool-type 9900
Router(config-register-pool-type)# description New Cisco SIP Phone 9900
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register pool-type</td>
<td>Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.</td>
</tr>
</tbody>
</table>
device-id (ephone-type)

To specify the device ID of a phone type, use the `device-id` command in ephone-type configuration mode. To reset to the default value, use the `no` form of this command.

```
device-id number
no device-id
```

**Syntax Description**

<table>
<thead>
<tr>
<th>number</th>
<th>Device ID of the phone type. Range: 1 to 2,147,483,647. Default: 0. See the table below for a list of supported device IDs.</th>
</tr>
</thead>
</table>

**Command Default**

Device ID is 0.

**Command Modes**

Ephone-type configuration (config-ephone-type)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3 Cisco Unified SRST 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies the device ID of the type of phone being added with the ephone-type template. If this command is set to the default value of 0, the ephone-type is invalid.

**Table 9: Supported Values for Ephone-Type Commands**

<table>
<thead>
<tr>
<th>Supported Device</th>
<th>device-id</th>
<th>device-type</th>
<th>num-buttons</th>
<th>max-presentation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified IP Phone 6901</td>
<td>547</td>
<td>6901</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 6911</td>
<td>548</td>
<td>6911</td>
<td>1</td>
<td>10</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7915 Expansion Module with 12 buttons</td>
<td>227</td>
<td>7915</td>
<td>12</td>
<td>0 (default)</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7915 Expansion Module with 24 buttons</td>
<td>228</td>
<td>7915</td>
<td>24</td>
<td>0</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7916 Expansion Module with 12 buttons</td>
<td>229</td>
<td>7916</td>
<td>12</td>
<td>0</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7916 Expansion Module with 24 buttons</td>
<td>230</td>
<td>7916</td>
<td>24</td>
<td>0</td>
</tr>
<tr>
<td>Cisco Unified Wireless IP Phone 7925</td>
<td>484</td>
<td>7925</td>
<td>6</td>
<td>4</td>
</tr>
<tr>
<td>Cisco Unified IP Conference Station 7937G</td>
<td>431</td>
<td>7937</td>
<td>1</td>
<td>6</td>
</tr>
<tr>
<td>Nokia E61</td>
<td>376</td>
<td>E61</td>
<td>1</td>
<td>1</td>
</tr>
</tbody>
</table>
Examples

The following example shows the device ID is set to 376 for the Nokia E61 when creating the ephone-type template:

Router(config)# ephone-type E61
Router(config-ephone-type)# device-id 376
Router(config-ephone-type)# device-name E61 Mobile Phone

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>device-name</td>
<td>Assigns a name to a phone type in an ephone-type template.</td>
</tr>
<tr>
<td>load</td>
<td>Associates a type of phone with a phone firmware file.</td>
</tr>
<tr>
<td>type</td>
<td>Assigns the phone type to a SCCP phone.</td>
</tr>
</tbody>
</table>
**device-name**

To assign a name to a phone type in an ephone-type template, use the `device-name` command in ephone-type configuration mode. To remove the name, use the no form of this command.

```
device-name name
no device-name
```

**Syntax Description**

| `name` | String that identifies this phone type. Value is any alphanumeric string up to 32 characters. |

**Command Default**

No name is assigned to this phone type.

**Command Modes**

Ephone-type configuration (config-ephone-type)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3 Cisco Unified SRST 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies a device name for the type of phone being added with the ephone-type template.

**Examples**

The following example shows that the name “E61 Mobile Phone” is assigned to a phone type when creating the ephone-type template:

```
Router(config)# ephone-type E61
Router(config-ephone-type)# device-id 376
Router(config-ephone-type)# device-name E61 Mobile Phone
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>device-id</td>
<td>Specifies the device ID for a phone type in an ephone-type template.</td>
</tr>
</tbody>
</table>
device-security-mode

To set the security mode for SCCP signaling for devices communicating with the Cisco Unified CME router globally or per ephone, use the `device-security-mode` command in telephony-service or ephone configuration mode. To return to the default, use the `no` form of this command.

```
device-security-mode {authenticated|none|encrypted}
no device-security-mode
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>authenticated</td>
<td>SCCP signaling between a device and Cisco Unified CME through the secure TLS connection on TCP port 2443.</td>
</tr>
<tr>
<td>none</td>
<td>SCCP signaling is not secure.</td>
</tr>
<tr>
<td>encrypted</td>
<td>SCCP signaling between a device and Cisco Unified CME through the secure TLS connection on TCP port 2443, and the media uses Secure Real-Time Transport Protocol (SRTP).</td>
</tr>
</tbody>
</table>

### Command Default

Device signaling is not secure.

### Command Modes

Telephony-service configuration (config-telephony)

Ephone configuration (config-ephone)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(15)XW</td>
<td>Cisco Unified CME 4.1</td>
<td>The <code>encrypted</code> keyword was added.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>The <code>encrypted</code> keyword was added.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>The <code>encrypted</code> keyword was added.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use this command with Cisco Unified CME phone authentication and encryption.

Set the SCCP signaling security mode globally using this command in telephony-service configuration mode or per ephone using this command in ephone configuration mode. If you use both commands, the per-phone setting overrides the global setting.

### Examples

The following example selects secure SCCP signaling for all ephones.

```
Router(config)# telephony-service
Router(config-telephony)# device-security-mode authenticated
```

The following example selects secure SCCP signaling for ephone 28:

```
Router(config-telephony)# device-security-mode authenticated 28
```
Router(config)# ephone 28
Router(config-ephone)# button 1:14 2:25
Router(config-ephone)# device-security-mode authenticated

The following example selects secure SCCP signaling for all ephones and then disables it for ephone 36:

Router(config)# telephony-service
Router(config-telephony)# device-security-mode authentication
Router(config)# ephone 36
Router(config-ephone)# button 1:15 2:16
Router(config-ephone)# device-security-mode none

The following example selects encrypted secure SCCP signaling and encryption through SRTP for all ephones:

Router(config)# telephony-service
Router(config-telephony)# device-security-mode encrypted
**device-type**

To specify the phone type, use the `device-type` command in ephone-type configuration mode. To reset to the default value, use the `no` form of this command.

```
device-type phone-type
no device-type
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>phone-type</code></td>
<td>Device type of the phone. See the table for a list of supported device types. Default value is the same value entered with the <strong>ephone-type</strong> command.</td>
</tr>
</tbody>
</table>

**Command Default**

Device type is the same value that is entered with the **ephone-type** command.

**Command Modes**

Ephone-type configuration (config-ephone-type)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3 Cisco Unified SRST 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies the device type of the phone being added with the ephone-type template. The device type is set to the same value as the **ephone-type** command unless you use this command to change the value.

This command must be set to one of the following supported values.

**Table 10: Supported Values for Ephone-Type Commands**

<table>
<thead>
<tr>
<th>Supported Device</th>
<th>device-id</th>
<th>device-type</th>
<th>num-buttons</th>
<th>max presentation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified IP Phone 7915 Expansion Module with 12 buttons</td>
<td>227</td>
<td>7915</td>
<td>12</td>
<td>0 (default)</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7915 Expansion Module with 24 buttons</td>
<td>228</td>
<td>7915</td>
<td>24</td>
<td>0</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7916 Expansion Module with 12 buttons</td>
<td>229</td>
<td>7916</td>
<td>12</td>
<td>0</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7916 Expansion Module with 24 buttons</td>
<td>230</td>
<td>7916</td>
<td>24</td>
<td>0</td>
</tr>
<tr>
<td>Cisco Unified IP Conference Station 7937G</td>
<td>431</td>
<td>7937</td>
<td>1</td>
<td>6</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 8941</td>
<td>586</td>
<td>8941</td>
<td>4</td>
<td>3</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 8945</td>
<td>585</td>
<td>8945</td>
<td>4</td>
<td>3</td>
</tr>
<tr>
<td>Nokia E61</td>
<td>376</td>
<td>E61</td>
<td>1</td>
<td>1</td>
</tr>
</tbody>
</table>
### Examples

The following example shows the device type set to 7915 in the ephone-type template for the Cisco Unified IP Phone 7915 Expansion Module with 12 buttons:

```
Router(config)# ephone-type 7915-12 addon
Router(config-ephone-type)# device-id 227
Router(config-ephone-type)# device-name 7915-12
Router(config-ephone-type)# device-type 7915
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>device-name</td>
<td>Assigns a name to a phone type in an ephone-type template.</td>
</tr>
<tr>
<td>ephone-type</td>
<td>Adds a Cisco Unified IP phone type by defining an ephone-type template.</td>
</tr>
<tr>
<td>load</td>
<td>Associates a type of phone with a phone firmware file.</td>
</tr>
<tr>
<td>type</td>
<td>Assigns the phone type to a SCCP phone.</td>
</tr>
</tbody>
</table>
dial-peer no-match isdn disconnect-cause

To disconnect the incoming ISDN call when no inbound voice dial peer is matched, use the dial-peer no-match disconnect-cause command in global configuration mode. To restore the default incoming call handling behavior, use the no form of this command.

```
dial-peer no-match isdn disconnect-cause cause-code
no dial-peer no-match isdn disconnect-cause cause-code
```

**Syntax Description**

<table>
<thead>
<tr>
<th>cause-code</th>
</tr>
</thead>
<tbody>
<tr>
<td>An ISDN cause code number. Range is from 1 to 188.</td>
</tr>
</tbody>
</table>

**Command Default**

Dial-peer no-match isdn disconnect-cause command is disabled. Incoming ISDN calls are not forced to disconnect if no inbound dial-peer is matched.

**Command Modes**

Global configuration

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to disconnect unauthorized ISDN calls when no inbound voice or modem dial peer is matched.

Refer to the ISDN Cause Values table in the Cisco IOS Debug Command Reference, for a list of ISDN cause codes.

**Examples**

The following example shows that ISDN cause code 28 has been specified to match inbound voice or modem dial peers:

```
Router# dial-peer no-match disconnect-cause 28
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show dial-peer voice</td>
<td>Displays configuration information for dial peers.</td>
</tr>
</tbody>
</table>
**dialplan**

To assign a dial plan to a SIP phone, use the `dialplan` command in voice register pool or voice register template configuration mode. To remove the dial plan from the phone, use the `no` form of this command.

```
dialplan  dialplan-tag
no  dialplan  dialplan-tag
```

**Syntax Description**

| `dialplan-tag` | Number that identifies the dial plan to use for this SIP phone. This is the `dialplan-tag` argument that was assigned to the dial plan with the voice register `dialplan` command. Range: 1 to 24. |

**Command Default**

No dial plan is assigned to the phone.

**Command Modes**

Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

You apply a dial plan to a SIP phone with this command after you create the dial plan with the voice register `dialplan` command. When the phone is reset or restarted, the dial plan file specified with this command is loaded to the phone. A phone can use only one dial plan.

A dial plan assigned to a SIP phone has priority over Key Press Markup Language (KPML), which is enabled by default on the phone.

If you use a voice register template to apply a command to a phone and you also use the same command in voice register pool configuration mode for the same phone, the value that you set in voice register pool configuration mode has priority.

After using the `no dialplan` command to remove a dial plan from a phone, use the `restart` command after creating a new configuration profile if the dial plan was defined with the `pattern` command. If the dial plan was defined using a custom XML file with the `filename` command, you must use the `reset` command for the change to take effect.

**Examples**

The following example shows that dial plan 5 is assigned to the SIP phone identified by pool 1:

```
Router(config)# voice register pool 1
Router(config-register-pool)# dialplan 5
```

The following example shows that dial plan 5 is assigned to voice register template 10:

```
Router(config)# voice register template 10
Router(config-register-temp)# dialplan 5
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>digit collect kpml</td>
<td>Enables KPML digit collection on a SIP phone.</td>
</tr>
<tr>
<td></td>
<td>filename</td>
<td>Specifies a custom XML file that contains the dial patterns to use for a SIP dial plan.</td>
</tr>
<tr>
<td></td>
<td>pattern</td>
<td>Defines a dial pattern for a SIP dial plan.</td>
</tr>
<tr>
<td></td>
<td>show voice register dialplan</td>
<td>Displays all configuration information for a specific SIP dial plan.</td>
</tr>
<tr>
<td></td>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
<tr>
<td></td>
<td>voice register dialplan</td>
<td>Enters voice register dialplan configuration mode to define a dial plan for SIP phones.</td>
</tr>
</tbody>
</table>
**dialplan-pattern**

To define a pattern that is used to expand extension numbers in Cisco Unified CME into fully qualified E.164 numbers, use the `dialplan-pattern` command in telephony-service configuration mode. To disable the `dialplan-pattern` command settings, use the `no` form of this command.

```
dialplan-pattern  tag  pattern  extension-length  extension-length  [{extension-pattern  extension-pattern[no-reg]}  [demote]]
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Tag Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>tag</td>
<td>Identifies this dial-plan pattern. The tag is a number from 1 to 10.</td>
</tr>
<tr>
<td>pattern</td>
<td>Dial-plan pattern, such as the area code, the prefix, and the first one or two digits of the extension number, plus wildcard markers or dots (.) for the remainder of the extension number digits.</td>
</tr>
<tr>
<td>extension-length</td>
<td>Sets the number of extension digits that will appear as a caller ID.</td>
</tr>
<tr>
<td>extension-length</td>
<td>Number of extension digits. The extension length must match the length of extensions for IP phones. Range: 1 to 32.</td>
</tr>
<tr>
<td>extension-pattern</td>
<td>(Optional) Sets an extension number’s leading digit pattern when it is different from the E.164 telephone number’s leading digits as defined in the <code>extension-pattern</code> argument.</td>
</tr>
<tr>
<td>extension-pattern</td>
<td>(Optional) Extension number’s leading digit pattern. Consists of one or more digits and wildcard markers or dots (.). For example, 5.. would include extension 500 to 599, and 5... would include 5000 to 5999. The length of the extension pattern must equal the value configured for the <code>extension-length</code> argument.</td>
</tr>
<tr>
<td>no-reg</td>
<td>(Optional) Prevents the E.164 numbers in the dial peer from registering with the gatekeeper.</td>
</tr>
<tr>
<td>demote</td>
<td>(Optional) Demotes the registered phone if it matches the pattern, extension-length, and extension pattern.</td>
</tr>
</tbody>
</table>

**Command Default**

No expansion pattern exists.

**Command Modes**

Telephony-service configuration

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1</td>
<td>The <code>extension-pattern</code> keyword was added.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
</tbody>
</table>
This command was modified. The demote keyword was added to the dialplan pattern command and the dialplan pattern tag value was increased to 1-10.

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was modified. The demote keyword was added to the dialplan pattern command and the dialplan pattern tag value was increased to 1-10.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command creates a pattern for expanding individual abbreviated extension numbers of calling numbers into fully qualified E.164 numbers.

Use this command when configuring a network with multiple Cisco Unified CMEs to ensure that the appropriate calling number, extension or E.164 number, is provided to the target Cisco Unified CME, and appears on the phone display of the called phone. In networks that have a single Cisco Unified CME, this command is not needed.

If multiple dial-plan patterns are defined, the system matches extension numbers against the patterns in sequential order, starting with with the lowest numbered dial-plan pattern tag first. Once a pattern matches an extension number, the pattern is used to generate an expanded number. If additional patterns subsequently match the extension number, they are not used.

The `dialplan-pattern` command builds additional dial peers for the expanded numbers it creates. For example, when the `phone-dn` with the number 1001 was defined, the following POTS dial peer was automatically created for it:

```bash
dial-peer voice 20001 pots
destination-pattern 1001
voice-port 50/0/2
```

When you define a dial-plan pattern that 1001 will match, such as 40855510.., a second dial peer is created so that calls to both the 1001 and 4085551001 numbers will be completed. In our example, the additional dial peer that is automatically created looks like the following:

```bash
dial-peer voice 20002 pots
destination-pattern 4085551001
voice-port 50/0/2
```

Both numbers are recognized by Cisco Unified CME as being associated with a SCCP phone.

Both dial peers can be seen with the `show telephony-service dial-peer` command.

In networks with multiple routers, you may need to use the `dialplan-pattern` command to expand extensions to E.164 numbers because local extension numbering schemes can overlap each other. Networks with multiple routers have authorities such as gatekeepers that route calls through the network. These authorities require E.164 numbers so that all numbers in the network will be unique. Use the `dialplan-pattern` command to expand extension numbers into unique E.164 numbers for registering with a gatekeeper.

Ephone-dn numbers for the Cisco IP phones must match the number in the `extension-length` argument; otherwise, the extension number cannot be expanded. For example, the following command maps all 3-digit extension numbers to the telephone number 40855501xx, so that extension 111 is expanded but the 4-digit extension 1011 is not.

```bash
dialplan-pattern 1 40855501.. extension-length 3
```

Using the `dialplan-pattern` command to expand extension numbers can sometimes result in the improper matching of numbers with dial peers. For example, the expanded E.164 number 2035550134 can match dial-peer destination-pattern 203, not 134, which would be the correct destination pattern for the desired extension. If it is necessary for you to use the `dialplan-pattern` command and you know that the expanded
numbers might match destination patterns for other dial peers, you can manually configure the E.164 expanded number for an extension as its secondary number using the `number` command, as shown in the following example:

```plaintext
ephone-dn 23
   number 134 secondary 2035550134
```

The pattern created by the `dialplan-pattern` command is also used to enable distinctive ringing for inbound calls. If a calling-party number matches a dial-plan pattern, the call is considered an internal call and has a distinctive ring that identifies the call as internal. Any call with a calling-party number that does not match a dial-plan pattern is considered an external call and has a distinctive ring that is different from the internal ringing.

When the `extension-pattern` keyword and `extension-pattern` argument are used, the leading digits of an extension pattern are stripped and replaced with the corresponding leading digits of the dial plan. For example, the following command maps all 4xx extension numbers to the E.164 number 40855501xx, so that extension 412 corresponds to 4085550112.

```
dialplan-pattern 1 4085550100 extension-length 3 extension-pattern 4..
```

When the demote keyword is used, the `dialplan-pattern` command tries to demote the registered phone if it matches the pattern, extension-length, and extension-pattern.

**Examples**

The following example shows how to create dial-plan pattern 1 for extension numbers 5000 to 5099 with a prefix of 408555. If an inbound calling party number (4085555044) matches dial-plan pattern 1, the recipient phone will display an extension (5044) as the caller ID and use an internal ringing tone. If an outbound calling party extension number (5044) matches the same dial-plan pattern 1, the calling-party extension will be converted to an E.164 number (4085555044). The E.164 calling-party number will appear as the caller ID.

```
Router(config)# telephony-service
Router(config-telephony)# dialplan-pattern 1 40855550.. extension-length 4 extension-pattern 50..
```

In the following example, the `dialplan-pattern` command creates dial-plan pattern 1 for extensions 800 to 899 with the telephone prefix starting with 408555. As each number in the extension pattern is declared with the `number` command, two POTS dial peers are created. In the example, they are 801 (an internal office number) and 4085579001 (an external number).

```
Router(config)# telephony-service
Router(config-telephony)# dialplan-pattern 1 40855590.. extension-length 3 extension-pattern 8..
```

The following example shows a configuration for two Cisco CME systems. One system uses 50.. and the other uses 60.. for extension numbers. Each is configured with the same two `dialplan-pattern` commands. Calls from the “50..” system to the “60..” system, and vice versa, are treated as internal calls. Calls that go across a H.323 network and calls that go to a PSTN through an ISDN interface on one of the configured Cisco CME routers are represented as E.164.

```
Router(config)# telephony-service
Router(config-telephony)# dialplan-pattern 1 40855550.. extension-length 4 extension-pattern 50..
```
Router(config-telephony)# dialplan-pattern 2 51055560.. extension-length 4 extension-pattern 60..

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>show telephony-service dial-peer</td>
<td>Displays dial peer information for extensions in a Cisco CME system.</td>
</tr>
</tbody>
</table>
To create a global prefix that can be used to expand the extension numbers of inbound and outbound calls into fully qualified E.164 numbers, use the `dialplan-pattern` command in call-manager-fallback configuration mode. To disable the `dialplan-pattern` command settings, use the `no` form of this command.

```markdown
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>tag</code></td>
<td>Dial-plan string tag used before a ten-digit telephone number. The tag number is from 1 to 10.</td>
</tr>
<tr>
<td><code>pattern</code></td>
<td>Dial-plan pattern, such as the area code, the prefix, and the first one or two digits of the extension number, plus wildcard markers or dots (.) for the remainder of the extension number digits.</td>
</tr>
<tr>
<td><code>extension-length</code></td>
<td>Sets the number of extension digits that will appear as a caller ID.</td>
</tr>
<tr>
<td><code>extension-length</code></td>
<td>The number of extension digits. The extension length must match the setting for IP phones in Cisco Unified CallManager mode. The range is from 1 to 32.</td>
</tr>
<tr>
<td><code>extension-pattern</code></td>
<td>(Optional) Sets an extension number’s leading digit pattern when it is different from the E.164 telephone number’s leading digits defined in the <code>pattern</code> variable.</td>
</tr>
<tr>
<td><code>extension-pattern</code></td>
<td>(Optional) The extension number’s leading digit pattern. Consists of one or more digits and wildcard markers or dots (.). For example, 5.. would include extensions 500 to 599; 5... would include extensions 5000 to 5999. The extension pattern must match the setting for IP phones in Cisco Unified CallManager mode.</td>
</tr>
<tr>
<td><code>no-reg</code></td>
<td>(Optional) Prevents the E.164 numbers in the dial peer from registering with the gatekeeper.</td>
</tr>
<tr>
<td><code>demote</code></td>
<td>(Optional) Demotes the registered phone if it matches the pattern, extension-length, and extension pattern.</td>
</tr>
</tbody>
</table>

### Command Default

No default behavior or values.

### Command Modes

Call-manager-fallback configuration

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco SRST 1.0</td>
<td>This command was introduced on the Cisco 2600 series and Cisco 3600 series multiservice routers and on the Cisco IAD2420 series.</td>
</tr>
<tr>
<td>12.2(2)XT</td>
<td>Cisco SRST 2.0</td>
<td>This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.</td>
</tr>
</tbody>
</table>
### Usage Guidelines

The `dialplan-pattern` command builds additional dial peers. For example, if a hidden POTS dial peer is created, such as the following:

```plaintext
Router(config)# dial-peer voice 20001 pots
Router(config-dial-peer)# destination-pattern 1001
Router(config-dial-peer)# voice-port 50/0/2
```

and a dial-plan pattern is created, such as 40855510.., then an additional dial peer will be created that allows calls to both the 1001 and 4085551001 numbers. For example:

```plaintext
Router(config)# dial-peer voice 20002 pots
Router(config-dial-peer)# destination-pattern 4085551001
Router(config-dial-peer)# voice-port 50/0/2
```

Both dial peers can be seen with the `show dial-peer voice` command.

The `dialplan-pattern` command also creates a global prefix that can be used by inbound calls (calls to an IP phone in a Cisco Unified SRST system) and outbound calls (calls made from an IP phone in a Cisco Unified SRST system) to expand their extension numbers to fully qualified E.164 numbers.

For inbound calls (calls to an IP phone in a Cisco Unified SRST system) where the calling party number matches the dial-plan pattern, the call is considered a local call and has a distinctive ring that identifies the call as internal. Any calling party number that does not match the dial-plan pattern is considered an external call and has a distinctive ring that is different from the internal ringing.

For outbound calls, the `dialplan-pattern` command converts the calling party’s extension number to an E.164 calling party number. Outbound calls that do not use an E.164 number and go through a PRI connection to the PSTN may be rejected by the PRI link as the calling party identifier.

If there are multiple patterns, called-party numbers are checked in numeric order, starting with pattern 1, until a match is found or until the last pattern has been checked. The valid dial-plan pattern with the lowest tag is used as a prefix to all local Cisco IP phones.

When `extension-pattern extension-pattern` keyword and argument are used, the leading digits of an extension pattern are stripped and replaced with the corresponding leading digits of the dial plan. For example, the following command maps all extension numbers 4xx to the PSTN number 40855501xx, so that extension 412 corresponds to 4085550112.
Router(config)# call-manager-fallback
Router(config-cm-fallback)# dialplan-pattern 1 4085550100 extension-length 3 extension-pattern 4..

The number of extension-pattern argument characters must match the number set for the extension-length argument. For example, if the extension-length is 3, the extension-pattern can be 8.., 1.., 51., and so forth.

A dial-plan pattern is required to register the Cisco IP phone lines with a gatekeeper. The no-reg keyword provides the option of not registering specific numbers to the gatekeeper so that those numbers can be used for other telephony services.

When the demote keyword is used, the dialplan-pattern command tries to demote the registered phone if it matches the pattern, extension-length, and extension-pattern.

Examples

The following example shows how to create dial-plan pattern 1 for extension numbers 5000 to 5099 with a prefix of 408555. If an inbound calling party number (408555044) matches dial-plan pattern 1, the recipient phone will display an extension (5044) as the caller ID and use an internal ringing tone. If an outbound calling party extension number (5044) matches dial-plan pattern 1, the calling party extension will be converted to an E.164 number (408555044). The E.164 calling party number will appear as the caller ID.

Router(config)# call-manager-fallback
Router(config-cm-fallback)# dialplan-pattern 1 4085550.. extension-length 4 extension-pattern 50..

In the following example, the dialplan-pattern command creates dial-plan pattern 1 for extensions 800 to 899 with the telephone prefix starting with 408555. As each number in the extension pattern is declared with the number command, two POTs dial peers are created. In the example, they are 801 (an internal office number) and 4085559001 (an external number).

Router(config)# call-manager-fallback
Router(config-cm-fallback)# dialplan-pattern 1 4085550.. extension-length 3 extension-pattern 8..

The following example shows a configuration for two Cisco Unified SRST systems. Each is configured with the same dialplan-pattern commands, but one system uses 50.. and the other uses 60.. for extension numbers. Calls from the “50..” system to the “60..” system, and vice versa, are treated as internal calls. Calls that go across an H.323 network and calls that go to a PSTN through an ISDN interface on one of the configured Cisco Unified SRST routers are represented as E.164.

Router(config)# call-manager-fallback
Router(config-cm-fallback)# dialplan-pattern 1 4085550.. extension-length 4 extension-pattern 50..
Router(config-cm-fallback)# dialplan-pattern 2 51055560.. extension-length 4 extension-pattern 60..

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-manager-fallback</td>
<td>Enables Cisco Unified SRST support and enters call-manager-fallback configuration mode.</td>
</tr>
<tr>
<td>show dial-peer voice</td>
<td>Displays information for voice dial peers.</td>
</tr>
</tbody>
</table>
dialplan-pattern (voice register)

To define a pattern that is used to expand extension numbers in Cisco Unified CME into fully qualified E.164 numbers, use the **dialplan-pattern** command in voice register global configuration mode. To disable the **dialplan-pattern** command settings, use the **no** form of this command.

```
dialplan-pattern tag pattern extension-length [extension-pattern |no-reg] [demote]
no dialplan-pattern tag
```

### Syntax Description

<table>
<thead>
<tr>
<th><strong>tag</strong></th>
<th>Unique number for identifying this dial-plan pattern. Range: 1 to 10.</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>pattern</strong></td>
<td>Dial-plan pattern to be matched, such as the area code, the prefix, and the first one or two digits of the extension number, plus wildcard markers or dots (.) for the remainder of the extension number digits.</td>
</tr>
<tr>
<td><strong>extension-length</strong></td>
<td>Number of extension digits that will appear as a caller ID.</td>
</tr>
<tr>
<td><strong>extension-length</strong></td>
<td>Number of digits in an extension. This variable must match the length of the directory numbers configured for SIP extensions in Cisco Unified CME. Range: 1 to 32.</td>
</tr>
<tr>
<td><strong>extension-pattern</strong></td>
<td>(Optional) Leading digit pattern to be configured for an extension when it is different from the leading digit pattern of the E.164 telephone number, as defined in the <strong>extension-pattern</strong> argument.</td>
</tr>
<tr>
<td><strong>extension-pattern</strong></td>
<td>(Optional) Leading digit pattern to be stripped from extension number when expanding an extension to an E.164 telephone number. Consists of one or more digits and wildcard markers or dots (.). For example, 5.. would include extension 500 to 599, and 5... would include 5000 to 5999. The length of the extension pattern must equal the value configured for the <strong>extension-length</strong> argument.</td>
</tr>
<tr>
<td><strong>no-reg</strong></td>
<td>(Optional) Prevents the E.164 numbers in the dial peer from registering with the gatekeeper.</td>
</tr>
<tr>
<td><strong>demote</strong></td>
<td>(Optional) Demotes the registered phone if it matches the pattern, extension-length, and extension pattern.</td>
</tr>
</tbody>
</table>

### Command Default

No expansion pattern exists.

### Command Modes

Voice register global configuration (config-register-global)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS 12.4(9)T.</td>
</tr>
</tbody>
</table>
This command creates a pattern for expanding individual abbreviated SIP extension numbers of calling numbers into fully qualified E.164 numbers.

Use this command when configuring a network with multiple Cisco Unified CMEs to ensure that the appropriate calling number, extension or E.164 number, is provided to the target Cisco Unified CME, and appears on the phone display of the called phone. In networks that have a single Cisco Unified CME, this command is not needed.

Up to five dial-plan patterns can be configured. If multiple dial-plan patterns are defined, the system matches extension numbers against the patterns in sequential order, starting with the lowest numbered dial-plan pattern tag first.

Dial peers for directory numbers are automatically created when SIP phones register in Cisco Unified CME. The `dialplan-pattern` command builds a second dial peer for the expanded number because an extension number matches the pattern. Both numbers are recognized by Cisco Unified CME as being associated with a SIP phone.

For example, the following POTS dial peer is automatically created for extension number 1001 when the associated SIP phone registers in Cisco Unified CME:

```plaintext
dial-peer voice 20001 pots
destination-pattern 1001
voice-port 50/0/2
```

If the extension number (1001) also matches a dial-plan pattern that is configured using the `dialplan-pattern` command, such as 40855510.., a second dial peer is dynamically created so that calls to both the 1001 and 4085551001 numbers can be completed. Based on the dial-plan pattern to be matched, the following additional POTS dial peer is created:

```plaintext
dial-peer voice 20002 pots
destination-pattern 4085551001
voice-port 50/0/2
```

Using the `no` form of this command will remove the dial peer that was created for the expanded number.

All dial peers can be displayed by using the `show dial-peer voice summary` command. All dial peers for numbers associated to SIP phones only can be displayed by using the `show voice register dial-peers` command. Dial peers created by using the `dialplan-expansion` command cannot be seen in the running configuration.

The value of the extension-length argument must be equal to the length of extension number to be matched, otherwise, the extension number cannot be expanded. For example, the following command maps all 3-digit extension numbers to the telephone number 40855501.., so that extension 111 is expanded but 4-digit extension number 1111 is not.

```plaintext
dialplan-pattern 1 40855501.. extension-length 3
```

When the `extension-pattern` keyword and `extension-pattern` argument are configured, the leading digits of the extension pattern variable are stripped away and replaced with the corresponding leading digits of the dial-plan pattern to create the expanded number. For example, the following command maps all 3-digit

---

**Usage Guidelines**

This command was modified. The `demote` keyword was added to the dialplan pattern command and the dialplan pattern tag value was increased to 1-10.

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was modified. The <code>demote</code> keyword was added to the dialplan pattern command and the dialplan pattern tag value was increased to 1-10.</td>
</tr>
</tbody>
</table>
extension numbers with the leading digit of “4” to the telephone number 40855501.., so that extension 434 corresponds to 4085550134.

dialplan-pattern 1 40855501.. extension-length 3 extension-pattern 4..

To apply dialplan-pattern expansion on a per-system basis to individual SIP redirecting numbers in a Cisco Unified CME system, including original called and last reroute numbers, use the call-forward command.

When the demote keyword is used, the dialplan-pattern command tries to demote the registered phone if it matches the pattern, extension-length, and extension-pattern

Examples

The following example shows how to create a dialplan-pattern for expanding extension numbers 60xxx to E.164 numbers 5105555xxx.

Router(config)# voice register global
Router(config-register-global)# dialplan-pattern 1 5105550... extension-length 5

The following example is output from the show dial-peer summary command displaying information for four dial peers, one each for extensions 60001 and 60002 and, because the dialplan-expansion command was configured to expand 6.... to 4085555...., one each for 4085550001 and 4085550002. The latter two dial peers will not appear in the running configuration.

Router# show dial-peer summary

<table>
<thead>
<tr>
<th>AD</th>
<th>TAG</th>
<th>TYPE</th>
<th>MIN</th>
<th>OPER</th>
<th>PREFIX</th>
<th>DEST-PATTERN</th>
<th>FER</th>
<th>THRU</th>
<th>SESS-TARGET</th>
<th>OUT</th>
<th>STATT</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>20010</td>
<td>pots</td>
<td>up</td>
<td>up</td>
<td>600025</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>20011</td>
<td>pots</td>
<td>up</td>
<td>up</td>
<td>600015</td>
<td>0</td>
<td>0</td>
<td>9</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>20012</td>
<td>pots</td>
<td>up</td>
<td>up</td>
<td>51055550015</td>
<td>0</td>
<td>9</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>20013</td>
<td>pots</td>
<td>up</td>
<td>up</td>
<td>51055550025</td>
<td>0</td>
<td>0</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-forward (voice register)</td>
<td>Applies dial-plan pattern expansion globally to redirecting number.</td>
</tr>
<tr>
<td>show dial-peer summary</td>
<td>Displays all dial peers created in Cisco Unified CME.</td>
</tr>
<tr>
<td>show voice register dial-peer</td>
<td>Displays dial-peer information for SIP extensions in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
digit collect kpml

To enable Key Press Markup Language (KPML) digit collection on a SIP phone, use the `digit collect kpml` command in voice register pool or voice register template configuration mode. To disable KPML, use the `no` form of this command.

```
digit collect kpml
no digit collect kpml
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
KPML digit collection is enabled.

**Command Modes**
Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1 Cisco Unified SRST 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1 Cisco Unified SRST 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
KPML is enabled by default for all directory numbers on the phone. A dial plan assigned to a phone has priority over KPML. Use the `no digit collect kpml` command to disable KPML on a phone.

If you use a voice register template to apply a command to a phone and you also use the same command in voice register pool configuration mode for the same phone, the value that you set in voice register pool configuration mode has priority.

KPML is not supported on the Cisco Unified IP Phone 7905, 7912, 7940, or 7960.

**Examples**
The following example shows KPML enabled on SIP phone 4:

```
Router(config)# voice register pool 4
Router(config-register-pool)# digit collect kpml
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dialplan</td>
<td>Assigns a dial plan to a SIP phone.</td>
</tr>
<tr>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a SIP phone.</td>
</tr>
<tr>
<td>voice register dialplan</td>
<td>Enters voice register dialplan configuration mode to define a dial plan for SIP phones.</td>
</tr>
</tbody>
</table>
**direct-inward-dial isdn**

To enable incoming ISDN enbloc dialing calls, use the `direct-inward-dial isdn` command in voice service voip mode. To disable incoming ISDN enbloc dialing calls use the `no` form of the command.

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

The direct inward dial `isdn` command is enabled.

**Command Modes**

voice service pots

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `direct-inward-dial isdn` command to enable the direct-inward-dial (DID) call treatment for an incoming ISDN call. When this feature is enabled, the incoming ISDN call is treated as if the digits were received from the DID trunk. The called number is used to select the outgoing dial peer. No dial tone is presented to the caller to collect dialed digits even if "no direct-inward-dial" of the selected inbound dial-peer is defined for an incoming ISDN call.

Use the `no` form of this command to turn off the global direct-inward-dial setting for incoming ISDN calls. When this command line is disabled, the “direct-inward-dial” setting of a selected inbound dial-peer is used to handle the incoming ISDN calls.'

**Examples**

The following is a sample output from this command displaying DID enabled for ISDN:

```
!
voice service voip
ip address trusted list
ipv4 172.19.245.1
ipv4 172.19.247.1
ipv4 172.19.243.1
ipv4 171.19.245.1
ipv4 171.19.10.1
allow-connections h323 to h323
allow-connections h323 to sip
allow-connections sip to h323
allow-connections sip to sip
supplementary-service media-renegotiate
sip
  registrar server expires max 120 min 120
!
!
dial-peer voice 1 voip
  destination-pattern 5511...
  session protocol sipv2
  session target ipv4:1.3.45.1
  incoming called-number 5522...
  direct-inward-dial
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice service</td>
<td>Enters voice service configuration mode.</td>
</tr>
</tbody>
</table>
directory

To define the order in which the names of Cisco IP phone users are displayed in the local directory, use the `directory` command in telephony-service configuration mode. To return to the default, use the `no` form of this command.

```
directory {first-name-first|last-name-first}
no directory {first-name-first|last-name-first}
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>first-name-first</code></td>
<td>First name is entered first in the Cisco IP phone directory name field.</td>
</tr>
<tr>
<td><code>last-name-first</code></td>
<td>Last name is entered first in the Cisco IP phone directory name field.</td>
</tr>
</tbody>
</table>

### Command Default

Default is `first-name-first`.

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command defines name order in the local directory. The directory itself is generated from entries made using the `name` command and the `number` command in ephone-dn configuration mode.

#### Note

The name information must be entered in the correct order in the `name` command.

The location for the file that is accessed when the Directories button is pressed is specified in the `url` (telephony-service) command.

### Examples

The following example shows how to configure the local directory with the last name first:

```
Router(config)# telephony-service
Router(config-telephony)# directory last-name-first
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>name</code></td>
<td>Specifies a name to be associated with an extension (ephone-dn).</td>
</tr>
<tr>
<td><code>number</code></td>
<td>Specifies a telephone number to be associated with an extension (ephone-dn).</td>
</tr>
<tr>
<td><code>url</code></td>
<td>Provisions URLs for the displays associated with buttons on Cisco IP phones.</td>
</tr>
</tbody>
</table>
directory entry

To add a system-wide phone directory and speed-dial definition, use the directory entry command in telephony-service configuration mode. To remove a definition, use the no form of this command.

```
directory entry {directory-tag number name name|clear}
no directory entry {directory-tag|clear}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Argument</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>directory-tag</td>
<td>Digit string that provides a unique identifier for this entry. Range: 1 to 250.</td>
</tr>
<tr>
<td>number</td>
<td>String of up to 32 digits that provides the full telephone number for this entry.</td>
</tr>
<tr>
<td>name name</td>
<td>String of up to 24 alphanumeric characters, including spaces. Cannot include opening or closing quotation marks (‘’, ’’, “, or ”).</td>
</tr>
<tr>
<td>clear</td>
<td>Removes all directory entries that were made with this command.</td>
</tr>
</tbody>
</table>

**Command Default**

Entries do not exist.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This feature was modified to enable systemwide speed-dialing of entries from 34 to 99.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>The maximum number of directory entries was increased from 100 to 250.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Cisco Unified CME automatically creates a local phone directory consisting of the telephone numbers and names that are entered during ephone-dn configuration. Additional directory entries can be made by administrators using the directory entry command. Phone number directory listings are displayed in the order in which they are entered.

A single entry can be removed using the no directory entry directory-tag command.

Directory entries that have directory-tag numbers from 34 to 99 also can be used as system-wide speed-dial numbers. That is, if you have the following definition for the headquarters office, any phone user can speed-dial the number:

```
Router(config)# telephony-service
directory entry {directory-tag number name name|clear}
```
Analog phone users press the asterisk (*) key and the speed-dial identifier (tag number) to dial a speed-dial number.

IP phone users follow this procedure to dial a speed-dial number:

1. With the phone on-hook, an IP phone user presses a two-digit speed-dial code (that is, 05 for the entry with tag 5). A new soft key, Abbr, appears in the phone display.
2. The phone user picks up the phone handset and presses the Abbr soft key. The full telephone number associated with the speed-dial tag is dialed.

**Examples**

The following example adds six telephone listings to the local directory. The last two entries, with the identifiers 50 and 51, can be speed-dialed by anyone on the system because their identifiers (directory-tags) are between 34 and 99.

```
Router(config)# telephony-service
Router(config-telephony)# directory entry 1 4045550110 name Atlanta
Router(config-telephony)# directory entry 2 3125550120 name Chicago
Router(config-telephony)# directory entry 4 2125550140 name New York City
Router(config-telephony)# directory entry 5 2065550150 name Seattle
Router(config-telephony)# directory entry 50 4085550123 name Corp Headquarters
Router(config-telephony)# directory entry 51 4085550145 name Division Headquarters
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show telephony-service</td>
<td>Displays the configured directory entries.</td>
</tr>
<tr>
<td>directory-entry</td>
<td></td>
</tr>
<tr>
<td>url directories</td>
<td>Provisions the directory URL to select an external directory resource and disables the Cisco Unified CME local directory service.</td>
</tr>
</tbody>
</table>
display-logout

To specify a message to display on phones in an ephone hunt group when all phones in the hunt group are logged out, use the `display-logout` command in ephone-hunt configuration mode. To return this value to the default, use the `no` form of this command.

```
display-logout string
no display-logout
```

**Syntax Description**

- `string` Character string to be displayed on hunt group member IP phones when all members are logged out.

**Command Default**

No logout message exists.

**Command Modes**

Ephone-hunt configuration (config-ephone-hunt)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command defines a plain-text message that displays on phones with ephone-dns that are members of a hunt group when all the members of the group are logged out. The message can be used to notify agents that no agents are available to take hunt group calls. It can also be used to tell agents about the disposition of any incoming calls to the hunt group when no agents are available to answer calls. For example, you could set the display to read “All Agents Unavailable,” or “Hunt Group Voice Mail” or “Hunt Group Night Service.”

**Examples**

The following example specifies a message to display when all agents are logged out of hunt group 3.

```
ephone-hunt 3 peer
pilot 4200
list 1001, 1002, 1003
display-logout All Agents Logged Out
hops 3
timeout 7, 10, 15
max-timeout 25
final 4500
```
dnd (voice register pool)

To enable the Do-Not-Disturb (DND) feature, use the `dnd-control` command in voice register pool configuration mode. To disable the DND, use the `no` form of this command.

```plaintext
   dnd
   no dnd
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
DND is disabled

**Command Modes**
Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Examples**
The following example shows how to enable DND:

```plaintext
Router(config)# voice register pool 1
Router(config-register-pool)# dnd
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dnd-control (voice register template)</td>
<td>Enables DND soft key in template to be assigned to SIP phones in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
**dnd feature-ring**

To disable ringing on phone buttons configured for feature ring when the phone is in do-not-disturb (DND) mode, use the `dnd feature-ring` command in ephone configuration mode. To allow lines configured for feature ring to ring when the phone is in DND mode, use the `no` form of this command.

```
dnd feature-ring
no dnd feature-ring
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Incoming calls to buttons configured for feature ring do not ring in DND mode.

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command applies only to phone lines that are configured for the feature-ring option with the `button` command.

Note that the affirmative form of the command is enabled by default and feature-ring lines will not ring when the phone is in DND mode. To enable feature-ring lines to ring when the phone is in DND mode, use the `no dnd feature-ring` command.

**Examples**

For the following example, when DND is active on ephone 1 and ephone 2, button 1 will ring, but button 2 will not.

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1001

Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 1002

Router(config)# ephone-dn 10
Router(config-ephone)# number 1110
Router(config-ephone)# preference 0
Router(config-ephone)# no huntstop

Router(config)# ephone-dn 11
Router(config-ephone)# number 1111
Router(config-ephone)# preference 1
Router(config-ephone)# no huntstop

Router(config)# ephone 1
Router(config-ephone)# button 1f1
Router(config-ephone)# button 2o10,11
Router(config-ephone)# no dnd feature-ring
```
Router(config-ephone-dn)# ephone 2
Router(config-ephone)# button 1f2
Router(config-ephone)# button 2010,11
Router(config-ephone)# no dnd feature-ring

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>button</td>
<td>Associates ephone-dns with individual buttons on a Cisco IP phone and specifies ring behavior.</td>
</tr>
</tbody>
</table>
dnd-control (voice register template)

To enable the Do-Not-Disturb (DND) soft key on SIP phones, use the dnd-control command in voice register template configuration mode. To disable the DND soft key on a SIP phone, use the no form of this command.

```
dnd-control
no dnd-control
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
DND soft key is enabled on SIP phones in Cisco Unified CME.

**Command Modes**
Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command enables a soft key for Do-Not-Disturb (DND) in the specified template which can then be applied to SIP phones. The DND soft key is enabled by default. To disable the DND soft key, use the dnd command. To apply a template to a SIP phone, use the template command in voice register pool configuration mode.

**Examples**
The following example shows how to disable the DND soft key:

```
Router(config)# voice register template 1
Router(config-register-template)# dnd-control
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dnd (voice register pool)</td>
<td>Enables DND feature.</td>
</tr>
</tbody>
</table>
**dn-webedit**

To enable the adding of extensions (ephone-dns) through the Cisco Unified CME graphical user interface (GUI), use the **dn-webedit** command in telephony-service configuration mode. To disable this feature, use the no form of this command.

```
dn-webedit
no  dn-webedit
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Extensions cannot be added through the Cisco Unified CME GUI.

**Command Modes**
Telephony-service configuration (config-telephony)

**Command History**
<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command is deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
The **dn-webedit** command enables the adding of extensions through the web-based GUI. If the **dn-webedit** command is enabled, a customer administrator or a system administrator can modify and assign extensions associated with the Cisco Unified CME router. If this ability is disabled, extensions must be added using Cisco IOS commands.

If the set of extension numbers used by the router is part of a larger telephone network, limitations on modification might be needed to ensure network integrity. Disabling the **dn-webedit** command prevents an administrator from allocating phone numbers and prevents assignment of numbers that may already be used elsewhere in the network.

**Examples**
The following example enables editing of directory numbers through the web-based GUI interface:

```
Router(config)# telephony-service
Router(config-telephony)# dn-webedit
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>time-webedit</td>
<td>Enables time setting through the web interface.</td>
</tr>
</tbody>
</table>
**dst (voice register global)**

To set the time period for daylight saving time on SIP phones, use the **dst** command in voice register global configuration mode. To disable daylight saving time, use the **no** form of this command.

```
**dst auto-adjust**
**no dst {start|stop}**
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>start</td>
<td>Sets beginning time for daylight saving time.</td>
</tr>
<tr>
<td>stop</td>
<td>Sets ending time for daylight saving time.</td>
</tr>
<tr>
<td>month</td>
<td>Abbreviated month. The following abbreviations are valid: <strong>jan, feb, mar, apr, may, jun, jul, aug, sep, oct, nov, dec.</strong></td>
</tr>
<tr>
<td>day day-of-month</td>
<td>Date of the month. Range is 1 to 31.</td>
</tr>
<tr>
<td>week week-number</td>
<td>Number identifying the week of the month. Range is 1 to 4, or 8, where 8 represents the last week of the month.</td>
</tr>
<tr>
<td>day day-of-week</td>
<td>Abbreviated day of the week. The following abbreviations are valid: <strong>sun, mon, tue, wed, thu, fri, sat.</strong></td>
</tr>
<tr>
<td>time hour:minutes</td>
<td>Beginning and ending time for daylight saving time, in HH:MM format using a 24-hour clock. The stop time must be greater than the start time. The value 24:00 is not valid. If you enter 00:00 for both start time and stop time, daylight saving time is enabled for the entire 24-hour period on the specified date.</td>
</tr>
</tbody>
</table>

**Command Default**

Default start time is first week of April, Sunday, 2:00 a.m and default stop time is last week of October, Sunday 2:00 a.m.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command sets the stop and start times for daylight saving time if the **dst auto-adjust** command is configured.

**Examples**

The following example shows how to set automatic adjustment of daylight saving time:

```
Router(config)# voice register global
Router(config-register-global)# dst start Jan day 1 time 00:00
Router(config-register-global)# dst stop Mar day 31 time 23:99
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>date-format (voice register global)</code></td>
<td>Sets the date display format on SIP phones in a Cisco CME system.</td>
</tr>
<tr>
<td><code>dst auto-adjust (voice register global)</code></td>
<td>Enables automatic adjustment of daylight saving time on SIP phones.</td>
</tr>
<tr>
<td><code>time-format (voice register global)</code></td>
<td>Selects a 12-hour clock or a 24-hour clock for the time display format on Cisco IP phones in a SIP CME system.</td>
</tr>
<tr>
<td><code>timezone (voice register global)</code></td>
<td>Sets the time zone used for SIP phones in a Cisco CME system.</td>
</tr>
</tbody>
</table>
dst auto-adjust (voice register global)

To enable automatic adjustment of daylight saving time on SIP phones, use the `dst auto-adjust` command in voice register global configuration mode. To disable daylight saving time auto adjustment, use the `no` form of this command.

```
dst auto-adjust
no dst auto-adjust
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Automatic adjustment of daylight saving time on SIP phones is enabled.

**Command Modes**
Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Automatic adjustment for daylight saving time is enabled by default. To disable auto adjusting for DST, use the `no dst auto-adjust` command. To set the start and stop times for DST, use the `dst` command.

**Examples**
The following example shows how to disable the automatic adjustment for daylight saving time:

```
Router(config)# voice register global
Router(config-register-global)# no dst auto-adjust
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>date-format (voice register global)</code></td>
<td>Sets the date display format on SIP phones in a Cisco CME system.</td>
</tr>
<tr>
<td><code>dst (voice register global)</code></td>
<td>Sets the start and stop time if using daylight saving time on SIP phones.</td>
</tr>
<tr>
<td><code>time-format (voice register global)</code></td>
<td>Selects a 12-hour clock or a 24-hour clock for the time display format on Cisco IP phones in a SIP CME system.</td>
</tr>
<tr>
<td><code>timezone (voice register global)</code></td>
<td>Sets the time zone used for SIP phones in a Cisco CME system.</td>
</tr>
</tbody>
</table>
dtmf-relay (voice register pool)

To specify the list of DTMF relay methods that can be used to relay dual-tone multifrequency (DTMF) audio tones between Session Initiation Protocol (SIP) endpoints, use the `dtmf-relay` command in voice register pool configuration mode. To send the DTMF audio tones as part of an audio stream, use the `no` form of this command.

```
dtmf-relay [cisco-rtp] [rtp-nte] [sip-notify] [sip-kpml]
no dtmf-relay
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>cisco-rtp</code></td>
<td>Forwards DTMF audio tones by using Real-Time Transport Protocol (RTP) with a Cisco proprietary payload type. This keyword is supported only for dial peers that are created by incoming REGISTERs from a SIP gateway. It is not supported for dial peers that are created by a SIP Cisco IP phone.</td>
</tr>
<tr>
<td><code>rtp-nte</code></td>
<td>Forwards DTMF audio tones by using Real-Time Transport Protocol (RTP) with a Named Telephone Event (NTE) payload.</td>
</tr>
<tr>
<td><code>sip-notify</code></td>
<td>Forwards DTMF audio tones by using SIP-NOTIFY messages. This keyword is supported only for dial peers that are created by incoming REGISTERs from a SIP gateway. It is not supported for dial peers that are created by a SIP Cisco IP phone.</td>
</tr>
<tr>
<td><code>sip-kpml</code></td>
<td>Forwards DTMF audio tones through Keypad Markup Language (KPML) messages.</td>
</tr>
</tbody>
</table>

### Command Default

DTMF tones are disabled and sent in-band. That is, they remain in the audio stream.

### Command Modes

Voice register pool configuration (config-register-pool)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(4)T</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was added to Cisco Unified CME.</td>
</tr>
<tr>
<td>15.1(1)T1</td>
<td>Cisco Unified CME 8.1 Cisco SIP SRST 8.1</td>
<td>The sip-kpml keyword was added to this command.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

During Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified CME registration, a dial peer is created and that dial peer has a default DTMF relay of in-band.

This command command allows you to change the default to a desired value. You must use one or more keywords when configuring this command.

DTMF audio tones are generated when you press a button on a Touch-Tone phone. The tones are compressed at one end of the call and when the digits are decompressed at the other end, there is a risk that they can become distorted. DTMF relay reliably transports the DTMF audio tones generated after call establishment out-of-band.
The SIP Notify method sends Notify messages bidirectionally between the originating and terminating gateways for a DTMF event during a call. If multiple DTMF relay mechanisms are enabled on a SIP dial peer and are negotiated successfully, the SIP Notify method takes precedence.

SIP Notify messages are advertised in an Invite message to the remote end only if the `dtmf-relay` command is set.

For SIP calls, the most appropriate methods to transport DTMF tones are RTP-NTE or SIP-NOTIFY.

**Note**

The `cisco-rtp` keyword is a proprietary Cisco implementation. If the proprietary Cisco implementation is not supported, the DTMF relay feature does not function, and the gateway sends DTMF tones in-band.

- The `sip-notify` keyword is available only if the VoIP dial peer is configured for SIP.

**Examples**

**Cisco Unified CME**

The following example shows how to enable the RTP-NTE and SIP-NOTIFY mechanisms for DTMF relay for SIP phone 4:

Router(config)# voice register pool 4
Router(config-register-pool)# dtmf-relay rtp-nte sip-notify

The following example shows sip-kpml option configured for dtmf-relay in voice register pool 5:

Router#show running config
voice register global
    mode cme
    source-address 10.32.153.49 port 5060
    max-dn 200
    max-pool 100
voice register pool 5
    id mac 0023.3319.8B7B
    type 7945
    number 1 dn 5
dtmf-relay sip-kpml
    username betaone password cisco
    codec g711ulaw
    no vad

**Cisco Unified SIP SRST**

The following is sample output from the `show running-config` command that shows that voice register pool 1 has been set up to send DTMF tones:

```
voice register pool 1
    application SIP.app
    incoming called-number 308
    voice-class codec 1
dtmf-relay rtp-nte
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>dtmf-relay (voice over IP)</code></td>
<td>Specifies how an H.323 or SIP gateway relays DTMF tones between telephony interfaces and an IP network.</td>
</tr>
</tbody>
</table>
dtmf-relay (voice register pool)
Cisco Unified CME Commands: E

• elin, on page 364
• elin (voice emergency response settings), on page 365
• em external, on page 367
• em keep-history, on page 368
• em logout, on page 369
• emadmin login, on page 370
• emadmin logout, on page 372
• emergency response callback, on page 373
• emergency response location, on page 374
• emergency response zone, on page 376
• encrypt password, on page 378
• ephone, on page 379
• ephone-dn, on page 381
• ephone-dn-template, on page 383
• ephone-dn-template (ephone-dn), on page 385
• ephone-hunt, on page 387
• ephone-hunt login, on page 390
• ephone-hunt statistics write-all, on page 391
• ephone-template, on page 393
• ephone-template (ephone), on page 396
• ephone-type, on page 398
• exclude, on page 400
• exclude (voice register), on page 402
• expiry, on page 403
• extension-assigner tag-type, on page 405
• extension-range, on page 407
• external-ring (voice register global), on page 409
elin

To create a PSTN number that replaces a 911 caller’s extension, use the `elin` command in voice emergency response location configuration mode. To remove the number, use the `no` form of this command.

`elin {1|2} number`
`no elin [{1|2}]`

**Syntax Description**

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>`{1</td>
<td>2}`</td>
</tr>
<tr>
<td><code>number</code></td>
<td>PSTN number that replaces a 911 caller’s extension.</td>
</tr>
</tbody>
</table>

**Command Default**

No replacement number is created.

**Command Modes**

Voice emergency response location configuration (cfg-emrgncy-resp-location)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1</td>
<td>This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)</td>
<td>This command was added for Cisco Unified CME.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to specify an ELIN, a PSTN number that will replace the caller’s extension.

The PSAP will see this number and use it to query the ALI database to locate the caller. The PSAP also uses this command for callbacks.

You can configure a second ELIN using the `elin 2` command. If two ELINs are configured, the system selects an ELIN using a round-robin algorithm. If an ELIN is not defined for the ERL, the PSAP sees the original calling number.

**Examples**

In the following example, all IP phones with the IP address of 10.X.X.X or 192.168.X.X are automatically associated with this ERL. If one of the phones dials 911, its extension is replaced with 408 555-0100 before it goes to the PSAP. The PSAP will see that the caller’s number is 408 555-0100.

```
voice emergency response location 1
elin 1 4085550100
subnet 10.0.0.0 255.0.0.0
subnet 2 192.168.0.0 255.255.0.0
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>subnet</code></td>
<td>Defines which IP phones are part of this ERL.</td>
</tr>
</tbody>
</table>
elin (voice emergency response settings)

To create a default ELIN that is used if no ERL has a subnet mask that matches the current 911 caller’s IP phone address, use the `elin` command in voice emergency response settings configuration mode. To remove the number, use the `no` form of this command.

```
elin  number
no  elin
```

**Syntax Description**

- `number` An E.164 number to be used as the default ELIN.

**Command Default**

No default ELIN number is created.

**Command Modes**

Voice emergency response settings configuration (cfg-emrgncy-resp-settings)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to specify an E.164 number to be the default ELIN if the 911 caller’s IP phone address does not match the subnet of any location in any ERL zone. The default ELIN can be an existing ELIN already defined in an ERL or it can be unique.

**Examples**

In this example, the ELIN (4085550101) defined in the voice emergency response settings configuration is used if the 911 caller’s IP phone address does not match any of the voice emergency response locations. After the 911 call is placed to the PSAP, the PSAP has 120 minutes to call back 408 555-0101 to reach the 911 caller. If the call history has expired (after 120 minutes), any callback is routed to extension 7500.

```
voice emergency response settings
callback 7500
elin 4085550101
expiry 120
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>callback</code></td>
<td>Default phone number to contact if a 911 callback cannot find the last 911 caller from the ERL.</td>
</tr>
<tr>
<td><code>expiry</code></td>
<td>Number of minutes a 911 call is associated to an ELIN in case of a callback from the 911 operator.</td>
</tr>
<tr>
<td><code>logging</code></td>
<td>Syslog informational message printed to the console each time an emergency call is made.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>voice emergency response settings</td>
<td>Creates a tag for identifying settings for E911 behavior.</td>
</tr>
</tbody>
</table>
em external

To remove the login page under the Extension Mobility option from the Services menu on IP phones in Cisco Unified CME, use the `em external` command in telephony-service configuration mode. To return to default, use the `no` form of this command.

```
em external
no em external
```

**Syntax Description**

This command has no keywords or arguments.

**Command Default**

Login page for Extension Mobility is accessible under the Extension Mobility option in the Services menu.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command removes the Extension Mobility login page from the Sevices menu on all IP phones registered in a Cisco Unified CME system on which Extension Mobility is enabled.

**Examples**

The following partial output shows the configuration for this command:

```
router# show running-configuration
.
.
.
telephony-service
   em external
   em logout 1:0
   max-ephones 10
   max-dn 100
   ip source-address 10.0.0.1 port 2000
   url authentication http://10.0.0.1/CCMCIP/authenticate.asp
   cnf-file location flash:
   cnf-file perphone
   max-conferences 8 gain -6
   transfer-system full-consult
   create cnf-files version-stamp Jan 01 2002 00:00:00
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ip http server</td>
<td>Enables the HTTP server on the Cisco Unified CME router that hosts the service URL for the Extension Mobility Login and Logout page.</td>
</tr>
</tbody>
</table>
**em keep-history**

To disable Automatic Clear Call History for Extension Mobility phones in Cisco Unified CME, use the `em keep-history` command in telphony-service configuration mode. To return to the default, use the `no` form of this command.

```
em keep-history
no em keep-history
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Call history record is automatically cleared when a user logs out from an Extension Mobility phone.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command disables Automatic Clear Call History for Extension Mobility phones in Cisco Unified CME. In Cisco Unified CME 4.3 and later versions, the EM manager in Cisco Unified CME sends commands to a phone to clear call history anytime a user is logs out from Extension Mobility. Use this command in telephony-service configuration mode to disable this feature at a system-level.

**Examples**

The following example shows how to configure Extension Mobility in Cisco Unified CME to keep, not clear, call histories after users log out from Extension Mobility phones:

```
Router(config)# telephony-service
Router(config-telephony)# em keep-history
Router(config-telephony)#
```
em logout

To configure up to three time-of-day based timers for automatically logging out all Extension Mobility users, use the em logout command in telephony-service configuration mode. To disable the timer, use the no form of this command.

`em logout time1 [time2] [time3]`
`no command time1 [time2] [time3]`

**Syntax Description**

| time | Time of day after which all users that are logged into Extension Mobility are logged out from Extension Mobility. Range: 00:00 to 24:00 on a 24-hour clock. |

**Command Default**

No time-of-day timer is created for automatically logging out Extension Mobility users.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
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<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command creates up to three time-of-day timers for automatically logging out all Extension Mobility users. If an Extension Mobility user is using the phone when automatic logout occurs, the user is logged out after the active call is completed.

The call history record is automatically cleared when a user logs out from an Extension Mobility phone. To disable Automatic Clear Call History on all Extension Mobility phones, use the em keep-history command in telephony-service configuration mode.

**Examples**

The following example shows how to configure two time-of-day timers to automatically log out all logged-on Extension Mobility users at 5:30 PM and again at midnight each day:

```
Router(config)# telephony-service
Router(config-telephony)# em logout 17:30 24:00
Router(config-telephony)#
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>em keep-history</td>
<td>Disables Automatic Clear Call History for Extension Mobility in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
emadmin login

To permit an external application to log into a Cisco Unified IP phone that is enabled for Extension Mobility in Cisco Unified CME, use the `emadmin login` command in privileged EXEC mode.

```
emadmin login name ephone-tag
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>name</td>
<td>Credential for Extension Mobility. This credential must be already configured by using the <code>user</code> command in voice-user-profile configuration mode.</td>
</tr>
<tr>
<td>ephone-tag</td>
<td>Unique identifier for IP phone that is enabled for Extension Mobility. This tag must already be configured by using the <code>ephone</code> command.</td>
</tr>
</tbody>
</table>

**Command Default**

External application cannot log into an Extension Mobility phone.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables an external application, such as a CSTA client application, to log into an Extension Mobility phone.

Before using this command, configure a credential in Extension Mobility by using the `user` command in voice-user-profile configuration mode.

The IP phone to be accessed must be enabled for Extension Mobility.

The application remains logged into the phone until it is manually or automatically logged out from the Extension Mobility phone.

This command does not have a `no` form.

**Examples**

The following example shows how to configure this command to log an application into an Extension Mobility phone (2) using the “user204” credential:

```
Router# login user204
2
Router#
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>emadmin logout</td>
<td>Logs out an external application from Extension Mobility.</td>
</tr>
<tr>
<td>em logout</td>
<td>Creates up to three time-of-day timers for automatically logging out all Extension Mobility users.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>logout-profile</td>
<td>Enables an IP phone for Extension Mobility.</td>
</tr>
<tr>
<td>max-idle-time</td>
<td>Creates an idle-duration timer for automatically logging out an Extension Mobility user.</td>
</tr>
<tr>
<td>user</td>
<td>Creates an authentication credential to be used by Extension Mobility.</td>
</tr>
</tbody>
</table>
emadmin logout

To manually log out an external application from Extension Mobility, use the emadmin logout command in privileged EXEC mode. To return to default, use the no form of this command.

```
emadmin logout name
no emadmin logout name
```

**Syntax Description**

- **name**: Already-configured credential in Extension Mobility user profile.

**Command Default**

Application remains logged into the Extension Mobility phone until logged out.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables an external application, such as a CSTA client application, to log out of an Extension Mobility phone.

**Examples**

The following example shows how to configure this command to log out an application that logged into an Extension Mobility phone using the “user204” credential:

```
Router# logout user204
Router#
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>user</td>
<td>Creates an authentication credential to be used by Extension Mobility.</td>
</tr>
</tbody>
</table>
emergency response callback

To define a dial peer that is used for 911 callbacks from the PSAP, use the emergency response callback command in voice dial-peer configuration mode. To remove the definition of the dial peer as an incoming link from the PSAP, use the no form of this command.

```
emergency response callback
no emergency response callback
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

The dial peer is not defined as an incoming link from the PSAP.

**Command Modes**

Dial-peer configuration (config-dial-peer)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1</td>
<td>This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)</td>
<td>This command was added for Cisco Unified CME.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to define which dial peer is used for 911 callbacks from the PSAP. You can define multiple dial peers with this command.

**Examples**

The following example shows a dial peer defined as an incoming link from the PSAP. If 408 555-0100 is configured as the ELIN for an ERL, this dial peer recognizes that an incoming call from 408 555-0100 is a 911 callback.

```
dial-peer voice 100 pots
    incoming called-number 4085550100
    port 1/1:D
    direct-inward-dial
    emergency response callback
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>emergency response location</td>
<td>Associates an ERL to either a SIP phone, ephone, or dial peer.</td>
</tr>
<tr>
<td>emergency response zone</td>
<td>Defines a dial peer that is used by the system to route emergency calls to the PSAP.</td>
</tr>
<tr>
<td>voice emergency response location</td>
<td>Creates a tag for identifying an ERL for the enhanced 911 service.</td>
</tr>
</tbody>
</table>
emergency response location

To associate an emergency response location (ERL) for Enhanced 911 Services with a dial peer, ephone, ephone-template, voice register pool, or voice register template, use the `emergency response location` command in dial peer, ephone, ephone-template, voice register pool, or voice register template configuration mode. To remove the association, use the `no` form of this command.

```
emergency response location tag
no emergency response location tag
```

**Syntax Description**

| Tag | Unique number that identifies an existing ERL tag defined by the `voice emergency response location` command. |

**Command Default**

No ERL tag is associated with a dial peer, ephone, ephone-template, voice register pool, or voice register template.

**Command Modes**

Dial-peer configuration (config-dial-peer)
Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)
Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1</td>
<td>This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)</td>
<td>This command was added to Cisco Unified CME in the ephone-template and voice register template configuration modes.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to assign an ERL to phones individually. Depending on the type of phones (endpoints) that you have, you can assign an ERL to a phone’s:

- Dial-peer configuration
- Ephone
- Ephone-template
- Voice register pool
- Voice register template

**Note**

The ephone-template and voice register template modes are not for Cisco Unified SRST or Cisco Unified SIP SRST. Voice register pool mode is not available for Cisco Unified SRST.
These methods of associating a phone with an ERL are alternatives to assigning a group of phones that are on the same subnet as an ERL.

The tag used by this command is an integer from 1 to 2147483647 and refers to an existing ERL tag that is defined by the voice emergency response location command. If the tag does not refer to a valid ERL configuration, the phone is not associated to an ERL. For Cisco Unified IP phones, the IP address is used to find the inclusive ERL subnet. For phones on a VoIP trunk or FXS/FXO trunk, the PSAP gets a reorder tone.

**Examples**

The following example shows how to assign an ERL to a phone’s dial peer:

```
dial-peer voice 12 pots
  emergency response location 18
```

The following example shows how to assign an ERL to a phone’s ephone:

```
ephone 41
  emergency response location 22
```

The following example shows how to assign an ERL to one or more SCCP phones:

```
ephone-template 6
  emergency response location 8
```

The following example shows how to assign an ERL to a phone’s voice register pool:

```
voice register pool 4
  emergency response location 21
```

The following example shows how to assign an ERL to one or more SIP phones:

```
voice register template 4
  emergency response location 8
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>emergency response callback</td>
<td>Defines a dial peer that is used for 911 callbacks from the PSAP.</td>
</tr>
<tr>
<td>emergency response zone</td>
<td>Defines a dial peer that is used by the system to route emergency calls to the PSAP.</td>
</tr>
<tr>
<td>voice emergency response location</td>
<td>Creates a tag for identifying an ERL for the enhanced 911 service.</td>
</tr>
</tbody>
</table>
emergency response zone

To define a dial peer that is used by the system to route emergency calls to a PSAP, use the emergency response zone command in voice dial-peer configuration mode. To remove the definition of the dial peer as an outgoing link to the PSAP, use the no form of this command.

**Syntax Description**

```
emergency response zone zone-tag
no emergency response zone
```

**Command Default**

The dial peer is not defined as an outgoing link to the PSAP. Therefore, E911 services are not enabled.

**Command Modes**

Dial-peer configuration (config-dial-peer)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1</td>
<td>This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)</td>
<td>The zone-tag argument was added and this command was added for Cisco Unified CME.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to specify that any calls using this dial peer are processed by the E911 software. To enable any E911 processing, the emergency response zone command must be enabled under a dial peer.

If no zone tag is specified, the system looks for a matching ELIN to the E911 caller’s phone by searching each emergency response location that was configured using the emergency response location command.

If a zone tag is specified, the system looks for a matching ELIN using sequential steps according to the contents of the configured zone. For example, if the E911 caller’s phone has an explicit ERL assigned, the system first looks for that ERL in the zone. If not found, it then searches each location within the zone according to assigned priority numbers, and so on. If all steps fail to find a matching ELIN, the default ELIN is assigned to the E911 caller’s phone. If no default ELIN is configured, the E911 caller’s automatic number identification (ANI) number is communicated to the Public Safety Answering Point (PSAP).

This command can be defined in multiple dial peers. The zone tag option allows only ERLs defined in that zone to be routed on this dial peer. Also, this command allows callers dialing the same emergency number to be routed to different voice interfaces based on the zone that includes their ERL.

**Examples**

The following example shows a dial peer defined as an outgoing link to the PSAP. Emergency response zone 10 is created and only calls from this zone are routed through 1/0/0.

```
dial-peer voice 911 pots
destination-pattern 9911
```
prefix 911
emergency response zone 10
port 1/0/0

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>emergency response callback</td>
<td>Defines a dial peer that is used for 911 callbacks from the PSAP.</td>
</tr>
<tr>
<td>emergency response location</td>
<td>Associates an ERL to either a SIP phone, ephone, or dial peer.</td>
</tr>
<tr>
<td>voice emergency response location</td>
<td>Creates a tag for identifying an ERL for E911 services.</td>
</tr>
<tr>
<td>voice emergency response zone</td>
<td>Creates an emergency response zone within which ERLs can be grouped.</td>
</tr>
</tbody>
</table>
encrypt password

To encrypt the password that is configured on Unified CME, use the `encrypt password` command in `telephony-service` configuration mode. To disable password encryption, use the `no` form of this command.

```
encrypt password
no encrypt password
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
This command is enabled by default.

**Command Modes**
Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a</td>
<td>Unified CME</td>
<td>The command is introduced.</td>
</tr>
<tr>
<td>Release</td>
<td>12.6</td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**
The CLI command `encrypt password` is enabled by default on Unified CME router. However, it is mandatory to configure `key config-key password-encrypt [Master key]` and `password encryption aes` along with `encrypt password` to support encryption on Unified CME router.

**Note**
If the key used to encrypt the password is replaced with a new key (replace key or re-key), then the password is re-encrypted with the new key.

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>password (auto-register)</code></td>
<td>Configures the mandatory password for automatic registration of SIP phones with Unified CME.</td>
</tr>
</tbody>
</table>
**ephone**

To enter Ethernet phone (ephone) configuration mode for an IP phone for the purposes of creating and configuring an ephone, use the `ephone` command in global configuration mode. To disable the ephone and remove the IP phone configuration, use the `no` form of this command.

```
ephone  phone-tag
no ephone  phone-tag
```

### Syntax Description

| **phone-tag** | Unique sequence number that identifies an ephone during configuration tasks. The maximum number is platform-dependent; refer to Cisco IOS command-line interface (CLI) help. |

### Command Default

No Cisco IP phone is configured.

### Command Modes

Global configuration (config)

### Command History

<table>
<thead>
<tr>
<th><strong>Cisco IOS Release</strong></th>
<th><strong>Cisco Product</strong></th>
<th><strong>Modification</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8).</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use the `ephone` command to enter ephone configuration mode. Use ephone configuration mode to create and configure Cisco Unified IP phones in Cisco Unified CME.

Before this command can be used for the first time, you must set the maximum number of ephones using the `max-ephones` command in telephony-service configuration mode. The maximum number of ephones varies by router platform and software version.

### Examples

The following example enters ephone configuration mode for a phone with the identifier 4 and assigns ephone-dn 1 to button 1:

```
Router(config)# ephone 4
Router(config-ephone)# button 1:1
```

### Related Commands

<table>
<thead>
<tr>
<th><strong>Command</strong></th>
<th><strong>Description</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><code>button</code></td>
<td>Assigns a button number to the Cisco IP phone directory number.</td>
</tr>
<tr>
<td><code>ephone-dn</code></td>
<td>Enters ephone-dn configuration mode.</td>
</tr>
<tr>
<td><code>mac-address</code></td>
<td>Configures the MAC address of a Cisco IP phone.</td>
</tr>
<tr>
<td><code>max-ephones</code></td>
<td>Configures the maximum number of Cisco IP phones that can be supported by a router.</td>
</tr>
<tr>
<td><code>restart (ephone)</code></td>
<td>Performs a fast reboot of a single phone associated with a Cisco CME router.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>restart all (telephony-service)</td>
<td>Performs a fast reboot of all phones associated with a Cisco CME router.</td>
</tr>
</tbody>
</table>
**ephone-dn**

To enter ephone-dn configuration mode to configure a directory number for an IP phone line, intercom line, paging line, voice-mail port, or message-waiting indicator (MWI), use the `ephone-dn` command in global configuration mode. To delete an ephone-dn, use the `no` form of this command.

```
ephone-dn  dn-tag  [{dual-line|octo-line}]
no  ephone-dn  dn-tag
```

<table>
<thead>
<tr>
<th><strong>Syntax Description</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>dn-tag</strong></td>
</tr>
<tr>
<td><strong>dual-line</strong></td>
</tr>
<tr>
<td><strong>octo-line</strong></td>
</tr>
</tbody>
</table>

**Command Default**
No ephone-dn is configured.

**Command Modes**
Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>The <code>dual-line</code> keyword was added.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.4</td>
<td>The <code>dual-line</code> keyword was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>The <code>octo-line</code> keyword was added.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to enter ephone-dn configuration mode to create directory numbers. In ephone-dn configuration mode, you assign an extension number using the `number` command, a name to appear in the local directory using the `name` command, and other parameters using various commands.

Before using the `ephone-dn` command, you must set the maximum number of ephone-dns to support in your system by using the `max-dn` command. The maximum number of ephone-dns that you can create depends on the software version, router platform, and amount of memory that you have installed.

A dual-line ephone-dn has one virtual voice port and two channels to handle two independent calls. This capacity allows call waiting, call transfer, and conference functions within a single directory number. Dual-line mode is supported on all phone types, but is not appropriate for voice-mail numbers, intercoms, or ephone-dns used for message-waiting indicators, paging, loopback, or hunt groups. Overlays of single-line hunt groups onto dual-line buttons are supported.
An octo-line directory number supports up to eight active calls, both incoming and outgoing, on a single phone button. Unlike a dual-line directory number, which is shared exclusively among phones, an octo-line directory number can split its channels among the phones sharing the directory number. All phones sharing the octo-line directory number are allowed to initiate or receive calls on the idle channels of the directory number.

Ephone-dns are created in single-line mode if the dual-line or octo-line keyword is not used. To change an ephone-dn from one mode to another, for example from dual-line mode to single-line mode, you must delete the ephone-dn and then re-create it.

**Examples**

The following example shows how to create directory number 1 with extension 5576:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5576
```

The following example shows an ephone-dn with the number 1001 in dual-line mode. The no huntstop command allows calls to continue to hunt to other ephone-dns if this one is busy or does not answer. The huntstop channel command disables call hunting to the second channel of this ephone-dn if the first channel is busy or does not answer.

```
Router(config)# ephone-dn 10 dual-line
Router(config-ephone-dn)# number 1001
Router(config-ephone-dn)# no huntstop
Router(config-ephone-dn)# huntstop channel
```

The following example shows an ephone-dn with the number 2001 in octo-line mode. The huntstop channel command enables call hunting to up to six channels of this ephone-dn. The remaining two channels are available for outgoing calls or features such as call transfer, call waiting, and conferencing.

```
Router(config)# ephone-dn 20 octo-line
Router(config-ephone-dn)# number 2001
Router(config-ephone-dn)# huntstop channel 6
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>huntstop (ephone-dn and ephone-dn-template)</td>
<td>Disables call hunting for directory numbers or channels.</td>
</tr>
<tr>
<td>max-dn</td>
<td>Sets the maximum number of ephone-dns that can be configured.</td>
</tr>
<tr>
<td>name</td>
<td>Associates a name with an extension (ephone-dn).</td>
</tr>
<tr>
<td>number</td>
<td>Associates a telephone or extension number with a directory number (ephone-dn).</td>
</tr>
</tbody>
</table>
**ephone-dn-template**

To enter ephone-dn-template configuration mode and create an ephone-dn template containing a standard set of ephone-dn features, use the `ephone-dn-template` command in global configuration mode. To delete an ephone-dn template, use the `no` form of this command.

```
ephone-dn-template template-tag
no ephone-dn-template template-tag
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>template-tag</code></td>
<td>Identifier for this ephone-dn template. Range is from 1 to 15.</td>
</tr>
</tbody>
</table>

**Command Default**

No ephone-dn template is created.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to create an ephone-dn template. An ephone-dn template contains a set of ephone-dn attributes that you can easily apply to one or more ephone-dns.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Type `?` in ephone-dn-template configuration mode to see the commands that are available in this mode. The following example shows CLI help for ephone-dn-template configuration mode:

```
Router(config-ephone-dn-template)# ?
Ephone Dn template configuration commands:
  call-forward       Define E.164 telephone number for call forwarding
  call-waiting       Config call-waiting option
  caller-id          Configure port caller id parameters
  corlist            Class of Restriction on dial-peer for this dn
  default            Set a command to its defaults
  description        dn desc, for DN Qualified Display Name
  exit               Exit from ephone-dn-template configuration mode
  hold-alert         Set Call On-Hold timeout alert parameters
  huntstop           Stop hunting on Dial-Peers
  mwi                set message waiting indicator options (mwi)
  no                 Negate a command or set its defaults
  pickup-group       set the call pickup group number for the DN
  translate          Translation rule
  translation-profile Translation profile
```

After creating an ephone-dn template, apply the template to one or more ephone-dns using the `ephone-dn-template` command in ephone-dn configuration mode. Even though you can define up to 15 different ephone templates, you cannot apply more than one template to a particular ephone-dn.
If you try to apply a second ephone-dn template to an ephone-dn that already has a template applied to it, the second template will overwrite the first ephone-dn template configuration after you use the `restart` command to reboot the phone.

To view your ephone-dn-template configurations, use the `show telephony-service ephone-dn-template` command. To see which ephone-dns have templates applied to them, use the `show running-config` command.

**Examples**

The following example creates ephone-dn template 3, which sets call forwarding on busy and no answer to forward calls to extension 4000 and sets the pickup group to 4. Ephone-dn template 3 is then applied to ephone-dn 23 and ephone-dn 33, which appear on phones 13 and 14, respectively.

```plaintext
ephone-dn-template 3
  call-forwarding busy 4000
call-forwarding noan 4000 timeout 30
pickup group 4
ephone-dn 23
timeout 33

ephone-dn-template 3

ephone-dn 33
  number 3333
ephone-dn-template 3

ephone 13
  button 1:23
ephone 14
  button 1:33
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ephone-dn-template (ephone-dn)</code></td>
<td>Applies an ephone-dn template to an ephone-dn.</td>
</tr>
<tr>
<td><code>restart (ephone)</code></td>
<td>Performs a fast reboot of a single phone associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td><code>restart (telephony-service)</code></td>
<td>Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td><code>show telephony-service ephone-dn-template</code></td>
<td>Displays ephone-dn-template configurations.</td>
</tr>
</tbody>
</table>
ephone-dn-template (ephone-dn)

To apply an ephone-dn template to an ephone-dn, use the **ephone-dn-template** command in ephone-dn configuration mode. To remove the ephone-dn template, use the **no** form of this command.

```
ephone-dn-template  template-tag
no ephone-dn-template  template-tag
```

**Syntax Description**
- **template-tag**: The template tag for a template created with the **ephone-dn-template** command in global configuration mode. Range is from 1 to 15.

**Command Default**
No ephone-dn template is applied to the ephone-dn.

**Command Modes**
Ephone-dn configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use the **ephone-dn-template** command in ephone-dn configuration mode to apply an ephone-dn template to an ephone. You cannot apply more than one ephone-dn template to an ephone-dn.

If you try to apply a second ephone-dn template to an ephone-dn that already has an ephone-dn template applied to it, the second template will overwrite the first ephone-dn template configuration.

To view your ephone-dn-template configurations, use the **show telephony-service ephone-dn-template** command.

**Examples**
The following example shows how to create ephone-dn template 3, which sets call forwarding on busy and no answer to forward calls to extension 4000 and sets the pickup group to 4, and apply the template to ephone-dn 23 and ephone-dn 33, which appear on ephones 13 and 14, respectively.

```
ephone-dn-template 3
call-forwarding busy 4000
call-forwarding noan 4000 timeout 30
pickup group 4
ephone-dn 23
number 2323
ephone-dn-template 3
ephone-dn 33
number 3333
ephone-dn-template 3
ephone 13
button 1:23
ephone 14
button 1:33
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>ephone-dn</td>
<td>Enters ephone-dn configuration mode.</td>
</tr>
<tr>
<td></td>
<td>ephone-dn-template</td>
<td>Creates an ephone-dn template and enters ephone-dn-template configuration mode.</td>
</tr>
<tr>
<td></td>
<td>show telephony-service ephone-dn-template</td>
<td>Displays ephone-dn template configurations.</td>
</tr>
</tbody>
</table>
ephone-hunt

To enter ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco Unified CME system, use the **ephone-hunt** command in global configuration mode. To delete a hunt group, use the **no** form of this command.

```
ephone-hunt  hunt-tag  {longest-idle|peer|sequential}
no  ephone-hunt  hunt-tag
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>hunt-tag</strong></td>
</tr>
<tr>
<td><strong>longest-idle</strong></td>
</tr>
<tr>
<td><strong>peer</strong></td>
</tr>
<tr>
<td><strong>sequential</strong></td>
</tr>
</tbody>
</table>

**Command Default**

No hunt group is defined.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.3(11)T</td>
<td>Cisco CME 3.2</td>
<td>The <strong>longest-idle</strong> keyword was added.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The maximum number of hunt groups was increased from 10 to 100.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command with the maximum number of hunt groups increased to 100 was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the **ephone-hunt** command to enter ephone-hunt configuration mode. Use ephone-hunt configuration mode to create ephone hunt groups in a Cisco Unified CME system.

A hunt group is a list of phone numbers that are assigned to take turns receiving incoming calls for one number, a pilot number that is defined with the **pilot** command. The list of numbers in the hunt group is defined using the **list** command. If a number in the list is busy or does not answer, the call is redirected to the next number in the list. The last number tried is the final number, which is defined using the **final** command.

The order in which the numbers are chosen can be longest-idle, peer, or sequential.
• If the order is longest-idle, each hop is directed to the ephone-dn that has been idle the longest. Idle time is determined from the last time that a phone registered, reregistered, or went on-hook.
• If the order is peer, the first number to which calls are directed is the number to the right of the number in the list that was the last number to ring on the previous occasion that the hunt group was called. If that number is busy or does not answer, the call is directed to the next number in the list and, in the process, circles back to the beginning of the list. In peer hunt groups, the hops command specifies how many times a call can hop from number to number before going to the final number, after which the call is no longer forwarded.
• If the order is sequential, the first number to which calls are directed is always the first number in the list. If that number is busy or does not answer, the call is redirected to the next available number in the list, from left to right.

Note

If the number of times that a call is redirected to a new number exceeds five, the max-redirect command must be used to increase the allowable number of redirects in the Cisco Unified CME system.

To configure a new hunt group, you must specify the longest-idle, peer, or sequential keyword. To change an existing ephone hunt group configuration, the keyword is not required. To change the type of hunt group from peer to sequential or sequential to peer, you must remove the existing hunt group first using the no form of the command and then recreate it.

Examples

The following example defines longest-idle ephone hunt group 1 with a pilot number 7501, a final number 8000, and 11 numbers in the list. After a call is redirected six times (makes six hops), it is redirected to the final number 8000.

ephone-hunt 1 longest-idle
    pilot 7501
    members logout
    list 7001, 7002, 7023, 7028, 7045, 7062, 7072, 7079, 7085, 7099
    final 8000
    preference 1
    hops 6
    timeout 20
    no-reg

The following example defines peer hunt group number 2. Callers dial the pilot number 5610 to reach the hunt group. The first extension to ring the first time that this hunt group is called is 5601. If 5601 does not answer, the hunt proceeds from left to right, beginning with the extension directly to the right of 5601, for four hops. If none of those extensions answers before the hops limit is reached, the call is forwarded to extension 6000, which is the number for the voice-mail service.

If extension 5601 answers the first call, then the second time someone calls the hunt group, the first extension to ring is 5602. If this call hops until extension 5617 answers it, then the third time someone calls the hunt group, the first extension to ring is 5633. If extension 5633 does not answer, the call is redirected to extension 5601, and so forth.

Router(config)# ephone-hunt 2 peer
Router(config-ephone-hunt)# pilot 5610
Router(config-ephone-hunt)# members logout
Router(config-ephone-hunt)# list 5601, 5602, 5617, 5633
Router(config-ephone-hunt)# final 6000
The following example defines sequential hunt group number 1. When callers dial extension 5601, the first phone to ring is 5001, then 5002, 5017, and 5028. If none of those extensions answers, the call is forwarded to extension 6000, which is the number for the voice-mail service.

Router(config)# ephone-hunt 1 sequential
Router(config-ephone-hunt)# pilot 5601
Router(config-ephone-hunt)# members logout
Router(config-ephone-hunt)# list 5001, 5002, 5017, 5028
Router(config-ephone-hunt)# final 6000
Router(config-ephone-hunt)# preference 1
Router(config-ephone-hunt)# timeout 30
Router(config-ephone-hunt)# exit

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>final</td>
<td>Defines the last ephone-dn in an ephone hunt group.</td>
</tr>
<tr>
<td>hops</td>
<td>Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.</td>
</tr>
<tr>
<td>list</td>
<td>Defines the ephone-dns that participate in an ephone hunt group.</td>
</tr>
<tr>
<td>max-redirect</td>
<td>Changes the current number of allowable redirects in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>members logout</td>
<td>Sets all static members initial state to logout.</td>
</tr>
<tr>
<td></td>
<td>The <code>members logout</code> command is rejected if configured after the <code>list</code> command.</td>
</tr>
<tr>
<td>no-reg (ephone-hunt)</td>
<td>Specifies that the pilot number of this ephone hunt group should not register with the H.323 gatekeeper.</td>
</tr>
<tr>
<td>pilot</td>
<td>Defines the ephone-dn that is dialed to reach an ephone hunt group.</td>
</tr>
<tr>
<td>preference (ephone-hunt)</td>
<td>Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.</td>
</tr>
<tr>
<td>timeout (ephone-hunt)</td>
<td>Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.</td>
</tr>
</tbody>
</table>
ephone-hunt login

To authorize an ephone-dn to dynamically join and leave an ephone hunt group, use the `ephone-hunt login` command in ephone-dn configuration mode. To disable this capability, use the `no` form of this command.

```
ephone-hunt login
no ephone-hunt login
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

An ephone-dn is not allowed to dynamically join and leave ephone hunt groups.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `show ephone-hunt` command to display current hunt group members, including those who joined the group dynamically.

**Examples**

The following example creates five ephone-dns and a hunt group that includes the first two ephone-dns and two wildcard slots. The last three ephone-dns are enabled for group hunt dynamic membership. Each of them can join and leave the hunt group whenever one of the slots is available.

```
ephone-dn 22
  number 4566
ephone-dn 23
  number 4567
ephone-dn 24
  number 4568
  ephone-hunt login
ephone-dn 25
  number 4569
  ephone-hunt login
ephone-dn 26
  number 4570
  ephone-hunt login
ephone-hunt 1 peer
  list 4566,4567,*,*
  final 7777
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>show ephone-hunt</code></td>
<td>Displays ephone-hunt group configuration, current status, and statistics.</td>
</tr>
</tbody>
</table>
ephone-hunt statistics write-all

Effective with Cisco Unified CME 9.0, the **ephone-hunt statistics write-all** command is replaced by the **hunt group statistics write-all** command in privileged EXEC mode. For more information, see the **hunt group statistics write-all** command.

To write ephone-hunt statistics information to a file, use the **ephone-hunt statistics write-all** command in privileged EXEC mode.

**Syntax**

```
ephone-hunt statistics write-all location
```

**Syntax Description**

- **location**: The URL or filename to which the statistics should be written.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>15.2(2)T</td>
<td>Cisco Unified CME 9.0</td>
<td>This command was replaced. See the <strong>hunt group statistics write-all</strong> command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to write out, in hourly increments, all the ephone hunt group statistics for the past seven days. This command is intended be used when normal hunt group statistics collection is interrupted, perhaps due to TFTP server failure. The commands that are normally used to provide hunt-group statistics are **hunt-group report delay hours**, **hunt-group report every hours**, **hunt-group report url**, and **statistics collect**. These commands allow you to specify shorter, more precise reporting periods and file-naming conventions.

**Note**

Each year on the day that daylight saving time adjusts the time back by one hour at 2 a.m., the original 1 a.m. to 2 a.m. statistics for that day are lost because they are overwritten by the new 1 a.m. to 2 a.m. statistics.

**Examples**

The following example writes the ephone hunt group statistics to a file in flash called “hunstats.”

See the **hunt-group report url** command for explanations of the output fields.

```
Router# ephone-hunt statistics write-all flash:hunstats
Writing out all ephone hunt statistics to tftp now.
11:13:58 UTC Fri Apr 29 2005,
01, Fri 11:00 - 12:00, HuntGp, 01, 01, 00000, 00000, 00000, 0000, 0000, 000000, 000000, 0000, 00000, 000000, 000000,
01, Fri 12:00 - 13:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000, 0000, 00000, 000000, 000000,
01, Fri 13:00 - 14:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000, 0000, 00000, 000000, 000000,
```
Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>hunt-group report delay</td>
<td>Delays hunt-group statistics collection for a specified number of hours.</td>
</tr>
<tr>
<td>every hours</td>
<td>Sets the hourly interval at which Cisco Unified CME B-ACD call statistics</td>
</tr>
<tr>
<td>hunt-group report url</td>
<td>Sets filename parameters and the URL path where Cisco Unified CME B-ACD</td>
</tr>
<tr>
<td>show ephone-hunt</td>
<td>Displays ephone hunt group information.</td>
</tr>
<tr>
<td>show ephone-hunt statistics</td>
<td>Displays ephone hunt group statistics.</td>
</tr>
<tr>
<td>statistics collection</td>
<td>Enables the collection of call statistics for an ephone hunt group.</td>
</tr>
</tbody>
</table>
**ephone-template**

To create an ephone template to configure a set of phone features and to enter ephone-template configuration mode, use the `ephone-template` command in global configuration mode. To delete an ephone template, use the `no` form of this command.

```
ephone-template  template-tag
no ephone-template  template-tag
```

**Syntax Description**

<table>
<thead>
<tr>
<th><em>template-tag</em></th>
<th>Identifier for this ephone template. Range is from 1 to 20.</th>
</tr>
</thead>
</table>

**Command Default**

No ephone template is created.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco CME 3.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The maximum number of templates that can be created was increased from 5 to 20.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>The modification to increase the maximum number of templates that can be created from 5 to 20 was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to create an ephone template containing a set of ephone commands. The template can then be easily applied to one or more ephones.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Type `?` in ephone-template configuration mode to see the commands that are available in this mode and that can be included in an ephone-template. The following example shows CLI help for ephone-template configuration mode at the time that this document was written:

```
Router(config-ephone-template)#?
Ephone template configuration commands:
after-hour  ephone exempt from after-hour blocking
codec      Set preferred codec for calls with other phones on this router
default    Set a command to its defaults
exit       Exit from ephone-template configuration mode
fastdial   Define ip-phone fastdial number
features   define features blocked
keep-conference Do not disconnect conference when conference initiator hang-up. Connect remaining parties together directly using call transfer.
keepalive  Define keepalive timeout period to unregister IP phone
keyphone   Identify an IP phone as keyphone
mtp        Always send media packets to this router
network-locale Select the network locale for this template.
night-service Define night-service bell
```
After creating an ephone template, apply the template to one or more ephones using the `ephone-template` command in ephone configuration mode. Even though you can define up to 20 different ephone templates, you cannot apply more than one template to a particular ephone.

After applying a template to an ephone or removing a template from an ephone, use the following commands:

- `restart`—Performs a fast reboot of the phone.
- `create cnf-files`—Rebuilds configuration files.

If you try to apply a second ephone template to an ephone that already has an ephone template applied to it, the second template will overwrite the first ephone template configuration after you use the `restart` command to reboot the phone.

To view your ephone-template configurations, use the `show telephony-service ephone-template` command. To see which ephones have templates applied to them, use the `show running-config` command.

### Examples

The following example creates two ephone templates. The `softkeys` commands in ephone-template configuration mode define what soft keys are displayed and their order. Template 1 is applied to ephone 32, which has the extension 2555, and template 2 is applied to ephone 38, which has the extension 2666.

```plaintext
ephone-template 1
  softkeys idle Dnd Redial Newcall Pickup Login
  softkeys seized Redial Cfwdall Gpickup Pickup
  softkeys alerting Callback Endcall
  softkeys connected Confrn Hold Endcall
ephone-template 2
  softkeys idle Redial Pickup
  softkeys seized Redial Pickup
  softkeys connected Hold Endcall
ephone-dn 25
  number 2555
ephone-dn 26
  number 2666
ephone 32
  button 1:25
  ephone-template 1
ephone 38
  button 1:26
  ephone-template 2
```

The following example creates an ephone template to block the use of Park and Transfer soft keys. It is applied to extension 2333.

```plaintext
ephone-template 15
  features blocked Park Transfer
ephone-dn 2
  number 2333
```
ephone 3
button 1:2
ephone-template 15

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>create cnf-files</td>
<td>Builds phone configuration files.</td>
</tr>
<tr>
<td></td>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone.</td>
</tr>
<tr>
<td></td>
<td>restart (ephone)</td>
<td>Performs a fast reboot of a single phone associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td></td>
<td>restart (telephony-service)</td>
<td>Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td></td>
<td>show telephony-service ephone-template</td>
<td>Displays ephone-template configurations.</td>
</tr>
</tbody>
</table>
To apply an ephone template to a particular SCCP phone in Cisco Unified CME, use the `ephone-template` command in ephone configuration mode. To remove the ephone template, use the `no` form of this command.

```
ephone-template  template-tag
no  ephone-template  template-tag
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Argument</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>template-tag</code></td>
<td>Unique identifier for a template created by using the <code>ephone-template</code> command in global configuration mode. Range is 1 to 20.</td>
</tr>
</tbody>
</table>

**Command Default**

No ephone template is applied to a phone.

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco CME 3.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified 4.0</td>
<td>The maximum number of ephone templates that can be created was increased from 5 to 20.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified 4.0</td>
<td>This command with an increased range for the <code>template-tag</code> argument was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was modified to specify that before an ephone template can be applied to a particular phone, the MAC address for that phone must be present in its configuration file.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command in ephone configuration mode applies an ephone template to a particular phone.

In Cisco Unified CME 4.3 and later versions, an ephone template cannot be applied to a particular phone unless its configuration file includes its MAC address. If you attempt to apply a template to a phone for which the MAC address in not configured, a message appears. To configure the MAC address for a Cisco Unified IP phone, use the `mac-address` command in ephone configuration mode.

After applying an ephone template, use the `restart` command to perform a fast reboot of the phone.

You cannot apply more than one ephone template at a time to any phone. If you attempt to apply a second ephone template to phone to which an ephone template is already applied, the second template will overwrite the first ephone template configuration after you reboot the phone.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value set in ephone configuration mode has priority over the value set in ephone-template configuration mode.

To view your ephone-template configurations, use the `show telephony-service ephone-template` command.
The following example defines ephone templates 1 and 2 and applies ephone template 1 to ephones 1 through 3 and ephone template 2 to ephone 4.

```plaintext
ephone-template 1
softkeys idle Dnd Redial Newcall Pickup Login
softkeys seized Redial Cfwdall Gpickup Pickup
softkeys alerting Callback Endcall
softkeys connected Confrn Hold Endcall
softkeys hold Newcall Resume
ephone-template 2
softkeys idle Redial Pickup
softkeys seized Redial Pickup
softkeys alerting Endcall
softkeys connected Hold Endcall
softkeys hold Resume
ephone 1
ephone-template 1
ephone 2
ephone-template 1
ephone 3
ephone-template 1
ephone 4
ephone-template 2
ephone 5
ephone-template 2
```

The following example creates an ephone template to block the use of Park and Transfer soft keys on extension 2333.

```plaintext
ephone-template 15
features blocked Park Trnsfer
ephone-dn 2
number 2333
ephone 3
button 1:2
ephone-template 15
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template</td>
<td>Creates an ephone-template and enters ephone-template configuration mode.</td>
</tr>
<tr>
<td>mac-address</td>
<td>Associates the MAC address of a Cisco Unified IP phone with an ephone configuration in Cisco Unified CME.</td>
</tr>
<tr>
<td>restart (ephone)</td>
<td>Performs a fast reboot of a single phone in Cisco Unified CME.</td>
</tr>
<tr>
<td>restart (telephony-service)</td>
<td>Performs a fast reboot of one or all phones in Cisco Unified CME.</td>
</tr>
<tr>
<td>show telephony-service</td>
<td>Displays ephone-template configurations.</td>
</tr>
<tr>
<td>ephone-template</td>
<td></td>
</tr>
</tbody>
</table>
**ephone-type**

To add a Cisco Unified IP phone type by defining an ephone-type template, use the `ephone-type` command in global configuration mode. To remove an ephone type, use the `no` form of this command.

```
ephone-type  phone-type  [addon]
no ephone-type  phone-type
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>phone-type</code></td>
<td>Unique label that identifies the type of phone. Value is any alphanumeric string up to 32 characters.</td>
</tr>
<tr>
<td><code>addon</code></td>
<td>(Optional) Phone type is an add-on module, such as the Cisco Unified IP Phone 7915 Expansion Module.</td>
</tr>
</tbody>
</table>

**Command Default**

Ephone type is not defined.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3 Cisco Unified SRST 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command adds a user-defined template for a phone type to a Cisco Unified CME system. This configuration template defines a set of attributes that describe the features of the new phone type. Use this command to add phone types that are not already defined in the system.

If you use a phone-type argument that matches a system-defined phone type, a message displays notifying you that the ephone-type is built-in and cannot be changed. For a list of system-defined phone types, see the `type` command.

Use the `create cnf-files` command for the new phone type to take effect.

**Examples**

The following example shows the Nokia E61 added with an ephone-type template, which is then assigned to ephone 2:

```
ephone-type  E61
device-id  376
device-name  E61 Mobile Phone
num-buttons  1
max-presentation  1
no utf8
no phoneload
!
!
telephony-service
load E61 SCCP61.8-2-2SR2S
max-ephones  100
```
max-dn 240
ip source-address 15.7.0.1 port 2000
cnf-file location flash:
cnf-file perphone
voicemail 8900
max-conferences 8 gain -6
transfer-system full-consult
create cnf-files version-stamp 7960 Sep 25 2007 21:25:47
!
ephone 2
mac-address 001C.821C.ED23
type E61
button 1:2

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>create cnf-files</td>
<td>Builds the eXtensible Markup Language (XML) configuration files that are required for IP phones.</td>
</tr>
<tr>
<td>device-id</td>
<td>Specifies the device ID for a phone type in an ephone-type template.</td>
</tr>
<tr>
<td>device-name</td>
<td>Assigns a name to a phone type in an ephone-type template.</td>
</tr>
<tr>
<td>load</td>
<td>Associates a type of Cisco Unified IP phone with a phone firmware file.</td>
</tr>
<tr>
<td>type</td>
<td>Assigns a phone type to an SCCP phone.</td>
</tr>
</tbody>
</table>
exclude

To exclude the availability of local services on a phone’s user interface such as, Extension Mobility (EM), My Phone Apps, and Local Directory from the phone’s configuration, use the exclude command in ephone or ephone-template mode.

```
exclude [{em|myphoneapp|directory|call-history}]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Argument</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>em</td>
<td>Extension mobility (EM) service.</td>
</tr>
<tr>
<td>myphoneapp</td>
<td>My Phone Apps service.</td>
</tr>
<tr>
<td>directory</td>
<td>Local directory service</td>
</tr>
<tr>
<td>call-history</td>
<td>Call history in the missed, received, or placed calls directory</td>
</tr>
</tbody>
</table>

**Command Default**

Local services are enabled.

**Command Modes**

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(4)M</td>
<td>Cisco Unified CME 8.6</td>
<td>This command was modified. Call-history option was added.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to exclude the availability of local services such as, EM, my phone apps, and local directory services from the phone configuration.

**Examples**

The following example shows directory and my phone apps excluded from ephone-template 8:

```
Router# conf t
Router(config)# ephone-template 8
Router(config-ephone-template)# exclude ?
directory local directory service	em extension mobility service
myphoneapp my phone apps service
<cr>
Router(config-ephone-template)# exclude directory
Router(config-ephone-template)# exclude myphoneapp!
```

The following example shows call-history as excluded from ephone 10:

```
!
telephony-service
max-ephones 40
max-dn 100
max-conferences 8 gain -6
transfer-system full-consult
!
```

```plaintext
ephone-template  5
exclude call-history
!
!
ephone  10
exclude call-history
device-security-mode none
!
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies template to an ephone.</td>
</tr>
<tr>
<td>show telephony-service ephone-template</td>
<td>Displays ephone-template configurations.</td>
</tr>
</tbody>
</table>
**exclude (voice register)**

To exclude from the Cisco Unified SIP IP phone’s user interface the availability of local services such as Extension Mobility (EM), My Phone Apps, and Local Directory, use the `exclude` command in voice register pool or voice register template configuration mode.

```
exclude [{em|myphoneapps|directory}]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>em</code></td>
<td>Extension Mobility service is excluded.</td>
</tr>
<tr>
<td><code>myphoneapps</code></td>
<td>My Phone Apps service is excluded.</td>
</tr>
<tr>
<td><code>directory</code></td>
<td>Local Directory service is excluded.</td>
</tr>
</tbody>
</table>

**Command Default**

Local services are enabled.

**Command Modes**

- Voice register pool configuration (config-register-pool)
- Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Examples**

The following example shows the Local Directory and My Phone Apps services excluded from voice register pool 33:

```
Router(config)# voice register pool 33
Router(config-register-pool)# exclude directory
Router(config-register-pool)# exclude myphoneapps
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>voice register pool</code></td>
<td>Enters voice register pool configuration mode and creates a pool configuration for Cisco Unified SIP IP phones in Cisco Unified CME.</td>
</tr>
<tr>
<td><code>voice register template</code></td>
<td>Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones.</td>
</tr>
</tbody>
</table>
expiry

To set the time after which emergency callback history expires, use the `expiry` command in voice emergency response settings configuration mode. To remove the number, use the `no` form of this command.

```
expiry  time
no expiry
```

**Syntax Description**

| Identifier (2-2880) in minutes for the maximum time the 911 caller history is available for callback. |

**Command Default**

The default expiry time is 180 minutes.

**Command Modes**

Voice emergency response settings configuration (cfg-emrgncy-resp-settings)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to specify the amount of time (in minutes) to keep emergency caller history for each ELIN. The time can be an integer in the range of 2 to 2880 minutes. The default value is 180 minutes.

**Examples**

In this example, the ELIN (4085550101) defined in the voice emergency response settings configuration is used if the 911 caller’s IP phone address does not match any of the voice emergency response locations. After the 911 call is placed to the PSAP, the PSAP has 120 minutes to call back 408 555-0101 to reach the 911 caller. If the call history has expired (after 120 minutes), any callback is routed to extension 7500.

```
voice emergency response settings
callback 7500
elin 4085550101
expiry 120
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>callback</td>
<td>Default phone number to contact if a 911 callback cannot find the last 911 caller from the ERL.</td>
</tr>
<tr>
<td>elin</td>
<td>E.164 number used as the default ELIN if no matching ERL to the 911 caller’s IP phone address is found.</td>
</tr>
<tr>
<td>logging</td>
<td>Syslog informational message printed to the console every time an emergency call is made.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>voice emergency response settings</td>
<td>Creates a tag for identifying settings for E911 behavior.</td>
</tr>
</tbody>
</table>
extension-assigner tag-type

To enable provision tags for identifying ephone configurations when using the extension assigner application, use the `extension-assigner tag-type` command in telephony-service configuration mode. To return to the default setting of using the ephone tag, use the `no` form of this command.

`extension-assigner tag-type {ephone-tag|provision-tag}`
`no extension-assigner tag-type`

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-tag</td>
<td>Ephone tags must be used to identify ephone configurations.</td>
</tr>
<tr>
<td>provision-tag</td>
<td>Provision tags must be used to identify ephone configurations.</td>
</tr>
</tbody>
</table>

**Command Default**

Ephone tags are used to identify ephone configurations for the extension assigner application.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC4</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables you to use provision tags for identifying ephone configurations to be assigned by the extension application application.

A provision tag is an unique number other than an ephone tag, such as a jack number or an extension number, for identifying the ephone configuration to be assigned to a particular IP phone by the extension assigner application.

Use this command to specify which type of tag, ephone tag or provision tag, is to be used to identify ephone configurations for the extension assigner application. The default configuration is ephone tag.

If you use this command with the `provision-tag` keyword, use the `provision-tag` command to create provision tags.

**Examples**

The following example shows that this command is configured to enable provision tags to be used for identifying the ephone configurations to be assigned by the extension assigner application. Note that provision tag 1001 is configured for ephone 1. During phone installation, the installation technician can press 1001 on the telephone keypad to assign the ephone 1 configuration, with extension number 1001 on button 1, to the IP phone being installed.

```
Telephony-service
   extension-assigner tag-type provision-tag
   auto assign 101-102
   auto-reg-ephone
   Ephone-dn 101
   number 1001
```
Ephone-dn 102
  number 1002
Ephone 1
  provision-tag 1001
  mac-address 02EA.EAEA.0001
  button 1:101
Ephone 2
  provision-tag 1002
  mac-address 02EA.EAEA.0002
  button 1:102

## Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>provision-tag</td>
<td>Creates a provision tag for identifying an ephone configuration.</td>
</tr>
</tbody>
</table>
extension-range

To define a range of extension numbers for a specific MOH group in Cisco Unified CME or Cisco Unified SRST, use the `extension-range` command in voice-moh-group configuration mode. To define a range of extension numbers for a specific directory number in Cisco Unified CME, use the `extension-range` command in ephone-dn configuration mode. To disable the extension-range command, use the `no` form of this command.

```
extension-range starting-extension to ending-extension
no extension-range starting-extension to ending-extension
```

**Syntax Description**

| starting-extension | Hexadecimal digits (0-9 or A-F) that define the starting extension number in an extension range. Maximum length: 32 digits. |
| ending-extension   | Hexadecimal digits (0-9 or A-F) that define the last extension number in an extension range. Value of the ending extension must be larger than value of the starting extension. Maximum length: 32 digits. |

**Command Default**

No extension-range is configured.

**Command Modes**

Voice MOH group configuration (config-voice-moh-group)
Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0 Cisco Unified SRST 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0 Cisco Unified SRST 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command configured in voice moh-group configuration mode identifies MOH clients calling extension numbers specified under the extension range configured for a MOH group in Cisco Unified CME or Cisco Unified SRST. This command in ephone-dn configuration mode identifies MOH clients calling extension numbers specified under the extension range configured for a directory number in Cisco Unified CME.

You can define multiple extension-ranges in the same MOH group or directory number.

The extension can be expressed in hexadecimal digits ranging from 0-9 or A-F and should not exceed the limit of 32 digits.

The starting-extension and ending-extension numbers must contain the same number of digits.

The ending extension number must be of a greater value than the starting extension number.

Extension-range for a MOH group must not overlap with any other extension-range configured in any other MOH group.

---

**Note**

If an extension range is defined in a MOH group and it is also defined under ephone-dn, the extension range defined under ephone-dn takes precedence.
The following example shows two extension ranges configured under voice moh-group 1:

Router(config)# voice moh-group 1
Router(config-voice-moh-group)# moh flash:/minuet.wav
Router(config-voice-moh-group)# description Marketing
Router(config-voice-moh-group)# extension range 1000 to 1999
Router(config-voice-moh-group)# extension range 3000 to 3999

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>moh</td>
<td>Enables music on hold from an audio file.</td>
</tr>
<tr>
<td>voice-moh-group</td>
<td>Enters voice moh-group configuration mode.</td>
</tr>
</tbody>
</table>
external-ring (voice register global)

To specify the type of ring sound used on Cisco Session Initiation Protocol (SIP) or Cisco SCCP IP phones for external calls, use the external-ring command in voice register global configuration mode. To return to the default ring sound, use the no form of this command.

external-ring \{bellcore-dr1|bellcore-dr2|bellcore-dr3|bellcore-dr4|bellcore-dr5\}
no external-ring

| Syntax Description | bellcore-dr1 bellcore-dr2 bellcore-dr3 bellcore-dr4 bellcore-dr5 | Standard distinctive ringing patterns as defined in the standard GR-506-CORE, LSSGR: Signaling for Analog Interfaces. |

| Command Default | The default ring sound is an internal ring pattern. |

| Command Modes | Voice register global configuration (config-register-global) |

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

When set, this command defines varying ring tones so that you can discriminate between internal and external calls from Cisco SIP or Cisco SCCP IP phones.

**Examples**

The following example shows how to specify that Bellcore DR1 be used for external ringing on Cisco SIP IP phones:

Router(config)# voice register global
Router(config-register-global)# external-ring bellcore-dr1
external-ring (voice register global)
Cisco Unified CME Commands: F

- fac, on page 412
- fac refer, on page 417
- fail-connect-time, on page 418
- fastdial, on page 419
- feature-button, on page 421
- feature-button (voice_register_pool), on page 423
- features blocked, on page 424
- feed, on page 426
- file text (voice register global), on page 428
- filename, on page 429
- final, on page 431
- final (voice hunt-group), on page 433
- forward local-calls, on page 434
- forward local-calls (voice hunt-group), on page 436
- forwarding local (voice register global), on page 438
- from-ring, on page 439
- fwd-final, on page 440
- fxo hook-flash, on page 441
To enable all standard feature access codes (FACs) or to create and enable individual custom FACs, use the `fac` command in telephony-service configuration mode. To disable FACs, use the `no` form of this command.

```
fac { standard | custom } { alias alias-tag feature }
fac refer
no fac { standard | custom } { alias alias-tag feature }
```

### Syntax Description

<table>
<thead>
<tr>
<th></th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>standard</strong></td>
<td>All predefined standard FACs are enabled.</td>
</tr>
<tr>
<td><strong>custom</strong></td>
<td>User-defined FAC for selecting a particular feature or function from the predefined set of features is enabled.</td>
</tr>
<tr>
<td><strong>alias</strong></td>
<td>Alternative FAC for dialing an existing FAC or existing FAC plus extra digits without removing the existing FAC is enabled.</td>
</tr>
<tr>
<td><strong>alias-tag</strong></td>
<td>Unique number that identifies this alias during configuration tasks. Range: 0 to 9.</td>
</tr>
<tr>
<td><strong>custom-fac</strong></td>
<td>User-defined code to dial using the keypad on an IP or analog phone. Code can be up to 256 characters and can contain numbers 0 to 9 and * and #. <strong>Note</strong> ## is not supported for FACs on SIP phones.</td>
</tr>
<tr>
<td><strong>to</strong></td>
<td>Maps custom FAC being configured to specified target.</td>
</tr>
<tr>
<td><strong>existing-fac</strong></td>
<td>Already configured custom FAC that is automatically dialed when the phone user dials the custom FAC being configured.</td>
</tr>
<tr>
<td><strong>extra-digits</strong></td>
<td>(Optional) Additional digits that are automatically dialed when the phone user dials the custom FAC being configured. Valid entries are:</td>
</tr>
<tr>
<td></td>
<td>• <strong>target extension</strong> — Telephone or extension number in Cisco Unified CME to which the incoming calls are forwarded. Used with the Call Forward feature.</td>
</tr>
<tr>
<td></td>
<td>• <strong>group number</strong> — Pickup group number, for a group other than the local group number. Used with the Pickup Group feature.</td>
</tr>
<tr>
<td></td>
<td>• <strong>pickup extension</strong> — Telephone or extension number in Cisco Unified CME to be picked up when ringing. To be used with the Pickup Direct feature.</td>
</tr>
<tr>
<td></td>
<td>• <strong>park-slot number</strong> — Number on which calls are to be temporarily parked. Use with the Call Park feature. Target park slot must be already configured in Cisco Unified CME.</td>
</tr>
<tr>
<td></td>
<td>• <strong>pilot number</strong> — Telephone or extension number configured as a the pilot number for an ephone hunt group to be joined. Hunt group to be joined must allow dynamic membership.</td>
</tr>
</tbody>
</table>
Predefined alphabetic string that identifies a particular feature or function. Valid entries are:

- **callfwd all** — Directs system to forward all incoming calls for this telephone or extension number.
- **callfwd cancel** — Directs system to cancel the call-forward-all selection.
- **ccw** — Disables the Call Waiting feature.
- **dnd** — Enables Do Not Disturb (DND) feature on SCCP phones. Not supported for SIP phones.
- **dpark-retrieval** — Enables Directed Call Park Retrieval feature. Applies to both SIP and SCCP phones.
- **ephone-hunt cancel** — Leaves an ephone hunt group that is configured to allow dynamic membership.
- **ephone-hunt hlog** — Activates or deactivates hunt group logout functionality, changing the status of the an ephone-dn for a hunt group agent from ready to not-ready or from not-ready to ready.
- **ephone-hunt hlog-phone** — Activates or deactivates phone-level hunt group logout functionality, changing the status of all the extensions on a hunt group member phone from ready to not-ready or from not-ready to ready.
- **ephone-hunt join** — Joins an ephone hunt group that is configured to allow dynamic membership. If multiple hunt groups have been created that allow dynamic membership, the hunt group to be joined is identified by its pilot number.
- **park** — Enables Call Park feature.
- **pickup direct** — Picks up a ringing call at any extension. Applies to both SIP and SCCP phones.
- **pickup group** — Picks up a ringing call in a different pickup group than yours. Applies to both SIP and SCCP phones.
- **pickup local** — Picks up a ringing call in your pickup group. Applies to both SIP and SCCP phones.
- **redial** — Redials the last number called.
- **transfvm** — Activates the Transfer to Voice-Mail feature.
- **voicemail** — Dials the voice-mail number.

**Feature Command Default**

FACs are disabled on IP phones.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>Standard FAC and <strong>transfvm</strong> keyword for a custom FAC were added for transfer to voice mail.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
<tr>
<td>Cisco IOS Release</td>
<td>Cisco Product</td>
<td>Modification</td>
</tr>
<tr>
<td>------------------</td>
<td>---------------</td>
<td>--------------</td>
</tr>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>The <strong>dpark-retrieval</strong> keyword was added and support for SIP phones was added for the <strong>park direct</strong>, <strong>park group</strong>, and <strong>park local</strong> keywords.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was modified. The <strong>ccw</strong> keyword was added for a custom FAC.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use this command to enable all predefined standard FACs or to create one or more custom FACs.

FACs enable phone users to use the keypad on an analog or IP phone registered in Cisco Unified CME to select or activate/deactivate a particular feature or function from a predefined set of features. For example, a phone user might press **1**, then press 2345 to forward all incoming calls to extension 2345.

Standard FACs and custom FACs are mutually exclusive. You can enable all standard FACs or create and enable one or more custom FACs.

Most FACs are valid only immediately after a phone user goes off-hook. The only exception is the call-park FAC. The call-park FAC actually invokes a call transfer to a park slot. To use the call-park FAC, a phone user must have an active call and must press the Transfer soft key (IP phone) or hook flash (analog phone) before dialing the call-park FAC. Dialing the FAC for the Call Park feature does not use the Park soft key function.

Use the **fac standard** command to enable all predefined standard FACs for all SCCP phones registered in Cisco Unified CME.

Use the **fac custom** command to create an individual custom FAC for selecting a particular feature or function from the predefined feature set.

Use the **fac custom** command with the **alias** keyword to create an alternative (custom) FAC for dialing an existing FAC, or existing FAC plus extra digits without removing the existing FAC. For example, an alias can be created to allow the phone user to press **1** to forward all incoming calls to a particular extension **without** requiring the phone user to dial the target extension number.

To disable all custom FACs, use the **fac standard** command, which enables all standard FACs. To disable all standard FACs or to disable an individual custom FAC, use the **no** form of the **fac** command.

Use the **show telephony-service fac** command to display a list of FACs that are configured on the Cisco Unified CME router.

### Examples

The following example shows how to enable standard FACs for all phones:

```
Router(config)# telephony-service
Router(config-telephony)# fac standard
fac standard is set!
```

The following example shows the output from the **show telephony-service fac** command when standard FACs are enabled:
Router# show telephony-service fac

telephony-service fac standard
callfwd all **1
callfwd cancel **2
pickup local **3
pickup group **4
pickup direct **5
park **6
dnd **7
redial **8
voicemail **9
ephone-hunt join *3
ephone-hunt cancel #3
ephone-hunt hlog *4
ephone-hunt hlog-phone *5
transfvm *6
dpark-retrieval **10
cancel call waiting *1

The following example shows how the standard FAC for the Call Forward All feature is changed to a custom FAC (#45). Then an alias is created to map a second custom FAC to #45 plus an extension (1111). The second custom FAC (#44) allows the phone user to press #44 to forward all calls all calls to extension 1111, without requiring the phone user to dial the extra digits that are the extension number.

Router(config)# telephony-service
Router(config-telephony)# fac custom callfwd all #45
callfwd all code has been configured to #45
Router(config-telephony)# fac custom alias 0 #44 to #451111
fac alias0 code has been configurated to #44!
alias0 map code has been configurated to #451111!

The following example shows how to create three aliases for the Group Pickup feature. The FAC for group pickup is **4. The three new custom FACs are #1, #2, and #4 to pickup groups 121, 122, and 124, respectively. This allows a phone user to press #1 to pick up calls in group 121, #2 to pick up calls in group 122, and #4 to pick up calls in group 124.

Router(config)# telephony-service
Router(config-telephony)# fac custom pickup group **4
fac pickup group code has been configured to **4
Router(config-telephony)# fac custom alias 1 #1 to **4121
cfac alias1 code has been configurated to #1!
alias1 map code has been configurated to **4121!
Router(config-telephony)# fac custom alias 2 #2 to **4122
cfac alias2 code has been configurated to #2!
alias2 map code has been configurated to **4122!
Router(config-telephony)# fac custom alias 4 #4 to **4124
cfac alias4 code has been configurated to #4!
alias4 map code has been configurated to **4124!

The following example shows the output from the show telephony-service fac command when custom FACs are configured:

Router# show telephony-service fac
telephony-service fac custom
callfwd all #45
alias 0 #44 to #451111
alias 1 #1 to **4121
alias 2 #2 to **4122
alias 4 #4 to **4124

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show telephony-service fac</td>
<td>Displays list of FACs that are configured on Cisco Unified CME.</td>
</tr>
</tbody>
</table>
fac refer

To send the SIP REFER to a SIP phone, use the **fac refer** command in voice register global configuration mode. To allow Cisco Unified CME to handle the SIP REFER internally, use the **no** form of this command.

```
fac refer
no fac refer
```

**Syntax Description**

- `lpcor-group` Name of the LPCOR resource group.

**Command Default**

Fac refer is enabled.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
</table>
| 15.1(3)T          | Cisco Unified CME 8.5  | This command was introduced.

**Usage Guidelines**

Use this command to control the SIP REFER to be sent to a SIP phone. The fac refer command is enabled in Cisco Unified CME by default to allow Cisco Unified CME to pass the REFER to the SIP phone, thereby enabling the phone to make a new call towards Cisco Unified CME. Cisco Unified CME accepts the new invite message as a new call and requires the call transferree to enter a forced authorization code (FAC) again. Use the no fac refer command to allow Cisco Unified CME to handle the SIP REFER internally instead of passing the call towards the SIP phone.

**Examples**

The following example shows no fac refer configured in voice register global:

```
Router#show run
!
voice register global
no fac refer
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>show voice register global</strong></td>
<td>Displays all global configuration parameters associated with SIP phones.</td>
</tr>
</tbody>
</table>
fail-connect-time

To specify the maximum time to wait for establishing VPN tunnel including establishing of SSL/DTLS and login or connect requests or responses, use the `fail-connect-time` command in vpn-profile configuration mode. To disable the fail-connect-time configuration, use the no form of this command.

```
fail-connect-time  seconds
```

**Syntax Description**

| seconds | Failure-to-connect time, in seconds. Range: 0 to 600 seconds. Default: 30 seconds. |

**Command Default**

Default fail-connect-time is 30 seconds.

**Command Modes**

Vpn-profile configuration (conf-vpn-profile)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to specify the fail-to-connect time for a vpn-profile. The fail-to-connect time specifies the maximum time to wait for establishing VPN tunnel including establishing of SSL/DTLS and login/connect request/response. The fail-to-connect time ranges from 0 seconds to 600 seconds. The default fail-to-connect time is 30 seconds.

**Examples**

The following example shows fail-connect-time set to 50 seconds for vpn-profile 4:

```
Router# show run
!
!
voice service voip
ip address trusted list
  ipv4 20.20.20.1
vpn-group 1
  vpn-gateway 1 https://9.10.60.254/SSLVPNphone
  vpn-trustpoint 1 trustpoint cme_cert root
  vpn-hash-algorithm sha-1
vpn-profile 1
  keepalive 50
  host-id-check disable
vpn-profile 2
  mtu 1300
  password-persistent enable
  host-id-check enable
vpn-profile 4
  fail-connect-time 50
  sip
!
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>vpn-profile</strong></td>
<td>Defines a VPN-profile.</td>
</tr>
</tbody>
</table>
fastdial

To create an entry for a personal speed-dial number, use the fastdial command in ephone or ephone-template configuration mode. To delete a personal speed-dial number, use the no form of this command.

```
fastdial dial-tag number name name-string
no fastdial dial-tag
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Fastdial Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>dial-tag</code></td>
<td>Unique sequence number that ranges from 1 to 100 and is used to identify a particular personal speed-dial number during configuration tasks.</td>
</tr>
<tr>
<td><code>number</code></td>
<td>Telephone number or extension to be dialed.</td>
</tr>
<tr>
<td><code>name</code></td>
<td>Label to appear in the Personal Speed Dial menu, containing a string of up to 24 alphanumeric characters. Personal speed dial is handled through an XML request, so characters that have special meaning to HTTP, such as ampersand (&amp;), percent sign (%), semicolon (;), angle brackets (&lt; &gt;), and vertical bars (</td>
</tr>
</tbody>
</table>

**Command Default**

No personal speed-dial numbers are present.

**Command Modes**

- Voice register pool configuration (config-register-pool)
- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was modified to increase the range from 24 to 100.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is supported only on certain Cisco Unified IP phones, such as the 7940, 7960, 7960G, 7970G, and 7971G-GE. To determine whether personal speed-dial menu is supported on your IP phone, see the Cisco Unified CME user documentation for your IP phone model.

Phone users access personal speed-dial numbers through the Directories > Local Services > Personal Speed Dial menu. Personal speed-dial numbers appear on this menu in the order in which they are entered during configuration.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.
The following example creates a directory of five personal speed-dial numbers for an IP phone:

Router(config)# ephone 1
Router(config-ephone)# fastdial 1 5001 name Front Register
Router(config-ephone)# fastdial 2 5002 name Security
Router(config-ephone)# fastdial 3 5003 name Rear Register
Router(config-ephone)# fastdial 4 5004 name Office
Router(config-ephone)# fastdial 5 912135550122 Accounting

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies a template to the ephone being configured.</td>
</tr>
<tr>
<td>show telephony-service ephone-template</td>
<td>Displays the contents of ephone templates.</td>
</tr>
</tbody>
</table>
**feature-button**

To enable feature button configuration on a line key, use the feature-button command in ephone, ephone-template, voice user profile, or voice logout profile configuration mode. To disable the feature button configuration on a line key, use the no form of this command.

```
feature-button index index <feature identifier> [label <label>]
no feature-button index index <feature identifier> [label <label>]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>index</td>
<td>Index number of a specific feature type. One from the total 24 feature IDs.</td>
</tr>
<tr>
<td>feature identifier</td>
<td>One of the following feature or stimulus IDs: Redial, Hold, Trnsfer, Cfwdall, Privacy, MeetMe, Confrn, Park, Pickup. Gpickup, Mobility, Dnd, Conflist, RmlstC, CallBack, NewCall, EndCall, HLog, NiteSrv, Acct, Flash, Login, TrnsfVM, LiveRcd.</td>
</tr>
<tr>
<td>label</td>
<td>Defines non-default text label for PLK button.</td>
</tr>
</tbody>
</table>

**Command Default**

No feature-button is configured.

**Command Modes**

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)
Voice user-profile configuration (config-user-profile)
Voice logout-profile configuration (config-logout-profile)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was modified to configure feature button on a phone’s line key. Feature button index number and feature ID keywords were added.</td>
</tr>
<tr>
<td>15.2(4)M</td>
<td>Cisco Unified CME 9.1</td>
<td>This command was modified to add label &lt;label&gt; for the PLK button.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to configure a DnD feature button as a short cut for the DnD softkey. This command with the privacy keyword takes precedence over the privacy-button command. If a feature-button command is configured for DnD, the privacy-button command will be ignored and the privacy button must be configured through the feature-button command to take effect.

In Cisco Unified CME 8.5 and later versions, the feature-button command allows you to program a phone’s line key to function as a feature button. You can configure one of the following 24 feature IDs: Redial, Hold, Trnsfer, Cfwdall, Privacy, MeetMe, Confrn, Park, Pickup. Gpickup, Mobility, Dnd, Conflist, RmlstC, CallBack, NewCall, EndCall, HLog, NiteSrv, Acct, Flash, Login, TrnsfVM, LiveRcd

**Examples**

The following example shows how to configure feature buttons:
Router(config)# ephone 1
Router(config-ephone) feature-button 1 privacy
Router(config-ephone) feature-button 2 dnd
Router(config-ephone) feature-button 3 Hlog label Agent Hlogout

The following example shows feature buttons configured in ephone template 9 and ephone template 10:
Router# show telephony-service ephone-template
ephone-template 9
conference drop-mode never
conference add-mode all
conference admin: No
max-calls-per-button 8
busy-trigger-per-button 0
privacy default
feature-button 1 Endcall
feature-button 3 Mobility
Always send media packets to this router: No
Preferred codec: g711ulaw
keepalive 30 auxiliary 30
User Locale: US
Network Locale: US
lpcor type:
lpcor (incoming):    (outgoing):
ephone-template 10
conference drop-mode never
conference add-mode all
conference admin: No
max-calls-per-button 8
busy-trigger-per-button 0
privacy default
feature-button 1 Park
feature-button 2 MeetMe
feature-button 3 CallBack
button-layout 1 line
button-layout 2-4 speed-dial
button-layout 5-6 blf-speed-dial
MLPP Service Domain Network none (0)
!

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>privacy-button</td>
<td>Enables the privacy feature button on an IP phone.</td>
</tr>
<tr>
<td>show telephony-service ephone</td>
<td>Displays the information about ephone configuration in a Cisco CallManager Express (Cisco CME) system.</td>
</tr>
<tr>
<td>show telephony-service ephone-dn-template</td>
<td>Displays the information about ephone-template’s configurations.</td>
</tr>
</tbody>
</table>
**feature-button (voice_register_pool)**

To configure feature button configuration on a line key, use the feature-button command in voice register pool or voice register template configuration mode. To disable the feature button configuration on a line key, use the no form of this command.

```
feature-button [index number feature identifier feature id]
no feature button [index number feature identifier feature id]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>index</th>
<th>Index number of a specific feature type. One from the total 24 feature IDs.</th>
</tr>
</thead>
<tbody>
<tr>
<td>feature identifier</td>
<td>One of the following feature or stimulus IDs: Redial, Hold, Trnsfer, Cfwdall, Privacy, MeetMe, Confm, Park, Pickup, Gpickup, Mobility, NewCall, EndCall, Dnd, ConfList, NewCall, HLog, Trnsfer.</td>
</tr>
</tbody>
</table>

**Command Default**

Feature-button configuration on a line key is disabled.

**Command Modes**

Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to program a phone’s line key to function as a feature button. You can configure one of the following 24 features IDs: Redial, Hold, Trnsfer, Cfwdall, Privacy, MeetMe, Confm, Park, Pickup, Gpickup, Mobility, NewCall, EndCall, Dnd, ConfList, NewCall, HLog, Trnsfer. The feature ID list for the command is incrementally updated across Unified CME releases.

**Examples**

The following example shows feature button configured in voice register pool 50:

```
voice register pool 50
  id mac 001E.7AC4.DC73
  feature-button 1 NewCall
  type 7965
  number 1 dn 65
  template 1
dtmf-relay rtp-nte
  speed-dial 1 2001 label "SD1-2001"
speed-dial 3 2003 label "SD3-2003"
blf-speed-dial 1 3001 label "BLFL1-3001"
!
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
</tbody>
</table>
features blocked

To prevent one or more features from being used on a Cisco Unified CME phone, use the `features blocked` command in ephone-template configuration mode. To allow all features to be used, use the `no` form of this command.

```
features blocked [CFwdAll] [Confrn] [GpickUp] [Park] [PickUp] [Trnsfer]
no features blocked
```

Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>CFwdAll</td>
<td>Call forward all calls.</td>
</tr>
<tr>
<td>Confrn</td>
<td>Conference.</td>
</tr>
<tr>
<td>GpickUp</td>
<td>Group call pickup.</td>
</tr>
<tr>
<td>Park</td>
<td>Call park.</td>
</tr>
<tr>
<td>PickUp</td>
<td>Directed or local call pickup. This includes pickup last-parked call and pickup from another extension or park slot.</td>
</tr>
<tr>
<td>Trnsfer</td>
<td>Call transfer.</td>
</tr>
</tbody>
</table>

Command Default

Features are not blocked.

Command Modes

Ephone-template configuration (config-ephone-template)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

Use this command to specify one or more features to be blocked in an ephone template, then apply the template in ephone configuration mode to one or more ephones to prevent the use of the specified features by those ephones. This feature can be used on IP phones and analog phones. After applying the template, any soft keys associated with the blocked features will still be visible but will not have any effect.

Use the `show telephony-service ephone-template` command to display the contents of ephone templates.

Examples

In the following example, call park and call transfer are blocked on ephone 3.

```
ephone-template 1
  features blocked Park Trnsfer
ephone-dn 2
  number 2333
ephone 3
  button 1:2
ephone-template 1
```

The following example blocks the use of the conference feature on ephone 3, which is an analog phone, by using a template.
```plaintext
ephone-template 1
features blocked Confrn
ephone-dn 78
number 2579
ephone 3
ephone-template 1
mac-address C910.8E47.1282
type anl
button 1:78
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies a template to the ephone being configured.</td>
</tr>
<tr>
<td>show telephony-service ephone-template</td>
<td>Displays the contents of ephone templates.</td>
</tr>
</tbody>
</table>
To enable an audio stream for multicast from an external live audio feed connected directly to the router by a foreign exchange office (FXO) or an E&M analog voice port, use the feed command in ephone-dn configuration mode. To disable the multicast audio stream, use the no form of this command.

```
feed ip ip-address port port-number [route ip-address] [out-call outcall-number]
no feed ip
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ip ip-address</td>
<td>Indicates that a particular audio stream is to be used as a multicast source and specifies the destination IP address for multicast.</td>
</tr>
<tr>
<td>port port-number</td>
<td>Specifies the media port for multicast. Range is from 2000 to 65535. Port 2000 is recommended because this port is already used for normal Real-Time Transport Protocol (RTP) media transmissions between IP phones and the Cisco CallManager Express (Cisco CME) router.</td>
</tr>
<tr>
<td>route ip-address</td>
<td>(Optional) Indicates the specific router interface on which to transmit the IP multicast packets. The default is that the audio stream is automatically output on the interface that corresponds to the address that was configured with the ip source-address command.</td>
</tr>
<tr>
<td>out-call outcall-number</td>
<td>(Optional) Sets up a call to the outcall number in order to connect to a live audio feed. If this keyword is not used, the live feed is assumed to derive from an incoming call to the ephone-dn that is being configured.</td>
</tr>
</tbody>
</table>

**Command Default**

No multicast audio stream is enabled on an extension.

**Command Modes**

Ephone-dn configuration

**Command History**

<table>
<thead>
<tr>
<th>Command History</th>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td></td>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

When this command is used, a connection for a live feed audio stream is established as an automatically connected voice call. If the out-call keyword is used, the Cisco CME system calls out to the specified number for the audio stream. If the out-call keyword is not used, it is assumed that the call is incoming to the ephone-dn. This includes VoIP calls if voice activity detection (VAD) is disabled. The typical operation is for the Cisco CME ephone-dn to establish a call to a local router E&M voice port.

Connection via E&M is the recommended mechanism because it requires minimal external components. The E&M port must be placed in 4-wire operation, using E&M immediate signaling and with the auto-cut-through option enabled. You directly connect a line-level audio feed (standard audio jack) to pins 3 and 6 of an E&M RJ-45 connector. The E&M WAN interface card (WIC) has a built-in audio transformer that provides appropriate electrical isolation for the external audio source. (The audio connection on the E&M port does not require loop current.) The signal immediate and auto-cut-through commands disable E&M signaling on this voice port. A G.711 audio packet stream is generated by the digital signal processor (DSP) on the E&M port.
If you are using an FXO voice port for live-feed audio stream instead of an E&M port, connect the source to the FXO voice port. This connection requires an external adapter to supply normal telephone company (telco) battery voltage with the correct polarity to the tip-and-ring leads of the FXO port. The adapter must also provide transformer-based isolation between the external audio source and the tip-and-ring leads of the FXO port.

If the `out-call` keyword is used, an outbound call to the live-feed source is attempted (or reattempted) every 30 seconds until the call is connected to the ephone-dn (extension) for which the `feed` command was configured. Note that this ephone-dn is not associated with a physical phone.

The related `moh (ephone-dn)` and `multicast moh` commands provide the ability to multicast an audio stream that is also being used as the source for Cisco CME system music on hold (MOH).

---

**Note**

IP phones do not support multicast at 224.x.x.x addresses.

---

**Examples**

The following example sets up a call to extension 7777 for a live audio stream and sends it via multicast:

```bash
Router(config)# ephone-dn 55
Router(config-ephone-dn)# feed ip 239.1.1.1 port 2000 route 10.10.23.3 out-call 7777
```

---

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>auto-cut-through</td>
<td>Enables call completion when an M-lead response is not provided.</td>
</tr>
<tr>
<td>ip source-address</td>
<td>Identifies the IP address and port through which IP phones communicate with a Cisco CME router.</td>
</tr>
<tr>
<td>moh (ephone-dn)</td>
<td>Enables music on hold from a live feed and multicast of the MOH audio stream.</td>
</tr>
<tr>
<td>moh (telephony-service)</td>
<td>Enables music on hold from an audio file.</td>
</tr>
<tr>
<td>multicast moh</td>
<td>Enables multicast of a music-on-hold audio stream.</td>
</tr>
<tr>
<td>signal</td>
<td>Specifies the type of signaling for a voice port.</td>
</tr>
</tbody>
</table>
file text (voice register global)

To generate ASCII text files of the configuration profiles for SIP phones, use the `file text` command in voice register global configuration mode. To return to the default, use the `no` form of this command.

```
file text
no file text
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
System directly generates only binary files for configuration profiles.

**Command Modes**
Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to generate an ASCII text file of the configuration profile for Cisco Unified IP Phone 7905s and 7905Gs, Cisco Unified IP Phone 7912s and 7912Gs, Cisco ATA-186s, or Cisco ATA-188s.

**Examples**
The following example shows how to generate an ASCII text file version of the configuration profiles for Cisco Unified IP Phone 7905s and 7905Gs, Cisco Unified IP Phone 7912s and 7912Gs, Cisco ATA-186s, or Cisco ATA-188s:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# file text
Router(config-register-global)# create profile
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>create profile (voice register global)</td>
<td>Generates the configuration profiles required for SIP phone.</td>
</tr>
<tr>
<td>show voice register profile</td>
<td>Displays the contents of configuration files that are in ASCII text format.</td>
</tr>
</tbody>
</table>
filename

To specify a custom XML file that contains the dial patterns to use for a SIP dial plan, use the `filename` command in voice register dialplan configuration mode. To remove the file, use the `no` form of this command.

`filename`  `filename`
`no`  `filename`

**Syntax Description**

| `filename` | Name of the XML file in flash memory. |

**Command Default**

A custom file is not used for the dial plan.

**Command Modes**

Voice register dialplan configuration (config-register-dialplan)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command selects a custom XML file containing dial patterns for the SIP dial plan. The file specified with this command must be loaded into flash memory. You must use the `type` command to specify the type of phone for which the dial plan is being defined before you can use this command. After you define a dial plan, assign it to a SIP phone by using the `dialplan` command.

The `pattern` command and `filename` command are mutually exclusive. You can use either the `pattern` command to define dial patterns manually, or the `filename` command to select a custom dial pattern file that is loaded in system flash.

If the custom XML file contains any errors, the dial plan might not work properly on the phone.

To remove a dial plan that is created using a custom XML file, use the `reset` command after removing the dial plan from the phone and creating a new configuration profile. Removing a dial plan that uses a dial pattern XML file does not take effect if you restart the phone with the `restart` command.

---

**Note**

This command is not supported for Cisco Unified IP Phone 7905 or 7912.

**Examples**

The following example shows that a custom file named sample.xml is specified for dial plan 2.

```
Router(config)# voice register dialplan 2
Router(config-register-dialplan)# type 7940-7960-others
Router(config-register-dialplan)# filename sample.xml
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>dialplan</code></td>
<td>Assigns a dial plan to a SIP phone.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------------</td>
<td>--------------------------------------------------</td>
</tr>
<tr>
<td><strong>pattern</strong></td>
<td>Defines a dial pattern for a SIP dial plan.</td>
</tr>
<tr>
<td><strong>show voice register dialplan</strong></td>
<td>Displays all configuration information for a specific SIP dial plan.</td>
</tr>
<tr>
<td><strong>type (voice register dialplan)</strong></td>
<td>Defines a phone type for a SIP dial plan.</td>
</tr>
</tbody>
</table>
final

To define the last extension (ephone-dn) in an ephone hunt group, use the `final` command in ephone-hunt configuration mode. To remove this number from the hunt group, use the `no` form of this command.

```
final number
no final
```

### Syntax Description

| number | Extension or phone number. Can be an ephone-dn primary or secondary number, voice-mail number, pilot number of another hunt group, or FXS caller-ID number. |

### Command Default

No final number is defined.

### Command Modes

Ephone-hunt configuration (config-ephone-hunt)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command defines the last extension in a hunt group and the destination of incoming calls to a hunt-group pilot number that are unanswered after being routed through the directory numbers in the hunt group list.

To avoid an infinite loop, use the `max-redirect` command.

### Examples

The following example defines ephone-dn 6000 as the last number of hunt group number 1:

```
Router(config)# ephone-hunt 1 sequential
Router(config-ephone-hunt)# final 6000
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>fwd-final</td>
<td>Specifies the final destination of an unanswered call that has been transferred into a hunt group.</td>
</tr>
<tr>
<td>hops</td>
<td>Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.</td>
</tr>
<tr>
<td>list</td>
<td>Defines the ephone-dns that participate in an ephone hunt group.</td>
</tr>
<tr>
<td>max-redirect</td>
<td>Changes the number of allowable redirects in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>no-reg (ephone-hunt)</td>
<td>Specifies that the pilot number of an ephone hunt group should not register with the H.323 gatekeeper.</td>
</tr>
<tr>
<td>pilot</td>
<td>Defines the ephone-dn that is dialed to reach an ephone hunt group.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------</td>
<td>----------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>preference (ephone-hunt)</td>
<td>Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.</td>
</tr>
<tr>
<td>timeout (ephone-hunt)</td>
<td>Sets the number of seconds after which a call that is not answered is redirected to the next number in the ephone-hunt-group list.</td>
</tr>
</tbody>
</table>
**final (voice hunt-group)**

To define the last extension in a voice hunt group, use the `final` command in voice hunt-group configuration mode. To remove this number from the hunt group, use the `no` form of this command.

### Syntax

**final number**

**no final**

### Syntax Description

<table>
<thead>
<tr>
<th><strong>Syntax</strong></th>
<th><strong>Description</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>number</strong></td>
<td>Telephone or extension number. Can be an E.164 number, voice-mail number, pilot number of another hunt group, or FXS caller-ID number.</td>
</tr>
</tbody>
</table>

### Command Default

No final number is defined in the voice hunt group.

### Command Modes

Voice hunt-group configuration (config-voice-hunt-group)

### Command History

<table>
<thead>
<tr>
<th><strong>Cisco IOS Release</strong></th>
<th><strong>Cisco Product</strong></th>
<th><strong>Modification</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command defines the last extension in a hunt group and the destination of incoming calls to a hunt-group pilot number that are unanswered after being routed through the directory numbers in the hunt group list.

To avoid an infinite loop, if a final number in one hunt group is configured as a pilot number of another hunt group, the pilot number of the first hunt group cannot be configured as a final number in any hunt group.

### Examples

The following example shows how to define extension 6000 as the last number of hunt group 1:

```
Router(config)# voice hunt-group 1 sequential
Router(config-voice-hunt-group)# final 6000
```

### Related Commands

<table>
<thead>
<tr>
<th><strong>Command</strong></th>
<th><strong>Description</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>hops (voice hunt-group)</strong></td>
<td>Defines the number of times that a call is redirected to the next number in a peer hunt-group list before proceeding to the final number.</td>
</tr>
<tr>
<td><strong>list (voice hunt-group)</strong></td>
<td>Defines the numbers that participate in a voice hunt group.</td>
</tr>
<tr>
<td><strong>max-redirect (voice register global)</strong></td>
<td>Changes the current number of allowable redirects in a Cisco CME system.</td>
</tr>
<tr>
<td><strong>timeout (voice hunt-group)</strong></td>
<td>Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.</td>
</tr>
</tbody>
</table>
forward local-calls

To allow internal (local) calls to be forwarded, use the `forward local-calls` command in ephone-dn or ephone-hunt configuration mode. To prevent internal calls from being forwarded, use the `no` form of this command.

```
forward local-calls
no forward local-calls
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Internal calls are forwarded as specified in the ephone-dn or ephone-hunt configuration of the called party.

**Command Modes**

- Ephone-dn configuration (config-ephone-dn)
- Ephone-hunt configuration (config-ephone-hunt)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Internal, or local, calls are defined as those calls that originate from other ephone-dns in the same Cisco Unified CME system.

When the `no forward local-calls` command is used in ephone-dn configuration mode, internal calls to that ephone-dn are not forwarded if the ephone-dn is busy or does not answer. If the ephone-dn is busy, the caller hears a busy signal. If the ephone-dn does not answer, the caller hears a ringback signal. The call is not forwarded even if call forwarding is enabled for the ephone-dn.

When the `no forward local-calls` command is used in ephone-hunt configuration mode, internal calls to a hunt-group pilot number are sent only to the first member of the group. If the first group member is busy, the caller hears a busy signal. If the first group member does not answer, the caller hears a ringback signal. The call is not forwarded to subsequent hunt group members.

**Examples**

In the following example, extension 2222 dials the pilot number 3000 and is forwarded to extension 3011. If 3011 is busy, the caller hears a busy tone. If 3011 does not answer, the caller hears ringback. The call is not forwarded, even after the timeout expires.

```
ephone-hunt 17 sequential
  pilot 3000
  list 3011, 3021, 3031
  timeout 10
  final 7600
  no forward local-calls
```

In the following example, extension 2222 calls extension 3675 and hears ringback or a busy signal. If an external caller reaches extension 3675 and there is no answer, the call is forwarded to extension 4000.
ephone-dn 25
  number 3675
  no forward local-calls
  call-forward noan 4000 timeout 30
**forward local-calls (voice hunt-group)**

To allow local calls to be forwarded, use the `forward local-calls` command in voice hunt-group configuration mode. To prevent local calls from being forwarded, use the `no` form of this command.

```plaintext
forward local-calls to-final
no forward local-calls to-final
```

**Syntax Description**

- `to-final` Prevents local calls from being forwarded to the final destination number.

**Command Default**

Local calls are forwarded as specified in the voice hunt-group configuration of the called party.

**Command Modes**

Voice hunt-group configuration (config-voice-hunt-group)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Local or internal calls are calls originating from a Cisco Unified SIP or Cisco Unified SCCP IP phone in the same Cisco Unified CME system.

Before Cisco Unified CME 9.5, the `no forward local-calls` command was configured in ephone-hunt group to prevent a local call from being forwarded to the next agent.

In Cisco Unified CME 9.5, local calls are prevented from being forwarded to the final destination using the `no forward local-calls to-final` command in parallel or sequential voice hunt-group configuration mode.

When the `no forward local-calls to-final` command is configured in sequential voice hunt-group configuration mode, local calls to the hunt-group pilot number are sent sequentially only to the list of members of the group using the rotary-hunt technique. In case all the group members of the voice hunt group are busy, the caller hears a busy tone. If any of the group members are available but do not answer, the caller hears a ringback tone and is eventually disconnected after the specified timeout. The call is not forwarded to the final destination number.

When the `no forward local-calls to-final` command is configured in parallel voice hunt-group configuration mode, local calls to the hunt-group pilot number are sent parallely to the list of members of the group using the blast technique. In case all the group members of the voice hunt group are busy, the caller hears a busy tone. If any of the group members are available but do not answer, the caller hears a ringback tone and is eventually disconnected after the specified timeout. The call is not forwarded to the final destination number.

**Examples**

The following example shows how to prevent the forwarding of local calls to the final destination in parallel voice hunt group 1:

```plaintext
Router# configure terminal
Router(config)# voice hunt-group 1 parallel
Router(config-voice-hunt-group)# no forward local-calls to-final
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice hunt-group</td>
<td>Enters voice hunt-group configuration mode to create a hunt group for phones in a Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
forwarding local (voice register global)

To use the forwarding-party number and name (the local number and name) in calls forwarded using local hairpin call routing on a SIP phone, use the forwarding local command in voice register global configuration mode. To return to the default, use the no form of this command.

forwarding local
no forwarding local

Syntax Description
This command has no arguments or keywords.

Command Default
Calling-party name and number used.

Command Modes
Voice register global configuration (config-register-global)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
</table>
| 12.4(4)T          | Cisco CME 3.4 | This command was introduced.

Usage Guidelines
This command replaces a calling-party number and name with the local forwarding-party number and name in hairpinned forwarded calls.

Examples
The following example shows how to enable local forwarding:

Router(config)# voice register global
Router(config-register-global)# forwarding local

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-forward b2bua all (voice register dn and voice register pool)</td>
<td>Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.</td>
</tr>
</tbody>
</table>
**from-ring**

To specify that on-hook time stamps for ephone hunt group agents should be updated when calls ring as well as when calls are answered in a longest-idle ephone hunt group, use the `from-ring` command in ephone-hunt configuration mode. To return to the default, use the `no` form of this command.

```
from-ring
no from-ring
```

**Syntax Description**

This command has no keywords or arguments.

**Command Default**

On-hook time stamps are updated only when calls are answered by agents.

**Command Modes**

Ephone-hunt configuration

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used only with longest-idle ephone hunt groups. In a longest-idle hunt group, the algorithm for choosing the next agent to receive a call is based on a comparison of on-hook time stamps. The agent with the smallest on-hook time stamp value is chosen when the next call comes to the hunt group.

This command can be used to specify that on-hook time stamps should be updated when calls ring agents as well as when calls are answered by agents.

The `show ephone-hunt` command displays on-hook time stamps.

**Examples**

The following example defines longest-idle ephone hunt group 1 with a pilot number 7501, a final number 8000, and five numbers in the list. Because the `from-ring` command is used, on-hook time stamps will be recorded when calls ring agents as well as when calls are answered. After a call is redirected three times (makes six hops), it is redirected to the final number, 8000.

```
ephone-hunt 1 longest-idle
pilot 7501
list 7001, 7002, 7023, 7028, 7045
final 8000
from-ring
hops 3
timeout 20
telephony-service
  max-redirect 8
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>show ephone-hunt</code></td>
<td>Displays configuration information, current status, and statistics for ephone hunt groups.</td>
</tr>
</tbody>
</table>
fwd-final

To specify the final destination of a call that has been transferred into a hunt group and is unanswered, use the `fwd-final` command in ephone-hunt configuration mode. To return to the default, use the `no` form of this command.

```
fwd-final {orig-phone|final}
no fwd-final {orig-phone|final}
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>orig-phone</th>
<th>Final destination is the phone that originally answered the call before transferring it to the pilot number of a hunt group.</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>final</td>
<td>Final destination is the last extension in the hunt group as specified in the hunt group configuration.</td>
</tr>
</tbody>
</table>

**Command Default**

Calls are sent to the final number that is specified in the hunt group configuration.

**Command Modes**

Ephone-hunt configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used for routing only unanswered transferred calls. Transferred calls are incoming calls to an ephone hunt group that were previously answered by a Cisco Unified CME extension and transferred into the hunt group.

The `orig-phone` keyword specifies that an unanswered transferred call is routed back to the extension that originally answered the call and transferred it to the hunt group.

The `final` keyword specifies that an unanswered transferred call is routed to the last extension in the hunt group as defined by using the `final` command.

**Examples**

The following example sets up a peer hunt group with three ephone-dns to answer calls. An unanswered transferred call will be routed to the ephone-dn that transferred it to the ephone hunt group pilot number. A DID call that dials the pilot number directly will be routed to extension 7600 if it is unanswered by the hunt group.

```
ephone-hunt 17 peer
  pilot 3000
  list 3011, 3021, 3031
  hops 3
  final 7600
  fwd-final orig-phone
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>final</code></td>
<td>Defines the last extension (ephone-dn) in an ephone hunt group.</td>
</tr>
</tbody>
</table>
fxo hook-flash

To enable display of a flash soft key on a Cisco IP Phones 7940 and 7940G or Cisco IP Phones 7960 and 7960G in a Cisco CallManager Express (Cisco CME) system, use the `fxo hook-flash` command in telephony-service configuration mode. To disable display of the flash soft key, use the `no` form of this command.

```
fxo hook-flash
no fxo hook-flash
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

The flash soft key is disabled.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Certain public switched telephony network (PSTN) services, such as three-way calling and call waiting, require hookflash intervention from the phone user. A soft key labeled flash provides this functionality for the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G users on foreign exchange office (FXO) lines attached to the Cisco CME system. The flash soft key is enabled using the `fxo hook-flash` command.

Once a flash soft key has been enabled on an IP phone, it is available to provide hookflash functionality during all calls except local IP-phone-to-IP-phone calls. Note that hookflash-controlled services can be activated only if they are supported by the PSTN connection that is involved in the call. The availability of the flash soft key does not guarantee that hookflash-based services are actually accessible to the phone user.

The flash soft key display is automatically disabled for local IP-phone-to-IP-phone calls.

This command must be followed by a quick reboot of the phones using the `restart all` command.

**Examples**

The following example enables the flash soft key on the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G:

```
Router(config)# telephony-service
Router(config-telephony)# fxo hook-flash
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>restart (ephone)</td>
<td>Performs a fast reboot of a single phone associated with a Cisco CME router.</td>
</tr>
<tr>
<td>restart (telephony-service)</td>
<td>Performs a fast reboot of one or all phones associated with a Cisco CME router.</td>
</tr>
</tbody>
</table>
fxo hook-flash
Cisco Unified CME Commands: G

- gsm-support, on page 444
- group (lpcor custom), on page 445
- group (telephony-service), on page 446
- group phone, on page 448
- group (voice register global), on page 450
- group (voice register pool), on page 451
To define the gsm support for a Cisco Unified SIP IP phone on Cisco Unified CME, use the `gsm-support` command in voice register pool-type mode. To remove the gsm support, use the `no` form of this command. The `no` form is typically used to override the inherited property of the reference ephone with default value.

```
gsm-support
no gsm-support
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
By default, the gsm-support is not enabled. When `reference-pooltype` is configured, the `gsm-support` value of the reference phone is inherited.

**Command Modes**
Voice Register Pool-type Configuration (config-register-pool-type)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(3)M</td>
<td>Cisco SIP CME 10.0</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to define the maximum number of addon modules for a Cisco Unified SIP IP phone on Cisco Unified CME. When you use the `no` form of this command, the inherited properties of the reference phone is takes precedence over the default value.

**Cisco Unified CME**

The following example shows how to enter voice register pool configuration mode and define the maximum number of addon modules for a Cisco Unified SIP IP phone on Cisco Unified CME:

```
Router(config)# voice register pool-type 9900
Router(config-register-pool-type)# gsm-support
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register pool-type</td>
<td>Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.</td>
</tr>
</tbody>
</table>
group (lpcor custom)

To add a logical partitioning class of restriction (LPCOR) resource group to the custom resource list, use the `group` command in LPCOR custom configuration mode. To remove a resource group, use the `no` form of this command.

```
file

Syntax Description

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>number</td>
<td>Group number of the LPCOR entry. Range: 1 to 64.</td>
</tr>
<tr>
<td>lpcor-group</td>
<td>Name of a LPCOR resource group.</td>
</tr>
</tbody>
</table>

Command Default

LPCOR resource group is not defined.

Command Modes

LPCOR custom configuration (cfg-lpcor-custom)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

Use this command to define all of the LPCOR resource groups that you are provisioning on the Cisco Unified CME router. You must logically partition the resources of the Cisco Unified CME router (trunks and phones) into different LPCOR resource groups so that you can apply the required call restrictions to each group.

Examples

The following example shows a LPCOR configuration with six resource groups:

```
voice lpcor custom
  group 1 sccp_phone_local
  group 2 sip_phone_local
  group 3 analog_phone_local
  group 4 sip_remote
  group 5 sccp_remote
  group 6 isdn_local
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice lpcor enable</td>
<td>Enables LPCOR functionality on the Cisco Unified CME router.</td>
</tr>
<tr>
<td>voice lpcor policy</td>
<td>Creates a LPCOR policy for a resource group.</td>
</tr>
</tbody>
</table>


**group (telephony-service)**

To create a (VRF) group for Cisco Unified CME users and phones, use the `group` command in telephony-service configuration mode. To remove a group, use the `no` form of this command.

```
group  group-tag  [vrf  vrfrname]
no  group
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>group-tag</code></td>
<td>Unique identifier for VRF group being configured. Range 1 to 5</td>
</tr>
<tr>
<td><code>vrf vrfrname</code></td>
<td>(Optional) Name of already-configured VRF to which this VRF group is associated.</td>
</tr>
</tbody>
</table>

### Command Default

No group is configured.

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

By default, VRF groups are associated with a global voice VRF unless you use the `vrf vrfrname` keyword and argument combination to specify otherwise.

If you configure this command, the `ip source-address`, `url` and `cnf-file location` commands in `telephony-service` configuration mode are automatically converted into `group 1` with a default global VRF for nvgen during system upgrade.

If you configure this command and the `cnf-file location` command is configured for `system:`, the per phone or per phone type file for an ephone in the VRF group is created in `system:/its/vrf<group-tag>/`. Local files are still created in `system:/its/`.

If you configure this command and the `cnf-file location` command is configured as `flash:` or `slot0:`, the per phone or per phone type file for an ephone in the VRF group is named `flash:/its/vrf<group-tag>_<filename>` or `slot0:/its/vrf<group tag>_filename`.

The location of the locale files is not affected by configuring a VRF group.

### Examples

The following example shows the configuration for three VRF groups. Group 1 is on a global voice VRF and the other two groups are on data VRFs.

```
telephony-service
sdspfarm conference mute-on # mute-off #
sdspfarm units 4
sdspfarm transcode sessions 10
sdspfarm tag 1 xcode101
sdspfarm tag 2 conf103
group  1
ip source-address 10.1.10.1 port 2000
url directories http://210.1.10.1/localdirectory
!
group  2 vrf data-vrf1
```
```
ip source-address 10.2.10.1 port 2000
!
group 3 vrf data-vrf2
  ip source-address 10.3.10.1 port 2000
!
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ip vrf</td>
<td>Defines a VRF for a router.</td>
</tr>
<tr>
<td>cnf-file location</td>
<td>Specifies a storage location for phone configuration files</td>
</tr>
</tbody>
</table>
**group phone**

To add a phone, including a TAPI-based client application, or a softphone on a PC to a VRF group for Cisco Unified CME, use the `group` command in ephone or ephone-template configuration mode. To remove the `group` configuration, use the `no` form of this command.

```
group
no group
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

By default, this feature is disabled.

**Command Modes**

- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Products</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)T</td>
<td>Cisco Unified CME 7.0(1)</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables to configure the voice VRF group for SIP phones. This command adds a softphone on a PC, an IP phone, or a TAPI client on an IP phone to a VRF group.

VRF groups for users and phones in Cisco Unified CME are created by using the `group` command in telephony-service configuration mode. All SCCP and SIP phones connected to Cisco Unified CME must register through the global voice VRF. TAPI-based client on an IP phone and softphones on a PC must register in Cisco Unified CME through a data VRF.

Before you can use this command, the MAC address for the IP phone being configured must be configured by using the `mac-address` command in ephone configuration mode.

If you use an ephone template to apply a command to an ephone and you also use the same command in ephone configuration mode, the value that you set in ephone configuration mode has priority over the ephone-template configuration.

**Examples**

The following example shows four phones in three VRF groups, two on data VRFs and one on a global voice VRF.

```
telephony-service
sds pfarm conference mute-on # mute-off #
sds pfarm units 4
sds pfarm transcode sessions 10
sds pfarm tag 1 xcode101
sds pfarm tag 2 conf103
group 1
  ip source-address 209.165.201.1 port 2000
  url directories http://209.165.201.1/localdirectory
!
group 2 vrf data-vrf1
  ip source-address 209.165.201.2 port 2000
!```
The following example shows four phones in three VRF groups, two on data VRFs and one on a global voice VRF.

Router(config)# voice register template
Router(config-telephony)# group <group-tag>

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode.</td>
</tr>
<tr>
<td></td>
<td>voice register template</td>
<td>Enters voice register template configuration mode.</td>
</tr>
<tr>
<td></td>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone configuration.</td>
</tr>
<tr>
<td></td>
<td>group (telephony-service)</td>
<td>Creates a VRF group for phones and users in Cisco Unified CME.</td>
</tr>
<tr>
<td></td>
<td>mac-address</td>
<td>Associates the MAC address of a Cisco IP phone with an ephone configuration.</td>
</tr>
</tbody>
</table>
group (voice register global)

To add a phone or a softphone on a PC to a Virtual routing and forwarding (VRF) group for Cisco Unified CME, use the `group` command in voice register global configuration mode. To remove the configuration, use the `no` form of this command. To configure SIP CME support for VRF by provisioning its source address under a group, use the `vrfname` command. To remove the configuration, use the `no` form of this command.

```
group group-tag  
no group  
group group-tag vrf vrfname  
no group vrf vrfname
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>group-tag</strong></td>
<td>Unique identifier of VRF group. Range is from 1 to 5.</td>
</tr>
<tr>
<td><strong>vrfname</strong></td>
<td>Specifies the name of the vrf group.</td>
</tr>
</tbody>
</table>

**Command Default**

By default, this feature is disabled.

**Command Modes**

voice register global (config-register global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables to configure the VRF group for SIP phones. This command is used to configure multiple VRF groups for SIP phones and soft phones on PC or mobile devices registering to CME. Each VRF group can be associated with a specific IP VRF. Phones in this VRF will use the source-address configured under this VRF group to register to CME.

**Example**

The following example shows three different VRF groups that have been configured, a voice VRF, a Data VRF, and a Voice VRF:

```
vovoice register global  
mode cme  
max-dn 100  
max-pool 100

group 1 vrf voice-vrf  
source-address 8.0.0.1

group 2 vrf data-vrf  
source-address 9.0.0.1

group 3 vrf voice-vrf1  
source-address 10.0.0.1
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>group (telephony-service)</code></td>
<td>Creates a VRF group for phones and users in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
group (voice register pool)

To configure multiple virtual routing and forwarding (VRF) groups for SIP phones and soft phones on PC or mobile, use the `group` command in voice register pool configuration or voice register template configuration modes. To remove the configuration, use the `no` form of this command. You can configure up to 5 VRF groups.

To add a phone or a softphone on a PC to a VRF group for Cisco Unified CME, use the `group` command in voice register pool or voice register template configuration modes. To remove the configuration, use the `no` form of this command.

```
group group-tag
no group
```

**Syntax Description**

- `group-tag`: Unique identifier of VRF group. Range is from 1 to 5.

**Command Default**

By default, this feature is disabled.

**Command Modes**

- voice register pool (config-register pool)
- voice register template (config-register template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables to configure the VRF group for SIP phones. The group id configured under voice register global can be associated to the voice register pool or template using the `group <group-tag>` command.

**Example**

The following example shows phones configured in different VRF groups under voice register pool and voice register template modes:

```
voice register pool 1
  group 1

voice register template 1
  group 3
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register global</td>
<td>Enters voice register global configuration mode.</td>
</tr>
<tr>
<td>voice register template</td>
<td>Enters voice register template configuration mode.</td>
</tr>
</tbody>
</table>
group (voice register pool)
Cisco Unified CME Commands: H

- headset auto-answer line, on page 454
- hfs enable, on page 456
- hfs home-path, on page 458
- hlog-block (voice hunt-group), on page 460
- hold-alert, on page 461
- hold-alert (voice register global), on page 464
- hops, on page 465
- hops (voice hunt-group), on page 467
- host-id-check, on page 468
- hunt-group report url, on page 470
- hunt-group statistics write-v2, on page 471
- hunt-group logout, on page 473
- hunt-group report delay hours, on page 476
- hunt-group report every hours, on page 478
- hunt-group statistics write-all, on page 480
- huntstop (ephone-dn and ephone-dn-template), on page 483
- huntstop (voice register dn), on page 487
headset auto-answer line

To enable auto-answer on the specified line when the headset key is engaged, use the **headset auto-answer** command in ephone configuration mode. To disable headset auto-answer for this line, use the **no** form of this command.

```
headset auto-answer line line-number
no headset auto-answer line line-number
```

**Syntax Description**

| **line-number** | Phone line that should be automatically answered. |

**Command Default**

Headset auto-answer is not enabled.

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th><strong>Cisco IOS Release</strong></th>
<th><strong>Cisco Product</strong></th>
<th><strong>Modification</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables headset auto-answer on a particular line. A line, as used in this command, is not identical to a phone button. A line, as used in this command, represents the ability for a call connection on this phone, and the line numbers generally follow a top-to-bottom sequence starting with the number 1.

The following examples represent common situations pertaining to a button:line relationship:

- button 1:1—A single ephone-dn is associated with a single ephone button. Counts as one line.
- button 1o1,2,3,4,5—Five ephone-dns are overlaid on a single ephone button. Counts as one line.
- button 2x1—An ephone button acts as an extension for an overlaid ephone button. Counts as one line.
- Button is unoccupied or programmed for speed-dial. Does not count as a line.

**Examples**

The following example shows how to enable headset auto-answer for line 1 (button 1) and line 4 (button 4), which has overlaid ephone-dns but counts as a single line in this context. In this example, four 1, 2, 3, and 4) buttons are defined for ephone 3.

```
ephone 3
button 1:2 2:4 3:6 4o21,22,23,24,25
headset auto-answer line 1
headset auto-answer line 4
```

The following example shows how to enable headset auto-answer for line 2 (button 2), which has overlaid ephone-dns, and line 3 (button 3), which is an overlay rollover line. In this example, three 1, 2, and 3) buttons are defined for ephone 17.

```
ephone 17
button 1:2 2o21,22,23,24,25 3x2
headset auto-answer line 2
```
headset auto-answer line 3

The following example shows how to enable headset auto-answer for line 2 (button 3) and line 3 (button 5). In this case, the button numbers do not match the line numbers because buttons 2 and 4 are not used.

ephone 25
button 1:2 3:4 5:6
headset auto-answer line 2
headset auto-answer line 3
**hfs enable**

To enable the HTTP File-Fetch Server (HFS) download service on an IP Phone in a Cisco Unified CME system, use the `hfs enable` command in telephony-service configuration mode. To disable the HFS download service, use the `no` form of this command.

```
no hfs enable [port port-number]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
</table>
| `port port-number` | (Optional) Specifies the port where the HFS download service is enabled. **Range is from 1024 to 65535.**

**Note**

If the entered custom HFS port clashes with the underlying IP HTTP port, an error message is displayed and the command is disallowed.

**Command Default**

An IP Phone is unable to download configuration and firmware files through the HFS infrastructure.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(1)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

To enable the HFS download service, the underlying HTTP server must be enabled first using the `ip http server` command because the HFS infrastructure is built on top of an existing IOS HTTP server.

This HFS infrastructure enables multiple HTTP services to co-exist. The HFS download service runs on custom port 6970 but can also share default port 80 with other services. Other HTTP services run on other non-standard ports like 1234.

Use the `hfs enable` command without keyword or argument to enable the HFS download service on the default HTTP server port.

**Examples**

The following example shows how to enable the HFS download service for Cisco Unified SIP IP Phone 7945 on port 65500:

```
Router(config)# ip http server
Router(config)# ip http port 1234
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# load 7945 SIP45.8.3.3S
Router(config-register-global)# create profile
Router(config-register-global)# exit
Router(config)# telephony-service
Router(config-telephony)# hfs enable port 65500
```

The following examples show how to enable the HFS service on default and custom ports.

For the default port:

```
Router(config)# ip http server
```
Router(config)# ip http port 1234
Router (config)# telephony-service
Router(config-telephony)# hfs enable

For the custom port:

Router(config)# ip http server
Router (config)# ip http port 1234
Router (config)# telephony-service
Router(config-telephony)# hfs enable port 6970

The following examples show how an entered custom HFS port clashes with the underlying ip http port. Port 6970 is configured as the IP HTTP port. When the HFS port is configured with the same value, an error message is displayed to show that the port is already in use.

Router(config)# ip http server
Router (config)# ip http port 6970
Router (config)# telephony-service
Router (config-telephony)# hfs enable port 6970
Invalid port number or port in use by other application

The HFS port number is already in use by the underlying IP HTTP server so an HFS port that is different from the underlying IP HTTP port must be used.

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>create profile (voice register global)</td>
<td>Generates the configuration profile files required for SIP phones.</td>
</tr>
<tr>
<td>ip http port</td>
<td>Specifies the port where the HTTP service is run.</td>
</tr>
<tr>
<td>ip http server</td>
<td>Enables the underlying IOS HTTP server of the the HFS infrastructure.</td>
</tr>
</tbody>
</table>
hfs home-path

To set up a home-path for IP phone firmware files, use the `hfs home-path` command in telephony-service configuration mode. To remove a directory as a home-path for phone files, use the `no` form of this command.

```
hfs home-path path
no hfs home-path path
```

**Syntax Description**

<table>
<thead>
<tr>
<th>path</th>
<th>Directory path where only IP phone firmware and configuration files are stored.</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Note</strong></td>
<td>The administrator must store the phone firmware files at the location set as the home path directory</td>
</tr>
</tbody>
</table>

**Command Default**

No directory path is specified for the storage of IP phone firmware and configuration files.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(1)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `hfs home-path` command to specify a directory path as the home-path to store IP phone firmware files.

**Examples**

The following example shows how to set up a home-path for IP phone firmware files in Cisco Unified CME:

```
Router(config)# telephony service
Router(config-telephony)# hfs home-path flash:/cme/loads/
```

The following example shows how a new directory called phone-load can be created under the root directory of the flash memory and set as the hfs home-path:

```
cassini-c2801# mkdir flash:phone-loads
Create directory filename [phone-loads]?
Created dir flash:phone-loads
cassini-c2801# sh flash:
-#- --length-- -----date/time------ path
1 13932728 Mar 22 2007 15:57:38 +00:00 c2801-ipbase-mz.124-1c.bin
2 33510140 Sep 18 2010 01:21:56 +00:00 rootfs9951.9-0-3.sebn
3 143604 Sep 18 2010 01:22:20 +00:00 aboot9951.111909R1-9-0-3.sebn
4 1249 Sep 18 2010 01:22:40 +00:00 sip9951.9-0-3.loads
5 66996 Sep 18 2010 01:23:00 +00:00 skern9951.022809R2-9-0-3.sebn
6 10724 Sep 18 2010 00:59:48 +00:00 dkern9951.100609R2-9-0-3.sebn
7 1507064 Sep 18 2010 01:00:24 +00:00 kern9951.9-0-3.sebn
8 0 Jan 5 2011 02:03:46 +00:00 phone-loads
14819328 bytes available (49192960 bytes used)
cassini-c2801# conf t
```

Enter configuration commands, one per line. End with CNTL/Z.

cassini-c2801(config)#tele
cassini-c2801(config)#telephony-service
cassini-c2801(config)#hfs home
cassini-c2801(config-telephony)#hfs home-path flash:?
WORD

cassini-c2801(config-telephony)#hfs home-path flash:phone-loads

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>hfs enable</td>
<td>Enables the HFS download service on an IP Phone in a Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
hlog-block (voice hunt-group)

To disable agent status control (logout/login) for voice hunt group on SIP or SCCP phones using Hlog softkey or by using FAC, use the `hlog-block` command in voice hunt-group configuration mode. To remove the configuration, use the `no` form of this command.

```
hlog-block
no hlog-block
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
By default, this command is disabled.

**Command Modes**
voice hunt group configuration (config-voice-hunt-group)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3)M5</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command disables the agent status control such that SIP or SCCP phones are not be able to logout/login from the voice hunt group using Hlog/FAC.

**Examples**
The following example shows how the voice hunt group hlog-block option is enabled for a phone:

```
Router(config)# voice hunt-group 1 parallel
Router(config-voice-hunt-group)# hlog-block
```
**hold-alert**

To set a repeating audible alert notification when a call is on hold on a Cisco Unified IP phone, use the `hold-alert` command in ephone-dn or ephone-dn-template configuration mode. To disable this feature, use the `no` form of this command.

```
hold-alert timeout {idle|originator|shared|shared-idle} [recurrence recurrence-timeout] [ring-silent-dn]
no hold-alert timeout {idle|originator|shared|shared-idle} [recurrence recurrence-timeout] [ring-silent-dn]
```

<table>
<thead>
<tr>
<th><code>timeout</code></th>
<th>Interval after which an audible alert notification is repeated, in seconds. Range is from 15 to 300. There is no default.</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>idle</code></td>
<td>Alerts only when the phone is idle.</td>
</tr>
<tr>
<td><code>originator</code></td>
<td>Alerts whether the phone is idle or busy.</td>
</tr>
<tr>
<td><code>shared</code></td>
<td>Alerts only when the extension is idle but alerts all phones that share the line.</td>
</tr>
<tr>
<td><code>shared-idle</code></td>
<td>Alerts all idle phones that share the line.</td>
</tr>
<tr>
<td><code>recurrence</code></td>
<td>Alerts recurrence after first timeout.</td>
</tr>
<tr>
<td><code>recurrence-timeout</code></td>
<td>Call on-hold recurrence timeout in seconds. Range is from 2 to 300.</td>
</tr>
<tr>
<td><code>ring-silent-dn</code></td>
<td>Rings the silent DN.</td>
</tr>
</tbody>
</table>

**Command Default**

Audible alert notification for on-hold calls is disabled. Only a visual indication is provided.

**Command Modes**

Ephone-dn configuration (config-ephone)
Ephone-dn-template configuration (config-ephone-dn-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-dn-template configuration mode. The shared-idle option and ring-silent-dn parameter were introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>15.1(3)T2</td>
<td>Cisco Unified CME 8.5</td>
<td>The recurrence parameter was introduced.</td>
</tr>
<tr>
<td>15.1(4)M1</td>
<td>Cisco Unified CME 8.6</td>
<td>The recurrence parameter was introduced.</td>
</tr>
</tbody>
</table>
Usage Guidelines

Use the **hold-alert** command to set an audible alert notification on a Cisco Unified IP phone to remind the phone user that a call is on hold. The **timeout** argument specifies the time interval in seconds from the time the call is placed on hold to the time the on-hold audible alert is generated. The alert is repeated every **timeout** seconds.

When the **idle** keyword is enabled, a one-second burst of ringing on the phone is generated on the IP phone that placed the call in the hold state, but only if the phone is in the idle state. If the phone is in active use, no on-hold alert is generated.

When the **originator** keyword is enabled, a one-second burst of ringing is generated on the phone that placed the call in the hold state, but only if the phone is in the idle state. If the phone is in use on another call, an audible beep (call-waiting tone) is generated.

When the **shared** keyword is enabled, a one-second ring burst is generated for all the idle phones that share the extension with the on-hold call. Phones that are in use do not receive an audio beep (call-waiting tone) alert. Only the phone that placed the call on hold hears a call-waiting beep if it is busy.

When the **shared-idle** keyword is enabled, a one-second ring burst is generated for all the idle phones that share the line with the on-hold call.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example sets audible alert notification to idle on extension 1111:

```bash
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1111
Router(config-ephone-dn)# name phone1
Router(config-ephone-dn)# hold-alert 100 idle
```

The following example uses an ephone-dn template to set audible alert notification for extension 1111 to only occur when the phone is idle:

```bash
Router(config)# ephone-dn-template 3
Router(config-ephone-dn-template)# hold-alert 100 idle
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1111
Router(config-ephone-dn)# name phone1
Router(config-ephone-dn)# ephone-dn-template 3
```

The following example uses an ephone-dn to set an additional timeout value between 2 and 300.

```bash
Router(config-ephone-dn)# hold-alert
  <15-300> call on-hold timeout in seconds
Router(config-ephone-dn)# hold-alert 15 idle
  alert on-hold timeout in seconds
Router(config-ephone-dn)# hold-alert 15 idle
  alert on-hold originator only if idle
Router(config-ephone-dn)# hold-alert 15 idle
  alert on-hold originator always
Router(config-ephone-dn)# hold-alert 15 idle
  alert all phones that share the line
Router(config-ephone-dn)# hold-alert 15 idle
  alert all idle phones that share the line
Router(config-ephone-dn)# hold-alert 15 idle
  alternate alert recurrence timeout after first
Router(config-ephone-dn)# hold-alert 15 idle recurrence
  alert recurrence timeout after first
Router(config-ephone-dn)# hold-alert 15 idle recurrence
  alternate alert recurrence timeout after first
Router(config-ephone-dn)# hold-alert 15 idle recurrence
  <2-300> call on-hold recurrence timeout in seconds
Router(config-ephone-dn)# hold-alert 15 idle recurrence
  <2-300> call on-hold recurrence timeout in seconds
Router(config-ephone-dn)# hold-alert 15 idle recurrence
  ring-silent-dn ring the silent DN
```
The following example uses an ephone-dn-template to set an additional timeout value between 2 and 300.

Router(config-ephone-dn-template)# hold-alert
<15-300> call on-hold timeout in seconds
Router(config-ephone-dn-template)# hold-alert 15
idle    alert on-hold originator only if idle
originator alert on-hold originator always
shared   alert all phones that share the line
shared-idle alert all idle phones that share the line
Router(config-ephone-dn-template)# hold-alert 15 idle
recurrence alternate alert recurrence timeout after first
ring-silent-dn ring the silent DN
Router(config-ephone-dn-template)# hold-alert 15 idle recurrence
<2-300> call on-hold recurrence timeout in seconds
Router(config-ephone-dn-template)# hold-alert 15 idle recurrence 3
ring-silent-dn ring the silent DN

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>ephone-dn-template (ephone-dn)</td>
<td>Applies ephone-dn-template to the ephone-dn being configured.</td>
</tr>
</tbody>
</table>
**hold-alert (voice register global)**

To enable a one-time audible alert notification for a call still on hold after a subsequent call has ended in Cisco Unified CME, use the command in voice register global configuration mode. To disable this feature, use the `no` form of this command.

```
hold-alert
no hold-alert
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Audible alert notification for on-hold calls is disabled.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables a one time audible alert notification on all supported SIP phones in a Cisco Unified CME system to remind the phone user that a call is on hold after a subsequent call has ended. The alert is not repeated and does not alert until a subsequent call ends after the original call was placed on hold. This applies globally to all SIP CME Phones.

**Note**

This command does not apply to Cisco ATAs that have been configured for SIP in Cisco Unified CME.

**Examples**

The following example shows how to set audible alert notification on SIP phones for on-hold calls:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# hold-alert
Router(config-register-global)# create profile
! Restart phone(s) to update config file change
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-waiting (voice register pool)</td>
<td>Enables call waiting on a SIP phone.</td>
</tr>
<tr>
<td>mode (voice register global)</td>
<td>Enables the mode for provisioning SIP phones in a Cisco CallManager Express (Cisco CME) system.</td>
</tr>
</tbody>
</table>
hops

To define the number of times that a call can proceed to the next ephone-dn in a peer or longest-idle ephone hunt group before the call proceeds to the final ephone-dn, use the **hops** command in ephone hunt configuration mode. To return to the default number of hops, use the **no** form of this command.

**hops number**

**no hops number**

**Syntax Description**

| number | Number of hops before the call proceeds to the final ephone-dn. Range is from 2 to 20, but the value must be less than or equal to the number of extensions that are specified in the **list** command. Default automatically adjusts to the number of hunt group members. |

**Command Default**

The number of hops automatically adjusts to the number of ephone hunt group members.

**Command Modes**

Ephone-hunt configuration (config-ephone-hunt)

**Command Modes**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.3(7)T</td>
<td>Cisco CME 3.1</td>
<td>The maximum number of hops was restricted to the number of extensions specified in the <strong>list</strong> command.</td>
</tr>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>Increased maximum number of hops to 20.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The default was changed from 2 hops to automatically adjust the number of hops to the number of ephone hunt group members.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>The modification to change the default was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is valid only for peer and longest-idle ephone hunt groups in Cisco Unified CallManager Express systems.

This command is required when you are configuring the **auto logout** command for peer and longest-idle hunt groups.

**Examples**

The following example sets the number of hops to 6 for peer hunt group 3:

```
Router(config)# ephone-hunt 3 peer
Router(config-ephone-hunt)# hops 6
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>auto logout</td>
<td>Enables the automatic change of an ephone hunt group agent's ephone-dn to not-ready status.</td>
</tr>
<tr>
<td>final</td>
<td>Defines the last ephone-dn in an ephone hunt group.</td>
</tr>
<tr>
<td>list</td>
<td>Defines the ephone-dns that participate in an ephone hunt group.</td>
</tr>
<tr>
<td>max-redirect</td>
<td>Changes the current number of allowable redirects in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>no-reg (ephone-hunt)</td>
<td>Specifies that the pilot number of this ephone hunt group should not register with the H.323 gatekeeper.</td>
</tr>
<tr>
<td>pilot</td>
<td>Defines the ephone-dn that is dialed to reach an ephone hunt group.</td>
</tr>
<tr>
<td>preference (ephone-hunt)</td>
<td>Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.</td>
</tr>
<tr>
<td>timeout (ephone-hunt)</td>
<td>Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.</td>
</tr>
</tbody>
</table>
hops (voice hunt-group)

To define the number of times that a call can hop to the next number in a peer hunt group before the call proceeds to the final number, use the `hops` command in voice hunt-group configuration mode. To return to the default number of hops, use the `no` form of this command.

```
hops number
no hops
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>number</code></td>
<td>Number of hops before the call proceeds to the final number. Range is 2 to 10, but the value must be less than or equal to the number of extensions that are specified in the <code>list</code> command. The default is the same number as there are destinations defined under the <code>list</code> command.</td>
</tr>
</tbody>
</table>

**Command Default**
The default is the number of `directory-number` arguments configured in the `list` command.

**Command Modes**
Voice hunt-group configuration (config-voice-hunt-group)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command is valid only for peer or longest-idle voice hunt groups in Cisco Unified CME systems.

**Examples**
The following example shows how to set the number of hops to 6 for peer voice hunt group 1:

```
Router(config)# voice hunt-group 1 peer
Router(config-voice-hunt-group)# list 1000, 1001, 1002, 1003, 1004, 1005, 1006, 006, 1007, 1008, 1009
Router(config-voice-hunt-group)# hops 6
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>final (voice hunt-group)</td>
<td>Defines the last extension in a voice hunt group.</td>
</tr>
<tr>
<td>list (voice hunt-group)</td>
<td>Defines the directory numbers that participate in a hunt group.</td>
</tr>
<tr>
<td>timeout (voice hunt-group)</td>
<td>Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list and defines the last directory number in the hunt group.</td>
</tr>
</tbody>
</table>
**host-id-check**

To configure host-id-check option for a vpn-profile, use the **host-id-check** command in vpn-profile configuration mode. To disable the host-id-check configuration, use the **no** form of this command.

```
host-id-check [{enable|disable}]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>enable</td>
<td>Enables host-id-check option in a vpn-profile.</td>
</tr>
<tr>
<td>disable</td>
<td>Disables host-id-check option in a vpn-profile.</td>
</tr>
</tbody>
</table>

**Command Default**

Host-id-check option is enabled.

**Command Modes**

Vpn-profile configuration (conf-vpn-profile)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to configure host-id-check option for a vpn-profile. This host ID check enhances the security by parsing the host name or the IP from latest URL of the VPN concentrator to check against the subjectAltNames field within the certificate, if the subjectAltNames existed. This check is performed by the phone.

**Examples**

The following example shows the host-id-check option enabled in vpn-profile 2 and disabled in vpn-profile 1:

```
Router# show run
!
voice service voip
ip address trusted list
ipv4 20.20.20.1
vpn-group 1
vpn-gateway 1 https://9.10.60.254/SSLVPNphone
vpn-trustpoint 1 trustpoint cme_cert root
vpn-hash-algorithm sha-1
vpn-profile 1
keepalive 50
host-id-check disable
vpn-profile 2
mtu 1300
password-persistent enable
host-id-check enable
sip
!
voice class media 10
media flow-around
!
!
voice register global
max-pool 10
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>vpn-profile</code></td>
<td>Defines a VPN-profile.</td>
</tr>
</tbody>
</table>
**hunt-group report url**

To set the filename parameters and the URL path where hunt group call statistics are sent using TFTP, use the `hunt-group report url` command in telephony service mode. To disable this feature, use the `no` form of this command.

```
hunt-group report url {prefix|suffix}
nohunt-group report url {prefix|suffix}
```

<table>
<thead>
<tr>
<th>Option</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>prefix</td>
<td>Provides the prefix of the filename to which the statistics are written. The prefix of the filename appears at the end of the URL.</td>
</tr>
<tr>
<td>suffix</td>
<td>Provides the suffix of the filename to which the statistics are written. Range is &lt;0-1&gt; to &lt;1-200&gt;.</td>
</tr>
</tbody>
</table>

**Command Default**

This command is disabled by default.

**Command Modes**

- telephony-service (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>hunt-group report delay hours</code></td>
<td>Delays the automatic transfer of Cisco CME B-ACD call statistics to a file.</td>
</tr>
<tr>
<td><code>hunt-group report every hours</code></td>
<td>Sets the hourly interval after which Cisco CME B-ACD call statistics are automatically transferred to a file.</td>
</tr>
</tbody>
</table>
**hunt-group statistics write-v2**

To write all the ephone hunt and voice hunt group statistics to a file along with total logged in and logged out time for agents, use the `hunt-group statistics write-v2` command in privileged EXEC mode.

**Syntax Description**

| Location | URL or filename to which the statistics is written. |

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(2)T</td>
<td>Cisco Unified CME 9.5</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.6(3)M</td>
<td>Cisco Unified CME 11.5</td>
<td>This command was enhanced to add statistics for total logged in and logged out time for voice hunt group.</td>
</tr>
<tr>
<td>16.3.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `hunt-group statistics write-v2` command to write out, in hourly increments, all the ephone and voice hunt group statistics for the past seven days, along with total logged in and logged out time for agents. This command is intended to be used when normal hunt group statistics collection is interrupted, perhaps due to TFTP server failure. If applicable, the TFTP statistics report consists of both ephone and voice hunt statistics.

**Note**

On the day that daylight saving time adjusts the time back by one hour at 2 a.m. each year, the original 1 a.m. to 2 a.m. statistics for that day are lost because they are overwritten by the new 1 a.m. to 2 a.m. statistics.

**Examples**

The following example shows how the `hunt-group statistics write-v2` command writes a combination of ephone and voice hunt group statistics to a file in TFTP server 202.153.144.25:

```
Router# hunt-group statistics write-v2 tftp://202.153.144.25/cmeteam/stats
Writing out all hunt group statistics to tftp://202.153.144.25/cmeteam/stats
01:47:08 UTC Mon Mar 21 2016,

EPHONE HUNT GROUP STAT,
01, Sat 00:00 - 01:00, HuntGp, 02, 02, 00001, 00001, 00000, 00007, 00007, 000001, 000001, 00000, 00000, 00000, 00000,
01, Sat 00:00 - 01:00, Agent, 5012, 00001, 000001, 000001, 000000, 000000, 000000, 000000, 000001, 000001, 000000, 000000,
01, Sat 00:00 - 01:00, Queue, 00000, 00001, 00000, 00006, 00006, 00000, 00000, 00000, 00000, 00000, 00000, 00000,
01, Sat 01:00 - 02:00, HuntGp, 00, 00, 00000, 00000, 00000, 00000, 00000, 00000, 000000, 000000,
01, Sat 01:00 - 02:00, Agent, 00, 00, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 000000, 000000,
01, Sat 01:00 - 02:00, Queue, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000,
01, Sat 02:00 - 03:00, HuntGp, 00, 00, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 000000, 000000,
01, Sat 02:00 - 03:00, Agent, 5012, 00000, 000001, 000001, 000000, 000000, 000000, 000000, 000000, 000000,
01, Sat 02:00 - 03:00, Queue, 00000, 00001, 00000, 00006, 00006, 00000, 00000, 00000, 00000, 00000, 00000, 00000,
01, Sat 03:00 - 04:00, HuntGp, 00, 00, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 000000, 000000,
01, Sat 03:00 - 04:00, Agent, 00, 00, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 000000, 000000,
01, Sat 03:00 - 04:00, Queue, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000,
01, Sat 04:00 - 05:00, HuntGp, 00, 00, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 000000, 000000,
01, Sat 05:00 - 06:00, HuntGp, 00, 00, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 000000, 000000,
01, Sat 05:00 - 06:00, Agent, 00, 00, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 000000, 000000,
01, Sat 05:00 - 06:00, Queue, 00000, 00001, 00000, 00006, 00006, 00000, 00000, 00000, 00000, 00000, 00000, 00000,
```
Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>hunt-group report delay hours</strong></td>
<td>Delays hunt group statistics collection for a specified number of hours.</td>
</tr>
<tr>
<td><strong>hunt-group report every hours</strong></td>
<td>Sets the hourly interval at which Cisco Unified CME B-ACD call statistics are automatically transferred to a file.</td>
</tr>
<tr>
<td><strong>hunt-group report url</strong></td>
<td>Sets filename parameters and the URL path where Cisco Unified CME B-ACD call statistics are to be sent using TFTP.</td>
</tr>
<tr>
<td><strong>hunt-group statistics write-all</strong></td>
<td>Writes all ephone and voice hunt group statistics to a file.</td>
</tr>
<tr>
<td><strong>show voicehunt-groupstatistics</strong></td>
<td>Displays voice hunt group statistics.</td>
</tr>
<tr>
<td><strong>show ephone-hunt statistics</strong></td>
<td>Displays ephone hunt group statistics.</td>
</tr>
</tbody>
</table>
hunt-group logout

To set the hunt-group logout options, use the `hunt-group logout` command with DND, Hlog, notify, and threshold keywords in telephony-service configuration mode. To return to the default, use the `no` form of this command.

```
hunt-group logout [{DND|HLog|notify|threshold number}]
no hunt-group logout [{DND|HLog|notify|threshold number}]
```

### Syntax Description

<table>
<thead>
<tr>
<th>DND</th>
<th>Agent phones do not answer the number of calls specified in the <code>auto logout</code> command and are automatically placed in both DND status and not-ready status. The HLog soft key is not displayed on phones.</th>
</tr>
</thead>
<tbody>
<tr>
<td>HLog</td>
<td>Agent phones do not answer the number of calls specified in the <code>auto logout</code> command and are automatically placed only in not-ready status. The HLog soft key is displayed on phones in addition to the DND soft key.</td>
</tr>
<tr>
<td>notify</td>
<td>Enables logout call in queue notification on HLog PLK button.</td>
</tr>
<tr>
<td>threshold number</td>
<td>Defines the boundary value by which how the Hlog PLK indicates the number of calls in queue on the logout agent’s phone. Range is 1 to 65535.</td>
</tr>
</tbody>
</table>

### Command Default

DND and HLog functionality is not separate and the HLog soft key will not be displayed on phones. The default is DND.

The default for threshold is disabled and the LED on the HLOG PLK blinks slow in amber as long as there are calls in queue.

The default for notify is disabled and has no LED display.

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>15.2(2)T2</td>
<td>Cisco Unified CME 9.1</td>
<td>The <code>notify</code> and <code>threshold</code> keywords were added.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

When Do Not Disturb (DND) functionality is activated, no calls are received at the phone, including ephone hunt group calls. DND is activated and canceled using the DND soft key or the DND feature access code (FAC).

When HLog functionality is activated, hunt-group agents are placed in not-ready status and hunt-group calls are blocked from the phone. Other calls that directly dial the phone’s extension numbers are still received at the phone. HLog is activated and canceled using the HLog soft key or an HLog FAC.

If the `auto logout` command is used, the Automatic Agent Status Not-Ready feature is invoked for an ephone hunt group. This feature is triggered when an ephone-dn member does not answer a specified number of ephone hunt group calls. The following actions take place:
If the `hunt-group logout HLog` command has been used, the agent is placed in not-ready status. The agent’s ephone-dn will not receive further hunt group calls but will receive calls that directly dial the ephone-dn’s extension numbers. An agent in not-ready status can return to ready status by pressing the HLog soft key or by using the HLog FAC.

If the `hunt-group logout HLog` command has not been used or if the `hunt-group logout DND` command has been used, the phone on which the ephone-dn appears is placed into DND mode, in which the ephone-dn does not receive any calls at all, including hunt-group calls. The red lamp on the phone lights to indicate DND status. An agent in DND mode can return to ready status by pressing the DND soft key or by using the DND FAC.

The DND and HLog keywords are exclusive and are used to enable separate handling of DND and HLog functionality for hunt-group agents and to display the HLog softkey on phones.

The notify and threshold keywords are used to enable the indication of the calls in queue for logout agents using the HLog Programmable Line Key.

If the threshold number is enabled, the LED on the Hlog PLK blinks slow in amber for the number of calls in queue less than the threshold and blinks fast in red for those equal or beyond the threshold. This command will not take effect if `hunt-group logout notify` is disabled.

When an agent who is a dynamic member of a hunt group is in not-ready status, the agent’s slot in the ephone hunt group is not relinquished. It remains reserved by the agent until the agent leaves the group.

The following example creates hunt group 3 with three agents (extensions 1001, 1002, and 1003). It specifies that after one unanswered call, an agent should be put into not-ready status but not into DND status.

```
Router(config)## telephony-service
Router(config-telephony)## hunt-group logout HLog
Router(config-telephony)## exit

Router(config)## ephone-hunt 3 peer
Router(config-ephone-hunt)## pilot 4200
Router(config-ephone-hunt)## list 1001, 1002, 1003
Router(config-ephone-hunt)## timeout 10
Router(config-ephone-hunt)## auto logout
Router(config-ephone-hunt)## final 4500
```

The following example sets the value of threshold to 2:

```
Router(config)## telephony-service
Router(config-telephony)## hunt-group logout ?
  DND    logout using DND softkey or PLK
  HLog   logout using HLog softkey or PLK
  notify enable logout call in queue notification on HLog PLK button
  threshold configure logout call in queue threshold
Router(config-telephony)## hunt-group logout threshold ?
  <1-65535> number of calls in queue
Router(config-telephony)## no hunt-group logout notify
Router(config-telephony)## no hunt-group logout threshold
  % Incomplete command.

Router(config-telephony)## no hunt-group logout threshold 2
```
Router(config-telephony)# **no hunt-group logout** ?
   DND logout using DND softkey or PLK

HLog logout using HLog softkey or PLK

   notify enable logout call in queue notification on HLog PLK button
   threshold configure logout call in queue threshold
Router(config-telephony)# **no hunt-group logout dnd**
Router(config-telephony)# **no hunt-group logout hlog**

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>auto logout</strong></td>
<td>Enables the automatic change of an agent’s ephone-dn to not-ready status after a specified number of hunt-group calls are not answered.</td>
</tr>
<tr>
<td><strong>feature-button</strong> index <code>&lt;feature identifier</code> &gt; <code>[label label ]</code></td>
<td>Enables the feature button configuration on a line key.</td>
</tr>
</tbody>
</table>
hunt-group report delay hours

To delay the automatic transfer of Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD) call statistics to a file, use the **hunt-group report delay hours** command in telephony-service configuration mode. To remove the delay setting, use the **no** form of this command.

### Syntax Description

**Syntax:**

```
hunt-group report delay number hours
no hunt-group report delay number hours
```

**number**
Number of hours by which the collection of statistics can be extended for the statistics collection periods configured with the **hunt-group report every hours** command. The range is from 1 to 10.

### Command Default

Hunt-group report is delayed for one hour.

### Command Modes

- **Telephony-service configuration (config-telephony)**

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command is used for Cisco CME basic automatic call distribution (B-ACD) and auto-attendant (AA) service only.

The **hunt-group report delay hours** command is used as part of a statistics reporting configuration that allows Cisco CME B-ACD call statistics to be sent automatically to files using TFTP. For detailed information, see [Cisco CME B-ACD and Tel Call-Handling Applications](#).

Statistics are collected and stored (**statistics collect** command and **hunt-group report url** command) in specified intervals (**hunt-group report every hours** command). The default is for the statistics to be collected one hour after the specified interval. Because calls are counted when they end, some of the longer calls may not be counted. For example, if there is a call from 1:35 p.m. to 3:30 p.m., the interval is 1 hour, and there is no delay, TFTP will write the 1 p.m. to 2 p.m. statistics at 3 p.m. However, at 3 p.m., the 1:35 p.m. call is still active, so the call will not be counted at that time as occurring in the 1 p.m. to 2 p.m. time slot. When the call finishes at 3:30 p.m., it will be counted as occurring from 1 p.m. to 2 p.m. The **show hunt-group** command will report it, but TFTP will have already sent out its report. To include the 1:35 p.m. call, you could use the **hunt-group report delay hours** command to delay TFTP statistics reporting for an extra hour so the 1 p.m. to 2 p.m. report will be written at 4 p.m. instead of 3 p.m.

### Examples

The following example shows a configuration in which statistics are reported for B-ACD calls that occur within three-hour time frames, but the collection of the statistic collection is extended for an extra hour to include calls that did not end within the three-hour time period:

```
Router(config)# telephony-service
Router(config-telephony)# hunt-group report every 3 hours
Router(config-telephony)# hunt-group report delay 1 hours
```
The following is an example of a report that the previous configuration might send to a file if the `statistics collect` command was entered at 18:20:

```
23:00:00 UTC Tue Dec 20 2004,
01, Tue 18:00 - 19:00, HuntGp, 02, 01, 00005, 00002, 0003, 0006, 00001, 00001, 0011,
01, Tue 19:00 - 20:00, HuntGp, 02, 02, 00000, 00000, 0000, 0000, 000000, 000000, 0000,
01, Tue 20:00 - 21:00, HuntGp, 02, 02, 00006, 00003, 0003, 0009, 000001, 000003, 0012,
```

Statistics collection has to take place for at least three hours for the statistics to be written to a file. The following is a chronology of events:

- At 19:00, the statistics collection was active for 40 minutes, so no statistics were written to file.
- At 20:00, the statistics collection was active for 1 hour and 40 minutes, so no statistics were written to file.
- At 21:00, the statistics collection was active for 2 hours and 40 minutes, so no statistics were written to file.
- At 22:00, the statistics collection was active for 3 hours and 40 minutes but there is a one-hour delay, so no statistics were written to file.
- At 23:00 the statistics were written to a file using TFTP.

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>hunt-group report every hours</code></td>
<td>Sets the hourly interval after which Cisco CME B-ACD call statistics are automatically transferred to a file.</td>
</tr>
<tr>
<td><code>hunt-group report url</code></td>
<td>Sets filename parameters and the URL path where Cisco CME B-ACD call statistics are to be sent using TFTP.</td>
</tr>
<tr>
<td><code>statistics collect</code></td>
<td>Enables the collection of Cisco CME B-ACD call data for an ephone hunt group.</td>
</tr>
</tbody>
</table>
**hunt-group report every hours**

To set the hourly interval at which Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD) call statistics are automatically transferred to a file, use the `hunt-group report every hours` command in telephony-service configuration mode. To remove the interval setting, use the `no` form of this command.

**Syntax Description**

| **number** | Number of hours after which auto-attendant (AA) call statistics are collected and reported. The range is from 1 to 84. |

**Command Default**

No hourly interval is configured.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used for Cisco CME basic automatic call distribution (B-ACD) and auto-attendant (AA) service only.

The `hunt-group report every hours` command is used as part of a statistics reporting configuration that allows Cisco CME B-ACD call statistics to be sent automatically to files by means of TFTP. For detailed information, see Cisco CME B-ACD and Tcl Call-Handling Applications.

Because calls are counted when they end, some of the longer calls may not be counted in the report. To delay the time in which statistics are collected and transferred you may configure a delay time with the `hunt-group report delay hours` command.

**Examples**

The following example sets the statistics collection to occur every three hours. There is no delay.

```
Router(config)# telephony-service
Router(config-telephony)# hunt-group report every 3 hours
```

The following is an example of a report that the previous configuration might send to a file if the `statistics collect` command was entered at 18:20:

```
22:00:00 UTC Tue Dec 20 2005,
01, Tue 18:00 - 19:00, HuntGp, 02, 01, 00005, 00002, 0003, 0006, 00001, 00001, 0011,
01, Tue 19:00 - 20:00, HuntGp, 02, 02, 00000, 00000, 0000, 0000, 00000, 00000, 0000,
01, Tue 20:00 - 21:00, HuntGp, 02, 02, 00006, 00003, 0003, 0009, 00001, 00003, 0012,
```

Statistics collection has to take place for at least three hours for the statistics to be written to a file. The following is a chronology of events:

- At 19:00, the statistics collection was active for 40 minutes, so no statistics were written to file.
• At 20:00, the statistics collection was active for 1 hour and 40 minutes, so no statistics were written to file.
• At 21:00, the statistics collection was active for 2 hours and 40 minutes, so no statistics were written to file.
• At 22:00, the statistics collection was active for 3 hours and 40 minutes, so statistics were written to a file using TFTP.

If the previous example were configured for a delay of one hour using the **hunt-group report delay 1 hours** command, the statistics would be written one hour later at 23:00.

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>hunt-group report delay hours</strong></td>
<td>Delays the automatic transfer of Cisco CME B-ACD call statistics to a file.</td>
</tr>
<tr>
<td><strong>hunt-group report url</strong></td>
<td>Sets filename parameters and the URL path where Cisco CME B-ACD call statistics are to be sent using TFTP.</td>
</tr>
<tr>
<td><strong>statistics collect</strong></td>
<td>Enables the collection of Cisco CME B-ACD call statistics for an ephone hunt group.</td>
</tr>
</tbody>
</table>
To write all the ephone and voice hunt group statistics to a file, use the **hunt-group statistics write-all** command in privileged EXEC mode.

### Syntax Description

```
location URL or filename to which the statistics should be written.
```

### Command Modes

Privileged EXEC (#)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>15.2(2)T</td>
<td>Cisco Unified CME 9.0</td>
<td>This command was introduced to replace the <strong>ephone-hunt statistics write-all</strong> command.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use the **hunt-group statistics write-all** command to write out, in hourly increments, all the ephone and voice hunt group statistics for the past seven days. This command is intended to be used when normal hunt group statistics collection is interrupted, perhaps due to TFTP server failure. If applicable, the TFTP statistics report consists of both ephone and voice hunt statistics.

The commands that are normally used to provide hunt group statistics are **hunt-group report delay hours**, **hunt-group report every hours**, **hunt-group report url**, and **statistics collect**. These commands allow you to specify shorter, more precise reporting periods and file naming conventions.

**Note**

On the day that daylight saving time adjusts the time back by one hour at 2 a.m. each year, the original 1 a.m. to 2 a.m. statistics for that day are lost because they are overwritten by the new 1 a.m. to 2 a.m. statistics.

### Examples

The following example shows how the **hunt-group statistics write-all** command writes a combination of ephone and voice hunt group statistics to a file in TFTP server 223.255.254.254:

```
Router# hunt-group statistics write-all tftp://223.255.254.254/ngm/huntgp/uc500/statall
Writing out all hunt group statistics to tftp://223.255.254.254/ngm/huntgp/uc500/statall
00:08:34 UTC Sat Feb 19 2011,

, EPHONE HUNT GROUP STAT,
01, Sat 00:00 - 01:00, HuntGp, 02, 02, 00001, 00001, 00000, 00007, 00007, 000001, 000001, 0000, 0000, 00000, 00000, 00000,
01, Sat 00:00 - 01:00, Agent, 5012, 00001, 000001, 00001, 00000, 00000, 00000, 000000, 000000, 000000, 000000, 000000, 000000,
01, Sat 00:00 - 01:00, Queue, 00000, 00001, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000,
01, Sat 00:00 - 01:00, Queue, 00000, 00001, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000,
01, Sat 01:00 - 02:00, HuntGp, 00, 00, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000,
01, Sat 02:00 - 03:00, HuntGp, 00, 00, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000, 00000,
Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>hunt-group report delay hours</strong></td>
<td>Delays hunt group statistics collection for a specified number of hours.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>---------</td>
<td>-------------</td>
</tr>
<tr>
<td><strong>hunt-group report every hours</strong></td>
<td>Sets the hourly interval at which Cisco Unified CME B-ACD call statistics are automatically transferred to a file.</td>
</tr>
<tr>
<td><strong>hunt-group report url</strong></td>
<td>Sets filename parameters and the URL path where Cisco Unified CME B-ACD call statistics are to be sent using TFTP.</td>
</tr>
<tr>
<td><strong>show ephone-hunt</strong></td>
<td>Displays ephone hunt group information.</td>
</tr>
<tr>
<td><strong>show ephone-hunt statistics</strong></td>
<td>Displays ephone hunt group statistics.</td>
</tr>
<tr>
<td><strong>statistics collection</strong></td>
<td>Enables the collection of call statistics for an ephone hunt group.</td>
</tr>
<tr>
<td><strong>statistics collection (voice hunt-group)</strong></td>
<td>Enables the collection of call statistics for a voice hunt group.</td>
</tr>
</tbody>
</table>
# huntstop (ephone-dn and ephone-dn-template)

To disable call hunting for directory numbers or channels, use the `huntstop` command in ephone-dn or ephone-dn-template configuration mode. To reset to the default, use the `no` form of this command.

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>huntstop [channel number-of-channels]</code></td>
<td>(Optional) For dual-line and octo-line directory numbers. Prevents incoming calls from hunting to the next channel if the first channel is busy or does not answer.</td>
</tr>
<tr>
<td><code>no huntstop [channel number-of-channels]</code></td>
<td>Supported for octo-line directory numbers only. Number of channels available to accept incoming calls. Remaining channels are reserved for outgoing calls or features such as call transfer, call waiting, and conferencing. Range: 1 to 8. Default: 8.</td>
</tr>
</tbody>
</table>

**Command Default**

Ephone-dn huntstop is enabled. Channel huntstop is disabled for dual-line directory numbers. Channel huntstop is set to 8 for octo-line directory numbers.

**Command Modes**

- Ephone-dn configuration (config-ephone-dn)
- Ephone-dn-template configuration (config-ephone-dn-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was implemented on the Cisco 1750 and Cisco 1751.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>The <strong>channel</strong> keyword was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This <strong>channel</strong> keyword was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was added to ephone-dn-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>The <strong>number-of-channels</strong> argument was added for octo-lines.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command with the <strong>number-of-channels</strong> argument for octo-lines was integrated into Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command without the **channel** keyword to disable call hunting for ephone-dns. An incoming call does not roll over (hunt) to another ephone-dn if the called number is busy or does not answer and a call hunt strategy has been established that includes this ephone-dn. A huntstop prevents hunt-on-busy from redirecting
a call from a busy phone into a dial-peer with a catch-all default destination. Use the no huntstop command to disable huntstop and allow hunting for ephone-dns.

Channel huntstop works in a similar way, but it affects call hunting behavior for the two channels of a dual-line ephone-dn. Use the channel keyword to prevent incoming calls from hunting to the second channel of an ephone-dn if the first channel is busy or does not answer. Incoming calls hunt forward to the next ephone-dn in the hunt sequence instead of to the next channel on the same ephone-dn.

For example, an incoming call might search through the following ephone-dns and channels:

ephone-dn 10 (channel 1) ephone-dn 10 (channel 2)
ephone-dn 11 (channel 1) ephone-dn 12 (channel 1) ephone-dn 12 (channel 2)

If the huntstop channel command is not enabled (the default), a call might ring for 30 seconds on ephone-dn 10 (channel 1) and then after 30 seconds move to ephone-dn 10 (channel 2), which is usually not the desired behavior. It is useful to reserve the second channel of a dual-line ephone-dn for call transfer, call waiting, or conferencing.

The number argument is required for an octo-line directory number when using the channel keyword. This argument limits the number of channels for incoming calls on an octo-line directory number and reserves the other channels for outgoing calls or features such as call transfer or conferencing. The router selects idle channels from the lowest number to the highest. This argument is supported only for an octo-line directory number.

In an ephone-dn template, you can apply separate huntstop channel commands for dual-line directory numbers and octo-line directory numbers.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example shows huntstop is disabled for ephone-dn 1. The huntstop attribute is set to OFF and allows calls to extension 5001 to hunt to another directory number when directory number 1 is busy.

```plaintext
ephone-dn 1  
  number 5001  
  no huntstop
```

The following example shows a typical configuration in which enabling huntstop (default) is required:

```plaintext
ephone-dn 1  
  number 5001

ephone 4  
  button 1:1  
  mac-address 0030.94c3.8724

dial-peer voice 5000 voip  
  destination-pattern 5...  
  session target ipv4:192.168.17.225
```

In the previous example, the huntstop attribute for the dial peer is set to ON by default and prevents calls to extension 5001 from being rerouted to the on-net H.323 dial peer for 5... when extension 5001 is busy (the three periods are used as wildcards).
The following example shows another configuration in which huntstop is not desired and is explicitly disabled. In this example, ephone 4 is configured with two lines, each with the same extension number 5001. This allows the second line to provide call-waiting notification for extension number 5001 when the first line is in use. Setting **no huntstop** on the first line (ephone-dn 1) allows incoming calls to hunt to the second line (ephone-dn 2) when the first line is busy.

Ephone-dn 2 has call forwarding set to extension 6000, which corresponds to a locally attached answering machine connected to a foreign exchange station (FXS) voice port. In this example, the plain old telephone system (POTS) dial peer for extension 6000 also has the dial-peer huntstop attribute explicitly set to prevent further hunting.

```plaintext
table

<table>
<thead>
<tr>
<th>Ephone-DN 1</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number 5001</td>
</tr>
<tr>
<td>No huntstop</td>
</tr>
<tr>
<td>Preference 1</td>
</tr>
<tr>
<td>Call-forward no an 6000</td>
</tr>
</tbody>
</table>

table

<table>
<thead>
<tr>
<th>Ephone-DN 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number 5001</td>
</tr>
<tr>
<td>Preference 2</td>
</tr>
<tr>
<td>Call-forward busy 6000</td>
</tr>
<tr>
<td>Call-forward no an 6000</td>
</tr>
</tbody>
</table>

table

<table>
<thead>
<tr>
<th>Ephone 4</th>
</tr>
</thead>
<tbody>
<tr>
<td>Button 1:1 2:2</td>
</tr>
<tr>
<td>Mac-address 0030.94c3.8724</td>
</tr>
<tr>
<td>Dial-peer Voice 6000 Pots</td>
</tr>
<tr>
<td>Destination-pattern 6000</td>
</tr>
<tr>
<td>Huntstop</td>
</tr>
<tr>
<td>Port 1/0/0</td>
</tr>
<tr>
<td>Description Answering-Machine</td>
</tr>
</tbody>
</table>

table

The following example shows a dual-line configuration in which an ephone-dn template is used to prevent calls from hunting to the second channel of any ephone-dn. The calls hunt through the first channels for each ephone-dn in the order 10, 11, 12.

```plaintext
table

<table>
<thead>
<tr>
<th>Ephone-DN-Template 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Huntstop Channel</td>
</tr>
</tbody>
</table>

table

<table>
<thead>
<tr>
<th>Ephone-DN 10 Dual-Line</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number 1001</td>
</tr>
<tr>
<td>No Huntstop</td>
</tr>
<tr>
<td>Ephone-DN-Template 2</td>
</tr>
</tbody>
</table>

table

<table>
<thead>
<tr>
<th>Ephone-DN 11 Dual-Line</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number 1001</td>
</tr>
<tr>
<td>No Huntstop</td>
</tr>
<tr>
<td>Ephone-DN-Template 2</td>
</tr>
<tr>
<td>Preference 1</td>
</tr>
</tbody>
</table>

table

<table>
<thead>
<tr>
<th>Ephone-DN 12 Dual-Line</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number 1001</td>
</tr>
<tr>
<td>No Huntstop</td>
</tr>
<tr>
<td>Ephone-DN-Template 2</td>
</tr>
<tr>
<td>Preference 2</td>
</tr>
</tbody>
</table>
```
The following example shows a configuration in which incoming calls to octo-line directory number 7 are limited to four, freeing the other four channels for outgoing calls or features such as call transfer or conferencing.

```
ephone-dn  7  octo-line
        number 2001
        name Smith, John
        huntstop channel 4
```

The following example shows an ephone-dn template configuration in which the huntstop is set for both dual-line and octo-line directory numbers.

```
ephone-dn-template  1
        huntstop channel
        huntstop channel 4
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>huntstop (dial-peer)</td>
<td>Disables further dial-peer hunting if a call fails using hunt groups.</td>
</tr>
<tr>
<td>number</td>
<td>Associates a telephone or extension number with a directory number (ephone-dn).</td>
</tr>
</tbody>
</table>
huntstop (voice register dn)

To disable call hunting behavior for a directory number on a SIP phone, use the **huntstop** command in voice register dn configuration mode. To reset to the default, use the **no** form of this command.

```
huntstop [channel number]
no huntstop [channel number]
```

**Syntax Description**

| channel number | (Optional) Number of channels available to accept incoming calls. Remaining channels are reserved for outgoing calls or features such as call transfer, call waiting, and conferencing. Range: 1 to 50. Default: 0 (disabled). |

**Command Default**

Call hunting is enabled for the directory number. Channel huntstop is disabled (0) for the directory number.

**Command Modes**

Voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>The <strong>channel</strong> keyword and <strong>number</strong> argument were added.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command has been integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command disables call hunting behavior for a directory number on a SIP IP phone so that an incoming call does not roll over (hunt) to another directory number if the called directory number is busy or does not answer and if a hunting strategy has been established that includes this directory number. A huntstop allows you to prevent hunt-on-busy from redirecting a call from a busy phone into a dial-peer setup with a catch-all default destination. Use the **no huntstop** command to disable huntstop and allow hunting for directory numbers (default).

The **channel** keyword and **number** argument limits the number of channels for incoming calls to a directory number and reserves the other channels for outgoing calls or features such as call transfer or conferencing. The router selects idle channels from the lowest number to the highest.

**Examples**

The following example shows a typical configuration in which huntstop is required. The **huntstop** command is enabled and prevents calls to extension 5001 from being rerouted to the on-net H.323 dial peer for 5... when extension 5001 is busy (three periods are used as wild cards).

```
voice register dn 1
  number 5001
  huntstop

voice register pool 4
  button 1:1
  mac-address 0030.94c3.8724

  dial-peer voice 5000 voip
```
The following example shows a configuration in which huntstop is not desired (default). In this example, directory number 4 is configured with two lines, each with the same extension number 5001. This is done to allow the second line to provide call-waiting notification for extension number 5001 when the first line is in use. Not enabling huntstop on the first line (directory number 1) allows incoming calls to hunt to the second line (directory number 2) on phone 4 when the directory number 1 line is busy.

Directory number 2 has call forwarding set to extension 6000, which corresponds to a locally attached answering machine connected to a foreign exchange station (FXS) voice port. In this example, the plain old telephone system (POTS) dial peer for extension 6000 has the dial-peer huntstop attribute explicitly set to prevent further hunting.

```
voice register dn 1
  number 5001
  preference 1
  call-forward noan 6000

voice register dn 2
  number 5001
  preference 2
  call-forward busy 6000
  call-forward noan 6000

voice register pool 4
  button 1:1 2:2
  mac-address 0030.94c3.8724

dial-peer voice 6000 pots
  destination-pattern 6000
  huntstop
  port 1/0/0
  description answering-machine
```

The following example shows a configuration in which incoming calls to directory number 23 are blocked if the total number of calls to extension 8123 exceeds 4. This frees the other channels for outgoing calls or features such as call transfer or conferencing.

```
voice register dn 23
  number 8123
  shared-line max-calls 4
  huntstop channel 4
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>huntstop (dial-peer)</strong></td>
<td>Disables all further dial-peer hunting if a call fails on the dial peer.</td>
</tr>
<tr>
<td><strong>shared-line</strong></td>
<td>Creates a directory number to be shared by multiple SIP phones.</td>
</tr>
</tbody>
</table>
Cisco Unified CME Commands: I

- ica, on page 490
- id (voice register pool), on page 491
- import certificate, on page 493
- index (lp cor ip-phone), on page 494
- index (lp cor ip-trunk), on page 496
- intercom (ephone-dn), on page 498
- intercom (voice register dn), on page 501
- internal-call, on page 503
- ip address trusted authenticate, on page 504
- ip address trusted call-block cause, on page 505
- ip address trusted list, on page 506
- ip qos dscp (telephony-service and voice register global), on page 507
- ip source-address (credentials), on page 509
- ip source-address (telephony-service), on page 511
ica

To specify the audio file used for the isolated code announcement, use the ica command in voice MLPP configuration mode. To disable use of this audio file, use the no form of this command.

`ica audio-url`
`no ica`

**Syntax Description**

| audio-url | Location of the announcement audio file in URL format. Valid storage locations are TFTP, FTP, HTTP, and flash memory. |

**Command Default**

No announcement is played.

**Command Modes**

Voice MLPP configuration (config-voice-mlpp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies the G.711 a-law or u-law 8-KHz encoded audio file (.wav or .au format) for the announcement that plays to callers when service or equipment problems prevent completion of their call.

The mlpp indication command must be enabled (default) for a phone to play precedence announcements.

This command is not supported by Cisco IOS help. If you type `?`, Cisco IOS help does not display a list of valid entries.

**Examples**

The following example shows that the audio file played for the isolated code announcement is named ica.au located in flash:

Router(config)# voice mlpp
Router(config-voice-mlpp)# ica flash:ica.au

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>bnea</td>
<td>Specifies the audio file used for the busy station not equipped for preemption announcement.</td>
</tr>
<tr>
<td>upa</td>
<td>Specifies the audio file used for the unauthorized precedence announcement.</td>
</tr>
<tr>
<td>vca</td>
<td>Specifies the audio file used for the vacant code announcement.</td>
</tr>
<tr>
<td>mlpp indication</td>
<td>Enables MLPP indication on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td>mlpp preemption</td>
<td>Enables preemption capability on an SCCP phone or analog FXS port.</td>
</tr>
</tbody>
</table>
id (voice register pool)

To explicitly identify a locally available individual Cisco SIP IP phone, or when running Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST), set of Cisco SIP IP phones, use the `id` command in voice register pool configuration mode. To remove local identification, use the `no` form of this command.

```
id {{ network address mask mask | address mask mask | ip address mask mask address mask || mac address | device-id-name devicename |
no id {{ network address mask mask | address mask mask | ip address mask mask address mask || mac address | device-id-name devicename
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>network address mask mask</td>
<td>This keyword/argument combination is used to accept SIP Register messages for the indicated phone numbers from any IP phone within the specified IPv4 and IPv6 subnets. <code>ipv6 address</code> can only be configured with an IPv6 address or a dual-stack mode.</td>
</tr>
<tr>
<td>ip address mask mask</td>
<td>This keyword/argument combination is used to identify an individual phones IPv4 or IPv6 address. <code>ipv6 address</code> can only be configured with an IPv6 address or a dual-stack mode.</td>
</tr>
<tr>
<td>mac address</td>
<td>The <code>mac address</code> keyword/argument combination is used to identify the MAC address of a particular Cisco IP phone.</td>
</tr>
<tr>
<td>device-id-name devicename</td>
<td>Defines the device name to be used to download the phone’s configuration file.</td>
</tr>
</tbody>
</table>

| Command Default | No SIP IP phone is configured. |
| Command Modes | Voice register pool configuration (config-register-pool) |

<table>
<thead>
<tr>
<th>Command History</th>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was introduced.</td>
<td></td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
<td></td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was added to Cisco CME.</td>
<td></td>
</tr>
<tr>
<td>15.3(3)T</td>
<td>Cisco Unified CME 10.0</td>
<td>This command was modified to add the <code>device-id-name devicename</code> keyword-argument combination.</td>
<td></td>
</tr>
<tr>
<td>Cisco IOS XE Everest 16.6.1</td>
<td>Unified SRST 12.0</td>
<td>This command was modified to add the following keyword-argument combinations for <code>network</code> and <code>ip</code> to include support for IPv6 address: <code>address mask mask</code>.</td>
<td></td>
</tr>
</tbody>
</table>
Configure this command before configuring any other command in voice register pool configuration mode. This command allows explicit identification of an individual Cisco SIP IP phone to support a degree of authentication, which is required to accept registrations, based upon the following:

- Verification of the local Layer 2 MAC address using the router’s Address Resolution Protocol (ARP) cache.
- Verification of the known single static IP address (or DHCP dynamic IP address within a specific subnet) of the Cisco SIP IP phone.

When the `mac address` keyword and argument are used, the IP phone must be in the same subnet as that of the router’s LAN interface, such that the phone’s MAC address is visible in the router’s ARP cache. Once a MAC address is configured for a specific voice register pool, remove the existing MAC address before changing to a new MAC address.

Note
For Cisco Unified SIP SRST, this command also allows explicit identification of locally available set of Cisco SIP IP phones.

Examples
The following is partial sample output from the `show running-config` command. The `id` command identifies the MAC address of a particular Cisco IP phone. The output shows that voice register pool 1 has been set up to accept SIP Register messages from a specific IP phone through the use of the `id` command.

```
voice register pool 1
id mac 0030.94C2.A22A
preference 5
cor incoming call 91 1 91011
translate-outgoing called 1
proxy 10.2.161.187 preference 1 monitor probe icmp-ping
alias 1 94... to 91011 preference 8
voice-class codec 1
```

The following is sample output from the `show running-config` command after configuring IPv6 address on Cisco Unified SRST router.

```
voice register pool 1
id network 2001:420:54FF:13::312:0/117
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>mode (voice register global)</code></td>
<td>Enables the mode for provisioning SIP phones in a Cisco Unified CallManager Express (Cisco Unified CME) system.</td>
</tr>
</tbody>
</table>
import certificate

To import a trusted certificate in PEM format from flash memory to the CTL file of an IP phone, use the import certificate command in ctl-client configuration mode. To return to the default, use the no form of this command.

import certificate  tag description flash:cert_name
no import certificate

**Syntax Description**

<table>
<thead>
<tr>
<th>Tag</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>tag</td>
<td>Identifier for the trusted certificate.</td>
</tr>
<tr>
<td>description</td>
<td>Descriptive name of the trusted certificate.</td>
</tr>
<tr>
<td>flash:cert_name</td>
<td>Specifies the filename of the trusted certificate stored in flash memory.</td>
</tr>
</tbody>
</table>

**Command Default**

None

**Command Modes**

CTL-client configuration (config-ctl-client)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(1)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

A CTLFile.tlv file should appear in the flash location after using the regenerate command in ctl-client configuration mode. If the file is missing, use the debug ctl-client command, followed by the regenerate command.

**Examples**

The following is an example of how the import certificate command is used to import the WebServer certificate with filename web_cer.cer from flash memory:

```plaintext
Router(config)# ctl-client
Router(config-ctl-client)# sast1 trustpoint primary-cme
Router(config-ctl-client)# sast2 trustpoint sast-secondary
Router(config-ctl-client)# import certificate 1 WebServer flash:web_cert.cer
Router(config-ctl-client)# regenerate
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ctl-client</td>
<td>Enters CTL-client configuration mode to set parameters for the CTL client.</td>
</tr>
</tbody>
</table>
index (lpcor ip-phone)

To add a logical partitioning class of restriction (LPCOR) group to the IP-phone subnet table, use the index command in LPCOR ip-phone subnet configuration mode. To remove a resource, use the no form of this command.

index index-number lpcor-group {ipv4-address network-mask [vrf vrf-name]|dhcp-pool pool-name}
no index index-number

Syntax Description

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>index-number</td>
<td>Number of the LPCOR subnet index entry. Range: 1 to 50.</td>
</tr>
<tr>
<td>lpcor-group</td>
<td>Name of a LPCOR resource-group policy.</td>
</tr>
<tr>
<td>ipv4-address</td>
<td>IPv4 address of the LPCOR policy.</td>
</tr>
<tr>
<td>network-mask</td>
<td>Subnet mask for the associated IPv4 address.</td>
</tr>
<tr>
<td>vrf vrf-name</td>
<td>(Optional) Dynamic Host Configuration Protocol (DHCP) server uses the VPN routing and forwarding (VRF) table that is associated with the access point name (APN).</td>
</tr>
<tr>
<td>dhcp-pool pool-name</td>
<td>User-defined name of the DHCP pool. The pool name can be a symbolic string (such as Sales) or an integer (such as 0).</td>
</tr>
</tbody>
</table>

Command Default

No index entry is configured.

Command Modes

LPCOR ip-phone subnet configuration (cfg-lpcor-ipphone-subnet)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

This command is used for mobility-type phones only, which can include Extension Mobility phones, teleworker remote phones, and Cisco IP Communicator softphones.

Two IP-phone subnet tables, containing up to 50 index entries, can be defined on each Cisco Unified CME router. One table is for incoming calls and the other table is for outgoing calls.

A LPCOR policy is dynamically associated with calls to and from a mobility-type phone by matching its current IP address or DHCP pool in the IP-phone subnet table. If the LPCOR policy cannot be provisioned from the IP-phone subnet table, the default LPCOR policy for mobility-type phones is used.

Entries in the IP-phone subnet tables are indexed in ascending order. The lookup of entries is in sequential ascending order. After Cisco Unified CME finds a matching entry, the corresponding LPCOR policy is associated with the call. Even if there are other entries that are a better match, only the first match is used.

For instance, in the example below, if a call originates from an IP phone with IP address 10.1.10.3, LPCOR policy local_g4 is associated with the incoming call instead of LPCOR policy local_g5 even though local_g5 is a better match.
Examples

The following example shows an IP-phone subnet table for incoming calls that has four entries:

```
voice lpcor ip-phone subnet incoming
index 1 local_g4 10.1.10.0 255.255.255.0
index 2 remote_g4 171.19.0.0 255.255.0.0
index 3 local_g5 10.1.10.2 255.255.255.255
index 4 local_g5 10.1.10.3 255.255.255.255
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>lpcor type</td>
<td>Specifies the LPCOR type for an IP phone.</td>
</tr>
<tr>
<td>voice lpcor ip-phone mobility</td>
<td>Sets the default LPCOR policy for mobility-type phones.</td>
</tr>
<tr>
<td>voice lpcor policy</td>
<td>Creates a LPCOR policy for a resource group.</td>
</tr>
</tbody>
</table>
To add a logical partitioning class of restriction (LPCOR) resource group to the IP trunk subnet table, use the `index` command in LPCOR IP-trunk subnet configuration mode. To remove a resource, use the `no` form of this command.

```
index number lpcor-group {ipv4-address network-mask|hostname host-name}
no index number
```

**Syntax Description**

- `number`: Number of the LPCOR subnet index entry. Range: 1 to 50.
- `lpcor-group`: Name of a LPCOR resource-group policy.
- `ipv4-address`: IPv4 address of the LPCOR policy.
- `network-mask`: Subnet mask of the associated IPv4 address.
- `hostname host-name`: User-defined IP host name.

**Command Default**
No index entry is configured.

**Command Modes**
LPCOR IP-trunk subnet configuration (cfg-lpcor-iptrunk-subnet)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

One IP-trunk subnet table, containing up to 50 index entries, can be defined on each Cisco Unified CME router for incoming VoIP trunk calls (H.323 or SIP).

An incoming VoIP trunk call is associated with a LPCOR policy by matching the remote IP address to an entry in the incoming IP-trunk subnet table. If that is not successful, the LPCOR policy in voice service configuration mode is applied.

Entries in the IP-trunk subnet table are indexed in ascending order. The lookup of entries is in sequential ascending order. After Cisco Unified CME finds a matching entry, it associates the corresponding LPCOR policy with the call. Even if there are other entries that are a better match, only the first match is used.

In the following example, an incoming VoIP call with a remote IP address of 172.19.22.25 is associated with sip_group1 even though voip_group2 is a better match.

**Examples**

The following example shows an IP-trunk subnet table with six index entries:

```
voice lpcor ip-trunk subnet incoming
index 1 h323_group1 172.19.33.0 255.255.255.0
index 2 sip_group1 172.19.22.0 255.255.255.0
index 3 voip_group2 172.19.33.25 255.255.255.255
index 4 voip_group3 172.19.22.26 255.255.255.255
```
Related Commands  |  Command  |  Description
--- | --- | ---
 | IP COR Incoming | Associates an incoming call with a LPCOR resource-group policy. |
 | voice IP COR Policy | Creates a LPCOR policy for a resource group. |
To create an intercom by programming a pair of extensions (ephone-dns) to automatically call and answer each other, use the `intercom` command in ephone-dn configuration mode. To remove an intercom, use the `no` form of this command.

```
intercom  extension-number  [[barge-in|no-mute] |no-auto-answer | no-mute] |[label|label]] |[paging number|ptt]
```

**Syntax Description**

<table>
<thead>
<tr>
<th><code>extension-number</code></th>
<th>Extension or telephone number to which calls are placed.</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>barge-in</code></td>
<td>(Optional) Allows inbound intercom calls to force an existing call into the call-hold state and the intercom call to be answered immediately.</td>
</tr>
<tr>
<td><code>label label</code></td>
<td>(Optional) Defines an alphanumeric label for the intercom, of up to 30 characters.</td>
</tr>
<tr>
<td><code>no-auto-answer</code></td>
<td>(Optional) Disables the intercom auto-answer feature.</td>
</tr>
<tr>
<td><code>no-mute</code></td>
<td>(Optional) Allows an intercom call to be answered without deactivating a speaker’s mute key.</td>
</tr>
<tr>
<td><code>paging number ptt</code></td>
<td>(Optional) Allows to set a paging number for push-to-talk (PTT) feature.</td>
</tr>
</tbody>
</table>

**Command Default**

Intercom functionality is disabled.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>The <code>no-mute</code> keyword was added.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>The paging number and ptt keywords and argument was added.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used to dedicate a pair of Cisco ephone-dns for use as a “press to talk” two-way intercom between Cisco IP phones. Intercom lines cannot be used in shared-line configurations. If an ephone-dn is configured for intercom operation, it must be associated with one Cisco IP phone only. The intercom attribute causes an IP extension (ephone-dn) to operate in autodial fashion for outbound calls and autoanswer-with-mute for inbound calls.

The `barge-in` keyword allows inbound intercom calls to force an existing call on the called phone into the call-hold state to allow the intercom call to be answered immediately. The `no-auto-answer` keyword creates for the IP phone line a connection that resembles a private line, automatic ringdown (PLAR). The `label` keyword defines a text label for the intercom.
Following this command, the intercom ephone-dns are assigned to ephones using the **button** command. Following the **button** command, the **restart** command must be used to initiate a quick reboot of the phones to which this intercom is assigned.

The default **intercom** command behavior is speakers are set to mute automatically when phones receive intercom calls. For example, if phone user 1 places an intercom call and connects to phone user 2, user 2 will hear user 1, but user 1 will not hear user 2. To be heard, user 2 must first disable the speaker’s mute function. The benefit is people who receive intercom calls can use the mute button to control when they will be heard initially.

The **no-mute** keyword deactivates the speaker mute function when IP phones receive intercom calls. For example, if phone user 1 makes an intercom call to phone user 2, both users will hear each other upon connection. The benefit is that people who receive intercom calls do not have to disable their speaker’s mute function to be heard, **but** their conversations and nearby background sounds will be heard the moment an intercom call to them is connected—regardless of whether they are ready to take a call or not.

The intercom command allows you to add a paging number to behave as a push-to-talk (ptt) feature. More information on the push-to-talk feature is available at this link: [http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmelabel.html#wpmkr1048855](http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmelabel.html#wpmkr1048855)

### Examples

The following example sets the intercom on Cisco IP phone directory number 1:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) number A5001
Router(config-ephone-dn) name "intercom"
Router(config-ephone-dn) intercom A5002 barge-in
```

The following example shows intercom configuration between two Cisco IP phones:

```
ephone-dn 18
   number A5001
   name "intercom"
   intercom A5002 barge-in

ephone-dn 19
   number A5002
   name "intercom"
   intercom A5001 barge-in
```

In the example, ephone-dn 18 and ephone-dn 19 are set as an intercom pair. Ephone-dn 18 is associated with button 3 of Cisco IP phone (ephone) 4, and ephone-dn 19 is associated with button number 3 of Cisco IP phone (ephone) 5. Button 3 on Cisco IP phone 4 and button 3 on Cisco IP phone 5 are set as a pair to provide intercom service to each other.

The intercom feature acts as a combination speed-dial PLAR and autoanswer with mute. If the **barge-in** keyword is set on the ephone-dn that receives the intercom call, the existing call is forced into the hold state, and the intercom call is accepted. If the phone user has the handset off hook (that is, not in speakerphone mode), the user hears a warming beep, and the intercom call is immediately connected with two-way audio. If the phone user is using speakerphone mode, the intercom connects with the microphone mute activated.
Any caller can dial into an intercom extension, and a call to an intercom extension that is originated by a nonintercom caller triggers an automatic answer exactly like a legitimate intercom call. To prevent nonintercom originators from manually dialing an intercom destination, you can use alphabetic characters when you assign numbers to intercom extensions using the `number` command. These characters cannot be dialed from a normal phone but can be dialed by preprogrammed intercom extensions whose calls are made by the router.

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>button</code></td>
<td>Associates ephone-dns with individual buttons on Cisco IP phones and specifies ring behavior per button.</td>
</tr>
<tr>
<td><code>number</code></td>
<td>Associates a telephone or extension number with an extension (ephone-dn).</td>
</tr>
<tr>
<td><code>restart (ephone)</code></td>
<td>Performs a fast reboot of a single phone associated with a Cisco CME router.</td>
</tr>
<tr>
<td><code>restart (telephony-service)</code></td>
<td>Performs a fast reboot of one or all phones associated with a Cisco CME router.</td>
</tr>
</tbody>
</table>
intercom (voice register dn)

To enable the intercom call option on a Cisco Unified SIP IP phone, use the `intercom` command in voice register dn configuration mode. To prevent a Cisco Unified SIP IP phone from making an intercom call, use the `no` form of this command.

```
intercom [speed-dial digit-string] [label label-text]
no intercom [speed-dial digit-string] [label label-text]
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>speed-dial</td>
<td>(Optional) Enables the intercom line user to place a call to a pre-configured destination. If the speed-dial is not configured, it simply initiates a new call on the intercom line and waits for the user to dial the destination number.</td>
</tr>
<tr>
<td>digit-string</td>
<td>Digits to be dialed when the speed-dial button is pressed on a Cisco Unified SIP IP phone. For Cisco Unified SIP IP phones, if the first character of the string is a plus sign (+), the speed-dial number is locked and cannot be changed at the phone. If the only character in the string is a pound sign (#), the user-programmable speed-dial button with no speed-dial number attached is defined.</td>
</tr>
<tr>
<td>label</td>
<td>(Optional) String that contains identifying text to be displayed next to the speed-dial button. Enclose the string in quotation marks if the string contains a space.</td>
</tr>
</tbody>
</table>

### Command Default

The Cisco Unified SIP IP phone cannot make or receive an intercom call.

### Command Modes

Voice register dn configuration (config-register-dn)

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(1)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

The intercom line cannot be the primary line of a Cisco Unified SIP IP phone and cannot be shared among Cisco Unified SIP IP phones.

When the intercom speed-dial option is not configured, the intercom line waits for the user to dial the destination number.

### Examples

The following example shows SIP intercom configured on extension 1001:

```
Router(config)# voice register dn 1
Router(config-register-dn) number 1001
Router(config-register-dn) intercom [speed-dial 1002] [label intercom1001]
Router(config)# voice register pool 1
Router(config-register-pool) id mac 001D.452D.580C
Router(config-register-pool) type 7962
Router(config-register-pool) number 1 dn 2
Router(config-register-pool) number 2 dn 1
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register dn</td>
<td>Enters voice register dn configuration mode.</td>
</tr>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode.</td>
</tr>
</tbody>
</table>
To assign an MOH group for calls from an internal directory number, use the `internal-call` command in telephony-service configuration mode. To disable the internal-call command, use the `no` form of this command.

```
internal-call moh-group-tag
no internal-call
```

### Syntax Description

- **moh-group-tag**: Specifies a MOH-group number to be used for calls from an internal directory number. Range is from 0 to 5, where 0 represents MOH configuration in telephony-service configuration mode.

### Command Default

No internal-call is configured.

### Command Modes

Telephony-service configuration (config-telephony-service)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Before using this command make sure you have MOH-groups configured under voice-moh-group configuration mode. This command allows you to assign a MOH-group for all calls from an internal directory number. MOH group tag identifies the unique number assigned to a MOH group. Range for MOH group tag is from 0 to 5, where 0 represents MOH configuration in telephony service.

### Examples

The following example shows MOH-group 4 assigned for an internal directory number:

```
telephony-service
  internal-call moh-group 4
  em logout 0:0 0:0 0:0
  max-ephones 58
  max-dn 192
  ip source-address 15.1.0.161 port 2000
  max-conferences 8 gain -6
  moh music-on-hold.au
  multicast moh 239.1.1.1 port 2000
  transfer-system full-consult
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice-moh-group</td>
<td>Enter voice-moh-group configuration mode.</td>
</tr>
<tr>
<td>moh filename</td>
<td>Enables music on hold from a flash audio feed</td>
</tr>
<tr>
<td>multicast moh</td>
<td>Enables multicast of the music-on-hold audio stream.</td>
</tr>
<tr>
<td>extension-range</td>
<td>Specifies the extension range for a clients calling a voice-moh-group.</td>
</tr>
</tbody>
</table>
ip address trusted authenticate

To enable ip address trusted authentication for incoming VoIP (H.323/SIP) calls, use the `ip address trusted authenticate` command in voice service voip mode. To disable ip address trusted authentication, use the no form of this command.

```
ip address trusted authenticate
no ip address trusted authenticate
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
IP address trusted list authenticate is enabled.

**Command Modes**
Voice Service Voip

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to enable the ip address trusted authentication for incoming H.323 or SIP trunk calls for toll fraud prevention on Cisco Unified CME.

**Examples**
The following is a sample output from this command displaying IP address trusted authentication enabled for incoming calls:

```
IP Address Trusted Authentication
Administration State: UP
Operation State: UP
IP Address Trusted Call Block Cause: call-reject (21)
VoIP Dial-peer IPv4 Session Targets:
Peer Tag Oper State Session Target
-------- ---------- --------------
11 DOWN ipv4:1.3.45.1
1 UP ipv4:1.3.45.1
IP Address Trusted List:
ipv4 172.19.245.1
ipv4 172.19.247.1
ipv4 172.19.243.1
ipv4 171.19.245.1
ipv4 171.19.10.1
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ip address trusted list</td>
<td>Allows to manually add additional valid IP addresses.</td>
</tr>
<tr>
<td>ip address trusted call- block cause</td>
<td>Allows to issues a cause-code when the incoming call is rejected by the IP address trusted authentication.</td>
</tr>
</tbody>
</table>
ip address trusted call-block cause

To issues a cause-code when the incoming call is rejected by the IP address trusted authentication, use the `ip address trusted call-block cause` command in voice service voip mode. To stop the IP address trusted authentication process from sending a call-block cause, use the `no` form of this command.

```
no ip address trusted call-block cause code-id
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>code-id</code></td>
<td>Q.850 call-disconnect cause code. Range is from 1 to 127.</td>
</tr>
</tbody>
</table>

**Command Default**

A call-reject (21) cause-code is issued to disconnect the incoming VoIP calls.

**Command Modes**

Voice Service voip.

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to issue a cause-code when the incoming call is rejected by the IP address trusted authentication. You can issue a specific call-block cause code using any one of the Q.850 call reject cause codes.

**Examples**

The following is a sample output from this command displaying the default call block cause code:

```
Router #show ip address trusted list
IP Address Trusted Authentication
  Administration State: UP
  Operation State:    UP
  IP Address Trusted Call Block Cause: call-reject (21)
  VoIP Dial-peer IPv4 Session Targets:
    Peer Tag  Oper State  Session Target
    --------  ----------  ------------
    11        DOWN       ipv4:1.3.45.1
    1         UP         ipv4:1.3.45.1
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ip address trusted list</code></td>
<td>Allows to manually add additional valid IP addresses.</td>
</tr>
<tr>
<td><code>ip address trusted authenticate</code></td>
<td>Enables IP address trusted authentication for incoming VoIP calls.</td>
</tr>
</tbody>
</table>
ip address trusted list

To manually add multiple IP addresses for incoming VoIP (H.323/SIP) calls, use the `ip address trusted list` command in voice service voip mode. To turn off the list, use the no form of this command.

```
ip address trusted list ipv4 ipv4 address network mask
no ip address trusted list ipv4 ipv4 address network mask
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ipv4-address</td>
<td>IPv4 address of the incoming H.323 or SIP calls.</td>
</tr>
<tr>
<td>network mask</td>
<td>Subnet IP address.</td>
</tr>
</tbody>
</table>

### Command Default

IP address trusted list is disabled.

### Command Modes

Voice Service Voip.

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use this command to manually add unique and multiple IP addresses to a list of trusted IP addresses. You can add up to 100 IPv4 addresses in the ip address trusted list. No duplicate IP addresses are allowed.

### Examples

The following is a sample output from this command displaying a list of trusted IP addresses:

```
Router #show ip address trusted list
IP Address Trusted Authentication
 Administration State: UP
 Operation State: UP
 IP Address Trusted Call Block Cause: call-reject (21)
 VoIP Dial-peer IPv4 Session Targets:
 Peer Tag Oper State Session Target
--------- -------------- -------------
11 DOWN ipv4:1.3.45.1
1 UP ipv4:1.3.45.1
 IP Address Trusted List:
 ipv4 172.19.245.1
 ipv4 172.19.247.1
 ipv4 172.19.243.1
 ipv4 171.19.245.1
 ipv4 171.19.10.1
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP address trusted authenticate</td>
<td>Enables IP address trusted authentication for incoming VoIP calls.</td>
</tr>
<tr>
<td>IP address trusted code-block cause</td>
<td>Allows to issues a cause-code when the incoming call is rejected by the IP address trusted authentication.</td>
</tr>
</tbody>
</table>
ip qos dscp (telephony-service and voice register global)

To set the Differentiated Services Code Point (DSCP) for marking the quality of service (QoS) requirements for each packet, use the `ip qos dscp` command in telephony-service or voice register global configuration mode. To reset to the default value, use the `no` form of this command.

```
ip qos dscp {number|default|ef} {media|service|signaling|video}
no ip qos dscp {number|default|ef} {media|service|signaling|video}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>number</th>
<th>DSCP value. Range: 0 to 63.</th>
</tr>
</thead>
<tbody>
<tr>
<td>af</td>
<td>Sets DSCP to assured forwarding bit pattern.</td>
</tr>
<tr>
<td></td>
<td>• af11—bit pattern 001010</td>
</tr>
<tr>
<td></td>
<td>• af12—bit pattern 001100</td>
</tr>
<tr>
<td></td>
<td>• af13—bit pattern 001110</td>
</tr>
<tr>
<td></td>
<td>• af21—bit pattern 010010</td>
</tr>
<tr>
<td></td>
<td>• af22—bit pattern 010100</td>
</tr>
<tr>
<td></td>
<td>• af23—bit pattern 010110</td>
</tr>
<tr>
<td></td>
<td>• af31—bit pattern 011010</td>
</tr>
<tr>
<td></td>
<td>• af32—bit pattern 011100</td>
</tr>
<tr>
<td></td>
<td>• af33—bit pattern 011110</td>
</tr>
<tr>
<td></td>
<td>• af41—bit pattern 100010</td>
</tr>
<tr>
<td></td>
<td>• af42—bit pattern 100100</td>
</tr>
<tr>
<td></td>
<td>• af43—bit pattern 100110</td>
</tr>
<tr>
<td>cs</td>
<td>Sets DSCP to class-selector codepoint.</td>
</tr>
<tr>
<td></td>
<td>• cs1—codepoint 1 (precedence 1)</td>
</tr>
<tr>
<td></td>
<td>• cs2—codepoint 2 (precedence 2)</td>
</tr>
<tr>
<td></td>
<td>• cs3—codepoint 3 (precedence 3)</td>
</tr>
<tr>
<td></td>
<td>• cs4—codepoint 4 (precedence 4)</td>
</tr>
<tr>
<td></td>
<td>• cs5—codepoint 5 (precedence 5)</td>
</tr>
<tr>
<td></td>
<td>• cs6—codepoint 6 (precedence 6)</td>
</tr>
<tr>
<td></td>
<td>• cs7—codepoint 7 (precedence 7)</td>
</tr>
<tr>
<td>default</td>
<td>Sets DSCP to default bit pattern of 000000.</td>
</tr>
<tr>
<td>ef</td>
<td>Sets DSCP to expedited forwarding bit pattern 101110.</td>
</tr>
<tr>
<td>media</td>
<td>Applies DSCP to media payload packets.</td>
</tr>
<tr>
<td>service</td>
<td>Applies DSCP to phone service including HTTP traffic.</td>
</tr>
<tr>
<td>signaling</td>
<td>Applies DSCP to signaling packets.</td>
</tr>
<tr>
<td>video</td>
<td>Applies DSCP to video stream.</td>
</tr>
</tbody>
</table>

**Command Default**

DSCP for media is `ef`. DSCP for service is `0`. DSCP for signaling is `cs3`. DSCP for video is `af41`. 
Command Modes

- Telephony-service configuration (config-telephony)
- Voice register global configuration (config-register-global)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

This command allows you to set different priority levels for different types of network traffic sent by the Cisco Unified CME router. Differentiated Services is a method of prioritizing specific network traffic based on the QoS specified by each packet. You can set different DSCP values, for example, for video and audio streams.

Cisco Unified CME downloads the configured DSCP value to the phones in their configuration files and all control messages and RTP streams are marked with the preferred DSCP value. Use this command in telephony-service mode to set the DSCP for SCCP phones. Use the command in voice register global mode to set the value for SIP phones.

If the DSCP is configured for the gateway interface using the `service-policy` command or in the dial peer using the `ip qos dscp` command, the value set with those commands takes precedence over the DSCP value configured with this command.

Examples

The following examples show the configuration of DSCP for different types of packets.

```
voice register global
  mode cme
  ip qos dscp af11 media
  ip qos dscp cs2 signal
  ip qos dscp af43 video
  ip qos dscp 25 service

telephony-service
  load 7960-7940 P00308000500
  max-ephones 100
  max-dn 240
  ip source-address 10.7.0.1 port 2000
  ip qos dscp af11 media
  ip qos dscp cs2 signal
  ip qos dscp af43 video
  ip qos dscp 25 service
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ip qos dscp</code></td>
<td>Sets the DSCP for QoS in a dial peer.</td>
</tr>
<tr>
<td><code>service-policy</code></td>
<td>Assigns a policy map to an interface that will be used as the service policy for the interface.</td>
</tr>
</tbody>
</table>
ip source-address (credentials)

To enable the Cisco Unified CME or Cisco Unified SRST router to receive credential service messages through the specified IP address and port, use the `ip source-address` command in credentials configuration mode. To disable the router from receiving messages, use the `no` form of this command.

```
ip source-address  ip-address  [port  [port]]
no ip source-address
```

**Syntax Description**

<table>
<thead>
<tr>
<th><code>ip-address</code></th>
<th>Router IP address, typically one of the addresses of the Ethernet port of the local router.</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>port port</code></td>
<td>(Optional) TCP port for credentials service communication. Range is from 2000 to 9999. Cisco Unified CME default is 2444. SRST default is 2445.</td>
</tr>
</tbody>
</table>

**Command Default**

Default port number in Cisco Unified CME is 2444. Default port number in Cisco Unified SRST is 2445.

**Command Modes**

Credentials configuration (config-credentials)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco SRST 3.3</td>
<td>This command was introduced for Cisco SRST.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced for Cisco Unified CME.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command for Cisco Unified CME was integrated in Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

**Cisco Unified CME**

This command is used with Cisco Unified CME phone authentication to identify a Cisco Unified CME router on which a CTL provider is being configured.

**Cisco Unified SRST**

The `ip source-address` command is a mandatory command to enable secure SRST. If the port number is not provided, the default value (2445) is used. The IP address is usually the IP address of the secure SRST router.

**Examples**

**Cisco Unified CME**

The following example creates a CTL provider on a Cisco Unified CME router that is not running the CTL client.

```
Router(config)# credentials
Router(config-credentials)# ip source-address 172.19.245.1 port 2444
Router(config-credentials)# trustpoint ctlpv
Router(config-credentials)# ctl-service admin user4 secret 0 c89L8o
```
Cisco Unified SRST

The following example enters credentials configuration mode and sets the IP source address and port:

```
Router(config)# credentials
Router(config-credentials)# ip source-address 10.6.21.4 port 2445
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ctl-service admin</code></td>
<td>Specifies a user name and password to authenticate the CTL client during the CTL protocol.</td>
</tr>
<tr>
<td><code>debug credentials</code></td>
<td>Sets debugging on the credentials service that runs between a Cisco Unified CME CTL provider and the CTL client or between an SRST router and Cisco Unified CallManager.</td>
</tr>
<tr>
<td><code>show credentials</code></td>
<td>Displays the credentials settings on a Cisco Unified CME or SRST router.</td>
</tr>
<tr>
<td><code>trustpoint (credentials)</code></td>
<td>Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with an SRST router certificate.</td>
</tr>
</tbody>
</table>
ip source-address (telephony-service)

To identify the IP address and port through which IP phones communicate with a Cisco Unified CME router, use the `ip source-address` command in telephony-service or group configuration mode. To disable the router from receiving messages from Cisco Unified IP phones, use the `no` form of this command.

```
ip { ipv4_address| ipv6_address} [port port] [secondary {ipv4 address|ipv6 address}] [rehome:seconds] [any-match|strict-match]
oip source-address
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ipv4_address</code></td>
<td>IPv4 address of the router, typically one of the addresses of the Ethernet port of the router.</td>
</tr>
<tr>
<td><code>ipv6_address</code></td>
<td>In Cisco Unified CME 8.0 and later versions: IPv6 address of the router, typically one of the addresses of the Ethernet port of the router.</td>
</tr>
<tr>
<td><code>port port</code></td>
<td>(Optional) TCP/IP port number to use for Skinny Client Control Protocol (SCCP). Default is 2000. For IPv4 only: Range is from 2000 to 9999. <strong>Note</strong> For IPv6, do not configure the port number to change from the default value (2000).</td>
</tr>
<tr>
<td><code>secondary</code></td>
<td>(Optional) Second Cisco Unified CME router with which phones can register if the primary Cisco Unified CME router fails. <strong>Note</strong> For dual-stack (IPv4 and IPv6) mode: Only an IPv4 address can be configured for a secondary router.</td>
</tr>
<tr>
<td><code>rehome seconds</code></td>
<td>(Optional) Used only by Cisco Unified IP phones that have registered with a Cisco Unified Survivable Remote Site Telephony (SRST) router. This keyword defines a delay that is used by phones to verify the stability of their primary SCCP controller (Cisco Unified Communications Manager or Cisco Unified CME) before the phones reregister with it. This parameter is ignored by phones unless they are registered to a secondary Cisco Unified SRST router. The range is from 0 to 65535 seconds. The default is 120 seconds. The use of this parameter is a phone behavior and is subject to change, based on the phone type and phone firmware version.</td>
</tr>
<tr>
<td><code>strict-match</code></td>
<td>(Optional) Requires strict IP address checking for registration.</td>
</tr>
</tbody>
</table>

**Command Default**

The IP address for communicating with phones is not defined.

**Command Modes**

Telephony-service configuration (config-telephony)  
Group configuration (conf-tele-group)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>
This command enables a router to receive messages from Cisco Unified IP phones through the specified IP address and port.

The Cisco Unified CME router cannot communicate with Cisco Unified CME phones if the IP address of the port to which they are attached is not configured. In Cisco Unified CME 8.0 and later versions, the Cisco Unified CME router can receive messages from IPv6-enabled or IPv4-enabled IP phones or from phones in dual-stack (both IPv6 and IPv4) mode.

- In Cisco Unified CME 8.0 and later versions: If the IP phones connected to Cisco Unified CME were configured for dual-stack mode by using `dual-stack` keyword with the `protocol mode` command, configure this command with the IPv6 address.
- In Cisco Unified CME 8.0 and later versions: If the IP phones to be connected to the port to be configured are IPv4-enabled only or IPv6-enabled only, configure this command with the corresponding IPv4 or IPv6 address.

For IPv6: Do not configure the `port port` keyword argument combination in this command to change the value from the default (2000). If you change the port number, IPv6 CEF packet switching engine will not be able to handle the IPv6 SCCP phones and various packet handling problems may occur when more than a dozen (approximately) calls in IPv6 are going on.

Use the `strict-match` keyword to instruct the router to reject IP phone registration attempts if the IP server address used by the phone does not match the source address.

Prior to Cisco IOS Telephony Services (Cisco ITS) V2.1, this command helped the router to autogenerate the SEPDEFAULT.cnf file, which was stored in the flash memory of the router. The SEPDEFAULT.cnf file contains the IP address of one of the Ethernet ports of the router to which the phone should register.

In ITS V2.1 and in Cisco CME 3.0 and later versions, the configuration files were moved to system:/its/. The file named Flash:SEPDEFAULT.cnf that was used with previous Cisco ITS versions is obsolete, but is retained as system:/its/SEPDEFAULT.cnf to support upgrades from older phone firmware.

For systems using Cisco ITS V2.1 or later versions, the IP phones receive their initial configuration information and phone firmware from the TFTP server associated with the router. In most cases, the phones obtain the IP address of their TFTP server using the `option 150` command and Dynamic Host Configuration Protocol (DHCP). For Cisco ITS or Cisco CME operation, the TFTP server address obtained by the Cisco Unified IP phones should point to the router IP address. The Cisco IP phones attempt to transfer a configuration file called XmlDefault.cnf.xml. This file is automatically generated by the router through the `ip source-address` command and is placed in router memory. The XmlDefault.cnf.xml file contains the IP address that the phones

---

**Modification**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The secondary <code>ip-address</code> and <code>rehome</code> <code>seconds</code> keyword-argument pairs were added.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>The secondary <code>ip-address</code> and <code>rehome</code> <code>seconds</code> keyword-argument pairs were added.</td>
</tr>
<tr>
<td>12.4(22)T</td>
<td>Cisco Unified CME 7.0(1)</td>
<td>This command was added to VRF group mode.</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was modified. Support for IPv6 was added and the <code>ipv4-address</code> and <code>ipv6-address</code> arguments replaced the generic <code>ip-address</code> argument.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

---

**Usage Guidelines**

This command enables a router to receive messages from Cisco Unified IP phones through the specified IP address and port.

The Cisco Unified CME router cannot communicate with Cisco Unified CME phones if the IP address of the port to which they are attached is not configured. In Cisco Unified CME 8.0 and later versions, the Cisco Unified CME router can receive messages from IPv6-enabled or IPv4-enabled IP phones or from phones in dual-stack (both IPv6 and IPv4) mode.

- In Cisco Unified CME 8.0 and later versions: If the IP phones connected to Cisco Unified CME were configured for dual-stack mode by using `dual-stack` keyword with the `protocol mode` command, configure this command with the IPv6 address.
- In Cisco Unified CME 8.0 and later versions: If the IP phones to be connected to the port to be configured are IPv4-enabled only or IPv6-enabled only, configure this command with the corresponding IPv4 or IPv6 address.

For IPv6: Do not configure the `port port` keyword argument combination in this command to change the value from the default (2000). If you change the port number, IPv6 CEF packet switching engine will not be able to handle the IPv6 SCCP phones and various packet handling problems may occur when more than a dozen (approximately) calls in IPv6 are going on.

Use the `strict-match` keyword to instruct the router to reject IP phone registration attempts if the IP server address used by the phone does not match the source address.

Prior to Cisco IOS Telephony Services (Cisco ITS) V2.1, this command helped the router to autogenerate the SEPDEFAULT.cnf file, which was stored in the flash memory of the router. The SEPDEFAULT.cnf file contains the IP address of one of the Ethernet ports of the router to which the phone should register.

In ITS V2.1 and in Cisco CME 3.0 and later versions, the configuration files were moved to system:/its/. The file named Flash:SEPDEFAULT.cnf that was used with previous Cisco ITS versions is obsolete, but is retained as system:/its/SEPDEFAULT.cnf to support upgrades from older phone firmware.

For systems using Cisco ITS V2.1 or later versions, the IP phones receive their initial configuration information and phone firmware from the TFTP server associated with the router. In most cases, the phones obtain the IP address of their TFTP server using the `option 150` command and Dynamic Host Configuration Protocol (DHCP). For Cisco ITS or Cisco CME operation, the TFTP server address obtained by the Cisco Unified IP phones should point to the router IP address. The Cisco IP phones attempt to transfer a configuration file called XmlDefault.cnf.xml. This file is automatically generated by the router through the `ip source-address` command and is placed in router memory. The XmlDefault.cnf.xml file contains the IP address that the phones...
use to register for service, using the SCCP. This IP address should correspond to a valid Cisco CME router IP address (and may be the same as the router TFTP server address).

Similarly, when an analog telephone adapter (ATA) such as the ATA-186 is attached to the Cisco Unified CME router, the ATA receives very basic configuration information and firmware from the TFTP server XmlDefault.cnf.xml file. The XmlDefault.cnf.xml file is automatically generated by the Cisco Unified CME router with the ip source-address command and is placed in the router’s flash memory.

By specifying a second Cisco Unified CME router in the ip source-address command, you improve the failover time for phones.

Examples

The following example sets the IP source address and port:

```
Router(config)# telephony-service
Router(config-telephony)# ip source-address 10.6.21.4 port 2000 strict-match
```

The following example establishes the router at 10.5.2.78 as a secondary router:

```
Router(config)# telephony-service
Router(config-telephony)# ip source-address 10.0.0.1 port 2000 secondary 10.5.2.78
```

Cisco Unified CME 8.0 and Later Versions

The following example shows how to configure this command with an IPv6 address. Do not change the port number from the default value (2000) when you configure an IPv6 address.

```
Router(config)# telephony-service
Router(config-telephony)# protocol mode ipv6
Router(config-telephony)# ip source-address 2001:10:10:10::3
```

The following example shows how to configure an IP address for dual-stack mode. When the IP phones are configured for dual-stack mode, the IP address of the router port to which the IP phones are connected must be an IPv6 address. For dual-stack mode, the address of the secondary router must be an IPv4 address.

```
Router(config)# telephony-service
Router(config-telephony)# protocol mode dual-stack
Router(config-telephony)# ip source address 2001:10:10:10::3 secondary 10.5.2.78
```

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>option</td>
<td>Configures DHCP server options.</td>
</tr>
<tr>
<td></td>
<td>protocol mode</td>
<td>Configures a preferred IP-address mode for SCCP IP phones in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
ip source-address (telephony-service)
Cisco Unified CME Commands: K

- keepalive (ephone and ephone-template), on page 516
- keepalive (telephony-service), on page 518
- keepalive (voice register global), on page 519
- keepalive (voice register session-server), on page 520
- keepalive (vpn-profile), on page 521
- keep-conference, on page 522
- keep-conference (voice register), on page 525
- keygen-retry, on page 527
- keypad-normalize, on page 528
- keyphone, on page 529
**keepalive (ephone and ephone-template)**

To set the length of the time interval between successive keepalive messages from the Cisco Unified CME router to a particular IP phone, use the `keepalive` command in ephone or ephone-template configuration mode. To reset this length to the default value, use the `no` form of this command.

```plaintext
keepalive  seconds
no  keepalive
```

**Syntax Description**

- `seconds`: Interval time, in seconds. Range is from 10 to 65535. Default is 30.

**Command Default**

Default is 30 seconds

**Command Modes**

- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)T</td>
<td>Cisco CME 2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command allows the keepalive interval to be set for individual phones, typically so that wireless phone batteries are not run down too quickly by overly frequent keepalive signals.

If the router fails to receive three successive keepalive messages, it considers the phone to be out of service until the phone reregisters.

If the `keepalive (telephony-service)` command and this command are set to different time intervals, the value that you set in ephone configuration mode has priority for the particular phone only.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

**Examples**

The following example sets the keepalive interval to 300 seconds:

```plaintext
Router(config)# ephone 1
Router(config-ephone)# keepalive 300
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies template to ephone being configured.</td>
</tr>
</tbody>
</table>

---

516 Cisco Unified Communications Manager Express Command Reference
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>keepalive (telephony-service)</td>
<td>Sets the time interval for keepalive messages between IP phones and the Cisco Unified CME router.</td>
</tr>
</tbody>
</table>
keepalive (telephony-service)

To set the length of the time interval between successive keepalive messages from the Cisco CallManager Express router to IP phones, use the `keepalive` command in telephony-service configuration mode. To reset this length to the default value, use the `no` form of this command.

```
keepalive seconds
no keepalive
```

**Syntax Description**

| seconds | Interval time, in seconds. Range is from 10 to 65535. Default is 30. |

**Command Default**

Default is 30 seconds.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

If the router fails to receive three successive keepalive messages, it considers the phone to be out of service until the phone reregisters.

If the `keepalive (telephony-service)` command and the `keepalive (ephone)` command are set to different time intervals, the value that you set in ephone configuration mode has priority for the particular phone only.

**Examples**

The following example sets the keepalive time interval to 40 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# keepalive 40
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>keepalive (ephone)</td>
<td>Sets the time interval for keepalive messages between a particular IP phone and the Cisco CME router.</td>
</tr>
</tbody>
</table>
keepalive (voice register global)

To set the length of time interval between successive keepalive messages from SIP phones to the Cisco Unified CME router, use the keepalive command in voice register global configuration mode. To reset this timer duration to the default value, use the no form of this command.

`keepalive seconds`
`no keepalive`

**Syntax Description**

| seconds | Sets the time interval, in seconds, between keepalive messages that are sent to the router by SIP Phones. If the interval is set to a larger value, it is possible for notification to be delayed when the primary router goes down. Range is from 120 to 65535. Default is 120 seconds. |

**Command Default**
Default is 120 seconds.

**Command Modes**
Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IOS XE Everest 16.4.1</td>
<td>Cisco Unified CME 11.6</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
If the primary router fails, a SIP phone will not receive an acknowledgment (200 OK) to its REGISTER message to the primary router, and it will immediately failover to the secondary Cisco Unified CME router.

**Examples**

The following example sets the keepalive time interval to 200 seconds:

```
Router(config)# voice register global
Router(config-register-global)# keepalive 200
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>keepalive (telephony-service)</td>
<td>Sets the length of the time interval between successive keepalive messages from the Cisco Unified CME router to SCCP phones.</td>
</tr>
</tbody>
</table>
keepalive (voice register session-server)

To define the duration for registrations of external feature servers after which the registration expires, use the keepalive command in voice register session-server configuration mode. To return to the default, use the no form of this command.

```
keepalive seconds
no keepalive
```

**Syntax Description**

| seconds | Duration for registration, in seconds. Range: 60 to 3600. Default: 300. |

**Command Default**

Default is 300 seconds.

**Command Modes**

Voice register session-server configuration (config-register-fs)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XW2</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(920)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command defines the duration for registration, in seconds, after which the registration expires unless the feature server reregisters before the registration expiry.

**Examples**

The following partial output shows the configuration for a session manager for an external feature server, including a keepalive expiry of 360 seconds:

```
router# show running-configuration
!
!
voice register session-server 1
  register-id CSR1
  keepalive 360
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>register id</td>
<td>Creates an ID for explicitly identifying an external feature server during Register requests.</td>
</tr>
</tbody>
</table>
**keepalive (vpn-profile)**

To specify the duration of time required to generate a keepalive message to the VPN concentrator, use the `keepalive` command in vpn-profile configuration mode.

```
keepalive  seconds
```

**Keyword Description**
- `seconds`: Duration for a vpn-profile session, in seconds. Range: 0 to 120. Default: 60.

**Command Default**
Default is 60 seconds.

**Command Modes**
Vpn-profile configuration (conf-vpn-profile)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to specify the amount of time required to generate a keepalive message to the VPN concentrator. The keepalive session ranges from 0 to 120 seconds. The default keepalive session is 60 seconds.

**Examples**
The following example shows the keepalive duration set to 50 seconds for vpn-profile 1.

```
Router#show run
!
!
voice service voip
  ip address trusted list
  ipv4 20.20.20.1
  vpn-group 1
  vpn-gateway 1 https://9.10.60.254/SSLVPNphone
  vpn-trustpoint 1 trustpoint cme_cert root
  vpn-hash-algorithm sha-1
  vpn-profile 1
  keepalive 50
  host-id-check disable
  vpn-profile 2
  mtu 1300
  password-persistent enable
  host-id-check enable
  sip
  !
  voice class media 10
  media flow-around
  !
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>vpn-profile</td>
<td>Defines a VPN-profile.</td>
</tr>
</tbody>
</table>
keep-conference

To allow conference initiators to exit from conference calls and to either end or maintain the conference for the remaining parties, use the keep-conference command in ephone or ephone-template configuration mode. To return to the default, use the no form of this command.

```
keep-conference [drop-last] [endcall] [local-only]
no keep-conference
```

**Syntax Description**

- **drop-last** (Optional) The action of the Confrn soft key is changed; the conference initiator can press the Confrn soft key (IP phone) or hookflash (analog phone) to drop the last party.
  
  **Note** Analog phones connected to the Cisco Unified CME system through a Cisco VG 224 require Cisco IOS Release 12.3(11)Y1 or a later release to use this feature.

- **endcall** (Optional) The action of the EndCall soft key is changed; the conference initiator can hang up or press the EndCall soft key to leave the conference and keep the other two parties connected.
  
  **Note** If this option is not enabled, pressing the EndCall soft key terminates the conference and disconnects all parties.

- **local-only** (Optional) The conference initiator can hang up to end the conference and leave the other two parties connected only if one of the remaining parties is local to the Cisco Unified CME system (an internal extension).

**Command Default**

A conference initiator can hang up or press the EndCall soft key to end a conference and disconnect all parties or press the Confrn soft key to drop only the last party that was connected to the conference.

**Command Modes**

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco CME 3.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The drop-last and local-only keywords were added, and this command was made available in ephone-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>The drop-last and local-only keywords, and this command in ephone-template configuration mode were integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

- **Note** This feature uses call transfer to connect the two remaining parties of a conference when a conference initiator leaves the conference. To use this feature, you must configure the transfer-system command using the full-blind, full-consult, or full-consult dss keywords.
If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

If the `keep-conference` command is configured with no keywords, a conference initiator can hang up to leave the conference and the other two parties will remain connected. Alternatively, the conference initiator can use the EndCall soft key to terminate the conference and disconnect all parties.

If the `keep-conference` command is configured with no keywords, a conference initiator can use the Confrn soft key (IP phone) or hookflash (analog phone) to break up the conference but stay connected to both parties. The oldest call will be put on hold, and the most recent call will be actively connected to the initiator. The conference initiator can navigate between the two parties by pressing the Hold soft key or the appropriate line button on the phone.

If the `endcall` keyword is used, the conference initiator can hang up or press the EndCall soft key to leave the conference with the other two parties remaining connected.

In Cisco CME 3.2.3 and later versions, if the `keep-conference` command is not configured (the default) or if the `no keep-conference` command is used, a conference initiator can drop the last party that was added to the conference by pressing the Confrn soft key (IP phone) or hookflash (analog phone).

---

**Note**

Analog phones connected to the Cisco Unified CME system through a Cisco VG 224 require Cisco IOS Release 12.3(11)YL1 or a later release to use this feature.

---

### Examples

In the following example, extension 3555 initiates a three-way conference. After the conference is established, extension 3555 can press the Confrn soft key to disconnect the last party that was connected and remain connected to the first party that was connected. If extension 3555 hangs up from the conference, the other two parties remain connected if one of them is local to the Cisco Unified CME system.

```plaintext
ephone-dn 35  
  number 3555  
ephone 24  
  button 1:35  
    keep-conference drop-last local-only
```

In the following example, extension 3666 initiates a three-way conference. After the conference is established, extension 3666 can press the Confrn soft key to disconnect the last party that was connected and remain connected to the first party that was connected. Also, extension 3666 can hang up from a three-way conference to terminate the conference and disconnect all parties or can press the EndCall soft key to leave the conference and keep the other two parties connected.

```plaintext
ephone-dn 36  
  number 3666  
ephone 25  
  button 1:36  
    keep-conference drop-last endcall
```

In the following example, extension 3777 initiates a three-way conference. After the conference is established, extension 3777 can press the Confrn soft key to disconnect the last party that was connected and remain connected to the first party that was connected. Also, extension 3777 can hang up from a three-way conference to terminate the conference and disconnect all parties or press the
EndCall soft key to leave the conference and keep the other two parties connected only if one of the two parties is local to the Cisco Unified CME system.

```plaintext
ephone-dn 38
  number 3777
  ephone 27
  button 1:38
  keep-conference drop-last endcall local-only
```

In the following example, extension 3999 initiates a three-way conference. After the conference is established, extension 3999 can hang up to terminate the conference and disconnect all parties or press the EndCall soft key to leave the conference and keep the other two parties connected only if one of the two parties is local to the Cisco Unified CME system.

```plaintext
ephone-dn 39
  number 3999
  ephone 29
  button 1:39
  keep-conference endcall local-only
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies template to ephone being configured.</td>
</tr>
<tr>
<td>max-conferences</td>
<td>Sets the maximum number of three-party conferences simultaneously supported</td>
</tr>
<tr>
<td></td>
<td>by the Cisco Unified CME router.</td>
</tr>
<tr>
<td>transfer-system</td>
<td>Specifies the call transfer method for IP phone extensions that use the ITU-T</td>
</tr>
<tr>
<td></td>
<td>H.450.2 standard.</td>
</tr>
</tbody>
</table>
keep-conference (voice register)

To allow IP phone conference initiators to exit from conference calls and keep the remaining parties connected, use the `keep-conference` command in voice register pool configuration mode or voice register template configuration mode. To disable the keep-conference feature, use the `no` form of this command.

```
keep-conference
no keep-conference
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Default is enabled.

**Command Modes**
Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>Cisco IOS XE Everest 16.5.1b</td>
<td>Unified CME 11.7</td>
<td>This command was supported in voice register template configuration mode.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
When the conference initiator hangs up, Cisco Unified Communications Manager Express (Cisco Unified CME) executes a call transfer to connect the two remaining lines. The remaining calls are transferred without consultation. To facilitate call transfer, the `transfer-attended` command or `transfer-blind` command must be enabled.

Conference initiators can disconnect from their conference calls by pressing the Confrn (conference) soft key. When an initiator uses the Confrn soft key to disconnect from the conference call, the oldest call leg is put on hold, leaving the initiator connected to the most recent call leg. The conference initiator can then navigate between the two separate parties by pressing either the Hold soft key or the line buttons to select the desired call.

**Examples**
The following example shows how to configure this command, if it was previously disabled, to keep remaining conference legs after the conference initiator hangs up.

```
Router(config)# voice register pool 1
Router(config-register-pool)# keep-conference
```

The following example shows how to configure this command under voice register template configuration mode.

```
Router(config)# voice register template 1
Router(config-register-template)# keep-conference
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>conference (voice register template)</td>
<td>Enables a soft key for conference in a SIP phone template.</td>
</tr>
<tr>
<td>max-conferences</td>
<td>Sets the maximum number of three-party conferences simultaneously supported by the Cisco CME router.</td>
</tr>
<tr>
<td>transfer-attended (voice register template)</td>
<td>Enables a soft key for attended transfer in a SIP phone template.</td>
</tr>
<tr>
<td>transfer-blind (voice register template)</td>
<td>Enables a soft key for blind transfer in a SIP phone template.</td>
</tr>
<tr>
<td>voice register template</td>
<td>Enters voice register template configuration mode and defines a template of common parameters for SIP phones.</td>
</tr>
</tbody>
</table>
**keygen-retry**

To specify the number of times that a CAPF server sends a key-generation request, use the **keygen-retry** command in CAPF-server configuration mode. To return to the default, use the **no** form of this command.

```
keygen-retry number
no keygen-retry
```

**Syntax Description**

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>number</strong></td>
<td>Number of retries. Range is from 0 to 100. Default is 3.</td>
</tr>
</tbody>
</table>

**Command Default**

Number of retries is 3.

**Command Modes**

CAPF-server configuration (config-capf-server)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication.

**Examples**

The following example specifies that the key generation process should be tried 5 times.

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>keygen-timeout</strong></td>
<td>Specifies the number of minutes that the CAPF server waits for a key-generation response from a phone.</td>
</tr>
</tbody>
</table>
### keypad-normalize

To impose a 200-millisecond delay before each keypad message from an IP phone, use the **keypad-normalize** command in ephone or ephone-template configuration mode. To return to the default, use the **no** form of this command.

**keypad-normalize**
**no keypad-normalize**

**Syntax Description**
This command has no keywords or arguments.

**Command Default**
Keypad messages are handled as fast as the system can handle them, without an imposed delay.

**Command Modes**
Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command normalizes the processing of incoming keypad messages from an IP phone so that one message is processed every 200 milliseconds. This is useful for handling the personal speed dial (fastdial) feature when the destination of the call tends to be slower in accepting the digits, or when converting keypad messages into appropriate digit events on the network side, such as RFC 2833 digits.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

**Examples**
The following example normalizes the sending of digits from ephone 43.

```
ephone 43
button 1:29
keypad-normalize
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies template to ephone being configured.</td>
</tr>
</tbody>
</table>
keyphone

To designate a Cisco Unified IP phone as a marked or “key” phone when using the Cisco Unified CME eXtensible Markup Language (XML) application program interface (API), use the `keyphone` command in `ephone` or `ephone-template` configuration mode. To remove the keyphone designation, use the `no` form of this command.

```
keyphone
no keyphone
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
The phone that is being configured is not a “key” phone.

**Command Modes**
Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command is used with the XML API to mark a Cisco Unified IP phone as a “key” phone to be tracked while using the XML API. The XML API can be instructed to report the status of only the “key” phones in the system for network management purposes, for example.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

**Examples**
The following example sets the phone with the phone tag of 1 as a “key” phone for the XML API:

```
Router(config)# ephone 1
Router(config-ephone)# keyphone
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template</td>
<td>Applies template to ephone being configured.</td>
</tr>
</tbody>
</table>
Cisco Unified CME Commands: L

- label, on page 532
- label (voice register dn), on page 533
- list (ephone-hunt), on page 534
- list (voice hunt-group), on page 537
- live-record, on page 539
- load (telephony-service), on page 540
- load (voice register global), on page 544
- load-cfg-file, on page 547
- loc2, on page 548
- location (voice emergency response zone), on page 549
- log password, on page 551
- log table, on page 552
- logging (voice emergency response settings), on page 553
- login (telephony-service), on page 555
- logo (voice register global), on page 557
- logout-profile, on page 558
- loopback-dn, on page 560
- lpcor incoming, on page 564
- lpcor outgoing, on page 566
- lpcor type, on page 568
**label**

To create a text identifier instead of a phone-number display for an extension on an IP phone console, use the label command in ephone-dn configuration mode. To delete a label, use the no form of this command.

```
label string
no label string
```

**Syntax Description**

- `string` | Alphanumeric string of up to 30 characters.

**Command Default**
No label is defined.

**Command Modes**
Ephone-dn configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

One label is allowed per extension (ephone-dn). The ephone-dn must already have a number that was set using the number command before a label can be created for it.

This command must be followed by a quick reboot of the phone on which the label appears, using the restart command.

**Examples**

The following example creates three phone labels to appear in place of three phone numbers on IP phone console displays:

```
Router(config)# ephone-dn 10
Router(config-ephone-dn)# label user10
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 20
Router(config-ephone-dn)# label user20
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 30
Router(config-ephone-dn)# label user30
Router(config-ephone-dn)# exit
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>number</td>
<td>Associates a telephone or extension number with an ephone-dn in a Cisco CME system.</td>
</tr>
<tr>
<td>restart (ephone)</td>
<td>Performs a fast reboot of a single phone associated with a Cisco CME router.</td>
</tr>
<tr>
<td>restart (telephony-service)</td>
<td>Performs a fast reboot of one or all phones associated with a Cisco CME router.</td>
</tr>
</tbody>
</table>
**label (voice register dn)**

To create a text identifier instead of a phone-number display for an extension on a SIP phone console, use the **label command** in voice register dn configuration mode. To delete a label, use the **no** form of this command.

```
label string
no label string
```

**Syntax Description**

| string | Alphanumeric string of up to 30 characters. |

**Command Default**

No label is created.

**Command Modes**

Voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

One label is allowed per extension (directory number). The directory number must already have a number that was set by using the **number** command before a label can be created for it.

After you configure this command, restart the phone by using the **reset** command.

**Examples**

The following example shows how to create three phone labels to appear in place of three phone numbers on Cisco IP phone console displays:

```
Router(config)# voice register dn 10
Router(config-register-dn)# label user10
Router(config-register-dn)# exit
Router(config)# voice register dn 20
Router(config-register-dn)# label user20
Router(config-register-dn)# exit
Router(config)# voice register dn 30
Router(config-register-dn)# label user30
Router(config-register-dn)# exit
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>number (voice register dn)</td>
<td>Associates a telephone or extension number with a directory number in a Cisco CME system.</td>
</tr>
<tr>
<td>reset (voice register pool)</td>
<td>Performs a compete reboot of a single SIP phone associated with a Cisco CME router.</td>
</tr>
<tr>
<td>reset (voice register global)</td>
<td>Performs a complete reboot of all SIP phones associated with a Cisco CME router.</td>
</tr>
</tbody>
</table>
list (ephone-hunt)

To create a list of extensions that are members of a Cisco Unified CME ephone hunt group, use the list command in ephone-hunt configuration mode. To remove a list from the router configuration, use the no form of this command.

```
list number[,number...]
no list
```

**Syntax Description**
- `number`: Preconfigured extension or E.164 number.
  An asterisk (*) can take the place of an extension number to represent a wildcard slot. An agent at an authorized ephone-dn can dynamically join and leave a hunt group if a wildcard slot is available. There can be up to 20 wildcard slots in a hunt group.

**Command Default**
No list is defined.

**Command Modes**
Ephone-hunt configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>The number of ephone-dns allowed was increased to 20.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The use of wildcard asterisks (*) in the <code>dn-number</code> argument was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>The use of wildcard asterisks (*) in the <code>dn-number</code> argument was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to create a list of member numbers for defining a hunt group.

List must contain 1 to 20 numbers.

A number cannot be added to a list unless it was already defined by using the `number` command.

Add or delete all numbers in a hunt group list at one time. You cannot add or single number to an existing list or remove one number from a list.

Any number in the list cannot be a pilot number of a parallel hunt group.

To allow dynamic membership in a hunt group, use asterisks to represent wildcard slots in the `list` command.

To allow an ephone-dn to use one of the wildcard slots to dynamically join a hunt group, use the `ephone-hunt login` command under that ephone-dn. Ephone-dns are disallowed from joining hunt groups by default, so you have to explicitly allow this behavior for each ephone-dn that you want to be able to log into hunt groups.
The `show ephone-hunt` command displays the numbers associated to ephone-dns that are joined to groups at the time that the command is run, in addition to static members of the hunt group. Static hunt group members are the numbers that are explicitly named in the `list` command.

**Examples**

The following example creates sequential hunt group number 7, which contains four static members (ephone-dns):

```plaintext
Router(config)# ephone-hunt 7 sequential
Router(config-ephone-hunt)# list 7711, 7712, 7713, 7714
```

The following example creates five ephone-dns and a hunt group that includes the first two ephone-dns as static members and two wildcard slots for dynamic hunt group members. The last three ephone-dns are enabled for dynamic membership in the hunt group. Any of them can join the hunt group whenever one of the wildcard slots is available. Once an ephone-dn has joined a hunt group, it can leave at any time.

```
ephone-dn 22
    number 4566
ephone-dn 23
    number 4567
ephone-dn 24
    number 4568
ephone-hunt login
ephone-dn 25
    number 4569
ephone-hunt login
ephone-dn 26
    number 4570
ephone-hunt login
ephone-hunt 1 peer
    list 4566,4567,*,*
final 7777
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ephone-hunt login</code></td>
<td>Allows an ephone-dn to dynamically join and leave an ephone hunt group.</td>
</tr>
<tr>
<td><code>final</code></td>
<td>Defines the last ephone-dn in an ephone hunt group.</td>
</tr>
<tr>
<td><code>hops</code></td>
<td>Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.</td>
</tr>
<tr>
<td><code>max-redirect</code></td>
<td>Changes the current number of allowable redirects in a Cisco CME system.</td>
</tr>
<tr>
<td><code>no-reg (ephone-hunt)</code></td>
<td>Specifies that the pilot number of this ephone hunt group should not register with the H.323 gatekeeper.</td>
</tr>
<tr>
<td><code>number (ephone-dn)</code></td>
<td>Associates an extension or telephone number with a directory number.</td>
</tr>
<tr>
<td><code>pilot</code></td>
<td>Defines the ephone-dn that is dialed to reach an ephone hunt group.</td>
</tr>
<tr>
<td><code>preference (ephone-hunt)</code></td>
<td>Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>show ephone-hunt</td>
<td>Displays ephone-hunt group configuration, current status, and statistics.</td>
</tr>
<tr>
<td>timeout (ephone-hunt)</td>
<td>Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.</td>
</tr>
</tbody>
</table>
list (voice hunt-group)

To define a list of extensions that are members of a voice hunt-group, use the list command in voice hunt-group configuration mode. To remove a list, use the no form of this command.

```
list number, number[, number...]
no list
```

### Syntax Description

| number | Extension or E.164 number assigned to a phone in Cisco Unified CME. List must contain 2 to 32 numbers. |

### Command Default

By default, hunt group list is not defined.

### Command Modes

Voice hunt-group configuration (config-voice-hunt-group)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>The maximum numbers allowed in a list was expanded from 10 to 32 and support for SCCP phones was added.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was modified to include support for wildcards which is indicated by &quot;*&quot;. symbol.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command creates the list of numbers to include in a voice hunt-group. Phones with these numbers ring when the hunt group pilot number is dialed. The numbers must be assigned to directory numbers with the number command.

All numbers in a hunt group list are added or deleted at one time. You cannot add a number to an existing list or remove a number from a list.

The pilot number of a parallel hunt group and shared-line numbers are not supported.

A phone number associated with an FXO port is not supported in parallel hunt groups.

### Examples

The following example shows how to create a sequential hunt group containing four extensions and a wildcard slot:

```
Router(config)# voice hunt-group 1 sequential
Router(config-voice-hunt-group)# list 7711, 7712, 7713, 7714, *
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>final (voice hunt-group)</td>
<td>Defines the last extension in a voice hunt group.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>hops (voice hunt-group)</td>
<td>Defines the number of times that a call is redirected to the next phone number in a peer hunt-group list before proceeding to the final number.</td>
</tr>
<tr>
<td>number (ephone-dn)</td>
<td>Associates an extension or telephone number with a directory number.</td>
</tr>
<tr>
<td>number (voice register dn)</td>
<td>Associates an extension or telephone number with a directory number.</td>
</tr>
<tr>
<td>pilot (voice hunt-group)</td>
<td>Defines the phone number that callers dial to reach a voice hunt group.</td>
</tr>
<tr>
<td>timeout (voice hunt-group)</td>
<td>Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list and defines the last number in the hunt group.</td>
</tr>
</tbody>
</table>
live-record

To define the extension number that is dialed when the LiveRcd soft key is pressed on a Cisco Unified IP Phone, use the `live-record` command in telephony-service configuration mode. To reset to the default value, use the `no` form of this command.

```
live-record  phone-number
no  live-record
```

**Syntax Description**

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>phone-number</strong></td>
<td>Telephone number that is dialed when the LiveRcd soft key is pressed.</td>
</tr>
</tbody>
</table>

**Command Default**

Live record is disabled.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies the telephone number that is speed-dialed to access the Live Record feature when the LiveRcd soft key on a Cisco Unified IP phone is pressed. This telephone number is used for all Cisco Unified IP phones connected to the router.

This telephone number must match the Live Record number configured in Cisco Unity Express.

**Examples**

The following example shows that the phone number 914085550100 is speed-dialed to record a call when the LiveRcd button is pressed:

```
Router(config)#  telephony-service
Router(config-telephony)#  live-record  914085550100
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone.</td>
</tr>
<tr>
<td>softkeys connected</td>
<td>Modifies the order and type of soft keys that display on an IP phone during the connected call state.</td>
</tr>
<tr>
<td>voicemail</td>
<td>Defines the telephone number that is speed-dialed when the Messages button is pressed on an IP phone.</td>
</tr>
</tbody>
</table>
To associate a type of Cisco Unified IP phone with a phone firmware file, use the `load` command in telephony-service configuration mode. To disassociate a type of phone from a phone firmware file, use the `no` form of this command.

`load` `phone-type` `firmware-file`
`no load` `phone-type`

### Syntax Description

<table>
<thead>
<tr>
<th><code>phone-type</code></th>
<th>Type of phone. The following phone types are predefined in the system:</th>
</tr>
</thead>
<tbody>
<tr>
<td>6945</td>
<td>Cisco Unified IP Phone 6945.</td>
</tr>
<tr>
<td>7902</td>
<td>Cisco Unified IP Phone 7902G.</td>
</tr>
<tr>
<td>7905</td>
<td>Cisco Unified IP Phone 7905G.</td>
</tr>
<tr>
<td>7910</td>
<td>Cisco Unified IP Phone 7910 and 7910G.</td>
</tr>
<tr>
<td>7911</td>
<td>Cisco Unified IP Phone 7911G.</td>
</tr>
<tr>
<td>7912</td>
<td>Cisco Unified IP Phone 7912G.</td>
</tr>
<tr>
<td>7914</td>
<td>Cisco Unified IP Phone 7914 Expansion Module.</td>
</tr>
<tr>
<td>7920</td>
<td>Cisco Unified Wireless IP Phone 7920.</td>
</tr>
<tr>
<td>7921</td>
<td>Cisco Unified Wireless IP Phone 7921.</td>
</tr>
<tr>
<td>7931</td>
<td>Cisco Unified IP Phone 7931G.</td>
</tr>
<tr>
<td>7935</td>
<td>Cisco Unified IP Conference Station 7935.</td>
</tr>
<tr>
<td>7936</td>
<td>Cisco Unified IP Conference Station 7936.</td>
</tr>
<tr>
<td>7941</td>
<td>Cisco Unified IP Phone 7941G.</td>
</tr>
<tr>
<td>7941GE</td>
<td>Cisco Unified IP Phone 7941G-GE.</td>
</tr>
<tr>
<td>7942</td>
<td>Cisco Unified IP Phone 7942.</td>
</tr>
<tr>
<td>7945</td>
<td>Cisco Unified IP Phone 7945.</td>
</tr>
<tr>
<td>7960-7940</td>
<td>Cisco Unified IP Phones 7960 and 7960G and Cisco Unified IP Phone 7940 and 7940G.</td>
</tr>
<tr>
<td>7961</td>
<td>Cisco Unified IP Phone 7961G.</td>
</tr>
<tr>
<td>7961GE</td>
<td>Cisco Unified IP Phone 7961G-GE.</td>
</tr>
<tr>
<td>7962</td>
<td>Cisco Unified IP Phone 7962.</td>
</tr>
<tr>
<td>7965</td>
<td>Cisco Unified IP Phone 7965.</td>
</tr>
<tr>
<td>7970</td>
<td>Cisco Unified IP Phone 7970G.</td>
</tr>
<tr>
<td>7971</td>
<td>Cisco Unified IP Phone 7971G-GE.</td>
</tr>
<tr>
<td>7975</td>
<td>Cisco Unified IP Phone 7975.</td>
</tr>
<tr>
<td>7985</td>
<td>Cisco Unified IP Phone 7985.</td>
</tr>
<tr>
<td>8941</td>
<td>Cisco Unified IP Phone 8941.</td>
</tr>
<tr>
<td>8945</td>
<td>Cisco Unified IP Phone 8945.</td>
</tr>
<tr>
<td>ata</td>
<td>Cisco ATA-186 and Cisco ATA-188.</td>
</tr>
</tbody>
</table>

**Note** You can also add a new phone type to your configuration by using the `ephone-type` command.
**firmware-file**  Filename of the IP phone firmware for a particular phone type.

- In Cisco Unified CME 7.0/4.3 and earlier versions, do not use the file suffix (.bin, .sbin, .loads) for any phone type except the Cisco ATA and Cisco Unified IP Phone 7905 and 7912.
- In Cisco Unified CME 7.0(1) and later versions, you must use the complete filename, including the file suffix, for phone firmware versions later than version 8-2-2 for all phone types.
- Filenames are case sensitive.

Firmware files are not associated with phone types.

Telephony-service configuration (config-telephony)

<table>
<thead>
<tr>
<th>Command History</th>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td></td>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td></td>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1</td>
<td>Support was added for the Cisco IP Phone 7914 Expansion Module.</td>
</tr>
<tr>
<td></td>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
<tr>
<td></td>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>The following keywords were added to this command: <strong>7902, 7905, and 7912</strong>.</td>
</tr>
<tr>
<td></td>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td></td>
<td>12.3(7)T</td>
<td>Cisco CME 3.1</td>
<td>The <strong>7920</strong> and <strong>7936</strong> keywords were added.</td>
</tr>
<tr>
<td></td>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>The <strong>7970</strong> keyword was added.</td>
</tr>
<tr>
<td></td>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>The <strong>7971</strong> keyword was added, and this command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td></td>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The <strong>7911, 7941, 7941GE, 7961, and 7961GE</strong> keywords were added.</td>
</tr>
<tr>
<td></td>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td></td>
<td>12.4(6)XE</td>
<td>Cisco Unified CME 4.0(2)</td>
<td>The <strong>7931</strong> keyword was added.</td>
</tr>
<tr>
<td></td>
<td>12.4(4)XC4</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>The <strong>7931</strong> keyword was added.</td>
</tr>
<tr>
<td></td>
<td>12.4(11)T</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>This command was integrated into Cisco IOS Release 12.4(11)T.</td>
</tr>
</tbody>
</table>
Modification
Cisco Product
Cisco IOS Release
Modification
---
12.4(11)XJ2
Cisco Unified CME 4.1
The 7921 and 7985 keywords were introduced.
12.4(15)T
Cisco Unified CME 4.1
This command was integrated into Cisco IOS Release 12.4(15)T.
12.4(15)T1
Cisco Unified CME 4.1(1)
The 7942, 7945, 7962, 7965, and 7975 keywords were introduced.
12.4(15)XZ
Cisco Unified CME 4.3
Support for user-defined phone types created with the ephone-type command was added.
12.4(20)T
Cisco Unified CME 7.0
This command was integrated into Cisco IOS Release 12.4(20)T.
12.4(20)YA
Cisco Unified CME 7.0(1)
Support for automatically creating bindings for firmware files only if the cnf-file location is flash or slot0 was added.
12.4(20)T1
Cisco Unified CME 7.0
The 7925 keyword was introduced.
12.4(22)T
Cisco Unified CME 7.0(1)
This command was integrated into Cisco IOS Release 12.4(22)T.
15.2(1)T
Cisco Unified CME 8.8
This command was modified. The 6945, 8941, and 8945 keywords were added.

Usage Guidelines
This command updates the Cisco Unified CME configuration file for the specified type of Cisco Unified IP phone to add the name of the firmware file to be loaded by a particular phone type. The firmware filename also provides the version number for the phone firmware that is in the file. When a phone is started up or rebooted, the phone reads the configuration file to determine which firmware file it must load and then looks for that firmware file on the TFTP server.

If applicable, Cisco Unified IP phones update themselves with new phone firmware whenever they are started up or rebooted.

A separate load command is needed for each type of phone. The Cisco Unified IP Phones 7940 and 7940G and the Cisco Unified IP Phones 7960 and 7960G have the same phone firmware and share the 7960-7940 keyword.

Before Cisco Unified CME 7.0(1):
- Do not include the file suffix (.bin, .sbin, .loads) for any phone type except Cisco ATA and Cisco Unified IP Phone 7905 and 7912 when you configure the load command in telephony-service configuration mode. For example:

```
Router(config-telephony)# load 7941 SCCP41.8-2-2SR2S
Router(config-telephony)#
```

- You must also configure the tftp-server command to enable TFTP access to the firmware files by Cisco Unified IP phones.

In Cisco Unified CME 7.0(1) and later versions:
• When specifying the load command for phone firmware versions later than version 8-2-2 for all phone types and you use the file suffix in the filename, the tftp-server bindings are automatically added for all the files forwarded for that load. For example:

Router(config-telephony)# load 7941 SCCP41.8-3-3S.loads
Router(config-telephony)#

• The load command is enhanced to automatically create TFTP bindings for phone firmware files if the cnf-file location command is configured with the flash or slot0 keyword. You are no longer required to configure the tftp-server command to create TFTP bindings only if the location of the cnf files is router flash or slot 0 memory. If the cnf-file location command is configured for something other than flash or slot 0, such as a TFTP server (url) or system memory (system:its/), you must still configure the tftp-server command to create TFTP bindings for phone firmware files. Use the complete filename, including the file suffix, when you configure the tftp-server command for phone firmware versions later than version 8-2-2 for all phone types.

To verify TFTP bindings, including the dictionary, language, and tone configuration files that are associated with the ISO-3166 codes that have been selected, use the show telephony-service tftp-bindings command.

After associating a firmware file with a Cisco Unified IP phone, use the reset command to reboot the phone.

Examples

Cisco Unified CME 7.0 and Earlier Versions

The following example shows how to identify the Cisco Unified IP phone firmware file to be used by the Cisco Unified IP Phones 7960 and 7960G and Cisco Unified IP Phone 7910G:

Router(config)# telephony-service
Router(config-telephony)# load 7960-7940 P00303020209
Router(config-telephony)# load 7910 P00403020209
Router(config-telephony)# exit
Router(config)# tftp-server flash:P00303020209.bin
Router(config)# tftp-server flash:P00403020209.bin

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>cnf-file location</td>
<td>Specifies a storage location for phone configuration files.</td>
</tr>
<tr>
<td>ephone-type</td>
<td>Adds a Cisco Unified IP phone type by defining a phone-type template.</td>
</tr>
<tr>
<td>reset</td>
<td>Resets a Cisco Unified IP phone.</td>
</tr>
<tr>
<td>show telephony-service tftp-bindings</td>
<td>Provides a list of configuration files that are accessible to IP phones using TFTP.</td>
</tr>
<tr>
<td>tftp-server</td>
<td>Enables TFTP access to firmware files on the TFTP server.</td>
</tr>
</tbody>
</table>
load (voice register global)

To associate a type of IP phone with a phone firmware file, use the load command in voice register global configuration mode. To disassociate a type of phone from a phone firmware file, use the no form of this command.

```
load  phone-type  firmware-file
no  load  phone-type  firmware-file
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>phone-type</td>
<td>Type of IP phone. The following choices are valid:</td>
</tr>
<tr>
<td>firmware-file</td>
<td>Filename for the Cisco Unified IP phone firmware to be associated with the IP phone type. Do not use the .bin or .load file extension, except for the Cisco Unified IP phone 7905, 7912, or ATA. Filenames are case sensitive.</td>
</tr>
</tbody>
</table>
The firmware file is not associated with the type of phone.

Voice register global configuration (config-register-global)

<table>
<thead>
<tr>
<th>Command History</th>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco MCE 3.4</td>
<td>This command was introduced.</td>
<td></td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified MCE 4.1</td>
<td>The 3951, 7911, 7941, 7941GE, 7961, 7961GE, 7970, and 7971 keyword were added.</td>
<td></td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified MCE 4.1</td>
<td>The 3951, 7911, 7941, 7941GE, 7961, 7961GE, 7970, and 7971 keyword were integrated into Cisco IOS Software Release 12.4(15)T.</td>
<td></td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified MCE 4.3</td>
<td>The 7942, 7945, 7962, 7965, and 7975 keyword were added.</td>
<td></td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified MCE 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
<td></td>
</tr>
<tr>
<td>15.2(1)T</td>
<td>Cisco Unified MCE 8.8</td>
<td>This command was modified. The 3905 keyword was added.</td>
<td></td>
</tr>
<tr>
<td>15.2(2)T</td>
<td>Cisco Unified MCE 9.0</td>
<td>This command was modified. The 6901, 6911, 6921, 6941, 6945, 6961, and ATA-187 keyword were added.</td>
<td></td>
</tr>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified MCE 10.5</td>
<td>This command was modified to provide support for Cisco Unified 7821, 7841, 7861 and DX650 IP phones.</td>
<td></td>
</tr>
</tbody>
</table>

Usage Guidelines

This command updates the Cisco Unified CME configuration file for the specified type of IP phone to add the name of the correct firmware file that the phone should load. This filename also provides the version number for the phone firmware that is in the file. Later, whenever a phone is started up or rebooted, the phone reads the configuration file to determine the name of the firmware file that it should load and then looks for that firmware file on the TFTP server.

A separate load command is needed for each type of phone. The Cisco Unified IP Phone 7940 and 7940G and Cisco Unified IP Phone 7960 and 7960G have the same phone firmware and share the 7960-7940 keyword. The Cisco Unified IP Phone 3911 and Cisco Unified IP Phone 3951 have the same phone firmware and share the 3951 keyword.

For certain IP phones, such as the Cisco Unified IP Phone 7906G, 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971G, there are multiple firmware files. For these phones, use the TERMnn.x-y-x-w.loads or SIPnn.x-y-x-w.loads firmware filename for the load command, without the .loads file extension. For these phones, you do not configure the load command for any firmware file other than the TERM.loads or SIP.loads firmware file.

Following the load command, use the tftp-server command to enable TFTP access to the file by Cisco Unified IP phones. The file extension is required when using the tftp-server command.

The load command must be followed by a reboot of the phones. Plug in a new IP phone or use the reset command to reboot an IP phone that is already connected to the Cisco router.

Examples

The following example shows how to configure the load command to indicate which phone firmware is to be used by a Cisco Unified IP Phone 7960 and 7960G, a Cisco Unified IP Phone 7912 and
7912G, and a Cisco Unified IP Phone 7941GEs. The `tftp-server` command is used to specify the location of the phone firmware files, including all firmware files for the Java-based Cisco Unified IP Phone 7941GE. Note that while no file extension is used with the `load` command, the file extension is required when using the `tftp-server` command.

```bash
Router(config)# voice register global
Router(config-register-global)# load 7960-7940 P00303020209
Router(config-register-global)# load 7912 P00403020209
Router(config-register-global)# load 7941 TERM41.7-0-3-0S
Router(config-register-global)# exit
Router(config)# tftp-server flash:P00303020209.bin
Router(config)# tftp-server flash:P00403020209.bin
Router(config)# tftp-server flash:SIP41.8-0-3-0S.loads
Router(config)# tftp-server flash:term61.default.loadsterm
Router(config)# tftp-server flash:41.default.loads
Router(config)# tftp-server flash:CVM41.2-0-2-26.sbn
Router(config)# tftp-server flash:cnu41.2-7-6-26.sbn
Router(config)# tftp-server flash:Jar41.2-9-2-26.sbn
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>reset (voice register global)</code></td>
<td>Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td><code>show voice register global</code></td>
<td>Displays all global configuration parameters associated with SIP phones.</td>
</tr>
<tr>
<td><code>tftp-server</code></td>
<td>Enables TFTP access to firmware files on the TFTP server.</td>
</tr>
<tr>
<td><code>type (voice register pool)</code></td>
<td>Defines a phone type for a SIP phone.</td>
</tr>
</tbody>
</table>
load-cfg-file

To load configuration files on the TFTP server and to sign configuration files that are not created by Cisco Unified CME, use the `load-cfg-file` command in telephony-service configuration mode. To return to the default, use the `no` form of this command.

```
load-cfg-file file-url alias file-alias [sign] [create]
no load-cfg-file file-url alias file-alias
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>file-url</code></td>
<td>Complete path of a configuration file in a local directory.</td>
</tr>
<tr>
<td><code>alias file-alias</code></td>
<td>Name of the file on the TFTP server.</td>
</tr>
<tr>
<td><code>sign</code></td>
<td>Signs the file and serves it on the TFTP server.</td>
</tr>
<tr>
<td><code>create</code></td>
<td>Creates the signed file in the local directory.</td>
</tr>
</tbody>
</table>

### Command Default

A file is not loaded on the TFTP server.

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command is used with Cisco Unified CME phone authentication to sign configuration files that are not created by Cisco Unified CME. This command also loads the signed and unsigned versions of the file on the TFTP server. To simply serve an already signed file on the TFTP server, use this command without the `sign` and `create` keywords.

The `create` keyword should be used with the `sign` keyword the first time that this command is used for each file. The `create` keyword is not maintained in the running configuration; this prevents signed files from being recreated during every reload.

### Examples

The following example creates a file called `ringlist.xml.sgn` in slot0 and serves both `ringlist.xml` and `ringlist.xml.sgn` on the TFTP server.

```
telephony-service
load-cfg-file slot0:Ringlist.xml alias Ringlist.xml sign create
```

The following example serves `P00307010200.sbn` on the TFTP server without creating a signed file.

```
telephony-service
load-cfg-file slot0:P00307010200.sbn alias P00307010200.sbn
```
**loc2**

To specify the audio file used for the loss of C2 features announcement, use the `loc2` command in voice MLPP configuration mode. To disable use of this audio file, use the `no` form of this command.

```
loc2 audio-url
no loc2
```

**Syntax Description**

`audio-url` Location of the announcement audio file in URL format. Valid storage locations are TFTP, FTP, HTTP, and flash memory.

**Command Default**

No announcement is played.

**Command Modes**

Voice MLPP configuration (config-voice-mlpp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies the G.711 a-law or u-law 8-KHz encoded audio file (.wav or .au format) for the announcement that plays to callers when the call leaves the Cisco Unified CME router on the trunk or when the user places a call to a different domain.

The `mlpp indication` command must be enabled (default) for a phone to play precedence announcements.

This command is not supported by Cisco IOS help. If you type `?`, Cisco IOS help does not display a list of valid entries.

**Examples**

The following example shows that the audio file played for the isolated code announcement is named ica.au located in flash:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# loc2 flash:loc2.au
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>bnea</td>
<td>Specifies the audio file used for the busy station not equipped for preemption announcement.</td>
</tr>
<tr>
<td>upa</td>
<td>Specifies the audio file used for the unauthorized precedence announcement.</td>
</tr>
<tr>
<td>vca</td>
<td>Specifies the audio file used for the vacant code announcement.</td>
</tr>
<tr>
<td>mlpp indication</td>
<td>Enables MLPP indication on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td>mlpp preemption</td>
<td>Enables preemption capability on an SCCP phone or analog FXS port.</td>
</tr>
</tbody>
</table>
location (voice emergency response zone)

To include a location within an emergency response zone, use the `location` command in voice emergency response zone mode. To assign specific priorities to the locations, use the priority tag. To remove the location, use the `no` form of this command.

```
location  location-tag[priority <1-100>]

no  location  location-tag
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Identifier for the emergency response zone location.</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>location-tag</code></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Identifier (1-100) for the priority ranking of locations, 1 being the highest priority.</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>priority 1-100</code></td>
</tr>
</tbody>
</table>

**Command Modes**

Voice emergency response zone configuration (cfg-emrgncy-resp-zone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to create locations within emergency response zones. The tag must be the same as the tag that is defined using the `voice emergency response location` command. This allows routing of 911 calls to different public safety answering points (PSAPs). Priority is optional and allows searching locations in a specified priority order. If there are locations with assigned priorities and locations configured without priorities, the prioritized locations are searched before those without an assigned priority.

**Examples**

The following example shows an assignment of emergency response location (ERLs) to two zones, 10 and 11, to route callers to two different PSAPs. The locations for ERLs in zone 10 are searched in sequential order for a phone address match. The calls from zone 10 have an emergency location identification number (ELIN) from ERLs 8, 9, and 10. The calls from zone 11 have an ELIN from ERLs 2, 3, 4, and 5. The locations for ERLs in zone 11 have priorities assigned and is searched in order of the assigned priority and not the ERL tag number.

```
voice emergency response zone 10
location 8
location 9
location 10
voice emergency response zone 11
location 5 priority 1
location 3 priority 2
location 4 priority 3
location 2 priority 10
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>emergency response callback</td>
<td>Defines a dial peer that is used for 911 callbacks from the PSAP.</td>
</tr>
<tr>
<td>emergency response location</td>
<td>Associates an ERL to either a SIP phone, ephone, or dial peer.</td>
</tr>
<tr>
<td>voice emergency response location</td>
<td>Creates a tag for identifying an ERL for the enhanced 911 service.</td>
</tr>
<tr>
<td>voice emergency response zone</td>
<td>Creates an emergency response zone within which ERLs can be grouped.</td>
</tr>
</tbody>
</table>
**log password**

Effective with Cisco Unified CME 4.0, the `log password` command was replaced by the `xml user` command in telephony-service configuration mode. See the `xml user` command for more information.

For Cisco CME 3.4 and earlier versions, to set a local password for an eXtensible Markup Language (XML) Application Programming Interface (API) query, use the `log password` command in telephony-service configuration mode. To remove the password definition, use the `no` form of this command.

```
log password password-string
no log password password-string
```

**Syntax Description**

<table>
<thead>
<tr>
<th>password-string</th>
<th>Character string that is a password for XML API queries. Maximum length is 28 characters. Longer strings are truncated.</th>
</tr>
</thead>
</table>

**Command Default**

No password is defined.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was replaced by the <code>xml user</code> command.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was replaced by the <code>xml user</code> command.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command is deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The local password is used to authenticate XML API requests on the network management server. If the password is not set, an XML API query fails local authentication.

The password string is stored as plain text. No encryption is supported.

**Examples**

The following example defines a local password for XML API requests:

```
Router(config)# telephony-service
Router(config-telephony)# log password ewvpil
```
log table

To set parameters for the table used to capture phone events used for the eXtensible Markup Language (XML) Application Programming Interface (API), use the **log table** command in telephony-service configuration mode. To reset parameters to their default values, use the **no** form of this command.

```
log table {max-size entries|retain-timer minutes}
no log table {max-size|retain-timer}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>max-size entries</td>
<td>Number of entries in the log table. Range is from 0 to 1000. Default is 150.</td>
</tr>
<tr>
<td>retain-timer minutes</td>
<td>Number of minutes to retain entries in the log table. Range is from 2 to 500. Default is 15.</td>
</tr>
</tbody>
</table>

**Command Default**

Default number of entries in table is 150. default number of minutes to retain entries in table is 15.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Cisco Unified CME captures and time-stamps events, such as phones registering and unregistering and extension status, and stores them in an internal buffer. This command sets the maximum number of events, or entries, that can be stored in the table. One event equals one entry. The **retain-timer** keyword sets the number of minutes that events are kept in the buffer before they are deleted.

The event table can be viewed using the **show fb-its-log** command.

**Examples**

The following example sets the maximum size of the table at 750 events and sets the retention time at 30 minutes:

```
Router(config)# telephony-service
Router(config-telephony)# log table max-size 750
Router(config-telephony)# log table retain-timer 30
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>show fb-its-log</strong></td>
<td>Displays information about the Cisco CME XML API configuration, statistics on XML API queries, and event logs.</td>
</tr>
</tbody>
</table>
logging (voice emergency response settings)

To enable syslog messages to capture emergency call data, use the `logging` command in voice emergency response settings configuration mode. To disable logging, use the `no` form of this command.

```
logging
no logging
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

This command is enabled by default.

**Command Modes**

Voice emergency response settings configuration (cfg-emrgncy-resp-settings)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to enable syslog messages to be announced for every 911 emergency call that is made. The syslog messages can be used by third party applications to send pager or e-mail notifications to an in-house support number. This optional command is enabled by default.

**Examples**

In this example, the ELIN (4085550101) defined in the voice emergency response settings configuration is used if the 911 caller’s IP phones address does not match any of the voice emergency response locations. After the 911 call is placed to the PSAP, the PSAP has 120 minutes to call back 408 555-0101 to reach the 911 caller. If the call history has expired (after 120 minutes), any callback is routed to extension 7500. The outbound 911 calls do not emit a syslog message to the logging facility (for example, a local buffer, console, or remote host).

```
voice emergency response settings
callback 7500
elin 4085550101
expiry 120
no logging
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>callback</code></td>
<td>Default phone number to contact if a 911 callback cannot find the last 911 caller from the ERL.</td>
</tr>
<tr>
<td><code>elin</code></td>
<td>E.164 number used as the default ELIN if no matching ERL to the 911 caller’s IP phone address is found.</td>
</tr>
<tr>
<td><code>expiry</code></td>
<td>Number of minutes a 911 call is associated with an ELIN in case of a callback from the 911 operator.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>------------------------------</td>
<td>------------------------------------------------------------------</td>
</tr>
<tr>
<td>voice emergency response settings</td>
<td>Creates a tag for identifying settings for E911 behavior.</td>
</tr>
</tbody>
</table>
login (telephony-service)

To define the timer for automatically deactivating user login on SCCP phones in a Cisco Unified CME system, use the `login` command in telephony-service configuration mode. To revert to the default values for automatic logout, use the `no` form of this command.

```
login [timeout [minutes]] [clear time]
no login
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>timeout</code></td>
<td>(Optional) Period of phone idleness after which user login is deactivated.</td>
</tr>
<tr>
<td><code>minutes</code></td>
<td>(Optional) Number of minutes for which an IP phone can be idle before the user is logged out automatically. Range: 1 to 1440. Default: 60.</td>
</tr>
<tr>
<td><code>clear</code></td>
<td>(Optional) Time of day after which user login for all IP phones is deactivated. Range: 00:00 to 24:00 on a 24-hour clock. Default: 24:00 (midnight).</td>
</tr>
</tbody>
</table>

### Command Default

User login is deactivated after a phone is idle for 60 minutes. User login for all phones is deactivated at 24:00.

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>Minimum value for the <code>minutes</code> argument was lowered from 5 minutes to 1 minute.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command with the modifications was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command defines the after-hours login timer. Individual users on specified phones can override call blocking by logging in using a personal identification number (PIN). The after-hours login timer deactivates user login on all phones at a specific time and deactivates a login session automatically after a phone is idle for a specified period of time.

The `login` command applies only to IP phones that have soft keys, such as the Cisco Unified IP Phone 7940 and 7940G and the Cisco Unified IP Phone 7960 and 7960G.

For this command to take effect, fast reboot and reregister all phones in Cisco Unified CME by using the `restart all` command in telephony-service configuration mode.

When a Cisco Unified CME router is rebooted, the login status for all phones is reset to the default.

### Examples

The following example sets the after-hours login timer to deactivate logged in phone users automatically after a 2-hour idle time and after 11:30 p.m.

```
Router(config)# telephony-service
```
Router(config-telephony)# login timeout 120 clear 2330

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>after-hour exempt</td>
<td>Specifies that an IP phone does not have any of its outgoing calls blocked even though call blocking has been defined.</td>
</tr>
<tr>
<td>after-hours block pattern</td>
<td>Defines a pattern of digits for blocking outgoing calls from IP phones.</td>
</tr>
<tr>
<td>after-hours date</td>
<td>Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
<tr>
<td>after-hours day</td>
<td>Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
<tr>
<td>pin</td>
<td>Sets a global/individual PIN for phone users to deactivate call blocking during call blocking periods.</td>
</tr>
<tr>
<td>restart (telephony-service)</td>
<td>Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td>show ephone login</td>
<td>Displays the login states of all phones.</td>
</tr>
</tbody>
</table>
**logo (voice register global)**

To specify a file to display on SIP phones, use the `logo` command in voice register global configuration mode. To disable the display of the file, use the `no` form of this command.

```
logo url
```

```
no logo
```

**Syntax Description**

- `url` URL as defined in RFC 2396.

**Command Default**

No file is specified for display on idle phones.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to define the URL for the file to be used by SIP phones connected in Cisco Unified CME. The file that is displayed must be encoded in eXtensible Markup Language (XML) by using the Cisco XML document type definition (DTD). For more information about Cisco DTD formats, see the *Cisco IP Phone Services Application Development Notes*.

After you configure this command, restart the phones by using the `reset` command.

**Examples**

The following example shows how to specify that the file logo.xml should be displayed on SIP phones:

```
Router(config)# voice register global
Router(config-voice-register-global)# logo http://mycompany.com/files/logo.xml
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>reset (voice register pool)</td>
<td>Performs a complete reboot of one phone associated with a Cisco CME router.</td>
</tr>
<tr>
<td>reset (voice register global)</td>
<td>Performs a complete reboot of one or all phones associated with a Cisco CME router.</td>
</tr>
</tbody>
</table>
logout-profile

To enable an IP phone for extension mobility and to apply a default logout profile to the phone, use the `logout-profile` command in ephone configuration mode. To disable extension mobility, use the `no` form of this command.

```
logout-profile  profile-tag
no logout-profile  profile-tag
```

**Syntax Description**

- `profile-tag`: Unique identifier for a default logout profile to be applied. Previously created by using the `voice logout-profile` command in voice logout-profile configuration mode. Range: 1 to maximum number of phones supported by platform.

**Command Default**

IP phone is not enabled for extension mobility.

**Command Modes**

Ephone configuration (config-ephone), Voice Register Pool configuration (voice register pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XW</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command is integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
<tr>
<td>15.1(4)M</td>
<td>Cisco Unified CME 8.6</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command in ephone configuration mode to enable a supported IP phone registered in Cisco Unified CME for extension mobility and to apply a default logout profile to the ephone being configured.

In Cisco Unified CME 4.2, extension mobility is supported only on SCCP IP phones.

In Cisco Unified CME 8.6 extension mobility is supported on SIP phones.

Extension mobility is not supported on non-display IP phones.

Extension mobility is not supported for analog devices.

Before using this command, you must create a logout profile to be applied to this phone by using the `voice logout-profile` command.

You cannot apply more than one logout profile to an ephone. If you attempt to apply a second logout profile to an ephone to which a profile has already been applied, the second profile will overwrite the first logout profile configuration.

**Examples**

The following example shows the ephone configuration for three different Cisco Unified IP phones. All three phones are enabled for extension mobility and share the same logout profile number (1), to be downloaded when these phones boot and when no phone user is logged into these phones:
ephone 1
   mac-address 000D.EDAB.3566
   type 7960
   logout-profile 1
ephone 2
   mac-address 0012.DA8A.C43D
   type 7970
   logout-profile 1
ephone 3
   mac-address 1200.80FC.9B01
   type 7911
   logout-profile 1

The following example shows the ephone configuration for two different Cisco Unified IP phones. Both phones are enabled for extension mobility and share the same logout profile number (22), to be downloaded when these phones boot and when no phone user is logged into these phones:

voice register pool 1
   logout-profile 22
   id mac 0012.0034.0056
   type 7960
voice register pool 2
   logout-profile 22
   id mac 0001.0023.0045
   type 7912

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>reset (voice logout-profile and voice user-profile)</td>
<td>Performs a complete reboot of all IP phones to which a particular logout-profile or user-profile is downloaded.</td>
</tr>
<tr>
<td>voice logout-profile</td>
<td>Enters voice profile configuration mode to configure a default logout profile for extension mobility.</td>
</tr>
</tbody>
</table>
loopback-dn

To create a virtual loopback voice port (loopback-dn) to establish a demarcation point for VoIP calls and supplementary services, use the `loopback-dn` command in ephone-dn configuration mode. To delete a loopback-dn configuration, use the `no` form of this command.

```
loopback-dn  dn-tag  [{forward  number-of-digits|strip  number-of-digits}]  [prefix  prefix-digit-string]  [suffix  suffix-digit-string]  [retry  seconds]  [auto-con]  [codec  {g711alaw|g711ulaw}]
```

Syntax Description

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>dn-tag</code></td>
<td>Unique sequence number that identifies the ephone-dn that is being paired for loopback with the ephone-dn that is currently being configured. The paired ephone-dn must be one that is already defined in the system.</td>
</tr>
<tr>
<td><code>forward number-of-digits</code></td>
<td>(Optional) Number of digits in the original called number to forward to the other ephone-dn in the loopback-dn pair. Range is from 1 to 32 digits. Default is to forward all digits.</td>
</tr>
<tr>
<td><code>strip number-of-digits</code></td>
<td>(Optional) Number of leading digits to be stripped from the original called number before forwarding to the other ephone-dn in the loopback-dn pair. Range is from 1 to 32 digits. Default is not to A-law strip any digits.</td>
</tr>
<tr>
<td><code>prefix prefix-digit-string</code></td>
<td>(Optional) Defines a string of digits to add in front of the forwarded called number. Maximum number of digits in the string is 32. Default is that no prefix is defined.</td>
</tr>
<tr>
<td><code>suffix suffix-digit-string</code></td>
<td>(Optional) Defines a string of digits to add to the end of the forwarded called number. Maximum number of digits in the string is 32. Default is that no suffix is defined. If you add a suffix that starts with the pound character (#), the string must be enclosed in quotation marks.</td>
</tr>
<tr>
<td><code>retry seconds</code></td>
<td>(Optional) Number of seconds to wait before retrying the loopback target when it is busy or unavailable. Range is from 0 to 32767. Default is that retry is disabled and appropriate call-progress tones are passed to the call originator.</td>
</tr>
<tr>
<td><code>auto-con</code></td>
<td>(Optional) Immediately connects the call and provides in-band alerting while waiting for the far-end destination to answer. Default is that automatic connection is disabled.</td>
</tr>
<tr>
<td><code>codec</code></td>
<td>(Optional) Explicitly forces the G.711 A-law or G.711 mu-law voice coding type to be used for calls that pass through the loopback-dn. This overrides the G.711 coding type that is negotiated for the call and provides mu-law to A-law conversion if needed. Default is that Real-Time Transport Protocol (RTP) voice packets are passed through the loopback-dn without considering the G.711 coding type negotiated for the calls.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Coding Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>g711alaw</td>
<td>G.711 A-law, 64000 bits per second, for T1.</td>
</tr>
<tr>
<td>g711ulaw</td>
<td>G.711 mu-law, 64000 bits per second, for E1.</td>
</tr>
</tbody>
</table>

Command Default

All calls are set to forward all digits and not to strip any digits. Prefix is not defined. Suffix is not defined. Retry is disabled. Automatic connection is disabled. RTP voice packets are passed through the loopback-dn without considering the G.711 coding type negotiated for the call.
Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(2)XT3</td>
<td>Cisco ITS 2.0</td>
<td>The suffix keyword was added.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T and the auto-con keyword was added.</td>
</tr>
<tr>
<td>12.2(11)T</td>
<td>Cisco ITS 2.01</td>
<td>The suffix keyword was added.</td>
</tr>
<tr>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1</td>
<td>The strip keyword was added.</td>
</tr>
<tr>
<td>12.2(11)YT2</td>
<td>Cisco ITS 2.1</td>
<td>The codec keyword was added.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

The loopback-dn command is used to configure two ephone-dn virtual voice ports as back-to-back-connected voice-port pairs. A call presented on one side of the loopback-dn pair is reoriginated as a new call on the opposite side of the loopback-dn pair. The forward, strip, prefix, and suffix keywords can be used to manipulate the original called number that is presented to the incoming side of the loopback-dn pair to generate a modified called number to use when reoriginating the call at the opposite side of the loopback-dn pair. For loopback-dn configurations, you must always configure ephone-dn virtual voice ports as cross-coupled pairs.

Note

Use of loopback-dn configurations within a VoIP network should be restricted to resolving critical network interoperability service problems that cannot otherwise be solved. Loopback-dn configurations are intended to be used in VoIP network interworking situations in which the only other alternative would be to make use of back-to-back-connected physical voice ports. Loopback-dn configurations emulate the effect of a back-to-back physical voice-port arrangement without the expense of the physical voice-port hardware. A disadvantage of loopback-dn configurations is that, because digital signal processors (DSPs) are not involved in a loopback-dn arrangement, the configuration does not support interworking or transcoding between calls that use different voice codecs. In many cases, the use of back-to-back physical voice ports that do use DSPs to resolve VoIP network interworking issues is preferred, because it introduces fewer restrictions in terms of supported codecs and call flows. Also, loopback-dns do not support T.38 fax relay.

Note

We recommend that you create the basic ephone-dn configuration for both ephone-dn entries before configuring the loopback-dn option under each ephone-dn. The loopback-dn mechanism should be used only in situations where the voice call parameters for the calls on either side of the loopback-dn use compatible configurations; for example, compatible voice codec and dual tone multifrequency (DTMF) relay parameters. Loopback-dn configurations should be used only for G.711 voice calls.

The loopback-dn arrangement allows an incoming telephone call to be terminated on one side of the loopback-dn port pair and a new pass-through outgoing call to be originated on the other side of the loopback-dn port pair. The loopback-dn port pair normally works with direct cross-coupling of their call states; the alerting call state on the outbound call segment is associated with the ringing state on the inbound call segment.
The loopback-dn mechanism allows for call operations (such as call transfer and call forward) that are invoked for the call segment on one side of the loopback-dn port pair to be isolated from the call segment that is present on the opposite side of the loopback-dn port pair. This approach is useful when the endpoint devices associated with the two different sides have mismatched call-transfer and call-forwarding capabilities. The loopback-dn arrangement allows for call-transfer and call-forward requests to be serviced on one side of the loopback-dn port pair by creating hairpin-routed calls when necessary. The loopback-dn arrangement avoids the propagation of call-transfer and call-forward requests to endpoint devices that do not support these functions.

The `loopback-dn` command provides options for controlling the called-number digits that are passed through from the incoming side to the outgoing side. The available digits can be manipulated with the `forward`, `strip`, `prefix`, and `suffix` keywords.

The `forward` keyword defines the number of digits in the original called number to forward to the other ephone-dn in the loopback-dn pair. The default is set to forward all digits. The `strip` keyword defines the number of leading digits to be stripped from the original called number before forwarding to the other ephone-dn in the loopback-dn pair. The default is set to not strip any digits. The `forward` and `strip` commands are mutually exclusive and can be used with any combination of the `prefix` and `suffix` keywords.

The `prefix` keyword defines a string of digits to add in front of the forwarded number.

The `suffix` keyword is most commonly used to add a terminating “#” (pound-sign) character to the end of the forwarded number to indicate that no more digits should be expected. The pound-sign character indicates to the call-routing mechanism that is processing the forwarded number that the forwarded number is complete. Providing an explicit end-of-number character also avoids a situation in which the call-processing mechanism waits for the interdigit timeout period to expire before routing the call onward using the forwarded number.

The Cisco IOS command-line interface (CLI) requires that arguments with character strings that start with the pound-sign (#) character be enclosed within quotation marks; for example, “#”.

The `retry` keyword is used to suppress a far-end busy indication on the outbound call segment. Instead of returning a busy signal to the call originator (on the incoming call segment), a loopback-dn presents an alerting or ringing tone to the caller and then periodically retries the call to the final far-end destination (on the outgoing call segment). This is not bidirectional. To prevent calls from being routed into the idle outgoing side of the loopback-dn port pair during the idle interval that occurs between successive outgoing call attempts, configure the outgoing side of the loopback-dn without a number so that there is no number to match for the inbound call.

The `auto-con` keyword is used to configure a premature trigger for a connected state for an incoming call segment while the outgoing call segment is still in the alerting state. This setup forces the voice path to open for the incoming call segment and support the generation of in-band call progress tones for busy, alerting, or ringback. The disadvantage of the `auto-con` keyword is premature opening of the voice path during the alerting stage and also triggering of the beginning of billing for the call before the call has been answered by the far end. These disadvantages should be considered carefully before you use the `auto-con` keyword.

The `codec` keyword is used to explicitly select the A-law or mu-law type of G.711 and to provide A-law to mu-law conversion if needed. Setting the codec type on one side of the loopback-dn forces the selection of A-law or mu-law for voice packets that are transmitted from that side of the loopback-dn. To force the A-law or mu-law G.711 codec type for both voice packet directions, set the codec type on both sides of the loopback-dn. Loopback-dn configurations are used only with G.711 calls. Other voice codec types are not supported.

The following example creates a loopback-dn configured with the `forward` and `prefix` keywords:
Router(config)# ephone-dn 7
Router(config-ephone-dn)# loopback-dn 15 forward 5 prefix 41

The following example creates a loopback-dn that appends the pound-sign (#) character to forwarded numbers to indicate the end of the numbers:

Router(config)# ephone-dn 7
Router(config-ephone-dn)# loopback-dn 16 suffix "#"

The following example shows a loopback-dn configuration that pairs ephone-dns 15 and 16. An incoming call (for example, from VoIP) to 4085550101 matches ephone-dn 16. The call is then reoriginated from ephone-dn 15 and sent to extension 50101. Another incoming call (for example, from a local IP phone) to extension 50151 matches ephone-dn 15. It is reoriginated from ephone-dn 16 and sent to 4085550151.

ephone-dn 15
number 5015.
  loopback-dn 16 forward 5 prefix 40855
caller-id local
no huntstop
!
ephone-dn 16
number 4085550101.
  loopback-dn 15 forward 5
caller-id local
no huntstop

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-dn</td>
<td>Enters ephone-dn configuration mode.</td>
</tr>
<tr>
<td>show ephone-dn loopback</td>
<td>Displays information about loopback ephone-dns that have been created in a Cisco CME system.</td>
</tr>
</tbody>
</table>
# lpcor incoming

To associate an incoming call with a logical partitioning class of restriction (LPCOR) resource-group policy, use the `lpcor incoming` command in ephone, ephone-template, trunk group, voice port, voice register pool, voice register template, or voice service configuration mode. To reset to the default, use the `no` form of this command.

```
lpcor incoming lpcor-group
no lpcor incoming
```

**Syntax Description**

| `lpcor-group` | Name of the LPCOR resource group. |

**Command Default**

LPCOR policy is not associated with the incoming call.

**Command Modes**

- Ephone configuration (config-ephone)
- Ephone template configuration (config-ephone-template)
- Trunk group configuration (config-trunk-group)
- Voice port configuration (config-voiceport)
- Voice register pool configuration (config-register-pool)
- Voice register template configuration (config-register-temp)
- Voice service configuration (conf-voi-serv)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

**Analog Phones**

An incoming call to an analog phone is associated with the LPCOR policy specified with this command in the voice port. Otherwise the LPCOR policy specified in the trunk group is used.

**SCCP IP Phones (Local or Remote)**

An incoming call to an SCCP IP phone is associated with the LPCOR policy specified with this command in ephone or ephone template configuration mode. The ephone configuration has precedence over the ephone-template configuration. All directory numbers on the phone share the same LPCOR setting.

**SIP IP Phones (Local or Remote)**

An incoming call to a SIP IP phone is associated with the LPCOR policy specified with this command in voice register pool or voice register template configuration mode. The voice register pool configuration has precedence over the voice register template configuration. All directory numbers on the phone share the same LPCOR setting.

**Note**

This command is not supported for phones configured with the `lpcor type mobility` command.
Phones that share a directory number must be configured with the same LPCOR policy. Different LPCOR settings on shared-line phones are not supported.

**PSTN Trunks**

An incoming call to the PSTN is associated with the LPCOR policy specified with this command in the voice port. Otherwise, the LPCOR policy specified in the trunk group is used. The voice port configuration takes precedence.

**VoIP Trunks (H.323 or SIP)**

An incoming call to a VoIP trunk is associated with the LPCOR policy specified with this command in voice service configuration mode if the remote IP address is not found in the IP trunk subnet table created with the `voice lpcor ip-trunk subnet incoming` command.

### Examples

The following example shows the command used in different configuration modes:

```
voice service voip
  lpcor incoming voip_group1
  trunk group analog1
    lpcor incoming analog_group1
    lpcor outgoing analog_group1
  voice-port 1/1/0
    lpcor incoming vp_group1
    lpcor outgoing vp_group1
  voice register pool 3
    lpcor type remote
    lpcor incoming sip_group3
    lpcor outgoing sip_group3
    id mac 001E.BE8F.96C0
    type 7940
    number 1 dn 3
  ephone 2
    mac-address 001C.821C.ED23
    type 7960
    button 1:2
    lpcor type remote
    lpcor incoming ephone_group2
    lpcor outgoing ephone_group2
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>lpcor outgoing</code></td>
<td>Associates an outgoing call with a LPCOR resource-group policy.</td>
</tr>
<tr>
<td><code>lpcor type</code></td>
<td>Specifies the LPCOR type for an IP phone.</td>
</tr>
<tr>
<td><code>voice lpcor ip-trunk subnet incoming</code></td>
<td>Creates a LPCOR IP-trunk subnet table for incoming calls from a VoIP trunk.</td>
</tr>
<tr>
<td><code>voice lpcor policy</code></td>
<td>Creates a LPCOR policy for a resource group.</td>
</tr>
</tbody>
</table>
lpcor outgoing

To associate an outgoing call with a logical partitioning class of restriction (LPCOR) resource-group policy, use the lpcor outgoing command in dial peer, ephone, ephone template, trunk group, voice port, voice register pool, or voice register template configuration mode. To reset to the default, use the no form of this command.

```
lpcor outgoing lpcor-group
no lpcor outgoing
```

**Syntax Description**

- **lpcor-group** Name of the LPCOR resource group.

**Command Default**

LPCOR policy is not associated with the outgoing call.

**Command Modes**

- Dial peer configuration (config-dial-peer)
- Ephone configuration (config-ephone)
- Ephone template configuration (config-ephone-template)
- Trunk group configuration (config-trunk-group)
- Voice port configuration (config-voiceport)
- Voice register pool configuration (config-register-pool)
- Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

**Analog Phones**

An outgoing call from an analog phone is associated with the LPCOR policy specified with this command in the voice port. Otherwise the LPCOR policy specified in the trunk group is used.

**SCCP IP Phones (Local or Remote)**

An outgoing call from an SCCP IP phone is associated with the LPCOR policy specified with this command in ephone configuration or ephone template configuration mode. The ephone configuration has precedence over the ephone-template configuration. All directory numbers on the phone share the same LPCOR setting.

**SIP IP Phones (Local or Remote)**

An outgoing call from a SIP IP phone is associated with the LPCOR policy specified with this command in voice register pool or voice register template configuration mode. The voice register pool configuration has precedence over the voice register template configuration. All directory numbers on the phone share the same LPCOR setting.

**Note**

This command is not supported for phones configured with the lpcor type mobility command.

- Phones that share a directory number must be configured with the same LPCOR policy. Different LPCOR settings on shared-line phones are not supported.
PSTN Trunks
An outgoing call from the PSTN uses the LPCOR policy specified with this command in the voice port if the outbound dial peer is configured with the `port` command. Otherwise the outgoing call uses the LPCOR policy specified with this command in the trunk group if the outbound dial peer is configured with the `trunkgroup` command.

VoIP Trunks (H.323 or SIP)
An outgoing VoIP call uses the LPCOR policy specified with this command in the outbound dial peer. Otherwise the outgoing call uses the default LPCOR policy.

Examples
The following example shows the command used in different configuration modes:

```
trunk group analog1
  lpcor incoming analog_group1
  lpcor outgoing analog_group1
!
voice-port 1/1/0
  lpcor incoming vp_group1
  lpcor outgoing vp_group1
!
dial-peer voice 2 voip
  destination-pattern 2...
  lpcor outgoing voip_group2
  session protocol sipv2
  session target ipv4:192.168.97.1
!
voice register pool 3
  lpcor type remote
  lpcor incoming sip_group3
  lpcor outgoing sip_group3
  id mac 001E.BE8F.96C0
  number 1 dn 3
!
ephone 2
  mac-address 001C.821C.ED23
  type 7960
  button 1:2
  lpcor type remote
  lpcor incoming ephone_group2
  lpcor outgoing ephone_group2
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>lpcor incoming</td>
<td>Associates an incoming call with a LPCOR resource-group policy.</td>
</tr>
<tr>
<td>lpcor type</td>
<td>Specifies the LPCOR type for an IP phone.</td>
</tr>
<tr>
<td>port (dial-peer)</td>
<td>Associates a dial peer with a voice port.</td>
</tr>
<tr>
<td>trunkgroup</td>
<td>Associates a dial peer with a trunk group.</td>
</tr>
<tr>
<td>voice lpcor policy</td>
<td>Creates a LPCOR policy for a resource group.</td>
</tr>
</tbody>
</table>
**lpcor type**

To specify the logical partitioning class of restriction (LPCOR) type for an IP phone, use the `lpcor type` command in ephone, ephone-template, voice register pool, or voice register template configuration mode. To reset to the default, use the `no` form of this command.

```
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>local</code></td>
<td>IP phone always registers to Cisco Unified CME through the LAN.</td>
</tr>
<tr>
<td><code>mobile</code></td>
<td>IP phone can register to Cisco Unified CME through the LAN or WAN.</td>
</tr>
<tr>
<td><code>remote</code></td>
<td>IP phone always registers to Cisco Unified CME through the WAN.</td>
</tr>
</tbody>
</table>

**Command Default**

LPCOR feature is disabled for the IP phone.

**Command Modes**

- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)
- Voice register pool configuration (config-register-pool)
- Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

According to the Telecom Regulatory Authority of India (TRAI) requirements, an IP phone can accept both PSTN and VoIP calls if it is locally registered to Cisco Unified CME through the LAN. Select the `local` keyword for this type of phone.

If an IP phone is registered remotely to Cisco Unified CME through the WAN, PSTN calls must be blocked from that remote IP phone. Select the `remote` keyword for this type of phone.

A static LPCOR policy is applied to an IP phone if the phone registers to Cisco Unified CME from the same region (local or remote) permanently.

If an IP phone moves between the local and remote regions, such as an Extension Mobility phone, Cisco IP Communicator softphone, or remote teleworker, select the `mobile` keyword. The LPCOR policy is assigned dynamically based on the phone’s currently registered IP address.

If you use a phone template to apply a command to a phone and you also use the same command in the phone configuration of the same phone, the value in phone configuration has priority.

**Examples**

The following example shows that SCCP IP phone 2 is set to the remote LPCOR type:

```
ephone 2
  mac-address 001C.821C.ED23
  type 7960
  button 1:2
```
lpcor type remote
lpcor incoming ephone_group2
lpcor outgoing ephone_group2

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>lpcor incoming</td>
<td>Associates a LPCOR resource-group policy with an incoming call.</td>
</tr>
<tr>
<td></td>
<td>lpcor outgoing</td>
<td>Associates a LPCOR resource-group policy with an outgoing call.</td>
</tr>
<tr>
<td></td>
<td>voice lpcor policy</td>
<td>Creates a LPCOR policy for a resource group.</td>
</tr>
</tbody>
</table>
lpcor type
Cisco Unified CME Commands: M

- mac-address (ephone), on page 573
- mac-address (voice-gateway), on page 575
- mailbox-selection (dial-peer), on page 576
- mailbox-selection (ephone-dn), on page 578
- max-calls-per-button, on page 579
- max-conferences, on page 581
- max-dn, on page 583
- max-dn (voice register global), on page 585
- max-ephones, on page 587
- max-idle-time, on page 589
- maximum bit-rate (video), on page 590
- max-pool (voice register global), on page 591
- max-presentation, on page 593
- max-redirect, on page 595
- max-subscription, on page 596
- max-timeout, on page 597
- media, on page 598
- members logout, on page 602
- members logout (voice hunt-group), on page 603
- missed-calls, on page 604
- mlpp indication, on page 605
- mlpp max-precedence, on page 607
- mlpp preemption, on page 609
- mlpp service-domain, on page 611
- mobility (ephone-dn), on page 613
- mobility (voice register dn), on page 614
- mode cme, on page 615
- moh (ephone-dn), on page 617
- moh (telephony-service), on page 620
- moh (voice moh-group), on page 622
- moh-file-buffer, on page 623
- moh-group (ephone-dn), on page 625
- mtp, on page 626
- mtu (vpn-profile), on page 628
- multicast moh, on page 629
- mwi (ephone-dn and ephone-dn-template), on page 631
- mwi (voice register dn), on page 633
- mwi expires, on page 634
- mwi prefix, on page 635
- mwi qsig, on page 637
- mwi reg-e164, on page 639
- mwi relay, on page 640
- mwi sip, on page 641
- mwi sip-server, on page 643
- mwi stutter (voice register global), on page 645
- mwi-line, on page 646
- mwi-type, on page 648
mac-address (ephone)

To associate the MAC address of a Cisco IP phone with an ephone configuration in a Cisco CallManager Express (Cisco CME) system, use the `mac-address` command in ephone configuration mode. To disassociate the MAC address from an ephone configuration, use the `no` form of this command.

```
mac-address [mac-address] [reserved]
no mac-address
```

**Syntax Description**

<table>
<thead>
<tr>
<th>mac-address</th>
<th>Identifying MAC address of an IP phone, which is found on a sticker located on the bottom of the phone.</th>
</tr>
</thead>
<tbody>
<tr>
<td>reserved</td>
<td>Identifies the reserved MAC address of the phone.</td>
</tr>
</tbody>
</table>

**Command Default**

There are no default behavior or values for this command.

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>The <code>mac-address</code> argument was made optional to enable automatic MAC address assignment after registration of phones.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to specify the MAC address of a specific Cisco IP phone in order to physically identify the Cisco IP phone in a Cisco CME configuration. The MAC address of each Cisco IP phone is printed on a sticker that is placed on the bottom of the phone.

If you choose to register phones before configuring them, the `mac-address` command can be used during configuration without entering the `mac-address` argument. The Cisco CME system detects MAC addresses and automatically populates phone configurations with their corresponding MAC addresses and phone types. This capability is not supported for voice-mail ports and is supported only by Cisco CME 3.0 and later versions. To use this capability, enable Cisco CME by using the following commands: `max-ephones`, `max-dn`, `create cnf-files`, and `ip source-address`. After these commands have been used, phones can start to register. Then, when you are configuring a registered ephone and you use the `mac-address` command with no argument, the MAC address of the phone is automatically read into the configuration. The equivalent functionality is available through the Cisco CME graphic user interface (GUI).

If you choose to configure phones before registering them, the MAC address for each ephone must be entered during configuration.

**Examples**

The following example associates the MAC address CFBA.321B.96FA with the IP phone that has phone-tag 22:
Router(config)# ephone 22
Router(config-ephone)# mac-address CFBA.321B.96FA

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>create cnf-files</td>
<td>Builds the XML configuration files that are required for IP phones used with</td>
</tr>
<tr>
<td></td>
<td>Cisco IOS Telephony Services V2.1, Cisco CallManager Express 3.0, or later</td>
</tr>
<tr>
<td></td>
<td>versions.</td>
</tr>
<tr>
<td>ip source-address</td>
<td>Identifies the IP address and port through which IP phones communicate with</td>
</tr>
<tr>
<td></td>
<td>a Cisco CME router.</td>
</tr>
<tr>
<td>max-dn</td>
<td>Sets the maximum number of ephone-dns to be supported by a Cisco CME router.</td>
</tr>
<tr>
<td>max-ephones</td>
<td>Sets the maximum number of ephones to be supported by a Cisco CME router.</td>
</tr>
<tr>
<td>show ephone registered</td>
<td>Displays status and information for registered IP phones.</td>
</tr>
</tbody>
</table>
mac-address (voice-gateway)

To define the MAC address of the voice gateway to autoconfigure, use the `mac-address` command in voice-gateway configuration mode. To remove the MAC address from the configuration, use the `no` form of this command.

```
mac-address  mac-address
no  mac-address
```

**Syntax Description**

| `mac-address` | MAC address of the voice gateway. |

**Command Default**

No MAC address is defined for the voice gateway to be autoconfigured.

**Command Modes**

Voice-gateway configuration (config-voice-gateway)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command defines the MAC address of the Cisco voice gateway that downloads its XML configuration file from Cisco Unified CME using the Autoconfiguration feature.

**Examples**

The following example associates the MAC address 001F.A30F.8331 for the Cisco VG224 voice gateway associated with tag 1:

```
voice-gateway system 1
network-locale FR
type VG224
mac-address 001F.A30F.8331
voice-port 0-23
create cnf-files
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>type (voice-gateway)</code></td>
<td>Defines the type of voice gateway to autoconfigure in Cisco Unified CME.</td>
</tr>
<tr>
<td><code>voice-port (voice-gateway)</code></td>
<td>Identifies the analog ports on the voice gateway that register to Cisco Unified CME.</td>
</tr>
</tbody>
</table>
mailbox-selection (dial-peer)

To set a policy for selecting a mailbox for calls from a Cisco Unified CME system that are diverted before being sent to a Cisco Unity Express or PBX voice-mail pilot number, use the `mailbox-selection` command in dial-peer configuration mode. To return to the default, use the `no` form of this command.

```
mailbox-selection  {last-redirect-num|orig-called-num}
no  mailbox-selection
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>last-redirect-num</td>
<td>(PBX voice mail only) The mailbox to which the call will be sent is the number that diverted the call to the voice-mail pilot number (the last number to divert the call).</td>
</tr>
<tr>
<td>orig-called-num</td>
<td>(Cisco Unity Express only) The mailbox to which the call will be sent is the number that was originally dialed before the call was diverted.</td>
</tr>
</tbody>
</table>

**Command Default**

Cisco Unity Express uses the last number to which the call was diverted before it was sent to voice mail as the mailbox number. Some legacy PBX systems use the originally called number as the mailbox number.

**Command Modes**

Dial-peer configuration (config-dial-peer)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

When Cisco Unified CME diverts a call, it captures the reroute information which will be used to compose a reroute request. A dial-peer match will be performed against the diverted-to number. If this is the voice mail pilot number and the `mailbox-selection` command has been used to install a policy, the reroute information will be amended as directed by the command. The originator will pick up the modified reroute request, build the diversion information and include it in the new diverted call to the voice-mail pilot number.

This command should be used on the outbound dial peer for the pilot number of the voice-mail system.

This command might not work properly in certain network topologies, including the following cases:

- When the last redirecting endpoint is not hosted on Cisco Unified CME. This rarely occurs with a PBX.
- When a call is forwarded across several SIP trunks. Multiple SIP Diversion Headers (stacking hierarchy) are not supported in Cisco IOS software.
- When a call is forwarded across non Cisco voice gateways that do not support the optional H450.3 originalCalledNr field.

**Examples**

The following example shows how to set a policy to select the mailbox of the originally called number when a call is diverted to a Cisco Unity Express or PBX voice-mail system with the pilot number 7000.

```
dial-peer voice 7000 voip
destination-pattern 7000
session target ipv4:10.3.34.211
codec g711ulaw
```
no vad
mailbox-selection orig-called-num
mailbox-selection (ephone-dn)

To set a policy for selecting a mailbox for calls that are diverted before being sent to a Cisco Unity voice-mail pilot number, use the `mailbox-selection` command in ephone-dn configuration mode. To return to the default, use the `no` form of this command.

```
mailbox-selection last-redirect-num
no mailbox-selection
```

**Syntax Description**

| last-redirect-num | The mailbox to which the call will be sent is the last number to divert the call. |

**Command Default**

Cisco Unity uses the originally called number as the mailbox number.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command sets the policy for selecting a mailbox for diverted calls.

This command is used on the ephone-dn associated with the voice-mail pilot number.

This command can only be used with SCCP phones.

This command might not work properly in certain network topologies, including the following cases:

- When the last redirecting endpoint is not hosted on Cisco Unified CME. This may rarely occur with a PBX.
- When a call is forwarded across several SIP trunks. Multiple SIP Diversion Headers (stacking hierarchy) are not supported in Cisco IOS software.
- When a call is forwarded across non Cisco voice gateways that do not support the optional H450.3 originalCalledNr field.

**Examples**

The following example sets a policy to select the mailbox of the last redirecting number when a call is diverted to a Cisco Unity voice-mail system with the pilot number 8000.

```
ephone-dn 2583
number 8000
mailbox-selection last-redirect-num
```
max-calls-per-button

To set the maximum number of calls allowed on an octo-line directory number on an SCCP phone, use the `max-calls-per-button` command in ephone or ephone-template configuration mode. To reset to the default, use the `no` form of this command.

```
max-calls-per-button number-of-calls
no max-calls-per-button
```

### Syntax Description

### Command Default
- Maximum number of calls allowed on an octo-line is 8.

### Command Modes
- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command limits the maximum number of calls, both incoming and outgoing, that can be active on each octo-line directory number on an SCCP phone. This command applies to all octo-line directory numbers on the phone.

This command must be set to a value that is more than or equal to the value set with the `busy-trigger-per-button` command.

For phones that do not support octo-line directory numbers such as the Cisco Unified IP Phone 7902, 7920, or 7931, and analog phones connected to the Cisco VG224 or Cisco ATA, we recommend that you set the `max-calls-per-button` command to 2. Otherwise, after the phone type is identified with either the `type` command or during phone registration, this command is automatically set to 2.

If you use an ephone template to apply a command to an ephone and you also use the same command in ephone configuration mode for the same ephone, the value that you set in ephone configuration mode has priority.

### Examples

The following example sets the maximum calls allowed on octo-lines to 4 on ephone 1.

```
Router(config)#
ephone 1
Router(config-ephone)# max-calls-per-button 4
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>busy-trigger-per-button</code></td>
<td>Sets the maximum number of incoming calls allowed on an octo-line directory number before it triggers Call Forward Busy on the phone.</td>
</tr>
</tbody>
</table>
### Command Reference

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-dn</td>
<td>Configures a directory number for SCCP phones.</td>
</tr>
<tr>
<td>type</td>
<td>Assigns a phone type to an SCCP phone.</td>
</tr>
</tbody>
</table>
max-conferences

To set the maximum number of three-party conferences that are supported simultaneously by the Cisco CallManager Express (Cisco CME) router, use the `max-conferences` command in telephony-service configuration mode. To reset this number to the default, use the `no` form of this command.

```
max-conferences [max-conference-number] [gain -6 | 0 | 3 | 6]
no max-conferences
```

**Syntax Description**

| **max-conference number** | Maximum number of three-party conferences that are supported simultaneously by the router. This number is platform-dependent, and the default is half the maximum for each platform. The following are the maximum values for this argument:
| - Cisco 1700 series, Cisco 2600 series, Cisco 2801—8
| - Cisco 2811, Cisco 2821, Cisco 2851, Cisco 3600 series, Cisco 3700 series—16
| - Cisco 3800 series—24 (requires Cisco IOS Release 12.3(11)XL or higher)

**Note** Each individual Cisco IP phone can host a maximum of one conference at a time. You cannot create a second conference on the phone if you already have an existing conference on hold.

| **gain** | (Optional) Increases the sound volume of VoIP and public switched telephony network (PSTN) parties joining a conference call. The allowable decibel units are -6 db, 0 db, 3 db, and 6 db. The default is -6 db.

**Command Default**

Default is half the maximum number of simultaneous three-party conferences for each platform.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th><strong>Cisco IOS Release</strong></th>
<th><strong>Cisco Product</strong></th>
<th><strong>Modification</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.3(11)XL1</td>
<td>Cisco CME 3.2.1</td>
<td>The <code>gain</code> keyword was added.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command supports three-party conferences for local and on-net calls only when all conference participants are using the G.711 codec. Conversion between G.711 mu-law and A-law is supported. Mixing of the media streams is supported by the Cisco IOS processor. The maximum number of simultaneous conferences is limited to the platform-specific maximums.

The `gain` keyword’s functionality is applied to inbound audio packets, so conference participants can more clearly hear a remote PSTN or VoIP caller joining their call. Note that this functionality cannot discriminate between a remote VoIP/foreign exchange office (FXO) source, which requires a volume gain, and a remote VoIP/IP phone, which does not require a volume gain and may therefore incur some sound distortions.
The following example sets the maximum number of conferences for a Cisco IP phone to 4 and configures a gain of 6 db for inbound audio packets from remote PSTN or VoIP calls joining a conference:

```
Router(config)# telephony-service
Router(config-telephony)# max-conferences 4 gain 6
```
max-dn

To set the maximum number of extensions (ephone-dns) to be supported by a Cisco Unified CME router, use the `max-dn` command in telephony-service configuration mode. To reset this number to the default value, use the `no` form of this command.

```
max-dn  max-directory-numbers  [ preference preference-order]  [ no-reg  { primary|both }]  
no  max-dn
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>maxdirectorynumbers</code></td>
<td>Maximum number of extensions (ephone-dns) to allow in the Cisco CME system.</td>
</tr>
<tr>
<td><code>preference preference-order</code></td>
<td>(Optional) Sets a preference value for the primary number of an ephone-dn.</td>
</tr>
<tr>
<td><code>no-reg</code></td>
<td>(Optional) Globally disables ephone registration with an H.323 gatekeeper or</td>
</tr>
<tr>
<td><code>primary</code></td>
<td>SIP proxy.</td>
</tr>
<tr>
<td><code>both</code></td>
<td>Primary ephone-dn numbers only.</td>
</tr>
<tr>
<td><code>both</code></td>
<td>Both primary and secondary ephone-dn numbers.</td>
</tr>
</tbody>
</table>

**Command Default**

The default is `0`.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified 4.0</td>
<td>The <code>preference</code>, <code>no-reg</code>, <code>primary</code>, and <code>both</code> keywords were introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified 4.0</td>
<td>The <code>preference</code>, <code>no-reg</code>, <code>primary</code>, and <code>both</code> keywords were integrated</td>
</tr>
<tr>
<td></td>
<td></td>
<td>into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `max-dn` command limits the number of extensions (ephone-dns) available in a Cisco Unified CME system. The maximum number of ephone-dns that you can create depends on the software version, router platform, and amount of memory that you have installed. Type `?` to display range.

The `max-ephones` command similarly limits the number of IP phones in a Cisco Unified CME system.
You can increase the number of allowable extensions to the maximum; but after the maximum allowable number is configured, you cannot reduce the limit without rebooting the router.

If registration with an H.323 gatekeeper or SIP proxy is enabled globally (the default), you can override the setting per extension by using the `no-reg` keyword in the `number` command for individual ephone-dn.

After using this command, you can provision individual extensions using the Cisco Unified CME graphic user interface (GUI) or the router CLI in ephone-dn configuration mode.

### Examples

The following example sets the maximum number of extensions (ephone-dns) to 12:

```
Router(config)# telephony-service
Router(config-telephony)# max-dn 12
```

The following example sets the maximum number of extensions to 150 and specifies that the primary number of each extension should receive a dial-peer preference order of 1:

```
Router(config)# telephony-service
Router(config-telephony)# max-dn 150 preference 1
```

The following example sets the maximum number of extensions to 200 and specifies that they should not register both primary and secondary numbers with the H.323 gatekeeper:

```
Router(config)# telephony-service
Router(config-telephony)# max-dn 200 no-reg both
```

The following example sets the maximum number of extensions to 200 and specifies that ephone-dn 36 should not register its primary number with the gatekeeper:

```
Router(config)# telephony-service
Router(config-telephony)# max-dn 200
Router(config-telephony)# exit
Router(config)# ephone-dn 36
Router(config-ephone-dn)# number 75373 no-reg primary
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-dn</td>
<td>Enters ephone-dn configuration mode.</td>
</tr>
<tr>
<td>max-ephones</td>
<td>Sets the maximum number of phones supported by the router.</td>
</tr>
<tr>
<td>number</td>
<td>Associates a telephone or extension number with an ephone-dn.</td>
</tr>
</tbody>
</table>
max-dn (voice register global)

To set the maximum number of SIP phone directory numbers (extensions) that are supported by a Cisco router, use the `max-dn` command in voice register global configuration mode. To reset to the default, use the `no` form of this command.

```
max-dn max-directory-numbers
no max-dn
```

### Syntax Description

<table>
<thead>
<tr>
<th>maxdirectorynumbers</th>
<th>Maximum number of extensions (ephone-dns) supported by the Cisco router. The maximum number is version and platform dependent; type <code>?</code> to display range.</th>
</tr>
</thead>
<tbody>
<tr>
<td>• In Cisco CME 3.4 to Cisco Unified CME 7.0 and in Cisco SIP SRST 3.4 to Cisco Unified SIP SRST 7.0: Default is maximum number supported by platform.</td>
<td></td>
</tr>
<tr>
<td>• In Cisco Unified CME 7.1 and Cisco Unified SIP SRST 7.1 and later versions: Default is 0.</td>
<td></td>
</tr>
</tbody>
</table>

### Command Default

Before Cisco Unified CME 7.1 and Cisco Unified SIP SRST 7.1, default is maximum number supported by platform.

In Cisco Unified CME 7.1 and Cisco Unified SIP SRST 7.1 and later versions, default is 0.

### Command Modes

Voice register global configuration (`config-register-global`) mode.

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1 Cisco Unified SIP SRST 7.1</td>
<td>The default value was changed to 0.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1 Cisco Unified SIP SRST 7.1</td>
<td>This command was integrated into Cisco IOS release 12.4(24)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command limits the number of SIP phone directory numbers (extensions) available in a Cisco Unified CME system. The `max-dn` command is platform specific. It defines the limit for the `voice register dn` command. The `max-pool` command similarly limits the number of SIP phones in a Cisco CME system.

You can increase the number of allowable extensions to the maximum; but after the maximum allowable number is configured, you cannot reduce the limit without rebooting the router. You cannot reduce the number of allowable extensions without removing the already-configured directory numbers with dn-tags that have a higher number than the maximum number to be configured.

**Note**

This command can also be used for Cisco Unified SIP SRST.

### Examples

The following example shows how to set the maximum number of directory numbers to 48:
Router(config)# voice register global
Router(config-register-global)# max-dn 48

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register dn</td>
<td>Enters voice register dn configuration mode to define an extension for a SIP phone line.</td>
</tr>
<tr>
<td>max-pool (voice register global)</td>
<td>Sets the maximum number of SIP voice register pools that are supported in a Cisco SIP SRST or Cisco CME environment.</td>
</tr>
</tbody>
</table>
max-ephones

To set the maximum number of Cisco IP phones to be supported by a Cisco CallManager Express (Cisco CME) router, use the `max-ephones` command in telephony-service configuration mode. To reset this number to the default value, use the `no` form of this command.

```
max-ephones max-phones
no max-ephones
```

### Syntax Description

| `maxphones` | Maximum number of phones supported by the Cisco CME router. The maximum number is version- and platform-dependent; refer to Cisco IOS command-line interface (CLI) help. Default is 0. |

### Command Default

Default is 0.

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was modified to set the maximum number of phones that can register to Cisco Unified CME.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

The `max-ephones` command limits the number of Cisco IP phones supported on the router. The maximum number you can set is platform- and version-dependent. Use CLI help to determine the maximum number of ephones you can set, as shown in this example:

```
Router(config-telephony)# max-ephones ?
<1-48> Maximum phones to support
```

The `max-dn` command similarly limits the number of extensions (ephone-dns) in a Cisco CME system.

**Note**

You can increase the number of phones; but after the maximum allowable number is configured, you cannot reduce the limit of the Cisco IP phones without rebooting the router.

After using this command, configure phones by using the Cisco CME graphic user interface (GUI) or the router CLI in ephone configuration mode.

### Examples

The following example sets the maximum number of Cisco IP phones in a Cisco CME system to 24:
Router(config)# telephony-service
Router(config-telephony)# max-ephones 24

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>ephone</td>
<td>Enters ephone configuration mode.</td>
</tr>
<tr>
<td></td>
<td>max-dn</td>
<td>Sets the maximum number of extensions (ephone-dns) that can be supported by the router.</td>
</tr>
</tbody>
</table>
max-idle-time

To create an idle-duration timer for automatically logging out an Extension Mobility user, use the **max-idle-time** command in voice user-profile configuration mode. To remove the timer, use the **no** form of this command.

```
max-idle-time minutes
no max-idle-time
```

**Syntax Description**
- **minutes**: Maximum number of minutes an Extension Mobility phone is idle after which the logged-in user is logged out from Extension Mobility. Range: 1 to 9999.

**Command Default**
No timer is created.

**Command Modes**
Voice user-profile configuration (config-user-profile)

**Command History**
- **Release**: 12.4(15)XZ Cisco Unified CME 4.3
- **Modification**: This command was introduced.
- **Release**: 12.4(20)T Cisco Unified CME 7.0
- **Modification**: This command was integrated into Cisco IOS Release 12.4(20)T.

**Usage Guidelines**
This command creates an idle-duration timer for automatically logging a user out from Extension Mobility. The timer monitors the phone and if the specified maximum idle time is exceeded, the EM manager logs out the user. Typically this command is used to log out users who fail to manually log out of Extension Mobility before leaving a phone.

The call history record is automatically cleared when a user logs out from an Extension Mobility phone. To disable Automatic Clear Call History on all Extension Mobility phones, use the **keep call-history** command in telephony-service configuration mode.

After creating or modifying a profile, use the **reset** command in voice user-profile configuration mode to reset all phones on which this profile is downloaded to propagate the modifications.

**Examples**
The following example shows how to create a 30-minute idle-duration timer in user profile 1:

```
Router(config)# voice user-profile 1
Router(config-user-profile)# max-idle-time 30
Router(config-user-profile)# reset
```

**Related Commands**
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>keep call-history</strong></td>
<td>Disables Automatic Clear Call History for Extension Mobility in Cisco Unified CME.</td>
</tr>
<tr>
<td><strong>reset (voice logout-profile and voice user-profile)</strong></td>
<td>Performs a complete reboot of all IP phones on which a particular logout profile or user profile is downloaded.</td>
</tr>
</tbody>
</table>
maximum bit-rate (video)

To modify the maximum IP phone video bandwidth in Cisco Unified CME, use the `maximum bit-rate` command in video configuration mode. To restore the default maximum bit-rate, use the `no` form of this command.

```
maximum bit-rate value
no maximum bit-rate
```

**Syntax Description**

| value | Video bandwidth in kb/s Range is 0 to 10000000. Default value is 10000000. |

**Command Default**

Maximum bit-rate of video bandwidth is 1,000,000 kb/s.

**Command Modes**

Video configuration (config-tele-video)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to modify the default value of the maximum video bandwidth for all video-capable phones associated with a Cisco Unified CME router. Default value is 1,000,000 kb/s.

**Examples**

The following example sets a maximum bit-rate of 256 kb/s.

```
Router(config)#
telephony-service
Router(config-telephony)# video
Router(config-tele-video)# maximum bit-rate 256
```
max-pool (voice register global)

To set the maximum number of Session Initiation Protocol (SIP) voice register pools that are supported in Cisco Unified SIP SRST or Cisco Unified CME, use the `max-pool` command in voice register global configuration mode. To reset the maximum number to the default, use the `no` form of this command.

```
max-pool max-voice-register-pools
no max-pool
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>max-voice-register-pools</code></td>
<td>Maximum number of SIP voice register pools supported by the Cisco router. The upper limit of voice register pools is version- and platform-dependent; type <code>?</code> for range.</td>
</tr>
<tr>
<td></td>
<td>• In Cisco CME 3.4 to Cisco Unified CME 7.0 and in Cisco SIP SRST 3.4 to Cisco Unified SIP SRST 7.0: Default is maximum number supported by platform.</td>
</tr>
<tr>
<td></td>
<td>• In Cisco Unified CME 7.1 and Cisco Unified SIP SRST 7.1 and later versions: Default is 0.</td>
</tr>
</tbody>
</table>

**Command Default**

Before Cisco Unified CME 7.1 and Cisco Unified SIP SRST 7.1, default is maximum number supported by platform. In Cisco Unified CME 7.1 and Cisco Unified SIP SRST 7.1 and later versions, default is 0.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1 Cisco Unified SIP SRST 7.1</td>
<td>The default value was changed to 0.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1 Cisco Unified SIP SRST 7.1</td>
<td>This command was integrated into Cisco IOS release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command limits the number of SIP phones supported by Cisco Unified CME. The `max-pool` command is platform specific and defines the limit for the `voice register pool` command.

The `max-dn` command similarly limits the number of directory numbers (extensions) in Cisco Unified CME.

You can increase the number of phones; but after the maximum allowable number is configured, you cannot reduce the limit of the SIP phones without rebooting the router.

**Note**

This command can also be used for Cisco Unified SIP SRST.

**Examples**

The following example shows how to set the maximum number of Cisco SIP IP phones in Cisco Unified SIP SRST or Cisco Unified CME to 24:

```
max-pool max-voice-register-pools 24
```
Router(config)# voice register global
Router(config-register-global)# max-pool 24

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>max-dn (voice register global)</td>
<td>Set the maximum number of SIP phone directory numbers (extensions) that are supported by a Cisco Unified CME router.</td>
</tr>
</tbody>
</table>
max-presentation

To set the number of call presentation lines supported by a phone type, use the max-presentation command in ephone-type configuration mode. To reset to the default, use the no form of this command.

```
max-presentation number
no max-presentation
```

**Syntax Description**

| number | Number of presentation lines. Range: 1 to 100. Default: 0. See the table for the number of presentation lines supported by each phone type. |

**Command Default**
No display lines are supported by the phone type.

**Command Modes**
Ephone-type configuration (config-ephone-type)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3 Cisco Unified SRST 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command defines the number of presentation lines that are supported for the type of phone being added with an ephone-type template.

**Table 11: Supported Values for Ephone-Type Commands**

<table>
<thead>
<tr>
<th>Supported Device</th>
<th>device-id</th>
<th>device-type</th>
<th>num-buttons</th>
<th>max-presentation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified IP Phone 6901</td>
<td>547</td>
<td>6901</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 6911</td>
<td>548</td>
<td>6911</td>
<td>1</td>
<td>10</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7915 Expansion Module with 12 buttons</td>
<td>227</td>
<td>7915</td>
<td>12</td>
<td>0 (default)</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7915 Expansion Module with 24 buttons</td>
<td>228</td>
<td>7915</td>
<td>24</td>
<td>0</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7916 Expansion Module with 12 buttons</td>
<td>229</td>
<td>7916</td>
<td>12</td>
<td>0</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7916 Expansion Module with 24 buttons</td>
<td>230</td>
<td>7916</td>
<td>24</td>
<td>0</td>
</tr>
<tr>
<td>Cisco Unified Wireless IP Phone 7925</td>
<td>484</td>
<td>7925</td>
<td>6</td>
<td>4</td>
</tr>
<tr>
<td>Cisco Unified IP Conference Station 7937G</td>
<td>431</td>
<td>7937</td>
<td>1</td>
<td>6</td>
</tr>
<tr>
<td>Nokia E61</td>
<td>376</td>
<td>E61</td>
<td>1</td>
<td>1</td>
</tr>
</tbody>
</table>
Examples

The following example shows that 1 presentation line is specified for the Nokia E61 when creating the ephone-type template.

Router(config)# ephone-type E61
Router(config-ephone-type)# max-presentation 1

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>device-id</td>
<td>Specifies the device ID for a phone type in an ephone-type template.</td>
</tr>
<tr>
<td>num-buttons</td>
<td>Sets the number of line buttons supported by a phone type.</td>
</tr>
<tr>
<td>type</td>
<td>Assigns the phone type to an SCCP phone.</td>
</tr>
</tbody>
</table>
max-redirect

To change the number of times that a call can be redirected by call forwarding or transfer within a Cisco Unified CME system, use the `max-redirect` command in telephony-service configuration mode. To reset to the default number of redirects, use the `no` form of this command.

```
max-redirect number
no max-redirect
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Number of permissible redirects. Range: 5 to 20. Default: 10.</th>
</tr>
</thead>
</table>

**Command Default**

Number of redirects is 10.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(24)T1</td>
<td>Cisco Unified CME 7.1</td>
<td>The default value was increased from 5 to 10.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command supports Cisco Unified CME ephone hunt groups by allowing calls to be redirected more than the default number of times.

**Examples**

The following example sets the maximum number of redirects to 8:

```
Router(config)# telephony-service
Router(config-telephony)# max-redirect 8
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-hunt</td>
<td>Creates an ephone hunt group in Cisco Unified CME.</td>
</tr>
<tr>
<td>hops</td>
<td>Sets the number of hops before a call proceeds to the final number.</td>
</tr>
</tbody>
</table>
**max-subscription**

To set the maximum number of concurrent watch sessions that are allowed, use the `max-subscription` command in presence configuration mode. To return to the default, use the `no` form of this command.

```
max-subscription number
no max-subscription
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>number</code></td>
<td>Maximum watch sessions. Range: 100 to 500. Default: 100.</td>
</tr>
</tbody>
</table>

**Command Default**

Maximum subscriptions is 100.

**Command Modes**

Presence configuration (config-presence)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command sets the maximum number of concurrent presence subscriptions for both internal and external subscribe requests.

**Examples**

The following example shows the maximum subscriptions set to 150:

```
Router(config)# presence
Router(config-presence)# max-subscription 150
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow watch</td>
<td>Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.</td>
</tr>
<tr>
<td>allow subscribe</td>
<td>Allows internal watchers to monitor external presence entities (directory numbers).</td>
</tr>
<tr>
<td>presence enable</td>
<td>Allows incoming presence requests from SIP trunks.</td>
</tr>
<tr>
<td>server</td>
<td>Specifies the IP address of a presence server for sending presence requests from internal watchers to external presence entities.</td>
</tr>
<tr>
<td>watcher all</td>
<td>Allows external watchers to monitor internal presence entities (directory numbers).</td>
</tr>
</tbody>
</table>
max-timeout

To set the maximum combined timeout for the no-answer periods for all ephone-dns in the ephone-hunt list, use the `max-timeout` command in ephone-hunt configuration mode. To return this value to the default, use the `no` form of this command.

```
max-timeout seconds
no max-timeout seconds
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>seconds</code></td>
<td>Number of seconds. Range is from 3 to 60000. Default is unlimited.</td>
</tr>
</tbody>
</table>

**Command Default**

Number of seconds is unlimited.

**Command Modes**

Ephone-hunt configuration (config-ephone-hunt)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following example shows how to set different no-answer timeouts for each ephone-dn in the hunt-group list and no maximum timeout. The first call to the hunt group rings extension 1001. If that extension does not answer in 7 seconds, the call is forwarded to extension 1002. If that extension does not answer after 10 seconds, the call is forwarded to extension 1003. However, if extension 1003 does not answer after 8 seconds, the call is sent to the final number, extension 4500, because the maximum timeout of 25 seconds has been reached.

```
ephone-hunt 3 peer
pilot 4200
list 1001, 1002, 1003
hops 3
timeout 7, 10, 15
max-timeout 25
final 4500
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-hunt</td>
<td>Defines an ephone hunt group and enters ephone-hunt configuration mode.</td>
</tr>
</tbody>
</table>
media

To enable media packets to pass directly between the endpoints, without the intervention of the Cisco Unified Border Element (Cisco UBE), and to enable the incoming and outgoing IP-to-IP call gain/loss feature for audio call scoring on either the incoming dial peer or the outgoing dial peer, enter the media command in dial peer, voice class, or voice service configuration mode. To return to the default IP/PGW behavior, use the no form of this command.

```
media [ {flow-around|flow-through|forking|monitoring [max-calls]|statistics|transcoder high-density} ]
no media [ {flow-around|flow-through|forking|monitoring [max-calls]|statistics|transcoder high-density} ]
```

**Syntax Description**

- **flow-around** (Optional) Enables media packets to pass directly between the endpoints, without the intervention of the Cisco UBE. The media packet is to flow around the gateway.
- **flow-through** (Optional) Enables media packets to pass through the endpoints, without the intervention of the Cisco UBE.
- **forking** (Optional) Enables the media forking feature for all calls.
- **monitoring** Enables the monitoring feature for all calls or a maximum number of calls.
- **max-calls** The maximum number of calls that are monitored.
- **statistics** (Optional) Enables media monitoring.
- **transcoder high-density** (Optional) Converts media codecs from one voice standard to another to facilitate the interoperability of devices using different media standards.

**Command Default**

The default behavior of the Cisco UBE is to receive media packets from the inbound call leg, terminate them, and then reoriginate the media stream on an outbound call leg.

**Command Modes**

Dial peer configuration (config-dial-peer)
Voice class configuration (config-class)
Voice service configuration (config-voi-serv)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(1)T</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)XJ2</td>
<td>This command was modified. The statistics keyword was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>This command was modified. The transcoder and high-density keywords were introduced.</td>
</tr>
<tr>
<td>15.0(1)M</td>
<td>This command was modified. The forking and monitoring keywords and the max-calls argument were introduced.</td>
</tr>
<tr>
<td>15.1(3)T</td>
<td>This command was modified. The media flow around is now supported for the SIP to SIP trunk calls in Cisco Unified CME 8.5.</td>
</tr>
</tbody>
</table>
With the default configuration, the Cisco UBE receives media packets from the inbound call leg, terminates them, and then reorigimates the media stream on an outbound call leg. Media flow-around enables media packets to be passed directly between the endpoints, without the intervention of the Cisco UBE. The Cisco UBE continues to handle routing and billing functions. Media flow-around for SIP-to-SIP calls is not supported.

The Cisco UBE must be running Cisco IOS Release 12.3(1) or a later release to support media flow-around.

You can specify media flow-around for a voice class, all VoIP calls, or individual dial peers.

The transcoder high-density keyword can be enabled in any of the configuration modes with the same command format. If you are configuring the transcoder high-density keyword for dial peers, make sure that the media transcoder high-density command is configured on both the in and out legs.

The software does not support configuring the transcoder high-density keyword on any dial peer that is to handle video calls. The following scenarios are not supported:

- Dial peers used for video at any time. Configuring the media transcoder high-density command directly under the dial-peer or a voice-class media configuration is not supported.
- Dial peers configured on a Cisco UBE used for video calls at any time. The global configuration of the media transcoder high-density command under voice service voip is not supported.

To enable the media command on a Cisco 2900 or Cisco 3900 series Unified Border Element voice gateway, you must first enter the mode border-element command. This enables the media forking and media monitoring commands. Do not configure the mode border-element command on the Cisco 2800 or Cisco 3800 series platforms.

**Examples**

**Media Flow-around Examples**

The following example shows media flow-around configured on a dial peer:

```
Router(config)# dial-peer voice 2 voip
Router(config-dial-peer) media flow-around
```

The following example shows media flow-around configured for all VoIP calls:

```
Router(config)# voice service voip
Router(config-voi-serv) media flow-around
```

The following example shows media flow-around configured for voice class calls:

```
Router(config)# voice class media 1
Router(config-class) media flow-around
```

**Media Flow-through Examples**

The following example shows media flow-around configured on a dial peer:

```
Router(config)# dial-peer voice 2 voip
Router(config-dial-peer) media flow-through
```

The following example shows media flow-around configured for all VoIP calls:
Router(config)# voice service voip
Router(config-voi-serv) media flow-through

The following example shows media flow-around configured for voice class calls:

Router(config)# voice class media 2
Router(config-class) media flow-through

Media Statistics Examples

The following example shows media monitoring configured for all VoIP calls:

Router(config)# voice service voip
Router(config-voi-serv) media statistics

The following example shows media monitoring configured for voice class calls:

Router(config)# voice class media 1
Router(config-class) media statistics

Media Transcoder High-density Examples

The following example shows the media transcoder keyword configured for all VoIP calls:

Router(config)# voice service voip
Router(config-voi-serv)# media transcoder high-density

The following example shows the media transcoder keyword configured for voice class calls:

Router(config)# voice class media 1
Router(config-voice-class)# media transcoder high-density

The following example shows the media transcoder keyword configured on a dial peer:

Router(config)# dial-peer voice 36 voip
Router(config-dial-peer)# media transcoder high-density

Media Monitoring on a Cisco UBE Platform

The following example shows how to configure audio call scoring for a maximum of 100 calls:

mode border-element
media monitoring 100

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dial-peer voice</td>
<td>Enters dial peer configuration mode.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------------------------------------------------------------</td>
</tr>
<tr>
<td><code>mode border-element</code></td>
<td>Enables the media monitoring capability of the <code>media</code> command.</td>
</tr>
<tr>
<td><code>voice class</code></td>
<td>Enters voice class configuration mode.</td>
</tr>
<tr>
<td><code>voice service</code></td>
<td>Enters voice service configuration mode.</td>
</tr>
</tbody>
</table>
members logout

To configure a Cisco Unified CallManager Express system for all non-shared static members or agents in an ephone-hunt with the Hlogout initial state, use the members logout command in ephone-hunt configuration mode. To return to the default, use the no form of this command.

This command is not allowed after list and hunt-group logout DND are configured or if DNs are shared.

members logout
no members logout

Syntax Description
This command has no arguments or keywords.

Command Default
All members are in Hlogin state.

Command Modes
ephone-hunt configuration (config-ephone-hunt)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(4)M</td>
<td>Cisco Unified CME 9.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

Usage Guidelines
All members configured under an ephone-hunt are initialized with HLogin. Use this command to initialize all non-shared static members to Hlogout.

Examples
The following example configures HLogin as the default for all non-shared ephone-hunt static members:

Router(config-telephony)# ephone-hunt 1
Router(config-ephone-hunt)# members logout

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-hunt</td>
<td>Enters ephone-hunt configuration mode to define a Cisco CME ephone-hunt group.</td>
</tr>
</tbody>
</table>
members logout (voice hunt-group)

To configure a Cisco Unified CME system for all non-shared static members or agents in a voice hunt group with the HLogin initial state, use the `members logout` command in voice hunt-group configuration mode. To return to the default state, use the `no members logout` form of this command.

This command is not allowed if the CLI command `list` is configured.

**members logout**

**no members logout**

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
All members are in HLogin state.

**Command Modes**
voice hunt-group

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IOS XE Everest 16.4.1</td>
<td>Cisco Unified CME 11.6</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.6(3)M1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

All members configured under a voice hunt-group are initialized with HLogin. Use this command to initialize all non-shared static members to HLogin. If any member of a hunt group in a SIP phone logs out using the CLI command `members logout`, all other DN's of that phone in any hunt group are also logged out. This is because SIP phones only support phone level logout. For SCCP phones, only the DN that is configured with the CLI command `members logout` is logged out from the hunt group. Other member DN's do not logout as SCCP phones support line level logout.

Members Logout is not supported if the CLI command `hunt-group logout DND` is configured. Also, you cannot configure the CLI command `members logout` if the command `list` is configured.

**Examples**

The following example configures HLogin as the default for all non-shared voice hunt-group static members:

```shell
Router(config-register-global)# voice hunt-group 1
Router(config-voice-hunt-group)# members logout
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>members logout</td>
<td>Enables members logout for ephone-hunt groups configured on a Cisco Unified CME.</td>
</tr>
</tbody>
</table>
missed-calls

To report missed calls to directory numbers on an IP phone, use the **missed-calls** command in ephone configuration mode. To suppress missed-calls reporting, use the **no** form of this command.

```plaintext
missed-calls [all]
no missed-calls
```

**Syntax Description**

- **all** (Optional) Displays all missed calls including those on overlay buttons.

**Command Default**

Missed calls are presented on the IP phone and listed in the missed-calls directory. Missed calls to overlay buttons are not reported.

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables Cisco Unified CME to report missed calls on the specified phone. Use the **all** keyword to report missed calls to overlaid directory numbers. Only calls to an overlay set that are visibly presented on the phone are reported as missed calls. Calls to an overlay that are terminated by the caller before they are displayed on the phone are not reported as missed calls.

If the unique extension number for a phone is assigned to an overlay set on the phone, missed calls to that extension number are not reported unless you enable this command using the **all** keyword.

**Examples**

The following example shows that all unanswered calls to 4001 are reported on phone 1.

```plaintext
ephone-dn 1 dual-line
number 4001
ephone 1
mac-address 0014.6AAC.24E3
type 7960
button 101,30,31 2:2 3:3
missed-calls all
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>button</strong></td>
<td>Associates directory numbers with individual buttons on a Cisco Unified IP Phone and specifies ring behavior.</td>
</tr>
</tbody>
</table>
mlpp indication

To enable MLPP indication on an SCCP phone or analog FXS port, use the `mlpp indication` command in ephone-template or voice-port configuration mode. To disable MLPP indication, use the `no` form of this command.

```
mlpp indication
no mlpp indication
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

MLPP indication is enabled on the phone.

**Command Modes**

Ephone-template configuration (config-ephone-template)
Voice-port configuration (config-voiceport)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Products</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

This command enables a phone to play precedence and preemption tones, and display precedence information for calls. If MLPP indication is disabled, calls on the phone can be preempted but there is no visual or audible indication.

To apply a template to an SCCP phone, use the `ephone-template` command in ephone configuration mode.

**Examples**

The following example shows MLPP indication is disabled in template 5 and applied to phone 12:

```
éphone-template 5
  mlpp max-precedence 0
  no mlpp indication
!
ephone 12
  mac-address 000F.9054.31BD
  ephone-template 5
  type 7960
  button 1:12
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ephone-template</code> (ephone)</td>
<td>Applies an ephone template to an SCCP phone.</td>
</tr>
<tr>
<td><code>mlpp max-precedence</code></td>
<td>Sets the maximum precedence (priority) level that a phone user can specify when making an MLPP call.</td>
</tr>
<tr>
<td><code>mlpp preemption</code></td>
<td>Enables preemption capability on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>preemption tone timer</td>
<td>Sets the amount of time the preemption tone plays on the called phone when a lower precedence call is being preempted.</td>
</tr>
</tbody>
</table>
mlpp max-precedence

To set the maximum precedence (priority) level that a phone user can specify when making an MLPP call, use the `mlpp max-precedence` command in ephone-template or voice-port configuration mode. To reset to the default, use the `no` form of this command.

```
mlpp max-precedence number
no mlpp max-precedence
```

### Syntax Description

| number | Number representing the maximum precedence level. Range: 0 to 4, where 0 is the highest priority. Default: 4. |

### Command Default

The MLPP precedence is 4 (routine).

### Command Modes

- Ephone-template configuration (config-ephone-template)
- Voice-port configuration (config-voiceport)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Products</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

This command sets the maximum precedence level that a user can select when making MLPP calls from a phone. The phone user can specify a precedence level that is less than or equal to this value. Cisco Unified CME rejects the call if a user selects a precedence level that is higher than the level set with this command and the user receives an error tone.

Emergency 911 calls are automatically assigned precedence level 0.

To apply a template to an SCCP phone, use the `ephone-template` command.

### Examples

The following example shows the precedence level set to 0 in template 5 and applied to phone 12:

```
ephone-template 5
  mlpp max-precedence 0
!
ephone 12
  mac-address 000F.9054.31BD
  ephone-template 5
  type 7960
  button 1:12
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>access-digits</td>
<td>Defines the access digit that phone users dial to request a precedence call.</td>
</tr>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an SCCP phone.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>mlpp indication</td>
<td>Enables MLPP indication on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td>mlpp preemption</td>
<td>Enables the preemption capability on an SCCP phone or analog FXS port.</td>
</tr>
</tbody>
</table>
**mlpp preemption**

To enable calls on an SCCP phone or analog FXS port to be preempted, use the **mlpp preemption** command in ephone-template or voice-port configuration mode. To disable preemption, use the **no** form of this command.

```plaintext
mlpp preemption
no mlpp preemption
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Preemption is enabled on the phone.

**Command Modes**

- Ephone-template configuration (config-ephone-template)
- Voice-port configuration (config-voiceport)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The command allows an SCCP IP phone or an FXS analog phone to have its calls preempted if it is busy with lower precedence calls.

A phone with preemption disabled can still receive precedence calls in an MLPP network, but the phone itself does not get preempted. The preemption-disabled phone can be connected to a call that is preempted (at another device), in which case, that device receives preemption.

To apply a template to an SCCP phone, use the **ephone-template** command in ephone configuration mode.

**Examples**

The following example shows preemption disabled in template 5 and applied to phone 12:

```
ephone-template 5
mlpp max-precedence 0
no mlpp preemption
!
ephone 12
mac-address 000F.9054.31BD
ephone-template 5
type 7960
button 1:12
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>ephone-template</strong> (ephone)</td>
<td>Applies an ephone template to an SCCP phone.</td>
</tr>
<tr>
<td><strong>mlpp indication</strong></td>
<td>Enables MLPP indication on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------</td>
<td>------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>mlpp max-precedence</td>
<td>Sets the maximum precedence (priority) level that a phone user can specify when making an MLPP call.</td>
</tr>
<tr>
<td>preemption tone timer</td>
<td>Defines the expiry time for the preemption tone for the call being preempted.</td>
</tr>
</tbody>
</table>
**mlpp service-domain**

To set the service domain and maximum precedence (priority) level for Multilevel Precedence and Preemption (MLPP) calls, use the `mlpp service-domain` command in ephone-template or voice-port configuration mode. To reset to the default, use the `no` form of this command.

```
mlpp service-domain {drsn|dsn} identifier domain-number max-precedence level
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>drsn</code></td>
<td>Phone belongs to Defense Red Switched Network (DRSN).</td>
</tr>
<tr>
<td><code>dsn</code></td>
<td>Phone belongs to Defense Switched Network (DSN).</td>
</tr>
<tr>
<td><code>domain-number</code></td>
<td>Number to identify the domain, in three-octet format. Range: 0x000000 to 0xFFFFFE.</td>
</tr>
<tr>
<td><code>level</code></td>
<td>Number representing the maximum precedence level, where 0 is the highest priority. Range is 0 to 4 (DSN) or 0 to 5 (DRSN).</td>
</tr>
</tbody>
</table>

**Command Default**

Phone uses global default configured with the `service-domain` command.

**Command Modes**

Ephone-template configuration (config-ephone-template)  
Voice-port configuration (config-voiceport)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command sets the MLPP domain type and number for the phone, and the maximum precedence level that a user can select when making MLPP calls from the phone.

The phone user can select a precedence level that is less than or equal to the value set with this command. Cisco Unified CME rejects the call if a user selects a precedence level that is higher than the level set with this command and the user receives an error tone.

If this command and the `service-domain` command are not enabled, the phone cannot make MLPP calls.

Emergency 911 calls are automatically assigned precedence level 0.

To apply a template to an SCCP phone, use the `ephone-template` command.

**Examples**

The following example shows the precedence level set to 1 in template 5 and applied to phone 15:

```
ephone-template 5
  mlpp service-domain dsn identifier 000010 max-precedence 1
  
ephone 15
  mac-address 000F.9054.31BD
  ephone-template 5
```
mlpp service-domain

type 7960
button 1:15

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>access-digit</td>
<td>Defines the access digit that phone users dial to request a precedence call.</td>
</tr>
<tr>
<td>ephone-template</td>
<td>Applies an ephone template to an SCCP phone.</td>
</tr>
<tr>
<td>mlpp indication</td>
<td>Enables MLPP indication on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td>mlpp preemption</td>
<td>Enables the preemption capability on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td>service-domain</td>
<td>Sets the global MLPP domain name and number.</td>
</tr>
</tbody>
</table>
mobility (ephone-dn)

To enable the Mobility feature on an extension of an SCCP IP phone, use the `mobility` command in ephone-dn configuration mode. To disable mobility on the extension, use the `no` form of this command.

```
mobility
no mobility
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Mobility is not enabled for the extension.

**Command Modes**
Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command enables the Mobility feature on the extension, which is required to enable the Single Number Reach (SNR) feature.

**Examples**
The following example shows extension 1001 is enabled for SNR. After a call rings at this number for 5 seconds, the call also rings at the remote number 4085550133. If the call is not answered after 20 seconds, the call no longer rings the phone and is forwarded to the voice-mail number 2001.

```
ephone-dn 10
number 1001
mobility
snr 4085550133 delay 5 timeout 15 cfwd-noan 2001
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>number</td>
<td>Associates a telephone or extension number with an ephone-dn.</td>
</tr>
<tr>
<td>snr</td>
<td>Enables Single Number Reach on an extension of an SCCP IP phone.</td>
</tr>
<tr>
<td>softkeys connected</td>
<td>Modifies the order and type of soft keys that display on an IP phone during the connected call state.</td>
</tr>
<tr>
<td>softkeys idle</td>
<td>Modifies the order and type of soft keys that display on an IP phone during the idle call state.</td>
</tr>
</tbody>
</table>
mobility (voice register dn)

To enable the Mobility feature on an extension of a Cisco Unified SIP IP phone, use the `mobility` command in voice register dn configuration mode. To disable the Mobility feature on the extension, use the `no` form of this command.

`mobility`

`no mobility`

Syntax Description
This command has no arguments or keywords.

Command Default
The Mobility feature is not enabled on the extension of a Cisco Unified SIP IP phone.

Command Modes
Voice register dn configuration (config-register-dn)

Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

Usage Guidelines
Use the `mobility` command to enable a Cisco Unified SIP IP phone to receive calls on an extension, which is required to enable the Single Number Reach (SNR) feature.

Examples
The following example shows how to enable the Mobility feature on directory number 25:

```
Router(config)# voice register dn 25
Router(config-register-dn)# mobility
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>snr (voice register dn)</td>
<td>Enables the SNR feature on an extension of a Cisco Unified SIP IP phone.</td>
</tr>
</tbody>
</table>
mode cme

To enable the mode for configuring SIP phones in a Cisco Unified CallManager Express (Cisco Unified CME) system, use the `mode cme` command in voice register global configuration mode. To return to the default, use the `no` form of this command.

```
mode cme
no mode
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>cme</code></td>
<td>Only valid keyword is <code>cme</code>. This mode determines the commands that are available to configure SIP phones.</td>
</tr>
<tr>
<td><code>esrst</code></td>
<td>Changes to the esrst mode and this mode determines the commands that are available to configure SIP phones.</td>
</tr>
</tbody>
</table>

**Command Default**

Default is SIP SRST mode.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.3(3)M</td>
<td>Cisco CME 10.0</td>
<td>This command was modified to add the <code>esrst</code> mode.</td>
</tr>
<tr>
<td>Cisco IOS XE Everest 16.5.1b</td>
<td>Unified CME 11.7 Unified SRST 11.7</td>
<td>The behavior of <code>no</code> form of this command was modified, to clear all voice register pools and voice register dns, along with mode specific configurations.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables Cisco Unified CME on the router for configuration purposes. The router is enabled for Cisco SIP SRST by default. Enable this command before configuring SIP phones in Cisco Unified CME to ensure that all required commands are available.

For releases prior to Unified CME/SRST 11.7, the `no` form of this command clears only the mode specific configurations (For example, `source-address` under voice register global configuration, and user credentials configured under voice register pool configuration). From Cisco IOS XE Everest 16.5.1 (Unified CME/SRST Release 11.7) onwards, the `no` form of this command clears all the voice register pools and voice register dns, along with mode specific configurations.

**Examples**

The following example shows how to set the mode to Cisco CME:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice register global</td>
<td>Displays all global configuration information associated with SIP phones.</td>
</tr>
</tbody>
</table>
moh (ephone-dn)

To enable music on hold (MOH) from an external live audio feed (standard line-level audio connection) connected directly to the router by an foreign office exchange (FXO) or an E&M analog voice port, use the moh command in ephone-dn configuration mode. To disable MOH from a live feed or to disable the outcall number or multicast capability, use the no form of this command.

moh [out-call outcall-number] [ip ip-address port port-number [route ip-address]]
no moh [{out-call outcall-number|ip;]}

Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>out-call</td>
<td>(Optional) Sets up a call to the outcall number in order to connect to the MOH feed. If this keyword is not used, the live feed is assumed to derive from an incoming call to the ephone-dn under which this command is used.</td>
</tr>
<tr>
<td>outcall-number</td>
<td></td>
</tr>
<tr>
<td>ip</td>
<td>(Optional) Indicates that this audio stream is to be used as a multicast source as well as the MOH source and specifies the destination IP address for multicast.</td>
</tr>
<tr>
<td>ip-address</td>
<td></td>
</tr>
<tr>
<td>port</td>
<td>(Optional) Specifies the media port for multicast. Range is from 2000 to 65535. Port 2000 is recommended because this port is already used for normal Real-Time Transport Protocol (RTP) media transmissions between IP phones and the Cisco CallManager Express router.</td>
</tr>
<tr>
<td>port-number</td>
<td></td>
</tr>
<tr>
<td>route</td>
<td>(Optional) Indicates the specific router interface on which to transmit the IP multicast packets. The default is that the MOH multicast stream is automatically output on the interface that corresponds to the address that was configured with the ip source-address command.</td>
</tr>
<tr>
<td>ip-address</td>
<td></td>
</tr>
</tbody>
</table>

Command Default

MOH is disabled on an extension.

Command Modes

Ephone-dn configuration (config-ephone)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>The ip, port, and route keywords were added.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

This command takes the specified live-feed audio stream and uses it as MOH for a Cisco Unified CME system. The connection for the live-feed audio stream is established as an automatically connected voice call. If the out-call keyword is used, the type of connection can include VoIP calls if voice activity detection (VAD) is disabled. The typical operation is for the MOH ephone-dn to establish a call to a local router E&M voice port.

Connection via E&M is the recommended mechanism because it requires minimal external components. The E&M port must be placed in 4-wire operation, using E&M immediate signaling and with the auto-cut-through option enabled. You directly connect a line-level audio feed (standard audio jack) to pins 3 and 6 of an E&M RJ-45 connector. The E&M WAN interface card (WIC) has a built-in audio transformer that provides...
appropriate electrical isolation for the external audio source. (The audio connection on the E&M port does not require loop current.) The **signal immediate** and **auto-cut-through** commands disable E&M signaling on this voice port. A G.711 audio packet stream is generated by the digital signal processor (DSP) on the E&M port.

If you are using an FXO voice port for live-feed MOH instead of an E&M port, connect the MOH source to the FXO voice port. This connection requires an external adapter to supply normal telephone company (telco) battery voltage with the correct polarity to the tip-and-ring leads of the FXO port. The adapter must also provide transformer-based isolation between the external audio source and the tip-and-ring leads of the FXO port.

Music from a live feed is continuously fed into the MOH playout buffer instead of being read from an audio file in flash memory. There is typically a two-second delay with live-feed MOH.

If the **out-call** keyword is used, an outbound call to the MOH live-feed source is attempted (or reattempted) every 30 seconds until the call is connected to the ephone-dn (extension) that has been configured for MOH. Note that this ephone-dn is not associated with any physical phone.

If the **moh** (ephone-dn) command is used without any keywords or arguments, the ephone-dn will accept an incoming call and use the audio stream from the call as the source for the MOH stream, displacing any audio stream that is available from a flash file. To accept an incoming call, the ephone-dn must have an extension or phone number configured for it. A typical usage would be for an external H.323-based server device to call the ephone-dn to deliver an audio stream to the Cisco CME system. Normally, only a single ephone-dn would be configured this way. If there is more than one ephone configured to accept incoming calls for MOH, the first ephone-dn that is successfully connected to a call (incoming or outgoing) is the MOH source for the system.

MOH can also be derived from an audio file when you use the **moh** command in telephony-service configuration mode with the **filename** argument. There can be only one MOH stream at a time in a Cisco CME system, and if both an audio file and a live feed have been specified for the MOH stream, the router seeks the live feed from the **moh** (ephone-dn) command first. If the live feed is found, the router displaces the audio file source. If the live feed is not found or fails at any time, the router falls back to the audio file source that was specified in the **moh** (telephony-service) command.

If you use the **ip** keyword to specify a multicast address in this command, the audio stream is sent to the multicast address in addition to serving as the MOH source. Additionally, if you specify a different multicast address using the **multicast moh** command under telephony-service configuration mode, the audio stream is also sent to the multicast address that you named in that command. It is therefore possible to send the live-feed audio stream to MOH and to two different multicast addresses: the one that is directly configured under the **moh** (ephone-dn) command and the one that is indirectly configured under the **multicast moh** (telephony-service) command.

A related command, the **feed** command, provides the ability to multicast an audio stream that is not the MOH audio stream.

---

**Note**

IP phones do not support multicast at 224.x.x.x addresses.

---

**Examples**

The following example establishes a live music-on-hold source by setting up a call to extension 7777:

```
Router(config)# ephone-dn 55
Router(config-ephone-dn)# moh out-call 7777
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>auto-cut-through</strong></td>
<td>Enables call completion when an M-lead response is not provided.</td>
</tr>
<tr>
<td><strong>ephone-dn</strong></td>
<td>Enters ephone-dn configuration mode to set directory numbers and parameters for individual Cisco IP phone extensions.</td>
</tr>
<tr>
<td><strong>feed</strong></td>
<td>Enables multicast of an audio stream that is different from the music-on-hold audio stream.</td>
</tr>
<tr>
<td><strong>ip source-address</strong></td>
<td>Identifies the IP address and port through which IP phones communicate with a Cisco CME router.</td>
</tr>
<tr>
<td><strong>moh (telephony-service)</strong></td>
<td>Enables music on hold from an audio file.</td>
</tr>
<tr>
<td><strong>multicast moh</strong></td>
<td>Enables multicast of the music-on-hold audio stream.</td>
</tr>
<tr>
<td><strong>signal</strong></td>
<td>Specifies the type of signaling for a voice port.</td>
</tr>
</tbody>
</table>
### moh (telephony-service)

To generate an audio stream from a file for music on hold (MOH) in a Cisco CallManager Express (Cisco CME) system, use the `moh` command in telephony-service configuration mode. To disable the MOH audio stream from this file, use the `no` form of this command.

```
moh filename
no moh
```

**Syntax Description**

<table>
<thead>
<tr>
<th>filename</th>
<th>Name of the audio file to use for the MOH audio stream. The file must be copied to flash memory on the Cisco CME router.</th>
</tr>
</thead>
</table>

**Command Default**

Tone on hold (a periodic beep is played to the caller)

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables MOH from .au and .wav format music files. MOH is played for G.711 callers and on-net VoIP and PSTN callers who are on hold in a Cisco CME system. Local callers within a Cisco CME system hear a repeating tone while they are on hold.

Audio files that are used for MOH must be copied to the Cisco CME router flash memory. A MOH file can be in .au or .wav file format; however, the file format must contain 8-bit 8-kHz data in A-law or mu-law data format. We recommend using a moh-file size greater than 100 KB.

If you want to replace or modify the audio file that is currently specified, you must first disable the MOH capability using the `no moh` command. The following example replaces file1 with file2:

```
Router(config-telephony)# moh file1
Router(config-telephony)# no moh
Router(config-telephony)# moh file2
```

If you specify a second file without first removing the original file, the MOH mechanism stops working and may require a router reboot to clear the problem.

A related command, the `moh` command in ephone-dn configuration mode, can be used to establish a MOH audio stream from a live feed. If you configure both commands, MOH falls back to playing music from the audio file if the live music feed is interrupted.

The `multicast moh` command allows you to use the MOH stream for a multicast broadcast.

When the `multicast moh` and `debug ephone moh` commands are both enabled, if you also use the `no moh` command, the debug output can be excessive and flood the console. Multicast MOH should be disabled before using the `no moh` command when the `debug ephone moh` command is enabled.

**Examples**

The following example enables music on hold and specifies a music file:
Router(config)# telephony-service
Router(config-telephony)# moh minuet.wav

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug ephone moh</td>
<td>Displays diagnostic information for music on hold.</td>
</tr>
<tr>
<td>moh (ephone-dn)</td>
<td>Enables music on hold from a live audio feed.</td>
</tr>
<tr>
<td>multicast moh</td>
<td>Enables multicast of the music-on-hold audio stream.</td>
</tr>
</tbody>
</table>
moh (voice moh-group)

To enable music on hold (MOH) for a MOH group, use the moh command in voice moh-group configuration mode. To disable music on hold, use the no form of this command.

```
moh  filename
no moh  filename
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>filename</td>
<td>Name of the music file. The music file must be in the system flash.</td>
</tr>
</tbody>
</table>

**Command Default**

No MOH is enabled

**Command Modes**

Voice moh-group configuration (config-voice-moh-group)

---

<table>
<thead>
<tr>
<th>Command History</th>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>15.0(1)XA</td>
<td>Cisco Unified CME/SRST/SIP SRST 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td></td>
<td>15.1(1)T</td>
<td>Cisco Unified CME/SRST/SIP SRST 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The moh command allows you to specify the .au and .wav format music files that are played to callers who have been put on hold. MOH works only for G.711 calls and on-net VoIP and PSTN calls. For all other calls, callers hear a periodic tone. You must provide the directory and filename of the MOH file in URL format. For example: moh flash:/minuet.au

**Note**

Music-on-hold files can be in .wav or .au file format; however, the file format must contain 8-bit 8-kHz data; for example, CCITT a-law or u-law data format.

**Examples**

The following example enables MOH for voice moh group 1 and specifies the music files:

```
Router(config)#
Router(config)#voice moh-group 1
Router(config-voice-moh-group)# moh flash:/minuet.wav
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice moh-group</td>
<td>Enters voice moh-group configuration mode.</td>
</tr>
<tr>
<td>extension-range</td>
<td>Defines extension range for a clients calling a voice-moh-group.</td>
</tr>
<tr>
<td>moh</td>
<td>Enables music on hold from a flash audio file.</td>
</tr>
<tr>
<td>multicast moh</td>
<td>Enables multicast of the music-on-hold audio stream.</td>
</tr>
</tbody>
</table>
moh-file-buffer

To specify a MOH file buffer size, use the **moh-file-buffer** command in telephony-service configuration mode. To delete the moh-file-buffer size, use the **no** form of this command.

```
moh-file-buffer file-size
no moh-file-buffer
```

**Syntax Description**

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>file-size</td>
<td>Specifies a numeric value for the buffer MOH file size between 64 KB and 10000 KB.</td>
</tr>
</tbody>
</table>

**Command Default**

No moh-file-buffer is configured.

**Command Modes**

Telephony-service configuration (config-telephony-service)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command allows to set a buffer MOH file size limit for new MOH files. You can allocate a MOH file buffer size between 64 KB (8 seconds) and 10000 KB (20 minutes, approximately). A large buffer size is desirable to cache the largest MOH file and better MOH performance. During memory allocation the buffer size is aligned to 16KB.

The default maximum file buffer size is 64 KB. If the MOH file size is too large, it cannot be cached and the buffer size falls back to 64 KB.

**Note**

When live-feed is enabled there is no file caching for MOH-group 0.

**Examples**

The following example shows a moh-file-buffer size of 180 KB assigned for future moh files under the telephony-service configuration mode.

```
! 
! 
telephony service
  max-conferences 8 gain -6
  transfer-system full-consult
  moh-file-buffer 180
  !
! 
line con 0
  exec-timeout 0 0
line aux 0
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice-moh-group</td>
<td>Enter voice-moh-group configuration mode.</td>
</tr>
<tr>
<td>moh filename</td>
<td>Enables music on hold from a flash audio feed</td>
</tr>
<tr>
<td>multicast moh</td>
<td>Enables multicast of the music-on-hold audio stream.</td>
</tr>
<tr>
<td>extension-range</td>
<td>Specifies the extension range for a clients calling a voice-moh-group.</td>
</tr>
</tbody>
</table>
moh-group (ephone-dn)

To assign a MOH group to a directory number, use the **moh-group** command in ephone-dn configuration mode. To remove the MOH group, use the **no** form of this command.

```
moh-group  tag
no  moh-group  tag
```

**Syntax Description**

- **tag**: A unique number that identifies a MOH group. Range is from 1 to 5.

**Command Default**

No MOH group is configured.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to assign a MOH group to a directory number in ephone-dn configuration mode. Use the number tag from 1 to 5 to specify the MOH group that you want to assign to this directory number.

**Examples**

The following example shows how to assign a MOH group to a directory number under ephone-dn mode.

```
Router(config)# ephone-dn 98
Router(config-ephone-dn)#moh-group 1
Router(config-ephone-dn)#
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>description</td>
<td>Displays a brief description about a voice moh-group in use.</td>
</tr>
<tr>
<td>extension-range</td>
<td>Defines extension range for a clients calling a voice-moh-group.</td>
</tr>
<tr>
<td>moh</td>
<td>Enables music on hold from a flash audio feed.</td>
</tr>
<tr>
<td>multicast moh</td>
<td>Enables multicast of the music-on-hold audio stream.</td>
</tr>
</tbody>
</table>
mtp

To send voice packets from an IP phone to the Cisco Unified CME router, use the `mtp` command in ephone or ephone-template configuration mode. To return to the default, use the `no` form of this command.

```
$mtp \{ \text{video-only} | \text{both} \}
$no \ mtp \{ \text{video-only} | \text{both} \}
```

### Syntax Description

<table>
<thead>
<tr>
<th>Option</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>video-only</td>
<td>Specifies that the video streams must be sent through the Cisco Unified CME route.</td>
</tr>
<tr>
<td>both</td>
<td>Specifies that both voice and video streams must be sent through the Cisco Unified CME route.</td>
</tr>
</tbody>
</table>

### Command Default

If no arguments are given, only voice packets are sent to the router.

An IP phone in a call with another IP phone in the same Cisco Unified CME system sends voice and video packets directly to the other phone.

### Command Modes

- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>15.1(4)M</td>
<td>Cisco Unified CME 8.6</td>
<td>Support for choosing video streams was added.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Normally, media packets (RTP packets) that are sent between IP phones in the same Cisco Unified CME system go directly to the other phone and do not travel through the Cisco Unified CME router. When these packets are sent from a remote IP phone to another IP phone in the same Cisco Unified CME system, they may be obstructed by a firewall. The `mtp` command instructs a phone to always send its media packets to the Cisco Unified CME router, which acts as a proxy and forwards the packets to the destination. Firewalls can then be easily configured to pass the RTP packets because the router uses a specified UDP port for media packets. In this way, RTP packets from remote IP phones can be delivered to IP phones on the same system. The default is that this function is off and that RTP packets that are sent from one IP phone to another IP phone in the same Cisco Unified CME system go directly to the other phone.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

### Examples

The following example sends video and audio packets from ephone 437 to the Cisco Unified CME router for all calls:

```
ephone 437
button 1:29
mtp both
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>ephone-template (ephone)</td>
<td>Applies template to ephone being configured.</td>
</tr>
</tbody>
</table>
**mtu (vpn-profile)**

To enter the mtu value in bytes, use the `mtu` command in vpn-profile configuration mode.

```
mtu  bytes
```

**Syntax Description**


**Command Default**

Default is 1290 bytes.

**Command Modes**

Vpn-profile configuration (conf-vpn-profile)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `mtu` command to define a value in bytes. The mtu value ranges from 256 to 1406 bytes. The default value is 1290 bytes.

**Examples**

The following example shows the mtu value set to 1300 bytes in vpn-profile 2:

```
Router# show run
!
!
!
voice service voip
ip address trusted list
ipv4 20.20.20.1
vpn-group 1
vpn-gateway 1 https://9.10.60.254/SSLVPNphone
vpn-trustpoint 1 trustpoint cme_cert root
vpn-hash-algorithm sha-1
vpn-profile 1
keepalive 50
host-id-check disable
vpn-profile 2
mtu 1300
password-persistent enable
host-id-check enable
sip
!
voice class media 10
media flow-around
!
voice register global
max-pool 10
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>vpn-profile</td>
<td>Defines a VPN-profile.</td>
</tr>
</tbody>
</table>
multicast moh

To use the music-on-hold (MOH) audio stream as a multicast source for Cisco Unified CME or for a MOH group, use the multicast moh command in telephony-service configuration mode or in voice-moh-group configuration mode. To disable multicast use of the MOH stream, use the no form of this command.

```
multicast moh ip-address port port-number [route ip-address-list]  
no multicast moh
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Argument</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ip-address</td>
<td>Specifies the destination IP address for multicast.</td>
</tr>
<tr>
<td>port port-number</td>
<td>Specifies the media port for multicast. Range is from 2000 to 65535. Port 2000 is recommended because this port is already used for normal Real-Time Transport Protocol (RTP) media transmissions between IP phones and the Cisco CME router.</td>
</tr>
<tr>
<td>route ip-address-list</td>
<td>(Optional) Indicates specific router interfaces over which to transmit the IP multicast packets. Up to four IP addresses can be listed, each separated from the other by a space. The default is that the MOH multicast stream is automatically output on the interfaces that correspond to the address that was configured with the ip source-address command.</td>
</tr>
</tbody>
</table>

**Command Default**

No multicast is enabled.

**Command Modes**

Telephony-service configuration (config-telephony)
Voice-moh-group configuration (config-voice-moh-group)

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was modified. Multicast MOH was enabled under voice moh-group configuration mode.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables multicast of the audio stream that is designated for MOH in a Cisco CME system. Use this command in voice moh-group configuration mode to enable multicast of audio stream for a specific MOH group.

A related command, the **moh (ephone-dn)** command, creates a MOH audio stream from an external live feed and optionally enables multicast on that stream. These two commands can be used concurrently to provide multicast of a live-feed MOH audio stream to two different multicast addresses.

Another related command, the **feed** command, enables multicast of an audio stream that is not the MOH audio stream.

When the **multicast moh** and **debug ephone moh** commands are both enabled, if you also use the no **moh** command, the debug output can be excessive and flood the console. Multicast MOH should be disabled before using the no **moh** command when the **debug ephone moh** command is enabled.
IP phones do not support multicast at 224.x.x.x addresses.

Multicast for live feed is not support in MOH groups.

Examples

The following example enables multicast of the MOH audio stream at multicast address 239.10.16.4 and names two router interfaces over which to send the multicast packets.

Example 1: Multicast enabled for MOH audio stream under telephony service.

```text
Router(config)# telephony-service
Router(config-telephony)# moh minuet.au
Router(config-telephony)# multicast moh 239.10.16.4 port 2000 route 10.10.29.17 10.10.29.33
```

Example 2: Multicast enabled for MOH audio stream under voice moh-group configuration mode.

```text
Router(config)# voice-moh-group 1
Router(config-voice-moh-group)# moh minuet.au
Router(config-voice-moh-group)# multicast moh 239.10.16.4 port 2000 route 10.10.29.17 10.10.29.33
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>feed</td>
<td>Enables multicast of an audio stream that is not the music-on-hold audio stream.</td>
</tr>
<tr>
<td>ip source-address</td>
<td>Identifies the IP address and port through which IP phones communicate with a Cisco CME router.</td>
</tr>
<tr>
<td>moh (ephone-dn)</td>
<td>Enables music on hold from a live audio feed.</td>
</tr>
<tr>
<td>moh (telephony-service)</td>
<td>Enables music on hold from an audio file.</td>
</tr>
<tr>
<td>voice moh-group</td>
<td>Enters voice moh-group configuration mode</td>
</tr>
</tbody>
</table>
mwi (ephone-dn and ephone-dn-template)

To enable a specific Cisco Unified IP phone extension to receive message-waiting indication (MWI) notification from an external voice-messaging system, use the mwi command in ephone-dn or ephone-dn-template configuration mode. To disable this feature, use the no form of this command.

```
mwi {off|on|on-off}
no mwi {off|on|on-off}
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>off</td>
<td>Sets a Cisco Unified IP phone extension to process MWI to OFF, using either</td>
</tr>
<tr>
<td>on</td>
<td>the main or secondary phone number.</td>
</tr>
<tr>
<td>on-off</td>
<td>Sets a Cisco Unified IP phone extension to process MWI to both ON and OFF,</td>
</tr>
<tr>
<td></td>
<td>using either the main or secondary phone number.</td>
</tr>
<tr>
<td>no off</td>
<td>Sets a Cisco Unified IP phone extension to process MWI to ON, using either</td>
</tr>
<tr>
<td></td>
<td>the main or secondary phone number.</td>
</tr>
<tr>
<td>no on</td>
<td>Sets a Cisco Unified IP phone extension to process MWI to both ON and OFF,</td>
</tr>
<tr>
<td></td>
<td>using either the main or secondary phone number.</td>
</tr>
<tr>
<td>no on-off</td>
<td>Sets a Cisco Unified IP phone extension to process MWI to OFF, using either</td>
</tr>
<tr>
<td></td>
<td>the main or secondary phone number.</td>
</tr>
</tbody>
</table>

### Command Default

MWI notification is disabled on an extension.

### Command Modes

Ephone-dn configuration (config-ephone-dn)

Ephone-dn-template configuration (config-ephone-dn-template)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-dn-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command enables a Cisco Unified IP phone extension to receive MWI notification from an external voice-messaging system for all the Cisco Unified IP phones connected to the Cisco Unified CME router. This extension is a “dummy” extension and is not associated with any physical phone. The external voice-messaging system is able to communicate MWI status by making telephone calls to the dummy extension number, with the MWI information embedded in either the called or calling-party IP phone number.

This command cannot be used unless the number command is already configured for this extension (ephone-dn).

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

### Examples

The following example sets MWI to on:

```
cisco-12.4(9)t#conf tel
 negotiate default
 negotiate ephone-dn
 negotiate ephone-dn-template
 mwi on
```

This sets MWI to ON on all the ephone-dn and ephone-dn-template configurations.
Router(config)# ephone-dn 1
Router(config-ephone-dn) number 8000
Router(config-ephone-dn) mwi on

The following example sets MWI to off:

Router(config)# ephone-dn 2
Router(config-ephone-dn) number 8001
Router(config-ephone-dn) mwi off

The following example sets MWI to both on and off for the primary and secondary number, where the MWI information is embedded in the calling-party number. A call placed by the voice-mail system to 8002 turns on the MWI light for the extension number indicated by the calling-party number for the MWI call. A call placed to 8003 turns off the MWI light.

Router(config)# ephone-dn 3
Router(config-ephone-dn) number 8002 secondary 8003
Router(config-ephone-dn) mwi on-off

The following example sets MWI to both on and off for the primary and secondary number, where the MWI information is embedded in the called-party number. A call placed by the voice-mail system to 8000*5001*1 turns on the MWI light for extension 5001. A call placed to 8000*5001*2 turns off the MWI light.

Router(config)# ephone-dn 20
Router(config-ephone-dn) number 8000*....*1 secondary 8000*....*2
Router(config-ephone-dn) mwi on-off

The following example uses an ephone-dn-template to set MWI to on:

Router(config)# ephone-dn-template 4
Router(config-ephone-dn-template) mwi on
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 8000
Router(config-ephone-dn)# ephone-dn-template 4

### Related Commands

<table>
<thead>
<tr>
<th>Description</th>
<th>Command</th>
</tr>
</thead>
<tbody>
<tr>
<td>Applies template to ephone-dn being configured.</td>
<td>ephone-dn-template (ephone-dn)</td>
</tr>
<tr>
<td>Sets the expiration timer for registration for either the client or the server.</td>
<td>mwi expires</td>
</tr>
<tr>
<td>Subscribes an extension in a Cisco Unified CME router to receive MWI notification from a SIP MWI server.</td>
<td>mwi sip (ephone-dn)</td>
</tr>
<tr>
<td>Configures the IP address and port number for an external SIP-based MWI server.</td>
<td>mwi sip-server (telephony-service)</td>
</tr>
<tr>
<td>Associates a telephone or extension number with an extension (ephone-dn) in a Cisco Unified CME system.</td>
<td>number</td>
</tr>
</tbody>
</table>
**mwi (voice register dn)**

To enable a specific Cisco IP phone extension (directory number) associated with a SIP phone to receive message-waiting indication (MWI) notification, use the `mwi` command in voice register dn configuration mode. To return to the default, use the `no` form of this command.

```
mwi
no mwi
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

This command is disabled

**Command Modes**

Voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables a particular extension on a SIP IP phone to receive MWI notification.

For Cisco Unified CME 4.1 and later versions, MWI requires that SIP phones must be configured with a directory number by using the `number` (voice register pool) command with the `dn` keyword; direct line numbers are not supported.

**Examples**

The following example shows how to enable MWI for a particular extension number associated with a SIP IP phone:

```
Router(config)# voice register dn 4
Router(config-register-dn)# mwi
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Configures number patterns for a voice register pool.</td>
</tr>
</tbody>
</table>

**number (voice register pool)**
**mwi expires**

To set the expiration timer for registration for the message-waiting indication (MWI) client or server, use the `mwi expires` command in telephony-service configuration mode. To disable the timer, use the `no` form of this command.

```
mwi expires seconds
no mwi expires seconds
```

**Syntax Description**

| seconds | Expiration time, in seconds. Range is from 600 to 99999. Default is 86400 (24 hours). |

**Command Default**

Default is 86400 seconds (24 hours).

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
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<td>This command was introduced.</td>
</tr>
<tr>
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<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following example sets the expiration timer to 1000 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# mwi expires 1000
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enables the Cisco CME router to relay MWI information to remote Cisco IP phones.</td>
</tr>
<tr>
<td>Configures the IP address and port number for the external SIP-based MWI server.</td>
</tr>
</tbody>
</table>
mwi prefix

To specify a prefix for an extension that will receive unsolicited message-waiting indication (MWI) from an external SIP-based MWI server, use the `mwi prefix` command in telephony-service configuration mode. To return to the default, use the `no` form of this command.

```plaintext
mwi prefix prefix-string
no mwi prefix
```

**Syntax Description**

- **prefix-string**: Digits at the beginning of a number that will be recognized as a prefix before a Cisco Unified CME extension number. The maximum prefix length is 32 digits.

**Command Default**

A prefix is not defined.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Central voice-messaging servers that provide mailboxes for several Cisco Unified CME sites may use site codes or prefixes to distinguish among similarly numbered ranges of extensions at different sites. In Cisco Unified CME 4.0 and later versions, you can specify that your Cisco Unified CME system should accept unsolicited SIP Notify messages for message-waiting indication (MWI) that include a prefix string as a site identifier.

For example, an MWI message might indicate that the central mailbox number 5551234 has a voice message. In this example, the digits 555 are set as the prefix string or site identifier using the `mwi prefix` command. The local Cisco Unified CME system is able to convert 5551234 to 1234 and deliver the MWI to the correct phone. Without this prefix string manipulation, the system would reject an MWI indication for 5551234 as not matching the local Cisco Unified CME extension 1234.

**Examples**

The following example identifies the SIP server for MWI notification at the IP address 172.16.14.22. It states that the Cisco Unified CME system will accept unsolicited SIP Notify messages for known mailbox numbers (extension numbers) that are prefixed with the digits 555.

```plaintext
sip-ua
mwi-server 172.16.14.22 unsolicited
telephony-service
mwi prefix 555
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Configures specific Cisco Unified IP phone directory numbers to receive MWI notification from an external voice-mail system.</td>
</tr>
<tr>
<td>Command</td>
</tr>
<tr>
<td>------------------</td>
</tr>
<tr>
<td><strong>mwi-server</strong></td>
</tr>
<tr>
<td><strong>mwi sip (ephone-dn)</strong></td>
</tr>
</tbody>
</table>
To enable Cisco Unified CME to interrogate a QSIG message center for the message-waiting indication (MWI) status of an IP phone extension, use the `mwi qsig` command in ephone-dn or ephone-dn-template configuration mode. To return to the default, use the `no` form of this command.

```
mwi qsig
no mwi qsig
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
An extension is not subscribed to receive MWI using QSIG.

**Command Modes**
Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
The `transfer-system` command must be used with the `full-consult` or `full-blind` keyword to enable H.450 call forwarding.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

**Examples**
In the following example, a voice mail extension (7000) and a normal extension (7582) are defined. Calls are forwarded to voice mail when extension 7582 is busy or does not answer. The message-waiting indicator (MWI) on extension 7582’s phone is subscribed to receive notifications from the QSIG message center.

```
ephone-dn 25
 number 7582
 mwi qsig
 call-forward busy 7000
call-forward noan 7000 timeout 20
telephony-service
 voicemail 7000
 transfer-system full-consult
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Applies a template to ephone-dn being configured.</td>
</tr>
<tr>
<td>Specifies the call transfer method for Cisco Unified CME extensions.</td>
</tr>
<tr>
<td>voicemail</td>
</tr>
</tbody>
</table>
mwi reg-e164

To register E.164 numbers rather than extension numbers with a Session Interface Protocol (SIP) proxy or registrar, use the mwi reg-e164 command in telephony-service configuration mode. To return to the default, use the no form of this command.

```
mwi reg-e164
no mwi reg-e164
```

**Syntax Description**

This command has no keywords or arguments.

**Command Default**

Registering extension numbers with the SIP proxy or registrar.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T7 12.4</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used when setting up extensions to use an external SIP-based message-waiting indication (MWI) server. The mwi-server command in SIP user-agent configuration mode specifies other settings for MWI service.

**Examples**

The following example specifies that E.164 numbers should be used for registration with the SIP proxy or registrar:

```
telephony-service
mwi reg-e164
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>mwi-server (SIP user-agent)</td>
</tr>
<tr>
<td>Specifies voice-mail server settings on a voice gateway or user agent (UA).</td>
</tr>
</tbody>
</table>
mwi relay

To enable a Cisco CallManager Express (Cisco CME) router to relay message-waiting indication (MWI) notification to remote Cisco IP phones, use the `mwi relay` command in telephony-service configuration mode. To disable MWI relay, use the `no` form of this command.

```
mwi relay
no mwi relay
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
MWI relay is disabled.

**Command Modes**
Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced command.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to enable the Cisco CME router to relay MWI notification to remote Cisco IP phones. The router at the central site acts as a notifier after this command is used.

**Examples**
The following example enables MWI relay:

```
Router(config)# telephony-service
Router(config-telephony)# mwi relay
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sets the expiration timer for registration for the client or the server.</td>
</tr>
<tr>
<td>Displays registration information for MWI relay clients.</td>
</tr>
</tbody>
</table>
**mwi sip**

To subscribe an extension in a Cisco Unified CME system to receive message-waiting indication (MWI) from a SIP-based MWI server, use the `mwi sip` command in ephone-dn or ephone-dn-template configuration mode. To remove the configuration, use the `no` form of this command.

```
mwi sip
no mwi sip
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

An extension is not subscribed to receive MWI.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-dn-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to subscribe an extension in a Cisco Unified CME router to receive MWI notification from a SIP-based MWI server, and use the `mwi sip-server` command to specify the IP address and port number for the external SIP-based MWI server. This function integrates a Cisco Unified CME router with a SIP-protocol-based MWI service.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

**Examples**

The following example subscribes extension 5001 to receive MWI notification from an external Session Initiation Protocol (SIP) MWI server and requests the SIP MWI server to send MWI notification messages through SIP to the Cisco Unified CME router for extension 5001:

```
Router(config) ephone-dn 1
Router(config-ephone-dn) number 5001
Router(config-ephone-dn) name MWI
Router(config-ephone-dn) mwi sip

Router(config) telephony-service
Router(config-telephony) mwi sip-server 172.30.0.5
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-dn</td>
<td>Enters ephone-dn configuration mode.</td>
</tr>
<tr>
<td>ephone-dn-template (ephone-dn)</td>
<td>Applies a template to an ephone-dn configuration.</td>
</tr>
<tr>
<td>mwi sip-server (telephony-service)</td>
<td>Configures the IP address and port number for the external SIP-based MWI server.</td>
</tr>
<tr>
<td>show mwi relay clients</td>
<td>Displays registration information for MWI relay clients.</td>
</tr>
</tbody>
</table>
**mwi sip-server**

To configure parameters associated with an external SIP-based message-waiting indication (MWI) server, use the `mwi sip-server` command in telephony-service configuration mode. To disable MWI server functionality, use the `no` form of this command.

```
mwi sip-server ip-address [{transport tcp|transport udp}] [port port-number] [reg-e164] [unsolicited [prefix prefix-string]]
no mwi sip-server ip-address [{transport tcp|transport udp}] [port port-number] [reg-e164] [unsolicited [prefix prefix-string]]
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ip-address</code></td>
<td>IP address of the MWI server.</td>
</tr>
<tr>
<td><code>transport tcp</code></td>
<td>(Optional) Selects TCP as the transport layer protocol. This is the default transport protocol.</td>
</tr>
<tr>
<td><code>transport udp</code></td>
<td>(Optional) Selects UDP as the transport layer protocol. The default if these keywords are not used is TCP.</td>
</tr>
<tr>
<td><code>port port-number</code></td>
<td>(Optional) Specifies port number for the MWI server. Range is from 2000 to 9999. Default is 5060 (SIP standard port).</td>
</tr>
<tr>
<td><code>reg-e164</code></td>
<td>(Optional) Registers an E.164 number with a Session Interface Protocol (SIP) proxy or registrar rather than an extension number. Registering with an extension number is the default.</td>
</tr>
<tr>
<td><code>unsolicited</code></td>
<td>(Optional) Sends SIP Notify message for MWI without any need to send a Subscribe message from the Cisco Unified CME router.</td>
</tr>
<tr>
<td><code>prefix prefix-string</code></td>
<td>(Optional) Allows the specified digits to be present before a recognized Cisco Unified CME extension number. The maximum prefix length is 32 digits.</td>
</tr>
</tbody>
</table>

**Command Default**

An external SIP-based MWI server is not defined.

**Command Modes**

Telephony-service configuration (config-telephony)

<table>
<thead>
<tr>
<th>Command History</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Cisco IOS Release</strong></td>
<td><strong>Cisco Product</strong></td>
</tr>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
</tr>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
</tr>
</tbody>
</table>
Usage Guidelines

Use this command to configure the IP address of an external SIP MWI server. This IP address is used with the `mwi sip` (ephone-dn) command to subscribe individual ephone-dn extension numbers to the notification list of the MWI SIP server. A SIP MWI client runs TCP by default.

The `transport tcp` keyword is the default setting. The `transport udp` keyword allows you to integrate with a SIP MWI client. The optional `port` keyword is used to specify a port number other than 5060, the default. The default registration is with an extension number, so the `reg-e164` keyword allows you to register with an E.164 ten-digit number.

Central voice-messaging servers that provide mailboxes for several Cisco Unified CME sites may use site codes or prefixes to distinguish among similarly numbered ranges of extensions at different sites. In Cisco CME 3.2.3 and later versions, you can specify that your Cisco Unified CME system should accept unsolicited SIP Notify messages for message-waiting indication (MWI) that include a prefix string as a site identifier.

Examples

The following example sets MWI for the SIP server and sets individual ephone-dn extension numbers to the MWI SIP server’s notification list:

```
Router(config) ephone-dn 1
Router(config-ephone-dn) number 5001
Router(config-ephone-dn) name Accounting
Router(config-ephone-dn) mwi sip
Router(config-ephone-dn) exit
Router(config) telephony-service
Router(config-telephony) mwi sip-server 192.168.0.5 transport udp
```

The following example identifies the SIP server for MWI notification at the IP address 172.16.14.22. It states that the Cisco Unified CME system will accept unsolicited SIP Notify messages that include the prefix 555 as a site identifier.

```
telephony-service
mwi sip-server 172.16.14.22 unsolicited prefix 555
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>mwi (ephone-dn)</td>
<td>Configures specific Cisco Unified IP phone directory numbers to receive MWI notification from an external voice-mail system.</td>
</tr>
<tr>
<td>mwi expires</td>
<td>Sets the expiration timer for registration for the client or the server.</td>
</tr>
<tr>
<td>mwi sip (ephone-dn)</td>
<td>Subscribes an extension in a Cisco Unified CME router to receive MWI notification from a SIP MWI server.</td>
</tr>
<tr>
<td>show mwi relay clients</td>
<td>Displays the registration information for MWI relay clients.</td>
</tr>
</tbody>
</table>
mwi stutter (voice register global)

To generate a stutter tone for message-waiting indication (MWI) in a Cisco CallManager Express (Cisco CME) system using SIP, use the `mwi stutter` command in voice register global configuration mode. To disable MWI stutter, use the `no` form of this command.

```
mwi stutter
no mwi stutter
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Stutter tone for MWI is disabled.

**Command Modes**
Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Examples**
The following example shows how to enable MWI stutter:

```
Router(config)# voice register global
Router(config-register-global)# mwi stutter
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>mwi reg-e164</code></td>
</tr>
</tbody>
</table>
mwi-line

To designate a line other than the primary line of an ephone to be associated with the ephone’s message waiting indicator (MWI) lamp, use the `mwi-line` command in ephone configuration mode. To return to the default, use the `no` form of this command.

```
mwi-line line-number
no mwi-line
```

**Syntax Description**

| line-number | Line number to be associated with the MWI lamp. Range is from 1 to 34. |

**Command Default**

A phone’s MWI lamp is lit only when there is a message waiting for the phone’s primary line (line 1).

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command designates a phone line other than the primary line to activate the MWI lamp on the phone. When a message is waiting for an ephone-dn associated with the designated line, the MWI lamp is turned on. When the message is heard, the MWI lamp is turned off. For phone lines other than the line that is designated to receive MWI, an envelope icon is displayed next to them when there is a message waiting.

Note that a logical phone “line” is not the same as a phone button. A line is a button that has one or more ephone-dns assigned to it. A button that has no ephone-dns assigned to it does not count as a line.

In most cases, one ephone-dn is assigned to one button on an ephone. When you set the `mwi-line` command to that button, the MWI lamp is turned on when there is a message waiting for that ephone-dn. When you set the `mwi-line` command to a button with a more complex configuration, the following rules apply:

- When a button has a single ephone-dn with primary and secondary numbers, the MWI lamp is turned on only when there is a message waiting for the primary number.
- When a button has several ephone-dns overlaid on it, the MWI lamp is turned on only when there is a message waiting for the first number in the list of ephone-dns.
- When a button is an overflow button for an overlay button, the MWI lamp is not turned on for any extension that might overflow to this button. If you set the `mwi-line` command to this button, the command is ignored.

**Examples**

The following example enables MWI on ephone 18 for line 2 (button 2), which has overlaid ephone-dns. The MWI lamp on this phone will be lit only if there is a message waiting for extension 2021. Button 4 is unused. The line numbers in this example are as follows:

- Line 1—Button 1—Extension 2020
- Line 2—Button 2—Extension 2021, 2022, 2023, 2024, 2025
- Line 3—Button 3—Extension 2021, 2022, 2023, 2024, 2025 (rollover line)
- Button 4—Unused
• Line 4—Button 5—Extension 2026

ephone-dn 20  
number 2020  
ephone-dn 21  
number 2021  
ephone-dn 22  
number 2022  
ephone-dn 23  
number 2023  
ephone-dn 24  
number 2024  
ephone-dn 25  
number 2025  
ephone-dn 26  
number 2026  
ephone 18  
button 1:20 2o21,22,23,24,25 3x2 5:26  
mwi-line 2

The following example enables MWI on ephone 17 for line 3 (extension 609). In this case, the button numbers do not match the line numbers because buttons 2 and 4 are not used.

• Line 1—Button 1—Extension 607  
• Button 2—Unused  
• Line 2—Button 3—Extension 608  
• Button 4—Unused  
• Line 3—Button 5—Extension 609

ephone-dn 17  
number 607  
ephone-dn 18  
number 608  
ephone-dn 19  
number 609  
ephone 25  
button 1:17 3:18 5:19  
mwi-line 3

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>button</td>
<td>Associates ephone-dns with individual buttons on an SCCP phone and to specify line type or ring behavior.</td>
</tr>
</tbody>
</table>
mwi-type

To specify the type of message-waiting indication (MWI) notification that a directory number can receive and process, use the `mwi-type` command in `ephone-dn` or `ephone-dn-template` configuration mode. To disable this feature, use the `no` form of this command.

```
mwi-type \{visual\|audio\|both\}
nm mwi-type \{visual\|audio\|both\}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>visual</td>
<td>Sets a directory number to process visual MWI, using either the main or secondary phone number.</td>
</tr>
<tr>
<td>audio</td>
<td>Sets a directory number to process audible MWI (AMWI), using either the main or secondary phone number.</td>
</tr>
<tr>
<td>both</td>
<td>Sets a directory number to process both visual and audible MWI, using either the main or secondary phone number.</td>
</tr>
</tbody>
</table>

**Command Default**

If MWI is enabled for a directory number, directory number will receive visual MWI.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(6)XE</td>
<td>Cisco Unified CME 4.0(2)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC4</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)T</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>This command was integrated into Cisco IOS Release 12.4(11)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables a directory number to receive audible, visual, or both audible and visual MWI notification from an external voice-messaging system. The external voice-messaging system is able to communicate MWI status by making telephone calls to the dummy extension, with the MWI information embedded in either the called or calling-party IP phone number.

Based on the capabilities of the IP phone and how the `mwi-type` command is configured, Message Waiting is communicated as follows:

- If the phone supports (visual) MWI and MWI is configured for the phone, Message Waiting light is lit.
- If the phone supports (visual) MWI only, Message Waiting light is lit regardless of the configuration.
- If the phone supports AMWI and AMWI is configured for the phone, stutter dial tone is sent to the phone when it goes off-hook.
- If the phone supports AMWI only and AMWI is configured, stutter dial tone is sent to the phone when it goes off-hook regardless of the configuration.
- If a phone supports (visual) MWI and AMWI and both options are configured for the phone, the Message Waiting light is lit and a stutter dial tone is sent to the phone when it goes off-hook.

Before using this command:

- Create the directory number to be configured by using the `number`
Enable MWI on this directory number by using the `mwi` command.

If you use an ephone-dn template to apply a command to a directory number and you also use the same command in ephone-dn configuration mode for the same number, the value that you set in ephone-dn configuration mode has priority.

### Examples

The following example shows how to enable AMWI on extension 8000, assuming that the phone to which this directory number is assigned supports AMWI. Otherwise, a call placed by the voice-mail system to 8001 turns on the MWI light for the extension number indicated by the calling-party number for the MWI call.

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) number 8000
Router(config-ephone-dn) MWI on
Router(config-ephone-dn) MWI-type audible
```

The following example shows how to enable both audible and visual MWI. A call placed by the voice-mail system to 8001 turns on the MWI light for the extension number indicated by the calling-party number for the MWI call. When the phone user takes the phone off hook, they hear a stutter dial tone:

```
Router(config)# ephone-dn 2
Router(config-ephone-dn) number 8001
Router(config-ephone-dn) MWI on
Router(config-ephone-dn) MWI-type both
```

The following example shows how to use an ephone-dn-template to set MWI type:

```
Router(config)# ephone-dn-template 4
Router(config-ephone-dn-template) MWI-type both
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 8000
Router(config-ephone-dn)# ephone-dn-template 4
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-dn-template (ephone-dn)</td>
<td>Applies a template to an ephone-dn configuration.</td>
</tr>
<tr>
<td>mwi (ephone and ephone template)</td>
<td>Enables a directory number to receive MWI.</td>
</tr>
<tr>
<td>number</td>
<td>Associates a telephone or extension number with a directory number in a Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
Cisco Unified CME Commands: N

- name (ephone-dn), on page 652
- name (ephone-hunt), on page 654
- name (voice emergency response location), on page 656
- name (voice hunt-group), on page 657
- name (voice register dn), on page 659
- network-locale (ephone-template), on page 660
- network-locale (telephony-service), on page 662
- network-locale (voice-gateway), on page 667
- night-service bell, on page 669
- night-service bell (ephone-dn), on page 671
- night-service code, on page 673
- night-service date, on page 675
- night-service day, on page 677
- night-service everyday, on page 679
- night-service weekday, on page 681
- night-service weekend, on page 683
- no-reg, on page 685
- no-reg (voice register dn), on page 687
- nte-end-digit-delay, on page 688
- ntp-server, on page 690
- number (ephone-dn), on page 691
  - night-service bell (voice register dn), on page 694
  - night-service bell (voice register pool), on page 696
- number (voice register template), on page 698
- number (voice register dn), on page 700
- number (voice register pool), on page 702
- number (voice user-profile and voice logout-profile), on page 704
- num-buttons, on page 708
- num-line, on page 710
name (ephone-dn)

To associate a name with a directory number in Cisco Unified CME, use the `name` command in ephone-dn configuration mode. To disassociate a name from an extension, use the `no` form of this command.

```
name  name
no   name
```

**Syntax Description**

<table>
<thead>
<tr>
<th>name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>name</td>
<td>Alphanumeric string of person or group associated with a directory number. Name must follow the order specified in the <code>directory (telephony-service)</code> command, either <code>first-name-first</code> or <code>last-name-first</code>. The two parts, first and last name or last and first name, of this argument must be separated with a space.</td>
</tr>
</tbody>
</table>

**Command Default**

This command has no default behavior or values.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `name` argument is used to provide caller ID for calls originating from a directory number in Cisco Unified CME and also generates local directory information that is accessed by using the Directories button on a Cisco IP phone.

The `name` argument combination must match the order specified in the `directory (telephony-service)` command, either `first-name-first` or `last-name-first`.

The `name` string must contain a space between the first and second parts of the string (first last or last first).

The `name` string cannot contain special characters such as an ampersand (&). The only special characters supported in the name string are the comma (,) and the percent sign (%).

To display a comma between the last and first names when the pattern is last-name-first, add a comma (,) to the end of the first part of the `name` string (last name), for example: last, first.

The second part of the `name` string can contain spaces, such as “and Handling.”

**Examples**

The following example configures the username John Smith with the pattern `first-name-first`:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) name John Smith
```

The following example configures the username Shipping and Handling with the pattern `first-name-first`:

```
Router(config)# ephone-dn 1
```

Cisco Unified Communications Manager Express Command Reference
The following example configures the username Jane Smith with the pattern `last-name-first` and with a comma:

```plaintext
Router(config)# ephone-dn 1
Router(config-ephone-dn) name Smith, Jane
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>directory (telephony-service)</td>
<td>Defines the name order for the local directory of Cisco IP phone users.</td>
</tr>
</tbody>
</table>
name (ephone-hunt)

To associate a name with a called voice hunt group, use the name command in ephone-hunt configuration mode. To dissociate the name of the called voice hunt group, use the no form of this command.

**name** "primary pilot name" [secondary "secondary "secondary pilot name"]

**no name** "primary pilot name" [secondary "secondary "secondary pilot name"]

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>&quot;primary pilot name&quot;</td>
<td>Name of primary pilot number.</td>
</tr>
<tr>
<td>secondary &quot;secondary pilot name&quot;</td>
<td>(Optional) Name of secondary pilot number.</td>
</tr>
</tbody>
</table>

**Command Default**

No name is associated with the called voice hunt group.

**Command Modes**

Ephone-hunt configuration (config-ephone-hunt)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

In Cisco Unified CME 9.5 and Cisco Unified SRST 9.5, when the secondary pilot name is not explicitly configured, the primary pilot name is applicable to both pilot numbers.

**Note**

Use quotes (" ) when input strings have spaces in between.

**Examples**

The following example configures the primary pilot name for both the primary and the secondary pilot numbers:

```
name SALES
```

The following example configures different names for the primary and secondary pilot numbers:

```
name SALES secondary SALES-SECONDARY
```

The following example associates a two-word name for the primary pilot number and a one-word name for the secondary pilot number:

```
name "CUSTOMER SERVICE" secondary CS
```

The following example associates a one-word name for the primary pilot number and a two-word name for the secondary pilot number:

```
name FINANCE secondary "INTERNAL ACCOUNTING"
```

The following example associates two-word names for the primary pilot number and the secondary pilot number:
name “INTERNAL CALLER” secondary “EXTERNAL CALLER”

When incoming call A reaches voice hunt group B and lands on final C, extension C does not show the name of the forwarder because the voice hunt group is not configured to display the name. To display the name of the forwarder and the final number, two separate names are required for the primary and secondary pilot numbers.

The following is a sample output of the show run command when the primary and secondary pilot names are configured in ephone-hunt configuration mode:

```plaintext
ephone-hunt 10 sequential
   pilot 1010 secondary 1020
   list 2004, 2005
   final 2006
   timeout 8, 8
   name “EHUNT PRIMARY” secondary “EHUNT SECONDARY”
ephone-hunt 11 peer
   pilot 1012 secondary 1022
   list 2004, 2005
   final 2006
   timeout 8, 8
   name EHUNT1 secondary EHUNT1-SEC
```

The following is a sample output of the show ephone-hunt command when the primary and secondary pilot names are configured in ephone-hunt configuration mode:

```plaintext
show ephone-hunt 10
Group 10
   type: sequential
   pilot number: 1010, peer-tag 20010
   pilot name: EHUNT PRIMARY
   secondary number: 1020, peer-tag 20011
   secondary name: EHUNT SECONDARY
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>voice hunt-group</code></td>
<td>Enters voice hunt-group configuration mode and creates a hunt group for phones in a Cisco Unified CME system.</td>
</tr>
<tr>
<td><code>show voice hunt-group</code></td>
<td>Displays configuration information associated with one or all voice hunt groups in a Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
name (voice emergency response location)

To describe or identify an emergency response location, use the `name` command in voice emergency response location mode. To remove this definition, use the `no` form of this command.

```
name string
no name
```

**Syntax Description**

- **string**: String (30 characters) used to describe or identify an ERL’s location.

**Command Default**

The location is not described.

**Command Modes**

Voice emergency response location configuration (cfg-emrgncy-resp-location)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to enable a word or description of the ERL for administrative purposes. The most common use of this command is to identify the location for the network administrator.

**Examples**

In this example, the location description is Your Company Incorporated.

```
voice emergency response location 60
subnet 1 209.165.200.224 255.255.0.0
elin 1 4085550101
name Your Company Incorporated,
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>address</td>
<td>Specifies a comma separated text entry (up to 247 characters) of an ERL’s civic address.</td>
</tr>
<tr>
<td>elin</td>
<td>Specifies a PSTN number to replace the caller’s extension.</td>
</tr>
<tr>
<td>subnet</td>
<td>Defines which IP phones are part of this ERL.</td>
</tr>
<tr>
<td>voice emergency response location</td>
<td>Creates a tag for identifying an ERL for E911 services.</td>
</tr>
</tbody>
</table>
name (voice hunt-group)

To associate a name with a called voice hunt group, use the `name` command in voice hunt-group configuration mode. To dissociate the name of the called voice hunt group, use the `no` form of this command.

```
name "primary pilot name" [secondary "secondary pilot name"]
no name "primary pilot name" [secondary "secondary pilot name"]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>&quot;primary pilot name&quot;</td>
<td>Name of primary pilot number.</td>
</tr>
<tr>
<td>secondary &quot;secondary pilot name&quot;</td>
<td>(Optional) Name of secondary pilot number.</td>
</tr>
</tbody>
</table>

**Command Default**

No name is associated with the called voice hunt group.

**Command Modes**

Voice hunt-group configuration (config-voice-hunt-group)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

In Cisco Unified CME 9.5 and Cisco Unified SRST 9.5, when the secondary pilot name is not explicitly configured, the primary pilot name is applicable to both pilot numbers.

**Note**

Use quotes (") when input strings have spaces in between.

**Examples**

The following example configures the primary pilot name for both the primary and the secondary pilot numbers:

```
name SALES
```

The following example configures different names for the primary and secondary pilot numbers:

```
name SALES secondary SALES-SECONDARY
```

The following example associates a two-word name for the primary pilot number and a one-word name for the secondary pilot number:

```
name "CUSTOMER SERVICE" secondary CS
```

The following example associates a one-word name for the primary pilot number and a two-word name for the secondary pilot number:

```
name FINANCE secondary "INTERNAL ACCOUNTING"
```

The following example associates two-word names for the primary and secondary pilot numbers:

```
name "INTERNAL CALLER" secondary "EXTERNAL CALLER"
```
When incoming call A reaches voice hunt group B and lands on final C, extension C does not show
the name of the forwarder because the voice hunt group is not configured to display the name. To
display the name of the forwarder and the final number, two separate names are required for the
primary and secondary pilots.

The following example shows how the primary and secondary pilot names are configured in voice
hunt-group configuration mode:

```
voice hunt-group 24 parallel
   final 097
   list 885,886,124,154
   timeout 20
   pilot 021 secondary 621
   name SALES secondary SALES-SECONDARY
```

The following is a sample output of the `show voice hunt-group` command when the primary and
secondary pilot names are configured in voice hunt-group configuration mode:

```
show voice hunt-group 1
Group 1
   type: parallel
   pilot number: 1000, peer-tag 2147483647
   secondary number: 2000, peer-tag 2147483646
   pilot name: SALES
   secondary name: SALES-SECONDARY
   list of numbers: 2004,2005
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>voice hunt-group</code></td>
<td>Enters voice hunt-group configuration mode and creates a hunt group for phones in a Cisco Unified CME system.</td>
</tr>
<tr>
<td><code>show voice hunt-group</code></td>
<td>Displays configuration information associated with one or all voice hunt groups in a Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
name (voice register dn)

To associate a name with a directory number in Cisco Unified CME, use the name command in voice register dn configuration mode. To disassociate a name from an extension, use the no form of this command.

**name name**

**no name**

<table>
<thead>
<tr>
<th>Syntax Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>name</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Command Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>This command has no default behavior or values.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Command Modes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice register dn configuration (config-register-dn)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Command History</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Cisco IOS Release</strong></td>
</tr>
<tr>
<td>12.4(4)T</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The name argument is used to provide caller ID for calls originating from a Cisco CME extension. This command also generates directory information for the local directory that is accessed from the Directories button on a Cisco IP phone.

**Examples**

The following example shows how to configure the username John Smith with the pattern first-name-first:

```
Router(config)# voice register dn 1
Router(config-register-dn) name John Smith
```

The following example shows how to configure the username Jane Smith with the pattern last-name-first:

```
Router(config)# voice register dn 1
Router(config-register-dn) name Smith, Jane
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>directory (telephony-service)</td>
</tr>
</tbody>
</table>
network-locale (ephone-template)

To specify a network locale in an ephone template, use the `network-locale` command in ephone-template configuration mode. To reset to the default network locale, use the `no` form of this command.

```
network-locale network-locale-tag
no network-locale
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>network-locale-tag</code></td>
<td>Locale identifier that was assigned to a network locale using the <code>network-locale</code> (telephony-service) command.</td>
</tr>
</tbody>
</table>

**Command Default**

The default network locale (network locale 0) is used.

**Command Modes**

Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

To apply network locales to individual ephones, you must specify per-phone configuration files using the `cnf-file perphone` command and identify the locales using the `network-locale (telephony-service) command`. After creating an ephone template that contains a locale tag, use the `ephone-template (ephone)` command to apply the template to individual ephones.

**Examples**

The following example defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales, ephone 12 uses FR, ephone 13 uses ES, and ephone 14 uses the default, US.

```
telephony-service
cnf-file location flash:
cnf-file perphone
create cnf-files
user-locale 1 JP
user-locale 2 FR
user-locale 3 ES
network-locale 1 JP
network-locale 2 FR
network-locale 3 ES
ephone-template 1
user-locale 1
network-locale 1
ephone-template 2
user-locale 2
network-locale 2
ephone-template 3
user-locale 3
network-locale 3
ephone 11
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>cnf-file</td>
<td>Specifies the type of configuration files that phones use.</td>
</tr>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone.</td>
</tr>
<tr>
<td>network-locale (telephony-service)</td>
<td>Sets the locale for geographically specific tones and cadences.</td>
</tr>
</tbody>
</table>
network-locale (telephony-service)

To select a code for a geographically specific set of tones and cadences on supported phone types, use the `network-locale` command in telephony-service configuration mode. To disable selection of a code, use the `no` form of this command.

```
network-locale [network-locale-tag] [user-defined-code] locale-code
no network-locale network-locale-tag
```

**Syntax Description**

<table>
<thead>
<tr>
<th>network-locale-tag</th>
<th>(Optional) Assigns a locale identifier to the locale code. Range is 0 to 4. Default is 0.</th>
</tr>
</thead>
<tbody>
<tr>
<td>user-defined code</td>
<td>(Optional) Assigns one of the user-defined codes to the specified locale code. Valid codes are U1, U2, U3, U4, and U5. There is no default.</td>
</tr>
<tr>
<td>locale-code</td>
<td>Locale files for the following ISO 3166 codes are predefined in system storage for supported phone types:</td>
</tr>
<tr>
<td></td>
<td>• AT—Austria</td>
</tr>
<tr>
<td></td>
<td>• CA—Canada</td>
</tr>
<tr>
<td></td>
<td>• CH—Switzerland</td>
</tr>
<tr>
<td></td>
<td>• DE—Germany</td>
</tr>
<tr>
<td></td>
<td>• DK—Denmark</td>
</tr>
<tr>
<td></td>
<td>• ES—Spain</td>
</tr>
<tr>
<td></td>
<td>• FR—France</td>
</tr>
<tr>
<td></td>
<td>• GB—United Kingdom</td>
</tr>
<tr>
<td></td>
<td>• IT—Italy</td>
</tr>
<tr>
<td></td>
<td>• JP—Japan</td>
</tr>
<tr>
<td></td>
<td>• NL—Netherlands</td>
</tr>
<tr>
<td></td>
<td>• NO—Norway</td>
</tr>
<tr>
<td></td>
<td>• PT—Portugal</td>
</tr>
<tr>
<td></td>
<td>• RU—Russian Federation</td>
</tr>
<tr>
<td></td>
<td>• SE—Sweden</td>
</tr>
<tr>
<td></td>
<td>• US—United States (default)</td>
</tr>
</tbody>
</table>

**Note**

You can also assign any valid ISO 3166 code that is not listed above to a user-defined code (U1 through U5), but you must first copy the appropriate XML tone files to flash, slot 0, or an external TFTP server and use the `cnf-files perphone` command to specify the use of per-phone configuration files.

**Command Default**

The default locale code is **US** (United States).

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>
This command was integrated into Cisco IOS Release 12.2(15)T.

The `network-locale-tag` and `user-defined-code` arguments were added.

The `network-locale-tag` and `user-defined-code` arguments were integrated into Cisco IOS Release 12.4(9)T.

### Usage Guidelines

This command designates a a network locale other than US as the locale for one or more phones in Cisco Unified CME.

Network locale 0 always holds the default locale, which is used for all phones that are not assigned alternative network locales or user-defined network locales. You can use this command to change the default locale.

The `show telephony-service tftp-bindings` command displays the locale-specific call-progress tone files that are accessible to IP phones using TFTP.

This command must be followed by a complete phone reboot using the `reset` command.

### Alternative Network Locales

The `network-locale-tag` argument allows you to specify up to five alternative network locales for use in a system using Cisco Unified CME 4.0 or a later release. For example, a company can specify network-locale France for phones A, B, and C; network-locale Germany for phones D, E, and F; and network-locale United States for phones G, H, and I.

Each one of the five alternative network locales that you can use in a multi-locale system is identified with a locale tag identifier. The identifier 0 always holds the default locale, although you can define this default to be any locale code that is supported in the system and is listed in the CLI help for the command. For example, if you define network locale 0 to be JP (Japanese), the default network locale for the router is JP. If you do not specify a locale for the identifier 0, the default is US (United States).

To apply alternative network locales to different phones, you must use the `cnf-files` command to specify per-phone configuration files. When you use per-phone configuration files, a phone's configuration file automatically uses the default locales in user locale 0 and network locale 0. You can override this default for individual ephones by assigning alternative locale tag identifiers to the alternative locale codes that you want to use and then creating ephone templates to assign the locale tag identifiers to individual ephones. For example, you can give the alternative locale tag of 2 to the locale code DK (Denmark).

After using the `network-locale (telephony-service)` command to associate a locale tag identifier with a locale code, use the `network-locale` command in ephone-template mode to apply the locale tag to an ephone template. Then use the `ephone-template` command in ephone configuration mode to apply the template to the ephones that should use the alternative network locale.

### User-Defined Network Locales

XML files for user locales and network locales that are not currently provided in the system must be downloaded to use this feature. Beginning in Cisco Unified CME 4.0, you can install the files to support a particular user and network locale in flash, slot 0, or an external TFTP server. You cannot install these files in the system location. These user-locale and network-locale files can then be used as default or alternative locales for all or some phones.
For example, if you have a site at which the phones should use the displays and tones for Traditional Chinese, which is not one of the predefined locales, you must download and install the XML files for Traditional Chinese on the phones that need to use this locale.

### Examples

The following example sets the default locale tag 0 to France:

```plaintext
telephony-service
  network-locale FR
```

The following example sets the default locale tag 0 to France. It shows another way to change the default network locale:

```plaintext
telephony-service
  network-locale 0 FR
```

The following example sets the alternative locale tag 1 to Germany:

```plaintext
telephony-service
  network-locale 1 DE
```

### Alternative Network Locale Example

The following example defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales, ephone 12 uses FR, ephone 13 uses ES, and ephone 14 uses the default, US.

```plaintext
telephony-service
  cnf-file location flash:
    cnf-file perphone
    user-locale 1 JP
    user-locale 2 FR
    user-locale 3 ES
    network-locale 1 JP
    network-locale 2 FR
    network-locale 3 ES
    create cnf-files
    ephone-template 1
      user-locale 1
      network-locale 1
    ephone-template 2
      user-locale 2
      network-locale 2
    ephone-template 3
      user-locale 3
      network-locale 3
    ephone 11
      button 1:25
      ephone-template 1
    ephone 12
      button 1:26
      ephone-template 2
    ephone 13
      button 1:27
      ephone-template 3
    ephone 14
```
User-Defined Network Locale Example

The following example applies the alternative locale tag 4 to the user-defined code U1, which is defined as ZH. ZH is the code that represents Traditional Chinese in ISO 639, the Language Code Reference. Because the code for Traditional Chinese is not one of those is provided in the system, the user must download the appropriate XML files to support this language.

In addition to the user-defined code, the example also defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales; ephone 12 uses FR; ephone 13 uses ES; ephone 14 uses the default, US; and ephone 15 uses the user-defined language, ZH (Traditional Chinese).

telephony-service
cnf-file location flash:
cnf-file perphone
user-locale 1 JP
user-locale 2 FR
user-locale 3 ES
user-locale 4 U1 ZH
network-locale 1 JP
network-locale 2 FR
network-locale 3 ES
network-locale 4 U1 ZH
create cnf-files
ephone-template 1
  user-locale 1
  network-locale 1
ephone-template 2
  user-locale 2
  network-locale 2
ephone-template 3
  user-locale 3
  network-locale 3
ephone-template 4
  user-locale 4
  network-locale 4
ephone 11
  button 1:25
  ephone-template 1
ephone 12
  button 1:26
  ephone-template 2
ephone 13
  button 1:27
  ephone-template 3
ephone 14
  button 1:28
  ephone 15
  button 1:29
  ephone-template 4
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>cnf-files</td>
<td>Specifies the type of phone configuration files to be created.</td>
</tr>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone.</td>
</tr>
<tr>
<td>network-locale (ephone-template)</td>
<td>Applies a locale tag identifier to an ephone template.</td>
</tr>
<tr>
<td>reset (ephone)</td>
<td>Performs a complete reboot of one phone associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td>reset (telephony-service)</td>
<td>Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td>show telephony-service tftp-bindings</td>
<td>Displays the current configuration files that are accessible to IP phones.</td>
</tr>
<tr>
<td>user-locale (telephony-service)</td>
<td>Sets the language for displays on supported phone types.</td>
</tr>
</tbody>
</table>
network-locale (voice-gateway)

To select a geographically specific set of tones and cadences for the voice gateway’s analog endpoints that register to Cisco Unified CME, use the `network-locale` command in voice-gateway configuration mode. To remove a code, use the `no` form of this command.

```
network-locale country-code
no network-locale country-code
```

**Syntax Description**

- **country-code**
  - The following ISO 3166 country codes are supported:
  - AE—United Arab Emirates
  - AR—Argentina
  - AT—Austria
  - AU—Australia
  - BE—Belgium
  - BR—Brazil
  - CA—Canada
  - CH—Switzerland
  - CN—China
  - CO—Colombia
  - CY—Cyprus
  - CZ—Czech Republic
  - DE—Germany
  - DK—Denmark
  - EG—Egypt
  - ES—Spain
  - FI—Finland
  - FR—France
  - GB—United Kingdom
  - GH—Ghana
  - GR—Greece
  - HK—Hong Kong
  - HU—Hungary
  - ID—Indonesia
  - IE—Ireland
  - IL—Israel
  - IN—India
  - IS—Iceland
  - IT—Italy
  - JO—Jordan
  - JP—Japan
  - KE—Kenya
  - KR—Korea Republic
  - KW—Kuwait
  - LB—Lebanon
  - LU—Luxembourg
  - MX—Mexico
  - MY—Malaysia
  - NG—Nigeria
  - NL—Netherlands
  - NO—Norway
  - NP—Nepal
  - NZ—New Zealand
  - OM—Oman
  - PA—Panama
  - PE—Peru
  - PH—Philippines
  - PK—Pakistan
  - PL—Poland
  - PT—Portugal
  - RU—Russian Federation
  - SA—Saudi Arabia
  - SE—Sweden
  - SG—Singapore
  - SI—Slovenia
  - SK—Slovakia
  - TH—Thailand
  - TR—Turkey
  - TW—Taiwan
  - US—United States (default)
  - VE—Venezuela
  - ZA—South Africa
  - ZW—Zimbabwe

**Command Default**

The default locale code is **US** (United States).

**Command Modes**

Voice-gateway configuration (config-voice-gateway)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>
Modification
Cisco Product Cisco IOS Release
12.4(24)T Cisco Unified CME 7.1 This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines
This command designates a network locale other than US as the locale for the analog endpoints registered to Cisco Unified CME. All voice ports are assigned the same network locale. If you want a different network locale on a specific phone, use the `cptone` command in voice-port configuration mode.

The `show telephony-service tftp-bindings` command displays the locale-specific call-progress tone files that are accessible to IP phones using TFTP.

After using this command, you must reboot the phones with the `reset` command.

Examples
The following example shows a voice gateway configuration where the network locale is set to France:

```
voice-gateway system 1
  network-locale FR
  type VG224
  mac-address 001F.A30F.8331
  voice-port 0-23
  create cnf-files
```

Related Commands
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>cptone</code></td>
<td>Specifies a regional analog voice-interface-related tone, ring, and cadence setting.</td>
</tr>
<tr>
<td><code>reset (voice-gateway)</code></td>
<td>Performs a complete reboot of all analog phones associated with the voice gateway and registered to Cisco Unified CME.</td>
</tr>
<tr>
<td><code>show telephony-service tftp-bindings</code></td>
<td>Displays the current configuration files accessible to IP phones.</td>
</tr>
<tr>
<td><code>voice-port (voice-gateway)</code></td>
<td>Identifies the ports on the voice gateway that will register to Cisco Unified CME.</td>
</tr>
</tbody>
</table>
To mark an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods, use the `night-service bell` command in ephone or ephone-template configuration mode. To remove night-service notification capability from a phone, use the `no` form of this command.

```
night-service bell
no night-service bell
```

### Syntax Description
This command has no arguments or keywords.

### Command Default
A phone is not marked for night-service bell notification.

### Command Modes
- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)

### Command History
<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines
When an ephone-dn is marked for night-service treatment using the `night-service bell` (ephone-dn) command, incoming calls that ring during the night-service time period on that ephone-dn send an alert indication to all IP phones that are marked to receive night-service bell notification with this command. The alert notification is in the form of a splash ring (not associated with any of the individual lines on the IP phone) and a visible display of the ephone-dn extension number. The phone user retrieves the call by pressing a PickUp or GPickUp soft key and dialing the appropriate digits.

Night-service periods are defined using the `night-service date` and `night-service day` commands. Night service can be manually disabled or reenabled from a phone configured with ephone-dns in night-service mode if the `night-service code` command has been set.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

### Examples
The following example designates the IP phone that is being configured as a phone that will receive night-service bell notification when ephone-dns marked for night service receive incoming calls during a night-service period:

```
Router(config)# ephone 4
Router(config-ephone)# night-service bell
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>ephone-template (ephone)</strong></td>
<td>Applies a template to an ephone configuration.</td>
</tr>
<tr>
<td><strong>night-service bell (ephone-dn)</strong></td>
<td>Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.</td>
</tr>
<tr>
<td><strong>night-service code</strong></td>
<td>Defines a code to disable or reenable night service on IP phones.</td>
</tr>
<tr>
<td><strong>night-service date</strong></td>
<td>Defines a recurring time period associated with a month and day during which night service is active.</td>
</tr>
<tr>
<td><strong>night-service day</strong></td>
<td>Defines a recurring time period associated with a day of the week during which night service is active.</td>
</tr>
</tbody>
</table>
# night-service bell (ephone-dn)

To mark an ephone-dn for night-service treatment, use the `night-service bell` command in ephone-dn configuration mode. To remove the night-service treatment from the ephone-dn, use the `no` form of this command.

```plaintext
night-service bell
no night-service bell
```

## Syntax Description
This command has no arguments or keywords.

## Command Default
An ephone-dn is not marked for night service.

## Command Modes
Ephone-dn configuration (config-ephone-dn)

## Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

## Usage Guidelines
When an ephone-dn is marked for night-service treatment using this command, incoming calls that ring during the night-service time period on that ephone-dn send an alert indication to all IP phones that are marked to receive night-service bell notification using the `night-service bell (ephone)` command. The alert notification is in the form of a splash ring (not associated with any of the individual lines on the IP phone) and a visible display of the ephone-dn extension number. The phone user retrieves the call by pressing a PickUp or GPickUp soft key and dialing the appropriate digits.

Night-service periods are defined using the `night-service date` and `night-service day` commands. Night service can be manually disabled or enabled from a phone configured with ephone-dns in night-service mode if the `night-service code` command has been set.

## Examples
The following example marks an ephone-dn as a line that will ring on IP phones designated to receive night-service bell notification when incoming calls are received on this ephone-dn during night-service periods:

```plaintext
Router(config) # ephone-dn 16
Router(config-ephone-dn) # night-service bell
```

## Related Commands

<table>
<thead>
<tr>
<th></th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>night-service bell (ephone)</code></td>
<td>Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.</td>
</tr>
<tr>
<td><code>night-service code</code></td>
<td>Defines a code to disable or reenable night service on IP phones.</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>---------------------------------------------------------------------------</td>
<td></td>
</tr>
<tr>
<td><strong>night-service date</strong></td>
<td></td>
</tr>
<tr>
<td>Defines a recurring time period associated with a month and day during</td>
<td></td>
</tr>
<tr>
<td>which night service is active.</td>
<td></td>
</tr>
<tr>
<td><strong>night-service day</strong></td>
<td></td>
</tr>
<tr>
<td>Defines a recurring time period associated with a day of the week during</td>
<td></td>
</tr>
<tr>
<td>which night service is active.</td>
<td></td>
</tr>
</tbody>
</table>
night-service code

To define a code to disable or reenable night service on IP phones, use the `night-service code` command in telephony-service configuration mode. To remove the code, use the `no` form of this command.

```
night-service code  digit-string
no  night-service code  digit-string
```

**Syntax Description**

| digit-string | Digit code that a user enters at an IP phone to disable or reenable night service. The code must begin with an asterisk (*). The maximum number of characters is 16, including the asterisk. |

**Command Default**

No code is defined.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>The action of this command was changed so that all night-service ephone-dns are activated or deactivated when the code is used rather than just the phone on which the code is input.</td>
</tr>
<tr>
<td>15.6(3)M</td>
<td>Cisco Unified CME 11.5</td>
<td>From Unified CME 11.5 onwards, this command can be used to activate or deactivate the night service from SIP phones as well.</td>
</tr>
<tr>
<td>16.3.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Night-service periods are defined with the `night-service date` and `night-service day` commands. When a dn is marked for night-service treatment using the `night-service bell` (ephone-dn or voice register dn) command, incoming calls that ring during the night-service time period on that dn send an alert indication to all IP phones that are marked to receive night-service bell notification using the `night-service bell` command. The alert notification is in the form of a burst ring for SCCP phones and message waiting tone for SIP phones (not associated with any of the individual lines on the IP phone). There is a visible display of the dn extension number. The phone user retrieves the call by pressing a PickUp or GPickUp soft key and dialing the appropriate digits.

When a night-service code has been defined using the `night-service code` command, night service for all night-service dns can be manually activated or deactivated from any phone that is configured with a night-service dn.

**Examples**

The following example defines a night-service code of *2985:

```
Router(config)# telephony-service
Router(config-telephony)# night-service code *2985
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>night-service bell (ephone)</strong></td>
<td>Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.</td>
</tr>
<tr>
<td><strong>night-service bell (ephone-dn)</strong></td>
<td>Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.</td>
</tr>
<tr>
<td><strong>night-service bell (voice register pool)</strong></td>
<td>Marks a SIP phone to receive night-service bell notification when incoming calls are received on voice register DNs that are marked for night service during night-service time periods.</td>
</tr>
<tr>
<td><strong>night-service bell (voice register dn)</strong></td>
<td>Marks a voice register dn to send night-service bell notification to designated SIP phones during night-service time periods.</td>
</tr>
<tr>
<td><strong>night-service bell (voice register template)</strong></td>
<td>Applies a template to a pool configuration.</td>
</tr>
<tr>
<td><strong>night-service date</strong></td>
<td>Defines a recurring time period associated with a month and day during which night service is active.</td>
</tr>
<tr>
<td><strong>night-service day</strong></td>
<td>Defines a recurring time period associated with a day of the week during which night service is active.</td>
</tr>
</tbody>
</table>
night-service date

To define a recurring time period associated with a date during which night service is active, use the **night-service date** command in telephony-service configuration mode. To delete the defined time period, use the **no** form of this command.

```
night-service date month day start-time stop-time
no night-service date month day start-time stop-time
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>month</strong></td>
<td>Abbreviated month. The following abbreviations for month are valid: jan, feb, mar, apr, may, jun, jul, aug, sep, oct, nov, dec.</td>
</tr>
<tr>
<td><strong>day</strong></td>
<td>Day of the month. Range is from 1 to 31.</td>
</tr>
<tr>
<td><strong>start-time stop-time</strong></td>
<td>Beginning and ending times for night service, in an HH:MM format using a 24-hour clock. The stop time must be greater than the start time. The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, night service is in effect for the entire 24-hour period on the specified date.</td>
</tr>
</tbody>
</table>

### Command Default

No time period based on date is defined for night service.

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

After you define night-service periods using this command and the **night-service day** command, use the **night-service bell** (ephone-dn) command to specify the extensions that will ring on other phones and the **night-service bell** (ephone) command to specify the phones on which the extensions will ring during the designated night-service periods.

### Examples

The following example defines a night-service time period for the entire day of January 1:

```
Router(config)# telephony-service
Router(config-telephony)# night-service date jan 1 00:00 00:00
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>night-service bell (ephone)</strong></td>
<td>Marks an IP phone to receive night-service-bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------------------</td>
<td>--------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>night-service bell (ephone-dn)</td>
<td>Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.</td>
</tr>
<tr>
<td>night-service code</td>
<td>Defines a code to disable or reenable night service on IP phones.</td>
</tr>
<tr>
<td>night-service day</td>
<td>Defines a recurring time period associated with a day of the week during which night service is active.</td>
</tr>
</tbody>
</table>
night-service day

To define a recurring time period associated with a day of the week during which night service is active, use the **night-service day** command in telephony-service configuration mode. To delete the defined time period, use the **no** form of this command.

```bash
night-service day day start-time stop-time
no night-service day day start-time stop-time
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>day</strong></td>
<td>Day of the week abbreviation. The following are valid day abbreviations: <strong>sun</strong>, <strong>mon</strong>, <strong>tue</strong>, <strong>wed</strong>, <strong>thu</strong>, <strong>fri</strong>, <strong>sat</strong>.</td>
</tr>
<tr>
<td><strong>start-time</strong></td>
<td>Beginning and ending times for night service, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, mon 19:00 07:00 means “from Monday at 7 p.m. until Tuesday at 7 a.m.”</td>
</tr>
<tr>
<td><strong>stop-time</strong></td>
<td>The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, night service is in effect for the entire 24-hour period on the specified day.</td>
</tr>
</tbody>
</table>

**Command Default**

No time period based on day of the week is defined for night service.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

After you define night-service periods using this command and the **night-service date** command, use the **night-service bell (ephone-dn)** command to specify the extensions that will ring on other phones and the **night-service bell (ephone)** command to specify the phones on which the extensions will ring during the designated night-service periods.

**Examples**

The following example defines a night-service time period from Monday at 7 p.m. to Tuesday at 9 a.m.:

```bash
Router(config)# telephony-service
Router(config-telephony)# night-service day mon 19:00 09:00
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>night-service bell (ephone)</strong></td>
<td>Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.</td>
</tr>
<tr>
<td><strong>Description</strong></td>
<td></td>
</tr>
<tr>
<td>-----------------</td>
<td>-------------------------------------------------</td>
</tr>
<tr>
<td><strong>night-service bell (ephone-dn)</strong></td>
<td>Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.</td>
</tr>
<tr>
<td><strong>night-service code</strong></td>
<td>Defines a code to disable or reenable night service on IP phones.</td>
</tr>
<tr>
<td><strong>night-service date</strong></td>
<td>Defines a recurring time period associated with a month and day during which night service is active.</td>
</tr>
</tbody>
</table>
night-service everyday

To define a recurring time period during which night service is active every day, use the `night-service everyday` command in telephony-service configuration mode. To delete the defined time period, use the `no` form of this command.

```
night-service everyday start-time stop-time
no night-service everyday
```

### Syntax Description

<table>
<thead>
<tr>
<th>start-time</th>
<th>stop-time</th>
</tr>
</thead>
</table>
| Beginning and ending times for night service, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, `mon 19:00 07:00` means “from Monday at 7 p.m. until Tuesday at 7 a.m.” The value `24:00` is not valid. If `00:00` is entered as a stop time, it is changed to `23:59`. If `00:00` is entered for both start time and stop time, night service is in effect for the entire 24-hour period on the specified day.

### Command Default

No recurring night-service time period is defined for every day.

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>
| 12.4(9)T          | Cisco Unified CME 4.0 | This command was integrated into Cisco IOS Release 12.4(9)T.

### Usage Guidelines

After you define recurring night-service time periods, use the `night-service bell (ephone-dn)` command to specify the extensions that will ring on other phones and the `night-service bell (ephone)` command to specify the phones on which the extensions will ring during the designated night-service periods.

### Examples

The following example defines a night-service time period to be in effect every day from 7 p.m. to 8 a.m.:

```
Router(config)# telephony-service
Router(config-telephony)# night-service everyday 19:00 08:00
```

### Related Commands

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>night-service bell (ephone)</code> Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.</td>
</tr>
<tr>
<td><code>night-service bell (ephone-dn)</code> Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>---------------</td>
</tr>
<tr>
<td>night-service code</td>
</tr>
<tr>
<td>night-service date</td>
</tr>
<tr>
<td>night-service day</td>
</tr>
<tr>
<td>night-service weekday</td>
</tr>
<tr>
<td>night-service weekend</td>
</tr>
</tbody>
</table>
night-service weekday

To define a recurring night-service time period to be in effect on all weekdays, use the night-service weekday command in telephony-service configuration mode. To delete the defined time period, use the no form of this command.

```
night-service weekday start-time stop-time
no night-service weekday
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>start-time</code></td>
<td>Beginning and ending times for night service, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, mon 19:00 07:00 means “from Monday at 7 p.m. until Tuesday at 7 a.m.”</td>
</tr>
<tr>
<td><code>stop-time</code></td>
<td>The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, night service is in effect for the entire 24-hour period on the specified day.</td>
</tr>
</tbody>
</table>

**Command Default**

No recurring night-service time period is defined for weekdays.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Weekdays are defined as Monday, Tuesday, Wednesday, Thursday, and Friday.

After you define night-service periods, use the night-service bell (ephone-dn) command to specify the extensions that will ring on other phones and the night-service bell (ephone) command to specify the phones on which the extensions will ring during the designated night-service periods.

**Examples**

The following example defines a night-service time period every weekday from 5 p.m. to 9 a.m.:

```
Router(config)# telephony-service
Router(config-telephony)# night-service weekday 17:00 09:00
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>night-service bell (ephone)</td>
<td>Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.</td>
</tr>
<tr>
<td>night-service bell (ephone-dn)</td>
<td>Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.</td>
</tr>
<tr>
<td>night-service weekday</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------</td>
<td>-------------</td>
</tr>
<tr>
<td>night-service code</td>
<td>Defines a code to disable or reenable night service on IP phones.</td>
</tr>
<tr>
<td>night-service date</td>
<td>Defines a recurring time period associated with a month and day during which night service is active.</td>
</tr>
<tr>
<td>night-service day</td>
<td>Defines a recurring time period associated with a day of the week during which night service is active.</td>
</tr>
<tr>
<td>night-service everyday</td>
<td>Defines a recurring night-service time period to be in effect everyday.</td>
</tr>
<tr>
<td>night-service weekend</td>
<td>Defines a recurring night-service time period to be in effect only on weekends.</td>
</tr>
</tbody>
</table>
**night-service weekend**

To define a recurring night-service time period to be active on weekends, use the `night-service weekend` command in telephony-service configuration mode. To delete the defined time period, use the `no` form of this command.

```
night-service weekend start-time stop-time
no night-service weekend
```

### Syntax Description

<table>
<thead>
<tr>
<th>start-time</th>
<th>stop-time</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>start-time</strong></td>
<td><strong>stop-time</strong></td>
</tr>
</tbody>
</table>

Beginning and ending times for night service, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, mon 19:00 07:00 means “from Monday at 7 p.m. until Tuesday at 7 a.m.”

The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, night service is in effect for the entire 24-hour period on the specified day.

### Command Default

No recurring night-service time period is defined for weekends.

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Weekend is defined as Saturday and Sunday.

After you define night-service periods, use the `night-service bell (ephone-dn)` command to specify the extensions that will ring on other phones and the `night-service bell (ephone)` command to specify the phones on which the extensions will ring during the designated night-service periods.

### Examples

The following example defines a night-service time period for all day Saturdays and Sundays:

```
Router(config) # telephony-service
Router(config-telephony)# night-service weekend 00:00 00:00
```

### Related Commands

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>night-service bell (ephone)</strong></td>
</tr>
<tr>
<td><strong>night-service bell (ephone-dn)</strong></td>
</tr>
<tr>
<td><strong>Description</strong></td>
</tr>
<tr>
<td>---</td>
</tr>
</tbody>
</table>
| **night-service code** | Defines a code to disable or reenable night service on IP phones.  
| **night-service date** | Defines a recurring time period associated with a month and day during which night service is active.  
| **night-service day** | Defines a recurring time period associated with a day of the week during which night service is active.  
| **night-service everyday** | Defines a recurring night-service time period to be in effect everyday.  
| **night-service weekday** | Defines a recurring night-service time period to be in effect only on weekdays.  
| **night-service weekend** |  

no-reg

To specify that the pilot number for a Cisco CallManager Express (Cisco CME) peer ephone hunt group not register with an H.323 gatekeeper, use the no-reg command in ephone-hunt configuration mode. To return to the default of the pilot number registering with an H.323 gatekeeper, use the no form of this command.

```
no-reg [{both|pilot}]
no no-reg [{both|pilot}]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>both</td>
<td>(Optional) Both the primary and secondary pilot numbers are not registered.</td>
</tr>
<tr>
<td>pilot</td>
<td>(Optional) Only the primary pilot number is not registered.</td>
</tr>
</tbody>
</table>

**Command Default**
The pilot number registers with the H.323 gatekeeper.

**Command Modes**
Ephone-hunt configuration (config-ephone-hunt)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.3(7)T</td>
<td>Cisco CME 3.1</td>
<td>The both and pilot keywords were introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command is valid only for Cisco CME peer ephone hunt groups.

**Examples**
The following example defines peer ephone hunt group 2 with a primary and secondary pilot number, and specifies that the secondary pilot number should not register with the H.323 gatekeeper:

```
Router(config)# ephone-hunt 2 peer
Router(config-ephone-hunt)# pilot 2222 secondary 4444
Router(config-ephone-hunt)# no-reg
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>final</td>
<td>Defines the last ephone-dn in an ephone hunt group.</td>
</tr>
<tr>
<td>hops</td>
<td>Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.</td>
</tr>
<tr>
<td>list</td>
<td>Defines the ephone-dns that participate in an ephone hunt group.</td>
</tr>
<tr>
<td>max-redirect</td>
<td>Changes the current number of allowable redirects in a Cisco CME system.</td>
</tr>
<tr>
<td>pilot</td>
<td>Defines the ephone-dn that is dialed to reach an ephone hunt group.</td>
</tr>
<tr>
<td>preference (ephone-hunt)</td>
<td>Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.</td>
</tr>
<tr>
<td>--------------------------</td>
<td>-----------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>timeout (ephone-hunt)</td>
<td>Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.</td>
</tr>
</tbody>
</table>
no-reg (voice register dn)

To specify that a voice DN for a SIP phone line in a Cisco CallManager Express (Cisco CME) system not register with an external proxy server, use the no-reg command in voice register dn configuration mode. To return to the default, use the no form of this command.

no-reg
no  no-reg

Syntax Description
This command has no arguments or keywords.

Command Default
This command is disabled.

Command Modes
Voice register dn configuration (config-register-dn)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

Usage Guidelines
This command specifies that a particular voice DN not register with the external proxy server. Configure the no-reg command per line. The default is to register all SIP lines in the Cisco CME system.

Examples
The following example shows how to configure bulk registration for registering a block of phone numbers starting with 408555 with an external registrar and specify that directory number 1, number 4085550100 not register with the external registrar:

```sh
Router(config)# voice register global
Router(voice-register-global)# mode cme
Router(voice-register-global)# bulk 408555....
Router(voice-register-global)# exit
Router(config)# voice register dn 1
Router(config-register-dn)# number 408550100
Router(config-register-dn)# no-reg
```

Related Commands

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Associates a telephone or extension number with a SIP phone in a Cisco CME system.</td>
</tr>
</tbody>
</table>

number (voice register dn)
nte-end-digit-delay

To specify the amount of time that each digit in the RTP NTE end event in an RFC 2833 packet is delayed before being sent, use the `nte-end-digit-delay` command in ephone or ephone-template configuration mode. To remove the delay amount, use the no form of this command.

```
nte-end-digit-delay [milliseconds]
no nte-end-digit-delay
```

**Syntax Description**


**Command Default**

All digits in the RTP NTE end event are sent in a single burst.

**Command Modes**

Ephone configuration (config-ephone)

Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

If your system is configured for RFC 2833 DTMF interworking and if the remote system cannot handle the RTP NTE end event sent in a single burst, use this command to delay sending each digit in the RTP NTE end event by the specified number of milliseconds. The default value for the delay is 100 ms.

This command only specifies the amount of time that each digit in the RTP NTE end event is delayed before being sent. To enable the delay, you must also configure the `dtmf-interworking rtp-nte` command in voice-service or dial-peer configuration mode.

If the phone user dials digits faster than the configured RTP NTE end-event delay, Cisco Unified CME will process the dialed digits and ignore the configured RTP NTE end-event delay unless you also configure the `keypad-normalize` command in ephone or ephone-template configuration mode.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

**Examples**

The following example shows the configuration for ephone 43 in which the `nte-end-digit-delay` command is configured for a 200 ms delay.

```
Router(config)# show running-config
.
.
.
ephone 43
button 1:29
nte-end-digit-delay 200
keypad-normalize
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>dtmf-interworking rtp-n-te</code></td>
<td>Introduces a delay between the dtmf-digit begin and dtmf-digit end events in RFC 2833 packets sent from the router.</td>
</tr>
<tr>
<td><code>ephone-template (ephone)</code></td>
<td>Applies a template to ephone being configured.</td>
</tr>
<tr>
<td><code>keypad-normalize</code></td>
<td>Ensures that the delay configured for a dtmf-end event is always honored.</td>
</tr>
</tbody>
</table>
ntp-server

To specify the IP address of the Network Time Protocol (NTP) server used by SIP phones in a Cisco Unified CME system, use the `ntp-server` command in voice register global configuration mode. To remove the NTP server, use the `no` form of this command.

```
ntp-server ip-address [mode {anycast|directedbroadcast|multicast|unicast}]
no ntp-server
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ip-address</td>
<td>IP address of the NTP server.</td>
</tr>
<tr>
<td>mode</td>
<td>(Optional) Enables the broadcast mode for the server.</td>
</tr>
<tr>
<td>anycast</td>
<td>Enables anycast mode.</td>
</tr>
<tr>
<td>directedbroadcast</td>
<td>Enables directed broadcast mode.</td>
</tr>
<tr>
<td>multicast</td>
<td>Enables multicast mode.</td>
</tr>
<tr>
<td>unicast</td>
<td>Enables unicast mode.</td>
</tr>
</tbody>
</table>

### Command Default

An NTP server is not used.

### Command Modes

Voice register global configuration (config-register-global)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command synchronizes all SIP phones to the specified NTP server.

This command is not supported on the Cisco Unified IP Phone 7905, 7912, 7940, or 7960.

### Examples

The following example shows the mode for the NTP server set to multicast:

```
Router(config)# voice register global
Router(config-register-global)# ntp-server 10.10.10.1 mode multicast
```

### Related Commands

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>create profile</td>
</tr>
<tr>
<td>restart (voice register)</td>
</tr>
</tbody>
</table>
number (ephone-dn)

To associate a telephone or extension number with an ephone-dn in a Cisco CallManager Express (Cisco CME) system, use the `number` command in ephone-dn configuration mode. To disassociate a number from an ephone-dn, use the `no` form of this command.

```
number number [secondary number] [no-reg [{both|primary}]]
no number
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Term</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>number</code></td>
<td>String of up to 16 characters that represents an E.164 telephone number. Normally the string is composed of digits, but the string may contain alphabetic characters when the number is dialed only by the router, as with an intercom number. One or more periods (.) can be used as wildcard characters. For details, see “Usage Guidelines.”</td>
</tr>
<tr>
<td><code>secondary</code></td>
<td>(Optional) Associates the number that follows as an additional number for this ephone-dn.</td>
</tr>
<tr>
<td><code>no-reg</code></td>
<td>(Optional) The E.164 numbers in the dial peer do not register with the gatekeeper. If you do not specify an option (both or primary) after the <code>no-reg</code> keyword, only the secondary number is not registered.</td>
</tr>
<tr>
<td><code>both</code></td>
<td>(Optional) Both primary and secondary numbers are not registered.</td>
</tr>
<tr>
<td><code>primary</code></td>
<td>(Optional) Primary number is not registered.</td>
</tr>
</tbody>
</table>

**Command Default**

No primary or secondary phone number is associated with the ephone-dn.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>The ability to use alphabetic characters as part of the number string was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command defines a valid number for an ephone-dn (extension) that is to be assigned to an IP phone. The `secondary` keyword allows you to associate a second telephone number with an ephone-dn so that it can be called by dialing either the main or secondary phone number. The secondary number may contain wildcards; for example, 50.. (the number 50 followed by periods, which stand for wildcards).

The `no-reg` keyword causes an E.164 number in the dial peer not to register with the gatekeeper. If you do not specify `both` or `primary` after the `no-reg` keyword, only the secondary number does not register.

A number normally contains only numeric characters, which allow it to be dialed from any telephone keypad. However, in certain cases, such as intercom numbers, which are normally dialed only by the router, you can insert alphabetic characters into the number to prevent phone users from dialing it and using the intercom function without authorization.
A number can also contain one or more periods (.) as wildcard characters that will match any dialed number in that position. For example, 51.. rings when 5100 is dialed, when 5101 is dialed, and so forth.

After you use the `number` command, assign the `ephone-dn` to an ephone using the `button` command. Following the use of the `button` command, the `restart` command must be used to initiate a quick reboot of the phone to which this number is assigned.

### Examples

The following example sets 5001 as the primary extension number for a Cisco IP phone and 0 as the secondary number. This configuration allows the telephone number 5001 to act as a regular extension number and also to act as the operator line such that callers who dial 0 are routed to the phone line with extension number 5001.

```plaintext
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001 secondary 0
```

The following example sets 5001 as the primary extension number for a Cisco IP phone and “500.” (the number 500 followed by a period) as the secondary number. This configuration allows any calls to extension numbers from the range 5000 to 5009 to be routed to extension 5001 if the actual extension number dialed cannot be found. For example, IP phones may be active in the system with lines that correspond to 5001, 5002, 5004, 5005, and 5009. A call to 5003 would be unable to locate a phone with extension 5003, so the call would be routed to extension 5001.

```plaintext
Router(config-ephone-dn)# number 5001 secondary 500.
```

The following example defines a pair of intercom ephone-dns that are programmed to call each other. The intercom numbers contain alphabetic characters to prevent anyone from dialing them from another phone. Ephone-dn 19 is assigned the number A5511 and is programmed to dial A5522, which belongs to ephone-dn 20. Ephone-dn 20 is programmed to dial A5511. No one else can dial these numbers.

```plaintext
Router(config)# ephone-dn 19
Router(config-ephone-dn)# number A5511
Router(config-ephone-dn)# name Intercom
Router(config-ephone-dn)# intercom A5522
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 20
Router(config-ephone-dn)# number A5522
Router(config-ephone-dn)# name Intercom
Router(config-ephone-dn)# intercom A5511
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>button</code></td>
<td>Associates ephone-dns with individual buttons on Cisco IP phones and specifies ring behavior per button.</td>
</tr>
<tr>
<td><code>intercom</code></td>
<td>Creates an intercom by programming a pair of extensions (ephone-dns) to automatically call and answer each other.</td>
</tr>
<tr>
<td><code>name</code></td>
<td>Configures a username associated with a directory number.</td>
</tr>
<tr>
<td><code>preference</code></td>
<td>Sets preference for the attached dial peer for a directory number.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>restart (ephone)</td>
<td>Performs a fast reboot of a single phone associated with a Cisco CME router.</td>
</tr>
<tr>
<td>restart (telephony-service)</td>
<td>Performs a fast reboot of one or all phones associated with a Cisco CME router.</td>
</tr>
</tbody>
</table>
**night-service bell (voice register dn)**

To mark a voice register dn for night-service treatment, use the `night-service bell` command in voice register dn configuration mode. To remove the night-service treatment from the voice register dn, use the `no` form of this command.

```
night-service bell
no night-service bell
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

By default, this command is disabled.

**Command Modes**

voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.6(3)M</td>
<td>Cisco Unified CME 11.5</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>16.3.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

When a voice register dn is marked for night-service treatment using this command, incoming calls that ring during the night-service time period on that voice register dn sends an alert indication to all IP phones that are marked to receive night-service bell notification. This is achieved using the `night-service bell (voice register pool)` command. The alert notification is in the form of a splash ring (not associated with any of the individual lines on the IP phone) and a visible display of the voice register extension number. The phone user retrieves the call by pressing a GPickUp soft key and dialing the appropriate digits.

Night-service periods are defined using the `night-service date` and `night-service day` commands. Night service can be manually disabled or enabled from a phone configured with voice register dn in night-service mode if the `night-service code` command has been set.

**Examples**

The following example marks a voice register dn as a line that will ring on IP phones designated to receive night-service bell notification when incoming calls are received on this voice register dn during night-service periods:

```
Router(config)# voice register dn 16
Router(config-register-dn)# night-service bell
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>night-service bell (voice register pool)</code></td>
<td>Marks a SIP phone to receive night-service bell notification when incoming calls are received on voice register DNs that are marked for night service during night-service time periods.</td>
</tr>
<tr>
<td><code>night-service code</code></td>
<td>Defines a code to disable or reenable night service on IP phones.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>night-service date</td>
<td>Defines a recurring time period associated with a month and day during which night service is active.</td>
</tr>
<tr>
<td>night-service day</td>
<td>Defines a recurring time period associated with a day of the week during which night service is active.</td>
</tr>
</tbody>
</table>
night-service bell (voice register pool)

To mark a SIP phone to receive night-service bell notification when incoming calls are received on voice register dn that are marked for night service during night-service time periods, use the `night-service bell` command in voice register pool configuration mode. To remove night-service notification capability from a phone, use the `no` form of this command.

```
night-service bell
no night-service bell
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

By default, this command is disabled.

**Command Modes**

voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.6(3)M</td>
<td>Cisco Unified CME 11.5</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>16.3.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

When a voice register dn is marked for night-service treatment using the `night-service bell (voice register dn)` command, incoming calls that ring during the night-service time period on that DN send an alert indication to all SIP phones marked to receive night-service bell notification. The alert notification is in the form of a splash ring (not associated with any of the individual lines on the IP phone) and a visible display of the voice register dn extension number. The phone user retrieves the call by pressing the GPickUp soft key and dialing the appropriate digits.

Night-service periods are defined using the `night-service date` and `night-service day` commands. Night service can be manually disabled or re-enabled from a phone configured with voice register dn in night-service mode if the `night-service code` command is set.

If you use a voice register template to apply a command to a SIP phone and you also use the same command in pool configuration mode for the same phone, the value that you set in pool configuration mode has priority.

**Examples**

The following example designates the SIP phone that is being configured as a phone that will receive night-service bell notification when voice register dns marked for night service receive incoming calls during a night-service period:

```
Router(config)# voice register pool 4
Router(config-register-pool)# night-service bell
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>night-service bell (voice register dn)</code></td>
<td>Marks a voice register dn to send night-service bell notification to designated SIP phones during night-service time periods.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>night-service bell (voice register template)</td>
<td>Applies a template to a pool configuration.</td>
</tr>
<tr>
<td>night-service code</td>
<td>Defines a code to disable or reenable night service on IP phones.</td>
</tr>
<tr>
<td>night-service date</td>
<td>Defines a recurring time period associated with a month and day during which night service is active.</td>
</tr>
<tr>
<td>night-service day</td>
<td>Defines a recurring time period associated with a day of the week during which night service is active.</td>
</tr>
</tbody>
</table>
night-service bell (voice register template)

To mark a SIP phone to receive night-service bell notification when incoming calls are received on voice register dn that are marked for night service during night-service time periods, use the `night-service bell` command in voice register template configuration mode. To remove night-service notification capability from a phone, use the `no` form of this command.

```
night-service bell
no night-service bell
```

### Syntax Description
This command has no arguments or keywords.

### Command Default
By default, this command is disabled.

### Command Modes
voice register template configuration (config-register-template)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.6(3)M</td>
<td>Cisco Unified CME 11.5</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>16.3.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Usage Guidelines
When a voice register dn is marked for night-service treatment using the `night-service bell (voice register dn)` command, incoming calls that ring during the night-service time period on that DN send an alert indication to all SIP phones marked to receive night-service bell notification. The alert notification is in the form of a splash ring (not associated with any of the individual lines on the IP phone) and a visible display of the voice register dn extension number. The phone user retrieves the call by pressing the GPickUp soft key and dialing the appropriate digits.

Night-service periods are defined using the `night-service date` and `night-service day` commands. Night service can be manually disabled or re-enabled from a phone configured with voice register dn in night-service mode if the `night-service code` command is set.

If you use a voice register template to apply a command to a SIP phone and you also use the same command in pool configuration mode for the same phone, the value that you set in pool configuration mode has priority.

### Examples
The following example designates the SIP phone that is being configured as a phone that will receive night-service bell notification when voice register dns marked for night service receive incoming calls during a night-service period:

```
Router(config)# voice register pool 4
Router(config-register-pool)# night-service bell
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>night-service bell (voice register dn)</code></td>
<td>Marks a voice register dn to send night-service bell notification to designated SIP phones during night-service time periods.</td>
</tr>
<tr>
<td><code>voice register pool</code></td>
<td>Applies a template to a pool configuration.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>night-service code</td>
<td>Defines a code to disable or reenable night service on IP phones.</td>
</tr>
<tr>
<td>night-service date</td>
<td>Defines a recurring time period associated with a month and day during which night service is active.</td>
</tr>
<tr>
<td>night-service day</td>
<td>Defines a recurring time period associated with a day of the week during which night service is active.</td>
</tr>
</tbody>
</table>
number (voice register dn)

To associate a telephone or extension number with a SIP phone in a Cisco CallManager Express (Cisco CME) system, use the **number** command in voice register dn configuration mode. To disassociate a number, use the **no** form of this command.

```
number number
no number
```

**Syntax Description**
```
number String of up to 16 characters that represents an E.164 telephone number. Normally the string is composed of digits, but the string may contain alphabetic characters when the number is dialed only by the router, as with an intercom number.
```

**Command Default**

This command has no default behavior or values.

**Command Modes**

Voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 and Cisco SIP SRST 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Valid characters in voice register DN include 0-9, ' ', '(', ')', '+' and '#'.

To allow insertion of '#' at any place in voice register dn, the CLI "allow-hash-in-dn" is configured in voice register global mode.

When the CLI "allow-hash-in-dn" is configured, the user is required to change the dial-peer terminator from '#' (default terminator) to another valid terminator in configuration mode. The other terminators that are supported include '0'-'9', 'A'-'F', and '*'.

This command defines a valid number for an extension that is to be assigned to a SIP phone. Use this command before using the other commands in voice register dn configuration mode.

A number normally contains only numeric characters which allows users to dial the number from any telephone keypad. However, in certain cases, such as the numbers for intercom extensions, you want to use numbers that can only be dialed internally from the Cisco CallManager Express router and not from telephone keypads.

The **number** command allows you to assign alphabetic characters to the number so that the extension can be dialed by the router for intercom calls but not by unauthorized individuals from other phones.

After you use the **number** command, use the **reset** command to initiate a quick reboot of the phone to which this number is assigned.

**Note**

This command can also be used for Cisco SIP SRST.

**Examples**

The following example shows how to set 5001 as the extension number for directory number 1 on a SIP phone.
Router(config)# voice register dn 1
Router(config-register-dn)# number 5001

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>reset (voice register global)</td>
<td>Performs a complete reboot of all SIP phones associated with a Cisco CME router.</td>
</tr>
<tr>
<td>reset (voice register pool)</td>
<td>Performs a complete reboot of a single SIP phone associated with a Cisco CME router.</td>
</tr>
</tbody>
</table>
number (voice register pool)

To indicate the E.164 phone numbers that the registrar permits to handle the Register message from a Cisco Unified SIP IP phone, use the number command in voice register pool configuration mode. To disable number registration, use the no form of this command.

```
number tag {number-pattern [preference value] [huntstop]dn dn-tag}
no number tag
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Tag</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>tag</code></td>
<td>Telephone number when there are multiple number commands. Range is 1 to 114.</td>
</tr>
<tr>
<td><code>number-pattern</code></td>
<td>Phone numbers (including wild cards and patterns) that are permitted by the registrar to handle the Register message from the Cisco Unified SIP IP phone.</td>
</tr>
<tr>
<td><code>preference value</code></td>
<td>(Optional) Defines the number list preference order. Range is 0 to 10. The highest preference is 0. There is no default.</td>
</tr>
<tr>
<td><code>huntstop</code></td>
<td>(Optional) Stops hunting when the dial peer is busy.</td>
</tr>
<tr>
<td><code>dn dn-tag</code></td>
<td>Identifies the directory number tag for this phone number as defined by the voice register <code>dn</code> command. Range is 1 to 288.</td>
</tr>
</tbody>
</table>

**Command Default**

Cisco Unified SIP IP phones cannot register in Cisco Unified CME.

**Command Modes**

Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was added to Cisco CME and the dn keyword was added.</td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1 Cisco Unified SRST 4.1</td>
<td>This command was modified. The number-pattern argument and preference and huntstop keywords were removed from Cisco Unified CME.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1 Cisco Unified SRST 4.1</td>
<td>The modifications to this command were integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>15.2(4)M</td>
<td>Cisco Unified CME 9.1 Cisco Unified SIP SRST 9.1</td>
<td>This command was modified to increase the valid value of the tag argument to 114.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The number command indicates the phone numbers that are permitted by the registrar to handle the Register message from the Cisco Unified SIP IP phone.

In Cisco Unified SRST, the keywords and arguments of this command allow for more explicit setting of user preferences regarding what number patterns should match the voice register pool.
Configure the **id (voice register pool)** command before any other voice register pool commands, including the **number** command. The **id** command identifies a locally available, individual Cisco Unified SIP IP phone or a set of Cisco Unified SIP IP phones.

### Examples

The following example shows three directory numbers assigned to Cisco Unified SIP IP phone 1 in Cisco Unified CME:

```plaintext
! voice register pool 1
  id mac 0017.E033.0284
  type 7961
  number 1 dn 10
  number 2 dn 12
  number 3 dn 13
  codec g711ulaw
!```

The following example shows directory numbers 10, 12, and 13 assigned to phone numbers 1, 2, and 55 of Cisco Unified SIP IP phone 2:

```plaintext
voice register pool 2
  id mac 0017.E033.0284
  type 7961
  number 1 dn 10
  number 2 dn 12
  number 55 dn 13
  codec g711ulaw
```

The following example shows a telephone number pattern set to 95... in Cisco Unified SRST. This means all five-digit numbers beginning with 95 are permitted by the registrar to handle the Register message.

```plaintext
voice register pool 3
  id network 10.2.161.0 mask 255.255.255.0
  number 1 95... preference 1
  cor incoming call95 1 95011
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>id (voice register pool)</strong></td>
<td>Explicitly identifies a locally available, individual Cisco Unified SIP IP phone or, when running Cisco Unified SIP SRST, a set of Cisco Unified SIP IP phones.</td>
</tr>
<tr>
<td><strong>voice register dn</strong></td>
<td>Enters voice register dn configuration mode to define an extension for a phone line, intercom line, voice-mail port, or a message-waiting indicator.</td>
</tr>
</tbody>
</table>
number (voice user-profile and voice logout-profile)

To create line definitions in a voice-user profile or voice-logout profile to be downloaded to a Cisco Unified IP phone that is enabled for extension mobility, use the **number** command in voice user-profile configuration mode or voice logout-profile configuration mode. To remove line definition from a profile, use the no form of this command.

```
number number[,...,number] type type

no number number[,...,number] type type
```

**Syntax Description**

<table>
<thead>
<tr>
<th><strong>number</strong></th>
<th>String of up to 16 characters that represents an E.164 telephone number to be associated with and displayed next to a line button on an IP phone. This directory number must be already configured by using the <strong>number</strong> command in ephone-dn or voice register dn configuration mode.</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>[...number]</strong></td>
<td>(Optional) For overlay lines only, with or without call waiting. Directory numbers to roll over to this line. Can contain up to 25 individual numbers separated by commas (.). This directory number must be already configured by using the <strong>number</strong> command in ephone-dn or voice register dn configuration mode.</td>
</tr>
<tr>
<td><strong>type</strong></td>
<td>Characteristics to be associated with this line button.</td>
</tr>
<tr>
<td><strong>type</strong></td>
<td>Word that describes characteristics to be associated with the line button being configured. Valid entries are as follows: • beep-ring • feature-ring • monitor-ring • silent-ring • overlay • cw-overlay</td>
</tr>
</tbody>
</table>

**Command Default**

No line definition is created.

**Command Modes**

Voice logout-profile configuration (config-logout-profile)
Voice user-profile configuration (config-user-profile)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XW</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>
Usage Guidelines

This command in voice user-profile configuration mode to creates a line button definition in a user profile to be downloaded to the IP phone when the user is logged into an IP phone that is enabled for extension mobility.

This command in voice logout-profile configuration mode creates a line button definition in a default profile to be downloaded to an IP phone when no user is logged into an IP phone that is enabled for extension mobility.

For button appearance, extension mobility will associate line definitions in the voiceLogout profile or voice-user profile to phone buttons in a sequential manner. If the profile contains more directory and speed-dial numbers than there are buttons on the physical phone to which the profile is downloaded, the remaining numbers in the profile are ignored.

On Cisco Unified IP phones, line definitions are assigned to available extension buttons before speed-dial definitions, in sequential order, starting with the lowest directory number first.

After creating or modifying a profile, use the reset (voice logout-profile or voice user-profile) command to reset all phones associated with the profile being configured to propagate the changes.

Type ? to list valid options for the type keyword. The following options are valid at the time that this document was written:

- beep-ring
  Beep but no ring. Audible ring is suppressed for incoming calls but call-waiting beeps are allowed. Visible cues are the same as those for a normal ring.

- feature-ring
  Differentiates incoming calls on a special line from incoming calls on other lines on the phone. The feature-ring cadence is a triple pulse, as opposed to a single-pulse ring for normal internal calls and a double-pulse ring for normal external calls.

- monitor-ring
  A line button that is configured for monitor mode on one phone provides visual line status for a line that also appears on another phone. When monitor mode is set for a button with a shared line, the line status indicates that the shared line is either idle or in use. The line and line button are available in monitor mode for visual status only. Calls cannot be made or received using a line button that has been set in monitor mode. Incoming calls on a line button that is in monitor mode do not ring and do not display caller ID or call-waiting caller ID. Monitor mode is intended to be used only in the context of shared lines so that one user, such as a receptionist, can visually monitor the in-use status of several users’ phone extensions (for example, as a busy-lamp field).

  The line button for a monitored line can be used as a direct-station-select for a call transfer when the monitored line is in an idle state. In this case, the phone user who transfers a call from a normal line can press the Transfer button and then press the line button of the monitored line, causing the call to be transferred to the phone number of the monitored line.

- silent-ring
  You can configure silent ring on any type of phone. However, you typically set silent ring only on buttons of a phone with multiple lines, such as a Cisco Unified IP Phone 7940 or Cisco Unified IP Phone 7960 and 7960G. The only visible cue is a flashing icon in the phone display.

  If you configure a button to have a silent ring, you will not hear a call-waiting beep or call-waiting ring regardless of whether the ephone-dn associated with the button is configured to generate a call-waiting beep or call-waiting ring.
In Cisco IOS Release 12.4(4)XC and later releases, the silent ringing behavior is overridden during active night-service periods. Silent ringing does not apply during designated night-service periods when the `s` keyword is used.

**overlay**

Overlay lines are directory numbers that share a single line button on a multibutton phone. When more than one incoming call arrives on lines that are set on a single button, the line (ephone-dn) that is the left most in the `number` command list is the primary line and is given the highest priority. If this call is answered by another phone or if the caller hangs up, the phone selects the next line in its overlay set to present as the ringing call. The caller ID display updates to show the caller ID for the currently presented call.

Directory numbers that are part of an overlay set can be single-line directory numbers or dual-line directory numbers, but the set must contain either all single-line or all dual-line directory numbers, and not a mixture of the two.

The primary directory number on each phone in a shared-line overlay set should be a unique ephone-dn. The unique ephone-dn guarantees that the phone will have a line available for outgoing calls, and ensures that the phone user can obtain dial-tone even when there are no idle lines available in the rest of the shared-line overlay set. Use a unique directory number in this manner to provide for a unique calling party identity on outbound calls made by the phone so that the called user can see which specific phone is calling.

The name of the first directory number in the overlay set is not displayed because it is the default directory number for calls to the phone, and the name or number is permanently displayed next to the phone’s button. For example, if there are ten numbers in an overlay set, only the last nine numbers are displayed when calls are made to them.

**cw-overlay**

The same configuration is used for overlaid lines both with and without call waiting.

Directory numbers can accept call interruptions, such as call waiting, by default. For call waiting to work, the default must be active. To ensure that this is the case, remove the `no call-waiting beep accept` command from the configurations of directory numbers for which you want to use call waiting.

Directory numbers that are part of a cw-overlay set can be single-line directory numbers or dual-line directory numbers, but the set must contain either all single-line or all dual-line directory numbers, and not a mixture of the two.

The Cisco Unified IP Phone 7931G cannot support overlays that contain directory numbers that are configured for dual-line mode.

**Examples**

The following example shows the configuration for a voice-user profile to be downloaded when the a phone user logs into a Cisco Unified IP phone that is enabled for extension mobility. The lines and speed-dial buttons in this profile that are configured on an IP phone after the user logs in depend on phone type. For example, if the user logs into a Cisco Unified IP Phone 7970, all buttons are configured according to voice-user profile1. However, if the phone user logs into a Cisco Unified IP Phone 7960, all six lines are mapped to phone buttons, and the speed dial is ignored because there is no button available for speed dial.

```
pin 12345
user me password pass123
number 2001 type silent-ring
```
number 2002 type beep-ring
number 2003 type feature-ring
number 2004 type monitor-ring
number 2005, 2006 type overlay
number 2007, 2008 type cw-overly
speed-dial 1 3001
speed-dial 2 3002 blf

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>logout-profile</td>
<td>Enables Cisco Unified IP phone for extension mobility and assigns a logout profile to this phone.</td>
</tr>
<tr>
<td>reset (voice logout-profile and voice user-profile)</td>
<td>Performs a complete reboot of all IP phones to which a particular logout-profile or user-profile is downloaded.</td>
</tr>
</tbody>
</table>
num-buttons

To set the number of line buttons supported by a phone type, use the num-buttons command in ephone-type configuration mode. To reset to the default, use the no form of this command.

```
num-buttons number
no num-buttons
```

**Syntax Description**

| number | Number of line buttons. Range: 1 to 100. Default: 0. See the table for the number of buttons supported by each phone type. |

**Command Default**

No line buttons are supported by the phone type.

**Command Modes**

Ephone-type configuration (config-ephone-type)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3 Cisco Unified SRST 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command defines the number of line buttons supported by the type of phone being added with an ephone-type template.

**Table 12: Supported Values for Ephone-Type Commands**

<table>
<thead>
<tr>
<th>Supported Device</th>
<th>device-id</th>
<th>device-type</th>
<th>num-buttons</th>
<th>max-presentation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified IP Phone 6901</td>
<td>547</td>
<td>6901</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 6911</td>
<td>548</td>
<td>6911</td>
<td>1</td>
<td>10</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7915 Expansion Module with 12 buttons</td>
<td>227</td>
<td>7915</td>
<td>12</td>
<td>0 (default)</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7915 Expansion Module with 24 buttons</td>
<td>228</td>
<td>7915</td>
<td>24</td>
<td>0</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7916 Expansion Module with 12 buttons</td>
<td>229</td>
<td>7916</td>
<td>12</td>
<td>0</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7916 Expansion Module with 24 buttons</td>
<td>230</td>
<td>7916</td>
<td>24</td>
<td>0</td>
</tr>
<tr>
<td>Cisco Unified Wireless IP Phone 7925</td>
<td>484</td>
<td>7925</td>
<td>6</td>
<td>4</td>
</tr>
<tr>
<td>Cisco Unified IP Conference Station 7937G</td>
<td>431</td>
<td>7937</td>
<td>1</td>
<td>6</td>
</tr>
<tr>
<td>Nokia E61</td>
<td>376</td>
<td>E61</td>
<td>1</td>
<td>1</td>
</tr>
</tbody>
</table>
Examples

The following example shows that 1 line button is specified for the Nokia E61 when creating the ephone-type template.

Router(config)# ephone-type E61
Router(config-ephone-type)# num-buttons 1

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>device-id</td>
<td>Specifies the device ID for a phone type.</td>
</tr>
<tr>
<td>max-presentation</td>
<td>Sets the number of call presentation lines supported by a phone type.</td>
</tr>
<tr>
<td>type</td>
<td>Assigns the phone type to an SCCP phone.</td>
</tr>
</tbody>
</table>
num-line

To define the maximum number of lines supported by new phone, use the **num-line** command in voice register pool-type mode. To remove the lines configured, use the **no** form of this command.

```
num-line  max-line
nonum-line max-line
```

**Syntax Description**

| Description        | Specific the number of lines supported by the phone model. Range is 1-114. |

**Command Default**

The default value of the addons is 1. When the **reference-pooltype** command is configured, the number of lines supported by the reference phone is inherited.

**Command Modes**

Voice Register Pool Configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(3)M</td>
<td>Cisco SIP CME 10.0</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to define the maximum number of lines for a Cisco Unified SIP IP phone on Cisco Unified CME. When you use the **no** form of this command, the inherited properties of the reference phone takes precedence over the default value.

**Cisco Unified CME**

The following example shows how to enter voice register pool-type configuration mode and define the maximum number of lines for a Cisco Unified SIP IP phone:

```plaintext
Router(config)# voice register pool-type 9900
Router(config-register-pool-type)# num-line 5
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>voice register pool-type</strong></td>
<td>Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.</td>
</tr>
</tbody>
</table>
Cisco Unified CME Commands: O

- olsontimezone, on page 712
- olsontimezone, on page 714
- overlap-signal, on page 716
- overwrite-dyn-stats (voice hunt-group), on page 719
olsontimezone

To set the Olson Timezone so that the correct local time is displayed on Cisco Unified SCCP IP phones or Cisco Unified SIP IP phones, use the `olsontimezone` command in telephony-service or voice register global configuration mode, respectively. To return to the default, use the `no` form of this command.

`olsontimezone timezone version number`
`no olsontimezone`

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>timezone</code></td>
<td>Olson Timezone names, which include the area (name of continent or ocean) and location (name of a specific location within that region, usually cities or small islands).</td>
</tr>
<tr>
<td><code>version number</code></td>
<td>Version of the tzupdater.jar or TzDataCSV.csv file. The version indicates whether the file needs to be updated or not.</td>
</tr>
</tbody>
</table>

**Note**: In Cisco Unified CME 9.0, the latest version is 2010o.

### Command Default

No Olson Timezone is set.

### Command Modes

- Telephony-service configuration (config-telephony)
- Voice register global configuration (config-register-global)

### Command History

**Release** | **Modification**
--- | ---
15.2(2)T | This command was introduced.

### Usage Guidelines

Use the `olsontimezone` command in either telephony-service or voice register global configuration mode, with the current version of Oracle’s Olson Timezone updater tool, tzupdater.jar, to set the correct Olson Timezone.

For Cisco Unified 3911 and 3951 SIP IP phones and Cisco Unified 6921, 6941, 6961, and 6945 SCCP and SIP IP phones, the correct Olson Timezone updater file is TzDataCSV.csv. The TzDataCSV.csv file is created based on the tzupdater.jar file.

To set the correct time zone, you must determine the Olson Timezone area/location where the Cisco Unified CME is located and download the latest tzupdater.jar or TzDataCSV.csv to a TFTP server (such as flash or slot 0) that is accessible to the Cisco Unified CME.

After a complete reboot, the phone checks if the version of its configuration file is earlier or later than 2010o. If it is earlier, the phone loads the latest tzupdater.jar and uses that updater file to calculate the Olson Timezone.

To make the Olson Timezone feature backward compatible, both the `time-zone` and `timezone` commands are retained as legacy time zones. Because the `olsontimezone` command covers approximately 500 time zones (Version 2010o of the tzupdater.jar file supports approximately 453 Olson Timezone IDs.), this command takes precedence when either the `time-zone` or the `timezone` command (that covers a total of 90 to 100 time zones only) is present at the same time as the `olsontimezone` command.

### Examples

The following example shows 7:29 p.m. as the time set on a Cisco Unified 7961 SCCP IP phone in Buenos Aires on May 13, 2011:
Router(config)# tftp-server flash:tzupdater.jar
Router(config)# tftp-server flash:TzDataCSV.csv
Router(config)# telephony-service
Router(config-telephony)# olsontimezone America/Argentina/Buenos Aires version 2010
Router(config-telephony)# create cnf-files
Router(config-telephony)# time-zone 21
Router(config-telephony)# exit
Router(config)# clock timezone CST -6
Router(config)# clock summer-time date 12 October 2010 2:00 26 April 2011 2:00
Router(config)# exit
Router# clock set 19:29:00 13 May 2011
Router# configure terminal
Router(config)# telephony-service
Router(config-telephony)# reset

The following example shows 3:25 p.m. as the time set on a Cisco Unified 6921 SIP IP phone in Buenos Aires on November 17, 2011:

Router(config)# tftp-server slot0:tzupdater.jar
Router(config)# tftp-server slot0:TzDataCSV.csv
Router(config)# voice register global
Router(config-register-global)# olsontimezone America/Argentina/Buenos Aires version 2010
Router(config-register-global)# create profile
Router(config-register-global)# time-zone 21
Router(config-register-global)# exit
Router(config)# clock timezone CST -6
Router(config)# clock summer-time date 12 October 2010 2:00 26 April 2011 2:00
Router(config)# exit
Router# clock set 15:25:00 17 November 2011
Router# configure terminal
Router(config)# voice register global
Router(config-register-global)# reset

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>time-zone</td>
<td>Sets the time zone so that the correct local time is displayed on Cisco Unified SCCP IP phones in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>timezone</td>
<td>Sets the time zone used for Cisco Unified SIP IP phones in a Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
To set the Olson Timezone so that the correct local time is displayed on Cisco Unified SCCP IP phones or Cisco Unified SIP IP phones, use the `olsontimezone` command in telephony-service or voice register global configuration mode, respectively. To return to the default, use the `no` form of this command.

`olsontimezone timezone version number`

`no olsontimezone`

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>timezone</code></td>
<td>Olson Timezone names, which include the area (name of continent or ocean) and location (name of a specific location within that region, usually cities or small islands).</td>
</tr>
<tr>
<td><code>version number</code></td>
<td>Version of the tzupdater.jar or TzDataCSV.csv file. The version indicates whether the file needs to be updated or not.</td>
</tr>
</tbody>
</table>

**Note** In Cisco Unified CME 9.0, the latest version is 2010o.

**Command Default**

No Olson Timezone is set.

**Command Modes**

Telephony-service configuration (config-telephony)

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(T)</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `olsontimezone` command in either telephony-service or voice register global configuration mode, with the current version of Oracle’s Olson Timezone updater tool, tzupdater.jar, to set the correct Olson Timezone.

For Cisco Unified 3911 and 3951 SIP IP phones and Cisco Unified 6921, 6941, 6961, and 6945 SCCP and SIP IP phones, the correct Olson Timezone updater file is TzDataCSV.csv. The TzDataCSV.csv file is created based on the tzupdater.jar file.

To set the correct time zone, you must determine the Olson Timezone area/location where the Cisco Unified CME is located and download the latest tzupdater.jar or TzDataCSV.csv to a TFTP server (such as flash or slot 0) that is accessible to the Cisco Unified CME.

After a complete reboot, the phone checks if the version of its configuration file is earlier or later than 2010o. If it is earlier, the phone loads the latest tzupdater.jar and uses that updater file to calculate the Olson Timezone.

To make the Olson Timezone feature backward compatible, both the `time-zone` and `timezone` commands are retained as legacy time zones. Because the `olsontimezone` command covers approximately 500 time zones (Version 2010o of the tzupdater.jar file supports approximately 453 Olson Timezone IDs.), this command takes precedence when either the `time-zone` or the `timezone` command (that covers a total of 90 to 100 time zones only) is present at the same time as the `olsontimezone` command.

**Examples**

The following example shows 7:29 p.m. as the time set on a Cisco Unified 7961 SCCP IP phone in Buenos Aires on May 13, 2011:
Router(config)# tftp-server flash:tzupdater.jar
Router(config)# tftp-server flash:TzDataCSV.csv
Router(config)# telephony-service
Router(config-telephony)# olsontimezone America/Argentina/Buenos Aires version 2010
Router(config-telephony)# create cnf-files
Router(config-telephony)# time-zone 21
Router(config-telephony)# exit
Router(config)# clock timezone CST -6
Router(config)# clock summer-time date 12 October 2010 2:00 26 April 2011 2:00
Router(config)# exit
Router# configure terminal
Router(config)# telephony-service
Router(config-telephony)# reset

The following example shows 3:25 p.m. as the time set on a Cisco Unified 6921 SIP IP phone in Buenos Aires on November 17, 2011:

Router(config)# tftp-server slot0:tzupdater.jar
Router(config)# tftp-server slot0:TzDataCSV.csv
Router(config)# voice register global
Router(config-register-global)# olsontimezone America/Argentina/Buenos Aires version 2010
Router(config-register-global)# create profile
Router(config-register-global)# time-zone 21
Router(config-register-global)# exit
Router(config)# clock timezone CST -6
Router(config)# clock summer-time date 12 October 2010 2:00 26 April 2011 2:00
Router(config)# exit
Router# configure terminal
Router(config)# voice register global
Router(config-register-global)# reset

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>time-zone</td>
<td>Sets the time zone so that the correct local time is displayed on Cisco Unified SCCP IP phones in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>timezone</td>
<td>Sets the time zone used for Cisco Unified SIP IP phones in a Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
overlap-signal

To configure overlap dialing in SCCP or SIP IP phones, use the overlap-signal command in ephone, ephone-template, telephony-service, voice register pool, voice register global, or voice register template configuration mode.

```
overlap-signal
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Overlap-signal is disabled.

**Command Modes**

- Call-manager-fallback
- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)
- Telephony-service configuration (config-telephony)
- Voice register pool (config-register-pool)
- Voice register global configuration (config-register-global)
- Voice register template (config-register-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5 Cisco Unified SRST 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

**SCCIP phones**

In SCCIP phones, overlap dialing is enabled when the overlap signal command is configured in ephone, ephone-template, and telephony-service configurations modes.

**SIP IP phones**

In SIP IP Phones, overlap dialing is enabled when the overlap signal command is configured in voice register pool, voice register global, and voice register template configuration modes.

**Cisco Unified SRST**

In Cisco Unified SRST, overlap dialing is enabled on SCCP IP phones when overlap signal command is configured in call-manager-fallback configuration mode.

**Examples**

The following example shows overlap-signal enabled on SCCP phones:

```
Router# show running config
!
!
telephony-service
max-ephones 25
max-dn 15
load 7906 SCCP11.8-5-3S.loads
load 7911 SCCP11.8-5-3S.loads
load 7921 CP7921G-1.3.3.LOADS
load 7941 SCCP41.8-5-3S.loads
load 7942 SCCP42.8-5-3S.loads
load 7961 SCCP41.8-5-3S.loads
```
load 7962 SCCP42.8-5-3S.loads
max-conferences 12 gain -6
web admin system name cisco password cisco
transfer-system full-consult
create cnf-files version-stamp Jan 01 2002 00:00:00
overlap-signal
!
ephone-template 1
button-layout 1 line
button-layout 3-6 blf-speed-dial
!
ephone-template 9
feature-button 1 Endcall
feature-button 3 Mobility
!
!
ephone-template 10
feature-button 1 Park
feature-button 2 MeetMe
feature-button 3 CallBack
button-layout 1 line
button-layout 2-4 speed-dial
button-layout 5-6 blf-speed-dial
overlap-signal
!
ephone 10
device-security-mode none
mac-address 02EA.EAEA.0010
overlap-signal
!

The following example shows overlap-signal configured in voice register global and voice register pool 10:

Router#show running config
!
!
voice service voip
ip address trusted list
 ipv4 20.20.20.1
media flow-around
allow-connections sip to sip
!
voice class media 10
media flow-around
!
!
voice register global
max-pool 10
overlap-signal
!
voice register pool 5
overlap-signal
!
!

The following example shows overlap-signal configured in call-manager-fallback mode:

Router# show run | sec call-manager
call-manager-fallback
max-conferences 12 gain -6
transfer-system full-consult
overlap-signal
**overwrite-dyn-stats (voice hunt-group)**

To overwrite statistics of previously joined dynamic agent with statistics of newly joined dynamic agents for voice hunt group, use the `overwrite-dyn-stats` command in voice hunt-group configuration mode. To remove the configuration, use the `no` form of this command.

`overwrite-dyn-stats`
`no overwrite-dyn-stats`

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

By default, this command is disabled.

**Command Modes**

voice hunt group configuration (config-voice-hunt-group)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.6(3)M</td>
<td>Cisco Unified CME 11.5</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>16.3.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is configured to overwrite statistics of previously joined dynamic agent with statistics of newly joined dynamic agents for voice hunt group. To remove the configuration, use the no form of this command. The statistics for the first 32 members (both dynamic and static members) joining in an hour are collected in the 32 statistic slots allotted. If any of the static members logout and login during the hour, that member is allotted the same slot as previous. In scenarios where free slots are available, free slots are used to write statistics of the newly joined dynamic agent. Once all the 32 slots are exhausted and a new dynamic member tries to join within the same hour, the `overwrite-dyn-stats` CLI takes effect. Using the CLI, the hunt statistic slot for the first dynamic member that joined the hunt-group is overwritten with the statistics of the newly joined dynamic member. The overwriting for statistics will continue at the same slot.

**Examples**

The following example shows how the voice hunt group overwrite-dyn-stats option is enabled:

```
Router(config)# voice hunt-group 1 parallel
Router(config-voice-hunt-group)# overwrite-dyn-stats
```
 overwrite-dyn-stats (voice hunt-group)
Cisco Unified CME Commands: P

• paging, on page 724
• paging group, on page 727
• paging-dn, on page 731
• paging-dn (voice register), on page 734
• param, on page 736
• param aa-hunt, on page 739
• param aa-pilot, on page 741
• param call-retry-timer, on page 743
• param co-did-max, on page 745
• param co-did-min, on page 747
• param dial-by-extension-option, on page 749
• param did-prefix, on page 751
• param drop-through-option, on page 753
• param drop-through-prompt, on page 755
• param ea-password, on page 757
• param handoff-string, on page 759
• param max-extension-length, on page 761
• param max-time-call-retry, on page 763
• param max-time-vm-retry, on page 766
• param menu-timeout, on page 768
• param number-of-hunt-grps, on page 770
• param queue-exit-extension, on page 772
• param queue-exit-option, on page 774
• param queue-len, on page 776
• param queue-manager-debugs, on page 778
• param queue-overflow-extension, on page 780
• param secondary-prefix, on page 782
• param second-greeting-time, on page 784
• param send-account true, on page 786
• param service-name, on page 788
• param store-did-max, on page 790
• param store-did-min, on page 792
• param voice-mail, on page 794
- param welcome-prompt, on page 796
- paramspace callsetup after-hours-exempt, on page 799
- park reservation-group, on page 801
- park-slot, on page 803
- password (auto-register), on page 808
- password-persistent, on page 810
- pattern (voice register dialplan), on page 811
- pattern direct, on page 813
- pattern ext-to-ext busy, on page 815
- pattern ext-to-ext no-answer, on page 817
- pattern trunk-to-ext busy, on page 819
- pattern trunk-to-ext no-answer, on page 821
- phone-display, on page 823
- phone-mode only, on page 824
- phone-key-size, on page 825
- phoneload, on page 826
- phoneload-support, on page 827
- phone-redirect-limit (voice register global), on page 828
- phone-ui park-list, on page 829
- phone-ui speeddial-fastdial, on page 830
- phone-ui voice-hunt-groups, on page 831
- pickup-call any-group, on page 832
- pickup-group, on page 833
- pilot, on page 835
- pilot (voice hunt-group), on page 837
- pin, on page 839
- pin (voice logout-profile and voice user-profile), on page 841
- pin (voice register pool), on page 843
- port (CAPF-server), on page 844
- preemption reserve timer, on page 845
- preemption tone timer (voice MLPP), on page 846
- preemption trunkgroup, on page 847
- preemption user, on page 848
- preference (ephone-dn), on page 849
- preference (ephone-hunt), on page 851
- preference (voice hunt-group), on page 853
- preference (voice register dn), on page 855
- preference (voice register pool), on page 857
- presence, on page 859
- presence call-list, on page 861
- presence enable, on page 863
- present-call, on page 864
- present-call (voice hunt-group), on page 866
- privacy (ephone), on page 867
- privacy (telephony-service), on page 869
- privacy (voice register global), on page 871
• privacy (voice register pool), on page 873
• privacy-button, on page 875
• privacy-button (voice register pool), on page 877
• privacy-on-hold, on page 879
• privacy-on-hold (voice register global), on page 880
• protocol mode, on page 881
• protocol-mode (telephony-service), on page 883
• provision-tag, on page 885
To define an extension (ephone-dn) as a paging extension that can be called to broadcast an audio page to a set of Cisco IP phones, use the `paging` command in ephone-dn configuration mode. To disable this feature, use the `no` form of this command.

```
paging [ip multicast-address port udp-port-number]
no paging [ip]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Argument</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ip multicast-address</code></td>
<td>(Optional) Uses an IP multicast address to multicast voice packets for audio paging; for example, 239.0.1.1. Note that IP phones do not support multicast at 224.x.x.x addresses. Default is that multicast is not used and IP phones are paged individually using IP unicast transmission (up to ten phones).</td>
</tr>
<tr>
<td><code>port udp-port-number</code></td>
<td>(Optional) Uses this UDP port for the multicast. Range is from 2000 to 65535. Default is 2000.</td>
</tr>
</tbody>
</table>

**Command Default**

No paging number is established.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

To configure a set of phones to receive an audio page, follow these steps:

1. Use the `paging` command in ephone-dn configuration mode to define a number that people can dial to send a page. The following example defines a paging-dn tag (21) and extension number (34455) to dial to send a page.

   ```
ephone-dn 21
   paging number 34455
   ```

1. Use the `paging-dn` command in ephone configuration mode to assign the same paging-dn tag that you defined in Step 1 to all the phones that you want to receive the page. This set of phones is called a “paging set.” You can have more than one paging set in a Cisco CME system. The following example assigns the paging-dn tag from Step 1 (21) to two phones (3 and 4) so that they will receive audio pages.

   ```
ephone 3
   paging-dn 21
   ephone 4
   paging-dn 21
   ```

The `paging` command configures an ephone-dn as an extension that people can dial to broadcast audio pages to a specified set of idle Cisco IP phones. The extension associated with this command does not appear on any ephone; it is a “dummy” extension. The dn-tag associated with this extension becomes the paging-dn tag for this paging set.
When a person dials the number assigned to the dummy extension and speaks into the phone, the audio stream is sent as a page to the paging set (the set of all phones that have been configured with this paging-dn tag as an argument to the `paging-dn` command). Idle phones in the paging set automatically answer the paging call in one-way speakerphone mode. Paging sets can be joined into a single combined paging group with the `paging group` command.

The optional `ip` keyword and `multicast-address` argument define a paging multicast address for this paging set. If an IP multicast address is not configured, IP phones are paged individually using IP unicast transmission (to a maximum of ten IP phones). The recommended operation is with an IP multicast address. When multiple paging-dn tags are configured using the `paging` command, each paging-dn tag should use a unique IP multicast address.

---

**Note**

IP phones do not support multicast at 224.x.x.x addresses.

---

Each ephone-dn and paging-dn tag that is used for paging can support a maximum of ten distinct targets (IP addresses and interfaces). A multicast address counts as a single target for each physical interface in use (regardless of the number of phones connected via the interface). Paging using a single IP multicast address that requires output on three different Ethernet interfaces represents use of three counts out of the maximum ten. Each unicast target counts as a single target, such that paging that does not use multicast at all is limited to paging ten phones. For example, ten IP phones paged through multicast on Fast Ethernet interface 0/1.1 plus five IP phones paged through multicast on Fast Ethernet interface 0/1.2 are counted as two targets.

For simultaneous paging to more than one paging ephone-dn, Cisco recommends that you use different IP multicast addresses (not just different port numbers) for paging configuration.

**Examples**

The following example creates a paging extension number that uses IP multicast paging:

```plaintext
Router(config)# ephone-dn 20
Router(config-ephone-dn) number 2000
Router(config-ephone-dn) paging ip 239.0.1.1 port 2000
```

A more complete configuration example follows, in which paging sets 20 and 21 are created. Pages to extension 2000 are multicast to ephones 1 and 2. Pages to extension 2001 are multicast to ephones 3 and 4.

```plaintext
ephone-dn 1
  number 2345
ephone-dn 2
  number 2346
ephone-dn 3
  number 2347
ephone-dn 4
  number 2348
ephone-dn 20
  number 2000
  paging ip 239.0.1.20 port 2000
ephone-dn 21
  number 2001
  paging ip 239.0.1.21 port 2000
ephone 1
  button 1:1
  paging-dn 20
  ephone 2
  button 1:2
```
paging-dn 20
ephone 3
button 1:3
paging-dn 21
ephone 4
button 1:4
paging-dn 21

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>paging-dn</td>
<td>Assigns audio paging reception capability to a Cisco IP phone.</td>
</tr>
<tr>
<td>paging group</td>
<td>Combines two or more paging sets into a combined paging group.</td>
</tr>
</tbody>
</table>
paging group

To create a combined paging group from two or more previously established paging sets, use the `paging group` command in ephone-dn configuration mode. To remove a paging group, use the `no` form of this command.

```
paging group paging-dn-tag, paging-dn-tag...
```

```
no paging group
```

**Syntax Description**

- `paging-dn-tag` Comma-separated list of paging-dn-tags (unique sequence numbers associated with paging ephone-dns) that have previously been associated with the paging extension of a paging set using the `paging-dn` or `paging-dn (voice register)` command. You can include up to ten paging-dn-tags separated by commas. For example, 4, 6, 7, 8.

**Command Default**
Paging is disabled on all Cisco IP phones.

**Command Modes**
Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>15.2(2)T</td>
<td>Cisco Unifide CME 9.0</td>
<td>This command was modified to include voice register pools in the ephone-dn and paging groups.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to combine previously defined sets of phones associated with individual paging extensions (ephone-dns) into a combined group to enable a single page to be sent to large numbers of phones at once. To remove a paging group, use the `no` form of the command. All paging-dn tags included in the list must have already been defined as paging-dns using the `paging` or `paging-dn (voice register)` command.

The use of paging groups not only allows phones to participate in a small local paging set (for example, paging to four phones in a company’s shipping and receiving department) but also supports company-wide paging when needed (for example, by combining the paging sets for shipping and receiving with the paging sets for accounting, customer support, and sales into a single paging group).

**Note**
The correct paging port for the paging-dn of Cisco Unified SIP IP phones in the `paging` command is an even number from 20480 to 32768. If you enter a wrong port number, a SIP REFER message request is sent to the IP phone but the Cisco Unified SIP IP phone is not paged.

**Examples**

In the following example, paging sets 20 and 21 are defined and then combined into paging group 22. Paging set 20 has a paging extension of 2000. When someone dials extension 2000 to deliver a page, the page is sent to Cisco IP phones (ephones) 1 and 2. Paging set 21 has a paging extension of 2001. When someone dials extension 2001 to deliver a page, the page is sent to ephones 3 and 4.
Paging group 22 combines sets 20 and 21, and when someone dials its paging extension, 2002, the page is sent to all the phones in both sets and to ephone 5, which is directly subscribed to the combined paging group.

```plaintext
ephone-dn 20
  number 2000
  paging ip 239.0.1.20 port 2000
ephone-dn 21
  number 2001
  paging ip 239.0.1.21 port 2000
ephone-dn 22
  number 2002
  paging ip 239.0.2.22 port 2000
paging group 20,21
ephone 1
  button 1:1
  paging-dn 20
ephone 2
  button 1:2
  paging-dn 20
ephone 3
  button 1:3
  paging-dn 21
ephone 4
  button 1:4
  paging-dn 21
ephone 5
  button 1:5
  paging-dn 22
```

The following example shows how the **paging group** command is used to configure combined paging groups composed of ephone and voice register directory numbers.

The first set of configuration tasks shows how to configure a combined paging group composed of Cisco Unified SCCP IP phone directory numbers only.

When extension 2000 is dialed, a page is sent to ephones 1 and 2 (first single paging group). When extension 2001 is dialed, a page is sent to ephones 3 and 4 (second single paging group). Finally, when extension 2002 is dialed, a page is sent to ephones 1, 2, 3, 4, and 5, producing the combined paging group (composed of the first single paging group, the second single paging group, and ephone 5).

Ephones 1 and 2 are included in paging ephone-dn 22 through the membership of ephone-dn 20 as paging group 20 in the combined paging group. Ephones 3 and 4 are included in paging ephone-dn 22 through membership of ephone-dn 21 as paging group 21 in the combined paging group. Ephone 5 is directly subscribed to paging-dn 22.

```plaintext
ephone-dn 20
  number 2000
  paging ip 239.0.1.20 port 20480
ephone-dn 21
  number 2001
  paging ip 239.1.1.21 port 20480
ephone-dn 22
  number 2002
  paging ip 239.1.1.22 port 20480
paging group 20,21
ephone-dn 6
  number 1103
```
The second set of configuration tasks shows how Cisco Unified SIP IP phone directory numbers can be configured and added to the previously established paging groups of the first set of configuration tasks to form a new combined paging group composed of ephone and voice register directory numbers.

When extension 2000 is dialed, a page is sent to ephones 1 and 2 and voice register pools 1 and 2 (new first single paging group). When extension 2001 is dialed, a page is sent to ephones 3 and 4 and voice register pools 3 and 4 (new second single paging group). Finally, when extension 2002 is dialed, a page is sent to ephones 1, 2, 3, 4, and 5 and voice register pools 1, 2, 3, 4, and 5 (new combined paging group).

Ephones 1 and 2 and voice register pools 1 and 2 are included in paging ephone-dn 22 through the membership of ephone-dn 20 as paging group 20 in the combined paging group. Ephones 3 and 4 and voice register pools 3 and 4 are included in paging ephone-dn 22 through membership of ephone-dn 21 as paging group 21 in the combined paging group. Ephone 5 and voice register pool 5 are directly subscribed to paging-dn 22.
Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>paging</td>
<td>Creates a paging extension (ephone-dn) that can be called in order to broadcast an audio page to a group of Cisco IP phones.</td>
</tr>
<tr>
<td>paging-dn</td>
<td>Assigns a paging extension (paging-dn) to a Cisco IP phone.</td>
</tr>
<tr>
<td>paging-dn (voice register)</td>
<td>Registers a Cisco Unified SIP IP phone to an ephone-dn paging directory number.</td>
</tr>
</tbody>
</table>
**paging-dn**

To create a paging extension (paging-dn) to receive audio pages on a Cisco Unified IP phone in a Cisco Unified CME system, use the **paging-dn** command in ephone or ephone-template configuration mode. To disable this feature, use the **no** form of this command.

**paging-dn paging-dn-tag {multicast|unicast}

**paging-dnno**

<table>
<thead>
<tr>
<th>Syntax Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>paging-dn-tag</strong> Dn-tag of an ephone-dn that was designated as a paging ephone-dn with the <strong>paging</strong> command.</td>
</tr>
<tr>
<td><strong>multicast</strong> Uses multicast if available. By default, audio paging is transmitted to the Cisco Unified IP phone using multicast.</td>
</tr>
<tr>
<td><strong>unicast</strong> Forces unicast paging for this phone. This keyword indicates that the Cisco Unified IP phone is not capable of receiving audio paging through multicast and requests that all pages to this phone be sent through unicast. The maximum number of phones supported through unicast is ten.</td>
</tr>
</tbody>
</table>

**Command Default**

Paging is disabled on all Cisco Unified IP phones.

**Command Modes**

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

<table>
<thead>
<tr>
<th>Command History</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Cisco IOS Release</strong></td>
</tr>
<tr>
<td>12.2(2)XT</td>
</tr>
<tr>
<td>12.2(8)T</td>
</tr>
<tr>
<td>12.4(4)XC</td>
</tr>
<tr>
<td>12.4(9)T</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

To configure a set of phones to receive an audio page, follow these steps:

1. Use the **paging** command in ephone-dn configuration mode to define a number that people can dial to send a page. The following example defines a paging-dn tag (21) and extension number (34455) to dial to send a page.

   ephone-dn 21
   paging
   number 34455

1. Use the **paging-dn** command in ephone configuration mode to assign the same paging-dn tag that you defined in Step 1 to all the phones that you want to receive the page. This set of phones is called a “paging set.” You can have more than one paging set in a Cisco Unified CME system. The following example assigns the paging-dn tag from Step 1 (21) to two phones (3 and 4) so that they will receive audio pages.
This command creates a paging extension (paging-dn) associated with an IP phone. Each phone can support only one paging-dn extension. This extension does not occupy a phone button and is therefore not configured on the phone with the button command. The paging-dn allows the phone to automatically answer audio pages in one-way speakerphone mode. There is no press-to-answer option as there is with an intercom extension.

The paging-dn-tag argument in this command takes the value of the dn-tag of an extension (ephone-dn) that has been made a paging ephone-dn using the paging command. This command is the extension that callers dial to deliver an audio page. All of the phones that are going to receive the same audio pages are configured with the same paging-dn-tag. These phones form a paging set.

An IP phone can belong to only one paging set, but any number of phones can belong to the same paging set using multicast. There can be any number of paging sets in a Cisco Unified CME system, and paging sets can be joined to create a combined paging group using the paging group command. For example, you may create separate paging sets for each department (sales, support, shipping) and combine them into a single combined paging group (all departments). Only single-level grouping is supported (no support for groups of groups).

Normal phone calls that are received while an audio page is in progress interrupt the page.

The paging mechanism supports audio distribution using IP multicast, replicated unicast, and a mixture of both (so that multicast is used where possible, and unicast is used with specific phones that cannot be reached through multicast).

For unicast paging to all phones, omit the IP multicast address in the ephone-dn configuration. For unicast paging to a specific phone using an ephone-dn configured for multicast, add the unicast keyword as part of the paging-dn command in ephone configuration mode.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

The following example creates paging number 5001 on ephone-dn 22 and adds ephone 4 as a member of the paging set. Multicast is set for the paging-dn. Note that IP phones do not support multicast at 224.x.x.x addresses.

```
ephone-dn 1
  number 5123
ephone-dn 22
  name Paging Shipping
  number 5001
  paging ip 239.1.1.10 port 2000
ephone 4
  mac-address 0030.94c3.8724
  button 1:1
  paging-dn 22 multicast
```

### Related Commands

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies a template to an ephone configuration.</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>-----------------------------</td>
<td>--------------------------------------</td>
</tr>
<tr>
<td><strong>number</strong></td>
<td>Configures a valid number for the Cisco Unified IP phone.</td>
</tr>
<tr>
<td><strong>paging</strong></td>
<td>Creates a paging extension (ephone-dn) that can be called in order to broadcast an audio page to a group of Cisco Unified IP phones.</td>
</tr>
<tr>
<td><strong>paging group</strong></td>
<td>Combines two or more paging sets into a combined paging group.</td>
</tr>
</tbody>
</table>
**paging-dn (voice register)**

To register a Cisco Unified SIP IP phone to an ephone-dn paging directory number (DN), use the `paging-dn` command in voice register pool or voice register template configuration mode. To unregister the Cisco Unified SIP IP phone from the paging directory number, use the `no` form of this command.

```
    paging-dn  paging-dn-tag  {multicast|unicast}
    no  paging-dn
```

### Syntax Description

<table>
<thead>
<tr>
<th><strong>paging-dn-tag</strong></th>
<th>Ephone-dn tag designated as the paging ephone-dn to which a Cisco Unified SIP IP phone is registered.</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>multicast</strong></td>
<td>Transmits audio paging to the Cisco Unified IP phone using multicast. This is the default.</td>
</tr>
<tr>
<td><strong>unicast</strong></td>
<td>Transmits audio paging to the Cisco Unified IP phone using unicast. This indicates that the Cisco Unified IP phone is not capable of receiving audio paging through multicast and requests that all pages to this phone be sent through unicast. The maximum number of phones supported through unicast is 12.</td>
</tr>
</tbody>
</table>

### Command Default

The Cisco Unified SIP IP phone is not registered to an ephone-dn paging DN and paging is disabled.

### Command Modes

Voice register pool configuration (config-register-pool)

Voice register template configuration (config-register-temp)

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

The `paging-dn` command applies to both voice register pool and voice register template configuration modes. When voice register pool is configured with the template and paging is configured in voice register pool configuration mode, paging in voice register pool configuration mode has higher precedence over paging in voice register template configuration mode.

The correct paging port for the paging-dn of Cisco Unified SIP phones in the `paging` command is an even number from 20480 to 32768. If you enter a wrong port number, a SIP REFER message request is sent to the IP phone but the Cisco Unified SIP IP phone is not paged.

### Examples

The following example shows how the Cisco Unified 7961 SIP IP phone is registered to both paging-dns 251 and 252:

```
ephone-dn  2  dual-line
    number  60012
ephone-dn  250
    number  7770
    paging ip 239.1.1.0 port 20480
    paging group 251,252
ephone-dn  251
    number  7771
    paging ip 239.1.1.1 port 20480
ephone-dn  252
```
number 7772
paging ip 239.1.1.2 port 20480
ephone-dn 253
number 7773
paging ip 239.1.1.3 port 20480
ephone 2
mac-address 001E.4A91.F27D
paging-dn 252
type 7961
button 1:2
voice register dn 1
number 60001
voice register dn 2
number 60002
voice register pool 1
id mac 0019.305D.82B8
type 7961
number 1 dn 1
codec g711ulaw
paging-dn 251
voice register pool 2
id mac 0019.305D.2153
type 7961
number 1 dn 2
codec g711ulaw
paging-dn 252

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>paging-dn</td>
<td>Creates a paging extension to receive audio pages on a Cisco Unified SCCP IP phone in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>paging group</td>
<td>Creates a combined paging group from two or more previously established paging sets.</td>
</tr>
</tbody>
</table>
**param**

To load and configure parameters in a package or a service (application) on the gateway, use the `param` command in application configuration mode. To reset a parameter to its default value, use the `no` form of this command.

```
param param-name [{param max-retries|param passwd|param passwd-prompt filename|param user-prompt filename|param term-digit|param abort-digit|param max-digits}]
```

**no param param-name**

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>param-name</code></td>
<td>Name of the parameter.</td>
</tr>
<tr>
<td><code>param max-retries</code></td>
<td>(Optional) Number of attempts to re-enter account or password. Value ranges from 0-10, default value is 0.</td>
</tr>
<tr>
<td><code>param passwd</code></td>
<td>(Optional) Character string that defines a predefined password for authorization.</td>
</tr>
<tr>
<td><code>param passwd-prompt filename</code></td>
<td>(Optional) Announcement URL to request password input. filename defines the name and location of the audio filename to be used for playing the password prompt.</td>
</tr>
<tr>
<td><code>param user-prompt filename</code></td>
<td>(Optional) Announcement URL to request authorization code. filename defines the name and location of the audio filename to be used for playing the username prompt.</td>
</tr>
<tr>
<td><code>param term-digit</code></td>
<td>Digit for terminating username or password digit input.</td>
</tr>
<tr>
<td><code>param abort-digit</code></td>
<td>Digit for aborting username or password digit input. Default value is *.</td>
</tr>
<tr>
<td><code>param max-digits</code></td>
<td>Maximum number of digits in a username or password. Range of valid value: 1 - 32. Default value is 32.</td>
</tr>
</tbody>
</table>

### Command Default

No default behavior or value.

### Command Modes

Application configuration

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(3)T</td>
<td>This command was modified. The following keywords and arguments were added: param max-retries, param passwd, param passwd-prompt filename, param user-prompt filename, param term-digit, param max-digit.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use this command in application parameter configuration mode to configure parameters in a package or service. A package is a linkable set of C or Tcl functions that provide functionality invoked by applications or other packages. A service is a standalone application.
The parameters available for configuration differ depending on the package or service that is loaded on the gateway. The **param register** Tcl command in a service or package registers a parameter and provides a description and default values which allow the parameter to be configured using the CLI. The **param register** command is executed when the service or package is loaded or defined, along with commands such as **package provide**, which register the capability of the configured module and its associated scripts. You must configure and load the Tcl scripts for your service or package and load the package in order to configure its parameters. See the *Tcl IVR API Version 2.0 Programming Guide* for more information.

When a package or service is defined on the gateway, the parameters in that package or service become available for configuration when you use this command. Additional arguments and keywords are available for different parameters. To see a list of available parameters, enter **param ?**.

To avoid problems with applications or packages using the same parameter names, the **parameter namespace**, or **parameterspace** concept is introduced. When a service or a package is defined on the gateway, its parameter namespace is automatically defined. This is known as the service or package’s local parameterspace, or “myparameterspace.” When you use this command to configure a service or package’s parameters, the parameters available for configuration are those contained in the local parameterspace. If you want to use parameter definitions found in different parameterspace, you can use the **paramspace parameter-namespace** command to map the package’s parameters to a different parameterspace. This allows that package to use the parameter definitions found in the new parameterspace, in addition to its local parameterspace.

Use this command in Cisco Unified Communication Manager Express 8.5 and later versions to define the username and password parameters to authenticate packages for Forced Authorization Code (FAC)

When a pre-defined password is entered using the param passwd keyword, callers are not requested to enter a password. You must define a filename for user-prompt to play an audio prompt requesting the caller to enter a valid username (in digits) for authorization. Similarly, you must define a filename for passwd-prompt to play an audio prompt requesting the caller to enter a valid password (in digits) for authorization.

### Examples

The following example shows how to configure a parameter in the httpios package:

```
application
package httpios
param paramA value4
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call application voice</td>
<td>Defines the name of a voice application and specify the location of the Tcl or VoiceXML document to load for this application.</td>
</tr>
<tr>
<td>param account-id-method</td>
<td>Configures an application to use a particular method to assign the account identifier.</td>
</tr>
<tr>
<td>param convert-discpi-after-connect</td>
<td>Enables or disables conversion of a DISCONNECT message with Progress Indicator set to PROG_INBAND (PI=8) to a regular DISCONNECT message when the call is in the active state.</td>
</tr>
<tr>
<td>param event-log</td>
<td>Enables or disables logging for linkable Tcl functions (packages).</td>
</tr>
<tr>
<td>param language</td>
<td>Configures the language parameter in a service or package on the gateway.</td>
</tr>
<tr>
<td>param mode</td>
<td>Configures the call transfer mode for a package.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>param pin-len</td>
<td>Defines the number of characters in the personal identification number (PIN) for an application.</td>
</tr>
<tr>
<td>param redirect-number</td>
<td>Defines the telephone number to which a call is redirected—for example, the operator telephone number of the service provider—for an application.</td>
</tr>
<tr>
<td>param reroutemode</td>
<td>Configures the call transfer reroutemode (call forwarding) for a package.</td>
</tr>
<tr>
<td>param retry-count</td>
<td>Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.</td>
</tr>
<tr>
<td>param security</td>
<td>Configures security for linkable Tcl functions (packages).</td>
</tr>
<tr>
<td>paramspace</td>
<td>Enables an application to use parameters from the local parameter space of another application.</td>
</tr>
<tr>
<td>param uid-length</td>
<td>Defines the number of characters in the UID for a package.</td>
</tr>
<tr>
<td>param warning-time</td>
<td>Defines the number of seconds of warning that a user receives before the allowed calling time expires.</td>
</tr>
</tbody>
</table>
param aa-hunt

To declare a Cisco Unified CME B-ACD menu number and associate it with the pilot number of an ephone hunt group, use the `param aa-hunt` command in application-parameter configuration mode. To remove the menu number and the ephone hunt group pilot number, use the `no` form of this command.

```
param aa-hunt menu-number pilot-number
no param aa-hunt menu-number pilot-number
```

**Syntax Description**

<table>
<thead>
<tr>
<th>menu-number</th>
<th>Number that callers must dial to reach the pilot number of an ephone hunt group. The range is from 1 to 10. The default is 1.</th>
</tr>
</thead>
<tbody>
<tr>
<td>pilot-number</td>
<td>Ephone hunt group pilot number.</td>
</tr>
</tbody>
</table>

**Command Default**

Menu number 1 is configured, but it is not associated with a pilot number.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the call application voice aa-hunt command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. It is configured under the `service` command for the call-queue script.

Up to ten aa-hunt menu options, or hunt groups, are allowed per call-queue service. You can use any of the allowable numbers in any order.

This command associates a menu option with the pilot number of an ephone hunt group. When a caller presses the digit of a menu option that has been associated with an ephone hunt group using this command, the call is routed to the pilot number of the hunt group.

Menu options for B-ACD services can be set up in many ways. For more information, see the Cisco Unified CallManager Express B-ACD and Tel Call-Handling Applications document for your release.

The highest aa-hunt number that you establish using this command also automatically maps to zero (0) and can therefore be used to represent operator services to your callers. In the following example, callers can dial either 8 or 0 to reach aa-hunt8, the hunt group with the pilot number 8888.

```
application
service queue flash:
  param aa-hunt1 1111
  param aa-hunt3 3333
  param aa-hunt8 8888
```

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

**Examples**

The following example configures a call-queue service called queue to associate three menu numbers with three pilot numbers of three ephone hunt groups:

- Pilot number 1111 for ephone hunt group 1 (sales)
• Pilot number 2222 for ephone hunt group 2 (customer service)
• Pilot number 3333 for ephone hunt group 3 (operator)

If a caller presses 2 for customer service, the call is transferred to 2222 and then is sent to the next available ephone-dn from the group of ephone-dns assigned to ephone hunt group 1: 2001, 2002, 2003, 2004, 2005, and 2006. The sequencing of ephone-dns within a hunt group is handled by the ephone hunt group itself, not by the B-ACD service. (Note that the configuration for ephone hunt groups used with Cisco Unified CME B-ACD services do not use the `final` command.)

```plaintext
ephone-hunt 1 peer
    pilot 1111
    list 1001, 1002, 1003, 1004, 1005, 1006, 1007, 1008, 1009, 1010

ephone-hunt 2 peer
    pilot 2222

ephone-hunt 3 peer
    pilot 3333
    list 3001, 3002, 3003, 3004

application
    service queue flash:
      param aa-hunt1 1111
      param aa-hunt2 2222
      param aa-hunt3 3333

```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
param aa-pilot

To assign a pilot number to a Cisco Unified CME B-ACD automated attendant (AA) service, use the `param aa-pilot` command in application-parameter configuration mode. To remove the AA pilot number, use the `no` form of this command.

```
param aa-pilot aa-pilot-number
no param aa-pilot aa-pilot-number
```

**Syntax Description**

| `aa-pilot-number` | Telephone number that callers dial in order to reach this AA service. |

**Command Default**

Cisco Unified CME B-ACD menu number 1 is configured, but it has no pilot number.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the <code>call application voice aa-pilot</code> command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. Each AA has one AA pilot number, although there may be more than one AA used with a B-ACD service.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

**Examples**

The following example sets up a B-ACD with two AAs, both in drop-through mode. The first AA is called acdaa and it has an AA pilot number of (800) 555-0121. The second AA is aa-bcd and has an AA pilot number of (800) 555-0123. Both AAs use the call-queue service named callq. Incoming POTS dial peers are established for both AA pilot numbers.

```
dial-peer voice 1010 pots
    service acdaa
    port 1/1/0
    incoming called-number 8005550121

dial-peer voice 1020 pots
    service aa-bcd
    port 1/1/1
    incoming called-number 8005550123

application
    service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
    param queue-manager-debugs 1
    param aa-hunt1 5071
    param aa-hunt2 5072
    param number-of-hunt-grps 2
    param queue-len 10
```

Cisco Unified Communications Manager Express Command Reference
! service acdaa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
paramspace english location tftp://192.168.254.254/user1/prompts/
paramspace english index 0
paramspace english language en
param aa-pilot 8005550121
param service-name callq
param max-time-vm-retry 2
param voice-mail 5007
param call-retry-timer 20
param number-of-hunt-grps 1
param drop-through-prompt _bacd_welcome.au
param drop-through-option 2
param second-greeting-time 45
param handoff-string acdaa
param max-time-call-retry 360
!
! service aa-bcd tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
paramspace english location tftp://192.168.254.254/user1/prompts/
paramspace english index 0
paramspace english language en
param aa-pilot 8005550123
param service-name callq
param second-greeting-time 60
param max-time-call-retry 180
param max-time-vm-retry 2
param voice-mail 5007
param call-retry-timer 5
param handoff-string aa-bcd
param drop-through-option 1
param number-of-hunt-grps 1

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>application</strong></td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td><strong>service</strong></td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
**param call-retry-timer**

To specify the time interval before each attempt to retry to connect a call to an ephone hunt group used with a Cisco CME B-ACD service, use the `param call-retry-timer` command in application-parameter configuration mode. To return to the default, use the `no` form of this command.

```
param call-retry-timer  seconds
no param call-retry-timer  seconds
```

**Syntax Description**

<table>
<thead>
<tr>
<th>seconds</th>
<th>Time that a call must wait before attempting or reattempting to transfer a call to an ephone hunt group pilot number, in seconds. Range is from 5 to 30 seconds.</th>
</tr>
</thead>
</table>

**Command Default**

Default is 15 seconds.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the <code>call application voice call-retry-timer</code> command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the `service` command for an AA service. A Cisco Unified CME B-ACD service can have more than one AA, and each AA can specify a different interval for retries to connect to ephone hunt group phones.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

**Examples**

The following example sets up a B-ACD with two AAs. The first AA is called acdaa and it has an AA pilot number of (800) 555-0121. The second AA is aa-bcd and has an AA pilot number of (800) 555-0123. Both AAs use the call-queue service named callq. The first AA has a call-retry timer set to 10 seconds, and the second AA has a call-retry timer set to 5 seconds.

```
dial-peer voice 1010 pots
  service acdaa
  port 1/1/0
  incoming called-number 8005550121

```

```
dial-peer voice 1020 pots
  service aa-bcd
  port 1/1/1
  incoming called-number 8005550123

```

```
application
  service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param aa-hunt2 5072
```
```plaintext
param number-of-hunt-grps 2	param queue-len 10
!
service acdaa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
paramspace english location tftp://192.168.254.254/user1/prompts/
paramspace english index 0
paramspace english language en
param aa-pilot 8005550121
param service-name callq
param max-time-vm-retry 2
param voice-mail 5007
param call-retry-timer 10
param number-of-hunt-grps 1
param drop-through-prompt _bacd_welcome.au
param drop-through-option 2
param second-greeting-time 45
param handoff-string acdaa
param max-time-call-retry 60
!
service aa-bcd tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
paramspace english location tftp://192.168.254.254/user1/prompts/
paramspace english index 0
paramspace english language en
param aa-pilot 8005550123
param service-name callq
param second-greeting-time 60
param max-time-call-retry 180
param max-time-vm-retry 2
param voice-mail 5007
param call-retry-timer 5
param handoff-string aa-bcd
param drop-through-option 1
param number-of-hunt-grps 1
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
**param co-did-max**

To set the upper boundary of the range of valid digits coming from the PSTN Central Office (CO) for use with the Direct Inward Dial (DID) Digit Translation Service, use the `param co-did-max` command in application-parameter configuration mode. To disable this option, use the `no` form of this command.

```
param co-did-max  max-co-value
no param co-did-max  max-co-value
```

| Syntax Description | max-co-value | Maximum value of digits coming from the CO. The digit string can be any length, but the string length must be the same in the `param co-did-min`, `param co-did-max`, `param store-did-min`, and `param store-did-max` commands. |

| Command Default | No maximum value is defined for the range of digits coming from the CO. |

| Command Modes | Application-parameter configuration (config-app-param) |

<table>
<thead>
<tr>
<th>Command History</th>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced to replace the <code>call application voice co-did-max</code> command.</td>
<td></td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command replaced the <code>call application voice co-did-max</code> command and was integrated into Cisco IOS Release 12.4(9)T.</td>
<td></td>
</tr>
</tbody>
</table>

| Usage Guidelines | This command defines the upper limit of the range of digits accepted from the CO when it is used with the Cisco Unified CallManager Express (Cisco Unified CME) DID Digit Translation Service. This service provides number translation for DID calls when the range of DID digits provided by the PSTN Central Office (CO) does not match the digits in the Cisco Unified CME extension numbers. The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers: a prompt is played and the calls are disconnected. |

| Examples | The following example configures DID Digit Translation Service on the Cisco Unified CME router. It sets a lower boundary of 00 and an upper boundary of 79 for the valid range of digits coming from the CO. |

```
application
  service didapp tftp://192.168.254.254/scripts/did/app-TND-DID-2.0.0.1.tcl
  paramspace english index 1
  paramspace english language en
  paramspace english location tftp://192.168.254.254/apps/dir25/
  param secondary-prefix 4
  param did-prefix 5
  param co-did-min 00
  param co-did-max 79
```
param store-did-min 00
param store-did-max 79

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>application</strong></td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td><strong>service</strong></td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
<tr>
<td><strong>param co-did-min</strong></td>
<td>Sets the lower boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the DID Digit Translation Service.</td>
</tr>
<tr>
<td><strong>param store-did-max</strong></td>
<td>Sets the upper boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the DID Digit Translation Service.</td>
</tr>
<tr>
<td><strong>param store-did-min</strong></td>
<td>Sets the lower boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the DID Digit Translation Service.</td>
</tr>
</tbody>
</table>
**param co-did-min**

To set the lower boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the Direct Inward Dial (DID) Digit Translation Service, use the `param co-did-min` command in application-parameter configuration mode. To disable this option, use the `no` form of this command.

```
param co-did-min min-co-value
no param co-did-min min-co-value
```

**Syntax Description**

| min-co-value | Minimum value of digits coming from the CO. The digit string can be any length, but the string length must be the same in the `param co-did-max`, `param co-did-max`, `param store-did-min`, and `param store-did-max` commands. |

**Command Default**

No minimum value is defined for the range of digits coming from the CO.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced to replace the <code>call application voice co-did-min</code> command.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command replaced the <code>call application voice co-did-min</code> command and was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command defines the upper limit of the range of digits accepted from the CO when it is used with the Cisco Unified CallManager Express (Cisco Unified CME) DID Digit Translation Service. This service provides number translation for DID calls when the range of DID digits provided by the PSTN Central Office (CO) does not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers: a prompt is played and the calls are disconnected.

**Examples**

The following example configures DID Digit Translation Service on the Cisco Unified CME router. It sets a lower boundary of 00 and an upper boundary of 79 for the valid range of digits coming from the CO.

```application
service didapp tftp://192.168.254.254/scripts/did/app-THD-DID-2.0.0.1.tcl
paramspace english index 1
paramspace english language en
paramspace english location tftp://192.168.254.254/apps/dir25/
param secondary-prefix 4
param did-prefix 5
param co-did-min 00
param co-did-max 79
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
<tr>
<td>param co-did-max</td>
<td>Sets the upper boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the DID Digit Translation Service.</td>
</tr>
<tr>
<td>param store-did-max</td>
<td>Sets the upper boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the DID Digit Translation Service.</td>
</tr>
<tr>
<td>param store-did-min</td>
<td>Sets the lower boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the DID Digit Translation Service.</td>
</tr>
</tbody>
</table>
**param dial-by-extension-option**

To assign a menu number to a Cisco CME B-ACD option by which callers can directly dial known extension numbers, use the **param dial-by-extension-option** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

**Syntax Description**

| **menu-number** | Menu option number to be associated with the dial-by-extension option. Range is from 1 to 9. There is no default. |

**Command Default**

Dial-by-extension option is not assigned.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the call application voice dial-by-extension-option command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

This command allows you to designate a menu option number for callers to press if they want to dial an extension number that they already know. This command also enables the playing of the `en_bacd_enter_dest.au` audio file after a caller presses the dial-by-extension menu number. The default announcement in this audio file is “Please enter the extension number you want to reach.”

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

**Examples**

The following example sets up a B-ACD with an AA called acd1, which has an AA pilot number of (800) 555-0121. The call-queue service used with this AA is named callq. Callers to (800) 555-0121 can press 1 to dial an extension number (**param dial-by-extension-option** under **service acd1**) or press 2 to be connected to the hunt group with the pilot number 5072 (**param aa-hunt2 5072** under **service callq**).

```
dial-peer voice 1010 pots
  service acd1
  port 1/1/0
  incoming called-number 8005550121
.
.
application
  service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt2 5072
```
param number-of-hunt-grps 1
param queue-len 10
!
service acd1 tftp://192.168.254.254/user1/CallQ/ACD/app-b-acd-aa.tcl
paramspace english location tftp://192.168.254.254/user1/prompts/
paramspace english index 0
paramspace english language en
param handoff-string acd1
param service-name callq
param aa-pilot 8005550121
param number-of-hunt-grps 1
param dial-by-extension-option 1
param second-greeting-time 45
param call-retry-timer 20
param max-time-call-retry 360
param max-time-vm-retry 2
param voice-mail 5007

<table>
<thead>
<tr>
<th>Related Commands</th>
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</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
**param did-prefix**

To set a prefix to add to digits coming from the PSTN Central Office (CO) to create valid extension numbers when using the Direct Inward Dial (DID) Digit Translation Service, use the `param did-prefix` command in application-parameter configuration mode. To disable this option, use the `no` form of this command.

```yaml
param did-prefix did-prefix
no param did-prefix did-prefix
```

**Syntax Description**

- `did-prefix` Prefix to add. Range is from 0 to 99.

**Command Default**

No prefix is defined.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME4.0</td>
<td>This command was introduced to replace the call application voice did-prefix command.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME4.0</td>
<td>This command replaced the call application voice did-prefix command and was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Cisco Unified CallManager Express (Cisco Unified CME) DID Digit Translation Service, which provides number translation for DID calls when the range of DID digits provided by the PSTN Central Office (CO) does not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers: a prompt is played and the calls are disconnected.

**Examples**

The following example configures DID Digit Translation Service on the Cisco Unified CME router. It specifies that a prefix of 5 should be applied to the digits coming from the CO in order to construct a valid extension number.

```bash
application
service didapp tftp://192.168.254.254/scripts/did/app-THD-DID=2.0.0.1.tcl
paramspace english index 1
paramspace english language en
paramspace english location tftp://192.168.254.254/apps/dir25/
param secondary-prefix 4
param did-prefix 5
param co-did-min 00
param co-did-max 79
param store-did-min 00
param store-did-max 79
```
<table>
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</tr>
</thead>
<tbody>
<tr>
<td><strong>application</strong></td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td><strong>service</strong></td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
param drop-through-option

To assign the drop-through option to a Cisco Unified CME B-ACD auto-attendant (AA) application, use the `param drop-through option` command in application-parameter configuration mode. To disable this option, use the `no` form of this command.

```
param drop-through-option menu-number
no param drop-through-option menu-number
```

### Syntax Description

| `menu-number` | Menu option number (aa-hunt number) to be associated with the drop-through option. |

### Command Default

Drop-through option is not assigned.

### Command Modes

Application-parameter configuration (config-app-param)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the <code>call application voice drop-through-option</code> command.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the `service` command for an AA service.

When an AA is configured for drop-through mode, the AA sends incoming calls directly to the call queue associated with the menu number specified in this command. Once in the queue, a caller hears ringback if an agent is available or music on hold (MOH) if all agents are busy. If a greeting prompt for drop-through mode is configured using the `param drop-through-prompt` command, a caller hears the prompt before being sent to the queue as described.

The menu option number is an aa-hunt number that is associated with an ephone hunt group using the `param aa-hunt` command.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

### Examples

The following example sets up a B-ACD with two AAs, both in drop-through mode. The first AA is called acdaa and it has an AA pilot number of (800) 555-0121. The second AA is aa-bcd and has an AA pilot number of (800) 555-0123. Both AAs use the call-queue service named callq. Callers to (800) 555-0121 drop directly through to the hunt group with the pilot number 5072 after hearing the greeting prompt in the audio file named en DTO welcome.au. Callers to (800) 555-0123 drop directly through to the hunt group with the pilot number 5071 without hearing any greeting.

```
dial-peer voice 1010 pots
  service acdaa
  port 1/1/0
  incoming called-number 8005550121

dial-peer voice 1020 pots
  service aa-bcd
  port 1/1/1
```
incoming called-number 8005550123

application

service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
    param queue-manager-debug 1
    param aa-hunt1 5071
    param aa-hunt2 5072
    param number-of-hunt-grps 2
    param queue-len 10

service acdaa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
    paramspace english language en
    param aa-pilot 8005550121
    param service-name callq
    param max-time-vm-retry 2
    param voice-mail 5007
    param call-retry-timer 20
    param number-of-hunt-grps 1
    param drop-through-prompt _bacd dto_welcome.au
    param drop-through-option 2
    param second-greeting-time 45
    param handoff-string acdaa
    param max-time-call-retry 360

service aa-bcd tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
    paramspace english language en
    param aa-pilot 8005550123
    param service-name callq
    param second-greeting-time 60
    param max-time-call-retry 180
    param max-time-vm-retry 2
    param voice-mail 5007
    param call-retry-timer 5
    param handoff-string aa-bcd
    param drop-through-option 1
    param number-of-hunt-grps 1

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
param drop-through-prompt

To associate an audio prompt file with the drop-through option for a Cisco Unified CME B-ACD automated attendant (AA) application, use the `param drop-through-prompt` command in application-parameter configuration mode. To disable the prompt, use the `no` form of this command.

```
param drop-through-prompt audio-filename
no param drop-through-prompt audio-filename
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>audio-filename</code></td>
<td>Identifying part of the filename of the prompt to be played when calls for the drop-through option are answered.</td>
</tr>
</tbody>
</table>

**Command Default**

No prompt is designated for the drop-through option.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
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<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the <code>call application voice drop-through-prompt</code> command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the `service` command for an AA service.

When an AA is configured for drop-through mode, the AA sends incoming calls directly to the call queue associated with the menu number specified in this command. Once in the queue, a caller hears ringback if an agent is available or music on hold (MOH) if all agents are busy. If an greeting prompt for drop-through mode is configured, a caller hears the prompt before being sent to the queue as described.

The menu option number is an aa-hunt number that is associated with an ephone hunt group using the `param aa-hunt` command.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

**Examples**

The following example sets up a B-ACD with two AAs, both in drop-through mode. The first AA is called acdaa and it has an AA pilot number of (800) 555-0121. The second AA is aa-bcd and has an AA pilot number of (800) 555-0123. Both AAs use the call-queue service named callq. Callers to (800) 555-0121 drop directly through to the hunt group with the pilot number 5072 after hearing the greeting prompt in the audio file named en_do_welcome.au. (The prefix en is specified in the `paramspace language` command and is automatically added to the filename provided in the `param drop-through-prompt` command.) Callers to (800) 555-0123 drop directly through to the hunt group with the pilot number 5071 without hearing any greeting.

```
dial-peer voice 1010 pots
service acdaa
port 1/1/0
incoming called-number 8005550121
```
dial-peer voice 1020 pots
  service aa-bcd
  port 1/1/1
  incoming called-number 8005550123

application
  service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param aa-hunt2 5072
  param number-of-hunt-grps 2
  param queue-len 10

  service acdaa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550121
  param service-name callq
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 20
  param number-of-hunt-grps 1
  param drop-through-prompt _bacd_dto_welcome.au
  param drop-through-option 2
  param second-greeting-time 45
  param handoff-string acdaa
  param max-time-call-retry 360

  service aa-bcd tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-bcd.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550123
  param service-name callq
  param second-greeting-time 60
  param max-time-call-retry 180
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 5
  param handoff-string aa-bcd
  param drop-through-option 1
  param number-of-hunt-grps 1

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>application</strong></td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td><strong>service</strong></td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
**param ea-password**

To create a password for accessing the extension assigner application, use the **param ea-password** command in application-parameter configuration mode.

```
param ea-password password
```

**Syntax Description**

| **password** | Numeric string to be used as password for the extension assigner application. Password string must be 2 to 10 characters long and can contain numbers 0 to 9. |

**Command Default**

No password is created.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th><strong>Cisco IOS Release</strong></th>
<th><strong>Cisco Product</strong></th>
<th><strong>Modification</strong></th>
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<tbody>
<tr>
<td>12.4(4)XC4</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command creates a password for using the extension assigner application. If this command is not configured, you cannot use the extension assigner application.

**Note**

There is no **no** form of this command. To change or remove the password for the extension assigner application, remove the service using the **no** form of the **service** command in application configuration mode.

**Examples**

The following example shows that a password (1234) is configured for the extension assigner application:

```
application
  service EA flash:ea/app-cme-ea-2.0.0.0.tcl
  paramspace english index 0
  paramspace english language en
  param ea-password 1234
  paramspace english location flash:ea/
  paramspace english prefix en
```

**Related Commands**

<table>
<thead>
<tr>
<th><strong>Description</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>application</strong></td>
</tr>
<tr>
<td>Description</td>
</tr>
<tr>
<td>-------------</td>
</tr>
<tr>
<td>service</td>
</tr>
</tbody>
</table>
param handoff-string

To specify the name of a Cisco Unified CME B-ACD auto-attendant (AA) to be passed to the call-queue script, use the **param handoff-string** command in application-parameter configuration mode. To disable the handoff string, use the no form of this command.

```
param handoff-string  aa-service-name
no param drop-through-prompt  aa-service-name
```

### Syntax Description

| **aa-service-name** | Service name that was assigned to the AA script with the `service` command. |

### Command Default

No string is designated to be passed to the call-queue service.

### Command Modes

Application-parameter configuration (config-app-param)

### Command History

<table>
<thead>
<tr>
<th><strong>Cisco IOS Release</strong></th>
<th><strong>Cisco Product</strong></th>
<th><strong>Modification</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the <strong>call application voice handoff-string</strong> command.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

The handoff string is used only when the call-queue script starts for the first time or restarts after a failure.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

### Examples

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number drop through to the ephone hunt group that has a pilot number of 5071 after hearing the initial prompt from the file `en_dt_prompt.au`. The AA name, aa, is passed to the call-queue service in the **param handoff-string** command.

```
dial-peer voice 1000 pots
  service aa
  port 1/1/0
  incoming called-number 8005550100
  ephone-hunt 10 sequential
  pilot 5071
  list 5011, 5012, 5013, 5014, 5015
!
application
  service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10
!
  service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
```
param handoff-string

```
paramspace english location tftp://192.168.254.254/user1/prompts/
paramspace english index 0
paramspace english language en
param aa-pilot 8005550100
param number-of-hunt-groups 1
param service-name callq
param handoff-string aa
param second-greeting-time 60
param drop-through-option 1
param drop-through-prompt _dt_prompt.au
param call-retry-timer 15
param max-time-call-retry 700
param voice-mail 5000
param max-time-vm-retry 2
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
param max-extension-length

To specify the maximum number of digits callers can dial when they choose the dial-by-extension option from the Cisco Unified CME B-ACD service, use the `param max-extension-length` command in application-parameter configuration mode. To return to the default, use the `no` form of this command.

```
param max-extension-length number
no param max-extension-length number
```

**Syntax Description**

| number | Number of digits. The lower limit is 0; there is no upper limit. The default is 5. |

**Command Default**

The default number of digits callers can dial using the dial-by-extension option is 5.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the <code>call application voice max-extension-length</code> command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the `service` command for an AA service.

Use this command to restrict the number of digits that callers can dial when using the dial-by-extension option.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

**Examples**

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits.

```
dial-peer voice 1000 pots
  service aa
  port 1/1/0
  incoming called-number 8005550100
  ephone-hunt 10 sequential
  pilot 5071
  list 5011, 5012, 5013, 5014, 5015
  !
application
  service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10
  !
  service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
```
Param Max-Extension-Length

```
paramspace english index 0
paramspace english language en
param aa-pilot 8005550100
param welcome-prompt _aa_welcome.au
param number-of-hunt-groups 1
param dial-by-extension-option 2
param max-extension-length 4
param service-name callq
param handoff-string aa
param second-greeting-time 60
param call-retry-timer 15
param max-time-call-retry 700
param voice-mail 5000
param max-time-vm-retry 2
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>application</strong></td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td><strong>service</strong></td>
<td>Enters application-parameter configuration mode and specifies a name for</td>
</tr>
<tr>
<td></td>
<td>the application and the location of the Tcl script to load for the</td>
</tr>
<tr>
<td></td>
<td>application.</td>
</tr>
</tbody>
</table>
**param max-time-call-retry**

To specify the maximum length of time for which calls to the Cisco Unified CME B-ACD service can stay in a call queue, use the `param max-time-call-retry` command in application-parameter configuration mode. To return to the default, use the `no` form of this command.

```
param max-time-call-retry  seconds
no param max-time-call-retry
```

**Syntax Description**

| seconds | Maximum length of time that the call-queue service can keep redialing a hunt group pilot number, in seconds. Range: 20 to 3600. Default: 600. |

**Command Default**

A call in a B-ACD call queue continues to try to connect to a hunt group for 600 seconds.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the call application voice max-time-call-retry command.</td>
</tr>
<tr>
<td>12.4(20)YA</td>
<td>Cisco Unified CME 7.0(1)</td>
<td>The minimum value of the <code>seconds</code> argument was increased from 0 to 20.</td>
</tr>
<tr>
<td>12.4(22)T</td>
<td>Cisco Unified CME 7.0(1)</td>
<td>This command was integrated into Cisco IOS Release 12.4(22)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) service. Configure this command under the `service` command for an AA service.

A call to a Cisco Unified CME B-ACD service is put into a call queue if the hunt group that the call tried to reach has no phones available to take the call because they are all busy. While the call is in the queue, a second greeting is played at intervals specified by the `param second-greeting-time` command. From the queue, the call makes retries to connect at intervals specified by the `param call-retry-timer` command until the maximum amount of time to be spent in the queue expires. The maximum amount of time is set by the `param max-time-call-retry` command. After the maximum amount of time expires, the call is routed to the alternate destination specified in the `param voice-mail` command. If the alternate destination number is busy, the call makes the number of retries to connect specified in the `param max-time-vm-retry` command. If the call is unable to connect to the alternate destination after the number of retries that has been specified, it is disconnected.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For configuration information, see the “Setting Up Call-Queue and AA Services” section in the Cisco Unified CME B-ACD and Tel Call-Handling Applications document.

**Examples**

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this
number can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits.

If a caller presses 2 and all the phones in ephone-hunt group 10 are busy, the call is put into a queue for hunt group 10. Every 60 seconds, the caller hears the second greeting, which is "Please continue to hold. An agent will be with you shortly." Every 15 seconds, the call-queue service tries again to connect the call to the hunt group. If no phones become available before 700 seconds expire, the call is routed to extension 5000. If that extension is busy, the call-queue service retries it 2 times more. If the call still cannot be connected, it is disconnected.

dial-peer voice 1000 pots
  service aa
  port 1/1/0
  incoming called-number 8005550100
ephone-hunt 10 sequential
  pilot 5071
  list 5011, 5012, 5013, 5014, 5015
application
  service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10
application
  service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550100
  param welcome-prompt _aa_welcome.au
  param number-of-hunt-groups 1
  param dial-by-extension-option 2
  param max-extension-length 4
  param service-name callq
  param handoff-string aa
  param second-greeting-time 60
  param call-retry-timer 15
  param max-time-call-retry 700
  param voice-mail 5000
  param max-time-vm-retry 2

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>call application voice load</td>
<td>Reloads the selected voice application script after it is modified.</td>
</tr>
<tr>
<td>param call-retry-timer</td>
<td>Specifies the time interval before each attempt to retry to connect a call to an ephone hunt group in a Cisco Unified CME B-ACD service.</td>
</tr>
<tr>
<td>param max-time-vm-retry</td>
<td>Specifies the maximum number of times that calls in a Cisco Unified CME B-ACD call queue can attempt to connect to the alternate destination number.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------------</td>
<td>--------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td><strong>param second-greeting-time</strong></td>
<td>Sets the length of the intervals between replays of the second greeting to calls waiting in hunt group call queues that are part of a Cisco Unified CME B-ACD service.</td>
</tr>
<tr>
<td><strong>param voice-mail</strong></td>
<td>Sets an alternate destination number to which to route calls that cannot be connected to a hunt group that is part of a Cisco Unified CME B-ACD service.</td>
</tr>
<tr>
<td><strong>service</strong></td>
<td>Enters application-parameter configuration mode to specify the name of the application and the location of the Tcl script to load.</td>
</tr>
</tbody>
</table>
param max-time-vm-retry

To specify the maximum number of times that calls in a Cisco Unified CME B-ACD call queue can attempt to connect to the alternate destination number, use the `param max-time-vm-retry` command in application-parameter configuration mode. To return to the default, use the `no` form of this command.

```
param max-time-vm-retry number
no param max-time-vm-retry number
```

**Syntax Description**
- `number` Number of times that the alternate destination number is redialed by the call-queue service. Range is from 1 to 3. Default is 1.

**Command Default**
A call in a B-ACD call queue tries to connect to an alternate destination number 1 time.

**Command Modes**
Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the <code>call application voice max-time-vm-retry</code> command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the `service` command for an AA service.

A call to a Cisco Unified CME B-ACD service is put into a call queue if the hunt group that the call tried to reach has no phones available to take the call because they are all busy. While the call is in the queue, a second greeting is played at intervals specified by the `param second-greeting-time` command. From the queue, the call makes retries to connect at intervals specified by the `param call-retry-timer` command until the maximum amount of time to be spent in the queue expires. The maximum amount of time is set by the `param max-time-call-retry` command. After the maximum amount of time expires, the call is routed to the alternate destination specified in the `param voice-mail` command. If the alternate destination number is busy, the call makes the number of retries to connect specified in the `param max-time-vm-retry command`. If the call is unable to connect to the alternate destination after the number of retries that has been specified, it is disconnected.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications document for your release.

**Examples**
The following example sets parameters for an AA application called `aa` and a call-queue application called `queue`. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits.

If a caller presses 2 and all the phones in ephone-hunt group 10 are busy, the call is put into a queue for hunt group 10. Every 60 seconds, the caller hears the second greeting, which is “Please continue to hold. An agent will be with you shortly.” Every 15 seconds, the call-queue service tries again to connect the call to the hunt group. If no phones become available before 700 seconds expire, the call
is routed to extension 5000. If that extension is busy, the call-queue service retries it 2 times more.
If the call still cannot be connected, it is now disconnected.

dial-peer voice 1000 pots
  service aa
  port 1/1/0
  incoming called-number 8005550100
  ephone-hunt 10 sequential
  pilot 5071
  list 5011, 5012, 5013, 5014, 5015
  
! application
  service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
    param queue-manager-debugs 1
    param aa-hunt1 5071
    param number-of-hunt-grps 1
    param queue-len 10
  !
  service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english language en
  param aa-pilot 8005550100
  param welcome-prompt _aa_welcome.au
  param number-of-hunt-groups 1
  param dial-by-extension-option 2
  param max-extension-length 4
  param service-name callq
  param handoff-string aa
  param second-greeting-time 60
  param call-retry-timer 15
  param max-time-call-retry 700
  param voice-mail 5000
  param max-time-vm-retry 2

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
**param menu-timeout**

To set the number of times the AA service will loop the menu prompt before connecting the caller to an operator if the caller does not select a menu option, use the `param menu-timeout` command in application-parameter configuration mode. To return to the default, use the `no` form of this command.

```
param menu-timeout number
no param menu-timeout
```

**Syntax Description**

<table>
<thead>
<tr>
<th>number</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>number</td>
<td>Times to replay menu prompt before connecting a caller to an operator. Range: 0 to 10. Default: 4.</td>
</tr>
</tbody>
</table>

**Command Default**

Auto-attendant service replays menu prompt 4 times before connecting the caller to an operator.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(22)YA</td>
<td>Cisco Unified CME 7.0(1)</td>
<td>The minimum value of the <code>number</code> argument was decreased from 1 to 0.</td>
</tr>
<tr>
<td>12.4(22)T</td>
<td>Cisco Unified CME 7.0(1)</td>
<td>This command was integrated into Cisco IOS Release 12.4(22)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) service.

If a caller does not select a menu option before the timeout set with this command expires, the call is transferred to the operator hunt group. The operator hunt-group is the hunt group with the highest aa-hunt number set with the `param aa-hunt` command.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For configuration information, see the “Setting Up Call-Queue and AA Services” section in the *Cisco Unified CME B-ACD and Tcl Call-Handling Applications* document.

**Examples**

The following example shows the menu timeout set to 5 replays for the AA application called order1-aa:

```
application
  service acme-aa1 tftp://192.168.254.254/acme/bacd/app-b-acd-aa-2.1.2.3.tcl
  paramspace english index 1
  param menu-timeout 5
  param handoff-string acme-aa1
  param dial-by-extension-option 2
  paramspace english language en
  param max-time-vmretry 2
  param max-extension-length 4
  param aa-pilot 8005550100
  paramspace english location flash:/bacd/
  param second-greeting-time 60
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>call application</td>
<td>Reloads the selected voice application script after it is modified.</td>
</tr>
<tr>
<td>voice load</td>
<td></td>
</tr>
<tr>
<td>param aa-hunt</td>
<td>Declares a Cisco Unified CME B-ACD menu number and associates it with the</td>
</tr>
<tr>
<td></td>
<td>pilot number of an ephone hunt group.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode to specify the name of the</td>
</tr>
<tr>
<td></td>
<td>application and the location of the Tcl script to load.</td>
</tr>
</tbody>
</table>
param number-of-hunt-grps

To specify the number of hunt groups used with a Cisco Unified CME B-ACD call-queue or AA service, use the `param number-of-hunt-grps` command in application-parameter configuration mode. To return to the default, use the `no` form of this command.

```
param number-of-hunt-grps number
no param number-of-hunt-grps number
```

**Syntax Description**

| `number` | Number of ephone hunt groups used by the service. Range is 1 to 10 for the call-queue service and 1 to 3 for an automated attendant (AA) service. |

**Command Default**

This parameter is not set.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
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<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the <code>call application voice number-of-hunt-grps</code> command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured both under the `service` command for the call-queue service and under the `service` command for an AA service.

The number of hunt groups specified for the call-queue service is the total of the number of hunt groups used with all the AAs with which it is associated. For example, if a B-ACD has three AAs, each with two hunt groups, the number of hunt groups for each AA is two and the number of hunt groups for the call-queue service is six.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

**Examples**

A call-queue service called CQ is set up to work with two AA services. CQ lists 4 as the number of hunt groups it uses. AA1 is associated with 3 hunt groups, and its callers hear the following prompt: “Press 1 for sales, press 2 for service, press 0 for operator.” AA2 uses drop-through mode. Its callers do not hear a prompt and are directly connected to the single hunt group that is associated with it.

```
application
  service CQ tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
    param queue-manager-debugs 1
    param aa-hunt1 1001
    param aa-hunt2 2001
    param aa-hunt3 3001
    param aa-hunt4 4001
    param number-of-hunt-grps 4
    param queue-len 10
  service AA1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
    paramspace english location tftp://192.168.254.254/user1/prompts/
```
paramspace english index 0
paramspace english language en
param aa-pilot 8005550111
param number-of-hunt-groups 3
param service-name CQ
param welcome-prompt _bacd_welcome.au
param handoff-string AA1
service AA2 tftp://192.168.254.254/user1/CallQ/ACD/app-b-acd-aa.tcl
paramspace english location tftp://192.168.254.254/user1/prompts/
paramspace english index 0
paramspace english language en
param aa-pilot 8005550122
param number-of-hunt-groups 1
param service-name CQ
param drop-through-option 4
param handoff-string AA2

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
param queue-exit-extension

To assign an extension number to a call-queue exit option, use the `param queue-exit-extension` command in application-parameter configuration mode. To remove an exit option, use the `no` form of this command.

```
param queue-exit-extension option-number extension-number
no param queue-exit-extension option-number
```

**Syntax Description**

<table>
<thead>
<tr>
<th>option-number</th>
<th>Number of the call-queue exit option. Range: 1 to 3. There is no default.</th>
</tr>
</thead>
<tbody>
<tr>
<td>extension-number</td>
<td>Extension number associated with the exit option.</td>
</tr>
</tbody>
</table>

**Command Default**

Call-queue exit option is not defined.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YA</td>
<td>Cisco Unified CME 7.0(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(22)T</td>
<td>Cisco Unified CME 7.0(1)</td>
<td>This command was integrated into Cisco IOS Release 12.4(22)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) service.

Use this command together with the `param queue-exit-option` command to enable callers to select from up to three different options to exit from a call queue. The `option-number` argument in this command corresponds to the `option-number` argument in the `param queue-exit-option` command.

You can record a customized second greeting that offers callers up to three options to exit from the call queue. For example, you might record a message that says, “To leave a message, press 6; to hear more options, press 7; to speak to an operator, press 8.”

This second greeting is stored in the audio file named en_bacd_allagentsbusy.au. You can record over the default message in this file, provided you do not change the name of the file.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For configuration information, see the “Setting Up Call-Queue and AA Services” section in the Cisco Unified CME B-ACD and Tcl Call-Handling Applications document.

**Examples**

The following example shows that the acme-aa1 application has three exit options defined for its call-queue service:

```
application
  service acme-aa1 tftp://192.168.254.254/acme/bacd/app-b-acd-aa-2.1.2.3.tcl
  param dial-by-extension-option 7
  param handoff-string acme-aa1
  paramspace english index 1
  param queue-exit-option2 7
  param max-time-vm-retry 2
```
param space english language en
param aa-pilot 801
param max-extension-length 4
param queue-overflow-extension 101
param queue-exit-extension2 101
param second-greeting-time 20
param queue-exit-option1 6
paramspace english location flash:/bacd/
param send-account true
param call-retry-timer 20
param queue-exit-option3 8
param voice-mail 444
param max-time-call-retry 60
param service-name sf-queue
param queue-exit-extension1 202
param number-of-hunt-grps 1
param drop-through-option 1
param queue-exit-extension3 444

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td></td>
<td>call application voice load</td>
<td>Reloads the selected voice application script after it is modified.</td>
</tr>
<tr>
<td></td>
<td>param queue-exit-option</td>
<td>Assigns a menu number to a call-queue exit option.</td>
</tr>
<tr>
<td></td>
<td>service</td>
<td>Enters application-parameter configuration mode to specify the name of the application and the location of the Tcl script to load.</td>
</tr>
</tbody>
</table>
**param queue-exit-option**

To assign a menu number to a call-queue exit option, use the `param queue-exit-option` command in application-parameter configuration mode. To disable this option, use the `no` form of this command.

```
param queue-exit-option option-number menu-number
no param queue-exit-option option-number
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Option</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>option-number</code></td>
<td>Number of the call-queue exit option. Range: 1 to 3. There is no default.</td>
</tr>
<tr>
<td><code>menu-number</code></td>
<td>Menu option number associated with the exit option.</td>
</tr>
</tbody>
</table>

**Command Default**

Call-queue exit option is not assigned.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YA</td>
<td>Cisco Unified CME 7.0(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(22)T</td>
<td>Cisco Unified CME 7.0(1)</td>
<td>This command was integrated into Cisco IOS Release 12.4(22)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) service.

Use this command together with the `param queue-exit-extension` command to enable callers to select from up to three different options to exit from a call queue. The `option-number` argument in this command corresponds to the `option-number` argument in the `param queue-exit-extension` command.

You can record a customized second greeting that offers callers up to three options to exit from the call queue. For example, you might record a message that says, “To leave a message, press 6; to hear more options, press 7; to speak to an operator, press 8.”

This second greeting is stored in the audio file named `en_bacd_allagentsbusy.au`. You can record over the default message in this file, provided you do not change the name of the file.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For configuration information, see the “Setting Up Call-Queue and AA Services” section in the *Cisco Unified CME B-ACD and Tcl Call-Handling Applications* document.

**Examples**

The following example shows that the acme-aa1 application has three exit options defined for its call-queue service:

```
application
  service acme-aa1 tftp://192.168.254.254/acme/bacd/app-b-acd-aa-2.1.2.3.tcl
  param dial-by-extension-option 7
  param handoff-string acme-aa1
  paramspace english index 1
  param queue-exit-option2 7
  param max-time-vm-retry 2
```
paramspace english language en
param aa-pilot 801
param max-extension-length 4
param queue-overflow-extension 101
param queue-exit-extension2 101
param second-greeting-time 20
param queue-exit-option1 6
paramspace english location flash:/bacd/
param send-account true
param call-retry-timer 20
param queue-exit-option3 8
param voice-mail 444
param max-time-call-retry 60
param service-name sf-queue
param queue-exit-extension1 202
param number-of-hunt-grps 1
param drop-through-option 1
param queue-exit-extension3 444

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>call application voice load</td>
<td>Reloads the selected voice application script after it is modified.</td>
</tr>
<tr>
<td>param queue-exit-extension</td>
<td>Assigns an extension number to a call-queue exit option.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode to specify the name of the</td>
</tr>
<tr>
<td></td>
<td>application and the location of the Tcl script to load.</td>
</tr>
</tbody>
</table>
param queue-len

To specify the number of calls that can be held in each call queue in a Cisco Unified CME B-ACD service, use the param queue-len command in application-parameter configuration mode. To return to the default, use the no form of this command.

**Syntax**

```
param queue-len number
no param queue-len number
```

**Syntax Description**

| number | Number of calls that can be held in a call queue. Range is 1 to 30. Default is 10. |

**Command Default**

The default queue length is 10.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the call application voice queue-len command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the service command for a call-queue service.

This command specifies the maximum number of calls that can be held in a call queue for a hunt group used with B-ACD when all of the hunt group member phones are busy.

Note that having calls in queue keeps PSTN ports occupied for a longer time, and you may want to plan for more ports if you have longer queues. The maximum number of calls allowed in the queues of ephone hunt groups must be based on the number of ports or trunks available. For example, if you have 20 foreign exchange office (FXO) ports and two ephone hunt groups, you can configure a maximum of ten calls per ephone hunt-group queue using the param queue-len 10 command. You can use the same configuration if you have a single T1 trunk, which supports 23 channels.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications document for your release.

**Examples**

A call-queue service called CQ is set up to work with two AA services. CQ lists four as the number of hunt groups it uses. AA1 is associated with three hunt groups, and its callers hear the following prompt: “Press 1 for sales, press 2 for service, press 0 for operator.” AA2 uses drop-through mode. Its callers do not hear a prompt and are directly connected to the single hunt group that is associated with it. Up to 12 calls can be held in the call queue for each hunt group if all the phones in the hunt group are busy.

```
application
  service CQ tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 1001
  param aa-hunt2 2001
  param aa-hunt3 3001
```
Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
param queue-manager-debugs

To enable the collection of call-queue debug information in a Cisco Unified CME B-ACD service, use the param queue-manager-debugs command in application-parameter configuration mode. To remove the setting, use the no form of this command with the keyword that was previously used.

```bash
param queue-manager-debugs [0|1]
no param queue-manager-debugs [0|1]
```

**Syntax Description**

- **0**: Disables collection of call-queue debug information.
- **1**: Enables collection of call-queue debug information.

**Command Default**

Collection of debug information is disabled.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the call application voice queue-manager-debugs command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the service command for the call-queue service.

This command enables the collection of data regarding call queue activity. It is used in conjunction with the debug voip ivr script command. Both commands must be enabled at the same time.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

**Examples**

A call-queue service called CQ is set up to work with two AA services. CQ lists four as the number of hunt groups it uses. AA1 is associated with three hunt groups, and its callers hear the following prompt: “Press 1 for sales, press 2 for service, press 0 for operator.” AA2 uses drop-through mode. Its callers do not hear a prompt and are directly connected to the single hunt group that is associated with it. Up to ten calls can be held in the call queue for each hunt group if all the phones in the hunt group are busy with other calls. Call-queue debugging is enabled.

```bash
application
service CQ tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
param queue-manager-debugs 1
param aa-hunt1 1001
param aa-hunt2 2001
param aa-hunt3 3001
param aa-hunt4 4001
param number-of-hunt-grps 4
param queue-len 10
service AA1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
param space english location tftp://192.168.254.254/user1/prompts/
```
paramspace english index 0
paramspace english language en
param aa-pilot 8005550111
param number-of-hunt-groups 3
param service-name CQ
param welcome-prompt _bacd_welcome.au
param handoff-string AA1
service AA2 tftp://192.168.254.254/user1/CallQ/ACD/app-b-acd-aa.tcl
paramspace english location tftp://192.168.254.254/user1/prompts/
paramspace english index 0
paramspace english language en
param aa-pilot 8005550122
param number-of-hunt-groups 1
param service-name CQ
param drop-through-option 4
param handoff-string AA2

## Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>application</strong></td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td><strong>service</strong></td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
**param queue-overflow-extension**

To set the extension number to route calls to when the call queue for the auto-attendant service is full, use the `param queue-overflow-extension` command in application-parameter configuration mode. To return to the default, use the `no` form of this command.

```
param queue-overflow-extension  extension-number
no param queue-overflow-extension
```

<table>
<thead>
<tr>
<th><strong>Syntax Description</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><code>extension-number</code></td>
<td>Extension number to which the auto-attendant service forwards calls when the call queue is full.</td>
</tr>
</tbody>
</table>

**Command Default**

No overflow extension is defined. Calls disconnect if the queue becomes full.

**Command Modes**

Application-parameter configuration `(config-app-param)`

<table>
<thead>
<tr>
<th><strong>Command History</strong></th>
<th><strong>Cisco IOS Release</strong></th>
<th><strong>Cisco Product</strong></th>
<th><strong>Modification</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>12.4(22)YA</td>
<td>Cisco Unified CME 7.0(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td></td>
<td>12.4(22)T</td>
<td>Cisco Unified CME 7.0(1)</td>
<td>This command was integrated into Cisco IOS Release 12.4(22)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) service.

This command specifies the extension number where calls are sent when the number of calls waiting in a B-ACD call queue exceeds the number set with the `param queue-len` command.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For configuration information, see the “Setting Up Call-Queue and AA Services” section in the *Cisco Unified CME B-ACD and Tcl Call-Handling Applications* document.

**Examples**

The following example shows that the AA application named acme-aa1 uses the call-queue service named CQ. When the number of calls in the queue exceeds 12, new calls that cannot be answered by an agent are sent to extension 5100.

```
application
  service CQ tftp://192.168.254.254/acme/bacd/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 1001
  param aa-hunt2 2001
  param aa-hunt3 3001
  param aa-hunt4 4001
  param number-of-hunt-grps 4
  param queue-len 12
!
application
  service acme-aa1 tftp://192.168.254.254/acme/bacd/app-b-acd-aa-2.1.2.3.tcl
  paramspace english index 1
```
param handoff-string acme-aa1
param dial-by-extension-option 2
paramspace english language en
param aa-pilot 8005550100
param queue-overflow-extension 5100
paramspace english location flash:/bacd/
param welcome-prompt _aa_welcome.au
param number-of-hunt-groups 1
param voice-mail 5000
param service-name CQ

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td></td>
<td>call application voice load</td>
<td>Reloads the selected voice application script after it is modified.</td>
</tr>
<tr>
<td></td>
<td>param queue-len</td>
<td>Specifies the number of calls that can be held in each call queue in a Cisco Unified CME B-ACD service.</td>
</tr>
<tr>
<td></td>
<td>service</td>
<td>Enters application-parameter configuration mode to specify the name of the application and the location of the Tcl script to load.</td>
</tr>
</tbody>
</table>
param secondary-prefix

To set a prefix to add to digits coming from the PSTN Central Office (CO) to route calls from a secondary Cisco Unified CME router to a primary Cisco Unified CME router when using the Direct Inward Dial (DID) Digit Translation Service, use the `param secondary-prefix` command in application-parameter configuration mode. To disable this option, use the `no` form of this command.

`param secondary-prefix  secondary-prefix
no param secondary-prefix  secondary-prefix`

**Syntax Description**

| secondary-prefix | Prefix to add to digits in order to route calls to the primary Cisco Unified CME router. Range is from 0 to 99. |

**Command Default**

No prefix is defined.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced to replace the call application voice secondary-prefix command.</td>
<td>This command replaced the call application voice secondary-prefix command and was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Cisco Unified CallManager Express (Cisco Unified CME) DID Digit Translation Service, which provides number translation for DID calls when the range of DID digits provided by the PSTN Central Office (CO) does not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers: a prompt is played and the calls are disconnected.

When calls are received by a secondary Cisco Unified CME router, they are routed to the primary router by configuring an H.323 VoIP dial peer and matching the destination pattern for that dial peer. The DID prefix that was configured for use with the DID script is appended to the incoming DID digits first. The secondary prefix is appended next. For example, if the incoming DID digits are 25, the DID prefix is 3, and the secondary prefix is 7, the transformed number will be 7325. The transformed number matches a VoIP dial peer, which uses the `forward-digits` command to send only the three relevant digits, the extension number, to the primary router.

See the `Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications` document for your release.
Examples

The following example configures a Basic DID application on the Cisco Unified CME router. It sets a prefix of 5 to apply to the digits coming from the CO in order to construct a valid extension number. Then the secondary prefix (4) is appended. If the incoming DID digits are 25, the DID prefix is 5, and the secondary prefix is 4, then the transformed number is 4525. The transformed number matches VoIP dial peer 1000. The VoIP dial peer sends calls to the primary Cisco Unified CME router using the IP address that is entered in the session target command. The dial peer uses the `forward-digits` command to send the extension number, 525, to the primary Cisco Unified CME router.

dial-peer voice 1000 voip
destination-pattern 45..
session target ipv4:10.1.1.1
dtmf-relay h245-alphanumeric
codec g711ulaw
forward-digits 3
application
service didapp tftp://192.168.254.254/scripts/did/app-THD-DID-2.0.0.1.tcl
paramspace english index 1
paramspace english language en
paramspace english location tftp://192.168.254.254/apps/dir25/
param secondary-prefix 4
param did-prefix 5
param co-did-min 00
param co-did-max 79
param store-did-min 00
param store-did-max 79

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>application</code></td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td><code>service</code></td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
param second-greeting-time

To set the length of the intervals between playouts of the second greeting to calls waiting in hunt group call queues that are part of a Cisco Unified CME B-ACD service, use the `param second-greeting-time` command in application-parameter configuration mode. To return to the default, use the `no` form of this command.

```
param second-greeting-time seconds
no param max-time-vm-retry seconds
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>seconds</th>
<th>Length of time intervals between playouts of the second greeting to calls in a B-ACD call queue, in seconds. Range is from 30 to 120. Default is 60.</th>
</tr>
</thead>
</table>

**Command Default**
The second greeting is played out every 60 seconds to calls in B-ACD call queues.

**Command Modes**
Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the <code>call application voice second-greeting-time</code> command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the `service` command for an AA service.

A call to a Cisco Unified CME B-ACD service is put into a call queue if the hunt group that the call tried to reach has no phones available to take the call because they are all busy. While the call is in the queue, a second greeting is played at intervals specified by the `param second-greeting-time` command. From the queue, the call retries to connect to the hunt group at intervals specified by the `param call retry timer` command until the maximum amount of time to be spent in the queue expires. The maximum amount of time is set by the `param max-time-call retry` command. After the maximum amount of time expires, the call is routed to the alternate destination specified in the `param voice mail` command. If the alternate destination number is busy, the call makes the number of retries to connect specified in the `param max-time-vm retry` command. If the call is unable to connect to the alternate destination after the number of retries that has been specified, it is disconnected.

The second greeting is stored in the audio file named `en_bacd_allagentsbusy.au`. You can re-record over the default message that is provided in the file, but you cannot change the name of the file.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

**Examples**

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits.

If a caller presses 2 and all the phones in ephone-hunt group 10 are busy, the call is put into a queue for hunt group 10. Every 60 seconds, the caller hears the second greeting, which is “Please continue...”
to hold. An agent will be with you shortly.” Every 15 seconds, the call-queue service tries again to connect the call to the hunt group. If no phones become available before 700 seconds expire, the call is routed to extension 5000. If that extension is busy, the call-queue service retries it 2 times more. If the call still cannot be connected, it is now disconnected.

```plaintext
dial-peer voice 1000 pots
  service aa
  port 1/1/0
  incoming called-number 8005550100
  ephone-hunt 10 sequential
  pilot 5071
  list 5011, 5012, 5013, 5014, 5015
!
application
  service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
    param queue-manager-debugs 1
    param aa-hunt1 5071
    param number-of-hunt-grps 1
    param queue-len 10
!
  service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
    paramspace english location tftp://192.168.254.254/user1/prompts/
    paramspace english language en
    param aa-pilot 8005550100
    param welcome-prompt _aa_welcome.au
    param number-of-hunt-groups 1
    param dial-by-extension-option 2
    param max-extension-length 4
    param service-name callq
    param handoff-string aa
    param second-greeting-time 60
    param call-retry-timer 15
    param max-time-call-retry 700
    param voice-mail 5000
    param max-time-vm-retry 2
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
param send-account true

To generate call detail records (CDRs) for calls that are handled by the Cisco Unified CME B-ACD service, use the `param send-account` command in application-parameter configuration mode. To return to the default, use the `no` form of this command.

```
param send-account true
no param send-account true
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

CDRs are not generated.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command captures CDRs in RADIUS format for calls handled by the Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) service. The call record includes the name of the AA service, hunt group pilot-number, and globally unique identifier (GUID).

For configuration information, see the “Setting Up Call-Queue and AA Services” section in the *Cisco Unified CME B-ACD and Tcl Call-Handling Applications* document.

For information on enabling RADIUS accounting, see the *CDR Accounting for Cisco IOS Voice Gateways* guide.

**Examples**

The following example shows that calls using the acme-aa1 service generate a call detail record:

```
application
  service acme-aa1 tftp://192.168.254.254/acme/bacd/
  app-b-acd-aa-2.1.2.3.tcl
  paramspace english index 1
  param handoff-string acme-aa1
  param dial-by-extension-option 2
  paramspace english language en
  param aa-pilot 8005550100
  paramspace english location flash:/bacd/
  param welcome-prompt _aa_welcome.au
  param send-account true
  param number-of-hunt-groups 1
  param voice-mail 5000
  param service-name callq
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>call application voice load</td>
<td>Reloads the selected voice application script after it is modified.</td>
</tr>
<tr>
<td>gw-accounting aaa</td>
<td>Enables the gateway to send accounting CDRs to the RADIUS server using VSAs (attribute 26).</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode to specify the name of the application and the location of the Tcl script to load.</td>
</tr>
</tbody>
</table>
param service-name

To specify a Cisco Unified CME B-ACD call-queue service to use with an automated attendant (AA) service, use the `param service-name` command in application-parameter configuration mode. To return to the default, use the `no` form of this command.

```
param service-name queue-service-name
no param service-name queue-service-name
```

**Syntax Description**

| queue-service-name | Name that was assigned to the B-ACD call-queue service with the `service` command. |

**Command Default**

No call-queue service is specified.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the <code>call application voice service-name command</code>.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the `service` command for an AA service.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

**Examples**

A call-queue service called CQ is set up to work with two AA services. CQ lists four as the number of hunt groups it uses. AA1 is associated with three hunt groups, and its callers hear the following prompt: “Press 1 for sales, press 2 for service, press 0 for operator.” AA2 uses drop-through mode. Its callers do not hear a prompt and are directly connected to the single hunt group that is associated with it. Up to ten calls can be held in the call queue for each hunt group if all the phones in the hunt group are busy with other calls. Call-queue debugging is enabled.

```
application
  service CQ tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 1001
  param aa-hunt2 2001
  param aa-hunt3 3001
  param aa-hunt4 4001
  param number-of-hunt-grps 4
  param queue-len 10
  service AA1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550111
  param number-of-hunt-groups 3
  param service-name CQ
  param welcome-prompt _bacd_welcome.au
```
param handoff-string AA1
service AA2 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
paramspace english location tftp://192.168.254.254/user1/prompts/
paramspace english index 0
param aa-pilot 8005550122
param number-of-hunt-groups 1
param service-name CQ
param drop-through-option 4
param handoff-string AA2

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>application</strong></td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td><strong>service</strong></td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
**param store-did-max**

To set the upper boundary of the range of digits that is valid in the Cisco Unified CME numbering plan used with the Direct Inward Dial (DID) Digit Translation Service, use the `param store-did-max` command in global configuration mode. To disable this option, use the `no` form of this command.

```
param store-did-max max-store-value
no param store-did-max max-store-value
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>max-store-value</code></td>
<td>Maximum value of digits in the Cisco Unified CME dial plan. The digit string can be any length, but the string length must be the same in the <code>param co-did-max</code>, <code>param co-did-min</code>, <code>param store-did-max</code>, and <code>param store-did-min</code> commands.</td>
</tr>
</tbody>
</table>

**Command Default**

No maximum value is defined for the range of digits in the dial plan.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced to replace the <code>call application voice store-did-max</code> command.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command replaced the <code>call application voice store-did-max</code> command and was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command defines the upper limit of the range of digits in the site dial plan for the Cisco Unified CallManager Express (Cisco Unified CME) Direct Inward Dial Digit Translation Service, which provides number translation for DID calls when the DID digits provided by the PSTN Central Office (CO) do not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers. A prompt is played and the calls are disconnected.

**Examples**

The following example configures Direct Inward Dial Digit Translation Service on the Cisco Unified CME router. It sets a lower boundary of 00 and an upper boundary of 79 for the range of digits in the Cisco Unified CME extension dial plan. Notice that the length of the digit string is the same (2 digits) for all related commands.

```
application
service didapp tftp://192.168.254.254/scripts/did/app-THD-DID-2.0.0.1.tcl
paramspace english index 1
paramspace english language en
paramspace english location tftp://192.168.254.254/apps/dir25/
param secondary-prefix 4
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
<tr>
<td>param co-did-max</td>
<td>Sets the upper boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the Direct Inward Dial Digit Translation Service.</td>
</tr>
<tr>
<td>param co-did-min</td>
<td>Sets the lower boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the Direct Inward Dial Digit Translation Service.</td>
</tr>
<tr>
<td>param store-did-min</td>
<td>Sets the lower boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the Direct Inward Dial Digit Translation Service.</td>
</tr>
</tbody>
</table>

```plaintext
param did-prefix 5
param co-did-min 00
param co-did-max 79
param store-did-min 00
param store-did-max 79
```
param store-did-min

To set the lower boundary of the range of digits that is valid in the Cisco Unified CME numbering plan used with the Direct Inward Dial (DID) Digit Translation Service, use the `param store-did-min` command in application-parameter configuration mode. To disable this option, use the `no` form of this command.

`param store-did-min min-store-value`

`no param store-did-min min-store-value`

**Syntax Description**

`min-store-value` Minimum value of digits in the Cisco Unified CME dial plan. The digit string can be any length, but the string length must be the same in the `param co-did-max`, `param co-did-min`, `param store-did-max`, and `param store-did-min` commands.

**Command Default**

No minimum value is defined for the range of digits in the dial plan.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced to replace the call application voice store-did-min command.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command replaced the call application voice store-did-min command and was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command defines the lower limit of the range of digits in the site dial plan when it is used with the Cisco Unified CallManager Express (Cisco Unified CME) DID Digit Translation Service. This service provides number translation for DID calls when the range of DID digits provided by the PSTN Central Office (CO) does not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers: a prompt is played and the calls are disconnected.

**Examples**

The following example configures DID Digit Translation Service on the Cisco Unified CME router. It sets a lower boundary of 00 and an upper boundary of 79 for the range of digits in the Cisco Unified CME extension dial plan.

```
application
  service didapp tftp://192.168.254.254/scripts/did/app=THD-DID=2.0.0.1.tcl
  paramspace english index 1
  paramspace english language en
  paramspace english location tftp://192.168.254.254/apps/dir25/
  param secondary-prefix 4
  param did-prefix 5
  param co-did-min 00
```
Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
<tr>
<td>param co-did-max</td>
<td>Sets the upper boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the DID Digit Translation Service.</td>
</tr>
<tr>
<td>param co-did-min</td>
<td>Sets the lower boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the DID Digit Translation Service.</td>
</tr>
<tr>
<td>param store-did-max</td>
<td>Sets the upper boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the DID Digit Translation Service.</td>
</tr>
</tbody>
</table>
**param voice-mail**

To set an alternate destination number to which to route calls that cannot be connected to a hunt group that is part of a Cisco Unified CME B-ACD service, use the `param voice-mail` command in application-parameter configuration mode. To return to the default, use the `no` form of this command.

```
param voice-mail number
no param voice-mail number
```

**Syntax Description**

| number | Extension number to which to route calls. The number must be associated with a dial peer that is reachable by the Cisco Unified CME system. |

**Command Default**

No alternate destination number is set.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was introduced to replace the <code>call application voice voice-mail</code> command.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the `service` command for an AA service.

Calls are diverted to an alternate destination only when one of the following criteria is met:

- The hunt group to which the call has been transferred is unavailable because all members are logged out.
- The call-queue maximum retry timer has expired.

The alternate destination can be any number at which you can assure call coverage, such as a voice-mail number, a permanently staffed number, or a number that rings an overhead night bell. Once a call is diverted to an alternate destination, it is no longer controlled by the B-ACD service. This parameter is set with the `param voice-mail` command.

If you send calls to a voice-mail system as an alternate destination, be sure to set up the voice-mail system as specified in the documentation for the system.

If you specify a number for an alternate destination, the number must be associated with a dial peer that is reachable by the Cisco Unified CME system.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information about B-ACD, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

**Examples**

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits.
If a caller presses 2 and all the phones in ephone-hunt group 10 are busy, the call is put into a queue for hunt group 10. Every 60 seconds, the caller hears the second greeting, which is “Please continue to hold. An agent will be with you shortly.” Every 15 seconds, the call-queue service tries again to connect the call to the hunt group. If no phones become available before 700 seconds expire, the call is routed to extension 5000, which is the alternate destination. If that extension is busy, the call-queue service retries it 2 times more. If the call still cannot be connected, it is disconnected.

dial-peer voice 1000 pots
  service aa
  port 1/1/0
  incoming called-number 8005550100
  ephone-hunt 10 sequential
  pilot 5071
  list 5011, 5012, 5013, 5014, 5015

application
  service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10

service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550100
  param welcome-prompt _aa_welcome.au
  param number-of-hunt-groups 1
  param dial-by-extension-option 2
  param max-extension-length 4
  param service-name callq
  param handoff-string aa
  param second-greeting-time 60
  param call-retry-timer 15
  param max-time-call-retry 700
  param voice-mail 5000
  param max-time-vm-retry 2

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
param welcome-prompt

To specify an audio file containing a prompt to be played as a welcome for callers to an automated attendant (AA) that is part of a Cisco Unified CME B-ACD service, use the param welcome-prompt command in application-parameter configuration mode. To return to the default, use the no form of this command.

**Syntax Description**

```
param welcome-prompt audio-filename
no param welcome-prompt audio-filename
```

| Identifier part of name of the audio file that contains the welcome greeting to be played when callers first reach the Cisco Unified CME B-ACD service. This name does not include the language prefix and it must begin with an underscore. Default is _bacd_welcome.au.

**Command Default**

The audio file named en_bacd_welcome.au is used as a welcome prompt.

**Command Modes**

Application-parameter configuration (config-app-param)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
</table>
|12.3(14)T|Cisco CME 3.3|This command was introduced to replace the call application voice voice-mail command.

**Usage Guidelines**

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the service command for an AA service.

Each AA service that is used with the Cisco Unified CME B-ACD service needs a welcome greeting to tell callers the destination they have reached and, sometimes, the options that they have. The en_bacd_welcome.au audio file is used by default. It announces “Thank you for calling,” and includes a two-second pause after the message. The filename of the welcome prompt audio file has two parts: a two-letter prefix that denotes a language code specified in the paramspace language command, and the identifying part that indicates the purpose of the file. In the default welcome prompt audio file, the prefix is en and the identifying part is _bacd_welcome.au. Note that the identifying part starts with an underscore.

If your Cisco Unified CME B-ACD service uses a single AA application, you can record a custom welcome greeting in the audio file named en_welcome_prompt.au and record instructions about menu choices in the audio file named en_bacd_options_menu.au.

If your Cisco Unified CME B-ACD service uses multiple AA applications, you will need separate greetings and menu options for each AA. Use the following guidelines:

- Record a separate welcome prompt for each AA application, using a different name for the audio file for each welcome prompt. For example, en_welcome_aa1.au and en_welcome_aa2.au. The welcome prompts that you record in these files should include both the greeting and the instructions about menu options.
- Record silence in the audio file en_bacd_options_menu.au. A minimum of one second of silence must be recorded. Note that you cannot change the identifier part of the name of this audio file.

For any Cisco Unified CME B-ACD configuration changes to take effect, you must reload the scripts.

For more information, see the Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications document for your release.
The following example sets parameters for two AA applications, called aa1 and aa2, and a call-queue application called queue. The direct-dial numbers to reach the AA services are (800) 555-0100 for aa1 and (800) 555-0110 for aa2. Callers to aa1 can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits. Callers to aa2 can press 2 to dial an extension number of 4 or fewer digits or press 3 to be connected to the ephone hunt group with the pilot number 5073. Both AAs share an operator hunt group, which is menu option 4.

The welcome prompt for aa1 is “Thank you for calling the Sales department. Press 1 to place an order. Press 2 if you know the extension of the party you want, or press 0 to speak to an operator.” The filename of the audio file that contains this welcome prompt is en_aa1_welcome.au.

The welcome prompt for aa2 is “Thank you for calling the Service department. Press 2 if you know the extension of the party you want. Press 3 to speak to a service technician or press 0 to speak to an operator.” The filename of the audio file that contains this welcome prompt is en_aa2_welcome.au.

dial-peer voice 1000 pots
  service aa1
  port 1/1/0
  incoming called-number 8005550100
dial-peer voice 1100 pots
  service aa2
  port 1/1/1
  incoming called-number 8005550110
ephone-hunt 10 sequential
  pilot 5071
  list 5011, 5012, 5013, 5014, 5015
ephone-hunt 11 sequential
  pilot 5073
  list 5021, 5022, 5023, 5024, 5025
! application
  service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
    param queue-manager-debugs 1
    param aa-hunt1 5071
    param aa-hunt3 5073
    param aa-hunt4 6000
    param number-of-hunt-grps 3
    param queue-len 10
! service aa1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550100
  param welcome-prompt _aa1_welcome.au
  param number-of-hunt-groups 2
  param dial-by-extension-option 2
  param max-extension-length 4
  param service-name callq
  param handoff-string aal
  param second-greeting-time 60
  param call-retry-timer 15
  param max-time-call-retry 700
  param voice-mail 5000
  param max-time-vm-retry 2
service aa2 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
param aa-pilot 8005550110
param welcome-prompt _aa2_welcome.au
param number-of-hunt-groups 2
param dial-by-extension-option 2
param max-extension-length 4
param service-name callq
param handoff-string aa2
param second-greeting-time 60
param call-retry-timer 15
param max-time-call-retry 700
param voice-mail 5000
param max-time-vm-retry 2

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application</td>
<td>Enters application configuration mode.</td>
</tr>
<tr>
<td>service</td>
<td>Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.</td>
</tr>
</tbody>
</table>
**paramspace callsetup after-hours-exempt**

To specify that an individual dial peer does not have any of its calls blocked by the Cisco router even though call blocking has been enabled, use the **paramspace callsetup after-hours-exempt** command in dial-peer configuration mode. To return to the default, use the **no paramspace callsetup after-hours-exempt** command.

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>true</strong></td>
<td>Dial peer is exempt from call-blocking configuration.</td>
</tr>
<tr>
<td><strong>false</strong></td>
<td>Dial peer is subject to call-blocking configuration. This is default.</td>
</tr>
</tbody>
</table>

**Command Default**

All dial peers are subject to call-blocking configuration.

**Command Modes**

Dial-peer configuration (config-dial-peer)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Products</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SRST 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is intended to allow H.323 and SIP trunk calls to utilize the voice gateway in spite of the the after-hours configuration in Cisco Unified CME or Cisco Unified SRST.

A Cisco voice gateway (session application) accesses the after-hours call-blocking configuration set by Cisco Unified CME or Cisco Unified SRST and blocks all SCCP, SIP, H.323, and POTS calls that go through the Cisco router regardless of whether the call is from a phone controlled by the Cisco router or from a phone controlled by some other call control application, such as Cisco Unified CallManager.

To disable the After Hours Call Blocking feature for incoming calls from phones other than those registered to a Cisco Unified CME or Cisco Unified SRST router, use this command to exempt an individual H.323, SIP, or POTS dial peer from the call blocking configuration.

To disable the After Hours Call Blocking feature for an individual IP phone registered in Cisco Unified CME or Cisco Unified SRST:

- In Cisco CME 3.4 and later, disable the After Hours Call Blocking feature for a directory number on a SIP phone by using the **after-hour exempt** command in voice register pool or voice register dn configuration mode.
- In Cisco CME 3.0 and later, disable the After Hours Call Blocking feature for an individual SCCP phone by using the **after-hour exempt** command in ephone or ephone-template configuration mode.
- In Cisco SIP SRST 3.4 and later, disable the After Hours Call Blocking feature for SIP phones in a voice register pool by using the **after-hour exempt** command in voice register pool configuration mode.
- In Cisco SRST, you cannot create an exemption for an individual phone from the call-blocking configuration.

**Examples**

The following example shows how to set the After Hours Call Blocking feature in Cisco Unified CME, and how to configure a particular dial peer (255) so that outgoing calls through this dial peer are exempt from this after-hours call blocking configuration:
Router(config)# telephony-service
Router(config-telephony)# after-hours block pattern 1 9011
Router(config-telephony)# exit
Router(config)# dial-peer voice 255 voip
Router(config-dial-peer)# paramspace callsetup after-hours-exempt true

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>after-hour exempt</td>
<td>Specifies that a SCCP phone does not have any of its outgoing calls blocked even though call blocking has been defined.</td>
</tr>
<tr>
<td>after-hour exempt (voice register dn)</td>
<td>Specifies that an individual SIP IP phone or a phone extension on a SIP IP phone does not have any of its outgoing calls blocked even though call blocking has been defined.</td>
</tr>
<tr>
<td>after-hour exempt (voice register pool)</td>
<td>Specifies that an individual SIP IP phone or phones in a voice register pool does not have any of its outgoing calls blocked even though call blocking has been defined.</td>
</tr>
<tr>
<td>after-hours block pattern</td>
<td>Defines a pattern of digits for blocking outgoing calls from IP phones.</td>
</tr>
<tr>
<td>after-hours date</td>
<td>Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
<tr>
<td>after-hours day</td>
<td>Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.</td>
</tr>
</tbody>
</table>
**park reservation-group**

To assign a call-park reservation group to a phone, use the `reservation-group` command in ephone, ephone-template, voice register pool, or voice register template configuration mode. To remove the group from the phone, use the `no` form of this command.

```
park reservation-group group-number
no park reservation-group
```

**Syntax Description**
- `group-number`: Unique number that identifies the reservation group. String can contain up to 32 digits.

**Command Default**
Extension does not belong to any reservation group.

**Command Modes**
- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)
- Voice register pool configuration (config-register-pool)
- Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command allows you to assign ownership to call-park slots by using Park Reservation Groups. A phone configured with a park reservation group can retrieve calls only from park slots configured with the same park reservation group. A phone without a park reservation group can retrieve calls from any park slot without an assigned park reservation group.

To assign a reservation group to a park-slot extension, use the `park-slot reservation-group` command.

If you use a template to apply a command to a phone and you also use the same command in ephone or voice register pool configuration mode for the same phone, the value that you set in the phone configuration mode has priority.

**Examples**
The following example shows park reservation-group 1 is assigned to phone 3 (SCCP). When calls for the Pharmacy are parked at extension 8126, phone 3 can retrieve them:

```
ephone-dn 26
  number 8126
  park-slot reservation-group 1 timeout 15 limit 2 transfer 8100
description park slot for Pharmacy
!
ephone 3
  park reservation-group 1
  mac-address 002D.264E.54FA
type 7962
  button 1:3
```
The following example shows park reservation-group 1 is assigned to phone 120 (SIP). When calls for the Pharmacy are parked at extension 8126, phone 120 can retrieve them:

```
voice register pool 120
park reservation-group 1
id mac 0030.94c2.A22A
type 7962
number 1 dn 20
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>call-park</strong></td>
<td>Defines system parameters for the call-park feature.</td>
</tr>
<tr>
<td><strong>system</strong></td>
<td></td>
</tr>
<tr>
<td><strong>park-slot</strong></td>
<td>Creates an extension (call-park slot) at which calls can be temporarily held (parked).</td>
</tr>
</tbody>
</table>
To create an extension (call-park slot) at which calls can be temporarily held (parked), use the **park-slot** command in ephone-dn configuration mode. To disable the extension, use the **no** form of this command.

```plaintext
park-slot [directed] [reservation-group group-number] [reserved-for extension-number] [(timeout seconds limit count) [notify extension-number [only]] [recall] [transfer extension-number] [alternate extension-number] [retry seconds limit count]]
no park-slot [directed] [reservation-group group-number] [reserved-for extension-number] [(timeout seconds limit count) [notify extension-number [only]] [recall] [transfer extension-number] [alternate extension-number] [retry seconds limit count]]
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax Element</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>directed</strong></td>
<td>(Optional) Enables Directed Call Park for this extension.</td>
</tr>
<tr>
<td><strong>reservation-group</strong></td>
<td>(Optional) Reserves this slot for phones configured with the same reservation group.</td>
</tr>
<tr>
<td><strong>group-number</strong></td>
<td></td>
</tr>
<tr>
<td><strong>reserved-for</strong></td>
<td>(Optional) Reserves this slot as a private park slot for the phone with the specified extension number as its primary line. All lines on that phone can use this park slot.</td>
</tr>
<tr>
<td><strong>extension-number</strong></td>
<td></td>
</tr>
<tr>
<td><strong>timeout</strong></td>
<td>(Optional) Sets the call-park reminder timeout in seconds. Range: 0 to 65535. This reminder sends a 1-second ring to the IP phone that parked the call and displays a message on the LCD panel of all phones in the Cisco Unified CME system, indicating that a call is on hold. By default, the reminder ring is sent only to the phone that parked the call.</td>
</tr>
<tr>
<td><strong>seconds</strong></td>
<td></td>
</tr>
<tr>
<td><strong>limit</strong></td>
<td>(Optional) Sets a limit on the number of reminder or retry timeouts. Range: 1 to 65535.</td>
</tr>
<tr>
<td><strong>count</strong></td>
<td></td>
</tr>
<tr>
<td><strong>notify</strong></td>
<td>(Optional) Sends a reminder ring to the specified extension in addition to the reminder ring that is sent to the phone that parked the call.</td>
</tr>
<tr>
<td><strong>extension-number</strong></td>
<td></td>
</tr>
<tr>
<td><strong>only</strong></td>
<td>(Optional) Sends a reminder ring only to the extension specified with the <strong>notify</strong> keyword and does not send a reminder ring to the phone that parked the call. This option allows all reminder rings for parked calls to be sent to a receptionist’s phone or an attendant’s phone, for example.</td>
</tr>
<tr>
<td><strong>recall</strong></td>
<td>(Optional) Returns the call to the phone that parked it after the timeout expires.</td>
</tr>
<tr>
<td><strong>transfer</strong></td>
<td>(Optional) Returns the call to the specified extension after the timeout expires.</td>
</tr>
<tr>
<td><strong>extension-number</strong></td>
<td></td>
</tr>
<tr>
<td><strong>alternate</strong></td>
<td>(Optional) Returns the call to this second target number if the recall or transfer target phone is in use on any of its extensions (ringing or connected).</td>
</tr>
<tr>
<td><strong>extension-number</strong></td>
<td></td>
</tr>
<tr>
<td><strong>retry</strong></td>
<td>(Optional) Sets the delay before another attempt to recall or transfer a parked call, in seconds. Range: 0 to 65535. Number of attempts is set by the <strong>limit</strong> keyword.</td>
</tr>
<tr>
<td><strong>seconds</strong></td>
<td></td>
</tr>
<tr>
<td><strong>limit</strong></td>
<td></td>
</tr>
<tr>
<td><strong>count</strong></td>
<td></td>
</tr>
</tbody>
</table>
Command Default
No call-park slot is defined.

Command Modes
Ephone-dn configuration (config-ephone-dn)

Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(7)T</td>
<td>Cisco CME 3.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The reserved-for, recall, transfer, alternate, and retry keywords were added.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>The directed and reservation-group keywords and support for SIP phones was added.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines
This command creates a call-park slot that is a floating extension, or ephone-dn that is not bound to a physical phone, at which phone users can place calls on hold for later retrieval from the same phone or from another phone.

At least one call-park slot must be defined with this command before the Park soft key displays on IP phones in a Cisco Unified CME system.

Phone users park calls using the Park soft key. A phone user can then retrieve a call by dialing the extension number of the call-park slot. On SCCP phones, the phone user who parks the call can also retrieve the call by using the PickUp soft key and an asterisk (*). Other SCCP phone users can retrieve the call by using the PickUp soft key and dialing the extension number of the call-park slot.

Calls can also be transferred to a call-park slot using the Transfer key; a transfer to a call-park slot is always a blind transfer. Calls can also be forwarded to a call-park slot, and callers can directly dial call-park slots.

When a call that uses a G.711 codec is parked, the caller hears the music-on-hold (MOH) audio stream; otherwise, the caller hears the on-hold tone.

The directed keyword enables the extension as a park slot for Directed Call Park. To retrieve a call from a directed call-park slot, you must define the retrieval prefix with the fac command. The retrieval prefix is supported for both SCCP and SIP phones.

The reservation-group keyword allows you to assign ownership to call-park slots by using Park Reservation Groups. A park slot configured with a park reservation group can only be used by phones configured with the same park reservation group. A park slot without a park reservation group can be used by any phone not assigned to a park reservation group.

A reminder ring can be sent to the extension that parked the call by using the timeout keyword, which sets the interval length to wait before sending call-park reminder rings. The number of time-out intervals and reminder rings are configured with the limit keyword and argument. For example, a limit of 3 timeout intervals sends 2 reminder rings (interval 1, ring 1, interval 2, ring 2, interval 3). The timeout and limit keywords and arguments also set the maximum time that calls stay parked. For example, a timeout interval of 10 seconds and a limit of 5 timeout intervals (park-slot timeout 10 limit 5) will park calls for approximately 50 seconds.

If the timeout keyword is not used with this command, no reminder ring is sent to the extension that parked the call. If the timeout keyword is used, a reminder ring is sent only to the extension that parked the call unless the notify keyword is also used to specify an additional extension number to receive a reminder ring.
When an additional extension number is specified using the `notify` keyword, the phone user at that extension can retrieve a call from this slot by pressing the PickUp soft key and an asterisk (*).

Each call-park slot can hold one call at a time, so the number of simultaneous calls that can be parked is equal to the number of slots that have been created. The `reserved-for` keyword creates a call-park slot that is dedicated for use by one extension so that extension always has a slot available at which to park a call. With nonreserved slots, multiple call-park slots can be created with the same extension number so that all the calls that are parked for a particular group can be parked at a known extension number. For example, at a hardware store, calls for the plumbing department can be parked at extension 101, calls for lighting can be parked at 102, and so forth. Then, anyone in the plumbing department can pick up calls from extension 101. When multiple calls are parked at the same extension number, they are picked up in the order in which they were parked; that is, the call that has been parked the longest is the first call picked up from that extension number.

IP phone users park calls at their dedicated call-park slots using the Park soft key. Phone users can also transfer calls to dedicated call-park slots using the Transfer soft key and a standard or custom feature access code (FAC) for call park. On analog phones, users transfer calls to dedicated call-park slots using hookflash and a standard or custom FAC for call park. The standard FAC for call park is **6. Custom FACs are created using the `fac` command.

If a dedicated park slot is not found for an ephone-dn attempting to park a call, Cisco Unified CME uses the standard call-park procedure; that is, the system searches for a preferred park slot (one with an ephone-dn number that matches the last two digits of the ephone-dn attempting to park the call) and if none is found, uses any available call-park slot.

If a name has been specified for a call-park slot, that name is displayed instead of an extension number on a recall or transfer of the call.

A parked call can have the following dispositions after its timeouts expire:

- **Recall**—If you specify that a call should be recalled to the parking phone after the timeout interval expires, the call is always returned to the phone's primary extension number, regardless of which extension on the phone did the parking.
- **Transfer**—If you specify a transfer target, the call is transferred to the specified number after the timeout intervals expire instead of returning to the primary number of the phone that did the parking.
- **Alternate**—You can also specify an alternate target extension to which to transfer a parked call if the recall or transfer target is in use. *In use* is defined as either ringing or connected to a call. For example, a call is parked at the dedicated park slot for the phone with the primary extension of 2001. After the timeouts expire, the system attempts to recall the call to extension 2001, but that line is now connected to a different call. The system transfers the call to the alternate target that was specified when the park slot was defined.
- **Disconnect**—When a timeout limit is set and no other disposition has been specified, a call parked at a call-park slot is disconnected after the number of reminder timeouts are reached.

### Examples

#### Basic Call Park

The following example shows a basic call-park slot at extension 1001. After a call is parked at this number, the system provides 10 reminder rings at intervals of 30 seconds to the extension that parked the call. Any phone can retrieve calls parked at this extension.

```
ephone-dn 45
  number 1001
  park-slot timeout 30 limit 10
```
Directed Call Park (Cisco Unified CME 4.4 and Later Versions)

The following example shows two call-park slots, extension 3110 and 3111, that can be used to park calls for the pharmacy using Directed Call Park.

```plaintext
ephone-dn 10
  number 3110
  park-slot directed
description park-slot for Pharmacy
!
ephone-dn 11
  number 3111
  park-slot directed
description park-slot for Pharmacy
```

Park Reservation Groups (Cisco Unified CME 4.4 and Later Versions)

The following example shows park reservation groups set up for two call-park slots. Extension 8126 is configured for group 1 and assigned to phones 3 and 4. Extension 8127 is configured for group 2 and assigned to phones 10 and 11. When calls for the Pharmacy are parked at extension 8126, only phones 3 and 4 can retrieve them.

```plaintext
ephone-dn 26
  number 8126
  park-slot reservation-group 1 timeout 15 limit 2 transfer 8100
description park slot for Pharmacy
!
ephone-dn 27
  number 8127
  park-slot reservation-group 2 timeout 15 limit 2 transfer 8100
description park slot for Auto
!
ephone 3
  park reservation-group 1
  mac-address 002D.264E.54FA
type 7962
button 1:3
!
ephone 4
  park reservation-group 1
  mac-address 0030.94C3.053E
type 7962
button 1:4
!
ephone 10
  park reservation-group 2
  mac-address 00E1.CB13.0395
type 7960
button 1:10
!
ephone 11
  park reservation-group 2
  mac-address 0016.9DEF.1A70
type 7960
button 1:11
```
Dedicated Park

The following example shows a dedicated call-park slot, 2558, that is reserved for the phone that has the primary extension of 2977. Both extension 2977 and 2976 are on the same phone, so they both can use this slot, which is reserved only for the extensions on that phone. After three timeout intervals of 60 seconds, a parked call is recalled to extension 2977. If extension 2977 is busy, the call is rerouted to extension 3754.

ephone-dn 24
  number 2977

ephone-dn 25
  number 2976

ephone-dn 27
  number 3754

ephone-dn 30
  number 2558
  name Park 2977
  park-slot reserved-for 2977 timeout 60 limit 3 recall alternate 3754

ephone 44
  button 1:24 2:25

ephone 45
  button 1:27

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td><strong>call-park system</strong></td>
<td>Defines system parameters for the call-park feature.</td>
</tr>
<tr>
<td></td>
<td><strong>fac</strong></td>
<td>Enables standard FACs or creates custom FACs.</td>
</tr>
<tr>
<td></td>
<td><strong>number</strong></td>
<td>Associates a telephone or extension number with a directory number.</td>
</tr>
<tr>
<td></td>
<td><strong>park reservation-group</strong></td>
<td>Assigns a call-park reservation group to a phone.</td>
</tr>
</tbody>
</table>
password (auto-register)

To configure the mandatory password for automatic registration of SIP phones with the Cisco Unified CME system, use the `password` command in voice auto register configuration mode. This command is a sub-mode CLI of the command `auto-register`. To disable configuring password for auto registration of SIP phones, use the `no` form of this command.

```
password [0|6] string
```

**Syntax Description**

- `password string`: The mandatory word string that administrator provides for auto registration of phones on Unified CME.

**Command Default**

By default, this command is disabled.

**Command Modes**

- voice auto register configuration (config-voice-auto-register)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.6(3)M</td>
<td>Cisco Unified CME 11.5</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>16.3.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command was enhanced for password encryption, based on Unified CME password policy.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables the administrator to configure the password credentials for SIP phones auto registering on Unified CME. It is mandatory that the password is configured before assigning the DN range.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters `[0|6]`. This in accordance with Unified CME Password Policy. The 0 in the parameter `[0|6]` mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

**Examples**

The following example shows how to configure password for auto registration of SIP phones:

```
Router(config)#voice register global
Router(config-register-global)#auto-register
Router(config-voice-auto-register)#?
```

**Examples (continued)**

```
VOICE auto register configuration commands:
  auto-assign Define DN range for auto assignment
default Set a command to its defaults
exit Exit from voice register group configuration mode
no Negate a command or set its defaults
password Default password for auto-register phones
service-enable Enable SIP phone Auto-Registration
template Default template for auto-register phones

Router(config-voice-auto-register)#password ?
WORD Password string
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>service-enable (auto-register)</td>
<td>Temporarily disables the auto registration process, but retains the password and DN range configurations. Once auto-register command is entered, the service is enabled by default.</td>
</tr>
<tr>
<td>auto-register</td>
<td>Enables automatic registration of SIP phones with the Cisco Unified CME system.</td>
</tr>
<tr>
<td>auto-assign (auto-register)</td>
<td>Configures the mandatory range of directory numbers for phones auto registering on Unified CME.</td>
</tr>
<tr>
<td>template (auto-register)</td>
<td>Creates a basic configuration template that supports all the configurations available on the voice register template.</td>
</tr>
<tr>
<td>auto-reg-ephone</td>
<td>Enables automatic registration of ephones with the Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
password-persistent

To configure password-persistent option for a vpn-profile, use the `password-persistent` command in vpn-profile configuration mode.

```
password-persistent [ { enabled | disable } ]
```

**Syntax Description**
- **enable** Enables password-persistent to authenticate.
- **disable** Disables password-persistent to authenticate.

**Command Default**
Password-persistent is disabled.

**Command Modes**
Vpn-profile configuration (conf-vpn-profile)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to enable or disable password-persistent option for a vpn-profile.

**Examples**
The following example shows the password-persistent command enabled for vpn-profile 2:

```
Router#show run
!
voice service voip
  ip address trusted list
    ipv4 20.20.20.1
  vpn-group 1
    vpn-gateway 1 https://9.10.60.254/SSLVPNphone
    vpn-trustpoint 1 trustpoint cme_cert root
    vpn-hash-algorithm sha-1
  vpn-profile 1
    keepalive 50
    host-id-check disable
  vpn-profile 2
    mtu 1300
    password-persistent enable
    host-id-check enable
    sip
    !
  voice class media 10
    media flow-around
  !
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>vpn-profile</code></td>
<td>Defines a VPN-profile.</td>
</tr>
</tbody>
</table>
pattern (voice register dialplan)

To define a dial pattern for a SIP dial plan, use the `pattern` command in voice register dialplan configuration mode. To remove the pattern, use the `no` form of this command.

```shell
pattern tag string [button button-number] [timeout seconds] [user {ip|phone}] 
no pattern tag
```

**Syntax Description**

- **tag**: Number that identifies the dial pattern. Range: 1 to 24.
- **string**: Dial pattern, such as the area code, prefix, and first one or two digits of the telephone number, plus wildcard characters or dots (.) for the remainder of the dialed digits.
- **button button-number**: (Optional) Button to which the dial pattern applies.
- **timeout seconds**: (Optional) Time, in seconds, that the system waits before dialing the number entered by the user. Range: 0 to 30. To have the number dialed immediately, specify 0. If this parameter is not used, the phone's default interdigit timeout value is used (10 seconds).
- **user**: (Optional) Tag that automatically gets added to the dialed number. Do not use this keyword if Cisco Unified CME is the only SIP call agent.
- **ip**: (Optional) Sets the value of the user tag to IP in the dialed number.
- **phone**: (Optional) Sets the value of the user tag to phone in the dialed number.

**Command Default**

No pattern is defined.

**Command Modes**

Voice register dialplan configuration (config-register-dialplan)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command defines a pattern of dialed digits that are matched by the phone and passed to Cisco Unified CME to initiate a call. Dial strings that match the pattern trigger the sending of a SIP INVITE message to Cisco Unified CME. Patterns are matched sequentially in order of the `tag` number.

You must first use the `type` command to specify the type of phone that the dial plan is being defined for before you can enter a pattern. Enter this command for each dial pattern that is part of the dial plan definition. After you define a dial plan, assign it to a SIP phone by using the `dialplan` command.

The `button` keyword specifies the button to which the dial pattern applies. If the user is initiating a call on line button 1, only the dial patterns specified for button 1 apply. If this keyword is not configured, the dial pattern applies to all lines on the phone. This keyword is not supported on Cisco Unified IP Phones 7905 or 7912. The button number corresponds to the order of the buttons on the side of the screen, from top to bottom, with 1 being the top button.
The **pattern** command and **filename** command are mutually exclusive. You can use either the **pattern** command to define dial patterns manually for a dial plan, or the **filename** command to select a custom dial pattern file that is loaded in system flash.

### Examples

The following example shows the dial patterns set for SIP dial plan 10:

```
Router(config)# voice register dialplan 10
Router(config-register-dialplan)# type 7905-7912
Router(config-register-dialplan)# pattern 52...
Router(config-register-dialplan)# pattern 91........
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dialplan</td>
<td>Assigns a dial plan to a SIP phone.</td>
</tr>
<tr>
<td>filename</td>
<td>Specifies a custom configuration file that contains dial patterns to use for the SIP dial plan.</td>
</tr>
<tr>
<td>show voice register dialplan</td>
<td>Displays all configuration information for a specific SIP dial plan.</td>
</tr>
<tr>
<td>type (voice register dialplan)</td>
<td>Defines a phone type for a SIP dial plan.</td>
</tr>
</tbody>
</table>
pattern direct

To configure the dual tone multifrequency (DTMF) digit pattern forwarding necessary to activate the voice-mail system when a user presses the Messages button on an IP phone in a Cisco CallManager Express (Cisco CME) system, use the **pattern direct** command in voice-mail integration configuration mode. To disable DTMF pattern forwarding when a user presses the Messages button on a phone, use the **no** form of this command.

```
pattern direct  tag1  {CDN|CGN|FDN}  [tag2  {CDN|CGN|FDN}]  [tag3  {CDN|CGN|FDN}]  [last-tag]  
no pattern direct
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>tag1</code></td>
<td>Alphanumeric string of fewer than four DTMF digits in length. The alphanumeric string can consist of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system’s integration file and immediately precede the number of the calling party, the number of the called party, or a forwarding number.</td>
</tr>
<tr>
<td><code>CDN</code></td>
<td>Called number (CDN) information is sent to the voice-mail system.</td>
</tr>
<tr>
<td><code>CGN</code></td>
<td>Calling number (CGN) information is sent to the voice-mail system.</td>
</tr>
<tr>
<td><code>FDN</code></td>
<td>Forwarding number (FDN) information is sent to the voice-mail system.</td>
</tr>
<tr>
<td><code>tag2</code>, <code>tag3</code></td>
<td>(Optional) Same as <code>tag1</code>. The router supports a maximum of four tags.</td>
</tr>
<tr>
<td><code>last-tag</code></td>
<td>(Optional) Same as <code>tag1</code>. This tag indicates the end of the pattern.</td>
</tr>
</tbody>
</table>

**Command Default**

This feature is disabled.

**Command Modes**

Voice-mail integration configuration (config-vm-integration)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>2.0</td>
<td>This command was introduced</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The **pattern direct** command is used to configure the sequence of dual tone multifrequency (DTMF) digits passed to a voice-mail system attached to the router through one or more voice ports. When a call is placed directly from a Cisco IP phone attached to the router, the voice-mail system expects to receive a sequence of DTMF digits at the beginning of the call to identify the user’s mailbox, accompanied by a string of digits to indicate that the caller is attempting to access the designated mailbox in order to retrieve messages.

Although it is unlikely that you will use multiple instances of the **CDN**, **CGN**, or **FDN** keywords in a single command line, it is permissible to do so.

**Examples**

The following example sets the DTMF pattern for a calling number (CGN) for a direct call to the voice-mail system:
**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>pattern ext-to-ext busy</strong></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to a busy extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td><strong>pattern ext-to-ext no-answer</strong></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension attempts to connect to an extension that does not answer and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td><strong>pattern trunk-to-ext busy</strong></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td><strong>pattern trunk-to-ext no-answer</strong></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td><strong>vm-integration</strong></td>
<td>Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and an analog voice-mail systems.</td>
</tr>
</tbody>
</table>
pattern ext-to-ext busy

To configure the dual tone multifrequency (DTMF) digit pattern forwarding necessary to activate a voice-mail system after an internal extension attempts to connect to a busy extension and the call is forwarded to voice mail, use the pattern **ext-to-ext busy** command in voice-mail integration configuration mode. To disable the feature, use the **no** form of this command.

```
pattern ext-to-ext busy tag1 {CDN|CGN|FDN} [tag2 {CDN|CGN|FDN}] [tag3 {CDN|CGN|FDN}] [last-tag]
no pattern ext-to-ext busy
```

**Syntax Description**

<table>
<thead>
<tr>
<th>tag</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>tag1</td>
<td>Alphanumeric string of fewer than four DTMF digits in length. The alphanumeric string can consist of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system’s integration file and immediately precede the number of the calling party, the number of the called party, or a forwarding number.</td>
</tr>
<tr>
<td>CDN</td>
<td>Called number (CDN) information is sent to the voice-mail system.</td>
</tr>
<tr>
<td>CGN</td>
<td>Calling number (CGN) information is sent to the voice-mail system.</td>
</tr>
<tr>
<td>FDN</td>
<td>Forwarding number (FDN) information is sent to the voice-mail system.</td>
</tr>
<tr>
<td>tag2, tag3</td>
<td>(Optional) Same as tag1. The router supports a maximum of four tags.</td>
</tr>
<tr>
<td>last-tag</td>
<td>(Optional) Same as tag1. This tag indicates the end of the pattern.</td>
</tr>
</tbody>
</table>

**Command Default**

This feature is disabled.

**Command Modes**

Voice-mail integration configuration (config-vm-integration)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.2(13)T</td>
<td>Cisco SRST 2.02</td>
<td>This command was added for Cisco SRST.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The pattern **ext-to-ext busy** command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the router through one or more voice ports. When a call is routed to the voice-mail system by call forward on busy from a Cisco IP phone attached to the router, the voice-mail system expects to receive digits that identify the mailbox associated with the forwarding phone together with digits that identify the extension number of the calling IP phone.

Although it is unlikely that you will use multiple instances of the CDN, CGN, or FDN keywords in a single command line, it is permissible to do so.
Examples

The following example sets the DTMF pattern for a local call forwarded on busy to the voice-mail system:

Router(config) vm-integration
Router(config-vm-integration) pattern ext-to-ext busy 7 FDN * CGN *

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>pattern direct</strong></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the Messages button on the phone.</td>
</tr>
<tr>
<td><strong>pattern ext-to-ext no-answer</strong></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension that does not answer and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td><strong>pattern trunk-to-ext busy</strong></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td><strong>pattern trunk-to-ext no-answer</strong></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td><strong>vm-integration</strong></td>
<td>Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and an analog voice-mail systems.</td>
</tr>
</tbody>
</table>
**pattern ext-to-ext no-answer**

To configure the dual tone multifrequency (DTMF) pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to a non answering extension and the call is forwarded to voice mail, use the `pattern ext-to-ext no-answer` command in voice-mail integration configuration mode. To disable this feature, use the `no` form of this command.

```
pattern ext-to-ext no-answer  tag1  {CDN|CGN|FDN}  [tag2  {CDN|CGN|FDN}]}  [tag3
{CDN|CGN|FDN}]}  [last-tag]  
no  pattern  ext-to-ext  no-answer
```

**Syntax Description**

| **tag1** | Alphanumeric string of fewer than four DTMF digits in length. The alphanumeric string can consist of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system’s integration file and immediately precede the number of the calling party, the number of the called party, or a forwarding number. |
| CDN | Called number (CDN) information is sent to the voice-mail system. |
| CGN | Calling number (CGN) information is sent to the voice-mail system. |
| FDN | Forwarding number (FDN) information is sent to the voice-mail system. |
| **tag2, tag3** | (Optional) Same as `tag1`. The router supports a maximum of four tags. |
| **last-tag** | (Optional) Same as `tag1`. This tag indicates the end of the pattern. |

**Command Default**

This feature is disabled.

**Command Modes**

Voice-mail integration configuration (config-vm-integration)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
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</tr>
</thead>
<tbody>
<tr>
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<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.2(13)T</td>
<td>Cisco SRST 2.02</td>
<td>This command was added for Cisco SRST.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `pattern ext-to-ext no-answer` command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the router through one or more voice ports. When a call is routed to the voice-mail system by call forward on no-answer from an IP phone attached to the router, the voice-mail system expects to receive digits that identify the mailbox associated with the forwarding phone together with digits that identify the extension number of the calling IP phone.

Although it is unlikely that you will use multiple instances of the **CDN**, **CGN**, or **FDN** keywords in a single command line, it is permissible to do so.
The following example sets the DTMF pattern for a local call forwarded on no-answer to the voice-mail system:

Router(config) vm-integration
Router(config-vm-integration) pattern ext-to-ext no-answer 5 FDN * CGN *

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>pattern direct</code></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the Messages button on the phone.</td>
</tr>
<tr>
<td><code>pattern ext-to-ext busy</code></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td><code>pattern trunk-to-ext busy</code></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td><code>pattern trunk-to-ext no-answer</code></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td><code>vm-integration</code></td>
<td>Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and an analog voice-mail systems.</td>
</tr>
</tbody>
</table>
pattern trunk-to-ext busy

To configure the dual tone multifrequency (DTMF) digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail, use the `pattern trunk-to-ext busy` command in voice-mail integration configuration mode. To return to the default, use the `no` form of this command.

```
pattern trunk-to-ext busy tag1 {CDN|CGN|FDN} [tag2 {CDN|CGN|FDN}] [tag3 {CDN|CGN|FDN}] [last-tag]
```

### Syntax Description

<table>
<thead>
<tr>
<th>tag1</th>
<th>Alphanumeric string of fewer than four DTMF digits in length. The alphanumeric string can consist of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system’s integration file and immediately precede the number of the calling party, the number of the called party, or a forwarding number.</th>
</tr>
</thead>
<tbody>
<tr>
<td>CDN</td>
<td>Called number (CDN) information is sent to the voice-mail system.</td>
</tr>
<tr>
<td>CGN</td>
<td>Calling number (CGN) information is sent to the voice-mail system.</td>
</tr>
<tr>
<td>FDN</td>
<td>Forwarding number (FDN) information is sent to the voice-mail system.</td>
</tr>
<tr>
<td>tag2,</td>
<td>(Optional) Same as tag1. The router supports a maximum of four tags.</td>
</tr>
<tr>
<td>tag3</td>
<td></td>
</tr>
<tr>
<td>last-tag</td>
<td>(Optional) Same as tag1. This tag indicates the end of the pattern.</td>
</tr>
</tbody>
</table>

### Command Default

This feature is disabled by default.

### Command Modes

Voice-mail integration configuration (config-vm-integration)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco SRST 2.02</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.2(13)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was added for Cisco SRST.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

The `pattern trunk-to-ext busy` command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the router through one or more voice ports. When a call is routed to the voice-mail system by call forward on busy from an IP phone attached to the router, the voice-mail system expects to receive a sequence of digits identifying the mailbox associated with the forwarding phone together with digits indicating that the call originated from a PSTN or VoIP caller.

Although it is unlikely that you will use multiple instances of the CDN, CGN, or FDN keywords in a single command line, it is permissible to do so.
The following example sets the DTMF pattern for call forwarding when an external trunk call reaches a busy extension and the call is forwarded to the voice-mail system:

```plaintext
Router(config) vm-integration
Router(config-vm-integration) pattern trunk-to-ext busy 6 FDN * CGN *
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>pattern direct</strong></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the Messages button on the phone.</td>
</tr>
<tr>
<td><strong>pattern ext-to-ext busy</strong></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td><strong>pattern ext-to-ext no-answer</strong></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td><strong>pattern trunk-to-ext no-answer</strong></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td><strong>vm-integration</strong></td>
<td>Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and an analog voice-mail systems.</td>
</tr>
</tbody>
</table>
**pattern trunk-to-ext no-answer**

To configure the dual tone multifrequency (DTMF) digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail, use the `pattern trunk-to-ext no-answer` command in voice-mail integration configuration mode. To disable this feature, use the `no` form of this command.

`pattern trunk-to-ext no-answer tag1 {CDN|CGN|FDN} [tag2 {CDN|CGN|FDN} ] [tag3 {CDN|CGN|FDN} ] [last-tag]`

`no pattern trunk-to-ext no-answer`

<table>
<thead>
<tr>
<th>Syntax Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>tag1</code></td>
</tr>
<tr>
<td>CDN</td>
</tr>
<tr>
<td>CGN</td>
</tr>
<tr>
<td>FDN</td>
</tr>
<tr>
<td><code>tag2</code>, <code>tag3</code> (Optional) Same as <code>tag1</code>. The router supports a maximum of four tags.</td>
</tr>
<tr>
<td><code>last-tag</code> (Optional) Same as <code>tag1</code>. This tag indicates the end of the pattern.</td>
</tr>
</tbody>
</table>

**Command Default**

This feature is disabled.

**Command Modes**

Voice-mail integration configuration (config-vm-integration)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco CME Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.2(13)T</td>
<td>2.02</td>
<td>This command was added for Cisco SRST.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `pattern trunk-to-ext no-answer` command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the router through one or more voice ports. When a call is routed to the voice-mail system by call forward on no-answer from an IP phone attached to the router, the voice-mail system expects to receive digits that identify the mailbox associated with the forwarding phone together with digits that indicate that the call originated from a PSTN or VoIP caller.

Although it is unlikely that you will use multiple instances of the CDN, CGN, or FDN keywords in a single command line, it is permissible to do so.
Examples

The following example sets the DTMF pattern for call forwarding when an external trunk call reaches an unanswered extension and the call is forwarded to a voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern trunk-to-ext no-answer 4 FDN * CGN *
```

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>pattern direct</td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the Messages button on the phone.</td>
</tr>
<tr>
<td>pattern ext-to-ext busy</td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td>pattern ext-to-ext no-answer</td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td>pattern trunk-to-ext busy</td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td>vm-integration</td>
<td>Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and an analog voice-mail systems.</td>
</tr>
</tbody>
</table>
phone-display

To enable a phone user to display voice hunt group information using the Services button on the phone, use the `phone-display` command in voice hunt group configuration mode. To remove the configuration, use the `no` form of this command.

```
phone-display
no phone-display
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
By default, this command is disabled.

**Command Modes**
ephone configuration mode

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command when configured, enables the user to view the information of a specific voice hunt group on the phone.

**Example**
The following example shows how the voice hunt group display option is enabled for a phone:

```
Router(config)# voice hunt-group 1 parallel
Router(config-voice-hunt-group)# phone-display
```
phone-mode only

To enable Jabber phone-only client support, use the `phone-mode only` command. To remove the configuration, use the `no` form of this command.

```
phone-mode phone only
nophone-mode phone only
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
By default, this feature is disabled.

**Command Modes**
voice register global (config-register global)
voice register pool (config-register pool)
voice register template (config-register template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command enables Jabber phone-only client support.

**Example**
The following example shows how phone-mode is enabled:

```
Router(config)# voice register pool
Router(config-register pool)# phone-mode phone-only
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register global</td>
<td>Enters voice register global configuration mode.</td>
</tr>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode.</td>
</tr>
<tr>
<td>voice register template</td>
<td>Enters voice register template configuration mode.</td>
</tr>
</tbody>
</table>
### phone-key-size

To specify the size of the RSA key pair that is generated on phones, use the `phone-key-size` command in CAPF-server configuration mode. To return the size to the default, use the `no` form of this command.

```
phone-key-size {512|1024|2048}
no phone-key-size
```

**Syntax Description**

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>512</strong></td>
<td>512 bits</td>
</tr>
<tr>
<td><strong>1024</strong></td>
<td>1024 bits. This is the default key size.</td>
</tr>
<tr>
<td><strong>2048</strong></td>
<td>2048 bits</td>
</tr>
</tbody>
</table>

**Command Default**

RSA key pair size is 1024.

**Command Modes**

CAPF-server configuration (config-capf-server)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication.

If you choose a higher key size than the default setting, the phones take longer to generate the entropy that is required to generate the keys. Key generation, which is set at low priority, allows the phone to function while the action occurs. Depending on the phone model, you may notice that key generation takes up to 30 or more minutes to complete.

**Examples**

The following example specifies a key size of 2048 bits.

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x80Wiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```
phoneload

To define the phone firmware support for a phone type, use the `phoneload` command in ephone-type configuration mode. To remove firmware support, use the `no` form of this command.

```
phoneload
no phoneload
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Phone type supports firmware configuration.

**Command Modes**

Ephone-type configuration (config-ephone-type)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3 Cisco Unified SRST 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies whether the phone type defined in the phone-type template supports firmware configuration using the `load` command.

**Examples**

The following example shows that support for phone firmware is disabled for the Nokia E61 phone type:

```
Router(config)# ephone-type E61
Router(config-ephone-type)# no phoneload
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>device-name</code></td>
<td>Assigns a name to a phone type in an ephone-type template.</td>
</tr>
<tr>
<td><code>load</code></td>
<td>Associates a type of Cisco Unified IP phone with a phone firmware file.</td>
</tr>
</tbody>
</table>
phoneload-support

To define the phone support for firmware download from CME, use the `phoneload-support` command in voice register pool-type mode. To disable phoneload support, use the `no` form of this command.

`phoneload-support`
`noponeload-support`

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

The phoneload support is disabled. When the `reference-pooltype` command is configured, phoneload support property of the reference phone is inherited.

**Command Modes**

Voice Register Pool Configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(3)M</td>
<td>Cisco SIP CME 10.0</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to define the default transport type. If the new phone supports the phoneload, you can use the `load` command in voice register global” mode to configure the corresponding phoneload for the new phone model. When you use the no form of this command, the inherited properties of the reference phone takes precedence over the default value.

**Example**

The following example shows how to define the phoneload support for a new phone model using the `phoneload-support` command:

```
Router(config)# voice register global
Router(config--register-global)# mode cme
Router(config-register-global)# load 9900 P0S3-06-0-00
Router(config-register-global)# phoneload-support
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>voice register pool-type</code></td>
<td>Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.</td>
</tr>
<tr>
<td><code>load</code></td>
<td>Associates a type of IP phone with a phone firmware file.</td>
</tr>
</tbody>
</table>
phone-redirect-limit (voice register global)

To set the number of 3XX responses an originating SIP phone in a Cisco CallManager Express (Cisco CME) system can accept for a single call, use the `phone-redirect-limit` command in voice register global configuration mode. To revert to the default, use the `no` form of this command.

```
phone-redirect-limit number
no phone-redirect-limit
```

**Syntax Description**

- `number` Maximum number of 3XX responses accepted for a single call. Range: 5 to 20. Default: 5.

**Command Default**

Default is 5

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to control how many subsequent 3XX responses an originating SIP phone can handle for a single call. The terminating side is any forwarding party which does not use B2BUA, but sends 3XX directly to the originating calling phone. When Cisco CME gets a 3XX from the terminating side, Cisco CME relays the 3XX to the originating SIP phone. The default number of 3XXs that the originating phone can accept is 5.

The following example shows how to set the maximum number of redirects to 6:

```
Router(config)# voice register global
Router(config-register-global)# phone-redirect-limit 6
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>voice register global</code> Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.</td>
</tr>
</tbody>
</table>
phone-ui park-list

To enable a phone user to view the list of active parked calls, use the `phone-ui park-list` command in the ephone configuration mode. To remove the configuration, use the `no` form of this command.

```
phone-ui park-list
no phone-ui park-list
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

By default, this feature is enabled for Skinny Call Control Protocol (SCCP) phones.

**Command Modes**

ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables the park-list menu option under My-Phone-Apps service button menu.

**Example**

The following example shows how to enable the park list display option for phone 7:

```
Router(config)# ephone 7
Router(config-ephone-type)# phone-ui park-list
```

**Example**

The following example shows how to disable park list display option for phone 7:

```
Router(config)# ephone 7
Router(config-ephone-type)# no phone-ui park-list
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>url button</td>
<td>Enables the configuration of the URL Services feature button on a line key.</td>
</tr>
<tr>
<td>url services</td>
<td>Associates a URL with the Programmable Services feature button on the supported Cisco Unified SCCP phones.</td>
</tr>
</tbody>
</table>
phone-ui speeddial-fastdial

To enable a phone user to configure speed-dial and fast-dial numbers from their phone, use the phone-ui speeddial-fastdial command in ephone configuration mode. To reset to the default value, use the no form of this command.

```
phone-ui speeddial-fastdial
no phone-ui speeddial-fastdial
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Enabled (speed-dial and fast-dial numbers are configurable from phone).

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables the speed-dial and fast-dial configuration menu on the phone so that users can configure these options directly.

The services URL must be configured using the url services command.

**Examples**

The following example shows that the speed-dial and fast-dial user interface is disabled for phone 7:

```
Router(config)# ephone 7
Router(config-ephone-type)# no phone-ui speeddial-fastdial
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>fastdial</td>
<td>Creates an entry for a personal speed-dial number.</td>
</tr>
<tr>
<td>speed-dial</td>
<td>Creates speed-dial definitions for a phone.</td>
</tr>
<tr>
<td>url services</td>
<td>Associates a URL with the programmable Services feature button on supported Cisco Unified IP phones.</td>
</tr>
</tbody>
</table>
To enable a Skinny Call Control Protocol (SCCP) phone user to display voice hunt group information using the Services button on a phone, use the `phone-ui voice-hunt-groups` command in ephone configuration mode. To remove the configuration, use the `no` form of this command.

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

By default, this command is enabled.

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables the Voice Hunt Groups menu option under the My-Phone-Apps service button menu.

**Example**

The following example shows how to disable the voice hunt group display option for phone 7:

```
Router(config)# ephone 7
Router(config-ephone-type)# no phone-ui voice-hunt-groups
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>url services</code></td>
<td>Associates a URL with the Programmable Services feature button on supported Cisco Unified IP phones.</td>
</tr>
</tbody>
</table>
pickup-call any-group

To enable a phone user to pickup a ringing call on extensions in any pickup group, use the **pickup-call any-group** command in ephone-dn or voice register dn configuration mode. To reset to the default value, use the **no** form of this command.

```plaintext
pickup-call any-group
no pickup-call any-group
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

User can pickup calls in other groups by pressing GPickUp soft key and dialing pickup group number.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)
Voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command allows a phone user to pickup any ringing call within the local Cisco Unified CME system by pressing the GPickUp soft key and asterisk (*), if the ringing extension is configured with a pickup group using the **pickup-group** command.

If this command is not configured, a phone user can pickup calls only from their local group by pressing the GPickUp soft key and *. To pickup calls in another group, the user must press the GPickUp soft key and dial the pickup group number.

**Examples**

The following example shows that extension 1020 can pick up calls ringing on extension 1030 by pressing the GPickUp softkey and *:

```plaintext
ephone-dn 102
  number 1020
  pickup-call any-group

ephone-dn 103
  number 1030
  pickup-group 5
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>pickup-group</td>
<td>Assigns an extension to a call-pickup group.</td>
</tr>
<tr>
<td>service directed-pickup</td>
<td>Enables Directed Call Pickup and modifies the function of the GPickUp and PickUp soft keys.</td>
</tr>
<tr>
<td>softkeys idle</td>
<td>Modifies the soft-key display on IP phones during the idle call state.</td>
</tr>
</tbody>
</table>
**pickup-group**

To assign an extension to a call-pickup group, use the `pickup-group` command in ephone-dn, ephone-dn-template, or voice register dn configuration mode. To remove the extension from a group, use the `no` form of this command.

```
pickup-group  group-number
no  pickup-group
```

**Syntax Description**

| group-number | String representing a pickup group. The string can contain up to 32 characters. |

**Command Default**

An extension does not belong to any pickup group.

**Command Modes**

- Ephone-dn configuration (config-ephone-dn)
- Ephone-dn-template configuration (config-ephone-dn-template)
- Voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was added to ephone-dn-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was added to voice register dn configuration mode for SIP directory numbers.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command allows you to assign an individual directory number to a call-pickup group. Phone users can pick up ringing calls within their own pickup group more easily than calls outside their group.

You can assign each directory number to only one pickup group. There is no limit to the number of directory numbers that can be assigned to a single pickup group, and there is no limit to the number of pickup groups that can be defined in a Cisco Unified CME system.

Pickup group numbers can vary in length, but must have unique leading digits. For example, you cannot define pickup group 17 and pickup group 177 in the same Cisco Unified CME system because a pickup in group 17 will always be triggered before the user can enter the final 7 for group 177. You can, however, define pickup groups 27 and 177 in the same Cisco Unified CME system.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.
Examples

The following examples assign extension 3242 to pickup group 25:

Router(config)# ephone-dn 4
Router(config-ephone-dn)# number 3242
Router(config-ephone-dn)# pickup-group 25

Router(config)# voice register dn 4
Router(config-register-dn)# number 3242
Router(config-register-dn)# pickup-group 25

The following example uses an ephone-dn-template to assign extension 3242 to pickup group 25:

Router(config)# ephone-dn-template 8
Router(config-ephone-dn-template)# pickup-group 25
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 4
Router(config-ephone-dn)# number 3242
Router(config-ephone-dn)# ephone-dn-template 8

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-dn-template (ephone-dn)</td>
<td>Applies a template to an ephone-dn configuration.</td>
</tr>
<tr>
<td>service directed-pickup</td>
<td>Enables Directed Call Pickup and modifies the function of the PickUp and GPickUp soft keys.</td>
</tr>
</tbody>
</table>
pilot

To define the ephone-dn that callers dial to reach a Cisco CallManager Express (Cisco CME) ephone hunt group, use the pilot command in ephone-hunt configuration mode. To remove the pilot number from the ephone Hunt group, use the no form of this command.

`pilot number [secondary number]`

`no pilot number [secondary number]`

**Syntax Description**

| `number` | String of up to 27 characters that represents an E.164 telephone number. Normally the string is composed of digits, but the string may contain alphabetic characters if the number is dialed only by the router, as with an intercom number, or is not intended to be dialed at all. Secondary numbers can contain wildcards in the string. For details, see “Usage Guidelines.” |
| `secondary` | (Optional) Defines the number that follows as an additional pilot number for the ephone Hunt group. |

**Command Default**

No pilot number is defined.

**Command Modes**

Ephone-hunt configuration (config-ephone-hunt)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.3(7)T</td>
<td>Cisco CME 3.1</td>
<td>The <code>secondary</code> secondary-number keyword-argument pair was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command defines a valid number for an ephone-dn (extension) that is to be assigned to an ephone hunt pilot group. The dial-plan pattern can be applied to the pilot number.

The `secondary` keyword allows you to associate a second telephone number with this ephone-dn so that the hunt group can be called by dialing either the main or secondary phone number. The secondary number may contain one or more wildcards instead of digits, even if the wildcard number overlaps the primary number. For example, 50.. (the number 50 followed by periods, which stand for wildcards) matches all four-digit extensions that start with 50. Wildcard characters cannot be used in the primary pilot number.

Alphabetic characters can be used to create a primary or secondary pilot number that cannot be dialed from a phone and is not part of the dial plan.

**Examples**

The following example sets the pilot number to 2345 for peer ephone hunt group number 5:

```
ephone-hunt 5 peer
  pilot 2345
  list 2346, 2347, 2348
  hops 3
  timeout 45
```
The following example sets the pilot number for ephone hunt group 3 to 2222 and the secondary pilot number to 4444:

```plaintext
ephone-hunt 3 sequential
   pilot 2222 secondary 4444
   list 2555, 2556, 2557
   final 6000
```

The following example uses wildcards in the secondary pilot number to create a hunt group that receives the calls made to all numbers that start with 555. The primary pilot number, A0, cannot be dialed.

```plaintext
ephone-hunt 1 longest-idle
   pilot A0 secondary 555....
   list 1000, 1001, 1002
   timeout 5
   hops 3
   final 1100
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-hunt</td>
<td>Enters ephone-hunt configuration mode to define a Cisco CME ephone hunt group.</td>
</tr>
<tr>
<td>final</td>
<td>Defines the last ephone-dn in an ephone hunt group.</td>
</tr>
<tr>
<td>hops</td>
<td>Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.</td>
</tr>
<tr>
<td>list</td>
<td>Lists the ephone-dns that participate in an ephone hunt group.</td>
</tr>
<tr>
<td>max-redirect</td>
<td>Changes the current number of allowable redirects in a Cisco CME system.</td>
</tr>
<tr>
<td>no-reg (ephone-hunt)</td>
<td>Specifies that the pilot number of an ephone hunt group should not register with the H.323 gatekeeper.</td>
</tr>
<tr>
<td>preference (ephone-hunt)</td>
<td>Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.</td>
</tr>
<tr>
<td>timeout (ephone-hunt)</td>
<td>Sets the number of seconds after which a call that is not answered is redirected to the next number in the ephone-hunt-group list.</td>
</tr>
</tbody>
</table>
pilot (voice hunt-group)

To define the number that callers dial to reach a Cisco Unified CME voice hunt group, use the pilot command in voice hunt-group configuration mode. To remove the pilot number from the voice hunt group, use the no form of this command.

```
pilot number [secondary number]
no pilot
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>number</td>
<td>String of up to 32 characters that represents an extension or E.164 telephone number.</td>
</tr>
<tr>
<td>secondary</td>
<td>(Optional) Defines an additional pilot number for the voice hunt group.</td>
</tr>
</tbody>
</table>

**Command Default**

No pilot number is defined.

**Command Modes**

Voice hunt-group configuration (config-voice-hunt-group)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command defines an extension that is assigned as the pilot number of a voice hunt group. The dial-plan pattern can be applied to the pilot number.

Normally the pilot number is composed of digits, but the string may contain alphabetic characters if the number is dialed only by the router, as with an intercom number, or is not intended to be dialed at all.

The `secondary` keyword allows you to associate a second telephone number so that the hunt group can be called by dialing either the primary or secondary phone number. The secondary number can contain one or more wild cards instead of digits, even if the wildcard number overlaps the primary number. For example, 50.. (the number 50 followed by periods, which stand for wild card) matches all four-digit extensions that start with 50. Wildcard characters cannot be used in the primary pilot number.

Alphabetic characters can be used to create a primary or secondary pilot number that cannot be dialed from a phone and is not part of the dial plan.

Voice hunt groups do not support the expansion of pilot numbers using the `dialplan-pattern` command. To enable external phones to dial the pilot number, you must configure a secondary pilot number using a fully qualified E.164 number.

**Examples**

The following example shows how to set the pilot number to 2345 for voice hunt group hunt group number 5:

```
voice-hunt 5 peer
pilot 2345
list 2346, 2347, 2348
hops 3
timeout 45
final 6000
```
The following example shows how to set the pilot number for voice hunt group 3 to 2222 and the secondary pilot number to 4444:

```
voice hunt-group 3 sequential
pilot 2222 secondary 4444
final 6000
```

The following example shows how to use wild cards in the secondary pilot number to create a voice hunt group that receives the calls made to all numbers that start with 55501. The primary pilot number, A0, cannot be dialed.

```
voice hunt-group 1 longest-idle
pilot A0 secondary 55501..
list 1000, 1001, 1002
timeout 5
hops 3
final 1100
```

The following example shows how to use a secondary pilot number in a parallel hunt group. Local phones can dial the primary pilot number, 1100. External phones (PSTN) must dial the full E.164 number, 4085550100.

```
voice hunt-group 4 parallel
final 1109
list 1101,1102,1103,1104
timeout 60
pilot 1100 4085550100
```

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dialplan-pattern</td>
<td>Defines a pattern that is used to expand extension numbers into fully qualified E.164 numbers.</td>
</tr>
<tr>
<td>final (voice hunt-group)</td>
<td>Defines the last extension in a voice hunt group.</td>
</tr>
<tr>
<td>hops (voice hunt-group)</td>
<td>Defines the number of times that a call is redirected to the next directory number in a peer voice hunt-group list before proceeding to the final directory number.</td>
</tr>
<tr>
<td>list (voice hunt-group)</td>
<td>Defines the directory numbers that participate in a hunt group.</td>
</tr>
<tr>
<td>voice hunt-group</td>
<td>Defines the type of hunt group.</td>
</tr>
</tbody>
</table>
pin

To set a personal identification number (PIN) for an IP phone in a Cisco CallManager Express (Cisco CME) system, use the `pin` command in ephone configuration mode. To remove a PIN, use the `no` form of this command.

```
pin  number
no  pin
```

**Syntax Description**

| `number` | PIN that will be used to log in to a Cisco IP phone. This is a numeric string from four to eight digits in length. |

**Command Default**

No PIN is set.

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `pin` command allows individual phone users to override call-blocking patterns that are associated with defined time periods. Call-blocking patterns that are in effect at all times (7 days a week, 24 hours a day) cannot be overridden using a PIN.

Call blocking on IP phones is defined in the following way. First, one or more patterns of outgoing digits to be blocked are defined using the `after-hours block pattern` command. Next, one or more time periods during which calls to those patterns are to be blocked are defined using the `after-hours date` or `after-hours day` command or both. By default, all IP phones in a Cisco CME system are restricted if at least one pattern and at least one time period are defined. Individual phones can be completely exempted from call blocking using the `after-hour exempt` command. An individual with a PIN can override call blocking by entering the PIN after pressing the Login soft key to log in to a phone that has been configured for that PIN using the `pin` command.

The PIN functionality applies only to IP phones that have soft keys, such as the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G.

**Examples**

The following example sets a PIN for an IP phone:

```
Router(config)# ephone 1
Router(config-ephone)# pin 1000
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Specifies that an IP phone does not have any of its outgoing calls blocked even though call blocking has been defined for a Cisco CME system.</td>
</tr>
<tr>
<td><strong>Description</strong></td>
</tr>
<tr>
<td>---</td>
</tr>
<tr>
<td><strong>after-hours block pattern</strong></td>
</tr>
<tr>
<td><strong>after-hours date</strong></td>
</tr>
<tr>
<td><strong>after-hours day</strong></td>
</tr>
<tr>
<td><strong>login</strong></td>
</tr>
<tr>
<td><strong>show ephone login</strong></td>
</tr>
</tbody>
</table>
pin (voice logout-profile and voice user-profile)

To configure a personal identification number (PIN) for accessing a particular IP phone that is enabled for extension mobility, use the `pin` command in voice logout-profile configuration mode or voice user-profile configuration mode. To remove a PIN, use the `no` form of this command.

```plaintext
pin [0|6] number
no pin [0|6] number
```

**Syntax Description**

<table>
<thead>
<tr>
<th></th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>number</td>
<td>Four- to eight-digit numeric string for accessing Cisco Unified IP phone.</td>
</tr>
<tr>
<td>[0</td>
<td>6]</td>
</tr>
</tbody>
</table>

**Command Default**

No PIN is configured.

**Command Modes**

Voice logout-profile configuration (config-logout-profile)

Voice user-profile configuration (config-user-profile)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XW</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2('1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command was enhanced for password encryption.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command in voice logout-profile configuration mode to create a PIN to be used by a phone user to disable the call blocking configuration for a Cisco Unified IP phone on which a logout profile is downloaded.

Use this command in voice user-profile configuration mode to create a PIN to be used by a phone user to disable the call blocking configuration for a Cisco Unified IP phone on which a user profile is downloaded.

PIN functionality applies only to IP phones that have soft keys, such as the Cisco Unified IP Phone 7940 and 7940G.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters [0|6] for this command. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

**Examples**

The following example shows the configuration for a user profile to be downloaded when a phone user logs into a Cisco Unified IP phone that is enabled for extension mobility, including a PIN of 12345:

```plaintext
pin 12345
user me password pass123
number 2001 type silent-ring
```
number 2002 type beep-ring
number 2003 type feature-ring
number 2004 type monitor-ring
number 2005,2006 type overlay
number 2007,2008 type cw-overly
speed-dial 1 3001
speed-dial 2 3002 blf

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>logout-profile</td>
<td>Enable an SCCP phone for Extension Mobility and apply logout profile to phone being configured.</td>
</tr>
<tr>
<td>reset (voice logout-profile and voice user-profile)</td>
<td>Performs complete reboot of all IP phones on which a particular logout-profile or user-profile is downloaded.</td>
</tr>
</tbody>
</table>
**pin (voice register pool)**

To set a personal identification number (PIN) to bypass the after-hour call block on a Cisco Unified SIP IP phone, use the `pin` command in voice register pool configuration mode. To remove the PIN, use the `no` form of the command.

```
pin [0|6] digits
no pin
```

**Syntax Description**

- **digits**
  - PIN to bypass the after-hour call block on the Cisco Unified SIP IP phone. Numeric string from four to eight digits in length.
  - The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

**Command Default**

No valid PIN is set.

**Command Modes**

Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>Unified CME 9.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command was enhanced for password encryption.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `pin` command allows individual Cisco Unified SIP IP phone users to override call-blocking patterns that are associated with defined time periods. Call-blocking patterns that are in effect at all times (7 days a week, 24 hours a day) cannot be overridden using a PIN.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters [0|6] for this command. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

**Examples**

The following example shows how to set a PIN to bypass the after-hour call block on a Cisco Unified SIP IP phone in voice register pool 80:

```
Router(config)# voice register pool 80
Router(config-register-pool)# pin 12345
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode and creates a pool configuration for Cisco Unified SIP IP phones in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
To define the TCP port number on which the CAPF server listens for incoming socket connections, use the `port` command in CAPF-server configuration mode. To use the default, use the `no` form of this command.

```
port tcp-port
no port
```

**Syntax Description**

<table>
<thead>
<tr>
<th>tcp-port</th>
<th>Port for secure communication. Range is from 2000 to 9999. Default is 3804.</th>
</tr>
</thead>
</table>

**Command Default**

TCP port number 3804.

**Command Modes**

CAPF-server configuration (config-capf-server)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication.

**Examples**

The following example specifies TCP port 3000 instead of the default port 3804:

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```
# preemption reserve timer

To set the amount of time to reserve a channel for a preemption call, use the `preemption reserve timer` command in voice MLPP configuration mode. To reset to the default, use the `no` form of this command.

```
preemption reserve timer  seconds
no  preemption reserve timer
```

## Syntax Description

<table>
<thead>
<tr>
<th>Description</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of seconds to reserve the channel. Range: 3 to 30. Default: 0.</td>
<td><code>seconds</code></td>
</tr>
</tbody>
</table>

## Command Default

Preemption reserve timer is disabled (0).

## Command Modes

Voice MLPP configuration (config-voice-mlpp)

## Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

When a channel on an SCCP phone is preempted by a higher priority MLPP call, the channel is reserved for the MLPP call so that other calls cannot use that channel before the call is connected.

## Examples

The following example shows the reserve timer set to 10 seconds.

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# preemption reserve timer 10
```

## Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>preemption enable</td>
<td>Enables preemption capabilities on a trunk group.</td>
</tr>
<tr>
<td>preemption tone timer</td>
<td>Sets the expiry time for the preemption tone for the outgoing call being preempted by a DDR backup call.</td>
</tr>
<tr>
<td>preemption user</td>
<td>Enables phones to preempt calls.</td>
</tr>
</tbody>
</table>
preemption tone timer (voice MLPP)

To set the amount of time the preemption tone plays on the called phone when a lower precedence call is being preempted, use the `preemption tone timer` command in voice MLPP configuration mode. To reset to the default, use the `no` form of this command.

`preemption tone timer seconds`
`no preemption tone timer`

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th><code>seconds</code></th>
<th>Length of preemption tone, in seconds. Range: 3 to 30. Default: 0.</th>
</tr>
</thead>
</table>

**Command Default**

Preemption tone timer is disabled (0).

**Command Modes**

Voice MLPP configuration (config-voice-mlpp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

This command sets how long a phone user hears the preemption tone play when a lower precedence call is being preempted by a higher priority call. The preemption tone stops playing when the timer expires or the user goes on-hook.

For calls to Cisco Unified IP phones, the called party can hang up immediately to connect to the new higher precedence call, or if the called party does not hang up, Cisco Unified CME forces the phone on-hook after the preemption tone timer expires and connects the call.

For FXS ports, the called party must acknowledge the preemption by going on-hook, before being connected to the new higher precedence call.

The `mlpp indication` command must be enabled (default) for a phone to play preemption tones.

**Examples**

The following example shows the tone timer is set to 15 seconds:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# preemption tone timer 15
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>mlpp indication</code></td>
<td>Enables MLPP indication on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td><code>mlpp preemption</code></td>
<td>Enables the preemption capability on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td><code>preemption reserve timer</code></td>
<td>Sets the amount time to reserve a channel for a preemption call.</td>
</tr>
<tr>
<td><code>preemption user</code></td>
<td>Enables the preemption capability for all supported phones.</td>
</tr>
</tbody>
</table>
preemption trunkgroup

To enable preemption capabilities for trunk groups, use the `preemption trunkgroup` command in voice MLPP configuration mode. To disable preemption capabilities, use the `no` form of this command.

```
preemption trunkgroup
no preemption trunkgroup
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Preemption is disabled for trunk groups.

**Command Modes**

Voice MLPP configuration (config-voice-mlpp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

The following example enables preemption capabilities for trunk groups:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# preemption trunkgroup
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>mlpp preemption</td>
<td>Enables calls on an SCCP phone or analog FXS port to be preempted.</td>
</tr>
<tr>
<td>preemption user</td>
<td>Enables phones to preempt calls.</td>
</tr>
</tbody>
</table>
preemption user

To enable phones to preempt calls, use the `preemption user` command in voice MLPP configuration mode. To disable preemption capabilities, use the `no` form of this command.

```
preemption user
no preemption user
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Preemption is disabled for phones.

**Command Modes**

Voice MLPP configuration (config-voice-mlpp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Products</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

This command enables SCCP and analog FXS phones in the system to preempt calls if the called party is busy with lower precedence calls.

**Examples**

The following example enables preemption capabilities for phones:

```
Router(config)# voice mlpp
Router(conf-voi-mlpp)# preemption user
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>mlpp preemption</code></td>
<td>Enables preemption capabilities on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td><code>preemption trunkgroup</code></td>
<td>Enables preemption capabilities on a trunk group.</td>
</tr>
</tbody>
</table>
preference (ephone-dn)

To set dial-peer preference order for an extension (ephone-dn) associated with a Cisco IP phone, use the `preference` command in ephone-dn configuration mode. To reset the preference order to the default, use the `no` form of this command.

```
preference preference-order [secondary secondary-order]
```

```
no preference
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>preference-order</code></td>
</tr>
<tr>
<td><code>secondary secondary-order</code></td>
</tr>
</tbody>
</table>

**Command Default**

Preference order for the primary number is 0 (highest preference). Preference order for the secondary number is 9 (lowest preference).

**Command Modes**

Ephone-dn configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>The <code>secondary secondary-order</code> keyword-argument pair was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

When you create an ephone-dn for an IP phone in a Cisco CallManager Express (Cisco CME) system, you automatically create a virtual voice port and one to four virtual dial peers to be used by that ephone-dn. This command sets a preference value for the primary and secondary numbers that are associated with the ephone-dn that you are creating. The preference values are passed transparently into the dial peer or dial peers created by the ephone-dn. The preference values allow you to control the selection of a desired dial peer when multiple dial peers are matched on the same destination-pattern (target) number value. In this way, the `preference` command can be used to establish a hunt strategy for incoming calls.

The `huntstop` command can be used to prevent further hunting for a dial-peer match when an ephone-dn is busy or does not answer.

**Examples**

The following example sets a preference of 2 for the directory number 3000:

```
ephone-dn 1
  number 3000
  preference 2
```

In the following example, the number 1222 under ephone-dn 4 has a higher preference than the number 1222 under ephone-dn 5.
The following example shows an ephone-dn with two numbers. The primary number has a higher preference than the secondary number.

```plaintext
ephone-dn 6
  number 2233 secondary 2234
  preference 0 secondary 1
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-dn</td>
<td>Enters ephone-dn configuration mode.</td>
</tr>
<tr>
<td>huntstop</td>
<td>Discontinues call hunting behavior for an extension (ephone-dn) or an extension channel.</td>
</tr>
</tbody>
</table>
preference (ephone-hunt)

To set preference order for the ephone-dn associated with an ephone-hunt-group pilot number in Cisco Unified CME, use the preference command in ephone-hunt configuration mode. To delete this preference order, use the no form of this command.

```
preference preference-order [secondary secondary-order]
no preference preference-order [secondary secondary-order]
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
</tr>
</thead>
<tbody>
<tr>
<td>preference-order</td>
</tr>
<tr>
<td>secondary secondary-order</td>
</tr>
</tbody>
</table>

### Command Default

Preference order for the primary number is 0 (highest preference). Preference order for the secondary number is 7 (lowest preference).

### Command Modes

Ephone-hunt configuration (config-ephone-hunt)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.3(7)T</td>
<td>Cisco CME 3.1</td>
<td>The secondary secondary-order keyword-argument pair was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command sets a preference value that is used for matching dial peers in a Cisco IP phone virtual dial-peer group. The preference value is associated with a pilot number for a Cisco CME ephone hunt group. The preference value is passed transparently into the dial peer created by the pilot number. Setting the preference enables the desired dial peer to be selected when multiple dial peers within a hunt group are matched for a dial string.

### Examples

The following example sets the preference for the pilot number of hunt group 23 to 1:

```
Router(config)# ephone-hunt 23 sequential
Router(config-ephone-hunt)# pilot 2355
Router(config-ephone-hunt)# preference 1
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>final</td>
<td>Defines the last ephone-dn in an ephone hunt group.</td>
</tr>
<tr>
<td>hops</td>
<td>Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.</td>
</tr>
</tbody>
</table>
### preference (ephone-hunt)

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>list</td>
<td>Lists the ephone-dns that participate in an ephone hunt group.</td>
</tr>
<tr>
<td>max-redirect</td>
<td>Changes the current number of allowable redirects in an Cisco CME system.</td>
</tr>
<tr>
<td>no-reg (ephone-hunt)</td>
<td>Specifies that the pilot number of an ephone hunt group not register with the H.323 gatekeeper.</td>
</tr>
<tr>
<td>pilot</td>
<td>Defines the ephone-dn that callers dial to reach an ephone hunt group.</td>
</tr>
<tr>
<td>timeout (ephone-hunt)</td>
<td>Sets the number of seconds after which a call that is not answered is redirected to the next number in the ephone-hunt-group list.</td>
</tr>
</tbody>
</table>
preference (voice hunt-group)

To set preference order for the voice dial peer associated with a voice hunt-group pilot number in Cisco Unified CME, use the **preference** command in voice hunt-group configuration mode. To delete this preference order, use the **no** form of this command.

```
preference preference-order [secondary secondary-order]
no preference preference-order [secondary secondary-order]
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>preference-order</strong></td>
<td>Preference order for the extension or telephone number associated with a dial peer. Range is 0 to 8. Default is 0.</td>
</tr>
<tr>
<td><strong>secondary secondary-order</strong></td>
<td>(Optional) Preference order for the secondary pilot number. Range is 1 to 8, where 1 is the highest preference and 8 is the lowest preference. Default is 7.</td>
</tr>
</tbody>
</table>

**Command Default**
Preference for primary number is 0 (highest preference). Preference for secondary number is 7 (lower preference).

**Command Modes**
Voice hunt-group configuration (config-voice-hunt-group)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command sets a preference value that is used for matching dial peers in a Cisco IP phone virtual dial-peer group. The preference value is associated with a pilot number for a Cisco CME voice hunt group. The preference value is passed transparently into the dial peer created by the pilot number. Setting the preference enables the desired dial peer to be selected when multiple dial peers within a hunt group are matched for a dial string.

**Note**
It is recommended that the parallel hunt-group pilot number be unique in the system. Parallel hunt groups may not work if there are more than one partial or exact dial-peer match. For example, this happens if the pilot number is “8000” and there is another dial peer that matches “8...”. If multiple matches cannot be avoided, give call parallel hunt group the highest priority to run by assigning a lower preference to the other dial peers. Note that 8 is the lowest preference value. By default, dial peers created by parallel hunt groups have a preference of 0.

**Examples**
The following is an example of a parallel voice hunt group. The pilot number is 6000 and the preference assigned to the pilot number is 1:

```
voice hunt-group 2 parallel
pilot 6000
preference 1
list 3000, 3010, 3020
final 9999
timeout 10
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>pilot (voice hunt-group)</code></td>
<td>Defines the voice dn that callers dial to reach a Cisco CallManager Express (Cisco CME) voice hunt group.</td>
</tr>
<tr>
<td><code>voice hunt-group</code></td>
<td>Defines the type of hunt group.</td>
</tr>
</tbody>
</table>
**preference (voice register dn)**

To set the dial-peer preference order for VoIP dial peer to be created for a directory number on a SIP phone, use the `preference` command in voice register dn configuration mode. To reset the preference order to the default, use the `no` form of this command.

```
preference preference-order
no preference
```

**Syntax Description**

| `preference-order` | Preference order for the extension or telephone number associated with a directory number. Range is 0 to 10. Default is 0. |

**Command Default**

Preference for primary number is 0 (highest preference).

**Command Modes**

Voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 and Cisco SIP SRST 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

When you create a directory number for a SIP phone in a Cisco CallManager Express (Cisco CME) or Cisco SIP SRST environment, you automatically create a virtual voice port and one to four virtual dial peers to be used by that directory number. This command sets a preference value for the extension or telephone number that is associated with the directory number hat you are creating. The preference value is passed transparently to dial peers created by the directory number. The preference value allows you to control the selection of a desired dial peer when multiple dial peers are matched on the same destination pattern (extension or telephone number). In this way, the `preference` command can be used to establish a hunt strategy for incoming calls. The `huntstop` command can be used to prevent further hunting for a dial-peer match when a number is busy or does not answer.

**Note**

This command can also be used for Cisco SIP SRST.

**Examples**

The following example shows how to set a preference of 2 for extension number 3000:

```
voice register dn 1
number 3000
preference 2
```

In the following example, extension number 1222 under voice register dn 4 has a higher preference than number 1222 under voice register dn 5.

```
voice register dn 4
number 1222
preference 0
!
```

Cisco Unified Communications Manager Express Command Reference
```plaintext
! 
voice register dn 5
number 1222
preference 1
```

## Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>huntstop (voice register dn)</strong></td>
<td>Discontinues call hunting behavior for an extension (directory number) or an extension channel.</td>
</tr>
<tr>
<td><strong>voice register dn</strong></td>
<td>Enters voice register dn configuration mode to define an extension for a SIP phone line.</td>
</tr>
</tbody>
</table>
preference (voice register pool)

To set the preference order for creating the VoIP dial peers created for a number associated with a voice pool, use the `preference` command in voice register pool configuration mode. To put the number in default preference order, use the `no` form of this command.

```
preference preference-order
no preference
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>preference-order</td>
<td>Preference order for the extension or telephone number associated with a pool. Range is 0 to 10. Default is 0, which is the highest preference.</td>
</tr>
</tbody>
</table>

**Command Default**

Preference for primary number is 0 (highest preference order).

**Command Modes**

Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was added to Cisco CallManager Express (Cisco CME).</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

When you create a voice register pool for a SIP phone or a group of SIP phones in a Cisco Unified CME or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) environment, you automatically create a virtual voice port and one to four virtual dial peers to be used by the number associated with that pool. The preference value is passed transparently to dial peers created for the number. The preference value allows you to control the selection of a desired dial peer when multiple dial peers are matched on the same destination pattern (extension or phone number) associated with the pool. In this way, the `preference` command can be used to establish a hunt strategy for incoming calls.

**Note**

Configure the `id` (voice register pool) command before any other voice register pool commands, including the preference command. The id command identifies a locally available individual SIP phone or set of Cisco SIP phones.

**Examples**

The following example shows how to set a preference of 2 for extension number 3000:

```
voice register pool 1
number 3000
preference 2
```

In the following example, extension number 1222 under voice register dn 4 has a higher preference than number 1222 under voice register pool 5.
voice register pool 4
    number 1222
    preference 0
!
voice register dn 5
    number 1222
    preference 1

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>id (voice register pool)</td>
<td>Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.</td>
</tr>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode for SIP phones.</td>
</tr>
</tbody>
</table>
presence

To enable presence service and enter presence configuration mode, use the `presence` command in global configuration mode. To disable presence service, use the `no` form of this command.

```
presence
no presence
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Presence service is disabled.

**Command Modes**
Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command enables the router to perform the following presence functions:

- Process presence requests from internal lines to internal lines. Notify internal subscribers of any status change.
- Process incoming presence requests from a SIP trunk for internal lines. Notify external subscribers of any status change.
- Send presence requests to external presentities on behalf of internal lines. Relay status responses to internal lines.

**Examples**
The following example shows how to enable presence and enter presence configuration mode to set the maximum subscriptions to 150:

```
Router(config) # presence
Router(config-presence) # max-subscription 150
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow watch</td>
<td>Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.</td>
</tr>
<tr>
<td>debug presence</td>
<td>Displays debugging information about the presence service.</td>
</tr>
<tr>
<td>max-subscription</td>
<td>Sets the maximum number of concurrent watch sessions that are allowed.</td>
</tr>
<tr>
<td>presence enable</td>
<td>Allows the router to accept incoming presence requests.</td>
</tr>
<tr>
<td>server</td>
<td>Specifies the IP address of a presence server for sending presence requests from internal watchers to external presence entities.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>show presence global</td>
<td>Displays configuration information about the presence service.</td>
</tr>
<tr>
<td>show presence subscription</td>
<td>Displays information about active presence subscriptions.</td>
</tr>
</tbody>
</table>
presence call-list

To enable Busy Lamp Field (BLF) monitoring for call lists and directories on phones registered to the Cisco Unified CME router, use the `presence call-list` command in ephone, presence, or voice register pool configuration mode. To disable BLF indicators for call lists, use the `no` form of this command.

```
presence call-list
no presence call-list
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
BLF monitoring for call lists is disabled.

**Command Modes**
Ephone configuration (config-ephone)
Presence configuration (config-presence)
Voice register pool configuration (config-register pool)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command enables a phone to monitor the line status of directory numbers listed in a directory or call list, such as a missed calls, placed calls, or received calls list. Using this command in presence mode enables the BLF call-list feature for all phones. To enable the feature for an individual SCCP phone, use this command in ephone configuration mode. To enable the feature for an individual SIP phone, use this command in voice register pool configuration mode.

If this command is disabled globally and enabled in voice register pool or ephone configuration mode, the feature is enabled for that voice register pool or ephone.

If this command is enabled globally, the feature is enabled for all voice register pools and ephones regardless of whether it is enabled or disabled on a specific voice register pool or ephone.

To display a BLF status indicator, the directory number associated with a telephone number or extension must have presence enabled with the `allow watch` command.

For information on the BLF status indicators that display on specific types of phones, see the Cisco Unified IP Phone documentation for your phone model.

**Examples**
The following example shows the BLF call-list feature enabled for ephone 1. The line status of a directory number that appears in a call list or directory is displayed on phone 1 if the directory number has presence enabled.

```
Router(config)# ephone 1
Router(config-ephone)# presence call-list
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>allow watch</strong></td>
<td>Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.</td>
</tr>
<tr>
<td><strong>blf-speed-dial</strong></td>
<td>Enables BLF monitoring for a speed-dial number on a phone registered to Cisco Unified CME.</td>
</tr>
<tr>
<td><strong>presence</strong></td>
<td>Enables presence service and enters presence configuration mode.</td>
</tr>
<tr>
<td><strong>show presence global</strong></td>
<td>Displays configuration information about the presence service.</td>
</tr>
</tbody>
</table>
presence enable

To allow incoming presence requests, use the presence enable command in SIP user-agent configuration mode. To block incoming requests, use the no form of this command.

```
presence enable
no presence enable
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Incoming presence requests are blocked.

**Command Modes**
SIP UA configuration (config-sip-ua)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command allows the router to accept incoming presence requests (SUBSCRIBE messages) from internal watchers and SIP trunks. It does not impact outgoing presence requests.

**Examples**
The following example shows how to allow incoming presence requests:

```
Router(config)# sip-ua
Router(config-sip-ua)# presence enable
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow subscribe</td>
<td>Allows internal watchers to monitor external presence entities (directory numbers).</td>
</tr>
<tr>
<td>allow watch</td>
<td>Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.</td>
</tr>
<tr>
<td>max-subscription</td>
<td>Sets the maximum number of concurrent watch sessions that are allowed.</td>
</tr>
<tr>
<td>show presence global</td>
<td>Displays configuration information about the presence service.</td>
</tr>
<tr>
<td>show presence subscription</td>
<td>Displays information about active presence subscriptions.</td>
</tr>
<tr>
<td>watcher all</td>
<td>Allows external watchers to monitor internal presence entities (directory numbers).</td>
</tr>
</tbody>
</table>
present-call

To present ephone-hunt-group calls only to member phones that are idle or onhook, use the **present-call** command in ephone-hunt configuration mode. To return to the default, use the **no** form of this command.

```
present-call {idle-phone|onhook-phone}
no present-call {idle-phone|onhook-phone}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>idle-phone</strong></td>
<td>Presents calls from the ephone-hunt group only if all lines are idle on the phone on which the hunt-group line appears. This option does not consider monitored lines that have been configured on the phone using the <strong>button m</strong> command.</td>
</tr>
<tr>
<td><strong>onhook-phone</strong></td>
<td>Presents calls from the ephone-hunt group only if the phone on which the number appears is in the onhook state. When this keyword is configured, calls in the ringing or hold state that are unrelated to the hunt group do not prevent the presentation of calls from the ephone-hunt group.</td>
</tr>
</tbody>
</table>

**Command Default**

Ephone hunt group calls are presented to lines (ephone-dn) that are not in use, regardless of the state of other lines on the same ephone.

**Command Modes**

Ephone-hunt configuration (config-ephone-hunt)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

If you do not use this command, an ephone hunt group presents calls to an ephone whenever the phone line (ephone-dn) that corresponds to a number in an ephone-hunt list is available. The status of other phone lines on the phone is not considered.

The **present-call** command adds additional controls that allow you to take into account the activity on all lines of a phone that has an ephone-dn that is assigned to an ephone hunt group. The **present-call** command allows you to specify that hunt groups should present calls to these phones only when they are on hook or are not busy with an active call. This keeps hunt group calls from possibly going unanswered because a phone is occupied with a call on a line other than the line assigned to the hunt group.

**Examples**

The following example sets up a peer hunt group with three ephone-dns to answer calls. Incoming calls are sent only to ephone-dns on phones that are on-hook.

```plaintext
ephone-hunt 17 peer
pilot 3000
list 3011, 3021, 3031
hops 3
final 7600
present-call onhook-phone
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ephone-hunt</code></td>
<td>Defines an ephone hunt group and enters ephone-hunt configuration mode.</td>
</tr>
</tbody>
</table>
present-call (voice hunt-group)

To present voice hunt-group calls only to member phones that are idle, use the `present-call` command in voice hunt group configuration mode. To return to the default, use the `no` form of this command.

```
present-call {idle-phone}
no present-call {idle-phone}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>idle-phone</th>
<th>Presents calls from the voice hunt group only if all lines are idle on the phone on which the hunt group line appears.</th>
</tr>
</thead>
</table>

**Command Default**

Voice hunt group calls are presented to lines (ephone-dns or voice register dns) that are not in use, regardless of the state of other lines on the same ephone or voice register pool.

**Command Modes**

voice hunt-group configuration (config-voice-hunt)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IOS XE Everest 16.4.1</td>
<td>Cisco Unified CME 11.6</td>
<td>This command was introduced for voice hunt group.</td>
</tr>
<tr>
<td>15.6(3)M1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

If you do not use this command, voice hunt group presents calls to an ephone or voice register pool whenever the phone line (ephone-dn or voice register dn) that corresponds to a number in a voice hunt group list is available. The status of other phone lines on the phone is not considered.

The `present-call` command adds additional controls that allow you to take into account the activity on all lines of a phone that has an ephone-dn or voice register dn that is assigned to a voice hunt group. The `present-call` command allows you to specify that hunt groups should present calls to these phones only when they are idle or not busy with an active call. This keeps hunt group calls from possibly going unanswered because a phone is occupied with a call on a line other than the line assigned to the hunt group.

**Examples**

The following example sets up a peer hunt group with three ephone-dns to answer calls. Incoming calls are sent only to ephone-dns or voice register dns on phones that are idle.

```
voice hunt-group 17 peer
pilot 3000
list 3011, 3021, 3031
final 7600
present-call idle-phone
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Defines a voice hunt group and enters voice hunt-group configuration mode.</td>
</tr>
</tbody>
</table>

voice hunt-group
privacy (ephone)

To modify privacy support on a specific phone, use the privacy command in ephone or ephone-template configuration mode. To reset to the default value, use the no form of this command.

```plaintext
privacy [{off|on}]
no privacy
```

**Syntax Description**
- **off** (Optional) Disables privacy on the phone.
- **on** (Optional) Enables privacy on the phone.

**Command Default**
Use system-level setting configured with the privacy command in telephony-service mode.

**Command Modes**
- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command modifies the privacy capability of individual phones. Privacy prevents other phone users from seeing call information or barging into a call on a shared octo-line directory number. Privacy is supported for calls on shared octo-line directory numbers only.

If only specific phones require access to privacy, disable privacy at the system-level by using the no privacy command in telephony-service configuration mode and enable privacy at the phone-level by using the privacy on command.

After a phone that is configured for privacy registers with Cisco Unified CME, the feature button on the phone is labeled “Privacy” and a status icon displays. If the button has a lamp, it lights. When the phone receives an incoming call, the user can make the call private by pressing the Privacy feature button. The privacy button toggles between on and off. The privacy state is applied to new calls and current calls that the user owns.

Users can dynamically enable privacy for shared-line calls by pressing the Privacy feature button on the phone if the privacy-button command is enabled.

The Privacy feature applies to all shared lines on a phone. If a phone has multiple shared lines and Privacy is enabled, other phones cannot view or barge into calls on any of the shared lines.

If you use an ephone template to apply a command to an ephone and you also use the same command in ephone configuration mode for the same ephone, the value that you set in ephone configuration mode has priority.

**Examples**
The following example shows privacy enabled on a specific phone and disabled at the system-level:

```plaintext
telephony-service
no privacy
```
privacy-on-hold
max-ephones 100
max-dn 240
!
ephone 10
privacy on
privacy-button
max-calls-per-button 3
busy-trigger-per-button 2
mac-address 00E1.CB13.0395
type 7960
button 1:7 2:10

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>privacy (telephony-service)</td>
<td>Enables privacy globally for all phones in the system.</td>
</tr>
<tr>
<td>privacy-button</td>
<td>Enables the privacy feature button on an IP phone.</td>
</tr>
<tr>
<td>privacy-on-hold</td>
<td>Enables privacy for calls that are on hold on shared octo-line directory numbers.</td>
</tr>
</tbody>
</table>
privacy (telephony-service)

To enable privacy at the system level for all phones, use the `privacy` command in telephony-service configuration mode. To disable privacy at the system level, use the `no privacy` form of this command.

```plaintext
privacy
no privacy
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Privacy is enabled at the system level for all phones.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables privacy for all phones in the system. Privacy prevents other phone users from seeing call information or joining a call on a shared octo-line directory number. Privacy is supported for calls on shared octo-line directory numbers only.

If only specific phones need access to privacy, disable privacy at the system-level by using the `no privacy` command and enable privacy at the phone level by using the `privacy on` command in ephone or ephone-template configuration mode.

After a phone that is configured for privacy registers with Cisco Unified CME, the feature button on the phone is labeled “Privacy” and a status icon displays. The button lamp, if available, lights to reflect the privacy setting of the phone. When the phone receives an incoming call, the user can make the call private by pressing the Privacy feature button.

**Examples**

The following example shows privacy disabled at the system-level and enabled on an individual phone:

```plaintext
telephony-service
no privacy
privacy-on-hold
max-ephones 100
max-dn 240
timeouts transfer-recall 60
voicemail 8900
max-conferences 8 gain -6
transfer-system full-consult
fac standard
!
!
ephone 10
privacy on
privacy-button
max-calls-per-button 3
```
busy-trigger-per-button 2
mac-address 00E1.CB13.0395
type 7960
button 1:7 2:10

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>privacy (ephone)</td>
<td>Modifies privacy support on a specific phone.</td>
</tr>
<tr>
<td>privacy-button</td>
<td>Enables the privacy feature button on an IP phone.</td>
</tr>
<tr>
<td>privacy-on-hold</td>
<td>Enables privacy for calls that are on hold on shared octo-line directory numbers.</td>
</tr>
</tbody>
</table>
privacy (voice register global)

To enable privacy at the system level for all SIP phones, use the `privacy` command in voice register global configuration mode. To disable privacy at the system level, use the `no` form of this command.

```
privacy
no privacy
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Privacy is enabled at the system level for all phones.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables privacy for all phones in the system. Privacy prevents other phone users from seeing call information or joining a call on a shared-line directory number. Privacy is supported for calls on shared-line directory numbers only.

If only specific phones need access to privacy, disable privacy at the system-level by using the `no privacy` command and enable privacy at the phone level by using the `privacy on` command in voice register pool or voice register template configuration mode.

After a phone that is configured with the `privacy-button` command registers with Cisco Unified CME, the feature button on the phone is labeled “Privacy” and a status icon displays. The button lamp, if available, lights to reflect the privacy setting of the phone. When the phone receives an incoming call, the user can make the call private by pressing the Privacy feature button.

**Examples**

The following example shows privacy disabled at the system-level and enabled on an individual phone:

```
voice register global
mode cme
privacy-on-hold
no privacy
max-dn 300
max-pool 150
voicemail 8900
call-feature-uri pickup http://10.4.212.11/pickup
call-feature-uri gpickup http://10.4.212.11/gpickup
!
voice register pool 130
id mac 001A.A11B.500E
type 7941
number 1 dn 30
template 6
```
**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>privacy (voice register pool)</strong></td>
<td>Modifies privacy support on a specific phone.</td>
</tr>
<tr>
<td><strong>privacy-button</strong></td>
<td>Enables the privacy feature button on an IP phone.</td>
</tr>
<tr>
<td><strong>privacy-on-hold</strong></td>
<td>Enables privacy for calls that are on hold on shared-line directory numbers.</td>
</tr>
<tr>
<td><strong>shared-line</strong></td>
<td>Creates a shared-line directory number for a SIP phone.</td>
</tr>
</tbody>
</table>
privacy (voice register pool)

To modify the phone-level privacy setting on a SIP phone, use the privacy command in voice register pool or voice register template configuration mode. To reset to the default value, use the no form of this command.

```
privacy {off|on}
no privacy
```

**Syntax Description**

<table>
<thead>
<tr>
<th>off</th>
<th>Disables privacy on the phone.</th>
</tr>
</thead>
<tbody>
<tr>
<td>on</td>
<td>Enables privacy on the phone.</td>
</tr>
</tbody>
</table>

**Command Default**

Use system-level setting configured with the privacy command in voice register global mode.

**Command Modes**

Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command modifies the privacy setting on the SIP phone. Privacy prevents other phone users from viewing call information or barging into a call on a shared-line directory number. Privacy is supported for calls on shared-line directory numbers only.

After a phone that is configured with the privacy-button command registers with Cisco Unified CME, the feature button on the phone is labeled “Privacy” and a status icon displays. If the button has a lamp, it lights. When the phone receives an incoming call, the user can make the call private by pressing the Privacy feature button. The privacy button toggles between on and off. The privacy state applies to new calls and current calls that the phone user owns.

The off and on keywords specify the initial Privacy state on the phone when the Privacy feature is enabled. The phone user can then toggle the privacy state on and off using the Privacy feature button.

The Privacy state applies to all shared lines on a phone. If a phone has multiple shared lines, other phones cannot view or barge into calls on any of the shared lines if the Privacy state is enabled.

If you use a voice register template to apply a command to a phone and you also use the same command in voice register pool configuration mode for the same phone, the value that you set in voice register pool configuration mode has priority.

**Examples**

The following example shows privacy enabled for a specific SIP phone:

```
Router(config) # voice register pool 123
Router(config-register-pool)# privacy on
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>privacy (voice register global)</td>
<td>Enables privacy at the system level for all SIP phones.</td>
</tr>
<tr>
<td>privacy-button</td>
<td>Enables the privacy feature button on an IP phone.</td>
</tr>
<tr>
<td>privacy-on-hold</td>
<td>Enables privacy for calls that are on hold on shared-line directory numbers.</td>
</tr>
<tr>
<td>shared-line</td>
<td>Creates a shared-line directory number for a SIP phone.</td>
</tr>
<tr>
<td>softkeys remote-in-use (voice register template)</td>
<td>Modifies the soft-key display during the remote-in-use call state on SIP phones.</td>
</tr>
<tr>
<td>template (voice register pool)</td>
<td>Applies a template to a SIP phone.</td>
</tr>
</tbody>
</table>
privacy-button

To enable the privacy feature button on an IP phone, use the `privacy-button` command in ephone, ephone-template, voice logout-profile, and voice user-profile configuration mode. To reset to the default value, use the `no` form of this command.

```bash
privacy-button
no privacy-button
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Privacy button is disabled.

**Command Modes**

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)
Voice logout-profile configuration (configlogout-profile)
Voice user-profile configuration (config-user-profile)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command allows phone users to dynamically enable or disable privacy for calls on shared octo-lines by pressing the Privacy feature button on the phone. Privacy prevents other phone users from viewing call information or joining calls on a shared octo-line directory number.

Privacy is supported only for calls on shared octo-line directory numbers so enable this command only on phones that share an octo-line directory number.

After a phone that is configured for privacy registers with Cisco Unified CME, the feature button on the phone is labeled “Privacy” and a status icon displays. The button lamp, if available, lights to reflect the privacy setting of the phone. The privacy feature button toggles between on and off. The privacy state is applied to new calls and current calls owned by the user.

Privacy is enabled on the phone with either the `privacy` command in ephone configuration mode or the `privacy` command in telephony-service mode.

If you use an ephone template to apply a command to an ephone and you also use the same command in ephone configuration mode for the same ephone, the value that you set in ephone configuration mode has priority.

**Examples**

The following example shows the privacy button is enabled for ephone 10:

```bash
ephone 10
privacy-button
max-calls-per-button 3
busy-trigger-per-button 2
mac-address 00E1.CB13.0395
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>privacy (ephone)</td>
<td>Modifies privacy support on a specific phone.</td>
</tr>
<tr>
<td>privacy (telephony-service)</td>
<td>Enables privacy globally for all phones in the system.</td>
</tr>
<tr>
<td>privacy-on-hold</td>
<td>Enables privacy for calls that are on hold on shared octo-line directory numbers.</td>
</tr>
</tbody>
</table>
privacy-button (voice register pool)

To enable the privacy feature button on an IP phone, use the `privacy-button` command in voice register pool or voice register template configuration mode. To reset to the default value, use the `no` form of this command.

```
privacy-button
no privacy-button
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Privacy button is disabled.

**Command Modes**

Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
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</thead>
<tbody>
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<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command allows phone users to dynamically enable or disable privacy for calls on shared lines by pressing the Privacy feature button on the phone. Privacy prevents other phone users from viewing call information or joining calls on a shared-line directory number.

Privacy is supported only for calls on shared-line directory numbers so enable this command only on phones that use a shared-line directory number.

After a phone that is configured with this command registers with Cisco Unified CME, the feature button on the phone is labeled “Privacy” and a status icon displays. The button lamp, if available, lights to reflect the privacy setting of the phone. The privacy feature button toggles between on and off. The privacy state is applied to new calls and current calls owned by the user.

Privacy is enabled on the phone with either the `privacy` command in voice register pool configuration mode or the `privacy` command in voice register global configuration mode.

If you use a voice register template to apply a command to a phone and you also use the same command in voice register pool configuration mode for the same phone, the value that you set in voice register pool configuration mode has priority.

**Examples**

The following example shows the privacy button is enabled for phone 124:

```
voice register pool 124
busy-trigger-per-button 5
id mac 0017.E033.0284
type 7965
number 1 dn 24
privacy-button
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>privacy (voice register global)</td>
<td>Enables privacy globally for all SIP phones in the system.</td>
</tr>
<tr>
<td>privacy (voice register pool)</td>
<td>Modifies privacy support on a specific phone.</td>
</tr>
<tr>
<td>privacy-on-hold (voice register global)</td>
<td>Enables privacy for calls that are on hold on shared-line directory numbers.</td>
</tr>
<tr>
<td>shared-line</td>
<td>Creates a shared-line directory number for a SIP phone.</td>
</tr>
</tbody>
</table>
privacy-on-hold

To enable privacy for calls that are on hold on shared octo-line directory numbers, use the `privacy-on-hold` command in telephony-service configuration mode. To disable privacy for calls on hold, use the `no` form of this command.

```
privacy-on-hold
no privacy-on-hold
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Privacy on hold is disabled.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
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<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command prevents other phone users from seeing or retrieving calls that are on hold on a shared octo-line directory number.

Privacy is enabled on the phone with either the `privacy` command in ephone configuration mode or the `privacy` command in telephony-service mode.

**Examples**

The following example shows how to enable privacy on hold for shared lines.

```
Router(config)# telephony-service
Router(config-telephony)# privacy-on-hold
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>privacy (ephone)</td>
<td>Modifies privacy support on a specific phone.</td>
</tr>
<tr>
<td>privacy (telephony-service)</td>
<td>Enables privacy globally for all phones in the system.</td>
</tr>
<tr>
<td>privacy-button</td>
<td>Enables the privacy feature button on an IP phone.</td>
</tr>
</tbody>
</table>
privacy-on-hold (voice register global)

To enable privacy for calls that are on hold on shared-line directory numbers, use the `privacy-on-hold` command in voice register global configuration mode. To disable privacy for calls on hold, use the `no` form of this command.

```
privacy-on-hold
no privacy-on-hold
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Privacy on hold is disabled.

**Command Modes**

Voice register global (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command prevents other phone users from seeing or retrieving calls that are on hold on a shared-line directory number.

Privacy is enabled on the phone with either the `privacy` command in voice register pool configuration mode or the `privacy` command in voice register global configuration mode.

**Examples**

The following example shows how to enable privacy on hold for shared lines.

```
Router(config)# voice register global
Router(config-register-global)# privacy-on-hold
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>privacy (voice register global)</code></td>
<td>Enables privacy globally for all phones in the system.</td>
</tr>
<tr>
<td><code>privacy (voice register pool)</code></td>
<td>Modifies privacy support on a specific phone.</td>
</tr>
<tr>
<td><code>privacy-button (voice register pool)</code></td>
<td>Enables the privacy feature button on an IP phone.</td>
</tr>
<tr>
<td><code>shared-line</code></td>
<td>Creates a shared-line directory number for a SIP phone.</td>
</tr>
</tbody>
</table>
To configure the Cisco IOS Session Initiation Protocol (SIP) stack, use the `protocol mode` command in SIP user-agent configuration mode. To disable the configuration, use the `no` form of this command.

**protocol mode {ipv4|ipv6|dual-stack [preference {ipv4|ipv6}]}
no protocol mode**

**Syntax Description**
- **ipv4**: Specifies the IPv4-only mode.
- **ipv6**: Specifies the IPv6-only mode.
- **dual-stack**: Specifies the dual-stack (that is, IPv4 and IPv6) mode.
- **preference {ipv4|ipv6}**: (Optional) Specifies the preferred dual-stack mode, which can be either IPv4 (the default preferred dual-stack mode) or IPv6.

**Command Default**
No protocol mode is configured. The Cisco IOS SIP stack operates in IPv4 mode when the `no protocol mode` or `protocol mode ipv4` command is configured.

**Command Modes**
SIP user-agent configuration (config-sip-ua)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)T</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
The `protocol mode` command is used to configure the Cisco IOS SIP stack in IPv4-only, IPv6-only, or dual-stack mode. For dual-stack mode, the user can (optionally) configure the preferred family, IPv4 or IPv6.

For a particular mode (for example, IPv6-only), the user can configure any address (for example, both IPv4 and IPv6 addresses) and the system will not hide or restrict any commands on the router. SIP chooses the right address for communication based on the configured mode on a per-call basis.

For example, if the domain name system (DNS) reply has both IPv4 and IPv6 addresses and the configured mode is IPv6-only (or IPv4-only), the system discards all IPv4 (or IPv6) addresses and tries the IPv6 (or IPv4) addresses in the order they were received in the DNS reply. If the configured mode is dual-stack, the system first tries the addresses of the preferred family in the order they were received in the DNS reply. If all of the addresses fail, the system tries addresses of the other family.

**Examples**
The following example configures dual-stack as the protocol mode:

```snippets
table
```

```
Router(config-sip-ua)# protocol mode dual-stack
```

The following example configures IPv6 only as the protocol mode:

```snippets
table
```

```
Router(config-sip-ua)# protocol mode ipv6
```

The following example configures IPv4 only as the protocol mode:
Router(config-sip-ua)# protocol mode ipv4
The following example configures no protocol mode:

Router(config-sip-ua)# no protocol mode

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>sip ua</td>
<td>Enters SIP user-agent configuration mode.</td>
</tr>
</tbody>
</table>
**protocol-mode (telephony-service)**

To configure a preferred IP address mode for SCCP IP phones in Cisco Unified CM Express, use the `protocol mode` command in telephony service configuration mode. To disable the router protocol mode, use the `no` form of this command.

```
protocol mode {ipv4|ipv6|dual-stack [preference {ipv4|ipv6}]}
no protocol mode {ipv4|ipv6|dual-stack [preference {ipv4|ipv6}]}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ipv4</code></td>
<td>IPv4-only mode.</td>
</tr>
<tr>
<td><code>ipv6</code></td>
<td>IPv6-only mode.</td>
</tr>
<tr>
<td><code>dual-stack</code></td>
<td>Dual-stack mode, ipv4 and ipv6 mode.</td>
</tr>
<tr>
<td><code>preference</code></td>
<td>Preference dual-stack mode, either ipv4 or ipv6 mode.</td>
</tr>
</tbody>
</table>

**Command Default**

No protocol mode is configured.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `protocol mode` command is used to configure SCCP IP phones in CUCME in IPv4-only, IPv6-only, or dual-stack mode. For dual-stack mode, the user can configure the preferred family, IPv4 or IPv6.

For a specific mode, the user is free to configure any address and the system will not hide or restrict any commands on the router. On a per-call basis, SCCCP phones choose the right address for communication based on the configured mode.

For example, if the DNS reply has both IPv4 and IPv6 addresses and the configured mode is IPv6-only (or IPv4-only), the system discards all IPv4 (or IPv6) addresses and tries the IPv6 (or IPv4) addresses in the order they were received in the DNS reply. If the configured mode is dual-stack, the system first tries the addresses of the preferred family in the order they were received in the DNS reply. If all of the addresses fail, the system tries addresses of the other family.

**Examples**

The following example configures dual-stack as the protocol mode:

```
Router(config)# telephony-service
Router(config-telephony)# protocol mode dual-stack preference ?
 ipv4  IPv4 address is preferred
 ipv6  IPv6 address is preferred
```

The following example configures IPv6-only mode as the protocol mode:

```
Router(config)# telephony-service
Router(config-telephony)# protocol mode ipv6
```

The following example configures IPv4-only mode as the protocol mode:
Router(config)# telephony-service
Router(config-telephony)# protocol mode ipv6

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>ip source-address</td>
<td>Identifies the IP address and port through which IP phones communicate with a Cisco Unified CME router.</td>
</tr>
<tr>
<td></td>
<td>shutdown</td>
<td>Allows to shut down SCCP server listening sockets.</td>
</tr>
</tbody>
</table>
provision-tag

To create a provision tag for identifying an ephone or voice register pool for the extension assigner application, use the `provision-tag` command in ephone configuration mode or voice register pool configuration mode. To remove the provision tag, use the `no` form of this command.

```
provision-tag tag
no provision-tag tag
```

**Syntax Description**

<table>
<thead>
<tr>
<th>tag</th>
<th>Unique number that identifies this provision tag. Range: 1 to 2147483647.</th>
</tr>
</thead>
</table>

**Command Default**

No provision tag is created.

**Command Modes**

- Ephone configuration (config-ephone)
- Voice register Pool (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC4</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Everest 16.4.1</td>
<td>Cisco Unified CME 11.6</td>
<td>This command was supported under voice register pool for SIP phones.</td>
</tr>
<tr>
<td>15.6(3)M1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command creates a provision tag.

For SCCP phones, a provision tag enables you to use some number other than an ephone tag, such as a jack number or an extension number, to identify an ephone configuration. The provision tag can be used with the extension assigner application to assign the corresponding ephone configuration to an IP phone. This command is ignored unless you also use the `extension-assigner tag-type` command with the `provision-tag` keyword.

**Examples**

The following example shows that provision tag 1001 is configured for ephone 1 and provision tag 1002 is configured for ephone 2:

```
Telephony-service
  extension-assigner tag-type provision-tag
  auto assign 101-102
  auto-reg-ephone
Ephone-dn 101
  number 1001
Ephone-dn 102
  number 1002
Ephone 1
  provision-tag 1001
  mac-address 02EA.EAEA.0001
```
For SIP phones, a provision tag enables you to assign any number within the range as an extension number. The provision tag is used with an extension assigner application to assign the corresponding voice register pool configuration to an IP phone.

The following example shows that provision tag 1001 is configured for voice register pool 1 and provision tag 1002 is configured for voice register pool 2:

```
Voice register global
  auto-register
  password xxxx
  auto assign 101-102
  voice register dn 101
    number 1001
  voice register dn 102
    number 1002
  voice register pool 1
    provision-tag 1001
    mac-address 02EA.EAEA.0001
    number 1 dn 101
  voice register pool 2
    provision-tag 1002
    mac-address 02EA.EAEA.0002
    number 2 dn 102
```
Cisco Unified CME Commands: R

- refer target dial-peer, on page 888
- refer-ood enable, on page 889
- reference-pooltype, on page 890
- regenerate (ctl-client), on page 891
- register-id, on page 892
- registrar server (SIP), on page 893
- reset (ephone), on page 895
- reset (telephony-service), on page 896
- reset (voice logout-profile and voice user-profile), on page 899
- reset (voice register global), on page 900
- reset (voice register pool), on page 901
- reset (voice-gateway), on page 902
- reset tapi, on page 903
- restart (ephone), on page 904
- restart (telephony-service), on page 905
- restart (voice register), on page 907
- restart (voice-gateway), on page 909
- ring (ephone-dn), on page 910
- route-code, on page 912
- rule (voice translation-rule), on page 913
**refer target dial-peer**

To populate the Refer To portion of a SIP Refer message with the address from the dial peer for the directory number being configured, use the `refer target dial-peer` command in voice register dn configuration mode. To return to the default, use the `no` form of this command.

```
refer target dial-peer
no refer target
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Call is transferred to the destination as specified in the SIP Refer message.

**Command Modes**

Voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XW2</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command in voice register dn configuration mode to specify that the destination address for this directory number be the dial peer. If this command is not configured, Cisco IOS software will transfer the call to the destination in the SIP Refer message and if that destination address is Cisco Unified CME, call SIP will send out and route back to CME before sending to the directory number, creating two extra call legs.

The following partial output from the `show working-configuration` command shows the configuration for three directory numbers. This configuration will populate the Refer To portion of the SIP Refer message with the address from the dial peer for each of the directory numbers.

```
voice register dn 1
  session-server 1
  number 8999
  allow watch
  refer target dial-peer
!
voice register dn 2
  session-server 1
  number 8001
  allow watch
  refer target dial-peer
!
voice register dn 3
  session-server 1
  number 8101
  allow watch
  refer target dial-peer
```
refer-ood enable

To enable out-of-dialog refer (OOD-R) processing, use the `refer-ood enable` command SIP user-agent configuration mode. To disable OOD-R, use the `no` form of this command.

```
refer-ood enable [request-limit]
no refer-ood enable
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>request-limit</td>
<td>(Optional) Maximum number of concurrent incoming OOD-R requests that the router can process. Range: 1 to 500. Default: 500.</td>
</tr>
</tbody>
</table>

**Command Default**

OOD-R processing is disabled.

**Command Modes**

SIP UA configuration (config-sip-ua)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Out of dialog Refer allows applications to establish calls using the SIP gateway or Cisco Unified CME. The application sets up the call and the user does not dial out from their own phone.

**Examples**

The following example shows how to enable OOD-R:

```
Router(config)# sip-ua
Router(config-sip-ua)# refer-ood enable
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
<th>Syntax</th>
</tr>
</thead>
<tbody>
<tr>
<td>Defines the authenticate mode for SIP phones in a Cisco Unified CME or Cisco Unified SRST system.</td>
<td>authenticate (voice register global)</td>
</tr>
<tr>
<td>Reloads a credential file into flash memory.</td>
<td>credential load</td>
</tr>
<tr>
<td>Displays all application debug messages.</td>
<td>debug voip application</td>
</tr>
</tbody>
</table>
reference-pooltype

To inherit the properties from the nearest supported phone model for a Cisco Unified SIP IP phone on Cisco Unified CME, use the `reference-pooltype` command in voice register pooltype mode. To remove the pooltype configuration, use the `no` form of this command.

```
reference-pooltype phone-type
noreference-pooltype phone-type
```

**Syntax Description**

- `phone-type` Unique number that represents the phone model.

**Command Default**

There is no reference phone to inherit the properties. This is the only command which has the `no` form as the default form.

**Command Modes**

Voice Register Pool Configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(3)M</td>
<td>Cisco SIP CME 10.0</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to inherit the properties from the nearest supported phone model for a Cisco Unified SIP IP phone on Cisco Unified CME.

**Example**

The following example shows how to enter voice register pool configuration mode and inherit the properties from the nearest supported phone model SIP phones on a Cisco Unified CME system:

```
Router# configure terminal
Router (config)# voice register pool-type 9900
Router (config--register-pool-type)# reference-pooltype 9971
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register pool-type</td>
<td>Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.</td>
</tr>
</tbody>
</table>
regenerate (ctl-client)

To create a new CTLFile.tlv file after making changes to the CTL client configuration, use the `regenerate` command in CTL-client configuration mode. The `no` form of this command has no effect in the configuration.

```
regenerate
no regenerate
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

A new CTLFile.tlv file is not created until this command is used.

**Command Modes**

CTL-client configuration (config-ctl-client)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication.

**Examples**

The following example gives the instruction to regenerate the CTL file with the current information.

```
Router(config)# ctl-client
Router(config-ctl-client)# server capf 10.2.2.2 trustpoint capftrust
Router(config-ctl-client)# server cmce 10.2.2.3 trustpoint cmetp
Router(config-ctl-client)# server tftp 10.2.2.4 trustpoint tftptp
Router(config-ctl-client)# sast1 trustpoint sast1tp
Router(config-ctl-client)# sast2 trustpoint sast2tp
Router(config-ctl-client)# regenerate
```
**register-id**

To create an ID for explicitly identifying an external feature server during Register requests, use the `register-id` command in voice register session-server configuration mode. To remove an ID, use the `no` form of this command.

```
register-id name
no register-id name
```

**Syntax Description**

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>name</strong></td>
<td>String of up to 30 alphanumeric characters.</td>
</tr>
</tbody>
</table>

**Command Default**

No identifier is created.

**Command Modes**

Voice register session-server configuration (config-register-fs)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XW2</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to create an ID for identifying a route point during Register requests. Cisco Unified CME challenges and authenticates the initial keepalive Register request and issues a system-wide unique Cisco-referenceID to be included in the response to the Register request from this route point.

**Examples**

The following partial output shows the configuration of a session manager for an external feature server, including the register ID of CSR1:

```
router# show running-configuration

!
!
voice register session-server 1
register-id CSR1
keepalive 360
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>keepalive</td>
<td>Duration for registration after which the registration expires unless the feature server reregisters before the registration expiry.</td>
</tr>
</tbody>
</table>
registrar server (SIP)

To enable SIP registrar functionality, use the registrar server command in SIP configuration mode. To disable SIP registrar functionality, use the no form of the command.

```
registrar server [expires [max sec] [min sec]]
no registrar server
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Argument</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>expires</td>
<td>(Optional) Sets the active time for an incoming registration.</td>
</tr>
<tr>
<td>max sec</td>
<td>(Optional) Maximum expires time for a registration, in seconds. The range is from 600 to 86400. The default is 3600.</td>
</tr>
<tr>
<td>min sec</td>
<td>(Optional) Minimum expires time for a registration, in seconds. The range is from 60 to 3600. The default is 60.</td>
</tr>
</tbody>
</table>

**Command Default**

SIP registrar functionality on the Cisco Unified CME router id disabled.

**Command Modes**

SIP configuration (config-voi-sip)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 and Cisco SIP SRST 3.4</td>
<td>This command was added to Cisco CME.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

When this command is entered, the router accepts incoming SIP Register messages. If SIP Register message requests are for a shorter expiration time than what is set with this command, the SIP Register message expiration time is used.

This command is mandatory for Cisco Unified SIP SRST or Cisco Unified CME and must be entered before any voice register pool or voice register global commands are configured.

If the WAN is down and you reboot your Cisco Unified CME or Cisco Unified SIP SRST router, when the router reloads it will have no database of SIP phone registrations. The SIP phones will have to register again, which could take several minutes, because SIP phones do not use a keepalive functionality. To shorten the time before the phones re-register, the registration expiry can be adjusted with this command. The default expiry is 3600 seconds; an expiry of 600 seconds is recommended.

**Examples**

The following partial sample output from the show running-config command shows that SIP registrar functionality is set:

```
voice service voip
allow-connections sip-to-sip
sip
```
**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>sip</td>
<td>Enters SIP configuration mode from voice service VoIP configuration mode.</td>
</tr>
<tr>
<td>voice register global</td>
<td>Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco Unified CME or Cisco Unified SIP SRST environment.</td>
</tr>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode for SIP phones.</td>
</tr>
</tbody>
</table>
**reset (ephone)**

To perform a complete reboot of a single phone associated with a Cisco CallManager Express (Cisco CME) router, use the `reset` command in ephone configuration mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

No reset is performed.

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

After you update information for one or more phones associated with a Cisco CME router, the phone or phones must be rebooted. There are two commands to reboot the phones: `reset` and `restart`. The `reset` command performs a “hard” reboot similar to a power-off-power-on sequence. It reboots the phone and contacts the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server to update from their information as well. The `restart` command performs a “soft” reboot by simply rebooting the phone without contacting the DHCP and TFTP servers. The `reset` command takes significantly longer to process than the `restart` command when you are updating multiple phones, but it must be used after updating phone firmware, user locale, network locale, or URL parameters. For simple button, line, or speed-dial changes, you can use the `restart` command.

Use the `reset (ephone)` command to perform a complete reboot of an IP phone when you are in ephone configuration mode. This command has the same effect as a `reset (telephony-service)` command that is used to reset a single phone.

This command has a no form, but the no form has no effect.

**Examples**

The following example resets the SCCP phone with a phone-tag of 1:

```
Router(config)# ephone 1
Router(config-ephone)# reset
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
<th>Command</th>
</tr>
</thead>
<tbody>
<tr>
<td>Performs a complete reboot of one or all phones associated with a Cisco CME router.</td>
<td><code>reset (telephony-service)</code></td>
</tr>
<tr>
<td>Performs a fast reboot of a single phone associated with a Cisco CME router.</td>
<td><code>restart (ephone)</code></td>
</tr>
<tr>
<td>Performs a fast reboot of one or all phones associated with a Cisco CME router.</td>
<td><code>restart (telephony-service)</code></td>
</tr>
</tbody>
</table>
reset (telephony-service)

To perform a complete reboot of one or all phones associated with a Cisco CallManager Express (Cisco CME) router, use the **reset** command in telephony-service configuration mode. To interrupt and cancel a sequential reset cycle, use the **no** form of the command with the **sequence-all** keyword.

```
reset {all [time-interval]|cancel|mac-address|sequence-all}
no reset {all [time-interval]|cancel|mac-address|sequence-all}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>all</td>
<td>Resets all Cisco IP phones served by the Cisco CME router. The router pauses for 15 seconds between the reset starts for each successive phone unless the <strong>time-interval</strong> argument is used to change that value.</td>
</tr>
<tr>
<td>time-interval</td>
<td>(Optional) Time interval, in seconds, between each phone reset. Range is from 0 to 60. Default is 15.</td>
</tr>
<tr>
<td>cancel</td>
<td>Interrupts a sequential reset cycle that was started with a <strong>reset sequence-all</strong> command.</td>
</tr>
<tr>
<td>mac-address</td>
<td>MAC address of a particular Cisco IP phone.</td>
</tr>
<tr>
<td>sequence-all</td>
<td>Resets all phones in strict one-at-a-time order by waiting for one phone to reregister before starting the reset for the next phone. The sequencing of resets prevents possible conflicts between phones trying to access TFTP services simultaneously. There is a reset timeout of 4 minutes, after which the router stops waiting for the currently registering phone to complete registration and starts to reset the next phone.</td>
</tr>
</tbody>
</table>

**Command Default**

No reset is performed.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1</td>
<td>The <strong>time-interval</strong> range maximum was increased from 15 to 60 and the default was changed from 0 to 15.</td>
</tr>
<tr>
<td>12.2(11)YT1</td>
<td>Cisco ITS 2.1</td>
<td>The <strong>cancel</strong> and <strong>sequence-all</strong> keywords were introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

After you update information for one or more phones associated with a Cisco CME router, the phone or phones must be rebooted using either the **reset** command or the **restart** command. The **reset** command performs a “hard” reboot similar to a power-off-power-on sequence and contacts the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server for updated information as well. The **restart** command performs a “soft” reboot by simply rebooting the phone without contacting the DHCP and TFTP servers. The **reset** command takes significantly longer to process than the **restart** command when you are updating multiple phones, but...
it must be used after you make changes to phone firmware, user locale, network locale, or URL parameters. For simple button, line, or speed-dial changes, you can use the `restart` command.

When you use the `reset` command, the default time interval of 15 seconds is recommended so that phone reset operations are staggered in order to avoid all phones attempting to access router system resources at the same time. A shorter interval may be used on systems with only a small number of phones or for cases where a simple reset of the phones is desired that does not result in the phones downloading updates to the phone firmware (using the router’s TFTP service).

When you use the `reset sequence-all` command, the router waits for one phone to complete its reset and reregister before starting to reset the next phone. The delay provided by this command prevents multiple phones from attempting to access the TFTP server simultaneously and therefore failing to reset properly. Each reset operation can take several minutes when you use this command. There is a reset timeout of 4 minutes, after which the router stops waiting for the currently registering phone to complete registration and starts to reset the next phone.

If the router configuration is changed so that the eXtensible Markup Language (XML) configuration files for the phones are modified (changes are made to user locale, network locale, or phone firmware), then whenever you use the `reset all` or `restart all` command, the router automatically executes the `reset sequence-all` command instead. The `reset sequence-all` command resets phones one at a time in order to prevent multiple phones from trying to contact the TFTP server simultaneously. This one-at-a-time sequencing can take a long time if there are many phones. To avoid this automatic behavior, use the `reset all time-interval` or the `restart all time-interval` with an explicit argument that is not equal to the default 15-second time interval; for example, set a time interval of 14 seconds. If a `reset sequence-all` command has been started in error, use the `reset cancel` command to interrupt and cancel the sequence of resets.

The `restart` command allows the system to perform quick phone resets in which only the button template, line information, and speed-dial information is updated. See the documentation for the `restart` command for more information.

The no form of this command has an effect only when used with the `all` or `sequence-all` keyword, when it interrupts and cancels the sequential resetting of phones.

### Examples

The following example resets all IP phones served by the Cisco CME router:

```bash
Router(config)# telephony-service
Router(config-telephony)# reset all
```

The following example resets the Cisco IP phone with the MAC address CFBA.321B.96FA:

```bash
Router(config)# telephony-service
Router(config-telephony)# reset CFBA.321B.96FA
```

The following example resets all IP phones in sequential, not-overlapping order:

```bash
Router(config)# telephony-service
Router(config-telephony)# reset sequence-all
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>reset (ephone)</code></td>
<td>Performs a complete reboot of a single phone associated with a Cisco CME router.</td>
</tr>
<tr>
<td><code>restart (ephone)</code></td>
<td>Performs a fast reboot of a single phone associated with a Cisco CME router.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td><strong>restart</strong> <em>(telephony-service)</em></td>
<td>Performs a fast reboot of one or all phones associated with a Cisco CME router.</td>
</tr>
<tr>
<td><strong>telephony-service</strong></td>
<td>Enters telephony-service configuration mode.</td>
</tr>
</tbody>
</table>
reset (voice logout-profile and voice user-profile)

To perform a complete reboot of all IP phones on which a particular extension-mobility profile is downloaded, use the reset command in voice logout-profile configuration mode or voice user-profile configuration mode.

reset

Syntax Description
This command has no arguments or keywords.

Command Default
No reset is performed.

Command Modes
Voice logout-profile configuration (voice-logout-profile)
Voice user-profile configuration (voice-user-profile)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XW2</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines
Use this command to perform a “hard” reboot similar to a power-off-power-on sequence, which includes downloading updated information from the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server.

Configure this command in voice logout-profile configuration mode after creating or modifying a logout profile for extension mobility.

Configure this command in voice user-profile configuration mode after creating or modifying an individual user’s profile for extension mobility.

This command has a no form, but the no form has no effect.

Examples
The following example shows how to modify a logout profile by adding speed-dial definitions and then reset all IP phones on which this logout profile is downloaded to propagate the modification:

Router# configure terminal
Router(config)# voice logout-profile 12
Router(config-user-profile)# speed-dial 1 3001
Router(config-user-profile)# speed-dial 2 3002 blf
Router (config/logout-profile)# reset
Router (config/logout-profile)# exit
Router (config)#
reset (voice register global)

To perform a complete reboot of all SIP phones associated with a Cisco CallManager Express (Cisco CME) router, use the reset command in voice register global configuration mode.

```
reset
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

No reset is performed.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

After you update information for one or more SIP phones associated with a Cisco CME router, reboot the phones by using the reset command. The reset command performs a “hard” reboot similar to a power-off-power-on sequence and contacts the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server for updated information as well. Configure the reset command after you make changes to phone firmware, user locale, network locale, or URL parameters.

The time interval between each phone reset is 15 seconds, thereby avoiding an attempt by all phones to access router system resources at the same time.

This command has a no form, but the no form has no effect.

**Examples**

The following example shows how to reset all SIP phones served by the Cisco CME router:

```
Router(config)# voice register global
Router(config-register-global)# reset
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>reset (voice register pool)</td>
<td>Performs a complete reboot of a single SIP phone associated with a Cisco CME router.</td>
</tr>
</tbody>
</table>
reset (voice register pool)

To perform a complete reboot of a specific SIP phone associated with a Cisco CallManager Express (Cisco CME) router, use the reset command in voice register pool configuration mode. To interrupt a reset cycle, use the no form of this command.

reset
no reset

Syntax Description
This command has no arguments or keywords.

Command Default
No reset is performed.

Command Modes
Voice register pool configuration (config-register-pool)

Command History
<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

Usage Guidelines
After you update information for one or more phones associated with a Cisco CME router, the phones must be rebooted by using the reset command. The reset command performs a “hard” reboot similar to a power-off-power-on sequence and contacts the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server for updated information as well. Configure the reset command after you make changes to phone firmware, user locale, network locale, or URL parameters.

Use this command to perform a complete reboot of an individual SIP phone when you are in voice register pool configuration mode. To reset all SIP phones, use the reset (voice register global) command.

This command has a no form, but the no form has no effect.

Examples

The following example shows how to reset SIP phone 1 served by the Cisco CME router:

Router(config)# voice register pool 1
Router(config-register-pool)# reset

Related Commands

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
</table>

reset (voice register global) | Performs a complete reboot of all SIP phones associated with a Cisco CME router.
reset (voice-gateway)

To perform a complete reboot of all analog phones associated with the voice gateway and registered to Cisco Unified CME, use the reset command in voice-gateway configuration mode.

reset

Syntax Description
This command has no arguments or keywords.

Command Default
No reset is performed.

Command Modes
Voice-gateway configuration (config-voice-gateway)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines
After you update information for one or more analog phones associated with the voice gateway, reboot the phones by using the reset command. The reset command performs a “hard” reboot similar to a power-off-power-on sequence and contacts the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server for updated information. Use the reset command after you make changes to phone firmware, user or network locales, or URL parameters.

The time interval between each phone reset is 15 seconds, to avoid an attempt by all phones to access system resources at the same time.

This command has a no form, but the no form has no effect.

Examples
The following example shows how to reset all analog phones associated with the voice gateway:

Router(config)# voice-gateway system 1
Router(config-voice-gateway)# reset

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>restart (voice-gateway)</td>
<td>Performs a fast restart of all analog endpoints associated with the voice gateway.</td>
</tr>
</tbody>
</table>
reset tapi

To reset the connection between a Telephony Application Programmer's Interface (TAPI) application and a particular SCCP phone in Cisco Unified CME, use the reset tapi command in ephone configuration mode.

**reset tapi**

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

No reset of the connection between the TAPI application and the router is performed.

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(20)YA</td>
<td>Cisco Unified CME 7.0(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(22)T</td>
<td>Cisco Unified CME 7.0(1)</td>
<td>This command was integrated into Cisco IOS Release 12.4(22)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command in ephone configuration mode resets the connection between a TAPI application and a particular SCCP phone. This command does not reset the Ethernet phone.

To disassociate and reestablish the connection without using this command, you must reboot the router.

This command has a no form, but the no form has no effect.

**Examples**

The following example shows how to reset the connection between a TAPI application and the SCCP phone associated with the ephone-tag of 1:

```
Router(config)# ephone 1
Router(config-ephone)# reset tapi
```
**restart (ephone)**

To perform a fast reboot of an IP phone associated with a Cisco CallManager Express (Cisco CME) router, use the `restart` command in ephone configuration mode. To cancel the reboot, use the `no` form of this command.

```
restart
no restart
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

No restart is performed.

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)YT1</td>
<td>Cisco ITS 2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command causes the system to perform a fast phone reboot in which only the button template, lines, and speed-dial numbers are updated on the phone. For updates related to phone firmware, user locale, network locale, or URL parameters, use the `reset` command. The `restart` command is much faster than the `reset` command because the phone does not need to access the DHCP or TFTP server.

To restart all phones in a Cisco CME system for quick changes to buttons, lines, and speed-dial numbers, use the `restart` command in telephony-service configuration mode.

This command has a `no` form, but the `no` form has no effect.

**Examples**

The following example restarts the phone with phone-tag 1:

```
Router(config)# ephone 1
Router(config-ephone)# restart
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>reset (ephone)</code></td>
<td>Performs a complete reboot of a Cisco IP phone associated with a Cisco CME router.</td>
</tr>
<tr>
<td><code>reset (telephony-service)</code></td>
<td>Performs a complete reboot of one or all phones associated with a Cisco CME router.</td>
</tr>
<tr>
<td><code>restart (telephony-service)</code></td>
<td>Performs a fast reboot of one or all phones associated with a Cisco CME router.</td>
</tr>
</tbody>
</table>
**restart (telephony-service)**

To perform a fast reboot of one or all phones associated with a Cisco CallManager Express (Cisco CME) router, use the `restart` command in telephony-service configuration mode. To cancel the reboot, use the `no` form of this command.

```
restart {all [time-interval]mac-address}
no restart {all [time-interval]mac-address}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>all</code></td>
<td>Restarts all phones associated with the Cisco CME router.</td>
</tr>
<tr>
<td><code>time-interval</code></td>
<td>(Optional) Time between each phone restart, in seconds. Range is from 0 to 60. Default is 15.</td>
</tr>
<tr>
<td><code>mac-address</code></td>
<td>MAC address of the phone to be restarted.</td>
</tr>
</tbody>
</table>

**Command Default**

Time-interval is 15 seconds.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)YT1</td>
<td>Cisco ITS 2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command causes the system to perform a fast phone reset in which only the button template, lines, and speed-dial numbers are updated on the phone. For updates related to phone firmware, user locale, network locale, or URL parameters, use the `reset` command.

Use the `restart` command to reboot IP phones after quick changes to buttons, lines, and speed-dial numbers. This command is much faster than the `reset` command because the phone does not access the DHCP or TFTP server.

To restart a single phone, use the `restart` command with the `mac-address` argument or use the `restart` command in ephone configuration mode.

If the router configuration is changed so that the eXtensible Markup Language (XML) configuration files for the phones are modified (changes are made to user locale, network locale, or phone firmware), then whenever you use the `reset all` or `restart all` command, the router automatically executes the `reset sequence-all` command instead. The `reset sequence-all` command resets phones one at a time in order to prevent multiple phones trying to contact the TFTP server simultaneously. This one-at-a-time sequencing can take a long time if there are many phones. To avoid this automatic behavior, use the `reset all time-interval` command or the `restart all time-interval` command with an explicit argument that is not equal to the default 15-second time interval; for example, set a time interval of 14 seconds. If a `reset sequence-all` command has been started in error, use the `reset cancel` command to interrupt and cancel the sequence of resets.

The `no` form of this command has an effect only when used with the `all` keyword, when it interrupts and cancels the sequential restarting of phones.

**Examples**

The following example performs a quick restart of all phones in a Cisco CME system:
Router(config)# telephony-service
Router(config-telephony)# restart all

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>reset (ephone)</code></td>
<td>Performs a complete reboot of a Cisco IP phone associated with a Cisco CME router.</td>
</tr>
<tr>
<td><code>reset (telephony-service)</code></td>
<td>Performs a complete reboot of one or all phones associated with a Cisco CME router.</td>
</tr>
<tr>
<td><code>restart (ephone)</code></td>
<td>Performs a fast reboot of a single phone associated with a Cisco CME router.</td>
</tr>
</tbody>
</table>
**restart (voice register)**

To perform a fast reset of one or all SIP phones associated with a Cisco Unified CME router, use the `restart` command in voice register global or voice register pool configuration mode. To cancel the reboot, use the `no` form of this command.

```
restart
no restart
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

SIP phones are not restarted.

**Command Modes**

Voice register global configuration (config-register-global)
Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command causes the system to perform a fast phone reset in which only the button template, lines, and speed-dial numbers are updated on the phone. For updates related to phone firmware, user locale, network locale, or URL parameters, use the `reset` command.

Use this command to reboot SIP phones after quick changes to buttons, lines, and speed-dial numbers. This command is much faster than the `reset` command because the phone does not access the DHCP or TFTP server.

To restart a single SIP phone, use the `restart` command in voice register pool configuration mode. To restart all SIP phones in a Cisco Unified CME system, use the `restart` command in voice register global configuration mode.

This command has a `no` form, however the `no` form has no effect.

**Note**

This command requires firmware load 8-0-2-14 or later versions; it is not supported in older SIP phone loads. To support this command on SIP phones using older firmware, you must upgrade all your phone firmware.

**Examples**

The following example performs a quick restart of all SIP phones in a Cisco Unified CME system:

```
Router(config) # voice register global
Router(config-register-global) # restart
```

The following example performs a quick restart of SIP phone 10:
Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>reset (voice register pool)</td>
<td>Performs a complete reboot of a single SIP phone associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td>reset (voice register global)</td>
<td>Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router.</td>
</tr>
</tbody>
</table>
**restart (voice-gateway)**

To perform a fast restart of all analog phones associated with the voice gateway and registered to Cisco Unified CME, use the `restart` command in voice-gateway configuration mode.

```
restart
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Analog phones are not restarted.

**Command Modes**

Voice-gateway configuration (config-voice-gateway)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command initiates a quick phone restart in which only the buttons, lines, and speed-dial numbers are updated on the phone. For updates related to phone firmware, user and network locales, or URL parameters, use the `reset` command.

Use this command to reboot all analog phones on the voice gateway after simple configuration changes to buttons, lines, and speed-dial numbers. This command is faster than the `reset` command because the phone does not access the DHCP server.

This command has a `no` form, although the `no` form has no effect.

**Examples**

The following example shows how to perform a quick restart of all analog phones:

```
Router(config)# voice-gateway system 1
Router(config-voice-gateway)# restart
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>reset (voice-gateway)</strong></td>
<td>Performs a complete reboot of all analog endpoints associated with the voice gateway.</td>
</tr>
</tbody>
</table>
**ring (ephone-dn)**

To set the ring pattern for all incoming calls to an ephone-dn, use the `ring` command in ephone-dn configuration mode. To return to the standard ring pattern, use the `no` form of this command.

```
ring {external|feature|internal} [ {primary|secondary} ]
no ring
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><code>external</code></td>
<td>External ring pattern is used for all incoming calls.</td>
<td></td>
</tr>
<tr>
<td><code>feature</code></td>
<td>Feature ring pattern is used for all incoming calls.</td>
<td></td>
</tr>
<tr>
<td><code>internal</code></td>
<td>Internal ring pattern is used for all incoming calls.</td>
<td></td>
</tr>
<tr>
<td><code>primary</code></td>
<td>(Optional) Ring pattern is used on primary number only.</td>
<td></td>
</tr>
<tr>
<td><code>secondary</code></td>
<td>(Optional) Ring pattern is used on secondary number only.</td>
<td></td>
</tr>
</tbody>
</table>

**Command Default**

Standard ring pattern is used.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command allows you to select one of the three ring styles supported by SCCP—internal, external, or feature ring. The ring pattern is used for all types of incoming calls to this directory number, on all phones on which the directory number appears. If the phone is already in use, an incoming call is presented as a call-waiting call and uses the distinctive call-waiting beep.

If the `primary` or `secondary` keyword is used, the distinctive ring is used only if the incoming called number matches the primary number or secondary number defined for the ephone-dn. If there is no secondary number defined for the ephone-dn, the `secondary` keyword has no effect.

By default, Cisco Unified CME uses the internal ring pattern for calls between local IP phones and uses the external ring pattern for all other types of calls.

You can associate the feature ring pattern with a specific button on a phone by using the `button f` command. This command assigns the ring pattern to the button on the phone so that different phones that share the same directory number can use a different ring style.

**Examples**

The following example sets external ringing for all incoming calls on extension 2389.

```
ephone-dn 24
number 2389
ring external
```
### Related Commands

| button | Associates ephone-dns with individual buttons on an IP phone and specifies line type or ring behavior. |

---

**Cisco Unified CME Commands: R**

related commands

Description

| button | Associates ephone-dns with individual buttons on an IP phone and specifies line type or ring behavior. |

---

**Cisco Unified Communications Manager Express Command Reference**

911
route-code

To enable phone users to dial a route code to specify special routing for a call, use the **route-code** command in voice MLPP configuration mode. To reset to the default, use the **no** form of this command.

```
route-code
no route-code
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Route code is disabled.

**Command Modes**

Voice MLPP configuration (config-voice-mlpp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables users to specify special routing for an MLPP call by dialing a route code. The route code is a two-digit number beginning with 1.

**Examples**

The following example shows how to enable users to dial a route code:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# route-code
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>access-digit</td>
<td>Defines the access digit that phone users dial to request a precedence call.</td>
</tr>
<tr>
<td>mlpp preemption</td>
<td>Enables preemption capability on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td>service-digit</td>
<td>Enables phone users to dial a service digit to request off-net services.</td>
</tr>
</tbody>
</table>
rule (voice translation-rule)

To define a translation rule, use the rule command in voice translation-rule configuration mode. To delete the translation rule, use the no form of this command.

**Match and Replace Rule**

```
rule precedence /match-pattern/ /replace-pattern/ [ type {match-type replace-type} ] [ plan {match-type replace-type} ]
```

**Reject Rule**

```
rule precedence reject /match-pattern/ /replace-pattern/ /match-pattern/ [ type {match-type } ] [ plan {match-type r} ]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>precedence</th>
<th>Priority of the translation rule. Range is from 1 to 15.</th>
</tr>
</thead>
<tbody>
<tr>
<td>/match-pattern/</td>
<td>Stream editor (SED) expression used to match incoming call information. The slash ‘/’ is a delimiter in the pattern.</td>
</tr>
<tr>
<td>/replace-pattern/</td>
<td>SED expression used to replace the match pattern in the call information. The slash ‘/’ is a delimiter in the pattern.</td>
</tr>
</tbody>
</table>

**type match-type replace-type** (Optional) Number type of the call. Valid values for the match-type argument are as follows:

- **abbreviated** — Abbreviated representation of the complete number as supported by this network.
- **any** — Any type of called number.
- **international** — Number called to reach a subscriber in another country.
- **national** — Number called to reach a subscriber in the same country, but outside the local network.
- **network** — Administrative or service number specific to the serving network.
- **reserved** — Reserved for extension.
- **subscriber** — Number called to reach a subscriber in the same local network.
- **unknown** — Number of a type that is unknown by the network.

Valid values for the replace-type argument are as follows:

- **abbreviated** — Abbreviated representation of the complete number as supported by this network.
- **international** — Number called to reach a subscriber in another country.
- **national** — Number called to reach a subscriber in the same country, but outside the local network.
- **network** — Administrative or service number specific to the serving network.
- **reserved** — Reserved for extension.
- **subscriber** — Number called to reach a subscriber in the same local network.
- **unknown** — Number of a type that is unknown by the network.
The match pattern of a translation rule is used for call-reject purposes.

| plan | (Optional) Numbering plan of the call. Valid values for the match-type argument are as follows:
|      | • any — Any type of dialed number.
|      | • data
|      | • ermes
|      | • isdn
|      | • national — Number called to reach a subscriber in the same country, but outside the local network.
|      | • private
|      | • reserved — Reserved for extension.
|      | • telex
|      | • unknown — Number of a type that is unknown by the network.
| replace-type | Valid values for the replace-type argument are as follows:
|      | • data
|      | • ermes
|      | • isdn
|      | • national — Number called to reach a subscriber in the same country, but outside the local network.
|      | • private
|      | • reserved — Reserved for extension.
|      | • telex
|      | • unknown — Number of a type that is unknown by the network.

| reject | The match pattern of a translation rule is used for call-reject purposes.

Command Default

No default behavior or values

Command Modes

Voice translation-rule configuration

Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)T</td>
<td>This command was introduced with a new syntax in voice-translation-rule configuration mode.</td>
</tr>
<tr>
<td>15.1(4)M</td>
<td>This command was introduced with an increase in the maximum value of the precidence variable from 15 to 100.</td>
</tr>
</tbody>
</table>

Usage Guidelines

Note

Use this command in conjunction after the voice translation-rule command. An earlier version of this command uses the same name but is used after the translation-rule command and has a slightly different command syntax. In the older version, you cannot use the square brackets when you are entering command syntax. They appear in the syntax only to indicate optional parameters, but are not accepted as delimiters in actual command entries. In the newer version, you can use the square brackets as delimiters. Going forward, we recommend that you use this newer version to define rules for call matching. Eventually, the translation-rule command will not be supported.
A translation rule applies to a calling party number (automatic number identification [ANI]) or a called party number (dialed number identification service [DNIS]) for incoming, outgoing, and redirected calls within Cisco H.323 voice-enabled gateways.

Number translation occurs several times during the call routing process. In both the originating and terminating gateways, the incoming call is translated before an inbound dial peer is matched, before an outbound dial peer is matched, and before a call request is set up. Your dial plan should account for these translation steps when translation rules are defined.

The below table shows examples of match patterns, input strings, and result strings for the rule (voice translation-rule) command.

<table>
<thead>
<tr>
<th>Match Pattern</th>
<th>Replacement Pattern</th>
<th>Input String</th>
<th>Result String</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>/.<em>\s</em></td>
<td>//</td>
<td>4085550100</td>
<td>4085550100</td>
<td>Any string to null string.</td>
</tr>
<tr>
<td>//</td>
<td>//</td>
<td>4085550100</td>
<td>4085550100</td>
<td>Match any string but no replacement. Use this to manipulate the call plan or call type.</td>
</tr>
<tr>
<td>/(^(.*))(\d\d\d))/</td>
<td>/\155\2/</td>
<td>4084560177</td>
<td>4085550177</td>
<td>Match from the middle of the input string.</td>
</tr>
<tr>
<td>/(.*)0120/</td>
<td>/\10155/</td>
<td>4081110120</td>
<td>4081110155</td>
<td>Match from the end of the input string.</td>
</tr>
<tr>
<td>/^1((.*)/</td>
<td>/\1/</td>
<td>1#2345</td>
<td>2345</td>
<td>Replace match string with null string.</td>
</tr>
<tr>
<td>/^408((\d\d\d))/</td>
<td>/\55\1/</td>
<td>4087770100</td>
<td>5550100</td>
<td>Match multiple patterns.</td>
</tr>
<tr>
<td>/1234/</td>
<td>/\00&amp;00/</td>
<td>5550100</td>
<td>55500010000</td>
<td>Match the substring.</td>
</tr>
<tr>
<td>/1234/</td>
<td>/\00\00/</td>
<td>5550100</td>
<td>55500010000</td>
<td>Match the substring (same as &amp;).</td>
</tr>
</tbody>
</table>

The software verifies that a replacement pattern is in a valid E.164 format that can include the permitted special characters. If the format is not valid, the expression is treated as an unrecognized command.

The number type and calling plan are optional parameters for matching a call. If either parameter is defined, the call is checked against the match pattern and the selected type or plan value. If the call matches all the conditions, the call is accepted for additional processing, such as number translation.

Several rules may be grouped together into a translation rule, which gives a name to the rule set. A translation rule may contain up to 15 rules. All calls that refer to this translation rule are translated against this set of criteria.

The precedence value of each rule may be used in a different order than that in which they were typed into the set. Each rule’s precedence value specifies the priority order in which the rules are to be used. For example, rule 3 may be entered before rule 1, but the software uses rule 1 before rule 3.

The software supports up to 128 translation rules. A translation profile collects and identifies a set of these translation rules for translating called, calling, and redirected numbers. A translation profile is referenced by trunk groups, source IP groups, voice ports, dial peers, and interfaces for handling call translation.
The following example applies a translation rule. If a called number starts with 5550105 or 70105, translation rule 21 uses the rule command to forward the number to 14085550105 instead.

Router(config)# voice translation-rule 21
Router(cfg-translation-rule)# rule 1 /^5550105/ /14085550105/
Router(cfg-translation-rule)# rule 2 /^70105/ /14085550105/

In the next example, if a called number is either 14085550105 or 014085550105, after the execution of translation rule 345, the forwarding digits are 50105. If the match type is configured and the type is not “unknown,” dial-peer matching is required to match the input string numbering type.

Router(config)# voice translation-rule 345
Router(cfg-translation-rule)# rule 1 /^14085550105/ /50105/ plan any national
Router(cfg-translation-rule)# rule 2 /^014085550105/ /50105/ plan any national

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice translation-rule</td>
<td>Displays the parameters of a translation rule.</td>
</tr>
<tr>
<td>voice translation-rule</td>
<td>Initiates the voice translation-rule definition.</td>
</tr>
</tbody>
</table>
Cisco Unified CME Commands: S1

- sast1 trustpoint, on page 919
- sast2 trustpoint, on page 920
- sdspfarm conference lecture mode on, on page 921
- sdspfarm conference mute-on mute-off, on page 922
- sdspfarm tag, on page 923
- sdspfarm transcode sessions, on page 925
- sdspfarm units, on page 926
- sdspfarm unregister force, on page 927
- secondary dialtone (voice port), on page 928
- secondary start, on page 929
- secondary-dialtone, on page 931
- secure-signaling trustpoint, on page 932
- semi-attended enable (voice register template), on page 933
- server (CTL-client), on page 934
- server (presence), on page 936
- server-security-mode, on page 937
- service directed-pickup, on page 939
- service dnis dir-lookup, on page 942
- service dnis overlay, on page 945
- service dss, on page 947
- service https (phone-template), on page 949
- service https (telephony-service), on page 950
- service https (voice register global), on page 951
- service https (voice register template), on page 952
- service local-directory, on page 953
- service phone, on page 956
- service profile, on page 966
- service-digit, on page 967
- service-enable (auto-register), on page 968
- service-domain, on page 970
- service-domain (voice class), on page 971
- service-domain midcall-mismatch, on page 972
- session-server, on page 973
• session-transport, on page 975
sast1 trustpoint

To specify the name of the SAST1 trustpoint, use the `sast1 trustpoint` command in CTL-client configuration mode. To return to the default, use the `no` form of this command.

```
sast1 trustpoint label
no sast1
```

**Syntax Description**

| label | Name of the SAST1 trustpoint. |

**Command Default**

No SAST1 trustpoint name is specified

**Command Modes**

CTL-client configuration (config-ctl-client)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication.

The `sast1 trustpoint` and `sast2 trustpoint` commands are used to set up the System Administrator Security Token (SAST) credentials, which are used to sign the CTL file. The SAST1 and SAST2 certificates must be different from each other, but to conserve memory either one of them can use the same certificate as Cisco Unified CME. The CTL file is always signed by SAST1 credentials. The SAST2 credentials are included in the CTL file so that if the SAST1 certificate is compromised, the CTL file can be signed by SAST2 to prevent the phones from being reset to their factory defaults.

**Examples**

The following example names sast1tp as the SAST1 trustpoint.

```
Router(config)# ctl-client
Router(config-ctl-client)# server capf 10.2.2.2 trustpoint capftrust
Router(config-ctl-client)# server cme 10.2.2.3 trustpoint cmctp
Router(config-ctl-client)# server tftp 10.2.2.4 trustpoint tftp
Router(config-ctl-client)# sast1 trustpoint sast1tp
Router(config-ctl-client)# sast2 trustpoint sast2tp
Router(config-ctl-client)# regenerate
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>sast2 trustpoint</td>
<td>Specifies the name of the SAST2 trustpoint.</td>
</tr>
</tbody>
</table>
**sast2 trustpoint**

To specify the name of the SAST2 trustpoint, use the `sast2 trustpoint` command in CTL-client configuration mode. To return to the default, use the `no` form of this command.

```
sast2 trustpoint  label
no sast2
```

**Syntax Description**

- **label**: Name of the SAST2 trustpoint.

**Command Default**

No SAST2 trustpoint name is specified.

**Command Modes**

CTL-client configuration (config-ctl-client)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication.

The `sast1 trustpoint` and `sast2 trustpoint` commands are used to set up the System Administrator Security Token (SAST) credentials, which are used to sign the CTL file. The SAST1 and SAST2 certificates must be different from each other, but to conserve memory either one of them can use the same certificate as Cisco CME. The CTL file is always signed by SAST1 credentials. The SAST2 credentials are included in the CTL file so that if the SAST1 certificate is compromised, the CTL file can be signed by SAST2 to prevent the phones from being reset to their factory defaults.

**Examples**

The following example names sast2tp as the SAST2 trustpoint.

```
Router(config)# ctl-client
Router(config-ctl-client)# server capf 10.2.2.2 trustpoint capftrust
Router(config-ctl-client)# server cme 10.2.2.3 trustpoint cmetp
Router(config-ctl-client)# server tftp 10.2.2.4 trustpoint tftptp
Router(config-ctl-client)# sast1 trustpoint sast1tp
Router(config-ctl-client)# sast2 trustpoint sast2tp
Router(config-ctl-client)# regenerate
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>sast1 trustpoint</td>
<td>Specifies the name of the SAST1 trustpoint.</td>
</tr>
</tbody>
</table>
**sdspfarm conference lecture mode on**

To permit a participant in a video conference call to switch back and forth between lecture mode and the configured default mode in DSP farm, use the `sdspfarm conference` command in telephony-service configuration mode. The participant who enters the FAC becomes the lecturer and is displayed on all other screens. The lecturer’s screen displays a scanning stream of the other participants.

To delete a tag generated by the `sdspfarm conference` command, use the `no` form of this command.

```
sdspfarm conference lecture mode on  F A C release  F A C
no sdspfarm conference lecture mode on  F A C release  F A C
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>FAC</td>
<td>Sets the Feature Access Codes (FAC) that a participant enters on the keypad to switch to the lecture mode. Valid values are the numbers on the keypad. Maximum 3 digits</td>
</tr>
<tr>
<td>release FAC</td>
<td>Sets the Feature Access Codes (FAC) that a participant enters on the keypad to exit lecture mode. Valid values are the numbers on the keypad. Maximum 3 digits</td>
</tr>
</tbody>
</table>

### Command Default

Lecture mode is not enabled by default.

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(4)M</td>
<td>Cisco Unified CME 8.6</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

You can define any three digits to be FAC for lecture mode. A participant cannot enter lecture mode on a phone with unsupported video formats, for example an audio-only phone. The lecture mode participant must exit lecture mode before anyone else can become the lecturer.

### Examples

The following example configure lecture mode to be activated when the user presses a FAC number of 111.

```
Router(config)# telephony-service
outer(config-telephony)# sdspfarm conference lecture-mode on 111 release 222
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>sdspfarm profile</code></td>
<td>Enters DSP farm profile configuration mode and defines a profile for DSP farm services.</td>
</tr>
<tr>
<td><code>sdspfarm transcode</code></td>
<td>Specifies the maximum number of transcoding sessions allowed per Cisco CME router.</td>
</tr>
<tr>
<td><code>sdspfarm units</code></td>
<td>Specifies the maximum number of DSP farms that are allowed to be registered to the SCCP server.</td>
</tr>
</tbody>
</table>
**sdspfarm conference mute-on mute-off**

To define mute-on and mute-off DTMF digits for use during conferencing, use the `sdspfarm conference mute-on mute-off` command in telephony-service configuration mode. To disable the mute-on and mute-off digits, use the `no` form of this command.

```
sdspfarm conference mute-on mute-on-digits mute-off mute-off-digits
no sdspfarm conference mute-on mute-on-digits mute-off mute-off-digits
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>mute-on</code></td>
<td>Defines the buttons you press on your phone to mute during a conference.</td>
</tr>
<tr>
<td><code>mute-on-digits</code></td>
<td>Maximum: 3 digits. Valid values are the numbers and symbols that appear on your</td>
</tr>
<tr>
<td></td>
<td>telephone keypad: 1, 2, 3, 4, 5, 6, 7, 8, 9, 0, *, and #.</td>
</tr>
<tr>
<td><code>mute-off</code></td>
<td>Defines the buttons you press on your IP phone to unmute during a conference.</td>
</tr>
<tr>
<td><code>mute-off-digits</code></td>
<td>Maximum: 3 digits. Valid values are the numbers and symbols that appear on your</td>
</tr>
<tr>
<td></td>
<td>telephone keypad: 1, 2, 3, 4, 5, 6, 7, 8, 9, 0, *, and #.</td>
</tr>
</tbody>
</table>

**Command Default**

No mute-on or mute-off digits are defined.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

You must define mute-on and mute-off digits to mute or unmute your phone using the keypad during a conference. The mute-on digits and mute-off digits can be the same or different. You can mute and unmute your phone using the phone’s mute button also. You must unmute the phone in the same way that you muted it, either with the keypad or the mute button.

**Examples**

The following example configures #5 as the buttons to press to mute and unmute the phone during a conference:

```
Router(config-telephony)# sdspfarm conference mute-on #5 mute-off #5
```
sdspfarm tag

To permit a digital-signal-processor (DSP) farm to be registered to Cisco Unified CME and associate it with the MAC address of a Skinny Client Control Protocol (SCCP) interface, use the `sdspfarm tag` command in telephony-service configuration mode. To delete a tag generated by the `sdspfarm tag` command, use the `no` form of this command.

```
sdspfarm tag number device-name
no sdspfarm tag number device-name
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>number</code></td>
<td>Numeric name for a DSP farm. Number from 1 to 10.</td>
</tr>
<tr>
<td><code>device-name</code></td>
<td>Word describing the device, such as the MAC address, of the SCCP client interface that is preceded by the Message Transfer Part (MTP).</td>
</tr>
</tbody>
</table>

**Command Default**

DSP farm is not created.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco Unified CME 3.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(4)M</td>
<td>Cisco Unified CME 8.6</td>
<td>Increased support for the number of DSP farms to 10.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

DSP farm profiles are sets of DSP resources used for conferencing and transcoding only. DSP farms do not include voice termination resources. Use the `show interface` command to find the MAC address of the SCCP client interface.

**Examples**

The following example declares tag 1 as the MAC address of mac000a.8aea.ca80. The `show interface` command is used to obtain the MAC address.

```
Router# show interface FastEthernet 0/0

FastEthernet0/0 is up, line protocol is up
Hardware is AmdFE, address is 000a.8aea.ca80 (bia 000a.8aea.ca80)

Router(config)# telephony-service

Router(config-telephony)# sdspfarm tag 1 mac000a.8aea.ca80
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>sdspfarm transcode</td>
<td>Specifies the maximum number of transcoding sessions allowed per Cisco CME router.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>sdspfarm</td>
<td>Specifies the maximum number of DSP farms that are allowed to be registered to the SCCP server.</td>
</tr>
<tr>
<td>units</td>
<td></td>
</tr>
</tbody>
</table>
### sdspfarm transcode sessions

To specify the maximum number of transcoding sessions allowed per Cisco CallManager Express (Cisco CME) router, use the `sdspfarm transcode sessions` command in telephony-service configuration mode. To return to the default transcode session of 0, use the `no` form of this command.

```
sdspfarm transcode sessions number
no sdspfarm transcode sessions number
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>number</code></td>
<td>Declares the number of DSP farm sessions. Valid values are numbers from 1 to 128.</td>
</tr>
</tbody>
</table>

| Command Default    | The default is 0. |

| Command Modes      | Telephony-service configuration (config-telephony) |

<table>
<thead>
<tr>
<th>Command History</th>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>12.3(11)T</td>
<td>Cisco Unified CME 3.2</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

| Usage Guidelines   | The transcoding is allowed between G.711 and G.729. A session consists of two transcode streams. To configure this information, you must know how many digital-signal-processor (DSP) farms are configured on the network module (NM) farms in your Cisco CME router. DSP farms are sets of DSP resources used for conferencing and transcoding only. DSP farms do not include voice termination resources. To learn how many DSP farms have been configured on your Cisco CME router, use the `show sdspfarm` command. |

<table>
<thead>
<tr>
<th>Examples</th>
<th>The following example sets the maximum number of transcoding sessions allowed on the Cisco CME router to 20:</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Router(config)# telephony-service</td>
</tr>
<tr>
<td></td>
<td>Router(config-telephony)# sdspfarm transcode sessions 20</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td><code>sdspfarm tag</code></td>
<td>Declares a DSP farm and associates it with an SCCP client interface’s MAC address.</td>
</tr>
<tr>
<td></td>
<td><code>sdspfarm unit</code></td>
<td>Specifies the maximum number of DSP farms that are allowed to be registered to the SCCP server.</td>
</tr>
<tr>
<td></td>
<td><code>show sdspfarm</code></td>
<td>Displays the status of the configured DSP farms and transcoding streams.</td>
</tr>
</tbody>
</table>
sdspfarm units

To specify the maximum number of digital-signal-processor (DSP) farm profiles that are allowed to be registered to the Skinny Client Control Protocol (SCCP) server, use the `sdspfarm units` command in telephony-service configuration mode. To set the number of DSP farm profiles to the default value of 0, use the `no` form of this command.

```
sdspfarm units number
no sdspfarm units number
```

**Syntax Description**

| number | Number of DSP farms. Valid values are numbers from 0 to 10. |

**Command Default**

The default number is 0.

**Command Modes**

Telephony-service configuration

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco Unified CME 3.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(4)M</td>
<td>Cisco Unified CME 8.6</td>
<td>Increased support for the number of DSP farms to 10.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

DSP farm profiles are sets of DSP resources used for conferencing and transcoding only. DSP farm profiles do not include voice termination resources.

**Examples**

The following example configures a Cisco CME router to register one DSP farm:

```
Router(config)# telephony-service
Router(config-telephony)# sdspfarm units 1
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>sdspfarm tag</td>
<td>Declares a DSP farm and associates it with the MAC address of an SCCP client interface.</td>
</tr>
<tr>
<td>sdspfarm transcode</td>
<td>Specifies the maximum number of transcoding sessions allowed per Cisco CME router.</td>
</tr>
</tbody>
</table>
**sdspfarm unregister force**

To remove all transcoding streams associated with active calls, use the `sdspfarm unregister force` command in telephony-service configuration mode. To deactivate the removal of transcoding streams, use the `no` form of this command.

```
  sdspfarm unregister force
  no sdspfarm unregister force
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

The default is transcoding streams are not removed.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco Unified CME 3.2</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

If any of the SCCP server’s associated streams are functioning in active calls, the default response for the `sdspfarm unregister force` command is to reject them. If no stream is used in a call, all of the transcoding streams associated with the DSP farm will be released, and SCCP server can recycle those streams for other DSP farms.

**Examples**

The following example removes all transcoding streams for active calls.

```
Router(config)# telephony-service
Router(config-telephony)# sdspfarm unregister force
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>sdspfarm tag</td>
<td>Declares a DSP farm and associates it with an SCCP client interface’s MAC address.</td>
</tr>
<tr>
<td>sdspfarm unit</td>
<td>Specifies the maximum number of DSP farms that are allowed to be registered to the SCCP server.</td>
</tr>
<tr>
<td>show sdspfarm</td>
<td>Displays the status of the configured DSP farms and transcoding streams.</td>
</tr>
</tbody>
</table>
**secondary dialtone (voice port)**

To allow dialed digits to be collected from the remote switch when “connection plar” is not defined from the analog FXO voice-port, use the secondary dialtone command in global configuration mode. To disable the secondary dialtone, use the no form of the command.

```
secondary dialtone
no secondary dialtone
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

The secondary dialtone command is disabled.

**Command Modes**

Global configuration.

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the secondary dialtone command to allow dialed digits to be collected from the remote switch when “connection plar” is not defined from the analog FXO voice-port.

The following is a sample output from this command:

```
Router(config)# voice-port 2/0/0
Router (config-voiceport)#no secondary ?
   dialtone    Secondary dialtone option for FXO port
Router (config-voiceport)#no secondary dialtone
"secondary dialtone" is used to enable 2-stage dialing for an incoming call
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice service</td>
<td>Enters voice service configuration mode.</td>
</tr>
</tbody>
</table>
**secondary start**

To specify the ephone hunt group agent to receive parked calls that are forwarded to the secondary pilot number, use the secondary start command in ephone-hunt configuration mode. To disable this setting, use the no form of this command.

```
secondary start [{current|next}list-position]
no secondary start [{current|next}list-position]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>current</td>
<td>The ephone-dn that parked this call.</td>
</tr>
<tr>
<td>next</td>
<td>The ephone-dn that follows the parking ephone-dn in the list specified by the list command.</td>
</tr>
<tr>
<td>list-position</td>
<td>The ephone-dn at the specified position in the list specified by the list command. Range is from 1 to 20.</td>
</tr>
</tbody>
</table>

**Command Default**

No hunt-group agent is specified for receiving parked calls that are forwarded to the secondary pilot number.

**Command Modes**

Ephone-hunt configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

When a call that has been parked by a hunt group agent meets either of these conditions, the hunt group agent to receive the call can be specified with the secondary start command:

- The call is recalled from call park to the hunt group secondary pilot number.
- The call is transferred from call park to an ephone-dn that forwards the call to the hunt group secondary pilot number.

**Examples**

The following example specifies that the third hunt group member (3031) will receive calls that are recalled or forwarded to the secondary group hunt pilot number (3001) after the calls have been parked by an ephone-dn.

```
ephone-hunt 17 sequential
  pilot 3000 secondary 3001
  list 3011, 3021, 3031
  timeout 10
  final 7600
  secondary start 3
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-hunt</td>
<td>Defines an ephone hunt group and enters ephone-hunt configuration mode.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>---------</td>
<td>-------------</td>
</tr>
<tr>
<td>list</td>
<td>Creates a list of extensions that are members of an ephone hunt group</td>
</tr>
</tbody>
</table>
secondary-dialtone

To activate a secondary dial tone when a Cisco IP phone user dials a defined public switched telephone network (PSTN) access prefix, use the `secondary-dialtone` command in telephony-service configuration mode. To disable the secondary dial tone, use the `no` form of this command.

```
secondary-dialtone  digit-string
no  secondary-dialtone
```

**Syntax Description**

- `digit-string`: String of up to 32 numbers that defines an access prefix.

**Command Default**

No secondary dial tone is enabled.

**Command Modes**

- **Telephony-service configuration (config-ephone)**

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The secondary dial tone is turned off when the next number after the access prefix is pressed. For example, if 8 is the access prefix and a person dials 8 555-0145, the secondary dial tone is turned off when the digit 5 is pressed.

**Note**

The symbol # is considered to be the terminating string of a dial string. Hence, it is not supported under `secondary-dialtone`, to avoid conflict with dial-peer matching.

**Examples**

The following example enables a secondary dial tone when a Cisco IP phone users press the digit 9 to get an outside line:

```
Router(config)# telephony-service
Router(config-telephony)# secondary-dialtone 9
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>telephony-service</code></td>
<td>Enters telephony-service configuration mode.</td>
</tr>
</tbody>
</table>
secure-signaling trustpoint

To specify the name of the PKI trustpoint with the certificate to use for TLS handshakes with IP phones on TCP port 2443, use the `secure-signaling trustpoint` command in telephony-service configuration mode. To return to the default, use the `no` form of this command.

```
secure-signaling trustpoint label
no secure-signaling trustpoint
```

**Syntax Description**

| label | Name of a configured PKI trustpoint with a valid certificate. |

**Command Default**

No trustpoint is specified.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication to name the trustpoint that enables handshaking between Cisco Unified CME and a phone to ensure secure SCCP signaling on TCP port 2443.

**Examples**

The following example names server25, the CAPF server, as the trustpoint to enable secure SCCP signaling:

```
Router(config)# telephony-service
Router(config-telephony)# device-security-mode authenticated
Router(config-telephony)# secure-signaling trustpoint server25
Router(config-telephony)# tftp-server-credentials trustpoint server12
Router(config-telephony)# load-cfg-file slot0:Ringlist.xml alias Ringlist.xml sign create
Router(config-telephony)# exit
```
semi-attended enable (voice register template)

To enable call transfer at the alert call stage for supported SIP phones in Cisco Unified CME, use the **semi-attended enable** command in the voice register template mode. To disable call transfer, use the no form of this command.

**semi-attended enable**  
**no semi-attended enable**

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Call transfer at the alert call stage is enabled.

**Command Modes**
Voice register template (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command enables a call transfer at the alert stage in the specified template which can then be applied to SIP phones in Cisco Unified CME. Semi-attended call transfer is enabled by default. To disable semi-attended call transfer, use the no **semi-attended** command.

To apply the template to a SIP phone, use the **template** command in voice register pool configuration mode.

**Examples**
The following partial output from the **show-running config** command shows that the semi-attended call transfer is disabled in voice register template 1:

```
Router# show running-config
!
.
.
.
!
voice register template 1
 no semi-attended enabled
!
```

The following example shows how to enable semi-attended call transfer in a template:

```
Router(config)# voice register template 1
Router(config-register-temp)# semi-attend enable
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>template (voice register pool)</td>
<td>Applies template to SIP IP phone being configured.</td>
</tr>
</tbody>
</table>
server (CTL-client)

To enter trustpoint information for the CAPF server, Cisco Unified CME router, or TFTP server into the router configuration, use the `server` command in CTL-client configuration mode. To return to the default, use the `no` form of this command.

```
server {capf|cme|cme-tftp|tftp} ip-address trustpoint label
no server {capf|cme|cme-tftp|tftp} ip-address
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>capf</td>
<td>CAPF server.</td>
</tr>
<tr>
<td>cme</td>
<td>Cisco Unified CME router.</td>
</tr>
<tr>
<td>cme-tftp</td>
<td>Combined Cisco Unified CME router and TFTP server.</td>
</tr>
<tr>
<td>tftp</td>
<td>TFTP server.</td>
</tr>
<tr>
<td>ip-address</td>
<td>IP address of the entity.</td>
</tr>
<tr>
<td>trustpoint label</td>
<td>Name of the PKI trustpoint for the entity.</td>
</tr>
</tbody>
</table>

**Command Default**

Trustpoint information about the CAPF server, Cisco Unified CME router, or TFTP server is not present in the security configuration.

**Command Modes**

CTL-client configuration (config-ctl-client)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication. Cisco IOS software stores credential information in a trustpoint. The trustpoint label in this command names the specified PKI trustpoint.

**Note**

Repeat this command for each entity that requires a trustpoint.

**Examples**

The following example defines trustpoint names and IP addresses for the CAPF server, the Cisco Unified CME router, and the TFTP server:

```
Router(config)# ctl-client
Router(config-ctl-client)# server capf 10.2.2.2 trustpoint capftrust
Router(config-ctl-client)# server cme 10.2.2.3 trustpoint cmets
Router(config-ctl-client)# server tftp 10.2.2.4 trustpoint tftptp
Router(config-ctl-client)# sast1 trustpoint sast1tp
Router(config-ctl-client)# sast2 trustpoint sast2tp
```
Router(config-ctl-client)# regenerate
To specify the IP address of a presence server for sending presence requests from internal watchers to external presence entities, use the `server` command in presence configuration mode. To remove the server, use the `no` form of this command.

```
server ip-address
no server
```

### Syntax Description

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ip-address</code></td>
<td>IP address of the remote presence server.</td>
</tr>
</tbody>
</table>

### Command Default

A remote presence server is not used.

### Command Modes

Presence configuration (config-presence)

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command specifies the IP address of a presence server that handles presence requests when the watcher and presence entity (presentity) are not collocated. The router acts as the presence server and processes all presence requests and status notifications when a watcher and presentity are both internal. If a subscription request is for an external presentity, the request is sent to the remote server specified by this command.

### Examples

The following example shows a presence server with IP address 10.10.10.1:

```
Router(config)# presence
Router(config-presence)# allow subscribe
Router(config-presence)# server 10.10.10.1
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow subscribe</td>
<td>Allows internal watchers to monitor external presence entities (directory numbers).</td>
</tr>
<tr>
<td>allow watch</td>
<td>Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.</td>
</tr>
<tr>
<td>max-subscription</td>
<td>Sets the maximum number of concurrent watch sessions that are allowed.</td>
</tr>
<tr>
<td>show presence global</td>
<td>Displays configuration information about the presence service.</td>
</tr>
<tr>
<td>show presence subscription</td>
<td>Displays information about active presence subscriptions.</td>
</tr>
<tr>
<td>watcher all</td>
<td>Allows external watchers to monitor internal presence entities (directory numbers).</td>
</tr>
</tbody>
</table>
server-security-mode

To change the security mode of the Cisco Unified CME phone authentication server, use the `server-security-mode` command in telephony-service configuration mode. To change the mode from secure to nonsecure, use the `no` form of this command.

```
server-security-mode {erase|non-secure|secure}
no server-security-mode
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>erase</code></td>
<td>Deletes the certificate trust list (CTL) file.</td>
</tr>
<tr>
<td><code>non-secure</code></td>
<td>Enables nonsecure mode.</td>
</tr>
<tr>
<td><code>secure</code></td>
<td>Secure mode.</td>
</tr>
</tbody>
</table>

**Command Default**

When the CTL file is initially generated, the mode is set to secure.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(22)T</td>
<td>Cisco Unified CME 7.0</td>
<td>The <code>erase</code> keyword was added.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication.

This command has no effect until the CTL file is initially generated by the CTL client. When the CTL file is successfully generated, the CTL client automatically sets the server security mode to secure. You can then toggle the mode from secure to nonsecure using this command.

After toggling between secure and non-secure mode, you must use the `regenerate` command in CTL-client configuration mode to generate the CTL file. This is necessary because if the security mode is nonsecure, its credentials are zeroed out in the CTL file. If the security mode is secure, the CTL file contains the server’s credentials.

The `no` version of this command sets the mode to non-secure; it does not remove the command from your configuration.

To remove this command from your configuration and revert to the state before the Cisco Unified CME security feature was activated, use the `erase` keyword and follow the instructions displayed on the console. When you use this command with the `erase` keyword, the router checks whether the Cisco IOS CTL-provider process is running, and if not, it deletes the CTL file from router storage. After using this command to delete the CTL file, you must manually delete the CTL file from any SCCP phones that had downloaded it previously.

**Examples**

The following example changes the mode of the server to non-secure.

```
telephony-service
```

Cisco Unified Communications Manager Express Command Reference
server-security-mode non-secure

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>regenerate</td>
<td>Creates a new CTLFile.tlv file after changes are made to the CTL client configuration.</td>
</tr>
</tbody>
</table>
**service directed-pickup**

To enable Directed Call Pickup and modify the function of the GPickUp and PickUp soft keys, use the `service directed-pickup` command in telephony-service configuration mode. To disable Directed Call Pickup, use the `no` form of this command.

```
service directed-pickup  [gpickup]
noservice directed-pickup  [gpickup]
```

**Syntax Description**

`gpickup` (Optional) Enables phone users to perform Directed Call Pickup using the GPickUp soft key.

**Command Default**

For SCCP phones, Directed Call Pickup using the PickUp soft key is enabled.

For SIP phones, Directed Call Pickup is not enabled.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>The <code>gpickup</code> keyword and support for SIP phones was added.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command modifies the function of the GPickUp and PickUp soft keys for the Directed Call Pickup and Local Group Pickup features.

To globally disable Directed Call Pickup on all phones, use the `no` form of this command. The `no` form of this command also changes the behavior of the PickUp soft key on IP phones so that a user pressing it invokes Local Group Pickup instead of Directed Call Pickup.

To selectively remove the PickUp soft key from one or more SCCP phones, use the `features blocked` command in ephone-template mode. The `features blocked` command removes the PickUp soft key from SCCP IP phones and blocks Directed Call Pickup on analog phones to which you apply the template.

The table describes the use of the GPickUp and PickUp soft keys for each feature depending on the setting of this command.
### Table 14: service directed-pickup Command Comparison

<table>
<thead>
<tr>
<th>Cisco IOS Command Syntax SIP Phones</th>
<th>SCCP Phones</th>
<th>SIP Phones</th>
</tr>
</thead>
<tbody>
<tr>
<td>service directed-pickup gpickup</td>
<td>GPickUp soft key and extension</td>
<td>Directed Call Pickup (Call on any ringing extension)</td>
</tr>
<tr>
<td></td>
<td>GPickUp soft key and * or PickUp soft key</td>
<td>Local Group Pickup (Call in same group)</td>
</tr>
<tr>
<td></td>
<td>GPickUp soft key and pickup group number</td>
<td>Other Group Pickup (Call in different group)</td>
</tr>
<tr>
<td>service directed-pickup (default)</td>
<td>GPickUp soft key and extension</td>
<td>Directed Call Pickup</td>
</tr>
<tr>
<td></td>
<td>GPickUp soft key and * or PickUp soft key</td>
<td>Local Group Pickup</td>
</tr>
<tr>
<td></td>
<td>GPickUp soft key and pickup group number</td>
<td>Other Group Pickup</td>
</tr>
<tr>
<td>no service directed-pickup</td>
<td>—</td>
<td>Directed Call Pickup</td>
</tr>
<tr>
<td></td>
<td>—</td>
<td>Local Group Pickup</td>
</tr>
<tr>
<td></td>
<td>GPickUp soft key and * or PickUp soft key</td>
<td>Other Group Pickup</td>
</tr>
</tbody>
</table>

#### Example

The following example shows that Directed Call Pickup is disabled globally:

```
telephony-service
no service directed-pickup
```

The following example shows that Directed Call Pickup, Group Pickup, and Local Group Pickup can be performed using the GPickUp soft key:

```
telephony-service
no service directed-pickup
service directed-pickup
```

---

1 Supported in Cisco Unified CME 7.1 and later versions.
telephony-service
service directed-pickup gpickup

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-feature-uri</td>
<td>Creates a new CTLFile.tlv file after changes are made to the CTL client configuration. Specifies the uniform resource identifier (URI) for soft keys on SIP phones registered to Cisco Unified CME.</td>
</tr>
<tr>
<td>features blocked</td>
<td>Prevents one or more features from being used on SCCP phones.</td>
</tr>
<tr>
<td>pickup-group</td>
<td>Assigns an extension to a call-pickup group.</td>
</tr>
</tbody>
</table>
service dnis dir-lookup

To allow the display of names associated with called numbers for incoming calls on IP phones, use the service dnis dir-lookup command in telephony-service configuration mode. To deactivate directory lookup, use the no form of this command.

```
service dnis dir-lookup
noservice dnis dir-lookup
```

**Command Default**
The default is directory service lookup is inactive.

**Command Modes**
Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco CME 3.2</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
The service dnis dir-lookup command provides a called number to the name-lookup service to support display of the name associated with the called number for incoming calls to IP phones. The display name is obtained from the Cisco CME system’s list of Cisco CME directory names created using the directory entry command in the telephony-service configuration mode.

Called numbers can be displayed for overlaid ephone-dn and for ephone-dns that are not overlaid. Secondary line are supported.

To allow a single ephone-dn to receive calls for multiple different called numbers (with different names), you must use wildcard “.” characters in the number field for the ephone-dn.

To use the service dnis dir-lookup command in conjunction with the ephone-hunt, you can configure the ephone-hunt group to use a pilot number that contains wildcard “.” characters. This command allows the ephone-hunt group to receive calls from different numbers. Individual ephone-dns that are configured as members of the hunt group with the ephone-hunt list must not have wildcard characters in their number fields.

If the service dnis dir-lookup is used at the same time as the service dnis overlay, the directory-lookup service takes precedence in resolving the name for the called number.

**Examples**
The following is an example of an overlaid ephone-dn configuration, where the service dnis dir-lookup allows one of three directory entry names to be displayed on three IP phones when a call is placed to a number declared in the directory entry command.

```
telephony-service
  service dnis dir-lookup
  directory entry 1 0001 name dept1
  directory entry 2 0002 name dept2
  directory entry 3 0003 name dept3
  ephone-dn 1
    number 0001
  ephone-dn 2
    number 0002
  ephone-dn 2
    number 0002
```
The following is an example of an ephone-dn configuration in which the overlay function is not in use. There are three IP phones, each with two buttons. Button 1 receives calls from user1, user2, and user3; button 2 receives calls from user4, user5, and user6.

telephony-service
  service dnis dir-lookup
  directory entry 1 5550001 name user1
  directory entry 2 5550002 name user2
  directory entry 3 5550003 name user3
  directory entry 4 5550010 name user4
  directory entry 5 5550011 name user5
  directory entry 6 5550012 name user6
  ephone-dn 1
    number 555000.
  ephone-dn 2
    number 5552001.
  ephone 1
    button 1:1
    button 2:2
    mac-address 1111.1111.1111
  ephone 2
    button 1:1
    button 2:2
    mac-address 2222.2222.2222

The following is an example of a hunt-group configuration. There are three phones, each with two buttons, and each button is assigned two numbers. When a person calls 5550341, Cisco CME matches the hunt-group pilot secondary number (555....) and rings button 1 on one of the two phones and displays “user1.” The selection of the phone is dependent on the ephone-hunt settings.

telephony-service
  service dnis dir-lookup
  max-redirect 20
  directory entry 1 5550341 name user1
  directory entry 2 5550772 name user2
  directory entry 3 5550263 name user3
  directory entry 4 5550150 name user4
  ephone-dn 1
    number 1001
  ephone-dn 2
    number 1002
  ephone-dn 3
    number 1003
  ephone-dn 4
    number 1004
  ephone 1
    button 1:1,2
    button 2:3,4
    mac-address 1111.1111.1111
  ephone 2
    button 1:1,2
    button 2:3,4
    mac-address 2222.2222.2222
  ephone-hunt 1 peer
    pilot 1000 secondary 555....
list 1001, 1002, 1003, 1004
final 5556000
hops 5
preference 1
timeout 20
no-reg

The following is an example of a secondary-line configuration. Ephone-dn 1 can accept calls from extension 1001 and from 5550000, 5550001, and 5550002.

telephony-service
service dnis dir-lookup
directory entry 1 5550000 name doctor1
directory entry 2 5550001 name doctor2
directory entry 3 5550002 name doctor3
ephone-dn 1
  number 1001 secondary 555000.
ephone 1
  button 1:1
  mac-address 2222.2222.2222

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>directory entry</td>
<td>Adds an entry to a local phone directory that can be displayed on IP phones.</td>
</tr>
<tr>
<td>ephone-hunt</td>
<td>Enters ephone-hunt configuration mode to create a hunt group for use in a Cisco CME system.</td>
</tr>
<tr>
<td>list</td>
<td>Creates a list of extensions that are members of a Cisco CME ephone hunt group.</td>
</tr>
<tr>
<td>service dnis overlay</td>
<td>Allows an ephone-dn name to appear on receiving IP phones’ displays when the ephone-dn’s number is called.</td>
</tr>
</tbody>
</table>
service dnis overlay

To allow incoming calls to an ephone-dn overlay to display called ephone-dn names, use the service dnis overlay command in telephony-service configuration mode. To deactivate the service dialed number identification service (DNIS) overlay, use the no form of this command.

service dnis overlay
no service dnis overlay

Command Default

The ephone-dn names in an ephone-dn overlay are not displayed on IP phones.

Command Modes

Telephony-service configuration (config-telephony)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco CME Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco Unified CME 3.2</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

Usage Guidelines

The service dnis overlay allows phone users to determine which ephone-dn within an overlay set is being called. Up to ten ephone-dns are allowed per overlay set. When an incoming call is presented under a service dnis overlay configuration, the phone displays the name of the individual ephone-dn according to the name configured under the ephone-dn configuration mode. Note that for an ephone-dn name to be displayed on IP phones, you must first assign ephone-dn names with the name command in ephone-dn configuration mode.

The number of the first ephone-dn listed in the button is the default display for all phones using the same set of overlaid ephone-dns. Calls to the first ephone-dn display the caller ID. Calls to the remaining ephone-dns display ephone-dn names. For example, if there are three phones with one overlay set containing five ephone-dns, the first ephone-dn’s number listed is the default display for all three phones. A call to the first ephone-dn displays the caller ID on all phones until the call is picked up. When the call is answered by phone 1, the displays in phone 2 and phone 3 return to the default display. Calls to the last four ephone-dns display ephone-dn names.

If the service dnis overlay is used at the same time as the service dnis dir-lookup, the service dnis dir-lookup takes precedence in determining the name to be displayed.

Examples

The following is an overlay configuration for two phones with button 1 assigned to pick up three 800 numbers from three ephone-dns that have been assigned names. The default display for button 1 is 18005550100. A call to 18005550100 displays the caller ID. Calls to 18005550001 and 18005550002 display “name1” and “name2,” respectively.

telephony-service
service dnis overlay
ephone-dn 1
 name mainnumber
 number 18005550100
ephone-dn 2
 name name1
 number 18005550001
ephone-dn 3
 name name2
 number 18005550002
ephone 1
 button 101,2,3
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>name</td>
<td>Associates a name with a Cisco CME extension (ephone-dn).</td>
</tr>
<tr>
<td>service dnis dir-lookup</td>
<td>Allows directory entry lookup for the display of directory entry names on IP phones.</td>
</tr>
</tbody>
</table>
service dss

To enable DSS (Direct Station Select) in a Cisco Unified CME system, use the `service dss` command in telephony-service configuration mode. To globally disable the DSS feature, use the `no` form of this command.

```
service dss
no service dss
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

DSS service is disabled.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(6)XE</td>
<td>Cisco Unified CME 4.0(2)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC4</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)T</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>This command is integrated into Cisco IOS Release 12.4(11)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables phone users to quickly transfer calls to an extension selected by a speed-dial or monitor line button without having to press the Transfer button. If this command is enabled, a user can transfer a call when the call is in the connected state by simply pressing a speed-dial or monitor line button to select the transfer destination. The transfer action is automatically implied by CME if the `service dss` is enabled.

This feature is supported only on phones on which monitor-line buttons for speed dial or speed-dial line buttons are configured.

Using the `no` form of the changes the behavior of the speed-dial line button on all IP phones so that a user pressing a speed-dial button in the middle of a connected call will play out the speed-dial digits into the call without transferring the call. If the `service dss` is disabled, the phone user must press the Transfer button before pressing the speed-dial line button or monitor line button to transfer the call.

For Cisco Unified CME 4.0 and earlier, the `transfer-system full-consult dss` is used to select between blind transfers and consult transfers for the DSS case.

**Examples**

The following example globally enables directed call pickup.

```
telephony-service
service dss
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>button</td>
<td>Associates ephone-dns with individual buttons on a Cisco Unified IP phone and to specify line type, such as monitor mode for a shared line.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>----------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>speed-dial</td>
<td>Defines a unique speed-dial identifier, a digit string to dial, and an optional label to display next to a line button.</td>
</tr>
</tbody>
</table>
service https (ephone-template)

To locally provision Hypertext Transfer Protocol Secure (HTTPS) services access from Cisco Unified SCCP IP phones on Cisco Unified CME, use the `service https` command in ephone-template configuration mode. To disable access to HTTPS services, use the `no` form of this command.

```
service https
no service https
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Cisco Unified SCCP IP phones are unable to access HTTPS services on Cisco Unified CME.

**Command Modes**

Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `service https` command to enable access to HTTPS services like local-directory lookup, My Phone Apps, and Extension Mobility.

**Examples**

The following example shows how to locally provision HTTPS services from Cisco Unified SCCP IP phones:

```
configure terminal
ephone-template 1
service https
```

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ephone-template</code></td>
<td>Enters ephone-template configuration mode and creates an ephone template to configure a set of phone features.</td>
</tr>
</tbody>
</table>
service https (telephony-service)

To globally provision Hypertext Transfer Protocol Secure (HTTPS) services access from Cisco Unified SCCP IP phones on Cisco Unified CME, use the `service https` command in telephony-service configuration mode. To disable access to HTTPS services, use the `no` form of this command.

```
service https
no service https
```

### Syntax Description

This command has no arguments or keywords.

### Command Default

Cisco Unified SCCP IP phones are unable to access HTTPS services on Cisco Unified CME.

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use the `service https` command to enable access to HTTPS services like local-directory lookup, My Phone Apps, and Extension Mobility.

### Examples

The following example shows how to globally provision HTTPS services from Cisco Unified SCCP IP phones:

```
configure terminal
telephony-service
cnf-file perphone
service https
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>telephony-service</code></td>
<td>Enters telephony-service configuration mode.</td>
</tr>
</tbody>
</table>
service https (voice register global)

To globally provision Hypertext Transfer Protocol Secure (HTTPS) services access from Cisco Unified SIP IP phones on Cisco Unified CME, use the service https command in voice register global configuration mode. To disable access to HTTPS services, use the no form of this command.

```
service https
no service https
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Cisco Unified SIP IP phones are unable to access HTTPS services on Cisco Unified CME.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the service https command to enable access to HTTPS services like local-directory lookup, My Phone Apps, and Extension Mobility.

**Examples**

The following example shows how to globally provision HTTPS services from Cisco Unified SIP IP phones:

```
configure terminal
voice register global
   service https
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register global</td>
<td>Enters voice register global configuration mode and sets global parameters for all supported Cisco Unified SIP IP phones in a Cisco Unified CME or Cisco Unified SIP SRST environment.</td>
</tr>
</tbody>
</table>
service https (voice register template)

To locally provision Hypertext Transfer Protocol Secure (HTTPS) services access from Cisco Unified SIP IP phones on Cisco Unified CME, use the service https command in voice register template configuration mode. To disable access to HTTPS services, use the no form of this command.

service https
no service https

Syntax Description
This command has no arguments or keywords.

Command Default
Cisco Unified SIP IP phones are unable to access HTTPS services on Cisco Unified CME.

Command Modes
Voice register template configuration (config-register-temp)

Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

Usage Guidelines
Use the service https command to enable access to HTTPS services like local-directory lookup, My Phone Apps, and Extension Mobility.

Examples
The following example shows how to globally provision HTTPS services from Cisco Unified SIP IP phones:

configure terminal
voice register template 1
  service https

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register template</td>
<td>Enters voice register template configuration mode and defines a template of common parameters for SIP phones.</td>
</tr>
</tbody>
</table>
service local-directory

To enable the availability of the local directory service on IP phones served by the Cisco Unified Communications Manager Express (Unified CME) router, use the service local-directory command in telephony service configuration mode. To disable the display, use the no form of this command.

```
service local-directory [authenticate] [username] [password] [0|6] password
no service local-directory [authenticate] [username] [password]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>authenticate</td>
<td>(Optional) Requires authentication for local directory search requests.</td>
</tr>
<tr>
<td>username</td>
<td>(Optional) Provides username for authentication of local directory server.</td>
</tr>
<tr>
<td>password [0</td>
<td>6]</td>
</tr>
</tbody>
</table>

The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

**Command Default**

Local directory service is available on IP Phones.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modifications</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>The authenticate keyword was introduced.</td>
</tr>
<tr>
<td>12.3(4)</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco IOS XE Everest 16.6.1</td>
<td>Unified CME 12.0</td>
<td>This command was enhanced to authenticate the username and password for accessing the local directory service.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command was enhanced for password encryption, based on Unified CME password policy.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command with Cisco IOS Telephony Services V2.1, Cisco CME 3.0, or a later version.

When you configure the url directories command with the URL and credentials of the server that hosts the local directory, the command takes precedence over service local-directory[authenticate] [username][password]. When you configure the url directories command with only the URL of the server that hosts the local directory, Unified CME tries to fetch the username and password credentials from the command service local-directory[authenticate] [username][password], if it is configured.
From Unified CME 12.6 onwards, you must configure password encryption using the parameters \([0|6]\). This in accordance with Unified CME Password Policy. The 0 in the parameter \([0|6]\) mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

**Example 1**

The following example specifies that the directory service should not be available on the IP phones served by this ITS router:

```
Router(config)# telephony-service
Router(config-telephony-service)# no service local-directory
```

**Example 2**

The following example configures the username and password for accessing the server that hosts the directory service. In this scenario, the command `url directories` is not configured.

```
Router(config)# telephony-service
Router(config-telephony-service)# service local-directory authenticate admin cisco12345
```

The output for this sample configuration in the CNF XML file is as follows:

```
```

**Example 3**

The following example specifies the configuration when the command `service local-directory authenticate [username][password]` is configured and the command `url directories` is configured without credentials. In this scenario, the server URL is updated with the credentials provided in the `service local-directory` CLI command.

```
Router(config)# telephony-service
Router(config-telephony-service)# service local-directory authenticate root cisco
```

The output for this sample configuration in the CNF XML file is as follows:

```
```

**Example 4**

The following example specifies the configuration when the CLI commands `url directories` and `service local-directory authenticate [username][password]` are configured with credentials. In this scenario, the server URL is updated with the credentials provided in the `url directories` CLI command.

```
Router(config)# telephony-service
Router(config-telephony-service)# service local-directory authenticate admin cisco
Router(config-telephony-service)# url directories http://root:cisco123@8.39.16.26:80/localdirectory
```

The output for this sample configuration in the CNF XML file is as follows:

```
<directoryURL>http://root:cisco123@8.39.16.26:80/localdirectory</directoryURL>
```
Example 5

The following example specifies the configuration when the CLI command `service local-directory` is configured and the commands `url directories` and `service local-directory[authenticate] [username][password]` are not configured. In this scenario, the local directory service is activated though the credentials are not configured. Hence, the XML files generated by tftp-bindings will contain only the URL information of the server without the username and password credentials.

```
Router(config)# telephony-service
Router(config-telephony-service)# service local-directory
```

The output for this sample configuration in the CNF XML file is as follows:

```
Router# show telephony-service tftp-bindings
more flash:/its/vrf1/SEP5057A88797E0.cnf.xml
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>telephony-service</code></td>
<td>Enters telephony-service configuration mode.</td>
</tr>
</tbody>
</table>
service phone

To modify the vendorConfig parameters in the configuration file, use the service phone command in telephony-service or ephone-template configuration mode. To disable a setting, use the no form of this command.

```
service phone parameter-name parameter-value
no service phone parameter-name parameter-value
```

**Syntax Description**
- **parameter-name**: Name of the vendorConfig parameter in the configuration file. For valid parameter names, see the table below. Parameter names are word and case-sensitive and must be entered exactly as shown.
- **parameter-value**: Value for the vendorConfig parameter. For valid values and defaults, see the table below.

**Command Default**
The vendorConfig parameters in phone configuration files are set to default values.

**Command Modes**
- Telephony-service configuration (config-telephony)
- Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-template configuration mode for certain parameters.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>15.1(4)M</td>
<td>Cisco Unified CME 8.6</td>
<td>This command was modified. The xml config file argument was added.</td>
</tr>
<tr>
<td>Cisco IOS XE Fuji 16.9.1</td>
<td>Unified CME 12.3</td>
<td>This command service phone lineMode 1 introduces support for Enhanced Line Mode (ELM) for Cisco IP Phone 8800 Series on Unified CME.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command in telephony-service configuration mode modifies vendorConfig parameters in configuration file for phones in a Cisco Unified CME system.

The vendorConfig section of a configuration file is read by a phone’s firmware when that Cisco Unified IP phone is booted. The number and type of parameters may vary from one firmware version to the next.

If a firmware version does not support a particular parameter, that parameter cannot be implemented. For example, Cisco phone firmware 8.2.1 or a later version is required to support the G.722-64K codec on G.722-capable SCCP phones and Cisco phone firmware 8.3.1 or a later version is required to support the G.722-64K codec on G.722-capable SIP phones. If your phones are loaded with an earlier version of phone...
firmware, they cannot support the G.722-64K codec regardless of how the `g722CodecSupport` parameter is configured.

The IP phone that downloads the configuration file will implement only those parameters that it can support and ignore configured parameters that it cannot implement. For example, a Cisco IP phone without a backlight display cannot implement backlight parameters regardless of how they are configured.

In Cisco Unified CME 4.0 and later versions, support for creating configuration files at a phone level was added for SCCP phones. This command in ephone-template configuration mode creates an template of vendorConfig parameters that can be applied to individual SCCP phones in Cisco Unified CME. This command in ephone-template configuration mode does not work for all vendorConfig parameters. See the table below for information about individual parameters.

In Cisco Unified CME 4.0 and later versions, if you use an ephone template to apply this command to one or more phones, you must also configure the `cnf-file perphone` command so that a separate configuration file is created for each phone, by MAC address. To apply this command in telephony-service mode to all phones of a particular type in Cisco Unified CME 4.0 and later versions, you can configure the `cnf-file perphonetype` command to specify that configuration files are generated by phone type.

To apply this command in telephony-service configuration mode to all phones in your Cisco Unified CME system, ensure that the system is configured for the default single per-system configuration file for all phones.

If you use an ephone template to apply this command to a phone and you also use this command in telephony-service configuration mode for the same phone, the value that you set in ephone-template configuration mode has priority.

After modifying the vendorConfig parameters, you must generate new configuration files.

After generating configuration files, reset or reboot the IP phone to be configured to download the new configuration file.

From Unified CME Release 12.3, you can enable Enhanced Line Mode on Unified for Cisco IP Phone 8800 Series (except 8821, 8831, 8832 models) by configuring the CLI command `service phonelineMode 1` under `telephony-service` configuration mode. The Cisco IP Phone 8800 Series configured on Unified CME uses the vendor config XML body in the CNF file to verify if the CLI command `service phonelineMode 1` is added to enable ELM mode. By default, ELM is not enabled on Unified CME. To disable ELM on the Unified CME router, you need to configure `no service phonelineMode`.

---

**Note**

The parameters for the `service phone` CLI command are case sensitive. For example, the command to configure ELM for Cisco IP Phone Series 8800 must be `service phone lineMode 1`. If the command input is `service phone LineMode 1`, `service phone linemode 1`, and so on, ELM is not configured.

Use the `show telephony-service tftp-binding` command to view the SEP*.cnf.xml files that are associated with individual phones. The following example entry from a Sep*.conf.xml file disables the PC port on a phone:

```xml
<vendorConfig>
  <pcPort>1</pcPort>
</vendorConfig>
```

The below table lists the basic vendorConfig parameters in alphabetical order.
Parameter names are word and case-sensitive and must be typed exactly as shown.

### Table 15: vendorConfig Parameter-Name and Parameter-Value Descriptions

<table>
<thead>
<tr>
<th>Parameter Name and Value</th>
<th>Description</th>
</tr>
</thead>
</table>
| actionableAlert {0 | 1} | Replaces the traditional incoming call pop-up notification with an alert that you must respond to.  
  • 0—Disabled.  
  • 1—Enabled (default). |
| adminPassword password | (For the Cisco Unified IP Phone 7921G only) Creates a password for accessing the web interface on a phone.  
  • password—String of up to 32 characters. |
| autoSelectLineEnable {0 | 1} | Enables and disables auto line selection.  
  • 0—Disabled.  
  • 1—Enabled (default). |
| backlightIdleTimeout HH:MM | Sets the length of time in hours and minutes after which the backlighting of the IP phone displays will switch off again once the phone is inactive.  
  • This parameter is applicable only on the days specified using the `daysBacklightNotActive` parameter.  
  • This parameter does not affect the display during the period of time specified using the `backlightOnDuration` parameter.  
  • Hour (HH) and minute (MM). You must enter all four characters. For example, 9:05 a.m. must be entered as 09:05. Default is one hour (01:00). |
| backlightOnDuration HH:MM | Sets the length of time in hours and minutes for which IP phone displays will be backlit.  
  • This parameter does not affect the display on the days specified using the `daysBacklightNotActive` parameter.  
  • Hour (HH) and minute (MM). You must enter all four characters. For example, 9:05 a.m. must be entered as 09:05. Default is 10 hours (10:00). |
| backlightOnTime HH:MM | Sets the time of day at which backlighting of the IP phone displays is switched on, using a 24-hour time format.  
  • This parameter does not affect the display on the days specified using the `daysBacklightNotActive` parameter.  
  • Hour (HH) and minute (MM). You must enter all four characters. For example, 9:05 a.m. must be entered as 09:05. Default is 07:30. |
| daysBacklightNotActive number,[number...] | Sets the days of the week on which backlighting of the IP phone displays is switched off unless there is user interaction with the IP phone.  
  • number—Represents the days of the week numerically, starting with Sunday (1) and ending with Saturday (7). Each number must be separated with a comma, without spaces:  
    ```
    daysBacklightNotActive 1,2,3.
    ```  
  • Default is no backlighting on Sun (1) and Sat (7). |
<table>
<thead>
<tr>
<th>Parameter Name and Value</th>
<th>Description</th>
</tr>
</thead>
</table>
| ```daysDisplayNotActive number[^number...[^number]]``` | Sets the days of the week on which IP phone displays will be blank.  
- **number**—Represents the days of the week numerically, starting with Sunday (1) and ending with Saturday (7). Each number must be separated with a comma, without spaces:  
  ```daysDisplayNotActive 1,2,3```  
- Default is an inactive display on Sun (1) and Sat (7).  
- To disable this parameter so that IP phone displays are always active, configure this parameter in telephony-service configuration mode using a space plus a comma (,): `daysDisplayNotActive ,`  
  for the **parameter-value.**  
  **Note**  
  This parameter is not supported in ephone-template configuration mode. |
| ```displayIdleTimeout HH:MM``` | Sets the length of time in hours and minutes for which IP phone displays will remain active, starting from the last time that the phone was used.  
- Hour (**HH**) and minute (**MM**). You must enter all four characters. For example, 9:05 a.m. must be entered as 09:05. Default is one hour (01:00).  
  **Note**  
  This parameter is not supported in ephone-template configuration mode. |
| ```displayOnDuration HH:MM``` | Sets the length of time in hours and minutes for which IP phone displays will be active.  
- Hour (**HH**) and minute (**MM**). You must enter all four characters. For example, 9:05 a.m. must be entered as 09:05. Default is 10 hours (10:00).  
  **Note**  
  This parameter is not supported in ephone-template configuration mode. |
| ```displayOnTime HH:MM``` | Sets the time of day at which IP phone displays are activated, using a 24-hour time format.  
- Hour (**HH**) and minute (**MM**). You must enter all four characters. For example, 9:05 a.m. must be entered as 09:05. Default is 07:30.  
  **Note**  
  This parameter is not supported in ephone-template configuration mode. |
| ```displayOnWhenIncomingCall {0 | 1}``` | Enables and disables an IP phone display to be activated when an incoming call is received (Line state is Ring in). The display will switch off again once the ringing stops if the user has not touched the phone and if the phone display is supposed to be off.  
- **0**—Disabled (default).  
- **1**—Enabled.  
  **Note**  
  This parameter is not supported in ephone-template configuration mode. |
| ```disableSpeaker {true | false}``` | Enables and disables the speakerphone.  
- **true**—Disabled.  
- **false**—Enabled (default). |
| ```disableSpeakerAndHeadset {true | false}``` | Enables and disables the speakerphone and headset.  
- **true**—Disabled.  
- **false**—Enabled (default). |
<table>
<thead>
<tr>
<th>Parameter Name and Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>enableGroupListen</strong> {true</td>
<td>Enables and disables Group Listen mode in which the handset and speaker are both active to allow multiple listeners to hear the conversation over the speaker while one user talks on the handset.</td>
</tr>
<tr>
<td>false}</td>
<td>• <strong>true</strong>—Enabled.</td>
</tr>
<tr>
<td></td>
<td>• <strong>false</strong>—Disabled (default).</td>
</tr>
<tr>
<td><strong>forwardingDelay</strong> {0</td>
<td>Enables and disables the activation of the IP phone’s PC Ethernet switch port when the IP phone boots to prevent Ethernet traffic from interfering with the bootup process.</td>
</tr>
<tr>
<td>1}</td>
<td>• 0—Disabled.</td>
</tr>
<tr>
<td></td>
<td>• 1—Enabled (default).</td>
</tr>
<tr>
<td><strong>garp</strong> {0</td>
<td>Enables and disables IP phone response to gratuitous Address Resolution Protocol (ARP) messages from the IP phone’s Ethernet interface.</td>
</tr>
<tr>
<td>1}</td>
<td>• 0—Disabled.</td>
</tr>
<tr>
<td></td>
<td>• 1—Enabled (default).</td>
</tr>
<tr>
<td><strong>g722CodecSupport</strong> {0</td>
<td>Enables and disables the registration of the G.722 codec on the IP phone.</td>
</tr>
<tr>
<td>1</td>
<td>2}</td>
</tr>
<tr>
<td></td>
<td>• 1—Disabled. Disables G.722-64K2 codec on phone.</td>
</tr>
<tr>
<td></td>
<td>• 2—Enabled. Enables G.722-64K codec on phone.</td>
</tr>
<tr>
<td><strong>handsetWidebandEnable</strong> {0</td>
<td>Enables or disables wideband handset option on supported IP phones.</td>
</tr>
<tr>
<td>1</td>
<td>2}</td>
</tr>
<tr>
<td></td>
<td>• 0—Phone default (default), equal to disabled or enabled and set by manufacturer.</td>
</tr>
<tr>
<td></td>
<td>• 1—Enabled. Enables wideband handset on phone.</td>
</tr>
<tr>
<td></td>
<td>• 2—Disabled. Disables wideband handset on phone.</td>
</tr>
<tr>
<td></td>
<td>• Wideband handset should only be used on supported IP phones with firmware version 8.3 or a later version.</td>
</tr>
<tr>
<td><strong>handsetWidebandUIControl</strong></td>
<td>Enables or disables control of handset options by phone user.</td>
</tr>
<tr>
<td>{0</td>
<td>1}</td>
</tr>
<tr>
<td></td>
<td>• 1—Disabled.</td>
</tr>
<tr>
<td><strong>headsetWidebandEnable</strong> {0</td>
<td>Enables or disables wideband headset option on supported IP phones.</td>
</tr>
<tr>
<td>1}</td>
<td>• If the <strong>headsetWidebandUIControl</strong> parameter is set to Enable (0), the option set in the phone UI, by the phone user, has priority over the value set for this parameter.</td>
</tr>
<tr>
<td></td>
<td>• 0—Enabled (default). Enables wideband headset on phone.</td>
</tr>
<tr>
<td></td>
<td>• 1—Disabled. Disables wideband headset on phone.</td>
</tr>
<tr>
<td></td>
<td>• Wideband handset should only be used on supported IP phones with firmware version 8.3 or a later version.</td>
</tr>
<tr>
<td>Parameter Name and Value</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------</td>
<td>-------------</td>
</tr>
<tr>
<td><strong>headsetWidebandUIControl</strong> {0</td>
<td>1}</td>
</tr>
<tr>
<td><strong>homeScreen</strong> {0</td>
<td>1}</td>
</tr>
<tr>
<td><strong>LineKeyBarge</strong> {0</td>
<td>2}</td>
</tr>
<tr>
<td><strong>lineMode</strong> { 1}</td>
<td>(For Cisco IP Phone 8800 Series only) Enables Enhanced Line Mode (ELM) for Unified CME. The no form of the command disables the ELM functionality. By default, ELM is disabled for Unified CME.</td>
</tr>
<tr>
<td><strong>loadServer</strong> [hostname</td>
<td>IPaddress]</td>
</tr>
</tbody>
</table>

**Note**

If the firmware file is not found, the firmware will not install. The phone will not be redirected to the TFTP server specified by the **option 150 ip** command.

- **hostname**—Name of the server from which the IP phone must retrieve phone firmware. Maximum length: 256 characters.
- **IPaddress**—IP address of server from which the IP phone must retrieve phone firmware.
- To disable this command and redirect the phone to use the TFTP server specified by the **option 150 ip** command to obtain its load files and upgrades, use this parameter name without the **hostname** or **IPaddress** argument.
<table>
<thead>
<tr>
<th>Parameter Name and Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>pcPort {0</td>
<td>1}</td>
</tr>
<tr>
<td></td>
<td>• 0—Enabled (default).</td>
</tr>
<tr>
<td></td>
<td>• 1—Disabled.</td>
</tr>
<tr>
<td>PushToTalkURL url</td>
<td>(For the Cisco Unified IP Phone 7921G only) Provisions the URL to be contacted for application services such as Push-To-Talk services.</td>
</tr>
<tr>
<td></td>
<td>• url—URL as defined in RFC 2396. Maximum length is 256 characters.</td>
</tr>
<tr>
<td>settingsAccess {0</td>
<td>1</td>
</tr>
<tr>
<td></td>
<td>• 0—Disabled.</td>
</tr>
<tr>
<td></td>
<td>• 1—Enabled (default). The phone user can modify features by using the Settings menu.</td>
</tr>
<tr>
<td></td>
<td>• 2—Restricted. The phone user is allowed to access User Preferences and volume settings only.</td>
</tr>
<tr>
<td>spanToPCPort {0</td>
<td>1}</td>
</tr>
<tr>
<td></td>
<td>• 0—Enabled (default).</td>
</tr>
<tr>
<td></td>
<td>• 1—Disabled.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> The path must be disabled to support Desktop Monitoring and Recording in a Cisco UCCX/Cisco Unified CME integration.</td>
</tr>
<tr>
<td>specialNumbers number[number...]</td>
<td>(For the Cisco Unified IP Phone 7921G only) Identifies a number that can be dialed on a phone regardless of whether the phone is locked or unlocked. For example, in the United States, the 911 emergency number is a good special number candidate to be dialed without unlocking the phone.</td>
</tr>
<tr>
<td></td>
<td>• number—Numerical string. Maximum length: 16 characters.</td>
</tr>
<tr>
<td></td>
<td>• To identify more than one special number, separate the numbers with a comma (,). Do not include spaces between numbers.</td>
</tr>
<tr>
<td></td>
<td>• The following example shows how to configure 411, 511, and 911 as special numbers:</td>
</tr>
</tbody>
</table>
|                         | ```
|                         | Router(config)# telephony-service
|                         | Router(config telephony-service)# service phone
<p>|                         | specialNumbers 411,511,911 |
| sshAccess {0 | 1}       | Enables and disables SSH access. |
|                         | • 0—Enabled (default). |
|                         | • 1—Disabled. |
| thumbButton1 PTTHbutton_number | (For Cisco Unified Wireless IP Phone 7921 and 7925) Associates thumb button on Cisco wireless IP phone with a phone button for one-way Push-To-Talk (PTT) functionality in Cisco Unified CME without requiring an external server. |
|                         | • button_number—Button on phone that is configured with an intercom dn that targets a paging number when user presses the thumb button. Range is 1 to 6. |
|                         | • The PTTHbutton_number keyword/argument combination is a contiguous character string and cannot contain spaces. |
|                         | • Implemented on supported phones with firmware version 1.0.4 or a later version. |</p>
<table>
<thead>
<tr>
<th>Parameter Name and Value</th>
<th>Description</th>
</tr>
</thead>
</table>
| videoCapability {0 | 1} | Enables and disables video capability for all applicable IP phones associated with a Cisco Unified CME router.  
- 0—Disabled (default).  
- 1—Enabled.  
- After using this parameter to enable video at a system level, you must configure the video command in ephone configuration mode for each video-capable phone.  
*Note* This parameter is not supported in ephone-template configuration mode. |
| voiceVlanAccess {0 | 1} | Enables and disables spanning, which is the IP phone’s access to the voice VLAN of the PC to which the IP phone’s Ethernet port is connected.  
- 0—Enabled (default).  
- 1—Disabled.  
*Note* For Cisco Unified IP Phone 7985, the default is Disabled (1). |
| webAccess {0 | 1 | 2} | Enables and disables web access that allows phone users to configure settings and features on User Option web pages.  
- 0—Enabled (default).  
- 1—Disabled.  
- 2—Read Only. For the Cisco Unified IP Phone 7921G only. The phone user can view only User Option web pages and cannot modify settings and features on the pages.  
*Note* For the Cisco Unified IP Phone 7921G, the default is Read Only (2). |
| WLanProfile tag {0 | 1} | (For Cisco Unified IP Phone 7921G only) Locks or unlocks a specific profile.  
- tag—Unique number assigned to profile. Range is 1 to 4.  
- 0—Locked (default).  
- 1—Unlocked. User can modify a profile.  
- Repeat this command for each profile to be locked or unlocked. |

### Examples

The following example shows how to configure multiple *service phone* parameters. This configuration is applied only in as much as IP phone firmware supports each parameter.

```
Router(config)# telephony-service
Router(config-telephony)# service phone disableSpeaker true
Router(config-telephony)# service phone disableSpeakerAndHeadset true
Router(config-telephony)# service phone forwardingDelay 1
Router(config-telephony)# service phone pcPort 1
Router(config-telephony)# service phone voiceVlanAccess 0
Router(config-telephony)# service phone settingsAccess 1
Router(config-telephony)# service phone videoCapability 1
Router(config-telephony)# service phone daysDisplayNotActive 1,7
Router(config-telephony)# service phone displayOnTime 07:30
Router(config-telephony)# service phone displayOnDuration 10:00
Router(config-telephony)# service phone displayIdleTimeout 01:00
Router(config-telephony)# service phone daysBacklightNotActive 1,7
Router(config-telephony)# service phone backlightOnTime 07:30
```
The following example shows how to set the default values for backlighting the phone display for all Cisco Unified IP phones with backlight capabilities in Cisco Unified CME:

Router(config) telephony-service
Router(config-telephony) service phone backlightOnDuration 10:00
Router(config-telephony) service phone backlightIdleTimeout 01:00
Router(config-telephony) create cnf-files
Router(config-telephony) reset all

The following example shows how to set the backlighting parameters so that there is no backlighting of the phone display for all Cisco Unified IP phones with backlight capabilities until there is user interaction with the phone. The backlightIdleTimeout parameter is configured so that the backlight will switch off again after 60 seconds of inactivity.

Router(config) telephony-service
Router(config-telephony) service phone daysBacklightNotActive 1,7
Router(config-telephony) service phone backlightOnTime 07:30
Router(config-telephony) service phone backlightOnDuration 10:00
Router(config-telephony) service phone backlightIdleTimeout 00.01
Router(config-telephony) create cnf-files
Router(config-telephony) reset all

The following examples show how to set the display parameters so that the phone display for all Cisco Unified IP phones with luminous displays are blank on Sunday (1), Monday (2), and Saturday (7):

Router(config) telephony-service
Router(config-telephony) service phone daysDisplayNotActive 1,2,7
Router(config-telephony) create cnf-files
Router(config-telephony) reset all

The following example shows how to disable the PC port on an individual IP phone (ephone 15) using an ephone template:

Router(config) ephone-template 8
Router(config-ephone-template) service phone pcPort 1
Router(config-ephone-template) exit
Router(config) ephone 15
Router(config-ephone) ephone-template 8
Router(config-ephone) exit
Router(config) telephony-service
Router(config-telephony) create cnf-files
Router(config-telephony) exit
Router(config) ephone 15
Router(config-ephone) reset

The following examples show how to enable ELM on Unified CME for Cisco IP Phones. Also, it provides steps to configure create profile and restart the phones under voice register global configuration mode to enable ELM for the Cisco IP Phone 8800 series phones on Unified CME:

Router(config) telephony-service
Router(config-telephony) service phone lineMode ?
   WORD enter the phone xml file parameter text for the previously entered parameter name
Router(config-telephony)#service phone lineMode 1
Router(config-telephony)#create cnf-files
Router(config-telephony)#end

Router(config)#voice register global
Router(config-register-global)#create profile
Router(config-register-global)#restart
Router(config-register-global)#end

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>cnf-file</td>
<td>Specifies that separate configuration files be generated for individual SCCP phones or types of SCCP phones.</td>
</tr>
<tr>
<td>create cnf-files</td>
<td>Builds XML configuration files that set IP phone displays and functionality.</td>
</tr>
<tr>
<td>create profile</td>
<td>Generates configuration profile files required for SIP phones.</td>
</tr>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies a template to the ephone being configured.</td>
</tr>
<tr>
<td>reset (telephony-service)</td>
<td>Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td>show telephony-service tftp-binding</td>
<td>Displays the current configuration files accessible to IP phones.</td>
</tr>
<tr>
<td>show voice register tftp-bind</td>
<td>Displays the current configuration files accessible to IP phones.</td>
</tr>
<tr>
<td>video (ephone)</td>
<td>Enables video capabilities on specified phones.</td>
</tr>
</tbody>
</table>
**service profile**

To set the parameters under the commonProfile section in IP phone SEP*.cnf.xml configuration files, use the `service profile` command in telephony-service configuration mode. To disable the settings, use the `no` form of this command.

```
service profile [{phonePassword password|callLogBlfEnabled|backgroundImageAccess false}]
```

**no service profile [{phonePassword password|callLogBlfEnabled|backgroundImageAccess false}]**

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>phonePassword password</td>
<td>Enters the phone password.</td>
</tr>
<tr>
<td>callLogBlfEnabled</td>
<td>Enables the call log.</td>
</tr>
<tr>
<td>backgroundImageAccess false</td>
<td>Disables the background image access.</td>
</tr>
</tbody>
</table>

**Command Default**

Parameters in the commonProfile section in IP phone SEP*.cnf.xml configuration files are not set.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T2</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

You can use the `service profile` command to set the parameters under the commonProfile section in IP phone SEP*.cnf.xml configuration files. Invoke the `create cnf-file` command to update phone configuration files.

**Examples**

The following example shows the `service profile` command is used at the router prompt:

```
Router# configure terminal
Router(config)# telephony-service
Router(config-telephony)# service profile phonePassword cisco
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>telephony-service</td>
<td>Enters telephony-service configuration mode.</td>
</tr>
</tbody>
</table>
**service-digit**

To enable phone users to dial a service digit to request off-net services, use the `service-digit` command in voice MLPP configuration mode. To reset to the default, use the `no` form of this command.

```
service-digit
no service-digit
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Service digit is disabled.

**Command Modes**

Voice MLPP configuration (config-voice-mlpp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables users to request off-net services by dialing a service digit, after dialing the MLPP access digit. The service digit provides information to the switch when connecting calls to government or public telephone services or networks that are not part of the Defense Switched Network (DSN).

Phone users request a service by dialing the access code NS, where N is the preconfigured MLPP access digit and S is the service digit. The service digit is a number from 5 to 9.

In Cisco Unified CME, the dial plan must be configured to play secondary dial-tone and the rest of the dialed digits are collected and passed to the off-net trunk. The digits that follow the prefix NS must be E.164 compliant.

**Examples**

The following example shows how to enable users to dial a service digit:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# service-digit
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>access-digit</td>
<td>Defines the access digit that phone users dial to request a precedence call.</td>
</tr>
<tr>
<td>mlpp preemption</td>
<td>Enables preemption capability on an SCCP phone or analog FXS port.</td>
</tr>
</tbody>
</table>
service-enable (auto-register)

To re-enable the auto-registration of SIP phones on Unified CME that is temporarily disabled, use the `service-enable` command in voice auto register configuration mode. This command is a sub-mode CLI of the command `auto-register`. To temporarily disable the auto registration process without losing configurations such as password and DN range, use the `no` form of this command.

```
  service-enable
  no service-enable
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>service-enable</code></td>
<td>Temporarily disables the auto registration process, but retains the password and DN range configurations. Once auto-register command is entered, the service is enabled by default.</td>
</tr>
<tr>
<td><code>no service-enable</code></td>
<td></td>
</tr>
</tbody>
</table>

**Command Default**

By default, this command is enabled.

**Command Modes**

voice auto register configuration (config-voice-auto-register)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.6(3)M</td>
<td>Cisco Unified CME 11.5</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>16.3.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is enabled by default.

If the administrator needs to temporarily disable or enable auto registration without losing configurations such as DN range, and password, the `no` form of this command, `no service-enable` is used.

**Examples**

The following example shows how to temporarily disable auto registration using the `no` form of the `service-enable` option:

```
Router(config)#voice register global
Router(config-register-global)#auto-register
Router(config-voice-auto-register)#?

VOICE auto register configuration commands:
  auto-assign Define DN range for auto assignment
  default Set a command to its defaults
  exit Exit from voice register group configuration mode
  no Negate a command or set its defaults
  password Default password for auto-register phones
  service-enable Enable SIP phone Auto-Registration
  template Default template for auto-register phones

Router(config-voice-auto-register)#no service-enable ?
<cr>
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>auto-register</td>
<td>Enables automatic registration of SIP phones with the Cisco Unified CME system.</td>
</tr>
<tr>
<td>password (auto-register)</td>
<td>Configures the mandatory password that administrator sets for auto registration of SIP phones on Unified CME.</td>
</tr>
<tr>
<td>auto-assign (auto-register)</td>
<td>Configures the mandatory range of directory numbers for phones auto registering on Unified CME.</td>
</tr>
<tr>
<td>template (auto-register)</td>
<td>Creates a basic configuration template that supports all the configurations available on the voice register template.</td>
</tr>
<tr>
<td>auto-reg-ephone</td>
<td>Enables automatic registration of ephones with the Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
**service-domain**

To set the global MLPP domain type and number, use the `service-domain` command in voice MLPP configuration mode. To reset to the default, use the `no` form of this command.

```
service-domain {drsn|dsn} identifier domain-number
no service-domain
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>drsn</td>
<td>Defense Red Switched Network (DRSN).</td>
</tr>
<tr>
<td>dsn</td>
<td>Defense Switched Network (DSN). This is the default value.</td>
</tr>
<tr>
<td>domain-number</td>
<td>Number to identify the global domain, in three-octet format. Range: 0x000000 to 0xFFFFFFFF. Default: 0.</td>
</tr>
</tbody>
</table>

**Command Default**

Domain type is `dsn`; domain number is 0.

**Command Modes**

Voice MLPP configuration (config-voice-mlpp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command sets the global domain type and number in Cisco Unified CME. Use the `mlpp service-domain` command to assign registered phones to different service domains. Any phone not configured with a specific service domain uses this global domain for MLPP calls.

**Examples**

The following example shows the global domain set to DSN with identifier 0010:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# service-domain dsn identifier 0010
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>mlpp service-domain</code></td>
<td>Sets the service domain and maximum precedence (priority) level for MLPP calls.</td>
</tr>
<tr>
<td><code>preemption trunkgroup</code></td>
<td>Enables preemption capabilities on a trunk group.</td>
</tr>
<tr>
<td><code>service-domain (voice class)</code></td>
<td>Sets the service domain name in the MLPP voice class.</td>
</tr>
</tbody>
</table>
service-domain (voice class)

To set the service domain name in the MLPP voice class, use the service-domain command in voice class configuration mode. To reset to the default, use the no form of this command.

```
service-domain {drsn|dsn}
no service-domain
```

**Syntax Description**

<table>
<thead>
<tr>
<th>dsn</th>
<th>Defense Red Switched Network (DRSN).</th>
</tr>
</thead>
<tbody>
<tr>
<td>dsn</td>
<td>Defense Switched Network (DSN).</td>
</tr>
</tbody>
</table>

**Command Default**

Domain name is dsn.

**Command Modes**

Voice class configuration (config-class)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command sets the domain name that is used for off-net MLPP calls.

After using this command, assign the voice class to an outbound POTS or VoIP dial peer by using the voice-class mlpp command.

**Examples**

The following example shows the domain name set to DSN:

```
Router(config)# voice class mlpp
Router(config-class)# service-domain dsn
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>mlpp service-domain</td>
<td>Sets the domain number and maximum precedence (priority) level for an MLPP call.</td>
</tr>
<tr>
<td>service-domain</td>
<td>Sets the global MLPP domain type and number.</td>
</tr>
<tr>
<td>voice-class mlpp</td>
<td>Assigns an MLPP voice class to a POTS or VoIP dial peer.</td>
</tr>
</tbody>
</table>
service-domain midcall-mismatch

To define the behavior when there is a domain mismatch between the two legs of a call, use the service-domain midcall-mismatch command in voice MLPP configuration mode. To reset to the default, use the no form of this command.

```
service-domain midcall-mismatch {method1|method2|method3|method4}
no service-domain midcall-mismatch
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>method1</td>
<td>Domain remains unchanged for each of the connections and the precedence level of the lower priority call changes to that of the higher priority call. This is the default value.</td>
</tr>
<tr>
<td>method2</td>
<td>Domain and precedence level of the lower priority call changes to that of the higher priority call.</td>
</tr>
<tr>
<td>method3</td>
<td>Domain remains unchanged for each of the connections and the precedence levels change to Routine for both calls.</td>
</tr>
<tr>
<td>method4</td>
<td>Domains change to that of the connection for which supplementary service was invoked (for example, transferee in case of transfer). Precedence levels change to Routine for both calls.</td>
</tr>
</tbody>
</table>

### Command Default

The default is **method1**.

### Command Modes

Voice MLPP configuration (config-voice-mlpp)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command determines the service domain and precedence level to apply in the case of a mismatch of these values between the two connections (call legs) of a call. This typically occurs when supplementary services such as Call Transfer or Conferencing are invoked during a call.

### Examples

The following example shows the domain mismatch method set to 2:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# service-domain midcall-mismatch method2
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>mlpp service-domain</td>
<td>Sets the domain number and maximum precedence (priority) level for an MLPP call.</td>
</tr>
<tr>
<td>preemption trunkgroup</td>
<td>Enables preemption capabilities on a trunk group.</td>
</tr>
<tr>
<td>service-domain</td>
<td>Sets the default MLPP domain name and number.</td>
</tr>
</tbody>
</table>
**session-server**

To specify a session manager to manage and monitor Register and Subscribe messages during a feature-server session, use the `session-server` command in voice register dn configuration mode, voice register pool configuration mode, or ephone-dn configuration mode. To return to the default, use the no form of this command.

```plaintext
session-server session-server-tag[,...session-server-tag]
no session-server session-server-tag
```

**Syntax Description**

| session-server-tag | Unique identifier of previously configured session manager in Cisco Unified CME. Range: 1 to 8. When configured in voice register dn configuration mode or in ephone-dn configuration mode, this argument can contain up to eight session-server-tags, separated by commas (,). |

**Command Default**

Session manager is not assigned.

**Command Modes**

- Ephone-dn configuration (ephone-dn)
- Voice register dn configuration (voice-register-dn)
- Voice register pool configuration (voice-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XW2</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Cisco Unified CME 4.2 and later versions provide a general interface for interoperating with external feature servers, such as the Cisco Unified CCX application on Cisco CRS, including call monitoring and device monitoring based on SIP presence and dialog event package. A session manager in Cisco Unified CME can manage and monitor Register and Subscribe messages.

Before configuring this command, a session manager must already configured in Cisco Unified CME by using the `voice register session-server` command.

Use the `session-server` command in voice register pool configuration mode to specify that Register and Subscribe messages for an external feature-server route point must contain a Cisco-referenceID field. Registration or subscription will be granted only for the specified route point. The route point for which Register and Subscribe messages are to be managed by this session manager must already be configured as a SIP endpoint in Cisco Unified CME. Typically, the configuration for the route point is provided from the feature server. If the configuration for the route point is deleted or must be modified, it can be reconfigured directly in Cisco Unified CME by using Cisco IOS commands. Each route point can be managed by only one session manager. Each session manager can manage multiple route points.
Use the `session-server` command in ephone-dn configuration mode or in voice register dn configuration mode to specify that Subscribe messages for a directory number must contain a Cisco-referenceID field. Registration or subscription will be granted only for the specified directory number. The directory number for which Subscribe messages are to be monitored by this session manager must already be configured in Cisco Unified CME. Each directory number can be monitored by up to eight session managers. Each session manager can subscribe for multiple directory numbers.

**Examples**

The following example shows the configuration for specifying that session manager 1 can control a route point (voice register pool) for an external feature server:

```plaintext
voice register pool 1
    session-server 1
```

The following example shows the configuration specifying which session managers can monitor Register and Subscribe messages to directory numbers associated with Cisco Unified CCX agent phones. Notice that several session managers (1, 3, 5, and 7) can subscribe for both directory numbers.

```plaintext
ephone-dn 1
    session-server 1,2,3,4,5,6,7,8
.
ephone-dn 2
    session-server 1,3,5,7
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>voice register session-server</code></td>
<td>Enters voice register session configuration mode for the purpose of configuring a session manager.</td>
</tr>
</tbody>
</table>
session-transport

To specify the transport layer protocol that a SIP phone uses to connect to Cisco Unified CME, use the `session-transport` command in voice register pool or voice register template configuration mode. To reset to the default value, use the `no` form of this command.

```
session-transport {tcp|udp}
no session-transport
```

**Syntax Description**

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>tcp</td>
<td>Transmission Control Protocol (TCP) is used.</td>
</tr>
<tr>
<td>udp</td>
<td>User Datagram Protocol (UDP) is used. This is the default.</td>
</tr>
</tbody>
</table>

**Command Default**

UDP is the default protocol.

**Command Modes**

Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command sets the transport layer protocol parameter in the phone’s configuration file.

If you use a voice register template to apply a to a phone and you also use the same command in voice register pool configuration mode for the same phone, the value that you set in voice register pool configuration mode has priority.

This command is not supported on the Cisco Unified IP Phone 7905, 7912, 7940, or 7960.

**Note**

Although this command is not supported for the Cisco Unified IP Phone 7905, 7912, 7940, or 7960, it can be used to assign TCP as the session transport type for these phones. If TCP is selected for an unsupported phone using this command, calls to that phone will not complete successfully. The phone can originate calls but it uses UDP, although TCP has been assigned.

**Examples**

The following example sets the transport layer protocol to TCP for SIP phone 10:

```
Router(config)# voice register pool 10
Router(config-register-pool)# session-transport tcp
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>create profile</td>
<td>Generates the configuration profile files required for SIP phones.</td>
</tr>
<tr>
<td>show sip-ua status</td>
<td>Displays the status of SIP call service on a SIP gateway.</td>
</tr>
<tr>
<td>template (voice register pool)</td>
<td>Applies template to voice register pool being configured.</td>
</tr>
</tbody>
</table>
Cisco Unified CME Commands: S2

- shared-line, on page 981
- shared-line sip, on page 982
- show capf-server, on page 984
- show credentials, on page 986
- show cti, on page 988
- show ctl-client, on page 991
- show ephone, on page 992
- show ephone attempted-registrations, on page 997
- show ephone cfa, on page 999
- show ephone dn, on page 1000
- show ephone dnd, on page 1001
- show ephone login, on page 1002
- show ephone moh, on page 1005
- show ephone offhook, on page 1006
- show ephone overlay, on page 1008
- show ephone phone-load, on page 1010
- show ephone registered, on page 1012
- show ephone registered summary, on page 1014
- show ephone remote, on page 1016
- show ephone ringing, on page 1017
- show ephone rtp connections, on page 1018
- show ephone socket, on page 1020
- show ephone summary brief, on page 1022
- show ephone summary, on page 1024
- show ephone summary types, on page 1026
- show ephone tapiclients, on page 1027
- show ephone telephone-number, on page 1028
- show ephone unregistered, on page 1029
- show ephone unregistered summary, on page 1030
- show ephone-dn, on page 1032
- show ephone-dn callback, on page 1040
- show ephone-dn conference, on page 1042
- show ephone-dn loopback, on page 1044
• show ephone-dn paging, on page 1046
• show ephone-dn park, on page 1049
• show ephone-dn statistics, on page 1050
• show ephone-dn summary, on page 1052
• show ephone-dn whisper, on page 1054
• show ephone-hunt, on page 1056
• show ephone-hunt statistics, on page 1063
• show fb-its-log, on page 1068
• show ip address trusted list, on page 1070
• show presence global, on page 1071
• show presence subscription, on page 1073
• show sdspfarm, on page 1077
• show shared-line, on page 1083
• show telephony-service admin, on page 1085
• show telephony-service all, on page 1087
• show telephony-service bulk-speed-dial, on page 1091
• show telephony-service conference hardware, on page 1093
• show telephony-service directory-entry, on page 1097
• show telephony-service ephone, on page 1098
• show telephony-service ephone-dn, on page 1101
• show telephony-service ephone-dn-template, on page 1103
• show telephony-service ephone-template, on page 1104
• show telephony-service fac, on page 1107
• show telephony-service security-info, on page 1108
• show telephony-service tftp-bindings, on page 1109
• show telephony-service voice-port, on page 1110
• show voice emergency, on page 1112
• show voice emergency addresses, on page 1113
• show voice emergency all, on page 1114
• show voice emergency callers, on page 1116
• show voice emergency zone, on page 1117
• show voice fac statistics, on page 1118
• show voice hunt-group, on page 1119
• show voice hunt-group statistics, on page 1124
• show voice register all, on page 1128
• show voice register credential, on page 1139
• show voice register dial-peers, on page 1141
• show voice register dialplan, on page 1143
• show voice register dn, on page 1145
• show voice register global, on page 1148
• show voice register hfs, on page 1152
• show voice register pool, on page 1153
• show voice register pool after-hour-exempt, on page 1161
• show voice register pool attempted-registrations, on page 1163
• show voice register pool cfa, on page 1165
• show voice register pool connected, on page 1167
- show voice register pool ip, on page 1170
- show voice register pool mac, on page 1172
- show voice register pool on-hold, on page 1174
- show voice register pool phone-load, on page 1177
- show voice register pool registered, on page 1178
- show voice register pool remote, on page 1184
- show voice register pool ringing, on page 1186
- show voice register pool telephone-number, on page 1188
- show voice register pool type, on page 1190
- show voice register pool type summary, on page 1193
- show voice register pool unregistered, on page 1194
- show voice register profile, on page 1196
- show voice register session-server, on page 1198
- show voice register statistics, on page 1200
- show voice register template, on page 1204
- show voice register tftp-bind, on page 1208
- shutdown(telephony-service), on page 1210
- sip-prefix, on page 1211
- snr, on page 1212
- snr (voice register dn), on page 1214
- snr answer-too-soon, on page 1216
- snr answer-too-soon (voice register dn), on page 1217
- snr calling-number local, on page 1218
- snr calling-number local (voice register dn), on page 1219
- snr mode, on page 1220
- snr ring-stop, on page 1221
- snr ring-stop (voice register dn), on page 1222
- softkeys alerting, on page 1223
- softkeys connected (voice register template), on page 1225
- softkeys connected, on page 1227
- softkeys hold, on page 1230
- softkeys idle, on page 1232
- softkeys idle (voice register template), on page 1235
- softkeys personal-conf-user (voice register template), on page 1237
- softkeys remote-in-use, on page 1239
- softkeys remote-in-use (voice register template), on page 1240
- softkeys ringin (voice register template), on page 1242
- softkeys ringing, on page 1244
- softkeys seized, on page 1246
- softkeys seized (voice register template), on page 1248
- source-addr, on page 1250
- source-address (voice register global), on page 1251
- speed-dial, on page 1253
- speed-dial (voice logout-profile and voice user-profile), on page 1256
- speed-dial (voice register pool), on page 1258
- srst dn line-mode, on page 1260
• srst dn template, on page 1262
• srst ephone description, on page 1263
• srst ephone template, on page 1264
• srst mode auto-provision, on page 1265
• standby username password, on page 1267
• statistics collect, on page 1268
• statistics collect (voice hunt-group), on page 1270
• subnet, on page 1271
• system message, on page 1272
shared-line

To create a directory number to be shared by multiple SIP phones, use the `shared-line` command in voice register dn configuration mode. To return to the default, use the `no` form of this command.

```
shared-line [max-calls number-of-calls]
no shared-line
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>max-calls number-of-calls</code></td>
<td>(Optional) Maximum number of active calls allowed on the shared line. Range: 2 to 16. Default: 2.</td>
</tr>
</tbody>
</table>

**Command Default**

Directory number is not a shared line. Maximum number of calls on a shared line is 2.

**Command Modes**

Voice register dn configuration

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables a shared line on an individual SIP phone directory number.

This command is supported only on the Cisco Unified IP Phone 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE.

**Examples**

The following example shows that extension 5001 associated with directory number 2 is defined as a shared line and can support up to four calls:

```
Router(config)# voice register dn 2
Router(config-register-dn)# number 5001
Router(config-register-dn)# shared-line max-calls 4
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>busy-trigger-per-button</code></td>
<td>Sets the maximum number of calls allowed on a SIP shared line before activating Call Forward Busy or a busy tone.</td>
</tr>
<tr>
<td><code>debug shared-line</code></td>
<td>Displays debugging information about shared lines on SIP phones.</td>
</tr>
<tr>
<td><code>huntstop</code></td>
<td>Disables call hunting behavior for a directory number on a SIP phone.</td>
</tr>
<tr>
<td><code>number (voice register dn)</code></td>
<td>Associates a telephone or extension number with a SIP phone.</td>
</tr>
<tr>
<td><code>show shared-line</code></td>
<td>Displays information about shared lines on SIP phones.</td>
</tr>
<tr>
<td><code>show voice register dn</code></td>
<td>Displays all configuration information associated with a specific voice register dn.</td>
</tr>
</tbody>
</table>
shared-line sip

To add an ephone-dn as a member of a shared directory number in the database of the Shared-Line Service Module for a mixed shared line between Cisco Unified SIP and Cisco Unified SCCP IP phones, use the `shared-line sip` command in ephone-dn configuration mode. To return to the default, use the `no` form of this command.

```
shared-line sip [max calls number-of-calls]
no shared-line sip
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>max calls number-of-calls</code></td>
<td>(Optional) Maximum number of active calls allowed on the shared line. Range: 2 to 16. Default: 2.</td>
</tr>
</tbody>
</table>

**Command Default**

Directory number is not a mixed shared line.

Maximum number of calls on a mixed shared line is 2.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `shared-line sip` command to add an ephone-dn as a member of a shared directory number in the database of the Shared-Line Service Module for a mixed shared line between Cisco Unified SIP IP phones and Cisco Unified SCCP IP phones. However, a mixed shared line is not enabled when an ephone-dn `nnnn` is the only shared directory number `nnnn` in the database of the Shared-Line Service Module. It is only enabled when a corresponding Cisco Unified SIP IP phone with a shared directory number `nnnn` is subscribed.

Mixed shared lines can only be configured on one of several common directory numbers. All attempts to add more are rejected.

**Note**

The secondary number of an ephone-dn cannot be used as a search key in the Shared-Line Service Module.

Features are effectively supported on a mixed shared line when dial-plan patterns have matching configurations in telephony-service and voice register global configuration modes using the `dialplan pattern` command.

**Examples**

The following example shows 1001 as the shared line between a Cisco Unified SCCP IP phone and a Cisco Unified SIP IP phone. The maximum number of active calls allowed on the mixed shared line is four.

```
voice register dn 1
   number 1001
shared-line max-calls 4
ephone-dn 1 octo-line
   number 1001
shared-line sip
```
The following example shows how configuring a mixed shared line on a second common directory number is rejected:

```plaintext
Router(config)# ephone-dn 14 octo-line
Router(config-ephone-dn)# number 2502
Router(config-ephone-dn)# shared-line sip
Router(config)# ephone-dn 20 octo-line
Router(config-ephone-dn)# number 2502
Router(config-ephone-dn)# shared-line sip
DN number already exists in the shared line database
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>dialplan pattern</code></td>
<td>Defines a pattern that is used to expand extension numbers in Cisco Unified CME into fully qualified E.164 numbers, in telephony-service configuration mode.</td>
</tr>
<tr>
<td><code>dialplan pattern (voice register)</code></td>
<td>Defines a pattern that is used to expand extension numbers in Cisco Unified CME into fully qualified E.164 numbers, in voice register global configuration mode.</td>
</tr>
<tr>
<td><code>shared-line</code></td>
<td>Creates a directory number to be shared by multiple Cisco Unified SIP IP phones.</td>
</tr>
</tbody>
</table>
show capf-server

To display CAPF server configuration and session information, use the `show capf-server` command in privileged EXEC configuration mode.

```
show capf-server {auth-string|sessions|summary}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>auth-string</td>
<td>Display authentication strings for ephones.</td>
</tr>
<tr>
<td>sessions</td>
<td>Display information about active CAPF sessions.</td>
</tr>
<tr>
<td>summary</td>
<td>Display CAPF server configuration details.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication.

**Examples**

The following example output displays CAPF server parameters:

```
Router# show capf-server summary
CAPF Server Configuration Details
  Trustpoint for TLS With Phone: cmeserver
  Trustpoint for CA operation: iosra
  Source Address: 10.1.1.1
  Listening Port: 3804
  Phone Key Size: 1024
  Phone KeyGen Retries: 100
  Phone KeyGen Timeout: 120 minutes
  Device Authentication Mode: Auth-String
```

The following example output displays the authentication strings that have been defined for the phones with the listed MAC addresses:

```
Router# show capf-server auth-string
Authentication Strings for configured Ephones
Mac-Addr     Auth-String
----------     ----------
000CCE3A817C  7012
001121116BDD  922
000D299D50DF  9182
000ED7B10DAC  3114
000F90485077  3328
0013C352E7F1  0678
```
The following example output displays active sessions between phones (identified by their MAC addresses) and the CAPF server. The phone ID field lists standard phone identifications, which include the letters “SEP” plus the MAC addresses of the phones. The below sample output defines the different session states that can appear in the output.

```
Router# show capf-server sessions
```

**Active CAPF Sessions**

<table>
<thead>
<tr>
<th>Phone ID</th>
<th>State</th>
</tr>
</thead>
<tbody>
<tr>
<td>SEP000CCE3A817C</td>
<td>AWAIT-KEYGEN-RES</td>
</tr>
</tbody>
</table>

**Table 16: show capf-server sessions State Descriptions**

<table>
<thead>
<tr>
<th>State</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>IDLE</td>
<td>Phone is idle.</td>
</tr>
<tr>
<td>AWAIT AUTH RES</td>
<td>A TLS connection was established on the TCP port that is specified in the configuration file. After a successful handshake verified the server certificate, a dialog was started between the CAPF server and the phone’s CAPF client. The server has challenged the phone by sending an authentication request and is waiting for a response.</td>
</tr>
<tr>
<td>AWAIT KEYGEN RESP</td>
<td>Phone authentication was successful. The CAPF server has sent a key generation request message to the phone and is waiting for a response.</td>
</tr>
<tr>
<td>AWAIT ENCRYPT MSG RESP</td>
<td>A key has been generated and the CAPF has used the phone’s public key to start the enrollment process with PKI. The CAPF sent an encrypt-message request to the phone and is waiting for a response.</td>
</tr>
<tr>
<td>AWAIT CA RESP</td>
<td>The phone has signed the received message using its private key and the CAPF has continued the enrollment process. PKI has forwarded the certificate request to the CA and is waiting for a response.</td>
</tr>
<tr>
<td>AWAIT STORE CERT RESP</td>
<td>Upon receiving an certificate issued from the CA, the CAPF has sent a store-certificate request message to the phone. The store-certificate request contains the certificate to be written to the phone’s flash memory. The CAPF is waiting for a store-certificate response message to confirm that the certificate has been stored.</td>
</tr>
</tbody>
</table>
show credentials

To display the credentials settings that have been configured for use during Cisco Unified CME phone authentication communications or secure Cisco Unified SRST fallback, use the `show credentials` command in privileged EXEC mode.

```
show credentials
```

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco SRST 3.3</td>
<td>This command was introduced for Cisco SRST.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced for Cisco Unified CME.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command for Cisco Unified CME was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

**Cisco Unified CME**

This command displays the credentials settings on a Cisco Unified CME router that has been configured with a CTL provider to be used with Cisco Unified CME phone authentication.

**Cisco Unified SRST**

This command displays the credentials settings on the Cisco Unified SRST router that are supplied to Cisco Unified CallManager for use during secure SRST fallback.

**Examples**

The following is sample output from the `show credentials`:

```
Router# show credentials
Credentials IP: 10.1.1.22
Credentials PORT: 2445
Trustpoint: srstca
```

The below table describes the fields in the sample output.

**Table 17: show credentials Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Credentials IP</td>
<td>Cisco Unified CME—IP address where the CTL provider is configured.</td>
</tr>
<tr>
<td></td>
<td>Cisco Unified SRST—The specified IP address where certificates from Cisco Unified CallManager to the SRST router are received.</td>
</tr>
</tbody>
</table>
### Field | Description
--- | ---
Credentials PORT | Cisco Unified CME—TCP port for credentials service communication. Default is 2444. Cisco Unified SRST—The port to which the SRST router connects to receive messages from the Cisco Unified IP phones. The port number is from 2000 to 9999. The default port number is 2445.

Trustpoint | Cisco Unified CME—CTL provider trustpoint label that will be used for TLS sessions with the CTL client. Cisco Unified SRST—The name of the trustpoint that is associated with the credentials service between the Cisco Unified CallManager client and the SRST router.

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>credentials</td>
<td>Enters credentials configuration mode to configure a Cisco Unified CME CTL provider certificate or a Cisco Unified SRST router certificate.</td>
</tr>
<tr>
<td>ctl-service admin</td>
<td>Specifies a user name and password to authenticate the CTL client during the CTL protocol.</td>
</tr>
<tr>
<td>debug credentials</td>
<td>Sets debugging on the credentials service that runs between a Cisco Unified CME CTL provider and the CTL client or between a Cisco Unified SRST router and Cisco Unified CallManager.</td>
</tr>
<tr>
<td>ip source-address (credentials)</td>
<td>Enables the Cisco Unified CME or SRST router to receive messages through the specified IP address and port.</td>
</tr>
<tr>
<td>trustpoint (credentials)</td>
<td>Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with a Cisco Unified SRST router certificate.</td>
</tr>
</tbody>
</table>
show cti

To display the status of the CTI subsystem, use the `show cti` command in privileged EXEC mode.

```
show cti {call|gecid|line node|session}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call</td>
<td>Details for active (ACT) calls only.</td>
</tr>
<tr>
<td>gcid</td>
<td>List of Global Call IDs for active calls only.</td>
</tr>
<tr>
<td>line node</td>
<td>List of line nodes.</td>
</tr>
<tr>
<td>session</td>
<td>Details for active CTI sessions.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>This command is deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This commands displays status information for the CTI subsystem in Cisco Unified CME.

**Examples**

The following sample output is for each command when there are no active calls.

```
Router#show cti gcid
GCID callIDs
- - -
no active GCID

Router#show cti call
DN CallID GCID Calling Called State
----- ----- ------------------- ------- ------- -----
201    204

A line-node is the internal data structure of a directory number. Once a line-node is created, the structure remains until the CTI interface is shut down.

Router#show cti line-node
line dn number of call instance
----- --------------------
1001    0
201     0
202     0
203     0
204     0
233     0
6789     0
A0001    0
```
The following is sample output from the `show cti gcid` command for one call. This sample contains a single Gcid with two callIDs, one for each call leg.

```
Router# show cti gcid
GCID          callIDs
-------------- -----------------
1E2E3483-5ACB11DE-BA9EF925-DF2AFB55  59291, 59292,
```

The following is sample output from the `show cti call` command. This samples shows that a call was placed from (DN) 201 to (DN) 204 and both directory numbers are now Active (ACT). Note that the Gcid and callIDs in this sample correspond to those in the output from the `show cti gcid` command.

```
Router# show cti call
DN     CallID     GCID       Calling  Called  State
-------- ------- -------------------------- ---------
201      59291   1E2E3483-5ACB11DE-BA9EF925-DF2AFB55  201     204     ACT
204      59292   1E2E3483-5ACB11DE-BA9EF925-DF2AFB55  201     204     ACT
```

The following is sample output from the `show cti line-node` command. In the following sample, there are eight line-nodes and two (201 and 204) are in use.

```
Router# show cti line-node
line  dn     number of call instance
------- -------- ------------------
1001    0
201     1
    callID 59291 (C7C ), *cg = 201, cd = 204
202     0
203     0
204     1
    callID 59292 (C7C ), cg = 201, *cd = 204
233     0
6789    0
A0001   0
```

<< Table number >> describes the significant fields shown in the display.

**Table 18: show xxx Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>GCID</td>
<td>Global Call ID (Gcid)—Unique identifier in for every call on an outbound leg of a VoIP dial peer for an endpoint. A single Gcid remains the same for the same call in the system, and is valid for redirect, transfer, and conference events.</td>
</tr>
<tr>
<td>CallID</td>
<td>Unique identifier for each call leg of a call.</td>
</tr>
</tbody>
</table>
## Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>clear cti</td>
<td>Clears the session between a CSTA client application and Cisco Unified CME.</td>
</tr>
<tr>
<td>session</td>
<td></td>
</tr>
</tbody>
</table>
show ctl-client

To display information about the certificate trust list (CTL) client, use the `show ctl-client` command in privileged EXEC configuration mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication.

**Examples**

The following example displays trustpoints and IP addresses known to the CTL client.

```
Router# show ctl-client

CTL Client Information
--------------------
SAST 1 Certificate Trustpoint: cmeserver
SAST 1 Certificate Trustpoint: sast2
List of Trusted Servers in the CTL
  CME  10.1.1.1  cmeserver
  TFTP 10.1.1.1  cmeserver
  CAPF 10.1.1.1  cmeserver
```
show ephone

To display information about registered Cisco Unified IP phones, use the show ephone command in user EXEC or privileged EXEC mode.

show ephone  [{mac-addressphone-type}]

**Syntax Description**

<table>
<thead>
<tr>
<th>mac-address</th>
<th>(Optional) Displays information for the phone with the specified MAC address.</th>
</tr>
</thead>
<tbody>
<tr>
<td>phone-type</td>
<td>(Optional) Displays information for phones of the specified phone type. Supported phone types are version-specific. Type ? to display a list of values.</td>
</tr>
</tbody>
</table>

**Command Modes**

User EXEC (>)
Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0 Cisco SRST 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0 Cisco SRST 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.2(11)T</td>
<td>Cisco ITS 2.01 Cisco SRST 2.01</td>
<td>The ata keyword was added and this command was implemented on the Cisco 1760.</td>
</tr>
<tr>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1 Cisco SRST 2.1</td>
<td>The 7914 keyword was added.</td>
</tr>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>The 7902, 7905, and 7912 keywords were added.</td>
</tr>
<tr>
<td>12.3(7)T</td>
<td>Cisco CME 3.1 Cisco SRST 3.1</td>
<td>The 7920 and 7936 keywords were added.</td>
</tr>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1 Cisco SRST 3.2.1</td>
<td>The 7970 keyword was added.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3 Cisco SRST 3.3</td>
<td>The 7971 keyword was added, and this command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0 Cisco Unified SRST 4.0</td>
<td>The 7911, 7941, 7941GE, 7961, and 7961GE keywords were added.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0 Cisco Unified SRST 4.0</td>
<td>The 7911, 7941, 7941GE, 7961, and 7961GE keywords were integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(6)XE</td>
<td>Cisco Unified CME 4.0(2)</td>
<td>The 7931 keyword was added for Cisco Unified CME.</td>
</tr>
<tr>
<td>12.4(4)XC4</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>The 7931 keyword was added for Cisco Unified CME.</td>
</tr>
<tr>
<td>12.4(11)T</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>The 7931 keyword for Cisco Unified CME was integrated into Cisco IOS Release 12.4(11)T.</td>
</tr>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1 Cisco Unified SRST 4.1</td>
<td>The 7921 and 7985 keywords were added.</td>
</tr>
</tbody>
</table>
The 7942, 7945, 7962, 7965, and 7975 keywords were added and this command was integrated into Cisco IOS Release 12.4(15)T1.

Emergency response location (ERL) information displays in the output.

Support for user-defined phone types created with the ephone-type command was added.

The 7915-12, 7915-24, 7916-12, 7916-24, and 7937 keywords were added. This command was modified. The IP-STE keyword was added and logical partitioning class of restriction (LPCOR) and Cancel Call Waiting information was added to the output.

The 7915-12, 7915-24, 7916-12, 7916-24, and 7937 keywords were added and this command was integrated into Cisco IOS Release 12.4(20)T.

This command was modified. The IP-STE keyword was added and logical partitioning class of restriction (LPCOR) and Cancel Call Waiting information was added to the output.

This command was integrated into Cisco IOS Release 15.1(1)T.

**Examples**

Significant fields in the output from this command are described in the table.

The following sample output shows general information for registered phones:

```
Router# show ephone
mediaActive:0 whisper_mediaActive:0 startMedia:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0 caps:8 privacy:0
IP:10.4.188.99 * 50007 Telecaster 7940 keepalive 8424 max_line 2 available_line 2
button 1: cw:1 ccw:(0 0)
dn 6 number 6006 CH1 IDLE CH2 IDLE overlay shared
button 2: cw:1 ccw:(0 0 0 0 0 0 0 0)
dn 42 number 6042 CH1 IDLE CH2 IDLE CH3 IDLE CH4 IDLE CH5 IDLE CH6 CH7 IDLE CH8 IDLE shared
overlay 1: 6(6006) 7(6007) 8(6008)
Preferred Codec: g711ulaw
Lpcor Type: local Incoming: ephone_group1 Outgoing: ephone_group1
```

The table describes significant fields in the output.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Active Call</td>
<td>An active call is in progress.</td>
</tr>
<tr>
<td>activeLine</td>
<td>Line (button) on the phone that is in use. Zero indicates that no line is in use.</td>
</tr>
<tr>
<td>auto-dial number</td>
<td>Intercom extension that automatically dials number.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>button number: dn number</td>
<td>Phone button number and the extension (ephone-dn) dn-tag number associated with that button.</td>
</tr>
<tr>
<td>bytes</td>
<td>Total number of voice data bytes sent or received by the phone.</td>
</tr>
<tr>
<td>Called Dn, Calling Dn</td>
<td>Ephone-dn tag numbers of the called and calling ephone-dn. Set to -1 if the call is not to or from an ephone-dn, or if there is no active call.</td>
</tr>
<tr>
<td>cfa number</td>
<td>Call-forward-all to number is enabled for this extension.</td>
</tr>
<tr>
<td>CH1 CH2</td>
<td>Status of channel 1 and, if this is a dual-line ephone-dn, the status of channel 2.</td>
</tr>
<tr>
<td>cw</td>
<td>1 indicates that Call Waiting is enabled. 0 indicates that Call Waiting is disabled.</td>
</tr>
<tr>
<td>debug</td>
<td>1 indicates that debug for the phone is enabled. 0 indicates that debug is disabled.</td>
</tr>
<tr>
<td>DnD</td>
<td>Do Not Disturb is set on this phone.</td>
</tr>
<tr>
<td>DP tag</td>
<td>Not used.</td>
</tr>
<tr>
<td>ephone-number</td>
<td>Unique sequence number used to identify this phone during configuration (phone-tag).</td>
</tr>
<tr>
<td>IP</td>
<td>Assigned IP address of the Cisco Unified IP phone.</td>
</tr>
<tr>
<td>Jitter</td>
<td>Amount of variation (in milliseconds) of the time interval between voice packets received by the Cisco Unified IP phone.</td>
</tr>
<tr>
<td>keepalive</td>
<td>Number of keepalive messages received from the Cisco Unified IP phone by the router.</td>
</tr>
<tr>
<td>Latency</td>
<td>Estimated playout delay for voice packets received by the Cisco Unified IP phone.</td>
</tr>
<tr>
<td>line number</td>
<td>Button number on an IP phone. Line 1 is the button nearest the top of the phone.</td>
</tr>
<tr>
<td>Lost</td>
<td>Number of voice packets lost, as calculated by the Cisco Unified IP phone, on the basis of examining voice packet time-stamp and sequence numbers during playout.</td>
</tr>
<tr>
<td>Lpcor Incoming</td>
<td>Setting of the <strong>lpcor incoming</strong> command.</td>
</tr>
<tr>
<td>Lpcor Outgoing</td>
<td>Setting of the <strong>lpcor outgoing</strong> command.</td>
</tr>
<tr>
<td>Lpcor Type</td>
<td>Setting of the <strong>lpcor type</strong> command.</td>
</tr>
<tr>
<td>Mac</td>
<td>MAC address.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Max Conferences</td>
<td>Maximum number of allowable conference calls and number of active conference calls.</td>
</tr>
<tr>
<td>max_line number</td>
<td>Maximum number of line buttons that can be configured on this phone.</td>
</tr>
<tr>
<td>mediaActive</td>
<td>1 indicates that an active conversation is in progress. 0 indicates that no conversation is ongoing.</td>
</tr>
<tr>
<td>monitor-ring</td>
<td>This button is set up as a monitor button.</td>
</tr>
<tr>
<td>number</td>
<td>Telephone or extension number associated with the Cisco Unified IP phone button and its dn-tag.</td>
</tr>
<tr>
<td>offhook</td>
<td>1 indicates that the phone is off-hook. 0 indicates that the phone is on-hook.</td>
</tr>
<tr>
<td>overlay</td>
<td>This button contains an overlay set. Use show ephone overlay to display the contents of overlay sets.</td>
</tr>
<tr>
<td>paging</td>
<td>1 indicates that the phone has received an audio page. 0 indicates that the phone has not received an audio page.</td>
</tr>
<tr>
<td>paging-dn</td>
<td>Ephone-dn that is dedicated for receiving audio pages on this phone. The paging-dn number is the number of the paging set to which this phone belongs.</td>
</tr>
<tr>
<td>Password</td>
<td>Authentication string that the phone user types when logging in to the web-based Cisco Unified CME GUI.</td>
</tr>
<tr>
<td>Port</td>
<td>Port used for TAPI transmissions.</td>
</tr>
<tr>
<td>REGISTERED</td>
<td>The Cisco Unified IP phone is active and registered. Alternative states are UNREGISTERED (indicating that the connection to the Cisco Unified IP phone was closed in a normal manner) and DECEASED (indicating that the connection to the Cisco Unified IP phone was closed because of a keepalive timeout).</td>
</tr>
<tr>
<td>reset</td>
<td>Pending reset.</td>
</tr>
<tr>
<td>reset_sent</td>
<td>Request for reset has been sent to the Cisco Unified IP phone.</td>
</tr>
<tr>
<td>ringing</td>
<td>1 indicates that the phone is ringing. 0 indicates that the phone is not ringing.</td>
</tr>
<tr>
<td>Rx Pkts</td>
<td>Number of received voice packets.</td>
</tr>
<tr>
<td>silent-ring</td>
<td>Silent ring has been set on this button and extension.</td>
</tr>
<tr>
<td>socket</td>
<td>TCP socket number used to connect to IP phone.</td>
</tr>
<tr>
<td>speed dial</td>
<td>This button is a speed-dial button, assigned to the speed-dial sequence number speed-tag. It dials digit-string and displays the text label-text next to the button.</td>
</tr>
<tr>
<td>speed-tag:digit-string</td>
<td></td>
</tr>
<tr>
<td>label-text</td>
<td></td>
</tr>
</tbody>
</table>
### Field | Description
--- | ---
sub=3, sub=4 | Subtype 3 means that one Cisco Unified IP Phone 7914 Expansion Module is attached to the main Cisco Unified IP Phones 7960 and 7960G, and subtype 4 means that two are attached.
Tag number | Dn-tag number, the unique sequence number that identifies an ephone-dn during configuration, followed by the type of ephone-dn it is.
TAPI Client IP Address | IP address of the PC running the TAPI client.
TCP socket | TCP socket number used to communicate with the Cisco Unified IP phone. This can be correlated with the output of other debug and show commands.
Telecaster model-number | Type and model of the Cisco Unified IP phone. This information is received from the phone during its registration with the router.
TxPkts | Number of transmitted voice packets.
Username | Username that the phone user types when logging in to the web-based Cisco Unified CME GUI.

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone-dn</td>
<td>Displays information about Cisco Unified IP phone extensions (ephone-dns).</td>
</tr>
<tr>
<td>show ephone login</td>
<td>Displays the login states of all local phones.</td>
</tr>
<tr>
<td>show telephony-service all</td>
<td>Displays systemwide status and information for a Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
show ephone attempted-registrations

To display the log of ephones that unsuccessfully attempt to register with Cisco Unified CME, use the show ephone attempted-registrations command in privileged EXEC mode.

show ephone attempted-registrations

Syntax Description
This command has no keywords or arguments.

Command Modes
Privileged EXEC (#)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines
The no auto-reg-ephone blocks the automatic registration of ephones whose MAC addresses are not explicitly listed in the configuration. When automatic registration is blocked, Cisco Unified CME records the MAC addresses of phones that attempt to register but cannot because they are blocked.

Use the show ephone attempted-registrations to view the list of phones that have attempted to register but have been blocked. The clear telephony-service ephone-attempted-registrations clears the list.

Examples
The following example displays ephones that unsuccessfully attempted to register with Cisco Unified CME:

```
Router# show ephone attempted-registrations
Attempting Mac address:
Num  Mac Address         DateTime               DeviceType
-----------------------------------------------
1     C863.8475.5417     22:52:05 UTC Thu Apr 28 2005  SCCP Gateway (AN)
2     C863.8475.5408     22:52:05 UTC Thu Apr 28 2005  SCCP Gateway (AN)
....
25    000D.2B8D7.7222    22:26:32 UTC Thu Apr 28 2005  Telecaster 7960
...
47    C863.94A8.D40F     22:52:17 UTC Thu Apr 28 2005  SCCP Gateway (AN)
49    C863.94A8.D400     22:52:15 UTC Thu Apr 28 2005  SCCP Gateway (AN)
```

The below table describes the significant fields shown in the display.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Num</td>
<td>Index number.</td>
</tr>
<tr>
<td>Mac Address</td>
<td>MAC address of the ephone.</td>
</tr>
<tr>
<td>DateTime</td>
<td>Date and time that the attempt to register was made.</td>
</tr>
</tbody>
</table>
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>auto-reg-ephone</strong></td>
<td>Enables automatic registration of ephones with the Cisco Unified CME system.</td>
</tr>
<tr>
<td><strong>clear telephony-service</strong></td>
<td></td>
</tr>
<tr>
<td><strong>ephone-attempted-registrations</strong></td>
<td>Empties the log of ephones that unsuccessfully attempt to register with Cisco Unified CME.</td>
</tr>
</tbody>
</table>

### Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DeviceType</td>
<td>Type of ephone.</td>
</tr>
</tbody>
</table>
show ephone cfa

To display status and information on the registered phones that have call-forward-all set on one or more of their extensions (ephone-dns), use the **show ephone cfa** command in privileged EXEC mode.

**show ephone cfa**

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following is sample output from the **show ephone cfa** command:

```
Router# show ephone cfa
ephone-1 Mac:0007.0EA6.353A TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:1.2.205.205 52491 Telecaster 7960 keepalive 14 max_line 6
button 1: dn 11 number 60011 cfa 60022 CH1 IDLE
button 2: dn 17 number 60017 cfa 60021 CH1 IDLE
```

The **show ephone** describes significant fields in this output.

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone</td>
<td>Displays statistical information about registered Cisco IP phones.</td>
</tr>
</tbody>
</table>
show ephone dn

To display phone information for specified dn-tag or for all dn-tags, use the show ephone dn command in privileged EXEC mode.

```
show ephone dn [dn-tag]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>dn-tag</strong></td>
<td>(Optional) Unique sequence number that is used during configuration to identify a particular extension (ephone-dn).</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to identify the phone on which a particular dn-tag has been assigned.

**Examples**

The following is sample output for the two appearances of DN 5:

```
Router# show ephone dn 5
Tag 5, Normal or Intercom dn
ephone 1, mac-address 0030.94c3.cAA2, line 2
ephone 2, mac-address 0030.94c2.9919, line 3
```

The show ephone describes significant fields in this output.

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone</td>
<td>Displays statistical information about registered Cisco IP phones.</td>
</tr>
</tbody>
</table>
show ephone dnd

To display information on the registered phones that have “do not disturb” set on one or more of their extensions (ephone-dns), use the `show ephone dnd` command in privileged EXEC mode.

Command: `show ephone dnd`

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command does not apply to Cisco Unified SRST.

**Examples**

The following is sample output from the `show ephone dnd` command:

```
Router# show ephone dnd
ephone-1 Mac:0007.0EA6.353A TCP socket:[1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:1.2.205.205 52486 Telecaster 7960 keepalive 2729 max_line 6 DnD
button 1: dn 11 number 60011 CH1 IDLE
```

The `show ephone` describes significant fields in this output.

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>show ephone</code></td>
<td>Displays statistical information about registered Cisco IP phones.</td>
</tr>
</tbody>
</table>
show ephone login

To display the login states of all local IP phones, use the **show ephone login** command in privileged EXEC mode.

**show ephone login**

**Syntax Description**

This command has no arguments or keywords.

**Privileged EXEC (#)**

**Command Modes**

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0 Cisco Unified SRST 8.0</td>
<td>This command was modified. LOCAL and GLOBAL replace TRUE in the output for “Pin enabled.”</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0 Cisco Unified SRST 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The **show ephone login** command displays whether an ephone has a personal identification number (PIN) and whether its owner is logged in.

In Cisco Unified CME 7.1 and earlier versions, FALSE is displayed if there is no PIN configured for the specified ephone. TRUE is displayed if there is a PIN configured for the specified ephone.

In Cisco Unified CME 8.0 and later versions, the show output is modified as follows:

- FALSE is displayed only if no PIN is defined, neither in an ephone configuration nor in the telephony-service configuration.
- LOCAL is displayed if an individual PIN is defined for the specific ephone.
- GLOBAL is displayed if a global PIN is defined.

**Cisco Unified CME 8.0 or Later Versions**

The following is sample output from the **show ephone login** command. It shows that a PIN is defined for ephone 1 and that its owner has not logged in. The other phones do not have PINs associated with them.

```
Router# show ephone login
ephone 1     Pin enabled:LOCAL    Logged-in:FALSE
ephone 2     Pin enabled:FALSE
ephone 3     Pin enabled:FALSE
```

The following is sample output from the **show ephone login** command. It shows that a PIN is defined for ephone 1 and that its owner has not logged in. A global PIN is defined also defined for this system.
If the `pin` command is configured in ephone configuration mode and telephony-service configuration mode, the command in ephone configuration mode takes precedence.

```bash
Router# show ephone login
ephone 1  Pin enabled:LOCAL    Logged-in:FALSE
ephone 2  Pin enabled:GLOBAL   Logged-in:TRUE
ephone 3  Pin enabled:GLOBAL   Logged-in:TRUE
```

The following is sample output from the `show ephone login` command. It shows that neither a local nor a global PIN is enabled for phones 1 to 3.

```bash
Router# show ephone login
ephone 1  Pin enabled:FALSE
ephone 2  Pin enabled:FALSE
ephone 3  Pin enabled:FALSE
```

**Cisco CME 3.0 to Cisco Unified CME 7.1**

The following is sample output from the `show ephone login` command. It shows that a PIN is enabled for phone 1 and that its owner has not logged in. The other phones do not have PINs associated with them.

```bash
Router# show ephone login
ephone 1  Pin enabled:TRUE    Logged-in:FALSE
ephone 2  Pin enabled:FALSE
ephone 3  Pin enabled:FALSE
ephone 4  Pin enabled:FALSE
ephone 5  Pin enabled:FALSE
ephone 6  Pin enabled:FALSE
ephone 7  Pin enabled:FALSE
ephone 8  Pin enabled:FALSE
ephone 9  Pin enabled:FALSE
```

The below table describes significant fields in this output.

**Table 21: show ephone login Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone phone-tag</td>
<td>Phone identified with its unique phone-tag sequence number.</td>
</tr>
<tr>
<td>Pin enabled</td>
<td>In Cisco Unified CME 7.1 and earlier versions:</td>
</tr>
<tr>
<td></td>
<td>• TRUE—A PIN is defined for this phone.</td>
</tr>
<tr>
<td></td>
<td>• FALSE—No PIN is defined for this phone.</td>
</tr>
<tr>
<td></td>
<td>In Cisco Unified CME 8.0 and later versions:</td>
</tr>
<tr>
<td></td>
<td>• LOCAL—A PIN has been defined for this phone.</td>
</tr>
<tr>
<td></td>
<td>• GLOBAL—A global PIN is defined for this Cisco Unified CME system.</td>
</tr>
<tr>
<td></td>
<td>• FALSE—No PIN is defined.</td>
</tr>
</tbody>
</table>
### Field Description

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Logged-in</td>
<td>• TRUE indicates that a phone user is currently logged in on this phone.</td>
</tr>
<tr>
<td></td>
<td>• FALSE indicates that no phone user is currently logged in on this phone.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>login</code> (telephony-service)</td>
<td>Defines when users of IP phones in a Cisco Unified CME system are logged out automatically.</td>
</tr>
<tr>
<td><code>pin</code></td>
<td>Sets a personal identification number (PIN) for an IP phone in a Cisco Unified CME system.</td>
</tr>
<tr>
<td><code>show ephone</code></td>
<td>Displays statistical information about registered Cisco IP phones.</td>
</tr>
</tbody>
</table>
show ephone moh

To display information about moh files in use, use the `show ephone moh` command in global configuration mode.

```
Router #show ephone moh
Skinny Music On Hold Status (moh-group 1)
  Active MOH clients 0 (max 830), Media Clients 0
  File flash:/minuet.au (not cached) type AU Media_Payload_G711Ulaw64k 160 bytes
  Moh multicast on 239.10.16.6 port 2000
Skinny Music On Hold Status (moh-group 2)
  Active MOH clients 0 (max 830), Media Clients 0
  File flash:/audio/hello.au type AU Media_Payload_G711Ulaw64k 160 bytes
  Moh multicast on 239.10.16.6 port 2000 via 0.0.0.0
Skinny Music On Hold Status (moh-group 3)
  Active MOH clients 0 (max 830), Media Clients 0
  File flash:/bells.au type AU Media_Payload_G711Ulaw64k 160 bytes
  Moh multicast on 239.10.16.5 port 2000 via 0.0.0.0
Skinny Music On Hold Status (moh-group 4)
  Active MOH clients 0 (max 830), Media Clients 0
  File flash:/3003.au type AU Media_Payload_G711Ulaw64k 160 bytes
  Moh multicast on 239.10.16.7 port 2000 via 0.0.0.0
Skinny Music On Hold Status (moh-group 5)
  Active MOH clients 0 (max 830), Media Clients 0
  File flash:/4004.au type AU Media_Payload_G711Ulaw64k 160 bytes
  Moh multicast on 239.10.16.8 port 2000 via 0.0.0.0
```

```
Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone-dn</td>
<td>Displays MOH group information for a phone directory number.</td>
</tr>
<tr>
<td>show ephone summary</td>
<td>Displays the information about the MOH files in use.</td>
</tr>
<tr>
<td>show voice moh-group statistics</td>
<td>Displays the MOH subsystem statistics information.</td>
</tr>
</tbody>
</table>
```
show ephone offhook

To display information and packet counts for the phones that are currently off hook, use the show ephone offhook command in privileged EXEC mode.

**show ephone offhook**

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command is introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command is integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a</td>
<td>Unified CME 12.6</td>
<td>This command is enhanced to display the keys that are in use per media stream, along with the SRTP Ciphers.</td>
</tr>
</tbody>
</table>

**Examples**

The following sample output is displayed when no phone is off hook:

Router# show ephone offhook
No ephone in specified type/condition.

The following sample output displays information for a phone that is off hook:

Router# show ephone offhook
ephone-5 Mac:000A.8A2C.8C6E TCP socket:[20] activeLine:1 REGISTERED
mediaActive:0 offhook:1 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.22.84.71 51228 Telecaster 7960 keepalive 43218 max_line 6
button 1:dn 9 number 59943 CH1 SIEZE silent-ring
button 2:dn 10 number 59943 CH1 IDLE
button 3:dn 42 number A4400 auto dial A4500 CH1 IDLE
button 4:dn 96 number 69943 auto dial 95259943 CH1 IDLE
button 5:dn 75 number 49943 auto dial 49943 CH1 IDLE
speed dial 1:57514 marketing
Active Call on DN 9 chan 1:59943 0.0.0.0 0 to 0.0.0.0 2000 via 172.30.151.1
G711Ulaw64k 160 bytes vad
Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Jitter 0 Latency 0 callingDn -1 calledDn -1
Username:user1 Password:newuser

The following is a sample output for the show command, show ephone offhook. The lines that are added to the show command output as part of the Unified CME 12.6 enhancement are local key and remote key.

ephone-1[0] Mac:549A.EEB5.8000 TCP socket:[1] activeLine:1 whisperLine:0 REGISTERED in SCCP
ver 21/17 max_streams=1 + Authentication + Encryption with TLS connection
mediaActive:1 whisper_mediaActive:0 startMedia:1 offhook:1 ringing:0 reset:0 reset_sent:0
paging 0 debug:0 caps:8
The following sample output displays information for a phone that has just completed a call:

```
Router# show ephone offhook
ephone-5 Mac:000A.8A2C.8C6E TCP socket:[20] activeLine:1 REGISTERED
mediaActive:1 offhook:1 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.22.84.71 51228 Telecaster 7960 keepalive 43224 max_line 6
button 1:dn 9 number 59943 CH1 CONNECTED silent-ring
button 2:dn 10 number 59943 CH1 IDLE
button 3:dn 42 number A4400 auto dial A4500 CH1 IDLE
button 4:dn 96 number 69943 auto dial 95259943 CH1 IDLE
button 5:dn 75 number 49943 auto dial 49943 CH1 IDLE
speed dial 1:57514 marketing
Active Call on DN 9 chan 1 :59943 10.23.84.71 22926 to 172.30.131.129 2000 via 172.30.151.1
G711Ulaw64k 160 bytes no vad
Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Jitter 0 Latency 0 callingDn -1 calledDn -1 (media path callID 19288 srcCallID 1 9289)
Username:user1 Password:newuser
```

The `show ephone` command describes significant fields in this output.

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>show ephone</code></td>
<td>Displays statistical information about registered Cisco IP phones.</td>
</tr>
</tbody>
</table>
show ephone overlay

To display information for the registered phones that have overlay ephone-dns associated with them, use the `show ephone overlay` command in privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command does not apply to Cisco Unified SRST.

**Examples**

The following is sample output from the `show ephone overlay` command:

```
Router# show ephone overlay
ephone-1 Mac:0007.0EA6.353A TCP socket:[1] activeLine:0 REGISTERED
IP:10.2.225.205 52486 Telecaster 7960 keepalive 2771 max_line 6
button 1: dn 11 number 60011 CH1 IDLE overlay
button 2: dn 17 number 60017 CH1 IDLE overlay
button 3: dn 24 number 60024 CH1 IDLE overlay
button 4: dn 30 number 60030 CH1 IDLE overlay
button 5: dn 36 number 60036 CH1 IDLE CH2 IDLE overlay
button 6: dn 39 number 60039 CH1 IDLE CH2 IDLE overlay
overlay 1: 11(60011) 12(60012) 13(60013) 14(60014) 15(60015) 16(60016)
overlay 2: 17(60017) 18(60018) 19(60019) 20(60020) 21(60021) 22(60022)
overlay 3: 23(60023) 24(60024) 25(60025) 26(60026) 27(60027) 28(60028)
overlay 4: 29(60029) 30(60030) 31(60031) 32(60032) 33(60033) 34(60034)
overlay 5: 35(60035) 36(60036) 37(60037) 38(60038) 39(60039) 40(60040)
```

The `show ephone` command describes significant fields in this output. The below table describes a field that is not in that table.

**Table 22: show ephone overlay Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>overlay number</td>
<td>Displays the contents of an overlay set, including each dn-tag and its associated extension number.</td>
</tr>
<tr>
<td>Related Commands</td>
<td>Command</td>
</tr>
<tr>
<td>------------------</td>
<td>-----------</td>
</tr>
<tr>
<td></td>
<td><strong>show ephone</strong></td>
</tr>
</tbody>
</table>
show ephone phone-load

To display information about the phone firmware that is loaded on registered phones, use the `show ephone phone-load` command in privileged EXEC mode.

**show ephone phone-load**

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following is sample output that displays the phone firmware versions for all phones in the system:

```
Router# show ephone phone-load

DeviceName | CurrentPhoneLoad | PreviousPhoneLoad | LastReset       |
-----------|------------------|------------------|-----------------|
SEP0002B9AFC49F | 3.2(2.14)      | 3.2(2.14)        | TCP-timeout     |
SEP003094C2D0B0 | 3.2(2.14)      | 3.2(2.14)        | TCP-timeout     |
SEP000C30F03707 | 3.2(2.14)      | 3.2(2.14)        | TCP-timeout     |
SEP003094C2999F | 3.2(2.14)      | 3.2(2.14)        | TCP-timeout     |
SEP000A8A2C8C6E | 3.2(2.14)      | 3.2(2.14)        | Initialized     |
SEP0002B9AFBB4D | 3.2(2.14)      | 3.2(2.14)        | TCP-timeout     |
SEP00075078627F | 3.2(2.14)      | 3.2(2.14)        | TCP-timeout     |
SEP0002F659E59  | 3.2(2.14)      | 3.2(2.14)        | TCP-timeout     |
SEP000248CCD626 | 3.2(2.14)      | 3.2(2.14)        | CM-closed-TCP   |
SEP000B215F88C1 | 3.2(2.14)      | 3.2(2.14)        | TCP-timeout     |
SEP000C30F0390C | 3.2(2.14)      | 3.2(2.14)        | TCP-timeout     |
SEP003094C30143 | 3.2(2.14)      | 3.2(2.14)        | TCP-timeout     |
```

The below table describes significant fields in this output.

**Table 23: show ephone phone-load Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DeviceName</td>
<td>Device name.</td>
</tr>
<tr>
<td>CurrentPhoneLoad</td>
<td>Current phone firmware version.</td>
</tr>
<tr>
<td>PreviousPhoneLoad</td>
<td>Phone firmware version before last phone load.</td>
</tr>
<tr>
<td>LastReset</td>
<td>Reason for last reset of phone.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>---------------</td>
<td>--------------------------------------------------</td>
</tr>
<tr>
<td>show ephone</td>
<td>Displays statistical information about registered Cisco IP phones.</td>
</tr>
</tbody>
</table>
show ephone registered

To display the status of registered phones, use the `show ephone registered` command in privileged EXEC mode.

**show ephone registered**

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was modified. The output was enhanced to include the setting of the feature-button command.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following is sample output from the show ephone registered command:

```
Router# show ephone registered
mediaActive:0 whisper_mediaActive:0 startMedia:0 offhook:0 reset:0 reset_sent:0 paging:0 debug:0 caps:7
IP:10.10.1.17 * 35177 6941 keepalive 3593 max_line 4 available_line 3
button 1: cw:1 dn 11 number 1001 CH1 IDLE CH2 IDLE
button 2: cw:1 dn 56 number 6971 auto dial 6970 CH1 IDLE
button 3: cw:1 dn 10 number 1000 CH1 IDLE CH2 IDLE
1 feature buttons enabled: dnd
Preferred Codec: g711ulaw
Lpcor Type: none
```

The below table describes significant fields in this output.

**Table 24: show ephone registered Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>active</td>
<td>Number of active parties registered.</td>
</tr>
<tr>
<td>ephone</td>
<td>Cisco IP phone.</td>
</tr>
<tr>
<td>mac-address</td>
<td>MAC address of the Cisco IP phone.</td>
</tr>
<tr>
<td>keepalive</td>
<td>Defines keepalive timeout period to unregister IP phone.</td>
</tr>
</tbody>
</table>
### Field Description

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>feature-buttons</td>
<td>Displays the type of feature button on the ephone.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone</td>
<td>Displays statistical information about registered Cisco IP phones.</td>
</tr>
</tbody>
</table>
show ephone registered summary

To display the details of all the registered Skinny Client Control Protocol (SCCP) phones that are sorted based on ephone tags, use the `show ephone registered summary` command in privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

This command has no default behavior or values.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to view the details of the registered phones configured in the SCCP mode sorted by ephone tags.

**Example**

The following is sample output of the registered phones configured in the SCCP mode.

```
router# show ephone registered summary

+------------+-------------+---------------+------------+------------+-------------+-------------------+
<table>
<thead>
<tr>
<th>PhoneType</th>
<th>Ephone</th>
<th>MacAddress</th>
<th>IpAddress</th>
<th>Ln</th>
<th>Dn</th>
<th>Number</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>8941</td>
<td>1</td>
<td>7081.050C.0927</td>
<td>9.51.0.71</td>
<td>1</td>
<td>1</td>
<td>3001</td>
<td>Registered</td>
</tr>
<tr>
<td></td>
<td>2</td>
<td>2*</td>
<td>3002</td>
<td>2</td>
<td>2*</td>
<td>3005</td>
<td>Registered</td>
</tr>
<tr>
<td></td>
<td>2</td>
<td>5*</td>
<td>3006</td>
<td>2</td>
<td>6*</td>
<td>3010</td>
<td>Registered</td>
</tr>
<tr>
<td>7970</td>
<td>2</td>
<td>001B.D52C.DF27</td>
<td>9.51.0.72</td>
<td>1</td>
<td>3</td>
<td>3003</td>
<td>Registered</td>
</tr>
<tr>
<td></td>
<td>2</td>
<td>4</td>
<td>3004</td>
<td>2</td>
<td>4</td>
<td>3010</td>
<td>Registered</td>
</tr>
<tr>
<td>7970</td>
<td>5</td>
<td>001B.D52C.4AAE</td>
<td>9.51.0.75</td>
<td>1</td>
<td>9</td>
<td>3009</td>
<td>Registered</td>
</tr>
<tr>
<td></td>
<td>2</td>
<td>10</td>
<td>3010</td>
<td>2</td>
<td>10</td>
<td>3010</td>
<td>Registered</td>
</tr>
</tbody>
</table>
+------------+-------------+---------------+------------+----|----|----------+-------------------+

Total ephones configured : 10
Total ephones registered : 3
Total ephones unregistered : 5
Total ephones deceased : 0
Ephones in unknown state : 2
```

**Note**

The * symbol adjacent to the Directory Number (DN) in the command output indicates that the Directory Number (DN) is an Overlay-dn.
### Table 25: show ephone registered summary field descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DN</td>
<td>Directory number of the phone.</td>
</tr>
<tr>
<td>Ephone</td>
<td>Total number of ephone tags configured.</td>
</tr>
<tr>
<td>IP Address</td>
<td>IP address of the phones.</td>
</tr>
<tr>
<td>LN</td>
<td>Line number of the phone.</td>
</tr>
<tr>
<td>MacAddress</td>
<td>Shows the MAC address of the SCCP phone.</td>
</tr>
<tr>
<td>Number</td>
<td>Number assigned to ephone.</td>
</tr>
<tr>
<td>PhoneType</td>
<td>Shows the type of Cisco IP phone.</td>
</tr>
<tr>
<td>Status</td>
<td>Shows the registration status.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone summary types</td>
<td>Displays the total number of registered and unregistered SCCP phones for each phone type.</td>
</tr>
<tr>
<td>show ephone unregistered summary</td>
<td>Displays the details of all the unregistered SCCP phones.</td>
</tr>
</tbody>
</table>
show ephone remote

To display nonlocal phones (phones with no Address Resolution Protocol [ARP] entry), use the `show ephone remote` command in privileged EXEC mode.

**show ephone remote**

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Phones without ARP entries are suspected not to be on the LAN. Use the `show ephone remote` command to identify phones without ARP entries that might have operational issues.

**Examples**

The following is sample output that identifies ephone 2 as not having an ARP entry:

```
Router# show ephone remote
ephone-2 Mac:0185.047C.993E TCP socket:[4] activeLine:0 REGISTERED
mediaActive:1 offhook:0 ringing:0 reset:0 reset_sent:0 paging 1 debug:0
IP:10.50.50.20 49231 Telecaster 7910 keepalive 112 max_line 2 dual-line
button 1:dn 3 number 95021 CH1 IDLE
paging-dn 25
```

The `show ephone` describes significant fields in this output.

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone</td>
<td>Displays statistical information about registered Cisco IP phones.</td>
</tr>
</tbody>
</table>
show ephone ringing

To display information on phones that are ringing, use the `show ephone ringing` command in privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following is sample output from the `show ephone ringing` command:

```
Router# show ephone ringing
ephone=1 Mac:0005.5E37.8090 TCP socket:[1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:1 reset:0 reset_sent:0 paging 0 debug:0
IP:10.50.50.10 49329 Telecaster 7960 keepalive 17602 max_line 6
button 1:dn 1 number 95011 CH1 RINGING CH2 IDLE
button 2:dn 2 number 95012 CH1 IDLE
```

The `show ephone` describes significant fields in this output.

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone</td>
<td>Displays statistical information about registered Cisco IP phones.</td>
</tr>
</tbody>
</table>
show ephone rtp connections

To display active Real-Time Transport Protocol (RTP) call information on ephone call legs, use the `show ephone rtp connections` command in privileged EXEC mode.

**show ephone rtp connections**

**Syntax Description**
This command has no arguments or keywords.

**Command Modes**
Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(1)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
The `show ephone rtp connections` command displays information on active RTP calls, including the ephone tag number of the phone with an active call, the channel of the ephone-dn, and the caller and called party's numbers for the connection for both local and remote endpoints. The output from this command provides an overview of all the connections in the system, narrowing the criteria for debugging pulse code modulation and Cisco Unified CME packets without a sniffer.

When an ephone to non-ephone call is made, information on the non-ephone does not appear in a `show ephone rtp connections` command output. To display the non-ephone call information, use the `show voip rtp connections` command.

**Examples**
The following sample output shows all the connected ephones in the Cisco Unified CME system. The sample output shows five active ephone connections with one of the phones having the `dspfarm-assist` keyword configured to transcode the code on the local leg to the indicated codec. The output also shows four ephone to ephone calls, represented in the CallID columns of both the RTP connection source and RTP connection destination by zero values.

Normally, a phone can have only one active connection but in the presence of a whisper intercom call, a phone can have two. In the sample output, ephone-40 has two active calls: it is receiving both a normal call and a whisper intercom call. The whisper intercom call is being sent by ephone-6, which has an invalid LocalIP of 0.0.0.0. The invalid LocalIP indicates that it does not receive RTP audio because it only has a one-way voice connection to the whisper intercom call recipient.

```
Router# show ephone rtp connections
Ephone RTP active connections :
Phone Line DN Chan SrcCallID DstCallID Codec (xcoded?)
SrcNum DstNum LocalIP RemoteIP
ephone-5 1 5 1 15 14 G729 (Y)
1005 1102 [192.168.1.100]:23192 [192.168.1.1]:2000
ephone-6 2 35 1 0 0 G711Ulaw64k (N)
1035 1036 [0.0.0.0]:0 [192.168.1.81]:21256
ephone-40 1 140 1 0 0 G711Ulaw64k (N)
1140 1141 [192.168.1.81]:21244 [192.168.1.70]:20664
ephone-40 2 36 1 0 0 G711Ulaw64k (N)
1035 1036 [192.168.1.81]:21256 [192.168.1.1]:2000
```
The below table explains the fields in the `show ephone rtp connections` command output.

### Table 26: show ephone rtp connections Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ephone</td>
<td>Ephone tag number with an active call.</td>
</tr>
<tr>
<td>Line</td>
<td>Line appearance of the phone.</td>
</tr>
<tr>
<td>DN</td>
<td>Ephone-dn tag.</td>
</tr>
<tr>
<td>Chan</td>
<td>Channel of the ephone-dn.</td>
</tr>
<tr>
<td>SrcCallID</td>
<td>CCAPI CallID for the RTP connection source. For ephone to ephone calls, this will be 0. SrcCallID compares to “CallId” in the <code>show voip rtp connections</code> command output.</td>
</tr>
<tr>
<td>DstCallID</td>
<td>CCAPI CallID for the RTP connection destination. For ephone to ephone calls, this will be 0. DstCallID compares to “dstCallId” in the <code>show voip rtp connections</code> command output.</td>
</tr>
<tr>
<td>Codec (xcoded)</td>
<td>Codec name used by the phone with the active call. If <code>xcoded</code> is ‘Y’, the phone has the <code>dspfarm-assist</code> keyword configured to transcode the code on the local leg to the indicated codec.</td>
</tr>
<tr>
<td>SrcNum</td>
<td>Caller’s number for the connection. This number is not necessarily the ephone’s DN.</td>
</tr>
<tr>
<td>DstNum</td>
<td>Called party’s number for the connection.</td>
</tr>
<tr>
<td>LocalIP</td>
<td>Call’s local IP address and port. This is usually the ephone’s IP address. The IP address in brackets is either in IPv4 or IPv6 format, followed by a colon and the port number. The port compares to the “LocalRTP” number in the <code>show voip rtp connections</code> command output.</td>
</tr>
<tr>
<td>RemoteIP</td>
<td>Call’s remote IP address and port. For flow-around ephone to ephone calls, this is usually the other ephone’s IP address. For flow-through trunk calls, this is usually the Cisco Unified CME’s IP address. The port compares to the “RmtRTP” number in the <code>show voip rtp connections</code> command output.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>show ephone registered</code></td>
<td>Displays the status of registered SCCP phones in Cisco Unified CME.</td>
</tr>
<tr>
<td><code>show voip rtp connections</code></td>
<td>Displays information about Real-Time Transport Protocol (RTP) named event packets.</td>
</tr>
</tbody>
</table>
show ephone socket

To display IP addresses (IPv4, IPv6, or dual-stack) being used by ephone sockets, use the `show ephone socket` command in privileged EXEC mode.

**show ephone socket**

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0 Cisco Unified SRST 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0 Cisco Unified SRST 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `show ephone socket` command to verify if IPv4 only, IPv6 only, or dual-stack (IPv4/IPv6) is configured on Cisco Unified CME. In the following example, `skinny_tcp_listen_socket fd = 0` and `skinny_tcp_listen_socket fd = 1` verify that dual-stack configuration. When IPv6 only is configured show ephone socket command displays `skinny_tcp_listen_socket fd = -1` and `skinny_tcp_listen_socket fd = 0` values. When IPv4 only is configured the show ephone socket command displays `skinny_tcp_listen_socket fd = 0` and `skinny_tcp_listen_socket (ipv6) fd = -1` values.

**Examples**

The following is sample output from the `show ephone socket` command:

```
Router# show ephone socket
skinny_tcp_listen_socket fd = 0
skinny_tcp_listen_socket (ipv6) fd = 1

skinny_secure_tcp_listen_socket fd = -1
skinny_secure_tcp_listen_socket (ipv6) fd = -1

skinny_open_sockets = 3:
Phone 3,
  skinny_sockets[0] fd = 1
    read_buffer 0x480061E8, read_offset 0, read_header N, read_length 0
    resend_queue 0x47CE8178, resend_offset 0, resend_flag N, resend_Q_depth 0
Phone 2,
  skinny_sockets[1] fd = 2
    read_buffer 0x48006A24, read_offset 0, read_header N, read_length 0
    resend_queue 0x47CE8104, resend_offset 0, resend_flag N, resend_Q_depth 0
Phone 1,
  skinny_sockets[2] fd = 3
    read_buffer 0x48007260, read_offset 0, read_header N, read_length 0
    resend_queue 0x47CE8090, resend_offset 0, resend_flag N, resend_Q_depth 0
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone summary</td>
<td>Displays information about Cisco IP phones.</td>
</tr>
</tbody>
</table>
show ephone summary brief

To display details of all the SCCP phones sorted by ephone-tag, use the `show ephone summary brief` command in privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

This command had no default behavior or values.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3)M</td>
<td>Cisco SIP CME 10.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The command output displays the status, IP address, and MAC address of the phones.

**Example**

The following is sample output of the `show ephone summary brief` command.

```
router# show ephone summary brief
-----------------------------------------------------------------------------------------------
PhoneType Ephone MacAddress IpAddress Ln Dn Number
-----------------------------------------------------------------------------------------------
8941     1    7081.050C.0927 9.51.0.71 1 1 3001
          2    2* 7081.050C.0927 9.51.0.71 2 2* 3002
          2    5* 7081.050C.0927 9.51.0.71 2 5* 3005
          2    6* 7081.050C.0927 9.51.0.71 2 6* 3006
7970     2    001B.D52C.DF27 9.51.0.72 1 3 3003
          2    4 001B.D52C.DF27 9.51.0.72 2 4 3004
7970     3    001B.54CA.43F7 9.51.0.72 1 5 3005
          2    6 001B.54CA.43F7 9.51.0.72 2 6 3006
          3    2* 001B.54CA.43F7 9.51.0.72 3 2* 3002
          3    3* 001B.54CA.43F7 9.51.0.72 3 3* 3003
          3    8* 001B.54CA.43F7 9.51.0.72 3 8* 3008
-----------------------------------------------------------------------------------------------
```

**Note**

The asterisk symbol (*) adjacent to the Directory Number (DN) in the command output indicates that the Directory Number (DN) is an Overlay-dn.
Total ephones configured : 10
Total ephones registered : 3
Total ephones unregistered: 5
Total ephones deceased : 0
Ephones in unknown state : 2

Table 27: show ephone summary brief field descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DN</td>
<td>Directory number of the phone.</td>
</tr>
<tr>
<td>Ephone</td>
<td>ephone tag.</td>
</tr>
<tr>
<td>IP Address</td>
<td>IP address of the phone.</td>
</tr>
<tr>
<td>LN</td>
<td>Line number of the phone.</td>
</tr>
<tr>
<td>MacAddress</td>
<td>Shows the MAC address of the SCCP phone.</td>
</tr>
<tr>
<td>Number</td>
<td>Number assigned to ephone.</td>
</tr>
<tr>
<td>PhoneType</td>
<td>Shows the type of Cisco IP phone.</td>
</tr>
<tr>
<td>Status</td>
<td>Shows the registration status.</td>
</tr>
</tbody>
</table>

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone summary types</td>
<td>Displays the total number of registered and unregistered SCCP phones for each phone type.</td>
</tr>
</tbody>
</table>
show ephone summary

To display brief information about Cisco IP phones, use the `show ephone summary` command in privileged EXEC mode.

**show ephone summary**

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco CME 1.0 Cisco SRST 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco CME 2.0 Cisco SRST 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0 Cisco Unified SRST 8.0</td>
<td>This command was modified. The output was enhanced to show IPv6 or IPv4 addresses configured on phones.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0 Cisco Unified SRST 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1 Cisco Unified SRST 8.1</td>
<td>This command was modified. The output was enhanced to show voice-class stun-usage information.</td>
</tr>
</tbody>
</table>

**Examples**

The following is sample output from the `show ephone summary` command:

```
Router# show ephone summary
hairpin_block:
ephone-1[0] Mac:FCAC.3BAE.0000 TCP socket:[17] activeLine:0 whisperLine:0 REGISTERED
mediaActive:0 whisper_mediaActive:0 startMedia:0 offhook:0 ringing:0 reset:0 reset_sent:0
debug:0  primary_dn: 1*
  IP:10.2.1.0 * SCCP Gateway (AN) keepalive 2966 music 0 1:1
  port 0/0/0
voice-class stun is enabled
mediaActive:0 whisper_mediaActive:0 startMedia:0 offhook:0 ringing:0 reset:0 reset_sent:0
debug:0  primary_dn: 2*
  IP:10.2.1.5 * SCCP Gateway (AN) keepalive 2966 music 0 1:2
  port 0/0/1
voice-class stun is enabled
ephone-4 Mac:0030.94C3.F43A TCP socket:[-1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
  IP:10.2.1.11 Telecaster 7960 keepalive 59
Max 48, Registered 1, Unregistered 0, Deceased 0, Sockets 1
Max Conferences 4 with 0 active (4 allowed)
Skinny Music On Hold Status
Active MOH clients 0 (max 72), Media Clients 0
No MOH file loaded
```
The `show ephone` command describes significant fields in this output.

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td><code>show ephone</code></td>
<td>Displays statistical information about registered Cisco IP phones.</td>
</tr>
</tbody>
</table>
show ephone summary types

To display the total count of registered and unregistered phones for each phone type operating in the Skinny Client Control Protocol (SCCP) mode, use the `show ephone summary types` command in privileged EXEC mode.

**show ephone summary types**

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

This command has no default behavior or values.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command displays the count of configured, registered, unregistered, and deceased phones.

**Example**

The following is an example of the `show ephone summary types` command:

```
Router# show ephone summary types
```

```
PhoneType Configured Registered Unregistered Deceased Other
Unknown Ephone type 2 0 0 0 2
6901 1 0 1 0 0
8945 1 0 1 0 0
7970 3 2 1 0 0
8941 3 1 2 0 0
```

```
Total Phones 10 3 5 0 2
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone summary brief</td>
<td>Displays the details of all the SCCP phones configured.</td>
</tr>
</tbody>
</table>
show ephone tapiclients

To display status of ephone Telephony Application Programming Interface (TAPI) clients, use the `show ephone tapiclients` command in privileged EXEC mode.

**show ephone tapiclients**

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following is sample output from the `show ephone tapiclients` command:

```
Router# show ephone tapiclients
ephone-4 Mac:0007.0EA6.39F8 TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:192.168.1.18 50291 Telecaster 7960 sub=3 keepalive 728 max_line 20
button 1:dn 6 number 1004 CH1 IDLE    CH2 IDLE
button 2:dn 1 number 1000 CH1 IDLE    shared
button 3:dn 2 number 1000 CH1 IDLE    shared
button 7:dn 3 number 1001 CH1 IDLE    CH2 IDLE    monitor-ring shared
button 8:dn 4 number 1002 CH1 IDLE    CH2 IDLE    monitor-ring shared
button 9:dn 5 number 1003 CH1 IDLE    CH2 IDLE    monitor-ring
button 10:dn 91 number A00 auto dial A01 CH1 IDLE
speed dial 1:2000 PAGE-Staff
speed dial 2:2001 Hunt-Staff
paging-dn 90
Username:userB Password:ge30qe
Tapi client information
Username:userB status:REGISTERED Socket :[5]
    Tapi Client IP address: 192.168.1.5 Port:2295
```

The `show ephone` command describes significant fields in this output.

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>show ephone</code></td>
<td>Displays statistical information about registered Cisco IP phones.</td>
</tr>
</tbody>
</table>
show ephone telephone-number

To display information for the phone associated with a specified number, use the `show ephone telephone-number` command in privileged EXEC mode.

`show ephone telephone-number number`

**Syntax Description**

- `number` Telephone number that is associated with an ephone.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to find the phone on which a particular telephone number appears.

**Examples**

The following is sample output from the `show ephone telephone-number`:

```
Router# show ephone telephone-number 91400
DP tag: 0, primary
Tag 1, Normal or Intercom dn
  ephone 1, mac-address 000A.0E51.19F0, line 1
```

The `show ephone` command describes significant fields in this output.

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone</td>
<td>Displays statistical information about registered Cisco IP phones.</td>
</tr>
</tbody>
</table>
**show ephone unregistered**

To display information about unregistered phones, use the `show ephone unregistered` command in privileged EXEC mode.

```
show ephone unregistered
```

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

There are two ways that an ephone can become unregistered. The first way is when an ephone is listed in the running configuration but no physical device has been registered for that ephone. The second way is when an unknown device was registered at some time after the last router reboot but has since unregistered.

**Examples**

The following is sample output from the `show ephone unregistered`:

```
Router# show ephone unregistered
ephone-1 Mac:0007.0E81.10F0 TCP socket:[-1] activeLine:0 UNREGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:0.0.0.0 0 Unknown 0 keepalive 0 max_line 0
```

The `show ephone` command describes significant fields in this output.

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone</td>
<td>Displays statistical information about registered Cisco IP phones.</td>
</tr>
</tbody>
</table>
**show ephone unregistered summary**

To display the details of all the unregistered Skinny Call Control Protocol (SCCP) phones sorted by ephone tag, use the `show ephone unregistered summary` command in privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

This command has no default behavior or values.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3)M</td>
<td>Cisco SIP CME 10.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to view the details of the unregistered phones configured in the SCCP mode.

**Example**

The following is a sample output of the `show ephone unregistered summary` command.

```
router# show ephone unregistered summary
```

```
PhoneType Ephone MacAddress IpAddress Ln Dn Number Status
7970 3 001B.54CA.43F7 1 5 3005 Unregistered
2 6 3006 Unregistered
3 2* 3002 Unregistered
3 3* 3003 Unregistered
3 8* 3008 Unregistered
8945 4 D48C.B5C9.D2E6 1 7 3007 Unregistered
2 8 3008 Unregistered
8941 6 1111.2222.3333 Unregistered
8941 10 1111.2222.3334 Unregistered
6901 11 1111.2222.3332 Unregistered
```

Total ephones configured : 10
Total ephones registered : 3
Total ephones unregistered: 5
Total ephones deceased : 0
Ephones in unknown state : 2

**Note**

The * symbol adjacent to the Directory Number (DN) in the command output indicates that the Directory Number (DN) is an Overlay-dn.
Table 28: show ephone unregistered summary field descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DN</td>
<td>Directory number of the phone.</td>
</tr>
<tr>
<td>Ephone</td>
<td>Total number of ephone tags configured.</td>
</tr>
<tr>
<td>IP Address</td>
<td>IP address of the phones.</td>
</tr>
<tr>
<td>LN</td>
<td>Line number of the phone.</td>
</tr>
<tr>
<td>MacAddress</td>
<td>Shows the MAC address of the SCCP phone.</td>
</tr>
<tr>
<td>Number</td>
<td>Number assigned to ephone.</td>
</tr>
<tr>
<td>PhoneType</td>
<td>Shows the type of Cisco IP phone.</td>
</tr>
<tr>
<td>Status</td>
<td>Shows the registration status.</td>
</tr>
</tbody>
</table>

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone registered summary</td>
<td>Displays the details of all the registered SCCP phones.</td>
</tr>
<tr>
<td>show ephone summary types pattern</td>
<td>Displays the total number of registered and unregistered SCCP phones for each phone type.</td>
</tr>
</tbody>
</table>
show ephone-dn

To display status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CallManager Express (Cisco Unified CME) or Cisco Unified Survivable Remote Site Telephony (SRST) environment, use the `show ephone-dn` command in privileged EXEC mode.

**show ephone-dn [dn-tag]**

**Syntax Description**

- **dn-tag** (Optional) For Cisco Unified CME, a unique sequence number that is used during configuration to identify a particular extension (ephone-dn).
- (Optional) For Cisco Unified SRST, a destination number tag. The destination number can be from 1 to 288.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco CME 1.0 Cisco SRST 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco CME 2.0 Cisco SRST 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T command.</td>
</tr>
</tbody>
</table>

**Examples**

**Cisco Unified CME**

The following Cisco Unified CME sample output displays status and information for all ephone-dns:

```
Router# show ephone-dn
50/0/1 CH1 DOWN
EFXS 50/0/1 Slot is 50, Sub-unit is 0, Port is 1
Type of VoicePort is EFXS
Operation State is UP
Administrative State is UP
No Interface Down Failure
Description is not set
Noise Regeneration is enabled
Non Linear Processing is enabled
Non Linear Mute is disabled
Non Linear Threshold is -21 dB
Music On Hold Threshold is Set to -38 dBm
In Gain is Set to 0 dB
Out Attenuation is Set to 0 dB
Echo Cancellation is enabled
Echo Cancellation NLP mute is disabled
Echo Cancellation NLP threshold is -21 dB
Echo Cancel Coverage is set to 8 ms
Playout-delay Mode is set to adaptive
Playout-delay Nominal is set to 60 ms
Playout-delay Maximum is set to 200 ms
Playout-delay Minimum mode is set to default, value 40 ms
Playout-delay Fax is set to 300 ms
Connection Mode is normal
```
Connection Number is not set
Initial Time Out is set to 10 s
Interdigit Time Out is set to 10 s
Call Disconnect Time Out is set to 60 s
Ringing Time Out is set to 180 s
Wait Release Time Out is set to 30 s
Companding Type is u-law
Region Tone is set for US
Station name None, Station number 91400
Caller ID Info Follows:
Standard BELLCORE
Translation profile (Incoming):
Translation profile (Outgoing):
Digit Duration Timing is set to 100 ms
50/0/2 CH1 IDLE CH2 IDLE
EFXS 50/0/2 Slot is 50, Sub-unit is 0, Port is 2
Type of VoicePort is EFXS
Operation State is DORMANT
Administrative State is UP
No Interface Down Failure
Description is not set
Noise Regeneration is enabled
Non Linear Processing is enabled
Non Linear Mute is disabled
Non Linear Threshold is -21 dB
Music On Hold Threshold is Set to -38 dBm
In Gain is Set to 0 dB
Out Attenuation is Set to 0 dB
Echo Cancellation is enabled
Echo Cancellation NLP mute is disabled
Echo Cancellation NLP threshold is -21 dB
Echo Cancel Coverage is set to 8 ms
Playout-delay Mode is set to adaptive
Playout-delay Nominal is set to 60 ms
Playout-delay Maximum is set to 200 ms
Playout-delay Minimum mode is set to default, value 40 ms
Playout-delay Fax is set to 300 ms
Connection Mode is normal
Connection Number is not set
Initial Time Out is set to 10 s
Interdigit Time Out is set to 10 s
Call Disconnect Time Out is set to 60 s
Ringing Time Out is set to 180 s
Wait Release Time Out is set to 30 s
Companding Type is u-law
Region Tone is set for US
Station name None, Station number 91450
Caller ID Info Follows:
Standard BELLCORE
Translation profile (Incoming):
Translation profile (Outgoing):
Digit Duration Timing is set to 100 ms

**Cisco Unified SRST**

The following SRST sample output displays status and information for all ephone-dns:

Router# show ephone-dn 7
50/0/7 INVALID
EFXS 50/0/7 Slot is 50, Sub-unit is 0, Port is 7
  Type of VoicePort is EFXS
Operation State is UP
Administrative State is UP
No Interface Down Failure
Description is not set
Noise Regeneration is enabled
Non Linear Processing is enabled
Non Linear Mute is disabled
Non Linear Threshold is -21 dB
Music On Hold Threshold is Set to -38 dBm
In Gain is Set to 0 dB
Out Attenuation is Set to 0 dB
Echo Cancellation is enabled
Echo Cancellation NLP mute is disabled
Echo Cancellation NLP threshold is -21 dB
Echo Cancel Coverage is set to 8 ms
Playout-delay Mode is set to default
Playout-delay Nominal is set to 60 ms
Playout-delay Maximum is set to 200 ms
Playout-delay Minimum mode is set to default, value 4 ms
Playout-delay Fax is set to 300 ms
Connection Mode is normal
Connection Number is not set
Initial Time Out is set to 10 s
Interdigit Time Out is set to 10 s
Call Disconnect Time Out is set to 60 s
Ringing Time Out is set to 8 s
Wait Release Time Out is set to 30 s
Companding Type is u-law
Region Tone is set for US
Station name None, Station number None
Caller ID Info Follows:
Standard BELLCORE
Voice card specific Info Follows:
Digit Duration Timing is set to 100 ms

The following table describes significant fields in the output from this command.

### Table 28: show ephone-dn Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Administrative State</td>
<td>Administrative (configured) state of the voice port.</td>
</tr>
<tr>
<td>alert</td>
<td>The number of calls that were disconnected by the far-end device when the local IP phone was in the call alerting state (for example, because the far-end phone rang but was not answered and the far-end system decided to drop the call rather than let the phone ring for too long).</td>
</tr>
<tr>
<td>answered (incoming)</td>
<td>The number of incoming calls that were actually answered (the phone goes off hook when ringing).</td>
</tr>
<tr>
<td>answered (outgoing)</td>
<td>The number of outgoing call attempts that were answered by the far end.</td>
</tr>
<tr>
<td>busy</td>
<td>The number of outgoing call attempts that got a busy response.</td>
</tr>
<tr>
<td>Call Disconnect Time Out</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>called, calling</td>
<td>Extension numbers of called and calling parties.</td>
</tr>
<tr>
<td>Caller ID Info Follows</td>
<td>Information about the caller ID.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Call Ref</td>
<td>A unique per-call identifier used by the SCCP protocol. The Call Ref values are assigned sequentially within the Cisco CME–SCCP interface, so this value also indicates the total number of SCCP calls since the router was last rebooted.</td>
</tr>
<tr>
<td>chan</td>
<td>Channel number of an ephone-dn.</td>
</tr>
<tr>
<td>CODEC</td>
<td>Codec type.</td>
</tr>
<tr>
<td>Companding Type</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>connect</td>
<td>The number of calls that were disconnected by the far-end device when the local IP phone was in the call connected state.</td>
</tr>
<tr>
<td>Connection Mode</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>Connection Number</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>Description</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>Digit Duration Timing</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>DN STATE</td>
<td>Ephone-dn tag number and state of the phone line associated with an extension.</td>
</tr>
<tr>
<td>Echo Cancellation...</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>Echo Cancel Coverage</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>EFXS</td>
<td>Voice port type.</td>
</tr>
<tr>
<td>Far-end disconnect at...</td>
<td>See connect, alert, hold, and ring.</td>
</tr>
<tr>
<td>Final Jitter</td>
<td>The final voice packet receive jitter reported by the IP phone at the end of the call.</td>
</tr>
<tr>
<td>hold</td>
<td>The number of calls that were disconnected by the far-end device when the local IP phone was in the call hold state (for example, if the caller was left on hold for too long and got tired of waiting).</td>
</tr>
<tr>
<td>incoming</td>
<td>The number of incoming calls presented (the phone rings).</td>
</tr>
<tr>
<td>In Gain</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>Initial Time Out</td>
<td>Amount of time the system waits for an initial input digit from the caller.</td>
</tr>
<tr>
<td>Interdigit Time Out</td>
<td>Amount of time the system waits for a subsequent input digit from the caller.</td>
</tr>
<tr>
<td>Last 64 far-end disconnect cause codes</td>
<td>See the Mappings of PSTN Cause Codes to SIP Event table for a list of public switch telephone network (PSTN) cause codes that can be sent as an ISDN cause information element (IE) and the corresponding Session Interface Protocol (SIP) event.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------</td>
<td>--------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Latency</td>
<td>The final voice packet receive latency reported by the IP phone at the end of the call.</td>
</tr>
<tr>
<td>Lost</td>
<td>Number of lost packets.</td>
</tr>
<tr>
<td>Music On Hold Threshold</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>No Interface Down Failure</td>
<td>State of the interface.</td>
</tr>
<tr>
<td>Noise Regeneration</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>Non Linear...</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>Operation State</td>
<td>Operational state of the voice port.</td>
</tr>
<tr>
<td>Out Attenuation</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>outgoing</td>
<td>The number of outgoing call attempts.</td>
</tr>
<tr>
<td>Playout-delay Maximum</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>Playout-delay...</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>Port</td>
<td>Port number for the interface associated with the voice interface card.</td>
</tr>
<tr>
<td>Region Tone</td>
<td>Not applicable to the Cisco IP phone.</td>
</tr>
<tr>
<td>ring</td>
<td>The number of calls that were disconnected by the far-end device when the local IP phone was in the ringing state (for example, if the call was not answered and the caller hung up).</td>
</tr>
<tr>
<td>Ringing Time Out</td>
<td>Duration, in seconds, for which ringing is to continue if a call is not answered. Set with the <code>timeouts ringing</code> command.</td>
</tr>
<tr>
<td>Rx Pkts, bytes</td>
<td>Number of packets and bytes received during the current or last call.</td>
</tr>
<tr>
<td>Signal Level to phone, peak</td>
<td>For G.711 calls only, this parameter indicates the most recent voice signal level in the voice IP packets sent from the router to the IP phone. This parameter is valid only for VoIP or PSTN G.711 calls to the IP phones. This parameter is not valid for calls between local IP phones, or calls that use codecs other than G.711. The peak field indicates the peak signal level seen during the entire call.</td>
</tr>
<tr>
<td>Slot</td>
<td>Slot used in the voice interface card for this port.</td>
</tr>
<tr>
<td>Station name</td>
<td>Station name.</td>
</tr>
<tr>
<td>Station number</td>
<td>Station number.</td>
</tr>
<tr>
<td>Stream Port</td>
<td>RTP port allocated by the given DN/channel.</td>
</tr>
<tr>
<td>Sub-unit</td>
<td>Subunit used in the voice interface card for this port.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Tx Pkts, bytes</td>
<td>Number of packets and bytes transmitted during the current call or last call.</td>
</tr>
<tr>
<td>Type of VoicePort</td>
<td>Voice port type.</td>
</tr>
<tr>
<td>VAD</td>
<td>Voice activity detection.</td>
</tr>
<tr>
<td>Voice card specific info</td>
<td>Information specific to the voice card.</td>
</tr>
<tr>
<td>VPM STATE</td>
<td>State indication for the VPM software component.</td>
</tr>
<tr>
<td>VTSP STATE</td>
<td>State indication for the VTSP software component.</td>
</tr>
<tr>
<td>Wait Release Time Out</td>
<td>Time that a voice port stays in the call-failure state while the router sends a busy tone, reorder tone, or out-of-service tone to the port.</td>
</tr>
</tbody>
</table>

The following table lists the PSTN cause codes that can be sent as an ISDN cause information element (IE) and the corresponding SIP event for each. These are the far-end disconnect cause codes listed in the output for the `show ephone-dn statistics command`.

**Table 30: Mappings of PSTN Cause Codes to SIP Events**

<table>
<thead>
<tr>
<th>PSTN Cause Code</th>
<th>Description</th>
<th>SIP Event</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Unallocated number</td>
<td>410 Gone</td>
</tr>
<tr>
<td>3</td>
<td>No route to destination</td>
<td>404 Not found</td>
</tr>
<tr>
<td>16</td>
<td>Normal call clearing</td>
<td>BYE</td>
</tr>
<tr>
<td>17</td>
<td>User busy</td>
<td>486 Busy here</td>
</tr>
<tr>
<td>18</td>
<td>No user responding</td>
<td>480 Temporarily unavailable</td>
</tr>
<tr>
<td>19</td>
<td>No answer from the user</td>
<td></td>
</tr>
<tr>
<td>21</td>
<td>Call rejected</td>
<td>603 Decline</td>
</tr>
<tr>
<td>22</td>
<td>Number changed</td>
<td>302 Moved temporarily</td>
</tr>
<tr>
<td>27</td>
<td>Destination out of order</td>
<td>404 Not found</td>
</tr>
<tr>
<td>28</td>
<td>Address incomplete</td>
<td>484 Address incomplete</td>
</tr>
<tr>
<td>29</td>
<td>Facility rejected</td>
<td>501 Not implemented</td>
</tr>
<tr>
<td>31</td>
<td>Normal unspecified</td>
<td>404 Not found</td>
</tr>
<tr>
<td>PSTN Cause Code</td>
<td>Description</td>
<td>SIP Event</td>
</tr>
<tr>
<td>-----------------</td>
<td>--------------------------------------------------</td>
<td>----------------------------------</td>
</tr>
<tr>
<td>34</td>
<td>No circuit available</td>
<td>503 Service unavailable</td>
</tr>
<tr>
<td>38</td>
<td>Network out of order</td>
<td></td>
</tr>
<tr>
<td>41</td>
<td>Temporary failure</td>
<td></td>
</tr>
<tr>
<td>42</td>
<td>Switching equipment congestion</td>
<td></td>
</tr>
<tr>
<td>44</td>
<td>Requested channel not available</td>
<td></td>
</tr>
<tr>
<td>47</td>
<td>Resource unavailable</td>
<td></td>
</tr>
<tr>
<td>55</td>
<td>Incoming class barred within CUG</td>
<td>603 Decline</td>
</tr>
<tr>
<td>57</td>
<td>Bearer capability not authorized</td>
<td>501 Not implemented</td>
</tr>
<tr>
<td>58</td>
<td>Bearer capability not presently available</td>
<td></td>
</tr>
<tr>
<td>63</td>
<td>Service or option unavailable</td>
<td>503 Service unavailable</td>
</tr>
<tr>
<td>65</td>
<td>Bearer cap not implemented</td>
<td>501 Not implemented</td>
</tr>
<tr>
<td>79</td>
<td>Service or option not implemented</td>
<td></td>
</tr>
<tr>
<td>87</td>
<td>User not member of CUG</td>
<td>603 Decline</td>
</tr>
<tr>
<td>88</td>
<td>Incompatible destination</td>
<td>400 Bad Request</td>
</tr>
<tr>
<td>95</td>
<td>Invalid message</td>
<td></td>
</tr>
<tr>
<td>102</td>
<td>Recover on timer expiry</td>
<td>408 Request timeout</td>
</tr>
<tr>
<td>111</td>
<td>Protocol error</td>
<td>400 Bad request</td>
</tr>
<tr>
<td>127</td>
<td>Interworking unspecified</td>
<td>500 Internal server error</td>
</tr>
<tr>
<td></td>
<td>Any code other than those listed above</td>
<td>500 Internal server error</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone-dn callback</td>
<td>Displays information about pending callbacks in a Cisco Unified CME or a Cisco Unified SRST environment.</td>
</tr>
<tr>
<td>show ephone-dn loopback</td>
<td>Displays information about loopback ephone-dns that have been created in a Cisco Unified CME or a Cisco Unified SRST environment.</td>
</tr>
<tr>
<td>show ephone-dn statistics</td>
<td>Displays display call statistics for a Cisco IP destination or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.</td>
</tr>
</tbody>
</table>
### Command Table

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>show ephone-dn summary</strong></td>
<td>Displays brief information about Cisco IP phone destination numbers or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.</td>
</tr>
</tbody>
</table>
show ephone-dn callback

To display information about pending callbacks in a Cisco Unified CallManager Express (Cisco Unified CME) or a Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) environment, use the `show ephone-dn callback` command in privileged EXEC mode.

### Syntax Description
This command has no arguments or keywords.

### Command Modes
Privileged EXEC

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

### Examples

The following sample output shows a callback placed by ephone-dn 1 against ephone-dn 3. Ephone-dn 3 has its channel 1 on hold and has just seized dial tone on its channel 2.

```plaintext
Router# show ephone-dn callback
DN 3 (95021) CallBack pending to DN 1 (95021) for ephone-1 age 7 seconds
State for DN 3 is CH1 HOLD  CH2 SIEZE
```

The following sample output shows a callback placed by ephone-dn 1 against ephone-dn 3. Ephone-dn 3 has a call in progress on channel 1.

```plaintext
Router# show ephone-dn callback
DN 3 (95021) CallBack pending to DN 1 (95021) for ephone-1 age 8 seconds
State for DN 3 is CH1 CONNECTED
```

Significant fields in the output from this command are described in the following table.

### Table 31: show ephone-dn callback Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DN 3 (95021) CallBack pending to DN 1 (95021)</td>
<td>Callback originator is the extension with the dn-tag 1 (in this example), and the callback has been placed on the extension with the dn-tag 3 and the number 95021.</td>
</tr>
<tr>
<td>age</td>
<td>Number of seconds since the callback was placed.</td>
</tr>
<tr>
<td>State for DN 3 is CH1... CH2...</td>
<td>Call states for channel 1 and channel 2, if any, of the extension that the callback is for.</td>
</tr>
</tbody>
</table>
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>show ephone-dn</code></td>
<td>Displays status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.</td>
</tr>
</tbody>
</table>
show ephone-dn conference

To display information about ad hoc and meet-me conferences in a Cisco Unified CallManager Express (Cisco Unified CME) environment, use the **show ephone-dn conference** command in privileged EXEC mode.

```
show ephone-dn conference [{ad-hoc [video]|meetme [video]|number number}]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ad-hoc</td>
<td>(Optional) Displays ad hoc conferences.</td>
</tr>
<tr>
<td>meetme</td>
<td>(Optional) Displays meet-me conferences.</td>
</tr>
<tr>
<td>video</td>
<td>(Optional) Displays video conferences.</td>
</tr>
<tr>
<td>number</td>
<td>(Optional) Displays the conference telephone or extension number.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was modified. The command output was enhanced to display the unlocked Meet-Me conference setting.</td>
</tr>
<tr>
<td>15.1(4)M</td>
<td>Cisco CME 8.6</td>
<td>This command was modified to display information on video conferences.</td>
</tr>
</tbody>
</table>

**Examples**

The following sample output displays information for the 1397 conference number. There are three directory numbers and six inactive parties. The number of unlocked DN tags are displayed at the end of each MeetMe conference.

```
Router# show ephone-dn conference number 1397
type    active inactive numbers
=================================================
Meetme  0       6        1397
DN tags: 10, 11, 12
Unlocked DN tags: 2/3
Meetme  0       4        2486
DN tags: 13, 14
All DN tags unlocked.
Meetme  0       4        1111
DN tags: 15, 16
Ad-hoc   0       4        7777
DN tags: 20, 21
Router# sh ephone-dn conference ad-hoc video
type    active inactive numbers
=================================================
Ad-hoc-video  3       3       2000
```
DN tags: 20, 21, 22
Router# sh ephone-dn conference meetme video
type active inactive numbers
-----------------------------------------------
Meetme-video 0 8 3000
1. DN tags: 25

The following table describes the significant fields shown in the display.

**Table 32: show ephone-dn conference Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>active</td>
<td>Number of active parties in the conference.</td>
</tr>
<tr>
<td>DN tags</td>
<td>Directory numbers (DNs) in the conference.</td>
</tr>
<tr>
<td>inactive</td>
<td>Number of inactive parties in the conference.</td>
</tr>
<tr>
<td>number</td>
<td>Conference telephone or extension number.</td>
</tr>
<tr>
<td>type</td>
<td>Type of conference: meet-me or ad hoc.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone-dn</td>
<td>Displays status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.</td>
</tr>
</tbody>
</table>
show ephone-dn loopback

To display information about loopback ephone-dns that have been created in a Cisco Unified CallManager Express (Cisco Unified CME) or a Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) environment, use the show ephone-dn loopback command in privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco CME 1.0 Cisco SRST 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco CME 2.0 Cisco SRST 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following example displays information for a loopback using ephone-dn 21 and ephone-dn 22:

```
Router# show ephone-dn loopback
LOOPBACK DN status (min 21, max 22):
DN 21 51... Loopback to DN 22 CH1 IDLE
CallingDn -1 CalledDn -1 Called  Calling  G711Ulaw64k
Strip NONE, Forward 2, prefix 10 retry 10 Media 0.0.0.0 0
callID 0 srcCallID 0 ssrc 0 vector 0
DN 22 11... Loopback to DN 21 CH1 IDLE
CallingDn -1 CalledDn -1 Called  Calling  G711Ulaw64k
Strip NONE, Forward 2, prefix 50 retry 10 Media 0.0.0.0 0
callID 0 srcCallID 0 ssrc 0 vector 0
```

Significant fields in the output from this command are described in the following table.

**Table 33: show ephone-dn loopback Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Called, Calling</td>
<td>Called number and calling number when there is a call present.</td>
</tr>
<tr>
<td>CalledDn, CallingDn</td>
<td>Ephone-dn tag numbers of the called and calling ephone-dn. Set to -1 if the call is not to or from an ephone-dn, or if there is no active call.</td>
</tr>
<tr>
<td>callID</td>
<td>Internal call reference. This usage is the same as in other Cisco IOS voice gateway commands.</td>
</tr>
<tr>
<td>DN</td>
<td>Ephone-dn tag (sequence number).</td>
</tr>
<tr>
<td>Forward</td>
<td>Number of digits in the original called number to forward to the other ephone-dn in the loopback-dn pair.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>G711...</td>
<td>G711Ulaw64k indicates G.711 codec, mu-law, 64000-bit stream. G711alaw64k indicates G.711 codec, A-law, 64000-bit stream.</td>
</tr>
<tr>
<td>Loopback to command...</td>
<td>Indicates the opposite ephone-dn in the loopback pair and the status of that ephone-dn.</td>
</tr>
<tr>
<td>Media</td>
<td>IP destination address, if any, for any voice packets that are passing through the loopback DN.</td>
</tr>
<tr>
<td>min, max</td>
<td>Lowest and highest dn-tag numbers of ephone-dns that are configured as loopback-dns.</td>
</tr>
<tr>
<td>prefix</td>
<td>Digit string to add to the beginning of forwarded called numbers.</td>
</tr>
<tr>
<td>retry</td>
<td>Number of seconds to wait before retrying the loopback target when is it busy.</td>
</tr>
<tr>
<td>srcCallID</td>
<td>Internal call reference for the destination.</td>
</tr>
<tr>
<td>ssrc</td>
<td>Real-time transport protocol (RTP) synchronization source (SSRC) of the most recent RTP packet.</td>
</tr>
<tr>
<td>Strip</td>
<td>Number of leading digits to strip before forwarding to the other extension in the loopback-dn pair.</td>
</tr>
<tr>
<td>vector</td>
<td>The following values describe the media path for voice packets that pass through the loopback-dn:</td>
</tr>
<tr>
<td></td>
<td>• 0—No media path or not a loopback-dn path (inactive).</td>
</tr>
<tr>
<td></td>
<td>• 1—Normal path. Loopback-dn has identified the final media destination as a local IP phone. The media IP address field shows a valid, non-zero value.</td>
</tr>
<tr>
<td></td>
<td>• 2—Hairpin. Media packets are routed back through paired loopback-dns. The final destination is not known. For example, this can be a VoIP-to-VoIP call path by a loopback-dn.</td>
</tr>
<tr>
<td></td>
<td>• 3—Hairpin. The final destination is an ephone-dn in a special mode such as paging.</td>
</tr>
<tr>
<td></td>
<td>• 4—Loopback-dn chain has been detected, in which two loopback-dn pairs have been connected together.</td>
</tr>
<tr>
<td></td>
<td>• 5—Loopback-dn chain has been detected in which more than two loopback-dn pairs are connected in series.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>loopback-dn</td>
<td>Creates a virtual loopback voice port (loopback-dn) to establish a demarcation point for VoIP voice calls and supplementary services.</td>
</tr>
<tr>
<td>show ephone-dn</td>
<td>Displays status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.</td>
</tr>
</tbody>
</table>
show ephone-dn paging

To display configuration information on paging groups, use the show ephone-dn paging command in user EXEC or privileged EXEC mode.

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC (>

Privileged EXEC (#)

Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
</table>
| 15.2(T) | This command was introduced.

Usage Guidelines

Use the show ephone-dn paging command to display which paging dn is specified and which phone is being paged.

Examples

The following is a sample output from the show ephone-dn paging command before paging. The output shows two parts: the static “Paging Configuration” part and the dynamic “Paging Control Info” part. The output of the show ephone-dn paging command should be exactly the same before and after paging.

Router# show ephone-dn paging
Paging Configuration
ephone-dn 250 ( IDLE )
  number 7770
  paging ip 239.1.1.0 port 20480
    ephone-2[1] paging-dn 250(OFF)
    ephone-7[6] paging-dn 250(OFF)
  paging group 251,252
    voice reg pool 1 pagingGrp 251(OFF)
    voice reg pool 2 pagingGrp 252(OFF)
  ephone-dn 251 ( IDLE )
    number 7771
    paging ip 239.1.1.1 port 20480
    voice reg pool 1 paging-dn 251(OFF)
  ephone-dn 252 ( IDLE )
    number 7772
    paging ip 239.1.1.2 port 20480
    voice reg pool 2 paging-dn 252(OFF)
  ephone-dn 253 ( IDLE )
    number 7773
    paging ip 239.1.1.3 port 20480
    ephone-8[7] paging-dn 253(OFF)

Paging Control Info
skinnypC[0] ephone-paging-dn 250 ( IDLE ) count 0
skinnypC[1] ephone-paging-dn 251 ( IDLE ) count 0
skinnypC[2] ephone-paging-dn 252 ( IDLE ) count 0
skinnypC[4] ephone-paging-dn 253 ( IDLE ) count 0
The following is a sample output from the `show ephone-dn paging` command during paging. In this output, the “Paging Configuration” part remains the same except for the changes in state from IDLE to ACTIVE and OFF to ON. However, the “Paging Control Info” part displays the changes in the paging control information.

```
Router# show ephone-dn paging
  Paging Configuration
  ephone-dn 250 (ACTIVE)
    number 7770
    paging ip 239.1.1.0 port 20480
      ephone-2[1] paging-dn 250(ON )
      ephone-7[6] paging-dn 250(OFF)
    paging group 251,252
      voice reg pool 1 pagingGrp 251(ON )
      voice reg pool 2 pagingGrp 252(ON )
  ephone-dn 251 ( IDLE )
    number 7771
    paging ip 239.1.1.1 port 20480
      voice reg pool 1 paging-dn 251(ON )
  ephone-dn 252 ( IDLE )
    number 7772
    paging ip 239.1.1.2 port 20480
      voice reg pool 2 paging-dn 252(ON )
  ephone-dn 253 ( IDLE )
    number 7773
    paging ip 239.1.1.3 port 20480
      ephone-8[7] paging-dn 253(ON )

  Paging Control Info
  skinnyPC[0] ephone-paging-dn 250 (ACTIVE) count 1
    phone ip address port
      ephone#[phone] 2[1] 239.1.1.0 20480
    sccp(ephone#[phone]): 2[1](mcast)
    group 251 (ephone#[phone]): None
    group 252 (ephone#[phone]): None
    sip (pool[peer tag]): None
      group 251 (pool[peer tag]): 1[40001](mcast)
      group 252 (pool[peer tag]): 2[40003](mcast)
  skinnyPC[1] ephone-paging-dn 251 ( IDLE ) count 0
  skinnyPC[2] ephone-paging-dn 252 ( IDLE ) count 0
  skinnyPC[4] ephone-paging-dn 253 ( IDLE ) count 0
```

The following is another sample output from the `show ephone-dn paging` command during paging:

```
Paging Configuration
  ephone-dn 250 ( IDLE )
    number 7770
    paging ip 239.1.1.0 port 20480
    paging group 251
      ephone-2[1] pagingGrp 251(ON )
      voice reg pool 3 pagingGrp 251(ON )
  ephone-dn 251 (ACTIVE)
    number 7771
    paging ip 239.1.1.1 port 20480
      ephone-2[1] paging-dn 251(ON )
      voice reg pool 3 paging-dn 251(ON )

  Paging Control Info
  skinnyPC[0] ephone-paging-dn 250 ( IDLE ) count 0
  skinnyPC[1] ephone-paging-dn 251 (ACTIVE) count 1
    phone ip address port
      ephone#[phone] 2[1] 239.1.1.1 20480
    sccp(ephone#[phone]): 2[1](m)
    sip (pool[peer tag]): 3[40007](m)
```
The following table describes the significant fields shown in the display.

**Table 34: show ephone-dn paging Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>phone</td>
<td>Indicates the ephone-dn and the paging-dn tag.</td>
</tr>
<tr>
<td>ip address</td>
<td>Indicates the IP multicast address to multicast voice packets for audio paging.</td>
</tr>
<tr>
<td>port</td>
<td>Indicates the UDP port for multicast paging. Range is from 2000 to 65535. The correct paging port for the paging-dn of Cisco Unified SIP IP phones is an even number from 20480 to 32768 only.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>paging-dn</td>
<td>Creates a paging extension (paging-dn) to receive audio pages on a Cisco Unified IP phone in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>paging-dn (voice register)</td>
<td>Registers a Cisco Unified SIP IP phone to an ephone-dn paging directory number.</td>
</tr>
<tr>
<td>paging group</td>
<td>Creates a combined paging group from two or more previously established paging sets.</td>
</tr>
</tbody>
</table>
show ephone-dn park

To display information about call-park slots in the system, use the **show ephone-dn park** command in privileged EXEC mode.

### Syntax Description
This command has no arguments or keywords.

### Command Modes
Privileged EXEC

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(7)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Examples
The following example shows information for a single call-park slot that uses an ephone-dn identifier of 50 and an extension number of 1560.

Router

```bash
# show ephone-dn park
DN 50 (1560) park-slot state IDLE
Notify to () timeout 15 limit 20
```

The following table describes the significant fields shown in the display.

**Table 35: show ephone-dn park Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DN</td>
<td>Ephone-dn tag (identifier) number for the call-park slot.</td>
</tr>
<tr>
<td>(1560)</td>
<td>Extension number associated with the call-park slot.</td>
</tr>
<tr>
<td>park-slot state</td>
<td>Whether the call-park slot is in use or idle.</td>
</tr>
<tr>
<td>Notify to ( )</td>
<td>Extension that has been specified for notification. Empty parentheses indicate that no extension was specified in the configuration.</td>
</tr>
<tr>
<td>timeout</td>
<td>Number of seconds between reminder rings, in seconds.</td>
</tr>
<tr>
<td>limit</td>
<td>Number of reminder rings before a call parked at this slot is disconnected.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>park-slot</td>
<td>Creates a floating extension (ephone-dn) at which calls can be temporarily held (parked).</td>
</tr>
</tbody>
</table>
show ephone-dn statistics

To display call statistics for a Cisco IP destination or for extensions (ephone-dns) in a Cisco Unified CallManager Express (Cisco Unified CME) or a Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) environment, use the `show ephone-dn` command in privileged EXEC mode.

```
show ephone-dn [dn-tag] statistics
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dn-tag</td>
<td>(Optional) Unique sequence number that is used during configuration to identify a particular extension (ephone-dn).</td>
</tr>
<tr>
<td>statistics</td>
<td>Displays voice quality statistics on calls for a specified extension or for all extensions.</td>
</tr>
</tbody>
</table>

**Command Modes**

- Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ1</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0 Cisco SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following sample output displays statistics for all extensions (ephone-dns) in a Cisco Unified CME system. There are two ephone-dns (DN1 and DN3) in this example.

```
Router# show ephone-dn statistics
Total Calls 103
Stats may appear to be inconsistent for conference or shared line cases
DN 1 chan 1 incoming 36 answered 21 outgoing 60 answered 30 busy 6
Far-end disconnect at: connect 29 alert 18 hold 7 ring 15
Last 64 far-end disconnect cause codes
17 17 17 17 17 17 17 17 17 17 17 17 17 17 17 17
16 16 16 16 16 16 16 16 16 16 16 16 16 16 16 16
local phone on-hook
DN 1 chan 1 (95011) voice quality statistics for last call
Call Ref 103 called 91500 calling 95011
Total Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Final Jitter 30 Latency 0 Lost 0
Packets counted by router 0
DN 1 chan 2 incoming 0 answered 0 outgoing 1 answered 0 busy 0
Far-end disconnect at: connect 0 alert 0 hold 0 ring 0
Last 64 far-end disconnect cause codes
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
local phone on-hook
DN 1 chan 2 (95011) voice quality statistics for last call
Call Ref 86 called calling
Total Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Final Jitter 0 Latency 0 Lost 0
```
Signal Level to phone 0 (-78 dB) peak 0 (-78 dB)
Packets counted by router 0
DN 3 chan 1 incoming 0 answered 0 outgoing 1 answered 1 busy 0
Far-end disconnect at: connect 0 alert 0 hold 0 ring 0
Last 64 far-end disconnect cause codes
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
DN 3 chan 1 (95021) voice quality statistics for current call
Call Ref 102 called 94011 calling 95021
Current Tx Pkts 241 bytes 3133 Rx Pkts 3304 bytes 515023 Lost 0
Jitter 30 Latency 0
Worst Jitter 30 Worst Latency 0
Signal Level to phone 201 (-39 dB) peak 5628 (-12 dB)
Packets counted by router 3305

The following sample output displays voice quality statistics for the ephone-dn with dn-tag 2:

Router# show ephone-dn statistics
DN 2 chan 1 incoming 0 answered 0 outgoing 2 answered 0 busy 0
Far-end disconnect at: connect 0 alert 0 hold 0 ring 0
Last 64 far-end disconnect cause codes
28 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
local phone on-hook
DN 2 chan 1 (91450) voice quality statistics for last call
Call Ref 2 called calling
Total Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Final Jitter 0 Latency 0 Lost 0
Signal Level to phone 0 (-78 dB) peak 0 (-78 dB)
Packets counted by router 0

The show ephone-dn command describes significant fields in the output from this command.

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show ephone-dn</td>
<td>Displays status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.</td>
</tr>
</tbody>
</table>
show ephone-dn summary

To display brief information about Cisco IP phone destination numbers or for extensions (ephone-dns) in a Cisco Unified CallManager Express (Cisco Unified CME) or a Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) environment, use the show ephone-dn summary command in privileged EXEC mode.

show ephone-dn summary

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)  

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco CME 1.0 Cisco SRST 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco CME 2.0 Cisco SRST 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

Examples

The following is example output from the show ephone-dn summary:

```
Router# show ephone-dn summary

+----------+--------+---------+--------+-------------------+---------+---------+----------+
| PORT     | DN STATE| CODEC   | VAD    | VTSP STATE        | VPM STATE|
|----------+--------+---------+--------+-------------------+---------+---------+----------+
| 50/0/1   | DOWN   | -       | -      | -                 | EFXS_ONHOOK |
| 50/0/2   | DOWN   | -       | -      | -                 | EFXS_ONHOOK |
| 50/0/3   | DOWN   | -       | -      | -                 | EFXS_ONHOOK |
| 50/0/4   | INVALID| -       | -      | -                 | EFXS_INIT |
| 50/0/5   | INVALID| -       | -      | -                 | EFXS_INIT |
| 50/0/6   | INVALID| -       | -      | -                 | EFXS_INIT |
```

The following table describes significant fields in the output from this command.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>CODEC</td>
<td>Type of codec.</td>
</tr>
<tr>
<td>DN STATE</td>
<td>Status of the ephone-dn.</td>
</tr>
<tr>
<td>EFXS</td>
<td>Voice port type.</td>
</tr>
<tr>
<td>PORT</td>
<td>Port number (virtual) for this interface. The number that follows the last slash in the port number is the ephone-dn tag. For example, if the port number is 50/0/1, the dn-tag is 1.</td>
</tr>
<tr>
<td>VAD</td>
<td>Voice activity detection status.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-----------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>VPM STATE</td>
<td>State indication for the voice port module (VPM) software component.</td>
</tr>
<tr>
<td>VTSP STATE</td>
<td>State indication for the voice telephony service provider (VTSP) software component.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>show ephone-dn</code></td>
<td>Displays status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.</td>
</tr>
</tbody>
</table>
**show ephone-dn whisper**

To display information about whisper intercom ephone-dns that have been created in Cisco Unified CME, use the `show ephone-dn whisper` command in privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

User EXEC (>)

Privileged EXEC (＃)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following is sample output from the `show ephone-dn whisper` command showing an active whisper intercom call between extension 6001 and 6002:

```
Router# show ephone-dn whisper

  DN  DN NUMBER  LABEL      SPEED DIAL  DN STATE  PHONE
     ---------  ---------  ----------  ----------  ----------  ----
  101  8881      wi_8881   -          IDLE  35 w36
  102  8882      -          -          IDLE  36       
  103  8883      wi_8883   -          IDLE  m35 38   
  104  8884      wi_8884   -          IDLE  38       
  105  8885      wi_8885_ad_8888882 IDLE       
  106  8886      wi_8886_ad_8888883 IDLE  36       
  107  8887      -          8888       IDLE  35       
  108  8888      Mary_sd_Peter 8887 IDLE  36       
  109  8889      -          -          IDLE       
  110  8890      wi_8890   -          IDLE       
  111  4441      4441_wi_sd_4444442 IDLE       
  112  4442      wi_4442   -          IDLE       
  113  4443      -          -          IDLE       
  114  4444      4444_sd-88882 8882 IDLE       
  141  5551      -          -          IDLE       
  142  5552      -          -          IDLE       
  143  5553      -          -          IDLE       
  144  5554      -          -          IDLE       
  145  5555      -          -          IDLE       
  161  6001      -          6002       WHISPER 1   
  162  6002      -          6001       WHISPER 2   
  163  6003      -          6001       IDLE       
  164  6004      -          6002       IDLE       
  166  6006      -          6003       IDLE       
  167  6007      -          6003       IDLE       
  168  6008      -          6002       IDLE       
  169  6009      -          6006       IDLE       
```
The following table describes the significant fields in the output from this command in alphabetical order.

**Table 37: show ephone-dn whisper Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DN</td>
<td>Directory number tag.</td>
</tr>
<tr>
<td>DN Number</td>
<td>Extension or telephone number assigned to directory number.</td>
</tr>
<tr>
<td>Label</td>
<td>Text string that identifies the whisper intercom line.</td>
</tr>
<tr>
<td>Speed Dial</td>
<td>Whisper intercom number to speed dial.</td>
</tr>
<tr>
<td>DN State</td>
<td>State of the directory number, either Idle or Busy.</td>
</tr>
<tr>
<td>Phone</td>
<td>Ephone that the directory number is assigned to.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug ephone whisper-intercom</td>
<td>Displays debugging messages for the Whisper Intercom feature.</td>
</tr>
<tr>
<td>show ephone-dn</td>
<td>Displays status and configuration information for phone extensions (ephone-dns) in Cisco Unified CME.</td>
</tr>
<tr>
<td>whisper-intercom</td>
<td>Enables the Whisper Intercom feature on a directory number.</td>
</tr>
</tbody>
</table>
show ephone-hunt

To display ephone-hunt configuration information and current status and statistics information, use the show ephone-hunt command in privileged EXEC mode.

```
show ephone-hunt [{tag|summary}]
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>tag</code></td>
<td>(Optional) Hunt-group number that was used to identify a hunt group in the ephone-hunt command. Range is 1 to 100.</td>
</tr>
<tr>
<td><code>summary</code></td>
<td>(Optional) Displays hunt group configuration information.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `show ephone-hunt` and `show ephone-hunt summary` commands display information for peer, sequential, and last-idle ephone hunt groups. Using the `tag` argument outputs data for a specific ephone hunt group.

The output is dependent on call activity. If there is no activity, no data is displayed.

**Examples**

The following examples are contained in this section:

**Verbose Output**

The following is a sample output from the `show ephone-hunt` command when no argument or keyword has been entered. The sample contains information for a peer hunt group, a sequential hunt group, and a longest-idle hunt group. See the table for descriptions of significant fields in the output.

```
Router# show ephone-hunt
Group 1
  type: peer
  pilot number: 450, peer-tag 20123
  list of numbers:
    451, aux-number A450A0900, # peers 5, logout 0, down 1
    peer-tag  dn-tag  rna  login/logout up/down
    [20122  42  0  login  up ]
    [20121  41  0  login  up ]
    [20120  40  0  login  up ]
    [20119  30  0  login  up ]
    [20118  29  0  login  down]
    452, aux-number A450A0901, # peers 4, logout 0, down 0
    peer-tag  dn-tag  rna  login/logout up/down
    [20127  45  0  login  up ]
    [20126  44  0  login  up ]
    [20125  43  0  login  up ]
    [20124  31  0  login  up ]
```
453, aux-number A450A0902, # peers 4, logout 0, down 0
peer-tag dn-tag rna login/logout up/down
[20131 48 0 login up ]
[20130 47 0 login up ]
[20129 46 0 login up ]
[20128 32 0 login up ]

477, aux-number A450A0903, # peers 1, logout 0, down 0
peer-tag dn-tag rna login/logout up/down
[20132 499 0 login up ]

preference: 0
members initial state: logout
preference (sec): 7
timeout: 3, 3, 3, 3
max timeout : 10
hops: 4
next-to-pick: 1
E.164 register: yes
auto logout: no
stat collect: no

Group 2
type: sequential
pilot number: 601, peer-tag 20098
list of numbers:
123, aux-number A601A0200, # peers 1, logout 0, down 0
peer-tag dn-tag rna login/logout up/down
[20097 56 0 login up ]

622, aux-number A601A0201, # peers 3, logout 0, down 0
peer-tag dn-tag rna login/logout up/down
[20101 112 0 login up ]
[20100 111 0 login up ]
[20099 110 0 login up ]

623, aux-number A601A0202, # peers 3, logout 0, down 0
peer-tag dn-tag rna login/logout up/down
[20104 122 0 login up ]
[20103 121 0 login up ]
[20102 120 0 login up ]
*, aux-number A601A0203, # peers 1, logout 0, down 1
peer-tag dn-tag rna login/logout up/down
[20105 0 0 - down]
*, aux-number A601A0204, # peers 1, logout 0, down 1
peer-tag dn-tag rna login/logout up/down
[20106 0 0 - down]

final number: 5255348
preference: 0
members initial state: logout
preference (sec): 9
timeout: 5, 5, 5, 5, 5
max timeout : 40
fwd-final: orig-phone
E.164 register: yes
auto logout: no
stat collect: no

Group 3
type: longest-idle
pilot number: 100, peer-tag 20142
list of numbers:
101, aux-number A100A9700, # peers 3, logout 0, down 3
on-hook time stamp 7616, off-hook agents=0
peer-tag dn-tag rna login/logout up/down
[20141 132 0 login down]
[20140 131 0 login down]
[20139 130 0 login down]
*, aux-number A100A9701, # peers 1, logout 0, down 1
on-hook time stamp 7616, off-hook agents=0
Summary Output

The following example shows a summary output. See the table for descriptions of significant fields in the output.

Router# `show ephone-hunt summary`

Group 1
- type: peer
- pilot number: 5000
- list of numbers:
  - 5001
  - 5002
  - 5003
  - 5004
  - 5005
- final number: 5006
- preference: 0
- members initial state: logout
- timeout: 180
- hops: 2
- E.164 register: yes

Group 2
- type: sequential
- pilot number: 6000
- list of numbers:
  - 5005
  - 5004
  - 5003
  - 5002
  - 5001
- final number: 5007
- preference: 5
- members initial state: logout
- timeout: 3
- E.164 register: no

Agent Status Control Conditions

A portion of the `show ephone-hunt` command output displays the ready and not-ready agent status of extensions in hunt groups. An extension that is ready is available to receive hunt-group calls. An
extension that is in not-ready status blocks hunt-group calls. An agent toggles an extension from ready to not ready and back to ready using the HLog soft key or a FAC.

The following examples display some output that reports different agent status not-ready conditions within a hunt group. In the hunt group used for these examples, there are four users: agent1 and agent4 share extension 8001, agent2 is on extension 8002, and agent3 is on extension 8003.

In the `show ephone-hunt` output, “logout 0” means that all instances of the extension are in ready status. Any number greater than zero next to “logout” indicates that at least one ephone using the extension has activated not-ready status.

If agent1 is in not-ready status, the `show ephone-hunt` command will display the following output. The logout value for extension 8001 is 1 because one phone is in not-ready status.

```
Router# show ephone-hunt
```

```
list of numbers:
8001, aux-number A8000A100, # peers 2, logout 1 ...
8002, aux-number A8000A101, # peers 1, logout 0...
8003, aux-number A8000A102, # peers 1, logout 0...
```

If agent1 and agent2 place their phones in not-ready status, the `show ephone-hunt` command will display the following output:

```
Router# show ephone-hunt
```

```
list of numbers:
8001, aux-number A8000A100, # peers 2, logout 1...
8002, aux-number A8000A101, # peers 1, logout 1...
8003, aux-number A8000A102, # peers 1, logout 0...
```

If all agents place their phones in not-ready status, the `show ephone-hunt` command displays the following output. Note that the logout value of 2 for extension 8001 indicates that both ephone-dns with that extension number (agent1 and agent4) are in not-ready status.

```
Router# show ephone-hunt
```

```
list of numbers:
8001, aux-number A8000A100, # peers 2, logout 2...
8002, aux-number A8000A101, # peers 1, logout 1...
8003, aux-number A8000A102, # peers 1, logout 1...
all agents logout!
```

**Automatic Agent Status Not-Ready Parameters**

The `show ephone-hunt` command displays the parameters that have been set using the `auto logout` command, which is used for the Automatic Agent Status Not-Ready feature. The table shows the possible values of the auto logout field. describes other fields in the output.

```
Router# show ephone-hunt 1
```
Group 1
  type: sequential
  pilot number: 8888, peer-tag 20029
  list of numbers:
  8001, aux-number A8888A000, # peers 1, logout 0, down 0
  peer-tag:dn-tag [ 20028:1]
  8003, aux-number A8888A001, # peers 1, logout 0, down 0
  peer-tag:dn-tag [ 20030:3]
  preference: 0
  members initial state: logout
  preference (sec): 9
  timeout: 5
  E.164 register: yes
  auto logout: no
  stat collect: yes

Table 38: show ephone-hunt Auto Logout Examples

<table>
<thead>
<tr>
<th>show ephone-hunt Output</th>
<th>Description</th>
<th>auto logout Command</th>
</tr>
</thead>
<tbody>
<tr>
<td>auto logout: no</td>
<td>The Automatic Agent Status Not-Ready feature is disabled. This is also the default if this command is not used.</td>
<td>no auto logout</td>
</tr>
<tr>
<td>auto logout: 1 type: both</td>
<td>The Automatic Agent Status Not-Ready feature is enabled and no options have been used with the auto logout command. The number of unanswered calls is 1 and the command applies to both static and dynamic hunt group members by default.</td>
<td>auto logout</td>
</tr>
<tr>
<td>auto logout: 2 type: both</td>
<td>Two unanswered calls will be sent to a hunt group agent before the agent’s status is automatically changed to not ready. The command applies to both static and dynamic hunt group members by default.</td>
<td>auto logout 2</td>
</tr>
<tr>
<td>auto logout: 3 type: static</td>
<td>Three unanswered calls will be sent to a hunt group agent before the agent’s status is automatically changed to not ready. The command applies to static hunt group members only.</td>
<td>auto logout 3 static</td>
</tr>
</tbody>
</table>

The table describes significant fields shown in **show ephone-hunt** command displays.

Table 39: show ephone-hunt Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>auto logout</td>
<td>Indicates whether the Automatic Agent Status Not-Ready feature has been enabled. See the table.</td>
</tr>
<tr>
<td>aux-number</td>
<td>Auxiliary number used to generate dial peers for a hunt group. This number is generated by the list command.</td>
</tr>
<tr>
<td>description</td>
<td>Description string entered for the ephone hunt group. This value is set using the description (ephone-hunt) command.</td>
</tr>
<tr>
<td>dn-tag</td>
<td>Directory number (DN) sequence number.</td>
</tr>
<tr>
<td>E.164 register</td>
<td>Displays whether a pilot number registers with an H.323 gatekeeper. This value is set by the no-reg command.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>final number</td>
<td>Last number in the ephone-hunt group, after which a call is no longer redirected. This value is set by the <strong>final</strong> command.</td>
</tr>
<tr>
<td>fwd-final</td>
<td>Final destination of an unanswered call that has been transferred into a hunt group: orig-phone means calls are returned to the transferring phone, and final means calls are sent to the final number specified in the configuration. This value is set by the <strong>fwd-final</strong> command.</td>
</tr>
<tr>
<td>hops</td>
<td>Number of hops before a call proceeds to the final number. This value is set by the <strong>hops</strong> command.</td>
</tr>
<tr>
<td>list of numbers</td>
<td>Extension numbers that are group members of the specified ephone hunt group. This value is set by the <strong>list</strong> command.</td>
</tr>
<tr>
<td>login/logout</td>
<td>Ready status of the agent: login means ready and accepting calls, and logout means not-ready and blocking hunt-group calls.</td>
</tr>
<tr>
<td>logout</td>
<td>Number of agents in the not-ready state (not accepting hunt-group calls).</td>
</tr>
<tr>
<td>max timeout</td>
<td>Maximum combined timeout for the no-answer periods for all ephone-dns in the ephone-hunt list. This value is set by the <strong>max-timeout</strong> command.</td>
</tr>
<tr>
<td>members initial state: logout/login</td>
<td>Sets all static members initial state to logout.</td>
</tr>
<tr>
<td>next-to-pick</td>
<td>(Peer hunt groups only) List number of the agent whose phone will ring when the next call comes in to the hunt group. (For example, if the order of agents in the <strong>list</strong> command is 451, 452, 453, 454, the list number 2 represents extension 452.)</td>
</tr>
<tr>
<td>off-hook agents</td>
<td>Number of agents who are currently off-hook.</td>
</tr>
<tr>
<td>on-hook time stamp</td>
<td>(Longest-idle hunt groups only) The last on-hook time of the agent, which is used to determine which agent to ring next time.</td>
</tr>
<tr>
<td>peers</td>
<td>Displays the number of ephone-dn dial peers.</td>
</tr>
<tr>
<td>peer-tag</td>
<td>Dial-peer sequence number.</td>
</tr>
<tr>
<td>pilot number</td>
<td>Number that callers dial to reach the ephone hunt group.</td>
</tr>
<tr>
<td>preference</td>
<td>Preference order set by the <strong>preference (ephone-hunt)</strong> command for the primary pilot number.</td>
</tr>
<tr>
<td>preference (sec)</td>
<td>Preference order set by the <strong>preference (ephone-hunt)</strong> command for the secondary pilot number.</td>
</tr>
<tr>
<td>rna</td>
<td>Number of unanswered hunt group calls (ring-no-answer) by this agent, used for the Automatic Agent Status Not-Ready feature.</td>
</tr>
<tr>
<td>stat collect</td>
<td>Indicates whether statistic are being Cisco Unified CME B-ACD data is being collected. See the <strong>statistics collect</strong> command.</td>
</tr>
</tbody>
</table>
timeout

Number of seconds after which a call that is not answered at one number is redirected to the next number in the hunt-group list. Multiple values in this field refer to the timeouts for the hops between ephone-dns in a hunt group as they appear in the list command. This value is set by the timeout command.

type

Type of ephone hunt group: longest-idle, peer, or sequential.

up/down

Dial peer is up or down.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>timeout</td>
<td>Number of seconds after which a call that is not answered at one number is redirected to the next number in the hunt-group list. Multiple values in this field refer to the timeouts for the hops between ephone-dns in a hunt group as they appear in the list command. This value is set by the timeout command.</td>
</tr>
<tr>
<td>type</td>
<td>Type of ephone hunt group: longest-idle, peer, or sequential.</td>
</tr>
<tr>
<td>up/down</td>
<td>Dial peer is up or down.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>auto logout</td>
<td>Enables automatic change of agent status to not-ready after a specified number of hunt-group calls are not answered.</td>
</tr>
<tr>
<td>ephone-hunt</td>
<td>Enters ephone-hunt configuration mode to create a hunt group for use in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>hunt-group logout</td>
<td>Enables separate handling of DND and HLog functionality for hunt-group agents and the display of an HLog soft key on phones.</td>
</tr>
<tr>
<td>members logout</td>
<td>Sets all static members initial state to logout.</td>
</tr>
<tr>
<td>show ephone-hunt statistics</td>
<td>Displays hunt group call statistics.</td>
</tr>
</tbody>
</table>
show ephone-hunt statistics

To display ephone-hunt statistics information, use the `show ephone-hunt statistics` command in privileged EXEC mode.

```
show ephone-hunt tag statistics {last hours hours|start day time [to day time]}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Tag</th>
<th>Hunt-tag number that was used to identify a hunt group in an <code>ephone-hunt</code> command. Range is 1 to 100.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Last</td>
<td>Displays information for the previous number of specified hours, counting backward from the current hour. Range is 1 to 167.</td>
</tr>
<tr>
<td>Hours</td>
<td>Number of hours for which to display call statistics.</td>
</tr>
<tr>
<td>Start</td>
<td>Defines the start of a period for which to display call statistics. Default duration is one hour.</td>
</tr>
<tr>
<td>Day</td>
<td>Day of week. Use <code>sun</code>, <code>mon</code>, <code>tue</code>, <code>wed</code>, <code>thu</code>, <code>fri</code>, or <code>sat</code>.</td>
</tr>
<tr>
<td>Time</td>
<td>Hour of day. Range is 0 to 23.</td>
</tr>
<tr>
<td>To</td>
<td>(Optional) Defines the stop time for display of call statistics.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco CME 3.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>Call hold statistics were added.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>Call hold statistics were integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>15.2(2)T</td>
<td>Cisco Unified CME 9.0</td>
<td>This command was modified to add the following fields: Calls handoff to IOS, Average time to handoff, Longest time to handoff, and Number of error calls.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `show ephone-hunt statistics` and `show ephone-hunt` statistics commands provide expanded information regarding extension (list of numbers) and pilot numbers.

The output is dependent on call activity. If there is no activity, no data is displayed.

If your Cisco Unified CME system is configured with the basic automatic call distribution (B-ACD) and auto-attendant service, you can enable the collection of call statistics per ephone hunt group with the `statistics collect` command. Additional data is displayed for all agents combined and for individual agents. The additional data includes statistics such as: the number of calls received, the amount of time the calls waited to be answered, and the amount of time the calls spent on hold or in a queue.
The `statistics collect` command can be used to obtain other call statistics, such as direct calls to hunt group pilot numbers. For more information, see the “Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant Service” chapter in the *Cisco Unified CME B-ACD and TCL Call-Handling Applications* guide.

Once you have enabled statistics collection, you can use the `show ephone-hunt statistics` command to display call statistics, or you can use the `hunt-group report every hours` and `hunt-group report url` commands to transfer the statistics to files using TFTP.

**Note**

On the day that daylight saving time adjusts the time back by one hour at 2 a.m. each year, the original 1 a.m. to 2 a.m. statistics for that day are lost because they are overwritten by the new 1 a.m. to 2 a.m. statistics.

**Examples**

The following is a sample output that displays call statistics for the past hour for hunt group 2, which is associated with a Cisco Unified CME B-ACD service:

```
Router# show ephone-hunt 2 stat last 1 h
Thu 02:00 - 03:00
 Max Agents: 3
 Min Agents: 3
 Total Calls: 9
 Answered Calls: 7
 Abandoned Calls: 2
 Average Time to Answer (secs): 6
 Longest Time to Answer (secs): 13
 Average Time in Call (secs): 75
 Longest Time in Call (secs): 161
 Average Time before Abandon (secs): 8
 Calls on Hold: 2
 Average Time in Hold (secs): 16
 Longest Time in Hold (secs): 21
 Per agent statistics:
 Agent: 8004
 From Direct Call:
  Total Calls Answered : 3:
  Average Time in Call (secs) : 70
  Longest Time in Call (secs) : 150
  Total Calls on Hold : 1:
  Average Hold Time (secs) : 21
  Longest Hold Time (secs) : 21
 From Queue:
  Total Calls Answered : 3
  Average Time in Call (secs) : 55
  Longest Time in Call (secs) : 78
  Total Calls on Hold : 2:
  Average Hold Time (secs) : 19
  Longest Hold Time (secs) : 26
 Agent: 8006
 From Direct Call:
  Total Calls Answered : 3:
  Average Time in Call (secs) : 51
  Longest Time in Call (secs) : 118
  Total Calls on Hold : 1:
  Average Hold Time (secs) : 11
  Longest Hold Time (secs) : 11
 From Queue:
  Total Calls Answered : 1
  Average Time in Call (secs) : 4
```
The following is a sample output from the `show ephone-hunt statistics` command. The output focuses on queue-related statistics.

Queue related statistics:
- Total calls presented to the queue: 5
- Calls handoff to IOS: 2
- Number of calls in the queue: 1
- Average time to handoff (secs): 3
- Longest time to handoff (secs): 3
- Number of abandoned calls: 0
- Average time before abandon (secs): 0
- Calls forwarded to voice mail: 0
- Calls answered by voice mail: 0
- Number of error calls: 0

The table describes the significant fields shown in the output of the `show ephone-hunt statistics` command, in alphabetical order.

**Table 40: show ephone-hunt statistics Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Abandoned calls</td>
<td>Total number of calls abandoned by hunt group agents. This does not include calls going to the final number.</td>
</tr>
<tr>
<td>Answered call</td>
<td>Total number of calls answered by hunt group agents.</td>
</tr>
<tr>
<td>Average time before abandon (secs)</td>
<td>Average length of time that unanswered calls waited before hanging up.</td>
</tr>
<tr>
<td>Average hold time (secs)</td>
<td>Average length of time that calls waited on hold for this agent.</td>
</tr>
<tr>
<td>Average time in call (secs)</td>
<td>Average length of time that unanswered calls waited before going to an agent.</td>
</tr>
<tr>
<td>Average time in hold (secs)</td>
<td>Average length of time that calls were kept on hold for all agents.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Average time to answer (secs)</td>
<td>Average length of time that all calls to Cisco Unified CME B-ACD waited before being answered.</td>
</tr>
<tr>
<td>Average time to handoff (secs)</td>
<td>Average length of time before a call was handed off to IOS.</td>
</tr>
<tr>
<td>Calls answered by voice mail</td>
<td>Total number of calls to Cisco Unified CME B-ACD that were answered by voice mail.</td>
</tr>
<tr>
<td>Calls exited the queue</td>
<td>Total number of calls to Cisco Unified CME B-ACD that exited queues.</td>
</tr>
<tr>
<td>Calls forwarded to voice mail</td>
<td>Total number of calls to Cisco Unified CME B-ACD that were forwarded to voice mail.</td>
</tr>
<tr>
<td>Calls handoff to IOS</td>
<td>Total number of calls handed off to IOS.</td>
</tr>
<tr>
<td>Calls on hold</td>
<td>Total number of calls that were placed on hold.</td>
</tr>
<tr>
<td>Longest hold time (secs)</td>
<td>Longest length of time that a call to this agent spent between being placed on hold and being picked up.</td>
</tr>
<tr>
<td>Longest time in call (secs)</td>
<td>Longest length of time in which calls to Cisco Unified CME B-ACD went to an agent and waited in a call queue.</td>
</tr>
<tr>
<td>Longest time in hold (secs)</td>
<td>Longest length of time that a call spent between being placed on hold and being picked up by agents.</td>
</tr>
<tr>
<td>Longest time to answer (secs)</td>
<td>Longest length of time before calls to Cisco Unified CME B-ACD were answered.</td>
</tr>
<tr>
<td>Longest time to handoff (secs)</td>
<td>Longest length of time before a call was handed off to IOS.</td>
</tr>
<tr>
<td>Max agent</td>
<td>Maximum number of hunt group agents.</td>
</tr>
<tr>
<td>Min agent</td>
<td>Minimum number of hunt group agents.</td>
</tr>
<tr>
<td>Number of abandoned calls:</td>
<td>Total number of calls to Cisco Unified CME B-ACD that hung up before being answered.</td>
</tr>
<tr>
<td>Number of error calls</td>
<td>Total number of misdialed calls.</td>
</tr>
<tr>
<td>Total calls answered</td>
<td>Total number of calls to Cisco Unified CME B-ACD that were answered by an agent.</td>
</tr>
<tr>
<td>Total calls on hold</td>
<td>Total number of calls that were on hold for this agent.</td>
</tr>
<tr>
<td>Total calls presented to the queue</td>
<td>Total number of calls made to Cisco Unified CME B-ACD.</td>
</tr>
<tr>
<td>Total calls</td>
<td>Total number of direct calls made to the hunt group.</td>
</tr>
</tbody>
</table>
From Cisco Unified CME Release 10.5 onwards, abandoned calls will not include the calls going to the final number. However, the total calls includes calls going to the final number. Use the formula 

"Final Calls = Total Calls - Answered Calls - Abandoned Calls" , to calculate the calls going to the final number.

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-hunt</td>
<td>Enters ephone-hunt configuration mode to create a hunt group for use in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>hunt-group report every hours</td>
<td>Sets the hourly interval at which Cisco Unified CME B-ACD call statistics are automatically transferred to a file.</td>
</tr>
<tr>
<td>hunt-group report url</td>
<td>Sets filename parameters and the URL path where Cisco Unified CME B-ACD call statistics are to be sent using TFTP.</td>
</tr>
<tr>
<td>statistics collect</td>
<td>Enables the collection of call statistics for an ephone hunt group.</td>
</tr>
</tbody>
</table>
show fb-its-log

To display information about the Cisco CallManager Express (Cisco CME) eXtensible Markup Language (XML) application program interface (API) configuration, statistics on XML API queries, and the XML API event logs, use the `show fb-its-log` command in privileged EXEC mode.

`show fb-its-log [summary]`

**Syntax Description**

| summary  | (Optional) Displays only the XML API configuration and the statistics for queries and logs, and not the logs themselves. |

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco CME Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following is sample output from the `show fb-its-log summary` command:

```
Router# show fb-its-log summary
    ---- Current Period ----
        extension events:4
        device events: 3
        overwrites:0
        missed:0
        deleted:0
    ---- History ------
        overwrites:0
        missed:0
        deleted:8
    ---- Threads ----
        max xml threads:2
        current thread:0
        read in process:FALSE
```

The following is sample output from the `show fb-its-log` command:

```
Router# show fb-its-log
    ---- Current Period ----
        extension events:4
        device events: 3
        overwrites:0
        missed:0
        deleted:0
    ---- History ------
        overwrites:0
        missed:0
        deleted:8
    ---- Threads ----
        max xml threads:2
```
The following table describes the significant fields in this output.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Current Period</td>
<td>The time between the last retain-timer-triggered cleanup to the next cleanup.</td>
</tr>
<tr>
<td>extension events</td>
<td>Events related to extensions that have been captured in the internal buffer.</td>
</tr>
<tr>
<td>device events</td>
<td>Events related to devices that have been captured in the internal buffer.</td>
</tr>
<tr>
<td>overwrites</td>
<td>Events that are written over previously recorded events in the buffer. Overwrites occur when the internal buffer size is too small; new events overwrite old ones. The internal buffer size is set using the <strong>max-size</strong> keyword in the <strong>log</strong> table command.</td>
</tr>
<tr>
<td>missed</td>
<td>Events that happen too quickly for the system to record.</td>
</tr>
<tr>
<td>deleted</td>
<td>Events removed from the internal buffer.</td>
</tr>
<tr>
<td>History</td>
<td>Information since the last system restart.</td>
</tr>
<tr>
<td>Threads</td>
<td>Current number of threads configured in the system.</td>
</tr>
<tr>
<td>max xml threads</td>
<td>Maximum number of concurrent XML threads allowed.</td>
</tr>
<tr>
<td>current thread</td>
<td>XML API query thread.</td>
</tr>
<tr>
<td>read in process</td>
<td>TRUE indicates that the xml-test.html file is being read now. FALSE indicates that the file is not being read.</td>
</tr>
<tr>
<td>UTC</td>
<td>Coordinated Universal Time, which is used by the system clock on the Cisco CME router.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>log table</strong></td>
<td>Sets the maximum size of the table used to capture phone events used for the Cisco CME XML API.</td>
</tr>
</tbody>
</table>
show ip address trusted list

To display a list of trusted ip addresses, use the **show ip address trusted list** command in privileged EXEC mode.

**show ip address trusted list**

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to display a list of trusted IP addresses.

**Examples**

The following is a sample output from this command displaying all statistical information:

```
Router #show ip address trusted list
IP Address Trusted Authentication
Administration State: UP
Operation State: UP
IP Address Trusted Call Block Cause: call-reject (21)
VoIP Dial-peer IPv4 Session Targets:
Peer Tag Oper State Session Target
-------- ---------- --------------
11 DOWN ipv4:1.3.45.1
1 UP ipv4:1.3.45.1
IP Address Trusted List:
ipv4 172.19.245.1
ipv4 172.19.247.1
ipv4 172.19.243.1
ipv4 171.19.245.1
ipv4 171.19.10.1
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ip address trusted list</td>
<td>Allows to add a list of trusted IP addresses.</td>
</tr>
</tbody>
</table>
show presence global

To display configuration information about the presence service, use the show presence global command in user EXEC or privileged EXEC mode.

Syntax Description
This command has no arguments or keywords.

Command Modes
User EXEC (>)
Privileged EXEC (#)

Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines
This command displays the configuration settings for presence.

Examples
The following example displays output from the show subscription global:

Router# show subscription global
Presence Global Configuration Information:
-------------------------------------------------------------
Presence feature enable : TRUE
Presence allow external watchers : FALSE
Presence max subscription allowed : 100
Presence number of subscriptions : 0
Presence allow external subscribe : FALSE
Presence call list enable : TRUE
Presence server IP address : 0.0.0.0
Presence sccp blfsd retry interval : 60
Presence sccp blfsd retry limit : 10
Presence router mode : CME mode

The table describes the significant fields shown in the display.

Table 42: show subscription global Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Presence feature enable</td>
<td>Indicates whether presence is enabled on the router with the presence command.</td>
</tr>
<tr>
<td>Presence allow external watchers</td>
<td>Indicates whether internal presentities can be watched by external watchers, as set by the watcher all</td>
</tr>
<tr>
<td>Presence max subscription allowed</td>
<td>Maximum number of presence subscriptions allowed by the max-subscription command.</td>
</tr>
<tr>
<td>Presence number of subscriptions</td>
<td>Current number of active presence subscriptions.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>----------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Presence allow external subscribe</td>
<td>Indicates whether internal watchers are allowed to subscribe to status notifications from external presentities, as set by the allow subscribe command.</td>
</tr>
<tr>
<td>Presence call list enable</td>
<td>Indicates whether the Busy Lamp Field (BLF) call-list feature is enabled with the presence call-list command.</td>
</tr>
<tr>
<td>Presence server IP address</td>
<td>Displays the IP address of an external presence server defined with the server command.</td>
</tr>
<tr>
<td>Presence sccp blfsd retry interval</td>
<td>Retry timeout, in seconds, for BLF speed-dial numbers on SCCP phones set by the sccp bl-speed-dial retry interval command.</td>
</tr>
<tr>
<td>Presence sccp blfsd retry limit</td>
<td>Maximum number of retries allowed for BLF speed-dial numbers on SCCP phones set by the sccp bl-speed-dial retry interval command.</td>
</tr>
<tr>
<td>Presence router mode</td>
<td>Indicates whether the configuration mode is set to Cisco Unified CME or Cisco Unified SRST by the mode command.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow watch</td>
<td>Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.</td>
</tr>
<tr>
<td>allow subscribe</td>
<td>Allows internal watchers to monitor external presence entities (directory numbers).</td>
</tr>
<tr>
<td>debug presence</td>
<td>Displays debugging information about the presence service.</td>
</tr>
<tr>
<td>presence enable</td>
<td>Allows the router to accept incoming presence requests.</td>
</tr>
<tr>
<td>server</td>
<td>Specifies the IP address of a presence server for sending presence requests from internal watchers to external presence entities.</td>
</tr>
<tr>
<td>show presence subscription</td>
<td>Displays information about active presence subscriptions.</td>
</tr>
<tr>
<td>watcher all</td>
<td>Allows external watchers to monitor internal presence entities (directory numbers).</td>
</tr>
</tbody>
</table>
show presence subscription

To display information about active presence subscriptions, use the `show presence subscription` command in user EXEC or privileged EXEC mode.

```
show presence subscription [ { details | presentity telephone-number | subid subscription-id | summary } ]
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax Element</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>details</code></td>
<td>(Optional) Displays detailed information about presentities, watchers, and presence subscriptions.</td>
</tr>
<tr>
<td><code>presentity telephone-number</code></td>
<td>(Optional) Displays information on the presentity specified by the destination telephone number.</td>
</tr>
<tr>
<td><code>subid subscription-id</code></td>
<td>(Optional) Displays information for the specific subscription ID.</td>
</tr>
<tr>
<td><code>summary</code></td>
<td>(Optional) Displays summary information about active subscription requests.</td>
</tr>
</tbody>
</table>

### Command Default

Information for all active presence subscriptions is displayed.

### Command Modes

- User EXEC (`>`)
- Privileged EXEC (`#`)

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command displays details about the currently active presence subscriptions.

### Examples

The following is sample output from the `show presence subscription details` command:

```
Presence Active Subscription Records Details:

Subscription ID : 1
Watcher : 6002@10.4.171.60
Presentity : 6005@10.4.171.34
Expires : 3600 seconds
Subscription Duration : 1751 seconds
line status : idle
watcher type : local
presentity type : local
Watcher phone type : SIP Phone
subscription type : Incoming Indication
retry limit : 0
sibling subID : 0
sdb : 0
dp : 6555346C
watcher dial peer tag : 40001
```

Cisco Unified Communications Manager Express Command Reference
number of presentity : 1

Subscription ID : 2
Watcher : 6002@10.4.171.60

Presence Active Subscription Records:

Subscription ID : 30
Watcher : 4085550103@10.4.171.34
Presentity : 5001@10.4.171.20
Expires : 3600 seconds
line status : idle
watcher type : local
presentity type : remote
Watcher phone type : SCCP [BLF Call List]
subscription type : Outgoing Request
retry limit : 0
sibling subID : 23
sdb : 0
dp : 0
watcher dial peer tag : 0

The following is sample output from the `show presence subscription summary` command:

```
Router# show presence subscription summary
Presence Active Subscription Records Summary: 15 subscription
Watcher Presentity SubID Expires SibID Status
-----------------------------------------------
6002@10.4.171.60 6005@10.4.171.34 1 3600 0 idle
6005@10.4.171.81 6002@10.4.171.34 6 3600 0 idle
6005@10.4.171.81 6003@10.4.171.34 8 3600 0 idle
6005@10.4.171.81 6002@10.4.171.34 9 3600 0 idle
6005@10.4.171.81 6003@10.4.171.34 10 3600 0 idle
6005@10.4.171.81 6001@10.4.171.34 12 3600 0 idle
6001@10.4.171.61 6003@10.4.171.34 15 3600 0 idle
6003@10.4.171.61 6002@10.4.171.34 17 3600 0 idle
6003@10.4.171.61 6003@10.4.171.34 19 3600 0 idle
6003@10.4.171.61 5001@10.4.171.34 23 3600 24 idle
6002@10.4.171.60 6003@10.4.171.34 121 3600 0 idle
6002@10.4.171.60 5002@10.4.171.34 128 3600 129 idle
6005@10.4.171.81 1001@10.4.171.34 130 3600 131 busy
6005@10.4.171.81 7005@10.4.171.34 132 3600 133 idle
```

The following is sample output from the `show presence subscription summary` command showing that device-based BLF monitoring is enabled on two phones:

```
Watcher Presentity SubID Expires SibID Status
-----------------------------------------------
D 2036@10.6.2.6 2038@10.6.2.254 33 3600 0 idle
2036@10.6.2.6 2038@10.6.2.254 35 3600 0 idle
D 2036@10.6.2.6 8883@10.6.2.254 37 3600 0 unknown
```

The following is sample output from the `show presence subscription subid` command:
Router# show presence subscription subid 133

Presence Active Subscription Records:
--------------------------------------------------
Subscription ID : 133
Watcher : 6005@10.4.171.34
Presentity : 7005@10.4.171.20
Expires : 3600 seconds
line status : idle
watcher type : local
presentity type : remote
Watcher phone type : SIP Phone
subscription type : Outgoing Request
retry limit : 0
sibling subID : 132
sdb : 0
dp : 0
watcher dial peer tag : 0

The following table describes the significant fields shown in the display.

*Table 43: show presence subscription Field Descriptions*

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Watcher</td>
<td>IP address of the watcher.</td>
</tr>
<tr>
<td>Presentity</td>
<td>IP address of the presentity.</td>
</tr>
<tr>
<td>Expires</td>
<td>Number of seconds until the subscription expires. Default is 3600.</td>
</tr>
<tr>
<td>line status</td>
<td>Status of the line: • Idle—Line is not being used. • In-use—User is on the line, whether or not this line can accept a new call. • Unknown—Phone is unregistered or this line is not allowed to be watched.</td>
</tr>
<tr>
<td>watcher type</td>
<td>Whether the watcher is local or remote.</td>
</tr>
<tr>
<td>presentity type</td>
<td>Whether the presentity is local or remote.</td>
</tr>
<tr>
<td>Watcher phone type</td>
<td>Type of phone, either SCCP or SIP.</td>
</tr>
<tr>
<td>subscription type</td>
<td>The type of presence subscription, either incoming or outgoing.</td>
</tr>
<tr>
<td>retry limit</td>
<td>Maximum number of times the router attempts to subscribe for the line status of an external SCCP phone when either the presentity does not exist or the router receives a terminated NOTIFY from the external presence server. Set with the <strong>sccp blf-speed-dial retry-interval</strong> command.</td>
</tr>
<tr>
<td>sibling subID</td>
<td>Sibling subscription ID if presentity is remote. If value is 0, presentity is local.</td>
</tr>
<tr>
<td>sdb</td>
<td>Voice port of the presentity.</td>
</tr>
<tr>
<td>dp</td>
<td>Dial peer of the presentity.</td>
</tr>
</tbody>
</table>
### Field Description

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>watcher dial peer tag</td>
<td>Dial peer tag of the watcher device.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow watch</td>
<td>Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.</td>
</tr>
<tr>
<td>blf-speed-dial</td>
<td>Enables BLF monitoring for a speed-dial number on a phone registered to Cisco Unified CME.</td>
</tr>
<tr>
<td>debug ephone blf</td>
<td>Displays debugging information for BLF presence features.</td>
</tr>
<tr>
<td>debug presence</td>
<td>Displays debugging information about the presence service.</td>
</tr>
<tr>
<td>presence</td>
<td>Enables presence service and enters presence configuration mode.</td>
</tr>
<tr>
<td>presence enable</td>
<td>Allows the router to accept incoming presence requests.</td>
</tr>
<tr>
<td>show presence global</td>
<td>Displays configuration information about the presence service.</td>
</tr>
</tbody>
</table>
show sdspfarm

To display the status of the configured digital signal processor (DSP) farms and transcoding streams, use the `show sdspfarm` command in privileged EXEC mode.

```
show sdspfarm [units|sessions {active|callID number|statistics|summary}]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>units</td>
<td>Displays the configured and registered DSP farms.</td>
</tr>
<tr>
<td>sessions</td>
<td>Displays the transcoding streams.</td>
</tr>
<tr>
<td>active</td>
<td>Displays all active sessions.</td>
</tr>
<tr>
<td>callID</td>
<td>Displays activities for a specific caller ID.</td>
</tr>
<tr>
<td>number</td>
<td>Displays caller ID number displayed by the <code>show voip rtp connection</code> command.</td>
</tr>
<tr>
<td>statistics</td>
<td>Displays session statistics.</td>
</tr>
<tr>
<td>summary</td>
<td>Displays summary information.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Products</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco CME 3.2</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Examples**

The following is sample output from the `show sdspfarm units`:

```
Router# show sdspfarm units
mtp-1 Device:MTP123456782012 TCP socket:[-1] UNREGISTERED
actual_stream:0 max_stream 0 IP:0.0.0.0 0 Unknown 0 keepalive 0
mtp-2 Device:MTP000a8eaca80 TCP socket:[5] REGISTERED
actual_stream:40 max_stream 40 IP:10.5.49.160 11001 MTP YOKO keepalive 12074
Supported codec:G711Ulaw
G711Alaw
G729
G729a
G729b
G729ab
max-mtps:2, max-streams:240, alloc-streams:40, act-streams:0
```

The following is sample output from the `show sdspfarm sessions active`:

```
Router# show sdspfarm sessions active
Stream-ID:3 mtp:2 1.5.49.160 20174 Local:2000 START
usage:MoH (DN=3, CH=1) FE=TRUE
codec=G729 duration:20 vad:0 peer Stream-ID:4
Stream-ID:4 mtp:2 1.5.49.160 17072 Local:2000 START
usage:MoH (DN=3, CH=1) FE=FALSE
codec=G711Ulaw64k duration:20 vad:0 peer Stream-ID:3
```

The following is sample output from the `show sdspfarm sessions callID`:

```
Router# show sdspfarm sessions callid 51M
Peer Stream-ID:5, srcCall-ID:52, codec:G711Ulaw64k , dur:20ms, vad:0, dstCall-ID:51, confID:5, mtp:2

Router-2015# show sdspfarm sessions callid 52
Stream-ID:5, srcCall-ID:52, codec:G711Ulaw64k , dur:20ms, vad:0, dstCall-ID:51, confID:5, mtp:2

The following is sample output from the show sdspfarm sessions statistics:

Router# show sdspfarm sessions statistics
Stream-ID:1 mtp:2 0.0.0.0 0 Local:0 IDLE
   codec:G711Ulaw64k duration:20 vad:0 peer Stream-ID:0
   recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:2 mtp:2 0.0.0.0 0 Local:0 IDLE
   codec:G711Ulaw64k duration:20 vad:0 peer Stream-ID:0
   recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:3 mtp:2 10.5.49.160 20174 Local:2000 START MoH (DN=3 , CH=1) FE=TRUE
   codec:G729 duration:20 vad:0 peer Stream-ID:4
   recv-pak:0 xmit-pak:0 out-pak:4780 in-pak:0 discard:0
Stream-ID:4 mtp:2 10.5.49.160 17072 Local:2000 START MoH (DN=3 , CH=1) FE=FALSE
   codec:G711Ulaw64k duration:20 vad:0 peer Stream-ID:3
   recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:5 mtp:2 0.0.0.0 0 Local:0 IDLE
   codec:G711Ulaw64k duration:20 vad:0 peer Stream-ID:0
   recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:6 mtp:2 0.0.0.0 0 Local:0 IDLE
   codec:G711Ulaw64k duration:20 vad:0 peer Stream-ID:0
   recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:7 mtp:2 0.0.0.0 0 Local:0 IDLE
   codec:G711Ulaw64k duration:20 vad:0 peer Stream-ID:0
   recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:8 mtp:2 0.0.0.0 0 Local:0 IDLE
   codec:G711Ulaw64k duration:20 vad:0 peer Stream-ID:0
   recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:9 mtp:2 0.0.0.0 0 Local:0 IDLE
   codec:G711Ulaw64k duration:20 vad:0 peer Stream-ID:0
   recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:10 mtp:2 0.0.0.0 0 Local:0 IDLE
   codec:G711Ulaw64k duration:20 vad:0 peer Stream-ID:0
   recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:11 mtp:2 0.0.0.0 0 Local:0 IDLE
   codec:G711Ulaw64k duration:20 vad:0 peer Stream-ID:0
   recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:12 mtp:2 0.0.0.0 0 Local:0 IDLE
   codec:G711Ulaw64k duration:20 vad:0 peer Stream-ID:0
   recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:13 mtp:2 0.0.0.0 0 Local:0 IDLE
   codec:G711Ulaw64k duration:20 vad:0 peer Stream-ID:0
   recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:14 mtp:2 0.0.0.0 0 Local:0 IDLE
   codec:G711Ulaw64k duration:20 vad:0 peer Stream-ID:0
   recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:15 mtp:2 0.0.0.0 0 Local:0 IDLE
   codec:G711Ulaw64k duration:20 vad:0 peer Stream-ID:0
   recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:16 mtp:2 0.0.0.0 0 Local:0 IDLE
   codec:G711Ulaw64k duration:20 vad:0 peer Stream-ID:0
   recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:17 mtp:2 0.0.0.0 0 Local:0 IDLE
show sdspfarm
The following is sample output from the `show dspsfarm sessions summary`:

```
Router# show dspsfarm sessions summary
max-mtps:2, max-streams:240, alloc-streams:40, act-streams:2

  ID    MTP  State   CallID confID Usage                Codec/Duration
  ----  ----  ------  ---------  ----    -------------------------------
   1    2    IDLE   -1          0                      G711Ulaw64k /20ms
   2    2    IDLE   -1          0                      G711Ulaw64k /20ms
   3    2  START   -1          3  MoH (DN=3 , CH=1) FE-TRUE G711Ulaw64k /20ms
   4    2  START   -1          3  MoH (DN=3 , CH=1) FE-FALSE G711Ulaw64k /20ms
   5    2    IDLE   -1          0                      G711Ulaw64k /20ms
   6    2    IDLE   -1          0                      G711Ulaw64k /20ms
   7    2    IDLE   -1          0                      G711Ulaw64k /20ms
   8    2    IDLE   -1          0                      G711Ulaw64k /20ms
   9    2    IDLE   -1          0                      G711Ulaw64k /20ms
  10    2    IDLE   -1          0                      G711Ulaw64k /20ms
  11    2    IDLE   -1          0                      G711Ulaw64k /20ms
  12    2    IDLE   -1          0                      G711Ulaw64k /20ms
  13    2    IDLE   -1          0                      G711Ulaw64k /20ms
  14    2    IDLE   -1          0                      G711Ulaw64k /20ms
  15    2    IDLE   -1          0                      G711Ulaw64k /20ms
  16    2    IDLE   -1          0                      G711Ulaw64k /20ms
  17    2    IDLE   -1          0                      G711Ulaw64k /20ms
  18    2    IDLE   -1          0                      G711Ulaw64k /20ms
  19    2    IDLE   -1          0                      G711Ulaw64k /20ms
  20    2    IDLE   -1          0                      G711Ulaw64k /20ms
  21    2    IDLE   -1          0                      G711Ulaw64k /20ms
  22    2    IDLE   -1          0                      G711Ulaw64k /20ms
  23    2    IDLE   -1          0                      G711Ulaw64k /20ms
  24    2    IDLE   -1          0                      G711Ulaw64k /20ms
  25    2    IDLE   -1          0                      G711Ulaw64k /20ms
  26    2    IDLE   -1          0                      G711Ulaw64k /20ms
  27    2    IDLE   -1          0                      G711Ulaw64k /20ms
  28    2    IDLE   -1          0                      G711Ulaw64k /20ms
  29    2    IDLE   -1          0                      G711Ulaw64k /20ms
  30    2    IDLE   -1          0                      G711Ulaw64k /20ms
  31    2    IDLE   -1          0                      G711Ulaw64k /20ms
  32    2    IDLE   -1          0                      G711Ulaw64k /20ms
  33    2    IDLE   -1          0                      G711Ulaw64k /20ms
  34    2    IDLE   -1          0                      G711Ulaw64k /20ms
  35    2    IDLE   -1          0                      G711Ulaw64k /20ms
  36    2    IDLE   -1          0                      G711Ulaw64k /20ms
  37    2    IDLE   -1          0                      G711Ulaw64k /20ms
  38    2    IDLE   -1          0                      G711Ulaw64k /20ms
  39    2    IDLE   -1          0                      G711Ulaw64k /20ms
  40    2    IDLE   -1          0                      G711Ulaw64k /20ms

The following table describes the fields shown in the show dspsfarm display.

Table 44: show dspsfarm Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>act-streams</td>
<td>Active streams that are currently involved in calls.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-------------</td>
<td>--------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>alloc-streams</td>
<td>Number of transcoding streams that are actually allocated to all DSP farms that are registered to Cisco CME.</td>
</tr>
<tr>
<td>callID</td>
<td>Caller ID that the active stream is in.</td>
</tr>
<tr>
<td>Codec</td>
<td>Codec in use.</td>
</tr>
<tr>
<td>confID</td>
<td>ConfID that is used to communicate with DSP farms.</td>
</tr>
<tr>
<td>discard</td>
<td>Number of packets that are discarded.</td>
</tr>
<tr>
<td>dstCall-ID</td>
<td>Caller ID of the destination IP call leg.</td>
</tr>
<tr>
<td>Duration or dur</td>
<td>Packet rates, in milliseconds.</td>
</tr>
<tr>
<td>ID</td>
<td>Transcoding stream sequence number in Cisco CME.</td>
</tr>
<tr>
<td>in-pak</td>
<td>Number of incoming packets from the source call leg.</td>
</tr>
<tr>
<td>Local</td>
<td>Local port for voice packets.</td>
</tr>
<tr>
<td>max-mtps</td>
<td>Maximum number of Message Transfer Parts (MTPs) that are currently allowed to register in Cisco CME.</td>
</tr>
<tr>
<td>max-streams</td>
<td>Maximum number of transcoding streams that are currently allowed in Cisco CME.</td>
</tr>
<tr>
<td>mtp or MTP</td>
<td>MTP sequence number where the transcoding stream is located.</td>
</tr>
<tr>
<td>out-pak</td>
<td>Number of outgoing packets sending to source call leg.</td>
</tr>
<tr>
<td>peer Stream-ID</td>
<td>Stream sequence number of the other stream paired in the same transcoding session. (Two transcoding streams make up a transcoding session).</td>
</tr>
<tr>
<td>recv-pak</td>
<td>Number of voice packets received from DSP farm.</td>
</tr>
<tr>
<td>srcCall-ID</td>
<td>Source caller ID of the source IP call leg.</td>
</tr>
<tr>
<td>State</td>
<td>Current state of the transcoding stream, could be IDLE, SEIZE, START, STOP, or END.</td>
</tr>
<tr>
<td>Stream-ID</td>
<td>Transcoding stream sequence number in Cisco CME.</td>
</tr>
<tr>
<td>TCP-socket</td>
<td>Socket number for DSP farm (similar to TCP socket for show ephone output).</td>
</tr>
<tr>
<td>usage</td>
<td>Current usage of the stream; for example, Ip-Ip (IP to IP transcoding), MOH (for MOH transcoding) and Conf (conference).</td>
</tr>
<tr>
<td>vad</td>
<td>Voice-activity detection (VAD) flag for the transcoding stream. It should always be 0 (False).</td>
</tr>
<tr>
<td>xmit-pak</td>
<td>Number of packets that are sent to DSP farm.</td>
</tr>
</tbody>
</table>

---

Cisco Unified Communications Manager Express Command Reference
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>sdspfarm tag</code></td>
<td>Permits a DSP farm to be registered to Cisco CME and be associated with an SCCP client interface’s MAC address.</td>
</tr>
<tr>
<td><code>sdspfarm transcode sessions</code></td>
<td>Specifies the maximum number of transcoding sessions allowed per Cisco CME router.</td>
</tr>
<tr>
<td><code>sdspfarm units</code></td>
<td>Specifies the maximum number of DSP farms that are allowed to be registered to Cisco CME.</td>
</tr>
</tbody>
</table>
show shared-line

To display information about the Session Initiation Protocol (SIP) shared lines, use the `show shared-line` command in user EXEC or privileged EXEC mode.

`show shared-line {call|details|subscription|summary}`

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call</td>
<td>Displays information about all active calls on shared lines.</td>
</tr>
<tr>
<td>details</td>
<td>Displays detailed information about each shared line.</td>
</tr>
<tr>
<td>subscription</td>
<td>Displays information for specific subscriptions to shared lines.</td>
</tr>
<tr>
<td>summary</td>
<td>Displays summary information about active subscriptions to shared lines.</td>
</tr>
</tbody>
</table>

**Command Modes**

User EXEC (>)
Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(24)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Examples**

The following is sample output from the `show shared-line call` command:

```
Router# show shared-line call
Shared-Line active call info:
Shared-Line: '20141', active calls: 3
Local User Local Address Remote User Remote Address CallID
---------- ========= =========== =========== ========= -----=---------
20141 20141@10.6.0.2 20143 20143@10.10.0.1 3168
20141 20141@10.6.0.1 Barge 20143@10.10.0.1 3209
20141 20141@10.6.0.2 20141 20141@10.10.0.1 3210
```

The following is sample output from the `show shared-line details` command:

```
Router# show shared-line details
Shared-Line info details:
Shared-Line: '20141', subscribed users: 2, max calls limit: 10
Index Users sub_id peer_tag Status
----- ========= ========= ========= ========= 
   1 20141@10.6.0.1 5 40001 ACTIVE
   2 20141@10.6.0.2 6 40002 ACTIVE
Free call queue size: 7, Active call queue size: 3
Message queue size: 20, Event queue size: 64
```

The following is sample output from the `show shared-line subscription` command:

```
Router# show shared-line subscription
```
Shared-Line Subscription Info:

Subscriptions to: '20141', total subscriptions: 2

<table>
<thead>
<tr>
<th>SubID</th>
<th>Subscriber</th>
<th>Expires</th>
<th>Sub-Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>5</td>
<td>20141@10.6.0.1</td>
<td>3600</td>
<td>NOTIFY_ACKED</td>
</tr>
<tr>
<td>6</td>
<td>20141@10.6.0.2</td>
<td>3600</td>
<td>NOTIFY_ACKED</td>
</tr>
</tbody>
</table>

The following is sample output from the `show shared-line summary` command:

Router# show shared-line summary
Shared-Line info summary:
Shared-Line: '20141', subscribed users: 2, max calls limit: 10

The following table describes the significant fields shown in the displays.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Expires</td>
<td>Number of seconds until the subscription expires.</td>
</tr>
<tr>
<td>Local Address</td>
<td>IP address of the local phone involved in the shared line call.</td>
</tr>
<tr>
<td>Local User</td>
<td>Extension number of the shared line.</td>
</tr>
<tr>
<td>Remote Address</td>
<td>IP address of the remote phone involved in the shared line call.</td>
</tr>
<tr>
<td>Remote User</td>
<td>Extension of the remote phone involved in the shared line call.</td>
</tr>
<tr>
<td>SubID</td>
<td>Subscription ID.</td>
</tr>
<tr>
<td>Subscriber</td>
<td>Extension number of the shared line and the IP address of the phone subscriber.</td>
</tr>
<tr>
<td>Sub-Status</td>
<td>Status of the subscription.</td>
</tr>
<tr>
<td>Users</td>
<td>IP addresses of the phones using the shared line.</td>
</tr>
</tbody>
</table>

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>debug shared-line</td>
<td>Displays debugging information about SIP shared lines.</td>
</tr>
</tbody>
</table>
show telephony-service admin

To display information about the Cisco CallManager Express (Cisco CME) system administrator, use the show telephony-service admin command in user EXEC or privileged EXEC mode.

show telephony-service admin

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco CME Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command is deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
</tbody>
</table>

Examples

The following is sample output from this command:

```
Router# show telephony-service admin
admin_username Admin
admin_password word
edit DN through Web: enabled.
edit TIME through Web: enabled.
```

The following table describes the significant fields in this output.

Table 46: show telephony-service admin Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>admin_username</td>
<td>Username of system administrator.</td>
</tr>
<tr>
<td>admin_password</td>
<td>Password of system administrator.</td>
</tr>
<tr>
<td>edit DN through Web</td>
<td>Whether editing of extensions through the GUI has been enabled using the dn-webedit command.</td>
</tr>
<tr>
<td>edit TIME through Web</td>
<td>Whether changing the router time through the GUI has been enabled using the time-webedit command.</td>
</tr>
</tbody>
</table>

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dn-webedit</td>
<td>Enables adding of extensions (ephone-dns) through the web interface.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-------------</td>
<td>--------------------------------------------------</td>
</tr>
<tr>
<td>time-webedit</td>
<td>Enables setting of time through the web interface.</td>
</tr>
</tbody>
</table>
show telephony-service all

To display detailed configuration for phones, voice ports, and dial peers in a Cisco Unified Communications Manager Express (Cisco Unified CME) system, use the `show telephony-service all` command in user EXEC or privileged EXEC mode.

**show telephony-service all**

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

User EXEC (>)

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco CME Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco Unified CME 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco Unified CME 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>15.2(2)T</td>
<td>Cisco Unified CME 9.0</td>
<td>This command was modified to display the total number of data collected from both ephone and voice hunt groups.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `show telephony-service all` command to display the total number of ephone and voice hunt groups that have statistics collection turned on.

**Examples**

The following is a sample output from the `show telephony-service all` command:

```
Router# show telephony-service all
CONFIG
-------
ip source-address 10.0.0.1 port 2000
max-ephones 24
max-dn 24
dialplan-pattern 1 408734....
voicemail 11111
transfer-pattern 510734....
keepalive 30
ephone-dn 1
number 5001
huntstop
ephone-dn 2
number 5002
huntstop
call-forward noan 5001 timeout 8
ephone-dn 3
number 5003
huntstop
ephone 1
mac-address 0080.6C.337CB
type 0
button 1:1
speed-dial 1 5002
speed-dial 2 5003
cos 0
```
show telephony-service all

ephone 2
  mac-address 0030.94C3.F96A
  type 0
  button 1:2 2:3 3:4
  speed-dial 1 5004
  speed-dial 2 5001
  cos 0

voice-port 50/0/1
  station-id number 5001

voice-port 50/0/2
  station-id number 5002
  timeout ringing 8

dial-peer voice 20025 pots
  destination-pattern 5001
  huntstop
  port 50/0/1
dial-peer voice 20026 pots
  destination-pattern 5002
  huntstop
  call-forward noan 5001
  port 50/0/2
dial-peer voice 20027 pots
  destination-pattern 5003
  huntstop
  port 50/0/3

The following is a sample output from the show telephony-service all command. The output shows that call statistics are collected for 14 hunt groups, including 6 ephone and 8 voice hunt groups.

Router# show telephony-service all
CONFIG (Version=8.7)
=====================
Version 8.7
Max phone load SCCP version 17
Max DSP farm SCCP version 18
Cisco Unified Communications Manager Express
For on-line documentation please see:
protocol mode default
ip source-address 1.4.190.80 port 2000
ip qos dscp:
  ef (the MS 6 bits, 46, in ToS, 0XB8) for media
  cs3 (the MS 6 bits, 24, in ToS, 0x60) for signal
  af41 (the MS 6 bits, 34, in ToS, 0x88) for video
  default (the MS 6 bits, 0, in ToS, 0x0) for service service directed-pickup
load 6921 SCCP69xx.9-0-3-0
load 6961 SCCP69xx.8-5-3-0
max-ephones 14
max-dn 56
max-conferences 4 gain -6
dspfarm units 0
dspfarm transcode sessions 0
conference software
privacy
no privacy-on-hold
hunt-group report url prefix tftp://223.255.254.254/ngm/huntgp/uc500/test
hunt-group report url suffix 0 to 20
hunt-group report every 1 hours
# of hunt-group collect data: 14
The following is another sample output from the `show telephony-service all` command. The output shows that call statistics are collected for seven hunt groups, including three ephone and four voice hunt groups.

Router# show telephony-service all

The following table describes significant fields in this output, in alphabetical order.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>button</td>
<td>Button on the Cisco IP phone.</td>
</tr>
<tr>
<td>call-forward noan</td>
<td>Call forward no answer is set.</td>
</tr>
<tr>
<td>cnf-file location</td>
<td>Storage location for phone configuration files. System (default), flash or slot 0 memory, and external TFTP server.</td>
</tr>
<tr>
<td>cnf-file option</td>
<td>Specifies the use of different phone configuration files by type of phone or by individual phone.</td>
</tr>
<tr>
<td>cos</td>
<td>Not applicable; unused.</td>
</tr>
<tr>
<td>destination-pattern</td>
<td>Destination pattern (telephone number) configured for this dial peer.</td>
</tr>
<tr>
<td>dial-peer voice</td>
<td>Voice dial peer.</td>
</tr>
<tr>
<td>dialplan-pattern</td>
<td>Dial-plan pattern is set to expand the abbreviated extension numbers to fully qualified E.164 numbers.</td>
</tr>
<tr>
<td>ephone</td>
<td>Cisco IP phone.</td>
</tr>
<tr>
<td>ephone-dn</td>
<td>Cisco IP phone directory number.</td>
</tr>
<tr>
<td>huntstop</td>
<td>Huntstop is set.</td>
</tr>
</tbody>
</table>
### Field

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ip source-address</td>
<td>IP address used by Cisco IP phones to register with the router for service.</td>
</tr>
<tr>
<td>keepalive</td>
<td>IP phone keepalive period, in seconds.</td>
</tr>
<tr>
<td>mac-address</td>
<td>MAC address.</td>
</tr>
<tr>
<td>max-dn</td>
<td>Maximum directory numbers.</td>
</tr>
<tr>
<td>max-ephones</td>
<td>Maximum numbers of Cisco IP phones.</td>
</tr>
<tr>
<td>number</td>
<td>Cisco IP phone number.</td>
</tr>
<tr>
<td>port</td>
<td>TCP port number used by Cisco IP phones to communicate with the router.</td>
</tr>
<tr>
<td>pots</td>
<td>POTS dial peer set.</td>
</tr>
<tr>
<td>speed-dial</td>
<td>Speed-dial is set.</td>
</tr>
<tr>
<td>station-id number</td>
<td>Number used for caller ID purposes when calls are made using the line.</td>
</tr>
<tr>
<td>timeout</td>
<td>Timeout is set.</td>
</tr>
<tr>
<td>timeout ringing</td>
<td>Maximum amount of time that the phone is allowed to ring before the call is disconnected.</td>
</tr>
<tr>
<td>transfer-pattern</td>
<td>Transfer pattern is set to allow transfer of calls to a specified number.</td>
</tr>
<tr>
<td>type</td>
<td>Not applicable; unused.</td>
</tr>
<tr>
<td>voicemail</td>
<td>A voice-mail (speed-dial) number is set.</td>
</tr>
<tr>
<td>voice-port</td>
<td>(Virtual) voice port designator.</td>
</tr>
<tr>
<td># of hunt-group collect data</td>
<td>Total number of data collected from both ephone and voice hunt groups.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show telephony dial-peer</td>
<td>Displays dial peers for extensions in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>show telephony voice-port</td>
<td>Displays virtual voice-port configuration for extensions in a Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
show telephony-service bulk-speed-dial

To display information about bulk speed-dial lists, use the `show telephony-service bulk-speed-dial` command in privileged EXEC mode.

```
show telephony-service bulk-speed-dial \{global list-id index-id [all]local phone-tag list-id index-id [all]summary\}
```

**Syntax Description**

- **global**
  - Global lists that can be accessed by all users.
- **local**
  - Personal lists that can be accessed by users configured to use the lists.
- **list-id**
  - Digit that identifies the list. Range is from 0 to 9.
- **index-id**
  - Identification number for an entry.
- **phone-tag**
  - Ephone identifier (phone-tag).
- **summary**
  - List of registered bulk speed-dial text files.

**Command Modes**

- Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following example displays the list of bulk speed-dial text files that have been configured in the system:

```
Router# show telephony-service bulk-speed-dial summary
List-id Entries Size Reference url
  0    40 3840 Global tftp://192.168.254.254/phonedirs/uut.csv
  1    20 1920 Global phoneBook.csv
  8    15 1440 Global tftp://192.168.254.254/phonedirs/big.txt
  1    20 1920 ephone-2 tftp://192.168.254.254/phonedirs/big.txt1
  7    20 1920 ephone-3 big.txt1
  7    20 1920 ephone-3 phoneBook.csv
```

The following example displays the single entry 1234 from list 9:

```
Router# show telephony-service bulk-speed-dial global 9 1234
Number: 1800 200 1345 name: Jay Smith Private: yes Extension: No
```

The following example displays all index entries starting with 1 for personal list number 7 for ephone 2:

```
Router# show telephony-service bulk-speed-dial local 2 7 1 all
```
The following table describes the significant fields shown in the display.

**Table 48: show telephony-service bulk-speed-dial Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>List-id</td>
<td>Digit that identifies the list. Range is from 0 to 9.</td>
</tr>
<tr>
<td>Entries</td>
<td>Number of entries in the speed-dial file.</td>
</tr>
<tr>
<td>Size</td>
<td>Size of the file, in KB.</td>
</tr>
<tr>
<td>Reference</td>
<td>Assignment of the list: global if assigned to all phones, or a specific ephone number.</td>
</tr>
<tr>
<td>url</td>
<td>Location of the text file, in URL format.</td>
</tr>
<tr>
<td>Index</td>
<td>Identification number for an entry.</td>
</tr>
<tr>
<td>Number</td>
<td>Number to be dialed and displayed on the phone.</td>
</tr>
<tr>
<td>Name</td>
<td>Name to be displayed on the phone.</td>
</tr>
<tr>
<td>Hide</td>
<td>Yes indicates that this number should not be displayed when it is dialed.</td>
</tr>
<tr>
<td>Append</td>
<td>Yes indicates that additional digits can be dialed by the user after this number has been speed-dialed before the call is completed.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>bulk-speed-dial list (ephone)</td>
<td>Enables a personal bulk speed-dial list for an ephone.</td>
</tr>
<tr>
<td>bulk-speed-dial list (telephony-service)</td>
<td>Enables a global bulk speed-dial list for all users of a Cisco Unified CME system.</td>
</tr>
<tr>
<td>bulk-speed-dial prefix</td>
<td>Sets the prefix code that phone users dial to access speed-dial numbers from a bulk speed-dial list.</td>
</tr>
</tbody>
</table>
show telephony-service conference hardware

To display information about hardware conferences in a Cisco Unified Communications Manager Express (Cisco CME) system, use the `show telephony-service conference hardware` command in privileged EXEC mode.

```
show telephony-service conference hardware [{ad-hoc [detail|video]}|detail [video]|meetme [{detail|video}]|number telephone-number]
```

**Syntax Description**

- `ad-hoc` (Optional) Ad-hoc hardware conferences.
- `detail` (Optional) Detailed information for all conferences.
- `video` (Optional) Video conferences.
- `meetme` (Optional) Meet-me hardware conferences.
- `number` (Optional) Conference number.
- `telephone-number` (Optional) Telephone or extension number.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>15.1(4)M</td>
<td>Cisco Unified CME 8.6</td>
<td>This command was modified to include the video option.</td>
</tr>
<tr>
<td>15.2(2)T</td>
<td>Cisco Unified CME9.0</td>
<td>This command was modified to add hardware conference information on Cisco Unified SIP IP phones to the output display.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `show telephony-service conference hardware` command to display ad-hoc and meet-me hardware conferences information, including which parties are still in the conference.

**Examples**

The following is a sample output that displays information for a four-party ad-hoc hardware conference. Extension 8044 created the conference by calling extension 8012, then adding extension 8004 to the conference. The conference administrator, extension 8006, called into the conference after it was established.

```
Router# show telephony-service conference hardware detail
Conference Type Active Max Peak Master MasterPhone Last cur(initial)

8893 Ad-hoc 4 8 4 8044 29 (29) 8006
```
Conference parties:
8006 (admin)
8004
8012
8044

The following is a sample output that displays information for a meet-me video conference:

```
Router# show telephony-service conference hardware detail video
Conference Type Active Max Peak Master MasterPhone Last cur(initial)
------------------------------------------------------------------------
9999 Meetme-Video 10 16 10 n/a 0 ( 0) 9012
Conference parties (number:phone)
  9012 :12 :Audio
  7001 :Video
  9003 :3 :Audio
  7047 :Audio
  7015 :Video
  3667 :Audio
  9024 :24 :Audio
  9023 :23 :Video
  3665 :Video
  9022 :22 :Video
```

The following is another sample output from the `show telephony-service conference hardware detail` command. The output shows an ad-hoc video hardware conference among three participants, two of which are Cisco Unified SIP IP phones.

```
Router# show telephony-service conference hardware detail
Conference Type Active Max Peak Master MasterPhone Last cur(initial)
------------------------------------------------------------------------
B000 Ad-hoc Video 3 4 3 3915 SIP3915 15(15) 5801 RM5801
Conference parties (number:phone)
  5801 5801 :Video
  3916 SIPPHONE3916 :16 :Video
  3915 SIPPHONE3915 :15 (admin):Video
```

The following is a sample output from the `show telephony-service conference hardware ad-hoc` command:

```
Router# show telephony-service conference hardware ad-hoc
Conference Type Active Max Peak Master MasterPhone Last cur(initial)
------------------------------------------------------------------------
B000 Ad-hoc Video 3 4 3 3915 SIP3915 15(15) 5801 RM5801
```

The following is a sample output from the `show telephony-service conference hardware meetme` command:
The following is a sample output from the `show telephony-service conference hardware number` command:

```
Router# show telephony-service conference hardware number B000
Conference  Type          Active  Max  Peak  Master       MasterPhone
Last        cur(initial)
-----------------------------------------------
B000         Ad-hoc Video  3       4     3  3915 SIP3915   15(15)
            5801          RM5801
```

The following is another sample output from the `show telephony-service conference hardware number` command:

```
Router# show telephony-service conference hardware number 7788
Conference  Type          Active  Max  Peak  Master       MasterPhone
Last        cur(initial)
-----------------------------------------------
7788         Meetme Video  4       4     4  3917 SIP3917   17(17)
            4801          SCCP4801
```

The following table describes the significant fields shown in the display, listed in alphabetical order.

**Table 49: `show telephony-service conference hardware` Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Active</td>
<td>Number of active parties in the conference.</td>
</tr>
<tr>
<td>admin</td>
<td>Ad hoc and meet-me hardware conference administrator. The administrator can:</td>
</tr>
<tr>
<td></td>
<td>• Dial in to any conference directly through the conference number.</td>
</tr>
<tr>
<td></td>
<td>• Use the ConfList soft key to list conference parties.</td>
</tr>
<tr>
<td></td>
<td>• Remove any party from any conference.</td>
</tr>
<tr>
<td>Conference</td>
<td>Conference directory number (DN).</td>
</tr>
<tr>
<td>Conference parties</td>
<td>DNs in the conference.</td>
</tr>
<tr>
<td>Last</td>
<td>Last participant to join the conference.</td>
</tr>
<tr>
<td>Master</td>
<td>Conference creator.</td>
</tr>
</tbody>
</table>
MasterPhone cur(initial)  cur—Current master phone. The phone that currently hosts the conference creator.
(initial)—Initial master phone. The phone that hosted the conference creator when the conference was created.
Because you can transfer the conference creator, the current master phone may be different from the initial master phone.

Max  Maximum number of participants allowed in the conference.

Peak  Maximum number of participants in the conference at any time.

Type  Type of conference: meet-me or ad hoc.
**show telephony-service directory-entry**

To display the entries made using the `directory entry`, use the `show telephony-service directory-entry` command in user EXEC or privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

- User EXEC
- Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco CME Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command lists directory entries that are made using the `directory entry` but does not list entries that are made using the `name` and `number` commands in ephone-dn configuration mode.

**Examples**

The following is sample output from this command:

```
Router# show telephony-service directory-entry
directory entry 1 4085550123 name Smith, John
```

The following table describes significant fields in this output, in alphabetical order.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>directory directory-tag (shown as 1 in the example)</td>
<td>Sequence number or unique identifier for a directory entry.</td>
</tr>
<tr>
<td>name (shown as Smith, John)</td>
<td>Name that appears in the directory associated with the number.</td>
</tr>
<tr>
<td>number (shown as 4085550123 in the example)</td>
<td>Telephone number or extension for the directory entry.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>directory entry</code></td>
<td>Adds an entry to a local phone directory that can be displayed on IP phones.</td>
</tr>
<tr>
<td><code>show telephony-service all</code></td>
<td>Displays detailed configuration of a Cisco CME system.</td>
</tr>
<tr>
<td><code>show telephony-service ephone-dn</code></td>
<td>Displays information for extensions (ephone-dns) in a Cisco CME system.</td>
</tr>
</tbody>
</table>
show telephony-service ephone

To display configuration for the Cisco IP phones, use the `show telephony-service ephone` command in user EXEC or privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

User EXEC (>)
Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco CME 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco CME 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>The conference add-mode, conference drop-mode, and conference admin fields were added.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was modified. The output was enhanced to include the setting of the feature-button command and information about logical partitioning class of restriction (LPCOR).</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following is sample output from this command:

```
Router# show telephony-service ephone

Number of Configured ephones 2 (Registered 2)
 ephone 1
 Device Security Mode: Non-Secure
 mac-address 1234.4321.7000
 type 7960
 button 1:1
 keepalive 30 auxiliary 30
 multicast-moh
 max-calls-per-button 8
 busy-trigger-per-button 0
 Always send media packets to this router: No
 Preferred codec: g711ulaw
 conference drop-mode never
 conference add-mode all
 conference admin: No
 privacy: Yes
 feature-button 1 Dnd
 user-locale US
```
The table describes significant fields in this output, in alphabetical order.

### Table 51: show telephony-service ephone Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>button</td>
<td>Button number on IP phone, separator to denote ring characteristics and ephone-dn tag. A colon (:) separator denotes a normal ring.</td>
</tr>
<tr>
<td>conference add-mode</td>
<td>Who can add parties to a conference:</td>
</tr>
<tr>
<td></td>
<td>• creator—Only the creator can add parties.</td>
</tr>
<tr>
<td></td>
<td>• all—Any party can add other parties if the creator remains in the conference.</td>
</tr>
<tr>
<td>conference drop-mode</td>
<td>When conferences are dropped:</td>
</tr>
<tr>
<td></td>
<td>• creator—Conference terminates when the creator hangs up.</td>
</tr>
<tr>
<td></td>
<td>• local—Conference terminates when the last local party in the conference hangs up or drops out of the conference.</td>
</tr>
<tr>
<td></td>
<td>• never—Conference is not dropped, even if the creator hangs up, as long as three parties remain in the conference.</td>
</tr>
<tr>
<td>conference admin</td>
<td>Ad hoc and meet-me hardware conference administrator. The administrator can:</td>
</tr>
<tr>
<td></td>
<td>• Dial in to any conference directly through the conference number</td>
</tr>
<tr>
<td></td>
<td>• Use the ConfList soft key to list conference parties</td>
</tr>
<tr>
<td></td>
<td>• Remove any party from any conference</td>
</tr>
<tr>
<td>ephone</td>
<td>Cisco IP phone.</td>
</tr>
<tr>
<td>feature-button</td>
<td>Displays the type of feature button on the ephone. Feature type can be configured with privacy or DND.</td>
</tr>
</tbody>
</table>
show telephony-service ephone

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>lpcor (incoming)</td>
<td>Setting of the lpcor incoming command.</td>
</tr>
<tr>
<td>lpcor (outgoing)</td>
<td>Setting of the lpcor outgoing command.</td>
</tr>
<tr>
<td>lpcor type</td>
<td>Setting of the lpcor type command.</td>
</tr>
<tr>
<td>mac-address</td>
<td>MAC address of the Cisco IP phone.</td>
</tr>
<tr>
<td>speed-dial</td>
<td>Speed-tag (unique identifier) and the number that is programmed for that speed-tag.</td>
</tr>
<tr>
<td>type</td>
<td>Model type of phone.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show telephony-service all</td>
<td>Displays detailed configuration for a Cisco Unified CME system.</td>
</tr>
<tr>
<td>show telephony-service dial-peer</td>
<td>Displays dial-peer information for extensions in Cisco Unified CME.</td>
</tr>
<tr>
<td>show telephony-service ephone-dn</td>
<td>Displays information for extensions in Cisco Unified CME.</td>
</tr>
<tr>
<td>show telephony-service voice-port</td>
<td>Displays configurations for virtual voice ports in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
**show telephony-service ephone-dn**

To display information about extensions (ephone-dns) in a Cisco CallManager Express (Cisco CME) system, use the `show telephony-service ephone-dn` command in user EXEC or privileged EXEC mode.

**show telephony-service ephone-dn**

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

User EXEC (>)

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco CME Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>1.0</td>
<td>This command was introduced</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following is sample output from this command:

```
Router# show telephony-service ephone-dn
ephone-dn 1
    number 5001
    huntstop
ephone-dn 2
    number 5002
    huntstop
call-forward noan 5001 timeout 8
ephone-dn 3
    number 5003
    huntstop
ephone-dn 4
    number 5004
    huntstop
```

The following table describes significant fields in this output, in alphabetical order.

**Table 52: show telephony-service ephone-dn Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-forward noan</td>
<td>Call forwarding is set to no answer. Other available options are call-forward busy and call-forward all.</td>
</tr>
<tr>
<td>ephone-dn</td>
<td>Cisco IP phone directory number.</td>
</tr>
<tr>
<td>huntstop</td>
<td>Huntstop is set.</td>
</tr>
<tr>
<td>number</td>
<td>Cisco IP phone number.</td>
</tr>
<tr>
<td>timeout</td>
<td>Timeout setting for call forwarding when an extension does not answer.</td>
</tr>
</tbody>
</table>
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show telephony-service all</td>
<td>Displays the detailed configuration of all the Cisco IP phones.</td>
</tr>
<tr>
<td>show telephony-service dial-peer</td>
<td>Displays dial peer information for extensions (ephone-dns) in a Cisco CME system.</td>
</tr>
<tr>
<td>show telephony-service voice-port</td>
<td>Displays configurations for virtual voice ports in a Cisco CME system.</td>
</tr>
</tbody>
</table>
**show telephony-service ephone-dn-template**

To display information about ephone-dn-template configurations, use the `show telephony-service ephone-dn-template` command in user EXEC or privileged EXEC mode.

```
show telephony-service ephone-dn-template
```

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

- User EXEC
- Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command displays contents of ephone-dn templates. Use the `show running-config` to display the association of templates to particular ephone-dns.

**Examples**

The following is sample output from this command:

```
Router# show telephony-service ephone-dn-template
ephone-template 1
  softkeys idle Newcall Redial Cfwdall Dnd Pickup Gpickup Login
  codec g711ulaw
  User Locale: US
  Network Locale: US
ephone-template 2
  softkeys idle Redial Newcall Dnd Cfwdall Pickup Gpickup Login
  codec g711ulaw
  User Locale: US
  Network Locale: US
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-dn-template</td>
<td>Creates an ephone-dn template and enters ephone-dn-template configuration mode.</td>
</tr>
</tbody>
</table>
show telephony-service ephone-template

To display the contents of ephone-templates, use the `show telephony-service ephone-template` command in user EXEC or privileged EXEC mode.

**show telephony-service ephone-template**

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

User EXEC (>)

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco CME 3.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>The conference add-mode, conference drop-mode, and conference admin fields were added.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>Emergency response location (ERL) information assigned to an ephone displays in the output.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was modified. Logical partitioning class of restriction (LPCOR) information was added to the output.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to display the contents of each ephone template that is defined. Use the `show running-config` command to display the association of templates to specific ephones.

**Examples**

The following is sample output from this command:

```
Router# show telephony-service ephone-template
ephone-template 1
  softkey idle  Cfwalli Dnd Gpickup Join Pickup RmLstC
  softkey connected  Acct ConfList Confrn Endcall Hold Join Park
  conference drop-mode never
  conference add-mode all
  conference admin: No
  max-calls-per-button 8
  busy-trigger-per-button 0
  privacy default
  MLPP max precedence level -1
  MLPP indication Enabled
```
MLPP preemption Enabled
Always send media packets to this router: No
Preferred codec: g711ulaw
keepalive 30 auxiliary 30
User Locale: US
Network Locale: US
Emergency Response Location: 6
lpcor type: remote
lpcor (incoming): local_sccp_phone_1 (outgoing): local_sccp_phone_1

The following table describes significant fields in this output.

### Table 53: show telephony-service ephone Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template</td>
<td>Identifier for the ephone template.</td>
</tr>
<tr>
<td>softkey hold</td>
<td>Soft keys displayed during the hold call stage.</td>
</tr>
<tr>
<td>softkey idle</td>
<td>Soft keys displayed during the call-idle call stage.</td>
</tr>
<tr>
<td>softkey seized</td>
<td>Soft keys displayed during the call-seized call stage.</td>
</tr>
<tr>
<td>softkey alerting</td>
<td>Soft keys displayed during the call-alerting call stage.</td>
</tr>
<tr>
<td>softkey connected</td>
<td>Soft keys displayed during the call-connected call stage.</td>
</tr>
<tr>
<td>conference drop-mode</td>
<td>When conferences are dropped:</td>
</tr>
<tr>
<td>conference add-mode</td>
<td>Who can add parties to a conference:</td>
</tr>
<tr>
<td>conference admin</td>
<td>Ad hoc and meet-me hardware conference administrator. The administrator can:</td>
</tr>
<tr>
<td>Preferred codec</td>
<td>Codec to use when initiating a call.</td>
</tr>
</tbody>
</table>

- creator: Conference terminates when the creator hangs up.
- local: Conference terminates when the last local party in the conference hangs up or drops out of the conference.
- never: Conference is not dropped, even if the creator hangs up, if three parties remain in the conference.

- creator: Only the creator can add parties.
- all: Any party can add other parties if the creator remains in the conference.
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>button-layout</td>
<td>Type of IP phone and number of fixed line or feature set.</td>
</tr>
<tr>
<td>User Locale</td>
<td>Locale that is associated with the phone user interface. The user locale identifies a set of detailed information, including language and font, to support users.</td>
</tr>
<tr>
<td>Network Locale</td>
<td>Locale that is associated with the phone. The network locale contains a definition of the tones and cadences that are used by the phones and gateways in the device pool in a specific geographic area.</td>
</tr>
<tr>
<td>Emergency response location</td>
<td>Identification of the ERL defined with the emergency response location command.</td>
</tr>
<tr>
<td>lpcor (incoming)</td>
<td>Setting of the lpcor incoming command.</td>
</tr>
<tr>
<td>lpcor (outgoing)</td>
<td>Setting of the lpcor outgoing command.</td>
</tr>
<tr>
<td>lpcor type</td>
<td>Setting of the lpcor type command.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template</td>
<td>Creates an ephone template.</td>
</tr>
</tbody>
</table>
**show telephony-service fac**

To display current feature access codes (FACs), use the `show telephony-service fac` command in privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Phone users dial FACs to access phone features. The set of standard FACs must be enabled using the `fac standard` before phone users can use them. Individual FACs can be changed using the `fac custom` command.

**Examples**

The following example displays the set of standard FACs:

```
Router# show telephony-service fac
telephony-service fac standard
callfwd all **1
callfwd cancel **2
pickup local **3
pickup group **4
pickup direct **5
park **6
dnd **7
redial **8
voicemail **9
ephone-hunt join *3
ephone-hunt cancel #3
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>fac</td>
<td>Enables standard FACs or creates a custom FAC.</td>
</tr>
</tbody>
</table>
show telephony-service security-info

To display the security-related information that is configured under telephony-service, use the show telephony-service security-info command in privileged EXEC configuration mode.

Syntax Description

show telephony-service security-info

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example displays security information that was configured under telephony-service.

Router# show telephony-service security-info
Skinny Server Trustpoint for TLS: cisco1
TFTP Credentials Trustpoint: cisco1
Server Security Mode: Secure
Global Device Security Mode: Authenticated
show telephony-service tftp-bindings

To display the current configuration files accessible to IP phones, use the `show telephony-service tftp-bindings` command in user EXEC or privileged EXEC mode.

**show telephony-service tftp-bindings**

**Syntax Description**
This command has no arguments or keywords.

**Command Modes**
User EXEC
Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco CME Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)YT</td>
<td>2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command with Cisco IOS Telephony Services V2.1, Cisco CallManager Express 3.0, or a later version.

This command provides a list of configuration files that are accessible to IP phones using TFTP, including the dictionary, language, and tone configuration files that are associated with the ISO-3166 codes that have been selected using the `user-locale` and `network-locale` commands.

**Examples**
The following is sample output from the `show telephony-service tftp-bindings` when the ISO-3166 code for Germany has been selected for both language and tones:

```
Router(config)# show telephony-service tftp-bindings
tftp-server system:/its/SEPDEFAULT.cnf
tftp-server system:/its/XMLDefault.cnf.xml alias XMLDefault.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEP00036B5B15.cnf.xml
tftp-server system:/its/germany/7960-font.xml alias Germany_Germany/7960-font.xml
tftp-server system:/its/germany/7960-dictionary.xml alias Germany_Germany/7960-dictionary.xml
tftp-server system:/its/germany/7960-kate.xml alias Germany_Germany/7960-kate.xml
tftp-server system:/its/germany/SCCP-dictionary.xml alias Germany_Germany/SCCP-dictionary.xml
tftp-server system:/its/germany/7960-tones.xml alias Germany_Germany/7960-tones.xml
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>network-locale</td>
<td>Sets the definition of the tones and cadences on the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G for a specific geographic area.</td>
</tr>
<tr>
<td>user-locale</td>
<td>Sets language for displays on the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G.</td>
</tr>
</tbody>
</table>
show telephony-service voice-port

To display configurations of virtual voice ports in a Cisco CallManager Express (Cisco CME) system, use the `show telephony-service voice-port` command in user EXEC or privileged EXEC mode.

### Syntax Description

This command has no arguments or keywords.

### Command Modes

- User EXEC (>)
- Privileged EXEC (#)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco CME Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command displays virtual voice-port configurations for a Cisco CME system. Each ephone-dn corresponds to a virtual voice port. For example, the ephone-dn with dn-tag 7 corresponds to virtual voice port 50/0/7. The virtual voice port provides the telephone line associated with the Cisco IP phone extension (ephone-dn).

### Examples

The following is sample output from this command:

```
Router# show telephony-service voice-port
voice-port 50/0/1
  station-id number 5001
!
voice-port 50/0/2
  station-id number 5002
  timeout ringing 8
!
voice-port 50/0/3
  station-id number 5003
!
voice-port 50/0/4
  station-id number 5004
!
```

The following table describes significant fields in this output, in alphabetical order.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>station-id number</td>
<td>Phone number used for caller ID purposes for calls made from this voice port.</td>
</tr>
<tr>
<td>timeout ringing</td>
<td>Maximum amount of time that a phone is allowed to ring before the call is disconnected.</td>
</tr>
<tr>
<td>voice-port</td>
<td>Virtual voice port.</td>
</tr>
</tbody>
</table>
## Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>show telephony-service all</code></td>
<td>Displays the detailed configuration of all the Cisco IP phones.</td>
</tr>
<tr>
<td><code>show telephony-service dial-peer</code></td>
<td>Displays dial-peer information for extensions in a Cisco CME system.</td>
</tr>
<tr>
<td><code>show telephony-service ephone-dn</code></td>
<td>Displays information for extensions (ephone-dns) in a Cisco CME system.</td>
</tr>
</tbody>
</table>
show voice emergency

To display the IP address, subnet mask, and ELIN for each emergency response location, use the `show voice emergency` command in user EXEC or privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

No default behavior or values

**Command Modes**

User EXEC (`>`)
Privileged EXEC (`#`)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to display the IP address, subnet mask, and ELIN for each emergency response location.

**Examples**

The following example shows sample output which includes IP mask and ELIN information for each ERL:

```
EEMERGENCY RESPONSE LOCATIONS
ERL  |  ELIN 1  |  ELIN2  |  SUBNET 1  |  SUBNET 2
1    | 6045550101 |         | 10.0.0.0   | 255.0.0.0
2    | 6045550102 | 6045550106 | 192.168.0.0 | 255.255.0.0
3    |       | 6045550107 | 172.16.0.0 | 255.255.0.0
4    | 6045550103 |         | 192.168.0.0 | 255.255.0.0
5    | 6045550105 |         | 209.165.200.224 | 255.0.0.0
6    | 6045550198 | 6045550109 | 209.165.201.0 | 255.255.255.224
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice emergency response location</td>
<td>Creates a tag for identifying an ERL for E911 services.</td>
</tr>
</tbody>
</table>
show voice emergency addresses

To display address information for each emergency response location, use the `show emergency addresses` command in user EXEC or privileged EXEC mode.

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

This command displays the physical address of each emergency response location.

Examples

The following example shows a sample output which includes physical address information for the ERL:

```
Router# show voice emergency addresses
3850 Zanker Rd, San Jose, 604, 5550101
225 W Tasman Dr, San Jose, 604, 5550102
275 W Tasman Dr, San Jose, 604, 5550103
518 Bellew Dr, Milpitas, 604, 5550104
400 Tasman Dr, San Jose, 604, 5550105
3675 Cisco Way, San Jose, 604, 5550106
```
show voice emergency all

To display all emergency response location information, use the **show voice emergency all** command in user EXEC or privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

No default behavior or values

**Command Modes**

User EXEC (>)  
Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to display all information configured for each emergency response location.

**Examples**

The following example shows a sample output, displaying all ERL-related information for ERL 1 and 3.

```
VOICE EMERGENCY RESPONSE SETTINGS
Callback Number: 6045550103
Emergency Line ID Number: 6045550155
Expiry: 2 minutes
Logging Enabled
EMERGENCY RESPONSE LOCATION 1
  Name: Cisco Systems 1
  Address: 3850 Zanker Rd, San Jose,elin.1.3,elin.4.10
  IP Address 1: 209.165.200.226 IP mask 1: 255.255.255.254
  IP Address 2: 209.165.202.129 IP mask 2: 255.255.0.0
  Emergency Line ID 1: 6045550180
  Emergency Line ID 2: 6045550188  [Jan 30 2007 16:05.52 PM]
  Last Caller: 6045550188  [Jan 30 2007 16:05.52 PM]
  Next ELIN For Emergency Call: 6045550166
EMERGENCY RESPONSE LOCATION 3
  Name: Cisco Systems 3
  Address: 225 W Tasman Dr, San Jose,elin.1.3,elin.4.10
  IP Address 1: 209.165.202.133 IP mask 1: 255.255.0.0
  IP Address 2: 209.165.202.130 IP mask 2: 255.0.0.0
  Emergency Line ID 1: 6045550150
  Emergency Line ID 2: 6045550150
  Last Caller: 6045550150  [Jan 30 2007 16:05.52 PM]
  Next ELIN For Emergency Call: 6045550151
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>address</td>
<td>Specifies a comma separated text entry (up to 250 characters) of an ERL’s civic address.</td>
</tr>
<tr>
<td>elin</td>
<td>Specifies a PSTN number that will replace the caller’s extension.</td>
</tr>
<tr>
<td>name</td>
<td>Specifies a string (up to 32 characters) used internally to identify or describe the emergency response location.</td>
</tr>
<tr>
<td>subnet</td>
<td>Defines which IP phones are part of this ERL.</td>
</tr>
<tr>
<td>voice emergency response location</td>
<td>Creates a tag for identifying an ERL for the E911 services.</td>
</tr>
</tbody>
</table>
show voice emergency callers

To display a list of 911 calls made over the last three hours, use the `show emergency callers` command in privileged EXEC mode.

### Syntax Description

This command has no arguments or keywords.

### Command Default

No list of 911 calls is displayed.

### Command Modes

Privileged EXEC (#)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1</td>
<td>This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)</td>
<td>This command was added to Cisco Unified CME.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use this command to display a list of all 911 calls made in the past three hours. The list shows the originating number, the ELIN used, and the time the call was placed.

### Examples

The following example shows sample output, which includes the originating number, the ELIN used, and the time the call was placed:

```
router# show voice emergency callers
EMERGENCY CALLS CALL BACK TABLE
ELIN | GALLER | TIME
6045550181 | 8155550151 | Oct 12 2006 04:05:21
6045550182 | 8155550152 | Oct 12 2006 04:05:21
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice emergency response location</td>
<td>Creates a tag for identifying an ERL for the enhanced 911 service.</td>
</tr>
</tbody>
</table>
show voice emergency zone

To display each emergency response zone’s list of locations in priority order, use the `show voice emergency zone` command in user EXEC or privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

No default behavior or values

**Command Modes**

User EXEC (>)
Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to display a list of the locations, in priority order, of all configured emergency response zones.

**Examples**

The following example shows a sample output which displays the ERL locations for emergency response zones 90 and 100.

```
EMERGENCY RESPONSE ZONES
zone 90
  location 4
  location 5
  location 6
  location 7
  location 2147483647
zone 100
  location 1 priority 1
  location 2 priority 2
  location 3 priority 3
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>location</td>
<td>Identifies locations within an emergency response zone.</td>
</tr>
<tr>
<td>voice emergency response location</td>
<td>Creates a tag for identifying an ERL for the enhanced 911 service.</td>
</tr>
<tr>
<td>voice emergency response zone</td>
<td>Creates an emergency response zone within which ERLs can be grouped.</td>
</tr>
</tbody>
</table>
show voice fac statistics

To display the FAC failure statistics collected by the system, use the **show voice fac statistics** command in privileged EXEC mode.

**show voice fac statistics**

Syntax Description
This command has no arguments or keywords.

Command Modes
Privileged EXEC

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

Usage Guidelines
Use this command to display the forced authentication code (FAC) success or failure statistics collected by the system.

Examples
The following is sample output from this command displaying all statistical information:

```
Router# show voice fac statistics
Voice FAC statistics for failure calls:
  Total basic calls:  5
  Total forward calls:  1
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show call active voice</td>
<td>Displays call information for voice calls that are in progress.</td>
</tr>
<tr>
<td>show call history voice</td>
<td>Displays the call history table for voice calls.</td>
</tr>
</tbody>
</table>
show voice hunt-group

To display configuration information associated with one or all voice hunt groups in a Cisco Unified Communications Manager Express (Cisco Unified CME) system, use the `show voice hunt-group` command in privileged EXEC mode.

```
show voice hunt-group hunt-group-tag [brief] {longest-idle|parallel|peer|sequential}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>hunt-group-tag</code></td>
<td>(Optional) Unique sequence number that identifies the voice hunt group. Range is 1 to 100.</td>
</tr>
<tr>
<td><code>brief</code></td>
<td>(Optional) Displays brief information on all voice hunt groups in a Cisco CME system.</td>
</tr>
<tr>
<td><code>longest-idle</code></td>
<td>(Optional) Displays summary of longest-idle voice hunt groups.</td>
</tr>
<tr>
<td><code>parallel</code></td>
<td>(Optional) Displays summary of parallel voice hunt groups.</td>
</tr>
<tr>
<td><code>peer</code></td>
<td>(Optional) Displays summary of peer voice hunt groups.</td>
</tr>
<tr>
<td><code>sequential</code></td>
<td>(Optional) Displays summary of sequential voice hunt groups.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(24)T</td>
<td>This command was introduced in a release earlier than Cisco IOS Release 12.4(24)T.</td>
</tr>
<tr>
<td>15.2(2)T</td>
<td>This command was modified to add stat collect as a field.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `show voice hunt-group` command to get information about voice hunt group configuration on the gateway as an alternative to the `show running-config` command.

Use the `show voice hunt-group` and `show voice hunt-group brief` commands to display hunt group configuration information for all voice hunt groups in a Cisco Unified CME system. Use `show voice hunt-group hunt-group-tag` to display data on a specific hunt-tag configuration created by the `voice hunt-group` command. Use the `longest-idle`, `parallel`, `peer`, or `sequential` keywords to display data on a specific type of voice hunt group configuration created by the `voice hunt-group` command.

**Examples**

The following is a sample output from the `show voice hunt-group` command, displaying all voice hunt groups configured on the router:

```
Router# show voice hunt-group
Group 1
  type: longest-idle
  preference: 0
  preference (sec): 0
  timeout: 0
  final_number: 1
Group 34
  type: parallel
  pilot number: 3, peer-tag 2147483647
  secondary number: 4, peer-tag 2147483646
```
The following is a sample output from the `show voice hunt-group` command, displaying the configuration for all the configured voice hunt groups:

```
Router# show voice hunt-group
Group 5
  type: parallel
  pilot number: 1234, peer-tag 1234
  list of numbers:
     MEMBER   USED_BY   STATE   LOGIN/LOGOUT
            -------     ------   ------------
     9498899994  9498899994  DOWN  Logout
     9498899993  9498899994  UP    Login
     *          -         -       -
  secondary number: 5678, peer-tag 5678
  list preference: 5
  preference (sec): 8
  timeout: 180
  final_number: 4444
Group 8
  type: longest-idle
  pilot number: 6666, peer-tag 6666
  list of numbers:
     MEMBER   USED_BY   STATE   LOGIN/LOGOUT
            -------     ------   ------------
     5106575902  5106575902  UP    Login
     4088531111  4088531111  UP    Login
     4083911375  4083911375  DOWN  Login
     4089306067  4089306067  DOWN  Logout
     8869395033  8869395033  DOWN  Logout
     88686619633 88686619633  DOWN  -
  preference: 0
  preference (sec): 0
  timeout: 180
  final_number:
  hops: 6
  phone-display: Yes
Group 10
  type: longest-idle
  pilot number: 7777777, peer-tag 7777777
  secondary number: 88888888, peer-tag 88888888
  list of numbers:
     MEMBER   USED_BY   STATE   LOGIN/LOGOUT
            -------     ------   ------------
     7654321   7654321  DOWN  Logout
     87654321   87654321  UP    Login
     987654321  987654321  UP    Logout
  preference: 0
  preference (sec): 0
  timeout: 180
  final_number:
  hops: 3
  phone-display: No
Group 15
  type: peer
  pilot number: 56789, peer-tag 56789
  list of numbers:
     MEMBER   USED_BY   STATE   LOGIN/LOGOUT
            -------     ------   ------------
```
The following is a sample output from the `show voice hunt-group` command, displaying information for a particular voice hunt group as specified by the `hunt-group-tag` number:

```
Router# show voice hunt-group 5
Group 5
  type: parallel
  pilot number: 1234, peer-tag 1234
  secondary number: 5678, peer-tag 5678
  list of numbers:
  MEMBER    USED_BY    STATE  LOGIN/LOGOUT
  -----------  -----------  -------  -----------
  9498889994  9498889994  UP      Logout
  9498889993  9498889993  DOWN     Login
  *           *           *        *
  preference: 5
  preference (sec): 8
  timeout: 20
  final_number: 4444
```

The following is a sample output from the `show voice hunt-group` command, displaying information about all the voice hunt groups of a particular type:

```
Router# show voice hunt-group longest-idle
Group 8
  type: longest-idle
  pilot number: 6666, peer-tag 6666
  list of numbers:
  MEMBER    USED_BY    STATE  LOGIN/LOGOUT
  -----------  -----------  -------  -----------
  5106575902  5106575902  UP      Logout
  4088531111  4088531111  UP      Login
  4083911375  4083911375  DOWN     -
  4089306067  4089306067  UP      Logout
  8869395033  8869395033  -        -
  88686619633  88686619633  UP      Login
  preference: 0
  preference (sec): 0
  timeout: 180
  final_number: 6
  phone-display: Yes
Group 10
  type: longest-idle
  pilot number: 7777777, peer-tag 7777777
  secondary number: 88888888, peer-tag 88888888
  list of numbers:
  MEMBER    USED_BY    STATE  LOGIN/LOGOUT
  -----------  -----------  -------  -----------
  7654321     7654321    UP      Logout
  87654321     87654321    UP      Login
  987654321    987654321  DOWN     Logout
  preference: 0
```
The following is a sample output from the `show voice hunt-group` command with the keyword `brief`:

```
Router# show voice hunt-group brief
TAG  TYPE  PILOT          LIST
---  ----  ---------------  ---------------------------------  
5    PAR   1234           9498889-, 9498889-
       5678           9498889-, 9498889-
8    LON   6666           5106575-, 4088531-, 4083911-, 4089306-, 8869395-,.....
10   LON   7777777        7654321, 8765432-, 9876543-
       8888888-       7654321, 8765432-, 9876543-
15   PER   56789         8765432-, 9876, 87654-
```

The following is a sample output from the `show voice hunt-group` command, indicating that call statistics is being collected:

```
Router# show voice hunt-group 1
Group 1
 type: parallel
 pilot number: 5000, peer-tag 2147483647
 list of numbers:
 MEMBER  USED_BY  STATE  LOGIN/LOGOUT
---------  -------  ------  ------------
5001  5001    UP   Logout
5002  5002    UP   Login
5011  5011   DOWN  -
5012  5012    UP   Logout
 preference: 0
 preference (sec): 0
 timeout: 12
 final_number: 5012
 stat collect: yes
 phone-display: Yes
```

The following is a sample output from the `show voice hunt-group` command when there is no voice hunt group configured:

```
Router# show voice hunt-group
no voice hunt-groups configured
Router# show voice hunt-group brief
no voice hunt-groups configured
Router# show voice hunt-group longest-idle
no voice hunt-groups configured
Router#
```

The following table describes the significant fields shown in the output.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Group</td>
<td>Tag number of voice hunt group.</td>
</tr>
</tbody>
</table>
### Field Description

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>type</td>
<td>Type of voice hunt group. The available voice hunt group types are: longest-idle, parallel, peer and sequential.</td>
</tr>
<tr>
<td>pilot number</td>
<td>Number that callers dial to reach the specified voice hunt group.</td>
</tr>
<tr>
<td>secondary-number</td>
<td>Alternate number for the specified voice hunt group.</td>
</tr>
<tr>
<td>list of numbers</td>
<td>Numbers of the extensions configured in the <code>voice hunt-group</code> command’s hunt-tag identifier.</td>
</tr>
<tr>
<td>preference</td>
<td>Preference order for the extension or telephone number associated with a dial peer. Range is 0 to 8. Default is 0.</td>
</tr>
<tr>
<td>preference (sec)</td>
<td>Preference order for the secondary pilot number. Range is from 0 to 10, where 0 is the highest preference and 10 is the lowest preference. Default is 9.</td>
</tr>
<tr>
<td>timeout</td>
<td>Number of seconds after which a call that is not answered at one number is redirected to the next number in the hunt group list.</td>
</tr>
<tr>
<td>final_number</td>
<td>Last number in the voice hunt group, after which a call is no longer redirected.</td>
</tr>
<tr>
<td>hops</td>
<td>Number of hops before a call proceeds to the final number.</td>
</tr>
<tr>
<td>stat collect</td>
<td>Yes indicates that call statistics are being collected for a voice hunt group.</td>
</tr>
<tr>
<td>phone-display</td>
<td>Displays the hunt group information on My Phone Apps service button.</td>
</tr>
<tr>
<td>hlog-block</td>
<td>Blocks the hlog functionality of voice hunt group on the phone.</td>
</tr>
<tr>
<td>peer-tag</td>
<td>Peer hunting tag.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>final (voice hunt-group)</code></td>
<td>Defines the last extension in a voice hunt group.</td>
</tr>
<tr>
<td><code>hops (voice hunt-group)</code></td>
<td>Defines the number of times that a call is redirected to the next directory number in a peer voice hunt group list before proceeding to the final directory number.</td>
</tr>
<tr>
<td><code>list (voice hunt-group)</code></td>
<td>Defines the directory numbers that participate in a directory number hunt group.</td>
</tr>
<tr>
<td><code>pilot (voice hunt-group)</code></td>
<td>Defines the voice dn that callers dial to reach a Cisco Unified Communications Manager Express (Cisco Unified CME) voice hunt group.</td>
</tr>
<tr>
<td><code>timeout (voice hunt-group)</code></td>
<td>Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list and defines the last directory number in the hunt group.</td>
</tr>
<tr>
<td><code>voice hunt-group</code></td>
<td>Configures voice hunt groups and the associated parameters.</td>
</tr>
</tbody>
</table>
show voice hunt-group statistics

To display call statistics from voice hunt groups, use the **show voice hunt-group statistics** command in privileged EXEC mode.

```
show voice hunt-group group-id statistics {last hours hours|start day time [to day time]}
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>group-id</code></td>
<td>Identifier for the voice hunt group. Range: 1 to 100.</td>
</tr>
<tr>
<td><code>last</code></td>
<td>Displays the latest call statistics for a voice hunt group for a specified number of hours, counting backward from the current hour. Range: 1 to 167.</td>
</tr>
<tr>
<td><code>hours</code></td>
<td>Number of hours that the call statistics are displayed.</td>
</tr>
<tr>
<td><code>start</code></td>
<td>Defines the start of the period for which the call statistics are displayed. Default duration is one hour.</td>
</tr>
<tr>
<td><code>day</code></td>
<td>Abbreviated day of the week. The following abbreviations are valid: sun, mon, tue, wed, thu, fri, sat.</td>
</tr>
<tr>
<td><code>time</code></td>
<td>Hour of the day. Range: 0 to 23.</td>
</tr>
<tr>
<td><code>to</code></td>
<td>(Optional) Defines the time the display of the call statistics ends.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the **show voice hunt-group statistics** command to display the average and longest times for a voice hunt group to answer a call, make a call, or put a call on hold. The command can also display the number of answered and abandoned calls, the number of calls forwarded to or answered by voice mail, and the number of error calls.

The output is dependent on call activity. If there is no activity, no data is displayed.

If your Cisco Unified CME system is configured with the basic automatic call distribution (B-ACD) and auto-attendant service, you can enable the collection of call statistics for every voice hunt group with the **voice hunt-group statistics collect** command. Additional data is displayed for all agents combined and for individual agents.

**Note**

On the day that daylight saving time adjusts the time back by one hour at 2 a.m. each year, the original 1 a.m. to 2 a.m. statistics for that day are lost because they are overwritten by the new 1 a.m. to 2 a.m. statistics.

For remote Cisco Unified SCCP IP phones in voice hunt groups, the hold and resume statistics are not updated.
The following is a sample output from the **show voice hunt-group statistics** command. The output includes direct calls to a voice hunt group number and calls from queue/B-ACD.

```
Router# show voice hunt-group 1 statistics last 1 h
Wed 04:00 - 05:00
Max Agents: 3
Min Agents: 3
Total Calls: 9
Answered Calls: 7
Abandoned Calls: 2
Average Time to Answer (secs): 6
Longest Time to Answer (secs): 13
Average Time in Call (secs): 75
Longest Time in Call (secs): 161
Average Time before Abandon (secs): 8
Calls on Hold: 2
Average Time in Hold (secs): 16
Longest Time in Hold (secs): 21
Per agent statistics:
Agent: 5012
From Direct Call:
Total Calls Answered: 3
Average Time in Call (secs): 70
Longest Time in Call (secs): 150
Totals Calls on Hold: 1
Average Hold Time (secs): 21
Longest Hold Time (secs): 21
From Queue:
Total Calls Answered: 3
Average Time in Call (secs): 55
Longest Time in Call (secs): 79
Total Calls on Hold: 2
Average Hold Time (secs): 19
Longest Hold Time (secs): 26
Agent: 5013
From Direct Call:
Total Calls Answered: 3
Average Time in Call (secs): 51
Longest Time in Call (secs): 118
Totals Calls on Hold: 1
Average Hold Time (secs): 11
Longest Hold Time (secs): 11
From Queue:
Total Calls Answered: 1
Average Time in Call (secs): 4
Longest Time in Call (secs): 4
Agent: 5014
From Direct Call:
Total Calls Answered: 1
Average Time in Call (secs): 161
Longest Time in Call (secs): 161
From Queue:
Total Calls Answered: 1
Average Time in Call (secs): 658
Longest Time in Call (secs): 658
Queue related statistics:
Total calls presented to the queue: 5
Calls handoff to IOS: 5
Number of calls in the queue: 0
Average time to handoff (secs): 2
Longest time to handoff (secs): 3
Number of abandoned calls: 0
```
The following is a sample output from the `show voice hunt-group statistics` command. The output focuses on queue-related statistics.

**Queue related statistics:**
- Total calls presented to the queue: 8
- Calls handoff to IOS: 3
- Number of calls in the queue: 1
- Average time to handoff (secs): 10
- Longest time to handoff (secs): 15
- Number of abandoned calls: 4
- Average time before abandon (secs): 7
- Calls forwarded to voice mail: 0
- Calls answered by voice mail: 0
- Number of error calls: 0

The following is a sample output from the `show voice hunt-group statistics` command. The output shows that no call statistics were collected from voice hunt group 1 from 2:00 to 4:00 on a Monday.

```
Router# show voice hunt-group 1 stat start Mon 2 to Mon 4
Mon 02:00 - 03:00
  No info
Mon 03:00 - 04:00
  No info
Mon 04:00 - 05:00
  No info
```

The following table describes the significant fields shown in the display.

**Table 56: show voice hunt-group statistics Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Abandoned calls</td>
<td>Total number of calls abandoned by hunt group agents. This does not include calls going to the final number.</td>
</tr>
<tr>
<td>Answered call</td>
<td>Total number of calls answered by hunt group agents.</td>
</tr>
<tr>
<td>Average time in call (secs)</td>
<td>Average length of time that unanswered calls waited before going to an agent.</td>
</tr>
<tr>
<td>Average time to answer (secs)</td>
<td>Average length of time that all calls to Cisco Unified CME B-ACD waited before being answered.</td>
</tr>
<tr>
<td>Average time in hold (secs)</td>
<td>Average length of time that calls were kept on hold for all agents.</td>
</tr>
<tr>
<td>Average hold time (secs)</td>
<td>Average length of time that calls waited on hold for this agent.</td>
</tr>
<tr>
<td>Average time to handoff (secs)</td>
<td>Average length of time before a call was handed off to IOS</td>
</tr>
<tr>
<td>Calls on hold</td>
<td>Total number of calls that were placed on hold.</td>
</tr>
<tr>
<td>Calls handoff to IOS</td>
<td>Total number of calls handed off to IOS.</td>
</tr>
</tbody>
</table>
### Field Description

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calls answered by voice mail</td>
<td>Total number of calls to Cisco Unified CME B-ACD that were answered by voice mail.</td>
</tr>
<tr>
<td>Calls forwarded to voice mail</td>
<td>Total number of calls to Cisco Unified CME B-ACD that were forwarded to voice mail.</td>
</tr>
<tr>
<td>Longest time to answer (secs)</td>
<td>Longest length of time before calls to Cisco Unified CME B-ACD were answered.</td>
</tr>
<tr>
<td>Longest time in call (secs)</td>
<td>Longest length of time that all calls to Cisco Unified CME B-ACD that went to an agent waited in a call queue.</td>
</tr>
<tr>
<td>Longest time in hold (secs)</td>
<td>Longest length of time that a call spent between being placed on hold and being picked up by agents.</td>
</tr>
<tr>
<td>Longest hold time (secs)</td>
<td>Longest length of time that a call to this agent was spent between being placed on hold and being picked up.</td>
</tr>
<tr>
<td>Longest time to handoff (secs)</td>
<td>Longest length of time before a call was handed off to IOS.</td>
</tr>
<tr>
<td>Max agent</td>
<td>Maximum number of hunt group agents.</td>
</tr>
<tr>
<td>Min agent</td>
<td>Minimum number of hunt group agents.</td>
</tr>
<tr>
<td>Number of abandoned calls</td>
<td>Total number of calls to Cisco Unified CME B-ACD that hung up before being answered.</td>
</tr>
<tr>
<td>Number of calls in the queue</td>
<td>Total number of calls in the queue.</td>
</tr>
<tr>
<td>Number of error calls</td>
<td>Total number of error calls.</td>
</tr>
<tr>
<td>Total calls</td>
<td>Total number of direct calls made to the hunt group.</td>
</tr>
<tr>
<td>Total calls answered</td>
<td>Total number of calls to Cisco Unified CME B-ACD that were answered by an agent.</td>
</tr>
<tr>
<td>Total calls on hold</td>
<td>Total number of calls that were placed on hold for this agent.</td>
</tr>
<tr>
<td>Total calls presented to the queue</td>
<td>Total number of calls made to Cisco Unified CME B-ACD.</td>
</tr>
</tbody>
</table>

### Note
From Cisco Unified CME Release 10.5 onwards, abandoned calls will not include the calls going to the final number. However, the total calls includes calls going to the final number. Use the formula "**Final Calls**= **Total Calls** - **Answered Calls** - **Abandoned Calls**", to calculate the calls going to the final number.

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice hunt-group statistics collect</td>
<td>Enables the collection of call statistics for voice hunt groups.</td>
</tr>
</tbody>
</table>
show voice register all

To display all Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified Communications Manager Express (Cisco Unified CME) configurations and register information, use the `show voice register all` command in privileged EXEC mode.

**show voice register all**

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was added to Cisco CME.</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco SIP SRST 8.0</td>
<td>This command was modified to display the signaling transport protocol.</td>
</tr>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1 Cisco Unified SIP SRST 8.1</td>
<td>The output display of this command was modified.</td>
</tr>
<tr>
<td>15.2(4)M</td>
<td>Cisco Unified CME 9.1</td>
<td>This command was modified to include Key Expansion Module (KEM) data in the output display.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

KEM data are displayed for Cisco Unified CME only. Cisco Unified SRST is unable to gather all the configuration details about KEMs from Cisco Unified CM.

**Examples**

**Cisco Unified SIP SRST**

The following is a sample output of the `show voice register all` command:

```
Router# show voice register all
VOICE REGISTER GLOBAL
-----------------------------------------------
CONFIG {Version=8.1}
-----------------------------------------------
Version 8.1
Mode is srst
Max-pool is 10
Max-dn is 10
Outbound-proxy is enabled and will use global configured value
Security Policy: DEVICE-DEFAULT
timeout interdigit 10
network-locale{0} US   (This is the default network locale for this box)
```
network-locale[1] US
user-locale(0) US (This is the default user locale for this box)
user-locale(1) US
user-locale(2) US
user-locale(3) US
user-locale(4) US  Active registrations : 0
Total SIP phones registered: 0
Total Registration Statistics
  Registration requests : 0
  Registration success : 0
  Registration failed : 0
  unRegister requests : 0
  unRegister success : 0
  unRegister failed : 0
  Attempts to register after last unregister : 0
  Last register request time :
  Last unregister request time :
  Register success time :
  Unregister success time :

VOICE REGISTER DN
----------------------
Dn Tag 1
Config:
  Number is 45111
  Preference is 0
  Huntstop is disabled
  Pool 1  has this DN configured for line 1
Dn Tag 2
Config:
  Number is 45112
  Preference is 0
  Huntstop is disabled
  Pool 2  has this DN configured for line 1
Dn Tag 3
Config:
  Number is 45113
  Preference is 0
  Huntstop is disabled
  Pool 3  has this DN configured for line 1, 2
Dn Tag 4
Config:
Dn Tag 7
Config:
  Number is 451110
  Preference is 0
  Huntstop is disabled
  Pool 1  has this DN configured for line 4
Dn Tag 8
Config:
  Pool 1  has this DN configured for line 3

VOICE REGISTER POOL
---------------------
Pool Tag 1
Config:
  Mac address is 001B.535C.D410
  Number list 1 : DN 1
  Number list 3 : DN 8
  Number list 4 : DN 7
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
kpml signal is disabled
Lpcor Type is none
Reason for unregistered state:
No registration request since last reboot/unregister
Dialpeers created:
Statistics:
Active registrations : 0
Total SIP phones registered: 0
Total Registration Statistics
  Registration requests : 0
  Registration success : 0
  Registration failed : 0
  unRegister requests : 0
  unRegister success : 0
  unRegister failed : 0
  Attempts to register after last unregister : 0
  Last register request time :
  Last unregister request time :
  Register success time :
  Unregister success time :

Pool Tag 2
Config:
  Mac address is 0015.C68E.6D13
  Number list 1 : DN 2
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  kpml signal is disabled
  Lpcor Type is none
Reason for unregistered state:
No registration request since last reboot/unregister
Dialpeers created:
Statistics:
Active registrations : 0
Total SIP phones registered: 0
Total Registration Statistics
  Registration requests : 0
  Registration success : 0
  Registration failed : 0
  unRegister requests : 0
  unRegister success : 0
  unRegister failed : 0
  Attempts to register after last unregister : 0
  Last register request time :
  Last unregister request time :
  Register success time :
  Unregister success time :

Pool Tag 3
Config:
  Mac address is 0021.5553.8998
  Number list 1 : DN 3
  Number list 2 : DN 3
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  kpml signal is enabled
  Lpcor Type is none
Reason for unregistered state:
No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations : 0
  Total SIP phones registered: 0
  Total Registration Statistics
Cisco Unified CME

The following is a sample output of the `show voice register all` command:

```
Router# show voice register all
1) show voice register all
VOICE REGISTER GLOBAL
-----------------------------
Version 8.1
Mode is cme
Max-pool is 10
Max-dn is 10
Outbound-proxy is enabled and will use global configured value
Security Policy: DEVICE-DEFAULT
Source-address is 8.3.3.5 port 5060
Time-format is 12
Date-format is M/D/Y
Time-zone is 5
Hold-alert is disabled
Mwi stutter is disabled
Mwi registration for full E.164 is disabled
Forwarding local is enabled
Privacy is enabled
Privacy-on-hold is disabled
Dst auto adjust is enabled
  start at Apr week 1 day Sun time 02:00
  stop at Oct week 8 day Sun time 02:00
Max redirect number is 5
IP QoS DSCP:
  ef (the MS 6 bits, 46, in ToS, 0xB8) for media
  cs3 (the MS 6 bits, 24, in ToS, 0x60) for signal
  af41 (the MS 6 bits, 34, in ToS, 0x88) for video
  default (the MS 6 bits, 0, in ToS, 0x0) for service
Telnet Level: 0
Tftp path is flash:
Generate text file is disabled
Tftp files are created, current syncinfo 0001140473454008
OS79XX.TXT is not created
timeout interdigit 10
network-locale[0] US  (This is the default network locale for this box)
network-locale[1] US
user-locale[0] US  (This is the default user locale for this box)
user-locale[1] US
```

Cisco Unified Communications Manager Express Command Reference
show voice register all

user-locale[2] US
user-locale[3] US
user-locale[4] US  Active registrations : 0
Total SIP phones registered: 0
Total Registration Statistics
  Registration requests : 0
  Registration success : 0
  Registration failed : 0
  unregister requests : 0
  unregister success : 0
  unregister failed : 0
Attempts to register after last unregister : 0
Last register request time :
Last unregister request time :
Register success time :
Unregister success time :

VOICE REGISTER DN

------------------
Dn Tag 1
Config:
  Number is 45111
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  Pool 1 has this DN configured for line 1
Dn Tag 2
Config:
  Number is 45112
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
call-forward b2bua noan 999 timeout 8
  after-hour exempt
  Pool 2 has this DN configured for line 1
  Pool 7 has this DN configured for line 1
Dn Tag 3
Config:
  Number is 45113
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
call-forward b2bua all 87687
  after-hour exempt
  Pool 3 has this DN configured for line 1, 2
Dn Tag 4
Config:
  Auto answer is disabled
Dn Tag 7
Config:
  Number is 451110
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  after-hour exempt
  Pool 1 has this DN configured for line 4
Dn Tag 8
Config:
  Auto answer is disabled
call-forward b2bua all 678
  after-hour exempt
  Pool 1 has this DN configured for line 3

VOICE REGISTER TEMPLATE

-----------------------
Temp Tag 1
Config:
Attended Transfer is enabled
Blind Transfer is enabled
Semi-attended Transfer is enabled
Conference is enabled
Caller-ID block is disabled
dnD control is enabled
Anonymous call block is disabled
Dialplan Tag is 1
Softkey connected Confrn
Lpcor type none
Pool 4 has this template configured

VOICE REGISTER DIALPLAN
-----------------------
Dialplan Tag 1
Config:
Type is 7905-7912
Template 1 has this dialplan configured
Pool 4 has this dialplan configured

VOICE REGISTER POOL
---------------------
Pool Tag 1
Config:
Mac address is 001B.535C.D410
Type is 7960
Number list 1 : DN 1
Number list 3 : DN 8
Number list 4 : DN 7
Proxy Ip address is 0.0.0.0
Dtmf Relay is disabled
Call Waiting is enabled
dnD is disabled
Busy trigger per button value is 0
call-forward phone all is 4566
call-forward h2bua all 4555
keep-conference is enabled
Lpcor Type is none
Transport type is udp
service-control mechanism is not supported
Privacy feature is not configured.
Privacy button is disabled
Reason for unregistered state:
No registration request since last reboot/unregister

Dialpeers created:
Statistics:
Active registrations : 0
Total SIP phones registered: 0
Total Registration Statistics
Registration requests : 0
Registration success : 0
Registration failed : 0
unRegister requests : 0
unRegister success : 0
unRegister failed : 0
Attempts to register
after last unregister : 0
Last register request time :
Last unregister request time :
Register success time :
Unregister success time :

Pool Tag 2
Config:
Mac address is 0015.C68E.6D13
Type is 7960
Number list 1 : DN 2
Proxy Ip address is 0.0.0.0
DTMF Relay is disabled
Call Waiting is enabled
DnD is disabled
Busy trigger per button value is 0
call-forward phone no an is 9886, timeout 98
keep-conference is enabled
username pool2 password lab
LpCor Type is none
Transport type is udp
service-control mechanism is not supported
Privacy feature is not configured.
Privacy button is disabled
Reason for unregistered state:
  No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations : 0
Total SIP phones registered: 0
Total Registration Statistics
  Registration requests : 0
  Registration success : 0
  Registration failed : 0
  unRegister requests : 0
  unRegister success : 0
  unRegister failed : 0
  Attempts to register after last unregister : 0
  Last register request time :
  Last unregister request time :
  Register success time :
  Unregister success time :
Pool Tag 3
Config:
  Mac address is 0021.5553.8998
  Type is 7975
Number list 1 : DN 3
Number list 2 : DN 3
Proxy Ip address is 0.0.0.0
DTMF Relay is disabled
Call Waiting is enabled
DnD is enabled
Busy trigger per button value is 0
call-forward phone all is 45112
call-forward b2bua all 45111
after-hour exempt
keep-conference is enabled
kpml signal is enabled
LpCor Type is none
Transport type is udp
service-control mechanism is not supported
Privacy feature is not configured.
Privacy button is disabled
Reason for unregistered state:
  No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations : 0
Total SIP phones registered: 0
Total Registration Statistics
  Registration requests : 0
  Registration success : 0
Registration failed : 0
unRegister requests : 0
unRegister success : 0
unRegister failed : 0
Attempts to register after last unregister : 0
Last register request time :
Last unregister request time :
Register success time :
Unregister success time :

Pool Tag 4
Config:
Mac address is 8989.9867.8769
Proxy Ip address is 0.0.0.0
DTMF Relay is disabled
Call Waiting is enabled
DnD is disabled
Busy trigger per button value is 0
keep-conference is enabled
template is 1
Lpcor Type is none
Transport type is udp
service-control mechanism is not supported
Privacy feature is not configured.
Privacy button is disabled
Reason for unregistered state:
No registration request since last reboot/unregister

Dialpeers created:
Statistics:
Active registrations : 0
Total SIP phones registered: 0
Total Registration Statistics
Registration requests : 0
Registration success : 0
Registration failed : 0
unRegister requests : 0
unRegister success : 0
unRegister failed : 0
Attempts to register after last unregister : 0
Last register request time :
Last unregister request time :
Register success time :
Unregister success time :

Pool Tag 7
Config:
Mac address is 0018.BAC8.D2B1
Number list 1 : DN 2
Proxy Ip address is 0.0.0.0
DTMF Relay is disabled
Call Waiting is enabled
DnD is disabled
Busy trigger per button value is 0
keep-conference is enabled
Lpcor Type is none
Transport type is udp
service-control mechanism is not supported
Privacy feature is not configured.
Privacy button is disabled
Reason for unregistered state:
No registration request since last reboot/unregister

Dialpeers created:
Statistics:
Active registrations : 0

Cisco Unified CME Commands: S2
show voice register all
Total SIP phones registered: 0
Total Registration Statistics
  Registration requests : 0
  Registration success : 0
  Registration failed : 0
  unRegister requests : 0
  unRegister success : 0
  unRegister failed : 0
  Attempts to register
    after last unregister : 0
  Last register request time :
  Last unregister request time :
  Register success time :
  Unregister success time :

The following is an example of a partial output of the show voice register all command, showing KEM data with the phone type information:

Router# show voice register all
Pool Tag 5
Config:
  Mac address is B4A4.E328.4698
  Type is 9971 addon 1 CKEM
  Number list 1 : DN 2
  Number list 2 : DN 3
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Video is enabled
  Camera is enabled
  Busy trigger per button value is 0
  keep-conference is enabled
  registration expires timer max is 200 and min is 60
  kpml signal is enabled
  Lpcor Type is none

The following is a sample output of the show voice register all command, showing the three KEMs configured with phone type 9971:

Router# show voice register all
Pool Tag 4
Config:
  Mac address is B4A4.E328.4698
  Type is 9971 addon 1 CKEM 2 CKEM 3 CKEM
  Number list 1 : DN 4
  Number list 2 : DN 5
  Number list 3 : DN 9

The following table describes the significant fields shown in this output.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pool Tag</td>
<td>Shows the assigned tag number of the current voice register pool.</td>
</tr>
<tr>
<td>Config</td>
<td>Shows the voice register pool.</td>
</tr>
<tr>
<td>Network address and Mask</td>
<td>Shows network address and mask information when the id command is configured.</td>
</tr>
</tbody>
</table>

Cisco Unified Communications Manager Express Command Reference

1136
### Field Description

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number list, Pattern, and Preference</td>
<td>Shows the <strong>number</strong> command configuration.</td>
</tr>
<tr>
<td>Proxy IP address</td>
<td>Shows the <strong>proxy</strong> command configuration.</td>
</tr>
<tr>
<td>Default preference</td>
<td>Shows the default preference value of this pool.</td>
</tr>
<tr>
<td>Incoming called number</td>
<td>Shows the <strong>incoming called-number</strong> command configuration.</td>
</tr>
<tr>
<td>Translate outgoing called tag</td>
<td>Shows the <strong>translate-outgoing</strong> command configuration.</td>
</tr>
<tr>
<td>Class of Restriction List Tag</td>
<td>Shows the COR tag.</td>
</tr>
<tr>
<td>Incoming corlist name</td>
<td>Shows the <strong>cor</strong> command configuration.</td>
</tr>
<tr>
<td>Application</td>
<td>Shows the <strong>application</strong> command configuration for this pool.</td>
</tr>
<tr>
<td>Dialpeers created</td>
<td>Lists all the dial peers created and their contents. Dial-peer contents differ for each application and are not described here.</td>
</tr>
<tr>
<td>Statistics</td>
<td>Shows the registration statistics for this pool.</td>
</tr>
<tr>
<td>Active registrations</td>
<td>Shows the current active registrations.</td>
</tr>
<tr>
<td>Total Registration Statistics</td>
<td>Shows the total registration statistics for this pool.</td>
</tr>
<tr>
<td>Registration requests</td>
<td>Shows the incoming registration requests.</td>
</tr>
<tr>
<td>Registration success</td>
<td>Shows the successful registrations.</td>
</tr>
<tr>
<td>Registration failed</td>
<td>Shows the failed registrations.</td>
</tr>
<tr>
<td>unRegister requests</td>
<td>Shows the incoming unregister/registration expire requests.</td>
</tr>
<tr>
<td>unRegister success</td>
<td>Reports the number of successful unregisters.</td>
</tr>
<tr>
<td>unRegister failed</td>
<td>Reports the number of failed unregisters.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>application (voice register pool)</strong></td>
<td>Selects the session-level application for the dial peer associated with an individual Cisco Unified SIP IP phone in a Cisco Unified CME environment or for a group of phones in a Cisco Unified SIP SRST environment.</td>
</tr>
<tr>
<td><strong>cor (voice register pool)</strong></td>
<td>Configures a class of restriction on the VoIP dial peers associated with directory numbers.</td>
</tr>
<tr>
<td><strong>id (voice register pool)</strong></td>
<td>Explicitly identifies a locally available, individual Cisco Unified SIP IP phone or, when running Cisco Unified SIP SRST, a set of Cisco Unified SIP IP phones.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------------------------------</td>
<td>-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>incoming called-number (dial peer)</td>
<td>Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer.</td>
</tr>
<tr>
<td>number (voice register pool)</td>
<td>Indicates the E.164 phone numbers that the registrar permits to handle the Register message from a Cisco Unified SIP IP phone.</td>
</tr>
<tr>
<td>proxy (voice register pool)</td>
<td>Autogenerates additional VoIP dial peers to reach the main proxy whenever a Cisco Unified SIP IP phone registers with a Cisco Unified SIP SRST gateway.</td>
</tr>
<tr>
<td>show sip-ua status registrar</td>
<td>Displays all the SIP endpoints currently registered with the contact address.</td>
</tr>
<tr>
<td>show voice register dial-peers</td>
<td>Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified SIP SRST or Cisco Unified CME register event.</td>
</tr>
<tr>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
<tr>
<td>translate-outgoing (voice register pool)</td>
<td>Allows an explicit setting of translation rules on the VoIP dial peer to modify a phone number dialed by any Cisco Unified IP phone user.</td>
</tr>
</tbody>
</table>
show voice register credential

To display configuration information associated with a credential file used for authorization, use the show voice register credential command in privileged EXEC mode.

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

Examples

The following is sample output from this command:

```
Router# show voice register credential
username: Jsmith, password: 1234abc, service: PRESENCE, file index 3
username: Ksample, password: xyz1234, service: PRESENCE, file index 3
username: Mmore, password: updwasc, service: PRESENCE, file index 3
username: Sstove, password: 12bms, service: PRESENCE, file index 3
username: Yjones, password: 357llvrus, service: PRESENCE, file index 5
username: Yjones2, password: 55rrtvu, service: PRESENCE OOD_REFER, file index 5
username: vtemp, password: 1234567, service: PRESENCE, file index 5
```

The table contains descriptions of fields shown in the output, listed in order of appearance.

Table 58: show voice register credential Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>username</td>
<td>Username that is authorized.</td>
</tr>
<tr>
<td>password</td>
<td>Password that is authorized.</td>
</tr>
<tr>
<td>service</td>
<td>Type of service for which the credential file is used; presence or Out-of-dialog REFER (OOD-R).</td>
</tr>
<tr>
<td>file index</td>
<td>Identification number of the credential file defined with the authenticate command.</td>
</tr>
</tbody>
</table>

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>authenticate (voice register global)</td>
<td>Defines the authenticate mode for SIP phones in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>credential load</td>
<td>Reloads a credential file into Flash memory.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>show voice register all</td>
<td>Displays all Cisco Unified CME and Cisco Unified SIP SRST configurations and register information.</td>
</tr>
</tbody>
</table>
show voice register dial-peers

To display details of all dynamically created VoIP dial peers associated with the Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified CallManager Express (Cisco Unified CME) register event, use the **show voice register dial-peers** command in privileged EXEC mode.

```bash
show voice register dial-peers [pool tag]
```

**Syntax Description**

| pool tag | Number of entries in attempted registrations table. Size range from 0 to 50. |

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was added to Cisco CME.</td>
</tr>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1 Cisco Unified SIP SRST 8.1</td>
<td>This command was modified. Pool tag keyword-argument was added. Command output display was also modified to display dial-peers specific to a pool.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to display the dial-peers associated with a pool. To display the dynamic dial-peers associated with a specific pool, use the pool keyword followed by the pool tag. When using the pool keyword, you must specify the pool tag.

**Examples**

**Cisco Unified CME and Cisco Unified SIP SRST**

The following is a sample output from this command displaying all dial-peers:

```bash
Router#show voice register dial-peers
Dial-peers for Pool 1
dial-peer voice 40001 voip
destination-pattern 45111
session target ipv4:8.3.3.111:5060
session protocol sipv2
call-fwd-all 4555
after-hours-exempt FALSE
dial-peer voice 40002 voip
destination-pattern 45113
session target ipv4:8.33.33.111:5060
session protocol sipv2
after-hours-exempt FALSE
Dial-peers for Pool 2
```

Cisco Unified Communications Manager Express Command Reference
Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying all statistical information related to pool 1:

Router# show voice register dial-peers pool 1
Dial-peers for Pool 1:
  dial-peer voice 40004 voip
  destination-pattern 1000
  redirect ip2ip
  session target ipv4:9.13.18.40:19633
  session protocol sipv2
dtmf-relay rtp-nte sip-notify
digit collect kpml
  codec g711ulaw bytes 160
  after-hours-exempt FALSE
  dial-peer voice 40001 voip
  destination-pattern 2000
  redirect ip2ip
  session target ipv4:9.13.18.40:19634
  session protocol sipv2
dtmf-relay rtp-nte sip-notify
digit collect kpml
  codec g711ulaw bytes 160

after-hours-exempt FALSE

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show sip-ua status registrar</td>
<td>Displays all the SIP endpoints currently registered with the contact address.</td>
</tr>
<tr>
<td>show voice register all</td>
<td>Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.</td>
</tr>
<tr>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
</tbody>
</table>
show voice register dialplan

To display all configuration information for a specific SIP dial plan, use the `show voice register dialplan` command in privileged EXEC mode.

```
show voice register dialplan {tag|all}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Tag</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>tag</td>
<td>Number that identifies the SIP dialplan. Range: 1 to 24.</td>
</tr>
<tr>
<td>all</td>
<td>(Optional) Displays all the dialplans defined in a system.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was modified. All keyword was added. Pools and templates that have dialplan configured are also displayed in the output.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to verify the configuration of SIP dial plans. You define a dial plan with the `voice register dialplan` command and assign it to a SIP phone with the `dialplan` command.

In Cisco Unified CME 8.1 and later, `show voice register dialplan` command also displays the pools and templates that have the dialplan configured. The pools which have the dialplan configured by virtue of inclusion of a template is also displayed as part of the pool list display. If a dialplan is configured under both template and pool, the dialplan under the pool takes precedence and the pool is displayed.

When used with the all keyword, the `show voice register dialplan` command displays configuration information for all the dialplans defined in a system.

**Examples**

The following is sample output from this command displaying information for dialplan 1:

```
Router# show voice register dialplan 1
Dialplan Tag 1
Config:
    Type is 7905-7912
    Template 1 has this dialplan configured
    Pool 4 has this dialplan configured
```

The following is a sample output from this command displaying information for all the dialplans configured in a system:

```
Router# show voice register dialplan all
Dialplan Tag 1
Config:
    Type is 7905-7912
```
Pattern 1 is 9879, timeout is 0, user option is phone, button is default
Pattern 24 is 908, timeout is 0, user option is phone, button is default
Dialplan Tag 2
Config:
  Type is 7940-7960-others
Pattern 3 is 9845, timeout is 0, user option is phone, button is default
Pattern 20 is 9098, timeout is 0, user option is phone, button is default

The table contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 59: show voice register dialplan Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Config</td>
<td>List of configuration options defined for this SIP dial plan.</td>
</tr>
<tr>
<td>Dialplan Tag</td>
<td>Tag number of the requested SIP dial plan.</td>
</tr>
<tr>
<td>Pattern</td>
<td>Dial pattern defined for a SIP dial plan with the <code>pattern</code> command in voice register dialplan configuration mode.</td>
</tr>
<tr>
<td>Type</td>
<td>Phone type defined for a SIP dial plan with the <code>type</code> command.</td>
</tr>
</tbody>
</table>

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dialplan</td>
<td>Assigns a dial plan to a SIP phone.</td>
</tr>
<tr>
<td>pattern (voice register dialplan)</td>
<td>Defines a dial pattern for a SIP dial plan.</td>
</tr>
<tr>
<td>show voice register all</td>
<td>Displays all Cisco Unified CME configurations and register information.</td>
</tr>
<tr>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
<tr>
<td>type (voice register dialplan)</td>
<td>Defines a phone type for a SIP dial plan.</td>
</tr>
<tr>
<td>voice register dialplan</td>
<td>Enters voice register dialplan configuration mode to define a dial plan for SIP phones.</td>
</tr>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode for SIP phones.</td>
</tr>
</tbody>
</table>
show voice register dn

To display all configuration information associated with a specific voice register dn, use the `show voice register dn` command in privileged EXEC mode.

```
show voice register dn {tag|all}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Tag</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>tag</td>
<td>Tag number of the voice register dn for which to display information. Range is 1 to 750.</td>
</tr>
<tr>
<td>all</td>
<td>(Optional) Displays configuration information associated with all voice register dns defined in a system.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 and Cisco SIP SRST 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(2)T</td>
<td>Cisco CME 8.1 and Cisco SIP SRST 8.1</td>
<td>This command was modified. The display output now shows pools that have DNs configured under them. All keyword was added to show configuration information for all voice register dns defined in system.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

In Cisco Unified CME 8.1 and Cisco Unified SIP SRST 8.1, the `show voice register dn` command displays the pools that have the DNs configured under them. When used with all keyword, the `show voice register dn` command displays configuration information for all the DNs defined in a system.

**Examples**

**Cisco Unified SIP CME**

The following is a sample output from this command:

```
Router# show voice register dn 1
Dn Tag 1
Config:
  Number is 11
  Preference is 10
  Huntstop is enabled
  Translation-profile incoming saaa
  Allow watch is enabled
  Pool 1 has this DN configured for line 1
```

**Cisco Unified SIP SRST**

The following is a sample output from this command:

```
Router# show voice register dn 2
Dn Tag 1
Config:
```
Number is 11
Preference is 10
Huntstop is enabled
Translation-profile incoming saaa
Allow watch is enabled
Pool 1 has this DN configured for line 1

Cisco Unified SIP SRST

The following is a sample output from this command displaying information for all the dns:

Dn Tag 1
Config:
  Number is 11
  Preference is 10
  Huntstop is enabled
  Translation-profile incoming saaa
  Allow watch is enabled
  Pool 1 has this DN configured for line 1

Dn Tag 2
Config:
  Number is 12
  Preference is 1
  Huntstop is enabled
  Allow watch is enabled
  Pool 2 has this DN configured for line 1, 2

Cisco Unified SIP CME

The following is a sample output from this command displaying information for all the dns:

Router# show voice register dn all
Dn Tag 1
Config:
  Number is 45111
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
Dn Tag 2
Config:
  Number is 45112
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  call-forward b2bua noan 999 timeout 8
  after-hour exempt
  Pool 2 has this DN configured for line 1
  Pool 7 has this DN configured for line 1
Dn Tag 3
Config:
  Number is 45113
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  call-forward b2bua all 87687
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  call-forward b2bua all 87687
Pool 1 has this DN configured for line 1
Pool 3 has this DN configured for line 1, 2

Dn Tag 4

Config:
Auto answer is disabled

Dn Tag 7

Config:
Number is 451110
Preference is 0
Huntstop is disabled
Auto answer is disabled
after-hour exempt

Pool 1 has this DN configured for line 4

Dn Tag 8

Config:
Auto answer is disabled
call-forward b2bua all 678
after-hour exempt

Pool 1 has this DN configured for line 3

The following table contains descriptions of significant fields shown in this output, listed in alphabetical order.

**Table 60: show voice register dn Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Auto answer</td>
<td>Status of auto-answer feature defined with the <strong>auto-answer</strong> command.</td>
</tr>
<tr>
<td>Config</td>
<td>List of configuration options defined for this voice register dn.</td>
</tr>
<tr>
<td>Dn Tag</td>
<td>Tag number of the requested voice register dn.</td>
</tr>
<tr>
<td>Huntstop</td>
<td>Status of huntstop behavior defined with the <strong>huntstop</strong> command.</td>
</tr>
<tr>
<td>Number</td>
<td>Telephone or extension number set with the <strong>number</strong> command in voice register dn configuration mode.</td>
</tr>
<tr>
<td>Preference</td>
<td>Preference order set with the <strong>preference</strong> command in voice register dn configuration mode.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>show voice register pool</strong></td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
<tr>
<td><strong>show voice register dn all</strong></td>
<td>Displays information associated with all the dns configured in a system.</td>
</tr>
<tr>
<td><strong>voice register dn</strong></td>
<td>Enters voice register dn configuration mode to define an extension for a SIP phone line.</td>
</tr>
</tbody>
</table>
show voice register global

To display all global configuration parameters associated with Cisco Unified SIP IP phones, use the `show voice register global` command in privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco SIP SRST 8.0</td>
<td>This command was modified to display the signaling transport protocol.</td>
</tr>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1 Cisco Unified SIP SRST 8.1</td>
<td>This command was modified to include global statistics in the output display.</td>
</tr>
<tr>
<td>15.2(2)T</td>
<td>Cisco Unified CME 9.0</td>
<td>This command was modified to include conference hardware in the output display.</td>
</tr>
</tbody>
</table>

**Examples**

**Cisco Unified CME**

The following is a sample output from the `show voice register global` command used in Cisco Unified CME:

```
Router# show voice register global
CONFIG [Version=8.1]
-------------------------------
Version 8.1
Mode is cme
Max-pool is 10
Max-dn is 10
Outbound-proxy is enabled and will use global configured value
Security Policy: DEVICE-DEFAULT
Source-address is 8.3.3.5 port 5060
Time-format is 12
Date-format is M/D/Y
Time-zone is 5
Hold-alert is disabled
Mwi stutter is disabled
Mwi registration for full E.164 is disabled
Forwarding local is enabled
Privacy is enabled
Privacy-on-hold is disabled
Dst auto adjust is enabled
start at Apr week 1 day Sun time 02:00
stop at Oct week 8 day Sun time 02:00
Max redirect number is 5
IP QoS DSCP:
```
ef (the MS 6 bits, 46, in ToS, 0xB8) for media
cs3 (the MS 6 bits, 24, in ToS, 0x60) for signal
af41 (the MS 6 bits, 34, in ToS, 0x88) for video
default (the MS 6 bits, 0, in ToS, 0x0) for service
Telnet Level: 0
Tftp path is flash:
Generate text file is disabled
Tftp files are created, current syncinfo 0001140473454008
OS79XX.TXT is not created
timeout interdigit 10
network-locale[0] US (This is the default network locale for this box)
network-locale[1] US
user-locale[0] US (This is the default user locale for this box)
user-locale[1] US
user-locale[2] US
user-locale[3] US
user-locale[4] US  Active registrations : 0
Total SIP phones registered: 0
Total Registration Statistics
  Registration requests : 0
  Registration success : 0
  Registration failed : 0
  unRegister requests : 0
  unRegister success : 0
  unRegister failed : 0
  Attempts to register after last unregister : 0
  Last register request time :
  Last unregister request time :
  Register success time :
  Unregister success time :

The following is a sample output from the show voice register global command. The output shows that hardware conferencing is enabled.

Router# show voice register global
CONFIG [Version=8.7]
------------------------
  Version 8.7
  Mode is cme
  Max-pool is 50
  Max-dn is 100
  Outbound-proxy is enabled and will use global configured value
  Security Policy: DEVICE-DEFAULT
  Forced Authorization Code Refer is enabled
  Source-address is 1.5.40.20 port 5060
  Time-format is 12
  Date-format is M/D/Y
  Time-zone is 5
  Hold-alert is disabled
  Mwi stutter is disabled
  Mwi registration for full E.164 is disabled
  Forwarding local is enabled
  Video is enabled
  Camera is enabled
  Privacy is enabled
  Privacy-on-hold is disabled
  Conference hardware is enabled
  Dst auto adjust is enabled
The following is a sample output from the \texttt{show voice register global} command used in Cisco Unified SIP SRST:

```
Router# show voice register global
CONFIG [Version=8.1]
-------------
Version 8.1
Mode is srst
Max-pool is 10
Max-dn is 10
Outbound-proxy is enabled and will use global configured value
Security Policy: DEVICE-DEFAULT
timeout interdigit 10
network-locale[0] US (This is the default network locale for this box)
network-locale[1] US
user-locale[0] US (This is the default user locale for this box)
user-locale[1] US
user-locale[2] US
user-locale[3] US
user-locale[4] US Active registrations : 0
Total SIP phones registered: 0
Total Registration Statistics
Registration requests : 0
Registration success : 0
Registration failed : 0
unRegister requests : 0
unRegister success : 0
unRegister failed : 0
Attempts to register
after last unregister : 0
Last register request time :
Last unregister request time :
Register success time :
Unregister success time :
```

The following table contains descriptions of significant fields shown in this output, listed in alphabetical order.

\textbf{Table 61: show voice register global Field Descriptions}

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Conference hardware</td>
<td>Shows whether the Cisco Unified SIP IP phone will perform local mixing on its own or request Cisco Unified CME to perform hardware conferencing using its DSP resource.</td>
</tr>
<tr>
<td>Date-format</td>
<td>Value of \texttt{date-format} command.</td>
</tr>
<tr>
<td>DST auto adjust</td>
<td>Setting of \texttt{dst auto-adjust} command.</td>
</tr>
<tr>
<td>Forwarding local</td>
<td>Setting of \texttt{forwarding local} command.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>----------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Generate text file</td>
<td>Setting of <strong>text file</strong> command.</td>
</tr>
<tr>
<td>Hold-alert</td>
<td>Setting of <strong>hold-alert</strong> command.</td>
</tr>
<tr>
<td>Load</td>
<td>Value of <strong>load</strong> command.</td>
</tr>
<tr>
<td>Max-dn</td>
<td>Reports the maximum number of SIP voice register directory numbers (DNs) supported by the Cisco Unified SIP CME or Cisco Unified SIP SRST router as configured with the <strong>max-dn</strong> command. The maximum possible number is platform-dependent.</td>
</tr>
<tr>
<td>Max-pool</td>
<td>Reports the maximum number of SIP voice register pools supported by the Cisco Unified SIP SRST or Cisco Unified CME router as configured with the <strong>max-pool</strong> command. The maximum possible number is platform-dependent.</td>
</tr>
<tr>
<td>Max redirect number</td>
<td>Maximum number of redirects set with the <strong>max-redirect</strong> command.</td>
</tr>
<tr>
<td>Mode</td>
<td>Reports the mode as configured with the <strong>mode</strong> command. Value can be either Cisco Unified CME or Cisco Unified SIP SRST.</td>
</tr>
<tr>
<td>MWI registration</td>
<td>Setting of <strong>mwi</strong> command.</td>
</tr>
<tr>
<td>MWI stutter</td>
<td>Setting of <strong>mwi stutter</strong> command.</td>
</tr>
<tr>
<td>Time-format</td>
<td>Value of <strong>time-format</strong> command.</td>
</tr>
<tr>
<td>Time-zone</td>
<td>Number of the timezone selected with the <strong>timezone</strong> command.</td>
</tr>
<tr>
<td>TFTP path</td>
<td>Directory location of provisioning files for Cisco Unified SIP IP phones that is specified with the <strong>tftp-path</strong> command.</td>
</tr>
<tr>
<td>Version</td>
<td>Reports the Cisco Unified SIP SRST or Cisco Unified CME version number.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>show sip-ua status registrar</strong></td>
<td>Displays all the Cisco Unified SIP IP phones currently registered with the contact address.</td>
</tr>
<tr>
<td><strong>show voice register all</strong></td>
<td>Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.</td>
</tr>
<tr>
<td><strong>show voice register dial-peers</strong></td>
<td>Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified SIP SRST or Cisco Unified CME register event.</td>
</tr>
<tr>
<td><strong>voice register global</strong></td>
<td>Enters voice register global configuration mode to set global parameters for all supported Cisco Unified SIP IP phones in a Cisco Unified CME or Cisco Unified SIP SRST environment.</td>
</tr>
</tbody>
</table>
**show voice register hfs**

To display the HTTP File-Fetch Server (HFS) file bindings of firmware files accessible to Cisco Unified SIP IP phones, use the `show voice register hfs` command in privileged EXEC mode.

### Syntax Description
This command has no arguments or keywords.

### Command Default
None

### Command Modes
Privileged EXEC

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(1)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines
Use the `show voice register hfs` command with Cisco Unified CME 8.8 or a later version. This command displays the bindings of firmware files that are accessible to Cisco Unified SIP IP phones using the HFS download service.

### Examples

```
Router(config)# show voice register hfs
Fetch Service Enabled = Y
  App enabled port = 6970
  Use default port = N
  Registered session-id = 19

Default home path = flash:/
  Ongoing fetches from home = 0

HTTP File Server Bindings
  No. of bindings = 11
  No. of url table entries = 9
  No. of alias table entries = 9
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>create profile (voice register global)</td>
<td>Generates the configuration profile files required for SIP phones.</td>
</tr>
<tr>
<td>hfs enable</td>
<td>Enables the HFS download service on an IP Phone in a Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
show voice register pool

To display all configuration information associated with a specific voice register pool, use the `show voice register pool` command in privileged EXEC mode.

`show voice register pool {pool-tag|all} [brief]`

**Syntax Description**

<table>
<thead>
<tr>
<th>pool-tag</th>
<th>Tag number of the voice register pool for which information is displayed. Range is 1 to 262.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Note</td>
<td>The maximum number of pools is version and platform dependent.</td>
</tr>
<tr>
<td>all</td>
<td>Displays the information of all the voice register pools.</td>
</tr>
<tr>
<td>brief</td>
<td>(Optional) Displays brief information of all voice register pools.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco SIP SRST</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco SIP SRST</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was added to Cisco CME.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SIP SRST 4.2(1)</td>
<td>This command was modified to include emergency response location information in the output display.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was modified to include logical partitioning class of restriction (LPCOR) information in the output display.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was modified. The all and brief keywords were added. Voice-class stun-usage information is displayed in the output.</td>
</tr>
<tr>
<td>15.2(2)T</td>
<td>Cisco Unified CME 9.0</td>
<td>This command was modified to include conference admin, conference add mode, and conference drop mode in the output display.</td>
</tr>
<tr>
<td>15.2(4)M</td>
<td>Cisco Unified CME 9.1 Cisco Unified SIP SRST 9.1</td>
<td>This command was modified to include Key Expansion Module (KEM) data in the output display.</td>
</tr>
</tbody>
</table>
### Examples

**Cisco Unified CME**

The following is a sample output of the `show voice register pool` command, displaying information for voice register pool 33 in Cisco Unified CME:

```
Router# show voice register pool 33
Pool Tag 33
Config:
  Mac address is 0009.B7F7.532E
  Type is 7960
  Number list 1 : DN 1
  Number list 2 : DN 2
  Number list 3 : DN 3
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  keep-conference is enabled
  template is 1
  Emergency response location 3
  Lpcor Type is local
  Lpcor Incoming is sip_group
  Lpcor Outgoing is sip_group

  Transport type is udp
  service-control mechanism is not supported
  Privacy feature is not configured.
  Privacy button is disabled

Dialpeers created:

Statistics:
  Active registrations : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests : 0
    Registration success : 0
    Registration failed : 0
    unRegister requests : 0
    unRegister success : 0
    unRegister failed : 0
```

The following is a sample output of the `show voice register pool` command. The output shows that a meet-me hardware conference administrator has been assigned, the conference creator or any of the participants can add a new participant, and the conference creator can terminate the active video hardware conference by hanging up.

```
Router# show voice register pool 15
Pool Tag 15
```
Config:
- Mac address is 1C17.D340.81F0
- Type is 9951
- Number list 1: DN 15
- Proxy IP address is 0.0.0.0
- Current Phone load version is Cisco-CP9951/9.0.1
- DTMF Relay is enabled, sip-notify
- Call Waiting is enabled
- DND is disabled
- Video is enabled
- Camera is enabled
- Busy trigger per button value is 0
- feature-button 5 DnD
- feature-button 6 MeetMe
- keep-conference is enabled
- registration expires timer max is 86400 and min is 60
- template is 1
- kpml signal is enabled
- Lpcor Type is none
- Transport type is udp
- service-control mechanism is supported
- registration Call ID is 1c17d340-81f00002-6c48fe8e-03013c10@1.5.40.105
- Registration method: per line
- Privacy feature is not configured.
- Privacy button is disabled
- active primary line is: 3915
- contact IP address: 1.5.40.105 port 5060
- Phone SIS Version: 5.0.0
- GW SIS Version: 1.0.0
- conference admin: yes
- conference add mode: all
- conference drop mode: creator
- paging-dn: config 0 [multicast] effective 0 [multicast]

The following is an example of a partial output of the `show voice register pool all` command, showing KEM data with the phone type information:

```
Router# show voice register pool all
Pool Tag 5
Config:
- Mac address is B4A4.E328.4698
- Type is 9971 addon 1 CKEM
- Number list 1: DN 2
- Number list 2: DN 3
- Proxy IP address is 0.0.0.0
- DTMF Relay is disabled
- Call Waiting is enabled
- DND is disabled
- Video is enabled
- Camera is enabled
- Busy trigger per button value is 0
- keep-conference is enabled
- registration expires timer max is 200 and min is 60
- kpml signal is enabled
- Lpcor Type is none
```

The following is a sample output of the `show voice register pool all` command, showing the three KEMs configured with phone type 9971:

```
Router# show voice register pool all
Pool Tag 4
Config:
```
Cisco Unified SIP SRST

The following is a sample output of the `show voice register pool` command, displaying all information for voice register pool 1 in Cisco Unified SIP SRST:

```
Router# show voice register pool 1

Pool Tag 1
Config:
Network address is 192.168.0.0, Mask is 255.255.0.0
Number list 1 : Pattern is 50.., Preference is 2
Proxy Ip address is 0.0.0.0
Default preference is 2
Incoming called number is
Translate outgoing called tag is 1
Class of Restriction List Tag: default
Incoming corlist name is allowall
Application is default.new

Dialpeers created:

dial-peer voice 40007 voip
   application default.new
   corlist incoming allowall
   preference 2
   incoming called-number 5001
   destination-pattern 5001
   redirect ip2ip
   session target ipv4:192.168.0.3
   session protocol sipv2
   translate-outgoing called 1
   voice-class codec 1

Statistics:
Active registrations : 2

Total Registration Statistics
Registration requests : 48
Registration success : 48
Registration failed : 0
unRegister requests : 46
unRegister success : 46
unRegister failed : 0

Emergency response location 6
```

The following is a sample output of the `show voice register pool brief` command, showing an IPv6 source address configured on a Cisco SIP IP Phone:

```
Router# show voice register pool brief

Pool ID IP Address Ln DN Number State
==== ======================== ========================== == === ========== ============
```

Cisco Unified Communications Manager Express Command Reference
voice class stun usage

The following is a sample output of the `show voice register pool` command, displaying voice-class stun-usage information for voice register pool 51:

Router# `show voice register pool 51`
Pool Tag 51
Config:
  Mac address is 0011.209F.5D60
  Type is 7960
  Number list 1 : DN 51
  Proxy Ip address is 0.0.0.0
  Current Phone load version is Cisco-SIPGateway/IOS-12.x
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  keep-conference is enabled
  template is 10
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is not supported
  registration Call ID is 2BA38EE3-17D311DB-800BCD81-A9AD11F0
  Privacy feature is not configured.
  Privacy button is disabled
  active primary line is: 16263646
  contact IP address: 192.168.0.87 port 5060
  Reason for unregistered state: No registration request since last reboot/unregister
  voice-class stun-usage is enabled. tag is 1
Dialpeers created:
Dial-peers for Pool 51:
Statistics:
  Active registrations : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests : 2
    Registration success : 2
    Registration failed : 0
    unRegister requests : 2
    unRegister success : 2
    unRegister failed : 0
  Attempts to register after last unregister : 0
  Last register request time : 13:43:27.839 IST Tue Apr 20 2010

The following table contains descriptions of significant fields shown in the Cisco Unified CME and Cisco Unified SIP SRST output, listed in alphabetical order.

Table 62: `show voice register pool` Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Active registrations</td>
<td>Shows the current active registrations.</td>
</tr>
<tr>
<td>Application</td>
<td>Shows the <code>application</code> command configuration for this pool.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------------------</td>
<td>------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Call Waiting</td>
<td>Shows the <code>call-waiting</code> command configuration.</td>
</tr>
<tr>
<td>Class of Restriction List Tag</td>
<td>Shows the COR tag.</td>
</tr>
<tr>
<td>Conference add mode</td>
<td>Shows the current setting of the hardware conference privilege for adding participants.</td>
</tr>
<tr>
<td>Conference admin</td>
<td>Shows whether the Cisco Unified SIP IP phone is assigned as the hardware conference administrator or not.</td>
</tr>
<tr>
<td>Conference drop mode</td>
<td>Shows who can terminate an active ad-hoc hardware conference by hanging up.</td>
</tr>
<tr>
<td>Config</td>
<td>Shows the voice register pool.</td>
</tr>
<tr>
<td>Default preference</td>
<td>Shows the default preference value of this pool.</td>
</tr>
<tr>
<td>Dialpeers created</td>
<td>Lists all the dial peers created and their contents. Dial-peer contents differ for each application and are not described here.</td>
</tr>
<tr>
<td>DnD</td>
<td>Shows the setting of the <code>dnd-control</code> command.</td>
</tr>
<tr>
<td>DTMF Relay</td>
<td>Shows the setting of the <code>dtmf-relay</code> command.</td>
</tr>
<tr>
<td>Emergency response location</td>
<td>Shows the ephone’s emergency response location to which an emergency response team is dispatched when an emergency call is made.</td>
</tr>
<tr>
<td>Incoming called number</td>
<td>Shows the <code>incoming called-number</code> command configuration.</td>
</tr>
<tr>
<td>Incoming corlist name</td>
<td>Shows the <code>cor</code> command configuration.</td>
</tr>
<tr>
<td>keep-conference</td>
<td>Shows the status of the <code>keep-conference</code> command.</td>
</tr>
<tr>
<td>Lpcor Incoming</td>
<td>Shows the setting of the <code>lpkor incoming</code> command.</td>
</tr>
<tr>
<td>Lpcor Outgoing</td>
<td>Shows the setting of the <code>lpkor outgoing</code> command.</td>
</tr>
<tr>
<td>Lpcor Type</td>
<td>Shows the setting of the <code>lpkor type</code> command.</td>
</tr>
<tr>
<td>Mac address</td>
<td>Shows the MAC address of the Cisco Unified SIP IP phone as defined by the <code>id</code> command.</td>
</tr>
<tr>
<td>Network address and Mask</td>
<td>Shows network address and mask information when the <code>id</code> command is configured.</td>
</tr>
<tr>
<td>Number list, Pattern, and Preference</td>
<td>Shows the <code>number</code> command configuration.</td>
</tr>
<tr>
<td>Pool Tag</td>
<td>Shows the assigned tag number of the current pool.</td>
</tr>
<tr>
<td>Proxy IP address</td>
<td>Shows the <code>proxy</code> command configuration; that is, the IP address of the external SIP server.</td>
</tr>
<tr>
<td>Registration failed</td>
<td>Shows the failed registrations.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------------</td>
<td>------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Registration requests</td>
<td>Shows the incoming registration requests.</td>
</tr>
<tr>
<td>Registration success</td>
<td>Shows the successful registrations.</td>
</tr>
<tr>
<td>Statistics</td>
<td>Shows the registration statistics for this pool.</td>
</tr>
<tr>
<td>Template</td>
<td>Shows the template-tag number for the template applied to the Cisco Unified SIP IP phone.</td>
</tr>
<tr>
<td>Total Registration Statistics</td>
<td>Shows the total registration statistics for this pool.</td>
</tr>
<tr>
<td>Translate outgoing called tag</td>
<td>Shows the <code>translate-outgoing</code> command configuration.</td>
</tr>
<tr>
<td>Type</td>
<td>Shows the phone type identified for the Cisco Unified SIP IP phone using the <code>type</code> command.</td>
</tr>
<tr>
<td>unRegister failed</td>
<td>Reports the number of failed unregisters.</td>
</tr>
<tr>
<td>unRegister requests</td>
<td>Shows the incoming unregister/registration expiry requests.</td>
</tr>
<tr>
<td>unRegister success</td>
<td>Reports the number of successful unregisters.</td>
</tr>
<tr>
<td>Username Password</td>
<td>Shows the values within the authentication credential.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>application (voice register pool)</strong></td>
<td>Selects the session-level application for the dial peer associated with an individual Cisco Unified SIP IP phone in a Cisco Unified CME environment or for a group of phones in a Cisco Unified SIP SRST environment.</td>
</tr>
<tr>
<td><strong>call-waiting (voice register pool)</strong></td>
<td>Enables the call-waiting option on a SIP phone.</td>
</tr>
<tr>
<td><strong>cor (voice register pool)</strong></td>
<td>Configures a class of restriction on the VoIP dial peers associated with directory numbers.</td>
</tr>
<tr>
<td><strong>dnd-control (voice register template)</strong></td>
<td>Enables the Do-Not-Disturb (DND) soft key on SIP phones.</td>
</tr>
<tr>
<td><strong>dtmf-relay (voice register pool)</strong></td>
<td>Specifies the list of dual-tone multifrequency (DTMF) relay methods that can be used to relay DTMF audio tones between SIP endpoints.</td>
</tr>
<tr>
<td><strong>id (voice register pool)</strong></td>
<td>Explicitly identifies a locally available, individual Cisco Unified SIP IP phone or, when running Cisco Unified SIP SRST, a set of Cisco Unified SIP IP phones.</td>
</tr>
<tr>
<td><strong>incoming called-number (dial peer)</strong></td>
<td>Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer.</td>
</tr>
<tr>
<td><strong>keep-conference (voice register pool)</strong></td>
<td>Allows IP phone conference initiators to exit from conference calls and keep the remaining parties connected.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>lpcor incoming</td>
<td>Associates an incoming call with a logical partitioning class of restriction (LPCOR) resource-group policy.</td>
</tr>
<tr>
<td>lpcor outgoing</td>
<td>Associates an outgoing call with an LPCOR resource-group policy.</td>
</tr>
<tr>
<td>lpcor type</td>
<td>Specifies the LPCOR type for an IP phone.</td>
</tr>
<tr>
<td>number (voice register pool)</td>
<td>Indicates the E.164 phone numbers that the registrar permits to handle the Register message from a Cisco Unified SIP IP phone.</td>
</tr>
<tr>
<td>proxy (voice register pool)</td>
<td>Autogenerates additional VoIP dial peers to reach the main proxy whenever a Cisco Unified SIP IP phone registers with a Cisco Unified SIP SRST gateway.</td>
</tr>
<tr>
<td>show sip-ua status registrar</td>
<td>Displays all the Cisco Unified SIP IP phones registered with the contact address.</td>
</tr>
<tr>
<td>show voice register all</td>
<td>Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.</td>
</tr>
<tr>
<td>show voice register dial-peer</td>
<td>Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified CME or Cisco Unified SIP SRST register event.</td>
</tr>
<tr>
<td>translate-outgoing (voice register pool)</td>
<td>Allows an explicit setting of translation rules on the VoIP dial peer to modify a phone number dialed by any Cisco Unified IP phone user.</td>
</tr>
<tr>
<td>type (voice register pool)</td>
<td>Defines a phone type for a SIP phone.</td>
</tr>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode for Cisco Unified SIP IP phones.</td>
</tr>
</tbody>
</table>
show voice register pool after-hour-exempt

To display the details of a phone that has after-hour-exempt enabled on it, use the **show voice register after-hour-exempt** command in privileged EXEC mode.

### Syntax Description

This command has no arguments or keywords.

### Command Modes

Privileged EXEC

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1 Cisco Unified SRST 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use this command to display the details of a phone that has after-hour-exempt enabled. Individual phones can be exempted from call blocking using the after-hour exempt.

### Cisco Unified CME

The following is a sample output from this command displaying information for phones with after after-hour-exempt:

```
Router# show voice register pool after-hour-exempt
Pool ID  IP Address   Ln DN Number  State
1 001B.535C.D410  8.3.3.111  3 8 451110 UNREGISTERED
2 0015.C68E.6D13  1 2 45112 UNREGISTERED
3 0021.5553.8998  1 3 45113 UNREGISTERED
7 0018.BAC8.D2B1  1 2 45112 UNREGISTERED
```

### Cisco Unified SRST

The following is a sample output from this command displaying information for phones with after after-hour-exempt:

```
Router# show voice register pool after-hour-exempt
Pool ID  IP Address   Ln DN Number  State
1 9.13.18.40  9.13.18.40  1 1 1000 REGISTERED
2 9.13.18.40  2 2 2000 REGISTERED
3 9.13.18.40  3 3 3000 REGISTERED
4 9.13.18.40  4 4 4000 REGISTERED
5 9.13.18.40  5 5 5000 UNREGISTERED
6 9.13.18.40  6 6 6000 UNREGISTERED
7 9.13.18.40  7 7 7000 UNREGISTERED
```

The table contains descriptions of significant fields shown in this output, listed in alphabetical order.
Table 63: show voice register pool after-hour exempt field descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DN</td>
<td>Directory number of the phone.</td>
</tr>
<tr>
<td>IP Address/port</td>
<td>IP address and port number of the phones.</td>
</tr>
<tr>
<td>LN</td>
<td>Line number of the phone.</td>
</tr>
<tr>
<td>Number</td>
<td>Number of the phones that have after-hour exempt enabled.</td>
</tr>
<tr>
<td>Pool</td>
<td>Shows the current pool.</td>
</tr>
<tr>
<td>State</td>
<td>Registration state.</td>
</tr>
</tbody>
</table>

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>after-hour exempt(voice register pool)</td>
<td>Specifies that an IP phone does not have any of its outgoing calls blocked although call blocking is defined.</td>
</tr>
<tr>
<td>show voice register all</td>
<td>Displays all Cisco SIP SRST and Cisco CME configurations and register information.</td>
</tr>
<tr>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode for SIP phones.</td>
</tr>
</tbody>
</table>
show voice register pool attempted-registrations

To display the details of phones that attempt to register with Cisco Unified CME or Cisco Unified SRST and fail, use the **show voice register pool attempted-registrations** command in privileged EXEC mode.

**show voice register pool attempted-registrations**

**Syntax Description**
This command has no arguments or keywords.

**Command Modes**
Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>Cisco Unified SRST 8.1</td>
</tr>
<tr>
<td></td>
<td></td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to display the details of the phones that attempt to register with Cisco Unified CME or Cisco Unified SRST and fail. If the phone registers successfully after some time, the attempted registration entry will still show up in the attempted-registration table. Use the clear voice register attempted-registrations command to remove the entry from the attempted registration table.

**Cisco Unified CME and Cisco Unified SRST**

The following is a sample output from this command displaying information for show voice register pool attempted-registrations:

```
Router# show voice register pool attempted-registrations
Phones that have attempted registrations and have failed:
  MAC address: 001b.535c.d410
  IP address : 8.3.3.111
  Attempts : 5
  Time of latest attempt: *10:50:00.886 UTC Wed Oct 14 2009
  Reason for failure : No pool match for the registration request
  MAC address: 0015.c68e.6d13
  IP address : 8.33.33.112
  Attempts : 4
  Time of latest attempt: *10:50:00.434 UTC Wed Oct 14 2009
  Reason for failure : No pool match for the registration request
  MAC address: 0009.43E9.0B35
  IP address : 9.13.40.83
  Attempts : 1
  Reason for failure : No pool match for the registration request

The following is a sample output from this command displaying information for show voice register pool attempted-registrations when none of the phones fail:

Router# show voice register pool attempted-registrations
Phones that have attempted registrations and have failed: NONE
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>attempted-registrations size</td>
<td>Allows to set the size of the table that stores information related to SIP phones that attempt to register and fail.</td>
</tr>
<tr>
<td></td>
<td>clear voice register attempted-registrations</td>
<td>Clears entries from the attempted-registration table.</td>
</tr>
</tbody>
</table>
show voice register pool cfa

To display the voice register pool details of a phone that has Call Forward All (CFA) enabled, use the `show voice register pool cfa` command in privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to display the voice register pool details of the phone with CFA feature enabled. When Call Forward All feature is enabled on Cisco Unified SIP IP phones such as 7940, 7941, 7941GE, 7942, 7945, 7960, 7961, 7961GE, 7962, 7965, 7970, 7971, 7975 through the CFA phone button. The `show voice register pool cfa` command displays only the call forward all B2BUA details.

The `show voice register pool cfa` command also displays the line number and DN number if available under the pool configuration. If call-forward-all is configured under both pool and DN, the configuration under DN takes precedence.

**Cisco Unified CME and Cisco Unified SRST**

The following is a sample output from this command displaying all statistical information:

```
Router# show voice register pool cfa
Pool Ln DN Number Call Forward All Number
---- -- ------ ------- -----------------------
 1 2 8 678                       
 0 1 45111 4555                  
 4 7 451110 4555                
 3 1 3 45113 87687              
 2 3 45113 87687
```

The table contains descriptions of significant fields shown in this output, listed in alphabetical order.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Forward All Number</td>
<td>Number to which the calls are forwarded.</td>
</tr>
<tr>
<td>DN</td>
<td>Voice register DN tag of the line.</td>
</tr>
<tr>
<td>LN</td>
<td>Line number of the telephone number.</td>
</tr>
<tr>
<td>Pool</td>
<td>Tag ID of the pool.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call forward b2bua all</td>
<td>Enables call forward all.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>show voice register all</td>
<td>Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.</td>
</tr>
<tr>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
<tr>
<td>show voice register pool detail all</td>
<td>Displays the details of all the pools defined in the system.</td>
</tr>
</tbody>
</table>
show voice register pool connected

To display the details of SIP phones that are in connected state, use the `show voice register pool connected` command in privileged EXEC mode.

```
show voice register pool connected [brief]
```

**Syntax Description**

| brief (Optional) Displays brief details of SIP phones that are in connected state. |

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1 Cisco Unified SRST 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to display the details of the phone that are currently in connected state (in conversation). The output for show voice register pool connected command shows details of both calls originating from the SIP phones and calls made towards SIP phones. When used with brief keyword, the show voice register pool connected command displays a brief detail of phones in connected state.

**Cisco Unified CME and Cisco Unified SRST**

The following is sample output from this command displaying all statistical information:

```
Router# show voice register pool connected
Outbound calls from SIP line phones:
Pool tag: 1
-------------
MAC Address : 001B.535C.D410
Contact IP : 8.3.3.111
Phone Number : 45111
Remote Number : 45112
Call 2
SIP Call ID : 001b535c-d4100010-79612b5a-336b0db5@8.3.3.111
State of the call : STATE_ACTIVE (7)
Substate of the call : SUBSTATE_NONE (0)
Calling Number : 45111
Called Number : 45112
Bit Flags : 0xC0401C 0x100 0x4
CC Call ID : 7
Source IP Address (Sig ) : 8.3.3.5
Destn SIP Req Addr:Port : [8.3.3.111]:5060
Destn SIP Resp Addr:Port: [8.3.3.111]:50076
Destination Name : 8.3.3.111
Number of Media Streams : 1
RTP Fork Object : 0x0
Media Mode : flow-through
Media Stream 1
State of the stream : STREAM_ACTIVE
Stream Call ID : 7
Stream Type : voice-only (0)
Stream Media Addr Type : 1
Negotiated Codec : g729r8 (20 bytes)
Codec Payload Type : 18
```
Negotiated Dtmf-relay : inband-voice
Dtmf-relay Payload Type : 0
QoS ID : -1
Local QoS Strength : BestEffort
Negotiated QoS Strength : BestEffort
Negotiated QoS Direction : None
Local QoS Status : None
Media Source IP Addr:Port: [8.3.3.5]:17580
Media Dest IP Addr:Port : [8.3.3.111]:26298
Options-Ping ENABLED:NO ACTIVE:NO
Inbound calls to SIP line phones:

Pool tag: 2

MAC Address : 0015.C68E.6D13
Contact IP : 8.33.33.112
Phone Number : 45112
Remote Number : 45111

Call 3
SIP Call ID : 4DA52F97-ADA311DE-8019803A-FF3E4CBC@8.3.3.5
State of the call : STATE_ACTIVE (7)
Substate of the call : SUBSTATE_NONE (0)
Calling Number : 45111
Called Number : 45112
Bit Flags : 0xC04018 0x100 0x80
CC Call ID : 8
Source IP Address (Sig ) : 8.3.3.5
Destn SIP Req Addr:Port : [8.33.33.112]:5060
Destn SIP Resp Addr:Port: [8.33.33.112]:5060
Destination Name : 8.33.33.112
Number of Media Streams : 1
Number of Active Streams: 1
RTP Fork Object : 0x0
Media Mode : flow-through
Media Stream 1
State of the stream : STREAM_ACTIVE
Stream Call ID : 8
Stream Type : voice-only (0)
Stream Media Addr Type : 1
Negotiated Codec : g729r8 (20 bytes)
Codec Payload Type : 18
Negotiated Dtmf-relay : inband-voice
Dtmf-relay Payload Type : 0
QoS ID : -1
Local QoS Strength : BestEffort
Negotiated QoS Strength : BestEffort
Negotiated QoS Direction : None
Local QoS Status : None
Media Source IPAddr:Port: [8.3.3.5]:16384
Media Dest IP Addr:Port : [8.3.3.112]:30040

The following is sample output from this command displaying brief statistical information:

Router# show voice register pool connected brief
Pool IP Address Number Remote Number
------------- -------------- -------------- 
1 8.3.3.111 45111 45112
Inbound calls to SIP line phones:
Pool IP Address Number Remote Number
------------- -------------- -------------- 
2 8.33.33.112 45111 45111
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>show sip-ua calls</strong></td>
<td>Displays active user agent client (UAC) and user agent server (UAS) information on SIP calls</td>
</tr>
<tr>
<td><strong>show voice register all</strong></td>
<td>Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.</td>
</tr>
<tr>
<td><strong>show voice register pool</strong></td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
</tbody>
</table>
show voice register pool ip

To display the details of a SIP phone with a specific IP address, use the **show voice register pool ip** command in privileged EXEC mode.

```
show voice register pool ip ip-address
```

**Syntax Description**
- `ip-address` IPv4 address of the SIP phone.

**Command Modes**
- Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1 Cisco Unified SRST 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to display the details of a phone with a specific IP-address. When the pool ID is configured as a mac address or an IP address the registered pools contain the IP address information. The pool information is displayed if the IP addresses match.

When the pool ID is IP and the pool is unregistered, IP address configured under pool is compared with the input IP. When the pool ID is network contact, the IP address of each phone that is registered is compared with the input IP address.

**Cisco Unified CME and Cisco Unified SRST**

The following is sample output from this command displaying all statistical information:

```
Router# show voice register pool ip 8.3.3.111
Pool ID IP Address Ln DN Number State
==== =============== =============== == === ==================== ============
1 001B.535C.D410 8.3.3.111 1 1 45111 REGISTERED
4 7 451110 UNREGISTERED
```

The table contains descriptions of significant fields shown in this output, listed in alphabetical order.

**Table 64: show voice register pool ip field descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DN</td>
<td>Voice register DN tag of the line.</td>
</tr>
<tr>
<td>ID</td>
<td>Phone identification (ID) address.</td>
</tr>
<tr>
<td>IP Address</td>
<td>IP address of the SIP phone.</td>
</tr>
<tr>
<td>LN</td>
<td>Line number of the telephone number.</td>
</tr>
<tr>
<td>Number</td>
<td>Number of the phones that have a mac address.</td>
</tr>
<tr>
<td>Pool</td>
<td>Tag ID of the pool.</td>
</tr>
</tbody>
</table>
### Field Description

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>State</td>
<td>Registration state of the line.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>show voice register all</code></td>
<td>Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.</td>
</tr>
<tr>
<td><code>show voice register pool</code></td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
</tbody>
</table>
**show voice register pool mac**

To display the details of voice register pool associated with a specific phone type, use the `show voice register pool mac` command in privileged EXEC mode.

```
show voice register pool mac H.H.H
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>H.H.H</strong></td>
<td>MAC address of the SIP phone attempting to register.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1 Cisco Unified SRST 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to display the details of the phone with the mac address H.H.H. The command displays only the pools that are configured with an ID as mac.

**Cisco Unified CME and Cisco Unified SRST**

The following is sample output from this command displaying all statistical information:

```
Router# show voice register pool mac 001B.535C.D410
Pool ID | IP Address | Ln | DN | Number | State         
--------|------------|----|----|--------|---------------|
        | 001B.535C.D410 | 8.3.3.111 | 45111 | 1 45111 | REGISTERED   
        |             | 4    | 451110 | 4 7 | UNREGISTERED |
```

The table contains descriptions of significant fields shown in this output, listed in alphabetical order.

**Table 65: show voice register pool mac field descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DN</td>
<td>Voice register DN tag of the line.</td>
</tr>
<tr>
<td>ID</td>
<td>Phone identification (ID) address.</td>
</tr>
<tr>
<td>IP Address</td>
<td>IP address of the SIP phone.</td>
</tr>
<tr>
<td>LN</td>
<td>Line number of the telephone number.</td>
</tr>
<tr>
<td>Number</td>
<td>Number of the phones that have a mac address.</td>
</tr>
<tr>
<td>Pool</td>
<td>Tag ID of the pool.</td>
</tr>
<tr>
<td>State</td>
<td>Registration state of the line.</td>
</tr>
</tbody>
</table>
## Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>show voice register all</code></td>
<td>Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.</td>
</tr>
<tr>
<td><code>show voice register pool</code></td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
</tbody>
</table>
show voice register pool on-hold

To display the details of phones that are currently on-hold, use the **show voice register pool on-hold** command in privileged EXEC mode.

**show voice register pool on-hold [brief]**

**Syntax Description**

| brief | (Optional) Displays brief details of SIP phones that are currently on-hold. |

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1 Cisco Unified SRST 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to display the details of the phone that are currently on-hold. The show voice register pool on-hold command output also displays a field to show if the hold was a locally initiated hold (initiated on the phone) or if the hold was initiated on the remote end. When used with brief keyword, the show voice register pool on-hold command displays a brief information of the phones that are currently put on hold by the remote caller or have put the remote caller on hold. The “Hold-Origin” field specifies the type of the hold, which can be either remote or local. Local indicates that the call is placed on hold by the local phone and remote indicates that call is placed on hold by the remote phone. In case of double-hold, the hold origin will display the value “Local and Remote”.

**Examples**

**Cisco Unified CME and Cisco Unified SRST**

The following is a sample output from this command displaying information for phones ringing in a voice register pool:

```plaintext
Router# show voice register pool on-hold brief
Outbound calls from SIP line phones:
Pool IP Address Number Remote Number Hold Origin
---- ---------------- -------------- ===============
1   8.3.3.111        45111        45112        Remote & Local

Inbound calls to SIP line phones:
Pool IP Address Number Remote Number Hold Origin
---- ---------------- -------------- ===============
2   8.33.33.112        45112        45131        Remote & Local
```

**Cisco Unified CME and Cisco Unified SRST**

The following is a sample output from this command displaying information for phones on-hold:

```plaintext
Router# show voice register pool on-hold
Outbound calls from SIP line phones:
Pool tag: 1
```
---

MAC Address : 001B.535C.D410
Contact IP : 8.3.3.111
Phone Number : 45111
Remote Number : 45112
Local Hold : CALL HOLD Pressed on SIP Phone

Call 4
SIP Call ID : 001b535c-d4100010-79612b5a-336b0db5@8.3.3.111
State of the call : STATE_ACTIVE (7)
Substate of the call : SUBSTATE_NONE (0)
Calling Number : 45111
Called Number : 45112
Bit Flags : 0xC0401C 0x10100 0x4
CC Call ID : 7
Source IP Address (Sig ) : 8.3.3.5
Destn SIP Req Addr:Port : [8.3.3.111]:5060
Destn SIP Resp Addr:Port: [8.3.3.111]:50076
Destination Name : 8.3.3.111
Number of Media Streams : 1
Number of Active Streams: 1
RTP Fork Object : 0x0
Media Mode : flow-through
Media Stream 1
State of the stream : STREAM_ACTIVE
Stream Call ID : 7
Stream Type : voice-only (0)
Stream Media Addr Type : 1
Negotiated Codec : g729r8 (20 bytes)
Codec Payload Type : 18
Negotiated Dtmf-relay : inband-voice
Dtmf-relay Payload Type : 0
QoS ID : -1
Local QoS Strength : BestEffort
Negotiated QoS Strength : BestEffort
Negotiated QoS Direction : None
Local QoS Status : None
Media Source IP Addr:Port: [8.3.3.5]:17580
Media Dest IP Addr:Port : [8.3.3.111]:26298

Options-Ping ENABLED:NO ACTIVE:NO
Inbound calls to SIP line phones:
Pool tag: 2
---

MAC Address : 0015.C68E.6D13
Contact IP : 8.33.33.112
Phone Number : 45112
Remote Number : 45111
Remote Hold : SIP Phone has received CALL HOLD

Call 5
SIP Call ID : 4DA52F97-ADA311DE-8019803A-FF3E4CBC@8.3.3.5
State of the call : STATE_ACTIVE (7)
Substate of the call : SUBSTATE_NONE (0)
Calling Number : 45111
Called Number : 45112
Bit Flags : 0xC04018 0x4100 0x80
CC Call ID : 8
Source IP Address (Sig ) : 8.3.3.5
Destn SIP Req Addr:Port : [8.33.33.112]:5060
Destn SIP Resp Addr:Port: [8.33.33.112]:5060
Destination Name : 8.33.33.112
Number of Media Streams : 1
Number of Active Streams: 1
RTP Fork Object : 0x0
Media Mode : flow-through
Media Stream 1
State of the stream : STREAM_ACTIVE
Stream Call ID : 8
Stream Type : voice-only (0)
Stream Media Addr Type : 1
Negotiated Codec : g729r8 (20 bytes)
Codec Payload Type : 18
Negotiated Dtmf-relay : inband-voice
Dtmf-relay Payload Type : 0
QoS ID : -1
Local QoS Strength : BestEffort
Negotiated QoS Strength : BestEffort
Negotiated QoS Direction : None
Local QoS Status : None
Media Source IP Addr:Port: [8.3.3.5]:16384
Media Dest IP Addr:Port : [8.33.33.112]:30040
Options-Ping  ENABLED:NO  ACTIVE:NO

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice register all</td>
<td>Displays all Cisco SIP SRST and Cisco CME configurations and register information.</td>
</tr>
<tr>
<td>show sip-ua calls</td>
<td>Displays active user agent client (UAC) and user agent server (UAS) information on SIP calls</td>
</tr>
<tr>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
</tbody>
</table>
show voice register pool phone-load

To display the details of phone-loads associated with phones that are registered to Cisco Unified CME, use the `show voice register pool phone-load` command in privileged EXEC mode.

Syntax Description
This command has no arguments or keywords.

Command Modes
Privileged EXEC

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

Usage Guidelines
Use this command to display the details of the phone-loads associated with phones that are registered with Cisco Unified CME. The phone-load information is taken from the REGISTER message sent by the phone.

Example
The following is a sample output from this command displaying information for voice register pool phone-load:

```
Router# show voice register pool phone-load
Pool  Device Name      Current Version Previous Version
----  ===========      =============== ===============
1     SEP01B535CD410  Cisco-CP7960G/8.0
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice register all</td>
<td>Displays all Cisco SIP SRST and Cisco CME configurations and register information.</td>
</tr>
<tr>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode for SIP phones.</td>
</tr>
</tbody>
</table>
show voice register pool registered

To display the details of phones that successfully register to Cisco Unified Communications Manager Express (Cisco Unified CME), use the `show voice register pool registered` command in privileged EXEC mode.

**show voice register pool registered**

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1 Cisco Unified SRST 8.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.2(4)M</td>
<td>Cisco Unified CME 9.1 Cisco Unified SIP SRST 9.1</td>
<td>This command was modified to display Key Expansion Module (KEM) details with the phone type information.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `show voice register pool registered` command to display the details of phones that are successfully registered to Cisco Unified CME and Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST).

**Cisco Unified CME**

The following is a sample output displaying information for a registered voice register pool in Cisco Unified CME:

```
Router# show voice register pool registered
Pool Tag 1
Config:
  Mac address is 001B.535C.D410
  Type is 7960
  Number list 1 : DN 1
  Number list 3 : DN 8
  Number list 4 : DN 7
  Proxy Ip address is 0.0.0.0
  Current Phone load version is Cisco-CP7960G/8.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  call-forward phone all is 4566
  call-forward b2bua all 4555
  keep-conference is enabled
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is supported
  registration Call ID is 001b535c-d410790d-17a6877e-5d04bbc588.3.3.111
  Privacy feature is not configured.
  Privacy button is disabled
  active primary line is: 45111
  contact IP address: 8.3.3.111 port 5060
Dialpeers created:
```
Dial-peers for Pool 1:
dial-peer voice 40001 voip
destination-pattern 45111
session target ipv4:8.3.3.111:5060
session protocol sipv2
call-fwd-all 4555
after-hours-exempt FALSE

Statistics:
Active registrations : 1
Total SIP phones registered: 1
Total Registration Statistics
Registration requests : 1
Registration success : 1
Registration failed : 0
unRegister requests : 0
unRegister success : 0
unRegister failed : 0
Attempts to register after last unregister : 0
Last register request time : *11:40:32.263 UTC Wed Oct 14 2009
Last unregister request time :
Unregister success time :

The following is a sample output displaying information for a registered voice register pool with a Cisco Unified 9971 Session Initiation Protocol (SIP) IP phone attached to a Cisco SIP IP Phone CKEM 36-Button Line Expansion Module:

Router# **show voice register pool registered**
Pool Tag 5
Config:
Mac address is B4A4.E328.4698
Type is 9971 addon 1 CKEM
Number list 1 : DN 2
Number list 2 : DN 3
Proxy Ip address is 0.0.0.0
DTMF Relay is disabled
Call Waiting is enabled
DnD is disabled
Video is enabled
Camera is enabled
Busy trigger per button value is 0
keep-conference is enabled
registration expires timer max is 200 and min is 60
kpml signal is enabled
Lpcor Type is none

Cisco Unified SRST

The following is a sample output displaying information for a registered voice register pool in Cisco Unified SRST:
Router# **show voice register pool registered**
Pool Tag 1
Config:
Ip address is 9.13.18.40, Mask is 255.255.0.0
Number list 1 : DN 1
Number list 2 : DN 2
Number list 3 : DN 3
Number list 4 : DN 4
Number list 5 : DN 5
Number list 6 : DN 6
Number list 7 : DN 7
Proxy Ip address is 0.0.0.0
DTMF Relay is enabled, rtp-nte, sip-notify
kpml signal is enabled
Lpcor Type is none
Dialpeers created:
Dial-peers for Pool 1:
dial-peer voice 40004 voip
destination-pattern 1000
redirect ip2ip
session target ipv4:9.13.18.40:19633
session protocol sipv2
dtmf-relay rtp-nte sip-notify
digit collect kpml
codec g711ulaw bytes 160
after-hours-exempt FALSE
dial-peer voice 40001 voip
destination-pattern 2000
redirect ip2ip
session target ipv4:9.13.18.40:19634
session protocol sipv2
dtmf-relay rtp-nte sip-notify
digit collect kpml
codec g711ulaw bytes 160
after-hours-exempt FALSE
dial-peer voice 40002 voip
destination-pattern 3000
redirect ip2ip
session target ipv4:9.13.18.40:19635
session protocol sipv2
dtmf-relay rtp-nte sip-notify
digit collect kpml
codec g711ulaw bytes 160
after-hours-exempt FALSE
dial-peer voice 40003 voip
destination-pattern 4000
redirect ip2ip
session target ipv4:9.13.18.40:19636
session protocol sipv2
dtmf-relay rtp-nte sip-notify
digit collect kpml
codec g711ulaw bytes 160
after-hours-exempt FALSE
Statistics:
Active registrations : 4
Total SIP phones registered: 1
Total Registration Statistics
  Registration requests : 4
  Registration success : 4
  Registration failed : 0
  unRegister requests : 0
  unRegister success : 0
  unRegister failed : 0
  Attempts to register after last unregister : 0
  Last register request time : .05:22:55.604 UTC Tue Oct 6 2009
  Last unregister request time :
  Register success time : .05:22:55.604 UTC Tue Oct 6 2009
  Unregister success time :

The following table contains descriptions of significant fields shown in the show voice register pool registered command output, listed in alphabetical order.
Table 66: show voice register pool registered Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Active registrations</td>
<td>Shows the current active registrations.</td>
</tr>
<tr>
<td>Application</td>
<td>Shows the application command configuration for this pool.</td>
</tr>
<tr>
<td>Call Waiting</td>
<td>Shows the setting of the call-waiting command.</td>
</tr>
<tr>
<td>Class of Restriction List Tag</td>
<td>Shows the COR tag.</td>
</tr>
<tr>
<td>Config</td>
<td>Shows the voice register pool.</td>
</tr>
<tr>
<td>Current phone-load</td>
<td>Shows the current version of the phone load.</td>
</tr>
<tr>
<td>Default preference</td>
<td>Shows the default preference value of this pool.</td>
</tr>
<tr>
<td>Dialpeers created</td>
<td>Results in a list of all dial peers created and their contents. Dial-peer contents differ for each application and are not described here.</td>
</tr>
<tr>
<td>DnD</td>
<td>Shows the setting of the dnd-control command.</td>
</tr>
<tr>
<td>DTMF Relay</td>
<td>Shows the setting of the dtmf-relay command.</td>
</tr>
<tr>
<td>Emergency response location</td>
<td>Shows the phone’s emergency response location to which an emergency response team is dispatched when an emergency call is made.</td>
</tr>
<tr>
<td>Incoming called number</td>
<td>Shows the incoming called-number command configuration.</td>
</tr>
<tr>
<td>Incoming corlist name</td>
<td>Shows the cor command configuration.</td>
</tr>
<tr>
<td>keep-conference</td>
<td>Shows the status of the keep-conference command.</td>
</tr>
<tr>
<td>Lpcor Incoming</td>
<td>Shows the setting of the lpcor incoming command.</td>
</tr>
<tr>
<td>Lpcor Outgoing</td>
<td>Shows the setting of the lpcor outgoing command.</td>
</tr>
<tr>
<td>Lpcor Type</td>
<td>Shows the setting of the lpcor type command.</td>
</tr>
<tr>
<td>Mac address</td>
<td>Shows the MAC address of this SIP phone as defined by the id command.</td>
</tr>
<tr>
<td>Network address and Mask</td>
<td>Shows network address and mask information when the id command is configured.</td>
</tr>
<tr>
<td>Number list, Pattern, and Preference</td>
<td>Shows the number command configuration.</td>
</tr>
<tr>
<td>Pool Tag</td>
<td>Shows the assigned tag number of the current pool.</td>
</tr>
<tr>
<td>Previous phone-load</td>
<td>Shows the version of the previous phone load.</td>
</tr>
<tr>
<td>Proxy IP address</td>
<td>Shows the proxy command configuration; that is, the IP address of the external SIP server.</td>
</tr>
</tbody>
</table>
### Field Description

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Registration failed</td>
<td>Shows the failed registrations.</td>
</tr>
<tr>
<td>Registration requests</td>
<td>Shows the incoming registration requests.</td>
</tr>
<tr>
<td>Registration success</td>
<td>Shows the successful registrations.</td>
</tr>
<tr>
<td>Statistics</td>
<td>Shows the registration statistics for this pool.</td>
</tr>
<tr>
<td>statistics time-stamps</td>
<td>Shows the registration statistics for this pool with specific time stamps.</td>
</tr>
<tr>
<td>Template</td>
<td>Shows the template-tag number for the template applied to this SIP phone.</td>
</tr>
<tr>
<td>Total Registration Statistics</td>
<td>Shows the total registration statistics for this pool.</td>
</tr>
<tr>
<td>Translate outgoing called tag</td>
<td>Shows the <code>translate-outgoing</code> command configuration.</td>
</tr>
<tr>
<td>Type</td>
<td>Shows the phone type identified for this SIP phone using the <code>type</code> command.</td>
</tr>
<tr>
<td>unRegister failed</td>
<td>Reports the number of failed unregisters.</td>
</tr>
<tr>
<td>unRegister requests</td>
<td>Shows the incoming unregister/registration expiry requests.</td>
</tr>
<tr>
<td>unRegister success</td>
<td>Reports the number of successful unregisters.</td>
</tr>
<tr>
<td>Username Password</td>
<td>Shows the values within the authentication credential.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>application (voice register pool)</td>
<td>Selects the session-level application for the dial peer associated with an individual Cisco Unified SIP IP phone in a Cisco Unified CME environment or for a group of phones in a Cisco Unified SIP SRST environment.</td>
</tr>
<tr>
<td>call-waiting (voice register pool)</td>
<td>Enables the call-waiting option on a SIP phone.</td>
</tr>
<tr>
<td>cor (voice register pool)</td>
<td>Configures a class of restriction on the VoIP dial peers associated with directory numbers.</td>
</tr>
<tr>
<td>dnd-control (voice register template)</td>
<td>Enables the Do-Not-Disturb (DND) soft key on SIP phones.</td>
</tr>
<tr>
<td>dtmf-relay (voice register pool)</td>
<td>Specifies the list of dual-tone multifrequency (DTMF) relay methods that can be used to relay DTMF audio tones between SIP endpoints.</td>
</tr>
<tr>
<td>id (voice register pool)</td>
<td>Explicitly identifies a locally available, individual Cisco Unified SIP IP phone or, when running Cisco Unified SIP SRST, a set of Cisco Unified SIP IP phones.</td>
</tr>
<tr>
<td>incoming called-number (dial peer)</td>
<td>Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>keep-conference (voice register pool)</td>
<td>Allows IP phone conference initiators to exit from conference calls and keep the remaining parties connected.</td>
</tr>
<tr>
<td>lpcor incoming</td>
<td>Associates an incoming call with a logical partitioning class of restriction (LPCOR) resource-group policy.</td>
</tr>
<tr>
<td>lpcor outgoing</td>
<td>Associates an outgoing call with an LPCOR resource-group policy.</td>
</tr>
<tr>
<td>lpcor type</td>
<td>Specifies the LPCOR type for an IP phone.</td>
</tr>
<tr>
<td>number (voice register pool)</td>
<td>Indicates the E.164 phone numbers that the registrar permits to handle the Register message from a Cisco Unified SIP IP phone.</td>
</tr>
<tr>
<td>proxy (voice register pool)</td>
<td>Autogenerates additional VoIP dial peers to reach the main proxy whenever a Cisco Unified SIP IP phone registers with a Cisco Unified SIP SRST gateway.</td>
</tr>
<tr>
<td>show voice register all</td>
<td>Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.</td>
</tr>
<tr>
<td>show voice register dial-peers</td>
<td>Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified SIP SRST or Cisco Unified CME register event.</td>
</tr>
<tr>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
<tr>
<td>show voice register pool unregistered</td>
<td>Displays the details of voice register pools that do not have any phones registered.</td>
</tr>
<tr>
<td>translate-outgoing (voice register pool)</td>
<td>Allows an explicit setting of translation rules on the VoIP dial peer to modify a phone number dialed by any Cisco Unified IP phone user.</td>
</tr>
<tr>
<td>type (voice register pool)</td>
<td>Defines a phone type for a SIP phone.</td>
</tr>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode for SIP phones.</td>
</tr>
</tbody>
</table>
show voice register pool remote

To display the details of phones that are at a remote location, use the `show voice register pool remote` command in privileged EXEC mode.

**show voice register pool remote**

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td></td>
<td>Cisco Unified SRST 8.1</td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to display the details of the phones that are at remote location and do not have an address resolution protocol (ARP) entry. If the pool id is MAC or IP, the entire pool detail is displayed in a brief format. If the pool id is network, only the line details with remote contact IP address are displayed. In Cisco Unified SRST, if the pool id is IP and if the pool is not registered, the configured IP is checked to see if it is a remote IP.

**Cisco Unified CME**

The following is a sample output from this command displaying information for remote phones:

```
Router# show voice register pool remote
Pool ID IP Address Ln DN Number State
----- =============== =============== == === ==================== ============
1  001B.535C.D410 8.3.3.111 1 1 45111 REGISTERED
3  8.3.0.0 1 3 45113 REGISTERED
4  8.3.0.0 1 3 45113 REGISTERED
2  8.3.3.112 1 2 45112 REGISTERED
3  8.3.0.0 1 3 45113 REGISTERED
```

**Cisco Unified SRST**

The following is a sample output from this command displaying information for remote phones:

```
Router# show voice register pool remote
Pool ID IP Address Ln DN Number State
----- =============== =============== == === ==================== ============
1  001B.535C.D410 8.3.3.111 1 1 45111 REGISTERED
3  8.3.0.0 1 3 45113 REGISTERED
4  8.3.0.0 1 3 45113 REGISTERED
2  8.3.3.112 8.3.3.112 1 2 45112 REGISTERED
3  8.3.0.0 8.3.44.116 1 3 45113 REGISTERED
```

The table contains descriptions of significant fields shown in this output, listed in alphabetical order.
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>show voice register all register all</strong></td>
<td>Displays all Cisco SIP SRST and Cisco CME configurations and register information.</td>
<td></td>
</tr>
<tr>
<td><strong>show voice register dial-peer</strong></td>
<td>Displays details of all dynamically created VoIP dial peers associated with the Cisco SIP SRST or Cisco CME register event.</td>
<td></td>
</tr>
<tr>
<td><strong>show voice register pool</strong></td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
<td></td>
</tr>
<tr>
<td><strong>voice register pool</strong></td>
<td>Enters voice register pool configuration mode for SIP phones.</td>
<td></td>
</tr>
</tbody>
</table>
show voice register pool ringing

To display the details of phones that are currently in ringing state, use the `show voice register pool ringing` command in privileged EXEC mode.

```
show voice register pool ringing [brief]
```

**Syntax Description**

`brief` (Optional) Displays brief details of SIP phones that are currently in ringing state.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>Cisco Unified SRST 8.1</td>
</tr>
<tr>
<td></td>
<td>This command was introduced.</td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to display the details of the phone that are currently in ringing state. When used with the `brief` keyword, the `show voice register pool ringing brief` command only displays information related to calls that are bound towards the SIP phones.

**Examples**

**Cisco Unified CME and Cisco Unified SRST**

The following is a sample output from this command displaying information for phones ringing in a voice register pool:

```
Router# show voice register pool ringing brief
Pool IP Address Number Remote Number
==== =============== ==================== ====================
2 8.33.33.112 45112 45111
```

**Cisco Unified CME and Cisco Unified SRST**

The following is a sample output from this command displaying information for phones ringing in a voice register pool:

```
Router# show voice register pool ringing
Pool tag: 2
---------------------
MAC Address: 0015.C68E.6D13
Contact IP: 8.33.33.112
Phone Number: 45112
Remote Number: 45111
Call 1
SIP Call ID: C0B5DA7-ADA311DE-8011803A-FF3E4CBC@8.3.3.5
State of the call: STATE_RECD_PROCEEDING (4)
Substate of the call: SUBSTATE_PROCEEDING_PROCEEDING (2)
Calling Number: 45111
Called Number: 45112
Bit Flags: 0xC00018 0x100 0x280
CC Call ID: 5
```
Source IP Address (Sig): 8.3.3.5
Destn SIP Req Addr:Port: [8.33.33.112]:5060
Destn SIP Resp Addr:Port: [8.33.33.112]:5060
Destination Name: 8.33.33.112
Number of Media Streams: 1
RTP Fork Object: 0x0
Media Mode: flow-through
Media Stream 1
  State of the stream: STREAM_ACTIVE
  Stream Call ID: 5
  Stream Type: voice+dtmf (1)
  Stream Media Addr Type: 1
  Negotiated Codec: No Codec (0 bytes)
  Codec Payload Type: 255 (None)
  Negotiated Dtmf-relay: inband-voice
  Dtmf-relay Payload Type: 0
  QoS ID: -1
  Local QoS Strength: BestEffort
  Negotiated QoS Strength: BestEffort
  Negotiated QoS Direction: None
  Local QoS Status: None
  Media Source IP Addr:Port: [8.3.3.5]:16882

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show sip-ua calls</td>
<td>Displays active user agent client (UAC) and user agent server (UAS) information on SIP calls</td>
</tr>
<tr>
<td>show voice register all</td>
<td>Displays all Cisco SIP SRST and Cisco CME configurations and register information.</td>
</tr>
<tr>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
</tbody>
</table>
show voice register pool telephone-number

To display the details of a phone line with a specific telephone-number, use the `show voice register pool telephone-number` command in privileged EXEC mode.

**show voice register pool telephone-number**

**Syntax Description**

| number | Number identifying a specific phone. |

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1 Cisco Unified SRST 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to display the details of the phone line with the specified telephone-number. If the line is registered, the contact ip address will be displayed. When the phone line is not registered and the pool ID type is network IP, the IP address is not displayed. When the phone line is not registered but some other line is registered for the same pool with MAC or IP address, then the IP address is displayed.

**Cisco Unified CME**

The following is a sample output from this command displaying all statistical information:

```
Router# show voice register pool telephone-number 45112
Pool ID   IP Address   Ln DN  Number   State
2         0015.C68E.6D13   1  2  45112  UNREGISTERED
7         0018.BAC8.D2B1   1  2  45112  UNREGISTERED
```

**Cisco Unified SRST**

```
Router# show voice register pool telephone-number 1000
Pool ID   IP Address   Ln DN  Number   State
1         9.13.18.40    9.13.18.40    1  1  1000  REGISTERED
```

The table contains descriptions of significant fields shown in this output, listed in alphabetical order.

**Table 67: show voice register pool telephone number field descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DN</td>
<td>Directory number of the phone.</td>
</tr>
<tr>
<td>ID</td>
<td>Phone identification (ID) address.</td>
</tr>
<tr>
<td>IP Address</td>
<td>IP address and port number of the phones</td>
</tr>
</tbody>
</table>
### Field Description

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>LN</td>
<td>Line number of the phone.</td>
</tr>
<tr>
<td>Number</td>
<td>Number of the phones.</td>
</tr>
<tr>
<td>Pool</td>
<td>Shows the current pool.</td>
</tr>
<tr>
<td>State</td>
<td>Registration state.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice register all</td>
<td>Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.</td>
</tr>
<tr>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
<tr>
<td>show voice register pool detail all</td>
<td>Displays the details of all the pools defined in the system.</td>
</tr>
</tbody>
</table>
show voice register pool type

To display the details of voice register pools associated with a specific phone type, use the `show voice register pool type` command in privileged EXEC mode.

```
show voice register pool type type
```

**Syntax Description**

| type | 3911, 3951, 7905, 7906, 7911, 7912, 7940, 7941, 7941GE, 7942, 7945, 7960, 7961, 7961GE, 7962, 7965, 7970, 7971, 7975, 7800 Series, 8800 Series, ATA (Cisco SIP Phone ATA), ATA-191, CKEM (Cisco SIP Key Expansion Module), CP-8800-Audio (Cisco SIP Key Expansion Module), CP-8800-Video (Cisco SIP Key Expansion Module), P100 (PingTel Xpressa 100), P600 (Polycom SoundPoint 600). |

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.2(4)M</td>
<td>Cisco Unified CME 9.1</td>
<td>This command was modified to add CKEM as a value for the <code>type</code> argument to display the details of voice register pools associated with all the phones configured with KEMs.</td>
</tr>
<tr>
<td>15.3(3)M</td>
<td>Cisco Unified CME 10.0</td>
<td>This command was enhanced to display the properties for new sip phone models configured using SIP fast track feature. New keyword option <code>all</code> was added to display all the phone models being used in the system along with the associated pools and registration details.</td>
</tr>
<tr>
<td>Cisco IOS XE</td>
<td>Unified CME 12.5</td>
<td>This command was modified to add <strong>CP-8800-Audio</strong> and <strong>CP-8800-Video</strong> as values for the <code>type</code> argument to display the details of voice register pools associated with A-KEMs and V-KEMs.</td>
</tr>
<tr>
<td>Gibraltar 16.10.1a</td>
<td>Unified CME 12.5</td>
<td>This command was modified to add <strong>ATA-191</strong> as the value for the <code>type</code> argument to display the details of voice register pools associated with Cisco ATA 191.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `show voice register pool type` command to display the details of voice register pools associated with a specific phone type.

The `show voice register pool type` command only takes the configured value of the phone type into consideration.

The CKEM value is available for Cisco Unified CME only and is not available for Cisco Unified SRST.

**Examples**

The following is a sample output of the `show voice register pool type` command for a Cisco Unified 7960 SIP IP phone, displaying all statistical information:
Router# `show voice register pool type 7960`

<table>
<thead>
<tr>
<th>Pool ID</th>
<th>IP Address</th>
<th>Ln DN</th>
<th>Number</th>
<th>State</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>001B.535C.D410</td>
<td>1</td>
<td>1</td>
<td>45111 REGISTERED</td>
</tr>
<tr>
<td>2</td>
<td>0015.C68E.6D13</td>
<td>1</td>
<td>2</td>
<td>45112 UNREGISTERED</td>
</tr>
</tbody>
</table>

The following is a sample output of the `show voice register pool type` command, showing all the phones configured with CP-8800-Audio:

Router# `show voice register pool type CP-8800-Audio`

<table>
<thead>
<tr>
<th>Pool ID</th>
<th>IP Address</th>
<th>Ln DN</th>
<th>Number</th>
<th>State</th>
</tr>
</thead>
<tbody>
<tr>
<td>2</td>
<td>38ED.18AF.8993</td>
<td>1</td>
<td>2</td>
<td>7001$ REGISTERED</td>
</tr>
</tbody>
</table>

The following is a sample output of the `show voice register pool type` command, showing all the 8865 phones configured with CP-8800-Video:

Router# `show voice register pool type CP-8800-Video`

<table>
<thead>
<tr>
<th>Pool ID</th>
<th>IP Address</th>
<th>Ln DN</th>
<th>Number</th>
<th>State</th>
</tr>
</thead>
<tbody>
<tr>
<td>4</td>
<td>00CC.FC99.8973</td>
<td>1</td>
<td>1</td>
<td>5001$ REGISTERED</td>
</tr>
</tbody>
</table>

The following is a sample output of the `show voice register pool type` command for a Cisco Unified 7821 SIP IP phone configured using SIP fast track feature, displaying all statistical information:

Router# `show voice register pool type 7821`

<table>
<thead>
<tr>
<th>FastTrack Phone Model : 7821</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pool type(index) representing the phone model : 7821</td>
</tr>
<tr>
<td>Reference pool type to inherit the properties from : 6921</td>
</tr>
<tr>
<td>Number of lines supported : 2 (inherited from 6921)</td>
</tr>
<tr>
<td>Number of addon modules supported : 0 (inherited from 6921)</td>
</tr>
<tr>
<td>Default session transport : UDP (inherited from 6921)</td>
</tr>
<tr>
<td>Description(helpstring) : Cisco IP Phone 7821</td>
</tr>
<tr>
<td>Phone supports GSM : NO (inherited from 6921)</td>
</tr>
<tr>
<td>Phone supports Telnet acess : NO (inherited from 6921)</td>
</tr>
<tr>
<td>Phone supports firmware download from CME : YES (inherited from 6921)</td>
</tr>
<tr>
<td>Phone specific XML tags :</td>
</tr>
<tr>
<td><code>&lt;maxNumCalls&gt;12&lt;/maxNumCalls&gt;</code> (inherited from 6921)</td>
</tr>
<tr>
<td><code>&lt;busyTrigger&gt;12&lt;/busyTrigger&gt;</code> (inherited from 6921)</td>
</tr>
<tr>
<td>Phone Family : RTL_PHONES</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Pool ID</th>
<th>IP Address</th>
<th>Ln DN</th>
<th>Number</th>
<th>State</th>
</tr>
</thead>
<tbody>
<tr>
<td>6</td>
<td>D824.BD27.9EAC</td>
<td>1</td>
<td>6</td>
<td>4080$ REGISTERED</td>
</tr>
</tbody>
</table>

The following is a sample output of the `show voice register pool type all` command, showing all the phone models used in the system:

Router# `show voice register pool type all`

<p>| Builtin Phone Model : 9971 |</p>
<table>
<thead>
<tr>
<th>Pool ID</th>
<th>IP Address</th>
<th>Ln DN Number</th>
<th>State</th>
</tr>
</thead>
<tbody>
<tr>
<td>3</td>
<td>A418.7529.93B0</td>
<td>1 3</td>
<td>4012$ REGISTERED</td>
</tr>
<tr>
<td>9</td>
<td>001E.7A25.D4EE</td>
<td>1 9</td>
<td>4006 UNREGISTERED</td>
</tr>
</tbody>
</table>

Built-in KEM Module: CKEM

<table>
<thead>
<tr>
<th>Pool ID</th>
<th>IP Address</th>
<th>Ln DN Number</th>
<th>State</th>
</tr>
</thead>
<tbody>
<tr>
<td>8</td>
<td>1234.1234.1234</td>
<td></td>
<td>UNREGISTERED</td>
</tr>
</tbody>
</table>

Built-in Phone Model: Jabber-MAC

FastTrack Phone Model: 8900
Pooltype(index) representing the phone model: 52
Reference pooltype to inherit the properties from: 8945
Number of lines supported: 4
Number of addon modules supported: 0 (inherited from 8945)
Default session transport: UDP (inherited from 8945)
Description(helpstring): Cisco SIP Phone 8945
Phone supports GSM: NO (inherited from 8945)
Phone supports Telnet access: NO (inherited from 8945)
Phone supports firmware download from CME: YES
Phone specific XML tags:
<maxNumCalls>24</maxNumCalls>
<busyTrigger>24</busyTrigger>
Phone family: GUMBO_PHONES

<table>
<thead>
<tr>
<th>Pool ID</th>
<th>IP Address</th>
<th>Ln DN Number</th>
<th>State</th>
</tr>
</thead>
<tbody>
<tr>
<td>7</td>
<td>D824.BD27.9EBD</td>
<td>1 7</td>
<td>4022$ REGISTERED</td>
</tr>
</tbody>
</table>

Router#

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice register all</td>
<td>Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.</td>
</tr>
<tr>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
</tbody>
</table>
show voice register pool type summary

To display the total count of registered and unregistered phones for each Session Initiation Protocol (SIP) phone type, use the `show voice register pool type summary` command in privileged EXEC mode.

**show voice register pool type summary**

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

This command has no default behavior or values.

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3) M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to view the count of the phones configured, registered and unregistered in the SIP mode.

**Example**

The following is a sample output of the `show voice register pool type summary` command:

```
router# show voice register pool type summary
PhoneType Configured Registered Unregistered
7970 1 1 0
8941 4 3 1
Unknown Phone type 4 0 4

Total Phones 9 4 5
```
show voice register pool unregistered

To display the details of the voice registration pools that do not have any phones registered, use the **show voice register pool unregistered** command in privileged EXEC mode.

**Syntax Description**
- This command has no arguments or keywords.

**Command Modes**
- Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1 Cisco Unified SRST 8.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
- Use this command to display the details of the pools that do not have any active registrations. In Cisco Unified SRST, if multiple phones are trying to register through the same pool and if one phone successfully registers and the others do not, the pool is not considered as an unregistered pool, as it does have an active registration of the registered phone.

**Examples**

**Cisco Unified CME and Cisco Unified SRST**

The following is a sample output from this command displaying information for pools with no active registration:

```
Router# show voice register pool unregistered
Pool Tag: 2
  MAC Address : 0015.C68E.6D13
  No. of attempts to register: 0
  Unregister time : 
  Last register request time : 
  Reason for state unregister:  No registration request since last reboot/unregister
Pool Tag: 3
  MAC Address : 0021.5553.8998
  No. of attempts to register: 0
  Unregister time : 
  Last register request time : 
  Reason for state unregister:  No registration request since last reboot/unregister
Pool Tag: 4
  MAC Address : 8999.9867.8769
  No. of attempts to register: 0
  Unregister time : 
  Last register request time : 
  Reason for state unregister:  No registration request since last reboot/unregister
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice register all</td>
<td>Displays all Cisco SIP SRST and Cisco CME configurations and register information.</td>
</tr>
<tr>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
<tr>
<td>show voice register pool registered</td>
<td>Displays details of phones that successfully register to Cisco Unified CME or Cisco Unified SRST.</td>
</tr>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode for SIP phones.</td>
</tr>
</tbody>
</table>
show voice register profile

To display the content of configuration files that are in ASCII text format, use the **show voice register profile** command in privileged EXEC mode.

**show voice register profile text tag**

**Syntax Description**

| tag | Unique identifier for voice register profile to be displayed. Range is 1–500.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
</table>
| 12.4(4)T          | Cisco CME 3.4 | This command was introduced.

**Usage Guidelines**

Use this command to display ASCII configuration files for the Cisco IP Phone 7905 and 7905G, Cisco IP Phone 7912 and 7912G, Cisco ATA-186, or Cisco ATA-188. To generate ASCII text files, use the **file text** command.

**Examples**

The following is sample output from this command displaying information in the configuration profile for voice register pool 4:

```
Router# show voice register profile text 4
Pool Tag: 4
txt
 AutoLookUp:0
 DirectoriesUrl:0
 ... CallWaiting:1
 CallForwardNumber:0
 Conference:1
 AttendedTransfer:1
 BlindTransfer:1
 ... SIPRegOn:1
 UseTftp:1
 UseLoginID:0
 UPassword:0
 NTPIP:0.0.0.0
 UID:2468
 ...
```

The following table contains descriptions of significant fields shown in this output, listed in alphabetical order.

**Table 68: show voice register profile Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Attended Transfer</td>
<td>Setting of soft key for attended transfer in a SIP phone template as defined by using the <strong>transfer-attended</strong> command. “1” indicates that the soft key is enabled; “0” indicates that the soft key is disabled.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Auto Lookup</td>
<td>1 indicates that Auto Lookup is enabled. 0 indicates that it is disabled.</td>
</tr>
<tr>
<td>Blind Transfer</td>
<td>Setting of soft key for blind transfer in a SIP phone template as defined by using the <code>transfer-blind</code> command. “1” indicates that the soft key is enabled; “0” indicates that the soft key is disabled.</td>
</tr>
<tr>
<td>Call Waiting</td>
<td>Setting of the call-waiting option on a SIP phone as defined by using the <code>call-waiting</code> command. “1” indicates that the soft key is enabled; “0” indicates that the soft key is disabled.</td>
</tr>
<tr>
<td>Call Forward Number</td>
<td>Number to which incoming calls are forwarded</td>
</tr>
<tr>
<td>Conference</td>
<td>Setting of soft key for conference in a SIP phone template as defined by using the <code>conference</code> command. “1” indicates that the soft key is enabled; “0” indicates that the soft key is disabled.</td>
</tr>
<tr>
<td>Directories URL</td>
<td>1 indicates that the Directories feature button for the phone is enabled. 0 indicates that it is disabled.</td>
</tr>
<tr>
<td>NTPIP</td>
<td>IP address for the NTP source</td>
</tr>
<tr>
<td>Pool tag</td>
<td>Pool tag of the configuration file being requested.</td>
</tr>
<tr>
<td>SIP Reg On</td>
<td>1 indicates that the registration with external proxy server for the phone is enabled. 0 indicates that it is disabled.</td>
</tr>
<tr>
<td>UI Password</td>
<td>1 indicates that the UI password is enabled on the phone. 0 indicates that dit is disabled.</td>
</tr>
<tr>
<td>UID</td>
<td>Authentication credential for SIP phone.</td>
</tr>
<tr>
<td>Use Login ID</td>
<td>1 indicates that “use login id” for phone is enabled. 0 indicates that it is disabled.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>create profile (voice register</code></td>
<td>Generates the configuration profiles required for SIP phone.</td>
</tr>
<tr>
<td><code>global)</code></td>
<td></td>
</tr>
<tr>
<td><code>file text (voice register</code></td>
<td>Generates ASCII text files for the Cisco IP Phone 7905 and 7905G, Cisco IP Phone 7912 and 79012G, Cisco ATA-186, or Cisco ATA-188.</td>
</tr>
<tr>
<td><code>global)</code></td>
<td></td>
</tr>
<tr>
<td><code>reset (voice register</code></td>
<td>Performs a complete reboot of all SIP phones associated with a Cisco CME router.</td>
</tr>
<tr>
<td><code>global)</code></td>
<td></td>
</tr>
<tr>
<td><code>voice register global</code></td>
<td>Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.</td>
</tr>
</tbody>
</table>
**show voice register session-server**

To display the call details of the registered session servers, use the `show voice register session-server` command in privileged EXEC mode.

### Syntax Description

This command has no arguments or keywords.

### Command Modes

Privileged EXEC (#)

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)T</td>
<td>This command was introduced in a release earlier than Cisco IOS Release 12.4(22)T.</td>
</tr>
</tbody>
</table>

### Examples

The following is sample output from the `show voice register session-server` command:

```
Router# show voice register session-server
Feature server 2, keepalive 60, register-uri CISCO-80NVCGATW_1259887561000
Session reg_number 569, refID 9B2783C0
Route point voice_reg_pool 28 reg_number 570
Route point voice_reg_pool 9 reg_number 571
Route point voice_reg_pool 19 reg_number 572
Route point voice_reg_pool 22 reg_number 573
Subscription sub_id 1133, calledNumber 1242
Subscription sub_id 1135, calledNumber 1054
Subscription sub_id 1138, calledNumber 1155
Subscription sub_id 1140, calledNumber 1188
Subscription sub_id 1142, calledNumber 261
Subscription sub_id 1146, calledNumber 1055
Subscription sub_id 1147, calledNumber 1100
Subscription sub_id 1149, calledNumber 1025
Subscription sub_id 1152, calledNumber 264
Subscription sub_id 1154, calledNumber 267
Subscription sub_id 1156, calledNumber 1185
Subscription sub_id 1157, calledNumber 1218
Subscription sub_id 1160, calledNumber 1056
Subscription sub_id 1161, calledNumber 263
Subscription sub_id 1163, calledNumber 1186
Subscription sub_id 1165, calledNumber 1243
Subscription sub_id 1167, calledNumber 1053
Subscription sub_id 1169, calledNumber 1120
Subscription sub_id 1171, calledNumber 1154
Subscription sub_id 1173, calledNumber 265
```

The following table describes the significant fields shown in this output.

**Table 68: show voice register session-server Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>Feature server</td>
<td>The number of active feature servers.</td>
</tr>
<tr>
<td>keepalive</td>
<td>Interval, in seconds, at which the peer sends keepalive messages. The range is from 1 to 60 seconds. The default is 60 seconds.</td>
</tr>
<tr>
<td>Field</td>
<td>Definition</td>
</tr>
<tr>
<td>------------------------------</td>
<td>--------------------------------------------------------------------------</td>
</tr>
<tr>
<td>register-uri</td>
<td>The registered Uniform Resource Identifier (URI) for the server.</td>
</tr>
<tr>
<td>Session reg_number</td>
<td>The registered number of the session.</td>
</tr>
<tr>
<td>Route point voice_reg_pool</td>
<td>Denotes the registered virtual device for application redirection.</td>
</tr>
<tr>
<td>Subscription sub_id</td>
<td>The subidentification number of the subscription.</td>
</tr>
</tbody>
</table>
**show voice register statistics**

To display statistics associated with the registration event, use the `show voice register statistics` command in privileged EXEC mode.

```
show voice register statistics [{global|pool tag}]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>global</td>
<td>(Optional) Displays aggregate statistics associated with the SIP phone registration event.</td>
</tr>
<tr>
<td>pool tag</td>
<td>(Optional) Displays registration pool statistics associated with a specific pool tag. The maximum number of pools is version and platform dependent. Type <code>?</code> to display a list of values.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC

<table>
<thead>
<tr>
<th>Command History</th>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>12.2(15)ZJ</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td></td>
<td>12.3(4)T</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td></td>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was added to Cisco CME.</td>
</tr>
<tr>
<td></td>
<td>15.1(2)T</td>
<td>Cisco CME 8.1 Cisco SIP SRST 8.1</td>
<td>This command was modified. The global and pool keywords and tag argument were added. The output display was also modified to show more information about pools in unregistered state and time-stamps of registration event.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

When using the `show voice register statistics` command, you can verify that the number of Registration and unRegister successes for global statistics are the sum of the values in the individual pools. Because some Registrations fail even before matching a voice register pool, for Registration and unRegister failed statistics the value is not the sum of the values in the individual pools. Immediate failures are accounted in the global statistics.

In Cisco Unified CME 8.1 and Cisco Unified SIP SRST 8.1, the time-stamps for the events is displayed along with other registration related statistics. The command output also displays the reason for pools in unregistered state. Use the `show voice register statistics` command with pool tag keyword to display registration pool statistics associated with a specific pool.

When using the global keyword, the `show voice register` command output displays the aggregate statistics associated with SIP phone registration. The output of this command also displays the attempted-registrations table.

**Examples**

**Cisco Unified CME and Cisco Unified SRST**

The following is a sample output from this command displaying all statistical information:
Router# **show voice register statistics**

**Sample Output:**

**Global statistics**
- Active registrations : 2
- Total SIP phones registered: 2

**Total Registration Statistics**
- Registration requests : 3
- Registration success : 2
- Registration failed : 1
- unRegister requests : 0
- unRegister success : 0
- unRegister failed : 0
- Attempts to register after last unregister : 1
- Last Unregister Request Time :
- Register Success Time : *11:11:56.707 UTC Wed Sep 16 2009
- Unregister Success Time :

**Register pool 1 statistics**
- Active registrations : 1
- Total SIP phones registered: 1

**Total Registration Statistics**
- Registration requests : 1
- Registration success : 1
- Registration failed : 0
- unRegister requests : 0
- unRegister success : 0
- unRegister failed : 0
- Attempts to register after last unregister : 0
- Last Register Request Time : *11:11:54.615 UTC Wed Sep 16 2009
- Last Unregister Request Time :
- Register Success Time : *11:11:54.623 UTC Wed Sep 16 2009
- Unregister Success Time :

**Register pool 2 statistics**
- Active registrations : 1
- Total SIP phones registered: 1

**Total Registration Statistics**
- Registration requests : 1
- Registration success : 1
- Registration failed : 0
- unRegister requests : 0
- unRegister success : 0
- unRegister failed : 0
- Attempts to register after last unregister : 0
- Last Register Request Time : *11:11:56.707 UTC Wed Sep 16 2009
- Last Unregister Request Time :
- Register Success Time : *11:11:56.707 UTC Wed Sep 16 2009
- Unregister Success Time :

---

**Cisco Unified CME and Cisco Unified SRST**

The following is a sample output from this command displaying all statistical information:

Router# **show voice register statistics global**

**Global Statistics:**
- Active registrations : 1
Total SIP phones registered: 2
Total Registration Statistics
  Registration requests : 97715
  Registration success : 3
  Registration failed : 97712
  unRegister requests : 1
  unRegister success : 1
  unRegister failed : 0
Attempts to register after last unregister : 97712
  Last register request time : *06:45:11.127 UTC Wed Oct 14 2009
  Last unregister request time : *11:56:22.179 UTC Tue Oct 13 2009
  Register success time : *12:10:37.263 UTC Tue Oct 13 2009
  Unregister success time : *11:56:22.182 UTC Tue Oct 13 2009
Phones that have attempted registrations and have failed:
  MAC address: 001b.535c.d410
  IP address : 8.3.3.111
  Attempts : 97712
  Time of first attempt : *12:20:32.775 UTC Tue Oct 13 2009
Reason for failure :
  Unauthorized registration request

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying all statistical information associated with pool 1:

Router# show voice register statistics pool 1
Pool 1 Statistics:
  Active registrations : 0
  Total SIP phones registered: 1
Total Registration Statistics
  Registration requests : 2
  Registration success : 2
  Registration failed : 0
  unRegister requests : 1
  unRegister success : 1
  unRegister failed : 0
Attempts to register after last unregister : 0
  Last register request time : *12:10:37.259 UTC Tue Oct 13 2009
  Last unregister request time : *11:56:22.179 UTC Tue Oct 13 2009
  Register success time : *12:10:37.263 UTC Tue Oct 13 2009
  Unregister success time : *11:56:22.182 UTC Tue Oct 13 2009
Reason for unregistered state:
  No registration request since last reboot/unregister

The following table describes the significant fields shown in this output.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Statistics:</td>
<td>Used with the all, pool, and statistics keywords. Shows the registration</td>
</tr>
<tr>
<td></td>
<td>statistics for this pool.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Active registrations</td>
<td>Used with the <strong>all, pool, and statistics</strong> keywords. Shows the current active registrations.</td>
</tr>
<tr>
<td>Last Register Request Time</td>
<td>Used with all, pool, and statistics keywords. Shows details such as day, date, and time when the phones requested to register the last time.</td>
</tr>
<tr>
<td>Last unRegister Request Time</td>
<td>Used with all, pool, and statistics keywords. Shows details such as day, date, and time when the phones requested to unregister the last time.</td>
</tr>
<tr>
<td>Total Registration Statistics</td>
<td>Used with the <strong>all, pool, and statistics</strong> keywords. Shows the total registration statistics for this pool.</td>
</tr>
<tr>
<td>Registration requests</td>
<td>Used with the <strong>all, pool, and statistics</strong> keywords. Shows the incoming registration requests.</td>
</tr>
<tr>
<td>Registration success</td>
<td>Used with the <strong>all, pool, and statistics</strong> keywords. Shows the successful registrations.</td>
</tr>
<tr>
<td>Registration failed</td>
<td>Used with the <strong>all, pool, and statistics</strong> keywords. Shows the failed registrations.</td>
</tr>
<tr>
<td>unRegister requests</td>
<td>Used with the <strong>all, pool, and statistics</strong> keywords. Shows the incoming unregister/registration expire requests.</td>
</tr>
<tr>
<td>unRegister success</td>
<td>Used with the <strong>all, pool, and statistics</strong> keywords. Reports the number of successful unregisters.</td>
</tr>
<tr>
<td>unRegister failed</td>
<td>Used with the <strong>all, pool, and statistics</strong> keywords. Reports the number of failed unregisters.</td>
</tr>
<tr>
<td>Global statistics</td>
<td>Used with the <strong>statistics</strong> keyword. Details all active registrations.</td>
</tr>
<tr>
<td>Register pool <em>number</em> statistics</td>
<td>Used with the <strong>statistics</strong> keyword. Details specific pool statistics.</td>
</tr>
</tbody>
</table>

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>show voice register all</strong></td>
<td>Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.</td>
</tr>
<tr>
<td><strong>show voice register pool</strong></td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
<tr>
<td><strong>show voice register pool</strong> attempted-registrations</td>
<td>Displays the details of phones that attempt to register with Cisco Unified CME or Cisco Unified SRST and fail.</td>
</tr>
</tbody>
</table>
show voice register template

To display all configuration information associated with a Cisco Unified SIP IP phone template, use the `show voice register template` command in privileged EXEC mode.

```
show voice register template {template-tag|all}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>template-tag</code></td>
<td>Number of the template for which to display information. Range is 1 to 5.</td>
</tr>
<tr>
<td><code>all</code></td>
<td>Displays all configuration information associated with all the Cisco Unified SIP IP phone templates.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC (#)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was modified to include emergency response location (ERL) information assigned to a Cisco Unified SIP IP phone in the output display.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was modified to include logical partitioning class of restriction (LPCOR) information in the output display.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was modified. All keyword was added. Pools that have the template defined are also displayed in the output. Voice-class stun-usage information is displayed in the output.</td>
</tr>
<tr>
<td>15.2(2)T</td>
<td>Cisco Unified CME 9.0</td>
<td>This command was modified to include conference admin, conference add mode, and conference drop mode in the output display.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `show voice register template` command to display all configuration information associated with a Cisco Unified SIP IP phone template defined in a system. Use the `all` keyword with the `show voice register template` command to display the details of all the templates defined in the system. A maximum of 10 templates can be configured and hence, the details of a maximum of 10 templates are displayed in the output.

**Examples**

The following is a sample output from the `show voice register template` command displaying information for a voice register template:
Router# show voice register template 1

Temp Tag 1
Config:
Attended Transfer is enabled
Blind Transfer is enabled
Semi-attended Transfer is enabled
Conference is enabled
Caller-ID block is disabled
DnD control is enabled
Anonymous call block is disabled
Voicemail is 56789, timeout 15
softkey connected Confrn Endcall Hold Trnsfer
softkey hold Newcall Resume
softkey idle Cfwdll Newcall Redial
softkey seized Cfwdll Endcall Redial
Emergency response location 6
Lpcor type local
Lpcor incoming sccp_phonel
Lpcor outgoing sccp_phonel

The following is a sample output from the `show voice register template` command displaying voice-class stun-usage information for voice register template 10:

Router# show voice register template 10
Temp Tag 10
Config:
Attended Transfer is enabled
Blind Transfer is enabled
Semi-attended Transfer is enabled
Conference is enabled
Caller-ID block is disabled
DnD control is enabled
Anonymous call block is disabled
softkey connected Confrn Endcall Hold Trnsfer
voice-class stun-usage is enabled. tag is 1
Lpcor type none
Pool 2 has this template configured
Pool 3 has this template configured
Pool 5 has this template configured
Pool 6 has this template configured
Pool 7 has this template configured
Pool 8 has this template configured
Pool 9 has this template configured
Pool 10 has this template configured
Pool 11 has this template configured
Pool 50 has this template configured

The following is a sample output from the `show voice register template` command. The output shows that a hardware conference administrator has been assigned, only the conference creator can add a new participant, and the conference creator can terminate the active video hardware conference by hanging up.

Router# show voice register template 5
Temp Tag 5
Config:
Attended Transfer is enabled
Blind Transfer is enabled
Semi-attended Transfer is enabled
Conference softkey is enabled
Caller-ID block is disabled
DnD control is enabled
Video is disabled
Camera is enabled
Anonymous call block is disabled
Lpcor type none
paging-dn 0 [multicast]
conference admin: yes
conference add mode: creator

conference drop mode: creator

The following table contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 71: show voice register template Field Descriptions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Anonymous call block</td>
<td>Status of anonymous caller blocking defined with the <strong>anonymous block</strong> command.</td>
</tr>
<tr>
<td>Attended Transfer</td>
<td>Status of attended transfer soft key defined with the <strong>transfer-attended</strong> command.</td>
</tr>
<tr>
<td>Blind Transfer</td>
<td>Status of blind transfer soft key defined with the <strong>transfer-blind</strong> command.</td>
</tr>
<tr>
<td>Caller-ID block</td>
<td>Status of caller-id feature defined with the <strong>caller-id block</strong> command.</td>
</tr>
<tr>
<td>Conference</td>
<td>Status of conference soft key defined with the <strong>conference</strong> command.</td>
</tr>
<tr>
<td>Conference admin</td>
<td>Shows whether the Cisco Unified SIP IP phone is assigned as the hardware conference administrator or not.</td>
</tr>
<tr>
<td>Conference add mode</td>
<td>Current setting of hardware conference privilege for adding participants.</td>
</tr>
<tr>
<td>Conference drop mode</td>
<td>Shows who can terminate an active ad-hoc hardware conference by hanging up.</td>
</tr>
<tr>
<td>Config:</td>
<td>List of configuration options defined for this template.</td>
</tr>
<tr>
<td>Dnd controls</td>
<td>Status of Do-Not-Disturb soft key defined with the <strong>dnd-control</strong> command.</td>
</tr>
<tr>
<td>Emergency response location</td>
<td>The ephone’s emergency response location to which an emergency response team is dispatched when an emergency call is made.</td>
</tr>
<tr>
<td>Lpcor incoming</td>
<td>Setting of the <strong>lpcor incoming</strong> command.</td>
</tr>
<tr>
<td>Lpcor outgoing</td>
<td>Setting of the <strong>lpcor outgoing</strong> command.</td>
</tr>
<tr>
<td>Lpcor type</td>
<td>Setting of the <strong>lpcor type</strong> command.</td>
</tr>
<tr>
<td>Temp Tag</td>
<td>Tag number of the requested template.</td>
</tr>
<tr>
<td>VAD</td>
<td>Status of voice activity detection defined with the <strong>vad</strong> command.</td>
</tr>
<tr>
<td>Voicemail</td>
<td>Voice-mail extension and timeout value defined with the <strong>voice-mail</strong> command.</td>
</tr>
</tbody>
</table>
Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice register all</td>
<td>Displays all voice register information, including statistics, pools, and dial peers.</td>
</tr>
<tr>
<td>voice register template</td>
<td>Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones.</td>
</tr>
</tbody>
</table>
show voice register tftp-bind

To display the current configuration files accessible to SIP phones, use the `show voice register tftp-bind` command in privileged EXEC mode.

**Syntax Description**

This command has no arguments or keywords.

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command provides a list of configuration files that are accessible to SIP phones using TFTP.

**Examples**

The following is sample output from this command:

```
Router(config)# show voice register tftp-bind
  tftp-server SIPDefault.cnf url system:/cme/sipphone/SIPDefault.cnf
  tftp-server syncinfo.xml url system:/cme/sipphone/syncinfo.xml
  tftp-server SIP0000B7F7532E.cnf url system:/cme/sipphone/SIP0000B7F7532E.cnf
  tftp-server SIP000ED7DF7932.cnf url system:/cme/sipphone/SIP000ED7DF7932.cnf
  tftp-server SIP0012D9DEOAA.cnf url system:/cme/sipphone/SIP0012D9DEOAA.cnf
  tftp-server gk123456789012 url system:/cme/sipphone/gk123456789012
  tftp-server gk123456789012.txt url system:/cme/sipphone/gk123456789012.txt
```

The following table contains descriptions of significant fields shown in this output, listed in alphabetical order.

**Table 72: show voice register tftp-bind Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ata&lt;mac-address&gt;</td>
<td>Cisco SIP configuration profile for a particular Cisco ATA-186 or Cisco ATA-188 as indicated by the <code>&lt;mac-address&gt;</code>. This file is generated by using the <code>create profile</code> command.</td>
</tr>
<tr>
<td>ata&lt;mac-address&gt;.txt</td>
<td>ASCII text file of a Cisco SIP configuration profile for a particular Cisco ATA-186 or Cisco ATA-188 as indicated by the <code>&lt;mac-address&gt;</code>. This file is generated by using the <code>file text</code> command.</td>
</tr>
<tr>
<td>gk&lt;mac-address&gt;</td>
<td>Cisco SIP configuration profile for a particular Cisco IP Phone 7912 or Cisco IP Phone 7912G as indicated by the <code>&lt;mac-address&gt;</code>. This file is generated by using the <code>create profile</code> command.</td>
</tr>
<tr>
<td>gk&lt;mac&gt;.txt</td>
<td>ASCII text file of a Cisco SIP configuration profile for a particular Cisco IP Phone 7912 or Cisco IP Phone 7912G as indicated by the <code>&lt;mac-address&gt;</code>. This file is generated by using the <code>file text</code> command.</td>
</tr>
</tbody>
</table>
Cisco SIP configuration profile for a particular Cisco IP Phone 7905 or Cisco IP Phone 7912G as indicated by the <mac-address>. This file is generated by using the create profile command.

ASCII text file of a Cisco SIP configuration profile for a particular Cisco IP Phone 7905 or Cisco IP Phone 7912G as indicated by the <mac-address>. This file is generated by using the file text command.

Configuration file to be shared by all Cisco SIP IP Phone 7940s and Cisco SIP IP Phone 7960s. This file is automatically generated by the router through the source-address and is placed in router memory. The SIPDefault.cnf file contains the IP address that the phones use to register for service, using the Session Initiation Protocol (SIP).

Cisco SIP configuration profile for a particular Cisco IP Phone 7940 or Cisco IP Phone 7960 as indicated by the <mac-address>. This file is generated by using the create profile command.

Configuration file to be shared by all Cisco SIP IP Phone 7940s and Cisco SIP IP Phone 7960s. This file is generated by using the create profile command.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Id&lt;mac-address&gt;</td>
<td>Cisco SIP configuration profile for a particular Cisco IP Phone 7905 or Cisco IP Phone 7912G as indicated by the &lt;mac-address&gt;. This file is generated by using the create profile command.</td>
</tr>
<tr>
<td>Id&lt;mac-address&gt;.txt</td>
<td>ASCII text file of a Cisco SIP configuration profile for a particular Cisco IP Phone 7905 or Cisco IP Phone 7912G as indicated by the &lt;mac-address&gt;. This file is generated by using the file text command.</td>
</tr>
<tr>
<td>SIPDefault.cnf</td>
<td>Configuration file to be shared by all Cisco SIP IP Phone 7940s and Cisco SIP IP Phone 7960s. This file is automatically generated by the router through the source-address and is placed in router memory. The SIPDefault.cnf file contains the IP address that the phones use to register for service, using the Session Initiation Protocol (SIP).</td>
</tr>
<tr>
<td>SIP&lt;mac-address&gt;.cnf</td>
<td>Cisco SIP configuration profile for a particular Cisco IP Phone 7940 or Cisco IP Phone 7960 as indicated by the &lt;mac-address&gt;. This file is generated by using the create profile command.</td>
</tr>
<tr>
<td>syncinfo.xml</td>
<td>Configuration file to be shared by all Cisco SIP IP Phone 7940s and Cisco SIP IP Phone 7960s. This file is generated by using the create profile command.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Description</th>
<th>Related Commands</th>
</tr>
</thead>
<tbody>
<tr>
<td>Generates the configuration profiles required for SIP phones.</td>
<td>create profile (voice register global)</td>
</tr>
<tr>
<td>Performs a complete reboot of one phone associated with a Cisco CME router.</td>
<td>reset (voice register dn)</td>
</tr>
<tr>
<td>Performs a complete reboot of one or all phones associated with a Cisco CME router.</td>
<td>reset (voice register pool)</td>
</tr>
<tr>
<td>Generates an ASCII format text file of the Cisco SIP configuration profile for Cisco IP Phone 7905s and 7905Gs, Cisco IP phone 7912s and 7912Gs, Cisco ATA-186s, and Cisco ATA-188s.</td>
<td>text file (voice register global)</td>
</tr>
<tr>
<td>Specifies the directory to which the provisioning file for SIP phones in a Cisco CallManager Express (Cisco CME) system will be written.</td>
<td>tftp-path (voice register global)</td>
</tr>
<tr>
<td>Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.</td>
<td>voice register global</td>
</tr>
</tbody>
</table>
shutdown(telephony-service)

To shut down the Skinny Client Control Protocol (SCCP) server listening socket, use the shutdown command in telephony-service configuration mode. To enable service, use the no form of this command.

```
shutdown
no shutdown
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

No shutdown is enabled

**Command Modes**

- Telephony-service configuration (config-telephony)
- Group configuration (conf-tele-group)
- Call-manager-fallback configuration (conf-cm-fallback)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0 Cisco Unified SRST 8.0</td>
<td>This command was introduced</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The shutdown command allows you to shut down the SCCP server listening sockets when you want to change or remove the IP address setup on your system. For example, if you have IPv6 address and you want to change the IP address setup to dual stack (IPv4 and IPv6) you can use the shutdown command.

**Examples**

The following example shows SCCP server listening sockets being shut down under telephony-service.

```
Router(config-telephony)#shutdown
```

The following example shows SCCP server listening sockets being shut down for group 2 (under group mode) in telephony service.

```
Router(config-telephony)#group 2
Router(conf-tele-group)#shutdown
```

The following example shows SCCP server listening sockets being shut down under call-manager-fallback mode.

```
Router(config-telephony)#group 2
Router(conf-cm-fallback)#shutdown
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>protocol-mode</td>
<td>Allows you to configure a preferred IP address mode for SCCP IP phones.</td>
</tr>
<tr>
<td>ip source address</td>
<td>Identifies the IP address and port through which IP phones communicate with a Cisco Unified CME router.</td>
</tr>
</tbody>
</table>
sip-prefix

To add "SIP_" prefix in the locale names while populating the configuration files for this phone type in fast track mode, use the **sip-prefix** command in global configuration mode. To disable, use the **no** form of this command.

**sip-prefix**
**no sip-prefix**

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
sip-prefix is enabled.

**Command Modes**
Router (config-register-pooltype) #sip-prefix

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was introduced</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
The sip-prefix command allows you to add “SIP_” prefix in the locale names of the configuration files. For example, DX650 phones do not require “SIP_” prefix in their locale names. However, other phone models require “SIP_” prefix in locale names. Use this command while adding new phones in fast-track mode based on their locale file format.

**Examples**
The following example shows how to configure “SIP_” prefix on the endpoint Cisco Unified IP Phone 7811.

Router(config)#voice register pool-type 7811
Router(config-register-pooltype)#sip-prefix

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>num -lines</td>
<td>Defines number of lines supported by the phone.</td>
</tr>
<tr>
<td>phoneload -support</td>
<td>Defines the phone support for Phoneload.</td>
</tr>
<tr>
<td>reference -pooltype</td>
<td>Reference pooltype to inherit the properties used in fast-track configuration.</td>
</tr>
</tbody>
</table>
To enable Single Number Reach (SNR) on an extension of an SCCP IP phone, use the `snr` command in ephone-dn configuration mode. To disable SNR on the extension, use the `no` form of this command.

```
snr e164-number delay seconds timeout seconds [ cfwd-noan extension-number ]
no snr
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>e164-number</code></td>
<td>E.164 telephone number to ring if IP phone extension does not answer.</td>
</tr>
<tr>
<td><code>delay seconds</code></td>
<td>Sets the number of seconds that the call rings the IP phone before ringing</td>
</tr>
<tr>
<td></td>
<td>the remote phone. Range: 0 to 10. Default: disabled.</td>
</tr>
<tr>
<td><code>timeout seconds</code></td>
<td>Sets the number of seconds that the call rings after the configured delay.</td>
</tr>
<tr>
<td></td>
<td>Call continues to ring for this length of time on the IP phone even if the</td>
</tr>
<tr>
<td></td>
<td>remote phone answers the call. Range: 5 to 60. Default: disabled.</td>
</tr>
<tr>
<td><code>cfwd-noan</code></td>
<td>(Optional) Forwards the call to this target number if the phone does not</td>
</tr>
<tr>
<td><code>extension-number</code></td>
<td>answer after both the <code>delay</code> and <code>timeout</code> seconds have expired. This is</td>
</tr>
<tr>
<td></td>
<td>typically the voice mail number.</td>
</tr>
</tbody>
</table>

### Command Default

Single Number Reach is not enabled on the extension.

### Command Modes

Ephone-dn configuration (config-ephone-dn)

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco UnifiedCME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command enables the SNR feature on the extension. The SNR feature allows users to answer incoming calls on their desktop IP phone or at a remote destination and to pick up in-progress calls on the desktop phone or the remote destination without losing the connection. If an incoming call to this extension is answered immediately, the call is treated as a normal call.

If the call is not answered within the number of seconds set with the `delay` keyword, Cisco Unified CME rings the remote number while continuing to ring the SNR extension. If the call is answered by the desktop IP phone within the number of seconds set with the `timeout` keyword, the call to the remote number is disconnected. If the call is answered on the IP phone, the user can send the call to the remote phone by pressing the Mobility soft key.

If the call is not answered by the IP phone within the number of seconds set with the `timeout` keyword, the ringing call appearance on the IP phone is deleted. This call is marked as hold state on the IP phone. If the user answers the call on the remote phone, the user can pull back the call to the IP phone by pressing the Resume soft-key.
**Examples**

The following example shows extension 1001 is enabled for SNR. After a call rings at this number for 5 seconds, the call also rings at the remote number 4085550133. The call continues ringing on both phones for 15 seconds. If the call is not answered after a total of 20 seconds, the call no longer rings and is forwarded to the voice-mail number 2001.

```plaintext
ephone-dn 10
data
number 1001
mobility
snr 4085550133 delay 5 timeout 15 cfwd-noan 2001
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>mobility</strong></td>
<td>Enables the Mobility feature on an extension of an SCCP IP phone.</td>
</tr>
<tr>
<td><strong>number</strong></td>
<td>Associates a telephone or extension number with an ephone-dn.</td>
</tr>
<tr>
<td><strong>softkeys connected</strong></td>
<td>Modifies the order and type of soft keys that display on an IP phone during the connected call state.</td>
</tr>
<tr>
<td><strong>softkeys idle</strong></td>
<td>Modifies the order and type of soft keys that display on an IP phone during the idle call state.</td>
</tr>
</tbody>
</table>
snr (voice register dn)

To enable the Single Number Reach (SNR) feature on an extension of a Cisco Unified SIP IP phone, use the `snr` command in voice register dn configuration mode. To disable the SNR feature on the extension, use the `no` form of the command.

```
snr e164-number delay seconds timeout seconds [cfwd-noan extension-number]
```

**no snr**

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>e164-number</strong></td>
<td>E.164 telephone number to call when the Cisco Unified SIP IP phone extension does not answer.</td>
</tr>
<tr>
<td><strong>delay seconds</strong></td>
<td>Sets the number of seconds that the Cisco Unified SIP IP phone rings when called. When the time delay is reached, the call is transferred to the PSTN phone and the SNR directory number. Range: 0 to 30. Default: 5.</td>
</tr>
<tr>
<td><strong>timeout seconds</strong></td>
<td>Sets the number of seconds that the Cisco Unified SIP IP phone rings after the configured time delay. When the timeout value is reached, no call is displayed on the phone. You have to use the Resume soft key to pull back or the Mobility soft key to send the call to a mobile phone. Range: 30 to 60. Default: 60. <strong>Note</strong> When the default is enabled, the Cisco Unified SIP IP phone continues to ring for 60 seconds even if the remote phone answers the call.</td>
</tr>
<tr>
<td><strong>cfwd-noan extension-number</strong></td>
<td>(Optional) Forwards the call to the extension number when the phone does not answer after both the time delay and timeout values are reached. The extension number is typically the voice mail number. <strong>Note</strong> This option is not supported for calls from FXO trunks because the calls connect immediately.</td>
</tr>
</tbody>
</table>

**Command Default**

The SNR feature is not enabled on the extension of a Cisco Unified SIP IP phone.

**Command Modes**

Voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `snr` command to enable the SNR feature on an extension of a Cisco Unified SIP IP phone.

The SNR feature allows you to answer incoming calls on your desktop IP phones or at a remote destination. It also allows you to pick up in-progress calls on a desktop phone or at a remote destination without losing the connection. If an incoming call to the extension is answered immediately, the call is treated as a normal call.

If the call is not answered within the number of seconds set with the `delay` keyword, Cisco Unified CME rings the remote number while continuing to ring the SNR extension. If the call is answered by the desktop IP phone within the number of seconds set with the `timeout` keyword, the call to the remote number is disconnected. If the call is answered on the IP phone, you can send the call to the remote phone by pressing the Mobility soft key.
If the call is not answered by the IP phone within the number of seconds set with the `timeout` keyword, the call is displayed on the IP phone as being in the hold state. If the user answers the call on the remote phone, the user can pull back the call to the IP phone by pressing the Resume soft key.

**Examples**

The following example shows that extension 1004 is enabled for SNR. After a call rings at this number for one second, the call also rings at the remote number 9900. The call continues ringing on both phones for 10 seconds. If the call is not answered after a total of 11 seconds, the call no longer rings and is forwarded to the voice-mail number 1007.

```
Router(config)# voice register dn 3
Router(config-register-dn)# number 1004
Router(config-register-dn)# name John Smith
Router(config-register-dn)# mobility
Router(config-register-dn)# snr calling-number local
Router(config-register-dn)# snr 9900 delay 1 timeout 10 cfwd-noan 1007
Router(config-register-dn)# snr ring-stop
Router(config-register-dn)# snr answer-too-soon 2
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>mobility (voice register dn)</code></td>
<td>Enables the Mobility feature on an extension of a Cisco Unified SIP IP phone.</td>
</tr>
<tr>
<td><code>snr answer-too-soon (voice register dn)</code></td>
<td>Sets the time in which SNR calls are prevented from being diverted to the voice mailbox of a mobile phone.</td>
</tr>
<tr>
<td><code>snr calling-number local (voice register dn)</code></td>
<td>Replaces the calling party number displayed on the configured mobile phone with the local SNR number.</td>
</tr>
<tr>
<td><code>snr ring-stop (voice register dn)</code></td>
<td>Ends the ringing on a Cisco Unified SIP IP phone after the Single SNR call is answered on the configured mobile phone.</td>
</tr>
</tbody>
</table>
snr answer-too-soon

To set the SNR answer to soon timer, use the snr answer-too-soon command in ephone-dn mode. To reset the default, use the no form of the command.

```
  snr answer-too-soon time
  no snr answer-too-soon
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>time</code></td>
<td>Time, in seconds. Range: 1 to 5.</td>
</tr>
</tbody>
</table>

**Command Default**

No answer too soon timer is set.

**Command Modes**

- Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to enable timer for answering the call on an SNR mobile phone. You can set a timer from 1 to 5 seconds. If the call is answered within the timer, the mobile leg is disconnected.

**Examples**

```
Router(config) ephone-dn 10
Router(config-ephone-dn)# snr answer-too-soon 4
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>snr</td>
<td>Enables SNR on the extension of an SCCP IP phone.</td>
</tr>
</tbody>
</table>
snr answer-too-soon (voice register dn)

To set the time in which Single Number Reach (SNR) calls are prevented from being diverted to the voice mailbox of a mobile phone, use the `snr answer-too-soon` command in voice register dn configuration mode. To allow SNR calls to be diverted to the voice mailbox, use the `no` form of the command.

```
    snr answer-too-soon time
    no snr answer-too-soon
```

**Syntax Description**

```
time
```

Time, in seconds. Range: 1 to 5.

**Command Default**

No answer-too-soon time is set. Calls may be diverted to the voice mailbox when a user’s mobile phone is not answered or is turned off.

**Command Modes**

Voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the `snr answer-too-soon` command to set the time in which SNR calls are prevented from being diverted to the voice mailbox of a mobile phone. When the call is diverted to the voice mailbox within the set time, the mobile phone call leg is disconnected.

**Examples**

The following example shows how SNR calls are prevented from being diverted to the voice mailbox of a mobile phone for 2 seconds:

```
Router(config)# voice register dn 3
Router(config-register-dn)# number 1004
Router(config-register-dn)# name John Smith
Router(config-register-dn)# mobility
Router(config-register-dn)# snr calling-number local
Router(config-register-dn)# snr 9900 delay 1 timeout 10
Router(config-register-dn)# snr ring-stop
Router(config-register-dn)# snr answer-too-soon 2
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>snr (voice register dn)</code></td>
<td>Enables the SNR feature on an extension of a Cisco Unified SIP IP phone.</td>
</tr>
</tbody>
</table>
**snr calling-number local**

To replace the calling-party number with the single number reach (SNR) extension number in calls forwarded to the remote phone, use the `snr calling-number local` command in ephone-dn configuration mode. To reset to the default, use the `no` form of this command.

```
snr calling-number local
no snr calling-number local
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Calling-party number is not replaced.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)

<table>
<thead>
<tr>
<th>Command History</th>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
<td></td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(T).</td>
<td></td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command replaces the original calling party number with the SNR extension number (local number) in the caller ID display for SNR calls forwarded to the remote phone. When the call is forwarded to the remote phone, such as a mobile phone, the caller ID shows the SNR number that the caller dialed, not the number of the original calling party.

**Examples**

The following example shows that the original calling party number is replaced by the SNR extension number 1234 when the call is forwarded to the mobile phone:

```
ephone-dn 1
number 1234
mobility
snr 408550123 delay 5 timeout 15 cfwd-noan 2001
snr calling-number local
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>calling-number local</td>
<td>Replaces a calling-party number and name with the forwarding-party number and name for all calls.</td>
</tr>
<tr>
<td>mobility</td>
<td>Enables the Mobility feature on an extension of an SCCP IP phone.</td>
</tr>
<tr>
<td>snr</td>
<td>Enables SNR on the extension of an SCCP IP phone.</td>
</tr>
</tbody>
</table>
snr calling-number local (voice register dn)

To replace the calling party number displayed on the configured mobile phone with the local Single Number Reach (SNR) number, use the `snr calling-number local` command in voice register dn configuration mode. To return to the default, use the `no` form of this command.

```
  snr calling-number local
  no snr calling-number local
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

The number of the calling party is displayed on the mobile phone configured to receive SNR calls.

**Command Modes**

Voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Examples**

The following example shows how the `snr calling-number local` command is used to display the local SNR number instead of the calling party’s number on the mobile phone:

```
Router(config)# voice register dn 3
Router(config-register-dn)# number 1004
Router(config-register-dn)# name John Smith
Router(config-register-dn)# mobility
Router(config-register-dn)# snr calling-number local
Router(config-register-dn)# snr 9900 delay 1 timeout 10
Router(config-register-dn)# snr ring-stop
Router(config-register-dn)# snr answer-too-soon 2
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>snr (voice register dn)</td>
<td>Enables the SNR feature on an extension of a Cisco Unified SIP IP phone.</td>
</tr>
</tbody>
</table>
snr mode

To set the mode for the Single Number Reach (SNR) directory number (DN), use the snr mode command in ephone-dn configuration mode. To return to the default, use the no form of this command.

**snr mode [virtual]**
**no snr mode**

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>virtual</strong></td>
<td>Enables the virtual mode for an SNR DN when it is unregistered or floating.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>Virtual mode is activated when the DN state remains up when it should be in the down state.</td>
</tr>
</tbody>
</table>

**Command Default**

No DN mode is set for the SNR feature.

**Command Modes**

Ephone-dn configuration (config-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

A virtual SNR DN is a DN not associated with any registered phone but is a number that can be called, have its calls forwarded to a preconfigured mobile phone, or put on an Auto Hold state when the mobile phone answers the call or the time delay is reached. In the Auto Hold state, the DN can either be floating or unregistered. A floating DN is a DN not configured with any phone while an unregistered DN is one associated with phones not registered to a Cisco Unified CME system.

A ringback tone is heard when a call is made to a virtual DN.

To enable the SNR feature, the SNR DN must be in the up state, the Mobility feature must be enabled, and the time delay or timeout value configured with the snr command must be reached.

**Examples**

The following example sets the virtual DN mode for SNR on ephone-dn 1:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# snr mode virtual
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-dn</td>
<td>Enters ephone-dn configuration mode to configure a DN for an IP phone line, intercom line, paging line, voice-mail port, or MWI.</td>
</tr>
<tr>
<td>snr</td>
<td>Enables SNR on an extension of a Cisco Unified SCCP IP phone.</td>
</tr>
</tbody>
</table>
**snr ring-stop**

To stop the IP phone from ringing after the SNR call is answered on a mobile phone, use the `snr ring-stop` command in ephone-dn configuration mode. To reset the default value, use the `no` form of the command.

```
  snr  ring-stop
  no  snr  ring-stop
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Phone continues to ring after the SNR call is answered on a mobile phone.

**Command Modes**

Ephone-dn configuration (conf-ephone-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to stop the IP phone from ringing after the SNR call is answered on a mobile phone.

**Examples**

```
Router(config-ephone-dn)10
Router(config-ephone-dn)#snr ring-stop
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>snr</td>
<td>Enables SNR on the extension of an SCCP IP phone.</td>
</tr>
</tbody>
</table>
**snr ring-stop (voice register dn)**

To end the ringing on a Cisco Unified SIP IP phone after the Single Number Reach (SNR) call is answered on the configured mobile phone, use the `snr ring-stop` command in voice register dn configuration mode. To allow the Cisco Unified SIP IP phone to continue ringing even after the SNR call has been answered, use the `no` form of the command.

```
  snr ring-stop
  no snr ring-stop
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

The Cisco Unified SIP IP phone continues to ring even after the SNR call is answered on a mobile phone.

**Command Modes**

Voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Examples**

The following example shows how to end the ringing on a Cisco Unified SIP IP phone:

```
Router(config)# voice register dn 3
Router(config-register-dn)# number 1004
Router(config-register-dn)# name John Smith
Router(config-register-dn)# mobility
Router(config-register-dn)# snr calling-number local
Router(config-register-dn)# snr 9900 delay 1 timeout 10
Router(config-register-dn)# snr ring-stop
Router(config-register-dn)# snr answer-too-soon 2
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>snr (voice register dn)</code></td>
<td>Enables the SNR feature on an extension of a Cisco Unified SIP IP phone.</td>
</tr>
</tbody>
</table>
softkeys alerting

To configure an ephone template for soft-key display during the alerting call stage, use the `softkeys alerting` command in ephone-template configuration mode. To remove a `soft key alerting` configuration, use the `no` form of this command.

```
softkeys alerting [Acct] [Callback] [Endcall]
no softkeys alerting
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Acct</td>
<td>(Optional) Soft-key name that appears on the IP phone during the alerting call stage. Short for “account code.” Provides access to configured accounts.</td>
</tr>
<tr>
<td>Callback</td>
<td>(Optional) Soft-key name that appears on the IP phone during the alerting call stage. Requests callback notification when a busy called line becomes free.</td>
</tr>
<tr>
<td>Endcall</td>
<td>(Optional) Soft-key name that appears on the IP phone during the alerting call stage. Ends the current call.</td>
</tr>
</tbody>
</table>

**Command Default**
The default soft keys for the alerting call stage and the order in which they appear on IP phones are, from first to last, Acct, Callback, and Endcall.

**Command Modes**
Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco CME Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>3.2</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
The alerting call stage is when the remote point is being notified of an incoming call, and the status of the remote point is being relayed to the caller as either ringback or busy.

The number and order of soft keys listed in the `softkeys alerting` correspond to the number and order of soft keys that will appear on IP phones.

**Examples**
In the following example, ephone template 1 is configured for the alerting stage and for the seized and connected call stages:

```
Router(config)# telephony-service
Router(config-telephony)# ephone-template 1
Router(config-ephone-template)# softkeys seized Redial Cfwdall Pickup
Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template</td>
<td>Applies an ephone template to an ephone.</td>
</tr>
<tr>
<td>(ephone)</td>
<td></td>
</tr>
<tr>
<td>softkeys connected</td>
<td>Configures an ephone template for soft-key display during the connected call stage.</td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>softkeys idle</td>
<td>Configures an ephone template for soft-key display during the idle call stage.</td>
</tr>
<tr>
<td>softkeys seized</td>
<td>Configures an ephone template for soft-key display during the seized call stage.</td>
</tr>
</tbody>
</table>
softkeys connected (voice register template)

To modify the soft key display during the connected call state on Cisco Unified SIP IP phones, use the `softkeys connected` command in voice register template configuration mode. To return to the default, use the `no` form of this command.

```
softkeys connected [ConfList] [Confrn] [Endcall] [Hold] [Park] [RmLstC] [Trnsfer] [iDivert] [HLog]
```

```
no softkeys connected
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>ConfList</th>
<th>(Optional) Lists all the participants in a conference.</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Confrn</td>
<td>(Optional) Connects callers to a conference call. This soft key also enables ad-hoc conference creators to initiate a conference.</td>
</tr>
<tr>
<td></td>
<td>Endcall</td>
<td>(Optional) Ends the current call.</td>
</tr>
<tr>
<td></td>
<td>Hold</td>
<td>(Optional) Places an active call on hold and resumes the call.</td>
</tr>
<tr>
<td></td>
<td>Park</td>
<td>(Optional) Places an active call on hold so it can be retrieved from another phone in the system.</td>
</tr>
<tr>
<td></td>
<td>RmLstC</td>
<td>(Optional) Removes the last conference participant.</td>
</tr>
<tr>
<td></td>
<td>Trnsfer</td>
<td>(Optional) Transfers active calls to another extension.</td>
</tr>
<tr>
<td></td>
<td>iDivert</td>
<td>(Optional) Immediately diverts a call to a voice-messaging system.</td>
</tr>
<tr>
<td></td>
<td>HLog</td>
<td>(Optional) Soft key that places a phone into not-ready status, in which it does not accept hunt-group calls. You must set the <code>hunt-group logout</code> command to HLog for this softkey to be functional. This key is a toggle; pressing it a second time returns the phone to ready status, in which it is available to receive calls.</td>
</tr>
</tbody>
</table>

**Command Default**

The default soft keys for the connected call state and the order in which they appear on Cisco Unified SIP IP phones are, from first to last: ConfList, Confrn, Endcall, Hold, Park, RmLstC, Trnsfer, iDivert, and HLog.

**Command Modes**

Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>The <code>Park</code> keyword was added.</td>
</tr>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>The <code>iDivert</code> keyword was added.</td>
</tr>
<tr>
<td>15.2(2)T</td>
<td>Cisco Unified CME 9.0</td>
<td>This command was modified. The syntax description for the Confrn soft key was updated. The <code>ConfList</code> and <code>RmLstC</code> keywords were added.</td>
</tr>
</tbody>
</table>
Usage Guidelines

The connected call state is when the connection to a remote point is established.

The number and order of soft keys used in this command correspond to the number and order of soft keys that appear on Cisco Unified SIP IP phones. Any soft key that is not explicitly specified with this command is disabled.

The ConfList and RmLastC soft keys are added in the connected state when hardware conference is enabled.

This command is not supported on the Cisco Unified 7905, 7912, 7940, and 7960 SIP IP Phones.

Examples

In the following example, Cisco Unified SIP IP phone template 1 is configured for the connected and seized call states:

```
Router(config)# voice register template 1
Router(config-register-temp)# softkeys seized Redial Cfwdall EndCall HLog
Router(config-register-temp)# softkeys connected Confrn Hold Endcall HLog
```

The following is a sample output from the `show voice register template` command. The output shows that the iDivert soft key is in connected state.

```
Router# show voice register template 1
Temp Tag 1
Config:
  Attended Transfer is enabled
  Blind Transfer is enabled
  Semi-attended Transfer is enabled
  Conference is enabled
  Caller-ID block is disabled
  DnD control is enabled
  Anonymous call block is disabled
  softkeys seized Redial Cfwdall EndCall HLog
  softkeys connected Confrn Hold Endcall HLog
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>softkeys hold (voice register template)</code></td>
<td>Modifies the soft key display on Cisco Unified SIP IP phones during the hold call state.</td>
</tr>
<tr>
<td><code>softkeys idle (voice register template)</code></td>
<td>Modifies the soft key display on Cisco Unified SIP IP phones during the idle call state.</td>
</tr>
<tr>
<td><code>softkeys seized (voice register template)</code></td>
<td>Modifies the soft key display on Cisco Unified SIP IP phones during the seized call state.</td>
</tr>
<tr>
<td><code>template (voice register pool)</code></td>
<td>Applies a phone template to a Cisco Unified SIP IP phone.</td>
</tr>
</tbody>
</table>
softkeys connected

To modify the order and type of soft keys that display on an IP phone during the connected call state, use the `softkeys connected` command in ephone-template configuration mode. To reset to the default, use the `no` form of this command.

```
softkeys connected [Acct] [ConfList] [Confrn] [Endcall] [Flash] [HLog] [Hold] [Join] [LiveRcd] [Mobility] [Park] [RmLstC] [Select] [TrnsfVM] [Trnsfer]
```

no softkeys connected

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Acct</td>
<td>(Optional) Soft key that provides access to configured accounts.</td>
</tr>
<tr>
<td>ConfList</td>
<td>(Optional) Soft key that lists all parties in a conference.</td>
</tr>
<tr>
<td>Confrn</td>
<td>(Optional) Soft key that connects callers to a conference call.</td>
</tr>
<tr>
<td>Endcall</td>
<td>(Optional) Soft key that ends the current call.</td>
</tr>
<tr>
<td>Flash</td>
<td>(Optional) Soft key that provides hookflash functionality for public switched telephony network (PSTN) services on calls connected to the PSTN via a foreign exchange office (FXO) port. Also called “hookflash.”</td>
</tr>
<tr>
<td>HLog</td>
<td>(Optional) Soft key that places a phone into not-ready status, in which it does not accept hunt-group calls. You must set the <code>hunt-group logout</code> command to HLog for this soft key to be visible. This key is a toggle; pressing it a second time returns the phone to ready status, in which it is available to receive calls.</td>
</tr>
<tr>
<td>Hold</td>
<td>(Optional) Soft key that places an active call on hold and resumes the call.</td>
</tr>
<tr>
<td>Join</td>
<td>(Optional) Soft key that joins an established call to conference.</td>
</tr>
<tr>
<td>LiveRcd</td>
<td>(Optional) Soft key that enables recording of a call.</td>
</tr>
<tr>
<td>Mobility</td>
<td>(Optional) Soft key that forwards the call to the PSTN number defined by the Single Number Reach (SNR) feature.</td>
</tr>
<tr>
<td>Park</td>
<td>(Optional) Soft key that places an active call on hold, so it can be retrieved from another phone in the system.</td>
</tr>
<tr>
<td>RmLstC</td>
<td>(Optional) Soft key that removes the last party added to the conference. This soft key only works for the conference creator.</td>
</tr>
<tr>
<td>Select</td>
<td>(Optional) Soft key that selects a call or a conference on which to take action.</td>
</tr>
<tr>
<td>TrnsfVM</td>
<td>(Optional) Soft key that transfers a call to a voice-mail extension number.</td>
</tr>
<tr>
<td>Trnsfer</td>
<td>(Optional) Soft key that transfers active calls to another extension.</td>
</tr>
</tbody>
</table>

### Command Default

The default soft keys for the connected call state and the order in which they appear on IP phones are, from first to last:

- With HLog support: Hold, EndCall, Trnsfer, Confrn, Acct, Flash, Park, HLog
• Without HLog support: Hold, EndCall, Trnsfr, Confrn, Acct, Flash, Park

**Command Modes**

Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco CME 3.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The HLog keyword was added.</td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>The ConfList, Join, RmLstC, and Select keywords were added.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>The LiveRcd and TrnsfVM keywords were added.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command with the LiveRcd and TrnsfVM keywords was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>The Mobility keyword was added.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The connected call state is when the connection to a remote point has been established.

Configure the ConfList, Join, and RmLstC soft keys for conferencing functions. These soft keys are supported for hardware-based conferencing only and require the appropriate DSP farm configuration.

**Note**

The ConfList (including the Remove, Update, and Exit soft keys within the ConfList function) and RmLstC soft keys do not work on the Cisco Unified IP Phone 7902, 7935, and 7936.

**Examples**

In the following example, ephone template 1 modifies the soft keys displayed for the seized, alerting, and connected call states:

```
Router(config)# ephone-template 1
Router(config-ephone-template)# softkeys seized Redial Cfwdall Pickup
Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone</td>
<td>Enters ephone configuration mode for an IP phone.</td>
</tr>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone.</td>
</tr>
<tr>
<td>fxo-hook-flash</td>
<td>Enables display of the Flash soft key.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td><strong>hunt-group logout</strong></td>
<td>Enables separate handling of DND and HLog functionality for hunt-group agents and the display of the HLog soft key on phones.</td>
</tr>
<tr>
<td><strong>softkeys alerting</strong></td>
<td>Modifies the soft-key display for the alerting call state.</td>
</tr>
<tr>
<td><strong>softkeys idle</strong></td>
<td>Modifies the soft-key display for the idle call state.</td>
</tr>
<tr>
<td><strong>softkeys ringing</strong></td>
<td>Modifies the soft-key display for the ringing call state.</td>
</tr>
<tr>
<td><strong>softkeys seized</strong></td>
<td>Modifies the soft-key display for the seized call state.</td>
</tr>
</tbody>
</table>
softkeys hold

To configure an ephone template to modify soft-key display during the call-hold call stage, use the `softkeys hold` command in ephone-template configuration mode. To remove a `softkeys hold` configuration, use the `no` form of this command.

```
softkeys hold [Join] [Newcall] [Resume] [Select]
no softkeys hold
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Join</td>
<td>(Optional) Soft-key name that appears on an IP phone during the hold call stage. Joins an established call to a conference.</td>
</tr>
<tr>
<td>Newcall</td>
<td>(Optional) Soft-key name that appears on an IP phone during the hold call stage. Opens a line on a speaker phone to place a new call.</td>
</tr>
<tr>
<td>Resume</td>
<td>(Optional) Soft-key name that appears on an IP phone during the hold call stage. Reconnects with the call on hold.</td>
</tr>
<tr>
<td>Select</td>
<td>(Optional) Soft-key name that appears on an IP phone during the hold call stage. Selects a call or a conference on which to take action.</td>
</tr>
</tbody>
</table>

**Command Default**

The default soft keys for the hold call stage and the order in which they appear on IP phones are alphabetical, from first to last, Join, Newcall, Resume, and Select.

**Command Modes**

Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>The <strong>Join</strong> and <strong>Select</strong> keywords were added.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command with the <strong>Join</strong> and <strong>Select</strong> keywords was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

You reach the call-hold state by pressing the Hold soft key while you are in the connected state. From the hold state, you can press Resume to return to the connected state or NewCall to start another call, leaving the original call in the call-hold state.

The number and order of soft keys listed in the `softkeys hold` correspond to the number and order of soft keys that will appear on IP phones.

Configure the Join and Select soft keys for conferencing functions. These soft keys are supported for hardware-based conferencing only and require the appropriate DSP farm configuration.

**Examples**

In the following example, ephone template 1 is configured for the idle, alerting, connected, and hold call stages. It is applied to ephone 25. When ephone 25 has a call on hold, the only soft key that will be available is the Resume soft key.
```bash
Router(config)# telephony-service
Router(config-telephony)# ephone-template 1

Router(config-ephone-template)# softkeys idle Redial Cfwdall Pickup
Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
Router(config-ephone-template)# exit
Router(config)# ephone 25
Router(config-ephone)# button 1:39
Router(config-ephone)# ephone-template 1
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone</td>
<td>Enters ephone configuration mode for an IP phone.</td>
</tr>
<tr>
<td>ephone-template</td>
<td>Declares and names an ephone template to configure IP phone soft-key display and enters ephone-template configuration mode</td>
</tr>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone.</td>
</tr>
<tr>
<td>softkeys alerting</td>
<td>Configures an ephone template for soft-key display during the alerting call stage.</td>
</tr>
<tr>
<td>softkeys connected</td>
<td>Configures an ephone template for soft-key display during the connected call stage.</td>
</tr>
<tr>
<td>softkeys idle</td>
<td>Configures an ephone template for soft-key display during the idle call stage.</td>
</tr>
<tr>
<td>softkeys seized</td>
<td>Configures an ephone template for soft-key display during the seized call stage.</td>
</tr>
</tbody>
</table>
softkeys idle

To modify the order and type of soft keys that display on an IP phone during the idle call state, use the `softkeys idle` command in ephone template configuration mode. To reset to the default, use the `no` form of this command.

```
softkeys idle [Cfwdall] [ConfList] [Dnd] [Gpickup] [HLog] [Join] [Login] [Mobility] [Newcall] [Pickup] [Redial] [RmLstC]
no softkeys idle
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>Cfwdall</code></td>
<td>(Optional) Soft key that forwards all calls.</td>
</tr>
<tr>
<td><code>ConfList</code></td>
<td>(Optional) Soft key that lists all parties in a conference.</td>
</tr>
<tr>
<td><code>Dnd</code></td>
<td>(Optional) Soft key that enables the Do-Not-Disturb features. This key is a toggle; pressing it a second time disables DND.</td>
</tr>
<tr>
<td><code>Gpickup</code></td>
<td>(Optional) Soft key that selectively picks up calls coming into a phone number that is a member of a pickup group.</td>
</tr>
<tr>
<td><code>HLog</code></td>
<td>(Optional) Soft key that places a phone into not-ready status, in which it does not accept hunt-group calls. You must set the <code>hunt-group logout</code> command to HLog for this soft key to be visible. This key is a toggle; pressing it a second time returns the phone to ready status, in which it is available to receive calls.</td>
</tr>
<tr>
<td><code>Join</code></td>
<td>(Optional) Soft key that joins an established call to a conference.</td>
</tr>
<tr>
<td><code>Login</code></td>
<td>(Optional) Soft key that provides personal identification number (PIN)-controlled access to restricted phone features.</td>
</tr>
<tr>
<td><code>Mobility</code></td>
<td>(Optional) Soft key that enables Single Number Reach (SNR) feature. This key is a toggle; pressing it a second time disables SNR.</td>
</tr>
<tr>
<td><code>Newcall</code></td>
<td>(Optional) Soft key that opens a line on a speaker phone to place a new call.</td>
</tr>
<tr>
<td><code>Pickup</code></td>
<td>(Optional) Soft key that selectively picks up calls coming into another extension.</td>
</tr>
<tr>
<td><code>Redial</code></td>
<td>(Optional) Soft key that redials the last number dialed.</td>
</tr>
<tr>
<td><code>RmLstC</code></td>
<td>(Optional) Soft key that removes the last party added to the conference. This soft key removes the last party only when the conference creator presses it.</td>
</tr>
</tbody>
</table>

### Command Default

The default soft keys for the idle call stage and the order in which they appear on IP phones are:

- FXO Trunk: Redial, NewCall, DoNotDisturb
- With HLog support: Redial, NewCall, CfwdAll, CallPickUp, GrpCallPickUp, DoNotDisturb, Login, HLog
- Without HLog support: Redial, NewCall, CfwdAll, CallPickUp, GrpCallPickUp, DoNotDisturb, Login

### Command Modes

Ephone-template configuration (config-ephone-template)
Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco CME 3.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The HLog keyword was added.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>The HLog keyword was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>The ConfList, Join, and RmLstC keywords were added.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command with the ConfList, Join, and RmLstC keywords was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>The Mobility keyword was added.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

The idle calling stage occurs before a call is made and after a call is complete.

The number and order of soft keys listed in the softkeys idle command correspond to the number and order of soft keys on IP phones.

Configure the ConfList, Join, and RmLstC soft keys for conferencing functions. These soft keys are supported for hardware-based conferencing only and require the appropriate DSP farm configuration.

Note

The ConfList (including the Remove, Update, and Exit soft keys within the ConfList function) and RmLstC soft keys do not work on the Cisco Unified IP Phone 7902 and Cisco Unified IP Phone 7935 and 7936.

Examples

In the following example, ephone template 1 is configured for the idle stage and for the alerting and connected call stages:

```plaintext
Router(config)# ephone-template 1
Router(config-ephone-template)# softkeys idle Redial Cfwdall Pickup
Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confmn Hold Endcall
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone</td>
<td>Enters ephone configuration mode for an IP phone.</td>
</tr>
<tr>
<td>ephone-template</td>
<td>Creates an ephone template.</td>
</tr>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone.</td>
</tr>
<tr>
<td>hunt-group logout</td>
<td>Enables separate handling of DND and HLog functionality for hunt-group agents and the display of an HLog soft key on phones.</td>
</tr>
<tr>
<td>softkeys alerting</td>
<td>Configures soft-key display during the alerting call state.</td>
</tr>
</tbody>
</table>
### Command Reference

#### softkeys idle

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>softkeys connected</td>
<td>Configures soft-key display during the connected call state.</td>
</tr>
<tr>
<td>softkeys seized</td>
<td>Configures soft-key display during the seized call state.</td>
</tr>
</tbody>
</table>
softkeys idle (voice register template)

To modify the soft-key display during the idle call state on SIP phones, use the `softkeys idle` command in voice register template configuration mode. To remove a `softkeys idle` configuration, use the `no` form of this command.

```
softkeys idle [Cfwdall] [DND] [Gpickup] [Newcall] [Pickup] [Redial] [HLog]
no softkeys idle
```

### Syntax Description

- **Cfwdall** (Optional) Soft key for “call forward all.” Forwards all calls.
- **DND** (Optional) Soft key that enables the Do-Not-Disturb feature.
- **Gpickup** (Optional) Soft key that allows a user to pickup a call that is ringing on another phone.
- **Newcall** (Optional) Soft key that opens a line on a speakerphone to place a new call.
- **Pickup** (Optional) Soft key that allows a user to pickup a call that is ringing on another phone that is a member of the same pickup group.
- **Redial** (Optional) Soft key that redials the last number dialed.
- **HLog** (Optional) Soft key that places a phone into not-ready status, in which it does not accept hunt-group calls. You must set the `hunt-group logout` command to HLog for this softkey to be functional. This key is a toggle; pressing it a second time returns the phone to ready status, in which it is available to receive calls.

### Command Default

The default soft keys for the idle call state and the order in which they appear on SIP phones are, from first to last, Redial, Newcall, Cfwdall, and HLog.

### Command Modes

Voice register template configuration (config-register-temp)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>The DND keyword was added.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Everest 16.4.1</td>
<td>Cisco Unified CME 11.6</td>
<td>HLog Softkey support was introduced.</td>
</tr>
<tr>
<td>15.6(3)M1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Usage Guidelines

The idle calling state occurs before a call is made and after a call is complete.
The number and order of soft keys used in this command correspond to the number and order of soft keys that appear on SIP phones. Any soft key that is not explicitly specified with this command is disabled if this command is used to change the default soft keys.

This command is not supported on the Cisco Unified IP Phone 7905, 7912, 7940, or 7960.

**Examples**

In the following example, SIP phone template 1 is configured for the idle and connected call states:

```
Router(config)# voice register template 1
Router(config-register-template)# softkeys idle Redial Cfwdall HLog
Router(config-register-template)# softkeys connected Confrn Hold Endcall HLog
```
**softkeys personal-conf-user (voice register template)**

To enable a personal user softkey template for Cisco IP Conference Phones 7832 and 8832, use the `softkeys personal-conf-user` command in voice register template configuration mode. To switch to a public user softkey template, use the `no` form of this command.

```
softkeys personal-conf-user
no softkeys personal-conf-user
```

**Command Default**

By default, the CLI command `softkeys personal-conf-user` is disabled. Hence, the Cisco IP Conference Phones 7832 and 8832 support the public user softkey template if the command is not configured.

**Command Modes**

Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IOS XE Fuji 16.9.1</td>
<td>Unified CME 12.3</td>
<td>A personal and public softkey template support was introduced for new Softkeys introduced as part of Unified CME 12.3 Release for Cisco IP Conference Phone 7832 and 8832.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The CLI command `softkeys personal-conf-user` is an optional configuration that is required only when the phone template is applied to Cisco IP Conference Phones 7832 and 8832. If no configuration is provided, then the default configuration of public user softkey template is applied. When the CLI command is enabled, the personal softkey template is applied to the conference phone (Only for Cisco IP Conference Phone 7832 and 8832). When the command is not enabled, the public softkey template is applied to the conference phone (Only for Cisco IP Conference Phone 7832 and 8832). As compared to public softkey user template, the following softkeys are additionally supported in a personal user softkey template for various phone states:

- Messages
- CfwdAll
- DND
- Redial

**Examples**

In the following example, Cisco IP Conference Phone 7832 is configured for the personal softkeys template:

```
Router(config)# voice register template 7
Router(config-register-template)# softkeys personal-conf-user
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>softkeys connected (voice register template)</code></td>
<td>Modifies the soft-key display on SIP phones during the connected call state.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>------------------------------------------------------------------</td>
</tr>
<tr>
<td>softkeys hold (voice register template)</td>
<td>Modifies the soft-key display on SIP phones during the hold call state.</td>
</tr>
<tr>
<td>softkeys idle (voice register template)</td>
<td>Modifies the soft-key display on SIP phones during the idle call state.</td>
</tr>
<tr>
<td>softkeys seized (voice register template)</td>
<td>Modifies the soft-key display on SIP phones during the seized call state.</td>
</tr>
<tr>
<td>template (voice register pool)</td>
<td>Applies a phone template to a SIP phone.</td>
</tr>
</tbody>
</table>
softkeys remote-in-use

To modify the order and type of soft keys that display on the IP phone during the remote-in-use call state, use the `softkeys remote-in-use` command in ephone-template configuration mode. To reset to the default, use the `no` form of this command.

```
softkeys remote-in-use [CBarge] [Newcall]
no softkeys remote-in-use
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>CBarge</td>
<td>(Optional) Soft key that allows a user to barge into a call on a shared octo-line directory number.</td>
</tr>
<tr>
<td>Newcall</td>
<td>(Optional) Soft key that opens a line on a speakerphone to place a new call.</td>
</tr>
</tbody>
</table>

**Command Default**

The default soft keys for the remote-in-use call state and the order in which they appear on IP phones are Newcall, CBarge.

**Command Modes**

Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The remote-in-use call state is when another phone is connected to a call on an octo-line directory number shared by this phone.

**Examples**

In the following example, ephone template 1 modifies the soft keys displayed for the alerting, connected, and remote-in-use call states:

```
Router(config)# ephone-template 1
Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
Router(config-ephone-template)# softkeys remote-in-use CBarge Newcall
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone.</td>
</tr>
<tr>
<td>softkeys alerting</td>
<td>Modifies the soft-key display for the alerting call state.</td>
</tr>
<tr>
<td>softkeys idle</td>
<td>Modifies the soft-key display for the idle call state.</td>
</tr>
<tr>
<td>softkeys seized</td>
<td>Modifies the soft-key display for the seized call state.</td>
</tr>
</tbody>
</table>
softkeys remote-in-use (voice register template)

To modify the soft-key display during the remote-in-use call state on SIP shared-line phones, use the softkeys remote-in-use command in voice register template configuration mode. To reset to the default, use the no form of this command.

softkeys remote-in-use [Barge] [Newcall] [cBarge]
no softkeys remote-in-use

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Barge</td>
<td>(Optional) Soft key that allows a user to join a call on a shared line.</td>
</tr>
<tr>
<td>Newcall</td>
<td>(Optional) Soft key that opens a line on a phone to place a new call.</td>
</tr>
<tr>
<td>cBarge</td>
<td>(Optional) Soft key that allows a user to join a call on a shared line and to turn the call into a conference call.</td>
</tr>
</tbody>
</table>

**Command Default**
The default soft keys for the remote-in-use call state and the order in which they appear on SIP phones are Barge, Newcall, cBarge.

**Command Modes**
Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
The remote-in-use call state is when another phone is connected to a call on a directory number shared by this phone.

**Examples**

In the following example, SIP phone template 1 modifies the soft keys displayed for the alerting, connected, and remote-in-use call states:

```
Router(config)# voice register template 1
Router(config-register-temp)# softkeys alerting Callback Endcall
Router(config-register-temp)# softkeys connected Confrn Hold Endcall
Router(config-register-temp)# softkeys remote-in-use CBarge Newcall
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>softkeys alerting (voice register template)</td>
<td>Modifies the soft-key display on SIP phones during the alerting call state.</td>
</tr>
<tr>
<td>softkeys idle (voice register template)</td>
<td>Modifies the soft-key display on SIP phones during the idle call state.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>softkeys seized (voice register template)</td>
<td>Modifies the soft-key display on SIP phones during the seized call state.</td>
</tr>
<tr>
<td>template (voice register pool)</td>
<td>Applies a phone template to a SIP phone.</td>
</tr>
</tbody>
</table>
softkeys ringin (voice register template)

To modify the soft-key display during the ringing call state on SIP phones, use the `softkeys ringIn` command in voice register template configuration mode. To remove the `softkeys ringIn` configuration, use the `no` form of this command.

```
softkeys ringIn [Answer] [DND] [iDivert] [HLog]
no softkeys ringIn
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Answer</td>
<td>(Optional) Soft key that picks up an incoming call.</td>
</tr>
<tr>
<td>DND</td>
<td>(Optional) Soft key that enables the Do Not Disturb feature.</td>
</tr>
<tr>
<td>HLog</td>
<td>(Optional) Soft key that places a phone into not-ready status, in which it does not accept hunt-group calls. You must set the <code>hunt-group logout</code> command to <code>HLog</code> for this soft key to be functional. This key is a toggle; pressing it a second time returns the phone to ready status, in which it is available to receive calls.</td>
</tr>
<tr>
<td>iDivert</td>
<td>(Optional) Immediately diverts a call to a voice-messaging system.</td>
</tr>
</tbody>
</table>

### Command Default

The following soft keys are displayed in alphabetical order, first to last, on IP phones during the ringIn call state: Answer, Dnd, HLog, and iDivert.

### Command Modes

Voice register template configuration (config-register-temp)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Everest 16.4.1</td>
<td>Cisco Unified CME 11.6</td>
<td>HLog Softkey support was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use this command to create a template in which you specify which soft keys are displayed, and in what order, on an IP phone during the ringIn call state. The ringing calling state is after a call is received and before the call is connected.

Any soft key that is not explicitly specified with this command is disabled if this command is used to change the default soft keys.

Configure the `Answer` keyword to enable a phone user to answer an incoming call on a line button that is unavailable; for example, if a line button is configured with a dual-line directory number and a call is holding on one channel of the directory number and another call is ringing on the second channel, the phone user can press the Answer soft key to pick up the incoming call on the second channel.

Configure the `DND` keyword to enable the phone user to place the phone into Do-Not-Disturb mode. Configure the Dnd soft key and the `hunt-group logout DND` command to enable the phone user to invoke DND mode and log the phone out of hunt groups in which it is a member.
Configure the **iDivert** keyword to immediately divert a call to a voice-messaging system.

Configure the **HLog** keyword to place a phone into not-ready status, in which it does not accept hunt-group calls.

To apply a voice register template to a phone, configure the **voice register template** command in voice register pool configuration mode.

### Examples

In the following example, SIP phone template 1 is configured for the ringing state, and for the alerting and connected call states:

```plaintext
Router(config)# voice register template 1
Router(config-register-template)# softkeys ringIn Answer Dnd Hlog iDivert

Router(config-register-template)# softkeys idle Newcall Redial Pickup Cfwdall HLog
Router(config-register-template)# softkeys connected Transfer Hold Endcall HLog
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>softkeys connected (voice register template)</code></td>
<td>Modifies the soft-key display on SIP phones during the connected call state.</td>
</tr>
<tr>
<td><code>softkeys hold (voice register template)</code></td>
<td>Modifies the soft-key display on SIP phones during the hold call state.</td>
</tr>
<tr>
<td><code>softkeys idle (voice register template)</code></td>
<td>Modifies the soft-key display on SIP phones during the idle call state.</td>
</tr>
<tr>
<td><code>softkeys seized (voice register template)</code></td>
<td>Modifies the soft-key display on SIP phones during the seized call state.</td>
</tr>
</tbody>
</table>
softkeys ringing

To configure an ephone template for soft-key display during the ringing call state, use the `softkeys ringing` command in ephone-template configuration mode. To remove the `softkeys ringing` configuration, use the `no` form of this command.

```
softkeys ringing [Answer] [Dnd] [HLog]
no softkeys ringing
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>Answer</code></td>
<td>(Optional) Soft-key name that appears on the IP phone during the ringing call state.</td>
</tr>
<tr>
<td><code>Dnd</code></td>
<td>(Optional) Soft-key name that appears on the IP phone during the ringing call state.</td>
</tr>
<tr>
<td><code>HLog</code></td>
<td>(Optional) Soft-key name that appears on the IP phone during the ringing call state.</td>
</tr>
</tbody>
</table>

### Command Default

The following soft keys are displayed in alphabetical order, first to last, on IP phones during the ringing call state: Answer, Dnd, HLog

### Command Modes

Ephone-template configuration (config-ephone-template)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XW</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use this command in ephone-template configuration mode to create a template in which you can specify which soft keys are displayed, and in what order, on an IP phone during the ringing call state. The ringing calling state is when a call is received and before the call is connected.

Any soft key that is not explicitly configured is disabled.

You can enter any of the keywords in any order. The number and order of soft keys listed in the `softkeys ringing` command corresponds to the number and order of soft keys that will appear on IP phones during the ringing call state.

Configure the `Answer` keyword with this command to enable a phone user to answer an incoming call on a line button that is unavailable; for example, if a line button is configured with a dual-line directory number and a call is holding on one channel of the directory number and another call is ringing on the second channel, the phone user can use the Answer soft key to pick up the incoming call on the second channel.

Configure the `HLog` keyword with this command to display the Hlog soft key during the ringing call state. To enable HLog softkey functionality during the call ringing state, you must also configure the `hunt-group logout HLog` command. If you configure the Hlog soft key and do not configure the `hunt-group logout HLog` command, the Hlog soft key appears on the phone screen but is not functional. The HLog softkey is a...
toggle for enabling or disabling the not-ready status, in which the directory number does not accept hunt-group calls.

Configure the Dnd keyword with this command to enable the phone user to place the phone into Do-Not-Disturb mode. Configure the Dnd soft key and the hunt-group logout DND command to enable the phone user to invoke DND mode and log the phone out of hunt groups in which it is a member.

To apply an ephone template to phone, configure the ephone-template (ephone) command in the ephone configuration mode.

Examples

In the following example, ephone template 1 is configured for the ringing state, and for the alerting and connected call states:

```
Router(config)# telephony-service
Router(config-telephony)# ephone-template 1

Router(config-ephone-template)# softkeys ringing Answer Dnd Hlog
Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
```

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dnd feature ring</td>
<td>Allows phone buttons configured with the feature-ring option to not ring when their phones are in do-not-disturb (DND) mode.</td>
</tr>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone.</td>
</tr>
<tr>
<td>hunt-group logout</td>
<td>Enables separate handling of DND and HLog functionality for hunt-group agents.</td>
</tr>
<tr>
<td>softkeys alerting</td>
<td>Configures an ephone template for soft-key display during the alerting call state.</td>
</tr>
<tr>
<td>softkeys connected</td>
<td>Configures an ephone template for soft-key display during the connected call state.</td>
</tr>
<tr>
<td>softkeys idle</td>
<td>Configures an ephone template for soft-key display during the idle call state.</td>
</tr>
<tr>
<td>softkeys seized</td>
<td>Configures an ephone template for the soft-key display during the seized call state.</td>
</tr>
</tbody>
</table>
softkeys seized

To modify the order and type of soft keys that display on an IP phone during the seized call state, use the `softkeys seized` command in ephone-template configuration mode. To remove a `softkeys seized` configuration, use the `no` form of this command.

```
softkeys seized [CallBack] [Cfwdall] [CWOff] [Endcall] [Gpickup] [HLog] [MeetMe] [Pickup] [Redial]
```

no softkeys seized

### Syntax Description

<table>
<thead>
<tr>
<th>Softkey</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>CallBack</td>
<td>(Optional) Soft key that requests callback notification when a busy called line becomes free.</td>
</tr>
<tr>
<td>Cfwdall</td>
<td>(Optional) Soft key that forwards all calls.</td>
</tr>
<tr>
<td>CWOff</td>
<td>(Optional) Soft key that disables Call Waiting.</td>
</tr>
<tr>
<td>Endcall</td>
<td>(Optional) Soft key that ends the current call.</td>
</tr>
<tr>
<td>Gpickup</td>
<td>(Optional) Soft key that selectively picks up calls coming into a phone number that is a member of a pickup group.</td>
</tr>
<tr>
<td>HLog</td>
<td>(Optional) Soft key that places a phone into not-ready status, in which it does not accept hunt-group calls. You must set the <code>hunt-group logout</code> command to HLog for this soft key to be visible. This key is a toggle; pressing it a second time returns the phone to ready status, in which it is available to receive calls.</td>
</tr>
<tr>
<td>MeetMe</td>
<td>(Optional) Soft key that initiates a meet-me conference.</td>
</tr>
<tr>
<td>Pickup</td>
<td>(Optional) Soft key that selectively picks up calls to another extension.</td>
</tr>
<tr>
<td>Redial</td>
<td>(Optional) Soft key that redials the last number dialed.</td>
</tr>
</tbody>
</table>

### Command Default

The default soft keys for the seized call stage and the order in which they appear on IP phones are:

- With HLog support: Redial, EndCall, CFwdAll, CallPickUp, GrpCallPickUp, CallBack, HLog
- Without HLog support: Redial, EndCall, CFwdAll, CallPickUp, GrpCallPickUp, CallBack

### Command Modes

Ephone-template configuration (config-ephone-template)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco CME 3.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The <code>HLog</code> keyword was added.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>The <code>HLog</code> keyword was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>The <code>MeetMe</code> keyword was added.</td>
</tr>
</tbody>
</table>
The seized calling stage is when a caller is attempting a call and has not yet been connected.

The number and order of soft keys listed in the `softkeys seized` command correspond to the number and order of soft keys on IP phones.

You must configure the MeetMe soft key to initiate a meet-me conference. Use this soft key for hardware conferencing only.

### Examples

In the following example, ephone template 1 modifies the soft keys in the seized, alerting, and connected call states:

```plaintext
Router(config)# telephony-service
Router(config-telephony)# ephone-template 1
Router(config-ephone-template)# softkeys seized Redial Cfwdall Pickup
Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone</td>
<td>Enters ephone configuration mode for an IP phone.</td>
</tr>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone.</td>
</tr>
<tr>
<td>hunt-group logout</td>
<td>Enables separate handling of DND and HLog functionality for hunt-group agents and the display of an HLog soft key on phones.</td>
</tr>
<tr>
<td>softkeys alerting</td>
<td>Modifies the soft keys that display during the alerting call stage.</td>
</tr>
<tr>
<td>softkeys connected</td>
<td>Modifies the soft keys that display during the connected call stage.</td>
</tr>
<tr>
<td>softkeys idle</td>
<td>Modifies the soft keys that display during the idle call stage.</td>
</tr>
</tbody>
</table>
softkeys seized (voice register template)

To modify the soft key display for the seized call state on Cisco Unified SIP IP phones, use the \texttt{softkeys seized} command in voice register template configuration mode. To return to the default, use the \texttt{no} form of this command.

\begin{verbatim}
softkeys seized [Cfwdall] [Endcall] [Gpickup] [MeetMe] [Pickup] [Redial]
no softkeys seized
\end{verbatim}

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cfwdall</td>
<td>(Optional) Appears on the Cisco Unified SIP IP phone during the seized call state. Short for “Call forward all.” Forwards all calls.</td>
</tr>
<tr>
<td>Endcall</td>
<td>(Optional) Appears on the Cisco Unified SIP IP phone during the seized call state. Ends the current call.</td>
</tr>
<tr>
<td>Gpickup</td>
<td>(Optional) Appears on the Cisco Unified SIP IP phone during the seized call state. Short for “Group call pick up.” Selectively picks up calls coming into a phone number that is a member of a pickup group.</td>
</tr>
<tr>
<td>MeetMe</td>
<td>(Optional) Appears on the Cisco Unified SIP IP phone during the seized call state. Short for “MeetMe conference.” Initiates a meet-me conference.</td>
</tr>
<tr>
<td>Pickup</td>
<td>(Optional) Appears on the Cisco Unified SIP IP phone during the seized call state. Short for “Call pick up.” Selectively picks up calls coming into another extension.</td>
</tr>
<tr>
<td>Redial</td>
<td>(Optional) Appears on the Cisco Unified SIP IP phone during the seized call state. Redials the last number dialed.</td>
</tr>
</tbody>
</table>

**Command Default**

The default soft keys for the seized call state and the order in which they appear on Cisco Unified SIP IP phones are, from first to last: Cfwdall, Endcall, Gpickup, MeetMe, Pickup, and Redial.

**Command Modes**

Voice register template configuration (config-register-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>15.2(2)T</td>
<td>Cisco Unified CME 9.0</td>
<td>This command was modified. The MeetMe keyword was added.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The seized calling state is when a caller goes offhook before any other action is taken.

The number and order of soft keys used in this command correspond to the number and order of soft keys that appear on Cisco Unified SIP IP phones. Any soft key that is not explicitly specified with this command is disabled.

The MeetMe soft key is added in the seized state when hardware conference is enabled.
This command is not supported on the Cisco Unified 7905, 7912, 7940, and 7960 SIP IP phones.

**Examples**

In the following example, Cisco Unified SIP IP phone template 1 is configured for the seized and connected call states:

```plaintext
Router(config)# voice register template 1
Router(config-register-template)# softkeys seized Redial Cfwdall
Router(config-register-template)# softkeys connected Confrn Hold Endcall
```
source-addr

To specify the IP address of the certification authority proxy function (CAPF) server on the Cisco Unified CME router, use the source-addr command in CAPF-server configuration mode. To return to the default, use the no form of this command.

```
source-addr  *ip-address*
no source-addr
```

**Syntax Description**

| *ip-address* | IP address of the Cisco Unified CME router. |

**Command Default**

No IP address is entered for the CAPF server in the router configuration.

**Command Modes**

CAPF-server configuration (config-capf-server)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication.

**Examples**

The following example identifies the IP address for the CAPF server as 10.10.10.1:

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```
**source-address (voice register global)**

To identify the IP address and port through which SIP phones communicate with a Cisco CallManager Express (Cisco Unified CME) router, use the `source-address` command in voice register global configuration mode. To disable the router from receiving messages from SIP phones, use the `no` form of this command.

```
source-address ip-address [port port | secondary ip-address]
no source-address ip-address
```

**Syntax Description**

- `ip-address` Preexisting router IP address, typically one of the addresses of the Ethernet port of the router.
- `port port` (Optional) TCP/IP port number to use for SIP. Range is 2000 to 9999. Default is 5060 for SIP phones.
- `secondary ip-address` Secondary router for Cisco Unified CME. TCP/IP port number is same as the primary Cisco Unified CME router.

**Command Default**

Port number for SIP: 5060

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>Cisco IOS XE Everest 16.4.1</td>
<td>Cisco Unified CME 11.6</td>
<td>This command was modified to add the keyword: secondary.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is a mandatory command. The Cisco CallManager Express router cannot communicate with the Cisco CME phones if the IP address is not provided. If the port number is not provided, the SIP default port for is 5060. The IP address is usually the IP address of the Ethernet port to which the phones are connected.

This command enables a router to receive messages from Cisco IP phones through the specified IP address and port.

For systems using ITS V2.1, Cisco CME 3.0, or later versions, the IP phones receive their initial configuration information and phone firmware from the TFTP server associated with the router. The TFTP server address obtained by the Cisco IP phones points to the router IP address. The Cisco IP phones transfer a configuration file called SIPDefault.cnf. This file is automatically generated by the router through the `source-address` and is placed in router memory. The SIPDefault.cnf file contains the IP address that the phones, using the Session Initiation Protocol (SIP), use to register for service. This IP address corresponds to a valid Cisco Unified CME router IP address (and may be the same as the router TFTP server address).

**Examples**

The following example shows how to set the IP source address and port:

```
Router(config)# voice register global
Router(config-register-global)# source-address 10.6.21.4 port 6000 secondary 10.6.50.6
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>create profile (voice register global)</strong></td>
<td>Generates the configuration profiles required for SIP phones.</td>
</tr>
<tr>
<td><strong>file text (voice register global)</strong></td>
<td>Generates ASCII text files for SIP phones.</td>
</tr>
<tr>
<td><strong>tftp-path (voice register global)</strong></td>
<td>Specifies the directory to which the provisioning file for SIP phones in a Cisco CallManager Express (Cisco Unified CME) system will be written.</td>
</tr>
<tr>
<td><strong>voice register global</strong></td>
<td>Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco Unified CME or Cisco SIP SRST environment.</td>
</tr>
</tbody>
</table>
speed-dial

To create speed-dial definitions for a Cisco Unified IP phone or analog phone that uses an analog telephone adaptor (ATA) in a Cisco Unified CME system, use the `speed-dial` command in ephone or ephone-template configuration mode. To disable a speed-dial definition, use the `no` form of this command.

```plaintext
speed-dial speed-tag digit-string [label label-text]
no speed-dial speed-tag
```

**Syntax Description**

<table>
<thead>
<tr>
<th>speed-tag</th>
<th>Unique sequence number that identifies a speed-dial definition during configuration tasks. Range is from 1 to 33.</th>
</tr>
</thead>
<tbody>
<tr>
<td>digit-string</td>
<td>Digits to be dialed when the speed-dial button is pressed on an IP phone or the digits to be dialed when the associated code is entered from an analog phone with an ATA device. For IP phones, if the first character of this string is the plus sign (+), this speed-dial number is locked and cannot be changed at the phone. If the only character in this string is a pound sign (#), a user-programmable speed-dial button with no speed-dial number attached is defined.</td>
</tr>
<tr>
<td>label</td>
<td>(Optional) String that contains identifying text to be displayed next to the speed-dial button. Enclose the string in quotation marks if the string contains a space.</td>
</tr>
</tbody>
</table>

**Command Default**

No speed-dial definitions are created.

**Command Modes**

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>The number of speed-dial definitions that can be created was increased from 4 to 33. The ability to program speed-dial numbers at the phone and the ability to lock speed-dial numbers were introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was modified to allow IP phones to access more speed-dial numbers than the number of available buttons on their phones and to allow analog phones to access up to 33 speed-dial numbers.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-template configuration mode.</td>
</tr>
</tbody>
</table>
Modification

This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Cisco Product

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

The speed-tag argument in this command is a unique identifier for a speed-dial definition on the phone that is being configured.

This command must be followed by a quick reboot of the phone using the restart command.

If you use an ephone template to apply a to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

This command defines speed-dial numbers that are local to the ephone that is being configured. The directory entry defines additional, systemwide speed-dial numbers.

**IP Phones**

For IP phones, speed-dial numbers can be defined by administrators using this command and the digit-string argument. The numbers are locked if the digit-string argument begins with a plus sign (+). Locked numbers cannot be changed at the phone. Speed-dial definitions without speed-dial numbers (those defined with only a pound sign) and speed-dial instances with unlocked digit-string arguments can be changed by users at their IP phones. Changes made to speed-dial definitions are saved in the router nonvolatile random-access memory (NVRAM) configuration after a timer-based delay.

On Cisco Unified IP phones, speed-dial definitions are assigned to available extension buttons that have not been assigned to extensions. Speed-dial definitions are assigned in the order of their identifier (tag) numbers. For example, if you define speed-dial 1, it is assigned to the first phone button that is available after the buttons that have been assigned to extensions. If you have used two buttons for extensions on a phone, speed-dial 1 is assigned to the third physical button on the phone. When you define speed-dial 2, it is assigned to the fourth physical button on the phone, and so on.

If more speed-dial definitions are created than are supported by the IP phone setup, the extra speed-dial configurations can be dialed from IP phones using this procedure:

1. With the phone on-hook, an IP phone user presses a two-digit speed-dial code (that is, 05 for the entry with tag 5). A new soft key, Abbr, appears in the phone display.
2. The phone user picks up the phone handset and presses the Abbr soft key. The full telephone number associated with the speed-dial tag is dialed.

Prior to Cisco IOS Releases 12.3(11)XL and 12.3(14)T, speed-dial entries that were in excess of the number of physical phone buttons available were ignored.

**Analog Phones**

Analog phone users who use a Cisco ATA-186, Cisco ATA-188, or Cisco VG 224 to connect to a Cisco Unified CME system use a different method to access speed-dial numbers. Analog phone users press the asterisk (*) key and the speed-dial identifier (tag number) to dial a speed-dial number. For instance, an analog phone user presses *1 to speed dial the number that has been programmed as speed-dial 1 on that ephone. Analog phones can have up to 33 local speed-dial numbers programmed by the system administrator. The numbers cannot be programmed from the phone.

Prior to Cisco IOS Releases 12.3(11)XL and 12.3(14)T, analog phones were limited to nine speed-dial numbers.)
Examples

The following example sets speed-dial button 2 to dial the phone user’s assistant at extension 5001 and locks the setting so that the phone user cannot change it at the phone:

```
Router(config)# ephone 23
Router(config-ephone)# speed-dial 2 +5001 label “Assistant”
```

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>directory entry</td>
<td>Adds a systemwide phone directory entry or speed-dial entry.</td>
</tr>
<tr>
<td>restart (ephone)</td>
<td>Performs a fast reboot of a single IP phone in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>restart (telephony-service)</td>
<td>Performs a fast reboot of one or all phones in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies template to ephone configuration.</td>
</tr>
</tbody>
</table>
speed-dial (voice logout-profile and voice user-profile)

To create speed-dial definitions in a user profile or logout profile for Extension Mobility in Cisco Unified CME, use the `speed-dial` command in voice user-profile configuration mode or voice logout-profile configuration mode. To disable a speed-dial definition, use the `no` form of this command.

```
speed-dial speed-tag number [label label] [blf]
no speed-dial speed-tag
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>speed-tag</strong></td>
</tr>
<tr>
<td><strong>number</strong></td>
</tr>
<tr>
<td><strong>label</strong></td>
</tr>
<tr>
<td><strong>blf</strong></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Command Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>No speed-dial definition is created.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Command Modes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice logout-profile configuration (config-logout-profile)</td>
</tr>
<tr>
<td>Voice user-profile configuration (config-user-profile)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Command History</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Cisco IOS Release</strong></td>
</tr>
<tr>
<td>12.4(11)XW</td>
</tr>
<tr>
<td>12.4(15)XY</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
</tr>
<tr>
<td>12.4(20)T</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Usage Guidelines</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use this command in voice user-profile configuration mode to create a speed-dial definition in a user profile for Extension Mobility. A user profile is downloaded to the IP phone when a user is logged into an IP phone that is registered in Cisco Unified CME and enabled for Extension Mobility.</td>
</tr>
<tr>
<td>Use this command in voice logout-profile configuration mode to create a speed-dial definition in a logout profile for Extension Mobility. A logout profile is downloaded to the IP phone when no user is logged into an IP phone that is registered in Cisco Unified CME and enabled for Extension Mobility.</td>
</tr>
<tr>
<td>For button appearance, Extension Mobility will associate directory numbers and then associates speed-dial definitions in the logout profile or user profile to phone buttons in a sequential manner. If the profile contains</td>
</tr>
</tbody>
</table>
more directory and speed-dial numbers than there are buttons on the physical phone to which the profile is downloaded, the remaining numbers in the profile are ignored.

On Cisco Unified IP phones, speed-dial definitions are assigned to available extension buttons that have not been assigned to extensions. Speed-dial definitions are assigned in the order of their identifier (tag) numbers, from 1 to 36.

**Examples**

The following example shows the configuration for a user profile to be downloaded when the a phone user logs into a Cisco Unified IP phone that is enabled for Extension Mobility. The lines and speed-dial buttons in this profile that are configured on an IP phone after the user logs in depend on the phone type. For example, if the user logs into a Cisco Unified IP Phone 7970, all buttons are configured according to voice-user profile 1. However, if the phone user logs into a Cisco Unified IP Phone 7960, all six lines are mapped to phone buttons and the speed dial is ignored because there is no button available for speed dial.

```
pin 12345
user me password pass123
number 2001 type silent-ring
number 2002 type beep-ring
number 2003 type feature-ring
number 2004 type monitor-ring
number 2005,2006 type overlay
number 2007,2008 type cw-overly
speed-dial 1 3001
speed-dial 2 3002 blf
```

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td><strong>logout-profile</strong></td>
<td>Enables Cisco Unified IP phone for Extension Mobility and assigns a logout profile to this phone.</td>
</tr>
<tr>
<td></td>
<td><strong>reset (voice logout-profile and voice user-profile)</strong></td>
<td>Performs complete reboot of all IP phones on which a particular logout-profile or user-profile is downloaded.</td>
</tr>
</tbody>
</table>
speed-dial (voice register pool)

To create a speed-dial definition for a Cisco Unified SIP IP phone or analog phone that uses an analog telephone adaptor (ATA) in a Cisco Unified Communications Manager Express (Cisco Unified CME) system, use the `speed-dial` command in voice register pool configuration mode. To disable a speed-dial definition, use the `no` form of this command.

```
speed-dial  speed-tag  digit-string  [label  label-text]
no  speed-dial  speed-tag
```

**Syntax Description**

<table>
<thead>
<tr>
<th>speed-tag</th>
<th>Unique sequence number that identifies a speed-dial definition during configuration tasks. Range is 1 to 113.</th>
</tr>
</thead>
<tbody>
<tr>
<td>digit-string</td>
<td>Digits to be dialed when the speed-dial button is pressed on an IP phone or the digits to be dialed when the associated code is entered from an analog phone with an ATA device. For IP phones, if the first character of this string is a plus sign (+), this speed-dial number is locked and cannot be changed at the phone. If the only character in this string is a pound sign (#), a user-programmable speed-dial button with no speed-dial number attached is defined.</td>
</tr>
<tr>
<td>label</td>
<td>(Optional) Text string that appears next to the speed-dial button. Enclose the string in quotation marks if the string contains a space.</td>
</tr>
</tbody>
</table>

**Command Default**

No speed-dial definition is created.

**Command Modes**

Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.2(4)M</td>
<td>Cisco Unified CME 9.1</td>
<td>This command was modified to increase the number of speed-dial configuration to 113.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `speed-dial` command creates a speed-dial definition for a Cisco Unified SIP IP phone being configured in Cisco Unified CME.

The `speed-tag` argument is a unique identifier for a speed-dial definition on the phone that is being configured. On Cisco Unified IP phones, speed-dial definitions are assigned to available extension buttons that have not been assigned to extensions. Speed-dial definitions are assigned in the order of their identifier numbers.

For example, if you define speed-dial 1, it is assigned to the first phone button that is available after the buttons that are assigned to extensions. If you used two buttons for extensions on a phone, speed-dial 1 is assigned to the third physical button on the phone. When you define speed-dial 2, it is assigned to the fourth physical button on the phone.

For Cisco Unified IP phones, speed-dial numbers can be assigned by the administrator using the `digit-string` argument and can be locked if the `digit-string` argument begins with a plus sign (+). Locked numbers cannot be changed at the phone. Speed-dial instances without speed-dial numbers (those defined with only a pound
sign) and speed-dial instances with unlocked digit-string arguments can be changed by users at their Cisco Unified IP phones.

If more speed-dial definitions are created than are supported by the IP phone setup, the extra speed-dial configurations are ignored.

Changes made to speed-dial buttons are saved in the router’s NVRAM configuration after a timer-based delay.

Analog phone users who use a Cisco ATA-186 or Cisco ATA-188 to connect to Cisco Unified CME systems use a different method to access speed-dial numbers. Instead of pressing a speed-dial button, phone users with ATA devices press the asterisk (*) key and a speed-tag number (speed-dial identifier) to dial a speed-dial number. For instance, a phone user with a Cisco ATA-186 presses *1 to dial the number that has been programmed as speed-dial 1 on that phone.

Phones with ATA devices are limited to a maximum of nine speed-dial numbers that must be programmed by the system administrator. The numbers cannot be programmed from the phone. With phones that use ATA devices, system administrators must be sure to tell phone users when speed-dial numbers have been programmed for their phones.

After you configure the speed-dial command, restart the phone using the reset command.

**Examples**

The following example shows how to set speed-dial button 2 to dial the head office at extension 5001 and lock the setting so that the phone user cannot change it at the phone:

```
Router(config)＃ voice register pool 23
Router(config-register-pool)＃ speed-dial 2 +5001 label "Head Office"
```

The following example shows how to set speed-dial button 13 to dial the sales office extension number (222):

```
Router(config)＃ voice register pool 3
Router(config-register-pool)＃ speed-dial 13 222 label "Sales Office"
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>reset (voice register global)</td>
<td>Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td>reset (voice register pool)</td>
<td>Performs a complete reboot of a specific SIP phone associated with a Cisco Unified CME system.</td>
</tr>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode for SIP phones.</td>
</tr>
</tbody>
</table>
srst dn line-mode

To specify line mode for the ephone-dns that are automatically created in Survivable Remote Site Telephony (SRST) mode on a Cisco Unified CME router, use the **srst dn line-mode** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

**srst dn line-mode** \{dual|dual-octo|octo|single\}

**no srst dn line-mode**

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>dual</th>
<th>SRST fallback ephone-dns are dual-line.</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>dual-octo</td>
<td>SRST fallback ephone-dns are dual-line or octo-line, depending on the phone type.</td>
</tr>
<tr>
<td></td>
<td>octo</td>
<td>SRST fallback ephone-dns are octo-line.</td>
</tr>
<tr>
<td></td>
<td>single</td>
<td>SRST fallback ephone-dns are single-line.</td>
</tr>
</tbody>
</table>

**Command Default**

Default is single-line mode.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>The <strong>dual-octo</strong> and <strong>octo</strong> keywords were added.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command with the <strong>dual-octo</strong> and <strong>octo</strong> keywords was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies whether ephone-dns that are created during fallback are dual-line, single-line, or octo-line ephone-dns. It applies only to the ephone-dns that are “learned” automatically from ephone configuration information, and not to ephone-dns that are manually configured in Cisco Unified CME.

If you use the **dual-octo** keyword, the type of ephone-dn that Cisco Unified CME in SRST mode creates depends on the phone type. It creates dual-line ephone-dns if the phone type is a Cisco Unified IP Phone 7902 or 7920, or an analog phone connected to the Cisco VG224 or Cisco ATA. It creates octo-line ephone-dns for all other phone types.

Use the **show telephony-service ephone-dn** command to display Cisco Unified CME parameters for ephone-dns.

**Examples**

The following example specifies dual-line mode for all SRST fallback ephone-dns.

```
telephony-service
    srst dn line-mode dual
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td><strong>show telephony-service ephone-dn</strong></td>
<td>Displays parameters for ephone-dns.</td>
</tr>
<tr>
<td></td>
<td><strong>srst mode auto-provision</strong></td>
<td>Enables SRST mode for a Cisco Unified CME router.</td>
</tr>
</tbody>
</table>
srst dn template

To specify an ephone-dn template to be used in Survivable Remote Site Telephony (SRST) mode on a Cisco Unified CallManager Express (Cisco Unified CME) router, use the `srst dn template` command in telephony-service configuration mode. To return to the default, use the `no` form of this command.

```
srst dn template template-tag
no srst dn template
```

**Syntax Description**
- `template-tag`: Identifying number of an existing ephone-dn template. Range is from 1 to 15.

**Command Default**
No ephone-dn template is specified.

**Command Modes**
Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command applies the specified ephone-dn template to all SRST fallback ephone-dns. Ephone-dn templates are created with the `ephone-dn-template` command.

Use the `show telephony-service ephone-dn-template` to display the contents of ephone-dn templates.

**Examples**

The following example applies ephone-dn template 2 to all SRST fallback ephone-dns.

```
telephony-service
srst dn template 2
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ephone-dn-template</code></td>
<td>Enters ephone-dn-template configuration mode to create an ephone-dn template.</td>
</tr>
<tr>
<td><code>show telephony-service ephone-dn-template</code></td>
<td>Displays the contents of ephone-dn templates.</td>
</tr>
</tbody>
</table>
srst ephone description

To specify a description to be associated with an ephone in Survivable Remote Site Telephony (SRST) mode on a Cisco Unified CallManager Express (Cisco Unified CME) router, use the `srst ephone description` command in telephony-service configuration mode. To return to the default, use the `no` form of this command.

```
srst ephone description string
no srst ephone description
```

**Syntax Description**
- `string`: Description to be associated with an ephone. Maximum string length is 100 characters.

**Command Default**
No description is specified.

**Command Modes**
Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use the `show telephony-service ephone` to display the ephone description to be associated with SRST fallback phones.

**Examples**

The following example applies a description to all SRST fallback ephones.

```
telephony-service
srst ephone description srst fallback auto-provision phone
```

The following excerpt displays a time-stamped SRST description for ephone 1:

```
Router# show running-config
ephone 1
description srst fallback auto-provision phone : Jul 07 2005 17:45:08
ephone-template 5
description srst fallback auto-provision phone : Jul 07 2005 17:45:08
mac-address 100A.7052.2AAE
button 1:1 2:2
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show telephony-service ephone</td>
<td>Displays ephone settings.</td>
</tr>
</tbody>
</table>
srst ephone template

To specify an ephone template to be used in Survivable Remote Site Telephony (SRST) mode on a Cisco Unified CallManager Express (Cisco Unified CME) router, use the `srst ephone template` command in telephony-service configuration mode. To return to the default, use the `no` form of this command.

```
srst ephone template template-tag
no srst ephone template
```

**Syntax Description**
- `template-tag` Identifying number of an existing ephone template. Range is from 1 to 20.

**Command Default**
No ephone template is specified.

**Command Modes**
Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Ephone templates are created with the `ephone-template` command. This command applies the specified ephone template to all SRST fallback ephones.

Use the `show telephony-service ephone-template` to display the contents of ephone templates.

**Examples**
The following example applies ephone template 3 to all SRST fallback ephones.

```
telephony-service
srst ephone template 3
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>ephone-template</code></td>
<td>Enters ephone-template configuration mode to create an ephone template.</td>
</tr>
<tr>
<td><code>show telephony-service ephone-template</code></td>
<td>Displays the contents of ephone templates.</td>
</tr>
</tbody>
</table>
srst mode auto-provision

To enable Survivable Remote Site Telephony (SRST) mode for a Cisco Unified CallManager Express (Cisco Unified CME) router, use the `srst mode auto-provision` command in telephony-service configuration mode. To return to the default, use the `no` form of this command.

```
srst mode auto-provision {all|dn|none}
no srst mode auto-provision
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>all</th>
<th>Includes information for learned ephones and ephone-dns in the running configuration.</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>dn</td>
<td>Includes information for learned ephone-dns in the running configuration.</td>
</tr>
<tr>
<td></td>
<td>none</td>
<td>Does not include information for learned ephones or learned ephone-dns in the running configuration. Use this keyword when you want Cisco Unified CME to provide SRST fallback services for Cisco Unified CallManager.</td>
</tr>
</tbody>
</table>

**Command Default**

SRST mode is disabled.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command puts a Cisco Unified CME router into SRST mode to provide fallback call-processing services for IP phones that have lost connection to their Cisco Unified CallManager. The phones can be preconfigured manually or the Cisco Unified CME-SRST router can dynamically learn their configuration. The keywords in this command allow you to specify how much of the learned phone configurations you want to include in the running configuration of the Cisco Unified CME-SRST router.

Use the `none` keyword to enable the Cisco Unified CME router to provide SRST fallback services for Cisco Unified CallManager. Use the `dn` or `all` keyword to enable the Cisco Unified CME router to learn the ephone-dn or ephone and ephone-dn configuration from Cisco Unified CallManager and include the information in its running configuration.

**Note**

We do not recommend that you use the `dn` or `all` keyword if you want Cisco Unified CME to provide SRST fallback services. After the Cisco Unified CME-SRST router learns the phone configuration from Cisco Unified CallManager and you save the configuration, the fallback phones are treated as locally configured phones on the Cisco Unified CME-SRST router which can adversely impact the fallback behavior of those phones.
Examples

The following example shows how to enable the Cisco Unified CME router to provide SRST fallback services for phones connected to Cisco Unified CallManager. Information for learned ephone-dns and ephones is not included in the running configuration.

telephony-service
  srst mode auto-provision none

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show telephony-service all</td>
<td>Displays detailed configuration for phones, voice ports, and dial peers in a Cisco Unified CME system.</td>
</tr>
<tr>
<td>srst dn line-mode</td>
<td>Specifies line mode for the ephone-dns that are automatically created in SRST mode on a Cisco Unified CME router.</td>
</tr>
</tbody>
</table>
standby username password

To specify that the standby (secondary backup) router XML interface is enabled, use the **standby username password** command in telephony-service configuration mode on the primary router. To disable the XML interface on the secondary backup router, use the **no** form of this command.

**Syntax**

```
standby username username password [0|6] password
no standby username username password [0|6] password
```

**Syntax Description**

- **username**
  - Specifies the username who is authorized to enable the XML interface.
- **password**
  - Specifies the password to use for access.

**Command Default**

An authorized user is not named.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command was enhanced for password encryption, based on Unified CME password policy.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to enable the XML interface on the secondary backup router. The username and password must be the same as that used for access to the primary router.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters [0|6]. This is in accordance with Unified CME Password Policy. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

**Examples**

The following example enables the XML interface on the secondary backup router:

```
Router(config)# telephony-service
Router(config-telephony)# standby username admin password 1234
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>username password</strong></td>
<td>To assign a login account username and password to a phone user so that the user can log in to the Cisco Unified CME router.</td>
</tr>
</tbody>
</table>
statistics collect

To enable the collection of call statistics for an ephone hunt group, use the `statistics collect` command in ephone-hunt configuration mode. To stop statistics collection and to delete statistics that have been collected, use the no form of this command.

```
statistics collect
no statistics collect
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

The default is no call statistics data is collected.

**Command Modes**

Ephone-hunt configuration (config-ephone-hunt)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used for the collection of call statistics, such as direct calls to hunt group pilot numbers, or calls to the Basic Automatic Call Distribution (B-ACD) and Auto Attendant service. For detailed information, see Cisco Unified CME B-ACD and Tcl Call-Handling Applications.

The `statistics collect` can be used to activate statistics collection for any number of ephone hunt groups.

Statistics collection begins at the time that the `statistics collect` is entered. A maximum of one week (168 hours) of statistics can be stored at a time. You can display the statistics with the `show hunt-group` or transfer statistics automatically to files using TFTP. The `no statistics collect` deletes all statistics that have been collected.

All or some of the statistics can be output with the `show hunt-group` or sent to files automatically using TFTP by using the `hunt-group report url` `hunt-group report every hours` commands.

**Examples**

The following example enables the collection of call statistics for ephone hunt group 1 and ephone hunt group 2:

```
Router(config)# ephone-hunt 1
Router(config-ephone-hunt)# statistics collect
Router(config)# ephone-hunt 2
Router(config-ephone-hunt)# statistics collect
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>hunt-group report delay hours</code></td>
<td>Delays the automatic transfer of Cisco CME B-ACD call statistics to a file.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td><strong>hunt-group report every hours</strong></td>
<td>Sets the hourly interval at which Cisco CME B-ACD call data is automatically transferred to a file.</td>
</tr>
<tr>
<td><strong>hunt-group report url</strong></td>
<td>Sets filename parameters and the URL path where Cisco CME B-ACD call statistics are to be sent using TFTP.</td>
</tr>
<tr>
<td><strong>show ephone-hunt statistics</strong></td>
<td>Displays ephone-hunt configuration information and current status and statistics information.</td>
</tr>
</tbody>
</table>
statistics collect (voice hunt-group)

To enable the collection of call statistics for a voice hunt group, use the statistics collect command in voice hunt-group configuration mode. To return to the default, use the no form of this command.

```
statistics collect
no statistics collect
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

No configuration statistics can be collected for voice hunt groups.

**Command Modes**

Voice hunt-group configuration (config-voice-hunt-group)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.2(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Examples**

The following example shows how to enable the collection of call statistics for voice hunt group 60:

```
Router(config)# voice hunt-group 60
Router(config-voice-hunt-group)# statistics collect
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>statistics collect</td>
<td>Enables the collection of call statistics for an ephone hunt group.</td>
</tr>
</tbody>
</table>
subnet

To define which IP phones are part of an emergency response location (ERL) for the enhanced 911 service, use the **subnet** command in voice emergency response location configuration mode. To remove the subnet definition, use the **no** form of this command.

**subnet** [{1|2}] *IPaddress mask*

**no subnet** [{1|2}]

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><em>IPaddress</em></td>
<td>IP address that identifies which IP phones are part of the ERL.</td>
</tr>
<tr>
<td><em>mask</em></td>
<td>IP subnet mask for the network segment that is part of the ERL.</td>
</tr>
</tbody>
</table>

**Command Default**

No subnets are defined.

**Command Modes**

Voice emergency response location configuration (cfg-emrgncy-resp-location)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1</td>
<td>This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)</td>
<td>This command was added to Cisco Unified CME.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to define the groups of IP phones that are part of an ERL. You can create up to 2 different subnets. To include all phones on a single ERL, you can set the subnet mask to 0.0.0.0 to indicate a “catch-all” subnet.

**Examples**

In the following example, all IP phones with the IP address of 10.X.X.X or 192.168.X.X are automatically associated with this ERL. If one of the phones dials 911, its extension is replaced with 408 555-0100 before it goes to the PSAP. The PSAP will see that the caller’s number is 408 555-0100.

```plaintext
voice emergency response location 1
elin 1 4085550100
subnet 10.0.0.0 255.0.0.0
subnet 2 192.168.0.0 255.255.0.0
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>elin</strong></td>
<td>Specifies a PSTN number that will replace the caller’s extension.</td>
</tr>
</tbody>
</table>
system message

To set a text message for display on idle Cisco IP Phones with display, such as Cisco IP Phone 7940 and Cisco IP Phone 7960, in a Cisco Unified Communications Manager Express (Cisco Unified CME) system, use the `system message` command in telephony-service configuration mode. To return to the default, use the `no` form of this command.

```
system message  text-message
no system message
```

**Syntax Description**

| text-message | Alphanumeric string of approximately 30 characters maximum to display when the phone is idle. |

**Command Default**

The message “Cisco Unified Communications Manager Express” is displayed.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco Unified CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco Unified CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The number of characters that can be displayed is not fixed because IP phones typically use a proportional (as opposed to a fixed-width) font. There is room for approximately 30 alphanumeric characters.

The display message is refreshed with a new message after any of the following events occurs:

- A busy phone goes back on-hook.
- An idle phone receives a keepalive message.
- A phone is restarted.

**Examples**

The following example sets the message “ABC Company” to display instead of “Cisco Unified Communications Manager Express” on idle Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G:

```
Router(config)# telephony-service
Router(config-telephony)# system message ABC Company
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>telephony-service</td>
<td>Enters telephony-service configuration mode.</td>
</tr>
</tbody>
</table>
Cisco Unified CME Commands: T

- telephony-service, on page 1275
- telnet-support, on page 1279
- template (auto-register), on page 1280
- template (voice register pool), on page 1282
- tftp-path (voice register global), on page 1283
- tftp-server-credentials trustpoint, on page 1284
- time-format, on page 1285
- time-format (voice register global), on page 1286
- timeout (phone-hunt), on page 1287
- timeout (voice hunt-group), on page 1289
- timeouts busy, on page 1290
- timeouts interdigit (telephony-service), on page 1291
- timeouts interdigit (voice register global), on page 1292
- timeouts night-service-bell, on page 1293
- timeouts ringing (telephony-service), on page 1295
- timeouts transfer-recall, on page 1296
- timeouts transfer-recall (voice register global), on page 1298
- timeouts transfer-recall (voice register dn), on page 1300
- time-webedit (telephony-service), on page 1302
- time-zone, on page 1303
- timezone (voice register global), on page 1306
- transfer max-length, on page 1309
- transfer-attended (voice register template), on page 1310
- transfer-blind (voice register template), on page 1311
- transfer-digit-collect, on page 1312
- transfer-mode, on page 1314
- transfer-park blocked, on page 1316
- transfer-pattern (telephony-service), on page 1318
- transfer-pattern blocked, on page 1320
- transfer-system, on page 1322
- translate (ephone-dn), on page 1325
- translate callback-number, on page 1327
- translate outgoing (voice register pool), on page 1329
• translation-profile, on page 1331
• translation-profile incoming, on page 1333
• transport (voice register pool-type), on page 1334
• trunk, on page 1335
• trustpoint (credentials), on page 1338
• trustpoint-label, on page 1340
• type, on page 1341
• type (voice register dialplan), on page 1346
• type (voice register pool), on page 1348
• type (voice-gateway), on page 1354
telephony-service

To enter telephony-service configuration mode for configuring Cisco Unified CME, use the `telephony-service` command in global configuration mode. To remove the entire Cisco Unified CME configuration for SCCP IP phones, use the `no` form of this command.

```
telephony-service [setup]
no telephony-service
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>setup</code></td>
<td>(Optional) Interactive setup tool for configuring Cisco Unified IP Phone 7910s, 7940s, and 7960s in Cisco Unified CME.</td>
</tr>
</tbody>
</table>

**Note**

This interactive Cisco CME setup tool is restricted to generating basic configuration files for Cisco Unified IP Phone 7910s, 7940s, and 7960s running SCCP protocol only.

**Command Default**

No Cisco Unified CME configuration for SCCP IP phones is present.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>The <code>setup</code> keyword was added.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enters the telephony-service configuration mode for configuring system wide parameters for SCCP IP phones in Cisco Unified CME.

The voice-gateway system is tied to the telephony service. The `telephony-service` command must be configured before the voice-gateway system is configured; otherwise, the voice gateway is hidden from the user.

Use the `setup` keyword to start the interactive setup tool to automatically configure only Cisco Unified IP Phone 7910s, 7940s, and 7960s in Cisco Unified CME.

For alternate methods of automatically configuring Cisco Unified CME, including Cisco Unified IP Phone 7910s, 7940s, and 7960s and other Cisco Unified IP phones, see the Cisco Unified CME Administrator Guide.

The `setup` keyword is not stored in the router nonvolatile random-access memory (NVRAM).

If you attempt to use the `setup` option for a system that already has a telephony-service configuration, the command is rejected. To use the `setup` option after an existing telephony-service configuration has been created, first remove the existing configuration using the `no telephony-service` command.

The table shows a sample dialog with the Cisco CME setup tool and explains possible responses to the Cisco CME setup tool prompts.
<table>
<thead>
<tr>
<th>Cisco CME Setup Tool Prompt</th>
<th>Description</th>
</tr>
</thead>
</table>
| **Do you want to setup DHCP service for your IP phones? [yes/no]:** | • Yes—Configures the Cisco Unified CME router to act as a Dynamic Host Configuration Protocol (DHCP) server, automatically providing IP addresses to your IP phones and provisioning the default gateway and TFTP IP addresses to be used by the phones. This method creates a single pool of IP addresses. If you need a pool for non-IP phones or if the Cisco router cannot act as the DHCP router, answer no and manually define the DHCP server.  
• No—Indicates that you have already configured DHCP or static IP addresses for the IP phones. |
| If you respond yes, you see the following prompts: |  |
| IP network for telephony-service DHCP Pool:  
Subnet mask for DHCP network :  
TFTP Server IP address (Option 150) :  
Default Router for DHCP Pool : |  |
| **Do you want to start telephony-service setup? [yes/no]:** | • Yes—Starts the interactive setup tool for configuring Cisco Unified IP Phone 7910s, 7940s, and 7960s.  
• No—Terminates the Cisco CME setup tool. |
| **Enter the IP source address for Cisco CallManager Express:** | IP address on which the router provides Cisco Unified CME services, usually the default gateway for the IP subnet that you are using for the IP phones, and the port for Skinny Client Control Protocol (SCCP) messages. |
| **Enter the Skinny Port for Cisco CallManager Express:** [2000]: | Enter the maximum number of IP phones that this Cisco Unified CME system will support. This number can be increased later, to the maximum allowed for this version and your router.  
**Note** The Cisco CME setup tool associates one number with each newly registering phone. If you want additional numbers on a phone, manually add them later. |
| **How many IP phones do you want to configure : [0]:** |  |
| **Do you want dual-line extensions assigned to phones? [yes for dual-line / no for single-line]:** | • Yes—Each newly registering IP phones is assigned a single number that is associated with a single phone button. The system generates a dual-line ephone-dn entry for each ephone-dn.  
• No—IP phones are linked directly to one or more PSTN trunk lines. Using keyswitch mode requires manual configuration in addition to using the Cisco CME setup tool. The system generates two ephone-dn entries for each ephone-dn, and they are both assigned to a single phone. |
<table>
<thead>
<tr>
<th>Cisco CME Setup Tool Prompt</th>
<th>Description</th>
</tr>
</thead>
</table>
| What language do you want on IP phones? | Language for IP phone displays, selected from the list.  
   - Default is 0, English.  
   - 0 English  
   - 1 French  
   - 2 German  
   - 3 Russian  
   - 4 Spanish  
   - 5 Italian  
   - 6 Dutch  
   - 7 Norwegian  
   - 8 Portuguese  
   - 9 Danish  
   - 10 Swedish  
   
| Which Call Progress tone set do you want on IP phones? | Locale for the tone set used to indicate call status or progress, selected from the list.  
   - Default is 0, United States.  
   - 0 United States  
   - 1 France  
   - 2 Germany  
   - 3 Russia  
   - 4 Spain  
   - 5 Italy  
   - 6 Netherlands  
   - 7 Norway  
   - 8 Portugal  
   - 9 UK  
   - 10 Denmark  
   - 11 Switzerland  
   - 12 Sweden  
   - 13 Austria  
   - 14 Canada  
   
| What is the first extension number you want to configure? | First number in pool of extension numbers to be created for IP phones connected to the Cisco router to be configured.  
   - Starting with this number, remaining extension numbers are automatically configured in a contiguous manner.  
   - This number must be compatible with your telephone number plan and if you use Direct Inward Dialing (DID) service, with public switched telephone network (PSTN) numbering requirements.  
   - 0:  
   
| Do you have Direct-Inward-Dial service for all your phones? | • Yes—If you have trunk access to public telephone service by ISDN or VoIP for all extension numbers. The system creates an appropriate dial plan.  
   • No—If you have simple analog phone lines only (for example, foreign exchange office [FXO] interfaces) or if you have trunk access for some lines but not all lines.  
| [yes/no]: | |

---

Cisco Unified Communications Manager Express Command Reference

1277
Cisco CME Setup Tool Prompt | Description
--- | ---
If you answer yes to the previous question, you see the following prompt: Enter the full E.164 number for the first phone: | Complete ten-digit telephone number, including area code, that corresponds to the first extension number.

Do you want to forward calls to a voice message service? [yes/no]: | • Yes—To forward calls to a single voice message service number when an IP phone is busy or does not answer. All phone extensions forward their calls to the same voice message service pilot number.  
• No—Not to forward calls to a single voice message service number. Answer no if you do not have a voice message system or if you want to customize call-forwarding behavior for each extension.

If you answer yes to the previous question, you see the following prompt: Enter the extension or pilot number of the voice message service: | Voice message service pilot number.  
• This step can be ignored during the setup dialog and manually configured later.

Call forward No Answer Timeout: [18]: | Timeout, in seconds, after which to forward calls to voice mail if they are not answered.  
• Default is 18.

Do you wish to change any of the above information? [yes/no]: | • Yes—Starts the dialog over again without implementing any of the answers that you previously gave.  
• No—Uses specified values to automatically build basic configuration for Cisco Unified IP Phone 7910s, 7940s, and 7960s in Cisco Unified CME.

Examples

The following example shows how to enter telephony-service configuration mode for manually configuring Cisco Unified CME. This example also includes the for configuring the maximum number of phones to 12:

Router(config)# telephony-service  
Router(config-telephony)# max-ephones 12

The following example shows how to start the Cisco CME setup tool:

Router(config)# telephony-service setup
**telnet-support**

To enable the telnet access for the phone, use the `telnet-support` command in voice register pool-type mode. To disable telnet support, use the `no` form of this command.

```
telnet-support
notelnet-support
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

The telnet support is not enabled. When the `reference-pooltype` command is configured, the telnet-support value of the reference phone is inherited.

**Command Modes**

Voice Register Pool Configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(3)M</td>
<td>Cisco SIP CME 10.0</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to enable the telnet access for the phone. When you use the no form of this command, the inherited properties of the reference phone takes precedence over the default value.

**Example**

The following example shows how to specify a description for a phone model using the `description` command:

```
Router(config)# voice register pool-type 9900
Router(config--register-pool-type)# telnet-support
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register pool-type</td>
<td>Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.</td>
</tr>
</tbody>
</table>
To create a basic configuration template that supports all the configurations available on the voice register template, use the template command in voice auto register configuration mode. This command is a sub-mode CLI of the command auto-register. To disable creation of the basic configuration template as part of the auto registration process, use the no form of this command.

```
template tag
no template
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>template</td>
<td>Creates a basic configuration template that supports all the configurations available on the voice register template. Range: 1 to 10.</td>
</tr>
</tbody>
</table>

### Command Default

By default, this command is disabled.

### Command Modes

voice auto register configuration (config-voice-auto-register)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.6(3)M</td>
<td>Cisco Unified CME 11.5</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>16.3.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command provides the option to create a basic configuration template that can be applied to all phones registering automatically on Unified CME. It is mandatory that voice register template is configured with the same template tag.

### Examples

The following example shows how to create a basic configuration template for auto registration of SIP phones:

```
Router(config)#voice register global
Router(config-register-global)#auto-register
Router(config-voice-auto-register)#?

VOICE auto register configuration commands:
auto-assign Define DN range for auto assignment
default Set a command to its defaults
exit Exit from voice register group configuration mode
no Negate a command or set its defaults
password Default password for auto-register phones
service-enable Enable SIP phone Auto-Registration
template Default template for auto-register phones

Router(config-voice-auto-register)#template ?
<1-10> template tag
Router(config-voice-auto-register)#template 10
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>service-enable (auto-register)</td>
<td>Temporarily disables the auto registration process, but retains the password and DN range configurations. Once auto-register command is entered, the service is enabled by default.</td>
</tr>
<tr>
<td></td>
<td>password (auto-register)</td>
<td>Configures the mandatory password that administrator sets for auto registration of SIP phones on Unified CME.</td>
</tr>
<tr>
<td></td>
<td>auto-assign (auto-register)</td>
<td>Configures the mandatory range of directory numbers for phones auto registering on Unified CME.</td>
</tr>
<tr>
<td></td>
<td>auto-register</td>
<td>Enables automatic registration of SIP phones with the Cisco Unified CME system.</td>
</tr>
<tr>
<td></td>
<td>auto-reg-ephone</td>
<td>Enables automatic registration of ephones with the Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
**template (voice register pool)**

To apply a template to a SIP phone, use the `template` command in voice register pool configuration mode. To remove the template, use the `no` form of this command.

```
template template-tag
no template template-tag
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>template-tag</td>
<td>The template tag that was created with the <code>voice register template</code> command in voice register global configuration mode. Range is 1 to 5.</td>
</tr>
</tbody>
</table>

**Command Default**

Template is not applied to a SIP IP phone.

**Command Modes**

Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Apply any one of five previously defined templates to a SIP phone. Only one template is applied to a SIP phone at one time.

**Examples**

The following example shows how to define templates 1 and 2 and apply template 1 to SIP phones 1, 2, and 3, and template 2 to SIP phone 4:

```
Router(config)# voice register template 1
Router(config-register-temp)# anonymous block
Router(config-register-temp)# caller-id block
Router(config-register-temp)# voicemail 5001 timeout 15
Router(config)# voice register template 2

Router(config-register-temp)# anonymous block
Router(config-register-temp)# caller-id block
Router(config-register-temp)# no conference
Router(config-register-temp)# no transfer-attended
Router(config-register-temp)# voicemail 5005 timeout 15
Router(config)# voice register pool 1
Router(config-register-pool)# template 1
Router(config)# voice register pool 2
Router(config-register-pool)# template 1
Router(config)# voice register pool 3
Router(config-register-pool)# template 1
Router(config)# voice register pool 4
Router(config-register-pool)# template 2
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register template</td>
<td>Enters voice register template configuration mode and defines a template of common parameters for SIP phones.</td>
</tr>
</tbody>
</table>
**tftp-path (voice register global)**

To specify the directory to which the configuring files for SIP phones in Cisco Unified CME are written, use the `tftp-path` command in voice register global configuration mode. To return to the default, use the `no` form of this command.

```
tftp-path {flash|slot0|tftp://url}
no tftp-path
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>flash:</td>
<td>Router flash memory.</td>
</tr>
<tr>
<td>slot0:</td>
<td>Router slot 0 memory.</td>
</tr>
<tr>
<td>tftp://</td>
<td>External TFTP server.</td>
</tr>
<tr>
<td>url</td>
<td>URL for external TFTP server.</td>
</tr>
</tbody>
</table>

**Command Default**

The default directory is system memory (system:/cme/sipphone/).

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command defines the location for configuration files that are generated by using the `create profile` command.

**Examples**

The following example shows how to set the path to an HTTP directory for an external TFTP server:

```
Router(config)# voice register global
Router(config-register-global)# tftp-path tftp://mycompany.com/files/
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>create profile</code> (voice register global)</td>
<td>Generates the configuration profiles required for SIP phones.</td>
</tr>
<tr>
<td><code>reset</code> (voice register global)</td>
<td>Performs a “hard” reboot similar to a power-off-power-on sequence for all SIP phones in Cisco Unified CME, including contacting the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server for updated information.</td>
</tr>
</tbody>
</table>
tftp-server-credentials trustpoint

To specify the PKI trustpoint that signs the phone configuration files, use the `tftp-server-credentials trustpoint` command in telephony-service configuration mode. To return to the default, use the `no` form of this command.

```
tftp-server-credentials trustpoint label
no tftp-server-credentials trustpoint
```

**Syntax Description**

- `label`: Name of a configured PKI trustpoint with a valid certificate.

**Command Default**

No trustpoint is defined for TFTP server communications.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication.

**Examples**

The following example names the CA trustpoint, server12, as the trustpoint that signs the phone configuration files.

```
Router(config)# telephony-service
Router(config-telephony)# device-security-mode authenticated
Router(config-telephony)# secure-signaling trustpoint server25
Router(config-telephony)# tftp-server-credentials trustpoint server12
Router(config-telephony)# load-cfg-file slot0:Ringlist.xml alias Ringlist.xml sign create
Router(config-telephony)# exit
```
**time-format**

To select a 12-hour clock or a 24-hour clock for the time display format on Cisco IP phones in a Cisco CallManager Express (Cisco CME) system, use the `time-format` command in telephony-service configuration mode. To return to the default, use the `no` form of this command.

```
  time-format {12|24}
  no  time-format
```

**Syntax Description**

| 12 | Selects a 12-hour clock. This is the default. |
| 24 | Selects a 24-hour clock. |

**Command Default**

Time is displayed in 12-hour clock format.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following example selects a 24-hour clock for the time display on Cisco IP phones:

```
Router(config)# telephony-service
Router(config-telephony)# time-format 24
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>date-format</code></td>
</tr>
</tbody>
</table>
time-format (voice register global)

To set the time display format on SIP phones in a Cisco CallManager Express (Cisco CME) system, use the `time-format` command in voice register global configuration mode. To display the time in the default format, use the **no** form of this command.

```
time-format {12|24}
no date-format
```

**Syntax Description**

- **12**: Sets time in a 12-hour (AM/PM) clock.
- **24**: Sets time in a 24-hour clock.

**Command Default**

Time is displayed in 12-hour clock format.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Examples**

The following example shows how to set the time format to a 24-hour clock so that 11:00PM is displayed as 2300.

```
Router(config)# voice register global
Router(config-register-global)# time-format 24
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>voice register global</code></td>
</tr>
</tbody>
</table>
timeout (ephone-hunt)

To define the number of seconds after which a call that is not answered is redirected to the next number in a hunt-group list in Cisco Unified CME, use the `timeout` command in ephone-hunt configuration mode. To return to the default, use the `no` form of this command.

```
timeout seconds[, seconds...]
no timeout seconds[, seconds...]
```

**Syntax Description**

- `seconds` Number of seconds. Range: 3 to 60000. You can enter a different value for each hop between ephone-dns in a hunt group. If you enter a single value, the value is applied to each hop between ephone-dns in a hunt group.

**Command Default**

Default is the value of the `timeout` `ringing` which has a default of 180 seconds if it is not set to another value.

**Command Modes**

Ephone-hunt configuration (config-ephone-hunt)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was modified to accept multiple arguments that correspond to the number of ephone-dns configured in the hunt group.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command with modifications up was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to set no-answer timeouts for each hop in a hunt group. You can enter a different value for each hop between ephone-dns in a hunt group or you enter a single value to be applied to each hop between ephone-dns in a hunt group list.

If you configure this command and you also configure the `max-timeout` for an ephone hunt group, the `max-timeout` takes precedence over this command.

**Examples**

The following example defines a no-answer timeout of 10 seconds for each hop between ephone-dns in hunt group 25. If extension 1001 does not answer in 10 seconds, the call is sent to 1002. If 1002 does not answer in 10 seconds, the call is sent to 1003. If 1003 does not answer in 10 seconds, the call is sent to the final number.

```
ephone-hunt 25 sequential
pilot 4200
list 1001, 1002, 1003
timeout 10
final 4500
```
The following example shows a hunt-group configuration with separate timeouts, one for each ephone in the hunt-group. If the first extension (1001) does not answer in 7 seconds, the call is sent to the second extension (1002). If the call is not answered by the second extension in 9 seconds, the call is forwarded to the third extension (1003). Extension 1003 has 15 seconds to answer before the call is sent to the final number.

```plaintext
ephone-hunt 3 peer
    pilot 4200
    list 1001, 1002, 1003
    timeout 7, 9, 15
    final 4500
```

The following example shows the configuration for an ephone hunt group for which the `max-timeout` command is also configured. Using this configuration, if the second number is busy, the third extension, 1003, has only 13 seconds to answer (20 - 7 = 13) because the value for max-timeout is 20 seconds.

```plaintext
ephone-hunt 3 peer
    pilot 4200
    list 1001, 1002, 1003
    timeout 7, 9, 15
    max-timeout 20
    final 4500
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>final</code></td>
<td>Defines the last ephone-dn in an ephone hunt group.</td>
</tr>
<tr>
<td><code>hops</code></td>
<td>Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.</td>
</tr>
<tr>
<td><code>list</code></td>
<td>Defines the ephone-dns that participate in an ephone hunt group.</td>
</tr>
<tr>
<td><code>max-redirect</code></td>
<td>Changes the current number of allowable redirects in a Cisco Unified CME system.</td>
</tr>
<tr>
<td><code>max-timeout</code></td>
<td>Sets the maximum combined timeout for the no-answer periods for all ephone-dns in an ephone-hunt list,</td>
</tr>
<tr>
<td><code>no-reg (ephone-hunt)</code></td>
<td>Specifies that the pilot number of an ephone hunt group should not register with the H.323 gatekeeper.</td>
</tr>
<tr>
<td><code>pilot</code></td>
<td>Defines the ephone-dn that callers dial to reach an ephone hunt group.</td>
</tr>
<tr>
<td><code>preference (ephone-hunt)</code></td>
<td>Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.</td>
</tr>
</tbody>
</table>
timeout (voice hunt-group)

To define the number of seconds after which a call that is not answered is redirected to the next number in a voice hunt-group list, use the `timeout` command in voice hunt-group configuration mode. To return to the default timeout, use the `no` form of this command.

```
timeout seconds
no timeout
```

### Syntax Description

<table>
<thead>
<tr>
<th><code>seconds</code></th>
<th>Number of seconds. Range is 3 to 60000. Default is 180.</th>
</tr>
</thead>
</table>

### Command Default

Timeout period is 180 seconds.

### Command Modes

Voice hunt-group configuration (config-voice-hunt-group)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

If Call Forward No Answer is configured for a directory number in the voice hunt group, set the timeout value of this command to a value that is less than the timeout value of the `call-forward noan` command.

### Examples

The following example shows how to define a no-answer timeout of 15 seconds for each hop between phones in peer hunt-group 25:

```
Router(config)# voice hunt-group 25 peer
Router(config-voice-hunt-group)# timeout 15
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>call-forward noan</code></td>
<td>Enables call forwarding so that incoming calls to an extension (ephone-dn) that does not answer are forwarded to another number.</td>
</tr>
<tr>
<td><code>final (voice hunt-group)</code></td>
<td>Defines the last extension in a voice hunt group.</td>
</tr>
<tr>
<td><code>hops (voice hunt-group)</code></td>
<td>Defines the number of times that a call is redirected to the next directory number in a peer voice hunt-group list before proceeding to the final directory number.</td>
</tr>
<tr>
<td><code>list (voice hunt-group)</code></td>
<td>Defines the directory numbers that participate in a hunt group.</td>
</tr>
</tbody>
</table>
timeouts busy

To set the amount of time after which a call is disconnected from a busy signal, use the `timeouts busy` command in telephony-service configuration mode. To return to the default value, use the `no` form of this command.

```
timeouts busy seconds
no timeouts busy
```

**Syntax Description**

| seconds | Number of seconds after connection before a call is disconnected from a busy signal. Range is from 0 to 30 seconds. Default is 10. |

**Command Default**

Timeout busy period is 10 seconds.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Examples**

The following example sets a busy timeout of 10 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# timeouts busy 10
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>telephony-service</code></td>
</tr>
</tbody>
</table>
**timeouts interdigit (telephony-service)**

To set the interdigit timeout value for all Cisco IP phones in a Cisco Unified CME system, use the `timeouts interdigit` command in telephony-service configuration mode. To return to the default value, use the `no` form of this command.

```
timeouts interdigit seconds
no timeouts interdigit
```

**Syntax Description**

<table>
<thead>
<tr>
<th>seconds</th>
<th>Interdigit timeout duration for Cisco IP phones, in seconds. Range is from 2 to 120. Default is 10.</th>
</tr>
</thead>
</table>

**Command Default**

Timeout period is 10 seconds.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XB</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8).</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The interdigit timeout timer is activated when the caller enters a digit and is restarted each time the caller enters subsequent digits until the destination address is identified. This command specifies how long, in seconds, the system waits after a caller enters an initial digit or a subsequent digit of a dialed string. If the configured timeout value is exceeded before the destination address is identified, a tone sounds and the call is terminated. The default is 10 seconds.

To disable the timeouts interdigit timer, set the `seconds` value to zero.

**Examples**

The following example sets the interdigit timeout value to 5 seconds for all Cisco IP phones:

```
Router(config)# telephony-service
Router(config-telephony)# timeouts interdigit 5
```

In this example, 5 seconds is also the elapsed time after which an incompletely dialed number times out. For example, if you dial nine digits (408555013) instead of the required ten digits (4085550134), you hear a busy tone after 5 “timeout” seconds.

**Related Commands**

<table>
<thead>
<tr>
<th>DESCRIPTION</th>
<th>Related Commands</th>
</tr>
</thead>
<tbody>
<tr>
<td>Configures the interdigit timeout value for a specified voice port.</td>
<td>timeouts interdigit (voice-port)</td>
</tr>
</tbody>
</table>
**timeouts interdigit (voice register global)**

To set the interdigit timeout value for all Cisco SIP phones in a Cisco Unified CME system, use the `timeouts interdigit` command in voice register global configuration mode. To return to the default value, use the `no` form of this command.

```
timeouts interdigit seconds
timeouts interdigit no
```

**Syntax Description**

| seconds | Interdigit timeout duration for Cisco SIP phones, in seconds. Range is from 2 to 120. Default is 10. |

**Command Default**

Timeout period is 10 seconds.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IOS XE Everest 16.4.1</td>
<td>Cisco Unified CME 11.6</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The interdigit timeout timer is activated when the caller enters a digit and is restarted each time the caller enters subsequent digits until the destination address is identified. This command specifies how long, in seconds, the system waits after a caller enters an initial digit or a subsequent digit of a dialed string. If the configured timeout value is exceeded before the destination address is identified, a tone sounds and the call is terminated. The default is 10 seconds.

To disable the timeouts interdigit timer, set the `seconds` value to zero.

**Examples**

The following example sets the interdigit timeout value to 5 seconds for all Cisco SIP phones:

```
Router(config)# voice register global
Router(config-register-global)# timeouts interdigit 5
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>timeouts interdigit (telephoy-service)</td>
<td>Configures the interdigit timeout value for a SCCP phone in Cisco Unified CME system.</td>
</tr>
</tbody>
</table>
**timeouts night-service-bell**

To specify the interval between two night-service notification bells, use the `timeouts night-service-bell` command in telephony-service configuration mode. To reset to the default value, use the `no` form of this command.

`timeouts night-service-bell seconds`

`no timeouts night-service-bell`

---

**Syntax Description**

| seconds | Duration, in seconds, between night-service notification bells. Range: 4 to 30. Default: 12 |

**Command Default**

Default is 12 seconds.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XW5</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command modifies the repeat interval between two night-service notification bells for the same call from the default (12 seconds) to the specified number of seconds.

When an ephone-dn is marked for night-service treatment, incoming calls that ring during the night-service time period on that directory number send a notification to all IP phones that are marked to receive night-service bell notification.

**Examples**

The following partial output shows that the night-service notification bell is configured for 4 seconds between bells for the same call:

```
Router# show running-configuration
.
.
.
telephony-service
.
.
    night-service code *1234
    night-service day Tue 00:00 23:00
    night-service day Wed 01:00 23:59
    timeouts night-service-bell 4
    !
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>night-service bell (ephone)</code></td>
<td>Marks an IP phone to receive night-service bell notification when incoming calls are received during night-service time periods on ephone-dns that are marked for night service.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>night-service bell (ephone-dn)</td>
<td>Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.</td>
</tr>
</tbody>
</table>
**timeouts ringing (telephony-service)**

To set the timeout value for ringing in a Cisco CallManager Express (Cisco CME) system, use the `timeouts ringing` command in telephony-service configuration mode. To reset the timeout value to the default value, use the `no` form of this command.

```
timeouts ringing seconds
no timeouts ringing
```

**Syntax Description**

| seconds | Duration, in seconds, for which the Cisco CME system allows ringing to continue if a call is not answered. Range is from 5 to 60000. Default is 180. |

**Command Default**

Timeout is 180 seconds.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
</tbody>
</table>

**Examples**

The following example allows incoming calls to ring for 600 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# timeouts ringing 600
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>telephony-service</code></td>
</tr>
</tbody>
</table>
**timeouts transfer-recall**

To enable Cisco Unified CME to recall a transferred call if the transfer-to party does not answer or is busy, use the `timeouts transfer-recall` command in ephone-dn, ephone-dn template, or telephony-service configuration mode. To reset to the default value, use the `no` form of this command.

```
timeouts transfer-recall seconds
no timeouts transfer-recall
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>seconds</th>
<th>Duration, in seconds, to wait before recalling a transferred call. Range: 1 to 1800. Default: 0 (disabled).</th>
</tr>
</thead>
</table>

**Command Default**

Transfer recall is disabled (0 seconds).

**Command Modes**

- Ephone-dn (config-ephone-dn)
- Ephone-dn template (config-ephone-dn-template)
- Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables Call Transfer Recall and sets the number of seconds that Cisco Unified CME waits before sending a transferred call back to the phone that initiated the transfer (transferor).

If the transfer-recall timer set with this command expires before the transfer-to party answers a call, the call is directed back to the transferor and the message, “Transfer Recall From xxxx” displays on the transferor phone. After the first recall, the timer restarts. The maximum number of retries is two if the transfer-to party remains busy or does not answer. The transferor and transfer-to party must be on the same Cisco Unified CME router; the transferee party can be remote.

Transfer recall is not supported if the transfer-to party has Call Forward Busy configured or is a member of any hunt group. If the transfer-to directory number has Call Forward No Answer (CFNA) enabled, Cisco Unified CME recalls a transferred call only if the transfer-recall timeout is less than the timeout set with the `call-forward noan` command. If the transfer-recall timeout is set to more than the CFNA timeout, the call is forwarded to the CFNA target number if the transfer-to party does not answer.

If the transferor is busy at the time of the recall, Cisco Unified CME attempts the recall again after the retry timer expires. The maximum number of retries is two. If the transferor phone remains busy, the call is disconnected after the third recall attempt.

Use this command in telephony-service configuration mode to enable the transfer-recall timer at the system level for all directory numbers. Use this command in ephone-dn configuration mode to enable the transfer-recall timer for a particular directory number, or use the command in ephone-dn template mode to apply it to one or more directory numbers.

If you use an ephone-dn template to apply a command to a directory number and you also use the same command in ephone-dn configuration mode for the same directory number, the value that you set in ephone-dn
configuration mode has priority. This command, set in telephony-service configuration mode, has the lowest priority.

**Examples**

The following example shows that transfer recall is enabled for extension 1030 (ephone-dn 103), which is assigned to ephone 3. If extension 1030 forwards a call and the transfer-to party does not answer, after 60 seconds the unanswered call is sent back to extension 1030 (transferor).

```
ephone-dn 103
    number 1030
    name Smith, John
    timeouts transfer-recall 60
    
ephone 3
    mac-address 002D.264E.54FA
    type 7962
    button 1:103
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-forward busy</td>
<td>Enables call forwarding so that incoming calls to a busy extension (ephone-dn) are forwarded to another extension.</td>
</tr>
<tr>
<td>call-forward noan</td>
<td>Enables call forwarding so that incoming calls to an extension (ephone-dn) that does not answer are forwarded to another number.</td>
</tr>
<tr>
<td>transfer-mode</td>
<td>Specifies the call transfer method for an individual directory number.</td>
</tr>
<tr>
<td>transfer-system</td>
<td>Specifies the call transfer method globally for all directory numbers.</td>
</tr>
<tr>
<td>trunk</td>
<td>Associates an ephone-dn with a foreign exchange office (FXO) port.</td>
</tr>
</tbody>
</table>
timeouts transfer-recall (voice register global)

To enable Cisco Unified CME to recall a transferred call if the transfer-to party does not answer or is busy, use the **timeouts transfer-recall** command in voice register global configuration mode. To reset to the default value, use the `no` form of this command.

**timeouts transfer-recall seconds**

**no timeouts transfer-recall**

---

**Syntax Description**

| seconds | Duration, in seconds, to wait before recalling a transferred call. Range: 1 to 1800. Default: 0 (disabled). |

**Command Default**

Transfer recall is disabled (0 seconds) on a Cisco Unified SIP IP phone.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IOS XE Everest 16.4.1</td>
<td>Cisco Unified CME 11.6</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables Call Transfer Recall and sets the number of seconds that Cisco Unified CME waits before sending a transferred call back to the phone that initiated the transfer (transferor).

If the transfer-recall timer set with this command expires before the transfer-to party answers a call, the call is directed back to the transferor and the message, “Transfer Recall From xxxx” displays on the transferor phone. If the transferor is busy after the recall, the timer restarts. The maximum number of retries is two if the transfer-to party remains busy or does not answer. The transferor and transfer-to party must be on the same Cisco Unified CME router; the transferee party can be remote.

Transfer recall is not supported if the transfer-to party has Call Forward Busy configured or is a member of any hunt group. The transferor phone and transfer-to phone must be registered to the same Cisco Unified CME, however the transferee phone can be remote. If the transfer-to directory number has Call Forward No Answer (CFNA) enabled, Cisco Unified CME recalls the call only if the transfer-recall timeout is set to less than the CFNA timeout. If the transfer-recall timeout is set to more than the CFNA timeout, the call is forwarded to the CFNA target number after the transfer-to party does not answer. If the transfer-recall timeout is equal to the CFNA timeout, the call is forwarded to the CFNA target number as the CFNA timeout expires before the transfer-recall timeout.

When Call Forward All is configured in Cisco Unified CME, the call is forwarded directly to call forward target number irrespective of whether the phone is busy or idle. In this scenario, transfer recall is not applicable after the call is forwarded.

If the transferor phone is busy, Cisco Unified CME attempts the recall again after the transfer-recall timeout value expires. Cisco Unified CME attempts a recall up to three times. If the transferor phone remains busy, the call is disconnected after the third recall attempt. Also, if the transferor phone is a shared line, and if one of the phones is idle, the transfer recall is directed to the transferor phone that is idle.

Use this command in voice register global configuration mode to enable the transfer-recall timer at the system level for all directory numbers.
The `timeouts transfer-recall` command in voice register global configuration mode has lesser priority than the value that you set in voice register dn configuration mode for the same directory number.

**Examples**

The following example shows that transfer recall is enabled for 20 seconds. If the transfer-to party does not answer after 20 seconds, the unanswered call is sent back to the (transferor).

```
Router(config)# voice register global
Router(config-register-global)# timeouts transfer-recall 20
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>timeouts transfer-recall</code> (Ephone-dn (config-ephone-dn) and Telephony-service configuration (config-telephony))</td>
<td>Enables Cisco Unified CME to recall a transferred call if the transfer-to party does not answer or is busy.</td>
</tr>
</tbody>
</table>
timeouts transfer-recall (voice register dn)

To enable Cisco Unified CME to recall a transferred call if the transfer-to party does not answer or is busy, use the timeouts transfer-recall command in voice register dn configuration mode. To reset to the default value, use the no form of this command.

```
timeouts transfer-recall seconds
no timeouts transfer-recall
```

**Syntax Description**

```
seconds  Duration, in seconds, to wait before recalling a transferred call. Range: 1 to 1800. Default: 0 (disabled).
```

**Command Default**

Transfer recall is disabled (0 seconds) on a Cisco Unified SIP IP phone.

**Command Modes**

Voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IOS XE Everest 16.4.1</td>
<td>Cisco Unified CME 11.6</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables Call Transfer Recall and sets the number of seconds that Cisco Unified CME waits before sending a transferred call back to the phone that initiated the transfer (transferor).

If the transfer-recall timer set with this command expires before the transfer-to party answers a call, the call is directed back to the transferor and the message, “Transfer Recall From xxxx” displays on the transferor phone. If the transferor is busy after the recall, the timer restarts. The maximum number of retries is two if the transfer-to party remains busy or does not answer. The transferor and transfer-to party must be on the same Cisco Unified CME router; the transferee party can be remote.

Transfer recall is not supported if the transfer-to party has Call Forward Busy configured or is a member of any hunt group. The transferor phone and transfer-to phone must be registered to the same Cisco Unified CME, however the transferee phone can be remote. If the transfer-to directory number has Call Forward No Answer (CFNA) enabled, Cisco Unified CME recalls the call only if the transfer-recall timeout is set to less than the CFNA timeout. If the transfer-recall timeout is set to more than the CFNA timeout, the call is forwarded to the CFNA target number after the transfer-to party does not answer. If the transfer-recall timeout is equal to the CFNA timeout, the call is forwarded to the CFNA target number as the CFNA timeout expires before the transfer-recall timeout.

When Call Forward All is configured in Cisco Unified CME, the call is forwarded directly to call forward target number irrespective of whether the phone is busy or idle. In this scenario, transfer recall is not applicable after the call is forwarded.

If the transferor phone is busy, Cisco Unified CME attempts the recall again after the transfer-recall timeout value expires. Cisco Unified CME attempts a recall up to three times. If the transferor phone remains busy, the call is disconnected after the third recall attempt. Also, if the transferor phone is a shared line, and if one of the phones is idle, the transfer recall is directed to the transferor phone that is idle.

Use this command in voice register dn configuration mode to enable the transfer-recall timer for a particular directory number.
If you use the **timeouts transfer-recall** command in voice register dn configuration mode for the same directory number, the value that you set in voice register dn configuration mode has priority than the value set in the voice register global configuration mode (this has the lowest priority).

**Examples**

The following example shows that transfer recall is enabled for extension 111 (voice register dn 1). If extension 111 forwards a call to voice register dn 2 and the transfer-to party does not answer, after 20 seconds the unanswered call is sent back to extension 1111 (transferor).

```
voice register dn 1
  timeouts transfer-recall 20
  number 111
voice register dn 2
  number 222
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>timeouts transfer-recall</strong> (Ephone-dn (config-ephone-dn) and Telephony-service configuration (config-telephony))</td>
<td>Enables Cisco Unified CME to recall a transferred call if the transfer-to party does not answer or is busy.</td>
</tr>
</tbody>
</table>
time-webedit (telephony-service)

To enable the system administrator to set time on the Cisco Unified CME router through the web interface, use the `time-webedit` command in telephony-service configuration mode. To disable this feature, use the `no` form of this command.

```
time-webedit
no time-webedit
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Time-setting through the web interface is disabled.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command is deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `time-webedit` allows a local administrator of the Cisco Unified CME router to change and set time through the web-based graphical user interface (GUI).

**Note**

Cisco discourages this method for setting network time. The router should be set up to automatically synchronize its router clock from a network-based clock source using Network Time Protocol (NTP). In the rare case that a network NTP clock source is not available, the `time-webedit` can be used to allow manual setting and resetting of the router clock through the Cisco CME GUI.

**Examples**

The following example enables web editing of time:

```
Router(config)# telephony-service
Router(config-telephony)# time-webedit
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dn-webedit</td>
<td>Enables adding of directory numbers through a web interface.</td>
</tr>
<tr>
<td>telephony-service</td>
<td>Enters telephony-service configuration mode.</td>
</tr>
</tbody>
</table>
time-zone

To set the time zone so that the correct local time is displayed on SCCP Cisco Unified IP phones, use the time-zone command in telephony-service configuration mode. To disable a time-zone setting configured with the time-zone command and return to the default time zone (Pacific Standard Time), use the no form of this command.

\texttt{time-zone\ number}
\texttt{no\ time-zone}

**Syntax Description**

<table>
<thead>
<tr>
<th>number</th>
<th>Numeric code for a named time zone. The following are the selections. The numbers in parentheses indicate the offset from Coordinated Universal Time (UTC) in minutes.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Note</td>
<td>The time shows incorrectly for phones configured in West Africa during the Summer Time. For West Africa, Summer Time or Daylight Savings Time (DST) is not used. There is no correct time zone in this time zone list to account for this time zone.</td>
</tr>
</tbody>
</table>

- 1 — Dateline Standard Time (-720)
- 2 — Samoa Standard Time (-660)
- 3 — Hawaiian Standard Time (-600)
- 4 — Alaskan Standard/Daylight Time (-540)
- 5 — Pacific Standard/Daylight Time (-480)
- 6 — Mountain Standard/Daylight Time (-420)
- 7 — United States (US) Mountain Standard Time (-420)
- 8 — Central Standard/Daylight Time (-360)
- 9 — Mexico Standard/Daylight Time (-360)
- 10 — Canada Central Standard Time (-360)
- 11 — SA Pacific Standard Time (-300)
- 12 — Eastern Standard/Daylight Time (-300)
- 13 — US Eastern Standard Time (-300)
- 14 — Atlantic Standard/Daylight Time (-240)
- 15 — South America (SA) Western Standard Time (-240)
- 16 — Newfoundland Standard/Daylight Time (-210)
- 17 — SA Standard/Daylight Time (-180)
- 18 — SA Eastern Standard Time (-180)
- 19 — Mid-Atlantic Standard/Daylight Time (-120)
- 20 — Azores Standard/Daylight Time (-60)
- 21 — UTC Standard/Daylight Time (+0)
- 22 — Greenwich Standard Time (+0)
- 23 — Western Europe Standard/Daylight Time (+60)
- 24 — GTB (Athens, Istanbul, Minsk) Standard/Daylight Time (+60)
- 25 — Egypt Standard/Daylight Time (+60)
- 26 — Eastern Europe Standard/Daylight Time (+60)
time-zone

<table>
<thead>
<tr>
<th>number continued</th>
</tr>
</thead>
<tbody>
<tr>
<td>27 — Romance Standard/Daylight Time (+120)</td>
</tr>
<tr>
<td>28 — Central Europe Standard/Daylight Time (+120)</td>
</tr>
<tr>
<td>29 — South Africa Standard Time (+120)</td>
</tr>
<tr>
<td>30 — Jerusalem Standard/Daylight Time (+120)</td>
</tr>
<tr>
<td>31 — Saudi Arabia Standard Time (+180)</td>
</tr>
<tr>
<td>32 — Russian Standard/Daylight Time (+180)</td>
</tr>
<tr>
<td>33 — Iran Standard/Daylight Time (+210)</td>
</tr>
<tr>
<td>34 — Caucasus Standard/Daylight Time (+240)</td>
</tr>
<tr>
<td>35 — Arabian Standard Time (+240)</td>
</tr>
<tr>
<td>36 — Afghanistan Standard Time (+270)</td>
</tr>
<tr>
<td>37 — West Asia Standard Time (+300)</td>
</tr>
<tr>
<td>38 — Ekaterinburg Standard Time (+300)</td>
</tr>
<tr>
<td>39 — India Standard Time (+330)</td>
</tr>
<tr>
<td>40 — Central Asia Standard Time (+360)</td>
</tr>
<tr>
<td>41 — Southeast Asia Standard Time (+420)</td>
</tr>
<tr>
<td>42 — China Standard/Daylight Time (+480)</td>
</tr>
<tr>
<td>43 — Taipei Standard Time (+480)</td>
</tr>
<tr>
<td>44 — Tokyo Standard Time (+540)</td>
</tr>
<tr>
<td>45 — Central Australia Standard/Daylight Time (+570)</td>
</tr>
<tr>
<td>46 — Australia Central Standard Time (+570)</td>
</tr>
<tr>
<td>47 — East Australia Standard Time (+600)</td>
</tr>
<tr>
<td>48 — Australia Eastern Standard/Daylight Time (+600)</td>
</tr>
<tr>
<td>49 — West Pacific Standard Time (+600)</td>
</tr>
<tr>
<td>50 — Tasmania Standard/Daylight Time (+600)</td>
</tr>
<tr>
<td>51 — Central Pacific Standard Time (+660)</td>
</tr>
<tr>
<td>52 — Fiji Standard Time (+720)</td>
</tr>
<tr>
<td>53 — New Zealand Standard/Daylight Time (+720)</td>
</tr>
<tr>
<td>54 — Venezuela Standard Time (-270)</td>
</tr>
<tr>
<td>55 — Pacific SA Daylight Time (-180)</td>
</tr>
<tr>
<td>56 — Pacific SA Standard Time (-240)</td>
</tr>
</tbody>
</table>

**Command Default**

The default is time-zone 5, Pacific Standard/Daylight Time (-480).

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>This command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command works with the vendorConfig section of the Sep*.cnf.xml configuration file, which is read by the phone firmware when the Cisco IP Phone is booted up. Certain phones, such as the Cisco Unified IP Phone...
7906, 7911, 7931, 7941, 7942, 7945, 7961, 7962, 7965, 7970, 7971, and 7975, obtain Coordinated Universal Time (UTC) from the clock of the Cisco router. To display the correct local time, the time display on these phones must be offset by using this command.

This command is not required for Cisco Unified IP Phone 7902G, 7905G, 7912G, 7920, 7921, 7935, 7936, 7940, 7960, or 7985G.

For changes to the time-zone settings take effect, the Sep*.cnf.xml file must be updated by using the `create cnf-files` command and the Cisco IP phones must rebooted by using the `reset` command.

**Examples**

The following example sets the Cisco IP Phone 7970 units to Fiji Standard Time:

```
Router(config)# telephony-service
Router(config-telephony)# time-zone 53
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>create cnf-files</code></td>
<td>Sets display and phone functionality for the Cisco IP Phone 7970 units using the vendorConfig parameters of the downloaded firmware’s Sep*.cnf.xml configuration file.</td>
</tr>
<tr>
<td><code>reset (telephony-service)</code></td>
<td>Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.</td>
</tr>
</tbody>
</table>
**timezone (voice register global)**

To set the time zone used for SIP phones in a Cisco Unified CME system, use the `timezone` command in voice register global configuration mode. To return to the default, use the `no` form of this command.

```
timezone number
no timezone
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Argument</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>number</td>
<td>Range is 1 to 53. Default is 5, Pacific Standard/Daylight Time</td>
</tr>
</tbody>
</table>

**Command Default**

Default is 5, Pacific Standard/Daylight Time.

**Command Modes**

Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The following table lists the supported time zone numbers and the corresponding description.

**Table 74: Time Zones**

<table>
<thead>
<tr>
<th>Number</th>
<th>Description</th>
<th>Offset in Minutes</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Dateline Standard Time</td>
<td>-720</td>
</tr>
<tr>
<td>2</td>
<td>Samoa Standard Time</td>
<td>-660</td>
</tr>
<tr>
<td>3</td>
<td>Hawaiian Standard Time</td>
<td>-600</td>
</tr>
<tr>
<td>4</td>
<td>Alaskan Standard/Daylight Time</td>
<td>-540</td>
</tr>
<tr>
<td>5</td>
<td>Pacific Standard/Daylight Time</td>
<td>-480</td>
</tr>
<tr>
<td>6</td>
<td>Mountain Standard/Daylight Time</td>
<td>-420</td>
</tr>
<tr>
<td>7</td>
<td>US Mountain Standard Time</td>
<td>-420</td>
</tr>
<tr>
<td>8</td>
<td>Central Standard/Daylight Time</td>
<td>-360</td>
</tr>
<tr>
<td>9</td>
<td>Mexico Standard/Daylight Time</td>
<td>-360</td>
</tr>
<tr>
<td>10</td>
<td>Canada Central Standard Time</td>
<td>-360</td>
</tr>
<tr>
<td>11</td>
<td>SA Pacific Standard Time</td>
<td>-300</td>
</tr>
<tr>
<td>12</td>
<td>Eastern Standard/Daylight Time</td>
<td>-300</td>
</tr>
<tr>
<td>13</td>
<td>US Eastern Standard Time</td>
<td>-300</td>
</tr>
<tr>
<td>14</td>
<td>Atlantic Standard/Daylight Time</td>
<td>-240</td>
</tr>
<tr>
<td>Number</td>
<td>Description</td>
<td>Offset in Minutes</td>
</tr>
<tr>
<td>--------</td>
<td>------------------------------------</td>
<td>-------------------</td>
</tr>
<tr>
<td>15</td>
<td>SA Western Standard Time</td>
<td>-240</td>
</tr>
<tr>
<td>16</td>
<td>Newfoundland Standard/Daylight Time</td>
<td>-210</td>
</tr>
<tr>
<td>17</td>
<td>South America Standard/Daylight Time</td>
<td>-180</td>
</tr>
<tr>
<td>18</td>
<td>SA Eastern Standard Time</td>
<td>-180</td>
</tr>
<tr>
<td>19</td>
<td>Mid-Atlantic Standard/Daylight Time</td>
<td>-120</td>
</tr>
<tr>
<td>20</td>
<td>Azores Standard/Daylight Time</td>
<td>-60</td>
</tr>
<tr>
<td>21</td>
<td>GMT Standard/Daylight Time</td>
<td>+0</td>
</tr>
<tr>
<td>22</td>
<td>Greenwich Standard Time</td>
<td>+0</td>
</tr>
<tr>
<td>23</td>
<td>W. Europe Standard/Daylight Time</td>
<td>+60</td>
</tr>
<tr>
<td>24</td>
<td>GTB Standard/Daylight Time</td>
<td>+60</td>
</tr>
<tr>
<td>25</td>
<td>Egypt Standard/Daylight Time</td>
<td>+60</td>
</tr>
<tr>
<td>26</td>
<td>E. Europe Standard/Daylight Time</td>
<td>+60</td>
</tr>
<tr>
<td>27</td>
<td>Romance Standard/Daylight Time</td>
<td>+120</td>
</tr>
<tr>
<td>28</td>
<td>Central Europe Standard/Daylight Time</td>
<td>+120</td>
</tr>
<tr>
<td>29</td>
<td>South Africa Standard Time</td>
<td>+120</td>
</tr>
<tr>
<td>30</td>
<td>Jerusalem Standard/Daylight Time</td>
<td>+120</td>
</tr>
<tr>
<td>31</td>
<td>Saudi Arabia Standard Time</td>
<td>+180</td>
</tr>
<tr>
<td>32</td>
<td>Russian Standard/Daylight Time</td>
<td>+180</td>
</tr>
<tr>
<td>33</td>
<td>Iran Standard/Daylight Time</td>
<td>+210</td>
</tr>
<tr>
<td>34</td>
<td>Caucasus Standard/Daylight Time</td>
<td>+240</td>
</tr>
<tr>
<td>35</td>
<td>Arabian Standard Time</td>
<td>+240</td>
</tr>
<tr>
<td>36</td>
<td>Afghanistan Standard Time</td>
<td>+270</td>
</tr>
<tr>
<td>37</td>
<td>West Asia Standard Time</td>
<td>+300</td>
</tr>
<tr>
<td>38</td>
<td>Ekaterinburg Standard Time</td>
<td>+300</td>
</tr>
<tr>
<td>39</td>
<td>India Standard Time</td>
<td>+330</td>
</tr>
<tr>
<td>40</td>
<td>Central Asia Standard Time</td>
<td>+360</td>
</tr>
<tr>
<td>41</td>
<td>SE Asia Standard Time</td>
<td>+420</td>
</tr>
<tr>
<td>42</td>
<td>China Standard/Daylight Time</td>
<td>+480</td>
</tr>
<tr>
<td>Number</td>
<td>Description</td>
<td>Offset in Minutes</td>
</tr>
<tr>
<td>--------</td>
<td>--------------------------------------------------</td>
<td>-------------------</td>
</tr>
<tr>
<td>43</td>
<td>Taipei Standard Time</td>
<td>+480</td>
</tr>
<tr>
<td>44</td>
<td>Tokyo Standard Time</td>
<td>+540</td>
</tr>
<tr>
<td>45</td>
<td>Cen. Australia Standard/Daylight Time</td>
<td>+570</td>
</tr>
<tr>
<td>46</td>
<td>AUS Central Standard Time</td>
<td>+570</td>
</tr>
<tr>
<td>47</td>
<td>E. Australia Standard Time</td>
<td>+600</td>
</tr>
<tr>
<td>48</td>
<td>AUS Eastern Standard/Daylight Time</td>
<td>+600</td>
</tr>
<tr>
<td>49</td>
<td>West Pacific Standard Time</td>
<td>+600</td>
</tr>
<tr>
<td>50</td>
<td>Tasmania Standard/Daylight Time</td>
<td>+600</td>
</tr>
<tr>
<td>51</td>
<td>Central Pacific Standard Time</td>
<td>+660</td>
</tr>
<tr>
<td>52</td>
<td>Fiji Standard Time</td>
<td>+720</td>
</tr>
<tr>
<td>53</td>
<td>New Zealand Standard/Daylight Time</td>
<td>+720</td>
</tr>
<tr>
<td>54</td>
<td>Venezuela Standard Time</td>
<td>-270</td>
</tr>
<tr>
<td>55</td>
<td>Pacific SA Daylight Time</td>
<td>-180</td>
</tr>
<tr>
<td>56</td>
<td>Pacific SA Standard Time</td>
<td>-240</td>
</tr>
</tbody>
</table>

**Examples**

The following example shows how to set the time zone to 8, Central Standard Daylight Time:

```plaintext
Router(config)# voice register global
Router(config-register-global)# timezone 8
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>dst (voice register global)</code></td>
<td>Sets the time period for daylight saving time on SIP phones.</td>
</tr>
<tr>
<td><code>dst auto-adjust (voice register global)</code></td>
<td>Enables automatic adjustment of daylight saving time on SIP phones.</td>
</tr>
<tr>
<td><code>time-format (voice register global)</code></td>
<td>Selects a 12-hour clock or a 24-hour clock for the time display format on SIP phones in a Cisco CME system</td>
</tr>
</tbody>
</table>
transfer max-length

To specify the maximum length of the transfer number, use the `transfer max-length` command in voice register pool or voice register template configuration mode. To disable the maximum length, use the `no` form of this command.

```
transfer max-length max-length
no transfer max-length max-length
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>max-length</code></td>
<td>Maximum length of the transfer number. Range is 3 to 16.</td>
</tr>
</tbody>
</table>

**Command Default**

No maximum length is specified for the transfer number.

**Command Modes**

Voice register pool configuration (config-register-pool)

Voice register template configuration ((config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `transfer max-length` command is used to indicate the maximum length of the number being dialed for a call transfer. When only a specific number of digits are to be allowed during a call transfer, a value between 3 and 16 is configured. When the number dialed exceeds the maximum length configured, then the call transfer is blocked.

**Examples**

The following example shows how to configure the maximum length of the transfer number under voice register pool 1. Because the maximum length is configured as 5, only call transfers to Cisco Unified SIP IP phones with a five-digit directory number are allowed. All call transfers to directory numbers with more than five digits are blocked.

```
Router# configure terminal
Router(config)# voice register pool 1
Router(config-register-pool)# transfer max-length 5
```

The following example shows how to configure the maximum length of the transfer number for a set of phones under voice register template 2:

```
Router# configure terminal
Router(config)# voice register template 2
Router(config-register-temp)# transfer max-length 10
```

**Command**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>voice register pool</code></td>
<td>Enters voice register pool configuration mode and creates a pool configuration for a SIP IP phone in Cisco Unified CME or for a set of SIP phones in Cisco Unified SIP SRST</td>
</tr>
<tr>
<td><code>voice register template</code></td>
<td>Enters voice register template configuration mode and defines a template of common parameters for SIP phones.</td>
</tr>
</tbody>
</table>
**transfer-attended (voice register template)**

To enable a soft key for attended transfer in a SIP phone template, use the `transfer-attended` command in voice register template configuration mode. To disable the soft key, use the `no` form of this command.

```
transfer-attended
no transfer-attended
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Soft key is enabled.

**Command Modes**

Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables a soft key for attended transfer in the specified template which can then be applied to SIP phones in Cisco CME. The attended transfer soft key is enabled by default. To disable the attended transfer soft key, use the `no transfer-attended` command. To apply the template to a SIP phone, use the `template` command in voice register pool configuration mode.

A blind transfer is one in which the transferring extension connects the caller to a destination extension before ringback begins. An attended transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

**Examples**

The following example shows how to disable attended transfer in template 1:

```
Router(config)# voice register template 1
Router(config-register-temp)# no transfer-attended
```

**Related Commands**

<table>
<thead>
<tr>
<th>Related Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>conference (voice register template)</td>
<td>Enables the soft key for conference in a SIP phone template.</td>
</tr>
<tr>
<td>template</td>
<td>Applies a template to a SIP phone.</td>
</tr>
<tr>
<td>transfer-blind (voice register template)</td>
<td>Enables a soft key for blind transfer in a SIP phone template.</td>
</tr>
</tbody>
</table>
transfer-blind (voice register template)

To enable a soft key for blind transfer in a SIP phone template, use the `transfer-blind` command in voice register template configuration mode. To disable the soft key, use the `no` form of this command.

```plaintext
transfer-blind

no transfer-blind
```

**Syntax Description**
- This command has no arguments or keywords.

**Command Default**
- Soft key is enabled.

**Command Modes**
- Voice register template configuration (config-register-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables a soft key for blind transfer in the specified template which can then be applied to SIP phones in Cisco CME. The blind transfer soft key is enabled by default. To disable the blind transfer soft key, use the `no transfer-blind` command. To apply the template to a SIP phone, use the `template` command in voice register pool configuration mode.

A blind transfer is one in which the transferring extension connects the caller to a destination extension before ringback begins. An attended transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

**Examples**

The following example shows how to disable blind transfer in template 1:

```plaintext
Router(config)# voice register template 1
Router(config-register-temp)# no transfer-blind
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>conference (voice register template)</td>
<td>Enables the soft key for conference in a SIP phone template.</td>
</tr>
<tr>
<td>template</td>
<td>Applies a template to a SIP phone.</td>
</tr>
<tr>
<td>transfer-attended (voice register template)</td>
<td>Enables the soft key for attended transfer on SIP phones.</td>
</tr>
</tbody>
</table>
transfer-digit-collect

To select the digit-collection method for consultative call-transfers, use the `transfer-digit-collect` command in telephony-service configuration mode for Cisco Unified CME or in call-manager-fallback configuration mode for Cisco Unified SRST. To reset to the default value, use the `no` form of this command.

```
transfer-digit-collect {new-call|orig-call}
no transfer-digit-collect
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>new-call</code></td>
<td>Dialed digits are collected from new call leg. Default value.</td>
</tr>
<tr>
<td><code>orig-call</code></td>
<td>Dialed digits are collected from original call leg.</td>
</tr>
</tbody>
</table>

**Command Default**

Digits are collected from the new consultative call-leg (`new-call` keyword).

**Command Modes**

- Telephony-service configuration (config-telephony)
- Call-manager-fallback configuration (config-cm-fallback)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3 Cisco Unified SRST 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies whether the dialed digits of the target number are collected on the original call leg or on the new call leg that is created when a phone user initiates a consultative call-transfer.

For consultative transfers, a local number is matched on the `number` command in ephone-dn configuration mode; a PSTN number is matched on the `transfer-pattern` command in telephony service mode.

The `orig-call` keyword selects the method used in versions before Cisco Unified CME 4.3 and Cisco Unified SRST 4.3. After a phone user presses the Transfer soft key, the dialed digits of the target number are collected on the original call leg and buffered until either a local ephone-dn or transfer-pattern is matched. When the transfer-to number is matched, the original call is put on hold and an idle line or channel is seized to send the dialed digits from the buffer.

The `new-call` keyword selects the default method that is used in Cisco Unified CME 4.3 and later versions and Cisco Unified SRST 4.3 and later versions. The transfer-to digits are collected on a new consultative call-leg that is created when the user presses the Transfer soft key. The consultative call-leg is seized and the dialed digits are passed on without being buffered until the digits are matched and the consultative call-leg moves to the alerting state.

The `new-call` keyword is supported only if:

- The `transfer-system full-consult` command (default) is set in telephony-service configuration mode.
- The `transfer-mode consult` command (default) is set for transferor's directory number (ephone-dn).
- An idle line or channel is available for seizing, digit collection, and dialing.
A consultative transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

### Examples

The following example shows the digit-collection set to the method used in versions before Cisco Unified CME 4.3 and Cisco Unified SRST 4.3:

```
Router(config)# telephony-service
Router(config-telephony)# transfer-digit-collect orig-call
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>transfer-mode</td>
<td>Specifies the type of call transfer for an individual directory number that uses the ITU-T H.450.2 standard.</td>
</tr>
<tr>
<td>transfer-pattern</td>
<td>Allows the transfer of calls to phones outside the local Cisco Unified CME network.</td>
</tr>
<tr>
<td>(telephony-service)</td>
<td></td>
</tr>
<tr>
<td>transfer-system</td>
<td>Specifies the call transfer method for all IP phones on a Cisco Unified CME router using the ITU-T H.450.2 standard.</td>
</tr>
</tbody>
</table>
## transfer-mode

To specify the type of call transfer for an individual IP phone extension that uses the ITU-T H.450.2 standard, use the `transfer-mode` command in ephone-dn configuration mode. To remove this specification, use the `no` form of this command.

```
transfer-mode {blind|consult}
no transfer-mode
```

### Syntax Description

<table>
<thead>
<tr>
<th>Mode</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>blind</td>
<td>Transfers calls without consultation using a single phone line.</td>
</tr>
<tr>
<td>consult</td>
<td>Transfers calls with consultation using a second phone line, if available.</td>
</tr>
</tbody>
</table>

### Command Default

The ephone-dn uses the transfer-system value that was set systemwide.

### Command Modes

Ephone-dn configuration (config-ephone)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco CME Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)YT</td>
<td>2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command specifies the type of call transfer for an individual Cisco IP phone extension that is using the ITU-T H.450.2 protocol. It allows you to override the system default `transfer-system` setting (full-consult or full-blind) for that extension.

Call transfers that use H.450.2 can be blind or consultative. A blind transfer is one in which the transferring phone connects the caller to a destination extension before ringback begins. A consultative transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

You can specify blind or consultative transfer on a system-wide basis by using the `transfer-system` command. The system-wide setting can then be overridden for individual phone extensions by using the `transfer-mode` command. For example, in a Cisco CallManager Express (Cisco CME) network that is set up for consultative transfer, a specific extension with an auto-attendant that automatically transfers incoming calls to specific extension numbers can be set to use blind transfer, because auto-attendants do not use consultative transfer.

Use this command with Cisco IOS Telephony Services V2.1, Cisco CallManager Express 3.0, or a later version.

### Examples

The following example sets blind mode for call transfers from this directory number:

```
Router(config)# ephone-dn 21354
Router(config-ephone-dn)# transfer-mode blind
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>ephone-dn</strong></td>
<td>Enters ephone-dn configuration mode to set directory numbers and parameters for individual Cisco IP phone lines.</td>
</tr>
<tr>
<td><strong>transfer-system</strong></td>
<td>Specifies the call transfer method for all IP phones on a Cisco ITS router using the ITU-T H.450.2 standard.</td>
</tr>
</tbody>
</table>
transfer-park blocked

To prevent extensions on an ephone from parking calls, use the `transfer-park blocked` command in ephone or ephone-template configuration mode. To return to the default, use the `no` form of this command.

```
transfer-park blocked
no transfer-park blocked
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Transfer to park is allowed.

**Command Modes**
- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command prevents transfers to park that use the Transfer soft key and a call-park slot number, while allowing call-parks that use only the Park soft key. To prevent use of the Park soft key, use an ephone template to remove it from the phone.

An exception to this is made for phones with dedicated park slots. If the `transfer-park blocked` command is used on an ephone that has a dedicated park slot, the phone is blocked from parking calls at park slots other than the dedicated park slot, but is still able to park calls at its own dedicated park slot. On an IP phone, the user presses the Transfer soft key and the call-park feature access code (FAC) to park a call at the phone's dedicated park slot. On an analog phone, the user presses hookflash and the call-park FAC.

When the `transfer-park blocked` command is used on an ephone that does not have a dedicated park slot, the phone is blocked from parking any calls.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

**Examples**
The following example prevents ephone 25 and extensions 234, 235, and 236 from parking calls at any call-park slot.

```
ephone-dn 11
  number 234
ephone-dn 12
  number 235
ephone-dn 13
  number 236
ephone 25
  button 1:11 2:12 3:13
transfer-park blocked
```

The following example uses an ephone template to prevent ephone 26 and extension 76589 from parking calls at any call-park slot.

```
ephone-dn 33
```

---

*Cisco Unified Communications Manager Express Command Reference*
number 76589
ephone-template 1
transfer-park blocked
ephone 26
button 1:33
ephone-template 1

The following example sets up a dedicated park slot for the extensions on ephone 6 and blocks transfers to call park from extensions 2977, 2978, and 2979 on that phone. Those extensions can still park calls at the phone's dedicated park slot by using the Park soft key or Transfer and the call-park FAC.

ephone-dn 3
number 2558
name Park 2977
park-slot reserved-for 2977 timeout 60 limit 3 recall alternate 3754
ephone-dn 4
number 2977
ephone-dn 5
number 2978
ephone-dn 6
number 2979
ephone 6
button 1:4 2:5 3:6
transfer-park blocked
**transfer-pattern (telephony-service)**

To allow transfer of telephone calls from Cisco IP phones to phones other than Cisco IP phones, use the `transfer-pattern` command in telephony-service configuration mode. To disable these transfers, use the `no` form of this command.

```
transfer-pattern transfer-pattern [blind]
no transfer-pattern
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Argument</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>transfer-pattern</code></td>
<td>String of digits for permitted call transfers. Wildcards are allowed. A maximum of 32 transfer patterns can be entered, using a separate command for each one.</td>
</tr>
<tr>
<td><code>blind</code></td>
<td>(Optional) When H.450.2 consultative call transfer is used, this keyword forces transfers that match the pattern to be executed as blind transfers. Overrides settings made using the <code>transfer-system</code> and <code>transfer-mode</code> commands.</td>
</tr>
</tbody>
</table>

**Command Default**

Transfer of calls is enabled only to local Cisco IP phones.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco CME Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>The <code>blind</code> keyword was added.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command allows you to transfer calls to “other” phones—that is, to non-IP phones and phones outside of your network. A call is then established between the transferred party and the new recipient. By default, all Cisco IP phone extension numbers are allowed as transfer targets.

The `blind` keyword is valid only for systems that use Cisco IOS Telephony Services V2.1, Cisco CallManager Express 3.0, or a later version and applies only to consultative transfers made using the H.450.2 standard. The `blind` keyword forces calls that are transferred to numbers that match the transfer pattern to be executed as blind or full-blind transfers, overriding any settings made using the `transfer-system` and `transfer-mode` commands.

When defining transfers to non-local numbers, it is important to note that transfer-pattern digit matching is performed before translation-rule operations. Therefore, you should specify in this command the digits actually entered by phone users before they are translated.

Use of the `.T` control character for the `transfer-pattern` argument is not recommended. The `.T` control character indicates a variable-length dial string, which causes Cisco CME to wait for an interdigit timeout (default is 10 seconds) before transferring a call. To avoid the interdigit timeout, a matching transfer pattern should be used with the `transfer-pattern` command. For example, use the `transfer-pattern 9........` command instead of the `transfer-pattern .T` command.
Examples

The following example sets a transfer pattern. A maximum of 32 transfer patterns can be entered. In this example, 55501.. (the two periods are wildcards) permits transfers to any number in the range from 555-0100 to 555-0199.

Router(config)# telephony-service
Router(config-telephony)# transfer-pattern 55501..

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>transfer-mode</td>
<td>Specifies the type of call transfer for an individual IP phone extension number that uses the ITU-T H.450.2 standard.</td>
</tr>
<tr>
<td>transfer-system</td>
<td>Specifies the call transfer method for all Cisco CME extensions that use the ITU-T H.450.2 standard.</td>
</tr>
</tbody>
</table>
**transfer-pattern blocked**

To block all call transfers for a specific Cisco Unified SIP IP phone or a set of Cisco Unified SIP IP phone, use the `transfer-pattern blocked` command in voice register pool and voice register template configuration mode. To allow call transfers, use the `no` form of this command.

```
transfer-pattern blocked
no transfer-pattern blocked
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Call transfers for a specific Cisco Unified SIP IP phone or a set of Cisco Unified SIP IP phone are allowed.

**Command Modes**
- Voice register pool configuration (config-register-pool)
- Voice register template configuration ((config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(2)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

When the `transfer-pattern blocked` command is configured for a specific phone, no call transfers are allowed from that phone over the trunk.

This feature forces unconditional blocking of all call transfers from a specific phone to any other non-local numbers (external calls from one trunk to another trunk). No call transfers from this specific phone are possible even when a transfer pattern matches the dialed digits for transfer.

**Examples**

The following example shows how to block all call transfers for voice register pool 5:

```
Router(config)# voice register pool 5
Router(config-register-pool)# transfer-pattern ?
   blocked  global  transfer pattern not allowed
Router(config-register-pool)# transfer-pattern blocked
```

The following example shows how to block all call transfers for a set of Cisco Unified SIP IP phones defined by voice register template 9:

```
Router(config)# voice register template 9
Router(config-register-temp)# transfer-pattern ?
   blocked  global  transfer pattern not allowed
Router(config-register-temp)# transfer-pattern blocked
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode and creates a pool configuration for a Cisco Unified SIP IP phone in Cisco Unified CME or for a set of Cisco Unified SIP IP phones in Cisco Unified SIP SRST.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td><strong>voice register template</strong></td>
<td>Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones.</td>
</tr>
</tbody>
</table>
transfer-system

To specify the call transfer method to be used by Cisco Unified IP phones in Cisco Unified CME, use the transfer-system command in telephony-service configuration mode. To disable the call transfer method, use the no form of this command.

```
transfer-system {blind|full-blind|full-consult [dss]|local-consult}
```

```
no transfer-system
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Option</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>blind</td>
<td>Transfers calls without consultation using a single phone line and the Cisco proprietary method. This is the default for Cisco CME 3.4 and earlier versions.</td>
</tr>
<tr>
<td>full-blind</td>
<td>Transfers calls without consultation using H.450.2 standard methods.</td>
</tr>
<tr>
<td>full-consult</td>
<td>Transfers calls using H.450.2 with consultation using a second phone line, if available. The calls fall back to full-blind if a second line is not available. This is the default for Cisco Unified CME 4.0 and later versions.</td>
</tr>
<tr>
<td>dss</td>
<td>Transfers calls with consultation to idle monitor lines. All other call-transfer behavior is identical to full-consult.</td>
</tr>
<tr>
<td>local-consult</td>
<td>Transfers calls with local consultation using a second phone line, if available, or the calls fall back to blind if the target for consultation or transfer is not local. This mode is intended for use primarily in Voice over Frame Relay (VoFR) networks, because the Cisco VoFR call transfer protocol does not support an end-to-end transfer-with-consultation mechanism. Not supported if transfer-to destination is on the Cisco ATA, Cisco VG224, or a SCCP-controlled FXS port.</td>
</tr>
</tbody>
</table>

**Command Default**

For Cisco Unified CME 4.0 and later versions, the transfer mode is full-consult. For Cisco CME 3.4 and earlier versions, the transfer mode is blind.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
<tr>
<td>12.3(11)T</td>
<td>Cisco CME 3.2</td>
<td>The dss keyword was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The command default was changed from blind to full-consult.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command with the default of full-consult was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Direct station select is a functionality that allows a multibutton phone user to transfer calls to an idle monitor line by pressing the Transfer key and the appropriate monitor button. The dss keyword permits consultative call transfer to monitored lines.
Call transfers can be blind or consultative. A blind transfer is one in which the transferring extension connects the caller to a destination extension before ringback begins. A consultative transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

The `transfer-system` command specifies whether the H.450.2 standard or a Cisco proprietary method will be used to communicate call transfer information across the network. When you specify use of the H.450.2 consultative or blind mode of transfer globally by using the `transfer-system` command (or by using the default), you can override this mode for individual phones by using the `transfer-mode` command. For example, in a system that is set up for consultative transfer, a specific extension with an auto-attendant that automatically transfers incoming calls to specific extension numbers can be set to use blind transfer, because auto-attendants do not use consultative transfer.

Prior to Cisco Unified CME 4.0, the default for this command specified the Cisco proprietary method. In Cisco Unified CME 4.0, the default was changed to specify the H.450.2 standard as the transfer method. Check the following table for configuration recommendations for different versions of Cisco Unified CME.

### Table 75: Transfer Method Recommendations

<table>
<thead>
<tr>
<th>Cisco Product</th>
<th>transfer-system Default</th>
<th>transfer-system to Use</th>
<th>Transfer Method Recommendation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified CME 4.0 and later versions</td>
<td>full-consult</td>
<td>full-consult or full-blind</td>
<td>Use H.450.2 for call transfer. Because this is the default for this version, you do not need to use the <code>transfer-system</code> command unless you want to use the <code>full-blind</code> or <code>dss</code> keyword. Optionally, you can use the proprietary Cisco method by using the <code>transfer-system</code> command with the <code>blind</code> or <code>local-consult</code> keyword.</td>
</tr>
<tr>
<td>Cisco CME 3.0 to 3.3</td>
<td>blind</td>
<td>full-consult or full-blind</td>
<td>Use H.450.2 for call transfer. You must explicitly configure the <code>transfer-system</code> command with the <code>full-consult</code> or <code>full-blind</code> keyword because H.450.2 is not the default for this version. Optionally, you can use the proprietary Cisco method by using the <code>transfer-system</code> command with the <code>blind</code> or <code>local-consult</code> keyword.</td>
</tr>
<tr>
<td>Cisco ITS 2.1 to 3.0</td>
<td>blind</td>
<td>blind or local-consult</td>
<td>Use the Cisco proprietary method. Because this is the default for this version, you do not need to use the <code>transfer-system</code> command unless you want to use the <code>local-consult</code> keyword. Optionally, you can use the H.450.2 standard for call transfer by using <code>transfer-system</code> command with the <code>full-consult</code> or <code>full-blind</code> keyword. You must also configure the router with a Tcl script that is contained in the file called app-h450-transfer.x.x.x.zip. This file is posted on the Cisco Unified CME software download website at <a href="http://www.cisco.com/cgi-bin/tablebuild.pl/ip-iostsp">http://www.cisco.com/cgi-bin/tablebuild.pl/ip-iostsp</a>.</td>
</tr>
</tbody>
</table>

### Examples

The following example sets full consultation as the call transfer method:

```
Router(config)# telephony-service
Router(config-telephony)# transfer-system full-consult
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>transfer-mode</td>
<td>Specifies the type of call transfer for an individual IP phone extension that uses the H.450.2 standard.</td>
</tr>
</tbody>
</table>
translate (ephone-dn)

To apply a translation rule in order to manipulate the digits that are dialed by users of Cisco Unified IP phones, use the translate command in ephone-dn or ephone-dn-template configuration mode. To disable the translation rule, use the no form of this command.

```
translate {called|calling} translation-rule-tag
no translate {called|calling}
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>called</code></td>
</tr>
<tr>
<td><code>calling</code></td>
</tr>
<tr>
<td><code>translation-rule-tag</code></td>
</tr>
</tbody>
</table>

**Command Default**

No translation rule is applied.

**Command Modes**

- Ephone-dn configuration (config-ephone-dn)
- Ephone-dn-template configuration (config-ephone-template-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-dn-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command allows you to select a preconfigured translation rule to modify the number dialed by a specific extension (Cisco Unified IP phone destination number, or ephone-dn). A translation rule is a general-purpose digit-manipulation mechanism that performs operations such as automatically adding telephone area and prefix codes to dialed numbers. The translation rules are applied to the voice ports created by the ephone-dn. The `called` keyword translates the called number, and the `calling` keyword translates the calling number.

The translation rule mechanism inserts a delay into the dialing process when digits are entered that do not explicitly match any of the defined translation rules. This delay is set by the interdigit timeout. The translation-rule mechanism uses the delay to ensure that it has acquired all of the digits from the phone user before making a final decision that there is no translation-rule match available (and therefore no translation operation to perform). To avoid this delay, it is recommended that you include a dummy translation rule to act as a pass-through rule for digit strings that do not require translation. For example, a rule like “^5 5” that maps a leading 5 digit into a 5 would be used to prevent the translation rule delay being applied to local extension numbers that started with a 5.
For this command to take effect, appropriate translation rules must have been created at the VoIP configuration level. Use the `show voice translation-rule` command to view the translation rules that you have defined. For information, see the Dial Peer Configuration on Voice Gateway Routers.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

**Examples**

The following example applies translation rule 20 to numbers called by extension 46839:

```bash
Router(config)# translation-rule 20
Router(config-translate)# rule 0 1234 2345 abbreviated
Router(config-translate)# exit
Router(config-ephone-dn)# number 46839
Router(config-ephone-dn)# translate called 20
```

The following example uses an ephone-dn-template to apply translation rule 20 to numbers called by extension 46839:

```bash
Router(config)# translation-rule 20
Router(config-translate)# rule 0 1234 2345 abbreviated
Router(config-translate)# exit
Router(config)# ephone-dn-template 1
Router(config-ephone-dn-template)# translate called 20
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 46839
Router(config-ephone-dn)# ephone-dn-template 1
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>rule</code></td>
<td>Defines a translation rule.</td>
</tr>
<tr>
<td><code>translation-rule</code></td>
<td>Creates a translation identifier and enters translation-rule configuration mode.</td>
</tr>
</tbody>
</table>
translate callback-number

To assign a translation profile for incoming or outgoing call legs on a Cisco IP phone, use the translate-profile command in call-manager-fallback configuration mode. To delete the translation profile from the voice port, use the no form of this command.

```
translate callback-number
no translate callback-number
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>incoming</td>
<td>Specifies that this translation profile handles incoming calls.</td>
</tr>
<tr>
<td>outgoing</td>
<td>Specifies that this translation profile handles outgoing calls.</td>
</tr>
<tr>
<td>name</td>
<td>Name of the translation profile.</td>
</tr>
</tbody>
</table>

| Command Default     | No default behavior or values. |
| Command Modes       | Voice translation-profile configuration (cfg-translation-profile) |

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use the translate callback-number command to translate a called number to E.164 format. The translated number allows a called or calling number to be presented in its local form. The translate callback-number command is applied when translation-profile is configured on dialpeers, ephone-dn, and voice register-dn. The translate callback-number command is effective when the configuration setup reached the SCCP and SIP IP phones.

**Examples**

The following example shows a configuration in which a translation profile called name1 is created with two voice translation rules. Rule1 consists of associated calling numbers, and rule2 consists of redirected called numbers. The Cisco IP phones in SRST mode are configured with name1.

```
voice translation-profile name1
translation calling rule1
translation called-direct rule2
call-manager-fallback
translation-profile incoming name1
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice translation-profile</td>
<td>Displays the configuration of a translation profile.</td>
</tr>
<tr>
<td>translate (call-manager-fallback)</td>
<td>Applies a translation rule to modify the phone number dialed or received by any Cisco IP phone user during CallManager fallback.</td>
</tr>
<tr>
<td>translation-rule</td>
<td>Creates a translation name and enters translation-rule configuration mode to apply rules to the translation name.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------------</td>
<td>--------------------------------------------------</td>
</tr>
<tr>
<td><code>voice translation-profile</code></td>
<td>Defines a translation profile for voice calls.</td>
</tr>
</tbody>
</table>
translate-outgoing (voice register pool)

To allow an explicit setting of translation rules on the VoIP dial peer in order to modify a phone number dialed by any Cisco IP phone user, use the `translate-outgoing` command in voice register pool configuration mode. To disable translation rules, use the `no` form of this command.

```
translate-outgoing {called|calling} rule-tag
no translate-outgoing {called|calling}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>called</td>
<td>Called party requires translation.</td>
</tr>
<tr>
<td>calling</td>
<td>Calling party requires translation.</td>
</tr>
<tr>
<td>rule-tag</td>
<td>The rule-tag is an arbitrarily chosen number by which the rule set is referenced. The range is from 1 to 2147483.</td>
</tr>
</tbody>
</table>

**Command Default**
Translation rules are enabled on the VoIP dial peer.

**Command Modes**
Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was added to Cisco CME.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Translation rules are a powerful general-purpose number-manipulation mechanism that perform operations such as automatically adding telephone area and prefix codes to dialed numbers. The translation rules are applied to VoIP dial peers created by Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified CallManager Express (Cisco Unified CME).

During registration, a dial peer is created, and that dial peer includes a default translation rule. The `translate-outgoing` command allows you to change the translation rule, if desired. The `translate-outgoing` command allows you to select a preconfigured number translation rule to modify the number dialed by a specific extension.

Translation rules must be set by using the `translate-outgoing` command before the `alias` command is configured in Cisco Unified SIP SRST.

Configure the `id` (voice register pool) command before any other voice register pool commands, including the `translate-outgoing` command. The `id` command identifies a locally available individual SIP phone or set of SIP phones.
Cisco Unified CME

The following is partial sample output from the `show running-config` command showing that called-party 1 requires translation.

```
voice register pool 1
  id mac 0030.94C2.A22A
  preference 5
  cor incoming call91 1 91011
  translate-outgoing called 1
```

Cisco Unified SIP SRST

The following is partial sample output from the `show running-config` command showing that called-party 1 requires translation.

```
voice register pool 1
  id mac 0030.94C2.A22A
  preference 5
  cor incoming call91 1 91011
  translate-outgoing called 1
  proxy 10.2.161.187 preference 1 monitor probe icmp-ping
  alias 1 94... to 91011 preference 8
  voice-class codec 1
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>alias (voice register pool)</td>
<td>Allows Cisco SIP IP phones to handle inbound PSTN calls to telephone numbers that are unavailable when the main proxy is not available.</td>
</tr>
<tr>
<td>id (voice register pool)</td>
<td>Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.</td>
</tr>
<tr>
<td>translate-outgoing (dial-peer)</td>
<td>Applies a translation rule to manipulate dialed digits on an outbound POTS or VoIP call leg.</td>
</tr>
<tr>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode for SIP phones.</td>
</tr>
</tbody>
</table>
translation-profile

To assign a translation profile for incoming or outgoing call legs on a Cisco Unified IP phone, use the translation-profile command in ephone-dn or ephone-dn-template configuration mode. To delete the translation profile from the voice port, use the no form of this command.

**translation-profile** {incoming|outgoing} name
no translation-profile {incoming|outgoing} name

Syntax Description

| incoming | Specifies that this translation profile handles incoming calls. |
| outgoing | Specifies that this translation profile handles outgoing calls. |
| name     | Name of the translation profile. |

Command Default

No default behavior or values

Command Modes

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco CME 3.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made available in ephone-dn-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

Usage Guidelines

Use the translation-profile command to assign a global predefined translation profile to an incoming or outgoing call leg.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example assigns the translation profile named call_in to handle translation of incoming calls and a translation profile named call_out to handle outgoing calls:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 2555
Router(config-ephone-dn)# translation-profile incoming call_in
Router(config-ephone-dn)# translation-profile outgoing call_out
```

The following example uses an ephone-dn-template to assign the translation profile named call_in to handle translation of incoming calls and the translation profile named call_out to handle outgoing calls:
Router(config)# ephone-dn-template 10
Router(config-ephone-dn-template)# translation-profile incoming call_in
Router(config-ephone-dn-template)# translation-profile outgoing call_out
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 2555
Router(config-ephone-dn)# ephone-dn-template 10

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice translation-profile</td>
<td>Displays the configuration of a translation profile.</td>
</tr>
<tr>
<td>translate</td>
<td>Applies a translation rule to modify the phone number dialed or received by any Cisco Unified IP phone user.</td>
</tr>
<tr>
<td>translation-rule</td>
<td>Creates a translation name and enters translation-rule configuration mode.</td>
</tr>
<tr>
<td>voice translation-profile</td>
<td>Defines a translation profile for voice calls.</td>
</tr>
<tr>
<td>voice translation-rule</td>
<td>Defines a translation rule for voice calls.</td>
</tr>
</tbody>
</table>
translation-profile incoming

To assign a translation profile for incoming call legs on a SIP phone, use the translation-profile incoming command in voice-register-dn configuration mode. To delete the translation profile from the directory number, use the no form of this command.

**translation-profile incoming name**
**no translation-profile incoming**

**Syntax Description**

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>name</td>
<td>Name of the translation profile to apply to incoming calls to this directory number. This is the name argument that was created for the profile with the voice translation-profile command.</td>
</tr>
</tbody>
</table>

**Command Default**

No translation profile is assigned to the directory number.

**Command Modes**

Voice register dn configuration (config-register-dn)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to assign a predefined translation profile to incoming call legs on the specified directory number. The translation profile that you assign is created by using the voice translation-profile command.

**Examples**

The following example shows that the translation profile named call_in is assigned to handle translation of incoming calls to directory number 1:

```
Router(config)# voice register dn 1
Router(config-register-dn)# number 2555
Router(config-register-dn)# translation-profile incoming call_in
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice translation-profile</td>
<td>Displays the configuration of a translation profile.</td>
</tr>
<tr>
<td>translate (translation profiles)</td>
<td>Associates a translation rule with a voice translation profile.</td>
</tr>
<tr>
<td>voice translation-profile</td>
<td>Defines a translation profile for voice calls.</td>
</tr>
<tr>
<td>voice translation-rule</td>
<td>Defines a translation rule for voice calls.</td>
</tr>
</tbody>
</table>
transport (voice register pool-type)

To define the default transport type supported by the new phone, use the `transport` command in voice register pool-type mode. To remove the description, use the `no` form of this command.

**Syntax Description**

- `udp` (Optional) Selects UDP as the transport layer protocol. This is the default transport protocol.
- `tcp` (Optional) Selects TCP as the transport layer protocol.

**Command Default**
The default transport protocol is UDP. When the reference-pooltype command is configured, the transport value of the reference phone is inherited.

**Command Modes**
Voice Register Pool-Type Configuration (config-register-pool-type)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(3)M</td>
<td>Cisco SIP CME 10.0</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to define the default transport type. If this parameter is not configured, UDP is used as default value. Currently, except the CiscoMobile-iOS and Jabber-Android, all other phone types uses UDP as default transport type. The default transport type will be ignored when the ‘session-transport {udp | tcp}’ command is configured for the pool.

**Example**

The following example shows how to specify a description for a phone model using the description command:

```bash
Router(config)# voice register pool-type 9900
Router(config-register-pool-type)# transport tcp
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register pool-type</td>
<td>Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.</td>
</tr>
</tbody>
</table>
To associate an ephone-dn with a foreign exchange office (FXO) port, use the **trunk** command in ephone-dn configuration mode. To disassociate the ephone-dn from the trunk number, use the **no** form of this command.

```
trunk digit-string [timeout seconds] [transfer-timeout seconds] [monitor-port port]
no trunk
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>digit-string</td>
<td>The number of the trunk line.</td>
</tr>
<tr>
<td>timeout seconds</td>
<td>(Optional) Interdigit timeout between dialed digits, in seconds. Range is 3 to 30. Default is 3.</td>
</tr>
<tr>
<td>transfer-timeout</td>
<td>(Optional) Number of seconds that Cisco Unified CME waits for the transfer-to party to answer a call after which the call is recalled to the phone that initiated the transfer. This keyword is supported for dual-line ephone-dns only. Range is 5 to 60000. Default is disabled.</td>
</tr>
<tr>
<td>monitor-port port</td>
<td>(Optional) Enables a button lamp or icon that shows that the specified port is in use. Port argument is platform-dependent; type ? to display syntax.</td>
</tr>
</tbody>
</table>

**Command Default**

Ephone-dns are not associated with FXO ports.

**Command Modes**

Ephone-dn configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(11)T</td>
<td>Cisco CME 3.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The monitor-port and transfer-timeout keywords were added and support for dual-line ephone-dns was added.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command with modifications was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to configure ephone-dns to support FXO lines that allow phones to have private lines connected directly to the PSTN. To bind the ephone-dn to the FXO port, use the destination pattern configured for the FXO line’s POTS dial peer for the **digit-string** argument.

The **timeout seconds** argument controls the interdigit delay period, during which digits are collected from the user, and the delay before the connection to the FXO port is established. The argument controls the amount of time that Cisco Unified CME waits to collect digits for the dialed number, so that the digits can be included in the redial buffer and the Placed Calls directory of the phone. Digits that are entered after the timeout period are not included in the redial buffer or in the Placed Calls directory on the phone. The timeout parameter does not affect the time used to cut through the connection from the phone’s trunk button to the FXO port. The phone user must either enter the pound (＃) key or wait for this interdigit timeout to complete digit collection.

The phone user also has the option to use the phone’s on-hook dialing feature so that the phone itself performs complete dial-string digit collection before signaling off-hook to the Cisco Unified CME. In this case all digits will be included in the Redial and Placed Calls Directory.
The `monitor-port` keyword enables direct status monitoring of the FXO port on the line button of the IP phone. The line button indicator, either a lamp or an icon depending on the phone, shows the in-use status of the FXO port during the duration of the call.

The `transfer-timeout` argument enables a transferred call to be automatically recalled if the transfer target does not answer after the specified number of seconds. The call is withdrawn from the transfer-to phone and the call resumes ringing on the phone that initiated the transfer.

The `monitor-port` and `transfer-timeout` keywords are not supported on ephone-dns for analog ports on the Cisco VG 224.

For dual-line ephone-dns, the second channel cannot receive incoming calls when the `trunk` command is configured.

### Examples

The following example shows the configuration for two phones that each have a private FXO line button and a shared-line button.

The shared line’s voice ports and dial peers are as follows:

```
Router(config)# voice-port 1/0/1
Router(config-voice-port)# connection plar-opx 1000
Router(config)# dial-peer voice 101 pots
Router(config-dial-peer)# destination-pattern 9
Router(config-dial-peer)# port 1/0/1
```

The private lines’ voice ports and dial peers are as follows:

```
Router(config)# voice-port 1/1/0
Router(config-voice-port)# connection plar-opx 5550111
Router(config)# dial-peer voice 110 pots
Router(config-dial-peer)# destination-pattern 80
Router(config-dial-peer)# port 1/1/0
Router(config)# voice-port 1/1/1
Router(config-voice-port)# connection plar-opx 5550112
Router(config)# dial-peer voice 111 pots
Router(config-dial-peer)# destination-pattern 81
Router(config-dial-peer)# port 1/1/1
```

The following is the configuration for the shared and private ephone-dns:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1000
Router(config-ephone-dn)# name Line1
Router(config-ephone-dn)# no huntstop
Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 5550111
Router(config-ephone-dn)# name Private line
Router(config-ephone-dn)# trunk 80
Router(config)# ephone-dn 3
Router(config-ephone-dn)# number 5550112
Router(config-ephone-dn)# name Private line
Router(config-ephone-dn)# trunk 81
```

The following is the configuration for ephones with button 1 as a shared line and button 2 a private line:

```
Router(config)# ephone 1
Router(config-ephone)# mac-address 1111.1111.1101
Router(config-ephone)# button 1:1 2:2
Router(config)# ephone 2
Router(config-ephone)# mac-address 1111.1111.1102
```
Router(config-ephone)# button 1;1 2;3

The following example shows that transferred calls are recalled after 30 seconds if the destination party does not answer and status monitoring is enabled for FXO port 1/1/1.

Router(config)# ephone-dn 5
Router(config-ephone-dn)# trunk 801 timeout 5 transfer-timeout 30 monitor-port 1/1/1

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>destination-number</td>
<td>Specifies a connection mode for a voice port.</td>
</tr>
</tbody>
</table>
trustpoint (credentials)

To specify the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with the Cisco Unified SRST router certificate, use the `trustpoint` command in credentials configuration mode. To change the specified trustpoint, use the `no` form of this command.

```
trustpoint  trustpoint-name
no  trustpoint
```

### Syntax Description

| trustpoint-name | Name of the trustpoint to be associated with the Cisco Unified CME CTL provider certificate or the Cisco Unified SRST device certificate. |

### Command Default

No default behavior or values.

### Command Modes

Credentials configuration (config-credentials)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.3(14)T</td>
<td>Cisco SRST 3.3</td>
<td>This command was introduced for Cisco SRST.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced for Cisco Unified CME.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command for Cisco Unified CME was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

**Cisco Unified CME**

This command is used with Cisco Unified CME phone authentication to define the trustpoint for the CTL provider. This trustpoint will be used for TLS sessions with the CTL client.

**Cisco Unified SRST**

The name of the trustpoint must be consistent with the trustpoint name of the Cisco Unified SRST router.

### Examples

**Cisco Unified CME**

The following example sets up a CTL provider on the Cisco Unified CME router with the IP address 172.19.245.1.

```
Router(config)# credentials
Router(config-credentials)# ip source-address 172.19.245.1 port 2444
Router(config-credentials)# trustpoint ctlpv
Router(config-credentials)# ctl-service admin user4 secret 0 c89L8o
```
Cisco Unified SRST

The following example enters credentials configuration mode, sets the IP source address and port, and specifies the trustpoint:

Router(config)# credentials
Router(config-credentials)# ip source-address 10.6.21.4 port 2445
Router(config-credentials)# trustpoint srstca

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>ctl-service admin</strong></td>
<td>Specifies a user name and password to authenticate the CTL client during the CTL protocol.</td>
</tr>
<tr>
<td><strong>debug credentials</strong></td>
<td>Sets debugging on the credentials service.</td>
</tr>
<tr>
<td><strong>ip source-address (credentials)</strong></td>
<td>Enables the router to receive messages through the specified IP address and port.</td>
</tr>
<tr>
<td><strong>show credentials</strong></td>
<td>Displays the credentials settings.</td>
</tr>
</tbody>
</table>
trustpoint-label

To specify the PKI trustpoint label to be used for the TLS connection between the CAPF server and the phone, use the trustpoint-label command in CAPF-server configuration mode. To return to the default, use the no form of this command.

trustpoint-label label
no trustpoint-label

**Syntax Description**

| label | Trustpoint name for the CAPF server. |

**Command Default**

No trustpoint label is specified for TLS connections.

**Command Modes**

CAPF-server configuration (config-capf-server)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used with Cisco Unified CME phone authentication to provide a PKI trustpoint name for the CAPF server. This trustpoint label is used for the TLS connection between the CAPF server and the phone.

**Examples**

The following example defines the CAPF server trustpoint name as server25.

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```
To assign a phone type to an SCCP phone, use the `type` command in ephone or ephone-template configuration mode. To remove a phone type, use the `no` form of this command.

```plaintext
type phone-type [addon 1 module-type [2 module-type]]
no type phone-type [addon 1 module-type [2 module-type]]
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>phone-type</code></td>
<td>Type of phone. The following phone types are predefined in the system:</td>
</tr>
<tr>
<td>12SP—12SP+ and 30VIP phones.</td>
<td></td>
</tr>
<tr>
<td>6901—Cisco Unified IP Phone 6901.</td>
<td></td>
</tr>
<tr>
<td>6911—Cisco Unified IP Phone 6911.</td>
<td></td>
</tr>
<tr>
<td>6921—Cisco Unified IP Phone 6921.</td>
<td></td>
</tr>
<tr>
<td>6941—Cisco Unified IP Phone 6941.</td>
<td></td>
</tr>
<tr>
<td>6945—Cisco Unified IP Phone 6945.</td>
<td></td>
</tr>
<tr>
<td>6961—Cisco Unified IP Phone 6961.</td>
<td></td>
</tr>
<tr>
<td>7902—Cisco Unified IP Phone 7902G.</td>
<td></td>
</tr>
<tr>
<td>7905—Cisco Unified IP Phone 7905G.</td>
<td></td>
</tr>
<tr>
<td>7906—Cisco Unified IP Phone 7906.</td>
<td></td>
</tr>
<tr>
<td>7910—Cisco Unified IP Phones 7910 and 7910G.</td>
<td></td>
</tr>
<tr>
<td>7911—Cisco Unified IP Phone 7911G.</td>
<td></td>
</tr>
<tr>
<td>7912—Cisco Unified IP Phone 7912G.</td>
<td></td>
</tr>
<tr>
<td>7920—Cisco Unified IP Phone 7920.</td>
<td></td>
</tr>
<tr>
<td>7921—Cisco Unified Wireless IP Phone 7921.</td>
<td></td>
</tr>
<tr>
<td>7925—Cisco Unified Wireless IP Phone 7925.</td>
<td></td>
</tr>
<tr>
<td>7931—Cisco Unified IP Phone 7931G.</td>
<td></td>
</tr>
<tr>
<td>7935—Cisco Unified IP Conference Station 7935.</td>
<td></td>
</tr>
<tr>
<td>7936—Cisco Unified IP Conference Station 7936.</td>
<td></td>
</tr>
<tr>
<td>7937—Cisco Unified IP Conference Station 7937.</td>
<td></td>
</tr>
<tr>
<td>7940—Cisco Unified IP Phone 7940G.</td>
<td></td>
</tr>
<tr>
<td>7941—Cisco Unified IP Phone 7941G.</td>
<td></td>
</tr>
<tr>
<td>7941GE—Cisco Unified IP Phone 7941G-GE.</td>
<td></td>
</tr>
<tr>
<td>Module Type</td>
<td>Description</td>
</tr>
<tr>
<td>-------------</td>
<td>-------------</td>
</tr>
<tr>
<td>7942</td>
<td>Cisco Unified IP Phone 7942.</td>
</tr>
<tr>
<td>7945</td>
<td>Cisco Unified IP Phone 7945.</td>
</tr>
<tr>
<td>7960</td>
<td>Cisco Unified IP Phone 7960G.</td>
</tr>
<tr>
<td>7961</td>
<td>Cisco Unified IP Phone 7961G.</td>
</tr>
<tr>
<td>7961GE</td>
<td>Cisco Unified IP Phone 7961G-GE.</td>
</tr>
<tr>
<td>7962</td>
<td>Cisco Unified IP Phone 7962.</td>
</tr>
<tr>
<td>7965</td>
<td>Cisco Unified IP Phone 7965.</td>
</tr>
<tr>
<td>7970</td>
<td>Cisco Unified IP Phone 7970G.</td>
</tr>
<tr>
<td>7970G</td>
<td>Cisco Unified IP Phone 7971G-GE.</td>
</tr>
<tr>
<td>7975</td>
<td>Cisco Unified IP Phone 7975.</td>
</tr>
<tr>
<td>7985</td>
<td>Cisco Unified IP Phone 7985.</td>
</tr>
<tr>
<td>8941</td>
<td>Cisco Unified IP Phone 8941.</td>
</tr>
<tr>
<td>8945</td>
<td>Cisco Unified IP Phone 8945.</td>
</tr>
<tr>
<td>8961</td>
<td>Cisco Unified IP Phone 8961.</td>
</tr>
<tr>
<td>9951</td>
<td>Cisco Unified IP Phone 9951.</td>
</tr>
<tr>
<td>9971</td>
<td>Cisco Unified IP Phone 9971.</td>
</tr>
<tr>
<td>anl</td>
<td>Analog.</td>
</tr>
<tr>
<td>ata</td>
<td>Cisco ATA-186 or Cisco ATA-188.</td>
</tr>
<tr>
<td>bri</td>
<td>SCCP Gateway (BR).</td>
</tr>
<tr>
<td>vgp</td>
<td>VG248 phone emulation for analog phone.</td>
</tr>
</tbody>
</table>

### Note
You can also add a new phone type to your configuration by using the `ephone-type` command.

### addon 1
(Required) Tells the router that an expansion module is being added to this Cisco Unified IP Phone and the type of module. Valid entries for `module-type` are:

<table>
<thead>
<tr>
<th>Module Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>7914</td>
<td>Cisco Unified IP Phone 7914 Expansion Module.</td>
</tr>
<tr>
<td>7915-12</td>
<td>Cisco Unified IP Phone 7915 12-Button Expansion Module.</td>
</tr>
<tr>
<td>7915-24</td>
<td>Cisco Unified IP Phone 7915 24-Button Expansion Module.</td>
</tr>
<tr>
<td>7916-12</td>
<td>Cisco Unified IP Phone 7916 12-Button Expansion Module.</td>
</tr>
<tr>
<td>7916-24</td>
<td>Cisco Unified IP Phone 7916 24-Button Expansion Module.</td>
</tr>
</tbody>
</table>

### Note
This keyword is not supported for user-defined phone types created with the `ephone-type` command.

### 2
(Required) Tells the router that a second expansion module is being added to this Cisco Unified IP Phone and the type of module. Valid entries for `module-type` are:

<table>
<thead>
<tr>
<th>Module Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>7914</td>
<td>Cisco Unified IP Phone 7914 Expansion Module.</td>
</tr>
<tr>
<td>7915-12</td>
<td>Cisco Unified IP Phone 7915 12-Button Expansion Module.</td>
</tr>
<tr>
<td>7915-24</td>
<td>Cisco Unified IP Phone 7915 24-Button Expansion Module.</td>
</tr>
<tr>
<td>7916-12</td>
<td>Cisco Unified IP Phone 7916 12-Button Expansion Module.</td>
</tr>
<tr>
<td>7916-24</td>
<td>Cisco Unified IP Phone 7916 24-Button Expansion Module.</td>
</tr>
</tbody>
</table>

### Note
This keyword is not supported for user-defined phone types created with the `ephone-type` command.

**Command Default**
No phone type or add-on expansion module is defined.
## Command Modes

- Ephone configuration (config-ephone)
- Ephone-template configuration (config-ephone-template)

## Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>The following keywords were added to this command: <strong>7902</strong>, <strong>7905</strong>, and <strong>7912</strong>.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.3(7)T</td>
<td>Cisco CME 3.1</td>
<td>The <strong>7920</strong> and <strong>7936</strong> keywords were added.</td>
</tr>
<tr>
<td>12.3(11)XL</td>
<td>Cisco CME 3.2(1)</td>
<td>The <strong>7970</strong> keyword was added.</td>
</tr>
<tr>
<td>12.3(14)T</td>
<td>Cisco CME 3.3</td>
<td>The <strong>7971</strong> keyword was added, and this command was integrated into Cisco IOS Release 12.3(14)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>The <strong>7911</strong>, <strong>7941</strong>, <strong>7941GE</strong>, <strong>7961</strong>, and <strong>7961GE</strong> keywords were added. This command was made available in ephone-template configuration mode.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>12.4(6)XE</td>
<td>Cisco Unified CME 4.0(2)</td>
<td>The <strong>7931</strong> keyword was added.</td>
</tr>
<tr>
<td>12.4(4)XC4</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>The <strong>7931</strong> keyword was added.</td>
</tr>
<tr>
<td>12.4(11)T</td>
<td>Cisco Unified CME 4.0(3)</td>
<td>This command was integrated into Cisco IOS Release 12.4(11)T.</td>
</tr>
<tr>
<td>12.4(11)XJ2</td>
<td>Cisco Unified CME 4.1</td>
<td>The <strong>7921</strong> and <strong>7985</strong> keywords were introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>12.4(15)T1</td>
<td>Cisco Unified CME 4.1(1)</td>
<td>The <strong>7942</strong>, <strong>7945</strong>, <strong>7962</strong>, <strong>7965</strong>, and <strong>7975</strong> keywords were introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>Support for user-defined phone types created with the <strong>ephone-type</strong> command was added.</td>
</tr>
<tr>
<td>12.4(15)XZ1</td>
<td>Cisco Unified CME 4.3</td>
<td>The <strong>7915-12</strong>, <strong>7915-24</strong>, <strong>7916-12</strong>, <strong>7916-24</strong>, and <strong>7937</strong> keywords were added.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>The <strong>7915-12</strong>, <strong>7915-24</strong>, <strong>7916-12</strong>, <strong>7916-24</strong>, and <strong>7937</strong> keywords were added and this command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>
### Cisco Unified CME Commands: T

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(20)T1</td>
<td>Cisco Unified CME 7.0</td>
<td>The 7925 keyword was added.</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was modified. The 6921, 6941, 6961, and IP-STE keywords were added.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was modified. The 6901 and 6911 keywords were added.</td>
</tr>
<tr>
<td>15.2(1)T</td>
<td>Cisco Unified CME 8.8</td>
<td>This command was modified. The 6945, 8941, and 8945 keywords were added.</td>
</tr>
<tr>
<td>15.3(3)M</td>
<td>Cisco Unified CME 10.0</td>
<td>This command was modified. The 7906, 8961, 9951, and 9971 keywords were added.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Not all phone types support add-on expansion modules. For support information, see User Documentation for Cisco Unified IP Phones.

This command must be followed by a phone reboot using the `reset` command.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

### Examples

The following example defines the IP phone with phone-tag 10 as a Cisco Unified IP Phone 7960G with two attached Cisco Unified IP Phone 7914 Expansion Modules:

```
Router(config)# ephone 10
Router(config-ephone)# type 7960 addon 1 7914 2 7914
```

The following example defines the IP phone with phone-tag 4 as a Cisco ATA device:

```
Router(config)# ephone 4
Router(config-ephone)# mac 1234.87655.234
Router(config-ephone)# type ata
```

The following example defines the IP phone with phone-tag 10 as a Cisco Unified IP Phone IP-STE:

```
Router(config)# ephone 10
Router(config-ephone)# type IPSTE
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-type</td>
<td>Adds a Cisco Unified IP phone type by defining a phone-type template.</td>
</tr>
<tr>
<td>reset (ephone)</td>
<td>Performs a complete reboot of one phone associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>reset (telephony-service)</td>
<td>Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.</td>
</tr>
</tbody>
</table>
**type (voice register dialplan)**

To specify a phone type for a SIP dial plan, use the `type` command in voice register dialplan configuration mode. To remove a phone type, use the `no` form of this command.

```
type  phone-type
no  type
```

**Syntax Description**

<table>
<thead>
<tr>
<th><code>phone-type</code></th>
<th>Type of SIP phone for which the dial plan is used. Values are:</th>
</tr>
</thead>
<tbody>
<tr>
<td>• 7905-7912—Cisco Unified IP Phone 7905, 7905G, 7912, or 7912G.</td>
<td></td>
</tr>
<tr>
<td>• 7940-7960-others—Cisco Unified IP Phone 7911, 7940, 7940G, 7941, 7942, 7941GE, 7945, 7960, 7960G, 7961, 7961GE, 7962, 7965, 7970, 7971, or 7975.</td>
<td></td>
</tr>
</tbody>
</table>

**Command Default**

The phone type is not defined.

**Command Modes**

Voice register dialplan configuration (config-register-dialplan)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>Support for Cisco Unified IP Phone 7942, 7945, 7962, 7965, and 7975 was added.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies the type of SIP phone for which the dial plan is defined. You must use this command before defining dial patterns with the `pattern` command or selecting a dial pattern file in flash with the `filename` command.

The phone type specified with this command must match the phone type specified with the `type` command in voice register pool mode. If the dial plan type does not match the type assigned to the phone, the dial-plan configuration file is not generated.

**Examples**

The following example shows a SIP dial plan being defined for a Cisco Unified IP Phone 7905 or Cisco Unified IP Phone 7912:

```
Router(config)# voice register dialplan 10
Router(config-register-dialplan)# type 7905-7912
Router(config-register-dialplan)# pattern 52...
Router(config-register-dialplan)# pattern 91.......
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>dialplan</code></td>
<td>Assigns a dial plan to a SIP phone.</td>
</tr>
<tr>
<td><code>filename</code></td>
<td>Specifies a custom XML file that contains the dial patterns to use for the SIP dial plan.</td>
</tr>
<tr>
<td><code>pattern (voice register dialplan)</code></td>
<td>Defines a dial pattern for a SIP dial plan.</td>
</tr>
<tr>
<td><code>show voice register dialplan</code></td>
<td>Displays configuration information for a specific SIP dial plan.</td>
</tr>
<tr>
<td><code>type (voice register pool)</code></td>
<td>Defines a phone type for a SIP phone.</td>
</tr>
</tbody>
</table>
**type (voice register pool)**

To define a phone type for a SIP phone, use the `type` command in voice register pool configuration mode. To remove a phone type, use the `no` form of this command.

```
type phone-type [addon 1 CKEM | CP-8800-Audio | CP-8800-Video] [addon 2 CKEM | CP-8800-Audio | CP-8800-Video] [addon 3 CKEM | CP-8800-Audio | CP-8800-Video]
no type
```
<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>phone-type</th>
</tr>
</thead>
</table>

**Cisco Unified CME Commands: T**

type (voice register pool)
Type of SIP phone that is being defined. Valid entries are as follows:

- 3905 — Cisco Unified IP Phone 3905.
- 3951 — Cisco Unified IP Phones 3911 and 3951.
- 6901 — Cisco Unified IP Phone 6901.
- 6911 — Cisco Unified IP Phone 6911.
- 6921 — Cisco Unified IP Phone 6921.
- 6922 — Cisco Unified IP Phone 6922.
- 6941 — Cisco Unified IP Phone 6941.
- 6945 — Cisco Unified IP Phone 6945.
- 6961 — Cisco Unified IP Phone 6961.
- 7821 — Cisco Unified IP Phones 7821.
- 7841 — Cisco Unified IP Phones 7841.
- 7861 — Cisco Unified IP Phones 7861.
- 7905 — Cisco Unified IP Phones 7905 and 7905G.
- 7906 — Cisco Unified IP Phone 7906G.
- 7911 — Cisco Unified IP Phone 7911G.
- 7912 — Cisco IP Phones 7912 and 7912G.
- 7940 — Cisco IP Phones 7940 and 7940G.
- 7941 — Cisco IP Phone 7941G.
- 7941GE — Cisco IP Phone 7941GE.
- 7942 — Cisco Unified IP Phone 7942.
- 7945 — Cisco Unified IP Phone 7945.
- 7960 — Cisco IP Phones 7960 and 7960G.
- 7961 — Cisco IP Phone 7961G.
- 7961GE — Cisco IP Phone 7961GE.
- 7962 — Cisco Unified IP Phone 7962.
- 7965 — Cisco Unified IP Phone 7965.
- 7970 — Cisco IP Phone 7970G.
- 7971 — Cisco IP Phone 7971GE.
- 7975 — Cisco Unified IP Phone 7975.
- 8851 — Cisco Unified IP Phone 8851.
- 8851NR — Cisco Unified IP Phone 8851NR.
- 8861 — Cisco Unified IP Phone 8861.
- 8865 — Cisco IP Phone 8865.
- 8961 — Cisco Unified IP Phone 8961.
- 9900 — Cisco Unified IP Phone 9900.
- 9951 — Cisco Unified IP Phone 9951.
- 9971 — Cisco Unified IP Phone 9971.
- ATA — Cisco ATA-186 or Cisco ATA-188.
- ATA-190 — Cisco ATA-190.
- ATA-191 — Cisco ATA-191.
- DX650 — Cisco DX650.
- P100 — PingTel Xpressa 100.
- **P600**—Polycom SoundPoint 600.
- **Jabber-CSF-Client**—Cisco Jabber CSF Client.

### addon 1 CKEM
(Optional) Tells the router that a Cisco SIP IP Phone CKEM 36-Button Line Expansion Module is being added to this Cisco Unified SIP IP phone.

**Note** This option is available to Cisco Unified 8961, 9951, and 9971 SIP IP phones only.

### 2 CKEM
(Optional) Tells the router that a second Cisco SIP IP Phone CKEM 36-Button Line Expansion Module is being added to this Cisco Unified SIP IP phone.

**Note** This option is available to Cisco Unified 9951 and 9971 SIP IP phones only.

### 3 CKEM
(Optional) Tells the router that a third Cisco SIP IP Phone CKEM 36-Button Line Expansion Module is being added to this Cisco Unified SIP IP phone.

**Note** This option is available to Cisco Unified 9971 SIP IP phones only.

### addon 1 CP-8800-Audio or addon 1 CP-8800-Video
(Optional) Tells the router that a Cisco SIP IP Phone 28-Button Line A-KEM or V-KEM is being added to this Cisco Unified SIP IP Phone.

### 2 CP-8800-Audio or 2 CP-8800-Video
(Optional) Tells the router that a second Cisco SIP IP Phone A-KEM or V-KEM is being added to this Cisco Unified SIP IP Phone.

### 2 CP-8800-Audio or 2 CP-8800-Video
(Optional) Tells the router that a second Cisco SIP IP Phone A-KEM or V-KEM is being added to this Cisco Unified SIP IP Phone.

---

**Command Default**
No phone type is defined.

**Command Modes**
Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was modified to add the 3951, 7911, 7941, 7941GE, 7961, 7961GE, 7970, and 7971 keywords.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>The 3951, 7911, 7941, 7941GE, 7961, 7961GE, 7970, and 7971 keywords were integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was modified to add the 7942, 7945, 7962, 7965, and 7975 keywords.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME8.5</td>
<td>This command was modified to add the 8961, 9951, and 9971 keywords.</td>
</tr>
<tr>
<td>Cisco IOS Release</td>
<td>Cisco Product</td>
<td>Modification</td>
</tr>
<tr>
<td>------------------</td>
<td>--------------</td>
<td>--------------</td>
</tr>
<tr>
<td>15.2(1)T</td>
<td>Cisco Unified CME 8.8</td>
<td>This command was modified to add the 3905 keyword.</td>
</tr>
<tr>
<td>15.2(2)T</td>
<td>Cisco Unified CME 9.0</td>
<td>This command was modified to add the 6901, 6911, 6921, 6941, 6945, 6961, ATA-187, and Jabber-Android keywords.</td>
</tr>
<tr>
<td>15.2(4)M</td>
<td>Cisco Unified CME 9.1</td>
<td>This command was modified to include the addon 1 CKEM, 2 CKEM, and 3 CKEM keywords.</td>
</tr>
<tr>
<td>15.3(3)M</td>
<td>Cisco Unified CME 10.0</td>
<td>This command was modified to add the 6922 and 9900 keywords.</td>
</tr>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was modified. The 78XX, DX650 and Jabber-CSF-Client keywords were added.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.10.1a Release</td>
<td>Unified CME 12.5</td>
<td>This command was modified. The ATA-191, CP-8800-Audio, and CP-8800-Video keywords were added.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The **addon 1 CKEM, 2 CKEM, and 3 CKEM** keywords increase the number of speed-dial, busy-lamp-field, and directory number keys that can be configured.

There are two options in removing a Key Expansion Module (KEM) when you have configured all three KEMs.

The first option is to use the **no** form of the **type** command, then use the **type** command to configure only the KEMs to be included. The following example shows how the second and third KEMs are removed from the configuration:

```
Router(config)# voice register pool 9
Router(config-register-pool)# type 9971 addon 1 CKEM 2 CKEM 3 CKEM
Router(config-register-pool)# no type 9971 addon 1 CKEM 2 CKEM 3 CKEM
Router(config-register-pool)# type 9971 addon 1 CKEM
```

The second option is to define the same phone type while excluding from the configuration the KEM to be removed. For example, you have configured the following:

```
Router(config)# voice register pool 3
Router(config-register-pool)# type 9971 addon 1 CKEM 2 CKEM 3 CKEM
```

To remove the third KEM, enter the following:

```
Router(config-register-pool)# type 9971 addon 1 CKEM 2 CKEM
```

To remove the second KEM, enter the following:

```
Router(config-register-pool)# type 9971 addon 1 CKEM
```

From Unified CME 12.5 Release, **type phone-type [addon 1 CKEM | CP-8800-Audio | CP-8800-Video [2 CKEM | CP-8800-Audio | CP-8800-Video [3 CKEM | CP-8800-Audio | CP-8800-Video ]] configuration**
is supported. The phone support is extended to Cisco IP Phone 8865 to support the Video KEM CP-8800-Video. Audio KEM CP-8800-Audio support is introduced for the Cisco IP Phone models 8851, 8851 NR, and 8861.

---

**Note**

All the configuration characteristics of CKEM discussed here is applicable to CP-8800-Audio and CP-8800-Video.

After configuring the phone type, use the `create profile` command in voice register global configuration mode to generate the configuration profile files required for the phone and then reset or restart the phone using the `reset` or `restart` command, respectively.

---

**Note**

Cisco Unified CME enables the `busy trigger-per-button (voice register pool)` command when phone-type 3905 is specified.

---

**Examples**

The following example shows how to define a SIP phone with phone-tag 10 as a Cisco Unified IP Phone 7960 or Cisco Unified IP Phone 7960G:

```plaintext
Router(config)# voice register pool 10
Router(config-register-pool)# type 7960
```

The following is a sample configuration for CP-8800-Audio and CP-8800-Video on the supported phone models for Unified CME 12.5 Release.

```plaintext
Router(config-register-pool)# type 8851 addon 1 CP-8800-Audio 2 CP-8800-Audio
Router(config-register-pool)# type 8851NR addon 1 CP-8800-Audio 2 CP-8800-Audio
Router(config-register-pool)# type 8861 addon 1 CP-8800-Audio 2 CP-8800-Audio 3 CP-8800-Audio
Router(config-register-pool)# type 8865 addon 1 CP-8800-Video 2 CP-8800-Video 3 CP-8800-Video
```

---

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>busy-trigger-per-button (voice register pool)</code></td>
<td>Sets the maximum number of calls allowed on a SIP directory number before activating Call Forward Busy or a busy tone.</td>
</tr>
<tr>
<td><code>load (voice register global)</code></td>
<td>Associates a type of Cisco Unified SIP IP phone with a phone firmware file.</td>
</tr>
<tr>
<td><code>reset (voice register global)</code></td>
<td>Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td><code>reset (voice register pool)</code></td>
<td>Performs a complete reboot of one SIP phone associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td><code>restart (voice register)</code></td>
<td>Performs a fast reset of one or all SIP phones associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td><code>voice register pool</code></td>
<td>Enters voice register pool configuration mode for SIP phones.</td>
</tr>
</tbody>
</table>
**type (voice-gateway)**

To define the type of voice gateway to autoconfigure, use the `type` command in voice-gateway configuration mode. To remove the type from the configuration, use the `no` form of this command.

```
type {vg202|vg204|vg224}
no type
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>vg202</td>
<td>Cisco VG202 Voice Gateway with 2 FXS ports.</td>
</tr>
<tr>
<td>vg204</td>
<td>Cisco VG204 Voice Gateway with 4 FXS ports.</td>
</tr>
<tr>
<td>vg224</td>
<td>Cisco VG224 Voice Gateway with 24 FXS ports.</td>
</tr>
</tbody>
</table>

**Command Default**

No type is defined for the voice gateway to be autoconfigured.

**Command Modes**

Voice-gateway configuration (config-voice-gateway)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies the type of Cisco voice gateway for which you are creating an XML configuration file.

**Examples**

The following example shows a configuration for the Cisco VG224 voice gateway:

```
voice-gateway system 1
network-locale FR
type VG224
mac-address 001F.A30F.8331
voice-port 0-23
create cnf-files
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>create cnf-files (voice-gateway)</td>
<td>Generates the XML configuration files that are required to autoconfigure the Cisco voice gateway.</td>
</tr>
<tr>
<td>mac-address (voice-gateway)</td>
<td>Defines the MAC address of the voice gateway to autoconfigure.</td>
</tr>
<tr>
<td>voice-port (voice-gateway)</td>
<td>Identifies the ports on the voice gateway that register to Cisco Unified CME.</td>
</tr>
</tbody>
</table>
Cisco Unified CME Commands: U

- upa, on page 1356
- upgrade (voice register global), on page 1357
- url (telephony-service), on page 1359
- url (voice register global), on page 1362
- url (voice register template), on page 1364
- url authentication, on page 1366
- url idle, on page 1368
- url services (ephone-template), on page 1369
- url-button, on page 1371
- url-button (voice-register-template), on page 1373
- user (voice logout-profile), on page 1374
- user (voice user-profile), on page 1376
- user-locale (ephone-template), on page 1378
- user-locale (telephony-service), on page 1380
- user-locale (voice register), on page 1386
- username (ephone), on page 1389
- username (voice register pool), on page 1391
- utf8, on page 1393
To specify the audio file used for the unauthorized precedence announcement, use the `upa` command in voice MLPP configuration mode. To disable use of this audio file, use the `no` form of this command.

```
upa  audio-url
no  upa
```

### Syntax Description

| `audio-url` | Location of the announcement audio file in URL format. Valid storage locations are TFTP, FTP, HTTP, and flash memory. |

### Command Default

No announcement is played.

### Command Modes

Voice MLPP configuration (config-voice-mlpp)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command specifies the G.711 a-law or u-law 8-KHz encoded audio file (.wav or .au format) for the announcement that plays to callers when they attempt to make a precedence call by using a higher level of precedence than the highest precedence level that is authorized for their line.

The `mlpp indication` command must be enabled (default) for a phone to play precedence announcements.

This command is not supported by Cisco IOS help. If you type `?`, Cisco IOS help does not display a list of valid entries.

### Examples

The following example shows the unauthorized precedence announcement plays the file named `upa.au` located in flash:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# upa flash:upa.au
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>bnea</code></td>
<td>Specifies the audio file used for the busy station not equipped for preemption announcement.</td>
</tr>
<tr>
<td><code>ica</code></td>
<td>Specifies the audio file used for the isolated code announcement.</td>
</tr>
<tr>
<td><code>vca</code></td>
<td>Specifies the audio file used for the vacant code announcement.</td>
</tr>
<tr>
<td><code>mlpp indication</code></td>
<td>Enables MLPP indication on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td><code>mlpp preemption</code></td>
<td>Enables preemption capability on an SCCP phone or analog FXS port.</td>
</tr>
</tbody>
</table>
upgrade (voice register global)

To generate a OS79XX.TXT file for firmware upgrades, use the upgrade command in voice register global configuration mode. To return to the default, use the no form of this command.

```
upgrade
no upgrade
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
No OS79XX.TXT file generated.

**Command Modes**
Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
The upgrade command performs the TFTP server alias binding, which can be verified with the `show voice register tftp-bind` command.

**Examples**
The following example shows the use of the `upgrade` command to upgrade Cisco SIP phone firmware from SIP [456].x to SIP [567].y with comments:

```
Router(config)# voice register global
Router(config-register-global)# load 7960-7940 P00x...
!Do not use file extension.
Router(config-register-global)# upgrade
!Generates OS79XX.txt file.
Router(config-register-global)# load 7960-7940 POSx...
!Do not use file extension. This is only required if you are upgrading to 7.y.

Router(config-register-global)# create profile
!Generates SIPDefault.cnf and other files.
Router(config-register-global)# reset
Router(config-register-global)# no upgrade
!Returns to default condition.
```

The P00x... and P0Sx... firmware filenames are required because the content in OS79XX.TXT is different from image_version tag in SIPDefault.cnf.

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Generates configuration profile files required for SIP IP phones in Cisco Unified CME.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Associates a type of IP phone with a phone firmware file.</td>
</tr>
<tr>
<td>Command</td>
</tr>
<tr>
<td>-------------------------------</td>
</tr>
<tr>
<td>mode cme</td>
</tr>
<tr>
<td>reset (voice register pool)</td>
</tr>
<tr>
<td>show voice register tftp-bind</td>
</tr>
</tbody>
</table>
url (telephony-service)

To provision uniform resource locators (URLs) for Cisco Unified IP phones connected to the Cisco Unified CME router, use the `url` command in telephony-service or group configuration mode. To remove a URL association, use the `no` form of this command.

```
url {authentication|directories|idle|information|messages|proxy-server|services} url [(line|root)]
no url {authentication|directories|idle|information|messages|proxy-server|services}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>authentication</code></td>
<td>Uses the information at the specified URL to validate requests made to the phone web server.</td>
</tr>
<tr>
<td><code>directories</code></td>
<td>Uses the information at the specified URL for the Directories button display.</td>
</tr>
<tr>
<td><code>idle</code></td>
<td>Information at the specified URL displays on the window of the IP phone during the idle state.</td>
</tr>
<tr>
<td><code>information</code></td>
<td>Uses the information at the specified URL for the Information button display. This button can be labeled “i” or “?”.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>Cisco Unified CME does not support the use of this URL.</td>
</tr>
<tr>
<td><code>messages</code></td>
<td>Uses the information at the specified URL for the Messages button display.</td>
</tr>
<tr>
<td><code>proxy-server</code></td>
<td>Specifies the host and port used to enable proxy HTTP requests for access to remote host addresses from the phone HTTP client.</td>
</tr>
<tr>
<td><code>services</code></td>
<td>Uses the information at the specified URL for the Services button display.</td>
</tr>
<tr>
<td><code>url</code></td>
<td>URL as defined in RFC 2396.</td>
</tr>
<tr>
<td><code>line</code></td>
<td>(optional) Supported only with <code>services</code> keyword. Alphanumeric string of 1 to 32 characters that is line-services name to be displayed under Services button.</td>
</tr>
<tr>
<td><code>root</code></td>
<td>(optional) Supported only with <code>services</code> keyword and supported in telephony-service mode only. Menu of root phone services supported by a CSTA client application is displayed under Services button.</td>
</tr>
</tbody>
</table>

**Command Default**

The router automatically uses the local directory service.

**Command Modes**

Telephony-service configuration (config-telephony)
Group configuration (conf-tele-group)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>12.4(22)T</td>
<td>Cisco Unified CME 7.0(1)</td>
<td>This command was added to VRF group configuration mode.</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager Express Command Reference

**Modification**

Cisco IOS Release | Cisco Product | Modification
--- | --- | ---
15.0(1)XA | Cisco Unified CME 8.0 | This command was modified. Support for the root keyword was added to this command.
15.1(1)T | Cisco Unified CME 8.0 | This command was integrated into Cisco IOS Release 15.1(1)T.

**Usage Guidelines**

Cisco Unified IP Phones can support URLs in association with the four programmable feature buttons on those IP phones: Directories, Information, Messages, and Services. The fifth button, Settings, is managed entirely by the phone. Operation of these services is determined by the Cisco Unified IP phone capabilities and the content of the referenced URL.

This command provisions URLs through the configuration file supplied by the Cisco Unified CME router to the Cisco Unified IP phones during phone registration.

**Note**

Cisco Unified CME does not support provisioning an information URL to access help using the i or ? buttons on a phone.

To use a Cisco Unified CallManager directory as an external directory source for Cisco Unified CME phones, the Cisco Unified CallManager must be made aware of the phones. You must list the MAC addresses of the Cisco Unified CME phones in the Cisco Unified CallManager and reset the phones from the Cisco Unified CallManager. It is not necessary for you to assign ephone-dns to the phones or for the phones to register with Cisco Unified CallManager.

**Note**

Provisioning of the directory URL to select an external directory resource disables the Cisco Unified CME local directory service.

You can disable the local directory by using the no service local-directory command.

This command must be followed by a complete phone reboot using the reset command.

**Examples**

The following example provisions the Directories and Services buttons. Note that the Messages button is configured by the voicemail command. The Messages button acts like a speed-dial key to retrieve messages from a specified telephone number.

Router(config)# telephony-service
Router(config-telephony)# url directories http://1.4.212.11/localdirectory
Router(config-telephony)# url services http://1.4.212.4/CCM/User/123456/urltest.html

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>group</td>
<td>Creates a virtual router forwarding (VRF) group for Cisco Unified CME users and phones.</td>
</tr>
<tr>
<td>reset (ephone)</td>
<td>Performs a complete reboot of one phone associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>------------------------</td>
<td>-----------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>reset (telephony-service)</td>
<td>Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td>service local-directory</td>
<td>Enables the availability of the local directory service on IP phones.</td>
</tr>
<tr>
<td>voicemail</td>
<td>Defines the telephone number that is speed-dialed when the Messages button on a Cisco Unified IP phone is pressed.</td>
</tr>
</tbody>
</table>
url (voice register global)

To provision uniform resource locators (URLs) for feature buttons on Cisco SIP IP phones connected to a Cisco Unified CME router, use the `url` command in voice register global configuration mode. To remove a URL association, use the `no` form of this command.

```
url {authentication|directory|service|idle} url
no url {authentication|directory|service|idle}
```

### Syntax Description
- **authentication**
  - Uses the information at the specified URL to validate requests made to the phone web server.
- **directory**
  - Uses the information at the specified URL for the Directories button display.
- **service**
  - Uses the information at the specified URL for the Services button display.
- **idle**
  - Uses the information at the specified URL to display on the IP phone during the idle state.
- **url**
  - URL as defined in RFC 2396.

### Command Default
The router automatically uses the local directory service.

### Command Modes
Voice register global configuration (config-register-global)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IOS XE Everest 16.6.1</td>
<td>Cisco Unified CME 12.0</td>
<td>The <code>idle</code> keyword was added.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines
The Cisco Unified IP Phones 7940 and 7940G and Cisco Unified IP Phones 7960 and 7960G can support two URLs in association with two programmable feature buttons: Directories and Services. Operation of these services is determined by the Cisco IP phone capabilities and the content of the specified URL. The `Settings button` is managed entirely by the phone. The Messages button is configured by the `voicemail` command.

The purpose of the `url` command is to provision the URLs through the configuration file supplied by the Cisco Unified CME router to the SIP phones during phone registration.

You can disable the local directory by specifying the string “none” instead of a URL with the `directory` keyword, as shown in the following example:

```
Router(config)# voice register global
Router(config-register-global)# url directory none
```

### Note
Provisioning the directory URL to select an external directory resource disables Cisco Unified CME local directory service.

After you configure this command, restart the phone by using the `reset` command.
Examples

The following example shows how to provision the Directories and Services buttons:

Router(config)# voice register global
Router(config-register-global)# url directory http://1.4.212.11/localdirectory
Router(config-register-global)# url service http://1.4.212.4/CCMUser/123456/urltest.html

The following example shows that the information at the specified URL is used to validate requests made to the phone web server.

Router(config)# voice register global
Router(config-register-global)# url authentication http://CME IP Address/CCMIP/authenticate.asp

The following example specifies that the file logo.xml should be displayed on IP phones when they are not being used and that the display should be refreshed every 12 seconds:

Router(config)# voice register global
Router(config-register-global)# url idle http://mycompany.com/files/logo.xml idle-timeout 12

Related Commands

<table>
<thead>
<tr>
<th>Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>reset (voice register pool)</td>
<td>Performs a complete reboot of one phone associated with a Cisco CME router.</td>
</tr>
<tr>
<td>reset (voice register global)</td>
<td>Performs a complete reboot of all SIP phones associated with a Cisco CME router.</td>
</tr>
<tr>
<td>telephony-service</td>
<td>Enters telephony-service configuration mode.</td>
</tr>
<tr>
<td>voicemail (voice register template)</td>
<td>Defines the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed.</td>
</tr>
</tbody>
</table>
url (voice register template)

To define SIP phone URLs to configure dial rules such as Application Dial Rule, Directory Lookup Dial Rule, and LDAP server, use `url AppDialRule`, `url DirLookupRule`, and `url ldapServer` commands in voice register template configuration mode. To specify a file to display on an IP phone that is not in use, use the `url idle` command in voice register template configuration mode. To define a URL for invoking phone services, use the `url service` command in voice register template configuration mode.

```
url {AppDialRule|DirLookupRule|ldapServer|idle|service} {string url}
```

**Syntax Description**

- `url AppDialRule string` Application dial rule URL.
- `url DirLookupRule string` Directory lookup rule URL.
- `url ldapServer string` LDAP server URL.
- `url idle url` Defines the location of a file to display on phones that are not in use and specifies the interval between refreshes of the display, in seconds.
- `url service url` Uses the information at the specified URL for invoking phone services.

**Command Default**

No file is specified.

**Command Modes**

Voice register template configuration (config-register-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IOS XE Everest 16.6.1</td>
<td>Unified CME 12.0</td>
<td>This command was modified to add <code>idle</code> and <code>service</code> keywords.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Cisco softphone SIP client uses the dial rules to integrate with the Lightweight Directory Access Protocol (LDAP) directory server. The Cisco softphone SIP client also uses dial rules such as application dial rule and directory lookup rule to translate the outgoing phone numbers and display the incoming phone numbers with a rich caller ID. A rich caller ID displays a caller’s name, caller’s picture, or caller’s phone number, or the information saved in the phone’s directory.

You can create the application dial rule or directory lookup rule xml files and add these files to a tftp server. The Cisco softphone SIP client can download the dial rules using the `url ldapServer string`, `url AppDialRule string`, and `url DirLookupRule string` commands.

You can define the location of a file to display on phones that are not in use, and specify the interval between refreshes of the display using `url idle` command. You can also define a URL for invoking phone services using the `url service` command.

The following example shows how to define SIP phone URLs to configure Application Dial Rule, Directory Lookup Dial Rule, LDAP server, idle url, and service url in voice register template configuration mode.

```
Router(config-register-temp)# url ldapServer ldap.abcd.com
```
Router(config-register-temp)# url AppDialRule tftp://10.1.1.1/AppDialRules.xml
Router(config-register-temp)# url DirLookupRule tftp://10.1.1.1/DirLookupRules.xml
Router(config-register-temp)# url idle http://www.mycompany.com/files/logo.xml idle-timeout 12
Router(config-register-temp)# url service http://10.0.0.4/CCMUser/123456/urltest.html

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>voice register pool</td>
<td>Enters voice register pool configuration mode.</td>
</tr>
<tr>
<td></td>
<td>voice register template</td>
<td>Enters voice register template configuration mode.</td>
</tr>
</tbody>
</table>
To instruct IP phones in Cisco Unified CME to send requests for authorization to a particular authentication server and include the specified credential in the requests, use the `url authentication` command in telephony-service configuration mode. To return to default, use the `no` form of this command.

```
url authentication url-address application-name password [0|6] password
no url authentication url-address application-name password [0|6] password
```

**Syntax Description**

- **url-address**: URL address of authentication server.

  The URL address for the authentication server in Cisco Unified CME is: `http://CME IP Address/CCMCIP/authenticate.asp`.

- **application-name**: Character string sent by application to identify itself to the server. Length of string: 1 to 15 characters.

  For applications other than Extension Mobility, the name portion of the credential must first be created in the application.

- **password [0|6]**: Character string sent by application to identify itself to the server. Length of string: 1 to 15 characters.

  The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

  For applications other than Extension Mobility, the password portion of the credential must first be created in the application.

**Command Default**

No authentication server or credential is specified for Cisco Unified CME to use for requesting authorization of commands from an application to a phone.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command was enhanced for password encryption, based on Unified CME password policy.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies to which authentication server an IP phone in Cisco Unified CME must send requests for authorization and what credential to send in the request.

For Extension Mobility, use this command to instruct Extension Mobility phones to send an HTTP GET/POST to request authorization from the Cisco Unified CME authentication server before clearing call history when a user logs out.
For Extension Mobility, no additional commands are required to create or save the credential. The credential for the EM manager in Cisco Unified CME is whatever values you specify by using this command.

For applications other than Extension Mobility, the requisite credential must be created in the application.

To use the authentication server in Cisco Unified CME 4.3 and later versions to authorize requests for applications other than Extension Mobility, you must also configure the `authentication credential` command in telephony-service configuration mode.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters [0|6]. This in accordance with Unified CME Password Policy. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

### Examples

The following example shows how to configure this command to instruct Extension Mobility phones in Cisco Unified CME to request authorization from the internal authentication server. The phones include the specified credential (extmob psswrd) in the requests.

```
Router(config)# telephony-service

Router(config-telephony)# url authentication http://192.0.2.0/CCMCIP/authenticate.asp extmob psswrd

Router(config-telephony)# exit

Router(config)#
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>authentication credential</code></td>
<td>Stores credentials in the database for the Cisco Unified CME authentication server.</td>
</tr>
<tr>
<td><code>keep call-history</code></td>
<td>Disables Automatic Clear Call History for Extension Mobility in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
url idle

To specify a file to display on an IP phone that is not in use, use the `url idle` command in telephony-service configuration mode. To disable display of the file, use the `no` form of this command.

```
url idle  url idle-timeout  seconds
no url idle
```

**Syntax Description**

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><code>url</code></td>
<td>URL as defined in RFC 2396.</td>
</tr>
<tr>
<td><code>idle-timeout seconds</code></td>
<td>Time interval between display refreshes, in seconds. Range is from 0 to 300.</td>
</tr>
</tbody>
</table>

**Command Default**

No file is specified for display on idle phones.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The file that is displayed must be encoded in eXtensible Markup Language (XML) using the Cisco XML document type definition (DTD). For more information about Cisco DTD formats, refer to Cisco IP Phone Services Application Development Notes.

This command must be followed by a complete phone reboot using the `reset` command.

**Examples**

The following example specifies that the file logo.xml should be displayed on IP phones when they are not being used and that the display should be refreshed every 12 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# url idle http://mycompany.com/files/logo.xml idle-timeout 12
```

**Related Commands**

<table>
<thead>
<tr>
<th></th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>reset (ephone)</code></td>
<td>Performs a complete reboot of one phone associated with a Cisco CME router.</td>
</tr>
<tr>
<td><code>reset (telephony-service)</code></td>
<td>Performs a complete reboot of one or all phones associated with a Cisco CME router.</td>
</tr>
</tbody>
</table>
url services (ephone-template)

To provision up to eight uniform resource locators (URLs) for the Services feature button on individual SCCP phones connected to Cisco Unified CME, use the `url services` command in ephone-template configuration mode. To reset to the default, use the `no` form of this command.

```
url services url-tag url url-name
no url services url-tag
```

**Syntax Description**

- **url-tag**: Identifier for url being configured for Services feature button. Range is 1 to 8.
- **url**: URL as defined in RFC 2396.
- **url-name**: Alpha-numerical string to appear for this URL in Services feature button display. Length of string is 1 to 256 contiguous characters (a-z, 0-9).

**Command Default**
The system-level configuration for the Services button is used.

**Command Modes**
Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XW</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command displays the information at up to eight URLs for the Services feature button display on a supported Cisco Unified IP phone. Operation of these services is determined by the capabilities of the Cisco Unified IP phone and the content of the specified URL.

If you use an ephone template to apply this command to one or SCCP phones and you also use the `url` command in telephony-service configuration mode to configure a services url for all SCCP phones, the value set in telephony-service configuration mode appears first in the list of options displayed when the phone user presses the Services feature button, before any URLs configured by using this command. Cisco Unified CME self-hosted services, such as Extension Mobility, always appears last in the list of options displayed for the Services feature button.

The number of `url-name` characters that appear on the IP phone display is not fixed because IP phones typically use a proportional font.

After creating an ephone template that contains a services URL, use the `ephone-template (ephone)` command to apply the template to individual phones.

**Examples**

The following example defines three urls for the Services feature button display, one for all SCCP phones and two others in an ephone-template that is applied to individual phones. Phones to which
the template is applied (ephones 11 and 13) will have a second and third option in the Services feature button display.

telephony-service
  url services http://10.0.0.4/CMEUser/123456/urlsupport.html
  .
  .
  create cnf-files
  .
  .
ephone-template 1
  url services 1 http://10.0.0.4/CMEUser/123456/cal.html Calendar
  url services 2 http://10.0.0.4/CMEUser/123456/quotes.html StockQuotes
ephone 11
  mac-address F00D.EDAB.1234
  type 7960
  button 1:25
ephone-template 1
ephone 12
  mac-address 0003.B0D5.6541
  type 7960
  button 1:26
  logout-profile 1
ephone 13
  mac-address 000D.3666.3D00
  type 7960
  ephone-template 1
  logout-profile 1

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ephone-template (phone)</td>
<td>Applies an ephone template to an SCCP phone.</td>
</tr>
<tr>
<td>url (telephony-service)</td>
<td>Provisions URLs for programmable feature buttons on supported Cisco Unified IP phones.</td>
</tr>
</tbody>
</table>
url-button

To configure service url feature on a line key, use the `url-button` command in ephone-template mode. To unconfigure the service url feature on a line key, use the `no` form of this command.

```
url-button index { type url [name] }
no url-button index type url [name]
```

Syntax Description

| `index` | Unique index number. The range is from 1 to 8. |
| `type` | Type of service URL button. The following types of URL service buttons are available: |
| | • myphoneapp: My phone application configured under phone user interface. |
| | • em: Extension Mobility |
| | • snr: Single Number Reach |
| | • voicehuntgroups: Displays a list of voice hunt groups |
| | • park-list: Displays a list of parked calls |
| `url-button` | Service url with maximum length of 31 characters. |

Command Default

By default, URL-button configuration on a line key is not configured.

Command Modes

ephone template (config-ephone-template)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was modified to add voice hunt groups and park-list as new types.</td>
</tr>
</tbody>
</table>

Usage Guidelines

You can configure url-button feature on a line key to function as an extension mobility (EM), My Phone Apps, or single number reach (SNR). You can also configure the url-button feature on a line button to function as a service URL by configuring a URL name of a maximum length of 31 characters.

Examples

The following examples shows three URL buttons configured on a line key:

```
! telephony-service
  max-phones 25
  max-conferences 12 gain -6
  transfer-system full-consult

! ephone-template 5
  url-button 1 em
  url-button 2 mphoneapp
  url-button 3 snr
  url-button 4 voicehuntgroups
  url-button 5 park-list
!```
ephone-template 6
conference drop-mode never
conference add-mode all
conference admin: No
max-calls-per-button 8
busy-trigger-per-button 0
privacy default
url-button 1 em
url-button 2 www.cisco.com www.cisco.com
url-button 3 snr
url-button 4 help help
url-button 7 myphoneapp
!
!

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>show telephony-service ephone-template</strong></td>
<td>Displays the contents of all the ephone templates defined.</td>
</tr>
</tbody>
</table>
url-button (voice-register-template)

To configure service url feature button on a line key, use the url-button command in voice register template mode. To disable the service url feature button configuration on a line key, use the no form of this command.

url-button [index number] [{url location|url name}]
no url-button [index number] [{url location|url name}]

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>index number</td>
<td>Unique index number. Range: 1 to 8.</td>
</tr>
<tr>
<td>url location</td>
<td>Location of the url.</td>
</tr>
<tr>
<td>url name</td>
<td>Service url with maximum length of 31 characters.</td>
</tr>
</tbody>
</table>

**Command Default**

URL-button configuration on a line key is disabled.

**Command Modes**

Voice register template configuration (config-register-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to configure a url-button on a phone’s line key. You can configure a line key to function as a url-button. You can also configure a line button to function as a service url by configuring a url name of a maximum length of 31 characters.

**Examples**

The following example shows url-button configured in voice register template 1:

```
Router# show run
!
!
voice register template 1
  url-button 1 http://www.cisco.com cisco
  button-layout 1 line
  button-layout 2,5 speed-dial
!
voice register pool 50
  id mac 001E.7AC4.DC73
  feature-button 1 NewCall
  type 7965
  number 1 dn 65
  template 1
dtmf-relay rtp-nte
  speed-dial 1 2001 label "SD1-2001"
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice register pool</td>
<td>Displays all configuration information associated with a particular voice register pool.</td>
</tr>
<tr>
<td>show voice register template</td>
<td>Displays all configuration information associated with a SIP phone template,</td>
</tr>
</tbody>
</table>
user (voice logout-profile)

To create an authentication credential for use by Telephone Application Programming Interface (TAPI) phone devices and certain other applications to log into Cisco Unified CME, use the `username` command in voice logout-profile configuration mode. To remove the credential, use the `no` form of this command.

```
user username password [0|6] password
no user name password [0|6] password
```

**Syntax Description**

<table>
<thead>
<tr>
<th>name</th>
<th>Unique alphanumeric string to be used by a TAPI phone device to log into Cisco Unified CME. String can contain a maximum of 15 alphanumeric characters.</th>
</tr>
</thead>
<tbody>
<tr>
<td>password</td>
<td>Password to be used with this username for authentication purposes.</td>
</tr>
<tr>
<td>password [0</td>
<td>6]</td>
</tr>
<tr>
<td></td>
<td>The 0 in the parameter [0</td>
</tr>
</tbody>
</table>

**Command Default**

No authentication credential is created.

**Command Modes**

Voice logout-profile configuration (voice-logout-profile)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XW</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command was enhanced for password encryption, based on Unified CME password policy.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command in voice logout-profile configuration mode to add an authentication credential to a logout profile for Extension Mobility. The authentication credential is used by TAPI phone devices and certain other applications to log into Cisco Unified CME via an IP phone that is enabled for Extension Mobility and on which the logout profile is downloaded.

The user name parameter of any authentication credential must be unique. Do not use the same value for a user name when you configure any two or more authentication credentials in Cisco Unified CME, such as the username for any Cisco United CME GUI account and the user name in a profile for Extension Mobility.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters [0|6]. This in accordance with Unified CME Password Policy. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.
The following example shows the configuration for a logout profile that defines the default appearance for a Cisco Unified IP phone that is enabled for Extension Mobility, including the username (23C2-8) and password (43214) to be used by a TAPI phone device to log into Cisco Unified CME:

```
pin 9999
user 23C2-8 password 43214
number 3001 type silent-ring
number 3002 type beep-ring
number 3003 type feature-ring
number 3004 type monitor-ring
number 3005,3006 type overlay
number 3007,3008 type cw-overly
speed-dial 1 2000
speed-dial 2 2001 blf
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>logout-profile</td>
<td>Enables a Cisco Unified IP phone for extension mobility and assigns a logout profile to this phone.</td>
</tr>
<tr>
<td>reset (voice logout-profile and voice user-profile)</td>
<td>Performs a complete reboot of all IP phones to which a particular logout-profile or user-profile is downloaded.</td>
</tr>
</tbody>
</table>
user (voice user-profile)

To create an authentication credential to be used by Extension Mobility in Cisco Unified CME, use the username command in voice user-profile configuration mode. To remove the credential, use the no form of this command.

```
user name password password
no user name password password
```

<table>
<thead>
<tr>
<th>name</th>
<th>Unique alphanumeric string to identify a user for this authentication credential only. String can contain a maximum of 15 alphanumeric characters.</th>
</tr>
</thead>
<tbody>
<tr>
<td>password</td>
<td>Password to be used with this username for authentication purposes.</td>
</tr>
<tr>
<td>password</td>
<td>Alphanumeric string.</td>
</tr>
</tbody>
</table>

**Command Default**

Credential is not created.

**Command Modes**

Voice user-profile configuration (config-user-profile)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XW</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
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<tr>
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<td>Cisco Unified CME 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command in voice user-profile configuration mode creates a credential to be authenticated by Cisco Unified CME before a phone user can log into a Cisco Unified IP phone that is enabled for Extension Mobility.

The user name parameter of any authentication credential must be unique. Do not use the same value for a user name when you configure any two or more authentication credentials in Cisco Unified CME, such as the username for any Cisco United CME GUI account and the user name in a profile for Extension Mobility.

When a user logs into an extension mobility enabled phone, Cisco Unified CME retrieves the appropriate user profile, based on username and password match, and replace the phone’s default logout profile with the user’s profile.

**Examples**

The following example shows the configuration to be downloaded after a user enters the username and password configured in this profile, and Cisco Unified CME matches the entry to the credentials in a user profile database:

```
voice user-profile 1
pin 12345
user me password pass123
number 2001 type silent-ring
number 2002 type beep-ring
```
number 2003 type feature-ring
number 2004 type monitor-ring
number 2005,2006 type overlay
number 2007,2008 type cw-overly
speed-dial 1 3001
speed-dial 2 3002 blf

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>reset (voice logout-profile and voice user-profile)</td>
<td>Performs complete reboot of all IP phones on which a particular logout-profile or user-profile is downloaded.</td>
</tr>
</tbody>
</table>
user-locale (ephone-template)

To specify a user locale in an ephone template, use the `user-locale` command in ephone-template configuration mode. To reset to the default user locale, use the `no` form of this command.

```
user-locale  user-locale-tag
no  user-locale
```

**Syntax Description**

| user-locale-tag | Locale identifier that was assigned to the user locale using the `user-locale (telephony-service)` command. |

**Command Default**

The default user locale (user-locale 0) is used.

**Command Modes**

Ephone-template configuration (config-ephone-template)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

To apply user locales to individual ephones, you must specify per-phone configuration files using the `cnf-file perphone` command and identify the locales using the `user-locale (telephony-service)` command.

After creating an ephone template that contains a locale tag, use the `ephone-template (ephone)` command to apply the template to individual ephones.

**Examples**

The following example defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales, ephone 12 uses FR, ephone 13 uses ES, and ephone 14 uses the default, US.

```
telephony-service
  cnf-file location flash:
  cnf-file perphone
  user-locale 1 JP
  user-locale 2 FR
  user-locale 3 ES
  network-locale 1 JP
  network-locale 2 FR
  network-locale 3 ES
  create cnf-files
  ephone-template 1
  user-locale 1
  network-locale 1
  ephone-template 2
  user-locale 2
  network-locale 2
  ephone-template 3
  user-locale 3
  network-locale 3
  ephone 11
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>cnf-file</td>
<td>Specifies the type of configuration files that phones use.</td>
</tr>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone.</td>
</tr>
<tr>
<td>user-locale (telephony-service)</td>
<td>Sets the language for displays on supported phone types.</td>
</tr>
</tbody>
</table>
user-locale (telephony-service)

To define languages for displays on supported phones, use the `user-locale` command in telephony-service configuration mode. To remove a locale configuration, use the `no` form of this command.

```
user-locale [user-locale-tag] [user-defined-code] country-code [load TAR-filename]
no user-locale [user-locale-tag] country-code
```

### Syntax Description

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>user-locale-tag</code></td>
<td>(Optional) Identifier for the specified locale. Required to configure multiple locales only. Range is 0 to 4. Default is 0.</td>
</tr>
<tr>
<td><code>user-defined-code</code></td>
<td>(Optional) Label for locale that is not one of the 12 standard ISO 366 locales. Use each label for only one <code>user-locale-tag</code> at a time. Values are U1, U2, U3, U4, and U5.</td>
</tr>
</tbody>
</table>
| `country-code`  | • DE—Germany  
                  • DK—Denmark  
                  • ES—Spain  
                  • FR—France  
                  • IT—Italy  
                  • JP—Japan  
                  • NL—Netherlands  
                  • NO—Norway  
                  • PT—Portual  
                  • RU—Russia  
                  • SE—Sweden  
                  • US—United States  
                  • Any valid ISO 639 code to be associated with the `user-defined-code` argument (U1 to U5) only. Code must be for a supported locale that is not listed above and for which the XML files can be downloaded to flash, slot 0, or a TFTP server.  
                  • U1, U2, U3, U4, U5—Only when used with the `load` keyword and where U1 to U5 corresponds to a user-defined locale for which the TAR file is downloaded to flash, slot 0, or a TFTP server. |
| `load`          | (Optional) Extracts contents of a TAR file to the location specified by using the `cnf-file location` command. This keyword is supported in Cisco Unified CME 7.0(1) and later versions. |
| `TAR-filename`  | TAR file containing the language JAR file and the tg3-tones.xml file for country-specific network tones and cadences. |

### Command Default

The default user-locale tag is 0 and the default locale is US (United States).

### Command Modes

Telephony-service configuration (config-telephony)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>
### Usage Guidelines

This command sets the language for displays on supported phone types. The `show telephony-service tftp-bindings` command displays the locale that is set using this command. This locale is associated with the dictionary and language files.

Follow this command with a complete phone reboot using the `reset` command.

User-locale 0 always holds the default language that is used for all SCCP phones that are not assigned alternative user locales or user-defined user locales. The system default is US (United States) unless you use this command to designate a different locale for user-locale 0.

#### Alternative User Locales

In Cisco Unified CME 4.0 or a later version, the `user-locale-tag` argument allows you to specify up to five alternative user locales. For example, a company can specify French for phones A, B, and C; German for phones D, E, and F; and United States for phones G, H, and I.

Each of the five user locales that you can use in a multi locale system is identified with the `user-locale-tag` argument. The identifier 0 always holds the default locale, although you can define this default to be any language code that is supported in the system and is listed in CLI help for the command. For example, if you define locale-tag 0 to be JP (Japanese), the default user locale for the router is JP. If you do not specify a locale for identifier 0, the default is US (United States). If you are using this command to configure a default locale for all SCCP phones in your system, you are not required to include `user-locale-tag 0` in the command.

To apply alternative user locales to different phones, you must use the `cnf-files` command to specify per-phone configuration files. When you use per-phone configuration files, a phone's configuration file automatically uses the default locales in user locale 0 and network locale 0. You can override this default for individual ephones by assigning locale tags to the alternative language codes that you want to use. Use ephone templates to assign the locale tag to individual ephones. For example, you can assign user-locale-tag 2 to the language code RU (Russian).
Use the **user-locale** command in ephone-template mode to apply a locale tag to an ephone template. Use the **ephone-template** command in ephone configuration mode to apply the template a phones that should use the alternative language.

**User-Defined User Locales**

In Cisco Unified CME 4.0 and later versions, you can install XML files to support up to five user and network locales that are not standard in your system. These files cannot be installed in the system storage location. To support user-defined locales, you must use the **cnf-files perphone** and **cnf-file location** commands and copy the appropriate XML language files into slot 0, flash, or TFTP memory. The user locales and network locales that are stored in this way can then be used as default or alternative entries for all or some phones.

For example, if you have a site at which the phones should use the displays and tones for Traditional Chinese, which is not one of the standard languages, you must download the XML files for Traditional Chinese to use this user-defined locale on a phone.

**Locale Installer**

In Cisco Unified CME 7.0(1) and later versions, this command with the **load** keyword is a locale installer that extracts the contents of the locale TAR file to the location specified by the **cnf-file location** command. Before Cisco Unified CME 7.0(1), you had to manually extract the files to flash, slot 0, or an external TFTP server.

Before using this command as a locale installer, you must manually create the required locale folders in the root directory of the external TFTP server.

**Examples**

The following example sets the default language tag for the IP phone display to French:

```bash
telephony-service
user-locale FR
```

The following example sets the default language tag for the IP phone display to French. It shows another way to change the default:

```bash
telephony-service
user-locale 0 FR
```

The following example sets the alternative language tag 1 to German:

```bash
telephony-service
user-locale 1 DE
```

**Cisco Unified CME 4.0 and Later Versions: Alternative User Locale**

The following example defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales, ephone 12 uses FR, ephone 13 uses ES, and ephone 14 uses the default, US.

```bash
telephony-service
cnf-file location flash:
cnf-file perphone
user-locale 1 JP
user-locale 2 FR
```
Cisco Unified CME 4.0 and Later Versions: User-Defined User Locale

The following example applies locale tag 4 to the user-defined code U1, which is defined as ZH. ZH is the code that represents Traditional Chinese in ISO 639, the Language Code Reference. Because the code for Traditional Chinese is not one of those that is provided in the system, the user must download the appropriate XML files to support this language.

In addition to the user-defined code, the example defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales; ephone 12 uses FR; ephone 13 uses ES; ephone 14 uses the default, US; and ephone 15 uses the user-defined language, ZH (Traditional Chinese).
Cisco Unified CME 7.0(1) and Later Versions: Using Locale Installer

The following example is the output from the `user-locale` command when the user-defined locale is on the default locale index (0) and the country-code is U2 for user-defined Finnish. The contents of the TAR file are extracted to flash, slot 0, or a TFTP server as previously specified by the `cnf-file` location command.

Router(config-telephone)# user-locale U2 load CME-locale-fi_FI-7.0.1.1.tar
Updating CNF files
LOCALE INSTALLER MESSAGE: VER:1
LOCALE INSTALLER MESSAGE: Langcode:fi
LOCALE INSTALLER MESSAGE: Language:Finnish
LOCALE INSTALLER MESSAGE: Filename: 7905-dictionary.xml
LOCALE INSTALLER MESSAGE: Filename: 7905-kate.xml
LOCALE INSTALLER MESSAGE: Filename: 7920-dictionary.xml
LOCALE INSTALLER MESSAGE: Filename: 7960-dictionary.xml
LOCALE INSTALLER MESSAGE: Filename: 7960-font.xml
LOCALE INSTALLER MESSAGE: Filename: 7960-kate.xml
LOCALE INSTALLER MESSAGE: Filename: 7960-tones.xml
LOCALE INSTALLER MESSAGE: Filename: mk-sccp.jar
LOCALE INSTALLER MESSAGE: Filename: tc-sccp.jar
LOCALE INSTALLER MESSAGE: Filename: td-sccp.jar
LOCALE INSTALLER MESSAGE: Filename: tags_file
LOCALE INSTALLER MESSAGE: Filename: utf8_tags_file
LOCALE INSTALLER MESSAGE: Filename: g3-tones.xml
LOCALE INSTALLER MESSAGE: Filename: SCCP-dictionary.utf-8.xml
LOCALE INSTALLER MESSAGE: Filename: SCCP-dictionary.xml
LOCALE INSTALLER MESSAGE: Filename: ipc-sccp.jar
LOCALE INSTALLER MESSAGE: Filename: gp-sccp.jar
LOCALE INSTALLER MESSAGE: New Locale configured
Processing file:flash:/its/user_define_2_tags_file
Processing file:flash:/its/user_define_2_utf8_tags_file
CNF-FILES: Clock is not set or synchronized, retaining old versionStamps
CNF files updating complete

Router(config-telephony)# create cnf-files
Router(config-telephony)# ephone 3
Router(config-ephone)# reset
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>cnf-file location</td>
<td>Specifies a storage location for XML configuration files.</td>
<td></td>
</tr>
<tr>
<td>cnf-files</td>
<td>Specifies the type of phone configuration files to be created.</td>
<td></td>
</tr>
<tr>
<td>ephone-template (ephone)</td>
<td>Applies an ephone template to an ephone.</td>
<td></td>
</tr>
<tr>
<td>network-locale (telephony-service)</td>
<td>Selects a code for a geographically specific set of tones and cadences on supported phone types.</td>
<td></td>
</tr>
<tr>
<td>reset (ephone)</td>
<td>Performs a complete reboot of one phone associated with a Cisco Unified CME router.</td>
<td></td>
</tr>
<tr>
<td>reset (telephony-service)</td>
<td>Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.</td>
<td></td>
</tr>
<tr>
<td>show telephony-service tftp-bindings</td>
<td>Displays the current configuration files that are accessible by IP phones.</td>
<td></td>
</tr>
<tr>
<td>user-locale (ephone-template)</td>
<td>Applies a user locale tag to an ephone template.</td>
<td></td>
</tr>
</tbody>
</table>
user-locale (voice register)

To define languages for display on supported Cisco Unified SIP IP phones, use the `user-locale` command in voice register global or voice register template configuration mode. To remove a locale configuration, use the no form of this command.

```
user-locale  [user-locale-tag] [user-defined-code] country-code [load TAR-filename]
```

```
no user-locale [user-locale-tag] [user-defined-code] country-code [load TAR-filename]
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>user-locale-tag</strong></td>
<td>(Optional) Identifier for the specified locale. Required to configure multiple locales only. Range is 0 to 4. Default is 0.</td>
</tr>
<tr>
<td><strong>user-defined-code</strong></td>
<td>(Optional) Label for locale that is not one of the 12 standard ISO 366 locales. Use each label for only one user-locale-tag at a time. Values are U1, U2, U3, U4, and U5. This option is only available after the <code>tftp-path</code> command is configured in voice register global configuration mode and the directory in which the configuration files are written is specified (flash, slot, or an external TFTP server).</td>
</tr>
</tbody>
</table>
| **country-code** | • DE—Germany  
• DK—Denmark  
• ES—Spain  
• FR—France  
• IT—Italy  
• JP—Japan  
• NL—Netherlands  
• NO—Norway  
• PT—Portugal  
• RU—Russia  
• SE—Sweden  
• U1—User defined user-locale 1  
• U2—User defined user-locale 2  
• U3—User defined user-locale 3  
• U4—User defined user-locale 4  
• U5—User defined user-locale 5  
• US—United States  
• Any valid ISO 639 code to be associated with the user-defined-code argument (U1 to U5) only. Code must be for a supported locale that is not listed above and for which the XML files can be downloaded to flash, slot 0, or a TFTP server.  
• U1, U2, U3, U4, U5—Only when U1 to U5 corresponds to a user-defined locale for which the TAR file is downloaded to flash, slot 0, or a TFTP server. |
| **load** | (Optional) Loads the specified localization package file. |
| **TAR-filename** | Note Use the complete filename, including the file suffix (.tar), when you configure the user-locale command for all Cisco Unified SIP IP phone types. |
The default user-locale tag is 0 and the default locale is US (United States).

Voice register global (config-register-global)
Voice register template (config-register-temp)

This command sets the language for displays on supported phone types.

The show voice register global command displays the language (locale) that is set using this command. This locale is associated with the dictionary and language files.

Follow this command with a complete phone reboot using the reset (voice register global) command.

User-locale 0 always holds the default language that is used for all SIP phones that are not assigned alternative user locales or user-defined user locales. The system default is US (United States) unless you use this command to designate a different locale for user-locale 0.

Alternative User Locales

The user-locale-tag argument allows you to specify up to five alternative user locales. For example, a company can specify French for phones A, B, and C; German for phones D, E, and F; and United States for phones G, H, and I.

Each of the five user locales that you can use in a multilocal system is identified with the user-locale-tag argument. The identifier 0 always holds the default locale, although you can define this default to be any language code that is supported in the system and is listed in CLI help for the command. For example, if you define locale-tag 0 to be JP (Japanese), the default user locale for the router is JP. If you do not specify a locale for identifier 0, the default is US (United States). If you are using this command to configure a default locale for all SIP phones in your system, you are not required to include user-locale-tag 0 in the command.

Use the user-locale command in voice register template configuration mode to apply a locale tag to a voice register template. Use the voice register template command in global configuration mode to apply the template to phones that should use the alternative language.

Example

The following example sets the default language tag for the IP phone display to French:

```plaintext
voice register global
user-locale 0 FR
```

The following example sets alternative language tag 2 as CH (Chinese):

```plaintext
Tftp path is flash:
Generate text file is disabled
Tftp files are created, current syncinfo 0202310605309206
OS79XX.TXT is not created
timeout interdigit 10
network-locale[0] US (This is the default network locale for this box)
network-locale[1] US
```
The following examples sets user-locale 2 and 3 for voice register template 5 and 6, respectively:

```plaintext
voice register template 1
softkeys hold Resume Newcall
softkeys idle Redial DND Gpickup Pickup Cfwdall
softkeys connected Endcall Hold Confrn Park Transf
softkeys remote-in-use Barge Newcall cBarge
no transfer-blind enable

voice register template 5
user-locale 2
!
voice register template 6
user-locale 3
!
```

The following example loads the locale package file for Germany:

```plaintext
Router(config)# voice register global
Router(config-register-global)# user-locale 2 DE load CME-locale-de_DE-German-8.6.3.0.tar
```

The following example loads the locale package file for Italy:

```plaintext
Router(config)# voice register global
Router(config-register-global)# user-locale IT load CME-locale-it_IT-Italian-8.6.2.4.tar
```

```
Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>reset (voice register global)</td>
<td>Performs a complete reboot of one phone associated with a Cisco Unified CME router.</td>
</tr>
<tr>
<td>show voice register global</td>
<td>Displays the current configuration files that are accessible to the Cisco Unified SIP IP phones.</td>
</tr>
<tr>
<td>voice register global</td>
<td>Sets global parameters for all supported Cisco SIP IP phones in a Cisco Unified CME environment.</td>
</tr>
<tr>
<td>voice register template</td>
<td>Defines a template of common parameters for Cisco Unified SIP IP phones.</td>
</tr>
</tbody>
</table>
```
username (ephone)

To assign a login account username and password to a phone user so that the user can log in to the Cisco Unified CME router through a web browser, use the `username` command in ephone configuration mode. To disable the username and password, use the `no` form of this command.

```
username username password [0|6] password
no username username password [0|6] password
```

**Syntax Description**

- `username`: Unique alphanumeric string to identify a user for this authentication credential only. String can contain a maximum of 28 alphanumeric characters. Default is Admin.
- `password`: Enables password for the Cisco Unified IP phone user.
- `password`: Password string.

**Command Default**

The default username for the administrator is Admin.

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command was enhanced for password encryption, based on Unified CME password policy.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command assigns a login account username and password for a phone user and establishes a login account for each Cisco Unified IP phone. This configuration can be completed only by the local system administrator of the Cisco Unified CME router.

You must also create a login account to allow Telephone Application Programming Interface (TAPI)-aware PC applications to register with the Cisco router and exercise remote-control operation of a Cisco Unified IP phone.

The user name parameter of any authentication credential must be unique. Do not use the same value for a user name when you configure any two or more authentication credentials in Cisco Unified CME, such as the `username` for any Cisco United CME GUI account and the user name in a profile for Extension Mobility.

This configuration permits the phone user to log in to Cisco Unified CME to view and change attributes associated only with the user’s IP phone.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters `[0|6]`. This is in accordance with Unified CME Password Policy. The 0 in the parameter `[0|6]` mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

**Examples**

The following example shows how to set the username and password:
Router(config)# ephone 1
Router(config-ephone)# username smith password 9golf

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>admin-password</td>
<td>Sets a password for the local system administrator of the Cisco IOS Telephony Service.</td>
</tr>
<tr>
<td>admin-username</td>
<td>Sets the username for the local system administrator of the Cisco IOS Telephony Service router.</td>
</tr>
</tbody>
</table>
username (voice register pool)

To assign an authentication credential to a phone user so that the SIP phone can register in Cisco CallManager Express (Cisco CME), use the `username` command in voice register pool configuration mode. To disable a username and password, use the `no` form of this command.

```
username username [password [0|6] password]
no username username [password [0|6] password]
```

**Syntax Description**

| username | Username of the local Cisco IP phone user. Default: Admin. |
| password | Enables password for the Cisco IP phone user. |
| password | Password string. |

**Command Default**

Authentication credential is not created.

**Command Modes**

Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command was enhanced for password encryption, based on Unified CME password policy.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Creates an authentication credential for SIP IP phone registration. This command is required if authentication is enabled with the `authenticate` command.

You must configure the voice register pool before configuring an authentication credential.

All lines in a phone share the same credential.

This configuration also permits the phone user to log in to Cisco Unified CME to view and change attributes associated only with the user’s SIP IP phone.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters `[0|6]`. This in accordance with Unified CME Password Policy. The 0 in the parameter `[0|6]` mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

**Note**

This command is not for SIP proxy registration.

**Examples**

The following example shows how to set the username and password:

```
Router(config)# voice register pool 1
Router(config-register-pool)# username smith password 0 9golf
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>authenticate (voice register global)</strong></td>
<td>Enables authentication for registration requests in which the MAC address cannot be identified by using other methods</td>
</tr>
</tbody>
</table>
utf8

To define Unicode UTF-8 support for a phone type, use the **utf8** command in ephone-type configuration mode. To reset to the default value, use the **no** form of this command.

```
utf8
no utf8
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Phone type supports Unicode UTF-8.

**Command Modes**

Ephone-type configuration (config-ephone-type)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3 Cisco Unified SRST 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies whether Unicode UTF-8 is supported by the type of phone that is being added with the phone-type template.

**Examples**

The following example shows that UTF-8 support is set to disabled for the Nokia E61 when creating the ephone-type template:

```
Router(config)# ephone-type E61
Router(config-ephone-type)# no utf8
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>device-id</td>
<td>Specifies the device ID for a phone type.</td>
</tr>
<tr>
<td>type</td>
<td>Assigns the phone type to an SCCP phone.</td>
</tr>
</tbody>
</table>
Cisco Unified CME Commands: V

- vad (voice register pool), on page 1397
- vad (voice register template), on page 1398
- vca, on page 1399
- video, on page 1401
- video (ephone), on page 1403
- video (telephony-service), on page 1404
- video screening (voice service sip), on page 1405
- video-bitrate (ephone), on page 1406
- vm-device-id (ephone), on page 1407
- vm-integration, on page 1408
- voice class mlpp, on page 1410
- voice emergency response location, on page 1411
- voice emergency response settings, on page 1413
- voice emergency response zone, on page 1415
- voice hunt-group, on page 1416
- voice-hunt-groups login, on page 1419
- voice lpco call-block cause, on page 1421
- voice lpco custom, on page 1425
- voice lpco enable, on page 1426
- voice lpco ip-phone mobility, on page 1427
- voice lpco ip-phone subnet, on page 1428
- voice lpco ip-trunk subnet incoming, on page 1430
- voice lpco policy, on page 1431
- voice mlpp, on page 1433
- voice moh-group, on page 1434
- voice register dialplan, on page 1435
- voice register dn, on page 1437
- voice register global, on page 1439
- voice register pool, on page 1441
- voice register pool-type, on page 1443
- voice register session-server, on page 1446
- voice register template, on page 1448
- voice user-profile, on page 1449
• voice-class codec (voice register pool), on page 1451
• voice-class mlpp (dial peer), on page 1453
• voice-class stun-usage, on page 1454
• voice-gateway system, on page 1455
• voicemail (telephony-service), on page 1456
• voicemail (voice register global), on page 1457
• voicemail (voice register template), on page 1458
• voice-port (voice-gateway), on page 1460
• vpn-gateway, on page 1461
• vpn-group, on page 1462
• vpn-hash-algorithm, on page 1463
• vpn-profile, on page 1464
• vpn-trustpoint, on page 1466
vad (voice register pool)

To enable voice activity detection (VAD) on a VoIP dial peer, use the `vad` command in voice register pool configuration mode. To disable VAD, use the `no` form of this command.

```
vad
no vad
```

Syntax Description

This command has no arguments or keywords.

Command Default

VAD is enabled.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td></td>
<td>Cisco SIP SRST 3.4</td>
<td></td>
</tr>
</tbody>
</table>

Usage Guidelines

VAD detects periods of silence in the voice signal and temporarily discontinues transmission of the signal during these periods to save bandwidth. Because VAD is enabled by default, there is no comfort noise during periods of silence. As a result, the call may seem to be disconnected and you may prefer to set `no vad` on the SIP phone pool.

Examples

The following example shows how to disable VAD for pool 1:

```
Router(config)# voice register pool 1
Router(config-register-pool)# no vad
```
vad (voice register template)

To enable voice activity detection (VAD) on SIP phones, use the `vad` command in voice register template configuration mode. To return to the default, use the `no` form of this command.

```
vad
no vad
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

VAD is disabled.

**Command Modes**

Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

VAD detects periods of silence in the voice signal and temporarily discontinues transmission of the signal during these periods to save bandwidth. To apply the template to a SIP phone, use the `template` command in voice register pool configuration mode.

**Examples**

The following example shows how to enable VAD:

```
Router(config)# voice register template 1
Router(config-register-temp)# vad
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>template (voice register pool)</td>
</tr>
</tbody>
</table>
vca

To specify the audio file used for the vacant code announcement, use the `vca` command in voice MLPP configuration mode. To disable use of this audio file, use the `no` form of this command.

```
vca audio-url voice-class cause-code tag
no vca
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>audio-url</code></td>
<td>Location of the announcement audio file in URL format. Valid storage locations are TFTP, FTP, HTTP, and flash memory.</td>
</tr>
<tr>
<td><code>tag</code></td>
<td>Number of the voice class that defines the cause codes for which the VCA is played. Range: 1 to 64.</td>
</tr>
</tbody>
</table>

**Command Default**

No announcement is played.

**Command Modes**

Voice MLPP configuration (config-voice-mlpp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command specifies the G.711 a-law or u-law 8-KHz encoded audio file (.wav or .au format) for the announcement that plays to callers when they dial an invalid or unassigned number.

The `mlpp indication` command must be enabled (default) for a phone to play precedence announcements.

The VCA plays for the cause codes defined with the `voice class cause-code` command.

This command is not supported by Cisco IOS help. If you type `?`, Cisco IOS help does not display a list of valid entries.

**Examples**

The following example shows that the audio file played for the vacant code announcement is named `vca.au` and is located in flash. The announcement plays for the unassigned-number and invalid-number cause codes, which are defined in the matching cause-code voice class.

```
voice class cause-code 1
  unassigned-number
  invalid-number
!
voice mlpp
vca flash:vca.au voice-class cause-code 1
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>bnea</strong></td>
<td>Specifies the audio file used for the busy station not equipped for preemption announcement.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td><strong>bpa</strong></td>
<td>Specifies the audio file used for the blocked precedence announcement.</td>
</tr>
<tr>
<td><strong>ica</strong></td>
<td>Specifies the audio file used for the isolated code announcement.</td>
</tr>
<tr>
<td><strong>mlpp indication</strong></td>
<td>Enables MLPP indication on an SCCP phone or analog FXS port.</td>
</tr>
<tr>
<td><strong>voice class cause-code</strong></td>
<td>Creates a voice class for defining a set of cause codes.</td>
</tr>
</tbody>
</table>
**video**

To enable video capability on Cisco Unified IP Phones 9951 and 9971, use the `video` command in voice register global, voice register template, and voice register pool configuration modes. To disable video capabilities on Cisco Unified IP Phones 9951 and 9971, use the `no` form of this command.

```
video
no video
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Video capability is disabled on Cisco Unified IP Phones 9951 and 9971.

**Command Modes**

- Voice register global
- Voice register template
- Voice register pool

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(4)M</td>
<td>Cisco Unified CME 8.6</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to enable video capability on Cisco Unified IP Phones 9951 and 9971. Video is supported on Cisco Unified IP phone 8961 through CUVA. You need to create profile and apply-config or restart to the phone to enable the video capability on phones.

**Examples**

The following example shows video command configured in voice register global:

```
Router#show run
!
!
voice service voip
  allow-connections sip to sip
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
!
!
voice register global
  mode cme
  bandwidth video tias-modifier 244 negotiate end-to-end
  max-pool 10
  video
!
voice register template 10
!
!
```

The following example shows video command configured under voice register pool 5, you can also configure the video command under voice register template:

```
Router#show run
!
!
```
voice service voip
  allow-connections sip to sip
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
!
voice register global
  mode cme
  bandwidth video tias-modifier 244 negotiate end-to-end
  max-pool 10
!
voice register pool 1
  id mac 1111.1111.1111
!
voice register pool 4
!
voice register pool 5
  logout-profile 58
  id mac 0009.A3D4.1234
  video
!

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>apply-config</td>
<td>Allows to dynamically apply the phone configuration on Cisco Unified SIP IP phones 8961, 9951, and 9971,</td>
</tr>
<tr>
<td>bandwidth video tias-modifier</td>
<td>Allows to set the maximum video bandwidth bytes per second (BPS) for SIP IP phones</td>
</tr>
</tbody>
</table>
**video (ephone)**

To enable video capabilities for an SCCP phone in Cisco Unified CME, use the `video` command in ephone configuration mode. To reset to default, use the `no` form of this command.

```plaintext
video
no video
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Video capabilities are disabled.

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables video capabilities in the ephone configuration for a particular phone.

Video capabilities for SCCP phones in Cisco Unified CME must be enabled globally as well as for individual phones. You must enable video for all video-capable SCCP phones associated with a Cisco Unified CME router by configuring the `video` capability parameter of the `service phone` command.

Video parameters, such as maximum bit rate, are set at a system-level in video configuration mode.

**Examples**

The following example shows the ephone portion from the `show running-configuration` command:

```plaintext
router# show running-configuration
.
.
. ephone 6
  video
    mac-address 000F.F7DE.CAA5
    type 7960
    button 1:6
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>service phone</code></td>
<td>Modifies the vendorConfig parameters in phone configuration files.</td>
</tr>
<tr>
<td><code>video (telephony-service)</code></td>
<td>Enters video configuration mode for modifying video parameters in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
video (telephony-service)

To enter video configuration mode for setting video parameters for all video-capable phones in Cisco Unified CME, use the `video` command in telephony-service configuration mode. To reset global video parameters, use the `no` form of this command.

```
video
no video
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
Defaults for global video parameters are configured.

**Command Modes**
Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command enters video configuration mode for setting video parameters for all video-capable Cisco Unified IP phones associated with a Cisco Unified CME router.

**Examples**
The following example shows how to enter video configuration mode for a Cisco Unified CME router. You must enter video configuration mode to set video parameters, such as maximum bit rate.

```
Router(config)#
telephony-service
Router(config-telephony)# video
Router(config-tele-video)# maximum bit-rate 256
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>maximum bit-rate</td>
<td>Sets the maximum video bandwidth for phones in Cisco unified CME.</td>
</tr>
<tr>
<td><code>show call active video</code></td>
<td>Displays call information for SCCP video calls in progress.</td>
</tr>
<tr>
<td><code>show call history video</code></td>
<td>Displays call history information for SCCP video calls.</td>
</tr>
</tbody>
</table>
video screening (voice service sip)

To enable transcoding and transsizing between two call legs when configuring SIP, use the `video screening` command in sip configuration mode. To disable transcoding and transsizing, use `no` form of this command.

```
video screening
no video screening
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Video screening is disabled

**Command Modes**

Sip

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(4)M</td>
<td>The command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to enable conversion of video streams if there is a mismatch between two call legs.

**Examples**

The following example enters the voice-card configuration mode and enables video screening:

```
Router(config)# voice service voip
Router(config-voicecard)# sip
Router((conf-serv-sip)# video screening
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>codec profile</td>
<td>Defines the video capabilities needed for video endpoints.</td>
</tr>
<tr>
<td>video codec</td>
<td>Assigns a video codec to a VoIP dial peer.</td>
</tr>
</tbody>
</table>
video-bitrate (ephone)

To specify the maximum IP phone video bandwidth in Cisco Unified CME, use the `video-bitrate` command in the ephone mode. To restore the default video bitrate or use the `no` form of this command.

```
video-bitrate value
no video-bitrate
```

**Syntax Description**

| value | Video bandwidth in kb/s. Range is from 64 to 102400 kbps. |

**Command Default**

Bit rate defaults to the maximum bit-rate configured under video configuration.

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(4)M</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to modify the value of the maximum video bandwidth for video-capable phones that support SIP, SCCP, and H.323.

**Examples**

The following example sets a bit-rate of 512 kb/s.

```
Router(config)# ephone
Router(config-ephone)# video-bitrate 512
```
vm-device-id (ephone)

To define a voice-messaging identification string, use the `vm-device-id` command in ephone configuration mode. To disable this feature, use the `no` form of this command.

```
vm-device-id  id-string
no vm-device-id
```

**Syntax Description**

| id-string | Voice-messaging device port identification (ID) string; for example, CiscoUM-VI1 for the first port and CiscoUM-VI2 for the second port. Note that the first two characters after the hyphen must be the uppercase letters V and I. |

**Command Default**

No voice-mail identification string is defined.

**Command Modes**

Ephone configuration (config-ephone)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to define a voice-messaging device ID string. A voice-messaging port registers with a device ID instead of a MAC address. To distinguish among different voice-messaging ports, the value of the voice-messaging device ID is used. The voice-messaging device ID is configured to a Cisco IP phone port, which maps to a corresponding voice-messaging port.

**Examples**

The following example shows how to set the voice-messaging device ID to CiscoUM-VI1:

```
Router(config) ephone 1
Router(config-ephone) vm-device-id CiscoUM-VI1
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Configures the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed.</td>
</tr>
</tbody>
</table>

```voicemail (telephony-service)`
vm-integration

To enter voice-mail integration configuration mode and enable voice-mail integration with dual tone multifrequency (DTMF) and analog voice-mail systems, use the `vm-integration` command in global configuration mode. To disable voice-mail integration, use the `no` form of this command.

```
vm-integration
no vm-integration
```

**Syntax Description**
This command has no arguments or keywords.

**Command Default**
DTMF integration with voice-mail system is disabled.

**Command Modes**
Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)YT</td>
<td>Cisco SRST 2.1</td>
<td>This command was introduced for Cisco Survivable Remote Site Telephony (SRST).</td>
</tr>
<tr>
<td>12.2(2)XT</td>
<td>Cisco ITS 2.0</td>
<td>This command was introduced Cisco ITS.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0 Cisco SRST 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
The `vm-integration` command is used to enter voice-mail integration configuration mode to enable in-band DTMF integration with a voice-mail system.

**Examples**
The following example shows how to enter the voice-mail integration configuration mode:

```
Router(config) vm-integration
Router(config-vm-integration) pattern direct 2 CGN *
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>pattern direct (vm-integration)</code></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when a user presses the Messages button on a phone.</td>
</tr>
<tr>
<td><code>pattern ext-to-ext busy (vm-integration)</code></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension reaches a busy extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td><code>pattern ext-to-ext no-answer (vm-integration)</code></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td><code>pattern trunk-to-ext busy (vm-integration)</code></td>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>-------------</td>
<td></td>
</tr>
<tr>
<td><strong>pattern trunk-to-ext no-answer (vm-integration)</strong></td>
<td></td>
</tr>
<tr>
<td>Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.</td>
<td></td>
</tr>
</tbody>
</table>
To create a voice class for the Multilevel Precedence and Preemption (MLPP) service, use the `voice class mlpp` command in global configuration mode. To remove the voice class, use the `no` form of this command.

```
voice class mlpp tag
no voice class mlpp tag
```

### Syntax Description

| Tag | Unique number to identify the voice class. Range: 1 to 10000. |

### Command Default

No voice class is configured for MLPP.

### Command Modes

Global configuration (config)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

This command creates the voice class for MLPP attributes. Use the `voice-class mlpp` (dial peer) command to assign the voice class to a dial peer.

### Examples

The following example shows the domain name set to DSN in the MLPP voice class:

```
Router(config)# voice class mlpp
Router(config-class)# service-domain dsn
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>service-domain (voice class)</td>
<td>Sets the service domain name in the MLPP voice class.</td>
</tr>
<tr>
<td>voice-class mlpp (dial peer)</td>
<td>Assigns an MLPP voice class to a POTS or VoIP dial peer.</td>
</tr>
</tbody>
</table>
voice emergency response location

To create a tag for identifying an emergency response location (ERL) for E911 services, use the `voice emergency response location` command in global configuration mode. To remove the ERL tag, use the `no` form of this command.

```
voice emergency response location tag
no voice emergency response location tag
```

**Syntax Description**

- `tag` Unique number that identifies this ERL tag.

**Command Default**

No ERL tag is created.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1 Cisco Unified SIP SRST 4.1</td>
<td>This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)</td>
<td>This command was added for Cisco Unified CME.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to create an ERL that identifies an area where emergency teams can quickly locate a 911 caller. The ERL definition optionally includes which ELINs are associated with the ERL and which IP phones are located in the ERL. You can define two or fewer unique IP subnets and two or fewer ELINs. If you define one ELIN, this ELIN is always used for phones calling from this ERL. If you define two ELINs, the system alternates between using each ELIN. If you define zero ELINs and phones use this ERL, the outbound calls do not have their calling numbers translated. The PSAP sees the original calling numbers for these 911 calls. You can optionally add the civic address using the `address` command and an address description using the `name` command.

**Examples**

In the following example, all IP phones with the IP address of 10.X.X.X or 192.168.X.X are automatically associated with this ERL. If one of the phones dials 911, its extension is replaced with 408 555-0100 before it goes to the PSAP. The PSAP will see that the caller’s number is 408 555-0100. The civic address, 410 Main St, Tooly, CA, and a descriptive identifier, Bldg 3 are included.

```
voice emergency response location 1
  elin 1 4085550100
  subnet 1 10.0.0.0 255.0.0.0
  subnet 2 192.168.0.0 255.255.0.0
  address 1,408,5550100,410,Main St.,Tooly,CA
  name Bldg 3
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>address</td>
<td>Specifies a comma separated text entry (up to 250 characters) of an ERL’s civic address.</td>
</tr>
<tr>
<td>elin</td>
<td>Specifies a PSTN number that will replace the caller's extension.</td>
</tr>
<tr>
<td>name</td>
<td>Specifies a string (up to 32-characters) used internally to identify or describe the emergency response location.</td>
</tr>
<tr>
<td>subnet</td>
<td>Defines which IP phones are part of this ERL.</td>
</tr>
</tbody>
</table>
**voice emergency response settings**

To define 911 call behavior settings, use the `voice emergency response settings` command in global configuration mode. To remove the settings, use the `no` form of this command.

```
voice emergency response settings
no voice emergency response settings
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

No default behavior or values

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to enable definition of the following 911 call behavior settings:

- **elin**: Default ELIN to use if a 911 caller’s IP phone’s address does not match the subnet of any location in any zone.
- **expiry**: Number of minutes a 911 call is associated to an ELIN in the case of a callback from the 911 operator.
- **callback**: Default number to contact if a 911 callback cannot find the last 911 caller.
- **logging**: Syslog informational message that is printed to the console each time an emergency call is made. This feature is enabled by default, however you can disable this feature by entering the `no` form of this command.

**Examples**

In the following example, if the 911 caller’s IP address does not match any of the voice emergency response locations, the ELIN defined in the `voice emergency response settings` configuration (4085550101) is used. After the 911 call is placed to the PSAP, the PSAP has 120 minutes (2 hours) to call back 408 555-0101 to reach the 911 caller. If during a callback, the last caller’s extension number cannot be found, the call is routed to extension 7500. The outbound 911 calls do not cause a syslog message to the logging facility (for example, to the local buffer, console, or remote host).

```
voice emergency response settings
callback 7500
elin 4085550101
expiry 120
no logging
```
### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>callback</td>
<td>Default phone number to contact if a 911 callback cannot find the last 911 caller from the ERL.</td>
</tr>
<tr>
<td>elin</td>
<td>E.164 number used as the default ELIN if no matching ERL to the 911 caller’s IP phone address is found.</td>
</tr>
<tr>
<td>expiry</td>
<td>Number of minutes a 911 call is associated to an ELIN in the case of a callback from the 911 operator.</td>
</tr>
<tr>
<td>logging</td>
<td>Syslog informational message printed to the console every time an emergency call is made.</td>
</tr>
</tbody>
</table>
voice emergency response zone

To create an emergency response zone, use the `voice emergency response zone` command in global configuration mode. To remove the created voice emergency response zone, use the `no` form of this command.

```
voice emergency response zone  tag
no voice emergency response zone  tag
```

**Syntax Description**

- `tag`: Identifier (1-100) for the voice emergency response zone.

**Command Default**

No default behavior or values

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to create voice emergency response zones that allow routing of 911 calls to different PSAPs.

**Examples**

The following example shows an assignment of ERLs to a voice emergency response zone. The calls have an ELIN from ERLs 8, 9, and 10. The locations for ERLs in zone 10 are searched in the order each CLI is entered for a phone address match because no priority order is assigned.

```
voice emergency response zone 10
location 8
location 9
location 10
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>location</code></td>
<td>Identifies locations within an emergency response zone and optionally assigns a priority order to the location.</td>
</tr>
</tbody>
</table>
voice hunt-group

To create a hunt group for phones in a Cisco Unified Communications Manager Express (Cisco Unified CME) or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) system, use the voice hunt-group command in global configuration mode. To delete a hunt group, use the no form of this command.

```
voice hunt-group hunt-tag {longest-idle|parallel|peer|sequential}
no voice hunt-group hunt-tag
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>hunt-tag</td>
<td>Unique sequence number that identifies the hunt group. Range is 1 to 100.</td>
</tr>
<tr>
<td>longest-idle</td>
<td>Allows an incoming call to go first to the number that has been idle the longest for the number of hops specified when the hunt group was defined. The longest-idle time is determined from the last time that a phone registered, reregistered, or went on-hook.</td>
</tr>
<tr>
<td>parallel</td>
<td>Allows an incoming call to simultaneously ring all the numbers in the hunt group member list.</td>
</tr>
<tr>
<td>peer</td>
<td>Allows a round-robin selection of the first extension to ring. Ringing proceeds in a circular manner from left to right. The round-robin selection starts with the number left of the number that answered when the hunt-group was last called.</td>
</tr>
<tr>
<td>sequential</td>
<td>Allows an incoming call to ring all the numbers in the left-to-right order in which they were listed when the hunt group was defined.</td>
</tr>
</tbody>
</table>

**Command Default**

By default, voice hunt group is not created.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was modified to add support for Cisco Unified SCCP IP phones.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
<tr>
<td>15.2(4)M</td>
<td>Cisco Unified SIP SRST 9.1</td>
<td>This command was introduced in Cisco Unified SIP SRST 9.1.</td>
</tr>
<tr>
<td>15.3(4)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was modified to include support for wildcards which is indicated by &quot;*&quot; symbol.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The voice hunt-group command enters voice hunt-group configuration mode to define a hunt group. A hunt group is a list of phone numbers that take turns receiving incoming calls to a specific number (pilot number), which is defined with the pilot command. The specific extensions included in the hunt group and the order and maximum number of extensions allowed in the list are defined with the list command.
If a number in the list is busy or does not answer, the call is redirected to the next number in the list. The last number tried is the final number, which is defined with the final command. If the number of times that a call is redirected to a new number exceeds 5, you must use the max-redirect command to increase the allowable number of redirects in the Cisco Unified CME or Cisco Unified SIP SRST system.

To configure a new hunt group, you must specify the longest-idle, parallel, peer, or sequential keyword. To change an existing hunt group configuration, the keyword is not required. To change the type of hunt group, for instance from peer to sequential or sequential to peer, you must remove the existing hunt group first by using the no form of this command and then re-create it.

The parallel keyword creates a dial peer to allow an incoming call to ring multiple phones simultaneously. The use of parallel hunt groups is also referred to as application-level forking because it enables the forking of a call to multiple destinations. A pilot dial peer cannot be used as a voice hunt group and a hunt group at the same time.

While ephone hunt groups only support Cisco Unified SCCP IP phones, a voice hunt group supports either a Cisco Unified SCCP IP phone or a Cisco Unified SIP IP phone.

With the voice hunt group feature preconfigured in the Cisco Unified SIP SRST router, voice hunt groups continue to be supported after phones fallback from a Cisco Unified Communications Manager (Cisco Unified CM) to a Cisco Unified SIP SRST router.

**Examples**

The following example shows how to define longest-idle hunt group 1 with pilot number 7501, final number 8000, and nine numbers in the list. After a call is redirected six times (makes 6 hops), it is redirected to the final number 8000.

```
Router(config)# voice hunt-group 1 longest-idle
Router(config-voice-hunt-group)# pilot 7501
Router(config-voice-hunt-group)# list 7001, 7002, 7023, 7028, 7045, 7062, 7067, 7072, 7079
Router(config-voice-hunt-group)# final 8000
Router(config-voice-hunt-group)# hops 6
Router(config-voice-hunt-group)# timeout 20
Router(config-voice-hunt-group)# exit
```

The following example shows how to define peer hunt group number 2. Callers dial the pilot number 5610 to reach the hunt group. The first extension to ring the first time that this hunt group is called is 5601. If 5601 does not answer, the hunt proceeds from left to right, beginning with the extension directly to the right. If none of those extensions answer, the call is forwarded to extension 6000, which is the number for the voice-mail service.

The second time someone calls the hunt group, the first extension to ring is 5602 if 5601 was answered during the previous call.

```
Router(config)# voice hunt-group 2 peer
Router(config-voice-hunt-group)# pilot 5610
Router(config-voice-hunt-group)# list 5601, 5602, 5617, 5633
Router(config-voice-hunt-group)# final 6000
Router(config-voice-hunt-group)# timeout 30
Router(config-voice-hunt-group)# exit
```
The following examples show how to define sequential hunt group number 3. When callers dial extension 5601, the first phone to ring is 5001, then 5002, 5017, and 5028. If none of those extensions answer, the call is forwarded to extension 6000, which is the number for the voice-mail service.

```
Router(config)# voice hunt-group 3 sequential
Router(config-voice-hunt-group)# pilot 5601
Router(config-voice-hunt-group)# list 5001, 5002, 5017, 5028
Router(config-voice-hunt-group)# final 6000
Router(config-voice-hunt-group)# timeout 30
Router(config-voice-hunt-group)# exit
```

The following example shows how to define a parallel hunt group. When callers dial extension 1000, extensions 1001, 1002, and so forth ring simultaneously. The first extension to answer is connected. All other call legs are disconnected. If none of the extensions answer, the call is forwarded to extension 2000, which is the number for the voice-mail service.

```
Router(config)# voice hunt-group 4 parallel
Router(config-voice-hunt-group)# pilot 1000
Router(config-voice-hunt-group)# list 1001, 1002, 1003, 1004
Router(config-voice-hunt-group)# final 2000
Router(config-voice-hunt-group)# timeout 20
Router(config-voice-hunt-group)# exit
```

The following example shows the support for wildcard slots in voice hunt-groups.

```
Router(config)# voice hunt-group 1 parallel
Router(config-voice-hunt-group)# pilot number 100
Router(config-voice-hunt-group)# list 1001, 1002, 1002, *, *
Router(config-voice-hunt-group)# exit
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>final (voice hunt-group)</td>
<td>Defines the last extension in a voice hunt group.</td>
</tr>
<tr>
<td>hops (voice hunt-group)</td>
<td>Defines the number of times that a call is redirected to the next phone number in a peer voice hunt-group list before proceeding to the final phone number.</td>
</tr>
<tr>
<td>list (voice hunt-group)</td>
<td>Defines the phone numbers that participate in a voice hunt group.</td>
</tr>
<tr>
<td>max-redirect</td>
<td>Changes the number of times that a call can be redirected by call forwarding or transfer within a Cisco Unified CME system.</td>
</tr>
<tr>
<td>pilot (voice hunt-group)</td>
<td>Defines the phone number that callers dial to reach a voice hunt group.</td>
</tr>
<tr>
<td>timeout (voice hunt-group)</td>
<td>Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list and defines the last phone number in the hunt group.</td>
</tr>
</tbody>
</table>
voice-hunt-groups login

To enable a voice register dn or ephone dn to join or unjoin voice hunt-groups dynamically, use the voice-hunt-groups login command in voice register dn configuration mode. To disable this capability, use the no form of this command.

voice-hunt-groups login
no voice-hunt-groups login

Syntax Description
This command has no arguments or keywords.

Command Default
A voice register dn or ephone dn is not allowed to dynamically join and unjoin voice hunt groups.

Command Modes
Voice register dn configuration (config-voice-register-dn)
Ephone dn configuration (config-ephone-dn)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.4(3)M</td>
<td>Cisco Unified CME 10.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

Usage Guidelines
Use the show voice hunt-groups command to display current hunt group members, including those who joined the group dynamically.

Examples

The following example creates five voice register dns and a hunt group that includes the first two voice register dn and two wildcard slots. The last three voice register dns are enabled for voice hunt group dynamic membership. Each of them can join and unjoin the hunt group whenever one of the slots is available.

```
voice register dn 22
   number 4566
voice register dn 23
   number 4567
voice register dn 24
   number 4568
voice-hunt-groups login
voice register dn 25
   number 4569
voice-hunt-groups login
voice register dn 26
   number 4570
voice-hunt-groups login
voice-hunt-groups 1 peer
   list 4566,4567,*,*
   final 7777
```

The following example creates three ephone dns and a hunt group that includes the first two ephone dn and two wildcard slots. The last one ephone dn is enabled for voice hunt group dynamic membership. Each of them can join and unjoin the hunt group whenever one of the slots is available.
Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice hunt-groups</td>
<td>Displays voice-hunt group configuration, current status, and statistics.</td>
</tr>
</tbody>
</table>
voice lpcor call-block cause

To define the cause code that is used when a call is blocked because LPCOR validation fails, use the `voice lpcor call-block cause` command in global configuration mode. To reset to the default, use the `no` form of this command.

```plaintext
voice lpcor call-block cause cause-code
no voice lpcor call-block cause
```

**Syntax Description**

- `cause-code`: Number of the cause code to generate when a call is blocked by the LPCOR validation process. Range: 1 to 180.

**Command Default**

Default cause code is 63 (serv/opt-unavail-unspecified).

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The following table lists the available cause codes.

*Table 76: Cause Codes for Calls Blocked by LPCOR Validation*

<table>
<thead>
<tr>
<th>Message</th>
<th>Description</th>
<th>Code Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>access-info-discard</td>
<td>access info discarded (43)</td>
<td>43</td>
</tr>
<tr>
<td>b-cap-not-implemented</td>
<td>bearer capability not implemented (65)</td>
<td>65</td>
</tr>
<tr>
<td>b-cap-restrict</td>
<td>restricted digital info bc only (70)</td>
<td>70</td>
</tr>
<tr>
<td>b-cap-unauthorized</td>
<td>bearer capability not authorized (57)</td>
<td>57</td>
</tr>
<tr>
<td>b-cap-unavail</td>
<td>bearer capability not available (58)</td>
<td>58</td>
</tr>
<tr>
<td>call-awarded</td>
<td>call awarded (7)</td>
<td>7</td>
</tr>
<tr>
<td>call-cid-in-use</td>
<td>call exists call id in use (83)</td>
<td>83</td>
</tr>
<tr>
<td>call-clear</td>
<td>call cleared (86)</td>
<td>86</td>
</tr>
<tr>
<td>call-reject</td>
<td>call rejected (21)</td>
<td>21</td>
</tr>
<tr>
<td>cell-rate-unavail</td>
<td>cell rate not available (37)</td>
<td>37</td>
</tr>
<tr>
<td>channel-unacceptable</td>
<td>channel unacceptable (6)</td>
<td>6</td>
</tr>
<tr>
<td>chantype-not-implement</td>
<td>chan type not implemented (66)</td>
<td>66</td>
</tr>
<tr>
<td>Message</td>
<td>Description</td>
<td>Code Number</td>
</tr>
<tr>
<td>--------------------------</td>
<td>--------------------------------------------</td>
<td>-------------</td>
</tr>
<tr>
<td>cid-in-use</td>
<td>call id in use (84)</td>
<td>84</td>
</tr>
<tr>
<td>codec-incompatible</td>
<td>codec incompatible (171)</td>
<td>171</td>
</tr>
<tr>
<td>cug-incalls-bar</td>
<td>cug incoming calls barred (55)</td>
<td>55</td>
</tr>
<tr>
<td>cug-outcalls-bar</td>
<td>cug outgoing calls barred (53)</td>
<td>53</td>
</tr>
<tr>
<td>dest-incompatible</td>
<td>incompatible destination (88)</td>
<td>88</td>
</tr>
<tr>
<td>dest-out-of-order</td>
<td>destination out of order (27)</td>
<td>27</td>
</tr>
<tr>
<td>dest-unroutable</td>
<td>no route to destination (3)</td>
<td>3</td>
</tr>
<tr>
<td>dsp-error</td>
<td>dsp error (172)</td>
<td>172</td>
</tr>
<tr>
<td>dtl-trans-not-node-id</td>
<td>dtl transit not my node id (160)</td>
<td>160</td>
</tr>
<tr>
<td>facility-not-implemented</td>
<td>facility not implemented (69)</td>
<td>69</td>
</tr>
<tr>
<td>facility-not-subscribed</td>
<td>facility not subscribed (50)</td>
<td>50</td>
</tr>
<tr>
<td>facility-reject</td>
<td>facility rejected (29)</td>
<td>29</td>
</tr>
<tr>
<td>glare</td>
<td>glare (15)</td>
<td>15</td>
</tr>
<tr>
<td>glaring-switch-pri</td>
<td>glaring switch PRI (180)</td>
<td>180</td>
</tr>
<tr>
<td>htspm-oos</td>
<td>HTSPM out of service (129)</td>
<td>129</td>
</tr>
<tr>
<td>ie-missing</td>
<td>mandatory ie missing (96)</td>
<td>96</td>
</tr>
<tr>
<td>ie-not-implemented</td>
<td>ie not implemented (99)</td>
<td>99</td>
</tr>
<tr>
<td>info-class-inconsistent</td>
<td>inconsistency in info and class (62)</td>
<td>62</td>
</tr>
<tr>
<td>interworking</td>
<td>interworking (127)</td>
<td>127</td>
</tr>
<tr>
<td>invalid-call-ref</td>
<td>invalid call ref value (81)</td>
<td>81</td>
</tr>
<tr>
<td>invalid-ie</td>
<td>invalid ie contents (100)</td>
<td>100</td>
</tr>
<tr>
<td>invalid-msg</td>
<td>invalid message (95)</td>
<td>95</td>
</tr>
<tr>
<td>invalid-number</td>
<td>invalid number (28)</td>
<td>28</td>
</tr>
<tr>
<td>invalid-transit-net</td>
<td>invalid transit network (91)</td>
<td>91</td>
</tr>
<tr>
<td>misdialed-trunk-prefix</td>
<td>misdialed trunk prefix (5)</td>
<td>5</td>
</tr>
<tr>
<td>msg-incomp-call-state</td>
<td>message in incomp call state (101)</td>
<td>101</td>
</tr>
<tr>
<td>msg-not-implemented</td>
<td>message type not implemented (97)</td>
<td>97</td>
</tr>
<tr>
<td>msgtype-incompatible</td>
<td>message type not compatible (98)</td>
<td>98</td>
</tr>
<tr>
<td>Message</td>
<td>Description</td>
<td>Code Number</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>-------------------------------------------</td>
<td>-------------</td>
</tr>
<tr>
<td>net-out-of-order</td>
<td>network out of order (38)</td>
<td>38</td>
</tr>
<tr>
<td>next-node-unreachable</td>
<td>next node unreachable (128)</td>
<td>128</td>
</tr>
<tr>
<td>no-answer</td>
<td>no user answer (19)</td>
<td>19</td>
</tr>
<tr>
<td>no-call-suspend</td>
<td>no call suspended (85)</td>
<td>85</td>
</tr>
<tr>
<td>no-channel</td>
<td>channel does not exist (82)</td>
<td>82</td>
</tr>
<tr>
<td>no-circuit</td>
<td>no circuit (34)</td>
<td>34</td>
</tr>
<tr>
<td>no-cug</td>
<td>non existent cug (90)</td>
<td>90</td>
</tr>
<tr>
<td>no-dsp-channel</td>
<td>no dsp channel (170)</td>
<td>170</td>
</tr>
<tr>
<td>no-req-circuit</td>
<td>no requested circuit (44)</td>
<td>44</td>
</tr>
<tr>
<td>no-resource</td>
<td>no resource (47)</td>
<td>47</td>
</tr>
<tr>
<td>no-response</td>
<td>no user response (18)</td>
<td>18</td>
</tr>
<tr>
<td>no-voice-resources</td>
<td>no voice resources available (126)</td>
<td>126</td>
</tr>
<tr>
<td>non-select-user-clear</td>
<td>non selected user clearing (26)</td>
<td>26</td>
</tr>
<tr>
<td>normal-call-clear</td>
<td>normal call clearing (16)</td>
<td>16</td>
</tr>
<tr>
<td>normal-unspecified</td>
<td>normal unspecified (31)</td>
<td>31</td>
</tr>
<tr>
<td>not-in-cug</td>
<td>user not in cug (87)</td>
<td>87</td>
</tr>
<tr>
<td>number-changeed</td>
<td>number changed (22)</td>
<td>22</td>
</tr>
<tr>
<td>param-not-implemented</td>
<td>non implemented param passed on (103)</td>
<td>103</td>
</tr>
<tr>
<td>perm-frame-mode-oos</td>
<td>perm frame mode out of service (39)</td>
<td>39</td>
</tr>
<tr>
<td>perm-frame-mode-oper</td>
<td>perm frame mode operational (40)</td>
<td>40</td>
</tr>
<tr>
<td>precedence-call-block</td>
<td>precedence call blocked (46)</td>
<td>46</td>
</tr>
<tr>
<td>preempt</td>
<td>preemption (8)</td>
<td>8</td>
</tr>
<tr>
<td>preempt-reserved</td>
<td>preemption reserved (9)</td>
<td>9</td>
</tr>
<tr>
<td>protocol-error</td>
<td>protocol error (111)</td>
<td>111</td>
</tr>
<tr>
<td>qos-unavail</td>
<td>qos unavailable (49)</td>
<td>49</td>
</tr>
<tr>
<td>rec-timer-exp</td>
<td>recovery on timer expiry (102)</td>
<td>102</td>
</tr>
<tr>
<td>redirect-to-new-destination</td>
<td>redirect to new destination (23)</td>
<td>23</td>
</tr>
<tr>
<td>req-vpci-vci-unavail</td>
<td>requested vpci vci not available (35)</td>
<td>35</td>
</tr>
</tbody>
</table>
### Message

<table>
<thead>
<tr>
<th>Message</th>
<th>Description</th>
<th>Code Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>send-infotone</td>
<td>send info tone (4)</td>
<td>4</td>
</tr>
<tr>
<td>serv-not-implemented</td>
<td>service not implemented (79)</td>
<td>79</td>
</tr>
<tr>
<td>serv/opt-unavail-unspecified</td>
<td>service or option not available unspecified (63)</td>
<td>63</td>
</tr>
<tr>
<td>stat-enquiry-resp</td>
<td>response to status enquiry (30)</td>
<td>30</td>
</tr>
<tr>
<td>subscriber-absent</td>
<td>subscriber absent (20)</td>
<td>20</td>
</tr>
<tr>
<td>switch-congestion</td>
<td>switch congestion (42)</td>
<td>42</td>
</tr>
<tr>
<td>temp-fail</td>
<td>temporary failure (41)</td>
<td>41</td>
</tr>
<tr>
<td>transit-net-unroutable</td>
<td>no route to transit network (2)</td>
<td>2</td>
</tr>
<tr>
<td>unassigned-number</td>
<td>unassigned number (1)</td>
<td>1</td>
</tr>
<tr>
<td>unknown-param-msg-discard</td>
<td>unrecognized param msg discarded (110)</td>
<td>110</td>
</tr>
<tr>
<td>unsupported-aal-parms</td>
<td>aal parms not supported (93)</td>
<td>93</td>
</tr>
<tr>
<td>user-busy</td>
<td>user busy (17)</td>
<td>17</td>
</tr>
<tr>
<td>vpci-vci-assign-fail</td>
<td>vpci vci assignment failure (36)</td>
<td>36</td>
</tr>
<tr>
<td>vpci-vci-unavail</td>
<td>no vpci vci available (45)</td>
<td>45</td>
</tr>
</tbody>
</table>

### Examples

The following example shows the cause code set to 79:

```
Router(config)# voice lpcor call-block cause 79
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>voice lpcor policy</strong></td>
<td>Creates a LPCOR policy for a resource group.</td>
</tr>
</tbody>
</table>
**voice lpcor custom**

To define the logical partitioning class of restriction (LPCOR) resource groups on the Cisco Unified CME router, use the `voice lpcor custom` command in global configuration mode. To remove the custom resource list, use the `no` form of this command.

```
voice lpcor custom
no voice lpcor custom
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Custom LPCOR resource list is not defined.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enters LPCOR custom configuration mode where you define the name of each of your resource groups using the `index` command. Only one custom resource list is allowed on a Cisco Unified CME router. After you add a resource group to this list, you must then create a LPCOR policy for each resource group that requires call restrictions.

**Examples**

The following example shows a LPCOR configuration with six resource groups:

```
voice lpcor custom
  group 1 sccp_phone_local
  group 2 sip_phone_local
  group 3 analog_phone_local
  group 4 sip_remote
  group 5 sccp_remote
  group 6 isdn_local
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>group (lpcor custom)</code></td>
<td>Adds a LPCOR resource group to the custom resource list.</td>
</tr>
<tr>
<td><code>voice lpcor enable</code></td>
<td>Enables LPCOR functionality on the Cisco Unified CME router.</td>
</tr>
<tr>
<td><code>voice lpcor policy</code></td>
<td>Creates a LPCOR policy for a resource group.</td>
</tr>
</tbody>
</table>
**voice lpcor enable**

To enable logical partitioning class of restriction (LPCOR) functionality on the Cisco Unified CME router, use the **voice lpcor enable** command in global configuration mode. To reset to the default, use the **no** form of this command.

```
voice lpcor enable
no voice lpcor enable
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

LPCOR capability is disabled.

**Command Modes**

Global configuration

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

After using this command, use the **voice lpcor custom** command to create a list of your LPCOR resource groups.

**Examples**

The following example shows a configuration with LPCOR enabled and a custom resource list:

```
voice lpcor enable
voice lpcor custom
  group 1 local_sccp_phone_1
  group 2 local_sip_phone_1
  group 3 local_analog_phone_1
  group 4 local_sccp_phone_2
!
voice lpcor policy local_sccp_phone_1
  accept local_sip_phone_1
  accept local_analog_phone_1
  accept local_sccp_phone_2
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice lpcor custom</td>
<td>Defines the LPCOR resource groups on the Cisco Unified CME router.</td>
</tr>
<tr>
<td>voice lpcor policy</td>
<td>Creates a LPCOR policy for a resource group.</td>
</tr>
</tbody>
</table>
voice lpcor ip-phone mobility

To set the default LPCOR policy for mobility-type phones, use the `voice lpcor ip-phone mobility` command in global configuration mode. To reset to the default, use the `no` form of this command.

```
voice lpcor ip-phone mobility {incoming|outgoing} lpcor-group
no voice lpcor ip-phone mobility {incoming|outgoing}
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>incoming</code></td>
<td>Sets default LPCOR policy for incoming calls.</td>
</tr>
<tr>
<td><code>outgoing</code></td>
<td>Sets default LPCOR policy for outgoing calls.</td>
</tr>
<tr>
<td><code>lpcor-group</code></td>
<td>Name of the LPCOR resource group.</td>
</tr>
</tbody>
</table>

**Command Default**

Default LPCOR policy is not defined for mobility-type phones.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command defines the default LPCOR policy for a mobility-type phone if the LPCOR policy cannot be provisioned using the LPCOR IP-phone subnet table.

**Examples**

The following example shows that the default LPCOR policy for mobility-type phones is set to remote_group1. Any mobility-type phones with a shared IP address from DHCP pool1 are considered local IP phones and are associated with the local_group1 LPCOR policy. Other mobility-type phones without a shared IP address are considered remote IP phones and are associated with the remote_group1 default LPCOR policy.

```
voice lpcor ip-phone subnet incoming
    index 1 local_group1 dhcp-pool pool1
! voice lpcor ip-phone subnet outgoing
    index 1 local_group1 dhcp-pool pool1
! voice lpcor ip-phone mobility incoming remote_group1
voice lpcor ip-phone mobility outgoing remote_group1
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>voice lpcor ip-phone subnet</code></td>
<td>Creates a LPCOR IP-phone subnet table for calls to or from a mobility-type phone.</td>
</tr>
</tbody>
</table>
voice lpcor ip-phone subnet

To create a logical partitioning class of restriction (LPCOR) IP-phone subnet table for calls to or from a mobility-type phone, use the `voice lpcor ip-phone subnet` command in global configuration mode. To reset to the default, use the `no` form of this command.

```
voice lpcor ip-phone subnet {incoming|outgoing}
no voice lpcor ip-phone subnet {incoming|outgoing}
```

**Syntax Description**

- `incoming` Creates IP-phone subnet table for incoming calls from mobility-type phone.
- `outgoing` Creates IP-phone subnet table for outgoing calls from mobility-type phone.

**Command Default**

IP-phone subnet table is not created.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(XA)</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(T).</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command is used for mobility-type phones only, which can include Extension Mobility phones, teleworker remote phones, and Cisco IP Communicator softphones.

This command enters LPCOR IP-phone subnet configuration mode to add LPCOR groups to the incoming or outgoing IP-phone subnet tables. Two IP-phone subnet tables, one for incoming calls and one for outgoing calls, can be defined on each Cisco Unified CME router and can include up to 50 IP address or DHCP pool entries.

A LPCOR policy is dynamically associated with calls to and from a mobility-type phone based on its current IP address or DHCP pool.

**Examples**

The following example shows:

```
voice lpcor ip-phone subnet incoming
index 1 local_g2 10.0.10.23 255.255.255.0 vrf vrf-group2
index 2 remote_g2 171.19.0.0 255.255.0.0
index 3 local_g1 dhcp-pool pool1

voice lpcor ip-phone subnet outgoing
index 1 local_g4 10.1.10.23 255.255.255.0 vrf vrf-group2
index 2 remote_g4 171.19.0.0 255.255.0.0
index 3 local_g5 dhcp-pool pool1
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>index (ip-phone)</td>
<td>Adds a LPCOR group to the IP-phone subnet table.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------------------</td>
<td>------------------------------------------------------------------</td>
</tr>
<tr>
<td><code>lpcor type</code></td>
<td>Specifies the LPCOR type for an IP phone.</td>
</tr>
<tr>
<td><code>voice lpcor ip-phone mobility</code></td>
<td>Sets the default LPCOR policy for mobility-type phones.</td>
</tr>
</tbody>
</table>
voice lpcor ip-trunk subnet incoming

To create a logical partitioning class of restriction (LPCOR) IP-trunk subnet table for incoming calls from a VoIP trunk, use the `voice lpcor ip-trunk subnet incoming` command in global configuration mode. To reset to the default, use the `no` form of this command.

`voice lpcor ip-trunk subnet incoming
no voice lpcor ip-trunk subnet incoming`

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

IP-trunk subnet table is not created.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enters LPCOR IP-trunk subnet configuration mode to add LPCOR groups to the IP-trunk subnet table. One IP-trunk subnet table, containing up to 50 index entries, can be defined on each Cisco Unified CME router for incoming calls from H.323 or SIP trunks.

Incoming VoIP trunk calls are associated with a LPCOR policy by matching the IP address or hostname in the IP-trunk subnet table first. If the IP address or hostname is not found in the table, the LPCOR policy specified with the `lpcor incoming` command in voice service configuration mode is applied.

**Examples**

The following example shows three resource groups are included in the IP-trunk subnet table:

```
voice lpcor ip-trunk subnet incoming
index 1 h323_group1 172.19.33.0 255.255.255.0
index 2 sip_group1 172.19.22.0 255.255.255.0
index 3 sip_group2 hostname sipexample
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>index (lpcor ip-trunk)</td>
<td>Adds a LPCOR resource group to the IP trunk subnet table.</td>
</tr>
<tr>
<td>lpcor incoming</td>
<td>Associates a LPCOR resource-group policy with an incoming call.</td>
</tr>
<tr>
<td>voice lpcor ip-phone subnet</td>
<td>Creates a LPCOR IP-phone subnet table for calls to or from a mobility-type phone.</td>
</tr>
</tbody>
</table>
voice lpcor policy

To create a logical partitioning class of restriction (LPCOR) policy for a resource group, use the `voice lpcor policy` command in global configuration mode. To reset to the default, use the `no` form of this command.

```
voice lpcor policy lpcor-group
no voice lpcor policy lpcor-group
```

**Syntax Description**

- `lpcor-group` Name of the LPCOR resource group.

**Command Default**

LPCOR policy is not defined.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

You can define one policy for each LPCOR resource group. The policy defines the other resource groups from which this resource group can accept calls. You must first name the policy by including it in the custom resource list using the `voice lpcor custom` command.

If you do not explicitly include any resource groups in the policy by using the `accept` command, that policy blocks all incoming calls that are associated with any LPCOR policy other than its own.

If a LPCOR policy is not defined for a target destination, the target can accept incoming calls from any resource group.

**Examples**

The following examples show a LPCOR configuration with four resource groups:

```
voice lpcor custom
index 1 siptrunk
index 2 h323trunk
index 3 pstn
index 4 voicemail
!
```

The LPCOR policy for h323trunk accepts calls from the voicemail group and rejects calls from the siptrunk and pstn groups:

```
voice lpcor policy h323trunk
  accept voicemail
!
```

The LPCOR policy for pstn blocks calls from the siptrunk, h323trunk, and voicemail groups:

```
voice lpcor policy pstn
!
```

The LPCOR policy for voicemail accepts calls from the siptrunk, h323trunk, and pstn groups:
The siptrunk group does not have a LPCOR policy defined so it can accept calls from any of the other resource groups.

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>accept</td>
<td>Allows a LPCOR resource group to accept incoming calls from another resource group.</td>
</tr>
<tr>
<td>show voice lpcor policy</td>
<td>Displays the LPCOR policy for the specified resource group.</td>
</tr>
<tr>
<td>voice lpcor custom</td>
<td>Defines the LPCOR resource groups on the Cisco Unified CME router.</td>
</tr>
</tbody>
</table>
voice mlpp

To enter MLPP configuration mode to enable MLPP service, use the voice service command in global configuration mode. To disable MLPP service, use the no form of this command.

\texttt{voice mlpp}

\texttt{no voice mlpp}

**Syntax Description**

This command has no keywords or arguments.

**Command Default**

No default behavior or values.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Products</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

Voice-mlpp configuration mode is used for the gateway globally.

**Examples**

The following example shows how to enter voice-mlpp configuration mode:

\texttt{Router(config)# voice mlpp}

\texttt{Router(config-voice-mlpp)# access-digit}

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>access-digit</td>
<td>Defines the access digit that phone users dial to request a precedence call.</td>
</tr>
<tr>
<td>mlpp preemption</td>
<td>Enables calls on an SCCP phone or analog FXS port to be preempted.</td>
</tr>
<tr>
<td>preemption trunkgroup</td>
<td>Enables preemption capabilities on a trunk group.</td>
</tr>
</tbody>
</table>
voice moh-group

To enter voice-moh-group configuration mode and set up music on hold (MOH) group parameters, use the `voice moh-group` command in global configuration mode. To remove the music on hold (MOH) group parameters from the configuration for SCCP IP phones, use the `no` form of this command.

`voice moh-group moh-group tag`
no voice moh-group tag

**Syntax Description**

| tag | Specifies a moh-group number tag (1-5) to be used for music on hold group parameters. |

**Command Default**

No voice-moh-group is enabled.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0 Cisco Unified SRST 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0 Cisco Unified SRST 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enters the voice-moh-group configuration mode for configuring music on hold (MOH) group parameters for SCCP IP phones in Cisco Unified CME or in Cisco Unified SRST.

**Examples**

The following example shows how to enter voice-moh-group configuration mode for configuring a moh group in Cisco Unified CME. This example also includes the command to configure a music on hold (MOH) flash file for this voice-moh-group.

```
Router(config)# voice-moh-group 1
Router(config-voice-moh-group)# moh minuet.wav
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>moh</td>
<td>Enables music on hold from a flash audio feed.</td>
</tr>
<tr>
<td>multicast moh</td>
<td>Enables multicast of the music-on-hold audio stream.</td>
</tr>
<tr>
<td>extension-range</td>
<td>Defines extension range for a clients calling a voice-moh-group.</td>
</tr>
</tbody>
</table>
voice register dialplan

To enter voice register dialplan configuration mode to define a dial plan for SIP phones, use the **voice register dialplan** command in global configuration mode. To remove the dialplan, use the **no** form of this command.

**voice register dialplan** *dialplan-tag*
**no voice register dialplan** *dialplan-tag*

**Syntax Description**
- **dialplan-tag** Number that identifies the dial plan. Range: 1 to 24.

**Command Default**
No dial plan is defined.

**Command Modes**
Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>This command was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
A dial plan allows a SIP phone to determine when enough digits are collected for call processing to take place. You define a dial plan using this command and then apply the dial plan to a SIP phone by using the **dialplan** command.

Dial plans allow SIP phones to perform pattern recognition as user input is collected. After a defined pattern is recognized, a SIP INVITE message is automatically sent to Cisco Unified CME and the user does not have to press the Dial key or wait for the interdigit timeout.

This command creates a dial plan file that is downloaded to the phone when the phone is reset or restarted.

**Examples**
The following example shows how to create dial plan 10 for a Cisco Unified IP Phone 7905:

```bash
Router(config)# voice register dialplan 10
Router(config-register-dialplan)# type 7905-7912
Router(config-register-dialplan)# pattern 52...
Router(config-register-dialplan)# pattern 91.......
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>dialplan</strong> Assigns a dial plan to a SIP phone.</td>
</tr>
<tr>
<td><strong>filename</strong> Specifies a custom XML configuration file that contains the dial patterns to use for a SIP dial plan.</td>
</tr>
<tr>
<td><strong>pattern</strong> (voice register dialplan) Defines a dial pattern for a SIP dial plan.</td>
</tr>
<tr>
<td><strong>show voice register dialplan</strong> Displays all configuration information for a specific SIP dial plan.</td>
</tr>
<tr>
<td>type (voice register dialplan)</td>
</tr>
</tbody>
</table>
voice register dn

To enter voice register dn configuration mode to define an extension for a phone line, intercom line, voice-mail port, or a message-waiting indicator (MWI), use the voice register dn command in global configuration mode. To remove the directory number, use the no form of this command.

voice register dn  dn-tag
no  voice register dn  dn-tag

Syntax Description

| dn-tag | Unique sequence number that identifies a particular directory number during configuration tasks. Range is 1 to 150, or the maximum defined by the max-dn command. |

Command Default

Directory number is not defined.

Command Modes

Global configuration (config)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 and Cisco SIP SRST 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

Usage Guidelines

Use this command to create directory numbers for SIP IP phones directly connected in Cisco Unified CME. In voice register dn configuration mode, you assign an extension number by using the number command, a name to appear in the local directory by using the name command, and other provisioning parameters by using various commands.

Before using this command, set the maximum number of directory numbers to appear in your system by using the max-dn command in voice register global configuration mode.

Note

This command can also be used for Cisco SIP SRST.

Note

The name or label associated with a directory number configured under voice register dn configuration mode cannot contain special characters such as quotes ("), angle brackets (<, >), ampersand (&), and percentage (%).

Examples

The following example shows how to enter voice register dn configuration mode for directory number 4 and forward calls to extension 8888 when extension 1001 does not answer:

Router(config)# voice register dn 4
Router(config-register-dn)# number 1001
Router(config-register-dn)# call-forward phone noan 8888
Router(config-register-dn)# call-forward b2bua all 5454
Router(config-register-dn)# call-forward b2bua busy 5705
Router(config-register-dn)# call-forward b2bua mbox 5550
Router(config-register-dn)# call-forward b2bua noan 5050 timeout 20
Router(config-register-dn)# after-hour exempt

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>max-dn (voice register global)</strong></td>
<td>Sets the maximum number of SIP phone directory numbers (extensions) supported by a Cisco CME router.</td>
</tr>
<tr>
<td><strong>mode (voice register global)</strong></td>
<td>Enables the mode for provisioning SIP phones in a Cisco CallManager Express (Cisco CME) system.</td>
</tr>
<tr>
<td><strong>number (voice register pool)</strong></td>
<td>Configures a valid number for a SIP phone.</td>
</tr>
</tbody>
</table>
voice register global

To enter voice register global configuration mode in order to set global parameters for all supported Cisco SIP IP phones in a Cisco Unified CME or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) environment, use the voice register global command in global configuration mode. To automatically remove the existing DNs, pools, and global dialplan patterns, use the no form of this command.

**voice register global**

**no voice register global**

### Syntax Description

This command has no arguments or keywords.

### Command Default

There are no system-level parameters configured for SIP IP phones.

### Command Modes

Global configuration (config)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.0(1)XA</td>
<td>Cisco SIP SRST 8.0</td>
<td>This command was updated to display the signaling transport protocol.</td>
</tr>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1 Cisco Unified SRST 8.1</td>
<td>The no form of the command was modified.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

**Cisco Unified CME**

Use this command to set provisioning parameters for all supported SIP phones in a Cisco Unified CME system.

**Cisco Unified SIP SRST**

Use this command to set provisioning parameters for multiple pools; that is, all supported Cisco SIP IP phones in a SIP SRST environment.

Cisco Unified CME 8.1 enhances the no form of voice register global command. The no voice register global command clears global configuration along with pools and DN configuration and also removes the configurations for voice register template, voice register dialplan, and voice register session-server. A confirmation is sought before the cleanup is made.

In Cisco Unified SRST 8.1 and later versions, the no voice register global command removes pools and DNs along with the global configuration.

### Note

The name or label associated with a directory number configured under voice register global configuration mode cannot contain special characters such as quotes ("), angle brackets (<, >), ampersand (&), and percentage (%).
Examples

Cisco Unified CME

The following is a partial sample output from the `show voice register global` command. All of the parameters listed were set under voice register global configuration mode:

```
Router# show voice register global
CONFIG [Version=4.0(0)]
------------------------
Version 4.0(0)
Mode is cme
Max-pool is 48
Max-dn is 48
Source-address is 10.0.2.4 port 5060
Load 7960-40 is POS3-07-4-07
Time-format is 12
Date-format is M/D/Y
Time-zone is 5
Hold-alert is disabled
Mwi stutter is disabled
Mwi registration for full E.164 is disabled
Dst auto adjust is enabled
   start at Apr week 1 day Sun time 02:00
   stop at Oct week 8 day Sun time 02:00
```

Examples

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from no voice register global command:

```
Router(config)# no voice register global
This will remove all the existing DNs, Pools, Templates, Dialplan-Patterns, Dialplans and Feature Servers on the system.
Are you sure you want to proceed? Yes/No? [no]:
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow connections sip to sip</td>
<td>Allows connections between SIP endpoints in a Cisco multiservice IP-to-IP gateway.</td>
</tr>
<tr>
<td>application (voice register global)</td>
<td>Selects the session-level application for all dial peers associated with SIP phones.</td>
</tr>
<tr>
<td>mode (voice register global)</td>
<td>Enables the mode for provisioning SIP phones in a Cisco Unified system.</td>
</tr>
</tbody>
</table>
## voice register pool

To enter voice register pool configuration mode and create a pool configuration for a SIP IP phone in Cisco Unified CME or for a set of SIP phones in Cisco Unified SIP SRST, use the `voice register pool` command in global configuration mode. To remove the pool configuration, use the `no` form of this command.

```plaintext
voice register pool  pool-tag
no voice register pool  pool-tag
```

### Syntax Description

<table>
<thead>
<tr>
<th>pool-tag</th>
<th>Unique number assigned to the pool. Range is 1 to 100.</th>
</tr>
</thead>
</table>

**Note**

For Cisco Unified CME systems, the upper limit for this argument is defined by the `max-pool` command.

### Command Default

There is no pool configured.

### Command Modes

Global configuration (config)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was added to Cisco CME.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

**Cisco Unified CME**

Use this command to set phone-specific parameters for SIP phones in a Cisco Unified CME system. Before using this command, enable the `mode cme` command and set the maximum number of SIP phones supported in your system by using the `max-pool` command.

**Cisco Unified SIP SRST**

Use this command to enable user control on which registrations are to be accepted or rejected by a SIP SRST device. The voice register pool command mode can be used for specialized functions and to restrict registrations on the basis of MAC, IP subnet, and number range parameters.

### Examples

#### Cisco Unified CME

The following example shows how to enter voice register pool configuration mode and forward calls to extension 9999 when extension 2001 is busy:

```plaintext
Router(config)# voice register pool 10
Router(config-register-pool)# type 7960
Router(config-register-pool)# number 1 2001
Router(config-register-pool)# call-forward busy 9999 mailbox 1234
```
Cisco Unified SIP SRST

The following partial sample output from the `show running-config` command shows that several voice register pool commands are configured within voice register pool 3:

```
voice register pool 3
 id network 10.2.161.0 mask 255.255.255.0
 number 1 95... preference 1
 cor outgoing call95 1 95011
 max registrations 5
 voice-class codec 1
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>max-pool (voice register global)</code></td>
<td>Sets the maximum number of SIP phones that are supported by a Cisco Unified CME system.</td>
</tr>
<tr>
<td><code>mode (voice register global)</code></td>
<td>Enables the mode for provisioning SIP phones in a Cisco Unified CME system.</td>
</tr>
<tr>
<td><code>number (voice register pool)</code></td>
<td>Configures a valid number for a SIP phone.</td>
</tr>
<tr>
<td><code>type (voice register pool)</code></td>
<td>Defines a Cisco IP phone type.</td>
</tr>
</tbody>
</table>
**voice register pool-type**

To enter voice register pool-type configuration mode and add a new Cisco Unified SIP IP phone to Cisco Unified CME, use the `voice register pool-type` command in global configuration mode. To return to the default, use the `no` form of this command.

```
voice register pool-type [device-reference supported phone-type]
[device-name name {device-type type}]
[addons max-addons]
[num-lines max-lines]
[transport-type {udp | tcp}]
[gsm-handoff][telnet][phoneload][xml-config xml-config value]
```

**novoiceregisterpool-type** [device-reference supported phone-type]
[device-name name {device-type type}]
[addons max-addons]
[num-lines max-lines]
[transport-type {udp | tcp}]
[gsm-handoff][telnet][phoneload][xml-config xml-config value]

### Syntax Description

<table>
<thead>
<tr>
<th>Option</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>device-reference</strong></td>
<td>Supported phone type for a new Cisco Unified SIP IP phone.</td>
</tr>
<tr>
<td><strong>supported phone-type</strong></td>
<td>For a list of the names, types, and corresponding properties of supported phones, see the below table.</td>
</tr>
<tr>
<td><strong>device-name</strong></td>
<td>Name of the device.</td>
</tr>
<tr>
<td><strong>type</strong></td>
<td>Phone type for a new Cisco Unified SIP IP phone.</td>
</tr>
<tr>
<td><strong>addons</strong></td>
<td>Maximum number of add-on modules supported by the new phone device.</td>
</tr>
<tr>
<td><strong>max-addons</strong></td>
<td>New add-on modules for an existing phone are not supported.</td>
</tr>
<tr>
<td><strong>num-lines</strong></td>
<td>Maximum number of lines supported by the phone. Range is 1 to ???.</td>
</tr>
<tr>
<td><strong>max-lines</strong></td>
<td></td>
</tr>
<tr>
<td>**transport-type {udp</td>
<td>tcp}**</td>
</tr>
<tr>
<td><strong>udp</strong></td>
<td>User Datagram Protocol (UDP) is used.</td>
</tr>
<tr>
<td><strong>tcp</strong></td>
<td>Transmission Control Protocol (TCP) is used.</td>
</tr>
<tr>
<td><strong>gsm-handoff</strong></td>
<td>Enables phone support for Global System for Mobile Communications (GSM) handoff.</td>
</tr>
<tr>
<td><strong>telnet</strong></td>
<td>Enables phone support for Telnet access.</td>
</tr>
<tr>
<td><strong>phoneload</strong></td>
<td>Enables support for phone loads.</td>
</tr>
<tr>
<td><strong>xml-config xml-config value</strong></td>
<td>Configuration XML file for the phone.</td>
</tr>
</tbody>
</table>

**Configuration Example**

```bash
voiceregisterpool-type
```
xml-config xml-tag value  (Optional) Defines the phone-specific XML tags to be used in the configuration file.
  • xml-tag—Phone-specific XML tag.
  • value—Value of the XML tag.

Command Default  The Cisco Unified 7965 SIP IP phone model is used as the default phone device reference.

Command Modes  Global configuration (config)

Command History  

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(3)M</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

Usage Guidelines  When the device-reference keyword is not configured and phone properties are not explicitly configured, the Cisco Unified 7965 SIP IP phone model is used as the default phone device reference and its corresponding phone properties are inherited by the new Cisco Unified SIP IP phone.

Table 1 lists the names, types, and corresponding properties of supported phones that can be entered as values for the device-reference keyword. The description string configured with the device-name keyword is displayed as a help string when the new phone type is listed with the supported device types for the type (voice register pool) command.

The description string configured with the device-name keyword is displayed as a help string when the new phone type is listed with the supported device types for the type (voice register pool) command.

With respect to the transport-type keyword, most Cisco Unified SIP IP phones use UDP as the default transport protocol to connect to Cisco Unified CME while CiscoMobile-iOS and Jabber-Android use TCP. These configurations can be changed using the session-transport {udp | tcp} command in voice register pool or voice register template configuration mode.

Example  The following example shows how to inherit the existing features of its phone model (9951) using the Fast-Track configuration approach. Phone model “9951” is used as the value of the reference-pooltype keyword. The maxNumCalls XML tag defines “3” as the maximum number of calls allowed per line while the busyTrigger XML tag defines "3" as the number of calls that triggers call forward busy per line on the phone.

voice register pool-type 9900
reference-pooltype 9951
device-name “SIP Phone 9900 addon module”
um-lines 24
addons 3
transport tcp
telnet
gsm-handoff
phoneload
xml-config maxNumCalls 3
xml-config busyTrigger 3
voice register pool 10
type 9900 addon 1 CKEM 2 CKEM 3 CKEM
id mac 1234.4567.7891
voice register global
mode cme
load 9900 POS3-06-00

The following example shows how to inherit the existing features of its parent phone type (Cisco Unified 6921 SIP IP phone) using the Fast-Track configuration approach. Parent phone model “6921” is used as the value of the reference-type keyword.

voice register pool-type 6922
reference-pooltype 6921
device-name “SIP Phone 6922”

voice register pool 11
type 6922
id mac 1234.4567.7890

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>session-transport</td>
<td>Specifies the transport layer protocol that a SIP phone uses to connect to Cisco Unified CME.</td>
</tr>
<tr>
<td>type (voice register pool)</td>
<td>Defines a phone type for a SIP phone.</td>
</tr>
</tbody>
</table>
voice register session-server

To enter voice register session-server configuration mode to enable and configure a session manager in Cisco Unified CME for an external feature server, use the `voice register session-server` command in global configuration mode. To remove a session manager, use the `no` form of this command.

```
voice register session-server session-server-tag
no voice register session-server session-server-tag
```

**Syntax Description**

- `session-server-tag` Explicitly identifies a session manager for configuration tasks. Range is 1 to the maximum number of Cisco IP phones supported by a Cisco Unified CME router as set by the `max-phones` command in telephony-service configuration mode.

**Command Default**

No session manager is created.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XW2</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>
| 15.2(1)T          | Cisco Unified CME 8.8       | This command was modified to allow the maximum number of Cisco IP phones supported by a Cisco Unified CME router to be identified as the session manager.

**Usage Guidelines**

Provisioning and configuration information in the Cisco Unified Contact Center Express (Cisco Unified CCX) is automatically provided to Cisco Unified CME. Use the `voice register session-server` command to enter voice register session-server configuration mode and reconfigure and enable a session manager for Cisco Unified CCX on a Cisco Carrier Routing System when the configuration from Cisco Unified CCX is deleted or must be modified.

A single Cisco Unified CME can support multiple session managers.

After creating one or more session managers, use the `session-server` command in voice register pool configuration mode to identify a session manager for controlling a route point.

After creating one or more session managers, use the `session-server` command in ephone-dn configuration mode to specify session managers for monitoring a directory numbers.

**Examples**

The following is a partial output from the `show running-configuration` command, showing the configuration for the session manager, session-server 1:

```
!```
voice register session-server 1
   keepalive 300
   register-id SB-SJ3-UCCX1_1164774025000

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>session-server</td>
<td>Specifies a session server to manage and monitor registration and subscription messages for an external feature server.</td>
</tr>
</tbody>
</table>
voice register template

To enter voice register template configuration mode and define a template of common parameters for SIP phones, use the `voice register template` command in global configuration mode. To remove a template, use the `no` form of this command.

```
voice register template template-tag
no voice register template template-tag
```

**Syntax Description**

- `template-tag`: Declares a template tag. Range: 1 to 10.

**Command Default**

No default behavior or values

**Command Modes**

- Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(11)XJ</td>
<td>Cisco Unified CME 4.1</td>
<td>The maximum number of templates was increased from 5 to 10.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Cisco Unified CME 4.1</td>
<td>The increase in the template number was integrated into Cisco IOS Release 12.4(15)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Up to ten different templates can be defined and applied to SIP phones. You create the template with this command and then apply the template to a phone by using the `template` command in voice register pool configuration mode.

**Examples**

In the following example, template 1 is created by using the `voice register template` command.

```
Router(config)# voice register template 1
Router(config-register-temp)# anonymous block
Router(config-register-temp)# caller-id block
Router(config-register-temp)# voicemail 5001 timeout 15
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>anonymous block (voice register template)</code></td>
<td>Enables anonymous call blocking in a SIP phone template.</td>
</tr>
<tr>
<td><code>caller-id block (voice register template)</code></td>
<td>Enables caller-ID blocking for outbound calls from a specific SIP phone.</td>
</tr>
<tr>
<td><code>template (voice register pool)</code></td>
<td>Applies a template to a SIP phone.</td>
</tr>
<tr>
<td><code>voicemail (voice register template)</code></td>
<td>Defines the extension that calls are forwarded to when an extension does not answer.</td>
</tr>
</tbody>
</table>
voice user-profile

To enter voice user-profile configuration mode and create a user profile for downloading by Extension Mobility for a particular individual phone user, use the `voice user-profile` command in global configuration mode. To delete an logout profile, use the `no` form of this command.

```
voice user-profile profile-tag
no voice user-profile profile-tag
```

**Syntax Description**

| profile-tag | Unique number that identifies this profile during configuration tasks. Range: 1 to three times the maximum number supported phones, where maximum is platform and version dependent and defined by the `max-ephone` command. |

**Command Default**

No user profile is created.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)XW</td>
<td>Cisco Unified CME 4.2</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XY</td>
<td>Cisco Unified CME 4.2(1)</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(15)XZ</td>
<td>Cisco Unified CME 4.3</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(20)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(20)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to create a user profile containing a user’s own personal settings, such as directory number, speed-dial lists, and services, for downloading to the IP phone when the individual phone user logs into a Cisco Unified IP phone that is registered in Cisco Unified CME and enabled for Extension Mobility.

Type `?` in voice profile configuration mode to see the commands that are available in this mode and that can be included in a user profile. The following example shows a list of commands that were available in voice user-profile configuration mode at the time that this document was written:

```
Router(config-user-profile)#?
Logout profile configuration commands:
  name    Define username and password for Extension Mobility.
  number  Create ip-phone line definition
  pin     Reset all phones associated with the profile being configured
  speed-dial Define ip-phone speed-dial number
```

All directory numbers to be included in a default logout profile or voice-user profile must already be configured in Cisco Unified CME.

After creating or modifying a profile, use the `reset (voice user-profile)` command to reset all phones on which this profile is downloaded to propagate the modifications.
Examples

The following example shows the configuration for a voice-user profile to be downloaded when a phone user logs into a Cisco Unified IP phone that is enabled for Extension Mobility. The lines and speed-dial buttons in this profile that are configured on a phone after the user logs in depend on the phone type. For example, if the user logs into a Cisco Unified IP Phone 7970, all buttons are configured according to voice-user profile 1. However, if the phone user logs into a Cisco Unified IP Phone 7960, all six lines are mapped to phone buttons and the speed dial is ignored because no button is available for speed dial.

```
pin 12345
user me password pass123
number 2001 type silent-ring
number 2002 type beep-ring
number 2003 type feature-ring
number 2004 type monitor-ring
number 2005,2006 type overlay
number 2007,2008 type cw-overlay
speed-dial 1 3001
speed-dial 2 3002 blf
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>logout-profile</td>
<td>Enables Cisco Unified IP phone for Extension Mobility and assigns a logout profile to this phone.</td>
</tr>
<tr>
<td>reset (voice logout-profile and voice user-profile)</td>
<td>Performs a complete reboot of all IP phones on which a particular logout profile or user profile is downloaded.</td>
</tr>
</tbody>
</table>
voice-class codec (voice register pool)

To assign a previously configured codec selection preference list, use the `voice-class codec` command in voice register pool configuration mode. To remove the codec preference assignment from the voice register pool, use the `no voice-class codec` form of this command.

```plaintext
voice-class codec tag
no voice-class codec
```

**Syntax Description**

- `tag`: Unique number assigned to the voice class. Range is from 1 to 10000. The tag number maps to the tag number created by using the `voice class codec` command in dial-peer configuration mode.

**Command Default**

There is no codec preference assignment in the voice register pool configuration.

**Command Modes**

Voice register pool configuration (config-register-pool)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco SIP SRST 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4 Cisco SIP SRST 3.4</td>
<td>This command was added to Cisco CME.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

During Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified CallManager Express (Cisco Unified CME) registration, a dial peer is created and that dial peer includes codec g729r8 by default. This command allows you to change the automatically selected default codec.

You can assign one voice class to each voice register pool. If you assign another voice class to a pool, the last voice class assigned replaces the previous voice class.

**Note**

The `id` (voice register pool) command is required and must be configured before any other voice register pool commands. The `id` command identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.

**Examples**

The following partial sample output from the `show running-config` command shows that voice register pool 1 has been set up to use the previously configured codec voice class 1:

```plaintext
voice register pool 1
  id mac 0030.94C2.A22A
  preference 5
  cor incoming call 91 1 91011
  translate-outgoing called 1
  proxy 10.2.161.187 preference 1 monitor probe icmp-ping
  alias 1 94... to 91011 preference 8
```
voice-class codec 1

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>codec (voice register pool)</td>
<td>Specifies the codec supported by a single Cisco SIP phone or a VoIP dial peer in a Cisco Unified SIP SRST or a Cisco Unified CME environment.</td>
</tr>
<tr>
<td>id (voice register pool)</td>
<td>Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.</td>
</tr>
<tr>
<td>voice class codec (dial-peer)</td>
<td>Assigns a previously configured codec selection preference list (codec voice class) to a VoIP dial peer.</td>
</tr>
</tbody>
</table>
voice-class mlpp (dial peer)

To assign a Multilevel Precedence and Preemption (MLPP) voice class to a POTS or VoIP dial peer, use the voice-class mlpp command in dial-peer configuration mode. To remove the voice class from the dial peer, use the no form of this command.

```
voice-class mlpp tag
no voice-class mlpp tag
```

**Syntax Description**

- **tag**: Unique number that identifies the voice class. Range: 1 to 10000.

**Command Default**

The dial peer does not use an MLPP voice class.

**Command Modes**

Dial-peer configuration (config-dial-peer)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.0(1)XA</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(1)T</td>
<td>Cisco Unified CME 8.0</td>
<td>This command was integrated into Cisco IOS Release 15.1(1)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The voice class that you assign to the dial peer must first be configured using the voice class mlpp command in global configuration mode.

You can assign one voice class to each dial peer. If you assign another voice class to a dial peer, the last voice class assigned replaces the previous voice class.

**Examples**

The following example shows that VoIP dial peer 36 is assigned MLPP class 2.

```
Router(config)# dial-peer voice 36 voip
Router(config-dial-peer)# voice-class mlpp 2
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>service-domain (voice class)</td>
<td>Sets the service domain name in the MLPP voice class.</td>
</tr>
<tr>
<td>show dial-peer voice</td>
<td>Displays the configuration for all dial peers configured on the router.</td>
</tr>
<tr>
<td>voice class mlpp</td>
<td>Creates an MLPP voice class.</td>
</tr>
</tbody>
</table>
voice-class stun-usage

To configure voice class, enter voice class configuration mode called stun-usage and use the voice-class stun-usage command in global, dial-peer, ephone, ephone template, voice register pool, or voice register pool template configuration mode. To disable the voice class, use the no form of this command.

voice-class stun-usage tag
no voice-class stun-usage tag

Syntax Description

| tag | Unique identifier in the range 1 to 10000.

Command Default

The voice class is not defined.

Command Modes

Global configuration (config)
Dial peer configuration (config-dial-peer)
Ephone configuration (config-ephone)
Ephone template configuration (config-ephone-template)
Voice register pool configuration (config-register-pool)
Voice register pool template configuration (config-register-pool)

Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)T</td>
<td>Cisco Unified CME 7.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>15.1(2)T</td>
<td>Cisco Unified CME 8.1</td>
<td>This command was modified. This command can be enabled in ephone summary, ephone template, voice register pool, or voice register pool template configuration mode.</td>
</tr>
</tbody>
</table>

Usage Guidelines

When the voice-class stun-usage is removed, the same is removed automatically from the dial-peer, ephone, ephone template, voice register pool, or voice register pool template configurations.

Examples

The following example shows how to set the voice class stun-usage tag to 10000:

Router(config)# voice class stun-usage 10000
Router(config-ephone)# voice class stun-usage 10000
Router(config-voice-register-pool)# voice class stun-usage 10000

Related Commands

- stun usage firewall-traversal flowdata: Enables firewall traversal using STUN.
- stun flowdata agent-id: Configures the agent ID.
**voice-gateway system**

To enter voice-gateway configuration mode and create a voice gateway configuration, use the `voice-gateway system` command in global configuration mode. To remove the configuration, use the `no` form of this command.

```
voice-gateway system tag
no voice-gateway system tag
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>tag</code></td>
<td>Unique number that identifies the voice gateway. Range: 1 to 25. There is no default value.</td>
</tr>
</tbody>
</table>

**Command Default**

Gateway configuration is not defined.

**Command Modes**

Global configuration (config)

**Command History**

<table>
<thead>
<tr>
<th>Command History</th>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td></td>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command has been integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enters voice-gateway configuration mode to define the parameters for a voice gateway using the auto-configuration feature. Define a configuration for each Cisco voice gateway whose analog FXS ports you want under the control of this Cisco Unified CME router.

**Examples**

The following example shows a voice gateway configuration:

```
voice-gateway system 1
network-locale FR
type VG224
mac-address 001F.A30F.8331
voice-port 0-23
create cnf-files
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>mac-address</td>
<td>Defines the MAC address of the Cisco voice gateway that downloads its configuration from Cisco Unified CME.</td>
</tr>
<tr>
<td>type</td>
<td>Defines the type of voice gateway to autoconfigure in Cisco Unified CME.</td>
</tr>
<tr>
<td>voice-port</td>
<td>Identifies the analog ports on the voice gateway that register to Cisco Unified CME.</td>
</tr>
</tbody>
</table>
voicemail (telephony-service)

To define the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed, use the voicemail command in telephony-service configuration mode. To disable the Messages button, use the no form of this command.

```
voicemail  phone-number
no  voicemail
```

**Syntax Description**

| phone-number | Phone number that is configured as a speed-dial number for retrieving messages. |

**Command Default**

No phone number is configure and the Messages button is disabled.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)YD</td>
<td>Cisco ITS 1.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(8)T</td>
<td>Cisco ITS 2.0</td>
<td>This command was integrated into Cisco IOS Release 12.2(8)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command configures the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed. The same telephone number is configured for voice messaging for all Cisco IP phones connected to the router.

**Examples**

The following example sets the phone number 914085550100 as the speed-dial number that is dialed to retrieve messages when the Messages button is pressed:

```
Router(config)# telephony-service
Router(config-telephony)# voicemail 914085550100
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>telephony-service</td>
</tr>
<tr>
<td>vm-device-id (ephone)</td>
</tr>
</tbody>
</table>
voicemail (voice register global)

To define the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed, use the `voicemail` command in voice register global configuration mode. To disable the Messages button, use the `no` form of this command.

```
voicemail  phone-number
no  voicemail
```

**Syntax Description**
- `phone-number` Telephone number that is speed-dialed for retrieving messages.

**Command Default**
No phone number is configure and the Messages button is disabled.

**Command Modes**
Voice register global configuration (config-register-global)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
This command configures the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed. The same telephone number is configured for voice messaging for all Cisco IP phones connected to the router.

**Examples**
The following example shows how to set telephone number 914085550100 as the speed-dial number to retrieve messages when the Messages button is pressed:

```
Router(config)# voice register global
Router(config-register-global)# voicemail 914085550100
```

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Provision uniform resource locators (URLs) for feature buttons on Cisco IP phones.</td>
</tr>
<tr>
<td>Defines the extension that calls are forwarded to when an extension does not answer.</td>
</tr>
<tr>
<td>Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.</td>
</tr>
</tbody>
</table>
voicemail (voice register template)

To define the extension that calls are forwarded to when an extension does not answer, use the `voicemail` command in voice register template configuration mode. To disable the voicemail extension, use the `no` form of this command.

```
voicemail  phone-number  timeout  timeout
no  voicemail
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>phone-number</code></td>
<td>Telephone number to which calls are forwarded when an extension does not answer.</td>
</tr>
<tr>
<td><code>timeout</code></td>
<td>Duration that a call can ring with no answer before the call is forwarded to the voicemail extension. Range is 5 to 60000. There is no default value.</td>
</tr>
</tbody>
</table>

**Command Default**

This command has no default behavior or values.

**Command Modes**

Voice register template configuration (config-register-temp)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>Cisco CME 3.4</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command defines the destination extension for voicemail when an extension on a SIP phone does not answer. To apply the template to a SIP phone, use the `template` command in voice register pool configuration mode.

**Examples**

The following example shows how to set telephone number 914085550100 as the number to be dialed to retrieve messages when the Messages button is pressed:

```
Router(config)# voice register template 1
Router(config-register-temp)# voicemail 50100 timeout 15
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>template (voice register pool)</code></td>
<td>Applies a template to a SIP phone.</td>
</tr>
<tr>
<td><code>url (voice register global)</code></td>
<td>Provisions uniform resource locators (URLs) for feature buttons on Cisco IP phones.</td>
</tr>
<tr>
<td><code>voice register global</code></td>
<td>Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.</td>
</tr>
<tr>
<td><code>voice register template</code></td>
<td>Enters voice register template configuration mode and defines a template of common parameters for SIP phones.</td>
</tr>
<tr>
<td><strong>Description</strong></td>
<td><strong>Description</strong></td>
</tr>
<tr>
<td>-----------------</td>
<td>----------------</td>
</tr>
<tr>
<td><strong>voicemail (voice register global)</strong></td>
<td>Defines the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed.</td>
</tr>
</tbody>
</table>
voice-port (voice-gateway)

To identify the analog ports on the voice gateway that register to Cisco Unified CME, use the `voice-port` command in voice-gateway configuration mode. To remove the ports, use the `no` form of this command.

```
voice-port port-range
no voice-port
```

**Syntax Description**

| `port-range` | Individual port number, or range of port numbers, on the voice gateway controlled by Cisco Unified CME. Enter individual port values separated by a comma (,) or enter a range using a hyphen (x-y). There is no default value. |

**Command Default**

No voice ports are supported.

**Command Modes**

Voice-gateway configuration (config-voice-gateway)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(22)YB</td>
<td>Cisco Unified CME 7.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(24)T</td>
<td>Cisco Unified CME 7.1</td>
<td>This command has been integrated into Cisco IOS Release 12.4(24)T.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command sets the total number of analog endpoints on the voice gateway that you intend to register to the Cisco Unified CME router. The Cisco VG202 supports two ports, Cisco VG204 supports four ports, and the Cisco VG224 supports 24 ports, numbered 0 to 23.

**Examples**

The following example shows a configuration for a Cisco VG224 voice gateway with 24 ports:

```
voice-gateway system 1
network-locale FR
type VG224
mac-address 001F.A30F.8331
voice-port 0-23
create cnf-files
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>network-locale (voice-gateway)</code></td>
<td>Selects a geographically specific set of tones and cadences for the voice gateway’s analog endpoints that register to Cisco Unified CME.</td>
</tr>
<tr>
<td><code>type (voice-gateway)</code></td>
<td>Defines the type of voice gateway to autoconfigure in Cisco Unified CME.</td>
</tr>
</tbody>
</table>
**vpn-gateway**

To enter vpn-gateway url, use the vpn-gateway command in vpn-group configuration mode. To disable the vpn-gateway configuration, use the *no* form of this command.

```
vpn-gateway number [url]
no vpn-group
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>number</strong></td>
<td>Vpn-gateway numbers. Range: 1-3.</td>
</tr>
<tr>
<td><strong>url</strong></td>
<td>VPN concentrator address url as https://&lt;IP&gt;/policy.</td>
</tr>
</tbody>
</table>

**Command Default**

vpn-gateway is not configured.

**Command Modes**

Vpn-group configuration (conf-vpn-group).

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to enter vpn-gateway urls. You can define up to 3 vpn-gateways urls for SSLVPN phones.

**Examples**

The following example shows vpn-gateway 1 configured for vpn-group 1:

```
Router# show run
!
!
!
voice-card 3
dspfarm
dsp services dspfarm
!
!
voice service voip
ip address trusted list
ipv4 20.20.20.1
vpn-group 1
vpn-gateway 1 https://9.10.60.254/SSLVPNphone
vpn-trustpoint 1 trustpoint cme_cert root
vpn-hash-algorithm sha-1
vpn-profile 1
host-id-check disable
sip
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>vpn-group</td>
<td>Specifies a vpn-group.</td>
</tr>
<tr>
<td>vpn-trustpoint</td>
<td>Specifies a vpn-gateway trustpoint.</td>
</tr>
</tbody>
</table>
**vpn-group**

To enter vpn-group mode, use the `vpn-group` command in voice service voip configuration mode. To delete all configurations associated with a vpn-group, use the `no` form of this command.

```
vpn-group tag
no vpn-group
```

### Syntax Description

| Tag | Vpn-group tag number. Range: 1-2 |

### Command Default

Vpn-group is not configured.

### Command Modes

Voice service voip (conf-voi-serv)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use this command to create vpn-groups. A vpn-group is a redundancy ordered list of up to 3 vpn-gateways that an SSL VPN client on a phone can connect to. You can create 2 vpn-groups.

### Examples

The following example shows vpn-group 1:

```
Router# show run
!
!
voice-card 3
dspfarm
dsp services dspfarm
!
!
voice service voip
ip address trusted list
ipv4 20.20.20.1
vpn-group 1
vpn-gateway 1 https://9.10.60.254/SSLVPNphone
vpn-trustpoint 1 trustpoint cme_cert root
vpn-hash-algorithm sha-1
vpn-profile 1
host-id-check disable
sip
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>vpn-gateway</td>
<td>Specifies a vpn-gateway URL.</td>
</tr>
<tr>
<td>vpn-trustpoint</td>
<td>Specifies a vpn-gateway trustpoint.</td>
</tr>
<tr>
<td>vpn-hash-algorithm</td>
<td>Specifies vpn hash encryption for the trustpoints.</td>
</tr>
</tbody>
</table>
**vpn-hash-algorithm**

To specify the algorithm to hash the VPN certificate provided in the configuration file downloaded to the phone, use the `vpn-hash-algorithm` command in `vpn-group` configuration mode. To disable `vpn-hash-encryption`, use the `no` form of this command.

```
vpn-hash-algorithm sha-1
no vpn-hash-algorithm
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>sha-1</code></td>
<td>Encryption algorithm.</td>
</tr>
</tbody>
</table>

**Command Default**

`vpn-hash-algorithm` is not configured

**Command Modes**

`Vpn-group configuration (conf-vpn-group)`

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command to specify the algorithm to hash the VPN certificate provided in the configuration file downloaded to the phone.

**Examples**

The following example shows `vpn-hash-algorithm` configured in `vpn-group 1`:

```
Router# show run
!
!
voice-card 3
dspfarm
dsp services dspfarm
!
!
voice service voip
ip address trusted list
ipv4 20.20.20.1
vpn-group 1
vpn-gateway 1 https://9.10.60.254/SSLVPNphone
vpn-trustpoint 1 trustpoint cme_cert root
vpn-hash-algorithm sha-1
vpn-profile 1
host-id-check disable
sip
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>vpn-group</code></td>
<td>Specifies a <code>vpn-group</code>.</td>
</tr>
<tr>
<td><code>vpn-trustpoint</code></td>
<td>Specifies a <code>vpn-gateway trustpoint</code>.</td>
</tr>
</tbody>
</table>
vpn-profile

To enter vpn-profile mode to configure vpn-profiles in Cisco Unified CME, use the **vpn-profile** command in voice service voip configuration mode. To remove the entire vpn-profile configuration, use the no form of this command.

```
vpn-profile tag
no vpn-profile
```

**Syntax Description**

**Command Default**
No vpn-profile is configured.

**Command Modes**
Voice service voip (conf-voi-serv)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**
Use this command to create one or more vpn-profiles on Cisco Unified CME. You can create 6 vpn-profiles.

**Examples**

The following example shows 3 vpn-profiles configured:

```
Router# show run
!
!
voice service voip
  ip address trusted list
  ipv4 20.20.20.1
  vpn-group 1
    vpn-gateway 1 https://9.10.60.254/SSLVPNphone
    vpn-trustpoint 1 trustpoint cme_cert root
    vpn-hash-algorithm sha-1
  vpn-profile 1
    keepalive 50
    host-id-check disable
    auto-network-detect enable
  vpn-profile 2
    mtu 1300
    password-persistent enable
    host-id-check enable
  vpn-profile 4
    fail-connect-time 50
  sip
    !
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice-service-voip</td>
<td>Enters voice-service configuration mode for Voice Over IP (VoIP) encapsulation.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-----------</td>
<td>-------------------------------------------</td>
</tr>
<tr>
<td>vpn-group</td>
<td>Enters vpn-group configuration mode.</td>
</tr>
</tbody>
</table>
vpn-trustpoint

To configure a vpn gateway trustpoint, use the vpn-trustpoint command in vpn-group configuration mode. To disable a vpn-gateway trustpoint associated with a vpn-group, use the no form of this command.

vpn-trustpoint number [{raw|trustpoint}] word [{leaf|root}]
no vpn-trustpoint

Syntax Description

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>number</td>
<td>Number of allowed trustpoints. Range is from 1 to 10.</td>
</tr>
<tr>
<td>raw</td>
<td>(Optional) Allows to enter VPN Gateway Trustpoint in raw form.</td>
</tr>
<tr>
<td>trustpoint</td>
<td>(Optional) Allows to enter VPN Gateway Trustpoint in IOS format.</td>
</tr>
<tr>
<td>leaf</td>
<td>Get the 1st leaf cert of the Trustpoint.</td>
</tr>
<tr>
<td>root</td>
<td>Get the root cert of the Trustpoint.</td>
</tr>
</tbody>
</table>

Command Default

Vpn-group (conf-vpn-group)

Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.1(3)T</td>
<td>Cisco Unified CME 8.5</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

Usage Guidelines

Use this command to create vpn-trustpoints for a vpn-group. You can configure as many as 10 vpn-trustpoints in a vpn-group. All vpn trustpoints must be entered in either raw or trustpoint (IOS) format.

Examples

The following example shows vpn-trustpoint 1 entered in trustpoint (IOS) format:

```
Router# show run
!
!
!
!
voice service voip
ip address trusted list
ipv4 20.20.20.1
vpn-group 1
vpn-gateway 1 https://9.10.60.254/SSLVPNphone
vpn-trustpoint 1 trustpoint cme_cert root
vpn-hash-algorithm sha-1
vpn-profile 1
host-id-check disable
sip
```

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>vpn-grouptrustpoint</td>
<td>Defines a vpn-group.</td>
</tr>
</tbody>
</table>
Cisco Unified CME Commands: W

- web admin customer, on page 1468
- web admin system, on page 1470
- web customize load, on page 1472
web admin customer

To define a username and password for a Cisco Unified CME customer administrator, use the `web admin customer` command in telephony-service configuration mode. To disable a customer administrator login, use the `no` form of this command.

`web admin customer name username {password string|secret {0|5} string}
no web admin customer`

**Syntax Description**

<table>
<thead>
<tr>
<th><strong>name username</strong></th>
<th>Username for the customer administrator. String can contain a maximum of 28 alphanumeric characters. Default is Customer.</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>password string</strong></td>
<td>Character string for login authentication, which will be stored in the running configuration as plain text. String can contain a maximum of 28 alphanumeric characters. Default is no password.</td>
</tr>
</tbody>
</table>
| **secret {0 | 5 } string** | Character string for login authentication, which will be stored in the running configuration as encrypted using Message Digest 5 (MD5). The digit 0 or 5 specifies whether the displayed string that follows is encrypted:
  • 0—Password that follows is not encrypted.
  • 5—Password that follows is encrypted. |

**Command Default**

Default is a customer administrator with username Customer and no password.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th><strong>Cisco IOS Release</strong></th>
<th><strong>Cisco Product</strong></th>
<th><strong>Modification</strong></th>
</tr>
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<tbody>
<tr>
<td>12.2(11)YT</td>
<td>Cisco ITS 2.1</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command is deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables customer administrator access for the Cisco Unified CME graphical user interface (GUI).

**Examples**

The following example defines a customer administrator named user22 whose password is pw567890:

```bash
Router(config)# telephony-service
Router(config-telephony)# web admin customer name user22 password pw567890
```
### Related Commands

<table>
<thead>
<tr>
<th>Description</th>
<th>Related Commands</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enters telephony-service configuration mode.</td>
<td><strong>telephony-service</strong></td>
</tr>
<tr>
<td>Loads and parses an XML file in router flash memory to customize a GUI for a customer administrator.</td>
<td><strong>web customize load</strong></td>
</tr>
</tbody>
</table>
web admin system

To define a username and password so that a system administrator can log in to the Cisco Unified CME router through a web browser, use the `web admin system` command in telephony-service configuration mode. To disable a system administrator login, use the `no` form of this command.

```
web admin system [name username] [[password string|secret {0|5} string]]
no web admin system
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Argument</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>name username</td>
<td>(Optional) Unique alphanumeric string to identify a user for this authentication credential only. String can contain a maximum of 28 alphanumeric characters. Default name is Admin.</td>
</tr>
<tr>
<td>password string</td>
<td>(Optional) Unique alphanumeric string to be used by the system administrator which will be stored in the running configuration as plain text. This password is typically known by more than one administrator user and may be created for the default system administrator username. String can contain a maximum of 28 alphanumeric characters. Default is no password.</td>
</tr>
</tbody>
</table>
| secret {0|5} string| (Optional) Unique alphanumeric string for a secret password which will be stored in the running configuration as unencrypted plain text or as encrypted using Message Digest 5 (MD5). This password is typically known by only a select number of administrator users.  
  - 0—Save string as unencrypted plain text in running configuration.
  - 5—Save encrypted string in running configuration. |

**Command Default**

Default is a system administrator with username Admin and no password.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
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</tr>
</thead>
<tbody>
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<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command is deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command enables system administrator access for the Cisco Unified CME graphical user interface (GUI).

The user name parameter of any authentication credential must be unique. Do not use the same value for a user name when you configure any two or more authentication credentials in Cisco Unified CME, such as the username for any Cisco United CME GUI account and the user name in a profile for Extension Mobility.

Use the `secret 5` keyword pair to instruct the system to encrypt the system administrator password with MD5 and to save the encrypted version in the running configuration.
Examples

The following example establishes a system administrator named user1 whose secret password will be encrypted in the running configuration:

Router(config)# telephony-service
Router(config-telephony)# web admin system name user1 secret 5 pw234567

An encrypted version of the preceding string is saved in the running configuration, as shown in the following partial example. The digit 5 that appears after the secret keyword in the running configuration indicates that the password that follows is shown in its encrypted version.

Router(config)# show running-config
!
!
!
web admin system name user1 secret 5 $1$TCyK$OU/NSQ/VtAU2ibHdi8Uau

<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>telephony-service</td>
<td>Enters telephony-service configuration mode.</td>
</tr>
</tbody>
</table>
web customize load

To load and parse an eXtensible Markup Language (XML) file in router flash memory to customize a Cisco CallManager Express graphic user interface (GUI) for a customer administrator, use the `web customize load` command in telephony-service configuration mode. To disable the customized GUI and use the system administrator GUI for the customer administrator, use the `no` form of this command.

```
web customize load filename
no web customize load
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>filename</code></td>
<td>Name of the XML file in router flash memory that defines the customer administrator GUI.</td>
</tr>
</tbody>
</table>

**Command Default**

The standard system administrator GUI is used.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
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<td>This command was introduced.</td>
</tr>
<tr>
<td>12.2(15)T</td>
<td>Cisco ITS 2.1</td>
<td>This command was integrated into Cisco IOS Release 12.2(15)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command is deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

Use this command with Cisco ITS V2.1 and later versions.

**Examples**

The following example specifies a file named `cust_admin_gui.xml` as the file that defines the GUI for Cisco CME customer administrators:

```
Router(config)# telephony-service
Router(config-telephony)# web customize load cust_admin_gui.xml
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>telephony-service</td>
<td>Enters telephony-service configuration mode.</td>
</tr>
</tbody>
</table>
Cisco Unified CME Commands: X

• xml-config, on page 1474
• xmlschema, on page 1475
• xmltest, on page 1476
• xmlthread, on page 1477
• xml user, on page 1478
xml-config

To define the phone-specific XML tags that can be used in the configuration file, use the `xml-config` command in the voice register pool type mode. To remove the XML tags, use the `no` form of this command.

```
xml-config [{maxNumcalls maxNumCalls|busyTrigger busyTrigger|custom custom}]
no xml-config [{maxNumcalls maxNumCalls|busyTrigger busyTrigger|custom custom}]
```

### Syntax Description

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>maxNumcalls</td>
<td>Defines the maximum number of calls allowed per line.</td>
</tr>
<tr>
<td>busyTrigger</td>
<td>Defines the number of calls that triggers call forward busy per line on the SIP phone.</td>
</tr>
<tr>
<td>custom</td>
<td>Defines the custom XML tags that can be appended at the end of the phone-specific CNF configuration profile using the custom option.</td>
</tr>
</tbody>
</table>

### Command Default

The phone-specific XML tags are not defined.

### Note

When the reference-pooltype command is configured, the XML configuration value of the reference phone is inherited.

### Command Modes

Voice Register Pool Configuration (config-register-pool)

### Command History

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>15.3(3)M</td>
<td>Cisco SIP CME 10.0</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### Usage Guidelines

Use this command to define the phone specific XML tags that can be used in the configuration file. The maximum number of calls allowed per line and the number of calls that triggers call forward busy per line information will be used while generating the XML file.

### Examples

The following example shows how to define the phone specific XML tags that can be used in the configuration file:

```
Router# configure terminal
Router(config)# voice register pool-type 9900
Router(config-register-pool-type)# xml-config maxNumCalls 3
Router(config-register-pool-type)# xml-config busyTrigger 3
Router(config-register-pool-type)# xml-config custom <custom-sftp>1</custom-sftp>
```

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice register pool-type</td>
<td>Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.</td>
</tr>
<tr>
<td>phoneload-support</td>
<td>Enables support for phone loads.</td>
</tr>
</tbody>
</table>
xmlschema

Effective with Cisco Unified CME 4.0, the xmlschema command was made obsolete.

For earlier releases, to specify the URL for a Cisco CME eXtensible Markup Language (XML) application program interface (API) schema, use the xmlschema command in telephony-service configuration mode. To set the URL for the XML API schema to the default, use the no form of this command.

xmlschema schema-url
no xmlschema

**Syntax Description**

- schema-url: Local or remote URL as defined in RFC 2396.

**Command Default**

Url for Cisco XML API schema is srst-its.xsd.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(15)ZJ</td>
<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>Cisco CME 3.0</td>
<td>This command was integrated into Cisco IOS Release 12.3(4)T.</td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made obsolete.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made obsolete in Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar</td>
<td>Unified CME 12.6</td>
<td>The command is deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
<tr>
<td>16.11.1a Release</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Examples**

The following example specifies a URL for an XML API schema:

Router(config)# telephony-service
Router(config-telephony)# xmlschema http://server2.example.com/schema/schema1.xsd

**Related Commands**

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>telephony-service</td>
</tr>
</tbody>
</table>
xmltest

Effective with Cisco Unified CME 4.0, the `xmltest` command was made obsolete.

For earlier releases, to specify that the HTTP payload in eXtensible Markup Language (XML) application program interface (API) queries be interpreted as having form format, use the `xmltest` command in telephony-service configuration mode. To specify that the HTTP payload should be interpreted as plain text (no form) format, use the `no` form of this command.

```plaintext
xmltest
no xmltest
```

**Syntax Description**

This command has no arguments or keywords.

**Command Default**

Default format is plain text (no form) format.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
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<tr>
<td>12.2(15)ZJ</td>
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<td>This command was made obsolete in Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar</td>
<td>Unified CME 12.6</td>
<td>The command is deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
<tr>
<td>16.11.1a Release</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Examples**

The following example specifies that the HTTP payload in XML API queries be interpreted as having form format:

```
Router(config)# telephony-service
Router(config-telephony)# xmltest
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>telephony-service</code></td>
<td>Enters telephony-service configuration mode.</td>
</tr>
</tbody>
</table>
xmlthread

Effective with Cisco Unified CME 4.0, the `xmlthread` command was made obsolete.

For earlier releases, to set the maximum number of concurrent Cisco CME eXtensible Markup Language (XML) application program interface (API) queries, use the `xmlthread` command in telephony-service configuration mode. To set the maximum number of queries to the default, use the `no` form of this command.

```
xmlthread number
no xmlthread
```

**Syntax Description**

| `number` | Maximum number of XML API queries. Range is from 1 to 5. Default is 2. |

**Command Default**

The maximum number of queries is 2.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
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<td>Cisco CME 3.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.3(4)T</td>
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<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was made obsolete in Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td></td>
<td>Unified CME 12.6</td>
<td>The command is deprecated. It is not supported on Unified CME 12.6 and later releases.</td>
</tr>
</tbody>
</table>

**Examples**

The following example sets the maximum number of XML API queries to 5:

```
Router(config)# telephony-service
Router(config-telephony)# xmlthread 5
```

**Related Commands**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>telephony-service</code></td>
<td>Enters telephony-service configuration mode.</td>
</tr>
</tbody>
</table>
xml user

To define a user who is authorized to use XML applications to execute commands, use the `xml user` command in telephony-service configuration mode. To delete the user, use the `no` form of this command.

```
xml user user-name password [0|6] password privilege-level
no xml user user-name password privilege-level
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>user-name</code></td>
<td>Unique string used by authorized user to access Cisco Unified CME. Maximum length of string: 19 alphanumeric characters.</td>
</tr>
<tr>
<td><code>password</code></td>
<td>Alphanumeric string to be used with this user name to provide access to Cisco Unified CME. Maximum length of string: 19 alphanumeric characters.</td>
</tr>
<tr>
<td><code>privilege-level</code></td>
<td>Level of access to Cisco IOS commands to be granted to this user. Only the commands with the same or a lower level can be executed via XML. Range is 0 to 15.</td>
</tr>
</tbody>
</table>

**Command Default**

User name is not defined.

**Command Modes**

Telephony-service configuration (config-telephony)

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco Product</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
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<td>Cisco Unified CME 4.0</td>
<td>This command was introduced.</td>
</tr>
<tr>
<td>12.4(9)T</td>
<td>Cisco Unified CME 4.0</td>
<td>This command was integrated into Cisco IOS Release 12.4(9)T.</td>
</tr>
<tr>
<td>Cisco IOS XE Gibraltar 16.11.1a Release</td>
<td>Unified CME 12.6</td>
<td>The command was enhanced for password encryption, based on Unified CME password policy.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

This command creates a credential be used by an authorized user to access Cisco Unified CME via XML and enable the user to execute all the Cisco IOS commands associated with a particular privilege level.

To change the default privilege level for one or more Cisco IOS commands, use the `privilege` command in global configuration mode.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters [0|6]. This in accordance with Unified CME Password Policy. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

**Examples**

The following example defines user23 as an authorized user at level 15:

```
Router(config)# telephony-service
Router(config-telephony)# xml user user23 password 3Rs92uzQ 15
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>privilege</td>
<td>Configures a new privilege level for users and associates commands with that privilege level.</td>
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</tbody>
</table>