Cisco Unified CME Commands: V

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

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.

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

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

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

	Description
template (voice register pool)	Applies a template to a SIP phone.

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

audio-url	Location of the announcement audio file in URL format. Valid storage locations are TFTP, FTP, HTTP, and flash memory.
tag	Number of the voice class that defines the cause codes for which the VCA is played. Range: 1 to 64.

Command Default

No announcement is played.

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

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

Command	Description
bnea	Specifies the audio file used for the busy station not equipped for preemption announcement.
bpa	Specifies the audio file used for the blocked precedence announcement.
ica	Specifies the audio file used for the isolated code announcement.
mlpp indication	Enables MLPP indication on an SCCP phone or analog FXS port.
voice class cause-code	Creates a voice class for defining a set of cause codes.

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

Cisco IOS Release	Cisco Product	Modification
15.1(4)M	Cisco Unified CME 8.6	This command was introduced.

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
!
```

Related Commands

Command	Description
apply-config	Allows to dynamically apply the phone configuration on Cisco Unified SIP IP phones 8961, 9951, and 9971,
bandwidth video tias-modifier	Allows to set the maximum video bandwidth bytes per second (BPS) for SIP IP phones

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.

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

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

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 videoCapability 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:

```
router# show running-configuration
.
.
.
ephone  6
 video
 mac-address 000F.F7DE.CAA5
 type 7960
 button  1:6
```

Related Commands

service phone	Modifies the vendorConfig parameters in phone configuration files.

video (telephony-service)	Enters video configuration mode for modifying video parameters in Cisco Unified CME.

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

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

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

	Description
maximum bit-rate	Sets the maximum video bandwidth for phones in Cisco unified CME.
show call active video	Displays call information for SCCP video calls in progress.

	Description
show call history video	Displays call history information for SCCP video calls.

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

Release	Modification
15.1(4)M	The command was introduced.

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

Command	Description
codec profile	Defines the video capabilities needed for video endpoints.
video codec	Assigns a video codec to a VoIP dial peer.

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 suse 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 Modes

Command History

Release	Modification
15.1(4)M	This command was introduced.

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

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

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

	Description
voicemail (telephony-service)	Configures the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed.

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

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco SRST 2.1	This command was introduced for Cisco Survivable Remote Site Telephony (SRST).
12.2(2)XT	Cisco ITS 2.0	This command was introduced Cisco ITS.
12.2(8)T	Cisco ITS 2.0 Cisco SRST 2.1	This command was integrated into Cisco IOS Release 12.2(8)T.

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

	Description
pattern direct (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when a user presses the Messages button on a phone.

	Description
pattern ext-to-ext busy (vm-integration)	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.
pattern ext-to-ext no-answer (vm-integration)	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.
pattern trunk-to-ext busy (vm-integration)	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.
pattern trunk-to-ext no-answer (vm-integration)	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.

voice class mlpp

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

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

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

Command	Description
service-domain (voice class)	Sets the service domain name in the MLPP voice class.
voice-class mlpp (dial peer)	Assigns an MLPP voice class to a POTS or VoIP dial peer.

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

Cisco IOS Release	Cisco Product	Modification
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1	This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was added for Cisco Unified CME.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

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
```

Related Commands

Command	Description
address	Specifies a comma separated text entry (up to 250 characters) of an ERL's civic address.
elin	Specifies a PSTN number that will replace the caller's extension.
name	Specifies a string (up to 32-characters) used internally to identify or describe the emergency response location.
subnet	Defines which IP phones are part of this ERL.

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

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

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 phone 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

Command	Description
callback	Default phone number to contact if a 911 callback cannot find the last 911 caller from the ERL.
elin	E.164 number used as the default ELIN if no matching ERL to the 911 caller's IP phone address is found.
expiry	Number of minutes a 911 call is associated to an ELIN in the case of a callback from the 911 operator.
logging	Syslog informational message printed to the console every time an emergency call is made.

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

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

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

Command	Description
location	Identifies locations within an emergency response zone and optionally assigns a priority order to the location.

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

hunt-tag	Unique sequence number that identifies the hunt group. Range is 1 to 100.
longest-idle	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.
parallel	Allows an incoming call to simultaneously ring all the numbers in the hunt group member list.
peer	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.
sequential	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.

Command Default

By default, voice hunt group is not created.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was modified to add support for Cisco Unified SCCP IP phones.

Cisco IOS Release	Cisco Product	Modification
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
15.2(4)M	Cisco Unified SIP SRST 9.1	This command was introduced in Cisco Unified SIP SRST 9.1.
15.3(4)M	Cisco Unified CME 10.5	This command was modified to include support for wildcards which is indicated by "*" . symbol.

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 example shows 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

Command	Description
final (voice hunt-group)	Defines the last extension in a voice hunt group.
hops (voice hunt-group)	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.

Command	Description
list (voice hunt-group)	Defines the phone numbers that participate in a voice hunt group.
max-redirect	Changes the number of times that a call can be redirected by call forwarding or transfer within a Cisco Unified CME system.
pilot (voice hunt-group)	Defines the phone number that callers dial to reach a voice hunt group.
timeout (voice hunt-group)	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.

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

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified CME 10.5	This command was introduced.

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
```

Examples

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

```
ephone-dn 22
 number 4566
ephone-dn 23
 number 4567
ephone-dn 24
 number 4568
 voice-hunt-groups login
voice-hunt-groups 1 peer
 list 4566,4567,*,*
 final 7777
```

Related Commands

Command	Description
show voice hunt-groups	Displays voice-hunt group configuration, current status, and statistics.

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.

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

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

The following table lists the available cause codes.

Table 1: Cause Codes for Calls Blocked by LPCOR Validation

Message	Description	Code Number
access-info-discard	access info discarded (43)	43
b-cap-not-implemented	bearer capability not implemented (65)	65
b-cap-restrict	restricted digital info bc only (70)	70
b-cap-unauthorized	bearer capability not authorized (57)	57
b-cap-unavail	bearer capability not available (58)	58

Message	Description	Code Number
call-awarded	call awarded (7)	7
call-cid-in-use	call exists call id in use (83)	83
call-clear	call cleared (86)	86
call-reject	call rejected (21)	21
cell-rate-unavail	cell rate not available (37)	37
channel-unacceptable	channel unacceptable (6)	6
chantype-not-implement	chan type not implemented (66)	66
cid-in-use	call id in use (84)	84
codec-incompatible	codec incompatible (171)	171
cug-incalls-bar	cug incoming calls barred (55)	55
cug-outcalls-bar	cug outgoing calls barred (53)	53
dest-incompatible	incompatible destination (88)	88
dest-out-of-order	destination out of order (27)	27
dest-unroutable	no route to destination (3)	3
dsp-error	dsp error (172)	172
dtl-trans-not-node-id	dtl transit not my node id (160)	160
facility-not-implemented	facility not implemented (69)	69
facility-not-subscribed	facility not subcribed (50)	50
facility-reject	facility rejected (29)	29
glare	glare (15)	15
glaring-switch-pri	glaring switch PRI (180)	180
htspm-oos	HTSPM out of service (129)	129
ie-missing	mandatory ie missing (96)	96
ie-not-implemented	ie not implemented (99)	99

Message	Description	Code Number
info-class-inconsistent	inconsistency in info and class (62)	62
interworking	interworking (127)	127
invalid-call-ref	invalid call ref value (81)	81
invalid-ie	invalid ie contents (100)	100
invalid-msg	invalid message (95)	95
invalid-number	invalid number (28)	28
invalid-transit-net	invalid transit network (91)	91
misdialled-trunk-prefix	misdialled trunk prefix (5)	5
msg-incomp-call-state	message in incomp call state (101)	101
msg-not-implemented	message type not implemented (97)	97
msgtype-incompatible	message type not compatible (98)	98
net-out-of-order	network out of order (38)	38
next-node-unreachable	next node unreachable (128)	128
no-answer	no user answer (19)	19
no-call-suspend	no call suspended (85)	85
no-channel	channel does not exist (82)	82
no-circuit	no circuit (34)	34
no-cug	non existent cug (90)	90
no-dsp-channel	no dsp channel (170)	170
no-req-circuit	no requested circuit (44)	44
no-resource	no resource (47)	47
no-response	no user response (18)	18
no-voice-resources	no voice resources available (126)	126
non-select-user-clear	non selected user clearing (26)	26

Message	Description	Code Number
normal-call-clear	normal call clearing (16)	16
normal-unspecified	normal unspecified (31)	31
not-in-cug	user not in cug (87)	87
number-changeed	number changed (22)	22
param-not-implemented	non implemented param passed on (103)	103
perm-frame-mode-oos	perm frame mode out of service (39)	39
perm-frame-mode-oper	perm frame mode operational (40)	40
precedence-call-block	precedence call blocked (46)	46
preempt	preemption (8)	8
preempt-reserved	preemption reserved (9)	9
protocol-error	protocol error (111)	111
qos-unavail	qos unavailable (49)	49
rec-timer-exp	recovery on timer expiry (102)	102
redirect-to-new-destination	redirect to new destination (23)	23
req-vpci-vci-unavail	requested vpci vci not available (35)	35
send-infotone	send info tone (4)	4
serv-not-implemented	service not implemented (79)	79
serv/opt-unavail-unspecified	service or option not available unspecified (63)	63
stat-enquiry-resp	response to status enquiry (30)	30
subscriber-absent	subscriber absent (20)	20
switch-congestion	switch congestion (42)	42
temp-fail	temporary failure (41)	41
transit-net-unroutable	no route to transit network (2)	2

Message	Description	Code Number
unassigned-number	unassigned number (1)	1
unknown-param-msg-discard	unrecognized param msg discarded (110)	110
unsupported-aal-parms	aal parms not supported (93)	93
user-busy	user busy (17)	17
vpci-vci-assign-fail	vpci vci assignment failure (36)	36
vpci-vci-unavail	no vpci vci available (45)	45

Examples

The following example shows the cause code set to 79:

```
Router(config)# voice lpcor call-block cause 79
```

Related Commands

Command	Description
voice lpcor policy	Creates a LPCOR policy for a resource group.

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

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

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

Command	Description
group (lpcor custom)	Adds a LPCOR resource group to the custom resource list.

Command	Description
voice lpcor enable	Enables LPCOR functionality on the Cisco Unified CME router.
voice lpcor policy	Creates a LPCOR policy for a resource group.

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

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

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

Command	Description
voice lpcor custom	Defines the LPCOR resource groups on the Cisco Unified CME router.

Command	Description
voice lpcor policy	Creates a LPCOR policy for a resource group.

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

incoming	Sets default LPCOR policy for incoming calls.
outgoing	Sets default LPCOR policy for outgoing calls.
lpcor-group	Name of the LPCOR resource group.

Command Default

Default LPCOR policy is not defined for mobility-type phones.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

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

Command	Description
voice lpcor ip-phone subnet	Creates a LPCOR IP-phone subnet table for calls to or from a mobility-type phone.

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

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

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

Command	Description
index (ip-phone)	Adds a LPCOR group to the IP-phone subnet table.
lpcor type	Specifies the LPCOR type for an IP phone.
voice lpcor ip-phone mobility	Sets the default LPCOR policy for mobility-type phones.

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

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

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

Command	Description
index (lpcor ip-trunk)	Adds a LPCOR resource group to the IP trunk subnet table.

Command	Description
lpcor incoming	Associates a LPCOR resource-group policy with an incoming call.
voice lpcor ip-phone subnet	Creates a LPCOR IP-phone subnet table for calls to or from a mobility-type phone.

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

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

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:

```
voice lpcor policy voicemail
 accept siptrunk
 accept h323trunk
 accept pstn
```

The siptrunk group does not have a LPCOR policy defined so it can accept calls from any of the other resource groups.

Related Commands

Command	Description
accept	Allows a LPCOR resource group to accept incoming calls from another resource group.
show voice lpcor policy	Displays the LPCOR policy for the specified resource group.
voice lpcor custom	Defines the LPCOR resource groups on the Cisco Unified CME router.

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.

voice mlpp

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

Cisco IOS Release	Cisco Products	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Voice-mlpp configuration mode is used for the gateway globally.

Examples

The following example shows how to enter voice-mlpp configuration mode:

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

Related Commands

Command	Description
access-digit	Defines the access digit that phone users dial to request a precedence call.
mlpp preemption	Enables calls on an SCCP phone or analog FXS port to be preempted.
preemption trunkgroup	Enables preemption capabilities on a trunk group.

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

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

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

moh	Enables music on hold from a flash audio feed.
multicast moh	Enables multicast of the music-on-hold audio stream.

extension-range	Defines extension range for a clients calling a voice-moh-group.

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

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

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:

```
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

	Description
dialplan	Assigns a dial plan to a SIP phone.

	Description
filename	Specifies a custom XML configuration file that contains the dial patterns to use for a SIP dial plan.
pattern (voice register dialplan)	Defines a dial pattern for a SIP dial plan.
show voice register dialplan	Displays all configuration information for a specific SIP dial plan.
type (voice register dialplan)	Defines a phone type for a SIP dial plan.

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

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 and Cisco SIP SRST 3.4	This command was introduced.

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.

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

	Description
max-dn (voice register global)	Sets the maximum number of SIP phone directory numbers (extensions) supported by a Cisco CME router.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco CallManager Express (Cisco CME) system.
number (voice register pool)	Configures a valid number for a SIP phone.

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

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
15.0(1)XA	Cisco SIP SRST 8.0	This command was updated to display the signaling transport protocol.
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	The no form of the command was modified.

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.

Examples

Examples

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 P0S3-07-4-07
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

Examples

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

Command	Description
allow connections sip to sip	Allows connections between SIP endpoints in a Cisco multiservice IP-to-IP gateway.
application (voice register global)	Selects the session-level application for all dial peers associated with SIP phones.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified system.

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.

voice register pool *pool-tag*

no voice register pool *pool-tag*

Syntax Description

pool-tag	Unique number assigned to the pool. Range is 1 to 100. **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

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME.

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

Examples

The following example shows how to enter voice register pool configuration mode and forward calls to extension 9999 when extension 2001 is busy:

```
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
```

Examples

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

	Description
max-pool (voice register global)	Sets the maximum number of SIP phones that are supported by a Cisco Unified CME system.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified CME system.
number (voice register pool)	Configures a valid number for a SIP phone.
type (voice register pool)	Defines a Cisco IP phone type.

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***}**

novoice 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***}**

Syntax Description

device-reference *supported phone-type*	(Optional) Defines the nearest-supported phone from which a new Cisco Unified SIP IP phone can inherit properties without the explicit configuration of the parameters. For a list of the names, types, and corresponding properties of supported phones, see the below table.
device-name *name*	(Optional) Defines the description string for the new phone device.
device-name *type*	Defines a phone type for a new Cisco Unified SIP IP phone.
addons *max-addons*	(Optional) Defines the maximum number of add-on modules supported by the new phone device. The maximum allowed value is 3. **Note** New add-on modules for an existing phone are not supported.
num-lines *max-lines*	Defines the maximum number of lines supported by the phone. Range is 1 to ???.
transport-type {udp | tcp}	(Optional) Defines the transport protocol supported by the phone. • udp**—**User Datagram Protocol (UDP) is used. • tcp**—**Transmission Control Protocol (TCP) is used.
gsm-handoff	(Optional) Enables phone support for Global System for Mobile Communications (GSM) handoff. **Note** For Cisco IOS Release 15.3(3)M, only CiscoMobile-iOS and Jabber-Android are supported.
telnet	(Optional) Enables phone support for Telnet access. **Note** For Cisco IOS Release 15.3(3)M, only Cisco Unified 3911, 3951, 7905, 7912, 7960, and 7940 SIP IP phones support Telnet access.
phoneload	(Optional) Enables support for phone loads.

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

Release	Modification
15.3(3)M	This command was introduced.

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.

.

Examples

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”
num-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 P0S3-06-0-00
```

Examples

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 referencetype 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
```

Related Commands

Command	Description
session-transport	Specifies the transport layer protocol that a SIP phone uses to connect to Cisco Unified CME.
type (voice register pool)	Defines a phone type for a SIP phone.

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-ephones** command in telephony-service configuration mode.

Command Default

No session manager is created.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW2	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
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 United 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
!
```

Related Commands

Command	Description
session-server	Specifies a session server to manage and monitor registration and subscription messages for an external feature server.

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

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	The maximum number of templates was increased from 5 to 10.
12.4(15)T	Cisco Unified CME 4.1	The increase in the template number was integrated into Cisco IOS Release 12.4(15)T.

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

	Description
anonymous block (voice register template)	Enables anonymous call blocking in a SIP phone template.
caller-id block (voice register template)	Enables caller-ID blocking for outbound calls from a specific SIP phone.
template (voice register pool)	Applies a template to a SIP phone.
voicemail (voice register template	Defines the extension that calls are forwarded to when an extension does not answer.

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

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

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    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 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 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-overly
speed-dial 1 3001
speed-dial 2 3002 blf
```

Related Commands

Command	Description
logout-profile	Enables Cisco Unified IP phone for Extension Mobility and assigns a logout profile to this phone.
reset (voice logout-profile and voice user-profile)	Performs a complete reboot of all IP phones on which a particular logout profile or user profile is downloaded.

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 form of this command.

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

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME.

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:

```
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

	Description
codec (voice register pool)	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.
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.
voice class codec (dial-peer)	Assigns a previously configured codec selection preference list (codec voice class) to a VoIP dial peer.

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

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

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

Command	Description
service-domain (voice class)	Sets the service domain name in the MLPP voice class.

Command	Description
show dial-peer voice	Displays the configuration for all dial peers configured on the router.
voice class mlpp	Creates an MLPP voice class.

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

Release	Cisco Product	Modification
12.4(22)T	Cisco Unified CME 7.0	This command was introduced.
15.1(2)T	Cisco Unified CME 8.1	This command was modified. This command can be enabled in ephone summary, ephone template, voice register pool, or voice register pool template configuration mode.

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

Command	Description
stun usage firewall-traversal flowdata	Enables firewall traversal using STUN.

Command	Description
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

tag	Unique number that identifies the voice gateway. Range: 1 to 25. There is no default value.

Command Default

Gateway configuration is not defined.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command has been integrated into Cisco IOS Release 12.4(24)T.

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

Command	Description
mac-address	Defines the MAC address of the Cisco voice gateway that downloads its configuration from Cisco Unified CME.

Command	Description
type	Defines the type of voice gateway to autoconfigure in Cisco Unified CME.
voice-port	Identifies the analog ports on the voice gateway that register to Cisco Unified CME.

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

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

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

	Description
telephony-service	Enters telephony-service configuration mode.
vm-device-id (ephone)	Defines the voice-mail ID string.

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

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

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

	Description
url (voice register global)	Provision uniform resource locators (URLs) for feature buttons on Cisco IP phones.
voicemail (voice register template)	Defines the extension that calls are forwarded to when an extension does not answer.

	Description
voice register global	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.

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

Syntax Description

phone-number	Telephone number to which calls are forwarded when an extension does not answer.
timeout *seconds*	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.

Command Default

This command has no default behavior or values.

Command Modes

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

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

	Description
template (voice register pool)	Applies a template to a SIP phone.

	Description
url (voice register global)	Provisions uniform resource locators (URLs) for feature buttons on Cisco IP phones.
voice register global	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.
voice register template	Enters voice register template configuration mode and defines a template of common parameters for SIP phones.
voicemail (voice register global)	Defines the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed.

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

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command has been integrated into Cisco IOS Release 12.4(24)T.

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

Command	Description
network-locale (voice-gateway)	Selects a geographically specific set of tones and cadences for the voice gateway's analog endpoints that register to Cisco Unified CME.
type (voice-gateway)	Defines the type of voice gateway to autoconfigure in Cisco Unified CME.

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

Syntax Description

number	Vpn-gateway numbers. Range: 1-3.
url	VPN concentrator address url as https://<IP>/policy.

Command Default

vpn-gateway is not configured.

Command Modes

Vpn-group configuration (conf-vpn-group).

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

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

Command	Description
vpn-group	Specifies a vpn-group.
vpn-trustpoint	Specifies a vpn-gateway trustpoint.

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

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

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

Command	Description
vpn-gateway	Specifies a vpn-gateway URL.
vpn-trustpoint	Specifies a vpn-gateway trustpoint.
vpn-hash-algorithm	Specifies vpn hash encryption for the trustpoints.

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

sha-1	Encryption algorithm.

Command Default

vpn-hash-algorithm is not configured

Command Modes

Vpn-group configuration (conf-vpn-group)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

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

Command	Description
vpn-group	Specifies a vpn-group.
vpn-trustpoint	Specifies a vpn-gateway trustpoint.

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

tag	Vpn-profile tag number. Range:1-6,

Command Default

No vpn-profile is configured.

Command Modes

Voice service voip (conf-voi-serv)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

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
  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
!
!
```

Related Commands

Command	Description
voice-service-voip	Enters voice-service configuration mode for Voice Over IP (VoIP) encapsulation.
vpn-group	Enters vpn-group configuration mode.

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

number	Number of allowed trustpoints. Range is from 1 to 10.
raw	(Optional) Allows to enter VPN Gateway Trustpoint in raw form.
trustpoint	(Optional) Allows to enter VPN Gateway Trustpoint in IOS format.
leaf	Get the 1st leaf cert of the Trustpoint.
root	Get the root cert of the Trustpoint.

Command Default

vpn-trustpoint is not configured.

Command Modes

Vpn-group (conf-vpn-group)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

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

Command	Description
vpn-grouptrustpoint	Defines a vpn-group.