AT&T IP Flexible Reach – Enhanced Features Service on MIS, MPLS PNT or AT&T VPN: Connecting Cisco Unified Communications Manager 9.0 with Cisco Unified Border Element Release 9.0 using SIP

April 25, 2013

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Introduction

Service Providers today, such as AT&T, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. AT&T IP Flexible Reach – Enhanced Features (IP Flexible Reach – EF) is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity via either analog or T1 lines. A demarcation device between these services and customer owned services is recommended. The Cisco Unified Border Element provides demarcation, security, interworking and session management services.

- This application note describes how to configure a Cisco Unified Communications Manager (CISCO UCM) 9.0 with Cisco Unified Border Element (CISCO UBE) for connectivity to AT&T’s IP Flexible Reach-Enhanced Features (IP Flexible Reach-EF) SIP trunk service. The application note also covers support and configuration example for Cisco Unity Connection (CUC) messaging integrated with the CISCO UCM. The deployment model covered in this application note is Customer Premises Equipment (CISCO UCM/CUC/CISCO UBE) to PSTN (AT&T IP Flexible Reach SIP). AT&T IP Flexible Reach-EF provides inbound and outbound call service.

- Testing was performed in accordance to AT&T’s IP Flexible Reach-EF test plan and all features were verified. Key features verified are: inbound and outbound basic call (including international calls), calling name delivery, calling number and name restriction, DNIS translations, CODEC negotiation, advanced 8YY call prompter, intra-site transfers, intra-site conferencing, call hold and resume, call forward (forward all, busy and no answer), leaving and retrieving voicemail (Cisco Unity Connection), auto-attendant, fax using T.38 and G.711 (G3 and SG3 speeds), teleconferencing, Simultaneous and Sequential Ring, failover of unresponsive SIP network to PSTN and outbound/inbound calls to/from TDM networks.

- The Cisco Unified Border Element function configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between AT&T SIP network and Cisco Unified Communications Manager. The configuration described in this document details the important commands for successful interoperability. Care must be taken by the network administrator deploying CISCO UBE to ensure these commands are set per each dial-peer required, to interoperate to AT&T SIP network.

- For IP Flexible Reach-EF service with MIS/MPLS PNT (and optionally with AT&T VPN) access, the Cisco Unified Border Element IP address (facing the Customer Edge Router) can be a private IP address. This will be NATed by the AT&T managed Customer Edge Router (or customer managed/MRS managed Customer Edge Router for AT&T VPN). Consult with AT&T provisioning engineer to resolve any IP addressing issues.

- Please refer to the Emergency 911/E911 Limitations and Restrictions section of this document for more information on Emergency 911/E911 services.

- This Application Note uses the Cisco 2921 Integrated Services Router (ISR) G2 to run CISCO UBE feature set however other Cisco voice gateways are also an option to use since CISCO UBE implementation does not depend on the platform. Here is a list of Cisco platforms capable of CISCO UBE functionality: (Note: not all platforms listed below may be able to support CUBE 9.0)

  - Cisco 3900 Series Integrated Services Routers
  - Cisco 2900 Series Integrated Services Routers
  - Cisco 2800 Series Integrated Services Routers
  - Cisco 3800 Series Integrated Services Routers
  - Cisco AS5350XM Universal Gateway
  - Cisco AS5400XM Universal Gateway
  - Cisco 1861 Integrated Services Router
  - Cisco 881 Integrated Services Router
  - Cisco 888 Integrated Services Router
  - Cisco IAD880 Series Integrated Access Devices
  - Cisco IAD2430 Integrated Access Device

- Consult your Cisco representative for the correct IOS image and for the specific application and Device Unit License and Feature License requirements for all your Cisco Unified Communications Manager with Cisco Unified Border Element components.
Network Topology

Basic Call Setup

Note: The Cisco Unified Border Element depicted in Figure 1 is not an AT&T managed device. It is recommended that the group responsible for the administration, management and configuration of the Cisco Unified Communications Manager, also manage and configure the Cisco Unified Border Element.

For Cisco supported deployment strategies using CISCO UBE, refer to the Cisco Unified Communications SRND Based on Cisco Unified Communications Manager 9.X:

System Components

Hardware Components

- Cisco Integrated Service Router G2. This solution was tested with C2921 but this application note applies to any ISR platforms. Refer to the following links provided in the Introduction section of this document for more information on ISR platforms:
- Cisco High-Density Packet Voice Digital Signal Processor Module (PVDM3). You will need to install DSP modules on an ISR platform if you require MTP, Transcoding or Conference Bridge resources for codecs other than G.711. DSPs are not required for basic calls. Click on the following link to access the Cisco DSP Calculator: http://www.cisco.com/web/applicat/dspcalc/dsp_calc.html
- Cisco MCS 7800 Series server to run Cisco Unified Communication Manager (CISCO UCM)
- Cisco IP Phones. This solution was tested with 7965, 7975 and 7961 phones, but any Cisco IP Phone model supporting RFC2833 can be used
- Cisco MCS 7800 Series server to run Cisco Unity Connection (CUC).
- VG224 Analog Gateway (Fax application). Alternatively, other Cisco IOS gateway(s) can be used. This component may be an H.323, SIP, or MGCP gateway. The protocol is optional, and the choice is left up to the customer’s network design.

Software Requirements

- Cisco Unified Communication Manager (CISCO UCM) Release 9.0. This solution was tested with 9.0.1.10000-37. This document is also applicable to all Cisco UCM 9.0 – 9.1 Enterprise and Business Edition 6000 software releases
- Cisco Unified Border Element (CISCO UBE) Release 9.0 with IOS version 15.2(4)M1. This configuration was tested with C2900-universalk9_npe-mz.SPA.152-4.M1; however, this document is also applicable to all IOS 15.2(4)M/CUBE 9.0 and 15.2(3)T/CUBE 9.0 software releases
- Cisco Unity Connection (CUC) Release 8.6. This solution was tested with 8.6.1.20002-109
- Cisco VG224 Gateway (IP-TDM) IOS version: 15.1(4)M4
- The documented CISCO UBE configuration can be supported with the following IOS feature sets: UNIVERSAL
- Consult your Cisco representative for the correct IOS image and for the specific application and Device Unit License and Feature License requirements for all your Cisco Unified Communications components. For reference, please follow this link: http://www.cisco.com/en/US/prod/collateral/voicesw/ps6790/gatecont/ps5640/order_guide_c07_462222.html
Features

Features Supported

- Basic Call using G.729
- Calling Party Number Presentation and Restriction
- Calling Name
- AT&T Advanced 8YY Call Prompter (8YY)
- Intra-site Call Transfer
- Intra-site Conference (See Caveat section for details)
- Call Hold and Resume
- Call Forward All, Busy and No Answer
- AT&T IP Teleconferencing
- Fax using T.38
- Fax over G.711
- Incoming DNIS Translation and Routing
- CISCO UBE Delayed-Offer-to-Early-Offer conversion of an initial SIP INVITE without SDP
- Outbound calls to AT&T’s IP and TDM networks
- CPE voicemail managed service, leave and retrieve voice messages via incoming AT&T SIP trunk (Cisco Unity Connection)
- Auto-attendant transfer-to service (See Caveat section for details)
- Failover (From non-responsive SIP network to legacy PSTN circuit)
- RTCP is supported on the following Cisco IP Phone models: 7906, 7911, 7912, 7931, 7941, 7942 7945, 7961, 7962, 7965, 7970, 7971, 8961 9951
- Network-based Call Forward Unconditional, Busy, No Answer and Not Reachable
- IP Flexible Reach – EF Network-based Simultaneous Ringing – Requires IOS version 15.2(4)M3
- IP Flexible Reach – EF Network-based Sequential Ringing – Requires IOS version 15.2(4)M3

Features Not Supported

- CISCO UCM/CISCO UBE Codec negotiation of G.722.1
- IP Flexible Reach – EF Network-based Blind Call Transfer
Caveats

Configuration considerations

Codec

- When using G.729 between AT&T IP Flexible Reach-EF and Cisco Unified Communication Manager (CISCO UCM) / Cisco Unified Border Element (CISCO UBE) SIP trunk, it is required to configure a conference bridge (CFB) resource to initiate a three-way conference between G.729 media end-points. (See configuration section for details).

- It is recommended to have a transcoder resource if the customer network will support more than one codec. (See configuration section for details)

- Cisco Unified IP phones using SIP as the registration protocol (SIP-line) does not support G.729 with annex B. This current SIP line side support causes failed call attempts when Cisco UBE is set for codec “g729br8” negotiation. Workaround is to remove “g729br8” from the preference codec list and only enable “g729r8”. (See configuration section for details)

- AT&T IP Flexible Reach-EF SIP service may offer G.729 annexB during a call and G.729 no annexB during another call. If this occurs during call transfer or call conference the service will fail due to incompatible codec negotiation. It is recommend to only advertise the codec g729r8 (g729 annexB=no) within the voice-class codec setting in CISCO UBE, in order to avoid this caveat.

- SIP Profiles on the CISCO UBE may also be employed to advertise desired RTP payload packet size. (See configuration section for details)

Diversion-Header for Diverted (Forwarded) Calls

CISCO UCM 8.5 introduces a new feature to modify SIP messages using normalization scripts to improve interoperability. For forwarded calls from CISCO UCM user to PSTN (out to AT&T’s IP Flexible Reach - EF service), some AT&T service areas require that the SIP Diversion header contain the full 10-digit DID number of the forwarding party. In this application note the assumption was made that a typical customer will utilize extension numbers (4-digit assignments in this example) and map 10-digit DID number using CISCO UCM translation patterns. Because 4-digit extensions were used, it is necessary to expand the 4-digit extension included in the Diversion header of a forwarding INVITE message to its full 10-digit DID number when the IP phone is set to call-forward. The requirement to expand the Diversion-Header has been achieved by the use of a SIP Normalization Script used by the SIP trunk to AT&T via CISCO UBE. Note that Diversion header contents may be generated by CTI ports and route points as well as endpoints. This script also removes the Diversion header generated by CISCO UCM with a “911” route point for integration with Cisco Emergency Responder. (See configuration section for details).

Unattended Call Transfer

- No Ringback tone - Caller does not hear ringback tone when call to another CISCO UCM phone is transferred (unattended) to AT&T. A workaround on the CISCO UBE is applied by issuing “disable-early-media 180” under sip-ua.

- Unable to complete unattended call transfer - Cisco UCM sends a SIP UPDATE message to the Cisco UBE for unattended call transfers. Since AT&T does not support SIP UPDATE, the CUBE is not able to pass this header to AT&T and it returns a 491 Request Pending message back to the Cisco UCM until the call session expires and the call transfer is not completed. As a workaround, a SIP profile can be applied on the Cisco UBE dial-peer towards the Cisco UCM to remove UPDATE as one of the allowed SIP header. This will prevent the Cisco UCM from sending any UPDATE message towards the CUBE. (See configuration section for details)

Fax

- For outbound calls, AT&T SIP network will always select the first choice codec presented in the initial SIP INVITE (unless the end-device does not support the listed preferred codec), and processes calls accordingly. Customers wishing to place G.711-only calls (e.g. Fax over G.711) must configure separate dial-peer(s) on Cisco UBE with G.711 codec assigned as the only choice. This dial-peer is
matched on CISCO UCM by a Route Pattern directing calls to it. Typically, this solution is used for fax/modem transmissions using G.711. To ensure that inbound fax/modem calls are established using G.711 codec, devices need to use dedicated range of DID numbers which use a dial-peer towards SIP/MGCP/H.323 gateways supporting fax/modem with destination pattern and incoming called number matching these DID range with the end-device using G.711 codec. Cisco UCM Region configurations and Region codec Relationships might be needed to support G.711 fax and G.729 voice calls simultaneously over the same SIP Trunk.

- For T.38 Fax calls, some SIP components within AT&T core do not support the “-0” as the Boolean value within the “T38FaxFillBitRemoval” parameter within the SDP header of a fax Re-INVITE. Thus, a sip profile is used to remove this attribute to achieve fax T.38 interoperability across AT&T SIP core.

SIP Provisional Acknowledgement

- Some PSTN network call prompters utilize early-media cut-through to offer menu options to the caller (DTMF select menu) before the call is connected. In order for CISCO UCM/CISCO UBE solution to achieve successful early-media cut-through the CISCO UCM to CISCO UBE call leg must be enabled with SIP PRACK. To enable SIP PRACK on the CISCO UCM, the SIP Profile “SIP Rel1XX Options” setting must be set to “Send PRACK”. The SIP Profile is found under Device>Device Settings>SIP Profile, You can assign this feature on a per SIP trunk basis using SIP profiles. SIP PRACK provisioning on CISCO UCM 8.X and newer software versions is enabled under SIP Profile configuration page while SIP PRACK support on CISCO UCM 7.X and older software versions is enabled under the Service Parameters configuration page.

IP Flexible Reach-EF Network-based Simultaneous Ringing and Sequential Ringing

- These network-based features are currently supported with IOS version 15.2(4)M3. SIP signaling analysis between the CUBE and AT&T IP Flexible Reach-EF network shows that CISCO UBE, when using IOS version 15.2(4)M1, does not handle properly the re-INVITE from the network to open the voice channel. The call drops once the call is answered in a simultaneous ringing or sequential ringing scenario. Please refer to Cisco DDTS CSCub35268 for more details. This limitation has been resolved on IOS software version 15.2(4)M3. Customers implementing these features must upgrade CISCO UBE to this software version.

CISCO UBE in High Availability (HA) mode

- For use of CUBE in High Availability (HA) mode using HSRP configuration, please use the following link for guidance:


HA was previously tested using CUBE 8.5.

Emergency 911/E911 Services Limitations and Restrictions

- Emergency 911/E911 Services Limitations and Restrictions - Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer configuration guide will properly operate with AT&T IP Flexible Reach-EF to complete 911/E911 calls; therefore, it is Customer's responsibility to ensure proper operation with its equipment/software vendor.

- While AT&T IP Flexible Reach-EF supports E911/911 calling capabilities under certain Calling Plans, there are circumstances when that E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found in the SG Library at http://new.serviceguide.att.com Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the Customer's location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Guide in detail to understand the limitations and restrictions.
Configuration

CISCO Unified Border Element (CISCO UBE) Software Version

CUBE_ISRG2_ATT#show version
Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9_NPE-M), Version 15.2(4)M1,
RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2012 by Cisco Systems, Inc.
Compiled Thu 26-Jul-12 20:54 by prod_rel_team

ROM: System Bootstrap, Version 15.0(1r)M1, RELEASE SOFTWARE (fc1)

CUBE_ISRG2_ATT uptime is 1 week, 5 days, 6 hours, 0 minutes
System returned to ROM by power-on
System image file is "flash:c2900-universalk9_npe-mz.SPA.152-4.M1.bin"
Last reload type: Normal Reload
Last reload reason: power-on

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:
If you require further assistance please contact us by sending email to export@cisco.com.

Cisco CISCO2921/K9 (revision 1.0) with 487424K/36864K bytes of memory.
Processor board ID FTX1348AHMN
3 Gigabit Ethernet interfaces
1 terminal line
DRAM configuration is 64 bits wide with parity enabled.
255K bytes of non-volatile configuration memory.
255488K bytes of ATA System CompactFlash 0 (Read/Write)

License Info:
License UDI:

-------------------------------------------------------------
Device# PID SN
-------------------------------------------------------------
*0 CISCO2921/K9 FTX1348AHMN

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Technology Package License Information for Module:'c2900'

<table>
<thead>
<tr>
<th>Technology</th>
<th>Technology-package</th>
<th>Type</th>
<th>Technology-package</th>
<th>Next reboot</th>
</tr>
</thead>
<tbody>
<tr>
<td>ipbase</td>
<td>ipbasek9</td>
<td>Permanent</td>
<td>ipbasek9</td>
<td></td>
</tr>
<tr>
<td>security</td>
<td>None</td>
<td>None</td>
<td>None</td>
<td>None</td>
</tr>
<tr>
<td>uc</td>
<td>uck9</td>
<td>Permanent</td>
<td>uck9</td>
<td></td>
</tr>
<tr>
<td>data</td>
<td>None</td>
<td>None</td>
<td>None</td>
<td>None</td>
</tr>
</tbody>
</table>

Configuration register is 0x2102

Configuring Cisco Unified Border Element (CISCO UBE)

CUBE_ISRG2_ATT#sho running
Building configuration...

Current configuration : 10206 bytes

! Last configuration change at 10:45:48 PST Mon Aug 27 2012
version 15.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service sequence-numbers
!
hostname CUBE_ISRG2_ATT
!
boot-start-marker
boot system flash:c2900-universalk9_npe-mz.SPA.152-4.M1.bin
boot-end-marker
!
!
logging queue-limit 10000
logging buffered 20000000
logging persistent filesize 20000000
logging rate-limit 10000
no logging console
enable secret 5 $1$4jgu$npJCRdswNO47pZhBy3fbi/
enable password cisco
!
no aaa new-model
clock timezone PST -8 0
!
!
!
ip domain name yourdomain.com
no ip cef
no ipv6 cef
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-110016895
enrollment selfsigned
subject-name cn=IOS-Self-Signed-Certificate-110016895
revocation-check none
rsakeypair TP-self-signed-110016895

crypto pki certificate chain TP-self-signed-110016895
certificate self-signed 01
30820254 308201BD A030201 02020101 300D0609 2A864886 F70D0101 04050030
31312F30 2D060355 04031326 494F532D 53656C66 2D5369676E65642D 43657274
69666963 6174652D 31313030 31363836 3935013E 170D3132 30383038 31363134
33365A17 0D323030 31033030 30303030 30303030 312F3230 06035504 03132649
4F532D53 656CC62D 5369676E 65624D43 65727469 66696361 74652331 31303031
36383639 3530819F 300D0609 2A864886 F70D0101 01050003 818D0030 81892081
8100DC2A 57584BA4 9E84CCB6 AE8FC0F9 4CC9D6A8 56FEED2D 83711442 0A45DCE7
875F5622 ES04025D 3BF6DD78 894E8DCC 67FA6E21 7EB2012F 551B8100 DBF61436
89CB7CA4 2C40EDCD 395AC7D0 F759F0C8 E94220B FDB9F6E9 34067A81 DE1BE5A9
DA7FB98E 533FB54E E3747FA3 758F19D9 BA886C9A 16FCD1A7 3B3DD80C 195110B2
A62B0203 010001A3 7C307A30 0F060355 1D130101 FF040530 031000FF 30270603
551D1104 20301E82 1C55434D 2D495332 47322D41 5452E79F 6F757264 6F6D6169
6E2E6E6F 6D301F06 03551D23 04183016 80148994 E1E21AE2 848D4387 C647F727
5DB0345F 0ACC301D 603551D 0E041604 148994E1 E21AE284 8D4387C6 47F7275D
B0345F0A CC300D06 92A86648 86F70D01 01040500 03811B00 050B77EA 0639D7C6
75C7E8E6 C256D569 86872149 ABD193ED DAE3220D F41FA5F0 867494BD D53F54A0
0BCD8990 2108F2E0 74D872AA 70386387 F1316AE6 A4DCB8EA 440B4483 DB11E722
F695EA4O 98762C08 34F3F0F9 0ADF2F99 516B51BF F285FB76 4A047C85 04DA0632
B55DCDF7 00F9BAC 22E789FF 50C2666C 7A2CCC7E 7A82FEA6

quit
voice-card 0
dspfarm
dsp services dspfarm

voice service voip
address-hiding
mode border-element
allow-connections sip to sip
no supplementary-service sip moved-temporarily
redirect ip2ip
h323
sip
error-passthru
asserted-id pai
early-offer forced
midcall-signaling passthru
privacy-policy passthru
g729 annexb-all

1 This command enables router to send P-Asserted ID within the SIP Message Header. Alternatively, this command can also be applied to individual dial-peers (voice-class sip asserted-id pai)
voice class codec 1
- codec preference 1 g729a8 bytes 30
- codec preference 2 g711ulaw

voice class sip-profiles 1
- response ANY sip-header Allow-Header modify "UPDATE," ""
- request INVITE sip-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"
- request REINVITE sip-header Attribute modify "a=T38FaxFillBitRemoval:0" ""
- request INVITE sip-header Audio-Attribute add "a=ptime:30"

license udi pid CISCO2921/K9 sn FTX1348AHMN

archive
log config
hidekeys
username Cisco password 0 cisco

redundancy

no ip ftp passive

translation-rule 1

interface Embedded-Service-Engine0/0
- no ip address
- shutdown

interface GigabitEthernet0/0
- description Connection to UC Interop lab network
- ip address 172.20.110.158 255.255.255.0
- duplex auto
- speed auto

interface GigabitEthernet0/1
- description connection to ATT Network
- ip address 99.136.XXX.XXX 255.255.255.0
- duplex auto
- speed auto

interface GigabitEthernet0/2

This command configures the codec preference to be assigned to dial-peers. Alternatively, single codec’s can be configured into individual dial-peers.

This SIP Profile removes UPDATE from SIP Message Header to/from Cisco UCM, as it can cause problems during unattended call transfers. By default, Cisco 6900-series IP phones use ptime value of 20 ms. AT&T networks prefer ptime value of 30 ms. This SIP profile modifies SDP ptime value from 20 to 30. It is also used to remove SDP media attribute "a=T38FaxFillBitRemoval:0", as it can cause T.38 fax relay transmissions to fail.
no ip address
shutdown
duplex auto
speed auto
!
ip default-gateway 172.20.110.1
ip forward-protocol nd
!
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
ip dns server view-group DNS
no ip pim dm-fallback
ip route 172.20.0.0 255.255.0.0 172.20.110.1
ip route 172.30.0.0 255.255.0.0 172.20.43.1
ip route 207.242.0.0 255.255.0.0 99.136.0.0
ip route 207.242.0.0 255.255.255.255 99.136.0.0
!
access-list 23 permit 10.10.10.0 0.0.0.7
!
!
control-plane
!
!
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
!
mgcp
no mgcp package-capability res-package
no mgcp package-capability fxr-package
mgcp fax t38 ecm
!
mgcp profile default
!
!
dial-peer voice 1999 voip
description Outgoing calls to AT&T - Facing AT&T network
destination-pattern 1T
session protocol sipv2
session target ipv4:207.242.xxx.xxx
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400 bytes 48
fax protocol t38 version 0 ls-redundancy 5 hs-redundancy 1 fallback pass-through g711ulaw

---

4 This command enables DTMF digit passing using RTP NTE (RFC2833) to calls matching this dial-peer
no vad
!
dial-peer voice 2000 voip
description Outgoing calls to AT&T - Facing Cisco UCM
session protocol sipv2
incoming called-number 1T
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip profiles 1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax protocol t38 version 0 ls-redundancy 5 hs-redundancy 1 fallback pass-through g711ulaw
no vad
!
dial-peer voice 732 voip
description Incoming calls from AT&T - Facing Cisco UCM
destination-pattern 00000[37][13][24]........
session protocol sipv2
session target ipv4:172.20.236.50
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip profiles 1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400 bytes 48
fax protocol t38 version 0 ls-redundancy 5 hs-redundancy 1 fallback pass-through g711ulaw
no vad
!
dial-peer voice 733 voip
description Incoming calls from AT&T - Facing AT&T network
destination-pattern 00000[37][13][24]........
session protocol sipv2
session target ipv4:172.20.236.50
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip profiles 1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 5 hs-redundancy 1 fallback pass-through g711ulaw
no vad
!
dial-peer voice 11 voip
description Outgoing Intl Calls to AT&T - Facing AT&T network
destination-pattern 011T
session protocol sipv2
session target ipv4:207.242.xxx.xxx
voice-class codec 1
voice-class sip asymmetric payload full

\* This command enables Cisco UBE to perform T.38 fax relay. To change fax protocol to pass-through using G.711ulaw, the command has to be changed to “fax protocol pass-through g711ulaw”
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400 bytes 48
fax protocol t38 version 0 ls-redundancy 5 hs-redundancy 1 fallback pass-through g711ulaw
no vad

! dial-peer voice 12 voip
description Outgoing Intl calls to AT&T - Facing Cisco UCM
session protocol sipv2
incoming called-number 011T
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip profiles 1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax protocol t38 version 0 ls-redundancy 5 hs-redundancy 1 fallback pass-through g711ulaw
no vad

! dial-peer voice 511 voip
description N11 Calls to AT&T - Facing AT&T network
destination-pattern [459]11
session protocol sipv2
session target ipv4:207.242.xxx.xxx
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
dtmf-relay rtp-nte
no vad

! dial-peer voice 512 voip
description N11 Calls to AT&T - Facing Cisco UCM
destination-pattern 3204...
session protocol sipv2
session target ipv4:172.20.236.50
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai

! dial-peer voice 2001 voip
description Incoming local 7-digit calls from AT&T - Facing Cisco UCM
destination-pattern 3204...
session protocol sipv2
session target ipv4:172.20.236.50
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
dtmf-relay rtp-n-te
fax-relay sg3-to-g3
fax rate 14400 bytes 48
fax protocol t38 version 0 ls-redundancy 5 hs-redundancy 1 fallback pass-through g711ulaw
no vad
!
dial-peer voice 2002 voip
description Incoming local 7-digit calls from AT&T - Facing AT&T network
session protocol sipv2
incoming called-number 3204...
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip assert-id pai
voice-class sip profiles 1
dtmf-relay rtp-n-te
fax-relay sg3-to-g3
fax rate 14400 bytes 48
fax protocol t38 version 0 ls-redundancy 5 hs-redundancy 1 fallback pass-through g711ulaw
no vad
!
dial-peer voice 2005 voip
description Outgoing call to AT&T Operator – Facing AT&T network
destination-pattern 0T
session protocol sipv2
session target ipv4:207.242.xxx.xxx
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip assert-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
dtmf-relay rtp-n-te
no vad
!
dial-peer voice 2006 voip
description Outgoing call to AT&T Operator – Facing Cisco UCM
destination-pattern 0T
session protocol sipv2
session target ipv4:207.242.xxx.xxx
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip assert-id pai
voice-class sip profiles 1
dtmf-relay rtp-n-te
no vad
!
dial-peer voice 72 voip
description Network-Based Call Forwarding Feature Access Codes plus Telephone Number – Facing AT&T network
destination-pattern *T
session protocol sipv2
session target ipv4:207.242.xxx.xxx
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
dtmf-relay rtp-nte

! dial-peer voice 73 voip
description Network-Based Call Forwarding Feature Access Codes plus Telephone Number - Facing Cisco UCM
session protocol sipv2
incoming called-number *T
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip profiles 1
dtmf-relay rtp-nte
no vad
!
sip-ua
no remote-party-id
disable-early-media 180^6
retry invite 2
!
!
!
!
!
!
!
!
!
!
line con 0
 exec-timeout 0 0
line aux 0
line 2
 no activation-character
 no exec
 transport preferred none
 transport input all
 transport output pad telnet rlogin lapb-ta mop udp v120 ssh
 stopbits 1
 line vty 0 4
 exec-timeout 15 0
 password cisco
 login
 transport input all
!
scheduler allocate 20000 1000
!
end

^6 This command allows Cisco UBE to disable early-media upon receiving 180 Ringing message. It is required in order to provide ringback tone during unattended call transfers
Configuring Cisco Unified Communication Manager (CISCO UCM)

CISCO UCM Version

Cisco Unified CM Administration
System version: 9.0.1.10000-37

Last Successful Login: Monday, August 27, 2012 11:07:34 AM EDT

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This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at our Export Compliance Product Report web site.

For information about Cisco Unified Communications Manager please visit our Unified Communications System Documentation web site.

For Cisco Technical Support please visit our Technical Support web site.
Note: CISCO UCM 9.0 introduced a new feature: Audio Codec Preference List. This feature allows for configuration of the order of audio codec preference, both for inter- and intra-Region calls. Add a new Audio Codec Preference List, with G.729 codecs configured above G.711 (higher priority). This new Audio Codec Preference list is assigned to the Region used by the Device Pool for Phones, as well as Conference Bridges. With this configuration in place, inbound calls, as well as call conferences initiated by Cisco IP phones, will use G.729 codec as their first choice codec. Audio codec preference for outbound calls is determined by CUBE (via configuration of voice-class codec)
### CISCO UCM Regions

#### Cisco Unified CM Administration

For Cisco Unified Communications Solutions

**Find and List Regions**

- **Add New**
- **Select All**
- **Clear All**
- **Delete Selected**

**Status**

- 2 records found

#### Regions (1 - 2 of 2)

<table>
<thead>
<tr>
<th>Name</th>
<th>Find Regions where name begins with</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Default</td>
</tr>
<tr>
<td></td>
<td>G711_R_01</td>
</tr>
</tbody>
</table>

**Region Configuration**

**Region Information**

- **Name**: Default

**Region Relationships**

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec Preference List</th>
<th>Maximum Audio Bit Rate</th>
<th>Maximum Session Bit Rate for Video Calls</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Use System Default</td>
<td>0 kbps (G.729)</td>
<td>000</td>
</tr>
<tr>
<td>G711_R_01</td>
<td>Codes Pref. List + Conf. Bridges and G711 endpoints</td>
<td>64 kbps (G.722, G.711)</td>
<td>004</td>
</tr>
</tbody>
</table>

**Modify Relationship to other Regions**

<table>
<thead>
<tr>
<th>Regions</th>
<th>Audio Codec Preference List</th>
<th>Maximum Audio Bit Rate</th>
<th>Maximum Session Bit Rate for Video Calls</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
<td></td>
</tr>
<tr>
<td>G711_R_01</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
<td></td>
</tr>
</tbody>
</table>
### Region Information

Name: G711 Region

### Region Relationships

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec Preference List</th>
<th>Maximum Audio Bit Rate</th>
<th>Maximum Session Bit Rate for Video Calls</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Custom Codec Preference - Conference Bridges</td>
<td>64 kbps (G.722, G.711)</td>
<td>394</td>
</tr>
<tr>
<td>G711Region</td>
<td>Use System Default (Factory Default low loss)</td>
<td>64 kbps (G.722, G.711)</td>
<td>394</td>
</tr>
</tbody>
</table>

NOTE: Regions not displayed

Use System Default | Use System Default | Use System Default

### Modify Relationship to other Regions

<table>
<thead>
<tr>
<th>Regions</th>
<th>Audio Codec Preference List</th>
<th>Maximum Audio Bit Rate</th>
<th>Maximum Session Bit Rate for Video Calls</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
<td></td>
</tr>
<tr>
<td>G711Region</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
<td></td>
</tr>
</tbody>
</table>
Note: Two Device Pools are configured: the **Default** Device Pool is assigned to the SIP Trunk connecting CISCO UCM to CISCO UBE, while the **G711 Pool** is assigned to phones and Conference Bridges. No special consideration needs to be taken when configuring the two Device Pools. Optionally, a Media Resource Group List can be added to the Device Pools, if needed, to assign selected Media Resources (Conference Bridges, Transcoders, MoH servers, Annunciators) to devices.
Media Resource Group

Note: The Media Resource Group (MRG) contains media resources, such as Conference Bridge, Transcoder, MoH server, and Annunciator. It will be assigned to a Media Resource Group List (MRGL), used to allocate media resources to groups of devices through Device Pools, or individually by configuring a valid MRGL at the device configuration page.
## Cisco 7965 IP Phone

### Phone Configuration

<table>
<thead>
<tr>
<th>Status</th>
<th>Status Ready</th>
</tr>
</thead>
</table>

### Association Information

<table>
<thead>
<tr>
<th>Line 1</th>
<th>Add a new DN</th>
</tr>
</thead>
<tbody>
<tr>
<td>Line 2</td>
<td>Add a new DN</td>
</tr>
<tr>
<td>Line 3</td>
<td>Add a new DN</td>
</tr>
<tr>
<td>Add a new SD</td>
<td></td>
</tr>
<tr>
<td>Add a new SD</td>
<td></td>
</tr>
</tbody>
</table>

### Phone Type

- **Product Type:** Cisco 7965
- **Device Protocol:** SCCP

### Device Information

- **Registration:** Registered with Cisco Unified Communications Manager CM-Polaris
- **IP Address:** 172.26.234.55
- **Active Load ID:** SCCP45.9-3-1-15
- **Download Status:** Unknown
- **Device is Active:**
- **Device is Trusted:**
- **MAC Address:** 00:04:40:7F:41
- **Description:** 7965 SCCP - ATT IP Enhanced Presence testing
- **Device Pool:** G711_Pool
- **Common Device Configuration:**
- **Phone Button Template:** SEP_02404BD7F41-SCCP-Individual Template
- **Softkey Template:** Standard User
- **Common Phone Profile:** Standard Common Phone Profile
- **Calling Search Space:** bj_phones_IP
- **AAA Calling Search Space:**
- **Media Resource Group List:** MRL_Polaris
- **User Hold MOS Audio Source:** 1-SampleAudioSource
- **Network Hold MOS Audio Source:** 1-SampleAudioSource
- **Location:** Hub_None
- **AAR Group:**
- **User Locale:**
- **Network Locale:**
- **Built In Bridge:** Default
- **Backup:**
| Page 26 of 84 |

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Owner User ID</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Phone Personalization</td>
<td>Default</td>
</tr>
<tr>
<td>Services Provisioning</td>
<td>Default</td>
</tr>
<tr>
<td>Phone Lead Name</td>
<td></td>
</tr>
<tr>
<td>Single Button Earpiece</td>
<td>Default</td>
</tr>
<tr>
<td>Join Across Lines</td>
<td>Default</td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>Default</td>
</tr>
<tr>
<td>BLF Audible Alert Setting (Phone Idle)</td>
<td>Default</td>
</tr>
<tr>
<td>BLF Audible Alert Setting (Phone Busy)</td>
<td>Default</td>
</tr>
<tr>
<td>Always Use Prime Line</td>
<td>Default</td>
</tr>
<tr>
<td>Always Use Prime Line for Voice Message</td>
<td>Default</td>
</tr>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

- Retry Video Call as Audio
- Ignore Presentation Indicators (internal calls only)
- Allow Control of Device from CTI
- Lagged into Hunt Group
- Remote Device
- Protected Device
- Hotline Device
- Require off-premise location

### Call Routing Information

#### Inbound Calls

<table>
<thead>
<tr>
<th>Calling Party Transformation CSS</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use Device Pool Callin Party Transformation CSS</td>
<td></td>
</tr>
</tbody>
</table>

#### Outbound Calls

<table>
<thead>
<tr>
<th>Calling Party Transformation CSS</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td></td>
</tr>
</tbody>
</table>
### Protocol Specific Information
- **Packet Capture Mode**: None
- **Packet Capture Duration**: 0
- **BLF Presence Group**: Standard Presence group
- **Device Security Profile**: Cisco 7800 - Standard SCCP Non-Secure Profile
- **SUBSCRIBE Calling Source Space**: < None >
- **Unattended Font**: [ ]
- **Requires DTMF Reception**: [ ]
- **RFC2833 Disabled**: [ ]

### Certification Authority Proxy Function (CAPF) Information
- **Certificate Operation**: No Pending Operation
- **Authentication Mode**: By null String
- **Authentication String**:  
- **Generate String**:  
- **Key Size (Bits)**: 1024
- **Operation Completes By**: 2012-07-12 (YYYY-MM-DD HH)
- **Certificate Operation Status**: None
- **Note**: Security Profile Contains Additional CAPF Settings.

### Expansion Module Information
- **Module 1**:  
  - **Module 1 Load Name**: < None >
- **Module 2**:  
  - **Module 2 Load Name**: < None >

### External Data Locations Information (Leave blank to use default)
- **Information**:  
- **Directory**:  
- **Messages**:  
- **Services**:  
- **Miscellaneous**:  
- **Optional**:  
- **Optional**:  
- **Optional**:  
- **Optional**:  
- **Optional**:  
- **Optional**:  
- **Optional**:  
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- **Opti...
<table>
<thead>
<tr>
<th>Authentication Server</th>
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</thead>
<tbody>
<tr>
<td>Proxy Server</td>
<td></td>
</tr>
<tr>
<td>Idle</td>
<td></td>
</tr>
<tr>
<td>Idle Timer (seconds)</td>
<td></td>
</tr>
<tr>
<td>Secure Authentication URL</td>
<td></td>
</tr>
<tr>
<td>Secure Directory URL</td>
<td></td>
</tr>
<tr>
<td>Secure Idle URL</td>
<td></td>
</tr>
<tr>
<td>Secure Information URL</td>
<td></td>
</tr>
<tr>
<td>Secure Messages URL</td>
<td></td>
</tr>
<tr>
<td>Secure Services URL</td>
<td></td>
</tr>
</tbody>
</table>

**Extension Information**

- Enable Extension Mobility
- Log Out Profile: Use Current Device Settings
- Log in Time: < None >
- Log out Time: < None >

**MLPP Information**

- MLPP Domain: < None >
- MLPP Indication: Default
- MLPP Preemption: Default

**Do Not Disturb**

- Do Not Disturb
- DND Option: Use Common Phone Profile Setting
- DND Incoming Call Alert: < None >

**Secure Shell Information**

- Secure Shell User: 
- Secure Shell Password: 
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Setting</th>
<th>Override Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disable Speakerphone</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Disable Speakerphone and Headset</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Forwarding Delay</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>PC Port</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Settings Access</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Gratuitous ARP</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>PC Voice VLAN Access</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Video Capabilities</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Auto Line Select</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Web Access</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Days Display Not Active</td>
<td>Sunday</td>
<td></td>
</tr>
<tr>
<td>Display On Time</td>
<td>07:30</td>
<td></td>
</tr>
<tr>
<td>Display On Duration</td>
<td>10:30</td>
<td></td>
</tr>
<tr>
<td>Display Idle Timeout</td>
<td>01:00</td>
<td></td>
</tr>
<tr>
<td>Enable Power Save Plus</td>
<td>Sunday</td>
<td></td>
</tr>
<tr>
<td>Phone On Time</td>
<td>09:00</td>
<td></td>
</tr>
<tr>
<td>Phone Off Time</td>
<td>24:00</td>
<td></td>
</tr>
<tr>
<td>Phone Idle Timeout</td>
<td>60</td>
<td></td>
</tr>
<tr>
<td>Enable Audible Alert</td>
<td></td>
<td></td>
</tr>
<tr>
<td>EnergyWise Domain</td>
<td></td>
<td></td>
</tr>
<tr>
<td>EnergyWise Endpoint Security Secret</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Allow EnergyWise Overrides</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Span to FC Port</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Logging Display</td>
<td>PC Controlled</td>
<td></td>
</tr>
</tbody>
</table>

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<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
<th>Enabled</th>
</tr>
</thead>
<tbody>
<tr>
<td>Display On When Incoming Call*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>RTCP*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>&quot;Incom&quot; Soft Key Timer</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Auto Call Select*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Advertise G.722 Codec*</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td>Wideband Headset UI Control*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Wideband Headset*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Peer Firmware Sharing*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): Switch Port*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): PC Port*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol - Media Endpoint Annex (LLDP-MED): Switch Port*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>LLDP Power Priority*</td>
<td>Unknown</td>
<td></td>
</tr>
<tr>
<td>Wireless Headset Hookswitch Control*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>IPv6 Load Server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>IPv6 Log Server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>802.1x Authentication*</td>
<td>User Controlled</td>
<td></td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure*</td>
<td>Normal</td>
<td></td>
</tr>
<tr>
<td>Minimum Ring Volume*</td>
<td>0-Silent</td>
<td></td>
</tr>
<tr>
<td>Headset Sidetone Level*</td>
<td>Normal</td>
<td></td>
</tr>
<tr>
<td>HTTPS Server*</td>
<td>Http and https Enabled</td>
<td></td>
</tr>
<tr>
<td>Handset/Headset Monitor*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Headset Recording*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>E911 Dialing*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Switch Port Remote Configuration*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>PC Port Remote Configuration*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Automatic Port Synchronization*</td>
<td>Disabled</td>
<td></td>
</tr>
</tbody>
</table>
Cisco IP Phone DN 4350

### Directory Number Configuration

<table>
<thead>
<tr>
<th>Status</th>
<th>Status: Ready</th>
</tr>
</thead>
</table>

### Directory Number Information

<table>
<thead>
<tr>
<th>Directory Number</th>
<th>4350</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Partition</td>
<td>phones</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>Alerting Name</td>
<td>CUCH 4350 (alert)</td>
</tr>
<tr>
<td>ASCII Alerting Name</td>
<td>CUCH 4350 (alert)</td>
</tr>
<tr>
<td>Allow Control of Device from CTI</td>
<td>✓</td>
</tr>
<tr>
<td>Associated Devices</td>
<td>SEP0024C40D7F41</td>
</tr>
</tbody>
</table>

**Edit Device**

**Edit Line Appearance**

### Directory Number Settings

<table>
<thead>
<tr>
<th>Voice Mail Profile</th>
<th>&lt; None &gt;</th>
<th>(Choose &lt;None&gt; to use system default)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>BLF Presence Group</td>
<td>Standard Presence group</td>
<td></td>
</tr>
<tr>
<td>User Hold MOH Audio Source</td>
<td>1-SampleAudioSource</td>
<td></td>
</tr>
<tr>
<td>Network Hold MOH Audio Source</td>
<td>1-SampleAudioSource</td>
<td></td>
</tr>
<tr>
<td>Auto Answer</td>
<td>Auto Answer Off</td>
<td></td>
</tr>
<tr>
<td>Reject Anonymous Calls</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Directory URIs

<table>
<thead>
<tr>
<th>Primary</th>
<th>URL</th>
<th>Partition</th>
<th>Remove</th>
</tr>
</thead>
<tbody>
<tr>
<td>✔️</td>
<td></td>
<td>&lt; None &gt;</td>
<td>-</td>
</tr>
</tbody>
</table>
### ACO Settings

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>AAR Destination Mask</th>
<th>AAR Group</th>
</tr>
</thead>
<tbody>
<tr>
<td>[ ] or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

☑ Retain this destination in the call forwarding history

### Call Forward and Call Pickup Settings

<table>
<thead>
<tr>
<th>Calling Search Space Activation Policy</th>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward All</td>
<td>[ ] or</td>
<td></td>
<td>sp_phones_rep</td>
</tr>
<tr>
<td>Secondary Calling Search Space for Forward All</td>
<td>[ ] or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Busy Internal</td>
<td>[ ] or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Busy External</td>
<td>[ ] or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Answer Internal</td>
<td>[ ] or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Answer External</td>
<td>[ ] or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Coverage Internal</td>
<td>[ ] or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Coverage External</td>
<td>[ ] or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward on CFI Failure</td>
<td>[ ] or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered Internal</td>
<td>[ ] or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered External</td>
<td>[ ] or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>No Answer Ring Duration (seconds)</td>
<td>[ ] or</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Call Pickup Group</td>
<td>[ ] or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Park Monitoring

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Park Monitoring Forward</td>
<td>[ ] or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>No Retrieve Destination External</td>
<td></td>
<td>A blank value means to call the parker's line.</td>
</tr>
<tr>
<td>Park Monitoring Forward</td>
<td>[ ] or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>No Retrieve Destination Internal</td>
<td></td>
<td>A blank value means to call the parker's line.</td>
</tr>
</tbody>
</table>
Note: Configure parameter **External Phone Number Mask** so as to provide the full 10-digit DID telephone number associated with the 4-digit DN.
VG224 Analog Gateway (FAX application)

<table>
<thead>
<tr>
<th>Gateway Details</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Product</td>
<td>VG224</td>
</tr>
<tr>
<td>Gateway</td>
<td>MGCF</td>
</tr>
<tr>
<td>Domain Name</td>
<td>VG224_Bench8</td>
</tr>
<tr>
<td>Description</td>
<td>VG224 on Bench 8</td>
</tr>
<tr>
<td>Cisco Unified Communications Manager Group</td>
<td>Default</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Configured Slots, VICS and Endpoints</th>
</tr>
</thead>
<tbody>
<tr>
<td>Module in Slot 2</td>
</tr>
<tr>
<td>Slot 0</td>
</tr>
<tr>
<td>2/6 2/7 2/8 2/9 2/10 2/11</td>
</tr>
<tr>
<td>2/12 2/13 2/14 2/25 2/26 2/27</td>
</tr>
<tr>
<td>2/10 2/19 2/20 2/21 2/22 2/23</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Specific Configuration Layout</th>
</tr>
</thead>
<tbody>
<tr>
<td>Modem Passthrough</td>
</tr>
<tr>
<td>Cisco Fax Relay</td>
</tr>
<tr>
<td>T38 Fax Relay</td>
</tr>
<tr>
<td>RTP Package Capability</td>
</tr>
<tr>
<td>MT Package Capability</td>
</tr>
<tr>
<td>RRS Package Capability</td>
</tr>
<tr>
<td>PRE Package Capability</td>
</tr>
<tr>
<td>SST Package Capability</td>
</tr>
<tr>
<td>RTP Unreachable OnOff</td>
</tr>
<tr>
<td>RTP Unreachable timeout (ms)</td>
</tr>
<tr>
<td>RTCP Report Interval (secs)</td>
</tr>
<tr>
<td>Simple SDP</td>
</tr>
</tbody>
</table>

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### VG224 Analog Gateway Endpoint

#### Gateway Configuration

**Status**
- **Status:** Ready

#### Directory Number Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Protocol</td>
<td>Analog Access</td>
</tr>
<tr>
<td>Gateway</td>
<td>VG224_Bench9</td>
</tr>
<tr>
<td>Product</td>
<td>Cisco MGCP FXS Port</td>
</tr>
<tr>
<td>Registration</td>
<td>Registered with Cisco Unified Communications Manager CM-Polaris</td>
</tr>
<tr>
<td>Bed-Point Name</td>
<td>AALN/SB/00/VG224_Bench9</td>
</tr>
<tr>
<td>Description</td>
<td>AALN/SB/00/VG224_Bench9</td>
</tr>
<tr>
<td>Device Pool</td>
<td>0711_Pool</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>PRQ_Medium</td>
</tr>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>10Phones_bp</td>
</tr>
<tr>
<td>AAR, Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Location</td>
<td>Hub_None</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Network Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>Default</td>
</tr>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Transmit Uifi</td>
<td>False</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Hot line Device</td>
<td>False</td>
</tr>
<tr>
<td>Device is trusted</td>
<td>False</td>
</tr>
</tbody>
</table>

#### Multilevel Precedence and Preemption (MLPP) Information

- **MLPP Domain:** < None >
- **MLPP Indication:** Not available on this device
- **MLPP Preemption:** Not available on this device

#### Port Information (POTS)

- **Port Direction:** Bothways
- **Prefix DN:**
- **Num Digits:** 0
- **Expected Digits:** 0
- **SMDI Port Number:**<none>
### AAR Settings

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>AAR Destination Mask</th>
<th>AAR Group</th>
</tr>
</thead>
<tbody>
<tr>
<td>AAR</td>
<td>□ or</td>
<td>□ or</td>
</tr>
</tbody>
</table>

- Select if you want to retain the destination in the call forwarding history.

### Call Forward and Call Pickup Settings

<table>
<thead>
<tr>
<th>Calling Search Space Activation Policy</th>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward All</td>
<td>□ or</td>
<td>□ or</td>
<td>ts_phones_rp</td>
</tr>
<tr>
<td>Secondary Calling Search Space for Forward All</td>
<td>□ or</td>
<td>□ or</td>
<td>ts_phones_rp</td>
</tr>
<tr>
<td>Forward Busy Internal</td>
<td>□ or</td>
<td>□ or</td>
<td>ts_phones_rp</td>
</tr>
<tr>
<td>Forward Busy External</td>
<td>□ or</td>
<td>□ or</td>
<td>ts_phones_rp</td>
</tr>
<tr>
<td>Forward No Answer Internal</td>
<td>□ or</td>
<td>□ or</td>
<td>ts_phones_rp</td>
</tr>
<tr>
<td>Forward No Answer External</td>
<td>□ or</td>
<td>□ or</td>
<td>ts_phones_rp</td>
</tr>
<tr>
<td>Forward No Coverage Internal</td>
<td>□ or</td>
<td>□ or</td>
<td>ts_phones_rp</td>
</tr>
<tr>
<td>Forward No Coverage External</td>
<td>□ or</td>
<td>□ or</td>
<td>ts_phones_rp</td>
</tr>
<tr>
<td>Forward on CTE Failure</td>
<td>□ or</td>
<td>□ or</td>
<td>ts_phones_rp</td>
</tr>
<tr>
<td>Forward Unregistered Internal</td>
<td>□ or</td>
<td>□ or</td>
<td>ts_phones_rp</td>
</tr>
<tr>
<td>Forward Unregistered External</td>
<td>□ or</td>
<td>□ or</td>
<td>ts_phones_rp</td>
</tr>
</tbody>
</table>

- No Answer Ring Duration (seconds)

### Park Monitoring

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Park Monitoring Forward No Retrieve Destination Internal</td>
<td>□ or</td>
<td>□ or</td>
</tr>
</tbody>
</table>

- Park Monitoring Forward No Retrieve Destination External

- A blank value means to call the parker's line.
Note: Configure parameter **External Phone Number Mask** so as to provide the full 10-digit DID telephone number associated with the 4-digit DN.

**IOS Configuration of VG224 Analog Gateway**

VG224_Bench8#sho run

Building configuration...

Current configuration : 2022 bytes

! Last configuration change at 00:30:21 UTC Mon Mar 1 1993
version 15.1
no service pad
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption

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hostname VG224_Bench8

! boot-start-marker
boot-end-marker
!
!
enable secret 5 $1$i0OSfyr2wLz/5awVuiQbES6Q.

! no aaa new-model
ip source-route
!
ip cef
ip host CM-Polaris 172.20.236.50
ip name-server 172.20.2.181
!
!
no ipv6 cef
!
!
voice-card 0
!
!
interface FastEthernet0/0
ip address 172.20.236.102 255.255.255.0
duplex full
speed 100
!
interface FastEthernet0/1
no ip address
shutdown
duplex auto
speed auto
!
!
ip forward-protocol nd
!
no ip http server
ip route 0.0.0.0 0.0.0.0 172.20.236.1
!
control-plane
!
!
voice-port 2/0
!
voice-port 2/1
!
voice-port 2/2
!
voice-port 2/3
!
voice-port 2/4
!
voice-port 2/5
!
voice-port 2/6
!
voice-port 2/7
!
voice-port 2/8
!
voice-port 2/9
!
voice-port 2/10
!
voice-port 2/11
!
voice-port 2/12
!
voice-port 2/13
!
voice-port 2/14
!
voice-port 2/15
!
voice-port 2/16
!
voice-port 2/17
!
voice-port 2/18
!
voice-port 2/19
!
voice-port 2/20
!
voice-port 2/21
!
voice-port 2/22
!
voice-port 2/23
!
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server 172.20.236.50
ccm-manager config
!
mgcp
mgcp call-agent CM-Polaris 2427 service-type mgcp version 0.1
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp default-package mt-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax rate 14400
mgcp fax t38 ecm7
mgcp fax t38 ls_redundancy 5
mgcp fax t38 hs_redundancy 1

7 This command enables T.38 Fax relay Error Correction Mode (ECM)
mgcp rtp payload-type g726r16 static
!
mgcp profile default
!
!
dial-peer voice 99920 pots
  service mgcpapp
  port 2/0
!
dial-peer voice 99921 pots
  service mgcpapp
  port 2/1
!
!
line con 0
  transport output all
line aux 0
  transport output all
line vty 0 4
  password cisco
  login
  transport input all
  transport output all
!
end
IOS Conference Bridge

Note: CISCO UCM requires a conference bridge resource for 3-way conference calls.
Note: Transcoder resources are required for calls using different codecs.

IOS Configuration of Conference Bridge and Transcoder

3825DSPfarm#sho run
Building configuration...

Current configuration : 3460 bytes
!
! Last configuration change at 17:56:03 UTC Wed Aug 29 2012
!
version 15.0
service config
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname 3825DSPfarm
!
boot-start-marker
boot system flash:c3825-ipvoicek9-mz.150-1.XA1.bin
boot-end-marker
!
! card type command needed for slot/vwic-slot 1/1
logging buffered 1000000
enable password cisco
!
no aaa new-model
no network-clock-participate slot 1
no network-clock-participate slot 2

voice-card 0
dspfarm
dsp services dspfarm

voice-card 1

voice-card 2
dspfarm

dot11 syslog
ip source-route
ip cef

ip domain name pbxlab.org
ip host CM-Polaris 172.20.236.50
ip host CM-Neptune 172.20.236.2
ip name-server 172.20.2.181
no ipv6 cef
multilink bundle-name authenticated

license udi pid CISCO3825 sn FTX0946A1BS
archive
log config
hidekeys
username cisco privilege 15 secret 5 $1sd9XF$5CFki/4rtzvcZ1iLmeghx/

interface GigabitEthernet0/0
description $ETH-LAN$ETH-SW-LAUNCH$INTF-INFO-GE 0/0$
ip address 172.20.236.101 255.255.255.0
duplex auto
speed auto
media-type rj45

interface GigabitEthernet0/1
no ip address
shutdown
duplex auto
speed auto
media-type rj45

ip default-gateway 172.20.236.1
ip forward-protocol nd

ip http server
ip http authentication local
no ip http secure-server
ip http timeout-policy idle 5 life 86400 requests 10000
!
ip route 0.0.0.0 0.0.0.0 GigabitEthernet0/0
!
control-plane
!
!
mgcp profile default
!
sccp local GigabitEthernet0/0
sccp ccm 172.20.236.50 identifier 2 version 6.0
sccp ccm 172.20.236.2 identifier 1 version 4.0
sccp ip precedence 3
sccp
!
sccp ccm group 10
associate ccm 2 priority 1
associate profile 10 register MTP0015f90d0970
associate profile 12 register CFB112233445566
keepalive retries 5
!
dspfarm profile 10 transcode
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
codec g729r8
codec g729br8
codec pass-through
maximum sessions 10
associate application SCCP
!
dspfarm profile 12 conference
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
codec g729r8
codec g729br8
maximum sessions 10
associate application SCCP
!
!
gateway
timer receive-rtp 1200
!
!
line con 0
line aux 0
line vty 0 4
password cisco
login
!
scheduler allocate 20000 1000
end
Music on Hold Server

### Device Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Registration</td>
<td>Registered with Cisco Unified Communications Manager CM-Polaris</td>
</tr>
<tr>
<td>IP Address</td>
<td>172.28.236.50</td>
</tr>
<tr>
<td>Description</td>
<td>MOH_2</td>
</tr>
<tr>
<td>Device Pool</td>
<td>MOH_CM-POLARIS</td>
</tr>
<tr>
<td>Host Server</td>
<td>CM-Polaris</td>
</tr>
<tr>
<td>Music On Hold Server Name</td>
<td>MOH_2</td>
</tr>
<tr>
<td>Description</td>
<td>MOH_CM-POLARIS</td>
</tr>
<tr>
<td>Device Pool</td>
<td>Default</td>
</tr>
<tr>
<td>Location</td>
<td>Hub, None</td>
</tr>
<tr>
<td>Maximum Multicast Streams</td>
<td>256</td>
</tr>
<tr>
<td>Maximum Multicast Connections</td>
<td>30</td>
</tr>
<tr>
<td>Fixed Audio Source Device</td>
<td></td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>Off</td>
</tr>
<tr>
<td>Run Flag</td>
<td>Yes</td>
</tr>
</tbody>
</table>

### Multi-cast Audio Source Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Base Multi-cast IP Address</td>
<td>0.0.0.0</td>
</tr>
<tr>
<td>Base Multi-cast Port Number</td>
<td>0</td>
</tr>
<tr>
<td>Increment Multi-cast on</td>
<td>0 Port Number</td>
</tr>
<tr>
<td>Increment Multi-cast IP Address</td>
<td>(Even numbers only)</td>
</tr>
</tbody>
</table>

### Selected Multi-cast Audio Sources

There are no Music On Hold Audio Sources selected for Multi-cast. Click Continue Audio Source in the top right corner of the page to select Multi-cast Audio Source.
### Annunciator

#### Annunciator Configuration

- **Status**: Ready

#### Annunciator Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP Address</td>
<td>172.10.236.50</td>
</tr>
<tr>
<td>Device Name</td>
<td>CM-Polaris</td>
</tr>
<tr>
<td>Description</td>
<td>ANN_CM_POLARIS</td>
</tr>
<tr>
<td>Device Port</td>
<td>Default</td>
</tr>
<tr>
<td>Location</td>
<td>Hub_None</td>
</tr>
<tr>
<td>Use Trusted Key Point</td>
<td>Off</td>
</tr>
</tbody>
</table>
### Music on Hold Service (IP Voice Media Streaming App) Parameter Settings

#### Cisco IP Voice Media Streaming App (Active) Parameters on server CM-Polaris (Active)

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Parameter Value</th>
<th>Suggested Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>- Annunciation (ANN) Parameters</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Call Count</td>
<td>46</td>
<td>48</td>
</tr>
<tr>
<td>Run Flag</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>- Conference Bridge (CFB) Parameters</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Call Count</td>
<td>48</td>
<td>49</td>
</tr>
<tr>
<td>Run Flag</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>- Media Termination Point (MTP) Parameters</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Call Count</td>
<td>48</td>
<td>48</td>
</tr>
<tr>
<td>Run Flag</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>- Clusterwide Parameters (Parameters that apply to all servers)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Supported MOH Codecs</td>
<td>711 mulaw</td>
<td>711 mulaw</td>
</tr>
<tr>
<td>MOH Fixed Audio Quality Level</td>
<td>Medium Quality</td>
<td>Medium Quality</td>
</tr>
<tr>
<td>IP DSCP to Cisco Unified Communications Manager</td>
<td>AFI2 DSCP (101010)</td>
<td>CS2(precedence 3) DSCP (101100)</td>
</tr>
<tr>
<td>Multicast MOH IP DSCP</td>
<td>AFI3 DSCP (101010)</td>
<td>EF DSCP (101110)</td>
</tr>
<tr>
<td>MTP DTMF Duration</td>
<td>100</td>
<td>100</td>
</tr>
</tbody>
</table>

Note: Make sure codecs G.729 Annex A and G.711 mulaw are configured in parameter **Supported MOH Codecs**.
### Route Pattern

#### Find and List Route Patterns

- Add New
- Select All
- Clear All
- Delete Selected

#### Status

1 record found

<table>
<thead>
<tr>
<th>Route Patterns (1 of 5)</th>
<th>Find Route Patterns where</th>
<th>Pattern</th>
<th>Description</th>
<th>Partition</th>
<th>Route Filter</th>
<th>Associated Device</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>9.*7911</td>
<td></td>
<td>9.*7911</td>
<td>Enhanced FlexReach feature deactivation code</td>
<td>route p</td>
<td></td>
<td>ATT SIP Trunk</td>
<td></td>
</tr>
<tr>
<td>9.<em>790</em></td>
<td></td>
<td>9.<em>790</em></td>
<td>Enhanced FlexReach feature activation access codes</td>
<td>route p</td>
<td></td>
<td>ATT SIP Trunk</td>
<td></td>
</tr>
<tr>
<td>9.*9</td>
<td></td>
<td>9.*9</td>
<td>To ATT Operator</td>
<td>route p</td>
<td></td>
<td>ATT SIP Trunk</td>
<td></td>
</tr>
<tr>
<td>9.*</td>
<td></td>
<td>9.*</td>
<td>To PSTN via ATT IP FlexReach SIP Trunk</td>
<td>route p</td>
<td></td>
<td>ATT SIP Trunk</td>
<td></td>
</tr>
<tr>
<td>9.*N11</td>
<td></td>
<td>9.*N11</td>
<td>Calls to N11 service over ATT SIP Trunk</td>
<td>route p</td>
<td></td>
<td>ATT SIP Trunk</td>
<td></td>
</tr>
</tbody>
</table>
Translation Pattern for 10 Incoming Called-Number to 4-Digit Extension (1 of 2)

**Translation Pattern Configuration**

**Status**
- Status: Ready

**Pattern Definition**
- Translation Pattern: 06030732320.XXXK
- Description: SIP trunk prefixes 10-digit DID numbers with a string of 5 zeros. Translation Pattern(s) used to convert incoming numbers to 4-digit DN's must contain the full prefix (including the zeros).

**Calling Party Transformations**
- Use Calling Party's External Phone Number Mask
- Calling Party Transform Mask
- Prefix Digits (Outgoing Calls)
- Calling Line ID Presentation: Default
- Calling Name Presentation: Default
- Calling Party Number Type: Cisco CallManager
- Calling Party Numbering Plan: Cisco CallManager

**Connected Party Transformations**
- Connected Line ID Presentation: Default
- Connected Name Presentation: Default

**Called Party Transformations**
- Discard Digits
- Called Party Transform Mask
- Prefix Digits (Outgoing Calls)
- Called Party Number Type: Cisco CallManager
- Called Party Numbering Plan: Cisco CallManager

Note: AT&T IP Flexible Reach-EF SIP trunk prefixes 10-digit DID numbers with a string of 5 zeros. Translation Pattern(s) used to convert incoming numbers to 4-digit DN's must contain the full prefix (including the zeros).
### Trunk Configuration

**Status**: Ready

<table>
<thead>
<tr>
<th>Device Information</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Trunk</td>
<td></td>
</tr>
<tr>
<td>SIP</td>
<td></td>
</tr>
<tr>
<td>Trunk Service Type</td>
<td>None (Default)</td>
</tr>
<tr>
<td>Device Name*</td>
<td>ATT_SIP_Trunk</td>
</tr>
<tr>
<td>Description</td>
<td>ATT SIP trunk to PSTN</td>
</tr>
<tr>
<td>Device Pool*</td>
<td>Default</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Call Classification*</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRLG_Peralta</td>
</tr>
<tr>
<td>Location*</td>
<td>Hub_Location</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Tunnelled Protocol*</td>
<td>None</td>
</tr>
<tr>
<td>QSIG Variant*</td>
<td>No Changes</td>
</tr>
<tr>
<td>ASTUN ROSE OID Encoding*</td>
<td>No Changes</td>
</tr>
<tr>
<td>Packet Capture Mode*</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
</tbody>
</table>

- Media Termination Point Required
- Rovi Video Call as Audio
- Path Replacement Support
- Transmit UTIF-8 for Calling Party Name
- Transmit UTIF-8 Names in QSIG AFDU
- Unattended Port
- SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end-to-end security. Failure to do so will expose keys and other information.
- Consider Traffic on This Trunk Secure* - When using both SRTP and TLS
- Route Class Signaling Enabled* - Default
- Use Trusted Relay Point* - Default
### Intercompany Media Engine (IME)

- E.164 Transformation Profile: `<Name>`

### Multilevel Precedence and Preemption (MLPP) Information

- MLPP Domain: `<None>`

### Call Routing Information

- Remote-Party-Id
- Asserted-Identity
- Assorted-Type: `[PST]`
- SIP Privacy: `[Default]`

### Inbound Cells

<table>
<thead>
<tr>
<th>Significant Digits</th>
<th><code>All</code></th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Line ID Presentation</td>
<td><code>[Default]</code></td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td><code>[Default]</code></td>
</tr>
<tr>
<td>Calling Search Space</td>
<td><code>tp_phones_wp</code></td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td><code>&lt;None&gt;</code></td>
</tr>
<tr>
<td>Prefix On</td>
<td><code>No</code></td>
</tr>
</tbody>
</table>

### Redirecting Diversion Header Delivery - Inbound

#### Incoming Calling Party Settings

If the administrator sets the prefix to `Default`, this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

<table>
<thead>
<tr>
<th>Prefix Settings</th>
<th>Default Prefix Settings</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Incoming Number</strong></td>
<td><code>Default</code></td>
<td>0</td>
</tr>
</tbody>
</table>

#### Connected Party Settings

- Connected Party Transformation CSS: `<None>`
- Use Device Pool Connected Party Transformation CSS: `Yes`
Note: A SIP Normalization Script is used to convert SIP Diversion headers from 4-digit DN’s to the full 10-digit E.164 telephone number, required for call diversions over AT&T IP Flexible Reach-EF SIP trunk.
SIP Normalization Script Configuration

M = {}

local mask = scriptParameters.getValue("Diversion-Mask")
-- handle the mask of the diversion header for non-911 calls

function M.outbound_INVITE(message)
  if mask
    then
      message:applyNumberMask("Diversion", mask)
    end
  end
return M

SIP Normalization Script (Text)

M = {}

local mask = scriptParameters.getValue("Diversion-Mask")
-- handle the mask of the diversion header for non-911 calls

function M.outbound_INVITE(message)
  if mask
    then
      message:applyNumberMask("Diversion", mask)
    end
  end
return M
Note: Inbound calls from AT&T need to have the prefix “stripped” in order to match 4-digit DN’s. This is accomplished by sending calls through a Translation Profile. Assign a Calling Search Space routing calls to the partition assigned to the Translation Pattern configured to strip the leading prefix.
SIP Profile used by SIP trunk to CISCO UBE

<table>
<thead>
<tr>
<th>SIP Profile Configuration</th>
<th>Related Links: Back To Find/List</th>
</tr>
</thead>
</table>

**SIP Profile Information**

- **Name**
- **Description**
- **Default MTP Telephony Event Payload Type**
- **Early Offer for G-Clear Calls**
- **SDP Session-level Bandwidth Modifier for Early Offer and Re-Invites**
- **User-Agent and Server header Information**
- **Accept Audio Codec Preferences in Received Offer**
- **Dial String Interpretation**
  - Redirect by Application
  - Disable Early Media on 180
  - Outgoing T.38 INVITE include audio mime
  - Enable ANAT
  - Require SDP Inactive Exchange for Mid-Call Media Change
  - Use Fully Qualified Domain Name in SIP Requests
  - Assured Services SIP conformance

**Parameters used in Phone**

- **Timer Invite Expires (seconds)**
- **Timer Register Delta (seconds)**
- **Timer Register Expires (seconds)**
- **Timer T1 (msec)**
- **Timer T2 (msec)**
- **Retry INVITE**
- **Retry Non-INVITE**
- **Start Media Port**
- **Stop Media Port**
- **Call Pickup URI**

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### Meet Me Service URI*

<table>
<thead>
<tr>
<th>User Info</th>
<th>Nominal</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Hold Ring Back*</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block*</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking*</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control*</td>
<td>User</td>
</tr>
<tr>
<td>TFTP Level for 7940 and 7960*</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Resource Priority Namespace</td>
<td></td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)*</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)*</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)*</td>
<td>0</td>
</tr>
<tr>
<td>Maximum Redirections*</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)*</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI*</td>
<td>x-cisco-serviceuri-meetme</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI*</td>
<td>x-cisco-serviceuri-default</td>
</tr>
</tbody>
</table>

#### Conference Join Enabled
- [x] RFC 2543 Hold
- [ ] Semi Attended Transfer
- [ ] Enable Wall
- [ ] Silent Message Warning
- [ ] MLPP User Authorization

#### Normalization Script

<table>
<thead>
<tr>
<th>Normalization Script</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable Trace</td>
<td>Parameter Name</td>
</tr>
<tr>
<td>1</td>
<td></td>
</tr>
</tbody>
</table>
Note: Some PSTN network call prompters utilize early-media cut-through to offer menu options to the caller (DTMF select menu) before the call is connected. In order for CISCO UCM/CISCO UBE solution to achieve successful early-media cut-through, the CISCO UCM to CISCO UBE call leg must be enabled with SIP PRACK. To enable SIP PRACK on the CISCO UCM, the SIP Profile “SIP Rel1XX Options” setting must be set to “Send PRACK”.
### Configuring Cisco Unified Communications Manager (CISCO UCM) for Cisco Unity Connection (CUC) SCCP integration

#### Voicemail Ports

![Cisco Unified CM Administration](image)

**Find and List Voicemail Ports**

- Status: 2 records found

**Voice Mail Port (1 - 2 of 2)**

<table>
<thead>
<tr>
<th>Device Name</th>
<th>Description</th>
<th>Device Pool</th>
<th>Device Security Mode</th>
<th>Calling Search Space</th>
<th>Extension</th>
<th>Partition</th>
<th>Status</th>
<th>IP Address</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>CiscoUM4-VTL</td>
<td>UC-PBXLab</td>
<td>0711 Pool</td>
<td>Non-Secure Voice Mail Port</td>
<td>voicemail</td>
<td>8053</td>
<td>phones</td>
<td>Registered with CM-Pools</td>
<td>172.20.236.109</td>
<td></td>
</tr>
<tr>
<td>CiscoUM4-VTR</td>
<td>UC-PBXLab</td>
<td>0711 Pool</td>
<td>Non-Secure Voice Mail Port</td>
<td>voicemail</td>
<td>8054</td>
<td>phones</td>
<td>Registered with CM-Pools</td>
<td>172.20.236.109</td>
<td></td>
</tr>
</tbody>
</table>
### Voicemail Port Configuration

<table>
<thead>
<tr>
<th><strong>Status</strong></th>
<th>Status: Ready</th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th><strong>Device Information</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Registration:</td>
<td>Registered with Cisco Unified Communications Manager GM-Polaris</td>
</tr>
<tr>
<td>IP Address</td>
<td>172.20.236.195</td>
</tr>
<tr>
<td>Device Name</td>
<td>CiscoUM4-V71</td>
</tr>
<tr>
<td>Description</td>
<td>UC-PBXLab</td>
</tr>
<tr>
<td>Device Pool</td>
<td>G711_Pool</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>voicemail</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Location</td>
<td>Hub_None</td>
</tr>
<tr>
<td>Device Security Mode</td>
<td>Non Secure Voice Mail Port</td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>Default</td>
</tr>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>Directory Number Information</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Directory Number</td>
<td>8500</td>
</tr>
<tr>
<td>Partition</td>
<td>Phones</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>voicemail</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Internal Caller ID Display</td>
<td>VoiceMail</td>
</tr>
<tr>
<td>Internal Caller ID Display (ASCII format)</td>
<td>VoiceMail</td>
</tr>
<tr>
<td>External Number Mask</td>
<td></td>
</tr>
</tbody>
</table>
## Message Waiting Numbers

**Cisco Unified CM Administration**

For Cisco Unified Communications Solutions

<table>
<thead>
<tr>
<th>Status</th>
<th>2 records found</th>
</tr>
</thead>
</table>

### Find and List Message Waiting Numbers

<table>
<thead>
<tr>
<th>Directory Number</th>
<th>Description</th>
<th>Partition</th>
<th>Calling Search Space</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>8001</td>
<td>MWI on for CUCM stations</td>
<td>phones</td>
<td>no_phones</td>
<td>Delete Selected</td>
</tr>
<tr>
<td>8002</td>
<td>MWI off for CUCM stations</td>
<td>phones</td>
<td>no_phones</td>
<td>Delete Selected</td>
</tr>
</tbody>
</table>
Voicemail Hunt Pilot

### Pattern Definition

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hunt Pilot*</td>
<td>5075</td>
</tr>
<tr>
<td>Route Partition</td>
<td>phones</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>&lt; Name &gt;</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt; Name &gt;</td>
</tr>
<tr>
<td>MLPP Precedence*</td>
<td>Default</td>
</tr>
<tr>
<td>Hunt List*</td>
<td>UC_PBXLab</td>
</tr>
<tr>
<td>Call Pickup Group</td>
<td>&lt; Name &gt;</td>
</tr>
<tr>
<td>Alerting Name</td>
<td>Unity Connection - PBXLab</td>
</tr>
<tr>
<td>ASCII Alerting Name</td>
<td>Unity Connection - PBXLab</td>
</tr>
<tr>
<td>Route Option</td>
<td></td>
</tr>
<tr>
<td>✓ Provide Outside Dial Time</td>
<td></td>
</tr>
<tr>
<td>✓ Urgent Priority</td>
<td></td>
</tr>
</tbody>
</table>

### Hunt Call Treatment Settings

- **Forward Hunt No Answer**
  - Do Not Forward Unanswered Calls
  - Use Forward Settings of Line Group Member
  - Forward Unanswered Calls to
    - Destination:
    - Calling Search Space: < Name >
    - Maximum Hunt Timer: 0
### Queueing

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Network Hold MOH Source &amp; Announcements</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Maximum Number of Callers Allowed in Queue*</td>
<td>82</td>
<td>(1-100)</td>
</tr>
<tr>
<td>When Queue is full:</td>
<td></td>
<td></td>
</tr>
<tr>
<td>- Disconnect the call</td>
<td></td>
<td></td>
</tr>
<tr>
<td>- Route the call to this destination</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Full Queue Calling Search Space</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Maximum Wait Time in Queue*</td>
<td>900</td>
<td>(10 - 3600 seconds)</td>
</tr>
<tr>
<td>When maximum wait time is met:</td>
<td></td>
<td></td>
</tr>
<tr>
<td>- Disconnect the call</td>
<td></td>
<td></td>
</tr>
<tr>
<td>- Route the call to this destination</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Maximum Wait Time Calling Search Space</td>
<td></td>
<td></td>
</tr>
<tr>
<td>When no hunt members are logged in or registered:</td>
<td></td>
<td></td>
</tr>
<tr>
<td>- Disconnect the call</td>
<td></td>
<td></td>
</tr>
<tr>
<td>- Route the call to this destination</td>
<td></td>
<td></td>
</tr>
<tr>
<td>No Hunt Members logged in or registered Calling Search Space</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
</tbody>
</table>

### Park Monitoring

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Park Monitoring</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Forward No Retrieve Destination</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>&lt; None &gt;</td>
<td>A blank value means to use the parker's Park Monitoring Forward No Retrieve Destination settings.</td>
</tr>
</tbody>
</table>

### Calling Party Transformations

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use Calling Party’s External Phone Number Mask</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Calling Party Transform Mask</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Prefix Digits</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Calling Line 1D Presentation*</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Calling Name Presentation*</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Calling Party Number Type*</td>
<td>Cisco CallManager</td>
<td></td>
</tr>
<tr>
<td>Calling Party Numbering Plan*</td>
<td>Cisco CallManager</td>
<td></td>
</tr>
</tbody>
</table>

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### Calling Party Transformations

- Use Calling Party's External Phone Number Mask
- Calling Party Transform Mask
- Prefix Digits
- Calling Line ID Presentation: Default
- Calling Name Presentation: Default
- Calling Party Number Type: Cisco CallManager
- Calling Party Numbering Plan: Cisco CallManager

### Connected Party Transformations

- Connected Line ID Presentation: Default
- Display Line Group Member DN as Connected Party
- Connected Name Presentation: Default

### Called Party Transformations

- Different Digits: < None >
- Called Party Transform Mask
- Prefix Digits
- Called Party Number Type: Cisco CallManager
- Called Party Numbering Plan: Cisco CallManager

### AAR Group Settings

- AAR Group: < None >
- External Number Mask
## Voicemail Line Group

### Line Group Configuration

<table>
<thead>
<tr>
<th>Line Group Information</th>
<th>Line Group Name*</th>
<th>UC_PNSLab</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>RNA Reversion Timeout*</td>
<td>10</td>
</tr>
<tr>
<td></td>
<td>Distribution Algorithm*</td>
<td>Longest Idle Time</td>
</tr>
</tbody>
</table>

#### Hunt Options

- **No Answer***: Try next member; then, try next group in Hunt List
- **Busy***: Try next member; then, try next group in Hunt List
- **Not Available***: Try next member; then, try next group in Hunt List

### Line Group Member Information

#### Find Directory Numbers to Add to Line Group

<table>
<thead>
<tr>
<th>Partition</th>
<th>Directory Number Contains</th>
<th>Find</th>
</tr>
</thead>
<tbody>
<tr>
<td>All</td>
<td></td>
<td></td>
</tr>
<tr>
<td>1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>2</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

#### Available DN/Route Partition

- 1232/phones
- 4000/phones
- 5000/phones
- 6855/phones
- 6854/phones

<table>
<thead>
<tr>
<th>Add to Line Group</th>
</tr>
</thead>
</table>

### Current Line Group Members

#### Reverse Order of Selected DN/Route Partitions

<table>
<thead>
<tr>
<th>Selected DN/Route Partition</th>
<th>Removed DN/Route Partition</th>
</tr>
</thead>
<tbody>
<tr>
<td>6855/phones</td>
<td></td>
</tr>
<tr>
<td>6854/phones</td>
<td></td>
</tr>
</tbody>
</table>

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Configuring Cisco Unity Connection (CUC) for Connection to CISCO UCM using SCCP Integration

CUC Version
CUC Telephony Integration with CISCO UCM

Phone System Basics (CM-Polaris-SCCP)

Phone System
Phone System Name*: CM-Polaris-SCCP

Message Waiting Indicators
- Send Message Counts
- Use Same Port for Enabling and Disabling MWIs
- Force All MWIs Off for this Phone System
- Synchronize All MWIs on This Phone System

Call Loop Detection by Using DTMF
- Enable for Supervised Transfers
- Enable for Forwarded Message Notification Calls (by Using DTMF)

DTMF Time To Use: A
Guard Time: 3500 milliseconds

Call Loop Detection by Using Extension
- Enable for Forwarded Message Notification Calls (by Using Extension)

Phone View Settings
- CTI Phone Access Username
- CTI Phone Access Password

Outgoing Call Restrictions
- Enable outgoing calls
- Disable all outgoing calls immediately
- Disable all outgoing calls between

Beginning Time: 12:00 AM
Ending Time: 12:00 AM

Save Delete Previous Next

Fields marked with an asterisk (*) are required.
CUC Port Group Settings

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Port Group Basics (CM-Polaris-SCCP-1)

- Port Group
  - Display Name*: CM-Polaris-SCCP-1
  - Integration Method: SCCP (Skinny)
  - Device Name Prefix: CiscoUCCX-200
  - Reset Status: Reset Not Required

Message Waiting Indicator Settings
- Enable Message Waiting Indicators
  - MWI On Extension: 8001
  - MWI Off Extension: 8002
  - Delay Between Requests: 0 milliseconds
  - Maximum Concurrent Requests: 0
  - Retries After Successful Attempt: 0
  - Retry Interval After Successful Attempt: 0 milliseconds

Related Links: Add Ports
IOS Fax Gateway Configuration (Optional)

Below are configuration details for deploying an IOS gateway for Fax applications instead of the VG224. In this example, the IOS gateway is configured as a SIP gateway, using a SIP trunk for connectivity to Cisco UCM.

**SIP Trunk to Fax Gateway**

![Cisco Unified CM Administration Interface](image)

- **Device Information**
  - Product: SIP Trunk
  - Device Protocol: SIP
  - Trunk Service Type: None (Default)
  - Device Name: FAX_GW
  - Description: SIP Trunk to FAX gateway
  - Common Device Configuration
    - Call Classification: Use System Default
    - Media Resource Group List: MRG_Fax
  - Location: Pub_None
  - AAR Group: < Name >
  - Tunnelled Protocol: None
  - QSIG Variant: No Changes
  - ASCI ROSE DID Encoding: No Changes
  - Packet Capture Mode: None
  - Packet Capture Duration: 0

- **Other Settings**
  - Media Termination Point Required
  - Retry Video Call as Audio
  - Path Replacement Support
  - Transmit UTF-8 for Calling Party Name
  - Transmit UTF-8 Names in QSIG AFDU
  - Unattended Port

- **SRTP Supported**
  - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end-to-end security. Failure to do so will expose keys and other information.

- **Consider Traffic on This Trunk Secure**
  - When using both SRTP and TLS

- **Route Class Signaling Enabled**
  - Default
### Trunk Configuration

**Intercompany Media Engine (IME)**
- E.164 Transformation Profile: <None>

**Multilevel Precedence and Preemption (MLPP) Information**
- MLPP Domain: <None>

### Call Routing Information
- Remote-Party-id
- Assured-Identity
- Asserted-Type: Default
- SIP Privacy: Default

#### Inbound Calls
- Significant Digits: All
- Connected Line ID Presentation: Default
- Connected Name Presentation: Default
- Calling Search Space: tp_phones_rp
- AAR Calling Search Space: tp_phones_rp
- Prefix DN: 

#### Redirecting Diversion Header Delivery - Inbound

### Incoming Calling Party Settings
- If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>None</td>
<td>&lt;None&gt;</td>
<td></td>
</tr>
</tbody>
</table>

### Connected Party Settings
- Connected Party Transformation CSS: <None>
- Use Device Pool Connected Party Transformation CSS: 

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Note: Unless analog lines on the IOS gateway are configured with valid 10-digit DID numbers, configure a valid 10-digit DID telephone number in parameter “Caller ID DN”. Outbound calls to IP Flexible Reach-EF SIP trunk will not be processed unless a valid AT&T-provided DID number is passed as the calling party ID.
### Trunk Configuration

**SIP Information**

<table>
<thead>
<tr>
<th>Destination Address</th>
<th>Destination Address</th>
<th>Destination Address IP</th>
<th>Destination Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>172.20.169.201</td>
<td></td>
<td></td>
<td>5060</td>
</tr>
</tbody>
</table>

- **MTP Preferred Originating Code**: 711-law
- **BLF Presence Group**: Standard Presence group
- **SIP Trunk Security Profile**: SIP Trunks
- **Routing Calling Search Space**: < None >
- **Out-Of-Box Refer Calling Search Space**: < None >
- **SUBSCRIBE Calling Search Space**: < None >
- **SIP Profile**: Standard SIP Profile
- **DTMF Signaling Method**: RFC 2833

**Normalization Script**

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Geolocation Configuration**

- **Geolocation**: < None >
- **Geolocation Filter**: < None >

**Send Geolocation Information**: [ ]
## Route Pattern to IOS Fax Gateway

**Cisco Unified CM Administration**
For Cisco Unified Communications Solutions

### Route Pattern Configuration

<table>
<thead>
<tr>
<th>Pattern Definition</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>#35[12]</td>
</tr>
<tr>
<td>Route Partition</td>
<td>route_p</td>
</tr>
<tr>
<td>Description</td>
<td>To ISR 2011 IOS Fax Gateway</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>MLPP Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Apply Call Blocking Percentage</td>
<td></td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Route Class</td>
<td>Default</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>FAX_GW</td>
</tr>
<tr>
<td>Route Option</td>
<td>Route this pattern</td>
</tr>
<tr>
<td>Call Classification</td>
<td>OnNet</td>
</tr>
</tbody>
</table>

- Allow Device Override
- Provide Outside Dial Tone
- Allow Overlap Sending
- Urgent Priority
- Require Forced Authorization Code
- Authorization Level | 0

- Require Client Matter Code
Configuring Cisco IOS Fax Gateway

FAX_GW#sh run
Building configuration...

Current configuration : 4530 bytes
!
! Last configuration change at 20:07:51 UTC Wed Apr 13 2011
!
version 15.0
service timestamps debug datime msec
service timestamps log datetime msec
no service password-encryption
!
hostname FAX_GW
!
boot-start-marker
boot-end-marker
!
logging buffered 10000000
enable password cisco
!
no aaa new-model
!
dot11 syslog
ip source-route
!
!
ip cef
!
!
no ip domain lookup
no ipv6 cef
multilink bundle-name authenticated
!
!
voice service voip
 h323
 sip
!
voice class codec 1
codec preference 1 g729a8
codec preference 2 g711ulaw
!
!
voice-card 0
dspfarm
dsp services dspfarm
!
!
license udi pid CISCO2811 sn FTCX040A1LY
archive
log config
hidekeys
username cisco privilege 15 secret 5 $1$1I1WY$fwla1ooHsE/ORF20GsQFz.
!
interface FastEthernet0/0
  ip address 172.20.109.201 255.255.255.0
duplex auto
  speed auto

interface FastEthernet0/1
  no ip address
  shutdown
duplex auto
  speed auto

ip forward-protocol nd

no ip http server
no ip http secure-server

ip route 0.0.0.0 0.0.0.0 172.16.100.1
ip route 0.0.0.0 0.0.0.0 172.20.109.1

control-plane

voice-port 0/0/0
  timeouts ringing infinity

voice-port 0/0/1
  timeouts ringing infinity

no mgcp package-capability res-package
no mgcp package-capability fxr-package
no mgcp timer receive-rcp

!dial-peer voice 4351 pots
destination-pattern 4351
  port 0/0/0
  forward-digits 0
!
dial-peer voice 4352 pots
destination-pattern 4352
  port 0/0/1
  forward-digits 0
!
dial-peer voice 38 voip
description dial-peer for outgoing fax call using T38
destinatoion-pattern 9T
  session protocol sipv2
  session target ipv4:172.20.236.50
  incoming called-number 435.
  voice-class codec 1
dtmf-relay rtp-nte
dtmf-relay sg3-to-g3

8 This parameter defines the destination IP address for the dial-peer. Enter the Cisco UCM IP address
fax rate 14400
fax protocol t38 ls-redundancy 5 hs-redundancy 1 fallback pass-through g711ulaw
no vad
!
dial-peer voice 711 voip
description dial-peer for outgoing fax call using G711 (no shut to test)
shutdown
destination-pattern 9T
translate-outgoing calling 2
session protocol sipv2
session target ipv4:172.20.236.50
incoming called-number 435.
dtmf-relay rtp-nte
playout-delay nominal 80
playout-delay mode fixed
codec g711ulaw
no vad
!
!
sip-ua
protocol mode ipv4
!
!
line con 0
exec-timeout 610 0
password cisco
login
line aux 0
line vty 0 4
exec-timeout 0 0
password cisco
login
!
scheduler allocate 20000 1000
end

IOSGW#

9 This command enables T.38 fax relay
10 This command is recommended for Fax using G.711 codec
11 This configuration is recommended for Fax using G.711 codec
12 This configuration is required for Fax using G.711 codec
## ACRONYMS

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>MGCP</td>
<td>Media Gateway Control Protocol</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>CISCO UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>CISCO UBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>SP</td>
<td>Service Provider</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public switched telephone network</td>
</tr>
<tr>
<td>IP</td>
<td>Internet Protocol</td>
</tr>
<tr>
<td>TDM</td>
<td>Time-division multiplexing</td>
</tr>
<tr>
<td>CODEC</td>
<td>Coder-Decoder (in this document a device used to digitize and undigitize voice signals)</td>
</tr>
</tbody>
</table>
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