AT&T IP Flexible Reach - Enhanced Features Service on MIS, MPLS PNT or AT&T VPN: Connecting Cisco Unified Communications Manager 8.6 with Cisco Unified Border Element Release 8.8 using SIP

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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 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) 8.6 with Cisco Unified Border Element (CISCO UBE) 8.8 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 provides inbound and outbound call service and IP Flexible Reach-EF network-based features.

- 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), CISCO auto-attendant, fax using T.38 and G.711 (G3 and SG3 speeds), teleconferencing, 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 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 3925 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 the routers listed below support Cisco UBE 8.8)

  - 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 8.x:

System Components

Hardware Components

- Cisco Integrated Service Router G2. This solution was tested with C3925 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:

  [http://www.cisco.com/cgi-bin/Support/DSP/cisco_dsp_calc.pl](http://www.cisco.com/cgi-bin/Support/DSP/cisco_dsp_calc.pl)

- Packet Voice Data Module (PVDM). You will need to install DSP modules (PVDM) 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. Follow the following link for system required DSP calculator.

- Cisco MCS 7800 Series server to run Cisco Unified Communication Manager (CISCO UCM)
  - Cisco IP Phones. This solution was tested with 9971, 7975, 7971 and 7962 phones, but any Cisco IP Phone model supporting RFC2833 can be used
- Cisco MCS 7800 Series server to run Cisco Unity Connection (CUC).

- Cisco IOS gateway (only needed if fax, analog phones or TDM systems are to interconnect). 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. Please refer to the IOS Fax Gateway Configuration section for details.

Software Requirements

- Cisco Unified Communication Manager Enterprise (Cisco UCM) Release 8.6. This solution was tested with 8.6.2.21900-5. This certification document covers all Cisco UCM 8.6 Enterprise and Business Edition 6000 8.6 maintenance releases.
- Cisco Unified Border Element Release 8.8 with IOS version 15.2.1T2 release. This configuration was tested with C3900-universalk9-mz.SPA.152-1.T2.bin; however, this document is applicable to all software versions 15.2(1)T/CUBE 8.8 and 15.2(2)T/CUBE 8.9 maintenance releases
- Cisco Unity Connection (CUC) Release 8.5. This solution was tested with 8.5.1.10000-206
- Cisco IOS Gateway (IP-TDM) version: 12.4 or later.

The documented CISCO UBE configuration can be supported with the following IOS feature sets: UNIVERSAL, IP VOICE, SP SERVICES, ADVANCED IP SERVICES, ADVANCED ENTERPRISE SERVICES, INT VOICE/VIDEO, IP/IPG, TDMIP GW INT VOICE/VIDEO, IP/IPG, TDMIP GW AES
- 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:
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 (See Caveat section for details)
- Fax over G.711 (See Caveat section for details)
- Incoming DNIS Translation and Routing
- CISCO UBE: performs 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
- IP Flexible Reach-EF Network-based Call Forward Unconditional, Busy, No Answer and Not Reachable
- IP Flexible Reach – EF Network-based Simultaneous Ringing – Requires IOS version 15.2(1)T4
- IP Flexible Reach – EF Network-based Sequential Ringing – Requires IOS version 15.2(1)T4

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 G729 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 g729 annexB during a call and g729 no annexB during another call, if this occurs during transfer call or conference call the service will fail due to incompatible codec negotiation. In order to avoid this situation we recommend the customer only advertise the codec g729r8 (g729 annexB=no) within the voice-class codec setting in Cisco UBE to avoid this caveat.

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

- For call scenarios with HOLD function such as call hold/resume / call transfer over NSN network, the NSN network sends multiple codec preference (18 0) in its 200 OK response to CUCM re-INVITES (DO). This causes the CUCM to change codec (already established at G729) to G711ulaw. A script was used on the CUCM to select first preference 18 (G729) from this list of multiple codecs.

Diversion-Header for Diverted Calls

- Since Release 8.5, CISCO UCM is able 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 serviced 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 we use 4-digit extensions on our CISCO UCM IP phones, 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 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. For example, 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. This workaround works for call transfer to AT&T via Sonus and NSN. However, since AT&T Cisco SIP GW responds with 183 with SDP, this workaround does not apply.

- Unable to complete unattended call transfer – The 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 is enabled under SIP Profile configuration page while SIP PRACK support on CISCO UCM 7.X 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 not supported. SIP signaling analysis between the CUBE and AT&T IP Flexible Reach-EF network shows that the CUBE does not properly handle 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 with 15.2(1)T4. 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. This document is posted on the interoperability portal"
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 Business Voice over IP Services found in the SG Library at [http://new.serviceguide.att.com](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 Business Voice over IP (BVoIP) Services Service Guide in detail to understand the limitations and restrictions.
**Configuration**

**Configuring Cisco Unified Border Element (CISCO UBE)**

UCM-ISRG2-ATT#sh run
Building configuration...

Current configuration : 9375 bytes

! No configuration change since last restart
version 15.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service sequence-numbers
!
hostname UCM-ISRG2-ATT
!
boot-start-marker
boot system flash:c3900-universalk9-mz.SPA.152-1.T2.bin
boot-end-marker
!
!
card type t1 1 1
logging queue-limit 10000
logging buffered 20000000
logging persistent filesize 20000000
logging rate-limit 10000
no logging console
enable secret 5 $1$4jgu$npJCRdswNO47pZhBy3fhi/
enable password cisco
!
no aaa new-model
clock timezone PST -8 0
no network-clock-participate slot 1
!
no ipv6 cef
!
!
!
!
ip domain name yourdomain.com
no ip cef
multilink bundle-name authenticated
!
!
!
!
!
isdn switch-type primary-ni
!
voice-card 0
dspfarm
This command enables DSP farming which allows DSP resources to register to Cisco UCM as MTP, CFB or Transcoder devices.

This command enables address-hiding feature which prevents the Cisco UBE from sending inside/customer network addressing.

This command is available and required in ISR G2 platforms only.

This command enables Cisco UBE’s basic IP-to-IP voice communication feature.

This command allows SIP error messages to pass-through end-to-end without modification through Cisco UBE.

This command must be enabled at a global level to maintain integrity of SIP signaling between AT&T network and CUCM across CUBE.

This command allows for privacy settings to be transparently passed across between AT&T network and Cisco UCM. This command can either be set at a global level, such as in this example, or it can be set at the dial-peer level.

This command allows Cisco UBE to negotiate all flavors of g729 codec and must be configured in order to interoperate seamlessly across AT&T’s BVoIP services. The command can be enable either globally, such as in this example, or per dial-peer basis using the “voice-class sip g729 annexb-all” command.

This command enables multiple codec support and performs codec filtering required for correct interoperability between AT&T SIP network and Cisco UCM. Payload packet size can also be configured here.

SIP Profiles can be used to manipulate SIP header attributes

This SIP Profile removes the SDP attribute “T38FaxFillBitRemoval:0” from Cisco IOS gateway upspeed Re-INVITE (inbound call to CPE.) Some SIP components within AT&T’s SIP core do not support the “0” as the Boolean value, instead some AT&T devices interpret the full attribute as the Boolean value (1=attribute present; 0=attribute not present). For this reason, we remove the attribute completely to achieve fax t.38 interoperability across AT&T’s entire SIP core.
voice class sip-profiles 2
   response ANY sip-header Allow-Header modify "UPDATE,"  

   !
   license udi pid C3900-SPE100/K9 sn FOC14400CH4
   hw-module pvdm 0/0
   !
   hw-module sm 1
   !
   !
   archive
   log config
   hidekeys
   username Cisco password 0 cisco
   !
   redundancy
   !
   controller T1 1/0
   pri-group timeslots 1-24
   !
   controller T1 1/1
   !
   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 99.136.103.65 255.255.255.0
   duplex auto
   speed auto
   !

13 This SIP Profile prevents the Cisco UBE from advertising to Cisco UCM support for UPDATE header. The Cisco UCM sends UPDATE message for unattended call transfers. Since AT&T does not support this SIP header, the Cisco UBE is unable to pass this header through to AT&T. The Cisco UBE then sends 491 Request Pending message back to Cisco UCM until the call session expires and the call transfer is not completed. Removing the UPDATE from the list of ALLOW header from the Cisco UBE prevents the Cisco UCM from sending any UPDATE messages.
interface GigabitEthernet0/2
no ip address
shutdown
duplex auto
speed auto
!
interface Serial1/0:23
no ip address
encapsulation hdlc
isdn switch-type primary-ni
isdn incoming-voice voice
no cdp enable
!
ip default-gateway 172.20.43.1
ip forward-protocol nd
!
no ip pim dm-fallback
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
ip route 0.0.0.0 0.0.0.0 172.20.66.1
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 207.242.225.200 255.255.255.255 99.136.103.70
ip route 207.242.225.210 255.255.255.255 99.136.103.70
!
!
ns resp-timeout 1
cpd cr-id 1
!
control-plane
!
voice-port 0/0/0
!
voice-port 0/0/1
!
voice-port 0/0/2
!
voice-port 0/0/3
!
voice-port 1/0:23
!
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
!
scmp local GigabitEthernet0/0
scmp ccm 172.20.66.254 identifier 1 version 7.0
scmp!

scmp ccm group 1
bind interface GigabitEthernet0/0
associate ccm 1 priority 1
associate profile 2 register MTP000F352F26E9
associate profile 1 register CFB000F352F26E9
signaling dscp default
audio dscp default
!
dspfarm profile 2 transcode
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
codec g729r8
codec g729br8
codec g722-64
maximum sessions 4
associate application SCCP
!
dspfarm profile 1 conference
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
codec g729r8
codec g729br8
codec g722-64
maximum sessions 4
associate application SCCP
!
dial-peer voice 1999 voip
description outgoing voice / t.38 fax call to AT&T facing AT&T network
destination-pattern 1T
session protocol sipv2
session target ipv4:207.242.225.210
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1

These scmp commands configure the shared DSP resources as conference bridge (CFB) and as transcoder device for CISCO UCM
This command sets the SIP server target for outgoing SIP calls to AT&T
This command assigns the voice-class codec list that is to be supported for calls terminating to this dial-peer
This command enables the support for sending/relaying the P-Asserted-ID header within the SIP INVITE
Allows for PAI privacy header setting to be passed through transparently
This command enables delay offer-to-early offer conversion of initial SIP INVITE message to calls matched to this dial-peer level
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400 bytes 48
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

! dial-peer voice 1998 voip
description outgoing voice / t.38 fax calls to AT&T facing AT&T network
session protocol sipv2
incoming called-number 1T
voice-class codec 1
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 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 1997 voip
description incoming voice / t.38 fax calls from AT&T facing CISCO UCM IPPBX
destination-pattern [37][13][24]........
session protocol sipv2
session target ipv4:172.20.66.154
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip profiles 2
dtmf-relay rtp-nte
fax rate 14400 bytes 48
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 1996 voip
description incoming voice / t.38 fax calls from AT&T facing CISCO UCM IPPBX
session protocol sipv2
session target ipv4:172.20.66.154
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip profiles 2
dtmf-relay rtp-nte
fax rate 14400 bytes 48
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 1995 voip
description Outgoing International calls to AT&T facing AT&T network
destination-pattern 011T
session protocol sipv2
session target ipv4:207.242.225.210
voice-class codec 1
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 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 1994 voip
description Outgoing International calls to AT&T facing AT&T network
session protocol sipv2
incoming called-number 011T
voice-class codec 1
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 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 1993 voip
description Outgoing N11 calls to AT&T facing AT&T network
destination-pattern [459]11
session protocol sipv2
session target ipv4:207.242.225.210
voice-class codec 1
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 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 1992 voip
description Outgoing N11 calls to AT&T facing AT&T network
session protocol sipv2
incoming called-number [459]11
voice-class codec 1
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 0 hs-redundancy 0 fallback none
no vad
!
!
sip-ua
no remote-party-id
disable-early-media 18023
retry invite 224
!
!

gatekeeper
shutdown
!
!
!
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 udptn v120 ssh
stopbits 1
line vty 0 4
exec-timeout 15 0
password cisco
login
transport input all
!

scheduler allocate 20000 1000
end

UCM-ISR2G2-ATT#

---

23 This command is used as a workaround for no ringback tone heard for unattended call transfer. This disables early media cut-through and plays local ringback.

24 This command sets the number of INVITE's CISCO UBE will send out without receiving a response before failing the call.
Configuring Cisco Unified Communication Manager (CISCO UCM)

CISCO UCM Version

Last Successful Logon: Jul 12, 2012 9:24:45 PM

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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.
CISCO UCM Regions

(1 of 3)

<table>
<thead>
<tr>
<th>Find Regions where Name begins with</th>
<th>Find</th>
<th>Clear Filter</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Name *</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Phone Reg</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- Default Region to handle G711
- Region assigned to SIP trunks and media resources handle G729
CISCO UCM Regions (2 of 3)

### Region Information
**Name:** Default

### Region Relationships
<table>
<thead>
<tr>
<th>Region</th>
<th>Max Audio Bit Rate</th>
<th>Max Video Call Bit Rate (Includes Audio)</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>64 kbps (G.722, G.711)</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Phones_Reg</td>
<td>8 kbps (G.729)</td>
<td>384</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

**NOTE:** Regions(s) not displayed
- Use System Default
- Use System Default
- Use System Default

### Modify Relationship to other Regions

<table>
<thead>
<tr>
<th>Regions</th>
<th>Max Audio Bit Rate</th>
<th>Max Video Call Bit Rate (Includes Audio)</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
<td></td>
</tr>
<tr>
<td>Phones_Reg</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
<td></td>
</tr>
</tbody>
</table>

* indicates required item.
### Region Information

Name: Phones_Reg

### Region Relationships

<table>
<thead>
<tr>
<th>Region</th>
<th>Max Audio Bit Rate</th>
<th>Max Video Call Bit Rate (Includes Audio)</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>8 kbps (G.711)</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Phones_Reg</td>
<td>64 kbps (G.722, G.711)</td>
<td>384</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

Note: Regions not displayed: Use System Default

### Modify Relationship to other Regions

<table>
<thead>
<tr>
<th>Regions</th>
<th>Max Audio Bit Rate</th>
<th>Max Video Call Bit Rate (Includes Audio)</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
<td></td>
</tr>
<tr>
<td>Phones_Reg</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

* indicates required item.
## Device Pool (List)

<table>
<thead>
<tr>
<th>Name</th>
<th>Cisco Unified CM Group</th>
<th>Region</th>
<th>Date/Time Group</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Default</td>
<td>Default</td>
<td>CMRocal</td>
<td><img src="image" alt="Unsure" /></td>
</tr>
<tr>
<td>Phones_DP</td>
<td>Default</td>
<td>Phones_Ren</td>
<td>CMRocal</td>
<td><img src="image" alt="Unsure" /></td>
</tr>
</tbody>
</table>

Add New | Select All | Clear All | Delete Selected |
Device Pool – Default

(1 of 4)
Device Pool – Default (2 of 4)

<table>
<thead>
<tr>
<th>Device Pool Configuration</th>
<th>Related Links: Back To Find/List</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Network Locale</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>SRST Reference</strong></td>
<td>Disable</td>
</tr>
<tr>
<td><strong>Connection Monitor Duration</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Single Button Dereg</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Join Across Lines</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Physical Location</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Device Mobility Group</strong></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

--- **Device Mobility Related Information**

<table>
<thead>
<tr>
<th>Device Mobility Calling Search Space</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Called Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

--- **Geolocation Configuration**

<table>
<thead>
<tr>
<th>Geolocation</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Geolocation Filter</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

--- **Call Routing Information**
Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Device Pool Configuration

- Call Routing Information

**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>Default Prefix Settings</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**Incoming Called 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>Default Prefix Settings</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

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Page 24 of 102
Device Pool – Default (4 of 4)

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td>3</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td>3</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>3</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td>3</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**Connected Party Settings**
Connected Party Transformation CSS: < None >

**Redirecting Party Settings**
Redirecting Party Transformation CSS: < None >

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Page 25 of 102
Device Pool - Phones_DP (2 of 4)

### Device Pool Configuration

<table>
<thead>
<tr>
<th>Device Pool Configuration Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Network Locate</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SRST Reference*</td>
<td>Disable</td>
</tr>
<tr>
<td>Connection Monitor Duration***</td>
<td></td>
</tr>
<tr>
<td>Single Button Range*</td>
<td>Default</td>
</tr>
<tr>
<td>Join Across Lines*</td>
<td>Default</td>
</tr>
<tr>
<td>Physical Location</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Device Mobility Group</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Device Mobility Related Information****

<table>
<thead>
<tr>
<th>Device Mobility Related Information Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Mobility Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Called Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Geolocation Configuration

<table>
<thead>
<tr>
<th>Geolocation Configuration Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Geolocation Filter</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Call Routing Information

<table>
<thead>
<tr>
<th>Call Routing Information Field</th>
<th>Value</th>
</tr>
</thead>
</table>
Device Pool - Phones_DP (3 of 4)

 Cisco Unified CM Administration  
 For Cisco Unified Communications Solutions

**Device Pool Configuration**

---

### 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>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Incoming Called 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>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>
Cisco Unified SIP IP Phone

(1 of 10)
Cisco Unified SIP IP Phone – (2 of 10)
Cisco Unified SIP IP Phone – (3 of 10)
### Phone Configuration

<table>
<thead>
<tr>
<th>Secure Information URL</th>
<th>Secure Messages URL</th>
<th>Secure Services URL</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Extension Information

- **Enable Extension Mobility**
  - Log Out Profile: [Use Current Device Settings] — `--` (dropdown)
  - Log In Time: `< None >`
  - Log Out Time: `< None >`

### MLPPP Information

- **MLPPP Domain**: `< None >`

### Do Not Disturb

- **Do Not Disturb**
  - DND Option: [Use Common Phone Profile Setting] — `--` (dropdown)
  - DND Incoming Call Alert: `< None >`

### Secure Shell Information

- **Secure Shell User**: `Any text input`
- **Secure Shell Password**: `Any text input`
RTCP enabled to send RTP
Cisco Unified SIP IP Phone – (9 of 10)

<table>
<thead>
<tr>
<th>Feature</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Discovery Protocol (CDP): Switch Port®</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): PC Port®</td>
<td>Enabled</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port®</td>
<td>Enabled</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP): PC Port®</td>
<td>Enabled</td>
</tr>
<tr>
<td>LLDP Asset ID</td>
<td></td>
</tr>
<tr>
<td>LLDP Power Priority*</td>
<td>Unknown</td>
</tr>
<tr>
<td>Wireless Headset Hookswitch Control®</td>
<td>Disabled</td>
</tr>
<tr>
<td>IPv6 Load Server</td>
<td></td>
</tr>
<tr>
<td>IPv6 Log Server</td>
<td></td>
</tr>
<tr>
<td>802.1x Authentication*</td>
<td>User Controlled</td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure*</td>
<td>Normal</td>
</tr>
<tr>
<td>Minimum Ring Volume*</td>
<td>S-Silent</td>
</tr>
<tr>
<td>HTTPS Server®</td>
<td>Http and Https Enabled</td>
</tr>
<tr>
<td>Handset/Headset Monitor*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Switch Port Remote Configuration®</td>
<td>Disabled</td>
</tr>
<tr>
<td>PC Port Remote Configuration®</td>
<td>Disabled</td>
</tr>
<tr>
<td>Automatic Port Synchronization®</td>
<td>Disabled</td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Cisco Unified SIP IP Phone – (10 of 10)

![Cisco Unified CM Administration](image)

**Phone Configuration**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>BLF Login Authentication*</td>
<td>User Controlled</td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure*</td>
<td>Normal</td>
</tr>
<tr>
<td>Minimum Ring Volume*</td>
<td>0-Silent</td>
</tr>
<tr>
<td>HTTPS Server*</td>
<td>http and https enabled</td>
</tr>
<tr>
<td>Handset/Headset Monitor*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Switch Port Remote Configuration*</td>
<td>Disabled</td>
</tr>
<tr>
<td>P3 Port Remote Configuration*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Automatic Port Synchronization*</td>
<td>Disabled</td>
</tr>
<tr>
<td>SSH Access*</td>
<td>Disabled</td>
</tr>
<tr>
<td>80-bit SRTP*</td>
<td>Disabled</td>
</tr>
</tbody>
</table>

* - indicates required item.

** - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

***Note: Security Profile Contains Additional CAPF Settings.

****Note: A protected device means it is capable of playing Secure and Non-Secure Tones. When the checkbox is checked, the user will hear a Secure or Non-Secure Tone when the call is connected.

*****Note: A custom Softkey template without supplementary service Softkeys must be used for a Hot line Device.
Cisco IP Phone DN 2717
(1 of 6)
Cisco IP Phone DN 2717 (4 of 6)

### Directory Number Configuration

#### MLPP Alternate Party Settings

<table>
<thead>
<tr>
<th>Target (Destination)</th>
<th>MLPP Calling Search Space</th>
<th>MLPP No Answer Ring Duration (seconds)</th>
</tr>
</thead>
</table>

#### Line Settings for All Devices

<table>
<thead>
<tr>
<th>Hold Reversion Ring Duration (seconds)</th>
<th>Hold Reversion Notification Interval (seconds)</th>
<th>Party Entrance Tone</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Setting the Hold Reversion Notification Interval to zero will disable the feature</td>
<td></td>
</tr>
</tbody>
</table>

#### Line 1 on Device SEP001D705FB383

<table>
<thead>
<tr>
<th>Display (Internal Caller ID)</th>
<th>ASCII Display (Internal Caller ID)</th>
<th>Line Text Label</th>
<th>ASCII Line Text Label</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Seven Sienna</td>
<td>Seven Sienna</td>
<td>Seven Sienna</td>
</tr>
</tbody>
</table>
Cisco Unified CM Administration

Directory Number Configuration

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>External Phone Number Mask</td>
<td></td>
</tr>
<tr>
<td>Visual Message Waiting Indicator Policy*</td>
<td>Use System Policy</td>
</tr>
<tr>
<td>Audible Message Waiting Indicator Policy*</td>
<td>Default</td>
</tr>
<tr>
<td>Ring Setting (Phone Idle)*</td>
<td></td>
</tr>
<tr>
<td>Ring Setting (Phone Active)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Idle)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Active)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Recording Option*</td>
<td>Call Recording Disabled</td>
</tr>
<tr>
<td>Recording Profile</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Monitoring Calling</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Log Missed Calls</td>
<td></td>
</tr>
</tbody>
</table>

---

Multiple Call/Call Waiting Settings on Device SEP001D705FB383

Note: The range to select the Max Number of calls is: 1-50

Maximum Number of Calls* | 4 |
IOS Conference Bridge for G729

Sample IOS configuration for conference bridge registered to Cisco UCM:

voice-card 0
dspfarm
dsp services dspfarm!
sccp local GigabitEthernet0/1
sccp ccm 172.X.X.X identifier 1 version 7.0
sccp

sccp ccm group 1
associate ccm1 priority 1
associate profile 1 register CFB000F352F26E9
dspfarm profile 1 conference
codec g711ulaw
codec g711alaw
codec g722ar8
codec g729abr8
codec g729r8
codec g729br8
codec g722-64
maximum sessions 4
associate application SCCP

Note: CISCO UCM requires a conference bridge resource for three-way conferencing that include g729 rtp streams.
IOS Transcoder

**Note:** If your network will support more than one codec, it is recommended to have a transcoder resource on Cisco Unified CM.

Sample IOS configuration for conference bridge registered to Cisco UCM

```plaintext
voice-card 0
dspfarm
dsp services dspfarm
	sccp local GigabitEthernet0/1
scss ccm 172.X.X.X identifier 1 version 7.0
sscp

sscp ccm group 1
associate ccm1 priority 1
associate profile 1 register MTP000F352F26E9
dspfarm profile 2 transcode
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
codec g729r8
codec g729br8
codec g722-64
maximum sessions 4
associate application SCCP
```
## Music on Hold

(1 of 2)

### Cisco Unified CM Administration

For Cisco Unified Communications Solutions

### Music On Hold (MOH) Server Configuration

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Registration</td>
<td>Registered with Cisco Unified Communications Manager CM-Sienna</td>
</tr>
<tr>
<td>IP Address</td>
<td>172.20.43.154</td>
</tr>
<tr>
<td>Host Server</td>
<td>CM-Sienna</td>
</tr>
<tr>
<td>Music On Hold Server Name*</td>
<td>MOH_1</td>
</tr>
<tr>
<td>Description</td>
<td>MOH_CM-Sienna</td>
</tr>
<tr>
<td>Device Pool*</td>
<td>Default</td>
</tr>
<tr>
<td>Location*</td>
<td>Multi_Name</td>
</tr>
<tr>
<td>Maximum Half Duplex Streams*</td>
<td>50</td>
</tr>
<tr>
<td>Maximum Multi-cast Connections*</td>
<td>250000</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

- Enable Multi-cast Audio Sources on this MOH Server: Yes
Music On Hold (2 of 2)
**Music On Hold Service Parameter (IP Voice Media Streaming App) Setting**

- **Conference Bridge (CFB) Parameters**
  - Call Count: 49
  - Run Flag: True

- **Media Termination Point (MTP) Parameters**
  - Call Count: 49
  - Run Flag: True

- **Clusterwide Parameters (Parameters that apply to all servers)**
  - **Supported Media Codes**
    - 711_mulaw
    - 711_alaw
    - 729_Ansik_SI

- **Multicast MCH IP DSCP**
  - EF DSCP (101110)

- **MTP DTMF Duration**
  - 0.00

- **MTP DTMF Power (volume)**
  - 9

Highlight and save applicable codecs to be supported.
Annunciator must be active and registered to support Cisco UCM local progress tones and announcements when early media is not available. Annunciator is part of the Cisco UCM IP Voice Media Streaming Application which must be enabled under Service Activation.
## Route Pattern

### Cisco Unified CM Administration
For Cisco Unified Communications Solutions

#### Find and List Route Patterns

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

### Status
7 records found

#### Route Patterns (1 - 7 of 7)

<table>
<thead>
<tr>
<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>10.911</td>
<td>Route Pattern used by CER for 911 calls</td>
<td>911</td>
<td>TO_CUBES5_ATT</td>
<td></td>
<td></td>
</tr>
<tr>
<td>11.911</td>
<td>Route Pattern used by CER for 911 calls</td>
<td>911</td>
<td>TO_CUBES5_ATT</td>
<td></td>
<td></td>
</tr>
<tr>
<td>13.911</td>
<td>Route Pattern used by CER for 911 calls</td>
<td>911</td>
<td>TO_CUBES5_ATT</td>
<td></td>
<td></td>
</tr>
<tr>
<td>2712</td>
<td>To IGS Fax GW - Incoming from SP</td>
<td>FAX_GW</td>
<td>TO_CUBES5_ATT</td>
<td></td>
<td></td>
</tr>
<tr>
<td>8.0110X000X00XX</td>
<td>Outgoing Intl calls to CUBE to ATT</td>
<td></td>
<td>TO_CUBES5_ATT</td>
<td></td>
<td></td>
</tr>
<tr>
<td>8.1XX000000X0XX</td>
<td>Outgoing National calls to CUBE to ATT</td>
<td></td>
<td>TO_CUBES5_ATT</td>
<td></td>
<td></td>
</tr>
<tr>
<td>145111</td>
<td>Outgoing N1 calls</td>
<td></td>
<td>TO_CUBES5_ATT</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

#### Actions
- **Add New**
- **Select All**
- **Clear All**
- **Delete Selected**
Route Pattern (1 of 3)

Route Pattern Configuration

- Status
  - Status: Ready

- Pattern Definition
  - Route Pattern: *Required*
  - Route Partition: <None>
  - Description:
  - Numbering Plan: NaNP
  - Route Filter: <None>
  - MLP Precedence: Default
  - Apply Cell Blocking Percentage:
  - Resource Priority Namespace Network Domain: <None>
  - Route Class: Default
  - Gateway/Route List: CUBE_TO_ATT (Edit)
  - Route Option:
    - Route this pattern
    - Block this pattern
    - No Error
  - Call Classification: Offline

- Options
  - Allow Device Override
  - Provide Outside Dial Tone
  - Allow Overlap Sending
  - Urgent Priority
  - Require Forced Authorization Code
Route Pattern (3 of 3)

---

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Route Pattern Configuration

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
- Called Party Numbering Plan

ISDN Network-Specific Facilities Information Element
- Network Service Protocol
- Carrier Identification Code

Related Links: Back To Find/List

Save  Delete  Copy  Add New

* - indicates required item.
Route Pattern to IOS Fax Gateway

(1 of 3)

<table>
<thead>
<tr>
<th>Pattern Definition</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>2714</td>
</tr>
<tr>
<td>Route Partition</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Description</td>
<td></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 (valid)</td>
</tr>
<tr>
<td>Route Option</td>
<td>Route this pattern</td>
</tr>
<tr>
<td>Cell Classification*</td>
<td>Offset</td>
</tr>
</tbody>
</table>

Allow Device Override: Provide Outside Dial Tone: Allow Overlap Sending: Urgent Priority: Require Forced Authorization Code:
Route Pattern to IOS Fax Gateway (2 of 3)

<table>
<thead>
<tr>
<th>Route Pattern Configuration</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Save</strong></td>
</tr>
<tr>
<td>Requires Password Authentication Code</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
</tbody>
</table>

---

**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)
Route Pattern to IOS Fax Gateway (3 of 3)

<table>
<thead>
<tr>
<th>Connected Party Transformations</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Line ID Presentation*</td>
<td>Default</td>
</tr>
<tr>
<td>Connected Name Presentation*</td>
<td>Default</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Called Party Transformations</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Discard Digits</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Called Party Transform Mask</td>
<td></td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td></td>
</tr>
<tr>
<td>Called Party Number Type*</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Called Party Numbering Plan*</td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>ISDN Network-Specific Facilities Information Element</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Network Service Protocol</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td>Carrier Identification Code</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Network Service</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>-- Not Selected --</td>
<td>-- Not Exist &gt;</td>
<td></td>
</tr>
</tbody>
</table>
Translation Pattern for 10 Incoming Called-Number to 4-Digit Extension

(1 of 2)
Translation Pattern for 10 Incoming Called-Number to 4-Digit Extension (2 of 2)

<table>
<thead>
<tr>
<th>Prefix Digits (Outgoing Calls)</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Line ID Presentation*</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Name Presentation*</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Party Number Type*</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Calling Party Numbering Plan*</td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

**Connected Party Transformations**

<table>
<thead>
<tr>
<th>Connected Line ID Presentation*</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Name Presentation*</td>
<td>Default</td>
</tr>
</tbody>
</table>

**Called Party Transformations**

<table>
<thead>
<tr>
<th>Discard Digits</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Called Party Transform Mask</td>
<td>XXXX</td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td></td>
</tr>
<tr>
<td>Called Party Number Type*</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Called Party Numbering Plan*</td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

* - Indicates required item.
### CISCO UCM SIP Trunks (List)

#### Find and List Trunks
- Add New
- Select All
- Clear All
- Delete Selected
- Reset Selected

#### Status
- 4 records found

#### Trunks (1 - 4 of 4)

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Calling Search Space</th>
<th>Device Pool</th>
<th>Route Pattern</th>
<th>Partition</th>
<th>Route Group</th>
<th>Priority</th>
<th>Trunk Type</th>
<th>SIP Trunk Security Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>CUBE_TO_ATT</td>
<td>Sip trunk to ATT via CUBE 8.8</td>
<td>Default</td>
<td>9.8</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>CUBE_TO_ATT</td>
<td>Sip trunk to ATT via CUBE 8.8</td>
<td>Default</td>
<td>9.218XXXX</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>FAX_GW</td>
<td>Sip Trunk to Fax_GW</td>
<td>Default</td>
<td>2714</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>UNITY</td>
<td>Sip Trunk to Unity</td>
<td>Default</td>
<td>8888</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Non Secure SIP Trunk Profile</td>
</tr>
</tbody>
</table>

- Add New
- Select All
- Clear All
- Delete Selected
- Reset Selected
SIP Trunk to CISCO UBE

(1 of 5)
SIP Trunk to CISCO UBE (2 of 5)

Trunk Configuration

- 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 APDU
- 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
- Route Class Signaling Enabled
- Use Trusted Relay Point
- PSTN Access
- Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile: <None>

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain: <None>

Call Routing Information
SIP Trunk to CISCO UBE (3 of 5)

![Cisco Unified CM Administration interface](image)

**Trunk Configuration**

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

### Inbound Calls
- Significant Digits: All
- Connected Line ID Presentation: Default
- Connected Name Presentation: Default
- Calling Search Space: <None>
- AAR Calling Search Space: <None>
- Prefix DN

**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></td>
<td>0</td>
<td>&lt;None&gt;</td>
<td></td>
</tr>
</tbody>
</table>
SIP Trunk to CISCO UBE (4 of 5)

### Trunk Configuration

<table>
<thead>
<tr>
<th>Action</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Save</td>
<td></td>
</tr>
<tr>
<td>Delete</td>
<td></td>
</tr>
<tr>
<td>Reset</td>
<td></td>
</tr>
<tr>
<td>Add New</td>
<td></td>
</tr>
</tbody>
</table>

**Connected Party Settings**

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

**Outbound Calls**

- Called Party Transformation CSS: `<None>`
- Use Device Pool Called Party Transformation CSS
- Calling Party Transformation CSS: `<None>`
- Use Device Pool Calling Party Transformation CSS
- Calling Party Selection
- Calling Line ID Presentation
- Calling Name Presentation
- Caller ID DN
- Caller Name

- Redirecting Diversion Header Delivery - Outbound
- Redirecting Party Transformation CSS: `<None>`
- Use Device Pool Redirecting Party Transformation CSS
SIP Trunk to CISCO UBE (5 of 5)

SIP Normalization Script used to normalize Diversion Header

Diversion Mask Parameter value used by the script
SIP Normalization Script Configuration

The following describes the procedure to apply a script to a SIP Trunk:

1. Copy and paste the SIP Normalization Script below to a text editor program (such as Notepad, TextEdit, etc.). Make sure that the script is not corrupted and save it as text file.

2. Under the Cisco UCM Administration page, go to Device > Device Settings > SIP Normalization Script configuration page. Assign a name and description for the script and click on “Add New” to add the SIP Normalization script saved. Adding the text of the script in the “Content” box can be done by:
   a. Clicking on “Import File” to upload the SIP Normalization script file saved
   OR
   b. Opening the saved text file, copying and pasting the text in the “Content” box.

3. Click on Save

4. Associate the SIP Normalization Script saved to the SIP Trunk requiring normalization (in this case, CUBE_to_ATT - SIP trunk to AT&T via CUBE)

For more information on Cisco UCM SIP Normalization configuration, please refer to:

SIP Normalization Script (Text)

(The following script is used to modify the Diversion header for forwarded calls toward AT&T so that the 4-digit extension included in the Diversion header of a forwarding INVITE message is expanded to its full 10-digit DID number)

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

(The following script narrows down the incoming multiple codec preference to G729 (18) only. This is usually the case for call flows with HOLD function such Hold/Resume, Call transfer over NSN network)

M = {}
function M.inbound_ANY_INVITE(message)
  local sdp = message:getSdp()
  if sdp then
    local m1 = sdp:getMediaDescription("audio")
    if m1 then
      m1 = m1:gsub("18 0", "18")
      m1 = m1:gsub("a=rtpmap:0 PCMU/8000%d*\n", ")
      sdp = sdp:modifyMediaDescription("audio", m1)
      message:setSdp(sdp)
    end
  end
end
return M
(Diversion header contents may be generated by CTI ports and route points as well as endpoints. For example, this script removes the Diversion header generated by Cisco UCM with a “911” route point for integration with Cisco Emergency Responder with a “911” route point. If other CTI ports or route points are present, for Cisco ER or other CTI applications, the script must be expanded to modify any other Diversion headers that may be added if calls are forwarded to AT&T)

-- This script is used to do one of 2 things
-- a) if an INVITE is sent to 911, remove the diversion header
-- b) otherwise mask the user part of the diversion header using the Diversion-Mask parameter

M = {}

local mask = scriptParameters.getValue("Diversion-Mask")

-- handle the mask of the diversion header for non-911 calls

function mask_diversion(msg)
    if mask
        then
            msg:applyNumberMask("Diversion", mask)
    end
end

-- return the user part of the specified uri

function getUserPartFromUri(uri)
    return string.match(uri, 'sip:(.*)@')
end

-- handle the diversion header change for the case of 911 calls

function handle_911_diversion(msg)
    msg:removeHeader("Diversion")
end

function M.outbound_INVITE(msg)

-- Determine whether this is a 911 call based on the user part in the reqUri

local method, requri, version = msg:getRequestLine()
if requri
    then
        -- found reqUri
        local userpart = getUserPartFromUri(requri)
    if userpart
        then
            -- found user part from reqUri
            if userpart == "911"
                then
                    -- handling as 911 call
                    handle_911_diversion(msg)
                    return
            end
        end
    end
end
end
end

-- handling as non-911 call
mask_diversion(msg)
end

return M
SIP Profile used for SIP trunk to CISCO UBE

(1 of 4)
### SIP Profile used for SIP trunk to CISCO UBE (2 of 4)

![SIP Profile Configuration](image)

<table>
<thead>
<tr>
<th>Parameters used in Phone</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)*</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)*</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)*</td>
<td>3600</td>
</tr>
<tr>
<td>Timer T1 (msec)*</td>
<td>500</td>
</tr>
<tr>
<td>Timer T2 (msec)*</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE*</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE*</td>
<td>10</td>
</tr>
<tr>
<td>Start Media Port*</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port*</td>
<td>32766</td>
</tr>
<tr>
<td>Call Pickup URI*</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI*</td>
<td>x-cisco-serviceuri-opickup</td>
</tr>
<tr>
<td>Call Pickup Group URI*</td>
<td>x-cisco-serviceuri-gpickup</td>
</tr>
<tr>
<td>Meet Me Service URI*</td>
<td>x-cisco-serviceuri-meetme</td>
</tr>
<tr>
<td>User Info*</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level*</td>
<td>Nominal</td>
</tr>
<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>
</tbody>
</table>
SIP Profile used for SIP trunk to CISCO UBE (3 of 4)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller ID Blocking</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control</td>
<td>User</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7960</td>
<td>Disabled</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>5</td>
</tr>
<tr>
<td>Maximum Redirects</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>x-cisco-serviceur-cfwdl</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>x-cisco-serviceur-abbrdiao</td>
</tr>
<tr>
<td>Conference Join Enabled</td>
<td></td>
</tr>
<tr>
<td>RFC 2543 Hold</td>
<td></td>
</tr>
<tr>
<td>Semi Attended Transfer</td>
<td></td>
</tr>
<tr>
<td>Enable VAD</td>
<td></td>
</tr>
<tr>
<td>Stutter Message Waiting</td>
<td></td>
</tr>
</tbody>
</table>

### Trunk Specific Configuration

- **Reroute Incoming Request to new Trunk based on**: Never
- **RSVP Over SIP**: Local RSVP
- **Fall back to local RSVP**: Yes
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”. 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 is enabled under SIP Profile configuration page while SIP PRACK support on CISCO UCM 7.X is enabled under the Service Parameters configuration page.
### SIP Trunk to Fax Gateway

(1 of 6)

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

#### Device Information

- **Product:** SIP Trunk
- **Device Protocol:** SIP
- **Trunk Service Type:** None (Default)
- **Device Name:** FAX_GW
- **Description:** SIP Trunk to FAX_GW
- **Device Pool:** Default
- **Common Device Configuration:** < None >
- **Call Classification:** Use System Default
- **Media Resource Group List:** < None >
- **Location:** Hub_None
- **AAR Group:** < None >
- **Tunneled Protocol:** None
- **QSIG Variant:** No Changes
- **ASN.1 ROSE OID Encoding:** No Changes
- **Packet Capture Mode:** None
- **Packet Capture Duration:** 0
### SIP Trunk to Fax Gateway (2 of 6)

#### Trunk Configuration

- **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 APDU**
- **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**
- **Route Class Signaling Enabled**
- **Use Trusted Relay Point**
- **PSTN Access**
- **Run On All Active Unified CM Nodes**

#### Intercompany Media Engine (IME)

- **E.164 Transformation Profile**

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

- **MLPP Domain**

#### Call Routing Information
SIP Trunk to Fax Gateway (3 of 6)
SIP Trunk to Fax Gateway (5 of 6)

![Cisco Unified CM Administration](image)

### SIP Information

<table>
<thead>
<tr>
<th>Destination Address</th>
<th>Destination Address IPv6</th>
<th>Destination Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>1* 172.20.109.201</td>
<td></td>
<td>5060</td>
</tr>
</tbody>
</table>

- **MTP Preferred Originating Codec**: ulaw
- **Presence Group**: Standard Presence group
- **SIP Trunk Security Profile**: Non Secure SIP Trunk Profile
- **Rerouting Calling Search Space**: None
- **Out-Of-Dialog 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>Normalization Script</th>
<th>Enable Trace</th>
<th>Parameter Name</th>
<th>Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>&lt; None &gt;</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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SIP Trunk to Fax Gateway (6 of 6)
Configuring Cisco IOS Fax Gateway

IOSGW#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 datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname IOSGW
!
boot-start-marker
boot-end-marker
!
logging buffered 1000000
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 g729r8
codec preference 2 g711ulaw
!
!
voice-card 0
dsfarm
dsp services dsfarm

license udi pid CISCO2811 sn FTX1040A1LY
archive
  log config
  hidekeys
username cisco privilege 15 secret 5 $1$IWy$fwla1ooHsE/ORF20G$sFz.

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-rtcp

!  
!  
dial-peer voice 2714 pots
  destination-pattern 2714
  port 0/0/0
  forward-digits 0

!  
dial-peer voice 2717 pots
  destination-pattern 2717
  port 0/0/1
  forward-digits 0

!  
dial-peer voice 38 voip
  description dial-peer for outgoing fax call using T38
  destination-pattern 9............
  translate-outgoing calling 2
  session protocol sipv2
  session target ipv4:172.20.43.154
  incoming called-number 27..
  voice-class codec 1
  dtmf-relay rtp-nte
  fax-relay sg3-to-g3
  fax rate 14400

fax protocol t38 is-redundancy 0 hs-redundancy 0 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 9............
  translate-outgoing calling 2
  session protocol sipv2
  session target ipv4:172.20.43.154
  incoming called-number 27..
  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#
Configuring Cisco Unified Communications Manager (CISCO UCM) for Cisco Unity Connection (CUC) SIP integration

![Cisco Unified CM Administration](image)

<table>
<thead>
<tr>
<th>Device Information</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Product</td>
<td>SIP Trunk</td>
</tr>
<tr>
<td>Device Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Trunk Service Type</td>
<td>None (Default)</td>
</tr>
<tr>
<td>Device Name</td>
<td>UNITY</td>
</tr>
<tr>
<td>Description</td>
<td>SIP Trunk to Unity</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>&lt; None &gt;</td>
</tr>
<tr>
<td>Location</td>
<td>Hub_None</td>
</tr>
<tr>
<td>AAR Group</td>
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<tr>
<td>Tunneled Protocol</td>
<td>None</td>
</tr>
<tr>
<td>QSIG Variant</td>
<td>No Changes</td>
</tr>
<tr>
<td>ASN.1 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>
# Trunk Configuration

- **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 APDU**
- **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**
- **Route Class Signaling Enabled**
- **Use Trusted Relay Point**
- **PSTN Access**
- **Run On All Active Unified CM Nodes**

### Intercompany Media Engine (IME)

- **E.164 Transformation Profile**

### Multilevel Precedence and Preemption (MLPP) Information

- **MLPP Domain**

### Call Routing Information
### Call Routing Information

- **Remote-Party-Id**
- **Asserted-Identity**
- **Asserted-Type**
- **SIP Privacy**

### Inbound Calls

- **Significant Digits**
- **Connected Line ID Presentation**
- **Connected Name Presentation**
- **Calling Search Space**
- **AAR Calling Search Space**
- **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>0</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
</tbody>
</table>
Trunk Configuration

- Redirecting Diversion Header Delivery - Outbound
- Redirecting Party Transformation CSS

SIP Information

- Destination Address: 172.20.238.250
- Destination Address IPv6: 5060
- Presence Group: Standard Presence group
- SIP Trunk Security Profile: Non Secure SIP Trunk Profile
- Out-Of-Dialog Refer Calling Search Space
- SUBSCRIBE Calling Search Space
- SIP Profile: Standard SIP Profile
- DTMF Signaling Method: No Preference

Normalization Script
Trunk Configuration

Out-Of-Dialing Refer Calling Search Space
- < None >

SUBSCRIBE Calling Search Space
- < None >

SIP Profile *
- Standard SIP Profile
- No Preference

DTMF Signaling Method *
- None
- S.657
- DTMF 3GPP

Normalization Script
- Normalization Script < None >

Enable Trace
- Yes

Parameter Name
Parameter Value

Geolocation Configuration

Geolocation
- < None >

Geolocation Filter
- < None >

Send Geolocation Information
- Yes

Save | Delete | Reset | Add New

* - Indicates required item.
** - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
Configuring Cisco Unity Connection (CUC) for Connection to CISCO UCM using SIP Integration

CUC Version
CUC Telephony Integration with CISCO UCM

(1 of 2)
CUC Telephony Integration with CISCO UCM (2 of 2)
CUC Port Group Settings

(1of 2)
CUC Port Group Settings

(2 of 2)
CUC Port Settings

(1 of 1)
CUC Sample User Basic Settings

(1 of 3)
CUC Sample User Basic Settings (2 of 3)
CUC Sample User Basic Settings (3 of 3)
CUC Sample User MWI Settings
### 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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</thead>
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<td>Cisco Systems, Inc. Capital Tower 168 Robinson Road #22-01 to #29-01 Singapore 068912</td>
</tr>
<tr>
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<td>Haarlerbergpark Haarlerbergweg 13-19 1101 CH Amsterdam The Netherlands www-europe.cisco.com</td>
<td></td>
<td></td>
</tr>
<tr>
<td>San Jose, CA 95134-1706 USA</td>
<td>Tel: 310 20 357 1000 Fax: 310 20 357 1100</td>
<td>Tel: 408 526-7660 Fax: 408 527-0883</td>
<td>Tel: +65 317 7777 Fax: +65 317 7799</td>
</tr>
<tr>
<td>Tel: 408 526-4000</td>
<td></td>
<td>Tel: 408 526-7660</td>
<td>Tel: +65 317 7777</td>
</tr>
<tr>
<td>800 553-NETS (6387)</td>
<td></td>
<td>Fax: 408 527-0883</td>
<td>Fax: +65 317 7799</td>
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<tr>
<td>Fax: 408 526-4100</td>
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