

AT&T IP Flexible Reach Service with Enhanced Features
Using MIS / PNT or AT&T Virtual Private Network Transport
with Cisco Unified Communications Manager v. 11.0 and
Cisco UBE v. 11.1.0 on an ISR 4431 Router with IPv6 SIP
Interface
MAR 2016



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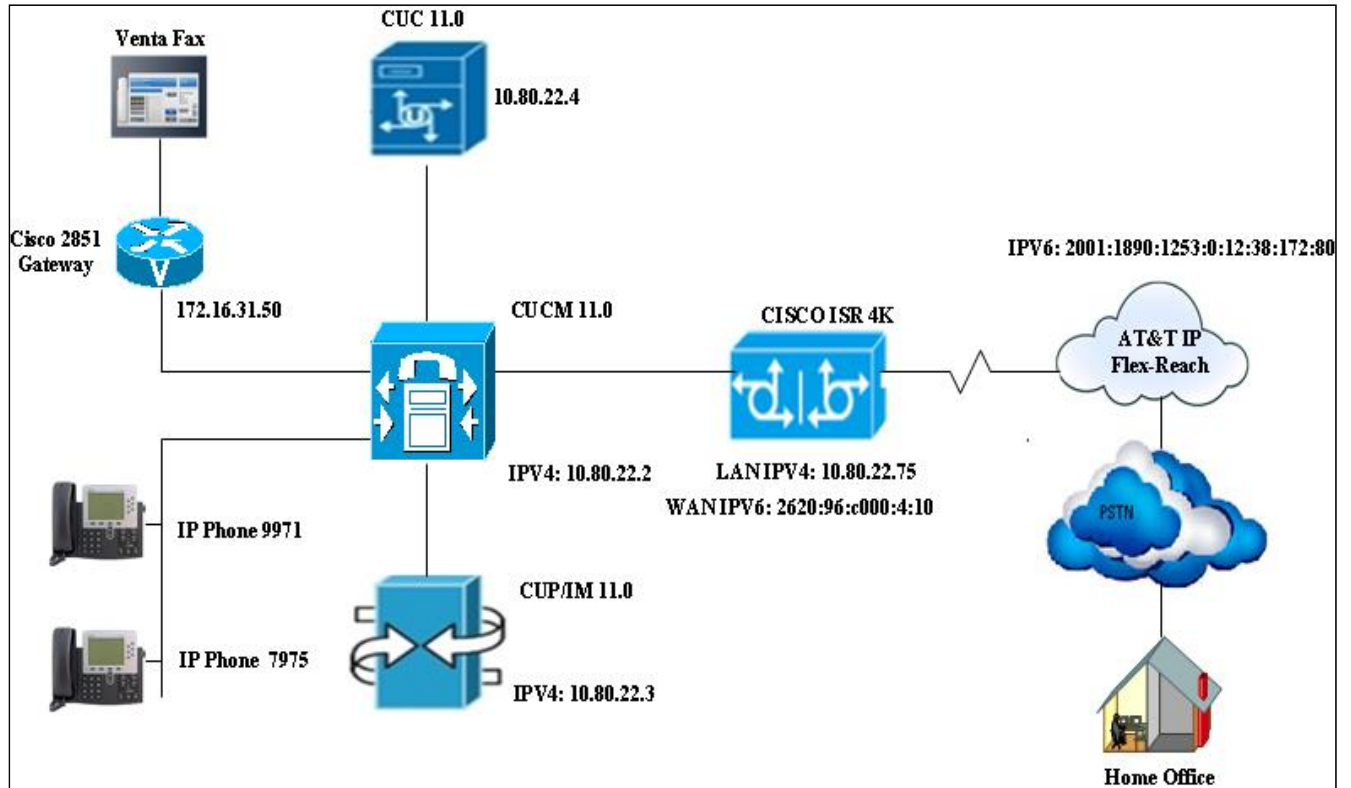


Introduction

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 (Cisco UBE) provides demarcation, security, interworking and session management services.

- This application note describes the necessary steps and configurations of Cisco Unified Communications Manager (Cisco UCM) 11.0, Cisco Unity Connection 11.0, Cisco Unified CM IM and Presence 11.0, Cisco Integrated Services Routers (ISR) Version 15.5(3) S1 with connectivity to AT&T's IP Flex-Reach SIP trunk service. It also covers support and configuration example of Cisco Unity Connection (CUC) messaging integrated with Cisco Unified Communications Manager (Cisco UCM). The deployment model covered in this application note is Cisco Integrated Services Routers (ISR) to PSTN (AT&T IP Flexible Reach SIP). AT&T IP Flexible Reach provides inbound and outbound call service.
- Testing was performed in accordance to AT&T's IP Flexible Reach 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, CODEC negotiation, 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 (BACD), fax G.711 and T38 (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 Integrated services router. The configurations described in this document details the important commands for successful interoperability. Care must be taken by the network administrator deploying Cisco ISR to ensure these commands are set per each dial-peer required, to interoperate done AT&T SIP network.
- 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 Communication Manager with Cisco Unified Border Element components.

Network Topology





Hardware Components

- UCS-C240 VMWare server running ESXi 5.5
- Cisco IP Phones. This solution was tested with Cisco 7975 & Cisco 9971 phones
- Cisco ISR4431/K9 (1RU) processor with 1659383K/6147K bytes of memory
- Processor board ID FTX1850ALVU
4 Gigabit Ethernet interfaces,
32768K bytes of non-volatile configuration memory,
4194304K bytes of physical memory,
7057407K bytes of flash memory at bootflash.

Software Requirements

- Cisco UCM: System version: 11.0.1.10000-10, including Business Edition 6000 and Business Edition 7000.
- ISR: ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.5(3) S1a, RELEASE SOFTWARE (fc1).
- Cisco UBE Software Release 11.1.0
- System image file is "isr4400-universalk9.03.16.01a.S.155-3.S1a-ext.SPA.bin".
- Cisco Unity Connection version: System version: 11.0.1.10000-10.
- Cisco Unified CM IM and Presence: System version: 11.0.1.10000-6.
- Cisco Jabber client version: 11.0.0 Build 65527
- VentaFax client version: 7.6.244.598 I



Features

Features – Supported

- Basic Call using G.729 and G711
- Calling Party Number Presentation and Restriction
- Calling Name Presentation
- AT&T Advanced 8YY Call Prompter (8YY)
- Cisco UBE Delayed-Offer-to-Early-Offer conversion of an initial SIP INVITE without SDP
- Intra-site Call Transfer
- Intra-site Conference
- Call Hold and Resume
- Call Forward All, Busy and No Answer
- AT&T IP Teleconferencing
- Fax over G.711
- Fax using T.38
- Incoming DNIS Translation and Routing
- Outbound calls to AT&T's IP and TDM networks
- Inbound calls from 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 ATT SIP network)
- Inbound & Outbound Calls using Cisco Jabber
- Emergency and 411 calls were terminated to a voicemail platform in lab environment within AT&T for test
- RTP

Network Based Features - Supported

- Call forward (Unconditional, Busy, No Answer, Not reachable)
- Sequential Ringing
- Simultaneous Ringing

NOTE: Using the AT&T IP Flexible Reach Portal, provision TN(s) on the CPE with the Sequential Ring and simultaneous feature. Provisioning is self-explanatory. Please contact your AT&T representative, if you need help with the provisioning Network based feature.

Features - Not Supported

- Cisco UCM Codec negotiation of G.722.1
- Network-Based Blind Call Transfer
- Network-Based Consultative Call Transfer



Caveats

Auto-Attendant

- The CUC auto-attendant feature was used to test attendant functionality using the default codec G711 for auto attendant prompts. G729 prompts can be used but was not tested.

Hold/Resume & Music on Hold (MOH)

- Re-invites for hold/resume from PSTN network is potentially dependent on the carrier/network through which the call is traversing.

PBX Based Call Forward Unconditional

- PBX Based Unconditional Call Forwarding test is temporarily blocked due to AT&T Flexible Reach network issue.

SIP Provisional Acknowledgement/Early media

- To play early media sent by ATT, Cisco UCM needs to be enabled with PRACK if 1XX contains SDP on Cisco UCM SIP Profile.
- 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, This feature can be assigned on a per SIP trunk basis using SIP profiles. SIP PRACK provisioning on Cisco UCM 9.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.

AT&T IP Teleconferencing (IPTC)

Following scenarios were not executed due to limitations on AT&T network

- IPTC - Hold & Resume
- IPTC - PBX-Based Attended Transfer
- IPTC - PBX-Based 3-way Call Conference

Configuration Considerations

- To enable conference on AT&T IP Flexible Reach and Cisco UCM SIP trunk, it is required to configure a conference bridge (CFB) resource to initiate a three-way conference between end-points. See configuration section for details.
- Forwarded calls from Cisco UCM user to PSTN (out to AT&T's IP Flexible Reach service), 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 has been made that a typical customer will utilize extension numbers (4-digit assignments in this example) and map 10-digit DID number using Cisco UBE translation profile. This is because the Cisco UCM uses 4-digit extensions on Cisco UCM IP phones and 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 profile in Cisco UBE (See configuration section for details).
- Upon receiving inbound calls, AT&T SIP network will always have 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/receive G.711-only calls must configure separate voice class codec on Cisco UBE with G.711 as the first choice.
- SIP Profiles may also be employed to advertise desired RTP payload packet size.
- "voice-class sip privacy id" needs to configure in Cisco UBE dial peer to make call From a CPE Phone to PSTN phone, Pass Calling Party Number (CPN), marked private.
- This test environment is not configured with Cisco UBE High Availability (HA)
- Cisco UCM sends a SIP UPDATE message to Cisco UBE for a call transfer. AT&T network does not support the SIP UPDATE message causing the Cisco UBE to timeout and the call transfer is not completed. As a workaround, SIP profile has been applied on the Cisco UBE to remove UPDATE from the allowed headers (See configuration section for details).

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 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 services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found 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



ISR Configuration

CISCO_4K_ROUTER2#sh version

Cisco IOS XE Software, Version 03.16.01a.S - Extended Support Release

Cisco IOS Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.5(3)S1a, RELEASE SOFTWARE (fc1)

Technical Support: <http://www.cisco.com/techsupport>

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ROM: IOS-XE ROMMON

CISCO_4K_ROUTER2 uptime is 2 weeks, 22 hours, 50 minutes

Uptime for this control processor is 2 weeks, 22 hours, 51 minutes

System returned to ROM by reload

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EDCS# xxx Rev #

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Note: Testing was conducted in tekVizion labs



System image file is "bootflash:/isr4400-universalk9.03.16.01a.S.155-3.S1a-ext.SPA.bi"

Last reload reason: Reload Command

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:

<http://www.cisco.com/wwl/export/crypto/tool/stqrg.html>

If you require further assistance please contact us by sending email to export@cisco.com.

Suite License Information for Module:'esg'

Suite	Suite Current	Type	Suite Next reboot
-------	---------------	------	-------------------



FoundationSuiteK9	None	None	None
-------------------	------	------	------

securityk9

appxk9

AdvUCSuiteK9	None	None	None
--------------	------	------	------

uck9

cme-srst

cube

Technology Package License Information:

Technology	Technology-package	Technology-package
Current	Type	Next reboot

appxk9	appxk9	RightToUse	appxk9
uck9	uck9	Evaluation	uck9
securityk9	securityk9	EvalRightToUse	securityk9
ipbase	ipbasek9	Permanent	ipbasek9

cisco ISR4431/K9 (1RU) processor with 1659383K/6147K bytes of memory.

Processor board ID FTX1850ALVU

4 Gigabit Ethernet interfaces

32768K bytes of non-volatile configuration memory.



4194304K bytes of physical memory.

7057407K bytes of flash memory at bootflash:.

Configuration register is 0x2102

CISCO_4K_ROUTER2#sh run

Building configuration...

Current configuration : 13184 bytes

!

! Last configuration change at 06:38:10 UTC Tue Mar 1 2016 by cisco

!

version 15.5

service timestamps debug datetime msec

service timestamps log datetime msec

no platform punt-keepalive disable-kernel-core

!

hostname CISCO_4K_ROUTER2

!

boot-start-marker

boot system flash isr4400-universalk9.03.16.01a.S.155-3.S1a-ext.SPA.bin

boot-end-marker

!

!



```
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
enable secret 5 $1$zQRB$CCbzfD1aYzk3kPvzAm2KU0
enable password cisco
!
aaa new-model
!
!
!
!
!
!
!
!
aaa session-id common
!
ipc zone default
association 1
no shutdown
!
!
```



```
ip name-server 8.8.8.8
```

```
no ip domain lookup
```

```
!
```

```
!
```

```
!
```

```
ipv6 unicast-routing
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
subscriber templating
```

```
!
```

```
multilink bundle-name authenticated
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
cts logging verbose
```

```
!
```




!

voice service voip

no ip address trusted authenticate

rtp-port range 16384 32766

address-hiding¹

mode border-element²

allow-connections sip to sip³

no supplementary-service sip handle-replaces

redirect ip2ip

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

no fax-relay sg3-to-g3

sip

header-passing

error-passthru⁴

asserted-id pai⁵

early-offer forced⁶

no silent-discard untrusted

midcall-signaling passthru⁷

privacy-policy passthru⁸

¹ Hide signaling and media peer addresses from endpoints other than gateway.

² If the mode border-element command is not entered, border-element-related commands are not available for Cisco Unified Border Element voice connections on the Cisco 2900 and Cisco 3900 series platforms with a universal feature set. The mode border-element command is not available on any other platforms.

³ This command enables Cisco UBE basic IP-to-IP voice communication feature.

⁴ This command allows SIP error messages to pass-through end-to-end without modification through Cisco UBE.

⁵ 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).

⁶ This command enables delay offer-to-early offer conversion of initial SIP INVITE message to calls matched to this dial-peer level.

⁷ This command must be enabled at a global level to maintain integrity of SIP signaling between AT&T network and Cisco Unified Communications Manager (Cisco UCM) across Cisco UBE.



```
g729 annexb-all
!
voice class codec 19
  codec preference 1 g729r8 bytes 30
  codec preference 2 g711ulaw
!
voice class codec 2
  codec preference 1 g711ulaw
  codec preference 2 g729r8 bytes 30
!
voice class codec 3
  codec preference 1 g711ulaw
!
!
voice class sip-profiles 1
  response ANY sip-header Allow-Header modify "UPDATE," ""
  request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:732216\1@\2>"10
  request INVITE sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"11
  response ANY sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"
  request INVITE sdp-header Audio-Attribute add "a=ptime:30"12
```

⁸ This command allows for privacy settings to be transparently passed 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.

⁹ This command configures the codec preference to be assigned to dial-peers. Alternatively, single code can be configured into individual dial-peers.

¹⁰ This SIP profile expands the Diversion header number from a 4-digit extension to a full 10-digit DID number in order to obtain interoperability with AT&T's served users during call-forward scenarios. The six digits in "sip: 732216" are variable and must be replaced with the first 6 digits of the DID's provisioned for the customer site.

¹¹ 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 ms and it should be applied to dial-peers where G729 is the preferred codec. If the customer creates a dial-peer specifically for G711, a sip-profile without modifying the ptime value should be applied. This is because G711 RTP was not defaulting to 20ms.



```
!  
!  
!  
!  
!  
!  
!  
  
license udi pid ISR4431/K9 sn FOC18232988  
  
license boot level appxk9  
  
license boot level uck9  
  
license boot level securityk9  
  
!  
  
spanning-tree extend system-id  
  
!  
  
username cisco privilege 15 secret 5 $1$AGR7$e7pQx6UI0be3bzRbc0lr81  
  
!  
  
redundancy  
  
mode none  
  
!  
  
!  
  
!  
  
!  
  
!  
  
vlan internal allocation policy ascending
```

¹² This SIP profile is required in order to advertise theptime=30 attribute in the outgoing SIP INVITE from Cisco UBE to AT&T. Currently RFC's do not have a standard method to advertise ptime values for each offered codec within a SDP offering with multiple codecs. This SIP profile allows for Cisco UBE to include the ptime attribute with a value of 30ms.



!

!

!

interface GigabitEthernet0/0/0

ip address 10.64.4.20 255.255.0.0

shutdown

media-type rj45

negotiation auto

!

interface GigabitEthernet0/0/1

no ip address

shutdown

media-type rj45

negotiation auto

!

interface GigabitEthernet0/0/2¹³

ip address 10.80.22.75 255.255.255.0¹⁴

media-type rj45

negotiation auto

!

interface GigabitEthernet0/0/3¹⁵

description Wan Interface

no ip address

media-type rj45

¹³ LAN interface to Cisco UCM

¹⁴ Cisco UBE LAN interface IPv4 Address

¹⁵ WAN interface to AT&T



```
negotiation auto

ipv6 address 2620:96:C000:4::10/64

!

interface GigabitEthernet0

vrf forwarding Mgmt-intf

no ip address

shutdown

negotiation auto

!

interface Vlan1

no ip address

shutdown

!

ip forward-protocol nd

no ip http server

no ip http secure-server

ip route 0.0.0.0 0.0.0.0 192.65.79.97

ip route 10.64.0.0 255.255.0.0 10.80.22.1

ip route 10.80.22.0 255.255.255.0 10.80.22.1

ip route 172.16.0.0 255.255.0.0 10.80.22.1

!

!

ipv6 route ::/0 2620:96:C000:4::1

!

!

control-plane
```



```
!  
mgcp behavior rsip-range tgcp-only  
mgcp behavior comedia-role none  
mgcp behavior comedia-check-media-src disable  
mgcp behavior comedia-sdp-force disable  
!  
mgcp profile default  
!  
!  
!  
!  
dial-peer voice 200 voip  
description "Outgoing To AT&T .IP PBX facing side"  
session protocol sipv2  
incoming called-number [2-9]T  
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  
voice-class sip bind control source-interface GigabitEthernet0/0/2  
voice-class sip bind media source-interface GigabitEthernet0/0/2  
dtmf-relay rtp-nte  
no fax-relay sg3-to-g3  
fax rate 14400  
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
```



```
no vad

!

dial-peer voice 214 voip16

description "Outgoing To AT&T"-AT&T facing side

destination-pattern [2-9]T

session protocol sipv217

session target ipv6:[2001:1890:1253:0:12:38:172:80]

voice-class codec 118

voice-class sip asymmetric payload full19

voice-class sip asserted-id pai

voice-class sip privacy-policy passthru20

voice-class sip early-offer forced

voice-class sip profiles 121

voice-class sip bind control source-interface GigabitEthernet0/0/322

voice-class sip bind media source-interface GigabitEthernet0/0/3

dtmf-relay rtp-nte

no fax-relay sg3-to-g3

fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

no vad
```

¹⁶ Dial peer for AT&T facing network

¹⁷ Session protocol SIPv2 is used for this testing.

¹⁸ Assigns voice class codec 1 settings to dial-peer (codec support and filtering).

¹⁹ Configures the dynamic SIP asymmetric payload support.

²⁰ This command allows for privacy settings to be transparently passed between AT&T network and Cisco UCM. In this example, the command is set at the dial-peer level, you can also set the command at a global level to affect all dial-peers without necessarily setting the command on each dial-peer.

²¹ This command enables the dial peer to use SIP profile 1

²² Configure the Cisco UBE SIP messaging to use the HSRP virtual address in SIP messaging. Once HSRP is configured under the physical interface and the bind command is issued, calls to the physical IP address will fail. This is because the SIP listening socket is now bound to the virtual IP address but the signaling packets use the physical IP address, and therefore cannot be handled.



!

dial-peer voice 800 voip

description " Incoming AT&T to IP-PBX . AT&T facing side "

translation-profile incoming test+1

huntstop

session protocol sipv2

incoming called-number [37][13][24][12].....

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

voice-class sip bind control source-interface GigabitEthernet0/0/3

voice-class sip bind media source-interface GigabitEthernet0/0/3

dtmf-relay rtp-nte²³

no fax-relay sg3-to-g3

fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none²⁴

no vad

!

dial-peer voice 700 voip

description " Incoming AT&T to IP-PBX - IP-PBX facing side "

huntstop

destination-pattern [37][13][24][12].....

no modem passthrough

²³ This command used to pass RTP NTE (RFC2833) DTMF with respect to the dial peers used for the call.

²⁴ This command enables T38 fax protocol for calls terminating on this dial-peer



```
session protocol sipv2
session target ipv4:10.80.22.2:5060
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
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
no fax-relay sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 300 voip
description " Int'l calls to AT&T - AT&T facing side "
destination-pattern 011T
no modem passthrough
session protocol sipv2
session target ipv6:[2001:1890:1253:0:12:38:172:80]
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
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 400 voip
description " Int'l calls to AT&T - IP-PBX facing side "
no modem passthrough
session protocol sipv2
incoming called-number 011T
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
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
```



```
dial-peer voice 500 voip
description " N11 Calls to AT&T - AT&T facing side "
destination-pattern .11
no modem passthrough
session protocol sipv2
session target ipv6:[2001:1890:1253:0:12:38:172:80]
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
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 600 voip
description " N11 Calls to AT&T - IP-PBX facing side "
no modem passthrough
session protocol sipv2
incoming called-number .11
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
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 141 voip
description "Network Feature"
destination-pattern *..
no modem passthrough
session protocol sipv2
session target ipv6:[2001:1890:1253:0:12:38:172:80]
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
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax rate 14400
```



```
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
```

```
no vad
```

```
!
```

```
dial-peer voice 2151 voip
```

```
description "Outgoing To AT&T"-AT&T facing side
```

```
destination-pattern 7323204...
```

```
session protocol sipv2
```

```
session target ipv6:[2001:1890:1253:0:12:38:172:80]
```

```
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
```

```
voice-class sip bind control source-interface GigabitEthernet0/0/3
```

```
voice-class sip bind media source-interface GigabitEthernet0/0/3
```

```
dtmf-relay rtp-nte
```

```
fax rate 14400
```

```
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
```

```
no vad
```

```
!
```

```
dial-peer voice 2152 voip
```

```
description "Outgoing To AT&T"-AT&T facing side -BVoip Number
```

```
destination-pattern 8772888362
```

```
session protocol sipv2
```

```
session target ipv6:[2001:1890:1253:0:12:38:172:80]
```



```
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
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
gateway
media-inactivity-criteria all
timer receive-rtcp 5
timer receive-rtp 86400
!
sip-ua
no remote-party-id
timers options 1000
connection-reuse
protocol mode dual-stack
!
!
```



```
line con 0
stopbits 1
line aux 0
stopbits 1
line vty 0
session-timeout 90
exec-timeout 960 0
password tekV1z10n
no activation-character
logging synchronous
transport preferred ssh
transport input all
stopbits 1
line vty 1 4
exec-timeout 960 0
password tekV1z10n
logging synchronous
transport input all
!
!
end
```

Cisco UCM Configuration

The configuration screen shots shows general over view of lab configuration for this interoperability testing.



Cisco UCM Version

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation **Cisco Unified CM Administration**
administrator | Search Documentation |

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Cisco Unified CM Administration

System version: 11.0.1.10000-10

VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 0
@ 2.70GHz, disk 1: 80Gbytes, 4096Mbytes RAM, Partitions
aligned

User administrator last logged in to this cluster on Wednesday, February 10, 2016 6:19:21 AM CST, to node 10.80.14.2, from 172.16.29.40 using HTTPS

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Cisco UCM Audio Codec Preference List

Navigation Path: System → Region Information → Audio codec preference list

Cisco UCM 11.0 has a feature called Audio Codec Preference List. This feature allows to configure the order of audio codec preference both for Inter and Intra Region calls. Audio Codec Preference list is assigned to the Region used by the Device Pool for Phones and by Conference Bridges. Based on user



requirement, different codec regions can be assigned as their first choice codec with this configuration for inbound calls as well as conferences initiated by Cisco IP phones. Audio codec preference for outbound calls is determined by Cisco UBE (via configuration of voice-class codec)

Cisco Unified CM Administration
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Navigation **Cisco Unified CM Administration**
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Audio Codec Preference List Configuration Related Links: **Back To Find/List** ▾ **Go**

Save **X** Delete Copy Add New

Audio Codec Preference List Information

Name* G729 Preferred Codec List

Description* G729 Preferred Codec List

Codecs in List*

- G.729a 8k
- G.729b 8k
- G.729ab 8k
- G.729 8k
- G.711 U-Law 64k
- G.711 A-Law 64k
- G.711 U-Law 56k
- G.711 A-Law 56k
- AMR-WB (7k-24k)
- AMR (5k-13k)
- MP4A-LATM 128k
- AAC-LD (MP4A Generic)
- MP4A-LATM 64k
- MP4A-LATM 56k
- L16 256k
- MP4A-LATM 48k
- ISAC 32k
- MP4A-LATM 32k
- MP4A-LATM 24k
- G.722.1 32k
- G.722 64k
- G.722.1 24k
- G.722 56k
- G.722 48k
- ILBC 16k
- G.728 16k
- GSM Enhanced Full Rate 13k
- GSM Full Rate 13k
- GSM Half Rate 6k
- G.723.1 7k

Save Delete Copy Add New

Cisco UCM Region Configuration

Navigation Path: System → Region Information → Region



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Region Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Region Information
Name*

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
G729 Region	G729 Preferred Codec List	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps

NOTE: Regions not displayed Use System Default Use System Default Use System Default Use System Default

Modify Relationship to other Regions

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
<div>Default G711 region G729 Region</div>	<div>Keep Current Setting</div>	<div><input checked="" type="radio"/> Keep Current Setting <input type="radio"/> <input type="text" value=""/> kbps</div>	<div><input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text" value=""/> kbps</div>	<div><input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text" value=""/> kbps</div>

Save Delete Reset Apply Config Add New

Device Pool Configuration

Navigation Path: System → Device Pool

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Note: Testing was conducted in tekVizion labs



“G729” Device Pool is configured for testing the interoperability. No special consideration needs to be taken when configuring the 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.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Device Pool Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Device Pool Information
Device Pool: G729 Pool (15 members**)

Device Pool Settings

Device Pool Name*	G729 Pool
Cisco Unified Communications Manager Group*	Default
Calling Search Space for Auto-registration	< None >
Adjunct CSS	< None >
Reverted Call Focus Priority	Default
Intercompany Media Services Enrolled Group	< None >

Roaming Sensitive Settings

Date/Time Group*	CMLocal
Region*	G729 Region
Media Resource Group List	MRGL_MTP
Location	< None >
Network Locale	< None >
SRST Reference*	Disable
Connection Monitor Duration***	

Single Button Barge* Default

Join Across Lines* Default

Physical Location < None >

Device Mobility Group < None >

Wireless LAN Profile Group < None > [View Details](#)

Local Route Group Settings

Standard Local Route Group	< None >
----------------------------	----------

Device Pool Configuration (continued...)

Device Mobility Related Information****

Device Mobility Calling Search Space < None >
AAR Calling Search Space < None >
AAR Group < None >
Calling Party Transformation CSS < None >
Called Party Transformation CSS < None >

Geolocation Configuration

Geolocation < None >
Geolocation Filter < None >

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.

Clear Prefix Settings
Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default		< None >
International Number	Default		< None >
Unknown Number	Default		< None >
Subscriber Number	Default		< None >

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.

Clear Prefix Settings
Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default	0	< None >
International Number	Default	0	< None >
Unknown Number	Default	0	< None >
Subscriber Number	Default	0	< None >



Device Pool Configuration (continued...)

Phone Settings
Caller ID For Calls From This Phone
Calling Party Transformation CSS <input type="text" value=" < None >"/>
Connected Party Settings
Connected Party Transformation CSS <input type="text" value=" < None >"/>
Redirecting Party Settings
Redirecting Party Transformation CSS <input type="text" value=" < None >"/>
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Copy"/> <input type="button" value="Reset"/> <input type="button" value="Apply Config"/> <input type="button" value="Add New"/>



Annunciator Configuration

Navigation: Media Resource → Annunciator

Set Name* = ANN_2.

Set Description = ANN_clus32pubsub. This is used for this example

Set Device Pool* = G729 Pool.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Annunciator Configuration Related Links: Back To Find/List ▾ Go

Save Reset Apply Config

Status

Status: Ready

Annunciator Information

Registration: Registered with Cisco Unified Communications Manager clus32pubsub

IPv4 Address: 10.80.22.2

☒ Device is trusted

Server* clus32pubsub ▾

Name* ANN_2

Description ANN_clus32pubsub

Device Pool* G729 Pool ▾

Location* Hub_None ▾

Use Trusted Relay Point* Off ▾

Save Reset Apply Config



Conference Bridge Configuration

Navigation: Media Resources → Conference Bridge

Set Conference Bridge Type* = Cisco Conference Bridge Software.

Set Host Server = clus32pubsub. This is used for this example.

Set Conference Bridge Name* = CFB_2.

Set Description = CFB_clus32pubsub. This is used in this example.

Set Device Pool* = G729.

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Conference Bridge Configuration **Related Links:**
Back To Find/List

Conference Bridge Information

Conference Bridge : CFB_2 (CFB_clus32pubsub)
Registration Registered with Cisco Unified Communications Manager clus32pubsub
IP Address 10.80.22.2

Software Conference Bridge Info

Conference Bridge Type* Cisco Conference Bridge Software
Host Server clus32pubsub

⚠ Device is not trusted

Conference Bridge Name* CFB_2
Description CFB_clus32pubsub
Device Pool* G729 Pool ▾
Common Device Configuration < None > ▾
Location* Hub_None ▾
Use Trusted Relay Point* Default ▾



Media Termination Point Configuration

Navigation: Media Resource → Media Termination Point

Set Media Termination Point Name* = MTP_2

Set Description = MTP_clus32pubsub. This is used for this example

Set Device pool* = G729

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Conference Bridge Configuration **Related Links:**
Back To Find/List

Conference Bridge Information

Conference Bridge : CFB_2 (CFB_clus32pubsub)
Registration Registered with Cisco Unified Communications Manager clus32pubsub
IP Address 10.80.22.2

Software Conference Bridge Info

Conference Bridge Type* Cisco Conference Bridge Software
Host Server clus32pubsub

⚠ Device is not trusted

Conference Bridge Name* CFB_2
Description CFB_clus32pubsub
Device Pool* G729 Pool ▾
Common Device Configuration < None > ▾
Location* Hub_None ▾
Use Trusted Relay Point* Default ▾



Music on Hold Server Configuration

Navigation: Media Resources → Music on Hold Server

Set Music on Hold Server Name* = MOH_2.

Set Description = MOH_clus32pubsub. This is used for this example.

Set Device Pool* = G729

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Music On Hold (MOH) Server Configuration Related Links: Back To Find/List ▾ Go

Save Reset Apply Config

Device Information

Registration: Registered with Cisco Unified Communications Manager clus32pubsub
IPv4 Address: 10.80.22.2
☒ Device is trusted

Host Server* clus32pubsub ▾
Music On Hold Server Name* MOH_2
Description MOH_clus32pubsub
Device Pool* G729 Pool ▾
Location* Hub_None ▾
Maximum Half Duplex Streams* 250
Maximum Multi-cast Connections* 250000
Fixed Audio Source Device
Use Trusted Relay Point* Off ▾
Run Flag* Yes ▾

Multi-cast Audio Source Information

☐ Enable Multi-cast Audio Sources on this MOH Server

Base Multi-cast IP Address* 0.0.0.0
Base Multi-cast Port Number* 0 (Even numbers only)
Increment Multi-cast on * ☒ Port Number ☐ IP Address

Selected Multi-cast Audio Sources

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

Save Reset Apply Config



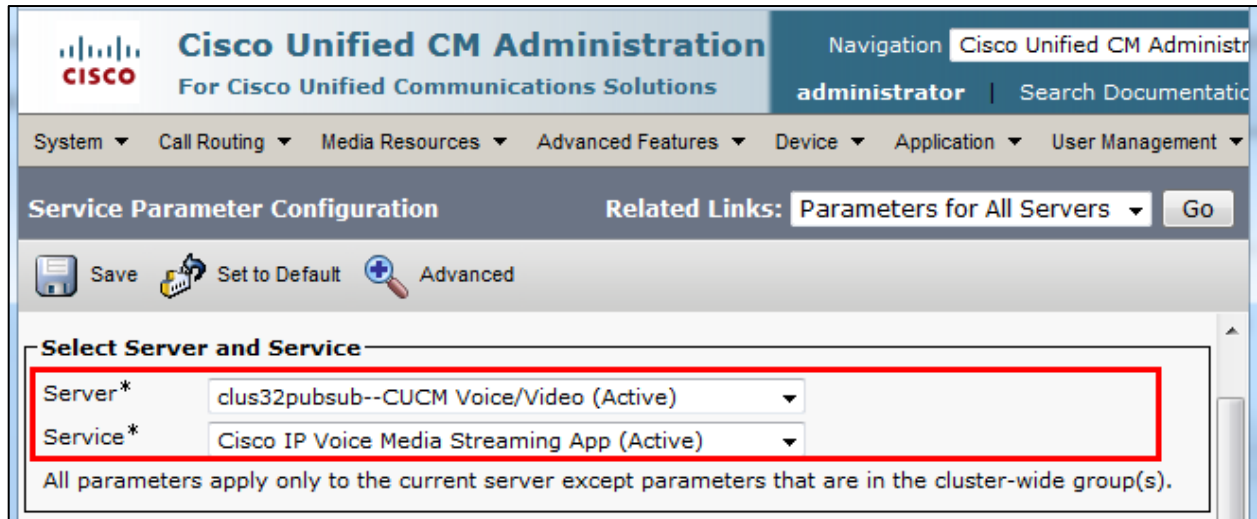
Music on Hold Service (IP Voice Media Streaming App) Parameter Settings

Navigation: System → Service Parameter

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

Select Server* = clus32pubsub--CUCM Voice/Video (Active). This is used in this example.

Select Service* = Cisco IP Voice Media Streaming App (Active)



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Service Parameter Configuration Related Links: Parameters for All Servers ▾ Go

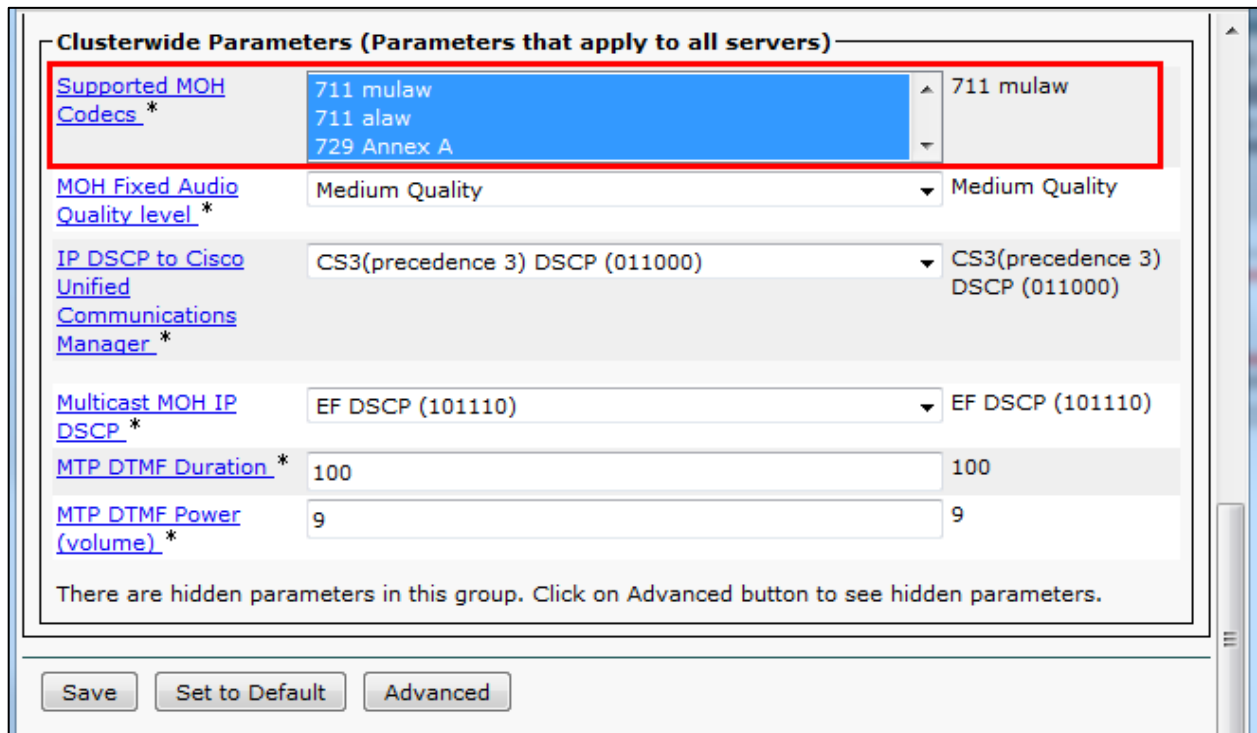
Save Set to Default Advanced

Select Server and Service

Server* clus32pubsub--CUCM Voice/Video (Active) ▾

Service* Cisco IP Voice Media Streaming App (Active) ▾

All parameters apply only to the current server except parameters that are in the cluster-wide group(s).



Clusterwide Parameters (Parameters that apply to all servers)

Supported MOH Codecs * 711 mulaw 711 alaw 729 Annex A

MOH Fixed Audio Quality level * Medium Quality ▾ Medium Quality

IP DSCP to Cisco Unified Communications Manager * CS3(precedence 3) DSCP (011000) ▾ CS3(precedence 3) DSCP (011000)

Multicast MOH IP DSCP * EF DSCP (101110) ▾ EF DSCP (101110)

MTP DTMF Duration * 100 100

MTP DTMF Power (volume) * 9 9

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Save Set to Default Advanced



Music on Hold Service (Duplex Streaming) Parameter Settings

Navigation: System → Service Parameter

Select Server* = clus32pubsub--CUCM Voice/Video (Active). This is used in this example.

Select Service* = Cisco CallManager (Active).

Select Duplex Streaming Enabled * = True

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the subtitle "For Cisco Unified Communications Solutions". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, and User Management. The main content area is titled "Service Parameter Configuration" and includes a "Related Links" section with a link to "Parameters for All Servers". Below this, there are buttons for "Save", "Set to Default", and "Advanced". The "Select Server and Service" section is highlighted with a red box, showing "Server*" set to "clus32pubsub--CUCM Voice/Video (Active)" and "Service*" set to "Cisco CallManager (Active)". Below this, a note states: "All parameters apply only to the current server except parameters that are in the cluster-wide group(s)." The "Clusterwide Parameters (Service)" section is also visible, showing a list of parameters. The "Duplex Streaming Enabled" parameter is highlighted with a red box, showing it is set to "True".

Clusterwide Parameters (Service)		
Default Network Hold MOH Audio Source ID *	1	1
Default User Hold MOH Audio Source ID *	1	1
Duplex Streaming Enabled *	True	False
Media Exchange Interface Capability Timer *	8	8
Send Multicast MOH in H.245 OLC Message *	True	True
Media Exchange Timer *	12	12
Media Exchange Stop Streaming Timer *	8	8
Open Video Channel Response Timer for SIP Interop *	500	500
Port Received Timer After Call Connection *	500	500
Media Resource Allocation Timer *	12	12
MTP and Transcoder Resource Throttling Percentage *	95	95



Media Resource Group Configuration

Navigation Path: Media Resources → Media Resources group

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) which is used to allocate media resources to groups of devices through Device Pools, or individually by configuring a valid MRGL at the device configuration page.

Set Name*= MRG_MTP - This is used for this example.

Set Description = MRG_MTP - This text is used to define this Media Resource Group List.

Set all Resources in the selected Media Resources Box.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documents

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Media Resource Group Configuration Related Links: Back To Find/List ▾ Go

Save Delete Copy Add New

Media Resource Group Status
Media Resource Group: MRG_MTP (used by 17 devices)

Media Resource Group Information

Name* MRG_MTP
Description MRG_MTP

Devices for this Group

Available Media Resources**

Selected Media Resources*
ANN_2 (ANN)
CFB_2 (CFB)
MOH_2 (MOH)
MTP_2 (MTP)

☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

Save Delete Copy Add New



Media Resource Group List Configuration

Navigation Path: Media Resources → Media Resource Group List

Set Name = MRGL_MTP.

Set selected Media Resource Groups = MRG_MTP.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documents

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Media Resource Group List Configuration Related Links: Back To Find/List Go

Save Delete Copy Add New

Media Resource Group List Status
Media Resource Group List: MRGL_MTP (used by 17 devices)

Media Resource Group List Information
Name * MRGL_MTP

Media Resource Groups for this List

Available Media Resource Groups

Selected Media Resource Groups: MRG_MTP

Save Delete Copy Add New



UC Service Configuration

Navigation: User Management → User Settings → UC Service

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation **Cisco Unified CM Administration** **Go**

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[System](#) ▾ [Call Routing](#) ▾ [Media Resources](#) ▾ [Advanced Features](#) ▾ [Device](#) ▾ [Application](#) ▾ [User Management](#) ▾ [Bulk Administration](#) ▾

Find and List UC Services

[+](#) Add New [Select All](#) [Clear All](#) [Delete Selected](#)

Status
 2 records found

UC Service (1 - 2 of 2) **Rows per Page** 50 ▾

Find UC Service where **Name** ▾ **begins with** ▾ [Find](#) [Clear Filter](#) [+](#) [-](#)

<input type="checkbox"/>	Name ▲	UC Service Type	Product Type	Host/IP Address	Port	Protocol
<input type="checkbox"/>	CTI_SRV	CTI	CTI	10.80.22.2	2748	TCP
<input type="checkbox"/>	IMP_SRV	IM and Presence	Unified CM (IM and Presence)	10.80.22.3		

[Add New](#) [Select All](#) [Clear All](#) [Delete Selected](#)



UC Service Configuration (Contd...)

Select UC Service Type: = CTI

Set Name* = CTI_SRV. This is used in this example.

Set Description = CTI for Jabber Clients. This is used in this example.

Set Host Name/IP Address* = 10.80.22.2 (Cisco UCM Address)

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UC Service Configuration

Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

Status
Status: Ready

Add a UC Service

UC Service Type: CTI
Product Type: CTI
Name* CTI_SRV
Description CTI for Jabber Clients
Host Name/IP Address* 10.80.22.2
Port 2748
Protocol: TCP

Save | Delete | Copy | Reset | Apply Config | Add New



UC Service Configuration (Contd...)

Select UC Service Type: = IM and Presence

Set Name* = IMP_SRV. This is used in this example.

Set Description = IM Presence. This is used in this example.

Set Host Name/IP Address* = 10.80.22.3 (Cisco UCM IM & Presence IP Address)

Cisco Unified CM Administration
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UC Service Configuration Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

Status
Status: Ready

Add a UC Service

UC Service Type: IM and Presence

Product Type*: Unified CM (IM and Presence)

Name*: IMP_SRV

Description: IM Presence

Host Name/IP Address*: 10.80.22.3

Save | Delete | Copy | Reset | Apply Config | Add New



Service Profile Configuration

Navigation: User Management → User Settings → Service Profile

Set Name* = Jabber_SVC_Profile. This is used in this example.

Set Description = Jabber Service Profile. This is used in this example.

Check - Make this the default service profile for the system.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documents

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Service Profile Configuration

Related Links: [Back To Find/List](#) ▾ Go

Save Delete Copy Add New

Status
 Status: Ready

Service Profile Information

Name*

Description

☒ Make this the default service profile for the system

Voicemail Profile

Primary

Secondary

Tertiary

[Credentials source for voicemail service](#)*

MailStore Profile

Primary

Secondary

Tertiary

[Inbox Folder](#)*

[Trash Folder](#)*

[Polling Interval \(in seconds\)](#)*

☒ [Allow dual folder mode](#)



Service Profile Configuration (Contd...)

Conferencing Profile
Primary
Secondary
Tertiary
Server Certificate Verification
[Credentials source for web conference service](#)*

Directory Profile
Primary
Secondary
Tertiary
☒ [Use UDS for Contact Resolution](#)
☒ [Use Logged On User Credential](#)
[Username](#)
[Password](#)
[Search Base 1](#)
[Search Base 2](#)
[Search Base 3](#)
☒ [Recursive Search on All Search Bases](#)
[Search Timeout \(seconds\)*](#)
[Base Filter \(Only used for Advance Directory\)](#)
[Predictive Search Filter \(Only used for Advance Directory\)](#)

IM and Presence Profile
Primary
Secondary
Tertiary

CTI Profile
Primary
Secondary
Tertiary

Video Conference Scheduling Portal Profile
Primary
Secondary
Tertiary



End User Configuration

Navigation: User Management → End User

Set User ID* = jabber1 – This is used in this example.

Set Password = Password for profile.

Set Directory URI = jabber1@lab.tekvizion.com.

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and a navigation menu with options like 'administrator', 'Search Documentation', 'About', and 'Lo'. Below this is a secondary navigation bar with categories like 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', and 'Bulk Administration'. The main content area is titled 'Find and List Users'. It features a toolbar with 'Add New', 'Select All', 'Clear All', and 'Delete Selected' buttons. A status box indicates '1 records found'. Below this is a table with columns: 'User ID', 'First Name', 'Last Name', 'Department', 'Directory URI', and 'User Status'. The table contains one row for the user 'jabber1', with 'cisco' as the department and 'jabber1@lab.tekvizion.com' as the directory URI. The user status is 'Enabled Local User'. The table is highlighted with a red border.

User ID	First Name	Last Name	Department	Directory URI	User Status
jabber1			cisco	jabber1@lab.tekvizion.com	Enabled Local User



End User Configuration (continued...)

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation [Cisco Unified CM Administration](#)
administrator | [Search Documentation](#)

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

End User Configuration **Related Links:** [Back to Find List Users ▾](#) [Go](#)

Save Delete Add New

User Information

User Status	Enabled Local User
User ID*	<input type="text" value="jabber1"/>
Password	<input type="password" value="....."/>
Confirm Password	<input type="password" value="....."/>
Self-Service User ID	<input type="text"/>
PIN	<input type="password" value="....."/>
Confirm PIN	<input type="password" value="....."/>
Last name*	<input type="text" value="cisco"/>
Middle name	<input type="text"/>
First name	<input type="text"/>
Title	<input type="text"/>
Directory URI	<input type="text" value="jabber1@lab.tekvizion.com"/>
Telephone Number	<input type="text"/>
Home Number	<input type="text"/>
Mobile Number	<input type="text"/>
Pager Number	<input type="text"/>
Mail ID	<input type="text"/>

[Edit Credential](#)

[Edit Credential](#)

Manager User ID	<input type="text"/>
Department	<input type="text"/>
User Locale	<input data-bbox="487 1396 1063 1423" type="text" value=" < None > "/>
Associated PC	<input type="text"/>
Digest Credentials	<input type="password" value="....."/>
Confirm Digest Credentials	<input type="password" value="....."/>
User Profile	<input "="" (factory="" data-bbox="487 1575 1047 1606" default)="" l="" standard="" type="text" value="Use System Default("/> View Details



End User Configuration (continued...)

Service Settings	
<input checked="" type="checkbox"/> Home Cluster	
<input checked="" type="checkbox"/> Enable User for Unified CM IM and Presence (Configure IM and Presence in the associated UC Service Profile)	
<input type="checkbox"/> Include meeting information in presence (Requires Exchange Presence Gateway to be configured on CUCM IM and Presence server)	
Presence Viewer for User	
UC Service Profile	Jabber_SVC_Profile View Details
Device Information	
Controlled Devices	CSFUser1 SEP00083031F2A8
Available Profiles	
v ^	
CTI Controlled Device Profiles	
v ^	
Extension Mobility	
Available Profiles	
v ^	
Controlled Profiles	
v ^	
Default Profile	-- Not Selected --
BLF Presence Group*	Standard Presence group
SUBSCRIBE Calling Search Space	< None >
<input checked="" type="checkbox"/> Allow Control of Device from CTI	
<input type="checkbox"/> Enable Extension Mobility Cross Cluster	
Directory Number Associations	
Primary Extension	< None >



Cisco IP Phone 7975 SCCP Configuration

Set MAC Address* = the below mac is used in this example.

Set Description = Cisco 7975 Phone. This text is used to identify this Phone.

Set Device Pool* = G729 Pool. This is used in this example.

Set Phone Button Template* = Standard 7975 SCCP. This is used in this example.

Set Media Resource Group List = MRGL_MTP. This is used in this example.

Set User Hold MOH Audio Source = 1-SampleAudioSource.

Set Network Hold MOH Audio Source = 1-SampleAudioSource

Association		Phone Type	
<div>Modify Button Items</div> <div>1 Line [1] - 2753 (no partition)</div> <div>2 Line [2] - Add a new DN</div> <div>3 Add a new SD</div> <div>4 Add a new SD</div> <div>5 Add a new SD</div> <div>6 Add a new SD</div> <div>7 Add a new SD</div> <div>8 Add a new SD</div> <div>9 ----- Unassigned Associated Items -----</div> <div>9 Add a new SD</div> <div>10 Add a new SURL</div> <div>11 Add a new BLF SD</div> <div>12 Add a new BLF Directed Call Park</div>		<div>Product Type: Cisco 7975</div> <div>Device Protocol: SCCP</div> <div>Real-time Device Status</div> <div>Registration: Registered with Cisco Unified Communications Manager clus32pubsub</div> <div>IPv4 Address: 172.16.31.122</div> <div>Active Load ID: SCCP75.9-4-2-1S</div> <div>Download Status: Unknown</div> <div>Device Information</div> <div><input checked="" type="checkbox"/> Device is Active</div> <div><input checked="" type="checkbox"/> Device is trusted</div> <div><div>MAC Address*00083031F5D4</div><div>DescriptionCisco 7975 Phone</div><div>Device Pool*G729 Pool View Details</div><div>Common Device ConfigurationATT_SIP_TRUNK View Details</div><div>Phone Button Template*Standard 7975 SCCP</div><div>Softkey TemplateStandard User</div></div>	
<div>13 CallBack</div> <div>14 Call Park</div> <div>15 Call Pickup</div> <div>16 Conference List</div> <div>17 Conference</div> <div>18 Do Not Disturb</div> <div>19 End Call</div> <div>20 Forward All</div> <div>21 Group Call Pickup</div> <div>22 Hold</div> <div>23 Hunt Group Logout</div> <div>24 Intercom [1] - Add a new Intercom</div> <div>25 Malicious Call Identification</div>		<div><div>Common Phone Profile*Standard Common Phone Profile View Details</div><div>Calling Search Space< None ></div><div>AAR Calling Search Space< None ></div><div><div>Media Resource Group ListMRGL_MTP</div><div>User Hold MOH Audio Source1-SampleAudioSource</div><div>Network Hold MOH Audio Source1-SampleAudioSource</div></div><div>Location*Hub_None</div><div>AAR Group< None ></div><div>User Locale< None ></div><div>Network Locale< None ></div></div>	



Cisco IP Phone 7975 SCCP Configuration (Continued...)

6 Meet Me Conference	Built In Bridge*	Default	
7 Mobility	Privacy*	Default	
8 New Call	Device Mobility	Default	View Current
9 Other Pickup	Mode*	Device Mobility Settings	
0 Quality Reporting Tool	Owner	<input type="radio"/> User <input checked="" type="radio"/> Anonymous (Public/Shared Space)	
1 Redial	Owner User ID		
2 Remove Last Participant	Phone	Default	
3 Transfer	Personalization*		
4 Video Mode	Services	Default	
5 Queue Status	Provisioning*		
6 Privacy	Phone Load Name		
7 None	Single Button Barge	Default	
	Join Across Lines	Default	
	Use Trusted Relay	Default	
	Point*		
	BLF Audible Alert	Default	
	Setting (Phone Idle)		
	*		
	BLF Audible Alert	Default	
	Setting (Phone		
	Busy)*		
	Always Use Prime	Default	
	Line*		
	Always Use Prime	Default	
	Line for Voice		
	**		
	*		

	Geolocation	< None >	
	<input checked="" type="checkbox"/> Retry Video Call as Audio		
	<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)		
	<input checked="" type="checkbox"/> Allow Control of Device from CTI		
	<input checked="" type="checkbox"/> Logged Into Hunt Group		
	<input type="checkbox"/> Remote Device		
	<input type="checkbox"/> Protected Device****		
	<input type="checkbox"/> Hot line Device*****		
	<input type="checkbox"/> Require off-premise location		
	Number Presentation Transformation		
	Caller ID For Calls From This Phone		
	Calling Party Transformation	< None >	
	CSS		
	<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)		
	Remote Number		
	Calling Party Transformation	< None >	
	CSS		
	<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)		



Cisco IP Phone 7975 SCCP Configuration (Continued...)

Protocol Specific Information	
Packet Capture Mode*	None
Packet Capture Duration	0
BLF Presence Group*	Standard Presence group
Device Security Profile*	Cisco 7975 - Standard SCCP Non-Secure Profile
SUBSCRIBE Calling Search Space	< None >
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Require DTMF Reception	
<input type="checkbox"/> RFC2833 Disabled	

Certification Authority Proxy Function (CAPF) Information	
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
<input type="button" value="Generate String"/>	
Key Size (Bits)*	2048
Operation Completes By	2015 3 27 12 (YYYY:MM:DD:HH)
Certificate Operation Status:	None
Note: Security Profile Contains Addition CAPF Settings.	

Expansion Module Information	
Module 1	< None >
Module 1 Load Name	
Module 2	< None >
Module 2 Load Name	
Module 3	< None >
Module 3 Load Name	



Cisco IP Phone 7975 SCCP Configuration (Continued...)

External Data Locations Information (Leave blank to use default)	
Information	<input type="text"/>
Directory	<input type="text"/>
Messages	<input type="text"/>
Services	<input type="text"/>
Authentication Server	<input type="text"/>
Proxy Server	<input type="text"/>
Idle	<input type="text"/>
Idle Timer (seconds)	<input type="text"/>
Secure Authentication URL	<input type="text"/>
Secure Directory URL	<input type="text"/>
Secure Idle URL	<input type="text"/>
Secure Information URL	<input type="text"/>
Secure Messages URL	<input type="text"/>
Secure Services URL	<input type="text"/>

Extension Information	
<input type="checkbox"/> Enable Extension Mobility	
Log Out Profile	-- Use Current Device Settings --
Log in Time	< None >
Log out Time	< None >


MLPP and Confidential Access Level Information	
MLPP Domain	< None >
MLPP Indication*	Default
MLPP Preemption*	Default
Confidential Access Mode	< None >
Confidential Access Level	< None >

Do Not Disturb	
<input type="checkbox"/> Do Not Disturb	
DND Option*	Use Common Phone Profile Setting
DND Incoming Call Alert	< None >



Cisco IP Phone 7975 SCCP Configuration (Continued...)

Secure Shell Information		
Secure Shell User	<input type="text" value="administrator"/>	
Secure Shell Password	<input type="password" value="....."/>	

Product Specific Configuration Layout		
	Parameter Value	Override Common Settings
		
<input type="checkbox"/> Disable Speakerphone		
<input type="checkbox"/> Disable Speakerphone and Headset		
PC Port *	<input type="text" value="Enabled"/>	
Back USB Port*	<input type="text" value="Enabled"/>	<input type="checkbox"/>
Side USB Port*	<input type="text" value="Enabled"/>	<input type="checkbox"/>
Cisco Camera *	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Console Access*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Video Capabilities*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Enable/Disable USB Classes	<input type="text" value="Mass Storage"/> <input type="text" value="Human Interface Device"/> <input type="text" value="Audio Class"/>	<input type="checkbox"/>
SDIO *	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Bluetooth *	<input type="text" value="Enabled"/>	<input type="checkbox"/>
Wifi *	<input type="text" value="Enabled"/>	<input type="checkbox"/>
Bluetooth Profiles*	<input type="text" value="Handsfree"/> <input type="text" value="Human Interface Device"/>	<input type="checkbox"/>

Settings Access*	<input type="text" value="Enabled"/>	<input type="checkbox"/>
Gratuitous ARP*	<input type="text" value="Disabled"/>	
PC Voice VLAN Access*	<input type="text" value="Enabled"/>	
Web Access*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Show All Calls on Primary Line*	<input type="text" value="Disabled"/>	
Days Display Not Active	<input type="text" value="Sunday"/> <input type="text" value="Monday"/> <input type="text" value="Tuesday"/>	<input type="checkbox"/>



Cisco IP Phone 7975 SCCP Configuration (Continued...)

Display On Time	07:30	<input type="checkbox"/>
Display On Duration	10:30	<input type="checkbox"/>
Display Idle Timeout	01:00	<input type="checkbox"/>
HTTPS Server*	http and https Enabled	<input type="checkbox"/>
Enable Power Save Plus	Sunday Monday Tuesday	<input type="checkbox"/>
Phone On Time	00:00	<input type="checkbox"/>
Phone Off Time	24:00	<input type="checkbox"/>
Phone Off Idle Timeout*	60	<input type="checkbox"/>
<input type="checkbox"/> Enable Audible Alert		<input type="checkbox"/>
EnergyWise Domain		<input type="checkbox"/>
EnergyWise Endpoint Security Secret		<input type="checkbox"/>
<input type="checkbox"/> Allow EnergyWise Overrides		<input type="checkbox"/>
Span to PC Port*	Disabled	
Logging Display*	Disabled	

Load Server		<input type="checkbox"/>
IPv6 Load Server		<input type="checkbox"/>
Recording Tone*	Disabled	
Recording Tone Local Volume*	100	
Recording Tone Remote Volume*	50	
Recording Tone Duration		
Display On When Incoming Call*	Enabled	<input type="checkbox"/>
RTCP*	Enabled	<input checked="" type="checkbox"/>
Log Server		<input checked="" type="checkbox"/>
IPv6 Log Server		<input type="checkbox"/>
Remote Log*	Disabled	<input type="checkbox"/>
Log Profile	Default Preset Telephony	<input type="checkbox"/>



Cisco IP Phone 7975 SCCP Configuration (Continued...)

Advertise G.722 and iSAC Codecs *	Use System Default	
Wideband Headset UI Control *	Enabled	
Wideband Headset *	Enabled	
Peer Firmware Sharing *	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): Switch Port *	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): PC Port *	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port *	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol (LLDP): PC Port *	Enabled	<input type="checkbox"/>

LLDP Asset ID		
LLDP Power Priority *	Unknown	
802.1x Authentication *	User Controlled	<input type="checkbox"/>
FIPS Mode *	Disabled	<input type="checkbox"/>
Detect Unified CM Connection Failure *	Normal	<input type="checkbox"/>
Switch Port Remote Configuration *	Disabled	<input type="checkbox"/>
PC Port Remote Configuration *	Disabled	<input type="checkbox"/>
Automatic Port Synchronization *	Disabled	<input type="checkbox"/>
Power Negotiation *	Enabled	<input type="checkbox"/>
Restrict Data Rates *	Disabled	
SSH Access *	Disabled	<input type="checkbox"/>
Incoming Call Toast Timer *	5	<input type="checkbox"/>
Provide Dial Tone from Release Button *	Disabled	<input type="checkbox"/>



Cisco IP Phone 7975 SCCP Configuration (Continued...)

Hide Video By Default*	Disabled	<input type="checkbox"/>
Background Image	<input type="text"/>	<input type="checkbox"/>
Simplified New Call UI*	Disabled	<input type="checkbox"/>
Enable VXC VPN for MAC	<input type="text"/>	
VXC VPN Option*	Dual Tunnel	<input type="checkbox"/>
VXC Challenge*	Challenge	<input type="checkbox"/>
VXC-M Servers	<input type="text"/>	<input type="checkbox"/>
Revert to All Calls*	Disabled	<input type="checkbox"/>
RTCP for Video*	Enabled	<input type="checkbox"/>
Record Call Log from Shared Line*	Disabled	<input type="checkbox"/>
Show Remote Private Calls*	Disabled	
Record Call Log For Remote Private Calls*	Enabled	
Show Call History for Selected Line Only.*	Disabled	<input type="checkbox"/>

Actionable Incoming Call Alert*	Disabled	<input type="checkbox"/>
DF bit*	0	<input type="checkbox"/>
Default Line Filter	<input type="text"/>	
Separate Audio and Video Mute*	Disabled	<input type="checkbox"/>
Softkey Control*	Feature Control Policy	<input type="checkbox"/>
Start Video Port	<input type="text"/>	<input type="checkbox"/>
Stop Video Port	<input type="text"/>	<input type="checkbox"/>
Lowest Alerting Line State Priority*	Disabled	<input type="checkbox"/>
TLS Resumption Timer*	3600	<input type="checkbox"/>
Audio EQ*	Default : Default	<input type="checkbox"/>

<input type="button" value="Save"/>	<input type="button" value="Delete"/>	<input type="button" value="Copy"/>	<input type="button" value="Reset"/>	<input type="button" value="Apply Config"/>	<input type="button" value="Add New"/>
-------------------------------------	---------------------------------------	-------------------------------------	--------------------------------------	---	--



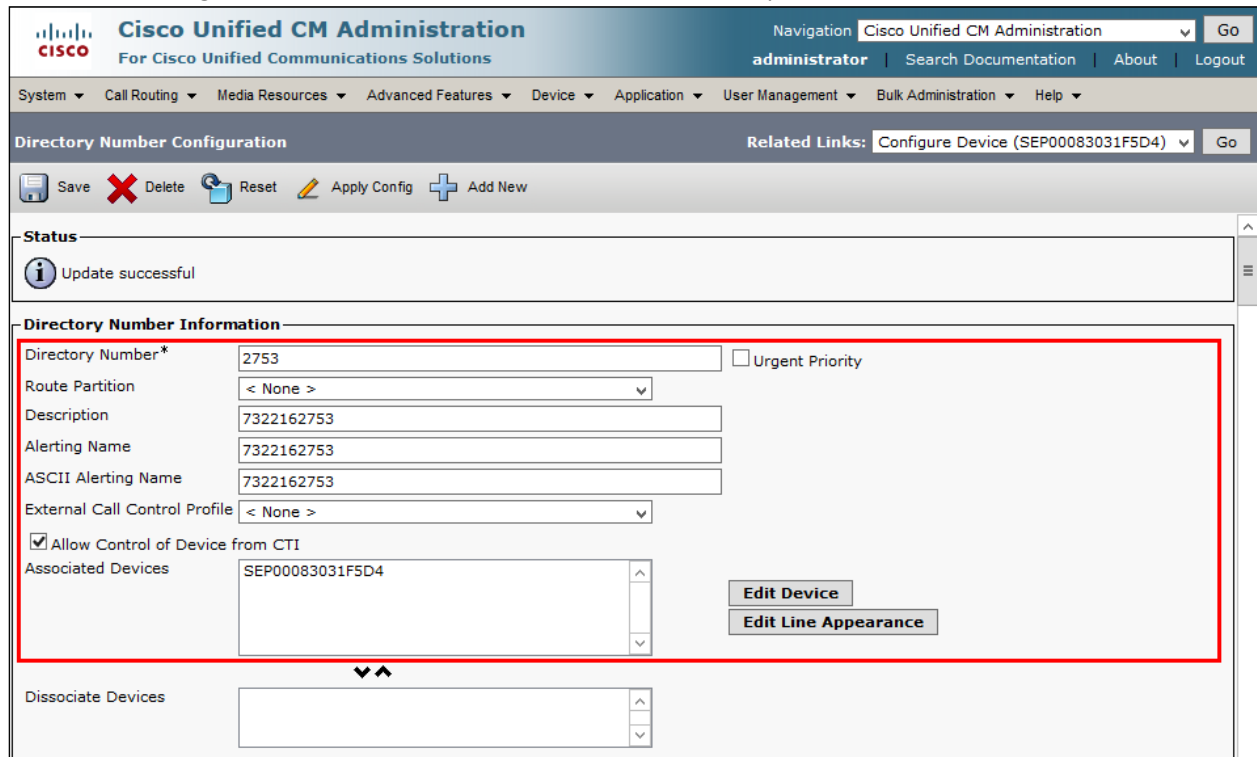
Cisco IP Phone 7975 SCCP Configuration (Continued...)

Set Directory Number* = 2753. This is used in this example.

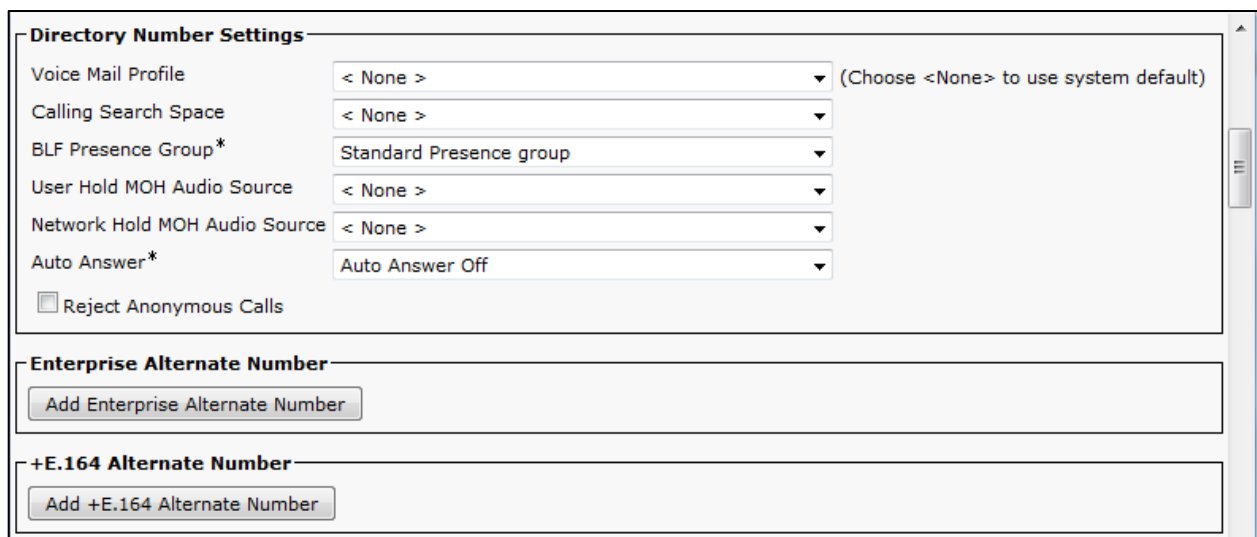
Set Description = 7322162753. This is used in this example.

Set Alerting Name = 7322162753. This is used in this example.

Set ASCII Alerting Name = 7322162753. This is used in this example.



The screenshot shows the Cisco Unified CM Administration interface for Directory Number Configuration. The page title is "Cisco Unified CM Administration" with the subtitle "For Cisco Unified Communications Solutions". The navigation bar includes "Navigation", "Cisco Unified CM Administration", "Go", "administrator", "Search Documentation", "About", and "Logout". The main navigation menu includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The "Directory Number Configuration" section is active, showing a "Related Links" section with "Configure Device (SEP00083031F5D4)" and "Go". The "Status" section shows "Update successful". The "Directory Number Information" section is highlighted with a red border and contains the following fields: "Directory Number*" (2753), "Route Partition" (< None >), "Description" (7322162753), "Alerting Name" (7322162753), "ASCII Alerting Name" (7322162753), "External Call Control Profile" (< None >), "Allow Control of Device from CTI" (checked), and "Associated Devices" (SEP00083031F5D4). There are "Edit Device" and "Edit Line Appearance" buttons. The "Dissociate Devices" section is at the bottom.



The screenshot shows the Cisco Unified CM Administration interface for Directory Number Settings. The "Directory Number Settings" section contains the following fields: "Voice Mail Profile" (< None >), "Calling Search Space" (< None >), "BLF Presence Group*" (Standard Presence group), "User Hold MOH Audio Source" (< None >), "Network Hold MOH Audio Source" (< None >), "Auto Answer*" (Auto Answer Off), and "Reject Anonymous Calls" (unchecked). The "Enterprise Alternate Number" section has an "Add Enterprise Alternate Number" button. The "+E.164 Alternate Number" section has an "Add +E.164 Alternate Number" button.



Cisco IP Phone 7975 SCCP Configuration (Continued...)

Directory URIs				
Primary	URI	Partition	Advertise Globally via ILS	Remove
<input checked="" type="radio"/>	<input type="text"/>	< None >	<input checked="" type="checkbox"/>	
<input type="button" value="Add Row"/>				

PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing	
Advertised Failover Number	< None >

AAR Settings		
	Voice Mail	AAR Destination Mask
AAR	<input type="checkbox"/> or	<input type="text"/>
<input checked="" type="checkbox"/> Retain this destination in the call forwarding history		

Call Forward and Call Pickup Settings		
	Voice Mail	Destination
Calling Search Space Activation Policy		Use System Default
Forward All	<input type="checkbox"/> or	<input type="text"/>
Secondary Calling Search Space for Forward All		< None >
Forward Busy Internal	<input type="checkbox"/> or	<input type="text"/>
Forward Busy External	<input type="checkbox"/> or	< None >
Forward No Answer Internal	<input type="checkbox"/> or	< None >
Forward No Answer External	<input type="checkbox"/> or	< None >
Forward No Coverage Internal	<input type="checkbox"/> or	< None >
Forward No Coverage External	<input type="checkbox"/> or	< None >



Cisco IP Phone 7975 SCCP Configuration (Continued...)

Forward on CTI Failure	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Unregistered Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Unregistered External	<input type="checkbox"/> or	<input type="text"/>	< None >
No Answer Ring Duration (seconds)	<input type="text"/>		
Call Pickup Group	< None >		

Park Monitoring			
	Voice Mail	Destination	Calling Search Space
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the parker's line.
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the parker's line.
Park Monitoring Reversion Timer		<input type="text"/>	A blank value will use value set in Park Monitoring

Reversion Timer service parameter

MLPP Alternate Party And Confidential Access Level Settings	
Target (Destination)	<input type="text"/>
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	<input type="text"/>
Confidential Access Mode	< None >
Confidential Access Level	< None >
Call Control Agent Profile	< None >

Line Settings for All Devices	
Hold Reversion Ring Duration (seconds)	<input type="text"/> Setting the Hold Reversion Ring Duration to zero will disable the feature
Hold Reversion Notification Interval (seconds)	<input type="text"/> Setting the Hold Reversion Notification Interval to zero will disable the feature
Party Entrance Tone*	Default



Cisco IP Phone 7975 SCCP Configuration (Continued...)

Set Display (caller ID) = 7322162753. This is used in this example.

Set ASCII Display (caller ID) = 7322162753. This is used in this example.

Set Line Text Label = 7322162753. This is used in this example.

Set External Phone Number Mask = 7322162753. This is used in this example.

Line 1 on Device SEP00083031F5D4

Display (Caller ID)	7322162753	Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Caller ID)	7322162753	
Line Text Label	7322162753	
External Phone Number Mask	7322162753	
Visual Message Waiting Indicator Policy*	Use System Policy	
Audible Message Waiting Indicator Policy*	Default	
Ring Setting (Phone Idle)*	Use System Default	
Ring Setting (Phone Active)	Use System Default	Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)	Use System Default	
Call Pickup Group Audio Alert Setting(Phone Active)	Use System Default	
Recording Option *	Call Recording Disabled	
Recording Profile	< None >	
Recording Media Source*	Gateway Preferred	
Monitoring Calling Search Space	< None >	
<input checked="" type="checkbox"/> Log Missed Calls		

Cisco IP Phone 7975 SCCP Configuration (Continued...)

Multiple Call/Call Waiting Settings on Device SEP00083031F2A8

Note: The range to select the Max Number of calls is: 1-200

Maximum Number of Calls*

Busy Trigger* (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEP00083031F2A8

☒ Caller Name

☐ Caller Number

☐ Redirected Number

☒ Dialed Number

Users Associated with Line

Associate End Users

Save

Delete

Reset

Apply Config

Add New

Cisco IP Phone 9971 SIP Configuration



Set MAC Address* = the below mac is used in this example.
Set Description = Cisco 9971. This text is used to identify this Phone.
Set Device Pool* = G729. This is used in this example.
Set Phone Button Template* = Standard 9971 SIP. This is used in this example.
Set Media Resource Group List = MRGL_MTP. This is used in this example.
Set User Hold MOH Audio Source = 1-SampleAudioSource.
Set Network Hold MOH Audio Source = 1-SampleAudioSource

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | Go
administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Phone Configuration | Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

Association

1 Line [1] - 2754 (no partition)
2 Line [2] - Add a new DN
3 Add a new SD
4 Add a new SD
5 Add a new SD
6 Add a new SD
----- Unassigned Associated Items -----
7 Add a new SD
8 All Calls
9 Add a new BLF Directed Call Park
10 Call Park
11 Call Pickup
12 CallBack
13 Group Call Pickup
14 Hunt Group Logout
15 Intercom [1] - Add a new Intercom
16 Malicious Call Identification
17 Meet Me Conference

Phone Type

Product Type: Cisco 9971
Device Protocol: SIP

Real-time Device Status

Registration: Registered with Cisco Unified Communications Manager clus32pubsub
IPv4 Address: 172.16.31.114
Active Load ID: sip9971.9-4-2-13
Inactive Load ID: sip9971.9-4-1-9
Download Status: Unknown

Device Information

Device is Active
Device is trusted

MAC Address*: C078BCA1B7DD
Description: Cisco 9971
Device Pool*: G729 Pool | View Details
Common Device Configuration: ATT_SIP_TRUNK | View Details
Phone Button Template*: Standard 9971 SIP
Softkey Template: Standard User
Common Phone Profile*: Standard Common Phone Profile | View Details
Calling Search Space: < None >

Common Phone Profile*: Standard Common Phone Profile | View Details
Calling Search Space: < None >
AAR Calling Search Space: < None >

Media Resource Group List: MRGL_MTP
User Hold MOH Audio Source: 1-SampleAudioSource
Network Hold MOH Audio Source: 1-SampleAudioSource

Location*: Hub_None
AAR Group: < None >
User Locale: < None >
Network Locale: < None >

Cisco IP Phone 9971 SIP Configuration (Continued...)

28	Alerting Calls	Built In Bridge*	Default
29	Queue Status	Privacy*	Default
30	Privacy	Device Mobility Mode*	Default View
31	None	Current Device Mobility Settings	
		Owner	<input type="radio"/> User <input checked="" type="radio"/> Anonymous (Public/Shared Space)
		Owner User ID	
		Phone Personalization*	Default
		Services Provisioning*	Default
		Phone Load Name	
		Use Trusted Relay Point*	Default
		BLF Audible Alert Setting (Phone Idle)*	Default
		BLF Audible Alert Setting (Phone Busy)*	Default
		Always Use Prime Line*	Default
		Always Use Prime Line for Voice Message*	Default
		Geolocation	< None >
		Geolocation	< None >
		Feature Control Policy	< None >
		<input type="checkbox"/> Ignore Presentation Indicators (internal calls only) <input checked="" type="checkbox"/> Allow Control of Device from CTI <input checked="" type="checkbox"/> Logged Into Hunt Group <input type="checkbox"/> Remote Device <input type="checkbox"/> Protected Device**** <input type="checkbox"/> Require off-premise location	
		Number Presentation Transformation	
		Caller ID For Calls From This Phone	
		Calling Party Transformation CSS	< None >
		<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)	
		Remote Number	
		Calling Party Transformation CSS	< None >
		<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)	

Cisco IP Phone 9971 SIP Configuration (Continued...)

Protocol Specific Information	
Packet Capture Mode*	None
Packet Capture Duration	0
BLF Presence Group*	Standard Presence group
SIP Dial Rules	< None >
MTP Preferred Originating Codec*	711ulaw
Device Security Profile*	Cisco 9971 - Standard SIP Non-Secure Profile
Rerouting Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile for ATT View Details
Digest User	< None >
<input type="checkbox"/> Media Termination Point Required <input type="checkbox"/> Unattended Port <input type="checkbox"/> Require DTMF Reception	

Certification Authority Proxy Function (CAPF) Information	
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
Generate String	
Key Size (Bits)*	2048
Operation Completes By	2015 6 20 12 (YYYY:MM:DD:HH)
Certificate Operation Status:	None
Note: Security Profile Contains Addition CAPF Settings.	

Expansion Module Information	
Module 1	< None >
Module 1 Load Name	
Module 2	< None >
Module 2 Load Name	
Module 3	< None >
Module 3 Load Name	

Cisco IP Phone 9971 SIP Configuration (Continued...)

External Data Locations Information (Leave blank to use default)

Information

Directory

Messages

Services

Authentication Server

Proxy Server

Idle

Idle Timer (seconds)

Secure Authentication URL

Secure Directory URL

Secure Idle URL

Secure Information URL

Secure Messages URL

Secure Services URL

Extension Information

☐ Enable Extension Mobility

Log Out Profile -- Use Current Device Settings --

Log in Time < None >

Log out Time < None >

MLPP and Confidential Access Level Information

MLPP Domain< None >

MLPP Indication*Default

MLPP Preemption*Default

Confidential Access Mode< None >

Confidential Access Level< None >

Do Not Disturb


☐ Do Not Disturb

DND Option*Use Common Phone Profile Setting

DND Incoming Call Alert< None >

Cisco IP Phone 9971 SIP Configuration (Continued...)

Secure Shell Information		
Secure Shell User	<input type="text" value="administrator"/>	
Secure Shell Password	<input type="password" value="....."/>	

Product Specific Configuration Layout		
<div>  <div>Parameter Value</div> <div>Override Common Settings</div> </div>		
<input type="checkbox"/> Disable Speakerphone <input type="checkbox"/> Disable Speakerphone and Headset		
PC Port *	<input type="text" value="Enabled"/>	
Back USB Port*	<input type="text" value="Enabled"/>	<input type="checkbox"/>
Side USB Port*	<input type="text" value="Enabled"/>	<input type="checkbox"/>
Cisco Camera*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Console Access*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Video Capabilities*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Enable/Disable USB Classes	<input type="text" value="Mass Storage"/> <input type="text" value="Human Interface Device"/> <input type="text" value="Audio Class"/>	<input type="checkbox"/>
SDIO *	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Bluetooth *	<input type="text" value="Enabled"/>	<input type="checkbox"/>
Wifi *	<input type="text" value="Enabled"/>	<input type="checkbox"/>
Bluetooth Profiles*	<input type="text" value="Handsfree"/> <input type="text" value="Human Interface Device"/>	<input type="checkbox"/>

Settings Access *	<input type="text" value="Enabled"/>	<input type="checkbox"/>
Gratuitous ARP*	<input type="text" value="Disabled"/>	
PC Voice VLAN Access*	<input type="text" value="Enabled"/>	
Web Access*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Show All Calls on Primary Line*	<input type="text" value="Disabled"/>	
Days Display Not Active	<input type="text" value="Sunday"/> <input type="text" value="Monday"/> <input type="text" value="Tuesday"/>	<input type="checkbox"/>

Cisco IP Phone 9971 SIP Configuration (Continued...)

Display On Time	07:30	<input type="checkbox"/>
Display On Duration	10:30	<input type="checkbox"/>
Display Idle Timeout	01:00	<input type="checkbox"/>
HTTPS Server*	http and https Enabled	<input type="checkbox"/>
Enable Power Save Plus	Sunday Monday Tuesday	<input type="checkbox"/>
Phone On Time	00:00	<input type="checkbox"/>
Phone Off Time	24:00	<input type="checkbox"/>
Phone Off Idle Timeout*	60	<input type="checkbox"/>
<input type="checkbox"/> Enable Audible Alert		<input type="checkbox"/>
EnergyWise Domain		<input type="checkbox"/>
EnergyWise Endpoint Security Secret		<input type="checkbox"/>
<input type="checkbox"/> Allow EnergyWise Overrides		<input type="checkbox"/>
Span to PC Port*	Disabled	<input type="checkbox"/>
Logging Display*	Disabled	<input type="checkbox"/>

Load Server		<input type="checkbox"/>
IPv6 Load Server		<input type="checkbox"/>
Recording Tone*	Disabled	<input type="checkbox"/>
Recording Tone Local Volume*	100	
Recording Tone Remote Volume*	50	
Recording Tone Duration		
Display On When Incoming Call*	Enabled	<input type="checkbox"/>
RTCP*	Enabled	<input checked="" type="checkbox"/>
Log Server		<input type="checkbox"/>
IPv6 Log Server		<input type="checkbox"/>
Remote Log*	Disabled	<input type="checkbox"/>
Log Profile	Default Preset Telephony	<input type="checkbox"/>

Cisco IP Phone 9971 SIP Configuration (Continued...)



Advertise G.722 and iSAC Codecs *	Use System Default ▼	
Wideband Headset UI Control*	Enabled ▼	
Wideband Headset*	Enabled ▼	
Peer Firmware Sharing*	Enabled ▼	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): Switch Port*	Enabled ▼	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): PC Port*	Enabled ▼	<input type="checkbox"/>
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled ▼	<input type="checkbox"/>
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled ▼	<input type="checkbox"/>

LLDP Asset ID	<input type="text"/>	
LLDP Power Priority*	Unknown ▼	
802.1x Authentication*	User Controlled ▼	<input type="checkbox"/>
FIPS Mode*	Disabled ▼	<input type="checkbox"/>
Detect Unified CM Connection Failure*	Normal ▼	<input type="checkbox"/>
Switch Port Remote Configuration*	Disabled ▼	<input type="checkbox"/>
PC Port Remote Configuration*	Disabled ▼	<input type="checkbox"/>
Automatic Port Synchronization*	Disabled ▼	<input type="checkbox"/>
Power Negotiation*	Enabled ▼	<input type="checkbox"/>
Restrict Data Rates*	Disabled ▼	
SSH Access*	Disabled ▼	<input type="checkbox"/>
Incoming Call Toast Timer*	5 ▼	<input type="checkbox"/>
Provide Dial Tone from Release Button*	Disabled ▼	<input type="checkbox"/>

Cisco IP Phone 9971 SIP Configuration (Continued...)

Hide Video By Default*	Disabled	<input type="checkbox"/>
Background Image		<input type="checkbox"/>
Simplified New Call UI*	Disabled	<input type="checkbox"/>
Enable VXC VPN for MAC		<input type="checkbox"/>
VXC VPN Option*	Dual Tunnel	<input type="checkbox"/>
VXC Challenge*	Challenge	<input type="checkbox"/>
VXC-M Servers		<input type="checkbox"/>
Revert to All Calls*	Disabled	<input type="checkbox"/>
RTCP for Video*	Enabled	<input type="checkbox"/>
Record Call Log from Shared Line*	Disabled	<input type="checkbox"/>
Show Remote Private Calls*	Disabled	<input type="checkbox"/>
Record Call Log For Remote Private Calls*	Enabled	<input type="checkbox"/>
Show Call History for Selected Line Only.*	Disabled	<input type="checkbox"/>

Actionable Incoming Call Alert*	Disabled	<input type="checkbox"/>
DF bit*	0	<input type="checkbox"/>
Default Line Filter		<input type="checkbox"/>
Separate Audio and Video Mute*	Disabled	<input type="checkbox"/>
Softkey Control*	Feature Control Policy	<input type="checkbox"/>
Start Video Port		<input type="checkbox"/>
Stop Video Port		<input type="checkbox"/>
Lowest Alerting Line State Priority*	Disabled	<input type="checkbox"/>
TLS Resumption Timer*	3600	<input type="checkbox"/>
Audio EQ*	Default : Default	<input type="checkbox"/>

Cisco IP Phone 9971 SIP Configuration (Continued...)



Set Directory Number* = 2754. This is used in this example.
Set Description = 7322162754. This is used in this example.
Set Alerting Name = 7322162754. This is used in this example.
Set ASCII Alerting Name = 7322162754. This is used in this example.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the subtitle "For Cisco Unified Communications Solutions". The user is logged in as "administrator". The main menu shows various system configuration options like System, Call Routing, Media Resources, etc. The current page is "Directory Number Configuration" for the device "SEPC07BBCA1B7DD".

The configuration form is divided into several sections:

- Status:** Shows "Status: Ready".
- Directory Number Information:** This section is highlighted with a red border. It contains fields for:
 - Directory Number*: 2754
 - Route Partition: < None >
 - Description: 7322162754
 - Alerting Name: 7322162754
 - ASCII Alerting Name: 7322162754
 - External Call Control Profile: < None >
 - ☒ Allow Control of Device from CTI
 - Associated Devices: SEPC07BBCA1B7DDButtons for "Edit Device" and "Edit Line Appearance" are also present.
- Directory Number Settings:** Contains settings for Voice Mail Profile, Calling Search Space, BLF Presence Group (Standard Presence group), User Hold MOH Audio Source, Network Hold MOH Audio Source, Auto Answer* (Auto Answer Off), and a checkbox for Reject Anonymous Calls.
- Enterprise Alternate Number:** Includes a button "Add Enterprise Alternate Number".
- +E.164 Alternate Number:** Includes a button "Add +E.164 Alternate Number".

Cisco IP Phone 9971 SIP Configuration (Continued...)



Directory URIs				
Primary	URI	Partition	Advertise Globally via ILS	Remove
<input checked="" type="radio"/>	<input type="text"/>	< None >	<input checked="" type="checkbox"/>	
<input type="button" value="Add Row"/>				

PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing	
Advertised Failover Number	< None >

AAR Settings		
	Voice Mail	AAR Destination Mask
AAR	<input type="checkbox"/> or <input type="text"/>	< None >
<input checked="" type="checkbox"/> Retain this destination in the call forwarding history		

Call Forward and Call Pickup Settings		
	Voice Mail	Destination
Calling Search Space Activation Policy		Use System Default
Forward All	<input type="checkbox"/> or <input type="text"/>	< None >
Secondary Calling Search Space for Forward All		< None >
Forward Busy Internal	<input type="checkbox"/> or <input type="text"/>	< None >
Forward Busy External	<input type="checkbox"/> or <input type="text"/>	< None >
Forward No Answer Internal	<input type="checkbox"/> or <input type="text"/>	< None >
Forward No Answer External	<input type="checkbox"/> or <input type="text"/>	< None >
Forward No Coverage Internal	<input type="checkbox"/> or <input type="text"/>	< None >
Forward No Coverage External	<input type="checkbox"/> or <input type="text"/>	< None >

Cisco IP Phone 9971 SIP Configuration (Continued...)



Forward on CTI Failure	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Unregistered Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Unregistered External	<input type="checkbox"/> or	<input type="text"/>	< None >
No Answer Ring Duration (seconds)	<input type="text"/>		
Call Pickup Group	< None >		

Park Monitoring			
	Voice Mail	Destination	Calling Search Space
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the parker's line.
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the parker's line.
Park Monitoring Reversion Timer	<input type="text"/>		A blank value will use value set in Park Monitoring

MLPP Alternate Party And Confidential Access Level Settings	
Target (Destination)	<input type="text"/>
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	<input type="text"/>
Confidential Access Mode	< None >
Confidential Access Level	< None >
Call Control Agent Profile	< None >

Line Settings for All Devices	
Hold Reversion Ring Duration (seconds)	<input type="text"/> Setting the Hold Reversion Ring Duration to zero will disable the feature
Hold Reversion Notification Interval (seconds)	<input type="text"/> Setting the Hold Reversion Notification Interval to zero will disable the feature
Party Entrance Tone*	Default

Cisco IP Phone 9971 SIP Configuration (Continued...)



Set Display (caller ID) = 7322162754. This is used in this example.

Set ASCII Display (caller ID) = 7322162754. This is used in this example.

Set Line Text Label = 7322162754. This is used in this example.

Set External Phone Number Mask = 7322162754. This is used in this example.

Line 1 on Device SEPC078BCA1B7DD

Display (Caller ID)	<input type="text" value="7322162754"/>	Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Caller ID)	<input type="text" value="7322162754"/>	
Line Text Label	<input type="text" value="7322162754"/>	
External Phone Number Mask	<input type="text" value="7322162754"/>	
Visual Message Waiting Indicator Policy*	<input type="text" value="Use System Policy"/>	
Audible Message Waiting Indicator Policy*	<input type="text" value="Default"/>	
Ring Setting (Phone Idle)*	<input type="text" value="Use System Default"/>	
Ring Setting (Phone Active)	<input type="text" value="Use System Default"/>	Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)	<input type="text" value="Use System Default"/>	
Call Pickup Group Audio Alert Setting(Phone Active)	<input type="text" value="Use System Default"/>	
Recording Option*	<input type="text" value="Call Recording Disabled"/>	
Recording Profile	<input type="text" value="< None >"/>	
Recording Media Source*	<input type="text" value="Gateway Preferred"/>	
Monitoring Calling Search Space	<input type="text" value="< None >"/>	
<input checked="" type="checkbox"/> Log Missed Calls		

Cisco IP Phone 9971 SIP Configuration (Continued...)

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Note: Testing was conducted in tekVizion labs

Multiple Call/Call Waiting Settings on Device SEPC07BBCA1B872

Note:The range to select the Max Number of calls is: 1-200

Maximum Number of Calls*

Busy Trigger* (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEPC07BBCA1B872

☒ Caller Name

☐ Caller Number

☐ Redirected Number

☒ Dialed Number

Users Associated with Line

SIP Trunk Security Profile Configuration used by SIP trunk to Cisco UBE

Navigation: System → Security → SIP Trunk Security Profile



Set Name* = ATT Non Secure SIP Trunk Profile. This is used in this example.

Set Description = Non Secure SIP Trunk Profile authenticated by null String. This is used in this example.

Set Device Security Mode = Non Secure.

Set Incoming Transport Type* = TCP+UDP.

Set Outgoing Transport Type = UDP.

SIP Trunk Security Profile Configuration

Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Reset Apply Config Add New

SIP Trunk Security Profile Information

Name* ATT Non Secure SIP Trunk Profile

Description Non Secure SIP Trunk Profile authenticated by null String

Device Security Mode Non Secure

Incoming Transport Type* TCP+UDP

Outgoing Transport Type UDP

☐ Enable Digest Authentication

Nonce Validity Time (mins)* 600

X.509 Subject Name

Incoming Port* 5060

☐ Enable Application level authorization

☐ Accept presence subscription

☐ Accept out-of-dialog refer**

☐ Accept unsolicited notification

☐ Accept replaces header

☐ Transmit security status

☐ Allow charging header

SIP V.150 Outbound SDP Offer Filtering* Use Default Filter

Save Delete Copy Reset Apply Config Add New

SIP Profile Configuration used by SIP trunk to Cisco UBE

Navigation: Device → Device Settings → SIP Profile

Set SIP profile Name * = Standard SIP Profile w/Early Media Disabled. This is used for this example



Check Disable Early Media on 180

Set SIP Rel1xx Options* = Send PRACK if 1xx contains SDP

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".

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration
administrator | Search Documentation | About

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration

SIP Profile Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

SIP Profile Information

Name* Standard SIP Profile for ATT

Description Standard SIP Profile for ATT

Default MTP Telephony Event Payload Type* 101

Early Offer for G.Clear Calls* Disabled

User-Agent and Server header information* Send Unified CM Version Information as User-Ager

Version in User Agent and Server Header* Major And Minor

Dial String Interpretation* Phone number consists of characters 0-9, *, #, an

Confidential Access Level Headers* Disabled

☐ Redirect by Application

☒ Disable Early Media on 180

☐ Outgoing T.38 INVITE include audio mline

☐ Use Fully Qualified Domain Name in SIP Requests

☐ Assured Services SIP conformance

SIP Profile Configuration used by SIP trunk to Cisco UBE (Continued...)

SDP Information	
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer*	Default
<input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change	
<input type="checkbox"/> Allow RR/RS bandwidth modifier (RFC 3556)	

Parameters used in Phone	
Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off

Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial



SIP Profile Configuration used by SIP trunk to Cisco UBE (Continued...)

<input checked="" type="checkbox"/>	Conference Join Enabled
<input type="checkbox"/>	RFC 2543 Hold
<input checked="" type="checkbox"/>	Semi Attended Transfer
<input type="checkbox"/>	Enable VAD
<input type="checkbox"/>	Stutter Message Waiting
<input type="checkbox"/>	MLPP User Authorization

Normalization Script

Normalization Script < None >

☐ Enable Trace

	Parameter Name	Parameter Value		
1	<input type="text"/>	<input type="text"/>	<input type="button" value="+"/>	<input type="button" value="-"/>

Incoming Requests FROM URI Settings

Caller ID DN

Caller Name

Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*

Never

RSVP Over SIP*

Local RSVP

Resource Priority Namespace List

< None >

☐ Fall back to local RSVP

SIP Rel1XX Options*

Send PRACK if 1xx Contains SDP

Video Call Traffic Class*

Mixed

Calling Line Identification Presentation*

Default

Session Refresh Method*

Invite

Early Offer support for voice and video calls*

Disabled (Default value)

☐ Enable ANAT

☐ Deliver Conference Bridge Identifier

☐ Allow Passthrough of Configured Line Device Caller Information

☐ Reject Anonymous Incoming Calls

☐ Reject Anonymous Outgoing Calls

☐ Send ILS Learned Destination Route String

SIP Profile Configuration used by SIP trunk to Cisco UBE (Continued...)

SIP OPTIONS Ping

☒ Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)*

Ping Interval for Out-of-service Trunks (seconds)*

Ping Retry Timer (milliseconds)*

Ping Retry Count*

SDP Information

☐ Send send-receive SDP in mid-call INVITE

☐ Allow Presentation Sharing using BFCP

☐ Allow iX Application Media

☐ Allow multiple codecs in answer SDP

SIP Trunk to Cisco UBE Configuration

Navigation: Device → Trunk



Set Device Name* = ATT_SIP_TRUNK. This is used for this example
Set Description = ATT SIP Trunk to PSTN. This is used for this example
Set Device Pool* = G729_pool. This is used for this example
Set Media Resource Group List = MRGL_MTP.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration
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System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ B

Trunk Configuration Related Links: Back To Find/List Go

Save X Delete Reset + Add New

SIP Trunk Status

Service Status: Full Service
Duration: Time In Full Service: 4 days 12 hours 44 minutes

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	ATT_SIP_TRUNK
Description	ATT SIP Trunk to PSTN
Device Pool*	G729 Pool
Common Device Configuration	ATT_SIP_TRUNK
Call Classification*	Use System Default
Media Resource Group List	MRGL_MTP
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None

SIP Trunk to Cisco UBE Configuration (Continued...)



Set Significant Digits* = 4. This is used in this example.

<input type="checkbox"/> Media Termination Point Required
<input checked="" type="checkbox"/> Retry Video Call as Audio
<input type="checkbox"/> Path Replacement Support
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU
<input type="checkbox"/> Unattended Port
<input type="checkbox"/> 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
<input checked="" type="checkbox"/> PSTN Access
<input type="checkbox"/> Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

MLPP and Confidential Access Level Information

MLPP Domain < None >

Confidential Access Mode < None >

Confidential Access Level < None >

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type* Default

SIP Privacy* Default

Inbound Calls

Significant Digits* 4

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

SIP Trunk to Cisco UBE Configuration (Continued...)

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.

[Clear Prefix Settings](#)

[Default Prefix Settings](#)

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

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.

[Clear Prefix Settings](#)

[Default Prefix Settings](#)

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input type="checkbox"/>

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* Originator

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Calling and Connected Party Info Format* Deliver DN only in connected party

☒ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

☐ Maintain Original Caller ID DN and Caller Name in Identity Headers



SIP Trunk to Cisco UBE Configuration (Continued...)

Set Destination Address = Set IP address of ISR-Cisco UBE.

Set SIP Trunk Security Profile* = ATT_Non Secure Sip Trunk Profile.

Set SIP Profile* = Standard SIP Profile w/Early Media Disabled. This is used in this example.

SIP Information		
Destination		
<input type="checkbox"/> Destination Address is an SRV		
Destination Address	Destination Address IPv6	Destination Port
1* 10.80.22.75		5060
MTP Preferred Originating Codec*	711ulaw	
BLF Presence Group*	Standard Presence group	
SIP Trunk Security Profile*	ATT 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 w/Early Media Disabled View Details	
DTMF Signaling Method*	No Preference	

Normalization Script	
Normalization Script < None >	
<input type="checkbox"/> Enable Trace	
Parameter Name	Parameter Value
1	
Recording Information	
<input checked="" type="radio"/> None	
<input type="radio"/> This trunk connects to a recording-enabled gateway	
<input type="radio"/> This trunk connects to other clusters with recording-enabled gateways	
Geolocation Configuration	
Geolocation	< None >
Geolocation Filter	< None >
<input type="checkbox"/> Send Geolocation Information	
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/>	



SIP Trunk to Fax Gateway Configuration.

Navigation: Device → Trunk

Set Device Name* = Trunk_SIP_FAX_Gateway. This is used for this example

Set Description = Trunk_SIP_FAX_Gateway. This is used for this example

Set Device Pool* = G729 pool. This is used for this example

Set Media Resource Group List = MRGL_MTP.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation menu with options like "Navigation", "Cisco Unified CM Administration", "administrator", "Search Documentation", and "A". Below this is a secondary navigation bar with tabs for "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Bulk A". The main content area is titled "Trunk Configuration" and includes a "Related Links" section with a "Back To Find/List" button and a "Go" button. Below the title bar are icons for "Save", "Delete", "Reset", and "Add New". The configuration form is divided into two sections: "SIP Trunk Status" and "Device Information". The "SIP Trunk Status" section shows "Service Status: Full Service" and "Duration: Time In Full Service: 0 day 2 hours 36 minutes". The "Device Information" section contains various fields: "Product" (SIP Trunk), "Device Protocol" (SIP), "Trunk Service Type" (None(Default)), "Device Name*" (Trunk_SIP_FAX_Gateway), "Description" (Trunk to SIP FAX Gateway), "Device Pool*" (G729 Pool), "Common Device Configuration" (< None >), "Call Classification*" (Use System Default), "Media Resource Group List" (MRGL_MTP), "Location*" (Hub_None), "AAR Group" (< None >), "Tunneled Protocol*" (None), "QSIG Variant*" (No Changes), "ASN.1 ROSE OID Encoding*" (No Changes), "Packet Capture Mode*" (None), and "Packet Capture Duration" (0). The fields "Device Name*", "Device Pool*", and "Media Resource Group List" are highlighted with red boxes.

SIP Trunk Status	
Service Status:	Full Service
Duration:	Time In Full Service: 0 day 2 hours 36 minutes

Device Information	
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Trunk_SIP_FAX_Gateway
Description	Trunk to SIP FAX Gateway
Device Pool*	G729 Pool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_MTP
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 Configuration (Continued...)

<input type="checkbox"/> Media Termination Point Required
<input checked="" type="checkbox"/> Retry Video Call as Audio
<input type="checkbox"/> Path Replacement Support
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU
<input type="checkbox"/> Unattended Port
<input type="checkbox"/> 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
<input checked="" type="checkbox"/> PSTN Access
<input checked="" type="checkbox"/> Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile	< None >
------------------------------	-----------------------------

MLPP and Confidential Access Level Information

MLPP Domain	< None >
Confidential Access Mode	< None >
Confidential Access Level	< None >

Call Routing Information

<input checked="" type="checkbox"/> Remote-Party-Id
<input checked="" type="checkbox"/> 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	<input type="text"/>
<input type="checkbox"/> Redirecting Diversion Header Delivery - Inbound	



SIP Trunk to Fax Gateway Configuration (Continued...)

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.

Clear Prefix Settings **Default Prefix Settings**

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input style="border: none;" type="text" value=" < None > "/>	<input checked="" type="checkbox"/>

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.

Clear Prefix Settings
Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input type="checkbox"/>

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* Originator

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Calling and Connected Party Info Format* Deliver DN only in connected party

☐ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Trunk to Fax Gateway Configuration (Continued...)



SIP Information

Destination
☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	172.16.31.50		5060

MTP Preferred Originating Codec*

711ulaw

BLF Presence Group*

Standard Presence group

SIP Trunk Security Profile*

ATT 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 for ATT

[View Details](#)

DTMF Signaling Method*

No Preference

Normalization Script
Normalization Script < None >
☐ Enable Trace**Recording Information**
☒ None
☐ This trunk connects to a recording-enabled gateway
☐ This trunk connects to other clusters with recording-enabled gateways**Geolocation Configuration**
Geolocation < None >
Geolocation Filter < None >
☐ Send Geolocation Information

Save

Delete

Reset

Add New



Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Set Route Pattern* = 9.@ This is used to route to AT&T via ISR Cisco UBE.

Set Description = To PSTN via ATT SIP Trunk. This text is used to identify this Route Pattern.

Set Gateway/Route List* = ATT_SIP_TRUNK. This is used for this example.

All other values are default

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation menu with options like "System", "Call Routing", "Media Resources", etc. The main content area is titled "Find and List Route Patterns". It features a status bar indicating "4 records found". Below this is a table of route patterns. The table has columns for "Pattern", "Description", "Partition", "Route Filter", "Associated Device", and "Copy". Four records are listed, with the last one, "9.@", highlighted by a red box. The "Associated Device" for the highlighted record is "ATT_SIP_TRUNK".

Pattern	Description	Partition	Route Filter	Associated Device	Copy
*XI	Network Based call forwarding			ATT_SIP_TRUNK	
2303	Voice Mail VIA SIP Trunk			UnityConnection	
4351	To the FAX Gateway			Fax_Gateway	
9.@	To PSTN via ATT SIP Trunk			ATT_SIP_TRUNK	



Route Pattern Configuration (Continued...)

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation **Cisco Unified CM Administration**
administrator | [Search Documentation](#) | [About](#)

[System](#) ▾ [Call Routing](#) ▾ [Media Resources](#) ▾ [Advanced Features](#) ▾ [Device](#) ▾ [Application](#) ▾ [User Management](#) ▾ [Bulk Adminis](#)

Route Pattern Configuration

Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Add New

Pattern Definition

Route Pattern*

9.@"

Route Partition

< None >

Description

To PSTN via ATT SIP Trunk

Numbering Plan*

NANP

Route Filter

< None >

MLPP Precedence*

Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

< None >

Route Class*

Default

Gateway/Route List*

ATT_SIP_TRUNK

[\(Edit\)](#)

Route Option

☒ Route this pattern

☐ Block this pattern

No Error

Call Classification*

OffNet

External Call Control Profile

< None >

☐ Allow Device Override

☒ Provide Outside Dial Tone

☐ Allow Overlap Sending

☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level*

0

☐ Require Client Matter Code

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



Route Pattern Configuration (Continued...)

Connected Party Transformations		
Connected Line ID Presentation *	Default	
Connected Name Presentation *	Default	
Called Party Transformations		
Discard Digits	PreDot	
Called Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Called Party Number Type *	Cisco CallManager	
Called Party Numbering Plan *	Cisco CallManager	
ISDN Network-Specific Facilities Information Element		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service
-- Not Selected --	< Not Exist >	
<div>Save Delete Copy Add New</div>		



Route Pattern Configuration (Continued...)

Set Route Pattern* = *X! This is used to route to AT&T via ISR Cisco UBE.

Set Description = Network-Based Call Forwarding. This text is used to identify this Route Pattern.

Set Gateway/Route List* = ATT_SIP_TRUNK. This is used for this example.

All other values are default

Note: This Route pattern is used to Activate/De-activate Network Based Call Forwarding Features.

Pattern Definition	
Route Pattern*	*X!
Route Partition	< None >
Description	Network Based call forwarding
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	ATT_SIP_TRUNK (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error
Call Classification*	OffNet
External Call Control Profile	< None >
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	0
<input type="checkbox"/> Require Client Matter Code	

Calling Party Transformations	
<input checked="" type="checkbox"/> 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



Route Pattern Configuration (Continued...)

Connected Party Transformations		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
Called Party Transformations		
Discard Digits	PreDot	
Called Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Called Party Number Type*	Cisco CallManager	
Called Party Numbering Plan*	Cisco CallManager	
ISDN Network-Specific Facilities Information Element		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service
-- Not Selected --	< Not Exist >	
<div>Save Delete Copy Add New</div>		



Route Pattern Configuration (Continued...)

Set Route Pattern* = 2753 this is used to route to Fax Client via Fax Gateway.

Set Description = To FAX. This text is used to identify this Route Pattern.

Set Gateway/Route List* FAX_Gateway. This is used for this example.

All other values are default

Route Pattern Configuration

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Related Links: Back To Find/List | Go

Save | Delete | Copy | Add New

Pattern Definition

Route Pattern* 2753

Route Partition < None >

Description To the Fax gateway

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence* Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain < None >

Route Class* Default

Gateway/Route List* Fax_Gateway (Edit)

Route Option

☒ Route this pattern

☐ Block this pattern No Error

Call Classification* OffNet

External Call Control Profile < None >

☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level* 0

☐ Require Client Matter Code

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



Route Pattern Configuration (Continued...)

Connected Party Transformations		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
Called Party Transformations		
Discard Digits	< None >	
Called Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Called Party Number Type*	Cisco CallManager	
Called Party Numbering Plan*	Cisco CallManager	
ISDN Network-Specific Facilities Information Element		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Servi
-- Not Selected --	< Not Exist >	
<div>Save Delete Copy Add New</div>		



Jabber Client Configuration

Navigation: Device → Phone

Select Phone Type* = Cisco Unified Client services framework

Set Device Name* = CSFUser1. This is used in this example.

Set Description = CSFUser1. This is used in this example.

Select Device Pool = G729. This is used in this example.

Select Phone Button Template* = Standard Client Services Framework.

Association		Phone Type	
<div>Modify Button Items</div> <div>1 Line [1] - 2753 (no partition)</div> <div>----- Unassigned Associated Items -----</div> <div>2 Line [2] - Add a new DN</div>		Product Type: Cisco Unified Client Services Framework Device Protocol: SIP	
		Real-time Device Status Registration: Unknown IPv4 Address: None	
		Device Information <input checked="" type="checkbox"/> Device is Active <input checked="" type="checkbox"/> Device is trusted <div><div>Device Name*</div><div>CSFUser1</div></div> <div><div>Description</div><div>CSFUser1</div></div> <div><div>Device Pool*</div><div>G729 Pool</div><div>View Details</div></div> <div><div>Common Device Configuration</div><div>< None ></div><div>View Details</div></div> <div><div>Phone Button Template*</div><div>Standard Client Services Framework</div></div> <div><div>Common Phone Profile*</div><div>Standard Common Phone Profile</div><div>View Details</div></div>	



Jabber Client Configuration (Contd...)

Media Resource Group List = MRGL_MTP

Set Owner check box

Set Owner user ID* = jabber1. This is used for this example

Calling Search Space	< None >
AAR Calling Search Space	< None >
Media Resource Group List	MRGL_MTP
User Hold MOH Audio Source	1-SampleAudioSource
Network Hold MOH Audio Source	1-SampleAudioSource
Location*	Hub_None
AAR Group	< None >
User Locale	< None >
Network Locale	< None >
Built In Bridge*	Default
Device Mobility Mode*	Default
	View Current Device Mobility Settings
Owner	<input checked="" type="radio"/> User <input type="radio"/> Anonymous (Public/Shared Space)
Owner User ID*	jabber1
Mobility User ID	< None >
Primary Phone	< None >
Use Trusted Relay Point*	Default

Always Use Prime Line*	Default
Always Use Prime Line for Voice Message*	Default
Geolocation	< None >
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)	
<input checked="" type="checkbox"/> Allow Control of Device from CTI	
<input checked="" type="checkbox"/> Logged Into Hunt Group	
<input type="checkbox"/> Remote Device	
<input type="checkbox"/> Require off-premise location	

Jabber Client Configuration (Contd...)

Number Presentation Transformation

Caller ID For Calls From This Phone

Calling Party Transformation CSS

< None >

☒ Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)

Remote Number

Calling Party Transformation CSS

< None >

☒ Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

Protocol Specific Information

Packet Capture Mode*

None

Packet Capture Duration

0

BLF Presence Group*

Standard Presence group

SIP Dial Rules

< None >

MTP Preferred Originating Codec*

711ulaw

Device Security Profile*

Cisco Unified Client Services Framework - Standar

Rerouting Calling Search Space

< None >

SUBSCRIBE Calling Search Space

< None >

SIP Profile*

Standard SIP Profile w/Early Media Disabled.

[View Details](#)

Digest User

< None >

☐ Media Termination Point Required

☐ Unattended Port

☐ Require DTMF Reception



Jabber Client Configuration (Contd...)

Certification Authority Proxy Function (CAPF) Information				
Certificate Operation*	No Pending Operation ▼			
Authentication Mode*	By Null String ▼			
Authentication String	<input type="text"/>			
<input type="button" value="Generate String"/>				
Key Size (Bits)*	2048 ▼			
Operation Completes By	2015	6	20	12 (YYYY:MM:DD:HH)
Certificate Operation Status:	None			
Note: Security Profile Contains Addition CAPF Settings.				


Extension Information	
<input type="checkbox"/>	Enable Extension Mobility
Log Out Profile	-- Use Current Device Settings -- ▼
Log in Time	< None >
Log out Time	< None >

MLPP and Confidential Access Level Information	
MLPP Domain	< None > ▼
Confidential Access Mode	< None > ▼
Confidential Access Level	< None > ▼

Do Not Disturb	
<input type="checkbox"/>	Do Not Disturb
DND Option*	Ringer Off ▼
DND Incoming Call Alert	< None > ▼

Jabber Client Configuration (Contd...)

Product Specific Configuration Layout



Parameter Value

Override Common Settings

Video Calling*

Enabled

▼

☐

Interactive Connectivity Establishment (ICE)

ICE

Enabled

▼

☐

Default Candidate Type

Host

▼

☐

Server Reflexive Address

Enabled

▼

☐

Primary TURN Server Host Name or IP Address

☐

Secondary TURN Server Host Name or IP Address

☐

TURN Server Transport Type

Auto

▼

☐

TURN Server Username

administrator

☒

TURN Server Password

.....

☒

Instant Messaging

File Types to Block in File Transfer

☐

URLs to Block in File Transfer

☐

Jabber Client Configuration (Contd...)

Desktop Client Settings			
Automatically Start in Phone Control*	Disabled		<input type="checkbox"/>
Automatically Control Tethered Desk Phone*	Disabled		<input type="checkbox"/>
Extend and Connect Capability*	Enabled		<input type="checkbox"/>
Display Contact Photos*	Enabled		<input type="checkbox"/>
Number Lookups on Directory*	Enabled		<input type="checkbox"/>
Jabber For Windows Software Update Server URL	user1@lab.tekvizion.com		<input checked="" type="checkbox"/>
Problem Report Server URL			<input type="checkbox"/>
Analytics Collection*	Disabled		<input type="checkbox"/>
Analytics Server URL			<input type="checkbox"/>
Cisco Support Field			<input type="checkbox"/>



Voicemail Port Configuration

Navigation: Advanced Feature → Voice Mail → Cisco Voice Mail Port

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

administrator | [Search Documentation](#) | [About](#) | [Logout](#)

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help

Find and List Voice Mail Ports

Add New Select All Clear All Delete Selected Reset Selected

Status
 2 records found

Voice Mail Port (1 - 2 of 2) Rows per Page 50

Find Voice Mail Port where Device Name ▾ begins with ▾ Find Clear Filter

Select item or enter search text ▾

<input type="checkbox"/>	Device Name ▲	Description	Device Pool	Device Security Mode	Calling Search Space	Extension	Partition	Status	IPv4 Address	Copy
<input type="checkbox"/>	CiscoUM1-VI1		G729 Pool	Non Secure Voice Mail Port		2295		Registered with clus32pubsub	10.80.22.4	
<input type="checkbox"/>	CiscoUM1-VI2		G729 Pool	Non Secure Voice Mail Port		2296		Registered with clus32pubsub	10.80.22.4	

Add New Select All Clear All Delete Selected Reset Selected Apply Config to Selected



Voicemail Port Configuration (Continued...)

Set Port Name = CiscoUM1-VI1. This is used for this example.

Set Description = VM Port. This is used for this example.

Set Device Pool = G729

Set Directory Number* = 2295. This is used in this example.

Device Information	
Registration:	Registered with Cisco Unified Communications Manager clus32pubsub
IPv4 Address:	10.80.22.4
<input checked="" type="checkbox"/> Device is trusted	
Port Name*	CiscoUM1-VI1
Description	Voice mail
Device Pool*	G729 Pool
Common Device Configuration	< None >
Calling Search Space	< None >
AAR Calling Search Space	< None >
Location*	Hub_None
Device Security Mode*	Non Secure Voice Mail Port
Use Trusted Relay Point*	Default
Geolocation	< None >

Directory Number Information	
Directory Number*	2295
Partition	< None >
Calling Search Space	< None >
AAR Group	< None >
Internal Caller ID Display	VoiceMail
Internal Caller ID Display (ASCII format)	VoiceMail
External Number Mask	

Save Delete Copy Reset Apply Config Add New



Message Waiting Numbers Configurations

Navigation: Advanced Features → Voice Mail → Message Waiting

Set Message Waiting Number* = 2298

Set Message Waiting Indicator* = On

Set Message Waiting Number* = 2399

Set Message Waiting Indicator* = Off

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation dropdown menu. Below the navigation bar is a breadcrumb trail: System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration. The main heading is "Find and List Message Waiting Numbers". Below this heading are buttons for "Add New", "Select All", "Clear All", and "Delete Selected". A status box indicates "2 records found". The table below shows the list of Message Waiting Numbers. The table has columns for "Directory Number", "Description", "Partition", "Calling Search Space", and "Copy". Two records are listed: 2298 (MWI ON) and 2299 (MWI OFF). The table is filtered by "Directory Number" and "begins with". The "Find" button is visible. The "Rows per Page" dropdown is set to 50.

	Directory Number ^	Description	Partition	Calling Search Space	Copy
<input type="checkbox"/>	2298	MWI ON			
<input type="checkbox"/>	2299	MWI OFF			



VoiceMail Pilot Configuration

Navigation: Advanced Features → Voice Mail → Voice Mail Pilot

Set Voice mail Pilot Number = 2300. This is used for this example

Set Description = VoiceMail Pilot number

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the subtitle "For Cisco Unified Communications Solutions". The user is logged in as "administrator". The main menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Bulk Administration. The "Find and List Voice Mail Pilots" section is active, showing 3 records found. A table lists the Voice Mail Pilots:

		Pilot Number ^	Description	Calling Search Space
<input type="checkbox"/>			No Voice Mail	
<input type="checkbox"/>			Default	
<input checked="" type="checkbox"/>		2300	VoiceMail Pilot-Default	

Buttons at the bottom of the table include Add New, Select All, Clear All, and Delete Selected.

The screenshot shows the "Voice Mail Pilot Information" configuration page. The "Voice Mail Pilot Number" field is set to 2300 and is highlighted with a red box. The "Calling Search Space" is set to "< None >". The "Description" is set to "VoiceMail Pilot-Default". The checkbox "Make this the default Voice Mail Pilot for the system" is checked. Buttons at the bottom include Save, Delete, and Add New.



FAX Gateway Configuration

```
voice service voip
no ip address trusted authenticate
allow-connections sip to sip
redirect ip2ip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no fax-relay sg3-to-g3
sip
midcall-signaling passthru
g729 annexb-all

voice class codec 1
codec preference 1 g729br8 bytes 30
codec preference 2 g711ulaw

voice class sip-profiles 1
response ANY sip-header Allow-Header modify "UPDATE," ""
request ANY sip-header Allow-Header modify "UPDATE," ""
response ANY sip-header Allow-Header modify "UPDATE," ""
response ANY sip-header Allow-Header modify "UPDATE," ""
```



```
voice-port 0/1/1
ring frequency 50
no echo-cancel enable
no vad
cptone IN
station-id number 2753
caller-id enable
dial-peer voice 101 pots
huntstop
service session
destination-pattern 2753
no digit-strip
port 0/0/1
forward-digits all

dial-peer voice 200 voip
description CUCM to Gateway
service session
session protocol sipv2
session transport udp
incoming called-number 2753
voice-class codec 1
voice-class sip profiles 1
```



```
dtmf-relay rtp-nte
no fax-relay sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
```

```
dial-peer voice 201 voip
description Gateway to CUCM
service session
destination-pattern [2-9]T
session protocol sipv2
session target ipv4:10.80.14.2
session transport udp
voice-class codec 1
voice-class sip profiles 1
dtmf-relay rtp-nte
no fax-relay sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
```




Cisco UCM SIP Integration with Cisco Unity Connection (CUC)

CUC Version

Cisco Unity Connection Administration

Version 11.0.1.10000-10

A photograph of a server room aisle, showing rows of blue server racks on both sides, receding into the distance under bright overhead lights.

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CUC Telephony Integration with Cisco UCM

Navigation: Telephony Integrations → Phone system

Set Phone System Name* = SIP. This is used for this example

The screenshot shows the Cisco Unity Connection Administration web interface. The left sidebar contains a navigation tree with 'Cisco Unity Connection' expanded, and 'Telephony Integrations' > 'Phone System' selected. The main content area is titled 'Search Phone Systems' and shows a status message 'Found 2 Phone System(s)'. Below this is a table of phone systems. The table has columns for 'Find Phone Systems', 'where Display Name', and 'Port Count'. Two rows are listed: 'Default' and 'SIP'. The 'SIP' row is highlighted with a red border. Below the table are buttons for 'Delete Selected' and 'Add New'.

Find Phone Systems	where Display Name	Port Count
<input type="checkbox"/>	Default	2
<input type="checkbox"/>	SIP	2



CUC Port Group

Navigation: Telephony Integration → Port Group

The screenshot displays the Cisco Unity Connection Administration web interface. The left sidebar shows the navigation tree with 'Telephony Integrations' expanded and 'Port Group' selected. The main content area is titled 'Search Port Groups' and shows a status message 'Found 2 Port Group(s)'. Below this is a table of port groups. The table has columns for 'Port Group Name', 'Phone System Display Name', 'Port Count', 'Integration Method', and 'Needs Reset'. Two port groups are listed: 'CiscoUM1-VI' and 'SIP-1'. The 'SIP-1' row is highlighted with a red border. Below the table are buttons for 'Delete Selected' and 'Add New'.

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administrator | Search Documentation | About

Search Port Groups Related Links: Check Telephony Configuration Go

Port Group Refresh Help

Status
Found 2 Port Group(s)

Port Groups (1 - 2 of 2) Rows per Page: 25

Find Port Groups where Port Group Name begins with

<input type="checkbox"/>	Port Group Name	Phone System Display Name	Port Count	Integration Method	Needs Reset
<input type="checkbox"/>	CiscoUM1-VI	Default	2	SCCP (Skinny)	No
<input type="checkbox"/>	SIP-1	SIP	2	SIP	No

Delete Selected Add New



CUC Port Group(continued...)

Set Display Name* = SIP-1. This is used in this example.

Check Register with SIP Server.

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration | administrator | Search Documentation | About

Cisco Unity Connection

- Schedules
 - Holiday Schedules
 - Global Nicknames
 - Subject Line Formats
 - Attachment Descriptions
 - Enterprise Parameters
 - Service Parameters
 - Plugins
 - Fax Server
- LDAP
 - SAML Single Sign on
 - Cross-Origin Resource Sharing (CORS)
 - SMTP Configuration
 - Advanced
- Telephony Integrations
 - Phone System
 - Port Group
 - Port
 - Speech Connect Port
 - Trunk
 - Security
- Tools
 - Task Management
 - Bulk Administration Tool
 - Custom Keypad Mapping
 - Migration Utilities
 - Grammar Statistics
 - SMTP Address Search

Port Group Basics (SIP-1)

Search Port Groups | Port Group Basics (SIP-1) | Related Links | Add Ports | Go

Port Group | Edit | Refresh | Help

Save | Delete | Previous | Next

Port Group

Display Name* SIP-1

Integration Method SIP

Reset Status Reset Not Required | Reset

Session Initiation Protocol (SIP) Settings

☒ Register with SIP Server

☐ Authenticate with SIP Server

Authentication Username

Authentication Password

Contact Line Name

SIP Security Profile 5060


SIP Transport Protocol TCP

Advertised Codec Settings

Change Advertising



CUC Port Settings

**Cisco Unity Connection Administration**
For Cisco Unified Communications Solutions


Navigation **Cisco Unity Connection Administration** Go
administrator | [Search Documentation](#) | [About](#) | [Sign Out](#)

Cisco Unity Connection

- Schedules
 - Holiday Schedules
 - Global Nicknames
 - Subject Line Formats
 - Attachment Descriptions
 - Enterprise Parameters
 - Service Parameters
 - Plugins
 - Fax Server
- LDAP
 - SAML Single Sign on
 - Cross-Origin Resource Sharing (CORS)
- SMTP Configuration
- Advanced
 - Telephony Integrations
 - Phone System
 - Port Group
 - Port**
 - Speech Connect Port
 - Trunk
 - Security
- Tools
 - Task Management

Search PortsSearch Ports

[Port](#) [Refresh](#) [Help](#)

Status
 Found 4 Port(s)

Port (1 - 4 of 4) Rows per Page 25

Find Port where

<input type="checkbox"/>	Display Name	Phone System Display Name	Extension	Server	Enabled	Answer Calls	Message Notification	Dialout MWI	TRAP Connection	Security Mode
<input type="checkbox"/>	CiscoUM1-VI-001 Default			clus24-unity	X	X	X	X	X	Non-secure
<input type="checkbox"/>	CiscoUM1-VI-002 Default			clus24-unity	X	X	X	X	X	Non-secure
<input type="checkbox"/>	SIP-1-001 SIP			clus24-unity	X	X	X	X	X	NA
<input type="checkbox"/>	SIP-1-002 SIP			clus24-unity	X	X	X	X	X	NA



CUC Sample User Basic Settings

Navigation: Cisco Unity connection → Users → Users

Set Alias = 4051. This is one of the extension used for this testing.

Set Extension = 4051. This is used for this example.

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration
administrator | Search Documentation | About | Sign

Cisco Unity Connection

- Users
 - Users
 - Import Users
 - Synch Users
- Class of Service
 - Class of Service
 - Class of Service Membership
- Templates
 - User Templates
 - Call Handler Templates
 - Contact Templates
 - Notification Templates
- Contacts
 - Contacts
- Distribution Lists
 - System Distribution Lists
- Call Management
 - System Call Handlers
 - Directory Handlers
 - Interview Handlers
 - Custom Recordings
 - Call Routing
- Message Storage
 - Mailbox Stores
 - Mailbox Stores Membership
 - Mailbox Quotas
 - Message Aging
- Networking

Edit User Basics (4351)

Search Users | Edit User Basics (4351)
Related Links: Bulk Edit By CSV | Go

User Edit Refresh Help

Save Delete Previous Next

Name

Alias* 4351

First Name Cisco

Last Name 4351

Display Name Cisco 4351

SMTP Address 4351 @clus24-unity.lab.tekvizion.com

Initials

Title

Employee ID

LDAP Integration Status

☐ Integrate with LDAP Directory

☒ Do Not Integrate with LDAP Directory

Phone

Extension* 4351

Cross-Server Transfer

Extension or URI



CUC Sample User Basic Settings (Continued...)

Set Partition = clus24-unity partition. This is used for this example.

Select Search Scope = clus24-unity Search Scope.

Select Phone System = SIP.

Cisco Unity Connection

- Users
 - Users
 - Import Users
 - Synch Users
- Class of Service
 - Class of Service
 - Class of Service Membership
- Templates
 - User Templates
 - Call Handler Templates
 - Contact Templates
 - Notification Templates
- Contacts
 - Contacts
- Distribution Lists
 - System Distribution Lists
- Call Management
 - System Call Handlers
 - Directory Handlers
 - Interview Handlers
 - Custom Recordings
 - Call Routing
- Message Storage
 - Mailbox Stores
 - Mailbox Stores Membership
 - Mailbox Quotas

Outgoing Fax Server: --- Not Selected ---

Partition: clus24-unity Partition

Search Scope: clus24-unity Search Space

Phone System: SIP

Class of Service: Voice Mail User COS

Active Schedule: Weekdays [View](#)

☐ Set for Self-enrollment at Next Sign-In

☒ List in Directory

☒ Send Non-Delivery Receipts on Failed Message Delivery

☒ Skip PIN When Calling From a Known Extension
Caution! Security risk. See Help for This Page for details.

☐ Use Short Calendar Caching Poll Interval

Recorded Name: [Play/Record](#)

Location

Address:

Building:

City:

State:

Postal Code:

Country: United States



<ul style="list-style-type: none">⊕ Contacts⊕ Distribution Lists⊕ Call Management⊕ Message Storage⊕ Networking⊕ Unified Messaging⊕ Video⊕ Dial Plan⊕ System Settings⊕ Telephony Integrations⊕ Tools	<input checked="" type="checkbox"/> Use System Default Time Zone
	Time Zone <input type="text" value="(GMT-06:00) America/Chicago"/>
	Language <input checked="" type="radio"/> Use System Default Language
	<input type="radio"/> <input type="text" value="English(United States)"/>
	Department <input type="text"/>
	Manager <input type="text"/>
	Billing ID <input type="text"/>
	Corporate Email Address <input type="text"/>
	<input type="checkbox"/> Generate SMTP Proxy Address From Corporate Email Address
	Directory URI <input type="text"/>
	Corporate Phone Number <input type="text"/>
	<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Previous"/> <input type="button" value="Next"/>
	Fields marked with an asterisk (*) are required.

Auto Attendant

Navigation: Call Management → System Call Handlers

Set Display Name = Demo auto attend. This is used for this example.

Set Phone System = SIP

Set Extension=2999. This number is used as Auto attendant on this set up.

Set Partition = Clus24-unity Partition. This is used for this example.



Cisco

Cisco Unity Connection Administration

For Cisco Unified Communications Solutions

Navigation Cisco Unity Connection Administration administrator Search Documentation About Sign

Cisco Unity Connection

Users

Users

Import Users

Sync Users

Class of Service

Class of Service

Class of Service Membership

Templates

User Templates

Call Handler Templates

Contact Templates

Notification Templates

Contacts

Contacts

Distribution Lists

System Distribution Lists

Call Management

System Call Handlers

Directory Handlers

Interview Handlers

Custom Recordings

Call Routing

Message Storage

Mailbox Stores

Search Call Handlers

Search Call Handlers

Call Handler Refresh Help

Status

Found 5 System Call Handler(s)

Search Limits

Limit search to All

System Call Handlers (1 - 5 of 5) Rows per Page 25

Find System Call where Display Name begins with Find

	Display Name ^	Extension
<input type="checkbox"/>	Auto attendant	4999
<input type="checkbox"/>	Demo Auto Attendant	2999
<input type="checkbox"/>	Goodbye	
<input type="checkbox"/>	Opening Greeting	
<input type="checkbox"/>	Operator	

Delete Selected Add New Bulk Edit Show Dependencies

Auto Attendant (Continued...)

Cisco Unity Connection	
<ul style="list-style-type: none"> Users <ul style="list-style-type: none"> Users Import Users Synch Users Class of Service <ul style="list-style-type: none"> Class of Service Class of Service Membership Templates <ul style="list-style-type: none"> User Templates Call Handler Templates Contact Templates Notification Templates Contacts <ul style="list-style-type: none"> Contacts Distribution Lists <ul style="list-style-type: none"> System Distribution Lists Call Management <ul style="list-style-type: none"> System Call Handlers Directory Handlers Interview Handlers Custom Recordings Call Routing Message Storage 	<h3>Call Handler</h3> <div>Display Name* Demo Auto Attendant</div> <div>Creation Time 2015-04-24 04:47:51.249</div> <div>Phone System SIP</div> <div>Active Schedule All Hours View</div> <div><input checked="" type="checkbox"/> Use System Default Time Zone</div> <div>Time Zone (GMT-06:00) America/Chicago</div> <div>Language <div><input checked="" type="radio"/> Use System Default Language</div> <div><input type="radio"/> Inherit Language from Caller</div> <div><input type="radio"/> English(United States)</div> </div> <div>Extension 2999</div> <div>Partition clus24-unity Partition</div> <div>Recorded Name Play/Record</div> <h4>Search Scope</h4> <div><input checked="" type="radio"/> Search Space clus24-unity Search Space</div> <div><input type="radio"/> Inherit Search Space from Call</div>

Cisco UCM Integration with Cisco Unified CM IM and Presence (CUP/IMP)




CUP/IMP Version

Cisco Unified CM IM and Presence Administration

System version: 11.0.1.10000-6

VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz, disk 1: 80Gbytes, 2048Mbytes RAM, Partitions aligned



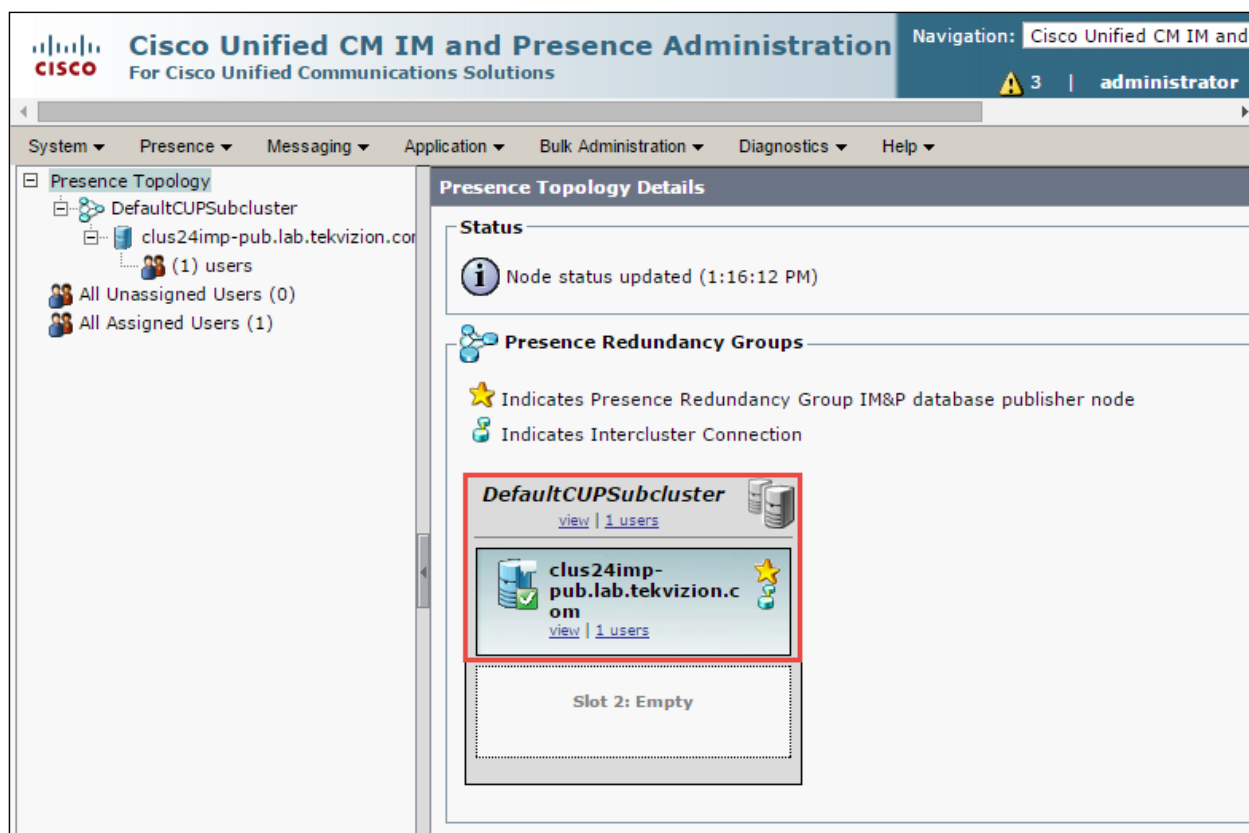
User administrator last logged in to this cluster on Wednesday, February 10, 2016 6:31:52 AM CST, to node 10.80.14.3, from 172.16.29.40 using HTTPS

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Presence Topology

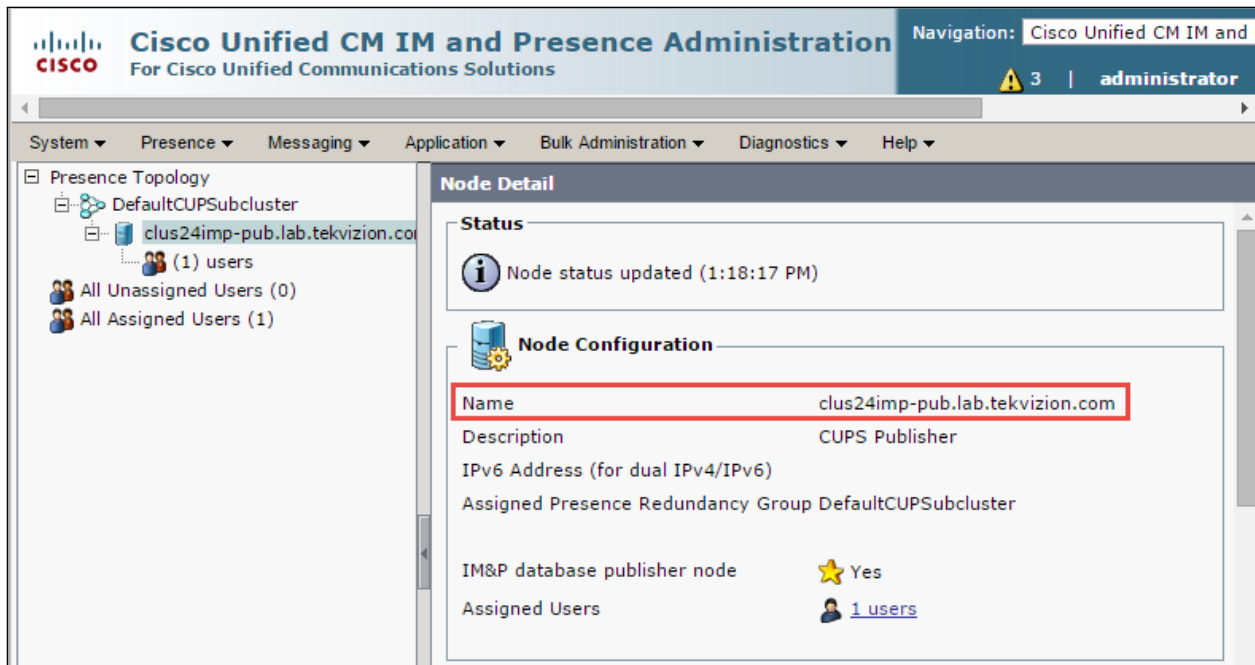
Navigation: System → Presence Topology



The screenshot displays the Cisco Unified CM IM and Presence Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM IM and Presence Administration For Cisco Unified Communications Solutions", and a navigation menu with "Cisco Unified CM IM and" and "administrator". Below the navigation bar, a menu bar contains "System", "Presence", "Messaging", "Application", "Bulk Administration", "Diagnostics", and "Help". The left sidebar shows a tree view under "Presence Topology" with "DefaultCUPSubcluster" expanded, showing "(1) users", "All Unassigned Users (0)", and "All Assigned Users (1)". The main content area is titled "Presence Topology Details" and contains a "Status" section with a message "Node status updated (1:16:12 PM)". Below this is a "Presence Redundancy Groups" section with a legend: a star icon for "Indicates Presence Redundancy Group IM&P database publisher node" and a connection icon for "Indicates Intercluster Connection". A red box highlights a node card for "DefaultCUPSubcluster" with a subcard for "clus24imp-pub.lab.tekvizion.com" (view | 1 users). Below the node card, it says "Slot 2: Empty".

Node Configuration

Navigation: System → Presence Topology → Fully Qualified Domain Name




The screenshot displays the Cisco Unified CM IM and Presence Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM IM and Presence Administration", and the subtitle "For Cisco Unified Communications Solutions". The right side of the navigation bar shows "Navigation: Cisco Unified CM IM and", a warning icon with the number 3, and the user "administrator". Below the navigation bar is a menu with options: System, Presence, Messaging, Application, Bulk Administration, Diagnostics, and Help. The main content area is divided into two sections. On the left, under "Presence Topology", there is a tree view showing "DefaultCUPSubcluster" and "clus24imp-pub.lab.tekvizion.com" (1 users). Below this, there are links for "All Unassigned Users (0)" and "All Assigned Users (1)". On the right, the "Node Detail" section for "clus24imp-pub.lab.tekvizion.com" is shown. It includes a "Status" section with an information icon and the text "Node status updated (1:18:17 PM)". Below this is the "Node Configuration" section, which contains a table with the following details:

Name	clus24imp-pub.lab.tekvizion.com
Description	CUPS Publisher
IPv6 Address (for dual IPv4/IPv6)	
Assigned Presence Redundancy Group	DefaultCUPSubcluster
IM&P database publisher node	★ Yes
Assigned Users	1 users

Users

Navigation: System → Cluster Topology → clus24imp.lab.tekvizion.com → Users


Cisco Unified CM IM and Presence Administration
 For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM IM and Presence Administration
 3 | administrator | Search | Logout

System ▾ | Presence ▾ | Messaging ▾ | Application ▾ | Bulk Administration ▾ | Diagnostics ▾ | Help ▾

Node User Assignment (clus24imp-pub.lab.tekvizion.com)

Status
 ⓘ 1 records found

User Assignment (1 - 1 of 1) Rows per Page 50 ▾

Find User Assignment where User ID ▾ begins with ▾ Find Clear Filter + -

User ID ▲	First Name	Last Name	IM Address	Directory URI	Failed Over	Node	Presence Redundancy Group
jabber1	cisco		jabber1@lab.tekvizion.com	jabber1@lab.tekvizion.com		clus24imp-pub.lab.tekvizion.com	DefaultCUPSubcluster

Presence gateway configuration

Navigation: Presence → Gateways



Set Presence Gateway Type *= CUCM
Set Description *= Cluster 24. This is used for this example.
Presence Gateway *= clus24pubsub.lab.tekvizion.com

Acronyms

AVPN	AT&T Virtual Private Network
CODEC	Coder-Decoder (in this document a device used to digitize and undigitize voice signals)



Cisco UBE	Cisco Unified Border Element
Cisco UCM	Cisco Unified Communications Manager
IP	Internet Protocol
ISR	Integrated Services Router
MGCP	Media Gateway Control Protocol
MIS	Managed Internet Services
PNT	Private Network Transport
PSTN	Public switched telephone network
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol
SP	Service Provider
TDM	Time-division multiplexing

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Note: Testing was conducted in tekVizion labs



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