

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 SIP Interface  
JAN 2016



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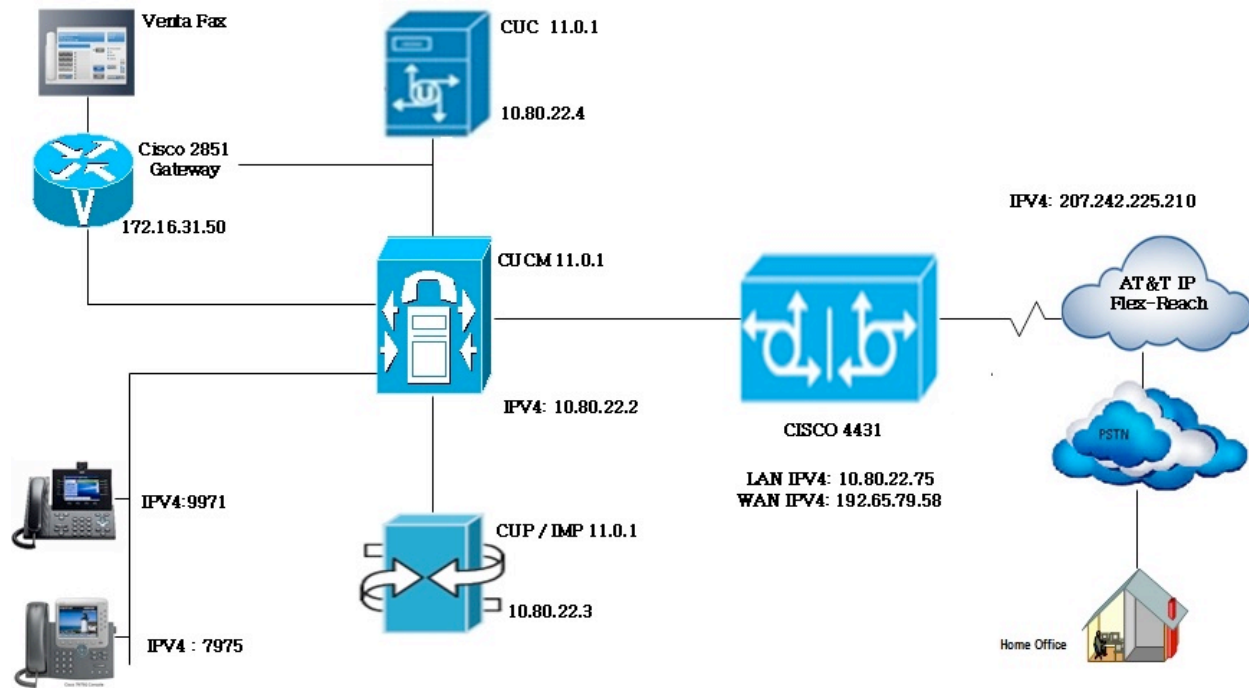


## Introduction

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)S1a 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.

## 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)
- 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.

### Ringback Tone on Early Unattended Transfer

- Caller does not hear ringback tone when a call is transferred to PSTN user.

### 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#**show 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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Compiled Wed 04-Nov-15 12:50 by mcpre

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ROM: IOS-XE ROMMON

CISCO\_4K\_ROUTER2 uptime is 2 weeks, 5 days, 8 hours, 17 minutes

Uptime for this control processor is 2 weeks, 5 days, 8 hours, 18 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](mailto: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	None	None	None
uck9	uck9	Evaluation	uck9
securityk9	None	None	None
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#**show running-config**

Building configuration...

Current configuration : 11328 bytes

!

! Last configuration change at 13:15:54 UTC Tue Dec 29 2015 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
!
!
!

no ip domain lookup
ip domain name tekvision.com
```



!

!

!

!

!

!

!

!

!

!

subscriber templating

multilink bundle-name authenticated

!

!

!

!

!

!

!

cts logging verbose

!

!

voice service voip

rtp-port range 16384 32766

address-hiding<sup>1</sup>

---

<sup>1</sup> Hide signaling and media peer addresses from endpoints other than gateway.





```
mode border-element 2
media disable-detailed-stats
allow-connections sip to sip 3
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
header-passing
error-passthru 4
asserted-id pai 5
no update-callerid
early-offer forced 6
midcall-signaling passthru 7
privacy-policy passthru 8
g729 annexb-all
!
voice class codec 1 9
```

---

<sup>2</sup> 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.

<sup>3</sup> This command enables Cisco UBE basic IP-to-IP voice communication feature.

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

<sup>5</sup> 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).

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

<sup>7</sup> 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.

<sup>8</sup> 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.



```
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:732320\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
!
!
```

---

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

<sup>10</sup> 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.

<sup>11</sup> 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.

<sup>12</sup> This SIP profile is required in order to advertise the ptime=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.



```
!  
!  
!  
voice translation-rule 113  
rule 1 /^.*\((40..\)\) / 732320\1/  
!  
!  
!  
voice translation-profile NPA  
translate calling 1  
!  
!  
!  
!  
license udi pid ISR4431/K9 sn FOC18232988  
license boot level appxk9  
license boot level uck9  
!  
spanning-tree extend system-id  
!  
username cisco privilege 15 secret 5 $1$AGR7$e7pQx6UI0be3bzRbc0lr81  
!  
redundancy  
mode none  
!
```

---

<sup>13</sup> This command used to convert 4 digit to 10 digit in contact header otherwise ATT will send 6xx error response while executing network related feature.



```
!  
vlan internal allocation policy ascending  
!  
!  
!  
!  
!  
!  
  
interface GigabitEthernet0/0/0  
ip address 10.64.4.20 255.255.0.0  
media-type rj45  
negotiation auto  
!  
  
interface GigabitEthernet0/0/1  
no ip address  
shutdown  
media-type rj45  
negotiation auto  
!  
  
interface GigabitEthernet0/0/214  
ip address 10.80.22.75 255.255.255.015  
media-type rj45  
negotiation auto  
!
```

---

<sup>14</sup> LAN interface to Cisco UCM

<sup>15</sup> Cisco UBE LAN interface IPv4 Address



```
interface GigabitEthernet0/0/316
description Wan Interface
ip address 192.65.79.58 255.255.255.224
media-type rj45
negotiation auto
!
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 0.0.0.0 192.65.79.33
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
ip route vrf Mgmt-intf 0.0.0.0 0.0.0.0 10.64.1.1
!
!
```

---

<sup>16</sup> WAN interface to AT&T



!

!

!

!

!

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"

no modem passthrough

session protocol sipv2



```
incoming called-number [1-9]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
fax-relay ecm disable
fax-relay sg3-to-g3
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 800 voip
description " Incoming AT&T to IP-PBX . AT&T facing side "
huntstop
no modem passthrough
session protocol sipv2
incoming called-number [37][13][24]32040..
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-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 700 voip
description " Incoming AT&T to IP-PBX - IP-PBX facing side "
huntstop
destination-pattern [37][13][24].....
no modem passthrough
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
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 100 voip<sup>17</sup>

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

destination-pattern 73236.....

no modem passthrough

session protocol sipv2<sup>18</sup>

session target ipv4:207.242.225.210

voice-class codec 1<sup>19</sup>

voice-class sip asymmetric payload full<sup>20</sup>

voice-class sip asserted-id pai

voice-class sip privacy-policy passthru<sup>21</sup>

voice-class sip profiles 1<sup>22</sup>

voice-class sip bind control source-interface GigabitEthernet0/0/3<sup>23</sup>

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

dtmf-relay rtp-nte<sup>24</sup>

---

<sup>17</sup> Dial peer for AT&T facing network

<sup>18</sup> Session protocol SIPv2 is used for this testing

<sup>19</sup> Assigns voice class codec 1 settings to dial-peer (codec support and filtering).

<sup>20</sup> Configures the dynamic SIP asymmetric payload support.

<sup>21</sup> 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.

<sup>22</sup> This command enables the dial peer to use SIP profile 1

<sup>23</sup> 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.



```
fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none25

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 ipv4:207.242.225.210

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
```

---

<sup>24</sup> This command used to pass RTP NTE (RFC2833) DTMF with respect to the dial peers used for the call.

<sup>25</sup> This command enables T38 fax protocol for calls terminating on this dial-peer



```
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 ipv4:207.242.225.210  
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 122 voip
description "OPERATOR TESTING"
destination-pattern 0
no modem passthrough
session protocol sipv2
session target ipv4:207.242.225.210
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 141 voip
description "Network Feature"
translation-profile outgoing NPA
destination-pattern *..
no modem passthrough
session protocol sipv2
session target ipv4:207.242.225.210
```



```
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 7323204607
no modem passthrough
session protocol sipv2
session target ipv4:207.242.225.210
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 214 voip
description "Outgoing To AT&T"-AT&T facing side
destination-pattern [2-9]T
no modem passthrough
session protocol sipv2
session target ipv4:207.242.225.210
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 ecm disable
fax-relay sg3-to-g3
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
```



```
no vad
!
!
gateway
timer receive-rtp 1200
!
!
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.

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Note: Testing was conducted in tekVizion labs



## Cisco UCM Version

**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 ▾

### 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

No successful backup has been taken after the Upgrade

User administrator last logged in to this cluster on Tuesday, January 5, 2016 3:59:32 AM CST, to node 10.80.22.2, from null using HTTPS

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

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

For information about Cisco Unified Communications Manager please visit our [Unified Communications System Documentation](#) web site.

For Cisco Technical Support please visit our [Technical Support](#) web site.

## 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)

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**Audio Codec Preference List Configuration** Related Links: [Back To Find/List](#) Go

Save Delete Copy Add New

**Audio Codec Preference List Information**

Name\* G729 Preferred Codec List

Description\* G729 Preferred Codec List

Codecs in List\*

- G.729b 8k
- G.729a 8k
- G.729ab 8k
- G.729 8k
- G.728 16k
- G.711 A-Law 64k
- G.711 U-Law 64k
- AMR (5k-13k)
- MP4A-LATM 128k
- AAC-LD (MP4A Generic)
- MP4A-LATM 64k
- MP4A-LATM 56k
- G.711 U-Law 56k
- G.711 A-Law 56k
- AMR-WB (7k-24k)
- 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
- GSM Enhanced Full Rate 13k
- GSM Full Rate 13k
- GSM Half Rate 6k
- G.723.1 7k
- OPUS (6k-510k)

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\* G729 Region

**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
Default G711 region G729 Region		Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> <input type="text"/> kbps <input type="radio"/> Use System Default <input type="radio"/> None <input type="text"/> kbps

Save Delete Reset Apply Config Add New

## Device Pool Configuration

**Navigation Path:** System → Device Pool

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“G729\_pool” 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.

The screenshot shows the Cisco Unified CM Administration interface for Device Pool 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", "administrator", "Search Documentation", "About", and "Logout". The main menu includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Bulk Administration". The "Device Pool Configuration" section is active, showing "Related Links: Back To Find/List" and "Go". The "Device Pool Information" section displays "Device Pool: G729 Pool (22 members\*\*)". The "Device Pool Settings" section is highlighted with a red box and contains the following fields:

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 >

The screenshot shows the "Roaming Sensitive Settings" section of the Cisco Unified CM Administration interface. The "Date/Time Group\*" and "Region\*" fields are highlighted with a red box. The "Date/Time Group\*" field is set to "CMLocal" and the "Region\*" field is set to "G729 Region". The "Media Resource Group List" field is set to "< None >". The "Location" field is set to "< None >". The "Network Locale" field is set to "< None >". The "SRST Reference\*" field is set to "Disable". The "Connection Monitor Duration\*\*\*" field is empty. The "Single Button Barge\*" field is set to "Default". The "Join Across Lines\*" field is set to "Default". The "Physical Location" field is set to "< None >". The "Device Mobility Group" field is set to "< None >". The "Wireless LAN Profile Group" field is set to "< None >" with a "View Details" link next to it. The "Local Route Group Settings" section is also visible, showing the "Standard Local Route Group" field set to "< 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 >

## Device Pool Configuration (continued...)

**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 >

**Phone Settings**

**Caller ID For Calls From This Phone**

Calling Party Transformation CSS
< None >

**Connected Party Settings**

Connected Party Transformation CSS
< None >

**Redirecting Party Settings**

Redirecting Party Transformation CSS
< None >

Save
Delete
Copy
Reset
Apply Config
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\_pool.

**Cisco Unified CM Administration**  
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**administrator** | Search Docu

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**Conference Bridge Configuration** Back To Find/List Go

Save Reset Apply Config

**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 ▾

Save Reset Apply Config



## 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 Pool

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**Media Termination Point Configuration** Related Links: Back To Find/List ▾ Go

Save Reset Apply Config

**Status**  
Status: Ready

**Media Termination Point Information**

Registration: Registered with Cisco Unified Communications Manager clus32pubsub  
IPv4 Address: 10.80.22.2  
Media Termination Point Type\*: Cisco Media Termination Point Software  
Host Server\*: clus32pubsub  
Media Termination Point Name\*: MTP\_2  
Description: MTP\_clus32pubsub  
Device Pool\*: G729 Pool ▾  
☐ Trusted Relay Point

Save Reset Apply Config




## 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\_pool.


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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).

The screenshot shows 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, Advanced Features, Device, Application, and User Management. The main content area is titled "Service Parameter Configuration" and includes a "Related Links" section with a dropdown for "Parameters for All Servers" and a "Go" button. 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 IP Voice Media Streaming App (Active)". A note below states: "All parameters apply only to the current server except parameters that are in the cluster-wide group(s)." The "Clusterwide Parameters (Parameters that apply to all servers)" section is also highlighted with a red box. It contains several parameters: "Supported MOH Codecs\*" with a list of "711 mulaw", "711 alaw", and "729 Annex A"; "MOH Fixed Audio Quality Level\*" set to "Medium Quality"; "IP DSCP to Cisco Unified Communications Manager\*" set to "CS3(precedence 3) DSCP (011000)"; "Multicast MOH IP DSCP\*" set to "EF DSCP (101110)"; "MTP DTMF Duration\*" set to "100"; and "MTP DTMF Power (volume)\*" set to "9". A note at the bottom of this section says: "There are hidden parameters in this group. Click on Advanced button to see hidden parameters." At the bottom of the page are buttons for "Save", "Set to Default", and "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


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

**Clusterwide Parameters (Service)**

<a href="#">Default Network Hold MOH Audio Source ID</a> *	1	1
<a href="#">Default User Hold MOH Audio Source ID</a> *	1	1
<a href="#">Duplex Streaming Enabled</a> *	True	False
<a href="#">Media Exchange Interface Capability Timer</a> *	8	8
<a href="#">Send Multicast MOH in H.245 OLC Message</a> *	True	True
<a href="#">Media Exchange Timer</a> *	12	12
<a href="#">Media Exchange Stop Streaming Timer</a> *	8	8
<a href="#">Open Video Channel Response Timer for SIP Interop</a> *	500	500
<a href="#">Port Received Timer After Call Connection</a> *	500	500
<a href="#">Media Resource Allocation Timer</a> *	12	12
<a href="#">MTP and Transcoder Resource Throttling Percentage</a> *	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.

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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\*   
Description

**Devices for this Group**  
Available Media Resources\*\*  
CFB\_2  
IVR\_2  
v v  
Selected Media Resources\*  
ANN\_2 (ANN)  
CFB\_c3825 (CFB)  
MOH\_2 (MOH)  
MTP\_2  
☐ 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.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and a user menu for "administrator". Below this is a secondary navigation bar with tabs for System, Call Routing, Media Resources, Advanced Features, Device, Application, and User Management. The main content area is titled "Media Resource Group List Configuration". It features a "Related Links" section with a "Back To Find/List" button and a "Go" button. Below this are icons for Save, Delete, Copy, and Add New. The configuration is divided into three sections: "Media Resource Group List Status" showing "MRGL\_MTP (used by 17 devices)", "Media Resource Group List Information" with a "Name\*" field containing "MRGL\_MTP", and "Media Resource Groups for this List". The latter section contains two list boxes: "Available Media Resource Groups" and "Selected Media Resource Groups", with "MRG\_MTP" selected in the latter. At the bottom are buttons for Save, Delete, Copy, and Add New.

## UC Service Configuration

**Navigation:** User Management → User Settings → UC Service








**Cisco Unified CM Administration**  
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Navigation **Cisco Unified CM Administration** **Go**  
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**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  ▾ begins with ▾  Find Clear Filter + -

<input type="checkbox"/>	Name ^	UC Service Type	Product Type	Host/IP Address	Port	Protocol
<input type="checkbox"/>	<a href="#">CTI SRV</a>	CTI	CTI	10.80.22.2	2748	TCP
<input type="checkbox"/>	<a href="#">IMP SRV</a>	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)

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and a 'Navigation' dropdown menu. Below this, a secondary navigation bar lists various system components like System, Call Routing, Media Resources, etc. The main content area is titled 'UC Service Configuration'. A toolbar with icons for Save, Delete, Copy, Reset, Apply Config, and Add New is present. The 'Status' section indicates the service is 'Ready'. The 'Add a UC Service' section is highlighted with a red box and contains the following configuration details:

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

At the bottom of the form, there are buttons for Save, Delete, Copy, Reset, Apply Config, and 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**  
For Cisco Unified Communications Solutions

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

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

**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

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Note: Testing was conducted in tekVizion labs



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

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

**Service Profile Configuration** Related Links: Back To Find/List Go

Save Delete Copy Add New

**Service Profile Information**

Name\* Jabber\_SVC\_Profile

Description Jabber Service Profile

☒ Make this the default service profile for the system

**Voicemail Profile**

Primary <None> ▾

Secondary <None> ▾

Tertiary <None> ▾

Credentials source for voicemail service\* Not set ▾

**MailStore Profile**

Primary <None> ▾

Secondary <None> ▾

Tertiary <None> ▾

Inbox Folder\* INBOX

Trash Folder\* Deleted Items

Polling Interval (in seconds)\* 60

☒ Allow dual folder mode

Service Profile Configuration (Contd...)

**Conferencing Profile**  
Primary <None> ▾  
Secondary <None> ▾  
Tertiary <None> ▾  
Server Certificate Verification Any ▾  
[Credentials source for web conference service](#)\* Not set ▾

**Directory Profile**  
Primary <None> ▾  
Secondary <None> ▾  
Tertiary <None> ▾  
☒ [Use UDS for Contact Resolution](#)  
☒ [Use Logged On User Credential](#)  
[Username](#) administrator  
[Password](#) .....  
[Search Base 1](#)  
[Search Base 2](#)  
[Search Base 3](#)  
☒ [Recursive Search on All Search Bases](#)  
[Search Timeout \(seconds\)\\*](#) 5  
[Base Filter \(Only used for Advance Directory\)](#)  
[Predictive Search Filter \(Only used for Advance Directory\)](#)

**IM and Presence Profile**  
Primary IMP\_SRV ▾  
Secondary <None> ▾  
Tertiary <None> ▾

**CTI Profile**  
Primary CTI\_SRV ▾  
Secondary <None> ▾  
Tertiary <None> ▾



## End User Configuration

**Navigation:** User Management → End User

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

Set Password = Password for profile.

Set Directory URI = jabber@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 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', and 'Bulk Administration'. The 'User Management' menu is expanded, showing 'Find and List Users'. Below this, there are buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'. A status box indicates '1 records found'. The main table lists users, with the following columns: 'User ID', 'Meeting Number', 'First Name', 'Last Name', 'Department', 'Directory URI', and 'User Status'. The first row, representing the user 'jabber', is highlighted with a red border. The 'User ID' column shows 'jabber', 'Meeting Number' shows 'cisco', 'First Name' shows 'cisco', 'Last Name' shows 'cisco', 'Department' shows 'cisco', 'Directory URI' shows 'jabber@lab.tekvizion.com', and 'User Status' shows 'Enabled Local User'. Below the table are buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'.

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



End User Configuration (continued...)

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**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

**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="jabber"/>	
Password	<input type="password" value="....."/>	<span>Edit Credential</span>
Confirm Password	<input type="password" value="....."/>	
Self-Service User ID	<input type="text"/>	
PIN	<input type="password" value="....."/>	<span>Edit Credential</span>
Confirm PIN	<input type="password" value="....."/>	
Last name*	<input type="text" value="cisco"/>	
Middle name	<input type="text"/>	
First name	<input type="text"/>	
Display name	<input type="text"/>	
Title	<input type="text"/>	
Directory URI	<input type="text" value="jabber@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"/>
Manager User ID	<input type="text"/>
Department	<input type="text"/>
User Locale	<span>&lt; None &gt;</span> ▾
Associated PC	<input type="text"/>
Digest Credentials	<input type="text"/>
Confirm Digest Credentials	<input type="text"/>
User Profile	<span>Use System Default( "Standard (Factory Default) U:</span> ▾ <span><a href="#">View Details</a></span>

**Service Settings**

☒ Home Cluster

☒ Enable User for Unified CM IM and Presence (Configure IM and Presence in the associated UC Service Profile)

☐ 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](#)



## End User Configuration(continued... )

**Device Information**

Controlled Devices

CSFuser1  
SEP00083031F5D4

^  
v

Device Association  
Line Appearance Association for Presence

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 >

☒ Allow Control of Device from CTI  
☐ 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 = Cisco7975\_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 Softkey Template = Standard User. This is used in this example.



**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 X Delete Copy Reset Apply Config Add New

**Association**

Modify Button Items

- 7975 Line [1] - 4086 (no partition)
- 7975 Line [2] - Add a new DN
- Add a new SD
- Add a new SD
- Add a new SD
- Add a new SD
- Add a new SD
- Add a new SD
- Unassigned Associated Items -----
- Add a new SD
- Add a new SURL
- Add a new BLF SD
- 7975 Add a new BLF Directed Call Park
- CallBack
- Call Park

**Phone Type**

Product Type: Cisco 7975  
Device Protocol: SCCP

**Real-time Device Status**

Registration: Registered with Cisco Unified Communications Manager clus32pubsub  
IPv4 Address: 172.16.31.184  
Active Load ID: SCCP75.9-4-2-1S  
Download Status: Unknown

**Device Information**

☒ Device is Active  
☒ Device is trusted

MAC Address\*: 00083031F5D4  
Description: 7323204086  
Device Pool\*: G729 Pool [View Details](#)  
Common Device Configuration: ATT\_SIP\_TRUNK [View Details](#)  
Phone Button Template\*: Standard 7975 SCCP  
Softkey Template: Standard User mobility  
Common Phone Profile\*: Standard Common Phone Profile [View Details](#)

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

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.

Check Owner = Anonymous (Public/Shared Space). This is used in this example.

15	Call Pickup	Calling Search Space	< None >
16	Conference List	AAR Calling Search Space	< None >
17	Conference	Media Resource Group List	MRGL_MTP
18	Do Not Disturb	User Hold MOH Audio Source	1-SampleAudioSource
19	End Call	Network Hold MOH Audio Source	1-SampleAudioSource
20	Forward All	Location*	Hub_None
21	Group Call Pickup	AAR Group	< None >
22	Hold	User Locale	< None >
23	Hunt Group Logout	Network Locale	< None >
24	<a href="#">7775 Intercom [1] - Add a new Intercom</a>	Built In Bridge*	Default
25	Malicious Call Identification	Privacy*	Default
26	Meet Me Conference	Device Mobility Mode	Default <a href="#">View Current Device Mobility Settings</a>
27	Mobility	Owner	<input type="radio"/> User <input checked="" type="radio"/> Anonymous (Public/Shared Space)
28	New Call	Owner User ID	
29	Other Pickup	Phone	Default
30	Quality Reporting Tool	Personalization*	
31	Redial	Services Provisioning*	Default
32	Remove Last Participant		
33	Transfer		
34	Video Mode		
35	Queue Status		

36	Privacy	Phone Load Name	
37	None	Single Button Barge	Default
		Join Across Lines	Default
		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 >

<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

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

**Number Presentation Transformation**

**Caller ID For Calls From This Phone**

Calling Party Transformation

< None >

CSS

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

**Remote Number**

Calling Party Transformation

< None >

CSS

☒ 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

Device Security Profile\*

Cisco 7975 - Standard SCCP Non-Secure Profile

SUBSCRIBE Calling Search Space

< None >

☐ Unattended Port

☐ Require DTMF Reception

☐ RFC2833 Disabled

**Certification Authority Proxy Function (CAPF) Information**

Certificate Operation\*

No Pending Operation

Authentication Mode\*

By Null String

Authentication String

Generate String

Key Order\*

RSA Only

RSA Key Size (Bits)\*

2048

EC Key Size (Bits)

< None >

Operation Completes By

2016

1

18

12

(YYYY:MM:DD:HH)

Certificate Operation

None

Status:

Note: Security Profile Contains Addition CAPF Settings.

**Expansion Module Information**

Module 1

< None >

Module 1 Load Name

Module 2

< None >

Module 2 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 >

Secure Shell Information	
Secure Shell User	administrator
Secure Shell Password	.....

Product Specific Configuration Layout		
	Parameter Value	Override Common Settings
<input type="checkbox"/> Disable Speakerphone		
<input type="checkbox"/> Disable Speakerphone and Headset		
Forwarding Delay*	Disabled	
PC Port *	Enabled	
Settings Access*	Enabled	<input type="checkbox"/>
Gratuitous ARP*	Disabled	
PC Voice VLAN Access*	Enabled	
Video Capabilities*	Disabled	<input type="checkbox"/>

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

Auto Line Select *	Disabled	
Web Access*	Disabled	<input type="checkbox"/>
Days Display Not Active	Sunday Monday Tuesday	<input type="checkbox"/>
Display On Time	07:30	<input type="checkbox"/>
Display On Duration	10:30	<input type="checkbox"/>
Display Idle Timeout	01:00	<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="text"/>	<input type="checkbox"/>
<input type="checkbox"/> Allow EnergyWise Overrides		<input type="checkbox"/>
Span to PC Port*	Disabled	
Logging Display*	PC Controlled	
Load Server	<input type="text"/>	<input type="checkbox"/>
Recording Tone*	Disabled	
Recording Tone Local Volume*	100	
Recording Tone Remote Volume*	50	
Recording Tone Duration	<input type="text"/>	
Display On When Incoming Call*	Disabled	<input type="checkbox"/>
RTCP*	Disabled	<input type="checkbox"/>
"more" Soft Key Timer	5	
Auto Call Select*	Enabled	
Log Server	<input type="text"/>	<input type="checkbox"/>
Advertise G.722 Codec*	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"/>

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

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	
Wireless Headset Hookswitch Control*	Disabled	
IPv6 Load Server		<input type="checkbox"/>
IPv6 Log Server		
802.1x Authentication*	User Controlled	<input type="checkbox"/>
Detect Unified CM Connection Failure*	Normal	<input type="checkbox"/>

Minimum Ring Volume*	0-Silent	
Headset Sidetone Level*	Default	
Headset Send Gain*	Default	
HTTPS Server*	http and https Enabled	<input type="checkbox"/>
Handset/Headset Monitor*	Enabled	
Headset Recording*	Disabled	
Enbloc Dialing*	Enabled	
Switch Port Remote Configuration*	Disabled	<input type="checkbox"/>
PC Port Remote Configuration*	Disabled	<input type="checkbox"/>
Automatic Port Synchronization*	Disabled	<input type="checkbox"/>
SSH Access*	Disabled	<input type="checkbox"/>
LOGIN Access*	Enabled	<input type="checkbox"/>
FIPS Mode*	Disabled	<input type="checkbox"/>
80-bit SRTCP*	Disabled	<input type="checkbox"/>

Customer Support Use	<input type="checkbox"/>
----------------------	--------------------------



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

Set Directory Number\* = 4086. This is used in this example.

Set Description = 7323204086. This is used in this example.

Set Alerting Name = 7323204086. This is used in this example.

Set ASCII Alerting Name = 7323204086. This is used in this example.

**Directory Number 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: [Configure Device \(SEP00083031F5D4\)](#) | Go

Save | Delete | Reset | Apply Config | Add New

**Directory Number Information**

Directory Number\* 4086 ☐ Urgent Priority

Route Partition < None >

Description 7323204086

Alerting Name 7323204086

ASCII Alerting Name 7323204086

External Call Control Profile < None >

☒ Allow Control of Device from CTI

Associated Devices SEP00083031F5D4

[Edit Device](#)

[Edit Line Appearance](#)

Dissociate Devices

**Directory Number Settings**

Voice Mail Profile VM\_profile (Choose <None> to use system default)

Calling Search Space < None >

BLF Presence Group\* Standard Presence group

User Hold MOH Audio Source 1-SampleAudioSource

Network Hold MOH Audio Source 1-SampleAudioSource

Auto Answer\* Auto Answer Off

☐ Reject Anonymous Calls

**Enterprise Alternate Number**

[Add Enterprise Alternate Number](#)

**+E.164 Alternate Number**

[Add +E.164 Alternate Number](#)



## Cisco IP Phone 7965 SCCP Configuration (Continued...)

Directory URIs			
Primary	URI	Partition	Advertise Globally via ILS
<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 checked="" type="checkbox"/> or	<input type="text"/>
Forward Busy External	<input checked="" type="checkbox"/> or	<input type="text"/>
Forward No Answer Internal	<input checked="" type="checkbox"/> or	<input type="text"/>
Forward No Answer External	<input checked="" type="checkbox"/> or	<input type="text"/>
Forward No Coverage Internal	<input checked="" type="checkbox"/> or	<input type="text"/>
Forward No Coverage External	<input checked="" type="checkbox"/> or	<input type="text"/>
Forward on CTI Failure	<input checked="" type="checkbox"/> or	<input type="text"/>
Forward Unregistered Internal	<input checked="" type="checkbox"/> or	<input type="text"/>
Forward Unregistered External	<input checked="" type="checkbox"/> or	<input type="text"/>
No Answer Ring Duration (seconds)		<input type="text"/>
Call Pickup Group		< None >



## Cisco IP Phone 7965 SCCP Configuration (Continued...)

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"/> Reversion Timer service parameter	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

Line 1 on Device SEP00083031F5D4	
Display (Caller ID)	7323204086 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)	7323204086
Line Text Label	7323204086
External Phone Number Mask	7323204086
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



## Cisco IP Phone 7965 SCCP Configuration (Continued...)

Call Pickup Group	Use System Default	▼
Audio Alert Setting(Phone Active)		
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		

**Multiple Call/Call Waiting Settings on Device SEP00083031F5D4**

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 SEP00083031F5D4**

☒ Caller Name  
☐ Caller Number  
☐ Redirected Number  
☒ Dialed Number

**Users Associated with Line**

Associate End Users



## Cisco IP Phone 9971 SIP Configuration

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

Set Description = 7323204085. this text is used to identify this Phone.

Set Device Pool\* = G729 Pool. 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

administrator | Search Documentation | About | Logout

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

**Phone Configuration** Related Links:

**Association**

- 1 [Line \[1\] - 4085 \(no partition\)](#)
- 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](#)

**Phone Type**

**Product Type:** Cisco 9971  
**Device Protocol:** SIP

**Real-time Device Status**

**Registration:** Registered with Cisco Unified Communications Manager clus32pubsub  
**IPv4 Address:** [172.16.31.110](#)  
**Active Load ID:** sip9971.9-4-2-13  
**Inactive Load ID:** sip9971.9-4-1-9  
**Download Status:** Unknown

----- 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
- 18 Mobility
- 19 Other Pickup
- 20 Quality Reporting Tool
- 21 Redial
- 22 [Add a new SURF](#)
- 23 [Add a new BLF SD](#)

**Device Information**

☒ Device is Active  
☒ Device is trusted

MAC Address\* C07BBCA1B7DD  
Description 7323204085  
Device Pool\* G729 Pool [View Details](#)  
Common Device Configuration < None > [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 >  
AAR Calling Search Space < None >  
Media Resource MRGL\_MTP  
Group List  
User Hold MOH Audio Source 1-SampleAudioSource  
Network Hold MOH Audio Source 1-SampleAudioSource

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

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24	Answer Oldest	Location*	Hub_None
25	Do Not Disturb	AAR Group	< None >
26	Services	User Locale	< None >
27	Record	Network Locale	< None >
28	Alerting Calls	Built In Bridge*	Default
29	Queue Status	Privacy*	Default
30	Privacy	Device Mobility Mode	Default
31	None		<a href="#">View Current Device Mobility Settings</a>
		Owner	<input checked="" type="radio"/> User <input type="radio"/> Anonymous (Public/Shared Space)
		Owner User ID*	jabber
		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 >
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	

<b>Number Presentation Transformation</b>	
<b>Caller ID For Calls From This Phone</b>	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)	
<b>Remote Number</b>	
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	711ulaw
Originating Codec*	
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 w/Early Media Disabled <a href="#">View Details</a>
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	
<a href="#">Generate String</a>	
Key Order*	RSA Only
RSA Key Size (Bits)*	2048
EC Key Size (Bits)	< None >
Operation Completes By	2016 1 21 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	<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 >

Secure Shell Information	
Secure Shell User	administrator
Secure Shell Password	.....



Product Specific Configuration Layout		
	Parameter Value	Override Common Settings
		
<input type="checkbox"/> Disable Speakerphone		
<input type="checkbox"/> Disable Speakerphone and Headset		
PC Port *	Enabled	
Back USB Port*	Enabled	<input type="checkbox"/>
Side USB Port*	Enabled	<input type="checkbox"/>
Cisco Camera*	Disabled	<input type="checkbox"/>
Console Access*	Disabled	<input type="checkbox"/>
Video Capabilities*	Disabled	<input type="checkbox"/>

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

Enable/Disable USB Classes	Mass Storage Human Interface Device Audio Class	<input type="checkbox"/>
SDIO *	Disabled	<input type="checkbox"/>
Bluetooth *	Enabled	<input type="checkbox"/>
Wifi *	Enabled	<input type="checkbox"/>
Bluetooth Profiles*	Handsfree Human Interface Device	<input type="checkbox"/>
Settings Access*	Enabled	<input type="checkbox"/>
Gratuitous ARP*	Disabled	
PC Voice VLAN Access*	Enabled	
Web Access*	Disabled	<input type="checkbox"/>
Show All Calls on Primary Line*	Disabled	
Days Display Not Active	Sunday Monday Tuesday	<input type="checkbox"/>

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
	RTCP*	Disabled
	Log Server	<input type="checkbox"/>
	IPv6 Log Server	<input type="checkbox"/>
	Remote Log*	Disabled

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

	Log Profile	Default Preset Telephony	<input type="checkbox"/>
	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):	Enabled	<input type="checkbox"/>
PC Port*		
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	<input type="checkbox"/>

SSH Access*	Disabled	<input type="checkbox"/>
Incoming Call Toast Timer*	5	<input type="checkbox"/>
Provide Dial Tone from Release Button*	Disabled	<input type="checkbox"/>
Hide Video By Default*	Disabled	<input type="checkbox"/>
Background Image		<input type="checkbox"/>
Simplified New Call UI*	Disabled	<input type="checkbox"/>

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

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"/>

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



Set Directory Number\* = 4084. This is used in this example.

Set Description = 7323204084. This is used in this example.

Set Alerting Name = Cisco 9971 Phone. This is used in this example.

Set ASCII Alerting Name = Cisco 9971 Phone. This is used in this example.

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**Directory Number Configuration** Related Links: Configure Device (SEPC07BBCA1B7DD) Go

Save Delete Reset Apply Config Add New

**Directory Number Information**

Directory Number\* 4085 ☐ Urgent Priority

Route Partition < None >

Description 7323204085

Alerting Name 7323204085

ASCII Alerting Name 7323204085

External Call Control Profile < None >

☒ Allow Control of Device from CTI

Associated Devices CSFuser1  
SEPC07BBCA1B7DD

Edit Device  
Edit Line Appearance

Dissociate Devices

**Directory Number Settings**

Voice Mail Profile < None > (Choose <None> to use system default)

Calling Search Space < None >

BLF Presence Group\* Standard Presence group

User Hold MOH Audio Source 1-SampleAudioSource

Network Hold MOH Audio Source 1-SampleAudioSource

Auto Answer\* Auto Answer Off

☐ Reject Anonymous Calls

**Enterprise Alternate Number**

Add Enterprise Alternate Number

**+E.164 Alternate Number**

Add +E.164 Alternate Number





Directory URIs			
Primary	URI	Partition	Advertise Globally via ILS
<input checked="" type="radio"/>	jabber@lab.tekvizion.com	< None >	<input checked="" type="checkbox"/>
<input type="button" value="Add Row"/>			

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

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 Group
AAR	<input type="checkbox"/> or		< 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
Calling Search Space Activation Policy			Use System Default
Forward All	<input type="checkbox"/> or		< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input type="checkbox"/> or		< None >
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	<input type="text"/>	< None >
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 >		

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

Park Monitoring			
	Voice Mail	Destination	Calling Search Space
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="&lt; None &gt;"/> A blank value means to call the parker's line.
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="&lt; None &gt;"/> 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

Line 1 on Device SEPC07BBCA1B7DD		
	Value	Update Shared Device Settings
Display (Caller ID)	<input type="text" value="7323204085"/> 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.	<input type="checkbox"/>
ASCII Display (Caller ID)	<input type="text" value="7323204085"/>	<input type="checkbox"/>
Line Text Label	<input type="text" value="7323204085"/>	<input type="checkbox"/>
External Phone Number Mask	<input type="text" value="7323204085"/>	<input type="checkbox"/>
Visual Message Waiting Indicator Policy*	Use System Policy	<input type="checkbox"/>
Audible Message Waiting Indicator Policy*	Default	<input type="checkbox"/>
Ring Setting (Phone Idle)*	Use System Default	<input type="checkbox"/>

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

Ring Setting (Phone Active)	Use System Default	Applies to this line when any line on the phone has a call in progress.	<input type="checkbox"/>
Call Pickup Group	Use System Default		
Audio Alert Setting(Phone Idle)			
Call Pickup Group	Use System Default		
Audio Alert Setting(Phone Active)			
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			<input type="checkbox"/>
			<input type="button" value="Submit"/> <input type="checkbox"/> Propagate Selected

**Multiple Call/Call Waiting Settings on Device SEPC07BBCA1B7DD**

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 SEPC07BBCA1B7DD**

☐ Caller Name  
☐ Caller Number  
☐ Redirected Number  
☐ Dialed Number

**Users Associated with Line**

	Full Name	User ID	Permission
<input checked="" type="checkbox"/>	cisco.	jabber	



## 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.

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**SIP Trunk Security Profile Configuration** Related Links: Back To Find/List ▾ Go

Save X 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 prompts 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".

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

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**SIP Profile Information**

Name\* Standard SIP Profile w/Early Media Disabled

Description Standard SIP Profile w/Early Media Disabled

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-Agen ▾

Version in User Agent and Server Header\* Major And Minor ▾

Dial String Interpretation\* Phone number consists of characters 0-9, \*, #, and ▾

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)*	1800
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
Media Port Ranges	<input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766
DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default
DSCP for Audio Portion of Video Calls	Use System Default
DSCP for TelePresence Calls	Use System Default
DSCP for Audio Portion of TelePresence Calls	Use System Default

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





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

Call Forward URI*	x-cisco-serviceuri-cfwdall				
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial				
<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					
<b>Normalization Script</b>					
Normalization Script	< None >				
<input type="checkbox"/> Enable Trace					
1	<table><thead><tr><th>Parameter Name</th><th>Parameter Value</th></tr></thead><tbody><tr><td></td><td></td></tr></tbody></table>	Parameter Name	Parameter Value		
Parameter Name	Parameter Value				

<b>Incoming Requests FROM URI Settings</b>	
Caller ID DN	
Caller Name	

<b>Trunk Specific Configuration</b>	
Reroute Incoming Request to new Trunk based on*	Never
Resource Priority Namespace List	< None >
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)
<input type="checkbox"/> Enable ANAT	
<input type="checkbox"/> Deliver Conference Bridge Identifier	
<input type="checkbox"/> Allow Passthrough of Configured Line Device Caller Information	
<input type="checkbox"/> Reject Anonymous Incoming Calls	
<input type="checkbox"/> Reject Anonymous Outgoing Calls	
<input type="checkbox"/> Send ILS Learned Destination Route String	

SIP OPTIONS Ping	
<input checked="" type="checkbox"/> 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)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6

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

SDP Information
<input type="checkbox"/> Send send-receive SDP in mid-call INVITE
<input type="checkbox"/> Allow Presentation Sharing using BFCP
<input type="checkbox"/> Allow iX Application Media
<input type="checkbox"/> Allow multiple codecs in answer SDP

Save

Delete

Copy

Reset

Apply Config

Add New



## 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.

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**Trunk Configuration** Related Links: Back To Find/List Go

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**SIP Trunk Status**

**Service Status:** Full Service  
**Duration:** Time In Full Service: 4 days 22 hours 36 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	< 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



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

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

Packet Capture Mode*	None
Packet Capture Duration	0
<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**

<input checked="" type="checkbox"/> Remote-Party-Id <input checked="" type="checkbox"/> 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	
<input checked="" type="checkbox"/> 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	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input style="width: 100%;" type="text" value=" &lt; None &gt; "/>	<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	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input style="width: 100%;" type="text" value=" &lt; None &gt; "/>	<input checked="" type="checkbox"/>

### Connected Party Settings

Connected Party Transformation CSS

☒ Use Device Pool Connected Party Transformation CSS

### Outbound Calls

Called Party Transformation CSS

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection\*

Calling Line ID Presentation\*

Calling Name Presentation\*

Calling and Connected Party Info Format\*

☒ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

☒ 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\* = ATT\_SIP\_Profile. This is used in this example.

Destination		
<input type="checkbox"/> Destination Address is an SRV		
Destination Address	Destination Address IPv6	Destination
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 <a href="#">View Details</a>	
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	
Save Delete Reset 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.

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

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

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

**Trunk Configuration** Related Links: Back To Find/List ▾ Go

Save Delete Reset Add New

**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* <input type="text" value="When using both sRTP and TLS"/> Route Class Signaling Enabled* <input type="text" value="Default"/> Use Trusted Relay Point* <input type="text" value="Default"/> <input checked="" type="checkbox"/> PSTN Access <input checked="" type="checkbox"/> Run On All Active Unified CM Nodes
<b>Intercompany Media Engine (IME)</b> E.164 Transformation Profile <input type="text" value=" &lt; None &gt;"/>

<b>MLPP and Confidential Access Level Information</b> MLPP Domain <input type="text" value=" &lt; None &gt;"/> Confidential Access Mode <input type="text" value=" &lt; None &gt;"/> Confidential Access Level <input type="text" value=" &lt; None &gt;"/>
<b>Call Routing Information</b> <input checked="" type="checkbox"/> Remote-Party-Id <input checked="" type="checkbox"/> Asserted-Identity Asserted-Type* <input type="text" value=" Default"/> SIP Privacy* <input type="text" value=" Default"/>
<b>Inbound Calls</b> Significant Digits* <input type="text" value=" All"/> Connected Line ID Presentation* <input type="text" value=" Default"/> Connected Name Presentation* <input type="text" value=" Default"/> Calling Search Space <input type="text" value=" &lt; None &gt;"/> AAR Calling Search Space <input type="text" value=" &lt; None &gt;"/> Prefix DN <input type="text"/> <input type="checkbox"/> Redirecting Diversion Header Delivery - Inbound



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

Incoming Calling Party Settings				
<p>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.</p>				
<div>Clear Prefix Settings    Default Prefix Settings</div>				
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 type="text" value=" &lt; None &gt;"/>	<input checked="" type="checkbox"/>

Incoming Called Party Settings				
<p>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.</p>				
<div>Clear Prefix Settings    Default Prefix Settings</div>				
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 type="text" value=" &lt; None &gt;"/>	<input type="checkbox"/>

Connected Party Settings	
Connected Party Transformation CSS	<input type="text" value=" &lt; None &gt;"/>
<input checked="" type="checkbox"/> Use Device Pool Connected Party Transformation CSS	



<b>Outbound Calls</b>	
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> 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
<input type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	
Redirecting Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Redirecting Party Transformation CSS	
<b>Caller Information</b>	
Caller ID DN	
Caller Name	
<input type="checkbox"/> 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 w/Early Media Disabled

[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



## 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 shows 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", "administrator", "Search Documentation", and "About". Below this is a secondary navigation bar with tabs for "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Bulk Administration". The main content area is titled "Find and List Route Patterns". It features a toolbar with buttons for "Add New", "Select All", "Clear All", and "Delete Selected". A status box indicates "3 records found". Below this is a table titled "Route Patterns (1 - 3 of 3)" with a "Rows per Page" dropdown set to 50. The table has columns for "Pattern", "Description", "Partition", "Route Filter", "Associated Device", and "Copy". Three records are listed: "4084" (To FAX, Trunk SIP FAX Gateway), "9.\*X!" (Network-Based Call Forwarding, ATT SIP TRUNK), and "9.@" (To PSTN via ATT SIP Trunk, ATT SIP TRUNK). The last two records are highlighted with a red box. At the bottom of the table are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

Pattern	Description	Partition	Route Filter	Associated Device	Copy
4084	To FAX			Trunk SIP FAX Gateway	
9.*X!	Network-Based Call Forwarding			ATT SIP TRUNK	
9.@	To PSTN via ATT SIP Trunk			ATT SIP TRUNK	



## Route Pattern Configuration (Continued...)

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**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...)

<b>Connected Party Transformations</b>		
Connected Line ID Presentation *	Default	
Connected Name Presentation *	Default	
<b>Called Party Transformations</b>		
Discard Digits	PreDot	
Called Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Called Party Number Type *	Cisco CallManager	
Called Party Numbering Plan *	Cisco CallManager	
<b>ISDN Network-Specific Facilities Information Element</b>		
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\* = 9.\*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*	9.*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 <a href="#">(Edit)</a>
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"/> Require Forced Authorization Code	<input type="checkbox"/> Allow Overlap Sending
Authorization Level*	0
<input type="checkbox"/> Require Client Matter Code	<input type="checkbox"/> Urgent Priority

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...)

<b>Connected Party Transformations</b>			
Connected Line ID Presentation*	Default		
Connected Name Presentation*	Default		
<b>Called Party Transformations</b>			
Discard Digits	PreDot		
Called Party Transform Mask			
Prefix Digits (Outgoing Calls)			
Called Party Number Type*	Cisco CallManager		
Called Party Numbering Plan*	Cisco CallManager		
<b>ISDN Network-Specific Facilities Information Element</b>			
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\* = 4084 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\* = Trunk\_SIP\_FAX\_Gateway. This is used for this example.





All other values are default


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**Route Pattern Configuration**
Related Links: [Back To Find/List](#) [Go](#)

 Save
  Delete
  Copy
  Add New

**Pattern Definition**

Route Pattern*	4084
Route Partition	< None >
Description	To FAX
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*	Trunk_SIP_FAX_Gateway <a href="#">(Edit)</a>
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**

☐ 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...)



<b>Connected Party Transformations</b>		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
<b>Called Party Transformations</b>		
Discard Digits	< None >	
Called Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Called Party Number Type*	Cisco CallManager	
Called Party Numbering Plan*	Cisco CallManager	
<b>ISDN Network-Specific Facilities Information Element</b>		
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

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Note: Testing was conducted in tekVizion labs



Select Phone Type\* = Cisco Unified Client services framework  
Set Device Name\* = CSFUser1. This is used in this example.  
Set Description = jabberclient. This is used in this example.  
Select Device Pool = G729 Pool. This is used in this example.  
Select Phone Button Template\* = Standard Client Services Framework.

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

Save Delete Copy Reset Apply Config Add New

**Association**

Modify Button Items

1 Line [1] - 4085 (no partition)  
----- Unassigned Associated Items -----

2 Line [2] - Add a new DN

**Phone Type**

Product Type: Cisco Unified Client Services Framework  
Device Protocol: SIP

**Real-time Device Status**

Registration: Unknown  
IPv4 Address: None

**Device Information**

☒ Device is Active  
☒ Device is trusted

Device Name\* CSFuser1  
Description jabberclient  
Device Pool\* G729 Pool View Details  
Common Device Configuration < None > View Details  
Phone Button Template\* Standard Client Services Framework  
Common Phone Profile\* Standard Common Phone Profile View Details

## Jabber Client Configuration (Contd...)

Media Resource Group List = MRGL\_MTP



Set Owner check box

Set Owner user ID\* = jabber. 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 <a href="#">View Current Device Mobility Settings</a>
Owner	<input checked="" type="radio"/> User <input type="radio"/> Anonymous (Public/Shared Space)
Owner User ID*	jabber
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	

<b>Number Presentation Transformation</b>	
<b>Caller ID For Calls From This Phone</b>	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)	

## Jabber Client Configuration (Contd...)

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Note: Testing was conducted in tekVizion labs

<b>Remote Number</b>	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)	

<b>Protocol Specific Information</b>	
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 - Standard
Rerouting Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile w/Early Media Disabled <a href="#">View Details</a>
Digest User	< None >
<input type="checkbox"/> Media Termination Point Required <input type="checkbox"/> Unattended Port <input type="checkbox"/> Require DTMF Reception	

<b>Certification Authority Proxy Function (CAPF) Information</b>	
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
<input type="button" value="Generate String"/>	
Key Order*	RSA Only
RSA Key Size (Bits)*	2048
EC Key Size (Bits)	< None >
Operation Completes By	2016 1 21 12 (YYYY:MM:DD:HH)
Certificate Operation None	
Status:	
Note: Security Profile Contains Addition CAPF Settings.	
<b>Extension Information</b>	
<input type="checkbox"/> Enable Extension Mobility	
Log Out Profile	-- Use Current Device Settings --
Log in Time	< None >
Log out Time	< None >





## Jabber Client Configuration (Contd...)

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 >

Product Specific Configuration Layout		
	Parameter Value	Override Common Settings
Video Calling *	Enabled	<input type="checkbox"/>
<b>Interactive Connectivity Establishment (ICE)</b>		
ICE	Enabled	<input type="checkbox"/>
Default Candidate Type	Host	<input type="checkbox"/>
Server Reflexive Address	Enabled	<input type="checkbox"/>
Primary TURN Server Host Name or IP Address		<input type="checkbox"/>
Secondary TURN Server Host Name or IP Address		<input type="checkbox"/>
TURN Server Transport Type	Auto	<input type="checkbox"/>

TURN Server Username	administrator	<input checked="" type="checkbox"/>
TURN Server Password	.....	<input checked="" type="checkbox"/>
<b>Instant Messaging</b>		
File Types to Block in File Transfer		<input type="checkbox"/>
URLs to Block in File Transfer		<input type="checkbox"/>



## Jabber Client Configuration (Contd...)

Desktop Client Settings	
Automatically Start in Phone Control*	<div>Disabled</div> <div></div>
Automatically Control Tethered Desk Phone*	<div>Disabled</div> <div></div>
Extend and Connect Capability*	<div>Enabled</div> <div></div>
Display Contact Photos*	<div>Enabled</div> <div></div>
Number Lookups on Directory*	<div>Enabled</div> <div></div>
Jabber For Windows Software Update	<div>jabber@lab.tekvizion.com</div> <div></div>
Server URL	
Problem Report	<div></div> <div></div>
Server URL	
Analytics Collection*	<div>Disabled</div> <div></div>

Analytics Server URL	<div></div> <div></div>
Cisco Support Field	<div></div> <div></div>

Save

Delete

Copy

Reset

Apply Config

Add New



## Voicemail Port Configuration

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

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**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"/>	<a href="#">CiscoUM1-VI1</a>		<a href="#">G729 Pool</a>	Non Secure Voice Mail Port		2295		Registered with clus32pubsub	10.80.22.4	
<input type="checkbox"/>	<a href="#">CiscoUM1-VI2</a>		<a href="#">G729 Pool</a>	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 = VoiceMail. This is used for this example.

Set Device Pool = G729 Pool

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	



## 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, there are tabs for "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Bulk Administration". The "Advanced Features" tab is selected, and the "Message Waiting Numbers" page is displayed.

The page title is "Find and List Message Waiting Numbers". Below the title, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected". A status bar indicates "2 records found".

The main content area shows a table of Message Waiting Numbers. The table has columns for "Directory Number", "Description", "Partition", "Calling Search Space", and "Copy". Two records are listed:

Directory Number	Description	Partition	Calling Search Space	Copy
2298	MWI ON			
2299	MWI OFF			

Below the table, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".



## 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-Default



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### Find and List Voice Mail Pilots

+ Add New   Select All   Clear All   Delete Selected

**Status**  
 3 records found

**Voice Mail Pilot (1 - 3 of 3)** Rows per Page 50 ▾

Voice  
Find Mail where Voice Mail Pilot Number ▾ begins with ▾  Find Clear Filter

Pilot

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

Add New Select All Clear All Delete Selected

**Voice Mail Pilot Information**

Voice Mail Pilot Number

Calling Search Space

Description

☒ Make this the default Voice Mail Pilot for the system

Save Delete Add New

## FAX Gateway Configuration

cme.in.tekvizion.com#sh version



Cisco IOS Software, 2800 Software (C2800NM-IPVOICEK9-M), Version 15.1(4)M5, RELEASE SOFTWARE (fc1)

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Compiled Tue 04-Sep-12 15:56 by prod\_rel\_team

ROM: System Bootstrap, Version 12.4(13r)T, RELEASE SOFTWARE (fc1)

cme.in.tekvizion.com uptime is 1 day, 5 hours, 59 minutes

System returned to ROM by reload at 14:27:25 IST Sun Jan 10 2016

System image file is "flash:c2800nm-ipvoicek9-mz.151-4.M5.bin"

Last reload type: Normal Reload

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<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.

Cisco 2851 (revision 1.0) with 249856K/12288K bytes of memory.

Processor board ID FHK1137F4LY

2 Gigabit Ethernet interfaces

62 Serial interfaces

2 terminal lines

2 Channelized E1/PRI ports

4 Voice FXS interfaces

2 cisco service engine(s)

DRAM configuration is 64 bits wide with parity enabled.

239K bytes of non-volatile configuration memory.

62720K bytes of ATA CompactFlash (Read/Write)

License Info:

License UDI:

```
-----  
Device# PID          SN  
-----  
*0      CISCO2851      FHK1137F4LY
```



Configuration register is 0x2102

cme.in.tekvizion.com#sh running-config

Building configuration...

Current configuration : 11391 bytes

!

! Last configuration change at 15:08:21 IST Sun Jan 10 2016 by cisco

version 15.1

service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

!

hostname cme.in.tekvizion.com

!

boot-start-marker

boot-end-marker

!

!

enable password tekV1z10n

!

aaa new-model

!

!



```
aaa authentication login local_auth local
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
aaa session-id common
```

```
clock timezone IST 5 30
```

```
network-clock-participate wic 2
```

```
network-clock-participate wic 3
```

```
!
```

```
dot11 syslog
```

```
ip source-route
```

```
!
```

```
!
```

```
ip cef
```

```
!
```

```
!
```

```
!
```

```
ip host Clus1-862-Pub 172.16.26.2
```

```
no ipv6 cef
```

```
multilink bundle-name authenticated
```

```
!
```

```
!
```

```
!
```

```
!
```



```
isdn switch-type primary-qsig
!
!
voice rtp send-recv
!
voice service pots
!
voice service voip
no ip address trusted authenticate
allow-connections sip to sip
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
g729 annexb-all
!
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711ulaw
!
voice class codec 2
codec preference 1 g711ulaw
codec preference 2 g729r8
!
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-card 0

!

crypto pki token default removal timeout 0

!

!

!

!

license udi pid CISCO2851 sn FHK1137F4LY

username cisco password 0 tekV1z10n

!

!

controller E1 0/2/0

  pri-group timeslots 1-31 service mgcp

!

controller E1 0/3/0

  clock source internal

  pri-group timeslots 1-31

!
```



```
ip tftp source-interface GigabitEthernet0/0
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
interface GigabitEthernet0/0
```

```
ip address 172.16.31.50 255.255.255.0
```

```
duplex auto
```

```
speed auto
```

```
!
```

```
interface Service-Engine0/0
```

```
no ip address
```

```
shutdown
```

```
!
```

```
interface GigabitEthernet0/1
```

```
no ip address
```

```
ip nat outside
```

```
ip virtual-reassembly in
```

```
shutdown
```

```
duplex auto
```

```
speed auto
```

```
!
```

```
interface Serial0/2/0:15
```

```
no ip address
```



```
encapsulation hdlc
isdn switch-type primary-qsig
isdn timer T310 120000
isdn protocol-emulate network
isdn incoming-voice voice
isdn map address .* plan isdn type national
isdn bind-l3 ccm-manager
isdn send-alerting
isdn sending-complete
no cdp enable
!
interface Serial0/3/0:15
no ip address
encapsulation hdlc
isdn switch-type primary-qsig
isdn timer T310 120000
isdn protocol-emulate network
isdn incoming-voice voice
no cdp enable
!
interface Service-Engine1/0
no ip address
shutdown
!
ip forward-protocol nd
!
```



```
ip http server
no ip http secure-server
!
ip route 0.0.0.0 0.0.0.0 172.16.31.1
!
access-list 1 permit 172.16.31.0 0.0.0.255
!
snmp-server community public RO
snmp-server location Chennai
!
!
!
!
control-plane
!
!
voice-port 0/0/0
no vad
shutdown
!
voice-port 0/0/1
no vad
shutdown
!
voice-port 0/3/0:15
!
```





voice-port 0/2/0:15

!

voice-port 0/1/0

no vad

shutdown

!

voice-port 0/1/1

cptone IN

station-id number 7323204084

caller-id enable

!

no mgcp timer receive-rtcp

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 777 pots

huntstop

service session



```
destination-pattern 4084
no digit-strip
port 0/1/1
forward-digits all
!
dial-peer voice 9224 voip
description CUCM to Gateway
service session
session protocol sipv2
session transport udp
incoming called-number 4084
voice-class codec 3
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 92240 voip
description Gateway to CUCM
service session
destination-pattern 9T
session protocol sipv2
session target ipv4:10.80.22.2
session transport udp
```



```
voice-class codec 3

dtmf-relay rtp-nte

fax-relay sg3-to-g3

fax rate 14400

fax nsf 000000

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

no vad

!

!

!

gateway

timer receive-rtp 1200

!

sip-ua

credentials username +19728522671 password 7 11584B5643475D realm bsrhelas.lab.tekvizion.com

no remote-party-id

retry register 5

timers connection aging 30

timers update 1000

no timers hold

timers register 1000

!

!

telephony-service
```



```
max-ephones 50
max-dn 60
ip source-address 172.16.31.50 port 2000
service phone sshAccess 0
cnf-file perphone
max-conferences 8 gain -6
web admin system name Administrator password tekV1z10n
transfer-system full-consult
create cnf-files version-stamp 7960 Nov 22 2013 19:05:58
!
!
ephone-dn 1
!
!
ephone-dn 2
!
!
ephone-dn 5
!
!
ephone-dn 6 dual-line
!
!
ephone-dn 10
!
!
```



```
ephone 1
mac-address 001D.A21A.3577
type 7961
button 1:1
```

```
!
!
!
```

```
ephone 2
mac-address 001D.A21A.291D
type 7961
button 1:2
```

```
!
!
!
```

```
ephone 5
mac-address 0008.21DF.CEC5
type 7940
ssh userid cisco password cisco
button 1:5
```

```
!
!
!
```

```
ephone 6
!
!
!
```



ephone 10

mac-address 0C27.2431.5FB9

button 1:10

!

!

!

banner login ^CC

=====

WELCOME to CISCO CME 8.6

=====

Cisco IOS Software (C2800NM-IPVOICEK9-M)

Version 15.1(4)M5, RELEASE SOFTWARE (fc1)

Cisco 2851 with 249856K/12288K bytes of memory

Processor board ID FHK1137F4LY

2 Gigabit Ethernet interfaces

2 terminal lines

2 Voice FXS interfaces

2 cisco service engine(s)

239K bytes of non-volatile configuration memory

62720K bytes of ATA CompactFlash (Read/Write)

Warning: Access is restricted.

All user activity is logged!

For support, contact 'kkumaraguru@tekvision.com'



=====

```
^C
!
line con 0
line aux 0
line 66
  no activation-character
  no exec
  transport preferred none
  transport input all
  transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
line 194
  no activation-character
  no exec
  transport preferred none
  transport input all
  transport output all
line vty 0 4
  session-timeout 180
  exec-timeout 0 0
  password tekV1z10n
  login authentication local_auth
  transport input all
!
scheduler allocate 20000 1000
ntp server 103.6.16.254
```



end

cme.in.tekvizion.com#

## Cisco UCM SCCP Integration with Cisco Unity Connection (CUC)

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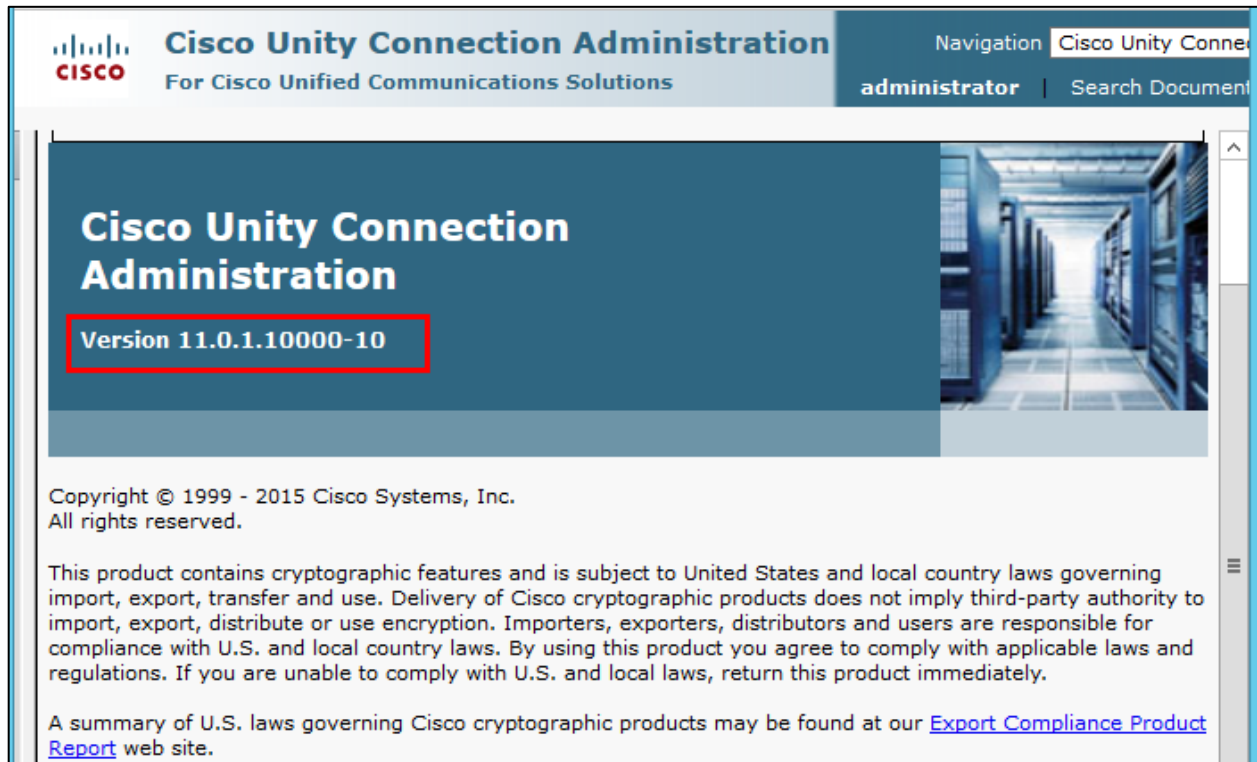
Page **136** of **154**

Note: Testing was conducted in tekVizion labs





CUC Version



## CUC Telephony Integration with Cisco UCM

**Navigation:** Telephony Integrations → Phone system

Set Phone System Name\* = cucmUM1. This is used for this example


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

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

**Cisco Unity**

- Users
- Class of Service
- Templates
- Contacts
- Distribution Lists
- Call Management
- Message Storage
- Networking
- Unified Messaging
- Video
- Dial Plan
- System Settings
- Telephony Integrations
  - Phone System**
  - Port Group
  - Port
  - Speech Connect Port
  - Trunk
- Security
- Tools

**Search Phone Systems**

Search Phone Systems

Related Links **Check Telephony Configuration** **Go**

Phone System Refresh Help

**Status**

Found 2 Phone System(s)

**Phone Systems (1 - 2 of 2)**

Rows per Page 25


Find Phone Systems where Display Name begins with  **Find**

<input type="checkbox"/>	Display Name ^	Port Count
<input type="checkbox"/>	<a href="#">CiscoUM1</a>	2
<input type="checkbox"/>	<a href="#">SIP_VM</a>	2

Delete Selected Add New

## CUC Port Group

**Navigation:** Telephony Integration → Port Group


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

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

**Cisco Unity**

- Users
- Class of Service
- Templates
- Contacts
- Distribution Lists
- Call Management
- Message Storage
- Networking
- Unified Messaging
- Video
- Dial Plan
- System Settings
- Telephony Integrations
  - Phone System
  - Port Group**
  - Port
  - Speech Connect Port
  - Trunk
- Security
- Tools

**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 Find

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

Delete Selected Add New

CUC Port Group(continued...)



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

Check Enable Message waiting indicators.

Set MWI on Extension = 2298. This is used in this example.

Set MWI off Extension= 299. This is used in this example.

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

Navigation: Cisco Unity Connection Admin | administrator | Search Documentation

**Port Group Basics (CiscoUM1-1)**  
Search Port Groups | Port Group Basics (CiscoUM1-1)  
Related Links: Add Ports | Go

Port Group | Edit | Refresh | Help

Save | Delete | Previous | Next

**Port Group**

Display Name\* | CiscoUM1-1

Integration Method | SCCP (Skinny)

Device Name Prefix\* | CiscoUM1-VI

Reset Status | Reset Not Required | Reset

**Message Waiting Indicator Settings**

☒ Enable Message Waiting Indicators

MWI On Extension | 2298

MWI Off Extension | 2299

Delay between Requests | 0 | milliseconds

Maximum Concurrent Requests | 0

Retries After Successful Attempt | 0

Retry Interval After Successful Attempt | 5 | milliseconds

Save | Delete | Previous | Next

## CUC Port Settings


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

Navigation **Cisco Unity Connection Administration**  
**administrator** | [Search Documentation](#)

**▼ Cisco Unity**

- ⊕ Users
- ⊕ Class of Service
- ⊕ Templates
- ⊕ Contacts
- ⊕ Distribution Lists
- ⊕ Call Management
- ⊕ Message Storage
- ⊕ Networking
- ⊕ Unified Messaging
- ⊕ Video
- ⊕ Dial Plan
- ⊕ System Settings
- ⊖ Telephony Integrations
  - Phone System
  - Port Group
  - Port**
  - Speech Connect Port
  - Trunk
- ⊕ Security
- ⊕ Tools

**Search Ports**
Related Links [Check Telephony Configuration](#) [Go](#)

Port Refresh Help

**Status**  
*i* 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
<input type="checkbox"/>	<a href="#">CiscoUM1-1-001</a>	<a href="#">CiscoUM1</a>		clus32unity	X	X	X
<input type="checkbox"/>	<a href="#">CiscoUM1-1-002</a>	<a href="#">CiscoUM1</a>		clus32unity	X	X	X
<input type="checkbox"/>	<a href="#">SIP_VM-1-001</a>	<a href="#">SIP_VM</a>		clus32unity	X	X	X
<input type="checkbox"/>	<a href="#">SIP_VM-1-002</a>	<a href="#">SIP_VM</a>		clus32unity	X	X	X

## CUC Sample User Basic Settings

**Navigation:** Cisco Unity connection → Users → Users



Set Alias = 4084. This is one of the extension used for this testing.  
Set Extension = 4084. This is used for this example.

The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar contains a navigation menu with options like Users, Class of Service, Templates, Contacts, Distribution Lists, Call Management, Message Storage, Networking, Unified Messaging, Video, Dial Plan, System Settings, Telephony Integrations, and Tools. The main content area is titled "Edit User Basics (4084)". It includes a search bar, a "Related Links" section with a "Bulk Edit By CSV" dropdown, and a "Go" button. Below this is a table with columns "User", "Edit", "Refresh", and "Help". The "User" column contains a "Save" button, and the "Edit" column contains "Delete", "Previous", and "Next" buttons. The form fields are organized into sections: "Name" (Alias\*, First Name, Last Name, Display Name, SMTP Address, Initials, Title, Employee ID), "LDAP Integration Status" (Integrate with LDAP Directory, Do Not Integrate with LDAP Directory), and "Phone" (Extension\*, Cross-Server Transfer Extension or URI). The "Alias\*" and "Extension\*" fields are highlighted with red boxes and contain the value "4084".

User	Edit	Refresh	Help
<a href="#">Save</a>	<a href="#">Delete</a> <a href="#">Previous</a> <a href="#">Next</a>		

**Name**

Alias*	4084
First Name	Cisco
Last Name	VM
Display Name	4084
SMTP Address	4084 @clus32unity.lab.tekvizion.com
Initials	
Title	
Employee ID	

**LDAP Integration Status**

☐ Integrate with LDAP Directory  
☒ Do Not Integrate with LDAP Directory

**Phone**

Extension*	4084
Cross-Server Transfer Extension or URI	

### CUC Sample User Basic Settings (Continued...)

Set Partition = clus32unity partition. This is used for this example.



Select Search Scope = clus32unity Search Scope.  
Select Phone System = cucmUM1.

**Cisco Unity**

- Users
  - Users
  - Import Users
  - Synch Users
- Class of Service
- Templates
- Contacts
- Distribution Lists
- Call Management
- Message Storage
- Networking
- Unified Messaging
- Video
- Dial Plan
- System Settings
- Telephony Integrations
- Tools

Outgoing Fax Number:

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

Partition: clus32unity Partition

Search Scope: clus32unity Search Space

Phone System: CiscoUM1

Class of Service: Voice Mail User COS

Active Schedule: Weekdays

☐ 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:

**Location**

Address:

Building:

City:

State:

Postal Code:

Country: United States

## CUC Sample User Basic Settings (Continued...)

<ul style="list-style-type: none"> <li>+ Distribution Lists</li> <li>+ Call Management</li> <li>+ Message Storage</li> <li>+ Networking</li> <li>+ Unified Messaging</li> <li>+ Video</li> <li>+ Dial Plan</li> <li>+ System Settings</li> <li>+ Telephony Integrations</li> <li>+ Tools</li> </ul>	<input checked="" type="checkbox"/> Use System Default Time Zone Time Zone <span>(GMT-06:00) America/Chicago</span> Language <input checked="" type="radio"/> Use System Default Language <input type="radio"/> <span>English(United States)</span> 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"/>





## Auto Attendant

**Navigation:** Call Management → System Call Handlers

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

Set Phone System = CUCM

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

Set Partition = Clus32unity Partition. This is used for this example.

The screenshot shows the Cisco Unity Connection Administration web interface. The left sidebar contains a navigation tree with 'Call Management' expanded, showing 'System Call Handlers' as the selected option. The main content area is titled 'Search Call Handlers' and displays a status message: 'Found 4 System Call Handler(s)'. Below this, there are search filters and a table of system call handlers. The table has columns for 'Display Name' and 'Extension'. The first row, 'Auto Attendant', is highlighted with a red box and has an extension of 2555. Other handlers listed are 'Goodbye', 'Opening Greeting', and 'Operator'. At the bottom of the table, there are buttons for 'Delete Selected', 'Add New', 'Bulk Edit', and 'Show Dependencies'.

Display Name	Extension
Auto Attendant	2555
Goodbye	
Opening Greeting	
Operator	0

## Auto Attendant (Continued...)

+

 Users

+

 Class of Service

+

 Templates

+

 Contacts

+

 Distribution Lists

-

 Call Management
 

System Call Handlers

Directory Handlers

Interview Handlers

Custom Recordings

+

 Call Routing

+

 Message Storage

+

 Networking

+

 Unified Messaging

+

 Video

+

 Dial Plan

+

 System Settings

+

 Telephony Integrations

+

 Tools

### Call Handler

Display Name\*

Auto\_Attendant

Creation Time

2015-12-22 00:38:39.136

Phone System

CiscoUM1

Active Schedule

All Hours

View

☒ Use System Default Time Zone

Time Zone

(GMT-06:00) America/Chicago

Language

☐ Use System Default Language
   
☒ Inherit Language from Caller
   
☐ English(United States)

Extension

2555

Partition

clus32unity Partition

Recorded Name

Play/Record

#### Search Scope

☐ Search Space
 

clus32unity Search Space

☒ Inherit Search Space from Call

Save

Delete

Previous

Next



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

CUP/IMP Version

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

Navigation: Cisco Unified CM IM

System ▼ Presence ▼ Messaging ▼ Application ▼ Bulk Administration ▼ Diagnostics ▼ Help ▼

**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, January 6, 2016 5:09:09 AM CST, to node 10.80.22.3, from 172.16.29.115 using HTTPS

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

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

For information about Cisco Unified CM IM and Presence please visit our [IM and Presence Documentation](#) web site.

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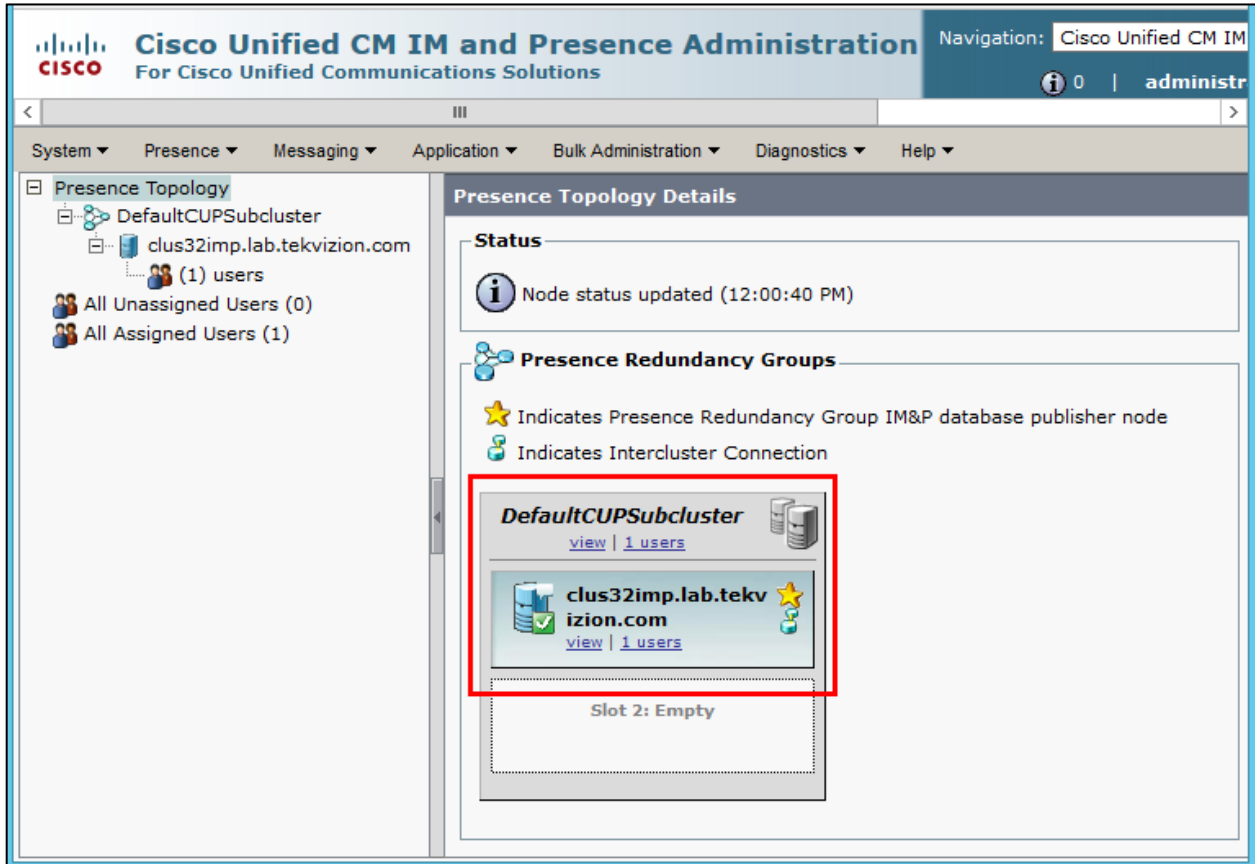
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Note: Testing was conducted in tekVizion labs

## Presence Topology

**Navigation:** System → Presence Topology



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

Navigation: Cisco Unified CM IM

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

**Presence Topology**

- DefaultCUPSubcluster
  - clus32imp.lab.tekvizion.com (1) users
    - All Unassigned Users (0)
    - All Assigned Users (1)

**Presence Topology Details**

**Status**

- Node status updated (12:00:40 PM)

**Presence Redundancy Groups**

- Indicates Presence Redundancy Group IM&P database publisher node
- Indicates Intercluster Connection

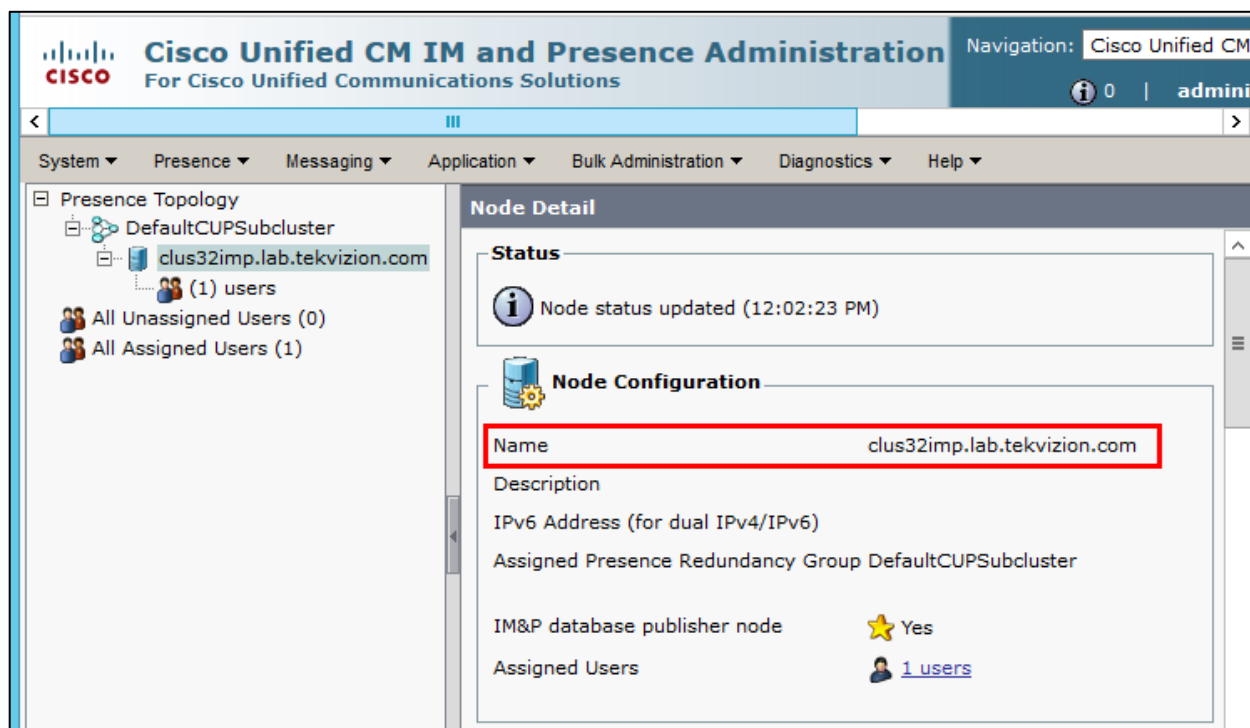
**DefaultCUPSubcluster**  
[view](#) | [1 users](#)

**clus32imp.lab.tekvizion.com**  
[view](#) | [1 users](#)

Slot 2: Empty

## Node Configuration

**Navigation:** System → Cluster Topology → Fully Qualified Domain Name



The screenshot displays the Cisco Unified CM IM and Presence Administration web interface. The left sidebar shows the navigation tree with 'Presence Topology' expanded, and 'clus32imp.lab.tekvizion.com' selected under 'DefaultCUPSubcluster'. The main content area is titled 'Node Detail' and shows the configuration for the selected node. The 'Name' field is highlighted with a red box and contains the value 'clus32imp.lab.tekvizion.com'. Other fields include 'Description', 'IPv6 Address (for dual IPv4/IPv6)', 'Assigned Presence Redundancy Group' (DefaultCUPSubcluster), 'IM&P database publisher node' (Yes), and 'Assigned Users' (1 users).

Node Detail	
<b>Status</b>	
Node status updated (12:02:23 PM)	
<b>Node Configuration</b>	
Name	clus32imp.lab.tekvizion.com
Description	
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 → clus32imp.lab.tekvizion.com → Users

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

Navigation: Cisco Unified CM IM and Presence Administration | 0 | administrator | Search

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

**Node User Assignment (clus32imp.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
jabber	cisco		jabber@lab.tekvizion.com	jabber@lab.tekvizion.com		clus32imp.lab.tekvizion.com	DefaultCUPSubcluster



## Presence gateway configuration

**Navigation:** Presence → Gateways

Set Presence Gateway Type \*= CUCM

Set Description \*= Cluster 32 9.1.2. This is used for this example.

Presence Gateway \*= clus23pubsub.lab.tekvizion.com

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

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

**Presence Gateway Configuration** Related Links: [Back To Find/List ▾](#) [Go](#)

Save Delete Add New

**Status**

Status: Ready

**Presence Gateway Settings (Cisco Unified Communications Manager)**

You can configure a Cisco Unified Communications Manager server as a presence gateway. The IM and Presence Service will then trigger the Cisco Unified Communications Manager to publish phone presence information (e.g. phone on/off hook status).

Presence Gateway Type\*

Description\*

Presence Gateway\*

Save Delete Add New



## 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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