

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. 10.5.2 and  
Cisco UBE v. 10.0.2 On an ISR G2 Router with SIP Interface  
MAR 2015



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## Introduction

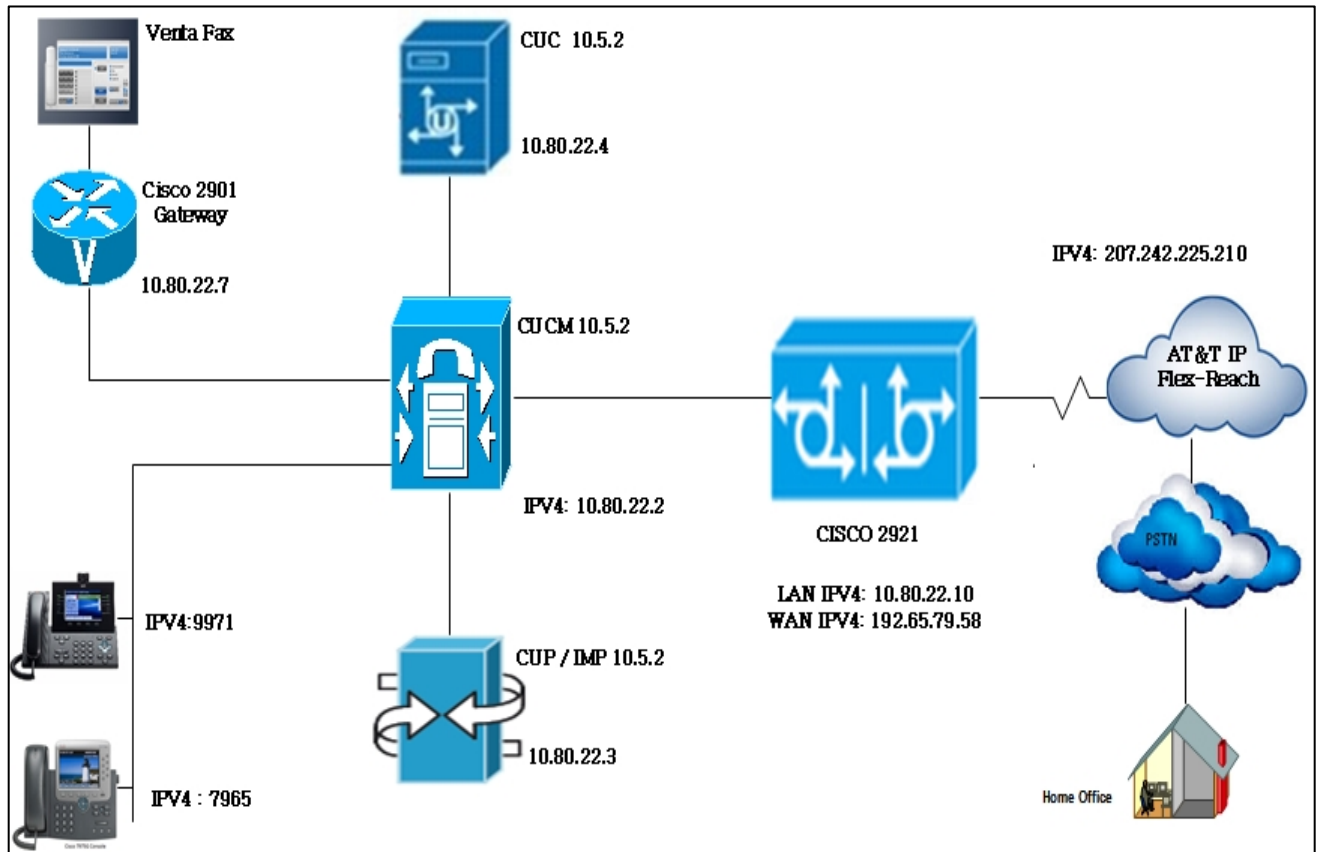
Service Providers today, such as AT&T, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. AT&T IP Flexible Reach is a service provider 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) 10.5.2, Cisco Unity Connection 10.5.2, Cisco Unified CM IM and Presence 10.5.2, Cisco Integrated Services Routers (ISR) Version 15.4(3) M1 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 7965 & Cisco 9971 phones
- Cisco integrated Service Router G2 - Cisco CISCO2921/K9 (revision 1.0) with 483328K/40960K bytes of memory
- Processor board ID FTX1746AJCC 3 Gigabit Ethernet interfaces, 1 terminal line, 1 Virtual Private Network (VPN) Module, DRAM configuration is 64 bits wide with parity enabled, 255K bytes of non-volatile configuration memory

## Software Requirements

- Cisco UCM: System version: 10.5.2.10000-5, including Business Edition 6000 and Business Edition 7000.
- ISR: C2900 Software (C2900-UNIVERSALK9-M), Version 15.4(3) M1, RELEASE SOFTWARE (fc1). System image file is "flash0:c2900-universalk9-mz.SPA.154-3.M1.bin".
- Cisco Unity Connection version: System version: 10.5.2.10000-5
- Cisco Unified CM IM and Presence: System version: 10.5.2.10000-9
- Cisco Jabber client version: V-9.1.3 Build 13181
- VentaFax client version: 7.3.233.582 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 (See Caveat section for details)
- 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
- RTCP

### 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

### Fax

- The maximum fax rate achieved using G711u (G3 or SG3) is only 14400 kbps.
- G711Passthrough test is achieved using "fax protocol pass-through g711ulaw".
- Fax protocol T38 has been tested.

### 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

topcube#**sh version**

Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9-M), Version 15.4(3)M1, RELEASE SOFTWARE (fc1)

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

Copyright (c) 1986-2014 by Cisco Systems, Inc.

Compiled Sat 25-Oct-14 03:34 by prod\_rel\_team

ROM: System Bootstrap, Version 15.0(1r)M16, RELEASE SOFTWARE (fc1)

topcube uptime is 3 days, 8 hours, 0 minutes

System returned to ROM by reload at 23:08:42 UTC Thu Mar 12 2015

System image file is "flash0:c2900-universalk9-mz.SPA.154-3.M1.bin"

Last reload type: Normal Reload

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).

Cisco CISCO2921/K9 (revision 1.0) with 483328K/40960K bytes of memory.

Processor board ID FTX1746AJCC

3 Gigabit Ethernet interfaces

1 terminal line

1 Virtual Private Network (VPN) Module

DRAM configuration is 64 bits wide with parity enabled.

255K bytes of non-volatile configuration memory.

250880K bytes of ATA System CompactFlash 0 (Read/Write)

License Info:

License UDI:

```
-----  
Device# PID SN  
-----  
*1 CISCO2921/K9 FTX1746AJCC
```



# Technology Package License Information for Module:'c2900'

Technology	Technology-package		Technology-package
Current	Type	Next reboot	
-----			
ipbase	ipbasek9	Permanent	ipbasek9
security	securityk9	Permanent	securityk9
uc	uck9	Permanent	uck9
data	None	None	None
NtwkEss	None	None	None
CollabPro	None	None	None

Configuration register is 0x2102

topcube#**sh running-configuration**

Building configuration...

Current configuration : 10923 bytes

!

! Last configuration change at 07:56:59 UTC Fri Mar 13 2015 by cisco

!

version 15.4



```
service tcp-keepalives-in
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname topcube
!
boot-start-marker
boot-end-marker
!
aqm-register-fnf
!
logging queue-limit 1000000000
logging buffered 10000000
logging rate-limit 10000
enable secret 4 Pe0NhiWw5IXZpE.k5VhTSCoGPcuVeRyrer9kEPz20Z6
!
no aaa new-model
!
!
!
!
!
!
!
!
```



!

!

!

!

!

!

no ip domain lookup

ip cef

no ipv6 cef

!

multilink bundle-name authenticated

!

!

!

!

!

!

cts logging verbose

!

!

voice-card 0

dspfarm

dsp services dspfarm

!

!

!



voice service voip  
no ip address trusted authenticate  
address-hiding<sup>1</sup>  
mode border-element<sup>2</sup>  
allow-connections sip to sip<sup>3</sup>  
redirect ip2ip  
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none  
sip  
header-passing  
error-passthru<sup>4</sup>  
asserted-id pai<sup>5</sup>  
no update-callerid  
early-offer forced<sup>6</sup>  
midcall-signaling passthru<sup>7</sup>  
privacy-policy passthru<sup>8</sup>  
g729 annexb-all

---

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

<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.



!

voice class codec 1<sup>9</sup>

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>"<sup>10</sup>

request INVITE sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"<sup>11</sup>

response ANY sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"

request INVITE sdp-header Audio-Attribute add "a=ptime:30"<sup>12</sup>

---

<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.



!

!

!

!

!

voice translation-rule 1<sup>13</sup>

rule 1 /^.\*\{(40..\)\} /732320\1/

!

voice translation-rule 2

rule 2 /^+\{(1\)\}(7.....\)\} / \2/

!

!

voice translation-profile NPA

translate calling 1

!

voice translation-profile test+1

translate called 2

!

!

!

license udi pid CISCO2921/K9 sn FTX1746AJCC

hw-module pvdm 0/0

!

---

<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.

<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.

```
!
!  
username cisco privilege 15 password 0 cisco  
!  
redundancy  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
interface Embedded-Service-Engine0/0  
no ip address  
shutdown  
!  
  
interface GigabitEthernet0/014
```

**14** WAN interface to AT&T



```
ip address 192.65.79.58 255.255.255.224
```

```
duplex auto
```

```
speed auto
```

```
!
```

```
interface GigabitEthernet0/115
```

```
ip address 10.80.22.10 255.255.255.016
```

```
duplex auto
```

```
speed auto
```

```
!
```

```
interface GigabitEthernet0/2
```

```
no ip address
```

```
shutdown
```

```
duplex auto
```

```
speed auto
```

```
!
```

```
ip forward-protocol nd
```

```
!
```

```
no ip http server
```

```
no ip http secure-server
```

```
!
```

```
ip route 0.0.0.0 0.0.0.0 192.65.79.33
```

```
ip route 10.0.0.0 255.0.0.0 10.80.22.1
```

```
ip route 10.64.0.0 255.255.0.0 10.80.22.1
```

```
ip route 172.16.0.0 255.255.0.0 10.80.22.1
```

---

<sup>15</sup> LAN interface to Cisco UCM

<sup>16</sup> Cisco UBE LAN interface IPv4 Address



```
!  
!  
!  
!  
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 100 voip17  
description "Outgoing To AT&T"-AT&T facing side  
destination-pattern 73236.....  
no modem passthrough
```

---

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



```
session protocol sipv218
session target ipv4:207.242.225.210
voice-class codec 119
voice-class sip asymmetric payload full20
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru21
voice-class sip profiles 122
voice-class sip bind control source-interface GigabitEthernet0/023
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte24
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none25
no vad
!
dial-peer voice 200 voip
description "Outgoing To AT&T .IP PBX facing side"
no modem passthrough
```

---

<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.

<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



```
session protocol sipv2
incoming called-number [27]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 GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
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 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
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 400 voip
description " Int'l calls to AT&T - IP-PBX facing side "
no modem passthrough
session protocol sipv2
incoming called-number 011T
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
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
voice-class sip bind media source-interface GigabitEthernet0/0
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 GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
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 800 voip
description " Incoming AT&T to IP-PBX . AT&T facing side "
translation-profile incoming test+1
huntstop
no modem passthrough
session protocol sipv2
incoming called-number +1[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
voice-class sip bind media source-interface GigabitEthernet0/0
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
voice-class sip bind media source-interface GigabitEthernet0/0
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 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
voice-class sip bind media source-interface GigabitEthernet0/0
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
voice-class sip bind media source-interface GigabitEthernet0/0
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 7323204292
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
voice-class sip bind media source-interface GigabitEthernet0/0
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 GigabitEthernet0/1

voice-class sip bind media source-interface GigabitEthernet0/1

dtmf-relay rtp-nte

fax-relay sg3-to-g3

fax rate 14400

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

no vad

!

!

gateway

media-inactivity-criteria all

timer receive-rtcp 5



```
timer receive-rtp 86400
!
sip-ua
no remote-party-id
retry invite 2
timers expires 1800000
connection-reuse
protocol mode ipv4
!
!
!
gatekeeper
shutdown
!
!
!
line con 0
logging synchronous
line aux 0
line 2
session-timeout 90
no activation-character
no exec
transport preferred none
transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
stopbits 1
```



```
line vty 0 4
session-timeout 90
exec-timeout 960 0
logging synchronous
login local
transport input all
!
scheduler allocate 20000 1000
!
end
```

## Cisco UCM Configuration

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



## Cisco UCM Version

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

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

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

### Cisco Unified CM Administration

**System version: 10.5.2.10000-5**

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

**Last Successful Backup: 13 day(s) ago**

User administrator last logged in to this cluster on Monday, March 16, 2015 1:49:44 AM CDT, to node 10.80.22.2, from 172.16.31.153 using HTTPS

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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 10.5.2 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

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Note: Testing was conducted in tekVizion labs



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)

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 navigation menu with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, and User Management. The main content area is titled "Audio Codec Preference List Configuration". Below this title, there are buttons for Save, Delete, Copy, and Add New. A red box highlights the "Name\*" and "Description\*" fields, both containing the text "G729 Preferred Codec List". Below these fields is a list of codecs under the heading "Codecs in List\*". The list includes various codecs such as G.729a 8k, G.729b 8k, G.729ab 8k, G.729 8k, G.711 U-Law 64k, G.711 A-Law 64k, G.711 U-Law 56k, G.711 A-Law 56k, AMR-WB (7k-24k), AMR (5k-13k), MP4A-LATM 128k, AAC-LD (MP4A Generic), MP4A-LATM 64k, MP4A-LATM 56k, L16 256k, MP4A-LATM 48k, ISAC 32k, MP4A-LATM 32k, MP4A-LATM 24k, G.722.1 32k, G.722 64k, G.722.1 24k, G.722 56k, and G.722 48k. A scroll bar is visible on the right side of the list. Below the list, there are buttons for Save, Delete, Copy, and Add New.

## Cisco UCM Region Configuration

**Navigation Path:** System → Region Information → Region



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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 ▾

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

Save Delete Reset Apply Config Add New

**Region Information**  
Name\*

**Region Relationships**

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

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

**Modify Relationship to other Regions**

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

Save Delete Reset Apply Config Add New

## Device Pool Configuration

**Navigation Path:** System → Device Pool

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Note: Testing was conducted in tekVizion labs



“G729\_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.

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

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

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

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

Save Delete Copy Reset Apply Config Add New

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

**Device Pool Settings**

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

**Roaming Sensitive Settings**

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

Single Button Barge\* Default

Join Across Lines\* Default

Physical Location < None >

Device Mobility Group < None >

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

**Local Route Group Settings**

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

Device Pool Configuration (continued...)

**Device Mobility Related Information\*\*\*\***

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

**Geolocation Configuration**

Geolocation < None >  
Geolocation Filter < None >

**Call Routing Information**

**Incoming Calling Party Settings**

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

Clear Prefix Settings
Default Prefix Settings

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

**Incoming Called Party Settings**

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

Clear Prefix Settings
Default Prefix Settings

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



## Device Pool Configuration (continued...)

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



## Annunciator Configuration

**Navigation:** Media Resource → Annunciator

Set Name\* = ANN\_2.

Set Description = ANN\_clus32pubsub. This is used for this example

Set Device Pool\* = G729\_pool.

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

Navigation **Cisco Unified CM Administration** | Search Document

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

**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

IP 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.

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**administrator** | Search Docu

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

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

**Conference Bridge Information**

Conference Bridge : CFB\_2 (CFB\_clus32pubsub)  
Registration Registered with Cisco Unified Communications Manager clus32pubsub  
IP Address 10.80.22.2

**Software Conference Bridge Info**

Conference Bridge Type\* Cisco Conference Bridge Software  
Host Server clus32pubsub

⚠ Device is not trusted

Conference Bridge Name\* CFB\_2  
Description CFB\_clus32pubsub  
Device Pool\* G729 Pool ▾  
Common Device Configuration < None > ▾  
Location\* Hub\_None ▾  
Use Trusted Relay Point\* Default ▾



## Media Termination Point Configuration

**Navigation:** Media Resource → Media Termination Point

Set Media Termination Point Name\* = MTP\_2

Set Description = MTP\_clus32pubsub. This is used for this example

Set Device pool\* = G729 Pool

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, and User Management. The main content area is titled "Media Termination Point Configuration" and includes a "Related Links" section with a "Back To Find/List" button. Below this is a "Status" section showing "Status: Ready". The "Media Termination Point Information" section contains the following fields:

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

There is also a checkbox for "Trusted Relay Point" which is currently unchecked. At the bottom of the form are buttons for "Save", "Reset", and "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.


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

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

**Music On Hold (MOH) Server Configuration**    Related Links: [Back To Find/List](#)    [Go](#)

 Save     Reset     Apply Config

**Device Information**

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

Host Server\* clus32pubsub  
Music On Hold Server Name\* MOH\_2  
Description MOH\_clus32pubsub  
Device Pool\* G729 Pool  
Location\* Hub\_None  
Maximum Half Duplex Streams\* 250  
Maximum Multi-cast Connections\* 250000  
Fixed Audio Source Device  
Use Trusted Relay Point\* Off  
Run Flag\* Yes

**Multi-cast Audio Source Information**

☐ Enable Multi-cast Audio Sources on this MOH Server  
Base Multi-cast IP Address\* 0.0.0.0  
Base Multi-cast Port Number\* 0 (Even numbers only)  
Increment Multi-cast on\* ☒ Port Number ☐ IP Address

**Selected Multi-cast Audio Sources**

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

[Save](#)    [Reset](#)    [Apply Config](#)

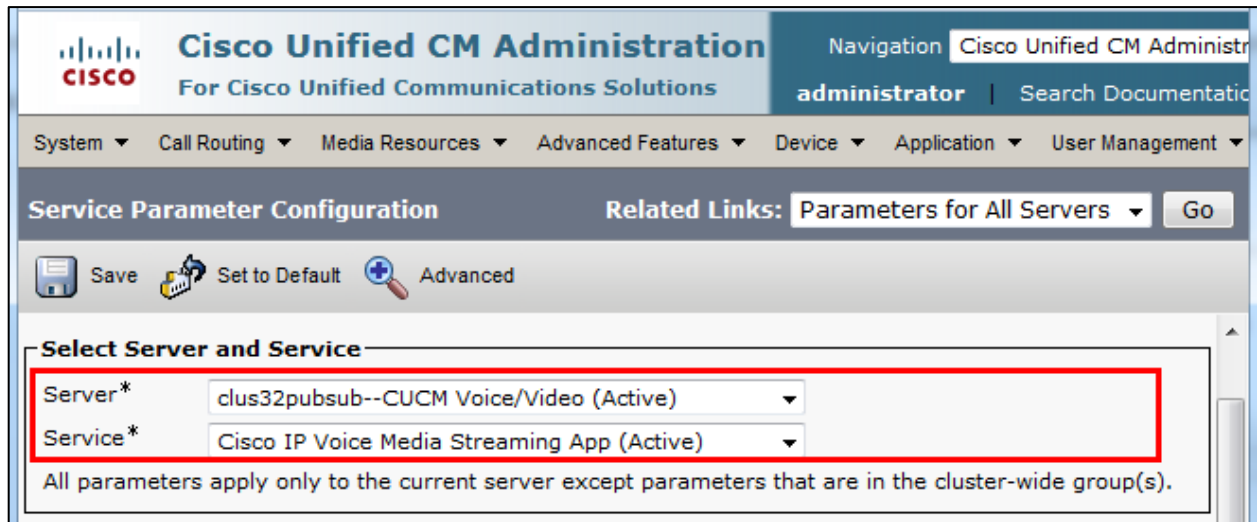
## Music on Hold Service (IP Voice Media Streaming App) Parameter Settings

**Navigation:** System → Service Parameter

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



Select Server\* = clus32pubsub--CUCM Voice/Video (Active). This is used in this example.  
Select Service\* = Cisco IP Voice Media Streaming App (Active).



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Navigation Cisco Unified CM Administration administrator | Search Documentation

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

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

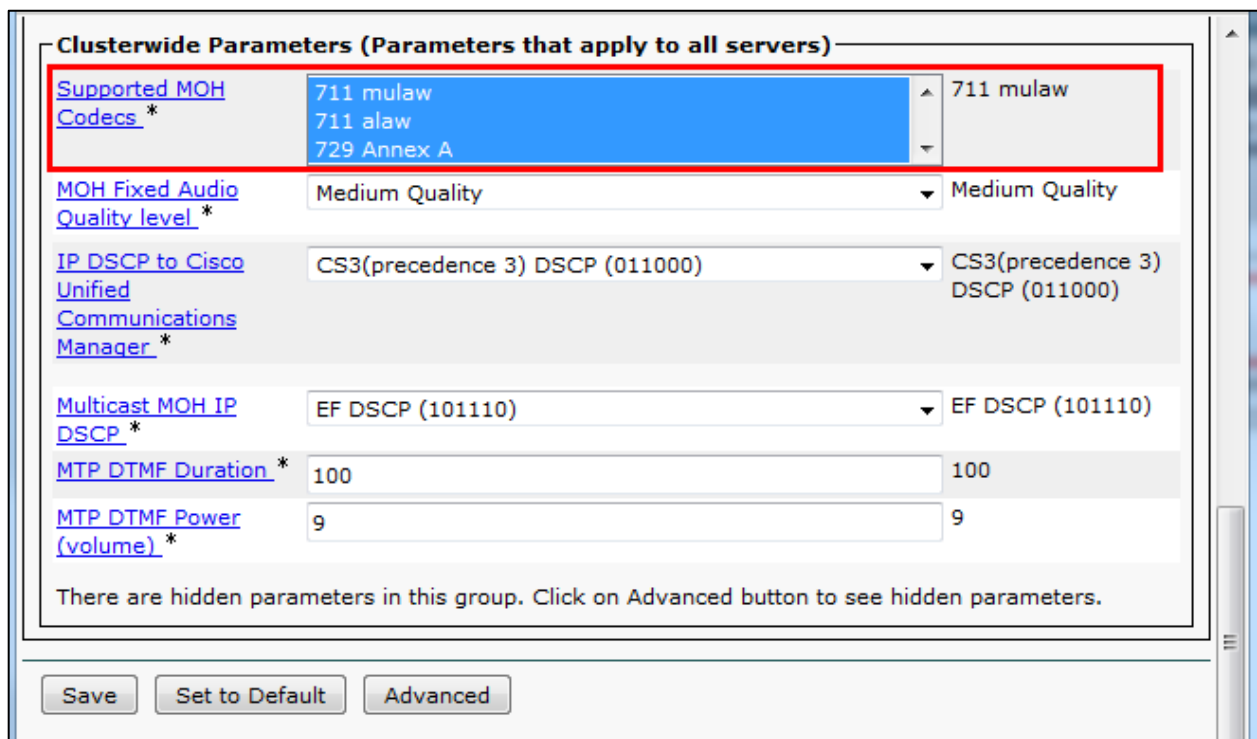
Save Set to Default Advanced

**Select Server and Service**

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

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

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



**Clusterwide Parameters (Parameters that apply to all servers)**

Supported MOH Codecs\* 711 mulaw 711 mulaw  
711 alaw  
729 Annex A

MOH Fixed Audio Quality level\* Medium Quality Medium Quality

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

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

MTP DTMF Duration\* 100 100

MTP DTMF Power (volume)\* 9 9

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

Save Set to Default Advanced

## Music on Hold Service (Duplex Streaming) Parameter Settings

**Navigation:** System → Service Parameter

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



Select Service\* = Cisco CallManager (Active).

Select Duplex Streaming Enabled \* = True

The screenshot shows the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and a navigation menu with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, and User Management. The user is logged in as "administrator".

The main section is titled "Service Parameter Configuration". It includes a "Related Links" section with a dropdown menu set to "Parameters for All Servers" and a "Go" button. Below this are icons for "Save", "Set to Default", and "Advanced".

The "Select Server and Service" section is highlighted with a red box. It contains two dropdown menus: "Server\*" set to "clus32pubsub--CUCM Voice/Video (Active)" and "Service\*" set to "Cisco CallManager (Active)". A note below these dropdowns states: "All parameters apply only to the current server except parameters that are in the cluster-wide group(s)."

The "Clusterwide Parameters (Service)" section is shown below. It contains a list of parameters with input fields and checkboxes. The "Duplex Streaming Enabled \*" parameter is highlighted with a red box and is set to "True". Other parameters include "Default Network Hold MOH Audio Source ID", "Default User Hold MOH Audio Source ID", "Media Exchange Interface Capability Timer", "Send Multicast MOH in H.245 OLC Message", "Media Exchange Timer", "Media Exchange Stop Streaming Timer", "Open Video Channel Response Timer for SIP Interop", "Port Received Timer After Call Connection", "Media Resource Allocation Timer", and "MTP and Transcoder Resource Throttling Percentage".

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

## Media Resource Group Configuration

**Navigation Path:** Media Resources → Media Resources group



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

Set Name\*= MRG\_MTP - This is used for this example.

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

Set all Resources in the selected Media Resources Box.

**Cisco Unified CM Administration**  
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Navigation: Cisco Unified CM Administration | administrator | Search Documents

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

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

Save Delete Copy Add New

**Media Resource Group Status**  
Media Resource Group: MRG\_MTP (used by 17 devices)

**Media Resource Group Information**  
Name\* MRG\_MTP  
Description MRG\_MTP

**Devices for this Group**  
Available Media Resources\*\*  
Selected Media Resources\*  
ANN\_2 (ANN)  
CFB\_2 (CFB)  
MOH\_2 (MOH)  
MTP\_2 (MTP)

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

Save Delete Copy Add New

## Media Resource Group List Configuration

**Navigation Path:** Media Resources → Media Resource Group List

Set Name = MRGL\_MTP.



Set selected Media Resource Groups = MRG\_MTP.

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 navigation menu with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, and User Management. The user is logged in as "administrator".

The main section is titled "Media Resource Group List Configuration". It includes a "Related Links" section with a "Back To Find/List" button and a "Go" button. Below this is a toolbar with icons for Save, Delete, Copy, and Add New.


The configuration is for the "Media Resource Group List Status" and "Media Resource Group List Information". The "Name\*" field is set to "MRGL\_MTP".

The "Media Resource Groups for this List" section shows two lists: "Available Media Resource Groups" and "Selected Media Resource Groups". The "Selected Media Resource Groups" list contains "MRG\_MTP".

At the bottom, there are buttons for "Save", "Delete", "Copy", and "Add New".

## UC Service Configuration

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



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 For Cisco Unified Communications Solutions

Navigation **Cisco Unified CM Administration** Go  
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[System](#) ▾ [Call Routing](#) ▾ [Media Resources](#) ▾ [Advanced Features](#) ▾ [Device](#) ▾ [Application](#) ▾ [User Management](#) ▾ [Bulk Administration](#) ▾

### Find and List UC Services

+ Add New ⌘ Select All ⌘ Clear All ✖ Delete Selected

**Status**  
 2 records found

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

Find UC Service where Name ▾ begins with ▾  Find Clear Filter + -

<input type="checkbox"/>	Name ^	UC Service Type	Product Type	Host/IP Address	Port	Protocol
<input type="checkbox"/>	<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 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" dropdown menu. Below the navigation bar is a secondary menu with links like "System", "Call Routing", "Media Resources", etc. The main content area is titled "UC Service Configuration". It features a toolbar with icons for Save, Delete, Copy, Reset, Apply Config, and Add New. Below the toolbar, the "Status" section shows "Status: Ready". The "Add a UC Service" section is highlighted with a red box and contains the following fields: "UC Service Type" (CTI), "Product Type" (CTI), "Name\*" (CTI\_SRV), "Description" (CTI for Jabber Clients), "Host Name/IP Address\*" (10.80.22.2), "Port" (2748), and "Protocol" (TCP). At the bottom of the form 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)

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 the subtitle 'For Cisco Unified Communications Solutions'. The user is logged in as 'administrator'. The main menu shows various categories like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Bulk Administration. The 'UC Service Configuration' section is active, showing a 'Status: Ready' message. Below this, the 'Add a UC Service' form is visible, with a red box highlighting the following fields:

- 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

At the bottom of the form, there are buttons for Save, Delete, Copy, Reset, Apply Config, and Add New.

## Service Profile Configuration

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

Set Name\* = Jabber\_SVC\_Profile. This is used in this example.



Set Description = Jabber Service Profile. This is used in this example.  
Check - Make this the default service profile for the system.

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

Navigation Cisco Unified CM Administration administrator | Search Documents

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

Secondary

Tertiary

Server Certificate Verification

[Credentials source for web conference service](#)\*

**Directory Profile**

Primary

Secondary

Tertiary

☒ [Use UDS for Contact Resolution](#)

☒ [Use Logged On User Credential](#)

[Username](#)

[Password](#)

[Search Base 1](#)

[Search Base 2](#)

[Search Base 3](#)

☒ [Recursive Search on All Search Bases](#)

[Search Timeout \(seconds\)\\*](#)

[Base Filter \(Only used for Advance Directory\)](#)

[Predictive Search Filter \(Only used for Advance Directory\)](#)

**IM and Presence Profile**

Primary

Secondary

Tertiary

**CTI Profile**

Primary

Secondary

Tertiary



## End User Configuration

**Navigation:** User Management → End User

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

Set Password = Password for profile.

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

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

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

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

### Find and List Users

+ Add New    Select All    Clear All    Delete Selected

**Status**  
1 records found

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


Find User where First name ▾ begins with ▾    Find    Clear Filter    +    -

<input type="checkbox"/>	User ID ^	First Name	Last Name	Department	Directory URI	User Status
<input type="checkbox"/>	jabber1		cisco		jabber1@lab.tekvizion.com	Enabled Local User

Add New    Select All    Clear All    Delete Selected






End User Configuration (continued...)


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

Navigation Cisco Unified CM Administra  
 administrator | Search Documentation

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

**End User Configuration**
Related Links: Back to Find List Users ▾ Go

 Save
  Delete
  Add New

**User Information**

User Status	Enabled Local User	
User ID*	jabber1	
Password	.....	Edit Credential
Confirm Password	.....	
Self-Service User ID		
PIN	.....	Edit Credential
Confirm PIN	.....	
Last name*	cisco	
Middle name		
First name		
Title		
Directory URI	jabber1@lab.tekvizion.com	
Telephone Number		
Home Number		
Mobile Number		
Pager Number		
Mail ID		

Manager User ID

Department

User Locale
 < None > ▾

Associated PC

Digest Credentials
 .....

Confirm Digest Credentials
 .....

User Profile
 Use System Default( "Standard (Factory Default) U ▾ [View Details](#)

## End User Configuration(continued... )

**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](#)

**Device Information**

Controlled Devices

CSFuser1  
SEPC07BBCA1B811

Device Association

Line Appearance Association for Presence

Available Profiles

CTI Controlled Device Profiles

**Extension Mobility**

Available Profiles

Controlled Profiles

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 >

## End User Configuration(continued... )

**Mobility Information**

☐ Enable Mobility  
☐ Enable Mobile Voice Access  
Maximum Wait Time for Desk Pickup\* 10000  
Remote Destination Limit\* 4  
Remote Destination Profiles  
[View Details](#)

**Multilevel Precedence and Preemption Authorization**

MLPP User Identification Number  
MLPP Password  
Confirm MLPP Password  
MLPP Precedence Authorization Level Default

**CAPF Information**

Associated CAPF Profiles  
[View Details](#)

**Permissions Information**

Groups
Standard Audit Users  
Standard CAR Admin Users  
Standard CCM Admin Users  
Standard CCM End Users  
Standard CCM Gateway Administration  
[View Details](#)

Roles
Standard AXL API Access  
Standard Admin Rep Tool Admin  
Standard Audit Log Administration  
Standard CCM Admin Users  
Standard CCM End Users  
[View Details](#)

Add to Access Control Group  
Remove from Access Control Group

Save Delete Add New



## Cisco IP Phone 7965 SCCP Configuration

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

Set Description = Cisco7965\_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 7965 SCCP. This is used in this example.

Set Softkey Template = Standard User. This is used in this example.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a search bar. Below the navigation bar, there are tabs for "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The "Device" tab is selected, and the "Phone Configuration" section is active. The "Related Links" section shows "Back To Find/List".

The main configuration area is divided into two columns. The left column, titled "Association", contains a list of 15 items, including "Line [1] - 4085 (no partition)", "Line [2] - Add a new DN", "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", "Add a new BLF Directed Call Park", "CallBack", "Call Park", "Call Pickup", "Conference List", and "Conference". The right column, titled "Phone Type", contains the following sections:

- Phone Type**
  - Product Type: Cisco 7965
  - Device Protocol: SCCP
- Real-time Device Status**
  - Registration: Registered with Cisco Unified Communications Manager clus32pubsub
  - IPv4 Address: 172.16.31.168
  - Active Load ID: SCCP45.9-3-1SR4-1S
  - Download Status: None
- Device Information**
  - ☒ Device is Active
  - ☒ Device is trusted
  - MAC Address\*: 44ADD9D56F39
  - Description: Cisco7965-Phone
  - Device Pool\*: G729 Pool [View Details](#)
  - Common Device Configuration: ATT\_Phones [View Details](#)
  - Phone Button Template\*: Standard 7965 SCCP
  - Softkey Template: Standard User
  - Common Phone Profile\*: Standard Common Phone Profile [View Details](#)



## Cisco IP Phone 7965 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.

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

35	None	Services Provisioning *	Default
		Phone Load Name	
		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	

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

<input type="checkbox"/> Remote Device <input type="checkbox"/> Protected Device**** <input type="checkbox"/> Hot line Device***** <input type="checkbox"/> Require off-premise location
<b>Number Presentation Transformation</b> <hr/> <b>Caller ID For Calls From This Phone</b> Calling Party Transformation <input type="text" value=" &lt; None &gt;"/> CSS <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 <input type="text" value=" &lt; None &gt;"/> CSS <input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

<b>Protocol Specific Information</b> <hr/> Packet Capture Mode* <input type="text" value="None"/> Packet Capture Duration <input type="text" value="0"/> BLF Presence Group* <input type="text" value="Standard Presence group"/> Device Security Profile* <input type="text" value="Cisco 7965 - Standard SCCP Non-Secure Profile"/> SUBSCRIBE Calling Search Space <input type="text" value=" &lt; None &gt;"/> <input type="checkbox"/> Unattended Port <input type="checkbox"/> Require DTMF Reception <input type="checkbox"/> RFC2833 Disabled
--

<b>Certification Authority Proxy Function (CAPF) Information</b> <hr/> Certificate Operation* <input type="text" value="No Pending Operation"/> Authentication Mode* <input type="text" value="By Null String"/> Authentication String <input type="text"/> <input type="button" value="Generate String"/> Key Size (Bits)* <input type="text" value="2048"/> Operation Completes By <input type="text" value="2015"/> <input type="text" value="3"/> <input type="text" value="27"/> <input type="text" value="12"/> (YYYY:MM:DD:HH) Certificate Operation Status: None Note: Security Profile Contains Addition CAPF Settings.
<b>Expansion Module Information</b> <hr/> Module 1 <input type="text" value=" &lt; None &gt;"/> Module 1 Load Name <input type="text"/> Module 2 <input type="text" value=" &lt; None &gt;"/> Module 2 Load Name <input type="text"/>



## Cisco IP Phone 7965 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	<input type="text"/>
Secure Shell Password	<input type="text"/>

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

Product Specific Configuration Layout		Override Common Settings
	Parameter Value	
<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"/>
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="checkbox"/>
<input type="checkbox"/> Allow EnergyWise Overrides		<input type="checkbox"/>
Span to PC Port*	Disabled	
Logging Display*	PC Controlled	
Load Server		<input type="checkbox"/>



	Recording Tone*	Disabled	
	Recording Tone	100	
	Local Volume*		
	Recording Tone	50	
	Remote Volume*		
	Recording Tone		
	Duration		
Display On When	Disabled		<input type="checkbox"/>
Incoming Call*			
RTCP*	Disabled		<input type="checkbox"/>

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

"more" Soft Key Timer	5	
Auto Call Select*	Enabled	
Log Server		<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"/>
Link Layer Discovery Protocol - Media Endpoint	Enabled	<input type="checkbox"/>

Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*		
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	



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

Headset	Default	
Sidetone Level*		
Headset Send	Default	
Gain*		
HTTPS Server*	http and https Enabled	<input type="checkbox"/>
Handset/Headset	Enabled	
Monitor*		
Headset	Disabled	
Recording*		
Enbloc Dialing*	Enabled	
Switch Port	Disabled	<input type="checkbox"/>
Remote		
Configuration*		
PC Port Remote	Disabled	<input type="checkbox"/>
Configuration*		
Automatic Port	Disabled	<input type="checkbox"/>
Synchronization*		
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		<input type="checkbox"/>
Support Use		

Save

Delete

Copy

Reset

Apply Config

Add New




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

Set Directory Number\* = 4085. This is used in this example.

Set Description = 7323204085. This is used in this example.

Set Alerting Name = Cisco 7965 Phone. This is used in this example.






Set ASCII Alerting Name = Cisco 7965 Phone. This is used in this example.


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

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

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

**Directory Number Configuration**
Related Links: **Configure Device (SEP44ADD9D56F39)** **Go**

 Save
 Delete
 Reset
 Apply Config
 Add New

**Directory Number Information**

Directory Number\* 4085  
Route Partition < None >  
Description 7323204085  
Alerting Name Cisco 7965 Phone  
ASCII Alerting Name Cisco 7965 Phone  
☒ Allow Control of Device from CTI  
Associated Devices SEP44ADD9D56F39  

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 < None >  
Network Hold MOH Audio Source < None >  
Auto Answer\* Auto Answer Off  
☐ Reject Anonymous Calls

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

**Enterprise Alternate Number**

**+E.164 Alternate Number**

**Directory URIs**

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

**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 >

☒ 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		< None >



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

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

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

MLPP Alternate Party And Confidential Access Level Settings	
Target (Destination)	<input type="text"/>
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	<input type="text"/>
Confidential Access Mode	< None >
Confidential Access Level	< None >
Call Control Agent Profile	< None >
Line Settings for All Devices	
Hold Reversion Ring Duration (seconds)	<input type="text"/> Setting the Hold Reversion Ring Duration to zero will disable the feature
Hold Reversion Notification Interval (seconds)	<input type="text"/> Setting the Hold Reversion Notification Interval to zero will disable the feature
Party Entrance Tone*	Default



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

**Line 1 on Device SEP44ADD9D56F39**

Display (Caller ID)	Cisco 7965 Phone	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)	Cisco 7965 Phone	
Line Text Label	Cisco 7965 Phone	
External Phone Number Mask	7323204085	
Visual Message Waiting Indicator Policy*	Use System Policy	
Audible Message Waiting Indicator Policy*	Default	
Ring Setting (Phone Idle)*	Use System Default	
Ring Setting (Phone Active)	Use System Default	Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)	Use System Default	

Call Pickup Group Audio Alert Setting(Phone Active)	Use System Default
Recording Option *	Call Recording Disabled
Recording Profile	< None >
Recording Media Source*	Gateway Preferred
Monitoring Calling Search Space	< None >
<input checked="" type="checkbox"/> Log Missed Calls	

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

### Multiple Call/Call Waiting Settings on Device SEP44ADD9D56F39

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 SEP44ADD9D56F39

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

### Users Associated with Line

Associate End Users

Save

Delete

Reset

Apply Config

Add New



## Cisco IP Phone 9971 SIP Configuration

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

Set Description = Cisco 9971 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 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**  
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Navigation: Cisco Unified CM Administration Go

administrator | Search Documentation | About | Logout

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

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

Save ✖ Delete 📄 Copy 🔄 Reset 🔧 Apply Config ➕ Add New

**Association** Modify Button Items

- 1 📞 [Line \[1\] - 4084 \(no partition\)](#)
- 2 📞 [Line \[2\] - Add a new DN](#)
- 3 📞 [Add a new SD](#)
- 4 📞 [Add a new SD](#)
- 5 📞 [Add a new SD](#)
- 6 📞 [Add a new SD](#)
- Unassigned Associated Items -----
- 7 📞 [Add a new SD](#)
- 8 All Calls
- 9 📞 [Add a new BLF Directed Call Park](#)
- 10 Call Park
- 11 Call Pickup
- 12 CallBack
- 13 Group Call Pickup
- 14 Hunt Group Logout
- 15 📞 [Intercom \[1\] - Add a new Intercom](#)

**Phone Type**

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

**Real-time Device Status**

**Registration:** Registered with Cisco Unified Communications Manager clus32pubsub  
**IPv4 Address:** [172.16.31.108](#)  
**Active Load ID:** sip9971.9-4-2-13  
**Inactive Load ID:** sip9971.9-4-1-9  
**Download Status:** None

**Device Information**

☒ Device is Active  
☒ Device is trusted

MAC Address\* C07BBCA1B811

Description Cisco 9971 Phone

Device Pool\* G729 Pool [View Details](#)

Common Device Configuration ATT\_Phones [View Details](#)

Phone Button Template\* Standard 9971 SIP

Softkey Template Standard User

16 Malicious Call Identification

17 Meet Me Conference

18 Mobility

19 Other Pickup

20 Quality Reporting Tool

21 Redial

22 📞 [Add a new SURL](#)

23 📞 [Add a new BLF SD](#)

24 Answer Oldest

25 Do Not Disturb

26 Services

27 Record

28 Alerting Calls

Common Phone Profile\* Standard Common Phone Profile [View Details](#)

Calling Search Space meetme\_css

AAR Calling Search Space < None >

Media Resource Group List MRGL\_MTP

User Hold MOH Audio Source 1-SampleAudioSource

Network Hold MOH Audio Source 1-SampleAudioSource

Location\* Hub\_None

AAR Group < None >

User Locale < None >

Network Locale < None >

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

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EDCS# xxx Rev #

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Note: Testing was conducted in tekVizion labs

29	Queue Status	Built In Bridge*	Default
30	Privacy	Privacy*	Default
31	None	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*	jabber1
		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 Originating Codec\*
711ulaw

Device Security Profile\*
Cisco 9971 - Standard SIP Non-Secure Profile

Rerouting Calling Search Space
< None >

SUBSCRIBE Calling Search Space
< None >

SIP Profile\*
Standard SIP Profile w/Early Media Disabled
[View Details](#)

Digest User
< None >

☐ Media Termination Point Required

☐ Unattended Port

☐ Require DTMF Reception

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

Certificate Operation\*
No Pending Operation

Authentication Mode\*
By Null String

Authentication String

Generate String

Key Size (Bits)\*
2048

Operation Completes By
2015 3 27 12 (YYYY:MM:DD:HH)

Certificate Operation Status:
None

Note: Security Profile Contains Addition CAPF Settings.

**Expansion Module Information**

Module 1
< None >

Module 1 Load Name

Module 2
< None >

Module 2 Load Name

Module 3
< None >

Module 3 Load Name



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

**External Data Locations Information (Leave blank to use default)**

Information	
Directory	
Messages	
Services	
Authentication Server	
Proxy Server	
Idle	
Idle Timer (seconds)	
Secure Authentication URL	
Secure Directory URL	
Secure Idle URL	
Secure Information URL	
Secure Messages URL	
Secure Services URL	

**Extension Information**

☐ Enable Extension Mobility

Log Out Profile -- Use Current Device Settings --

Log in Time < None >

Log out Time < None >

**MLPP and Confidential Access Level Information**

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

**Do Not Disturb**

☐ Do Not Disturb

DND Option\* Use Common Phone Profile Setting


DND Incoming Call Alert < None >



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

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

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

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



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

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

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



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

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

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



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

Hide Video By Default*	Disabled	<input type="checkbox"/>
Background Image		<input type="checkbox"/>
Simplified New Call UI*	Disabled	<input type="checkbox"/>
Enable VXC VPN for MAC		
VXC VPN Option*	Dual Tunnel	<input type="checkbox"/>
VXC Challenge*	Challenge	<input type="checkbox"/>
VXC-M Servers		<input type="checkbox"/>
Revert to All Calls*	Disabled	<input type="checkbox"/>
RTCP for Video*	Enabled	<input type="checkbox"/>
Record Call Log from Shared Line*	Disabled	<input type="checkbox"/>
Show Remote Private Calls*	Disabled	
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		
Separate Audio and Video Mute*	Disabled	<input type="checkbox"/>
Softkey Control*	Feature Control Policy	<input type="checkbox"/>
Start Video Port		<input type="checkbox"/>
Stop Video Port		<input type="checkbox"/>
Lowest Alerting Line State Priority*	Disabled	<input type="checkbox"/>
TLS Resumption Timer*	3600	<input type="checkbox"/>
Audio EQ*	Default : Default	<input type="checkbox"/>

Save

Delete

Copy

Reset

Apply Config

Add New




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

**Directory Number Configuration** Related Links: Configure Device (SEPC07BBCA1B811) Go

Save Delete Reset Apply Config Add New

**Directory Number Information**

Directory Number\* 4084 ☐ Urgent Priority  
Route Partition < None >  
Description 7323204084  
Alerting Name Cisco 9971 Phone  
ASCII Alerting Name Cisco 9971 Phone  
External Call Control Profile < None >  
☒ Allow Control of Device from CTI  
Associated Devices SEPC07BBCA1B811  

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 < None >  
Network Hold MOH Audio Source < None >  
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 9971 SIP Configuration (Continued...)



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

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

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

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



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

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

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

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

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



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

Set Display (caller ID) = Cisco9971-Phone. This is used in this example.

Set ASCII Display (caller ID) = Cisco9971-Phone. This is used in this example.

Set Line Text Label = Cisco9971-Phone. This is used in this example.

Set External Phone Number Mask = 7323204084. This is used in this example.

**Line 1 on Device SEPC07BBCA1B811**

Display (Caller ID)	Cisco 9971-Phone	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)	Cisco 9971-Phone	
Line Text Label	Cisco 9971-Phone	
External Phone Number Mask	7323204084	
Visual Message Waiting Indicator Policy*	Use System Policy	
Audible Message Waiting Indicator Policy*	Default	
Ring Setting (Phone Idle)*	Use System Default	
Ring Setting (Phone Active)	Use System Default	Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)	Use System Default	

Call Pickup Group Audio Alert Setting(Phone Active)	Use System Default
Recording Option *	Call Recording Disabled
Recording Profile	< None >
Recording Media Source*	Gateway Preferred
Monitoring Calling Search Space	< None >
<input checked="" type="checkbox"/> Log Missed Calls	

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

### Multiple Call/Call Waiting Settings on Device SEPC07BBCA1B811

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 SEPC07BBCA1B811

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

### Users Associated with Line

	Full Name	User ID	Permission
<input type="checkbox"/>	cisco,	jabber1	



## 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

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**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

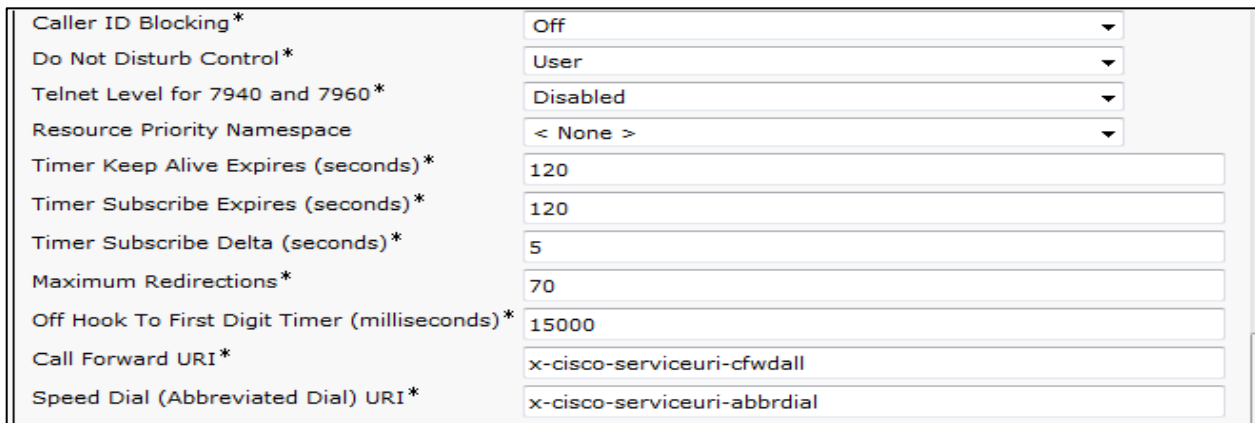
☐ Assured Services SIP conformance



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

<b>SDP Information</b>	
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)	

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



- ☒ Conference Join Enabled
- ☐ RFC 2543 Hold
- ☒ Semi Attended Transfer
- ☐ Enable VAD
- ☐ Stutter Message Waiting
- ☐ MLPP User Authorization

**Normalization Script**

Normalization Script < None >

☐ Enable Trace

	Parameter Name	Parameter Value		
1			+	-

Incoming Requests FROM URI Settings

Caller ID DN

Caller Name

Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on\*

Never

RSVP Over SIP\*

Local RSVP

Resource Priority Namespace List

< None >

☐ Fall back to local RSVP

SIP Rel1XX Options\*

Send PRACK if 1xx Contains SDP

Video Call Traffic Class\*

Mixed

Calling Line Identification Presentation\*

Default

Session Refresh Method\*

Invite

Early Offer support for voice and video calls\*

Disabled (Default value)

☐ Enable ANAT

☐ Deliver Conference Bridge Identifier

☐ Allow Passthrough of Configured Line Device Caller Information

☐ Reject Anonymous Incoming Calls

☐ Reject Anonymous Outgoing Calls

☐ Send ILS Learned Destination Route String

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

**SIP OPTIONS Ping**  
☒ Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"  
Ping Interval for In-service and Partially In-service Trunks (seconds)\*   
Ping Interval for Out-of-service Trunks (seconds)\*   
Ping Retry Timer (milliseconds)\*   
Ping Retry Count\*

**SDP Information**  
☐ Send send-receive SDP in mid-call INVITE  
☐ Allow Presentation Sharing using BFCP  
☐ Allow iX Application Media  
☐ Allow multiple codecs in answer SDP

## SIP Trunk to Cisco UBE Configuration

**Navigation:** Device → Trunk



Set Device Name\* = ATT\_SIP\_TRUNK. This is used for this example  
Set Description = ATT SIP Trunk to PSTN. This is used for this example  
Set Device Pool\* = G729\_pool. This is used for this example  
Set Media Resource Group List = MRGL\_MTP.

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

Save X Delete Reset + Add New

**SIP Trunk Status**

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

**Device Information**

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

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



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

<input type="checkbox"/> Media Termination Point Required
<input checked="" type="checkbox"/> Retry Video Call as Audio
<input type="checkbox"/> Path Replacement Support
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU
<input type="checkbox"/> Unattended Port
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.
Consider Traffic on This Trunk Secure* <span>When using both sRTP and TLS</span>
Route Class Signaling Enabled* <span>Default</span>
Use Trusted Relay Point* <span>Default</span>
<input checked="" type="checkbox"/> PSTN Access
<input type="checkbox"/> Run On All Active Unified CM Nodes

---

**Intercompany Media Engine (IME)**

---

E.164 Transformation Profile	<span>&lt; None &gt;</span>
------------------------------	-----------------------------

---

**MLPP and Confidential Access Level Information**

---

MLPP Domain	<span>&lt; None &gt;</span>
Confidential Access Mode	<span>&lt; None &gt;</span>
Confidential Access Level	<span>&lt; None &gt;</span>

---

**Call Routing Information**

---

<input checked="" type="checkbox"/> Remote-Party-Id
<input checked="" type="checkbox"/> Asserted-Identity
Asserted-Type* <span>Default</span>
SIP Privacy* <span>Default</span>

---

**Inbound Calls**

---

Significant Digits*	<span>4</span>
Connected Line ID Presentation*	<span>Default</span>
Connected Name Presentation*	<span>Default</span>
Calling Search Space	<span>&lt; None &gt;</span>
AAR Calling Search Space	<span>&lt; None &gt;</span>
Prefix DN	<input type="text"/>
<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 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 type="text" value=" &lt; None &gt;"/>	<input 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	<input type="text"/>
Caller Name	<input type="text"/>
<input type="checkbox"/> 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.



SIP Information		
<b>Destination</b>		
<input type="checkbox"/> Destination Address is an SRV		
<b>Destination Address</b>	<b>Destination Address IPv6</b>	<b>Destination Port</b>
1* 10.80.22.10		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	

<b>Normalization Script</b>	
Normalization Script < None >	
<input type="checkbox"/> Enable Trace	
<b>Parameter Name</b>	<b>Parameter Value</b>
1	
<b>Recording Information</b>	
<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	
<b>Geolocation Configuration</b>	
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	

Outbound Calls	
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	
Caller Information	
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 *	10.80.22.7		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

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

**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 interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the subtitle "For Cisco Unified Communications Solutions". The navigation menu includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Bulk Administration". The "Call Routing" menu is expanded, showing "Find and List Route Patterns".

Below the navigation bar, there is a section titled "Find and List Route Patterns". It includes a toolbar with buttons for "Add New", "Select All", "Clear All", and "Delete Selected". A status box indicates "3 records found".

The main content area displays a table of Route Patterns. The table has columns for "Pattern", "Description", "Partition", "Route Filter", "Associated Device", and "Copy". The table shows three records:

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	

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



## Route Pattern Configuration (Continued...)

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

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

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

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

Save Delete Copy Add New

**Pattern Definition**

**Route Pattern\***

9.@"

Route Partition

< None >

Description

To PSTN via ATT SIP Trunk

Numbering Plan\*

NANP

Route Filter

< None >

MLPP Precedence\*

Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

< None >

Route Class\*

Default

Gateway/Route List\*

ATT\_SIP\_TRUNK

[\(Edit\)](#)

Route Option

☒ Route this pattern

☐ Block this pattern

No Error

Call Classification\*

OffNet

External Call Control Profile

< None >

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

☐ Require Forced Authorization Code

Authorization Level\*

0

☐ Require Client Matter Code

**Calling Party Transformations**

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\*

Default

Calling Name Presentation\*

Default

Calling Party Number Type\*

Cisco CallManager

Calling Party Numbering Plan\*

Cisco CallManager



## Route Pattern Configuration (Continued...)

<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"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	0
<input type="checkbox"/> Require Client Matter Code	

**Calling Party Transformations**

<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

Route Pattern Configuration (Continued...)



<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

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

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

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

**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

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain < None >

Route Class\* Default

Gateway/Route List\* Trunk\_SIP\_FAX\_Gateway (Edit)

Route Option  
☒ Route this pattern  
☐ Block this pattern No Error ▾

Call Classification\* OffNet ▾

External Call Control Profile < None > ▾

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

☐ Require Forced Authorization Code

Authorization Level\* 0

☐ Require Client Matter Code

**Calling Party Transformations**

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\* Default ▾

Calling Name Presentation\* Default ▾

Calling Party Number Type\* Cisco CallManager ▾

Calling Party Numbering Plan\* Cisco CallManager ▾



## Route Pattern Configuration (Continued...)

<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

Select Phone Type\* = Cisco Unified Client services framework

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

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

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

Select Phone Button Template\* = Standard Client Services Framework.

**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  
1 Line [1] - 4084 (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: Unregistered  
IPv4 Address: 172.16.31.153  
Active Load ID: image\_a  
Download Status: None

**Device Information**  
☒ Device is Active  
☒ Device is trusted  
Device Name\* CSFuser1  
Description CSFuser1  
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\* = jabber1. This is used for this example

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

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

## Jabber Client Configuration (Contd...)

**Number Presentation Transformation**

**Caller ID For Calls From This Phone**

Calling Party Transformation CSS

< None >

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

**Remote Number**

Calling Party Transformation CSS

< None >

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

**Protocol Specific Information**

Packet Capture Mode\*

None

Packet Capture Duration

0

BLF Presence Group\*

Standard Presence group

SIP Dial Rules

< None >

MTP Preferred Originating Codec\*

711ulaw

Device Security Profile\*

Cisco Unified Client Services Framework - Standard

Rerouting Calling Search Space

< None >

SUBSCRIBE Calling Search Space

< None >

SIP Profile\*

Standard SIP Profile w/Early Media Disabled

[View Details](#)

Digest User

< None >

☐ Media Termination Point Required

☐ Unattended Port

☐ Require DTMF Reception

## Jabber Client Configuration (Contd...)

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

Certificate Operation

Authentication Mode

Authentication String

Key Size (Bits)\*

Operation Completes     (YYYY:MM:DD:HH)

By

Certificate Operation None

Status:

Note: Security Profile Contains Addition CAPF Settings.

**Extension Information**

☐ Enable Extension Mobility

Log Out Profile

Log in Time

Log out Time

**MLPP and Confidential Access Level Information**

MLPP Domain

Confidential Access Mode

Confidential Access Level

**Do Not Disturb**

☐ Do Not Disturb

DND Option\*

DND Incoming Call Alert

## Jabber Client Configuration (Contd...)

Product Specific Configuration Layout

?

Parameter Value

Override Common Settings

Video Calling

Enabled

\*

Interactive Connectivity Establishment (ICE)

ICE

Enabled

Default Candidate Type

Host

Server Reflexive Address

Enabled

Primary TURN Server Host Name or IP Address

Secondary TURN Server Host Name or IP Address

TURN Server Transport Type

Auto

TURN Server Username

administrator

TURN Server Password

.....

Instant Messaging

File Types to Block in File Transfer

URLs to Block in File Transfer

## Jabber Client Configuration (Contd...)

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

Analytics Server URL		<input type="checkbox"/>
Cisco Support Field		<input type="checkbox"/>

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



## Voicemail Port Configuration

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

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

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

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

**Find and List Voice Mail Ports**

Add New

Select All

Clear All

Delete Selected

Reset Selected

Apply Config to 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**

<input type="checkbox"/>	Device Name ▲	Description	Device Pool	Device Security Mode	Calling Search Space	Extension	Partition	Status	IP Address	Copy
<input type="checkbox"/>	<a href="#">CiscoUM1-VI1</a>	VoiceMail	<a href="#">G729 Pool</a>	Non Secure Voice Mail Port		2301		Registered with clus32pubsub	10.80.22.4	
<input type="checkbox"/>	<a href="#">CiscoUM1-VI2</a>	VoiceMail	<a href="#">G729 Pool</a>	Non Secure Voice Mail Port		2302		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\* = 2301. 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	VoiceMail
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*	2301
Partition	< None >
Calling Search Space	< None >
AAR Group	< None >
Internal Caller ID Display	VoiceMail
Internal Caller ID Display (ASCII format)	VoiceMail
External Number Mask	

Save Delete Copy Reset Apply Config Add New



## Message Waiting Numbers Configurations

**Navigation:** Advanced Features → Voice Mail → Message Waiting

Set Message Waiting Number\* = 2298

Set Message Waiting Indicator\* = On

Set Message Waiting Number\* = 2399

Set Message Waiting Indicator\* = Off

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and a navigation menu with options like "administrator", "Search Documentation", and "About". Below this is a secondary navigation bar with categories like "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Bulk Administration".

The main content area is titled "Find and List Message Waiting Numbers". It includes a toolbar with buttons for "Add New", "Select All", "Clear All", and "Delete Selected". Below the toolbar is a "Status" section indicating "2 records found".

The "Message Waiting Numbers" section shows a list of two records, highlighted with a red box. The records are as follows:

Message Waiting Numbers	Directory Number	Description	Partition	Calling Search Space	Copy
<input type="checkbox"/>	<a href="#">2298</a>	MWI ON			
<input type="checkbox"/>	<a href="#">2299</a>	MWI OFF			

Below the table is another toolbar with 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

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

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

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

### 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 Pilot    Voice Mail Pilot Number ▾ begins with ▾    Find    Clear Filter    +    -

		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	VoiceMail Pilot-Default	

Add New    Select All    Clear All    Delete Selected

### Voice Mail Pilot Information

Voice Mail Pilot Number 2300

Calling Search Space < None > ▾

Description VoiceMail Pilot-Default

☒ Make this the default Voice Mail Pilot for the system

Save    Delete    Add New



## FAX Gateway Configuration

FAX-GATEWAY2#**sh version**

Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9-M), Version 15.4(3)M1, RELEASE SOFTWARE (fc1)

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

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Compiled Sat 25-Oct-14 03:34 by prod\_rel\_team

ROM: System Bootstrap, Version 15.0(1r)M16, RELEASE SOFTWARE (fc1)

FAX-GATEWAY2 uptime is 1 week, 16 hours, 39 minutes

System returned to ROM by reload at 14:38:17 UTC Tue Mar 10 2015

System image file is "flash0:c2900-universalk9-mz.SPA.154-3.M1.bin"

Last reload type: Normal Reload

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

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EDCS# xxx Rev #

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Note: Testing was conducted in tekVizion labs



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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).

Cisco CISCO2901/K9 (revision 1.0) with 483328K/40960K bytes of memory.

Processor board ID FTX174081SJ

2 Gigabit Ethernet interfaces

1 terminal line

2 Voice FXS interfaces

DRAM configuration is 64 bits wide with parity enabled.

255K bytes of non-volatile configuration memory.

250880K bytes of ATA System CompactFlash 0 (Read/Write)

License Info:

License UDI:

-----  
Device# PID SN  
-----



\*1 CISCO2901/K9 FTX174081SJ

Technology Package License Information for Module:'c2900'

-----			
Technology	Technology-package	Technology-package	
	Current	Type	Next reboot
-----			
ipbase	ipbasek9	Permanent	ipbasek9
security	None	None	None
uc	uck9	Permanent	uck9
data	None	None	None
NtwkEss	None	None	None
CollabPro	None	None	None

Configuration register is 0x2102

FAX-GATEWAY2#sh run

Building configuration...



Current configuration : 7131 bytes

!

! Last configuration change at 14:41:28 UTC Wed Mar 25 2015 by cisco

!

version 15.4

service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

!

hostname FAX-GATEWAY2

!

boot-start-marker

boot-end-marker

!

aqm-register-fnf

!

logging queue-limit 1000000000

logging buffered 10000000

logging rate-limit 10000

no logging console

no logging monitor

enable secret 4 iR3uUX3Bo6oYbT6ajhFwJe39FR4g.1QCmm7yYduKGZI

!

no aaa new-model

!



!

!

!

!

!

!

!

!

!

!

!

!

!

ip domain name lab.tekvizion.com

ip name-server 10.64.1.3

ip cef

no ipv6 cef

multilink bundle-name authenticated

!

!

!

!

stcapp feature access-code

!

stcapp feature speed-dial

!



!

!

!

!

cts logging verbose

!

crypto pki trustpoint TP-self-signed-2189441908

enrollment selfsigned

subject-name cn=IOS-Self-Signed-Certificate-2189441908

revocation-check none

rsakeypair TP-self-signed-2189441908

!

!

crypto pki certificate chain TP-self-signed-2189441908

certificate self-signed 01

3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030  
31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274  
69666963 6174652D 32313839 34343139 3038301E 170D3133 31303031 32303234  
30325A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649  
4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D32 31383934  
34313930 3830819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281  
810092C7 1982BC36 792DA64E 8FB4D8BC 1DDD4D7A 0882107F B14FCB24 699A35A9  
D521C88A 5B43F4FC D394E945 81A1380A 2E753478 93190ADE 75AA8971 883E9214  
C607CCDF 6FCCDE9C E95CE01A AEE4FCBE 3E91A43C D11C638F FC3E4ED2 57569523  
70A8D7C6 EFAD6688 C6244C79 5B955391 BF75EE61 DC4D0ADE 8D897AE2 CE76A938  
983F0203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603



```
551D2304 18301680 14279B59 09E3EB37 0AE0DCE0 F8075BB6 DF93858A 45301D06
03551D0E 04160414 279B5909 E3EB370A E0DCE0F8 075BB6DF 93858A45 300D0609
2A864886 F70D0101 05050003 8181006E CF10B11F 9D8B59A9 AEACDEB8 26649CBB
0F6C9690 12EAE870 4BF5703D 98D2665A CD1B27D2 9B29351D 3ADF0B97 3C41F59A
0DD82FF8 66CE4689 2D089FE8 EF3FFE54 5C85608C EE45908F D1160BDE A9185D58
D3DA8795 428A7CE7 B9522F7C 60796800 485EDA2F B6C86F7A DF66B562 74942705
C81F1883 7D4E29FC 8E999F7E EAE070
```

```
quit
```

```
voice-card 0
```

```
dsp services dspfarm
```

```
!
```

```
!
```

```
!
```

```
voice service voip
```

```
allow-connections sip to sip
```

```
no supplementary-service sip handle-replaces
```

```
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," ""
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
license udi pid CISCO2901/K9 sn FTX174081SJ
```

```
hw-module pvdm 0/0
```

```
!
```

```
!
```

```
!
```

```
username cisco privilege 15 secret 4 tnhtc92DXBhelxjYk8LWJrPV36S2i4ntXrpb4RFmfqY
```

```
!
```

```
redundancy
```

```
!
```

```
!
```

```
!
```



```
!  
!  
!  
interface Embedded-Service-Engine0/0  
  no ip address  
  shutdown  
!  
interface GigabitEthernet0/0  
  ip address 10.80.22.7 255.255.255.0  
  duplex auto  
  speed auto  
!  
interface GigabitEthernet0/1  
  no ip address  
  shutdown  
  duplex auto  
  speed auto  
!  
ip forward-protocol nd  
!  
ip http server  
ip http authentication local  
ip http secure-server  
ip http timeout-policy idle 60 life 86400 requests 10000  
!  
ip route 0.0.0.0 0.0.0.0 10.80.22.1
```



```
ip route 10.64.0.0 255.255.0.0 10.80.22.1
ip route 10.80.0.0 255.255.0.0 10.80.22.1
ip route 172.16.0.0 255.255.0.0 10.80.22.1
!
!
!
!
control-plane
!
!
voice-port 0/0/0
no vad
shutdown
!
voice-port 0/0/1
no echo-cancel enable
no vad
station-id name fax test
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
!
!
ccm-manager music-on-hold
!
no ccm-manager fax protocol cisco
!
dial-peer voice 110 pots
service session
destination-pattern 4084
port 0/0/1
!
dial-peer voice 200 voip
description CUCM to Gateway
service session
session protocol sipv2
session transport udp
incoming called-number 4084
voice-class codec 1
voice-class sip profiles 1
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 201 voip

description Gateway to CUCM

service session

destination-pattern [2-9]T

session protocol sipv2

session target ipv4:10.80.22.2

session transport udp

voice-class codec 1

voice-class sip profiles 1

dtmf-relay rtp-nte

fax rate 14400

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

no vad

!

!

gateway

timer receive-rtp 1200

!

!

!

gatekeeper

shutdown
```



!

!

banner exec ^C

% Password expiration warning.

-----

Cisco Configuration Professional (Cisco CP) is installed on this device and it provides the default username "cisco" for one-time use. If you have already used the username "cisco" to login to the router and your IOS image supports the "one-time" user option, then this username has already expired. You will not be able to login to the router with this username after you exit this session.

It is strongly suggested that you create a new username with a privilege level of 15 using the following command.

```
username <myuser> privilege 15 secret 0 <mypassword>
```

Replace <myuser> and <mypassword> with the username and password you want to use.

-----

^C

banner login ^C

-----

Cisco Configuration Professional (Cisco CP) is installed on this device.



This feature requires the one-time use of the username "cisco" with the password "cisco". These default credentials have a privilege level of 15.

YOU MUST USE CISCO CP or the CISCO IOS CLI TO CHANGE THESE PUBLICLY-KNOWN CREDENTIALS

Here are the Cisco IOS commands.

```
username <myuser> privilege 15 secret 0 <mypassword>
no username cisco
```

Replace <myuser> and <mypassword> with the username and password you want to use.

IF YOU DO NOT CHANGE THE PUBLICLY-KNOWN CREDENTIALS, YOU WILL NOT BE ABLE TO LOG INTO THE DEVICE AGAIN AFTER YOU HAVE LOGGED OFF.

For more information about Cisco CP please follow the instructions in the QUICK START GUIDE for your router or go to <http://www.cisco.com/go/ciscocp>

-----

^C

!

line con 0

login local

line aux 0

line 2



```
no activation-character
no exec
transport preferred none
transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
stopbits 1
line vty 0 4
exec-timeout 0 0
login local
transport input telnet ssh
line vty 5 15
login local
transport input telnet ssh
!
scheduler allocate 20000 1000
!
end
```



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

CUC Version

### Cisco Unity Connection Administration

Version 10.5.2.10000-5

A photograph of a server room aisle, showing rows of server racks on both sides, with a bright light at the end of the aisle.

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## CUC Telephony Integration with Cisco UCM

**Navigation:** Telephony Integrations → Phone system

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

The screenshot shows the Cisco Unity Connection Administration web interface. The left sidebar contains a navigation tree with 'Telephony Integrations' expanded, showing 'Phone System' as the selected item. The main content area is titled 'Search Phone Systems' and displays a status message: 'Found 2 Phone System(s)'. Below this is a table of phone systems. The first row, 'CUCM', is highlighted with a red box and has a port count of 2. The second row, 'PhoneSystem', has a port count of 0. The interface also includes a search bar, a 'Find' button, and 'Delete Selected' and 'Add New' buttons.

Display Name	Port Count
CUCM	2
PhoneSystem	0



## CUC Port Group

**Navigation:** Telephony Integration → Port Group

The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar contains a navigation menu with the following items: Cisco Unity Connection, Users, Class of Service, Templates, Contacts, Distribution Lists, Call Management, Message Storage, Networking, Unified Messaging, Video, Dial Plan, System Settings, Telephony Integrations (expanded), Phone System, Port Group (highlighted), Port, Speech Connect Port, Trunk, Security, and Tools. The main content area is titled 'Search Port Groups' and includes a search bar with the text 'Find Port Groups where Port Group Name begins with'. Below the search bar is a table of port groups. The table has the following columns: Port Group Name, Phone System Display Name, Port Count, Integration Method, and Needs Reset. The table contains one row for the 'CUCM-1' port group, which is highlighted with a red box. The row data is: CUCM-1, CUCM, 2, SCCP (Skinny), and No. Below the table are buttons for 'Delete Selected' and 'Add New'.

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

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

**Search Port Groups** Search Port Groups

Port Group Refresh Help

**Status**  
Found 1 Port Group(s)

**Port Groups (1 - 1 of 1)** Rows per Page 25

Find Port Groups where Port Group Name begins with

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

Delete Selected Add New



## CUC Port Group(continued...)

Set Display Name\* = CUCM-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.



## CUC Port Settings

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

Navigation Cisco Unity Connection  
administrator | Search Documentation

Cisco Unity Connection


- 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

 Found 2 Port(s)

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

Find Port where Display Name begins with

<input type="checkbox"/>	Display Name ^	Phone System Display Name	Extension	Server	Enabled	Answer Calls	Message Notification
<input type="checkbox"/>	<a href="#">CUCM-1-001</a>	<a href="#">CUCM</a>		clus32unity	X	X	X
<input type="checkbox"/>	<a href="#">CUCM-1-002</a>	<a href="#">CUCM</a>		clus32unity	X	X	X

Delete Selected Add New

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EDCS# xxx Rev #

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Note: Testing was conducted in tekVizion labs



## 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.

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

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

**Cisco Unity Connection**

- 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

**Edit User Basics (4084)**

Search Users ▶ Edit User Basics (4084)  
Related Links Bulk Edit By CSV Go

User Edit Refresh Help

Save Delete Previous Next

**Name**

Alias\* 4084

First Name

Last Name

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 = CUCM.

<b>▼ Cisco Unity Connection</b> <ul style="list-style-type: none"> <li>Users           <ul style="list-style-type: none"> <li>Users</li> <li>Import Users</li> <li>Sync Users</li> </ul> </li> <li>Class of Service</li> <li>Templates</li> <li>Contacts</li> <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>	Outgoing Fax Number <input type="text"/> Outgoing Fax Server --- Not Selected --- <div style="border: 2px solid red; padding: 2px;">           Partition clus32unity Partition            Search Scope clus32unity Search Space            Phone System CUCM         </div> Class of Service Voice Mail User COS Active Schedule Weekdays <input type="button" value="View"/> <input type="checkbox"/> Set for Self-enrollment at Next Sign-In <input checked="" type="checkbox"/> List in Directory <input checked="" type="checkbox"/> Send Non-Delivery Receipts on Failed Message Delivery <input checked="" type="checkbox"/> Skip PIN When Calling From a Known Extension <b>Caution!</b> Security risk. See Help for This Page for details. <input type="checkbox"/> Use Short Calendar Caching Poll Interval Recorded Name <input type="button" value="Play/Record"/> <b>Location</b> Address <input type="text"/> Building <input type="text"/> City <input type="text"/> State <input type="text"/> Postal Code <input type="text"/> Country United States
---	---

<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 (GMT-06:00) America/Chicago Language <input checked="" type="radio"/> Use System Default Language <input type="radio"/> English(United States) Department <input type="text"/> Manager <input type="text"/> Billing ID <input type="text"/> Corporate Email Address <input type="text"/> <input type="checkbox"/> Generate SMTP Proxy Address From Corporate Email Address Directory URI <input type="text"/> Corporate Phone Number <input type="text"/> <input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Previous"/> <input type="button" value="Next"/> Fields marked with an asterisk (*) are required.
---	---

## Auto Attendant

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



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

Set Phone System = 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 'System Call Handlers' selected. 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 results. The table has columns for 'Display Name' and 'Extension'. The first row, 'Demo Auto attend', is highlighted with a red box and has an extension of 2999. The other three rows are 'Goodbye', 'Opening Greeting', and 'Operator', all with an extension of 0. At the bottom of the table are buttons for 'Delete Selected', 'Add New', 'Bulk Edit', and 'Show Dependencies'.

	Display Name ^	Extension
<input checked="" type="checkbox"/>	<a href="#">Demo Auto attend</a>	2999
<input type="checkbox"/>	<a href="#">Goodbye</a>	0
<input type="checkbox"/>	<a href="#">Opening Greeting</a>	0
<input type="checkbox"/>	<a href="#">Operator</a>	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\*

Demo Auto attend

Creation Time

2015-03-05 05:25:48.800

Phone System

CUCM

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

2999

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

**System version: 10.5.2.10000-9**

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



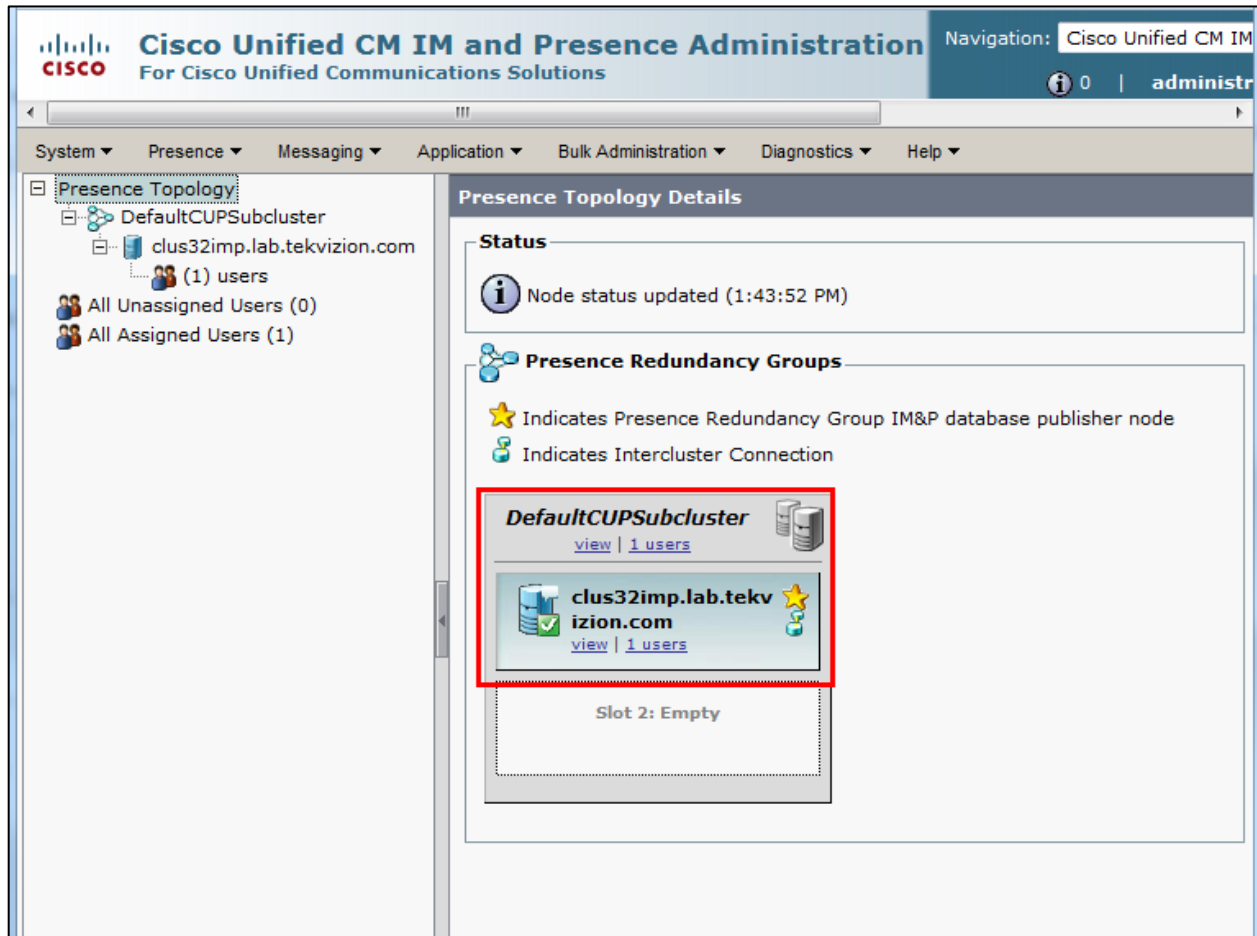
User administrator last logged in to this cluster on Monday, March 16, 2015 2:03:05 AM CDT, to node 10.80.22.3, from 172.16.31.153 using HTTPS

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## Presence Topology

**Navigation:** System → Presence Topology



The screenshot displays the Cisco Unified CM IM and Presence Administration web interface. The top navigation bar includes links for System, Presence, Messaging, Application, Bulk Administration, Diagnostics, and Help. The left sidebar shows the 'Presence Topology' section expanded, with sub-items for 'DefaultCUPSubcluster', 'clus32imp.lab.tekvizion.com' (1 users), 'All Unassigned Users (0)', and 'All Assigned Users (1)'. The main content area is titled 'Presence Topology Details' and contains the following sections:

- Status:** A message indicating 'Node status updated (1:43:52 PM)'.
- Presence Redundancy Groups:** A section explaining the star icon as the 'IM&P database publisher node' and the intercluster connection icon.
- DefaultCUPSubcluster:** A highlighted box containing details for the 'clus32imp.lab.tekvizion.com' subcluster, including a 'view' link and '1 users' count. Below this, 'Slot 2: Empty' is indicated.



## Node Configuration

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

The screenshot displays the Cisco Unified CM IM and Presence Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM IM and Presence Administration", and the subtitle "For Cisco Unified Communications Solutions". The right side of the navigation bar shows "Navigation: Cisco Unified CM" and a user profile "admini". Below the navigation bar is a menu with tabs: System, Presence, Messaging, Application, Bulk Administration, Diagnostics, and Help. The main content area is divided into two sections. On the left, under "Presence Topology", there is a tree view showing "DefaultCUPSubcluster" and "clus32imp.lab.tekvizion.com" (1 users). Below this, there are links for "All Unassigned Users (0)" and "All Assigned Users (1)". On the right, the "Node Detail" section is displayed. It has a "Status" section with a message "Node status updated (1:45:19 PM)". Below that is the "Node Configuration" section, which contains a table with the following details:

Node Configuration	
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**

**i** 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
<a href="#">jabber1</a>	cisco		jabber1@lab.tekvizion.com	jabber1@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



## Important Information

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