

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 Sep 2015



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Introduction

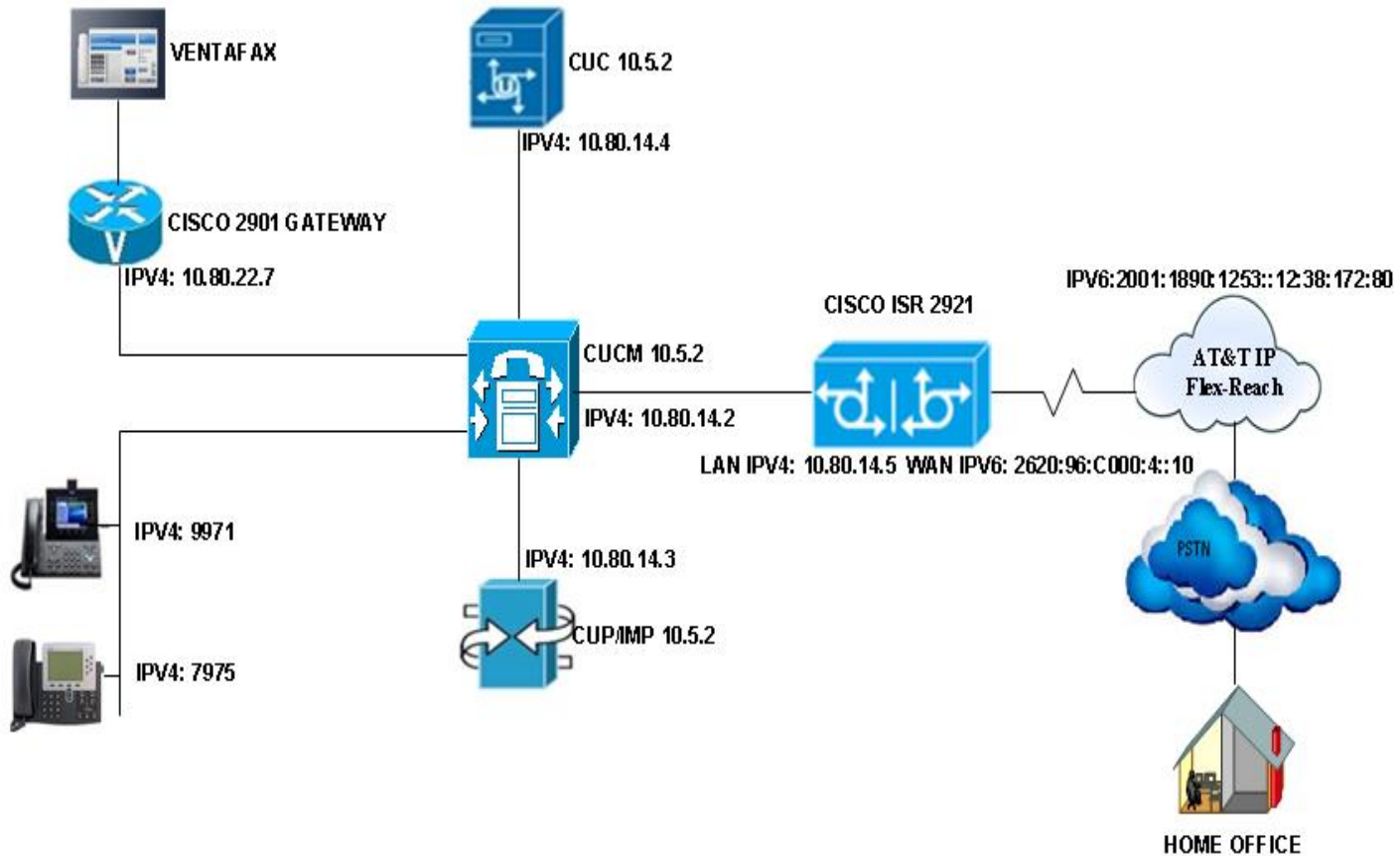
Service Providers today, such as AT&T, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. AT&T IP Flexible Reach is a service provider 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. Network Topology between Cisco ISR and AT&T Flexible reach SIP is deployed using IPV6.
- 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

- Cisco UCS-C240 VMWare server running ESXi 5.5
- Cisco IP Phones. This solution was tested with Cisco 7965, Cisco 7975 and Cisco 9971 phones
- Cisco Integrated Services Router - Cisco CISCO2921/K9 (revision 1.0) with 483328K/40960K bytes of memory.
- Processor board ID FTX1746AJCB
 - 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)

Software Requirements

- Cisco UCM: System version: 10.5.2.11900-3, including Business Edition 6000 and Business Edition 7000.
- Cisco ISR: C2900 Software (C2900-UNIVERSALK9-M), Version 15.4(3)M1, RELEASE SOFTWARE (fc1)
- System image file is "flash:c2900-universalk9-mz.SPA.154-3.M1.bin"
- Cisco Unity Connection version: System version: 10.5.2.11900-3
- Cisco Unified CM IM and Presence: System version: 10.5.2.10000-9
- Cisco Jabber client version: 10.5.0 Build 37889
- VentaFax client version: 7.4.237.590



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 T.38 and G.711
- 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 911 calls were terminated to a voicemail platform in lab environment within AT&T for test

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 Ringing and Simultaneous Ringing features. Provisioning is self-explanatory. AT&T representative can be contacted for any help required in this regard.

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 (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.

Ring back Tone on Early Unattended Transfer

- Caller does not hear ring back 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 executed with limitations on AT&T network

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



Configuration Considerations

- To enable conference using G729 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.
- 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 SIP UPDATE message. This causes the Cisco UBE to timeout and hence 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

ATT-IPV6#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)

ATT-IPV6 uptime is 4 days, 4 hours, 47 minutes

System returned to ROM by reload at 06:50:49 UTC Thu Sep 24 2015

System image file is "flash: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.

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

Processor board ID FTX1746AJCB

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	FTX1746AJCB
----	--------------	-------------



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 0x21024 Gigabit Ethernet interfaces

32768K bytes of non-volatile configuration memory.

4194304K bytes of physical memory.

7741439K bytes of eUSB flash at bootflash:.

Configuration register is 0x2102



ATT-IPV6#sh run

Building configuration...

Current configuration : 11243 bytes

!

! Last configuration change at 08:17:43 UTC Mon Sep 28 2015 by cisco

!

version 15.4

service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

service sequence-numbers

!

hostname ATT-IPV6

!

boot-start-marker

boot system flash c2900-universalk9-mz.SPA.154-3.M1.bin

boot-end-marker

!

aqm-register-fnf

!

logging queue-limit 1000000000

logging buffered 30000000

logging rate-limit 10000

no logging console

no logging monitor



```
enable secret 4 Pe0NhiWw5IXZpE.k5VhTSCoGPcuVeRyrer9kEPz20Z6!
```

```
no aaa new-model
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
no ip domain lookup
```

```
ip cef
```

```
ipv6 unicast-routing
```

```
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-hiding1
mode border-element2
media disable-detailed-stats
allow-connections sip to sip3
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
header-passing
error-passthru4
asserted-id pai5
early-offer forced6
```

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

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

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

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

⁵ This command enables router to send P-Asserted ID within the SIP Message Header. Alternatively, this command can also be applied to individual dial-peers (voice-class sip asserted-id pai).

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



```
no silent-discard untrusted
midcall-signaling passthru7
privacy-policy passthru8
g729 annexb-all
!
voice class codec 19
  codec preference 1 g729r8 bytes 30
  codec preference 2 g711ulaw
!
voice class codec 2
  codec preference 1 g711ulaw
  codec preference 2 g729r8 bytes 30
!
voice class sip-profiles 1
  response ANY sip-header Allow-Header modify "UPDATE," ""
  request INVITE sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"10
  response ANY sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"
  request INVITE sdp-header Audio-Attribute add "a=ptime:30"11
!
```

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

⁸ This command allows for privacy settings to be transparently passed between AT&T network and Cisco UCM. This command can either be set at a global level, such as in this example, or it can be set at the dial-peer level.

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

¹⁰ 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.

¹¹ 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.



!

!

!

!

license udi pid CISCO2921/K9 sn FTX1746AJCB

hw-module pvdn 0/0

!

!

!

username cisco privilege 15 password 0 cisco

!

redundancy

!

!

!

!

!

!

!

!

interface Embedded-Service-Engine0/0

no ip address

shutdown

!



```
interface GigabitEthernet0/012
```

```
description Wan Interface
```

```
no ip address
```

```
negotiation auto
```

```
ipv6 address 2620:96:C000:4::10/6413
```

```
ipv6 enable
```

```
cdp enable
```

```
!
```

```
interface GigabitEthernet0/114
```

```
description Lan Interface
```

```
ip address 10.80.14.5 255.255.255.015
```

```
negotiation auto
```

```
cdp enable
```

```
!
```

```
interface GigabitEthernet0/2
```

```
no ip address
```

```
shutdown
```

```
duplex auto
```

```
speed auto
```

```
!
```

```
ip forward-protocol nd
```

```
!
```

```
no ip http server
```

¹² Cisco UBE WAN interface to AT&T

¹³ Cisco UBE WAN interface IPv6 Address

¹⁴ Cisco UBE LAN interface to Cisco UCM

¹⁵ Cisco UBE LAN interface IPv4 Address



```
no ip http secure-server
!
ip route 10.64.0.0 255.255.0.0 10.80.14.1
ip route 10.80.0.0 255.255.0.0 10.80.14.1
ip route 172.16.0.0 255.255.0.0 10.80.14.1
!
ipv6 route ::/0 2620:96:C000:4::1
!
!
!
control-plane
!
!
!
!
!
!
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
!
```



!

```
dial-peer voice 300 voip
description " Int'l calls to AT&T - AT&T facing side "
destination-pattern 011T
session protocol sipv2
session target ipv6:[2001:1890:1253:0:12:38:172:80]
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
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 800 voip
description " Incoming AT&T to IP-PBX . AT&T facing side "
huntstop
session protocol sipv2
incoming called-number [27][37][27].....
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 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
session protocol sipv2
session target ipv6:[2001:1890:1253:0:12:38:172:80]
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/016
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
```

¹⁶ 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. By enabling these commands, Cisco UBE Wan Interface is bound.



```
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

session protocol sipv2

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

voice-class codec 1

voice-class sip asymmetric payload full

voice-class sip asserted-id pai

voice-class sip privacy-policy passthru

voice-class sip early-offer forced

voice-class sip profiles 1

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

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

dtmf-relay rtp-nte

no vad

!

dial-peer voice 141 voip

description "Network Feature"

destination-pattern *..

session protocol sipv2

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

voice-class codec 1
```



```
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2151 voip
description " Incoming AT&T to IP-PBX - IP-PBX facing side "
destination-pattern 7322162...
session protocol sipv2
session target ipv4:10.80.14.2:5060
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 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 700 voip¹⁷

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

huntstop

destination-pattern 7323680...

session protocol sipv2¹⁸

session target ipv4:10.80.14.2:5060

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 early-offer forced

voice-class sip bind control source-interface GigabitEthernet0/1²²

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

dtmf-relay rtp-nte²³

fax rate 14400

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

no vad

!

¹⁷ Dial-peer facing AT&T Network

¹⁸ Session protocol SIPv2 is used for this testing.

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

²⁰ Configures the dynamic SIP asymmetric payload support.

²¹ 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.

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

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

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



```
dial-peer voice 400 voip
description " Int'l calls to AT&T - IP-PBX facing side "
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 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
session protocol sipv2
session target ipv6:[2001:1890:1253:0:12:38:172:80]
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
```



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

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

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

```
no vad
```

```
!
```

```
dial-peer voice 600 voip
```

```
description " N11 Calls to AT&T - IP-PBX facing side "
```

```
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 bind control source-interface GigabitEthernet0/1
```

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

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

```
no vad
```

```
!
```

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

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

```
session protocol sipv2
```

```
incoming called-number 7322162...
```

```
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 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
!
!
gateway
media-inactivity-criteria all
timer receive-rtcp 5
timer receive-rtp 86400
!
sip-ua
no remote-party-id
timers expires 1800000
connection-reuse
protocol mode dual-stack
!
!
!
gatekeeper
shutdown
!
!
```



```
!  
line con 0  
  logging synchronous  
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 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 Go

administrator | Search Documentation | About | Logout

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

Cisco Unified CM Administration

System version: 10.5.2.11900-3

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

Last Successful Backup: 0 day(s) ago

User administrator last logged in to this cluster on Monday, September 28, 2015 7:25:24 AM CDT, to node 10.80.14.2, from 172.16.31.250 using HTTPS

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

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

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

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



Cisco UCM Audio Codec Preference List

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

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

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

Save Delete Copy Add New

Audio Codec Preference List Information

Name* G729 Preferred Codec List

Description* G729 Preferred Codec List

Codecs in List*

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

Save Delete Copy Add New



Cisco UCM Region Configuration

Navigation Path: System → Region Information → Region

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 ▾

Region Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Region Information

Name * G729

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	G729 Preferred codec list	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
G729	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
Default G711 G729	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> kbps	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> kbps	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> kbps

Save Delete Reset Apply Config Add New



Device Pool Configuration

Navigation Path: System → Device Pool

“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
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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

Status
Status: Ready

Device Pool Information
Device Pool: G729 (18 members**)

Device Pool Settings

Device Pool Name *	G729
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 ▾
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](#)



Device Pool Configuration (Contd.)

Local Route Group Settings
Standard Local Route Group < None >

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

Geolocation Configuration
Geolocation < None >
Geolocation Filter < None >

Call Routing Information

Incoming Calling Party Settings

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

Clear Prefix Settings **Default Prefix Settings**

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

Device Pool Configuration (Contd.)

Incoming Called Party Settings

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

Clear Prefix Settings
Default Prefix Settings

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

Phone Settings

Caller ID For Calls From This Phone

Calling Party Transformation CSS
< None >

Connected Party Settings

Connected Party Transformation CSS
< None >

Redirecting Party Settings

Redirecting Party Transformation CSS
< None >

Save
Delete
Copy
Reset
Apply Config
Add New



Annunciator Configuration

Navigation: Media Resource → Annunciator

Set Name* = ANN_2.

Set Description = ANN_clus24pubsub. This is used for this example

Set Device Pool* = G729

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified administrator | Search

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

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

Save Reset Apply Config

Status

i Status: Ready

Annunciator Information

Registration: Registered with Cisco Unified Communications Manager
clus24pubsub.lab.tekvizion.com

IPv4 Address: 10.80.14.2

☒ Device is trusted

Server* clus24pubsub.lab.tekvizion.com ▾

Name* ANN_2

Description ANN_clus24pubsub

Device Pool* G729 ▾

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 = clus24pubsub.lab.tekvizion.com. This is used for this example.

Set Conference Bridge Name* = CFB_2.

Set Description = CFB_clus24pubsub. This is used in this example.

Set Device Pool* = G729

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation

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

Conference Bridge Configuration Back To Find/List Go

Save Reset Apply Config

Conference Bridge Information

Conference Bridge CFB_2 (CFB_clus24pubsub)
:
Registration: Registered with Cisco Unified Communications Manager
clus24pubsub.lab.tekvizion.com
IPv4 Address: 10.80.14.2

Software Conference Bridge Info

Conference Bridge Type* Cisco Conference Bridge Software
Host Server clus24pubsub.lab.tekvizion.com

⚠ Device is not trusted

Conference Bridge Name* CFB_2
Description CFB_clus24pubsub
Device Pool* G729 ▾
Common Device Configuration < None > ▾
Location* Hub_None ▾
Use Trusted Relay Point* Default ▾

Save Reset Apply Config



Media Termination Point Configuration

Navigation: Media Resource → Media Termination Point

Set Media Termination Point Name* = MTP_2

Set Description = MTP_clus24pubsub. This is used for this example

Set Device pool* = G729

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation **Cisco Unified**
administrator | Search

System > Call Routing > Media Resources > Advanced Features > Device > Application > Use

Media Termination Point Configuration Related Links: [Back To Find/List](#)

Status
 Status: Ready

Media Termination Point Information

Registration: Registered with Cisco Unified Communications Manager
clus24pubsub.lab.tekvizion.com

IPv4 Address: 10.80.14.2

Media Termination Point Type*: Cisco Media Termination Point Software

Host Server*: clus24pubsub.lab.tekvizion.com

Media Termination Point Name*:

Description:

Device Pool*:

☐ Trusted Relay Point

*- indicates required item.



Music on Hold Server Configuration

Navigation: Media Resources → Music on Hold Server

Set Music on Hold Server Name* = MOH_2.

Set Description = MOH_clus24pubsub. This is used for this example.

Set Device Pool* = G729

Music On Hold (MOH) Server Configuration

Save Reset Apply Config

Status

Status: Ready

Device Information

Registration: Registered with Cisco Unified Communications Manager clus24pubsub.lab.tekvizion.com

IPv4 Address: 10.80.14.2

☒ Device is trusted

Host Server* clus24pubsub.lab.tekvizion.com

Music On Hold Server Name* MOH_2

Description MOH_clus24pubsub

Device Pool* G729

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 mu-law are configured in parameter Supported MOH Codecs.

Select Server* = clus24pubsub.lab.tekvizion.com (Active). This is used in this example.

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

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 "Navigation", "administrator", "Search Documentation", and "About". Below this is a secondary navigation bar with tabs for "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Bulk Administration". The main content area is titled "Service Parameter Configuration" and includes a "Related Links" section with a dropdown for "Parameters for All Servers" and a "Go" button. The interface is divided into two main sections: "Status" and "Select Server and Service". The "Status" section shows "Status: Ready". The "Select Server and Service" section contains two dropdown menus: "Server*" set to "clus24pubsub.lab.tekvizion.com--CUCM Voice/Vide" and "Service*" set to "Cisco IP Voice Media Streaming App (Active)". Below these is a note: "All parameters apply only to the current server except parameters that are in the cluster-wide group(s)". The second section, "Clusterwide Parameters (Parameters that apply to all servers)", contains a table of parameters. The first row, "Supported MOH Codecs*", is highlighted with a red box and shows a list of codecs (711 mulaw, 711 alaw, 729 Annex A) with "711 mulaw" selected. Other parameters include "MOH Fixed Audio Quality level*" (Medium Quality), "IP DSCP to Cisco Unified Communications Manager*" (CS3(precedence 3) DSCP (011000)), "Multicast MOH IP DSCP*" (EF DSCP (101110)), "MTP DTMF Duration*" (100), and "MTP DTMF Power (volume)*" (9). At the bottom of this section is a note: "There are hidden parameters in this group. Click on Advanced button to see hidden parameters." The bottom of the interface features three buttons: "Save", "Set to Default", and "Advanced".

Service Parameter Configuration

Related Links: Parameters for All Servers Go

Save Set to Default Advanced

Status

Status: Ready

Select Server and Service

Server* clus24pubsub.lab.tekvizion.com--CUCM Voice/Vide

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)

<u>Supported MOH Codecs</u> *	711 mulaw 711 alaw 729 Annex A	711 mulaw
<u>MOH Fixed Audio Quality level</u> *	Medium Quality	Medium Quality
<u>IP DSCP to Cisco Unified Communications Manager</u> *	CS3(precedence 3) DSCP (011000)	CS3(precedence 3) DSCP (011000)
<u>Multicast MOH IP DSCP</u> *	EF DSCP (101110)	EF DSCP (101110)
<u>MTP DTMF Duration</u> *	100	100
<u>MTP DTMF Power (volume)</u> *	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* = clus24pubsub.lab.tekvizion.com (Active). This is used in this example.

Select Service* = Cisco Call Manager (Active).

Select Duplex Streaming Enabled * = True

Service Parameter Configuration

Save Set to Default Advanced

Status
 Status: Ready

Select Server and Service
Server* clus24pubsub.lab.tekvizion.com--CUCM Voice/Vider
Service* Cisco CallManager (Active)
All parameters apply only to the current server except parameters that are in the cluster-wide group(s).

Clusterwide Parameters (Service)

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
Intercluster Capabilities Mismatch Timer *	1000	1000
Silence Suppression *	False	False



Media Resource Group Configuration

Navigation Path: Media Resources → Media Resources group

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

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

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

Set all Resources in the selected Media Resources Box.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration
administrator | Search Documentation

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

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 20 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 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 items like System, Call Routing, Media Resources, Advanced Features, Device, Application, and User Management. The "Media Resources" menu is expanded, showing "Media Resource Group List Configuration". Below the navigation bar, there is a "Related Links" section with a "Back To Find/List" button and a "Go" button. The main content area has a toolbar with "Save", "Delete", "Copy", and "Add New" buttons. The "Media Resource Group List Status" section shows "Media Resource Group List: MRGL_MTP (used by 20 devices)". The "Media Resource Group List Information" section has a "Name*" field with the value "MRGL_MTP". The "Media Resource Groups for this List" section has two list boxes: "Available Media Resource Groups" and "Selected Media Resource Groups". The "Selected Media Resource Groups" list box contains the value "MRG_MTP". The bottom of the page has a footer with copyright information and a note about testing.

Media Resource Group List Configuration Related Links: [Back To Find/List](#) [Go](#)

Save **X** Delete Copy Add New

Media Resource Group List Status

Media Resource Group List: MRGL_MTP (used by 20 devices)

Media Resource Group List Information

Name* MRGL_MTP

Media Resource Groups for this List

Available Media Resource Groups

Selected Media Resource Groups MRG_MTP

Save Delete Copy Add New



UC Service Configuration

Navigation: User Management → User Settings → UC Service

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

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Find and List UC Services

Add New Select All Clear All Delete Selected

Status
 3 records found

UC Service (1 - 3 of 3) **Rows per Page** 50 ▾

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

<input type="checkbox"/>	Name ^	UC Service Type	Product Type	Host/IP Address	Port	Protocol
<input type="checkbox"/>	CTI_SRV	CTI	CTI	10.80.14.2	2748	TCP
<input type="checkbox"/>	IMP_SRV	IM and Presence	Unified CM (IM and Presence)	10.80.14.3		
<input type="checkbox"/>	Unity_Connection	Voicemail	Unity	10.80.14.4	443	HTTP

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 Client. This is used in this example.

Set Host Name/IP Address* = 10.80.14.2 (Cisco UCM Address)

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

Save Delete Copy Reset Apply Config Add New

Status
 Status: Ready

UC Service Information

UC Service Type:	CTI
Product Type:	CTI
Name*	CTI_SRV
Description	CTI for Jabber Client
Host Name/IP Address*	10.80.14.2
Port	2748
Protocol:	TCP

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



UC Service Configuration (Contd.)

Select UC Service Type: = IM and Presence

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

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

Set Host Name/IP Address* = 10.80.14.3 (Cisco UCM IM & Presence IP Address)

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

Save Delete Copy Reset Apply Config Add New

Status

Status: Ready

UC Service Information

UC Service Type: **IM and Presence**

Product Type*

Name*

Description

Host Name/IP Address*

*- indicates required item.



UC Service Configuration (Contd.)

Select UC Service Type: = Voicemail

Set Name* = Unity Connection. This is used in this example.

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

Set Host Name/IP Address* = 10.80.14.4 (Cisco Unity Connection Administration)

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation **Cisco Unified**
administrator | Search

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ Use

UC Service Configuration Related Links: [Back To Find/List](#)

Status
 Status: Ready

UC Service Information

UC Service Type:	Voicemail
Product Type*	<input type="text" value="Unity"/>
Name*	<input type="text" value="Unity Connection"/>
Description	<input type="text" value="Voicemail"/>
Host Name/IP Address*	<input type="text" value="10.80.14.4"/>
Port	<input type="text" value="443"/>
Protocol	<input type="text" value="HTTP"/>

*- indicates required item.



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

Save Delete Copy Add New

Status
 Status: Ready

Service Profile Information

Name*

Description

☒ Make this the default service profile for the system

Voicemail Profile

Primary ▾

Secondary ▾

Tertiary ▾

[Credentials source for voicemail service](#)* ▾

MailStore Profile

Primary ▾

Secondary ▾

Tertiary ▾

[Inbox Folder](#)*

[Trash Folder](#)*

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

☒ [Allow dual folder mode](#)



Service Profile Configuration (Contd.)

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

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

IM and Presence Profile
Primary IMP_SRV ▾
Secondary <None> ▾
Tertiary <None> ▾

CTI Profile
Primary CTI_SRV ▾
Secondary <None> ▾
Tertiary <None> ▾



Service Profile Configuration (Contd.)

Video Conference Scheduling Portal Profile

Primary

<None> ▾

Secondary

<None> ▾

Tertiary


<None> ▾

Save

Delete

Copy

Add New

 *- indicates required item.

▾



End User Configuration

Navigation: User Management → End User

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

Set Password = Password for profile.

Set Directory URI = jabber1@lab.tekvizion.com


The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and a navigation menu with options like 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', and 'Bulk Administration'. The 'User Management' menu is expanded, showing 'Find and List Users'. Below this, there are buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'. A status box indicates '1 records found'. The main table shows a single user entry with the following details:

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

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





End User Configuration (Contd.)

**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 ▾

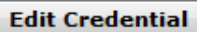
End User Configuration Related Links: Back to Find List Users ▾ Go

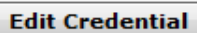
Save  Delete  Add New

User Information

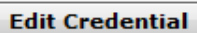
User Status Enabled Local User

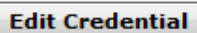
User ID* jabber1

Password 

Confirm Password 

Self-Service User ID

PIN 

Confirm PIN 

Last name* cisco

Middle name

First name

Title

Directory URI jabber1@lab.tekvizion.com

Telephone Number

Home Number

Mobile Number

Pager Number

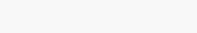
Mail ID

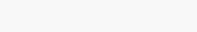
Manager User ID

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 (Contd.)

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
SEP00083031F2A8

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 (Contd.)

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 Admin Rep Tool Admin Standard Audit Log Administration Standard CCM Admin Users Standard CCM End Users Standard CCM Gateway Management	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 = SCCP_Chen_7965. This text is used to identify this Phone.

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

Set Phone Button Template* = Standard 7965 SCCP. This is used in this example.

Set Soft key 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 panels. The left panel, titled "Association", lists 18 items for configuration, including "Line [1] - 0461 (no partition)", "Line [2] - Add a new DN", "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", "Conference", "Do Not Disturb", "End Call", and "Forward All". The right panel, titled "Phone Type", shows the configuration for a "Cisco 7965" device using the "SCCP" protocol. The "Real-time Device Status" section indicates the device is registered with the Cisco Unified Communications Manager and provides details on its IPv4 address, active load ID, and download status. The "Device Information" section includes checkboxes for "Device is Active" and "Device is trusted", and a list of configuration fields: "MAC Address*", "Description", "Device Pool*", "Common Device Configuration", "Phone Button Template*", "Softkey Template", "Common Phone Profile*", and "Calling Search Space". The fields "MAC Address*", "Description", "Device Pool*", "Phone Button Template*", and "Softkey Template" are highlighted with a red box.

Association	Phone Type
1 Line [1] - 0461 (no partition)	Product Type: Cisco 7965
2 Line [2] - Add a new DN	Device Protocol: SCCP
3 Add a new SD	Real-time Device Status
4 Add a new SD	Registration: Registered with Cisco Unified Communications Manager
5 Add a new SD	clus24pubsub.lab.tekvizion.com
6 Add a new SD	IPv4 Address: 172.16.31.130
7 Add a new SD	Active Load ID: term65.default
8 Add a new SURL	Download Status: None
9 Add a new BLF SD	Device Information
10 Add a new BLF Directed Call Park	Device is Active
11 CallBack	Device is trusted
12 Call Park	MAC Address*: 08CC68E9FAB3
13 Call Pickup	Description: SCCP_Chen_7965
14 Conference List	Device Pool*: G729
15 Conference	Common Device Configuration: < None >
16 Do Not Disturb	Phone Button Template*: Standard 7965 SCCP
17 End Call	Softkey Template: Standard User
18 Forward All	Common Phone Profile*: Standard Common Phone Profile
	Calling Search Space: < None >



Cisco IP Phone 7965 SCCP Configuration (Contd.)

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.

Phone Configuration

Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

18 Forward All

19 Group Call Pickup

20 Hold

21 Hunt Group Logout

22 [Intercom \[1\] - Add a new Intercom](#)

23 Malicious Call Identification

24 Meet Me Conference

25 Mobility

26 New Call

27 Other Pickup

28 Quality Reporting Tool

29 Redial

30 Remove Last Participant

31 Transfer

32 Video Mode

33 Queue Status

34 Privacy

35 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

Privacy* Default

Device Mobility Mode Default [View Current](#)

[Device Mobility Settings](#)

Owner ☐ User ☒ Anonymous (Public/Shared Space)

Owner User ID

Phone Personalization* Default

Personalization

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 >

☒ Retry Video Call as Audio

☐ Ignore Presentation Indicators (internal calls only)

☒ Allow Control of Device from CTI

☒ Logged Into Hunt Group



Cisco IP Phone 7965 SCCP Configuration (Contd.)

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

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

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



Cisco IP Phone 7965 SCCP Configuration (Contd.)

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

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

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

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

Secure Shell Information	
Secure Shell User	administrator
Secure Shell Password



Cisco IP Phone 7965 SCCP Configuration (Contd.)

Product Specific Configuration Layout		
	Parameter Value	Override Common Settings
<input type="checkbox"/> Disable Speakerphone		
<input type="checkbox"/> Disable Speakerphone and Headset		
Forwarding Delay*	Disabled	
PC Port *	Enabled	
Settings Access*	Enabled	<input type="checkbox"/>
Gratuitous ARP*	Disabled	
PC Voice VLAN Access*	Enabled	
Video Capabilities*	Disabled	<input type="checkbox"/>
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		<input type="checkbox"/>
Security Secret		
<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 Local Volume*	100	
Recording Tone Remote Volume*	50	
Recording Tone Duration		
Display On When Incoming Call*	Disabled	<input type="checkbox"/>
RTCP*	Disabled	<input type="checkbox"/>



Cisco IP Phone 7965 SCCP Configuration (Contd.)

	<p>"more" Soft Key Timer <input type="text" value="5"/></p> <p>Auto Call Select* <input type="button" value="Enabled"/></p> <p>Log Server <input type="text"/></p> <p>Advertise G.722 Codec* <input type="button" value="Use System Default"/></p> <p>Wideband Headset UI Control* <input type="button" value="Enabled"/></p> <p>Wideband Headset* <input type="button" value="Enabled"/></p> <p>Peer Firmware Sharing* <input type="button" value="Enabled"/></p> <p>Cisco Discovery Protocol (CDP): Switch Port* <input type="button" value="Enabled"/></p> <p>Cisco Discovery Protocol (CDP): PC Port* <input type="button" value="Enabled"/></p> <p>Link Layer Discovery Protocol - Media Endpoint <input type="button" value="Enabled"/></p>
	<p>Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port* <input type="button" value="Enabled"/></p> <p>Link Layer Discovery Protocol (LLDP): PC Port* <input type="button" value="Enabled"/></p> <p>LLDP Asset ID <input type="text"/></p> <p>LLDP Power Priority* <input type="button" value="Unknown"/></p> <p>Wireless Headset Hookswitch Control* <input type="button" value="Disabled"/></p> <p>IPv6 Load Server <input type="text"/></p> <p>IPv6 Log Server <input type="text"/></p> <p>802.1x Authentication* <input type="button" value="User Controlled"/></p> <p>Detect Unified CM Connection Failure* <input type="button" value="Normal"/></p> <p>Minimum Ring Volume* <input type="button" value="0-Silent"/></p>



Cisco IP Phone 7965 SCCP Configuration (Contd.)

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



Cisco IP Phone 7965 SCCP Configuration (Contd.)

Set Directory Number* = 0461. This is used in this example.

Set Description = Cisco 7965 Phone. 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 Admin ▾

Directory Number Configuration Related Links: [Configure Device \(SEP08CC68E9FAB3\)](#) Go

Save Delete Reset Apply Config Add New

Status
Status: Ready

Directory Number Information

Directory Number*	0461	<input type="checkbox"/> Urgent Priority
Route Partition	< None >	
Description	Cisco 7965 Phone	
Alerting Name	Cisco 7965 Phone	
ASCII Alerting Name	Cisco 7965 Phone	
External Call Control Profile	< None >	
<input checked="" type="checkbox"/> Allow Control of Device from CTI		
Associated Devices	SEP08CC68E9FAB3	
Edit Device Edit Line Appearance		
Dissociate Devices		



Cisco IP Phone 7965 SCCP Configuration (Contd.)

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

Directory URIs

Primary	URI	Partition	Advertise Globally via ILS	Remove
<input checked="" type="radio"/>		< None >	<input checked="" type="checkbox"/>	
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		< None >
<input checked="" type="checkbox"/> Retain this destination in the call forwarding history			



Cisco IP Phone 7965 SCCP Configuration (Contd.)

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 >

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

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



Cisco IP Phone 7965 SCCP Configuration (Contd.)

Set Display (Caller ID) = Cisco 7965-Phone. This is used in this example.

Set ASCII Display (Caller ID) = Cisco 7965-Phone. This is used in this example.

Set Line Text Label = Cisco 7965-Phone. This is used in this example.

Set External Phone Number Mask = 7323680461. This is used in this example.

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

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

Line 1 on Device SEP08CC68E9FAB3	
Display (Caller ID)	<input type="text" value="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)	<input type="text" value="Cisco 7965-Phone"/>
Line Text Label	<input type="text" value="Cisco7965-Phone"/>
External Phone Number Mask	<input type="text" value="7323680461"/>
Visual Message Waiting Indicator Policy*	Use System Policy
Audible Message Waiting Indicator Policy*	Default
Ring Setting (Phone Idle)*	Use System Default
Ring Setting (Phone Active)	Use System Default Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)	Use System Default



Cisco IP Phone 7965 SCCP Configuration (Contd.)

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

Multiple Call/Call Waiting Settings on Device SEP08CC68E9FAB3

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

Maximum Number of Calls*	4	
Busy Trigger*	2	(Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEP08CC68E9FAB3

<input checked="" type="checkbox"/> Caller Name
<input type="checkbox"/> Caller Number
<input type="checkbox"/> Redirected Number
<input checked="" type="checkbox"/> Dialed Number

Users Associated with Line

Associate End Users

Save Delete Reset Apply Config Add New

i *- indicates required item.

i **- Changes to Line or Directory Number settings require restart.



Cisco IP Phone 7975 SIP Configuration

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

Set Description = SIP_Chen_7975. This text is used to identify this Phone.

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

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

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

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

Set User Hold MOH Audio Source = 1-SampleAudioSource

Set Network Hold MOH Audio Source = 1-SampleAudioSource

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

administrator | Search Documentation | About | Logout

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

Phone Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Association

Modify Button Items

1 Line [1] - 2754 (no partition)
2 Line [2] - Add a new DN
3 Add a new SD
4 Add a new SD
5 Add a new SD
6 Add a new SD
7 Add a new SD
8 Add a new SD
----- Unassigned Associated Items -----
9 Add a new SD
10 Add a new BLF Directed Call Park
11 Do Not Disturb
12 Intercom [1] - Add a new Intercom
13 Add a new SURL
14 Add a new BLF SD
15 Call Park

Phone Type

Product Type: Cisco 7975
Device Protocol: SIP

Real-time Device Status

Registration: Registered with Cisco Unified Communications Manager
clus24pubsub.lab.tekvizion.com

IPv4 Address: 172.16.31.155
Active Load ID: SIP75.9-4-2-1S
Download Status: None

Device Information

☒ Device is Active
☒ Device is trusted

MAC Address* 00083031F49B
Description SIP_Chen_7975
Device Pool* G729 [View Details](#)
Common Device Configuration < None > [View Details](#)
Phone Button Template* Standard 7975 SIP
Softkey Template Standard User
Common Phone Profile* Standard Common Phone Profile [View Details](#)

17 CallBack
18 Conference List
19 Conference
20 End Call
21 Forward All
22 Group Call Pickup
23 Hold
24 Hunt Group Logout

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



Cisco IP Phone 7975 SIP Configuration (Contd.)

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

Save Delete Copy Reset Apply Config Add New	
25 Malicious Call Identification	AAR Group < None >
26 Meet Me Conference	User Locale < None >
27 Mobility	Network Locale < None >
28 New Call	Built In Bridge* Default
29 Other Pickup	Privacy* Off
30 Quality Reporting Tool	Device Mobility Mode* Default View Current
31 Redial	Owner <input type="radio"/> User <input checked="" type="radio"/> Anonymous (Public/Shared Space)
32 Remove Last Participant	Owner User ID
33 Transfer	Phone Personalization* Default
34 Queue Status	Services Provisioning* Default
35 Privacy	
36 None	

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 type="checkbox"/> Ignore Presentation Indicators (internal calls only)	
<input checked="" type="checkbox"/> Allow Control of Device from CTI	
<input checked="" type="checkbox"/> Logged Into Hunt Group	
<input type="checkbox"/> Remote Device	
<input type="checkbox"/> Protected Device****	
<input type="checkbox"/> Hot line Device*****	
<input type="checkbox"/> Require off-premise location	



Cisco IP Phone 7975 SIP Configuration (Contd.)

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

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 7975 - Standard SIP Non-Secure Profile
Rerouting Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile for ATT View Details
Digest User	< None >
<input type="checkbox"/> Media Termination Point Required	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Require DTMF Reception	

Certification Authority Proxy Function (CAPF) Information	
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
<input type="button" value="Generate String"/>	
Key Size (Bits)*	2048
Operation Completes By	2015 7 12 12 (YYYY:MM:DD:HH)
Certificate Operation Status:	None
Note: Security Profile Contains Addition CAPF Settings.	



Cisco IP Phone 7975 SIP Configuration (Contd.)

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

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

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

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



Cisco IP Phone 7975 SIP Configuration (Contd.)

Secure Shell Information	
Secure Shell User	administrator
Secure Shell Password

Product Specific Configuration Layout	
	Parameter Value
<input type="checkbox"/> Disable Speakerphone	
<input type="checkbox"/> Disable Speakerphone and Headset	
Forwarding Delay*	Disabled
PC Port *	Enabled
Settings Access*	Enabled
Gratuitous ARP*	Disabled
PC Voice VLAN Access*	Enabled
Auto Line Select *	Disabled
Web Access*	Disabled
Days Display Not Active	Sunday Monday Tuesday
Display On Time	07:30
Display On Duration	10:30
Display Idle Timeout	01:00
Span to PC Port*	Disabled
Logging Display*	PC Controlled

Load Server		<input type="checkbox"/>
Recording Tone*	Disabled	
Recording Tone Local Volume*	100	
Recording Tone Remote Volume*	50	
Recording Tone Duration		
Display On When Incoming Call*	Disabled	<input type="checkbox"/>
RTCP*	Disabled	<input type="checkbox"/>
"more" Soft Key Timer	5	



Cisco IP Phone 7975 SIP Configuration (Contd.)

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 Discover (LLDP-MED): Switch Port*	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled	<input type="checkbox"/>

LLDP Asset ID		
LLDP Power Priority*	Unknown	
Wireless Headset Hookswitch Control*	Disabled	
IPv6 Load Server		<input type="checkbox"/>
IPv6 Log Server		
802.1x Authentication*	User Controlled	<input type="checkbox"/>
Detect Unified CM Connection Failure*	Normal	<input type="checkbox"/>
Minimum Ring Volume*	0-Silent	
Headset Sidetone Level*	Default	
Headset Send Gain*	Default	
HTTPS Server*	http and https Enabled	<input type="checkbox"/>

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Note: Testing was conducted in tekVizion labs



Cisco IP Phone 7975 SIP Configuration (Contd.)

	Handset/Headset Monitor*	Enabled
	Headset Recording*	Disabled
	Switch Port Remote Configuration*	Disabled
	PC Port Remote Configuration*	Disabled
	Automatic Port Synchronization*	Disabled
	SSH Access*	Disabled
	LOGIN Access*	Enabled
	80-bit SRTCP*	Disabled
	Customer Support Use	

Save Delete Copy Reset Apply Config Add New



Cisco IP Phone 7975 SIP Configuration (Contd.)

Set Directory Number* = 2754. This is used in this example.

Set Description = Cisco 7975-Phone. This is used in this example.

Set Alerting Name = Cisco 7975-Phone. This is used in this example.

Set ASCII Alerting Name = Cisco 7975-Phone. This is used in this example.

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 "administrator", "Search Documentation", "About", and "Logout". Below the navigation bar, there's a breadcrumb trail: "System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help". The main heading is "Directory Number Configuration". To the right, there's a "Related Links" section with a link to "Configure Device (SEP00083031F49B)". Below the heading, there's a toolbar with icons for "Save", "Delete", "Reset", "Apply Config", and "Add New". The "Status" section shows "Status: Ready". The "Directory Number Information" section contains several fields: "Directory Number*" (2754), "Route Partition" (< None >), "Description" (Cisco 7975-Phone), "Alerting Name" (Cisco 7975-Phone), "ASCII Alerting Name" (Cisco 7975-Phone), and "External Call Control Profile" (< None >). There's a checkbox for "Allow Control of Device from CTI" which is checked. Below this, the "Associated Devices" list shows "SEP00083031F49B". To the right of this list are buttons for "Edit Device" and "Edit Line Appearance". At the bottom of this section is a "Dissociate Devices" list.

Directory Number*	2754	<input type="checkbox"/> Urgent Priority
Route Partition	< None >	
Description	Cisco 7975-Phone	
Alerting Name	Cisco 7975-Phone	
ASCII Alerting Name	Cisco 7975-Phone	
External Call Control Profile	< None >	
<input checked="" type="checkbox"/> Allow Control of Device from CTI		
Associated Devices	SEP00083031F49B	
Dissociate Devices		

The screenshot shows the "Directory Number Settings" section of the Cisco Unified CM Administration interface. It contains several dropdown menus: "Voice Mail Profile" (< None >), "Calling Search Space" (< None >), "BLF Presence Group*" (Standard Presence group), "User Hold MOH Audio Source" (< None >), "Network Hold MOH Audio Source" (< None >), and "Auto Answer*" (Auto Answer Off). There's a checkbox for "Reject Anonymous Calls" which is unchecked. Below this section is the "Enterprise Alternate Number" section with an "Add Enterprise Alternate Number" button. At the bottom is the "+E.164 Alternate Number" section with an "Add +E.164 Alternate Number" button.

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	
<input type="checkbox"/> Reject Anonymous Calls		

Enterprise Alternate Number

Add Enterprise Alternate Number

+E.164 Alternate Number

Add +E.164 Alternate Number



Cisco IP Phone 7975 SIP Configuration (Contd.)

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 >
Forward on CTI Failure	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Unregistered Internal	<input type="checkbox"/> or	<input type="text"/>	< None >

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



Cisco IP Phone 7975 SIP Configuration (Contd.)

Set Display (Caller ID) = Cisco 7975-Phone. This is used in this example.

Set ASCII Display (Caller ID) = Cisco 7975-Phone. This is used in this example.

Set Line Text Label = Cisco 7975-Phone. This is used in this example.

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

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

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

Line Settings for All Devices	
Hold Reversion Ring Duration (seconds)	Setting the Hold Reversion Ring Duration to zero will disable the feature
Hold Reversion Notification Interval (seconds)	Setting the Hold Reversion Notification Interval to zero will disable the feature
Party Entrance Tone*	Default

Line 1 on Device SEP00083031F49B	
Display (Caller ID)	Cisco 7975-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 7975-Phone
Line Text Label	Cisco 7975-Phone
External Phone Number Mask	7322162754
Visual Message Waiting Indicator Policy*	Use System Policy
Audible Message Waiting Indicator Policy*	Default
Ring Setting (Phone Idle)*	Use System Default



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

Ring Setting (Phone Active)	<input type="text" value="Use System Default"/>	Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)	<input type="text" value="Use System Default"/>	
Call Pickup Group Audio Alert Setting(Phone Active)	<input type="text" value="Use System Default"/>	
Recording Option*	<input type="text" value="Call Recording Disabled"/>	
Recording Profile	<input type="text" value="< None >"/>	
Recording Media Source*	<input type="text" value="Gateway Preferred"/>	
Monitoring Calling Search Space	<input type="text" value="< None >"/>	
<input checked="" type="checkbox"/> Log Missed Calls		

Multiple Call/Call Waiting Settings on Device SEP00083031F49B

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

Maximum Number of Calls*

Busy Trigger*

(Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEP00083031F49B

☒ Caller Name

☐ Caller Number

☐ Redirected Number

☒ Dialed Number

Users Associated with Line

Associate End Users

Save

Delete

Reset

Apply Config

Add New

*- indicates required item.

**- Changes to Line or Directory Number settings require restart.



Cisco IP Phone 9971 SIP Configuration

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

Set Description = SIP_Chen_9971. This text is used to identify this Phone.

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

Set Phone Button Template* = Standard 9971 SIP. This is used in this example.

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

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

Set User Hold MOH Audio Source = 1-SampleAudioSource.

Set Network Hold MOH Audio Source = 1-SampleAudioSource.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

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

Phone Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Association

Modify Button Items

- Line [1] - 2753 (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
- All Calls
- Add a new BLF Directed Call Park
- Call Park
- Call Pickup
- CallBack
- Group Call Pickup
- Hunt Group Logout
- Intercom [1] - Add a new Intercom
- Malicious Call Identification
- Meet Me Conference

Phone Type

Product Type: Cisco 9971
Device Protocol: SIP

Real-time Device Status

Registration: Registered with Cisco Unified Communications Manager
clus24pubsub.lab.tekvizion.com

IPv4 Address: 172.16.28.112
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* C07BCCA1B846
Description SIP_Chen_9971
Device Pool* G729 [View Details](#)
Common Device Configuration < None > [View Details](#)
Phone Button Template* Standard 9971 SIP
Softkey Template Standard User
Common Phone Profile* Standard Common Phone Profile [View Details](#)

Calling Search Space < None >
AAR Calling Search Space < None >
Media Resource Group List MRGL_MTP
User Hold MOH Audio Source 1-SampleAudioSource
Network Hold MOH Audio Source 1-SampleAudioSource
Location* Hub_None



Cisco IP Phone 9971 SIP Configuration (Contd.)

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

27	Record	AAR Group	< None >
28	Alerting Calls	User Locale	< None >
29	Queue Status	Network Locale	< None >
30	Privacy	Built In Bridge*	Default
31	None	Privacy*	Off
		Device Mobility Mode*	Default View Current Device Mobility Settings
		Owner	<input type="radio"/> User <input checked="" type="radio"/> Anonymous (Public/Shared Space)
		Owner User ID	

Phone Personalization*	Default
Services Provisioning*	Default
Phone Load Name	
Use Trusted Relay Point*	Default
BLF Audible Alert Setting (Phone Idle)*	Default
BLF Audible Alert Setting (Phone Busy)*	Default
Always Use Prime Line*	Default
Always Use Prime Line for Voice Message*	Default
Geolocation	< None >
Feature Control Policy	< None >
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)	
<input checked="" type="checkbox"/> Allow Control of Device from CTI	
<input checked="" type="checkbox"/> Logged Into Hunt Group	
<input type="checkbox"/> Remote Device	
<input type="checkbox"/> Protected Device****	
<input type="checkbox"/> Require off-premise location	

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



Cisco IP Phone 9971 SIP Configuration (Contd.)

Protocol Specific Information

Packet Capture Mode*

None

Packet Capture Duration

0

BLF Presence Group*

Standard Presence group

SIP Dial Rules

< None >

MTP Preferred Originating Codec*

711ulaw

Device Security Profile*

Cisco 9971 - Standard SIP Non-Secure Profile

Rerouting Calling Search Space

< None >

SUBSCRIBE Calling Search Space

< None >

SIP Profile*

Standard SIP Profile for ATT

[View Details](#)

Digest User

< None >

☐ 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

201571312 (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 (Contd.)

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

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

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

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

Secure Shell Information	
Secure Shell User	administrator
Secure Shell Password



Cisco IP Phone 9971 SIP Configuration (Contd.)

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

PC Voice VLAN Access*	Enabled	
Web Access*	Disabled	<input type="checkbox"/>
Show All Calls on Primary Line*	Disabled	
Days Display Not Active	Sunday Monday Tuesday	<input type="checkbox"/>
Display On Time	07:30	<input type="checkbox"/>
Display On Duration	10:30	<input type="checkbox"/>
Display Idle Timeout	01:00	<input type="checkbox"/>
HTTPS Server*	http and https Enabled	<input type="checkbox"/>
Enable Power Save Plus	Sunday Monday Tuesday	<input type="checkbox"/>
Phone On Time	00:00	<input type="checkbox"/>
Phone Off Time	24:00	<input type="checkbox"/>
Phone Off Idle Timeout*	60	<input type="checkbox"/>
<input type="checkbox"/> Enable Audible Alert		<input type="checkbox"/>



Cisco IP Phone 9971 SIP Configuration (Contd.)

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

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



Cisco IP Phone 9971 SIP Configuration (Contd.)

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"/>
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"/>



Cisco IP Phone 9971 SIP Configuration (Contd.)

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"/>

i *- indicates required item.

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

i ***Note: Security Profile Contains Addition CAPF Settings.

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

i *****Note: A custom Softkey template without supplementary service Softkeys must be used for a Hot line Device.



Cisco IP Phone 9971 SIP Configuration (Contd.)

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

Set Description = Cisco 9971-Phone. 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.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

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

Directory Number Configuration Related Links: [Configure Device \(SEPC07BBCA1B846\)](#) Go

Save Delete Reset Apply Config Add New

Status
Status: Ready

Directory Number Information

Directory Number*	2753	<input type="checkbox"/> Urgent Priority
Route Partition	< None >	
Description	Cisco 9971-Phone	
Alerting Name	Cisco 9971-Phone	
ASCII Alerting Name	Cisco 9971-Phone	

External Call Control Profile: < None >

☒ Allow Control of Device from CTI

Associated Devices: SEPC07BBCA1B846

[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 9971 SIP Configuration (Contd.)

Enterprise Alternate Number				
<input type="button" value="Add Enterprise Alternate Number"/>				
+E.164 Alternate Number				
<input type="button" value="Add +E.164 Alternate Number"/>				
Directory URIs				
Primary	URI	Partition	Advertise Globally via ILS	Remove
<input type="radio"/>	<input type="text"/>	< None >	<input checked="" type="checkbox"/>	<input type="button" value="Remove"/>
<input type="button" value="Add Row"/>				
PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing				
Advertised Failover Number <input type="text" value="< 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 >

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



Cisco IP Phone 9971 SIP Configuration (Contd.)

Set Display (Caller ID) = Cisco 9971-Phone. This is used in this example.

Set ASCII Display (Caller ID) = Cisco 9971-Phone. This is used in this example.

Set Line Text Label = Cisco 9971-Phone. This is used in this example.

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

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

Line 1 on Device SEPC07BBCA1B846	
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	7322162753
Visual Message Waiting Indicator Policy*	Use System Policy



Cisco IP Phone 9971 SIP Configuration (Contd.)

Audible Message Waiting Indicator Policy*	Default	▼
Ring Setting (Phone Idle)*	Use System Default	▼
Ring Setting (Phone Active)	Use System Default	▼
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		

Multiple Call/Call Waiting Settings on Device SEPC07BBCA1B846
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 SEPC07BBCA1B846
☒ Caller Name
☐ Caller Number
☐ Redirected Number
☒ Dialed Number

Users Associated with Line

Associate End Users

Save

Delete

Reset

Apply Config

Add New

*- indicates required item.

**- Changes to Line or Directory Number settings require restart.



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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration
administrator | Search Documentation

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

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

Save ✕ Delete Copy Reset Apply Config + Add New

SIP Trunk Security Profile Information

Name* ATT Non Secure SIP Trunk Profile

Description Non Secure SIP Trunk Profile authenticated by null String

Device Security Mode Non Secure ▾

Incoming Transport Type* TCP+UDP ▾

Outgoing Transport Type UDP ▾

☐ Enable Digest Authentication

Nonce Validity Time (mins)* 600

X.509 Subject Name

Incoming Port* 5060

☐ Enable Application level authorization

☐ Accept presence subscription

☐ Accept out-of-dialog refer**

☐ Accept unsolicited notification

☐ Accept replaces header

☐ Transmit security status

☐ Allow charging header

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

Save Delete Copy Reset Apply Config Add New



SIP Profile Configuration used by SIP trunk to Cisco UBE

Navigation: Device → Device Settings → SIP Profile

Set SIP profile Name * = Standard SIP Profile for ATT. This is used for this example

Check Disable Early Media on 180

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

Note*= Some PSTN network call prompters utilize early-media cut-through to offer menu options to the caller (DTMF select menu) before the call is connected. In order for Cisco UCM/Cisco UBE solution to achieve successful early-media cut-through, the Cisco UCM to Cisco UBE call leg must be enabled with SIP PRACK. To enable SIP PRACK on the Cisco UCM, the SIP Profile “SIP Rel1XX Options” setting must be set to “Send PRACK”.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

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

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

Save Delete Copy Reset Apply Config Add New

SIP Profile Information

Name* Standard SIP Profile for ATT

Description Standard SIP Profile for ATT

Default MTP Telephony Event Payload Type* 101

Early Offer for G.Clear Calls* Disabled

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

Version in User Agent and Server Header* Major And Minor

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

Confidential Access Level Headers* Disabled

☐ Redirect by Application

☒ Disable Early Media on 180

☐ Outgoing T.38 INVITE include audio mline

☐ Use Fully Qualified Domain Name in SIP Requests

☐ Assured Services SIP conformance



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

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

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

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



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

<input checked="" type="checkbox"/> Conference Join Enabled <input type="checkbox"/> RFC 2543 Hold <input checked="" type="checkbox"/> Semi Attended Transfer <input type="checkbox"/> Enable VAD <input type="checkbox"/> Stutter Message Waiting <input type="checkbox"/> MLPP User Authorization								
Normalization Script Normalization Script < None > <input type="checkbox"/> Enable Trace <table><thead><tr><th></th><th>Parameter Name</th><th>Parameter Value</th><th></th></tr></thead><tbody><tr><td>1</td><td><input type="text"/></td><td><input type="text"/></td><td><input type="button" value="+"/> <input type="button" value="-"/></td></tr></tbody></table>		Parameter Name	Parameter Value		1	<input type="text"/>	<input type="text"/>	<input type="button" value="+"/> <input type="button" value="-"/>
	Parameter Name	Parameter Value						
1	<input type="text"/>	<input type="text"/>	<input type="button" value="+"/> <input type="button" value="-"/>					
Incoming Requests FROM URI Settings Caller ID DN <input type="text"/> Caller Name <input type="text"/>								
Trunk Specific Configuration Reroute Incoming Request to new Trunk based on* Never RSVP Over SIP* Local RSVP Resource Priority Namespace List < None > <input type="checkbox"/> Fall back to local RSVP <div>SIP Rel1XX Options* Send PRACK if 1xx Contains SDP</div> Video Call Traffic Class* Mixed Calling Line Identification Presentation* Default Session Refresh Method* Invite Early Offer support for voice and video calls* Disabled (Default value) <input type="checkbox"/> Enable ANAT <input type="checkbox"/> Deliver Conference Bridge Identifier <input type="checkbox"/> Allow Passthrough of Configured Line Device Caller Information <input type="checkbox"/> Reject Anonymous Incoming Calls <input type="checkbox"/> Reject Anonymous Outgoing Calls <input type="checkbox"/> Send ILS Learned Destination Route String								



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

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. This is used for this example

Set Media Resource Group List = MRGL_MTP.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation dropdown menu. Below this, a secondary navigation bar lists various system components like System, Call Routing, Media Resources, etc. The main content area is titled "Trunk Configuration" and contains a "SIP Trunk Status" section and a "Device Information" section. The "Device Information" section is the primary focus, showing a form for configuring a SIP Trunk. Key fields are highlighted with red boxes: "Device Name*" (ATT_SIP_TRUNK), "Description" (ATT SIP Trunk to PSTN), "Device Pool*" (G729), and "Media Resource Group List" (MRGL_MTP). Below the "Device Information" section, there is a "Packet Capture Mode*" dropdown set to "None" and a "Packet Capture Duration" input field set to "0". A series of checkboxes are present, including "Media Termination Point Required", "Retry Video Call as Audio" (checked), "Path Replacement Support", "Transmit UTF-8 for Calling Party Name", "Transmit UTF-8 Names in QSIG APDU", "Unattended Port", and "SRTP Allowed". A note explains that SRTP Allowed requires TLS configuration for end-to-end security. Further down, there are dropdown menus for "Consider Traffic on This Trunk Secure*" (set to "When using both sRTP and TLS"), "Route Class Signaling Enabled*" (set to "Default"), and "Use Trusted Relay Point*" (set to "Default"). At the bottom, there are checkboxes for "PSTN Access" and "Run On All Active Unified CM Nodes".

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

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 16 hours 11 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
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

☐ Media Termination Point Required
☒ Retry Video Call as Audio
☐ Path Replacement Support
☐ Transmit UTF-8 for Calling Party Name
☐ Transmit UTF-8 Names in QSIG APDU
☐ Unattended Port
☐ SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure*: When using both sRTP and TLS
Route Class Signaling Enabled*: Default
Use Trusted Relay Point*: Default

☐ PSTN Access
☐ Run On All Active Unified CM Nodes



SIP Trunk to Cisco UBE Configuration (Contd.)

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

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

MLPP and Confidential Access Level Information

MLPP Domain < None >

Confidential Access Mode < None >

Confidential Access Level < None >

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type* Default

SIP Privacy* Default

Inbound Calls

Significant Digits* 4

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

☐ Redirecting Diversion Header Delivery - Inbound

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.

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

Incoming Called Party Settings

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

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



SIP Trunk to Cisco UBE Configuration (Contd.)

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

Set SIP Trunk Security Profile* = ATT Non Secure Sip Trunk Profile.

Set SIP Profile* = Standard SIP Profile for ATT. This is used in this example.

Connected Party Settings		
Connected Party Transformation CSS	< None >	
<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 checked="" 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	<input type="text"/>
Caller Name	<input type="text"/>
<input type="checkbox"/> Maintain Original Caller ID DN and Caller Name in Identity Headers	

SIP Information		
Destination		
<input type="checkbox"/> Destination Address is an SRV		
Destination Address	Destination Address IPv6	Destination Port
1* 10.80.14.5		5060
MTP Preferred Originating Codec*	711ulaw	
BLF Presence Group*	Standard Presence group	
SIP Trunk Security Profile*	ATT Non Secure SIP Trunk Profile	
Rerouting Calling Search Space	< None >	
Out-Of-Dialog Refer Calling Search Space	< None >	
SUBSCRIBE Calling Search Space	< None >	
SIP Profile*	Standard SIP Profile for ATT View Details	
DTMF Signaling Method*	No Preference	



SIP Trunk to Cisco UBE Configuration (Contd.)

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



SIP Trunk to Fax Gateway Configuration

Navigation: Device → Trunk

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

Set Description = SIP Trunk to FAX Gateway. This is used for this example

Set Device Pool* = G729. 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 Go

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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: 3 days 15 hours 37 minutes

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Trunk_SIP_Fax_Gateway
Description	SIP Trunk to Fax Gateway
Device Pool*	G729
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 (Contd.)

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

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

MLPP and Confidential Access Level Information

MLPP Domain < None >

Confidential Access Mode < None >

Confidential Access Level < None >

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type* Default

SIP Privacy* Default

Inbound Calls

Significant Digits* All

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

☐ Redirecting Diversion Header Delivery - Inbound



SIP Trunk to Fax Gateway Configuration (Contd.)

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.				
<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=" < None >"/>	<input checked="" type="checkbox"/>

Incoming Called Party Settings				
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.				
<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=" < None >"/>	<input type="checkbox"/>

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

Outbound Calls	
Called Party Transformation CSS	<input type="text" value=" < None >"/>
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	<input type="text" value=" < None >"/>
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
Calling Party Selection*	<input type="text" value=" Originator"/>
Calling Line ID Presentation*	<input type="text" value=" Default"/>
Calling Name Presentation*	<input type="text" value=" Default"/>
Calling and Connected Party Info Format*	<input type="text" value=" Deliver DN only in connected party"/>
<input type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	
Redirecting Party Transformation CSS	<input type="text" value=" < None >"/>
<input checked="" type="checkbox"/> 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 Fax Gateway Configuration (Contd.)

Set Destination Address = Set IP address of Fax Gateway.

Set SIP Trunk Security Profile* = ATT Non Secure Sip Trunk Profile.

Set SIP Profile* = Standard SIP Profile for ATT. This is used in this example.

SIP Information		
Destination		
<input type="checkbox"/> Destination Address is an SRV		
Destination Address	Destination Address IPv6	Destination Port
1 * 10.80.22.7		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 for ATT View Details	
DTMF Signaling Method*	No Preference	

Normalization Script		
Normalization Script < None >		
<input type="checkbox"/> Enable Trace		
Parameter Name	Parameter Value	
1		<input type="button" value="+"/> <input type="button" value="-"/>

Recording Information	
<input checked="" type="radio"/> None	
<input type="radio"/> This trunk connects to a recording-enabled gateway	
<input type="radio"/> This trunk connects to other clusters with recording-enabled gateways	

Geolocation Configuration	
Geolocation	< None >
Geolocation Filter	< None >
<input type="checkbox"/> Send Geolocation Information	



Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

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Find and List Route Patterns

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

Status
 5 records found

Route Patterns (1 - 5 of 5) Rows per Page 50 ▾

Find Route Patterns where Pattern ▾ begins with ▾ Find Clear Filter + -

<input type="checkbox"/>	Pattern ^	Description	Partition	Route Filter	Associated Device	Copy
<input type="checkbox"/>	*X!	Network Based call forwarding			ATT SIP TRUNK	
<input type="checkbox"/>	01191XXXXXXXXXX	International Calling			ATT SIP TRUNK	
<input type="checkbox"/>	7323680461				Trunk SIP Fax Gateway	
<input type="checkbox"/>	9.@				ATT SIP TRUNK	
<input type="checkbox"/>	9.XXX				ATT SIP TRUNK	

Add New Select All Clear All Delete Selected



Route Pattern Configuration (Contd.)

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 Numbering Plan* = NANP. This text is used to identify the Numbering Plan.

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

Check the Use Calling party's External Phone Number Mask check box, for external phone number to be used for calling line identification (CLID) on outgoing calls

All other values are default.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation menu with options like "Navigation", "Search Documentation", "About", and "Logout". Below this, a breadcrumb trail shows the path: "System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help". The main heading is "Route Pattern Configuration", with a "Related Links" section containing "Back To Find/List" and "Go".

The "Pattern Definition" section contains the following fields:

- 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 (with an "Edit" link)
- Route Option: ☒ Route this pattern, ☐ Block this pattern (with a "No Error" dropdown)
- 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

The "Calling Party Transformations" section includes:

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

The "Connected Party Transformations" section includes:

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



Route Pattern Configuration (Contd.)

Set Discard Digits = PreDot.

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



Route Pattern Configuration (Contd.)

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

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

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

All other values are default.

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

Cisco Unified CM Administration
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Route Pattern Configuration

Related Links: Back To Find/List | Go

Save | Delete | Copy | Add New

Status
Update successful

Pattern Definition

Route Pattern*	*X!
Route Partition	< None >
Description	Network-Based call forwarding
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	ATT_SIP_TRUNK (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error
Call Classification*	OffNet
External Call Control Profile	< None >

☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level* 0
☐ Require Client Matter Code



Route Pattern Configuration (Contd.)

Check the Use Calling party's External Phone Number Mask check box, for external phone number to be used for calling line identification (CLID) on outgoing calls

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

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

Called Party Transformations
Discard Digits < None >
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type* Cisco CallManager
Called Party Numbering Plan* Cisco CallManager

ISDN Network-Specific Facilities Information Element
Network Service Protocol -- Not Selected --
Carrier Identification Code
Network Service -- Not Selected -- Service Parameter Name < Not Exist > Service Parameter Value

Save Delete Copy Add New

i *- indicates required item.



Route Pattern Configuration (Contd.)

Set Route Pattern* = 7323680461. 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.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation menu with options like "Navigation", "administrator", "Search Documentation", "About", and "Logout". Below this is a secondary menu with categories like "System", "Call Routing", "Media Resources", etc. The main content area is titled "Route Pattern Configuration" and includes a "Related Links" section with "Back To Find/List" and "Go".

The "Pattern Definition" section contains the following fields and values:

- Route Pattern*: 7323680461
- 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 (with an [Edit](#) link)
- 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

The "Calling Party Transformations" section includes:

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



Route Pattern Configuration (Contd.)

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



Jabber Client Configuration

Navigation: Device → Phone

Select Phone Type* = Cisco Unified Client Services Framework

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

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

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

Select Phone Button Template* = Standard Client Services Framework.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

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

Phone Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

Association
Modify Button Items
1 7715 Line [1] - 2754 (no partition)
----- Unassigned Associated Items -----
2 7715 Line [2] - Add a new DN

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

Real-time Device Status
Registration: Unknown
IPv4 Address: None

Device Information
☒ Device is Active
☒ Device is trusted
Device Name* CSFUser1
Description CSFUser1
Device Pool* G729 [View Details](#)
Common Device Configuration < None > [View Details](#)
Phone Button Template* Standard Client Services Framework
Common Phone Profile* Standard Common Phone Profile [View Details](#)
Calling Search Space < None >



Jabber Client Configuration (Contd.)

Media Resource Group List = MRGL_MTP

Check Owner = User. This is used in this example.

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

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

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



Jabber Client Configuration (Contd.)

Number Presentation Transformation

Caller ID For Calls From This Phone

Calling Party Transformation < None >

CSS

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

Remote Number

Calling Party Transformation < None >

CSS

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

Protocol Specific Information

Packet Capture Mode* None

Packet Capture Duration 0

BLF Presence Group* Standard Presence group

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 for ATT [View Details](#)

Digest User < None >

☐ Media Termination Point Required

☐ Unattended Port

☐ Require DTMF Reception



Jabber Client Configuration (Contd.)

Certification Authority Proxy Function (CAPF) Information	
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
<input type="button" value="Generate String"/>	
Key Size (Bits)*	2048
Operation Completes By	2015 7 12 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None	
Note: Security Profile Contains Addition CAPF Settings.	

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

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

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



Jabber Client Configuration (Contd.)

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

Instant Messaging	
File Types to Block in File Transfer	<input type="checkbox"/>
URLs to Block in File Transfer	<input type="checkbox"/>



Jabber Client Configuration (Contd.)

Desktop Client Settings		
Automatically Start in Phone Control*	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

i *- indicates required item.

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

i ***Note: Security Profile Contains Addition CAPF Settings.

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

i *****Note: A custom Softkey template without supplementary service Softkeys must be used for a Hot line Device.



Voicemail Port Configuration

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

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

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

Find and List Voice Mail Ports

Add New Select All Clear All Delete Selected Reset Selected 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
Select item or enter search text ▾

<input type="checkbox"/>	Device Name ▲	Description	Device Pool	Device Security Mode	Calling Search Space	Extension	Partition	Status	IPv4 Address	Copy
<input type="checkbox"/>	CiscoUM1-VI1	VM Port	G729	Non Secure Voice Mail Port		2501		Registered with clus24pubsub.lab.tekvizion.com	10.80.14.4	
<input type="checkbox"/>	CiscoUM1-VI2	VM Port	G729	Non Secure Voice Mail Port		2502		Registered with clus24pubsub.lab.tekvizion.com	10.80.14.4	

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



Voicemail Port Configuration (Contd.)

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

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

Set Device Pool = G729

Set Directory Number* = 2501. This is used in this example.

Device Information	
Registration:	Registered with Cisco Unified Communications Manager clus24pubsub.lab.tekvizion.com
IPv4 Address:	10.80.14.4
<input checked="" type="checkbox"/> Device is trusted	
Port Name*	CiscoUM1-VI1
Description	VM Port
Device Pool*	G729
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*	2501
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* = 2511

Set Message Waiting Indicator* = On

Set Message Waiting Number* = 2512

Set Message Waiting Indicator* = Off

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation menu with options like "Navigation", "administrator", "Search Documentation", "About", and "Logout". Below this, a secondary menu lists various system components: "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help".

The main content area is titled "Find and List Message Waiting Numbers". It features a toolbar with icons for "Add New", "Select All", "Clear All", and "Delete Selected". Below the toolbar, a status box indicates "2 records found".

The "Message Waiting Numbers" table is shown with a "Rows per Page" dropdown set to 50. The table has columns for "Directory Number", "Description", "Partition", "Calling Search Space", and "Copy". Two records are listed:

	Directory Number	Description	Partition	Calling Search Space	Copy
<input type="checkbox"/>	2511	MWI ON			
<input type="checkbox"/>	2512	MWI OFF			

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



Voicemail Pilot Configuration

Navigation: Advanced Features → Voice Mail → Voice Mail Pilot

Set Voice Mail Pilot Number = 2302. This is used for this example

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

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

Find and List Voice Mail Pilots

+ Add New Select All Clear All Delete Selected

Status
4 records found

Voice Mail Pilot (1 - 4 of 4) Rows per Page 50 ▾

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

		Pilot Number ^	Description	Calling Search Space
<input type="checkbox"/>			Default	
<input type="checkbox"/>			No Voice Mail	
<input checked="" type="checkbox"/>		2302	Voice Mail	
<input type="checkbox"/>		2303	Voice Mail Pilot number with SIP	

Add New Select All Clear All Delete Selected

Voice Mail Pilot Information

Voice Mail Pilot Number 2302

Calling Search Space < None > ▾

Description Voice Mail

☒ Make this the default Voice Mail Pilot for the system

Save Delete Add New



FAX Gateway Configuration

FAX-GATEWAY2# **show 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)

FAX-GATEWAY2 uptime is 1 week, 6 days, 19 hours, 19 minutes

System returned to ROM by power-on

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

Last reload type: Normal Reload

Last reload reason: power-on

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

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EDCS# xxx Rev #

Page **120** of **151**

Note: Testing was conducted in tekVizion labs



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.

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 running-config**

Building configuration...

Current configuration : 8536 bytes

!

! Last configuration change at 11:58:36 UTC Wed Jul 1 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

rsa-keypair 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

no ip address trusted authenticate

address-hiding

mode border-element

allow-connections sip to sip

no supplementary-service sip handle-replaces

redirect ip2ip

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

sip

midcall-signaling passthru

g729 annexb-all

!

voice class codec 1

codec preference 1 g729r8

codec preference 2 g711ulaw

!

!

voice class sip-profiles 1

response ANY sip-header Allow-Header modify "UPDATE," ""



```
request ANY sip-header Allow-Header modify "UPDATE," ""
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

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

```
hw-module pvdn 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

no vad

shutdown

!

voice-port 0/1

no echo-cancel enable

no vad

cptone IN

station-id name fax test

station-id number 7323680461

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 7323680461  
no digit-strip  
port 0/1  
forward-digits all  
!  
dial-peer voice 200 voip  
description CUCM to Gateway  
service session  
session protocol sipv2  
session transport udp  
incoming called-number 7323680461  
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 9T
session protocol sipv2
session target ipv4:10.80.14.2
session transport udp
voice-class codec 1
voice-class sip profiles 1
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
!
gateway
timer receive-rtp 1200
!
sip-ua
!
!
!
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 10.5.2.11900-3



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Note: Testing was conducted in tekVizion labs



CUC Telephony Integration with Cisco UCM

Navigation: Telephony Integrations → Phone system

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

The screenshot shows the Cisco Unity Connection Administration interface. The left navigation menu is expanded to 'Telephony Integrations' > 'Phone System'. The main content area displays the 'Search Phone Systems' page. A status message indicates 'Found 1 Phone System(s)'. Below this, a table titled 'Phone Systems (1 - 1 of 1)' shows a single entry: 'Default' with a port count of 2. The 'Default' entry is highlighted with a red box. The table has columns for 'Display Name' and 'Port Count'. Below the table are buttons for 'Delete Selected' and 'Add New'.

Display Name	Port Count
Default	2



CUC Port Group

Navigation: Telephony Integration → Port Group

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

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

Cisco Unity

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

Search Port Groups Search Port Group

Related Links: Check Telephony Configuration Go

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 Find

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

Delete Selected Add New



CUC Port Group (Contd.)

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

Check Enable Message waiting indicators.

Set MWI On Extension = 2511. This is used in this example.

Set MWI Off Extension= 2512. This is used in this example.

The screenshot displays the Cisco Unity Connection Administration web interface. The left sidebar shows the navigation menu with 'Cisco Unity' expanded, and 'Port Group' selected under 'Telephony Integrations'. The main content area is titled 'Port Group Basics (CUCM-1)'. It includes a 'Port Group' section with fields for 'Display Name*' (CUCM-1), 'Integration Method' (SCCP (Skinny)), and 'Device Name Prefix*' (CiscoUM1-VI). Below this is the 'Message Waiting Indicator Settings' section, which has a red box highlighting the 'Enable Message Waiting Indicators' checkbox (checked), 'MWI On Extension' (2511), and 'MWI Off Extension' (2512). Other settings include 'Delay between Requests' (0 milliseconds), 'Maximum Concurrent Requests' (0), 'Retries After Successful Attempt' (0), and 'Retry Interval After Successful Attempt' (5 milliseconds). Navigation buttons (Save, Delete, Previous, Next) are present at the top and bottom of the configuration area. A note at the bottom states: 'Fields marked with an asterisk (*) are required.'



CUC Port Settings

Navigation: Telephony Integration → Port

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

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

Cisco Unity

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

Search Ports

Search Port

Related Links **Check Telephony Configuration** **Gg**

Port Refresh Help

Status

i Found 2 Port(s)

Port (1 - 2 of 2)

Rows per Page 25

Find Port where **Display Name** begins with **Find**

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

Delete Selected **Add New**



CUC Sample User Basic Settings

Navigation: Cisco Unity Connection → Users → Users

Set Alias = 0461. This is one of the extension used for this testing.

Set Extension = 0461. This is used for this example.

Cisco Unity

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

Edit User Basics (0461) Search Users Edit User Basics (0461)
Related Links Bulk Edit By CSV Go

User Edit Refresh Help

Save Delete Previous Next

Name

Alias* 0461

First Name

Last Name

Display Name 0461

SMTP Address 0461 @clus24-unity.lab.tekvizion.com

Initials

Title

Employee ID

LDAP Integration Status

☐ Integrate with LDAP Directory

☒ Do Not Integrate with LDAP Directory

Phone

Extension* 0461

Cross-Server Transfer

Extension or URI

Outgoing Fax Number



CUC Sample User Basic Settings (Contd.)

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

Select Search Space = clus24-unity Search Space.

Select Phone System = Default.

Cisco Unity <ul style="list-style-type: none">Users<ul style="list-style-type: none">UsersImport UsersSynch UsersClass of ServiceTemplatesContactsDistribution ListsCall ManagementMessage StorageNetworkingUnified MessagingVideoDial PlanSystem SettingsTelephony IntegrationsTools	Outgoing Fax Server --- Not Selected --- Partition clus24-unity Partition Search Scope clus24-unity Search Space Phone System Default Class of Service Voice Mail User COS Active Schedule All Hours 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 Caution! Security risk. See Help for This Page for details. <input type="checkbox"/> Use Short Calendar Caching Poll Interval Recorded Name Play/Record Location Address Building City State Postal Code Country United States
	<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 Manager Billing ID Corporate Email Address <input type="checkbox"/> Generate SMTP Proxy Address From Corporate Email Address Directory URI Corporate Phone Number Save Delete Previous Next Fields marked with an asterisk (*) are required.



Auto Attendant

Navigation: Call Management → System Call Handlers

Set Display Name = Demo Auto Attendant. This is used for this example.

Set Phone System = Default

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

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

Set Search Space = clus24-unity Search Space.

Cisco Unity Connection

- 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

Search Call Handlers Search Call Handlers

Call Handler Refresh Help

Status

Found 5 System Call Handler(s)

Search Limits

Limit search to All

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

Find System where Display Name begins with Find

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

Delete Selected Add New Bulk Edit Show Dependencies



Auto Attendant (Contd.)

Cisco Unity Connection

⊕ 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

It is recommended you backup report data prior to renaming the Call Handler Display Name

Call Handler

Display Name* Demo Auto Attendant

Creation Time 2015-04-24 04:47:51.249

Phone System Default

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 clus24-unity Partition

Recorded Name Play/Record

⊕ Unified Messaging

⊕ Video

⊕ Dial Plan

⊕ System Settings

⊕ Telephony Integrations

⊕ Tools

Search Scope

☒ Search Space clus24-unity Search Space

☐ Inherit Search Space from Call

Save Delete Previous Next

Fields marked with an asterisk (*) are required.
All dates and times displayed in (GMT-06:00) Central Time (US & Canada)



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

CUP/IMP Version

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

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

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

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 Wednesday, July 1, 2015 2:21:40 AM CDT, to node 10.80.14.3, from 172.16.29.182 using HTTPS

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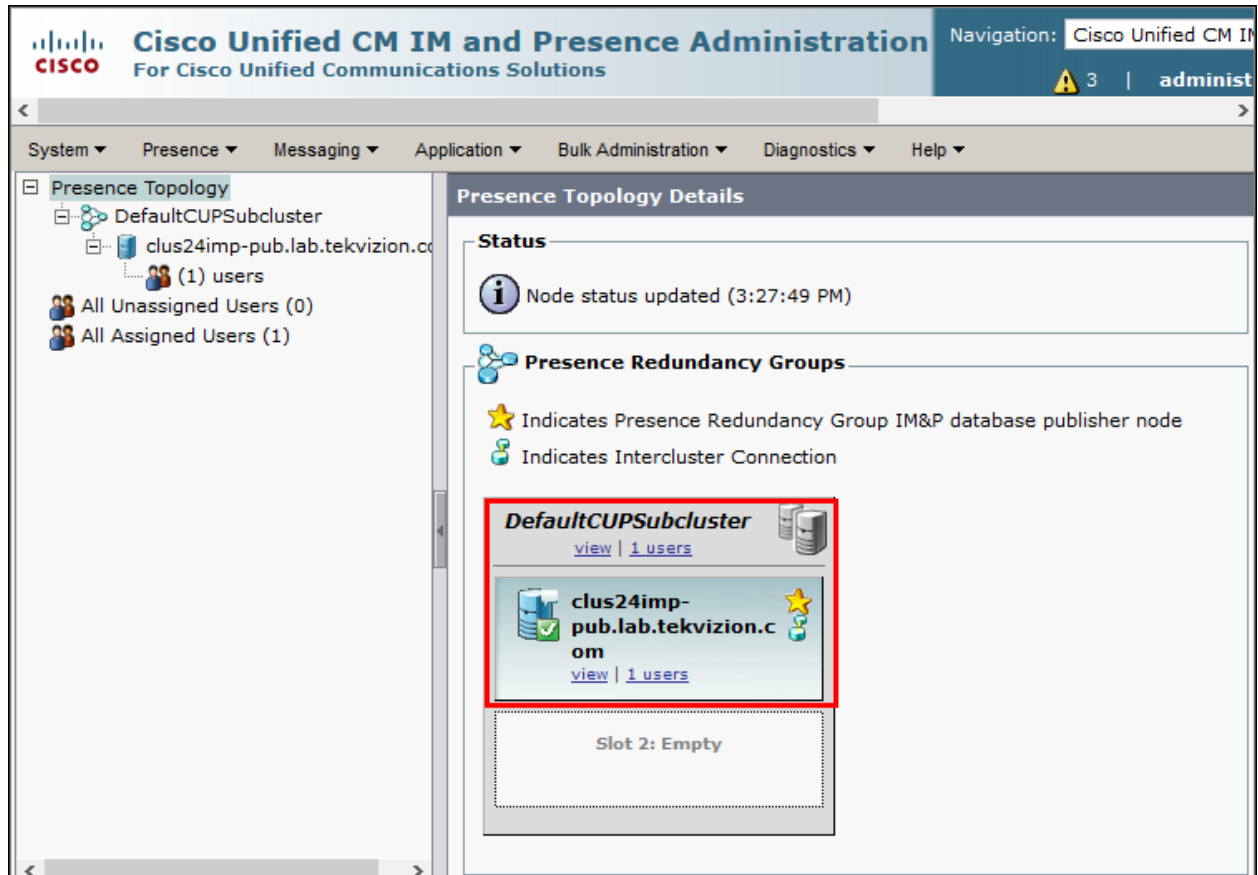
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Note: Testing was conducted in tekVizion labs

Presence Topology

Navigation: System → Presence Topology



The screenshot displays the Cisco Unified CM IM and Presence Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM IM and Presence Administration", and the subtitle "For Cisco Unified Communications Solutions". The right side of the navigation bar shows "Navigation: Cisco Unified CM IM" and a user profile "administ". Below the navigation bar is a menu with options: System, Presence, Messaging, Application, Bulk Administration, Diagnostics, and Help. The main content area is divided into two sections. On the left is a tree view under "Presence Topology" showing "DefaultCUPSubcluster" with a sub-entry "clus24imp-pub.lab.tekvizion.com" (1 users). Below this are "All Unassigned Users (0)" and "All Assigned Users (1)". The right section is titled "Presence Topology Details". It contains a "Status" section with a message "Node status updated (3:27:49 PM)". Below that is a "Presence Redundancy Groups" section with a legend: a star icon for "Indicates Presence Redundancy Group IM&P database publisher node" and a connection icon for "Indicates Intercluster Connection". A red box highlights a cluster entry for "DefaultCUPSubcluster" with a "view" link and "1 users". Below this, a specific node "clus24imp-pub.lab.tekvizion.com" is shown with a checkmark icon, a star icon, and a connection icon, along with its own "view" link and "1 users" count. At the bottom of the cluster entry, it says "Slot 2: Empty".



Node Configuration

Navigation: System → Presence Topology → Fully Qualified Domain Name


The screenshot displays the Cisco Unified CM IM and Presence Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM IM and Presence Administration For Cisco Unified Communications Solutions", and a navigation path: "Navigation: Cisco Unified CM IM and Presence Administration". Below this, a secondary navigation bar shows tabs for System, Presence, Messaging, Application, Bulk Administration, Diagnostics, and Help. The left sidebar shows the "Presence Topology" tree with "DefaultCUPSubcluster" expanded, and "clus24imp-pub.lab.tekvizion.com" selected. The main content area is titled "Node Detail" and shows the configuration for the selected node. The "Status" section indicates "Node status updated (3:29:03 PM)". The "Node Configuration" section lists the following details:

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



Users

Navigation: System → Presence Topology → clus24imp-pub.lab.tekvizion.com → Users

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

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

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

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

Status
1 records found

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

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

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



Presence gateway configuration

Navigation: Presence → Gateways

Set Presence Gateway Type *= CUCM

Set Description *= Cluster 24. This is used for this example.

Presence Gateway *= clus24pubsub.lab.tekvizion.com

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

Navigation: Cisco Unified CM IM | 3 | administr

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

Presence Gateway Configuration Related Links: Back To Find/List ▾ Go

Save ✕ Delete + Add New

Status
i Add successful

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* CUCM ▾
Description* Cluster 24
Presence Gateway* clus24pubsub.lab.tekvizion.com

Save Delete Add New



Acronyms

AVPN	AT&T Virtual Private Network
CODEC	Coder-Decoder (in this document a device used to digitize and undigitize voice signals)
Cisco UBE	Cisco Unified Border Element
Cisco UCM	Cisco Unified Communications Manager
IP	Internet Protocol
ISR	Integrated Services Router
MGCP	Media Gateway Control Protocol
MIS	Managed Internet Services
PNT	Private Network Transport
PSTN	Public Switched Telephone Network
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol
SP	Service Provider
TDM	Time-Division Multiplexing



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