Application Note

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

JAN 2016

Note: Testing was conducted in tekVizion labs

Table of Contents

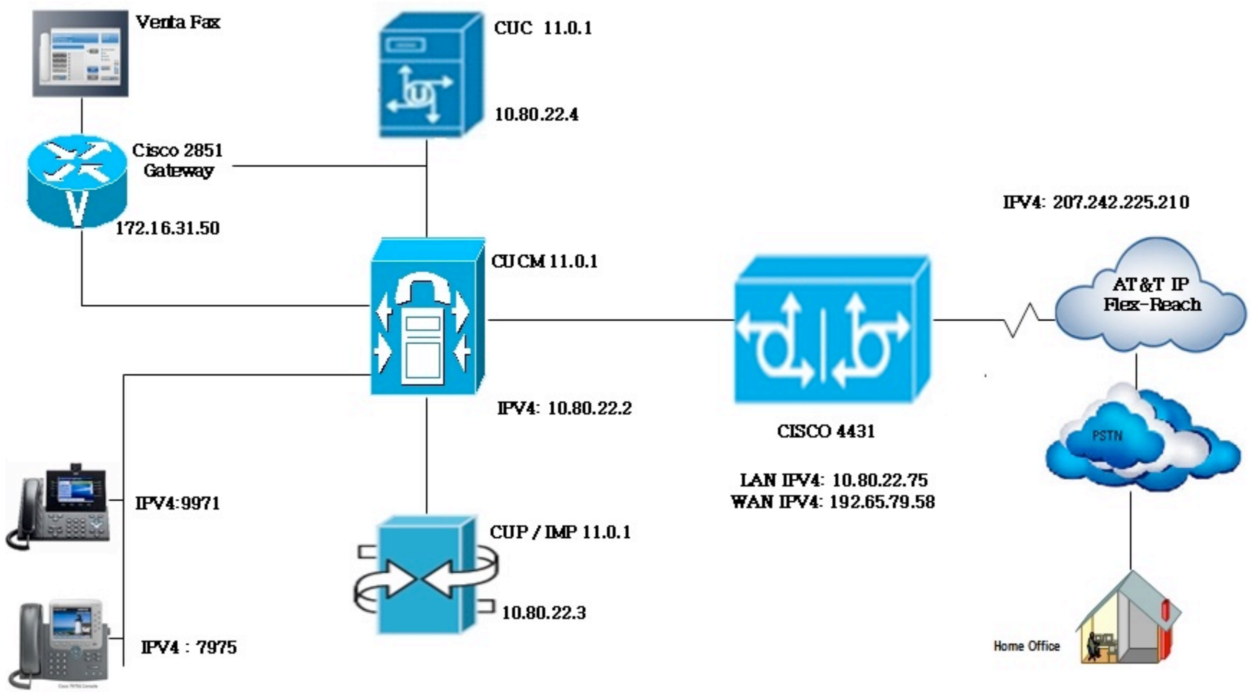
Introduction

Service Providers today, such as AT&T, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. AT&T IP Flexible Reach is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity via either analog or T1 lines. A demarcation device between these services and customer owned services is recommended. The Cisco Unified Border Element (Cisco UBE) provides demarcation, security, interworking and session management services.

- This application note describes the necessary steps and configurations of Cisco Unified Communications Manager (Cisco UCM) 11.0, Cisco Unity Connection 11.0, Cisco Unified CM IM and Presence 11.0, Cisco Integrated Services Routers (ISR) Version 15.5(3)S1a with connectivity to AT&T's IP Flex-Reach SIP trunk service. It also covers support and configuration example of Cisco Unity Connection (CUC) messaging integrated with Cisco Unified Communications Manager (Cisco UCM). The deployment model covered in this application note is Cisco Integrated Services Routers (ISR) to PSTN (AT&T IP Flexible Reach SIP). AT&T IP Flexible Reach provides inbound and outbound call service.

- Testing was performed in accordance to AT&T's IP Flexible Reach test plan and all features were verified. Key features verified are: inbound and outbound basic call (including international calls), calling name delivery, calling number and name restriction, CODEC negotiation, intra-site transfers, intra-site conferencing, call hold and resume, call forward (forward all, busy and no answer), leaving and retrieving voicemail (Cisco Unity Connection), CISCO auto-attendant (BACD), fax G.711 and T38 (G3 and SG3 speeds), teleconferencing, failover of unresponsive SIP network to PSTN and outbound/inbound calls to/from TDM networks.
- The Cisco Unified Border Element function configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between AT&T SIP network and Cisco Integrated services router. The configurations described in this document details the important commands for successful interoperability. Care must be taken by the network administrator deploying Cisco ISR to ensure these commands are set per each dial-peer required, to interoperate done AT&T SIP network.
- Consult your Cisco representative for the correct IOS image and for the specific application and Device Unit License and Feature License requirements for all your Cisco Unified Communication Manager with Cisco Unified Border Element components.

Network Topology



Note: Testing was conducted in tekVizion labs

Hardware Components

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

Software Requirements

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

Features

Features – Supported

- Basic Call using G.729 and G711
- Calling Party Number Presentation and Restriction
- Calling Name Presentation
- AT&T Advanced 8YY Call Prompter (8YY)
- Cisco UBE Delayed-Offer-to-Early-Offer conversion of an initial SIP INVITE without SDP
- Intra-site Call Transfer
- Intra-site Conference
- Call Hold and Resume
- Call Forward All, Busy and No Answer
- AT&T IP Teleconferencing
- Fax over G.711
- Fax using T.38
- Incoming DNIS Translation and Routing
- Outbound calls to AT&T's IP and TDM networks
- Inbound calls from AT&T's IP and TDM networks
- CPE voicemail managed service, leave and retrieve voice messages via incoming AT&T SIP trunk (Cisco Unity Connection)
- Auto-attendant transfer-to service (See Caveat section for details)
- Failover (From non-responsive SIP network to ATT SIP network)
- Inbound & Outbound Calls using Cisco Jabber
- Emergency and 411 calls were terminated to a voicemail platform in lab environment within AT&T for test
- RTCP

Network Based Features - Supported

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

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

Features - Not Supported

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

Caveats

Auto-Attendant

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

Hold/Resume & Music on Hold (MOH)

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

Ringback Tone on Early Unattended Transfer

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

PBX Based Call Forward Unconditional

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

SIP Provisional Acknowledgement/Early media

- To play early media sent by ATT, Cisco UCM needs to be enabled with PRACK if 1XX contains SDP on Cisco UCM SIP Profile.
- Some PSTN network call prompters utilize early-media cut-through to offer menu options to the caller (DTMF select menu) before the call is connected. In order for Cisco UCM/Cisco UBE solution to achieve successful early-media cut-through, the Cisco UCM to Cisco UBE call leg must be enabled with SIP PRACK. To enable SIP PRACK on the Cisco UCM, the SIP Profile “SIP Rel1XX Options” setting must be set to “Send PRACK”. The SIP Profile is found under Device>Device Settings>SIP Profile, This feature can be assigned on a per SIP trunk basis using SIP profiles. SIP PRACK provisioning on Cisco UCM 9.X and newer software versions is enabled under SIP Profile configuration page, while SIP PRACK support on Cisco UCM 7.X and older software versions is enabled under the Service Parameters configuration page.

AT&T IP Teleconferencing (IPTC)

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

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

Configuration Considerations

- To enable conference on AT&T IP Flexible Reach and Cisco UCM SIP trunk, it is required to configure a conference bridge (CFB) resource to initiate a three-way conference between end-points. See configuration section for details.
- Forwarded calls from Cisco UCM user to PSTN (out to AT&T's IP Flexible Reach service), AT&T serviced areas require that the SIP Diversion header contain the full 10-digit DID number of the forwarding party. In this application note the assumption has been made that a typical customer will utilize extension numbers (4-digit assignments in this example) and map 10-digit DID number using Cisco UBE translation profile. This is because the Cisco UCM uses 4-digit extensions on Cisco UCM IP phones and it is necessary to expand the 4-digit extension included in the Diversion header of a forwarding INVITE message to its full 10-digit DID number when the IP phone is set to call-forward. The requirement to expand the Diversion-Header has been achieved by the use of a SIP profile in Cisco UBE (See configuration section for details).
- Upon receiving inbound calls, AT&T SIP network will always have the first choice codec presented in the initial SIP INVITE (unless the end-device does not support the listed preferred codec), and processes calls accordingly. Customers wishing to place/receive G.711-only calls must configure separate voice class codec on Cisco UBE with G.711 as the first choice.
- SIP Profiles may also be employed to advertise desired RTP payload packet size.
- "voice-class sip privacy id" needs to configure in Cisco UBE dial peer to make call From a CPE Phone to PSTN phone, Pass Calling Party Number (CPN), marked private.
- This test environment is not configured with Cisco UBE High Availability (HA)
- Cisco UCM sends a SIP UPDATE message to Cisco UBE for a call transfer. AT&T network does not support the SIP UPDATE message causing the Cisco UBE to timeout and the call transfer is not completed. As a workaround, SIP profile has been applied on the Cisco UBE to remove UPDATE from the allowed headers (See configuration section for details).

Emergency 911/E911 Services Limitations and Restrictions

- Emergency 911/E911 Services Limitations and Restrictions - Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer configuration guide will properly operate with AT&T IP Flexible Reach to complete 911/E911 calls; therefore, it is Customer's responsibility to ensure proper operation with its equipment/software vendor
- While AT&T IP Flexible Reach services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found at http://new.serviceguide.att.com. Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power and delays that may occur in updating the Customer's location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Guide in detail to understand the limitations and restrictions

ISR Configuration

CISCO_4K_ROUTER2#**show version**

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

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

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

Copyright (c) 1986-2015 by Cisco Systems, Inc.

Compiled Wed 04-Nov-15 12:50 by mcpre

Cisco IOS-XE software, Copyright (c) 2005-2015 by cisco Systems, Inc.

All rights reserved. Certain components of Cisco IOS-XE software are

licensed under the GNU General Public License ("GPL") Version 2.0. The

software code licensed under GPL Version 2.0 is free software that comes

with ABSOLUTELY NO WARRANTY. You can redistribute and/or modify such

GPL code under the terms of GPL Version 2.0. For more details, see the

documentation or "License Notice" file accompanying the IOS-XE software,

or the applicable URL provided on the flyer accompanying the IOS-XE

software.

ROM: IOS-XE ROMMON

CISCO_4K_ROUTER2 uptime is 2 weeks, 5 days, 8 hours, 17 minutes

Uptime for this control processor is 2 weeks, 5 days, 8 hours, 18 minutes

System returned to ROM by reload

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

Last reload reason: Reload Command

This product contains cryptographic features and is subject to United

States and local country laws governing import, export, transfer and

use. Delivery of Cisco cryptographic products does not imply

third-party authority to import, export, distribute or use encryption.

Importers, exporters, distributors and users are responsible for

compliance with U.S. and local country laws. By using this product you

agree to comply with applicable laws and regulations. If you are unable

to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:

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

If you require further assistance please contact us by sending email to

export@cisco.com.

Suite License Information for Module:'esg'

Suite Suite Current Type Suite Next reboot

FoundationSuiteK9 None None None

securityk9

appxk9

AdvUCSuiteK9 None None None

uck9

cme-srst

cube

Technology Package License Information:

Technology Technology-package Technology-package

Current Type Next reboot

----

appxk9 None None None

uck9 uck9 Evaluation uck9

securityk9 None None None

ipbase ipbasek9 Permanent ipbasek9

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

Processor board ID FTX1850ALVU

4 Gigabit Ethernet interfaces

32768K bytes of non-volatile configuration memory.

4194304K bytes of physical memory.

7057407K bytes of flash memory at bootflash:.

Configuration register is 0x2102

CISCO_4K_ROUTER2#**show running-config**

Building configuration...

Current configuration : 11328 bytes

!

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

!

version 15.5

service timestamps debug datetime msec

service timestamps log datetime msec

no platform punt-keepalive disable-kernel-core

!

hostname CISCO_4K_ROUTER2

!

boot-start-marker

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

boot-end-marker

!

!

```
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
enable secret 5 $1$zQRB$CCbzfD1aYzk3kPvzAm2KU0
enable password cisco
!
aaa new-model
!
!
!
!
!
!
!
aaa session-id common
!
!
!

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

```
!
!
!
!
!
!
!
!
!
!
subscriber templating
multilink bundle-name authenticated
!
!
!
!
!
!
!
cts logging verbose
!
!
voice service voip
 rtp-port range 16384 32766
 address-hiding [1]
```

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

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

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

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

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

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

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

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

[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.

codec preference 1 g729r8 bytes 30

codec preference 2 g711ulaw

!

voice class codec 2

codec preference 1 g711ulaw

codec preference 2 g729r8 bytes 30

!

voice class codec 3

codec preference 1 g711ulaw

!

!

voice class sip-profiles 1

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

request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:732320\1@\2>" [10]

request INVITE sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30" [11]

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

request INVITE sdp-header Audio-Attribute add "a=ptime:30" [12]

!

!

---

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

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

[11] Cisco 6900-series IP phones use ptime value of 20 ms. AT&T networks prefer ptime value of 30 ms. This SIP profile modifies SDP ptime value from 20 to 30 ms and it should be applied to dial-peers where G729 is the preferred codec. If the customer creates a dial-peer specifically for G711, a sip-profile without modifying the ptime value should be applied. This is because G711 RTP was not defaulting to 20ms.

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

```
!

!

!

voice translation-rule 1 [13]

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

!

!

!

voice translation-profile NPA

translate calling 1

!

!

!

!

license udi pid ISR4431/K9 sn FOC18232988

license boot level appxk9

license boot level uck9

!

spanning-tree extend system-id

!

username cisco privilege 15 secret 5 $1$AGR7$e7pQx6UI0be3bzRbc0lr81

!

redundancy

 mode none

!
```

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

```
!
vlan internal allocation policy ascending
!
!
!
!
!
!
interface GigabitEthernet0/0/0
 ip address 10.64.4.20 255.255.0.0
 media-type rj45
 negotiation auto
!
interface GigabitEthernet0/0/1
 no ip address
 shutdown
 media-type rj45
 negotiation auto
!
interface GigabitEthernet0/0/2 [14]
 ip address 10.80.22.75 255.255.255.0 [15]
 media-type rj45
 negotiation auto
!
```

[14] LAN interface to Cisco UCM
[15] Cisco UBE LAN interface IPv4 Address

```
interface GigabitEthernet0/0/3 [16]
 description Wan Interface
 ip address 192.65.79.58 255.255.255.224
 media-type rj45
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 shutdown
 negotiation auto
!
interface Vlan1
 no ip address
 shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.33
ip route 10.80.22.0 255.255.255.0 10.80.22.1
ip route 172.16.0.0 255.255.0.0 10.80.22.1
ip route vrf Mgmt-intf 0.0.0.0 0.0.0.0 10.64.1.1
!
!
```

[16] WAN interface to AT&T

```
!
!
!
!
!
control-plane
!
 !
 !
 !
 !
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
!
!
dial-peer voice 200 voip
 description "Outgoing To AT&T .IP PBX facing side"
 no modem passthrough
 session protocol sipv2
```

```
 incoming called-number [1-9]T
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax-relay sg3-to-g3
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 800 voip
 description " Incoming AT&T to IP-PBX . AT&T facing side "
 huntstop
 no modem passthrough
 session protocol sipv2
 incoming called-number [37][13][24]32040..
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
```

```
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/3
 voice-class sip bind media source-interface GigabitEthernet0/0/3
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 700 voip
 description " Incoming AT&T to IP-PBX - IP-PBX facing side "
 huntstop
 destination-pattern [37][13][24].......
 no modem passthrough
 session protocol sipv2
 session target ipv4:10.80.22.2:5060
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax rate 14400
```


Note: Testing was conducted in tekVizion labs

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

no vad

!

dial-peer voice 100 voip [17]

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

destination-pattern 73236.....

no modem passthrough

session protocol sipv2 [18]

session target ipv4:207.242.225.210

voice-class codec 1 [19]

voice-class sip asymmetric payload full [20]

voice-class sip asserted-id pai

voice-class sip privacy-policy passthru [21]

voice-class sip profiles 1 [22]

voice-class sip bind control source-interface GigabitEthernet0/0/3 [23]

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

dtmf-relay rtp-nte [24]

---

[17] Dial peer for AT&T facing 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] This command enables the dial peer to use SIP profile 1

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

```
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none [25]
 no vad
!
dial-peer voice 300 voip
 description " Int'l calls to AT&T - AT&T facing side "
 destination-pattern 011T
 no modem passthrough
 session protocol sipv2
 session target ipv4:207.242.225.210
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/3
 voice-class sip bind media source-interface GigabitEthernet0/0/3
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 400 voip
```

[24] This command used to pass RTP NTE (RFC2833) DTMF with respect to the dial peers used for the call.
[25] This command enables T38 fax protocol for calls terminating on this dial-peer

```
 description " Int'l calls to AT&T - IP-PBX facing side "
 no modem passthrough
 session protocol sipv2
 incoming called-number 011T
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 500 voip
 description " N11 Calls to AT&T - AT&T facing side "
 destination-pattern .11
 no modem passthrough
 session protocol sipv2
 session target ipv4:207.242.225.210
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
```

```
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/3
 voice-class sip bind media source-interface GigabitEthernet0/0/3
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 600 voip
 description " N11 Calls to AT&T - IP-PBX facing side "
 no modem passthrough
 session protocol sipv2
 incoming called-number .11
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
```

```
dial-peer voice 122 voip
 description "OPERATOR TESTING"
 destination-pattern 0
 no modem passthrough
 session protocol sipv2
 session target ipv4:207.242.225.210
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/3
 voice-class sip bind media source-interface GigabitEthernet0/0/3
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 141 voip
 description "Network Feature"
 translation-profile outgoing NPA
 destination-pattern *..
 no modem passthrough
 session protocol sipv2
 session target ipv4:207.242.225.210
```

```
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/3
 voice-class sip bind media source-interface GigabitEthernet0/0/3
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 2151 voip
 description "Outgoing To AT&T"-AT&T facing side
 destination-pattern 7323204607
 no modem passthrough
 session protocol sipv2
 session target ipv4:207.242.225.210
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/3
```

```
 voice-class sip bind media source-interface GigabitEthernet0/0/3
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 214 voip
 description "Outgoing To AT&T"-AT&T facing side
 destination-pattern [2-9]T
 no modem passthrough
 session protocol sipv2
 session target ipv4:207.242.225.210
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/3
 voice-class sip bind media source-interface GigabitEthernet0/0/3
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax-relay sg3-to-g3
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
```

```
 no vad
!
!
gateway
 timer receive-rtp 1200
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0
 session-timeout 90
 exec-timeout 960 0
 password tekV1z10n
 no activation-character
 logging synchronous
 transport preferred ssh
 transport input all
 stopbits 1
line vty 1 4
 exec-timeout 960 0
 password tekV1z10n
 logging synchronous
 transport input all
!
```

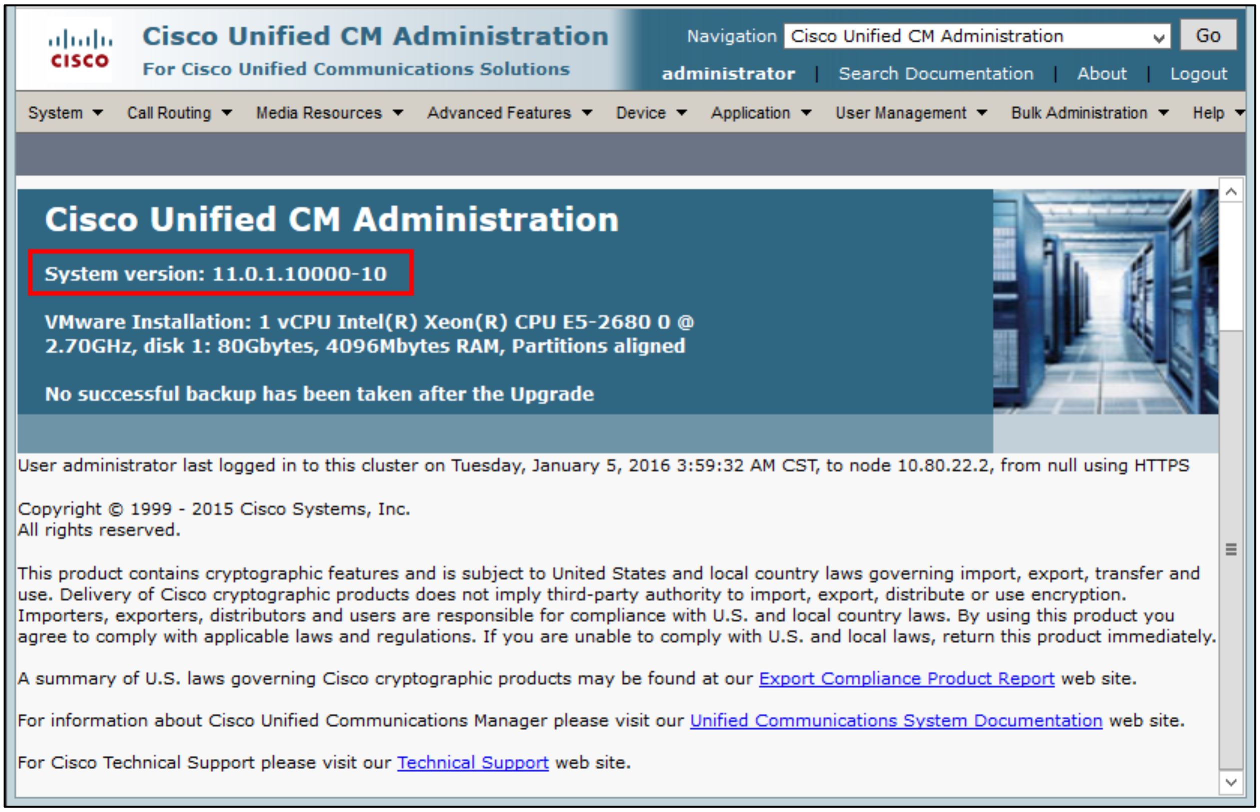
!

end

Cisco UCM Configuration

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

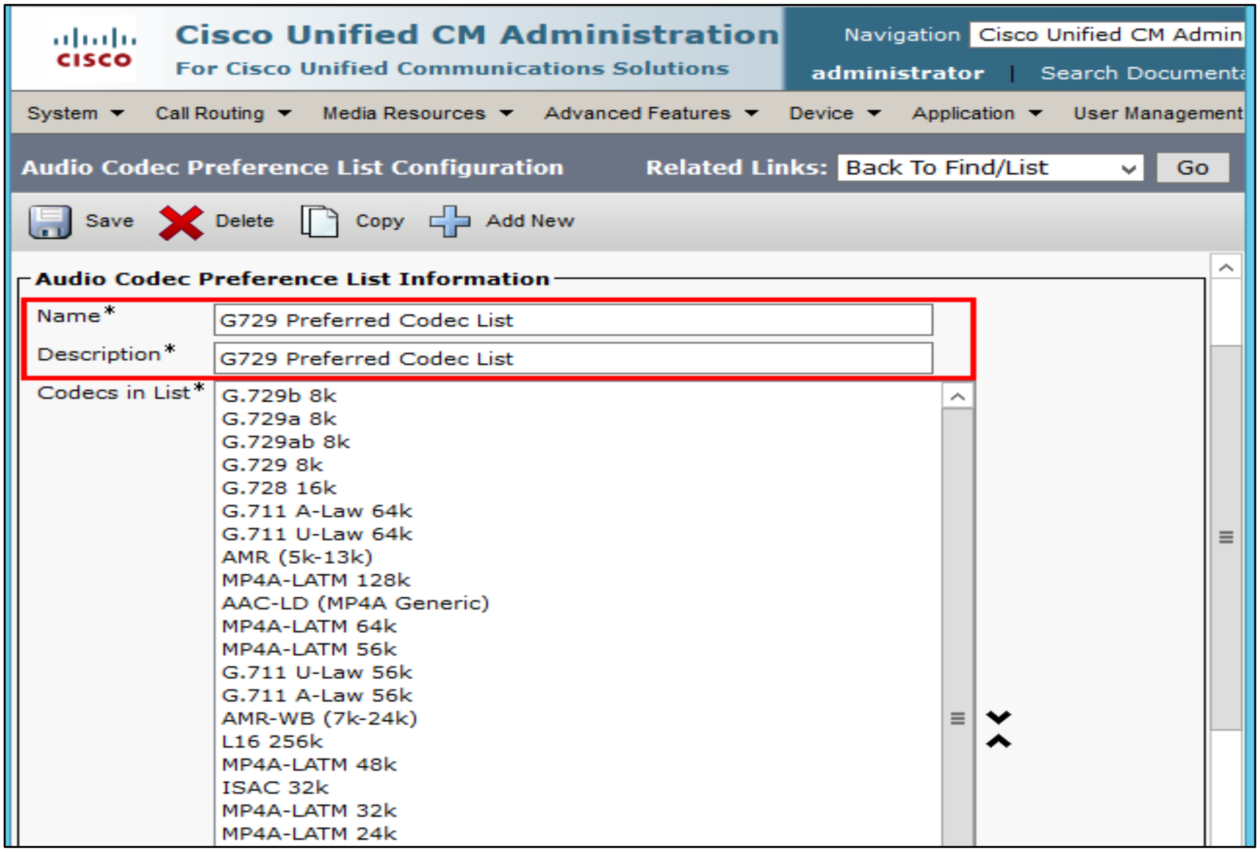
Cisco UCM Version

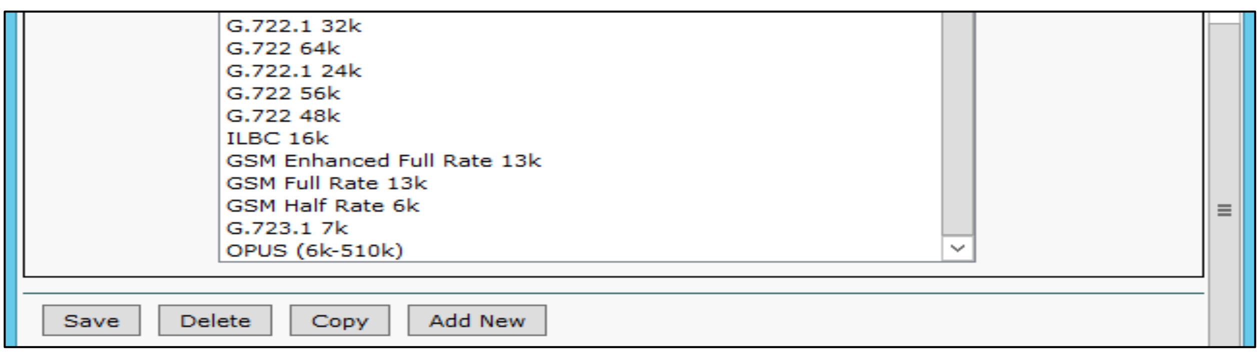


Cisco UCM Audio Codec Preference List

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

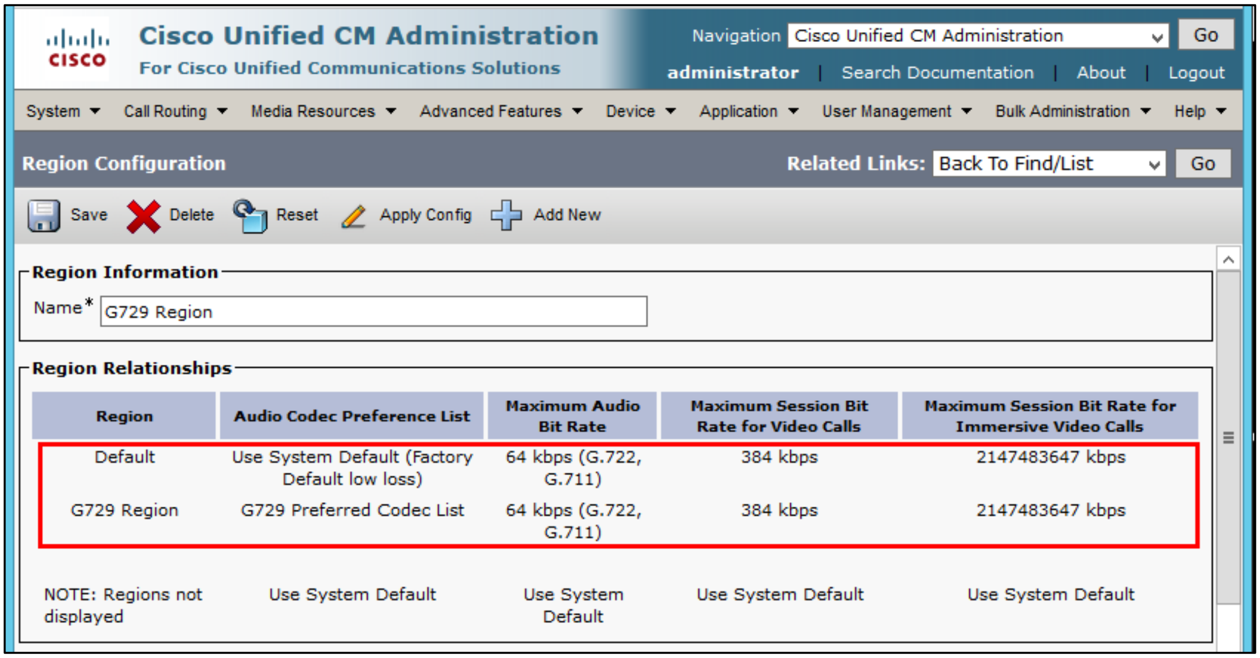
Cisco UCM 11.0 has a feature called Audio Codec Preference List. This feature allows to configure the order of audio codec preference both for Inter and Intra Region calls. Audio Codec Preference list is assigned to the Region used by the Device Pool for Phones and by Conference Bridges. Based on user requirement, different codec regions can be assigned as their first choice codec with this configuration for inbound calls as well as conferences initiated by Cisco IP phones. Audio codec preference for outbound calls is determined by Cisco UBE (via configuration of voice-class codec)

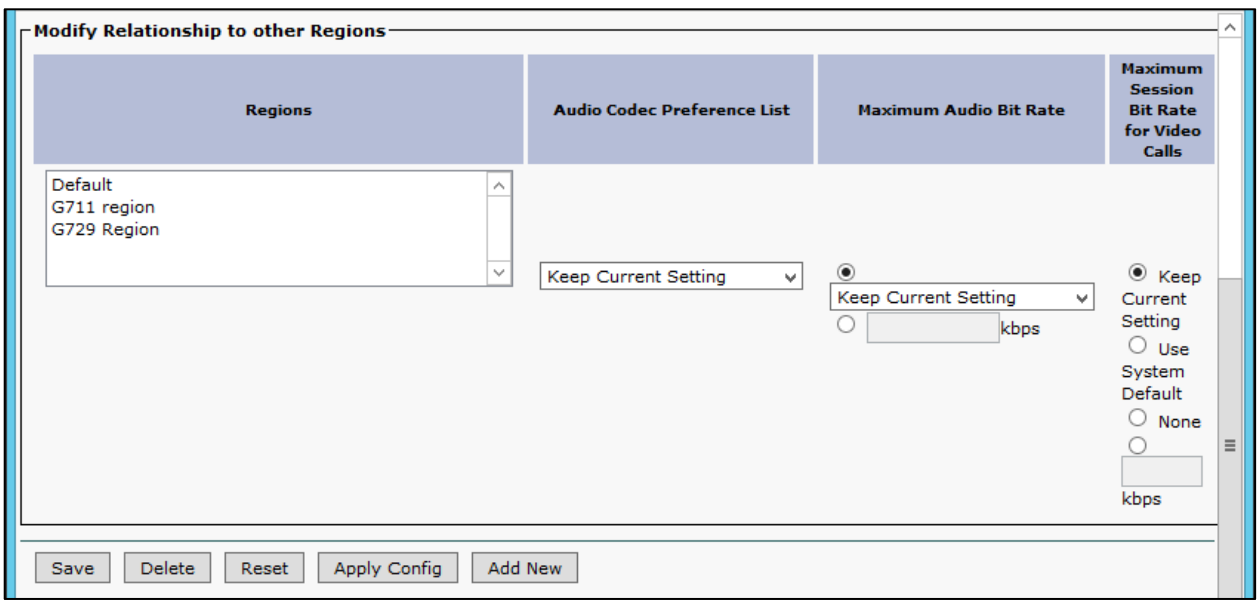




Cisco UCM Region Configuration

Navigation Path: System → Region Information → Region

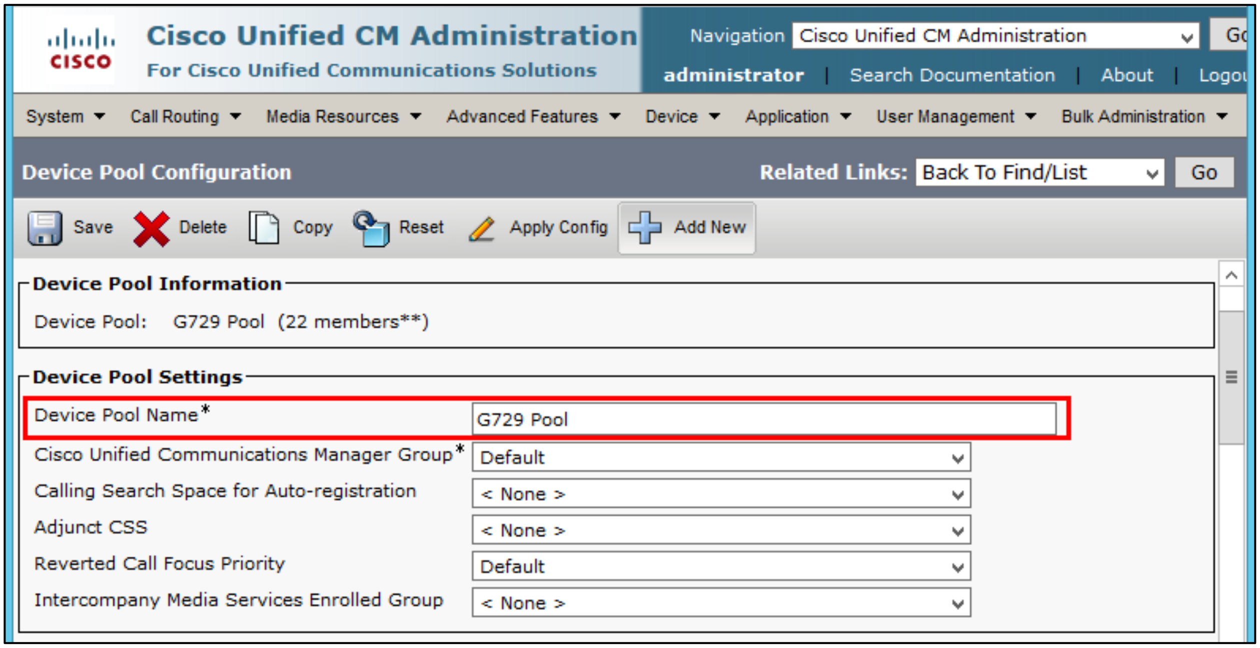


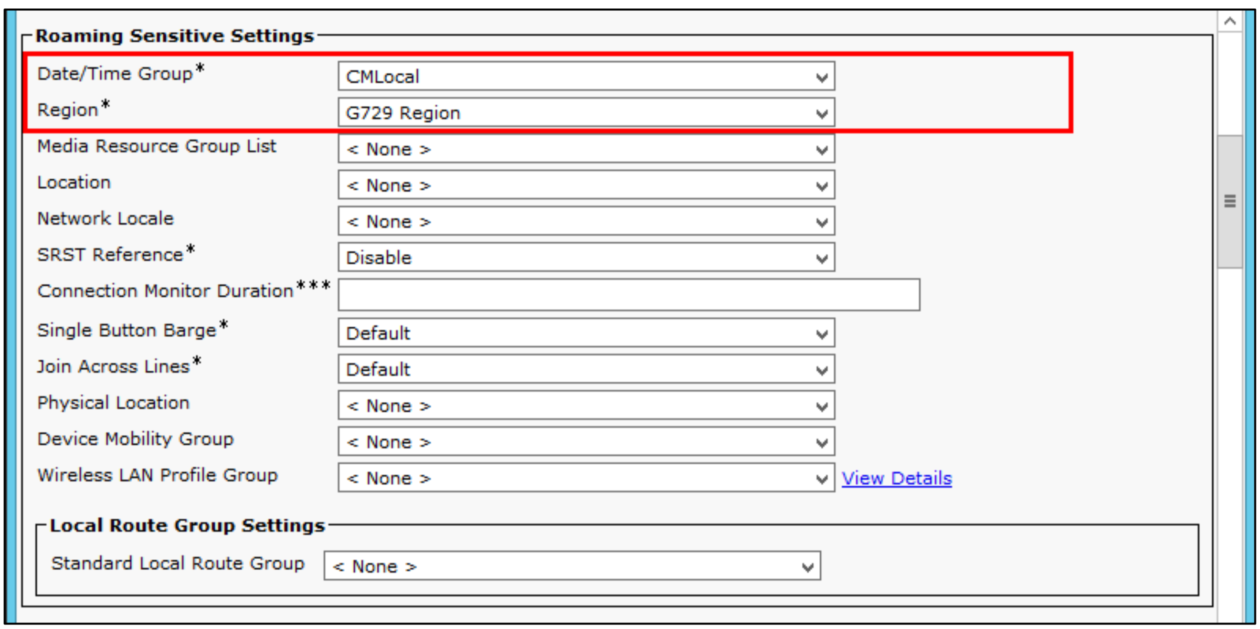


Device Pool Configuration

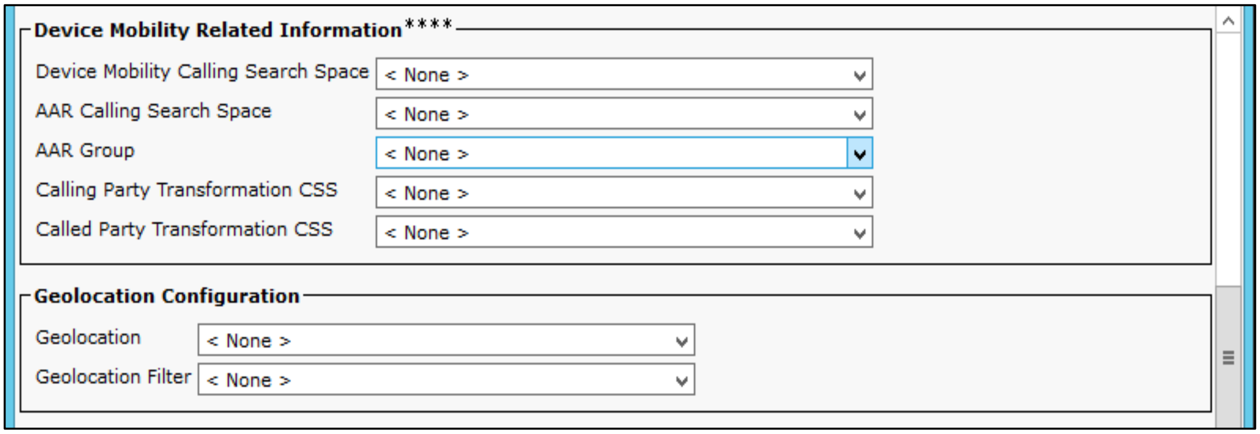
Navigation Path: System → Device Pool

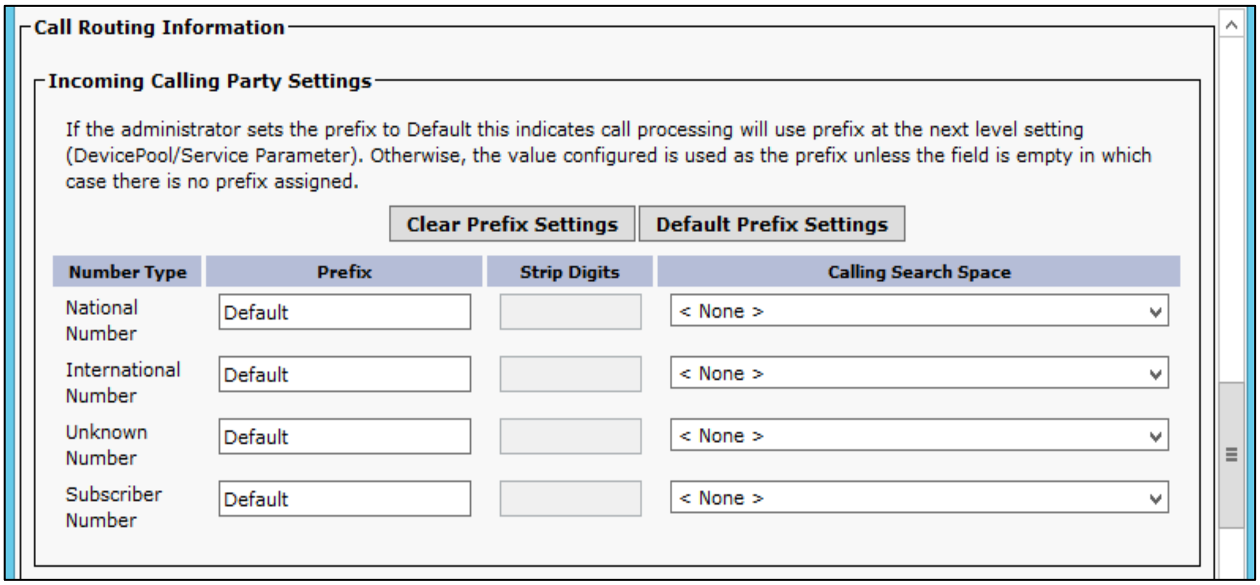
“G729_pool” Device Pool is configured for testing the interoperability. No special consideration needs to be taken when configuring the Device Pools. Optionally, a Media Resource Group List can be added to the Device Pools, if needed, to assign selected Media Resources (Conference Bridges, Transcoders, MoH servers, Annunciators) to devices.





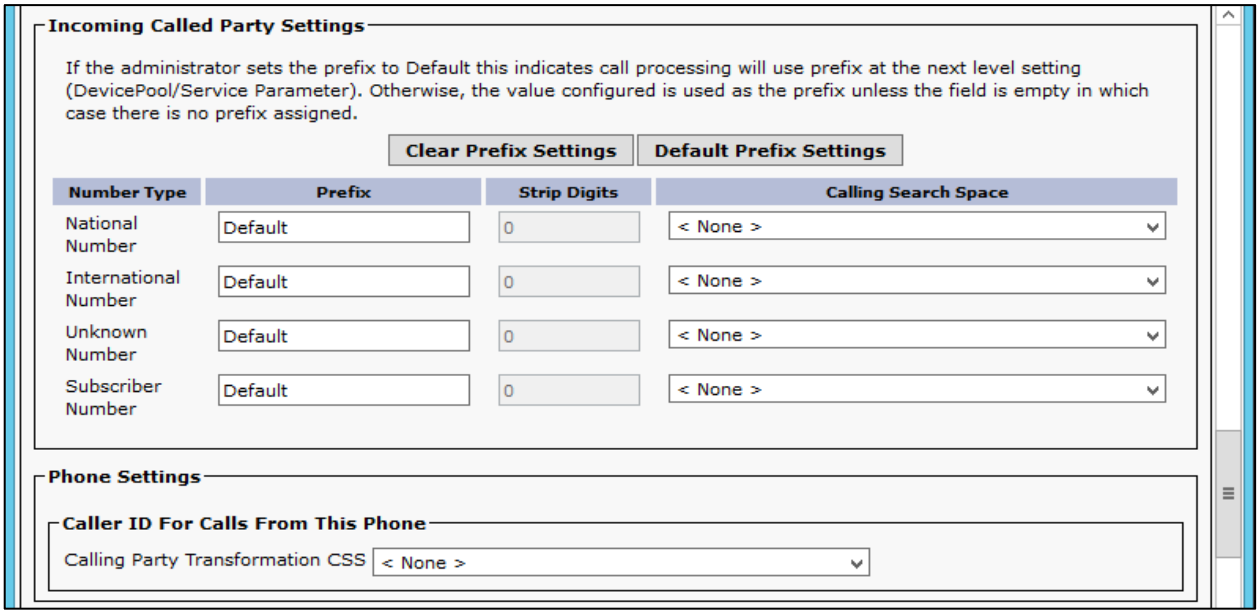
Device Pool Configuration (continued…)





Note: Testing was conducted in tekVizion labs

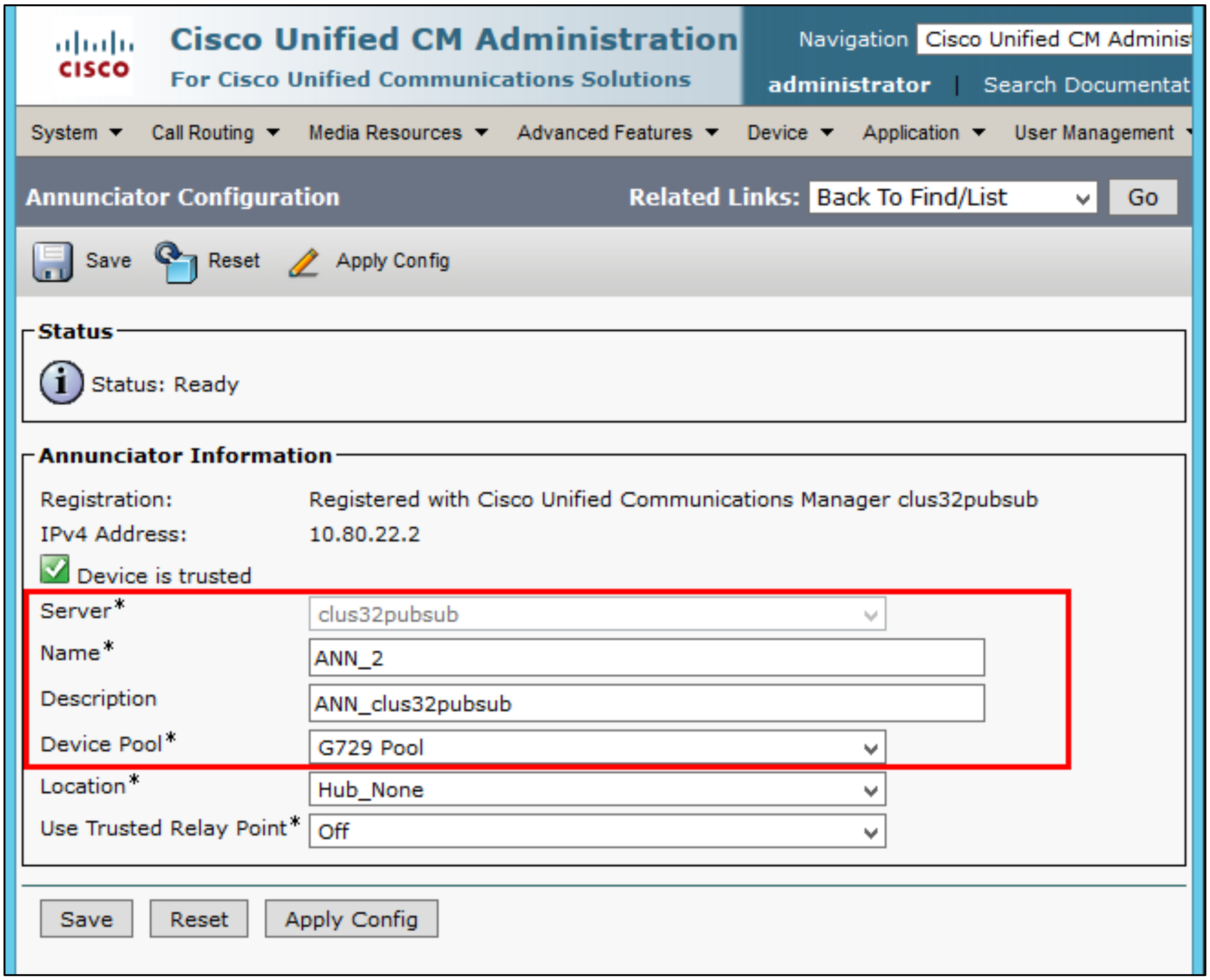
Device Pool Configuration (continued…)



Annunciator Configuration

Navigation: Media Resource → Annunciator

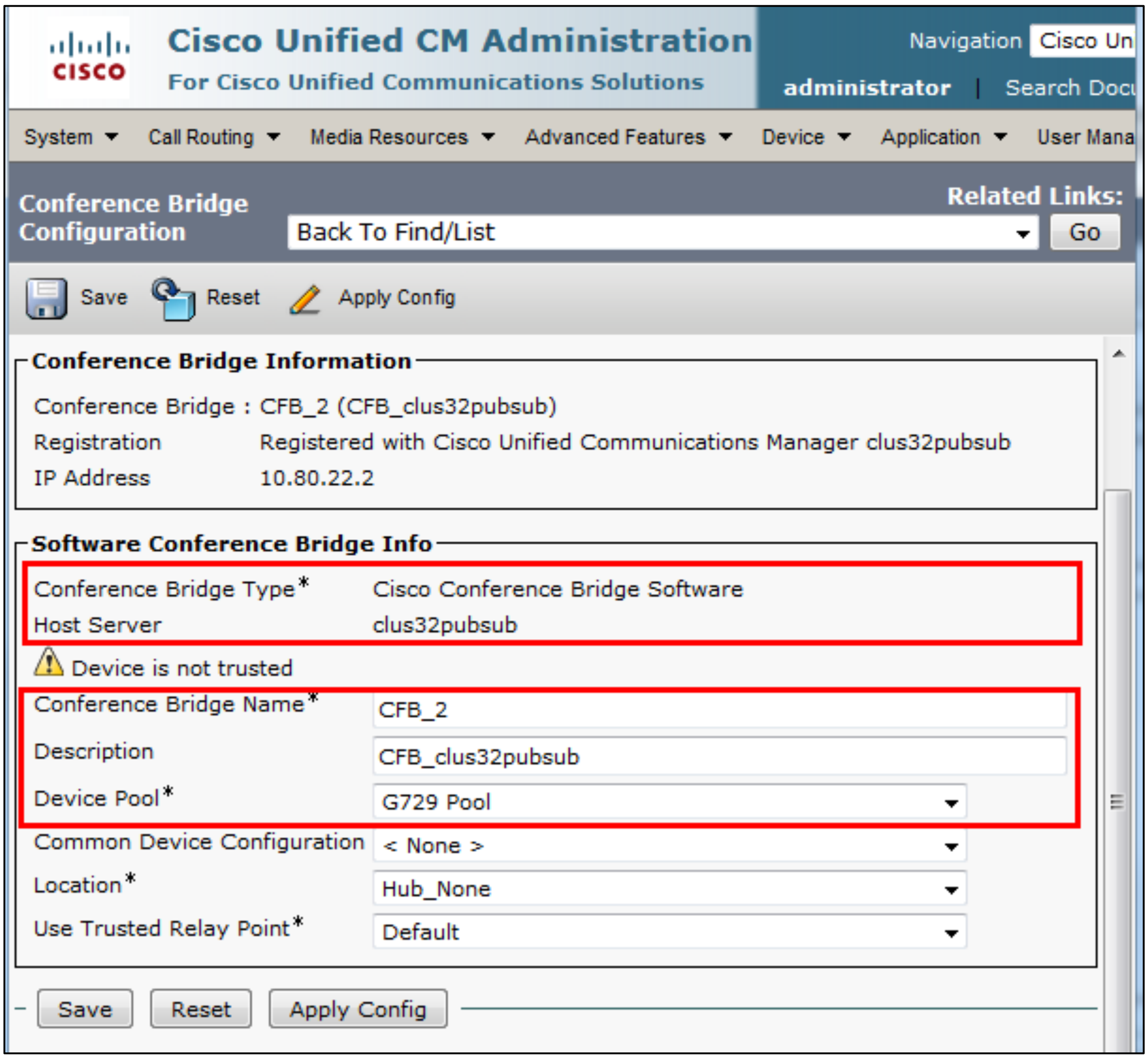
Set Name* = ANN_2.
Set Description = ANN_clus32pubsub. This is used for this example
Set Device Pool* = G729_pool.



Conference Bridge Configuration

Navigation: Media Resources → Conference Bridge

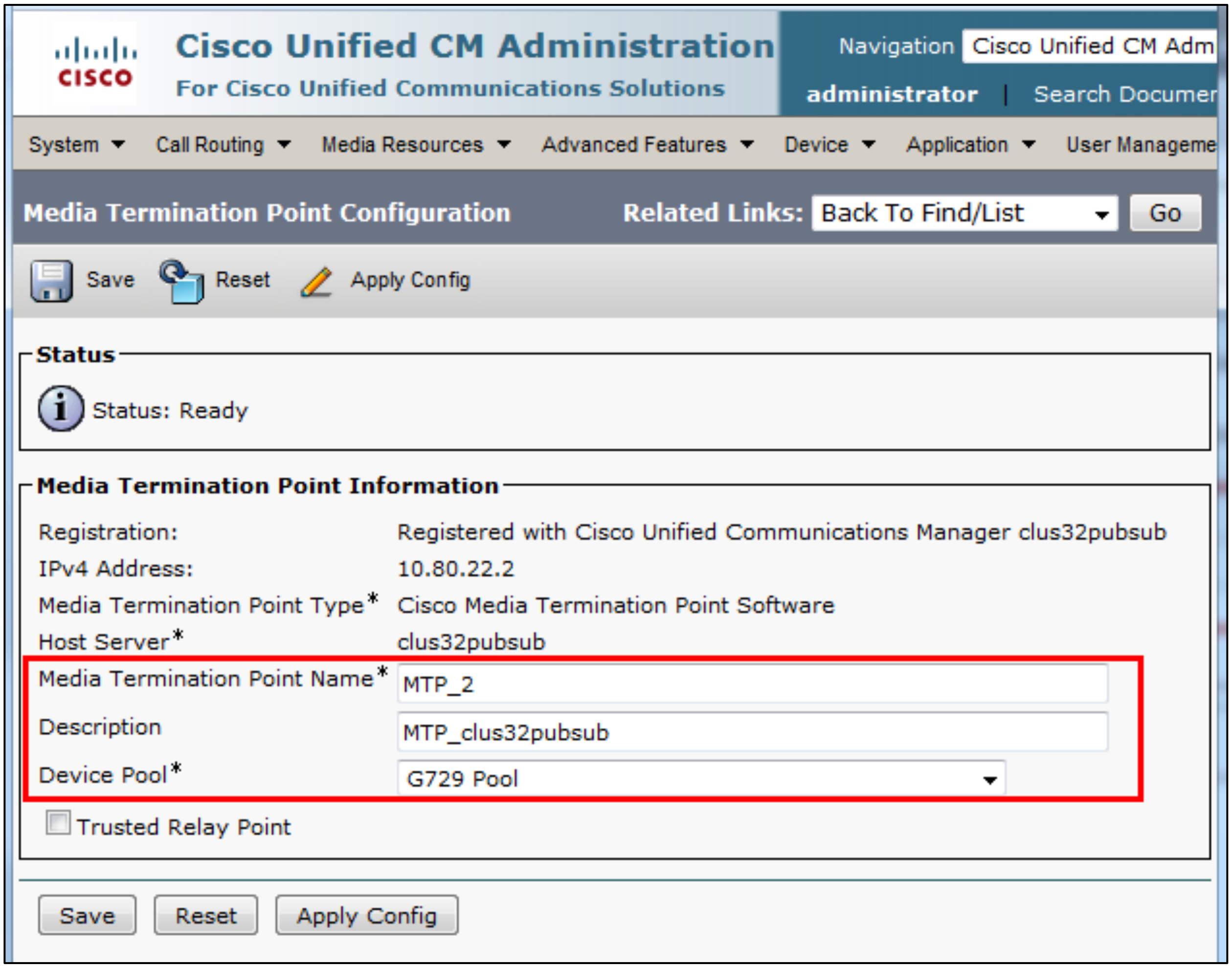
Set Conference Bridge Type* = Cisco Conference Bridge Software.
Set Host Server = clus32pubsub. This is used for this example.
Set Conference Bridge Name* = CFB_2.
Set Description = CFB_clus32pubsub. This is used in this example.
Set Device Pool* = G729_pool.



Media Termination Point Configuration

Navigation: Media Resource→Media Termination Point

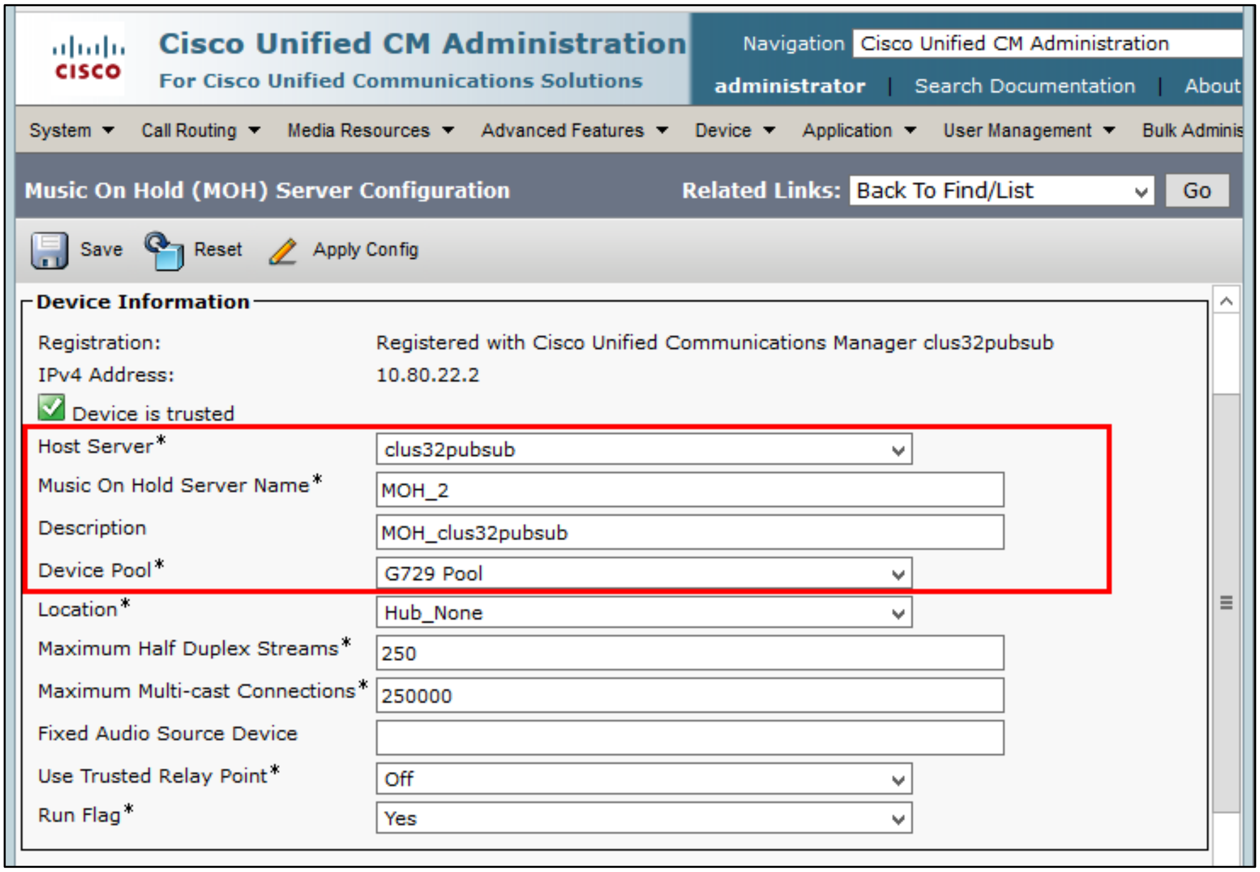
Set Media Termination Point Name* = MTP_2
Set Description = MTP_clus32pubsub. This is used for this example
Set Device pool* = G729 Pool

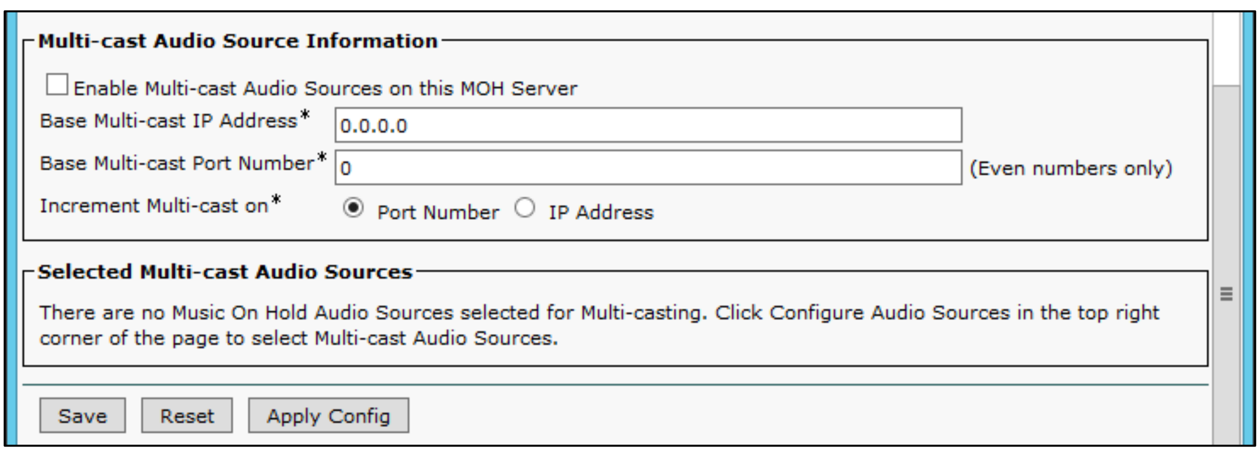


Music on Hold Server Configuration

Navigation: Media Resources → Music on Hold Server

Set Music on Hold Server Name* = MOH_2.
Set Description = MOH_clus32pubsub. This is used for this example.
Set Device Pool* = G729_pool.





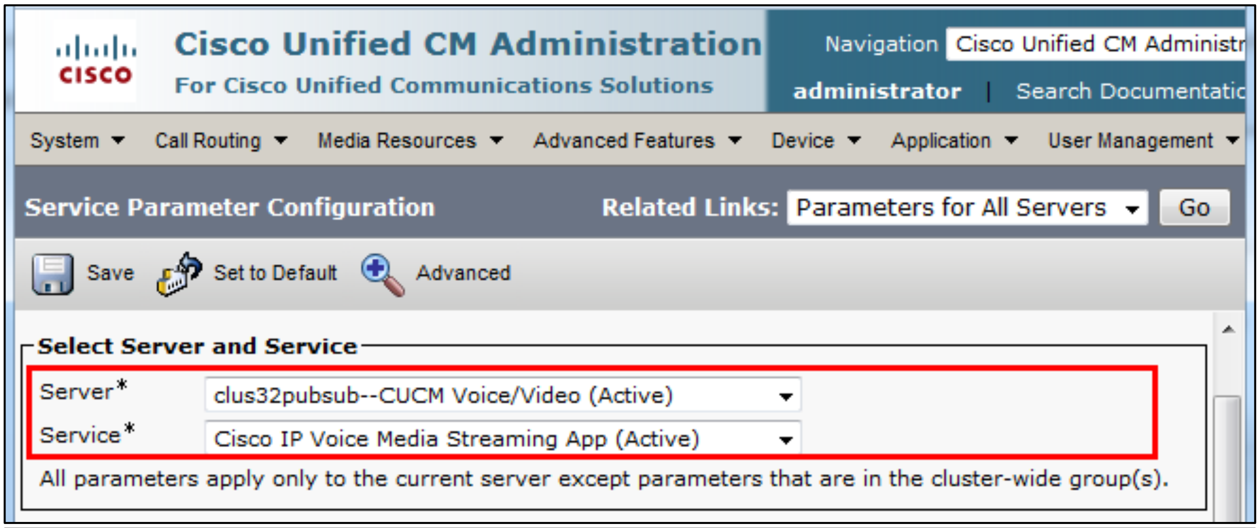
Music on Hold Service (IP Voice Media Streaming App) Parameter Settings

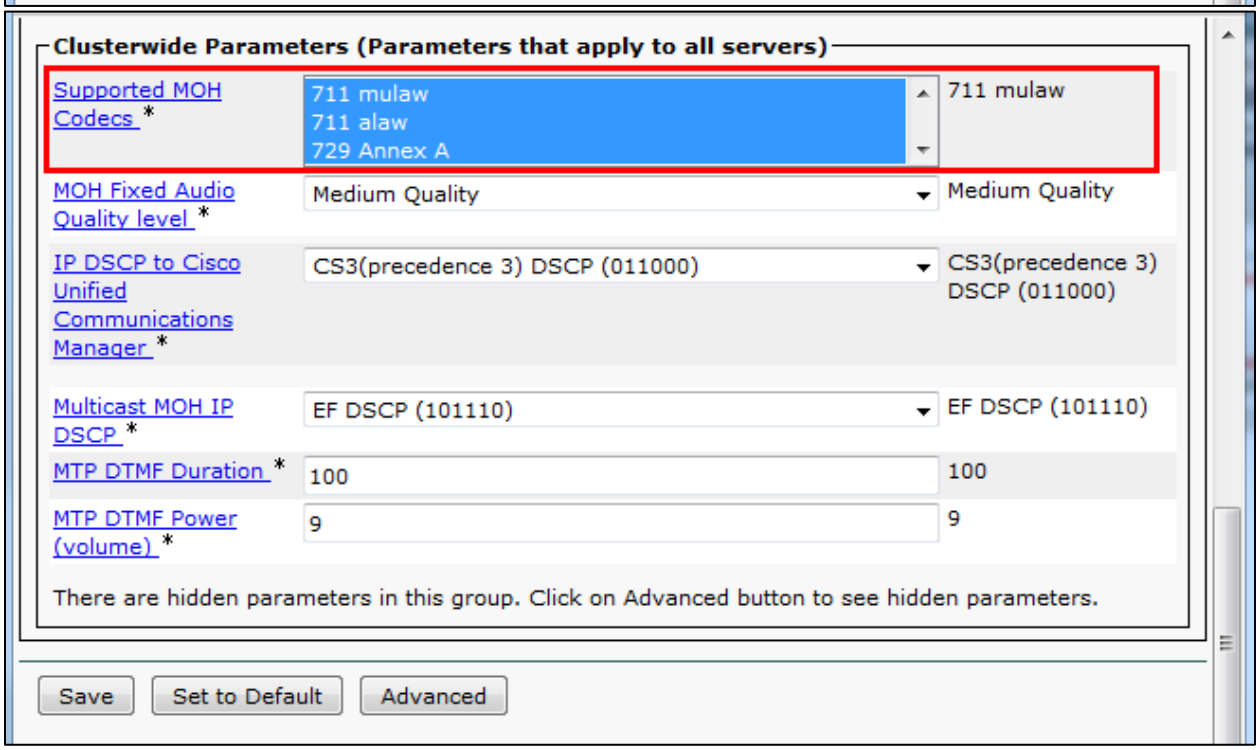
Navigation: System → Service Parameter

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

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

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

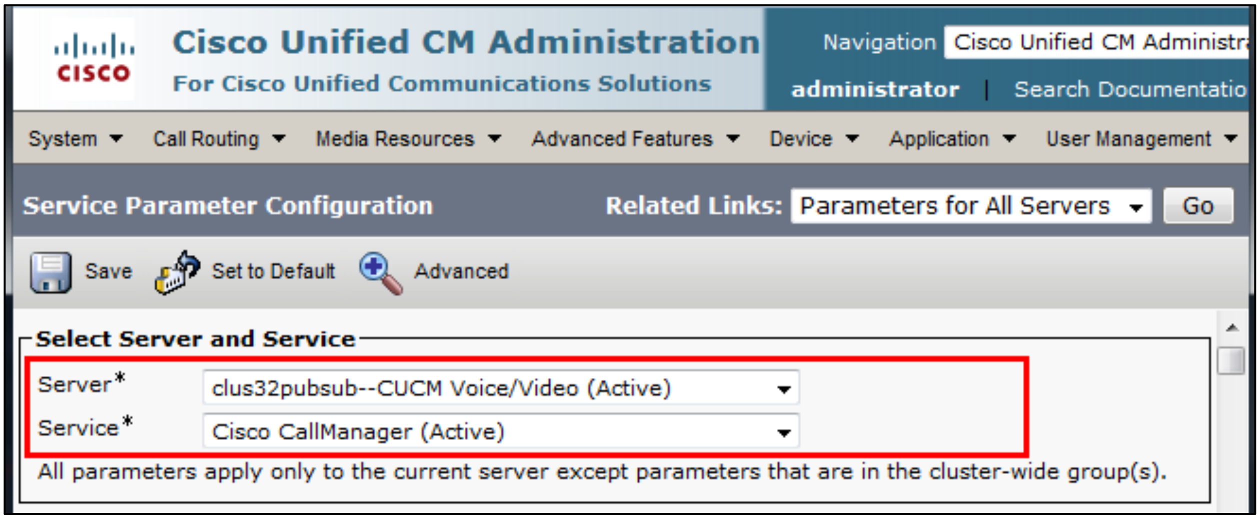


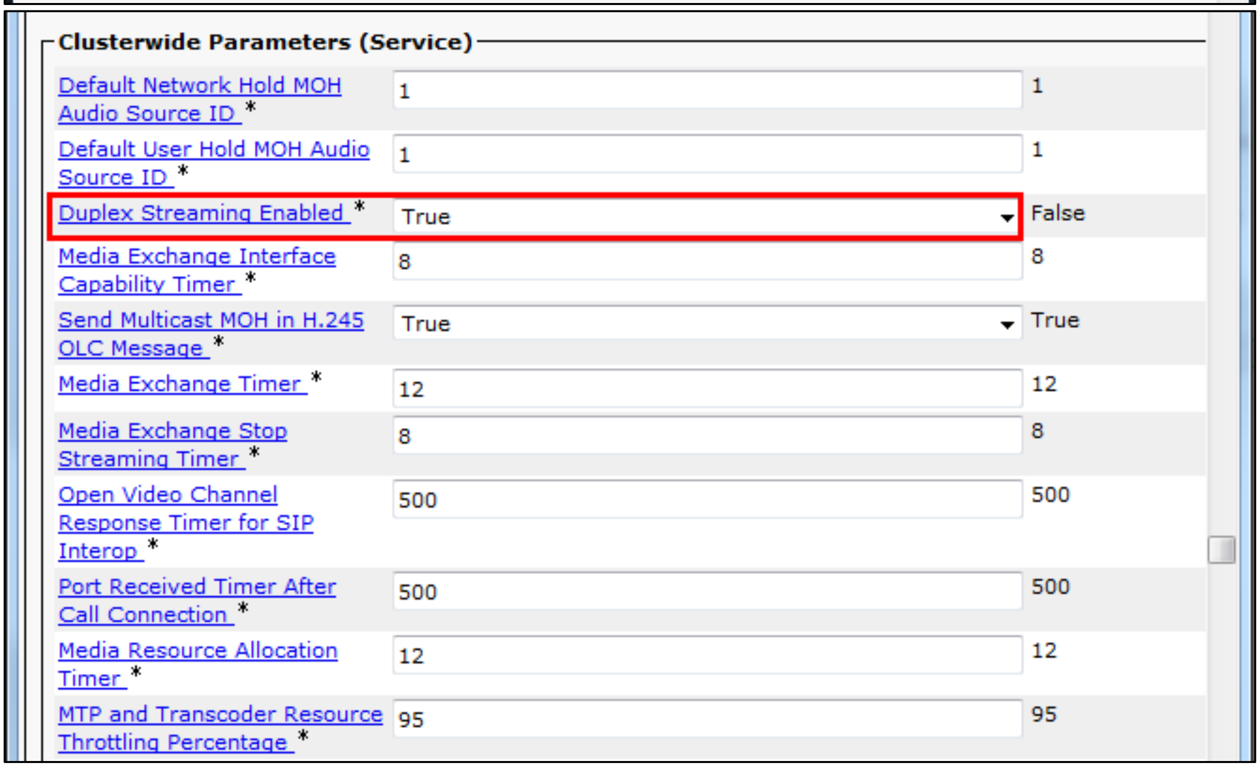


Music on Hold Service (Duplex Streaming) Parameter Settings

Navigation: System → Service Parameter

Select Server* = clus32pubsub--CUCM Voice/Video (Active). This is used in this example.
Select Service* = Cisco CallManager (Active).
Select Duplex Streaming Enabled * = True



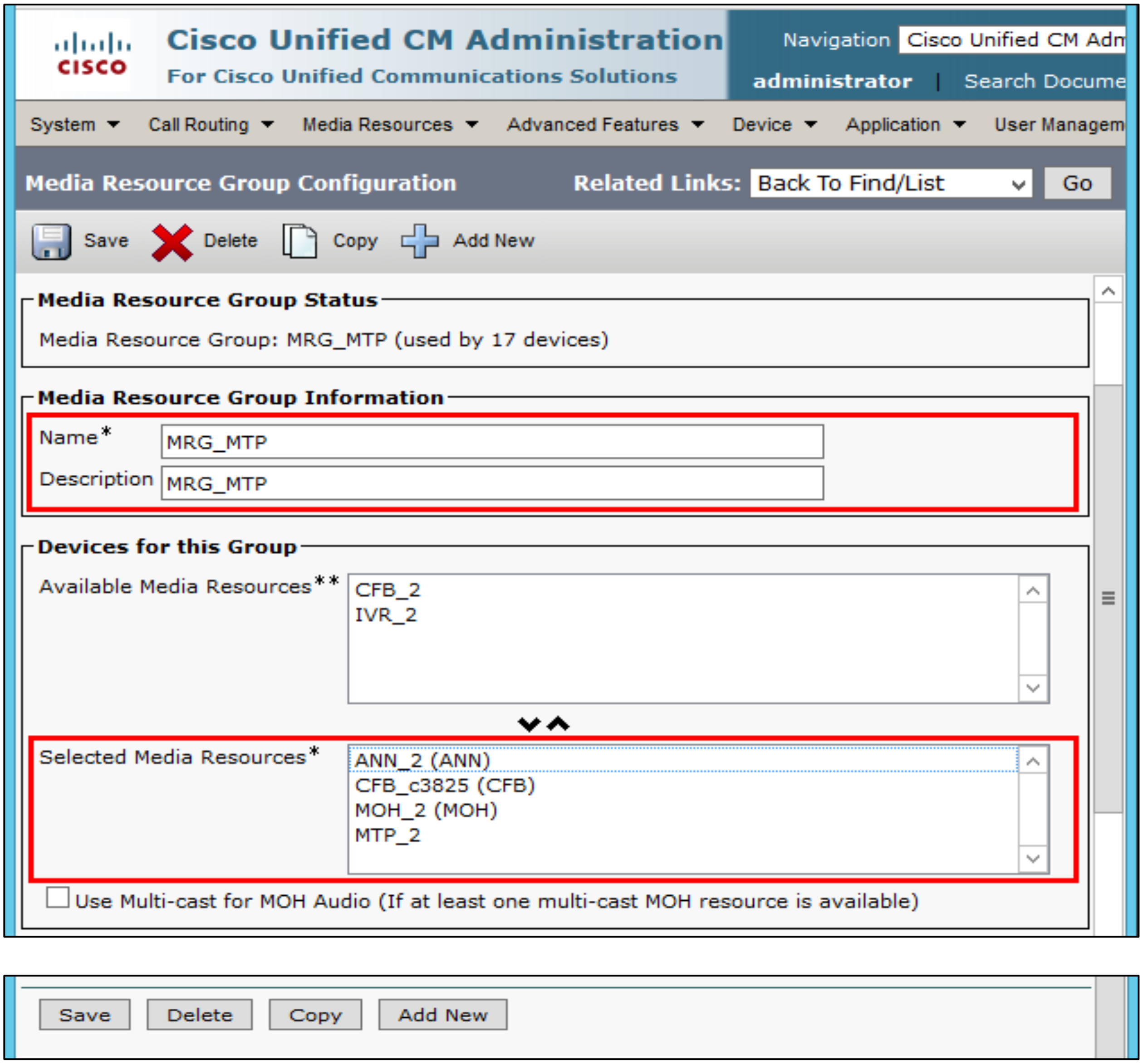


Media Resource Group Configuration

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.

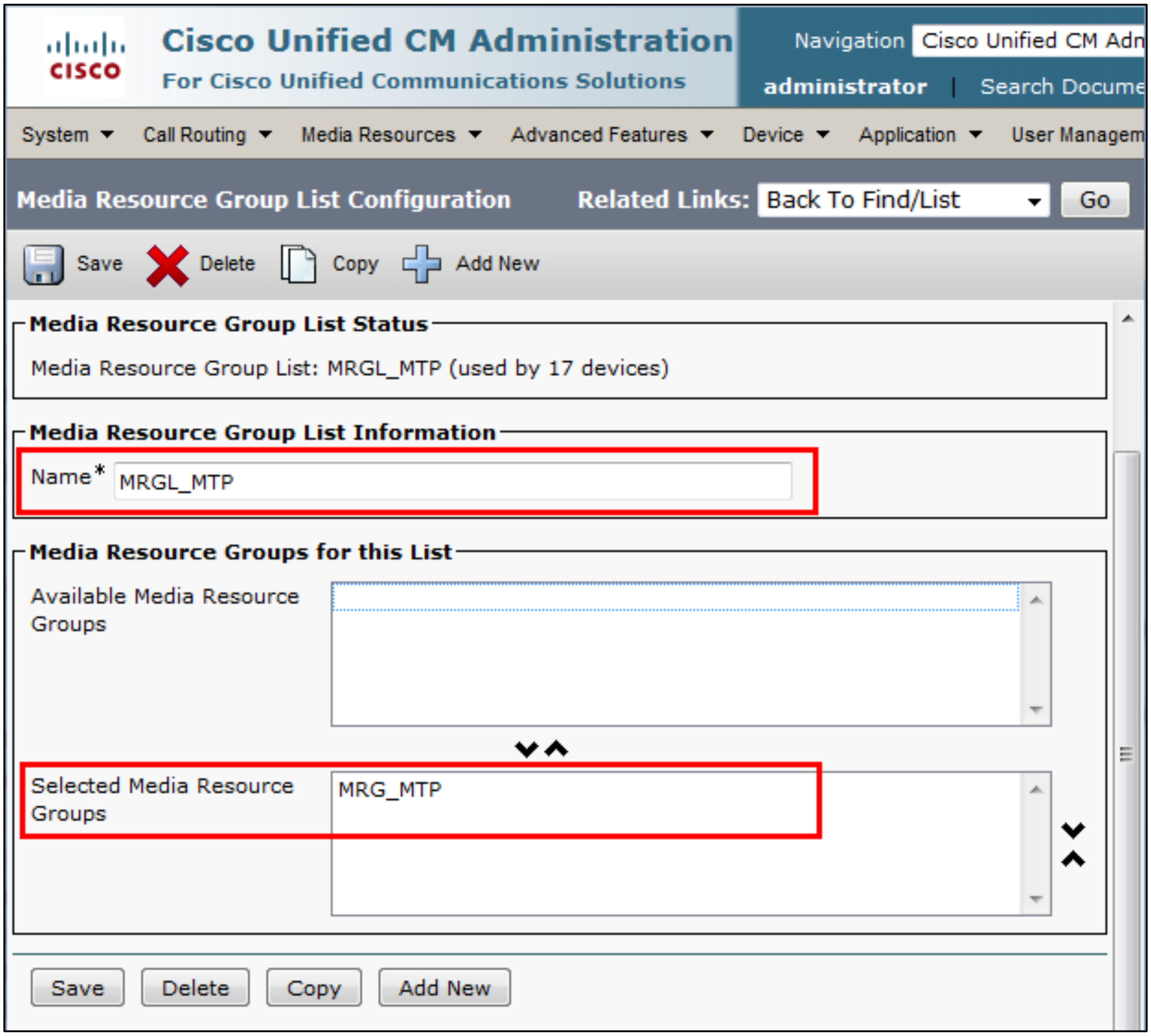
Set Name*= MRG_MTP - This is used for this example.
Set Description = MRG_MTP - This text is used to define this Media Resource Group List.
Set all Resources in the selected Media Resources Box.



Media Resource Group List Configuration

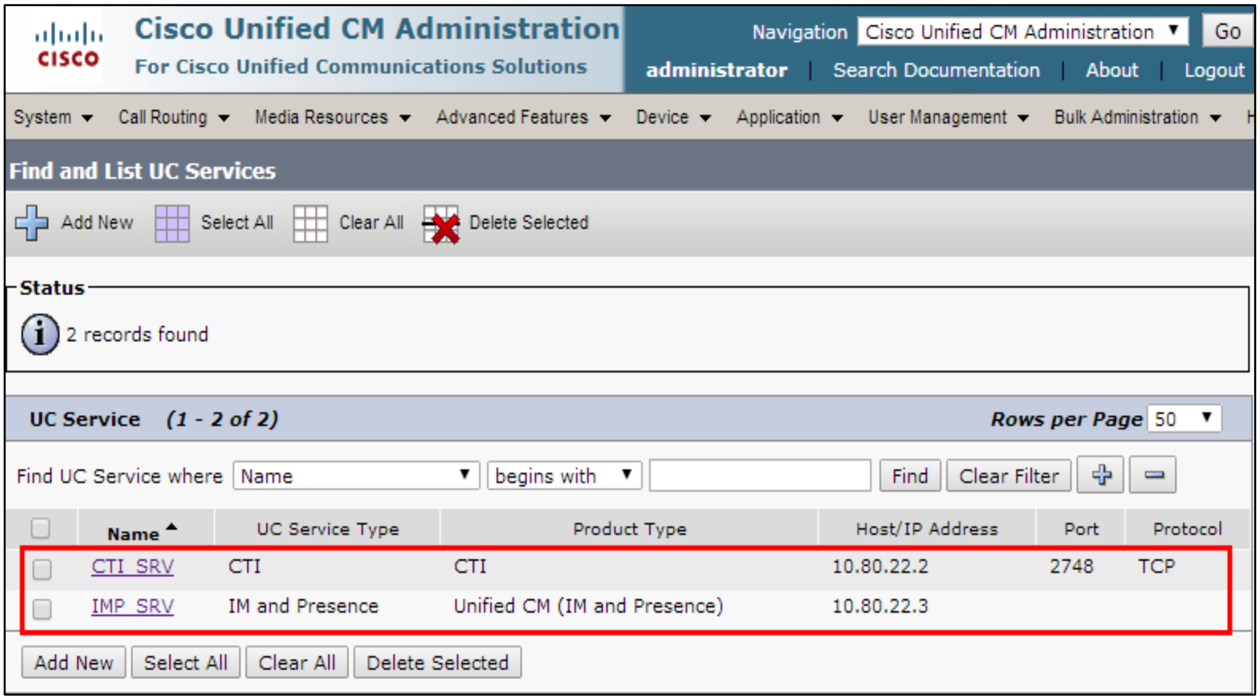
Navigation Path: Media Resources → Media Resource Group List

Set Name = MRGL_MTP.
Set selected Media Resource Groups = MRG_MTP.



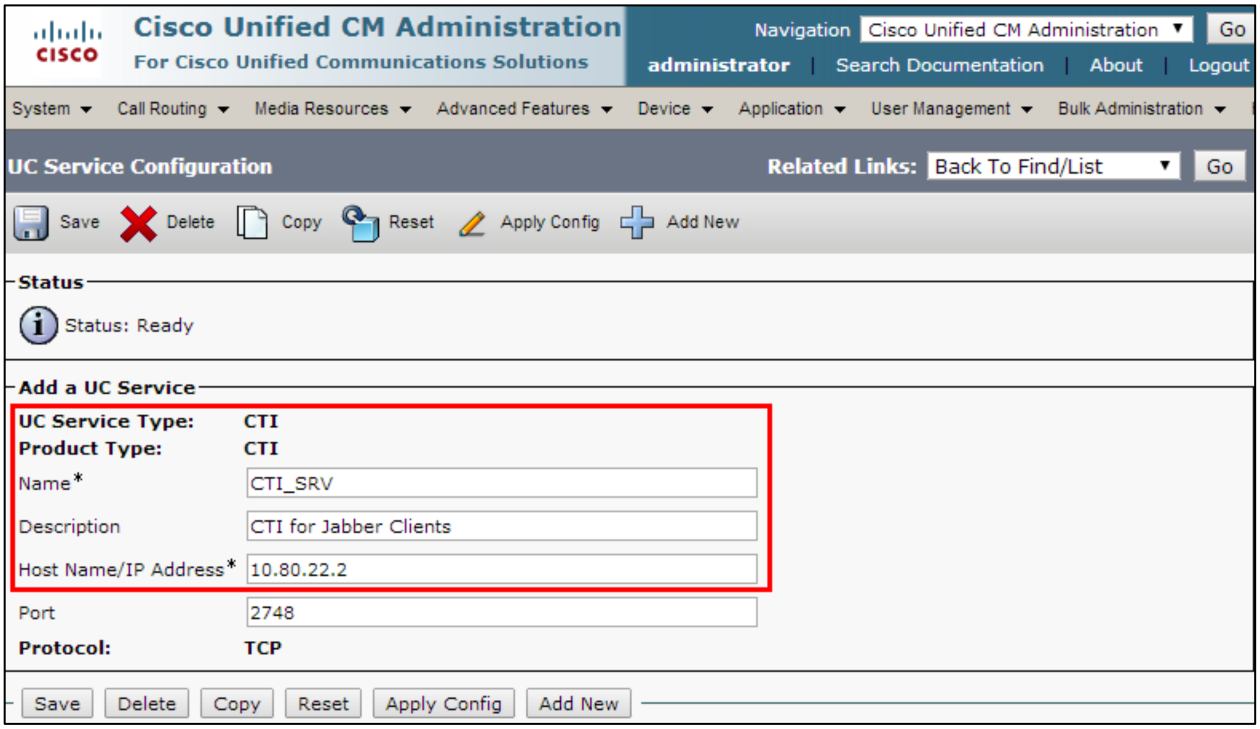
UC Service Configuration

Navigation: User Management → User Settings → UC Service



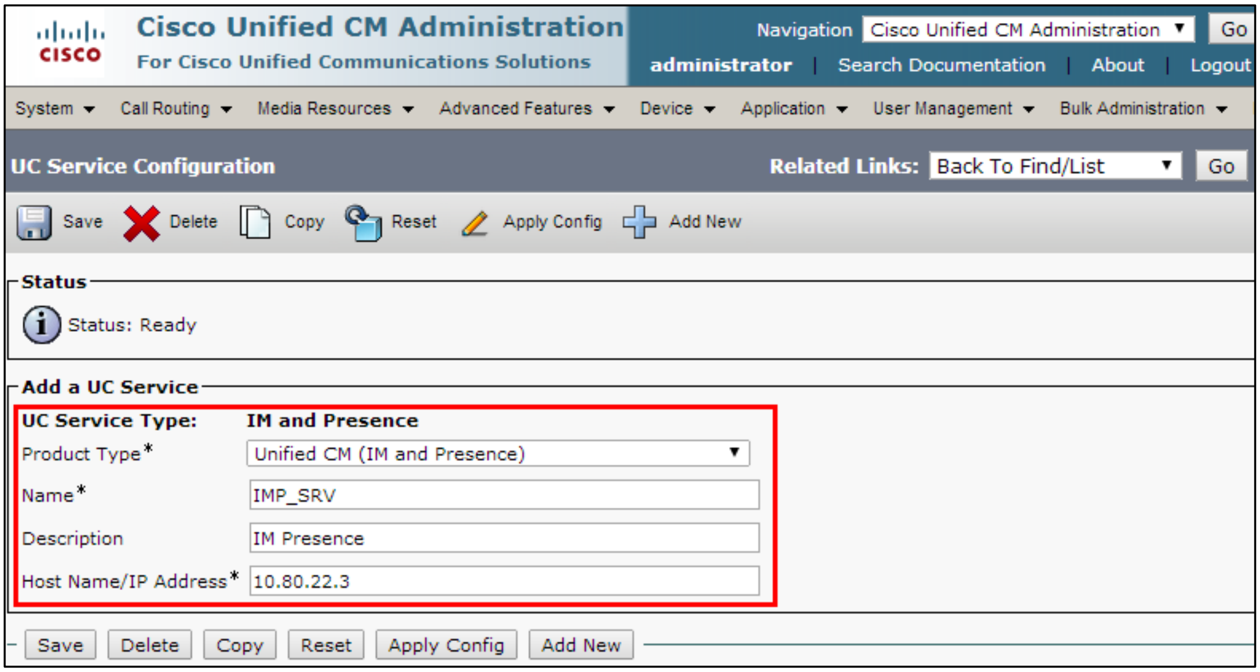
UC Service Configuration (Contd…)

Select UC Service Type: = CTI
Set Name* = CTI_SRV. This is used in this example.
Set Description = CTI for Jabber Clients. This is used in this example.
Set Host Name/IP Address* = 10.80.22.2 (Cisco UCM Address)



UC Service Configuration (Contd…)

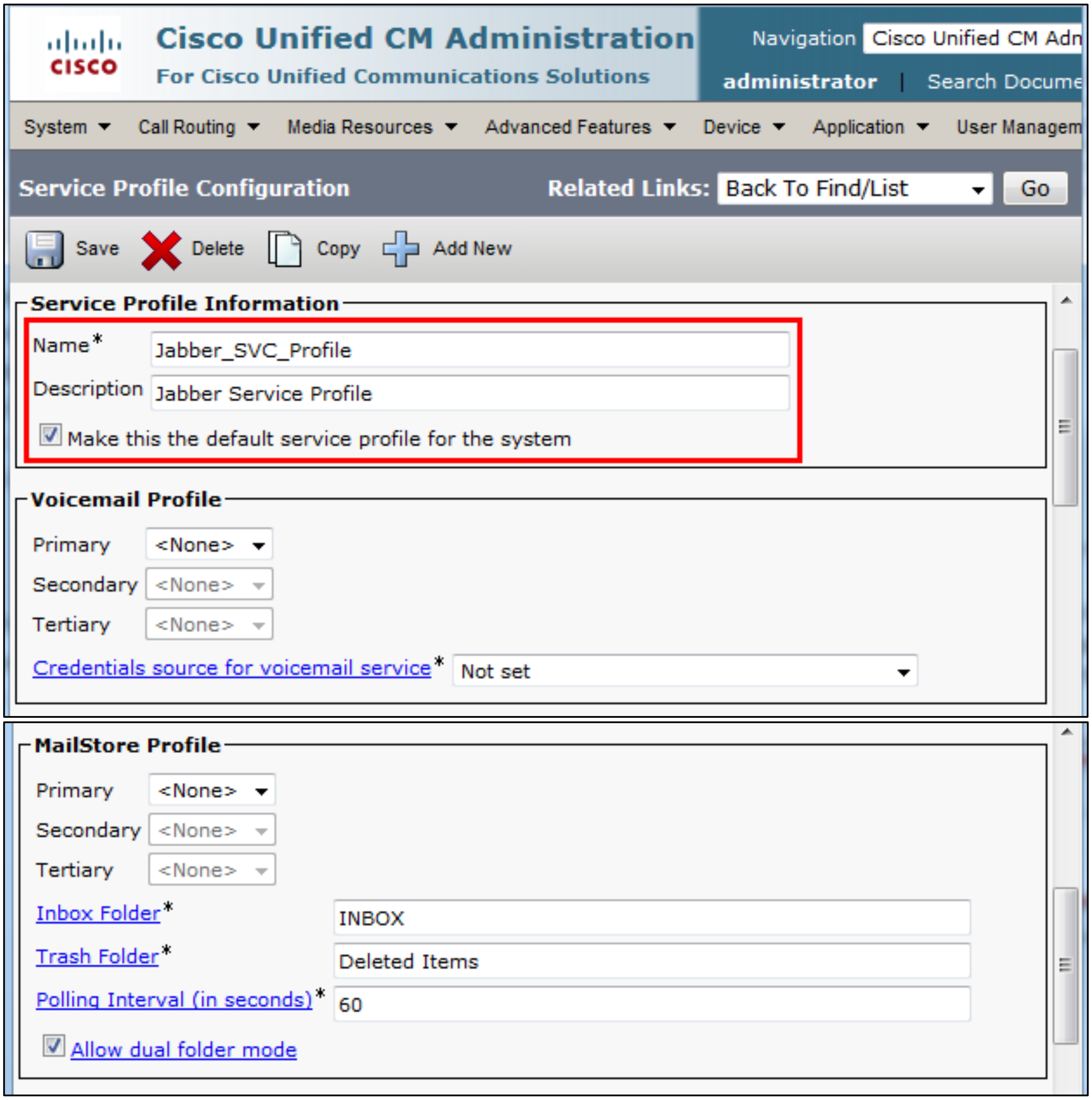
Select UC Service Type: = IM and Presence
Set Name* = IMP_SRV. This is used in this example.
Set Description = IM Presence. This is used in this example.
Set Host Name/IP Address* = 10.80.22.3 (Cisco UCM IM & Presence IP Address)



Service Profile Configuration

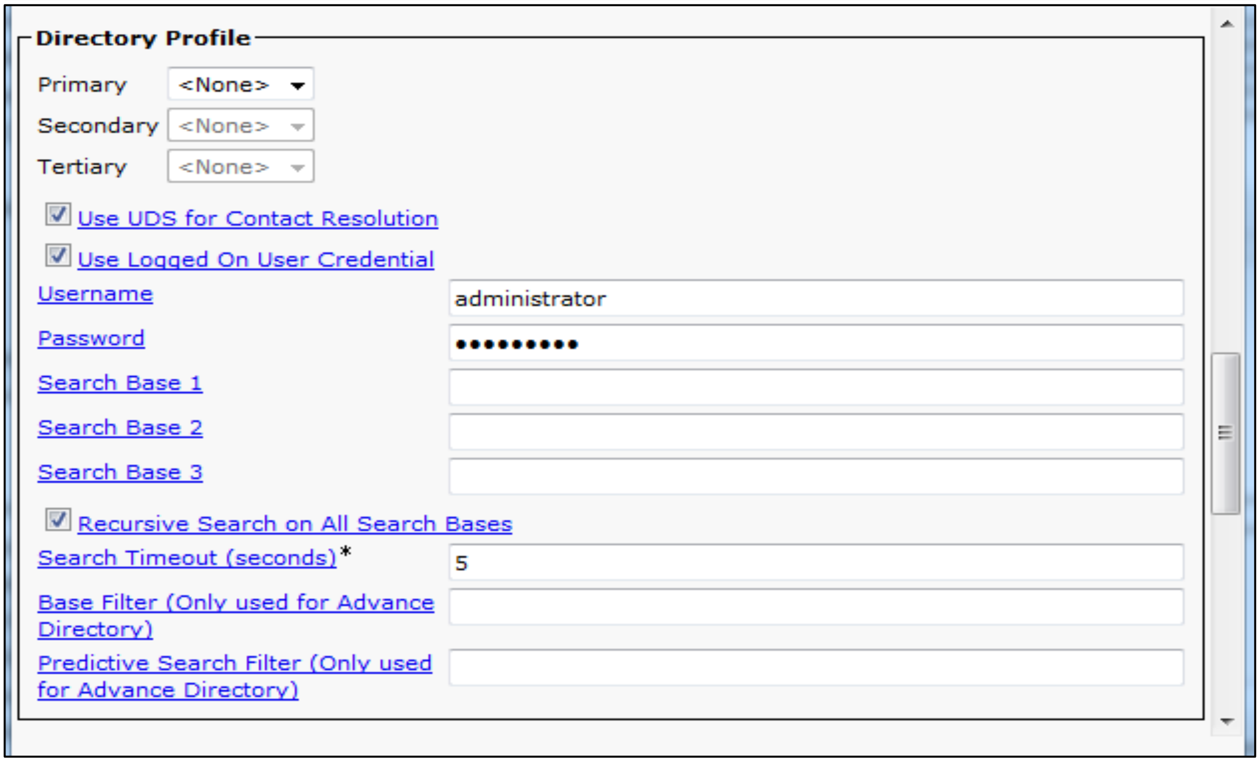
Navigation: User Management → User Settings → Service Profile

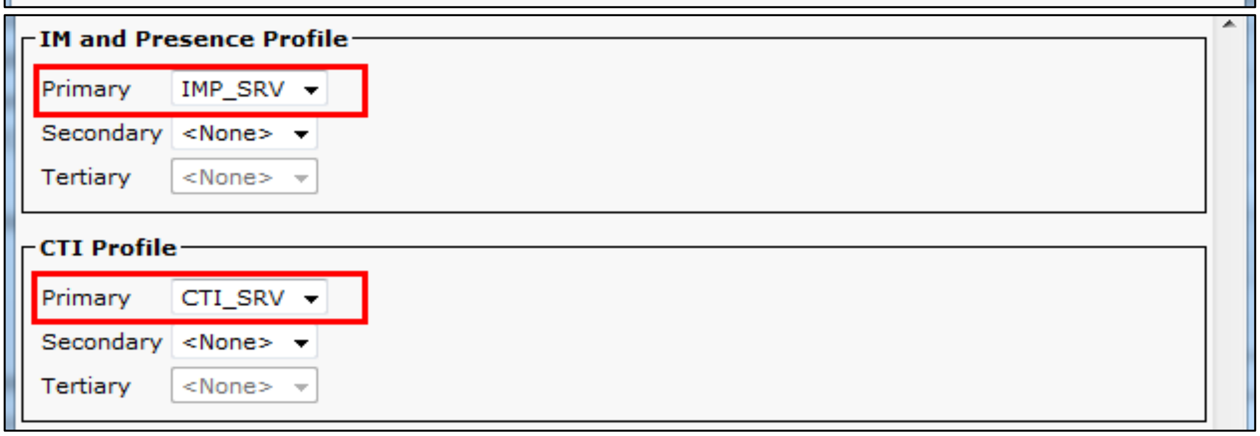
Set Name* = Jabber_SVC_Profile. This is used in this example.
Set Description = Jabber Service Profile. This is used in this example.
Check - Make this the default service profile for the system.



Service Profile Configuration (Contd…)


Conferencing Profile
Primary
<None>
Secondary
<None>
Tertiary
<None>
Server Certificate Verification
Any
Credentials source for web conference service*
Not set

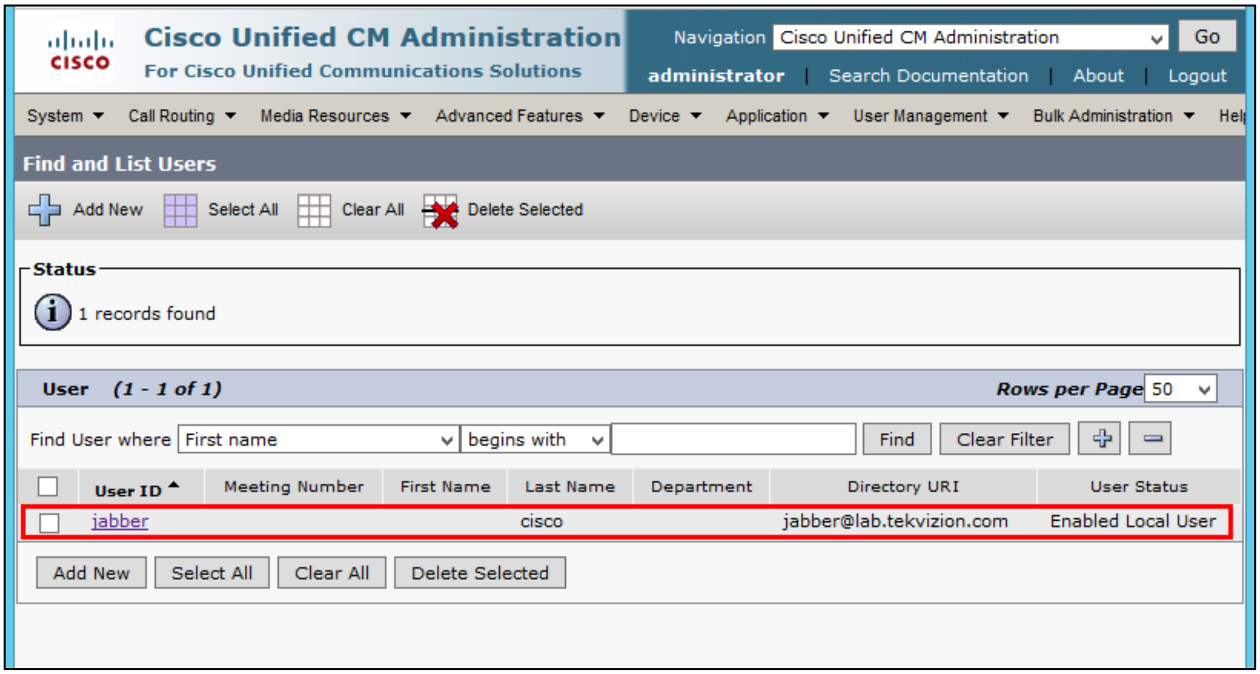

Directory Profile
Primary
<None>
Secondary
<None>
Tertiary
<None>
Use UDS for Contact Resolution
Use Logged On User Credential
Username
administrator
Password
••••••••••
Search Base 1
Search Base 2
Search Base 3
Recursive Search on All Search Bases
Search Timeout (seconds)*
5
Base Filter (Only used for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


IM and Presence Profile
Primary
IMP_SRV
Secondary
<None>
Tertiary
<None>
CTI Profile
Primary
CTI_SRV
Secondary
<None>
Tertiary
<None>

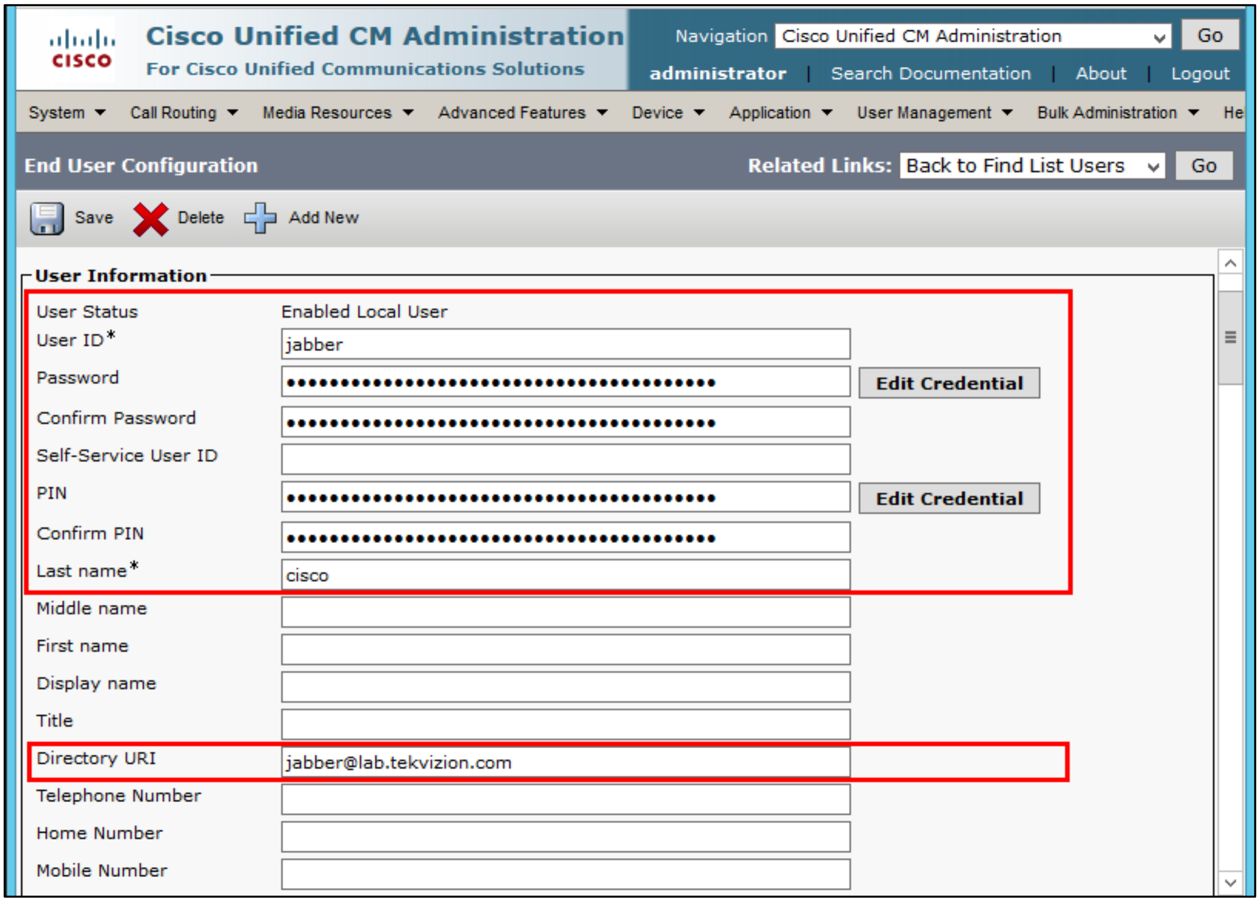
End User Configuration

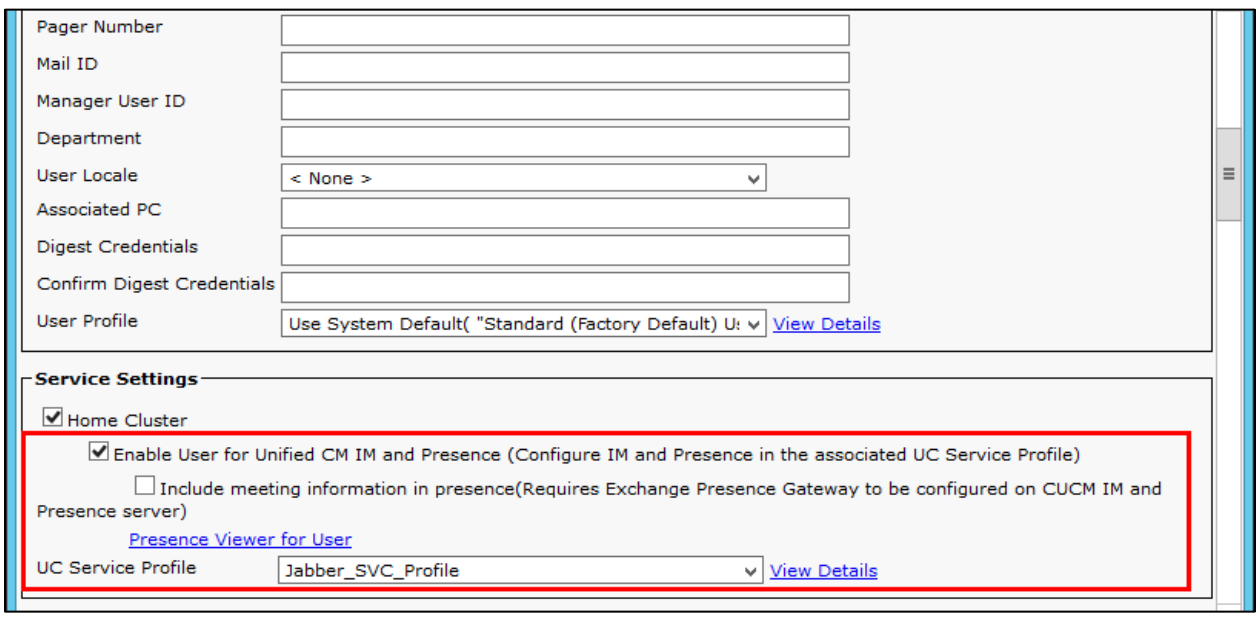
Navigation: User Management → End User

Set User ID* = jabber – This is used in this example.
Set Password = Password for profile.
Set Directory URI = jabber@lab.tekvizion.com.

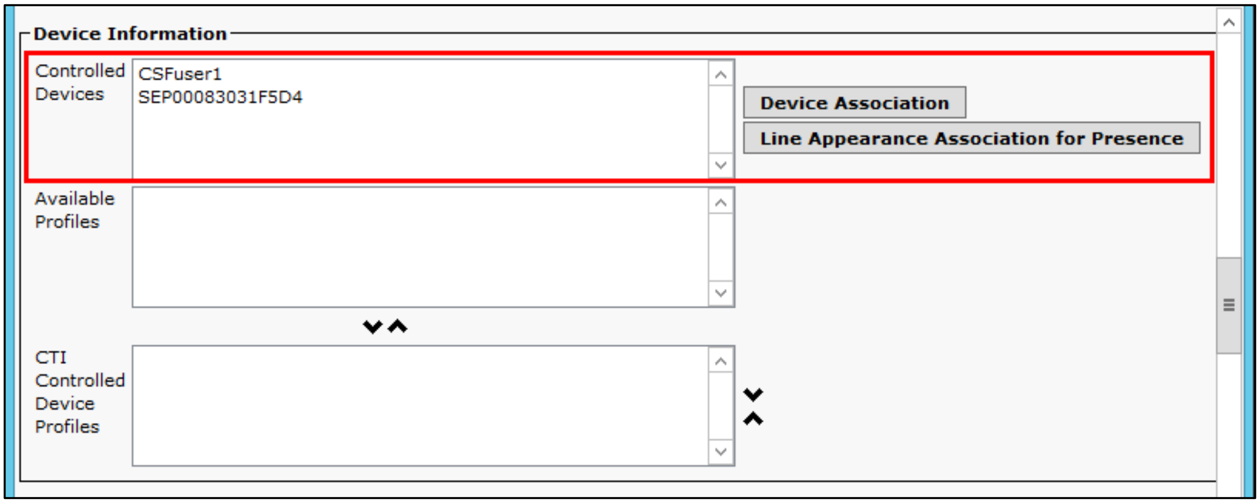


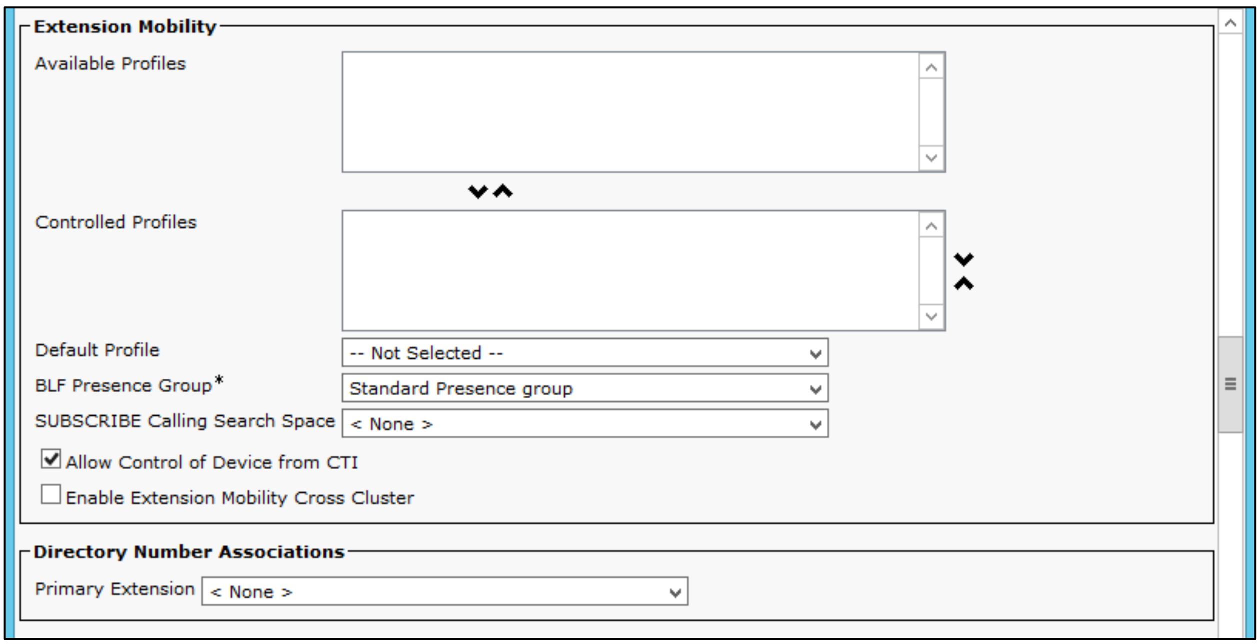
End User Configuration (continued…)


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 He
End User Configuration
Related Links: Back to Find List Users Go
Save Delete Add New
User Information
User Status Enabled Local User
User ID* jabber
Password
Edit Credential
Confirm Password
Self-Service User ID
PIN
Edit Credential
Confirm PIN
Last name* cisco
Middle name
First name
Display name
Title
Directory URI jabber@lab.tekvizion.com
Telephone Number
Home Number
Mobile Number

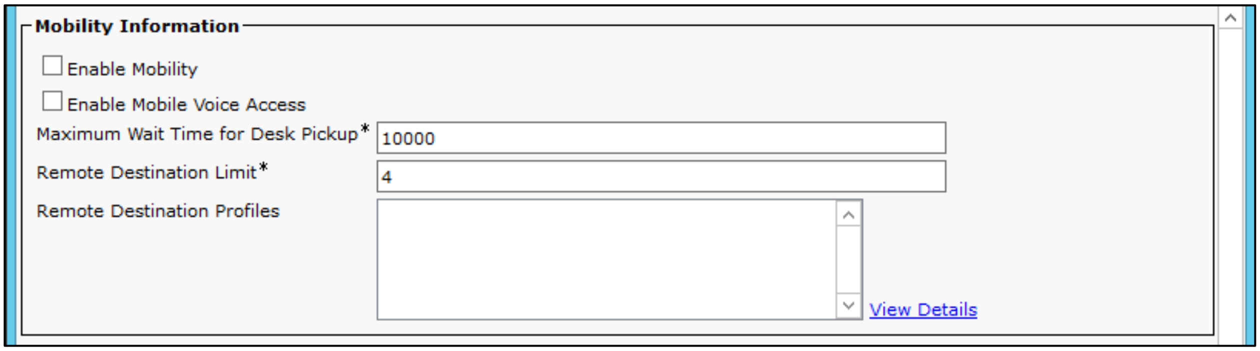

Pager Number
Mail ID
Manager User ID
Department
User Locale < None >
Associated PC
Digest Credentials
Confirm Digest Credentials
User Profile Use System Default("Standard (Factory Default) U: View Details
Service Settings
Home Cluster
Enable User for Unified CM IM and Presence (Configure IM and Presence in the associated UC Service Profile)
Include meeting information in presence(Requires Exchange Presence Gateway to be configured on CUCM IM and Presence server)
Presence Viewer for User
UC Service Profile Jabber_SVC_Profile View Details

End User Configuration(continued…)

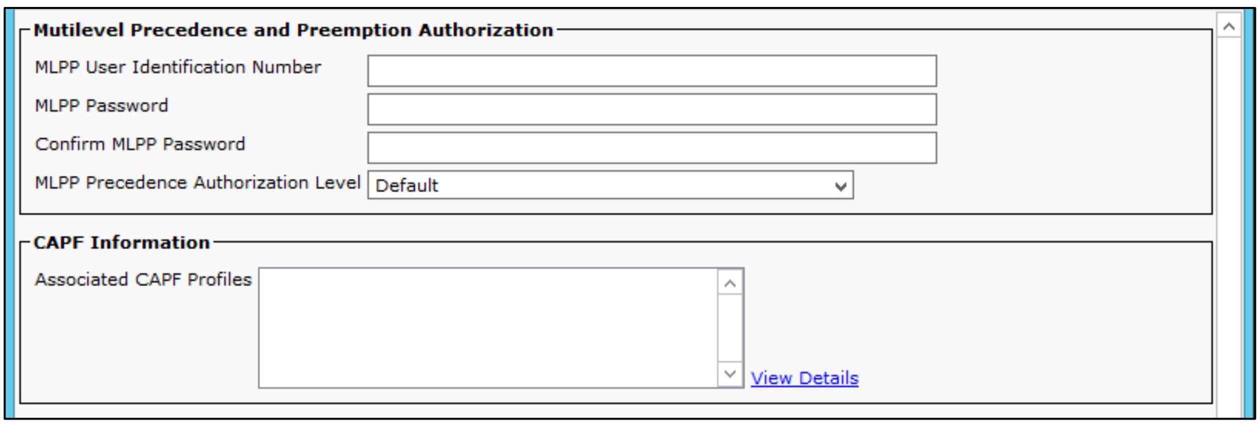


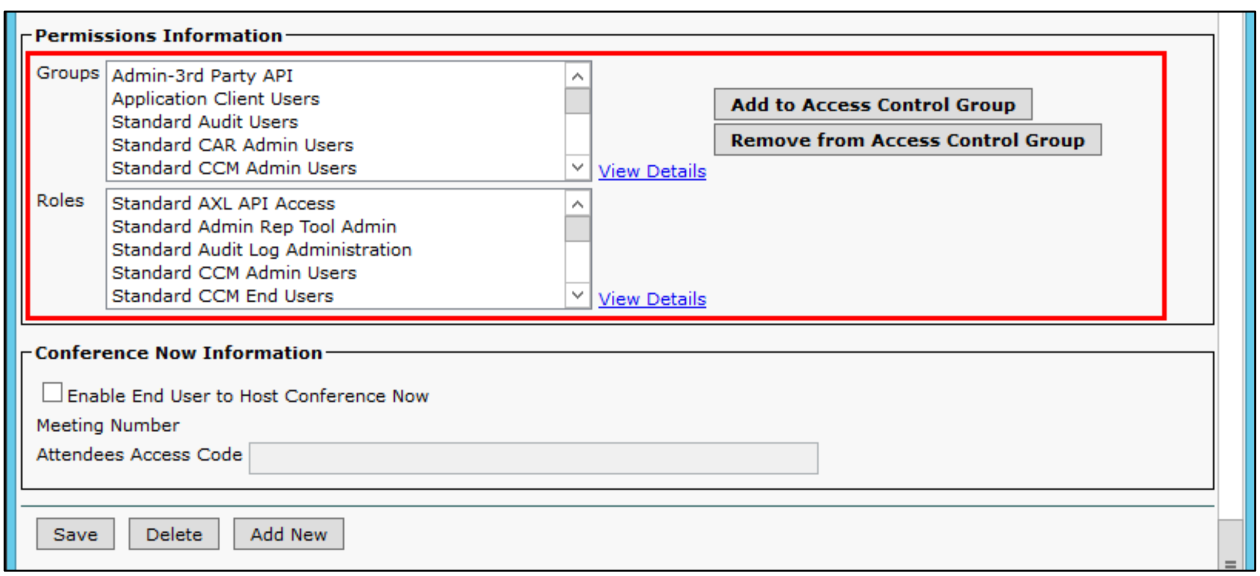


Note: Testing was conducted in tekVizion labs



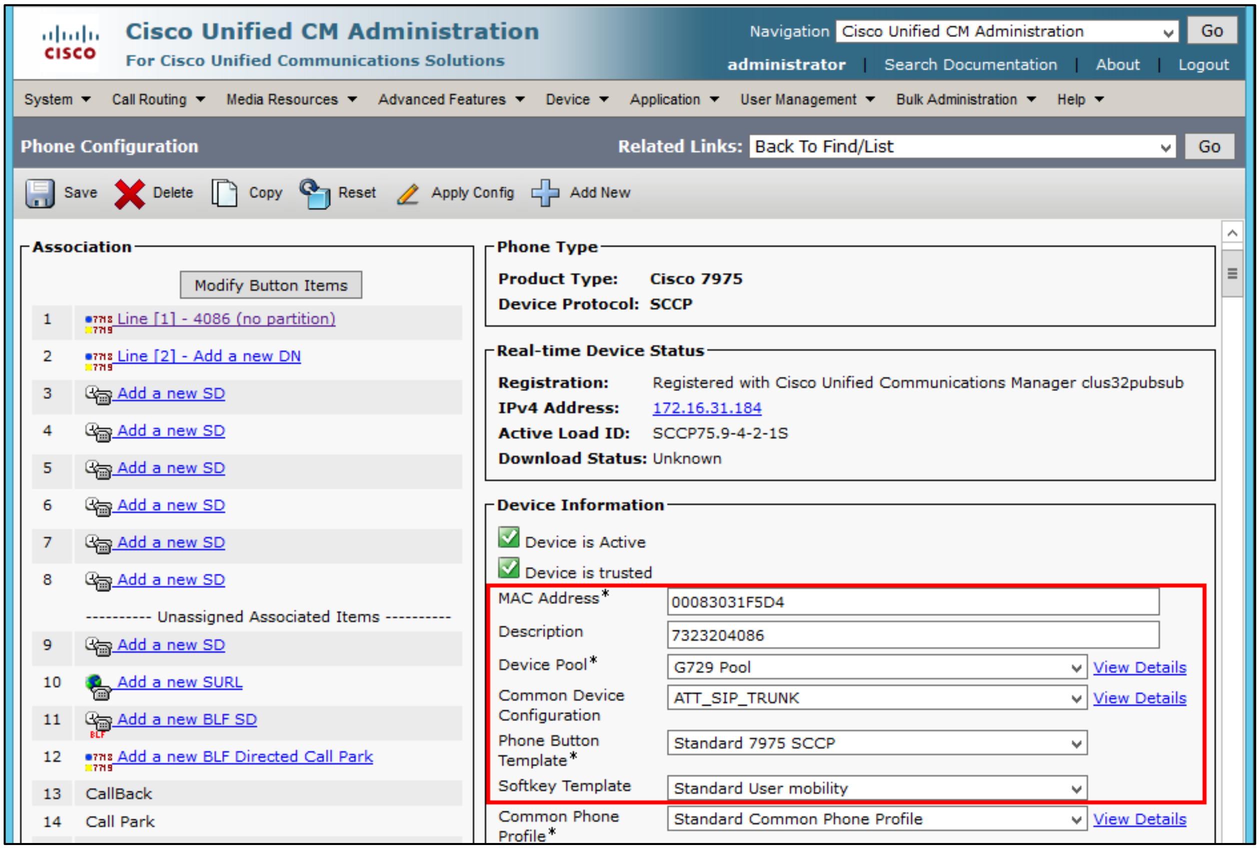
End User Configuration(continued…)





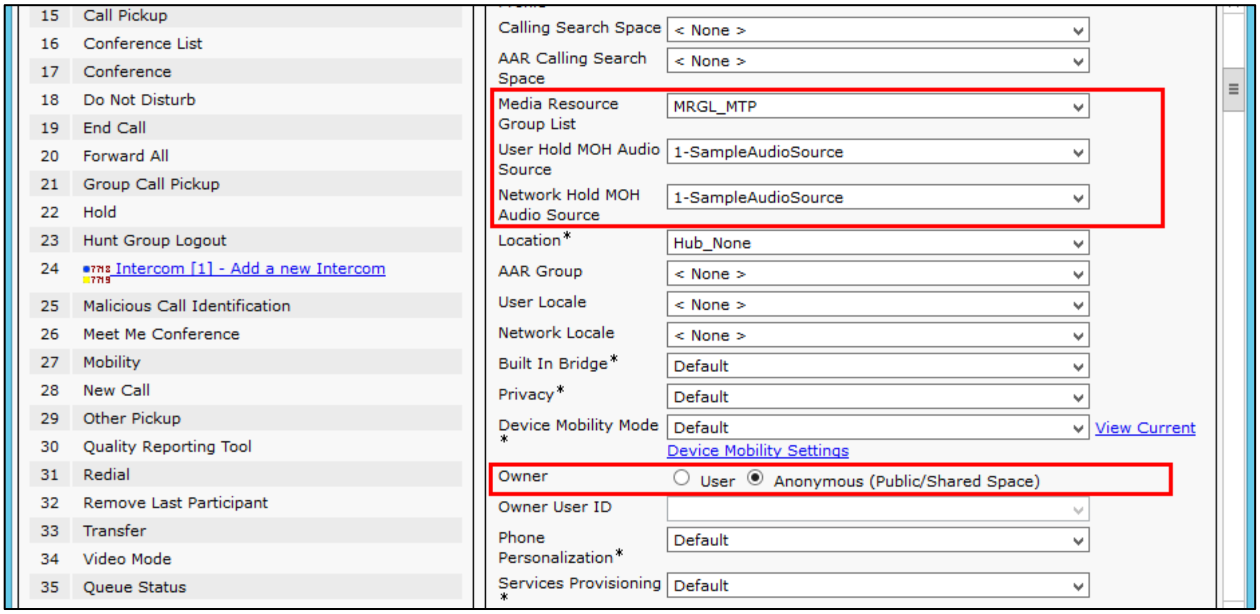
Cisco IP Phone 7975 SCCP Configuration

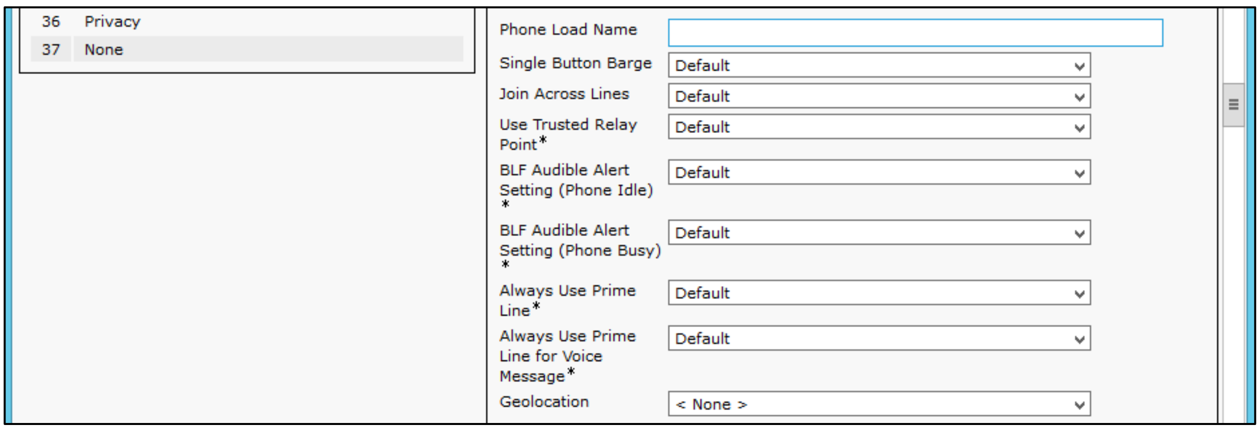
Set MAC Address* = the below mac is used in this example.
Set Description = Cisco7975_Phone. this text is used to identify this Phone.
Set Device Pool*= G729 pool. This is used in this example.
Set Phone Button Template*= Standard 7975 SCCP. This is used in this example.
Set Softkey Template = Standard User. This is used in this example.

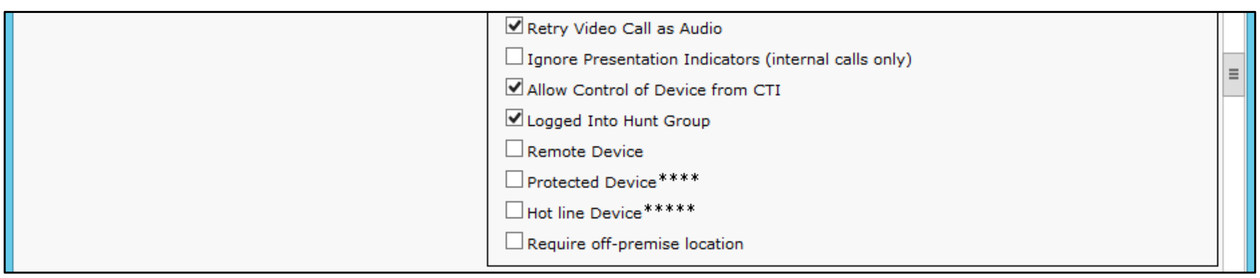


Cisco IP Phone 7975 SCCP Configuration (Continued…)

Set Media Resource Group List = MRGL_MTP. This is used in this example.
Set User Hold MOH Audio Source = 1-SampleAudioSource.
Set Network Hold MOH Audio Source = 1-SampleAudioSource.
Check Owner = Anonymous (Public/Shared Space). This is used in this example.

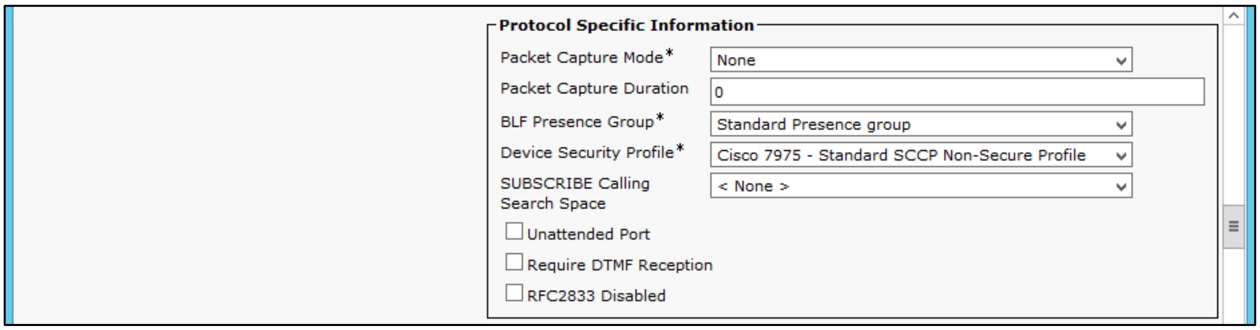


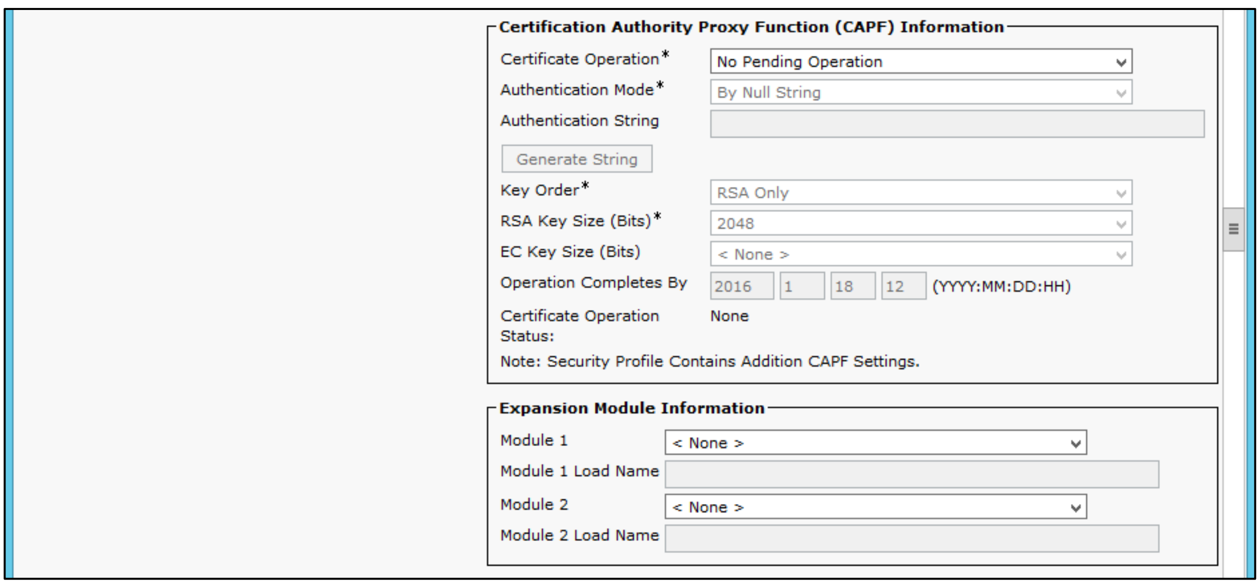




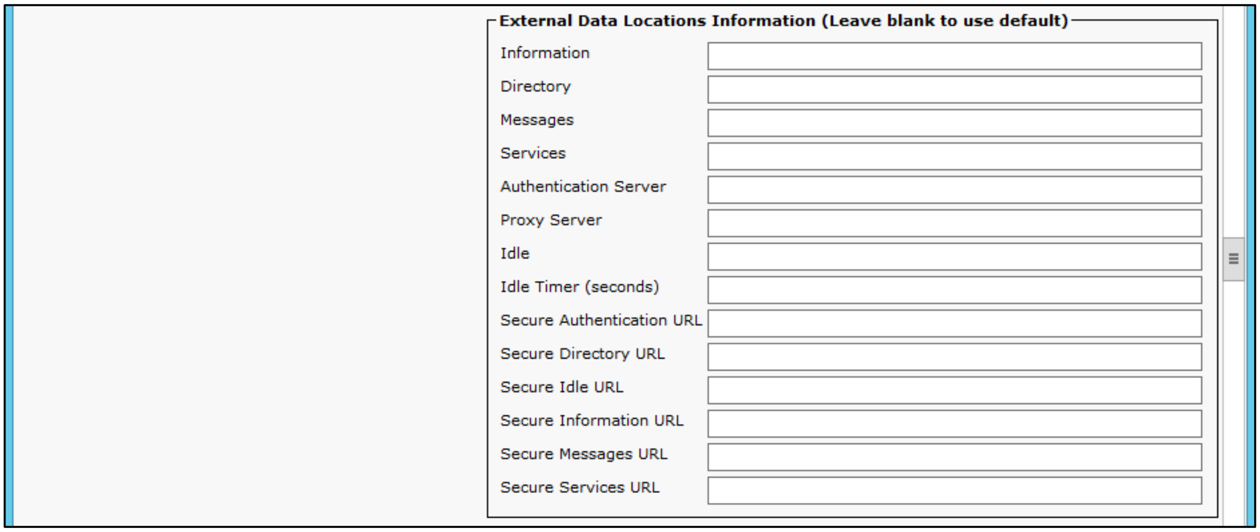
Cisco IP Phone 7975 SCCP Configuration (Continued…)

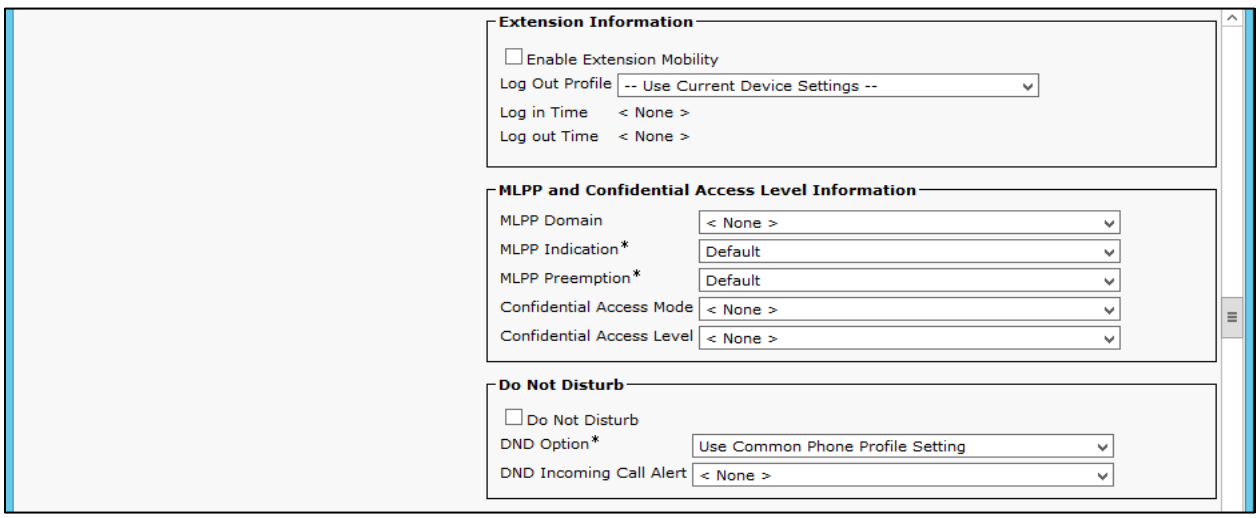

Number Presentation Transformation
Caller ID For Calls From This Phone
Calling Party Transformation CSS
< None >
Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)
Remote Number
Calling Party Transformation CSS
< None >
Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)


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

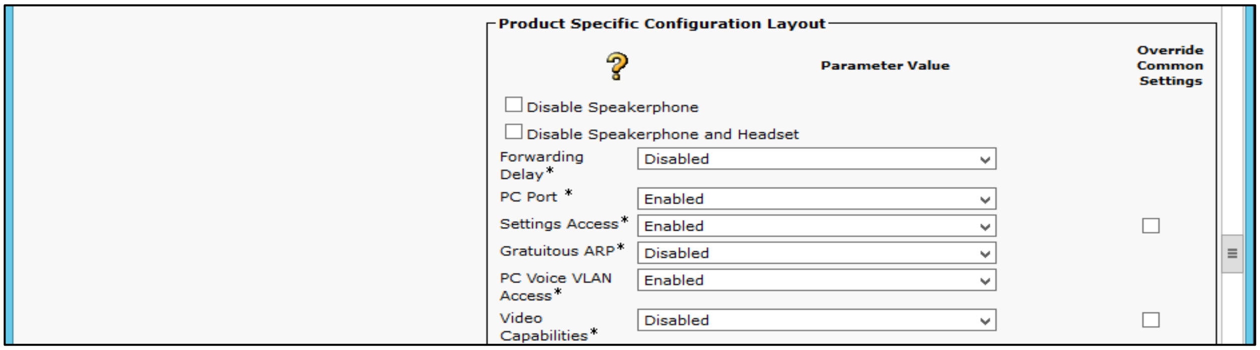

Certification Authority Proxy Function (CAPF) Information
Certificate Operation*
No Pending Operation
Authentication Mode*
By Null String
Authentication String
Generate String
Key Order*
RSA Only
RSA Key Size (Bits)*
2048
EC Key Size (Bits)
< None >
Operation Completes By
2016
1
18
12
(YYYY:MM:DD:HH)
Certificate Operation 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 7975 SCCP Configuration (Continued…)

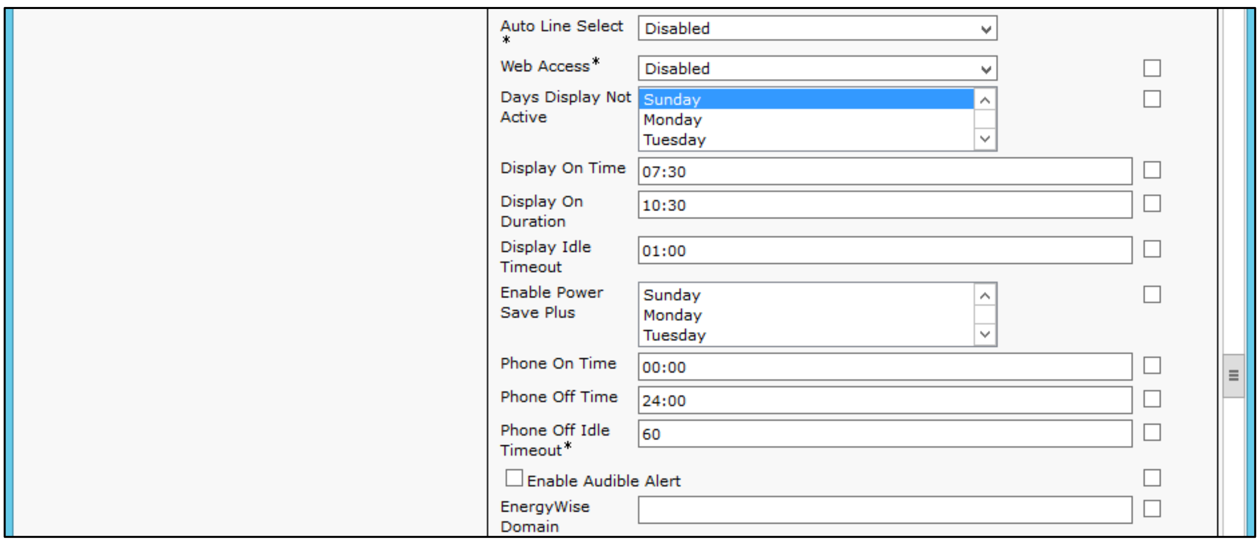


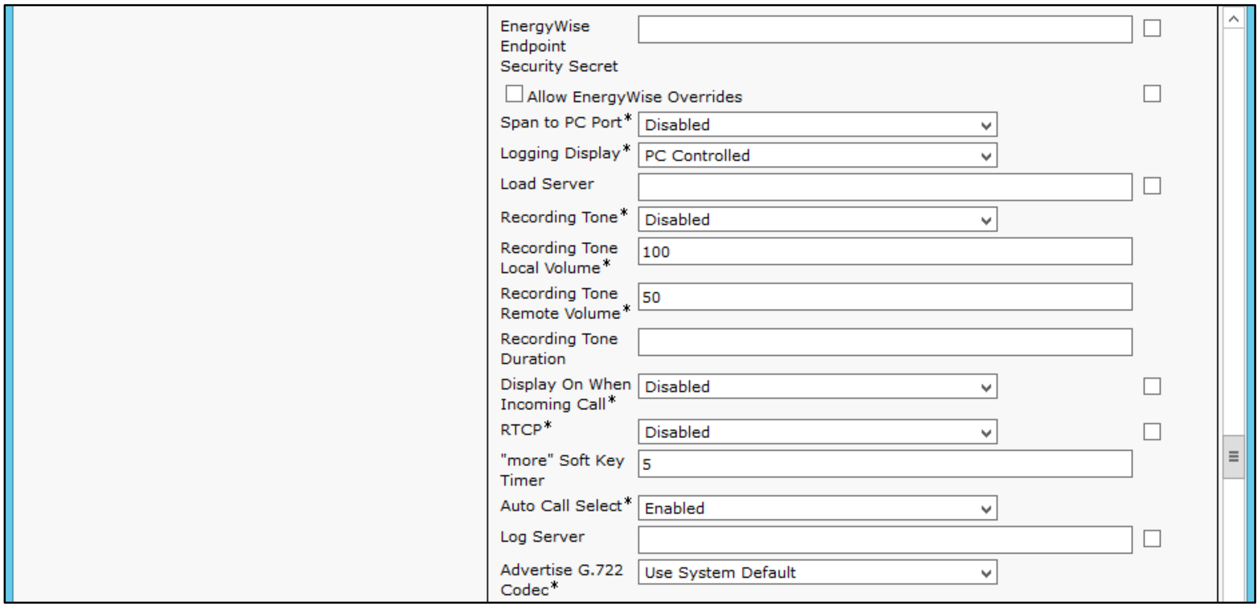


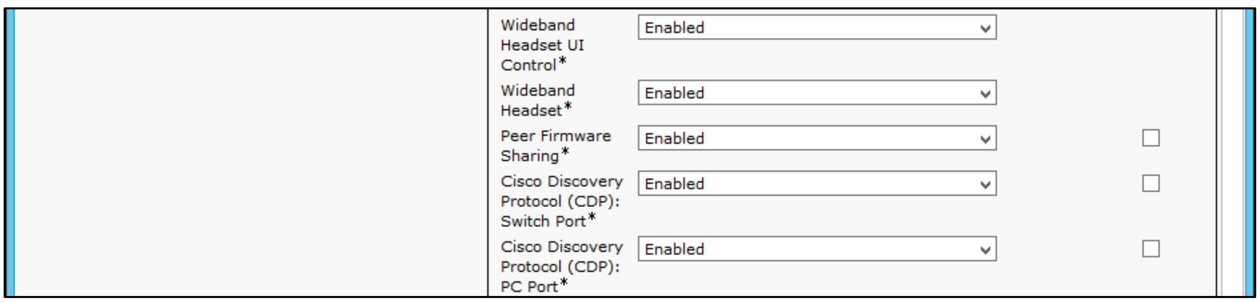
Note: Testing was conducted in tekVizion labs



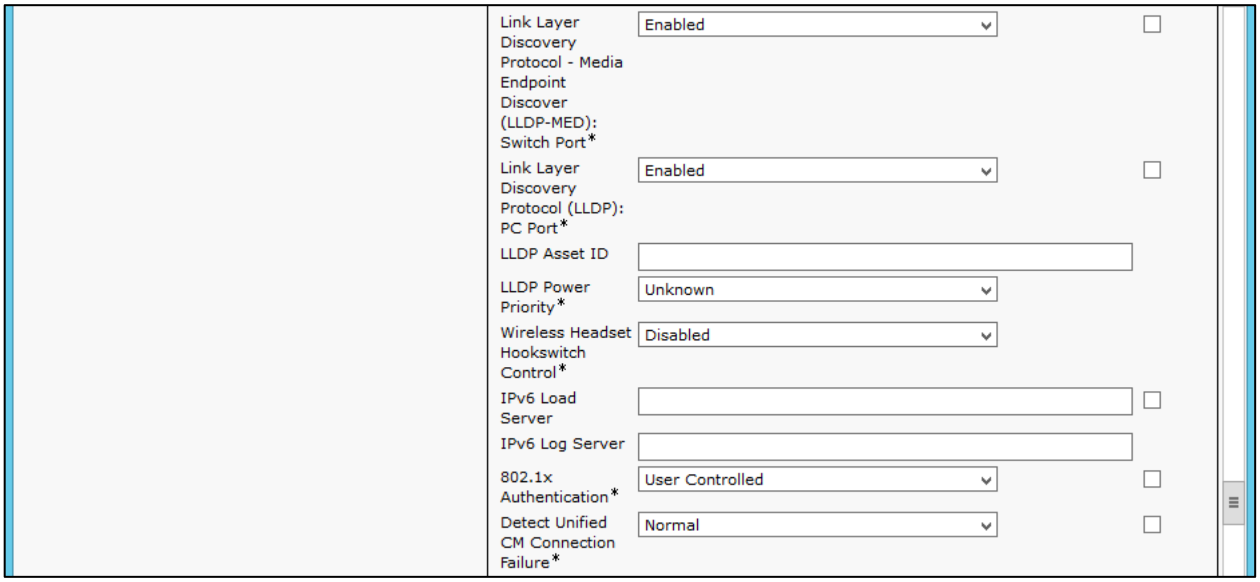
Cisco IP Phone 7975 SCCP Configuration (Continued…)

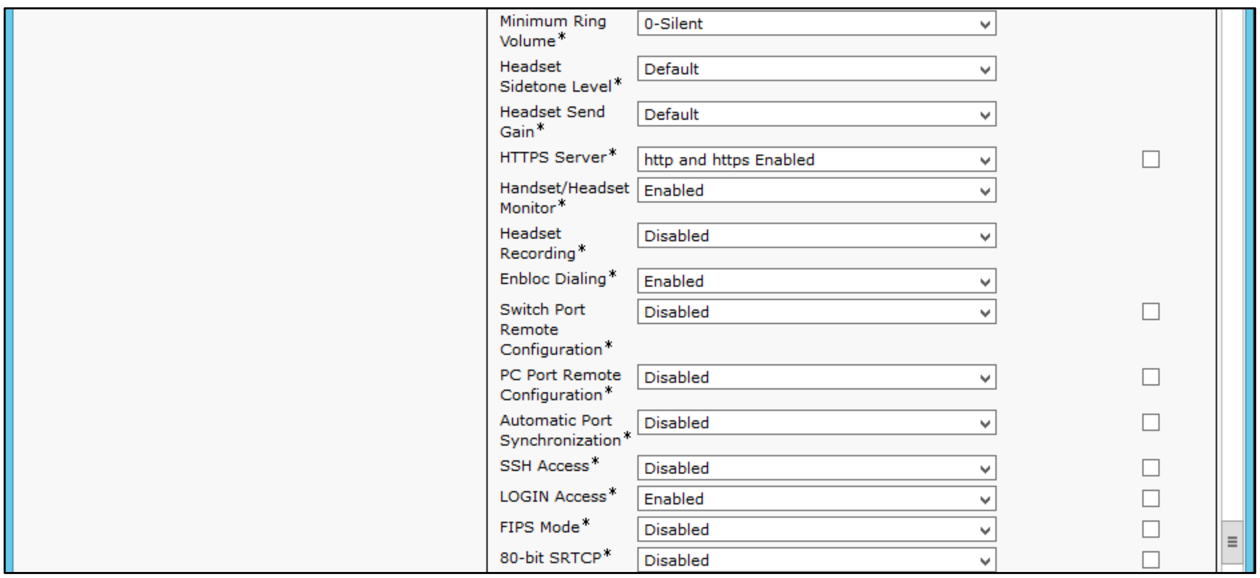


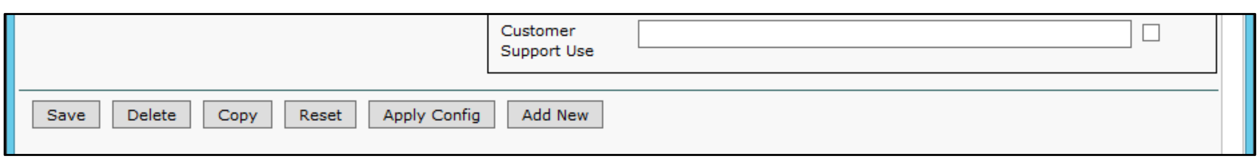




Cisco IP Phone 7975 SCCP Configuration (Continued…)

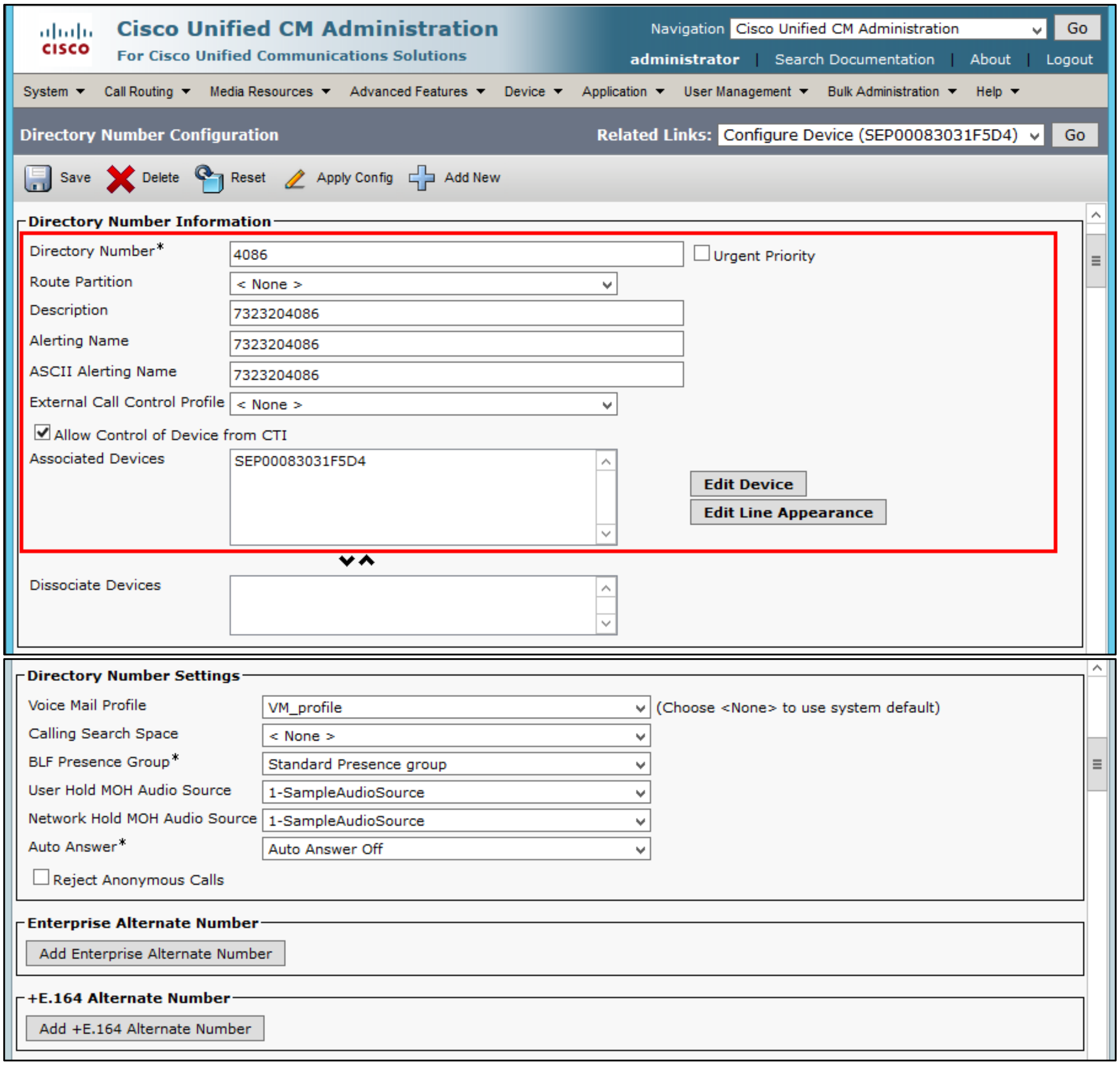

Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*
Enabled
Link Layer Discovery Protocol (LLDP): PC Port*
Enabled
LLDP Asset ID
LLDP Power Priority*
Unknown
Wireless Headset Hookswitch Control*
Disabled
IPv6 Load Server
IPv6 Log Server
802.1x Authentication*
User Controlled
Detect Unified CM Connection Failure*
Normal


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

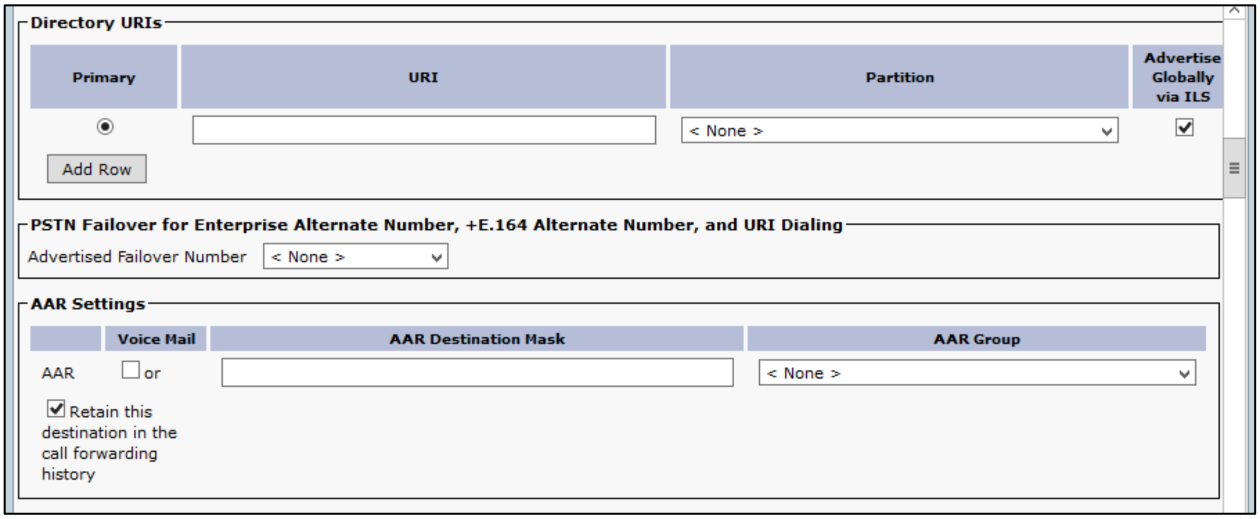

Customer Support Use
Save
Delete
Copy
Reset
Apply Config
Add New

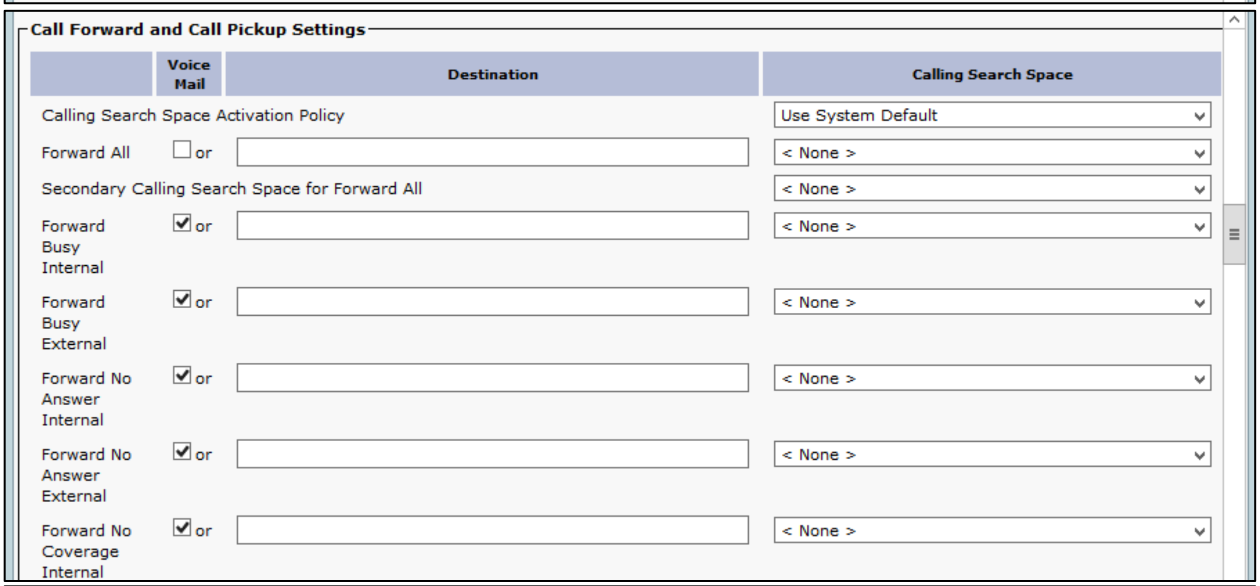
Cisco IP Phone 7975 SCCP Configuration (Continued…)

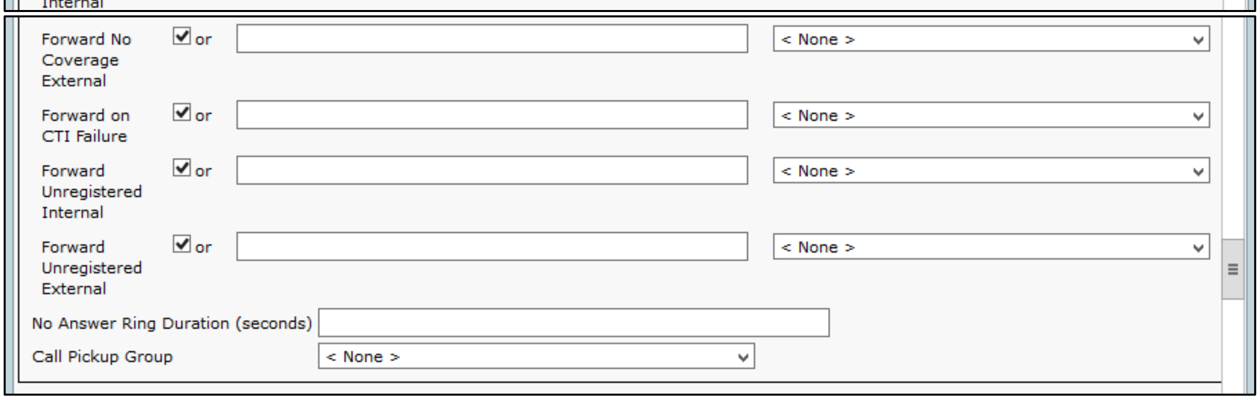
Set Directory Number* = 4086. This is used in this example.
Set Description = 7323204086. This is used in this example.
Set Alerting Name = 7323204086. This is used in this example.
Set ASCII Alerting Name = 7323204086. This is used in this example.



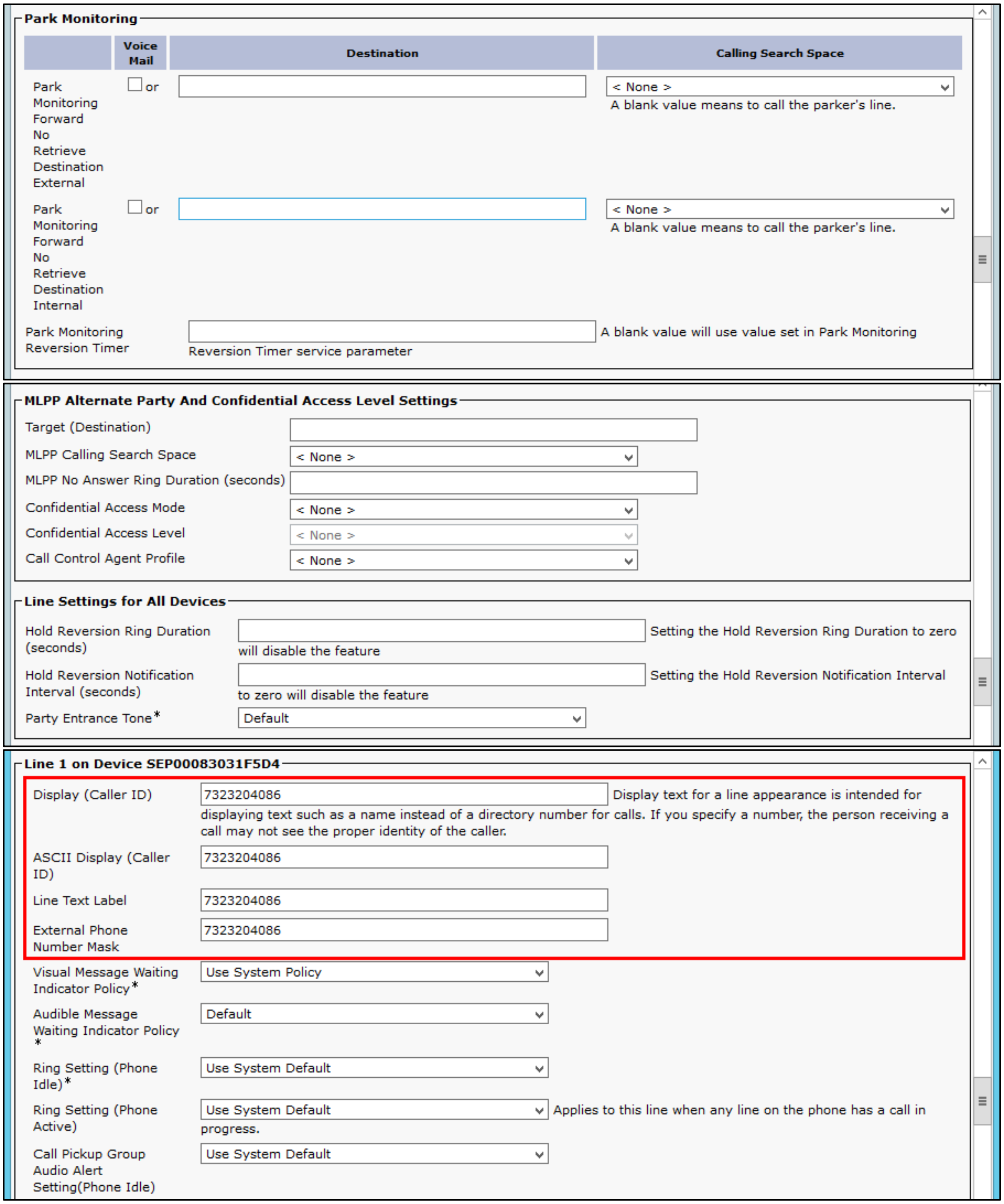
Cisco IP Phone 7965 SCCP Configuration (Continued…)







Cisco IP Phone 7965 SCCP Configuration (Continued…)



Park Monitoring

	Voice Mail	Destination	Calling Search Space
Park Monitoring Forward No Retrieve Destination External	or		< None > A blank value means to call the parker's line.
Park Monitoring Forward No Retrieve Destination Internal	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 Reversion Timer service parameter

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 >

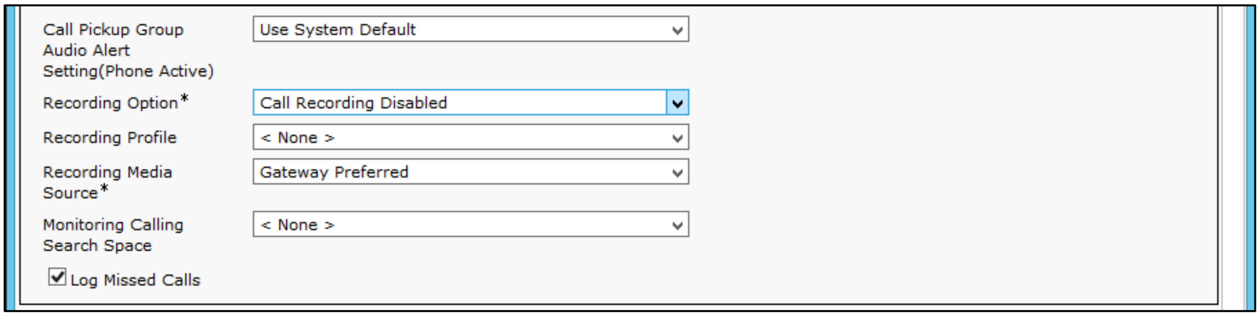
Line Settings for All Devices

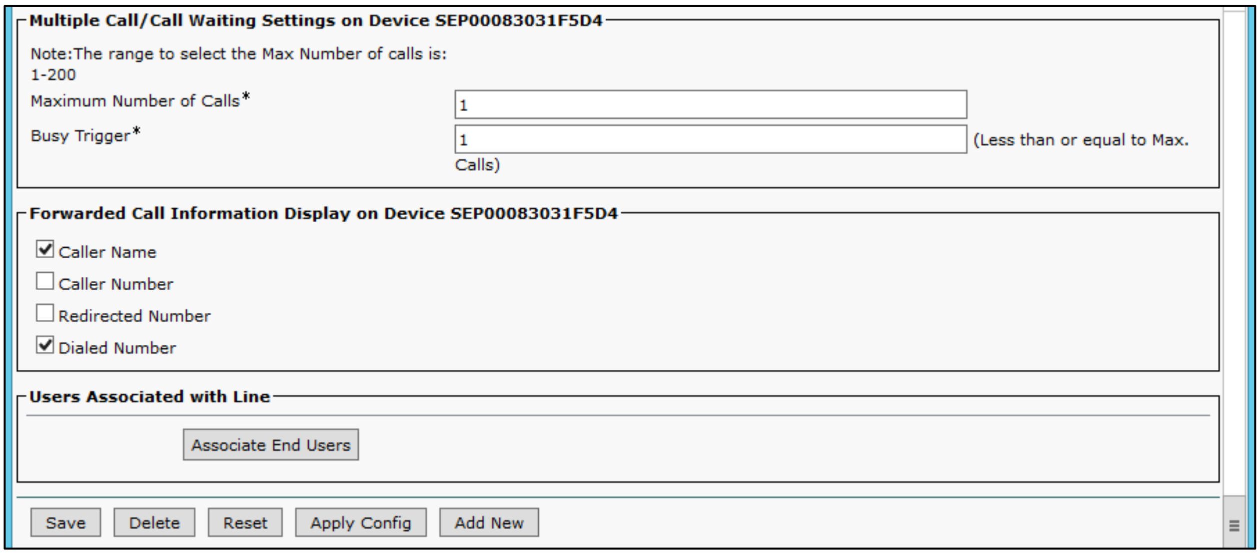
Hold Reversion Ring Duration (seconds)	Setting the Hold Reversion Ring Duration to zero will disable the feature
Hold Reversion Notification Interval (seconds)	Setting the Hold Reversion Notification Interval to zero will disable the feature
Party Entrance Tone*	Default

Line 1 on Device SEP00083031F5D4

Display (Caller ID)	7323204086 Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Caller ID)	7323204086
Line Text Label	7323204086
External Phone Number Mask	7323204086
Visual Message Waiting Indicator Policy*	Use System Policy
Audible Message Waiting Indicator Policy *	Default
Ring Setting (Phone Idle)*	Use System Default
Ring Setting (Phone Active)	Use System Default Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)	Use System Default

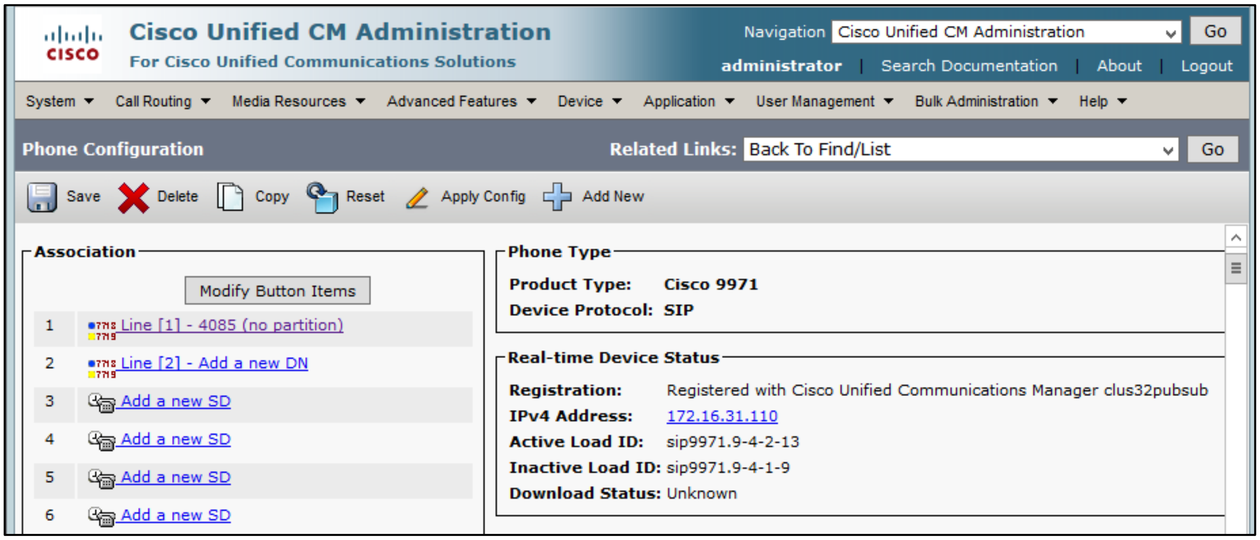
Cisco IP Phone 7965 SCCP Configuration (Continued…)

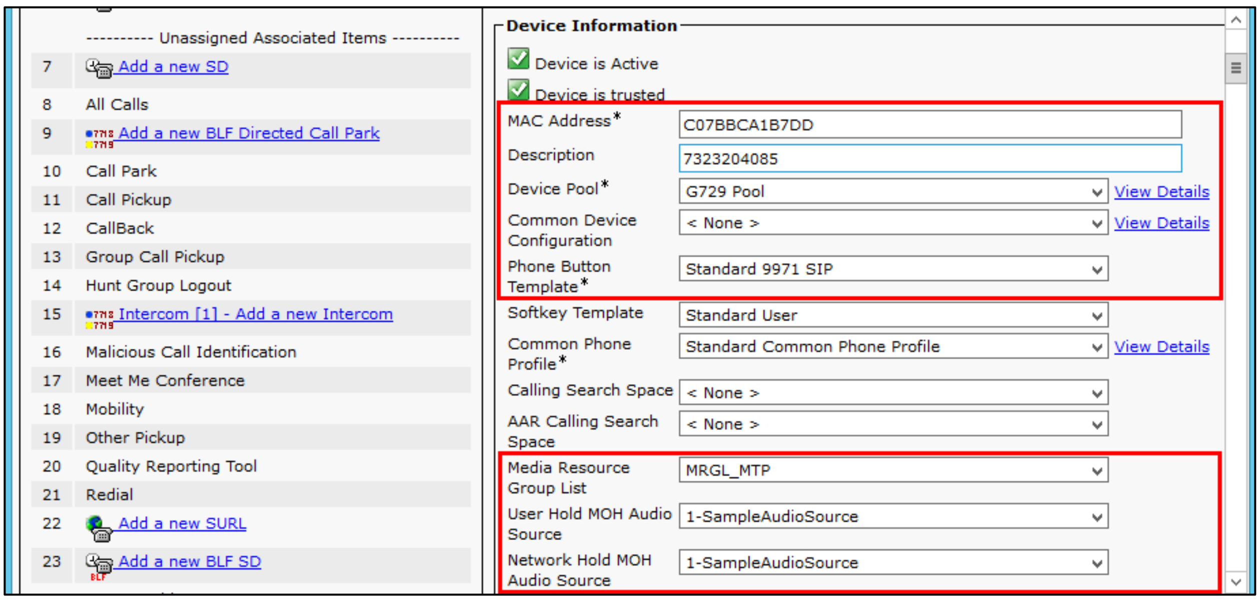




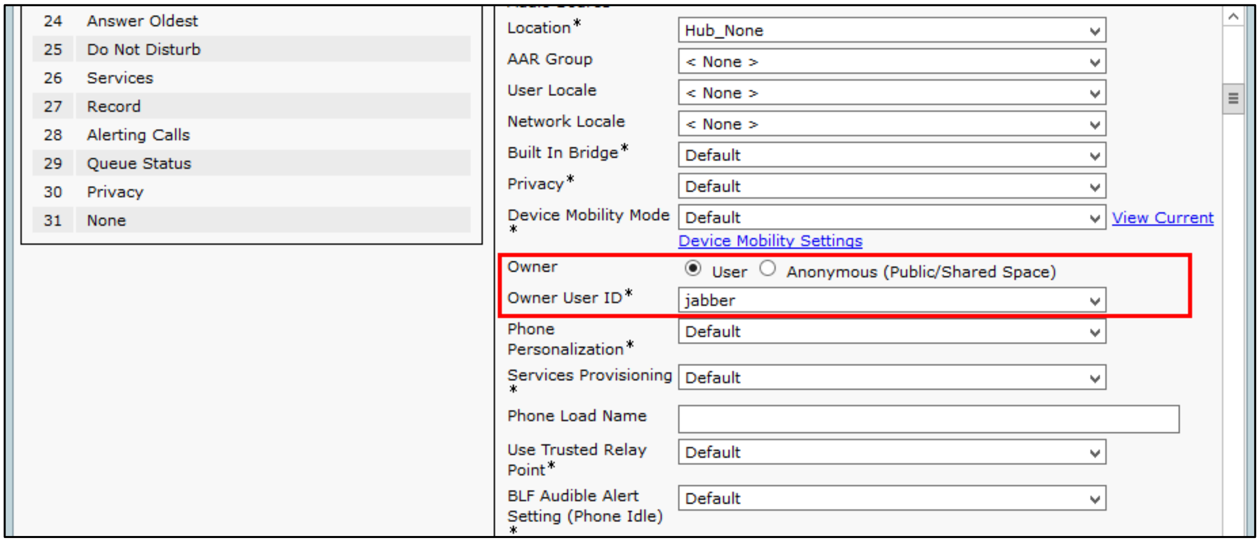
Cisco IP Phone 9971 SIP Configuration

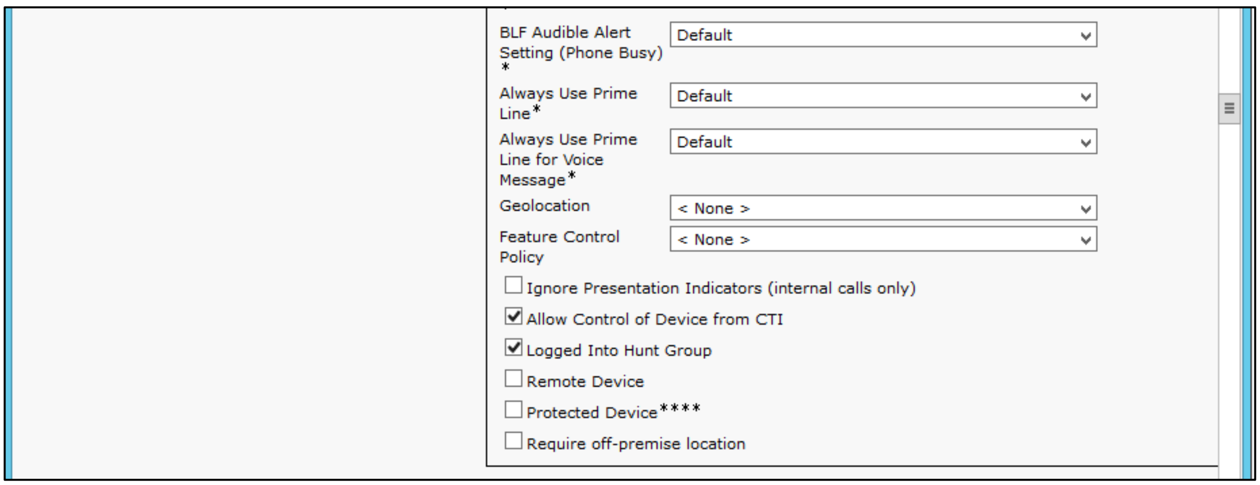
Set MAC Address* = the below mac is used in this example.
Set Description = 7323204085. this text is used to identify this Phone.
Set Device Pool*= G729 Pool. This is used in this example.
Set Phone Button Template*= Standard 9971 SIP. This is used in this example.
Set Media Resource Group List = MRGL_MTP. This is used in this example.
Set User Hold MOH Audio Source = 1-SampleAudioSource.
Set Network Hold MOH Audio Source = 1-SampleAudioSource

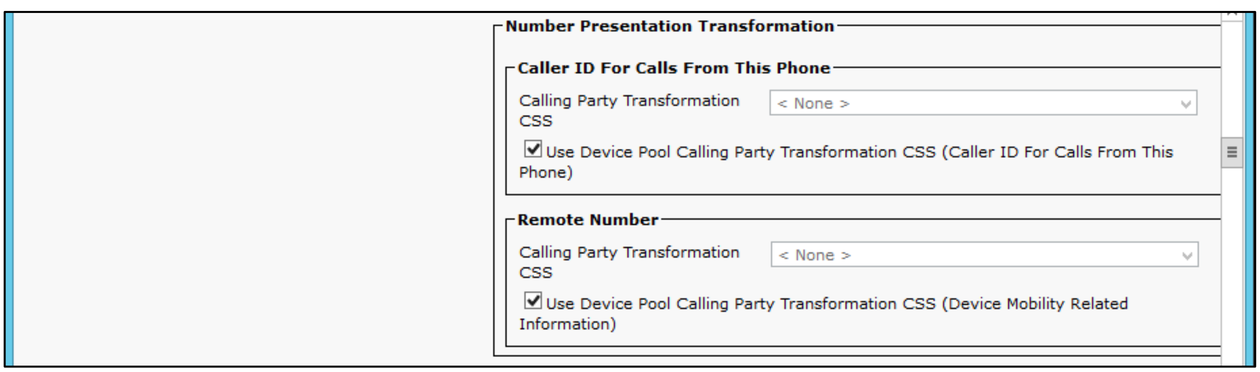




Cisco IP Phone 9971 SIP Configuration (Continued…)


24 Answer Oldest
25 Do Not Disturb
26 Services
27 Record
28 Alerting Calls
29 Queue Status
30 Privacy
31 None
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* jabber
Phone Personalization* Default
Services Provisioning* Default
Phone Load Name
Use Trusted Relay Point* Default
BLF Audible Alert Setting (Phone Idle)* Default


BLF Audible Alert Setting (Phone Busy)* Default
Always Use Prime Line* Default
Always Use Prime Line for Voice Message* Default
Geolocation < None >
Feature Control Policy < None >
Ignore Presentation Indicators (internal calls only)
Allow Control of Device from CTI
Logged Into Hunt Group
Remote Device
Protected Device****
Require off-premise location


Number Presentation Transformation
Caller ID For Calls From This Phone
Calling Party Transformation CSS < None >
Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)
Remote Number
Calling Party Transformation CSS < None >
Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

Cisco IP Phone 9971 SIP Configuration (Continued…)

Protocol Specific Information

Packet Capture Mode*	None
Packet Capture Duration	0
BLF Presence Group*	Standard Presence group
SIP Dial Rules	< None >
MTP Preferred Originating Codec*	711ulaw
Device Security Profile*	Cisco 9971 - Standard SIP Non-Secure Profile
Rerouting Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile w/Early Media Disabled View Details
Digest User	< None >

- [ ] Media Termination Point Required
- [ ] Unattended Port
- [ ] Require DTMF Reception

Certification Authority Proxy Function (CAPF) Information

Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
Generate String	
Key Order*	RSA Only
RSA Key Size (Bits)*	2048
EC Key Size (Bits)	< None >
Operation Completes By	2016 1 21 12 (YYYY:MM:DD:HH)
Certificate Operation Status:	None

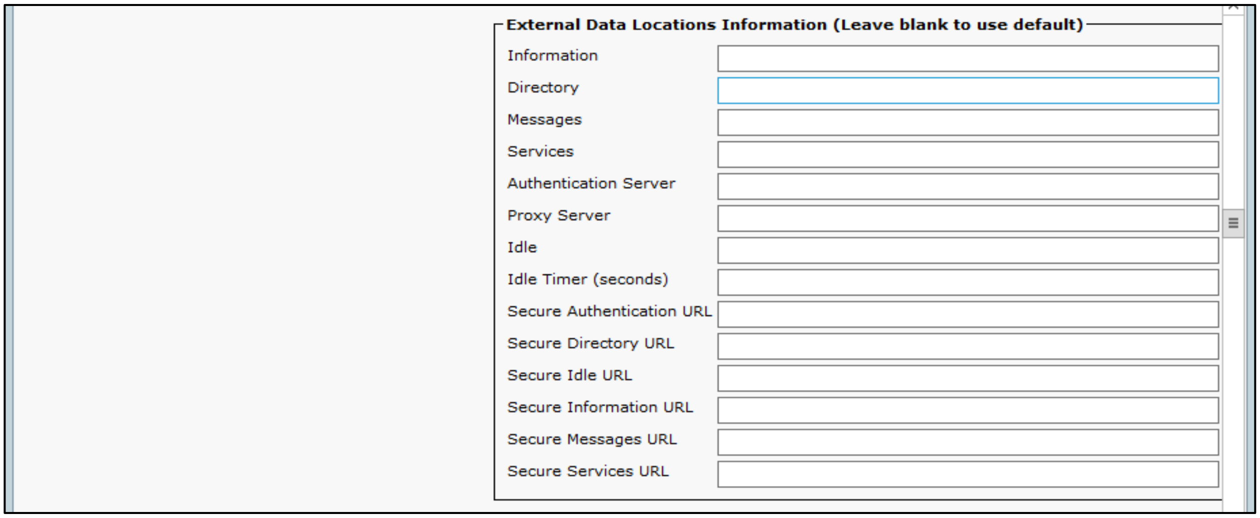
Note: Security Profile Contains Addition CAPF Settings.

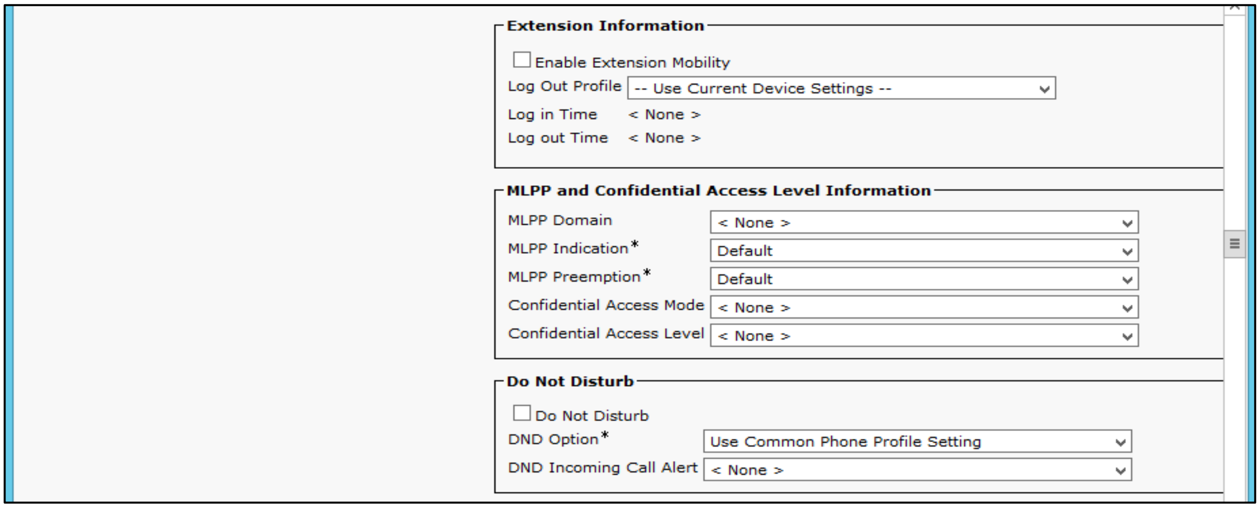
Expansion Module Information

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

Note: Testing was conducted in tekVizion labs

Cisco IP Phone 9971 SIP Configuration (Continued…)





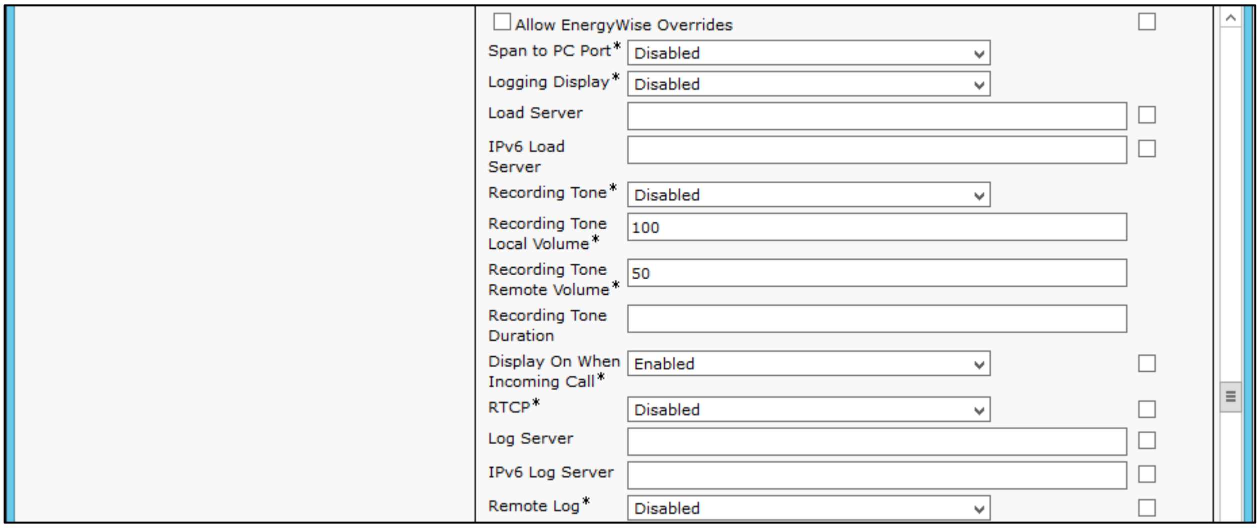
Product Specific Configuration Layout

	Parameter Value	Override Common Settings
☐ Disable Speakerphone		
☐ Disable Speakerphone and Headset		
PC Port *	Enabled	
Back USB Port*	Enabled	☐
Side USB Port*	Enabled	☐
Cisco Camera*	Disabled	☐
Console Access*	Disabled	☐
Video Capabilities*	Disabled	☐

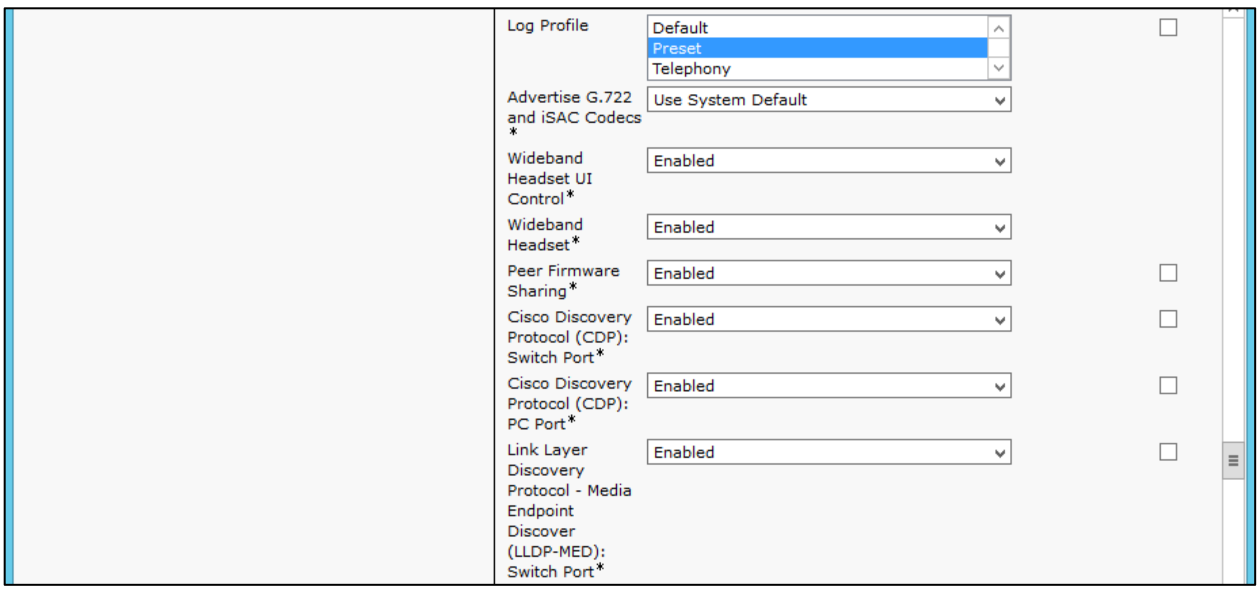
Cisco IP Phone 9971 SIP Configuration (Continued…)

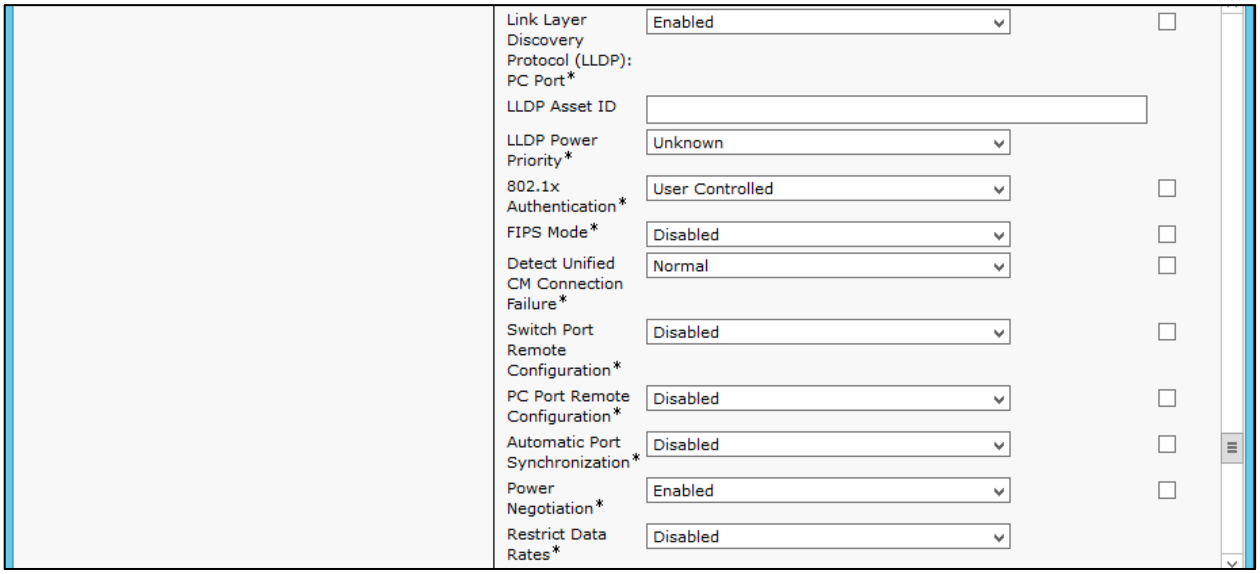
Enable/Disable USB Classes	Mass Storage Human Interface Device Audio Class	☐
SDIO *	Disabled	☐
Bluetooth *	Enabled	☐
Wifi *	Enabled	☐
Bluetooth Profiles*	Handsfree Human Interface Device	☐
Settings Access*	Enabled	☐
Gratuitous ARP*	Disabled	☐
PC Voice VLAN Access*	Enabled	
Web Access*	Disabled	☐
Show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

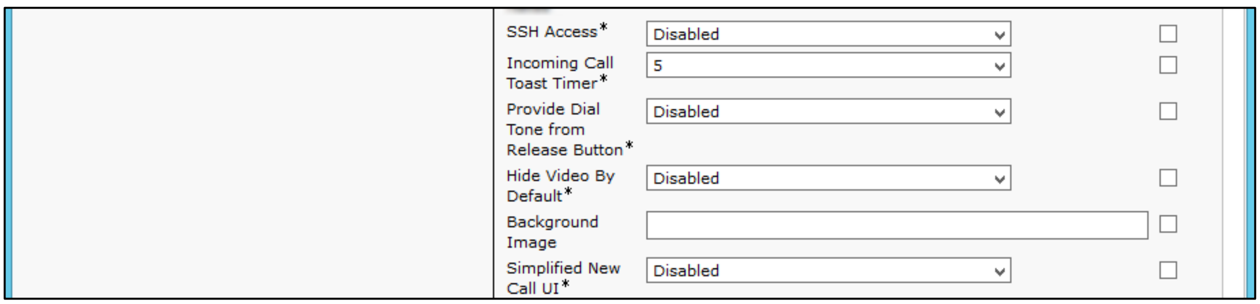
Display On Time	07:30	☐
Display On Duration	10:30	☐
Display Idle Timeout	01:00	☐
HTTPS Server*	http and https Enabled	☐
Enable Power Save Plus	Sunday Monday Tuesday	☐
Phone On Time	00:00	☐
Phone Off Time	24:00	☐
Phone Off Idle Timeout*	60	☐
☐ Enable Audible Alert		☐
EnergyWise Domain		☐
EnergyWise Endpoint Security Secret		☐



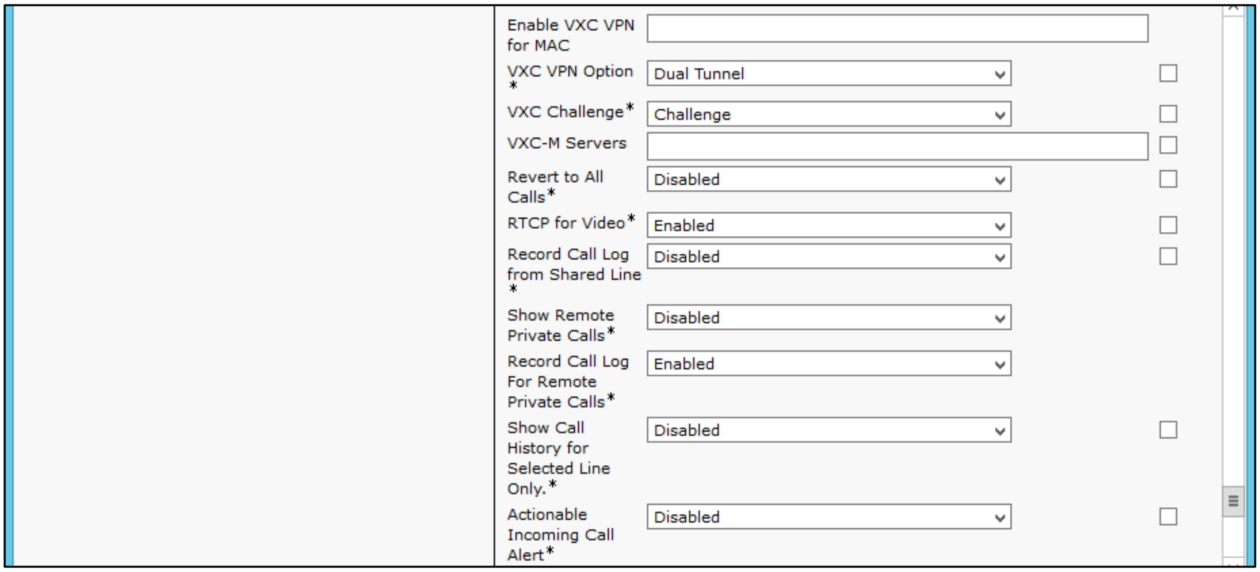
Cisco IP Phone 9971 SIP Configuration (Continued…)

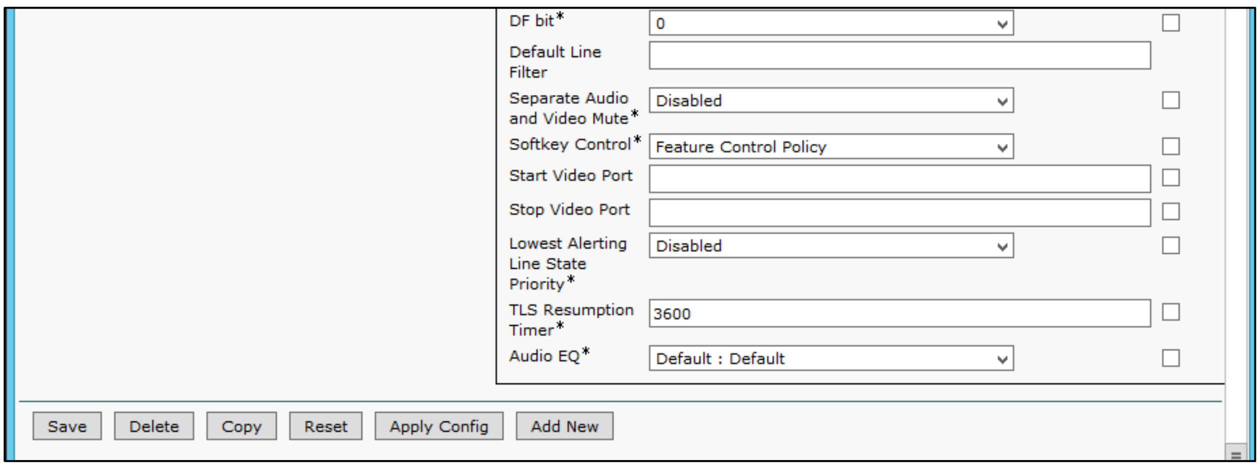






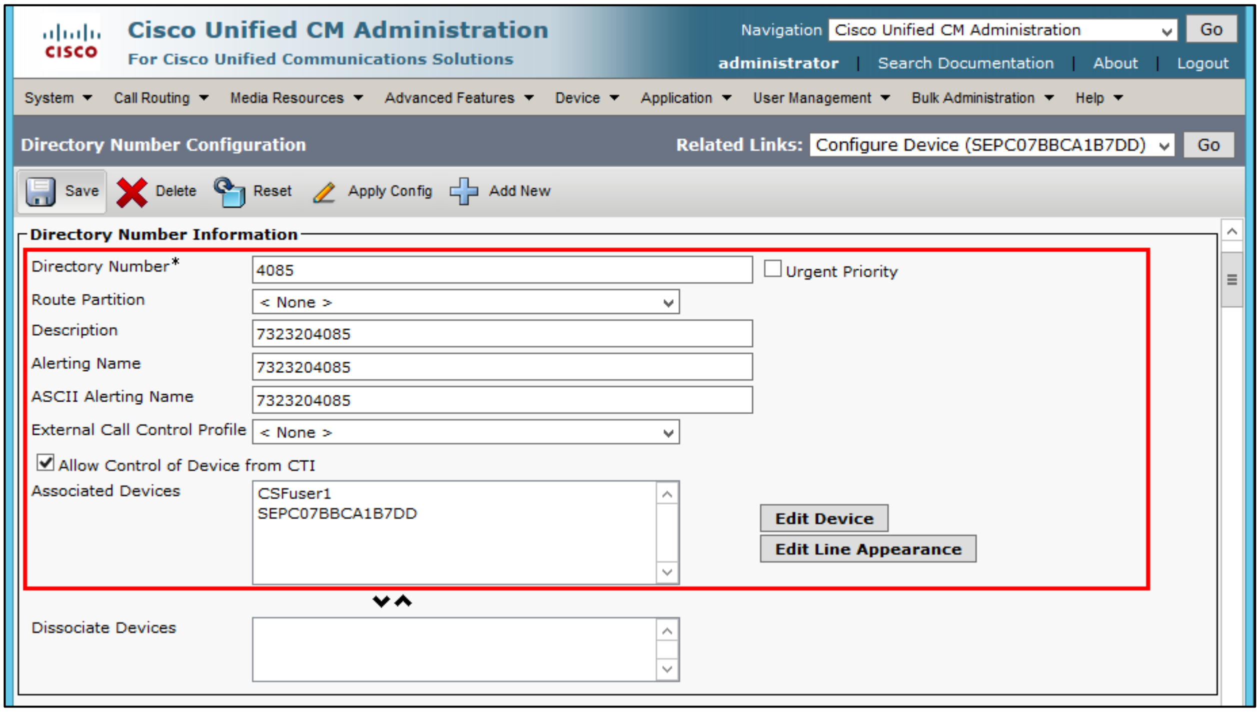
Cisco IP Phone 9971 SIP Configuration (Continued…)

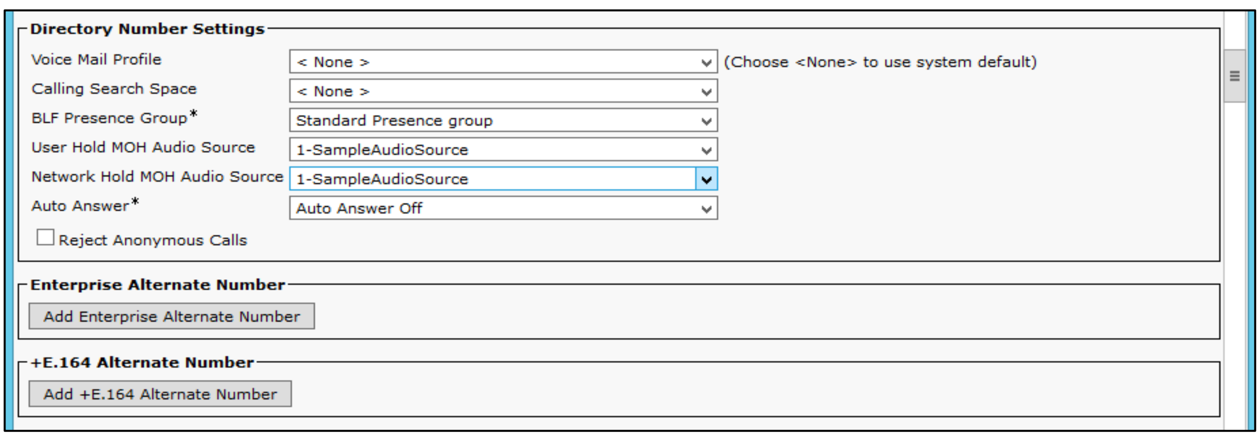


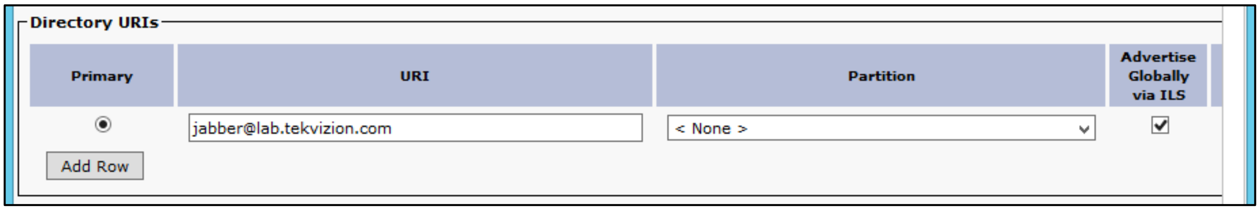


Cisco IP Phone 9971 SIP Configuration (Continued…)

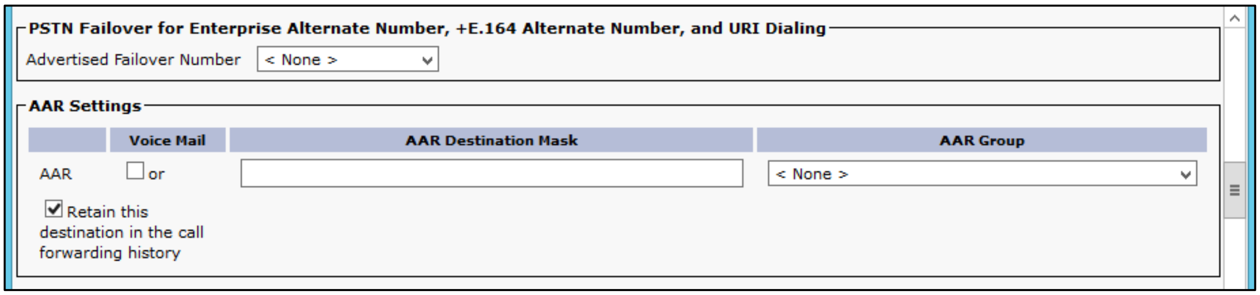
Set Directory Number* = 4084. This is used in this example.
Set Description = 7323204084. This is used in this example.
Set Alerting Name = Cisco 9971 Phone. This is used in this example.
Set ASCII Alerting Name = Cisco 9971 Phone. This is used in this example.

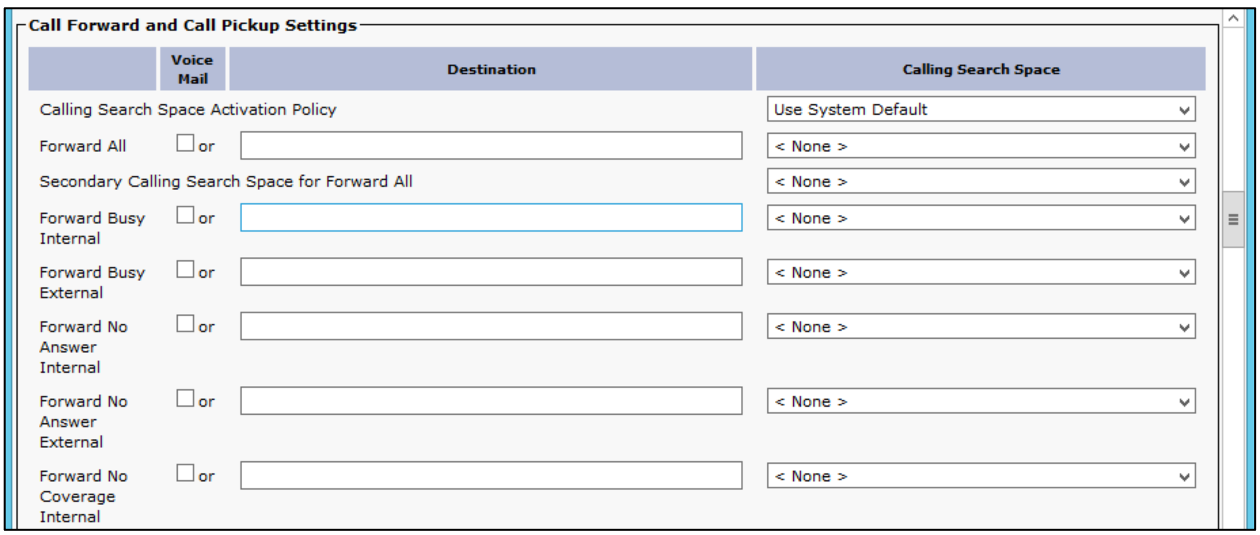


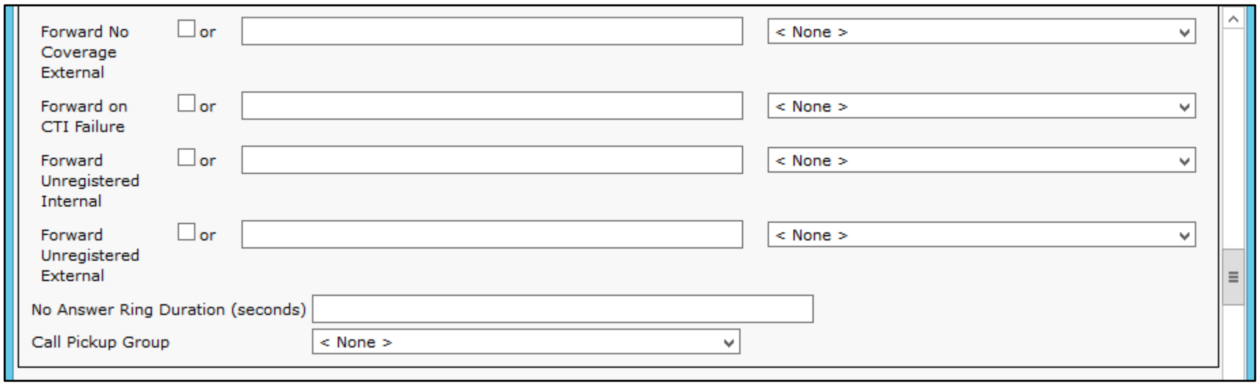




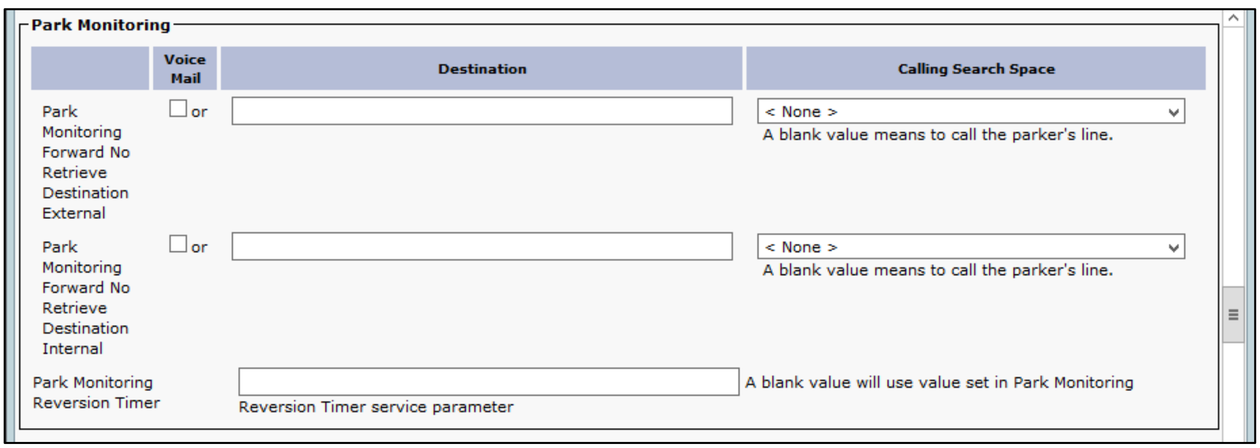
Cisco IP Phone 9971 SIP Configuration (Continued…)

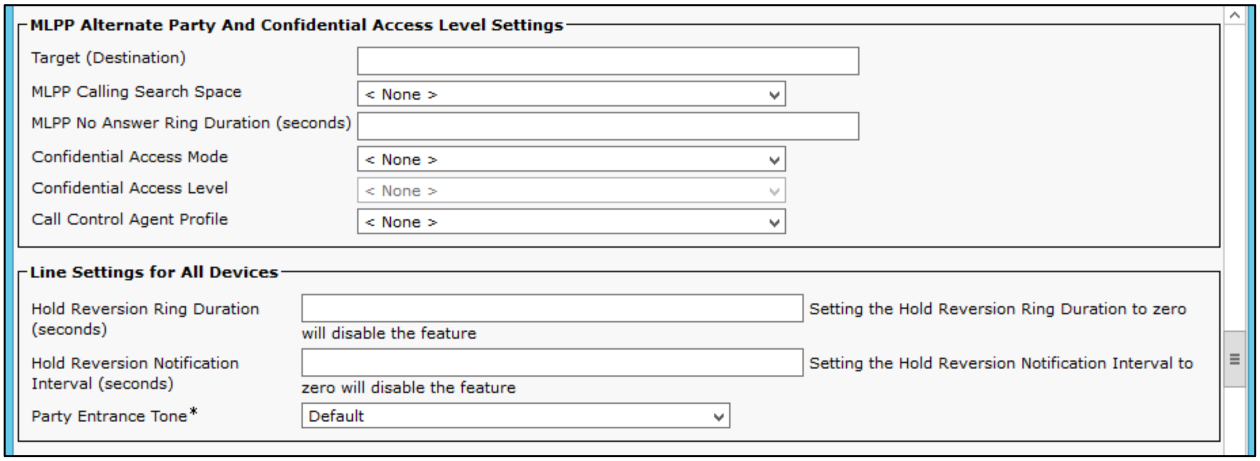


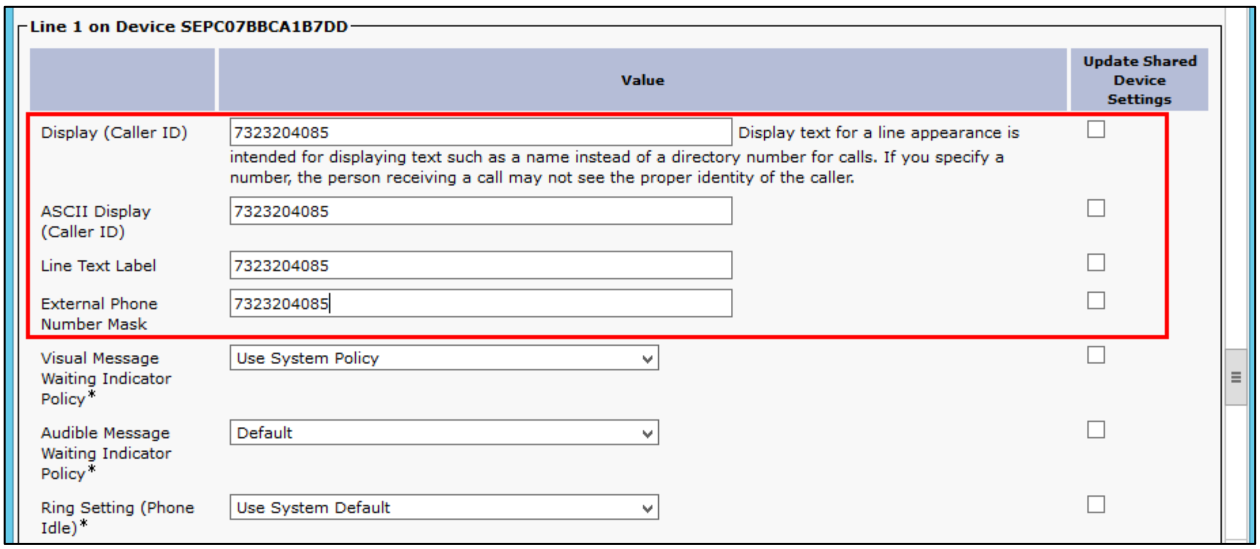




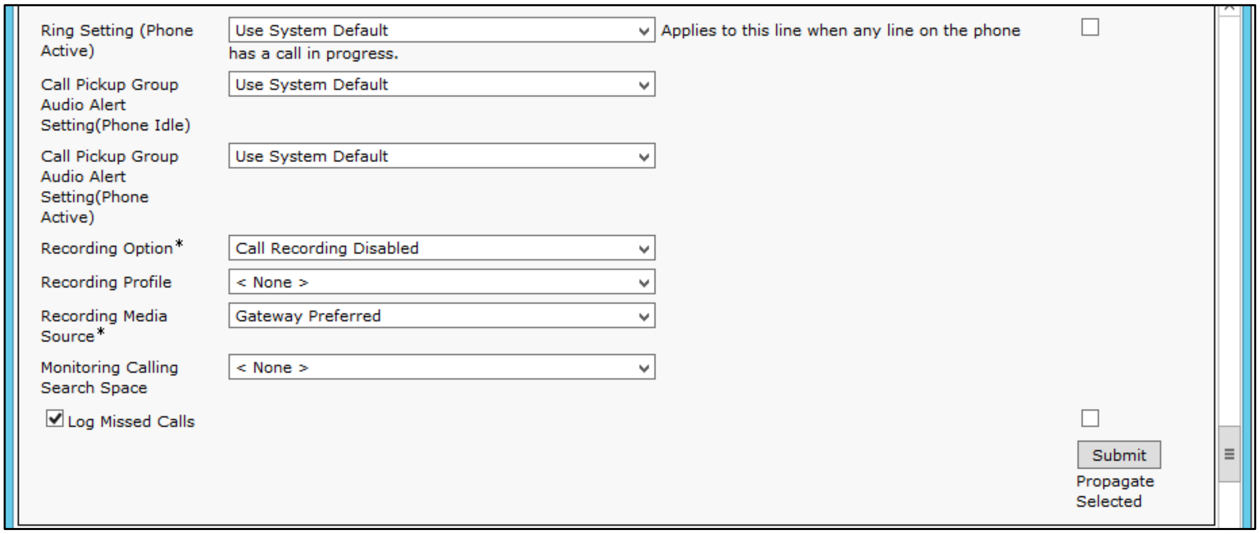
Cisco IP Phone 9971 SIP Configuration (Continued…)

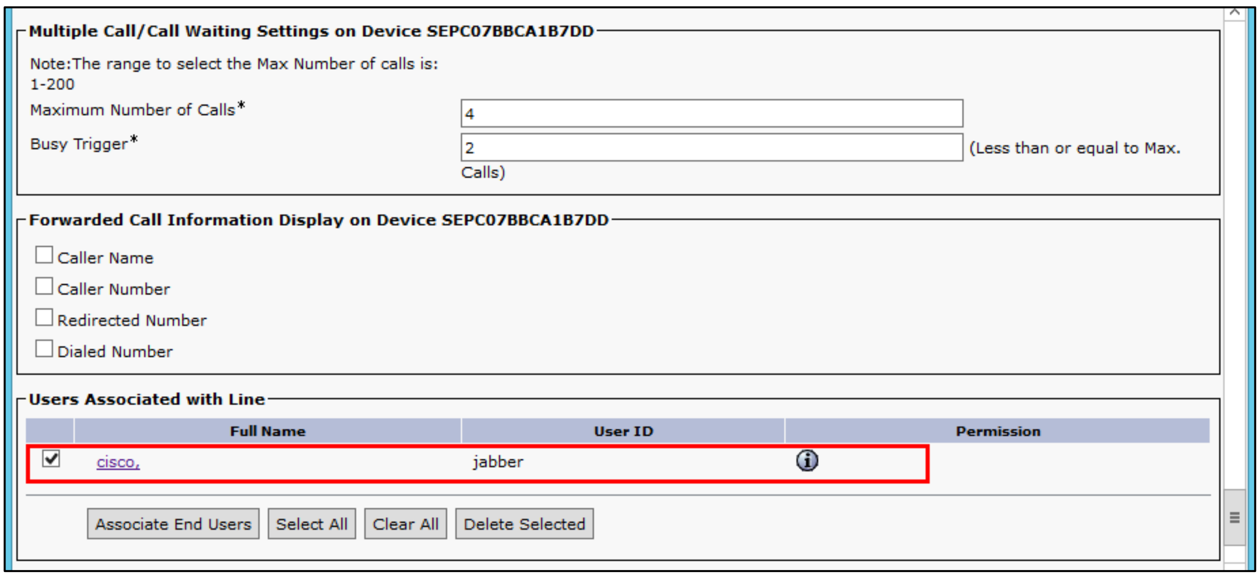


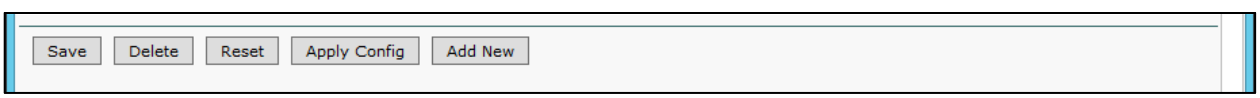




Cisco IP Phone 9971 SIP Configuration (Continued…)


Ring Setting (Phone Active)
Use System Default
Applies to this line when any line on the phone has a call in progress.
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 >
Log Missed Calls
Submit
Propagate Selected

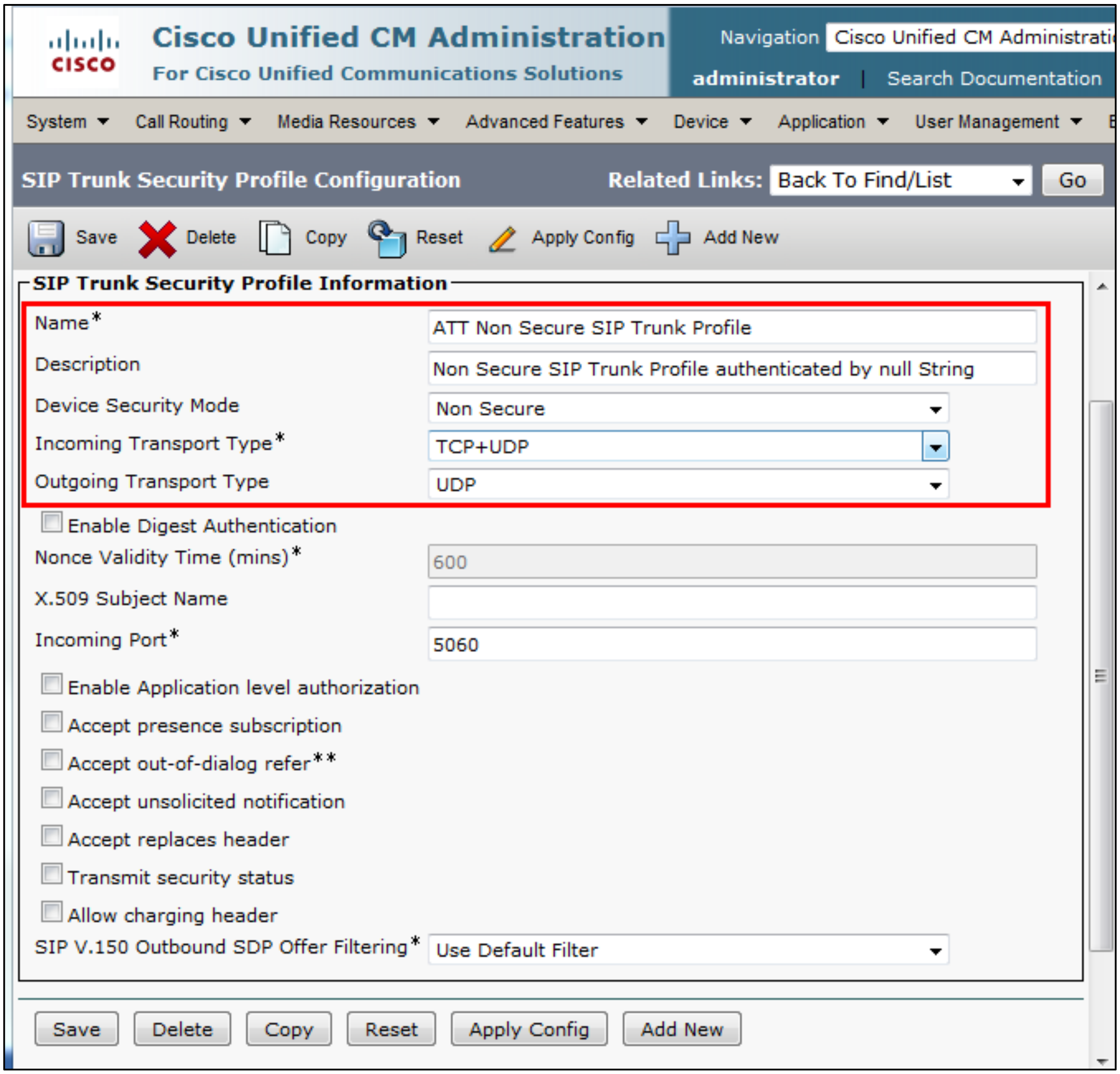

Multiple Call/Call Waiting Settings on Device SEPC07BBCA1B7DD
Note:The range to select the Max Number of calls is: 1-200
Maximum Number of Calls*
4
Busy Trigger*
2
(Less than or equal to Max. Calls)
Forwarded Call Information Display on Device SEPC07BBCA1B7DD
Caller Name
Caller Number
Redirected Number
Dialed Number
Users Associated with Line
Full Name
User ID
Permission
cisco,
jabber
Associate End Users
Select All
Clear All
Delete Selected


Save
Delete
Reset
Apply Config
Add New

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

Navigation: System → Security → SIP Trunk Security Profile

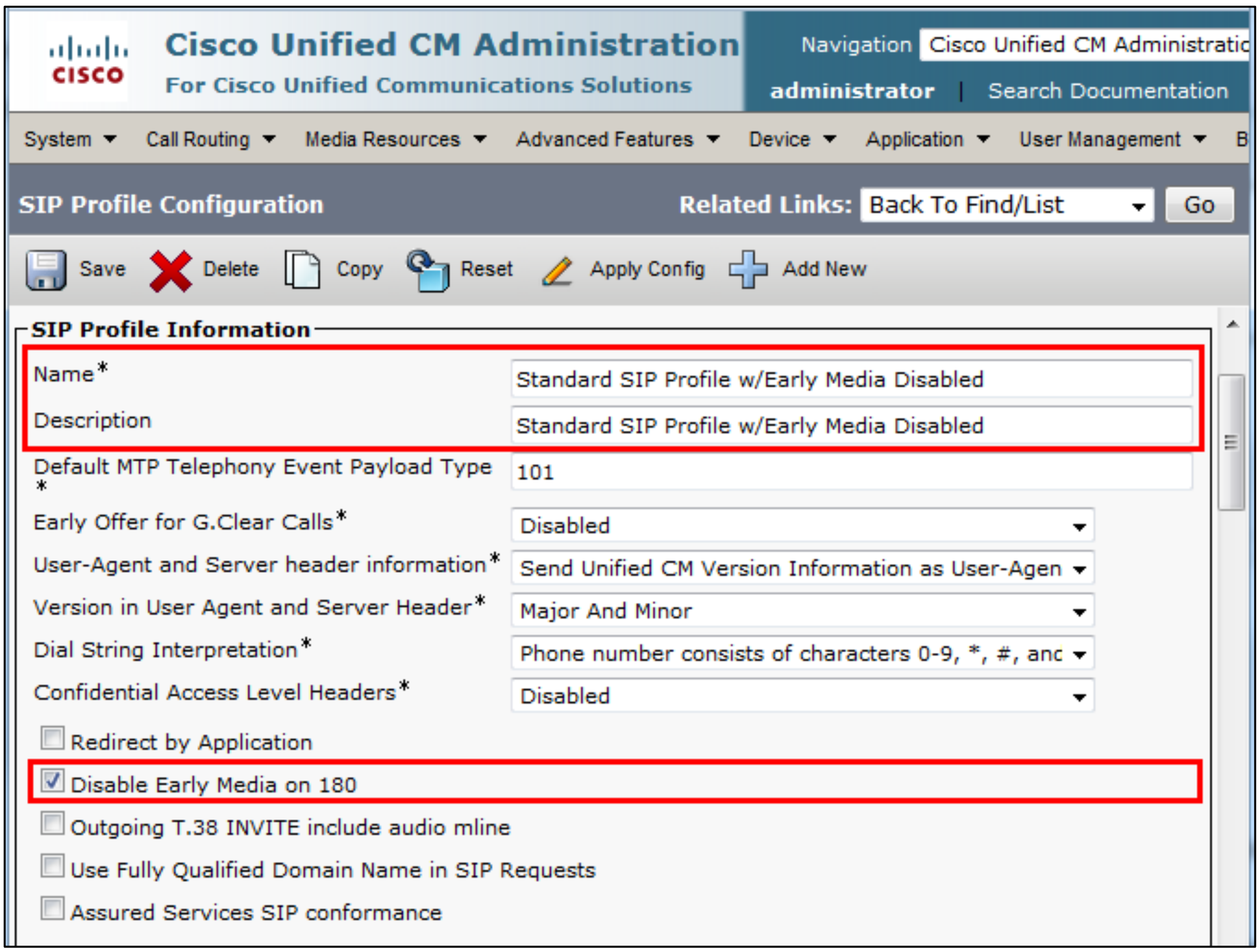
Set Name* = ATT Non Secure SIP Trunk Profile. This is used in this example.
Set Description = Non Secure SIP Trunk Profile authenticated by null String. This is used in this example.
Set Device Security Mode = Non Secure.
Set Incoming Transport Type* = TCP+UDP.
Set Outgoing Transport Type = UDP.



SIP Profile Configuration used by SIP trunk to Cisco UBE

Navigation: Device → Device Settings → SIP Profile

Set SIP profile Name * = Standard SIP Profile w/Early Media Disabled. This is used for this example
Check Disable Early Media on 180
Set SIP Rel1xx Options* = Send PRACK if 1xx contains SDP
Note*= Some PSTN network call prompters utilize early-media cut-through to offer menu options to the caller (DTMF select menu) before the call is connected. In order for Cisco UCM/Cisco UBE solution to achieve successful early-media cut-through, the Cisco UCM to Cisco UBE call leg must be enabled with SIP PRACK. To enable SIP PRACK on the Cisco UCM, the SIP Profile "SIP Rel1XX Options" setting must be set to "Send PRACK".



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

SDP Information

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer*	Default

- [ ] Require SDP Inactive Exchange for Mid-Call Media Change
- [ ] Allow RR/RS bandwidth modifier (RFC 3556)

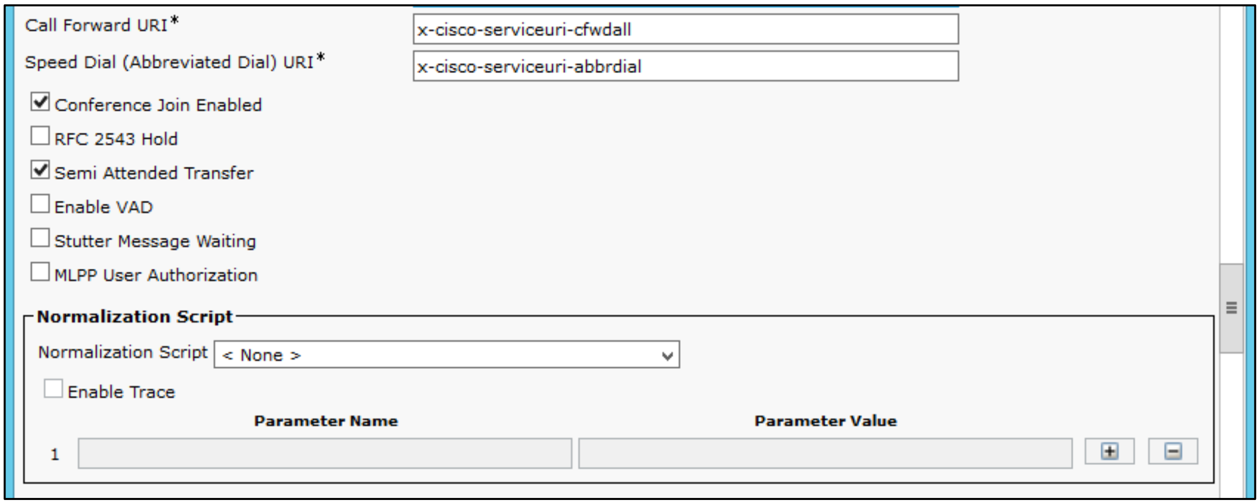
Parameters used in Phone

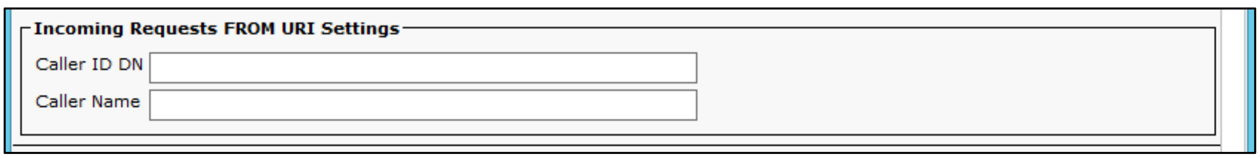
Timer Invite Expires (seconds)*	1800
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Media Port Ranges	(•) Common Port Range for Audio and Video ( ) Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766
DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default
DSCP for Audio Portion of Video Calls	Use System Default
DSCP for TelePresence Calls	Use System Default
DSCP for Audio Portion of TelePresence Calls	Use System Default

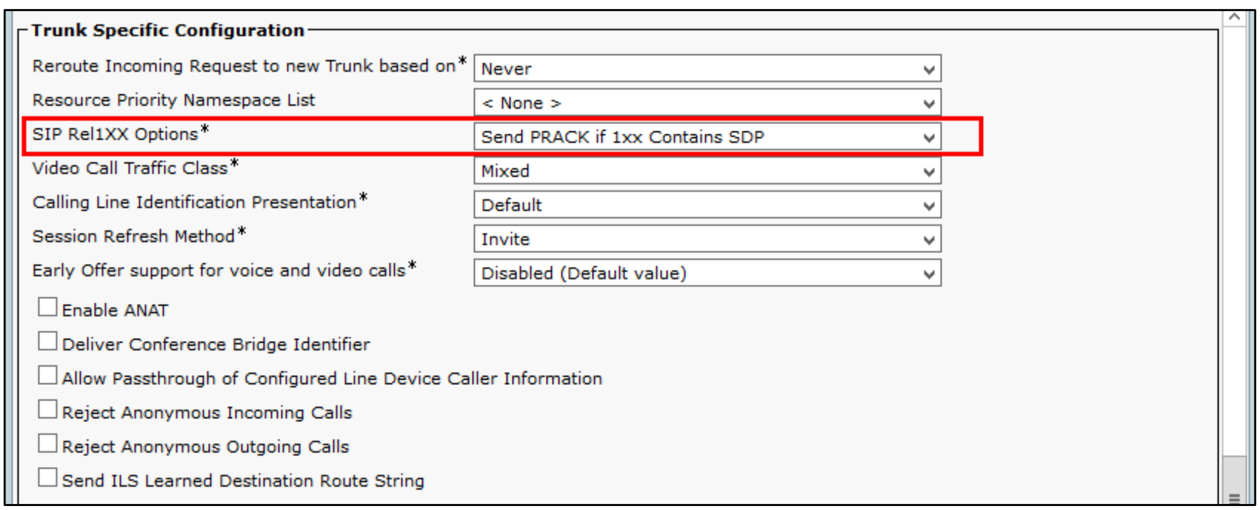
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000

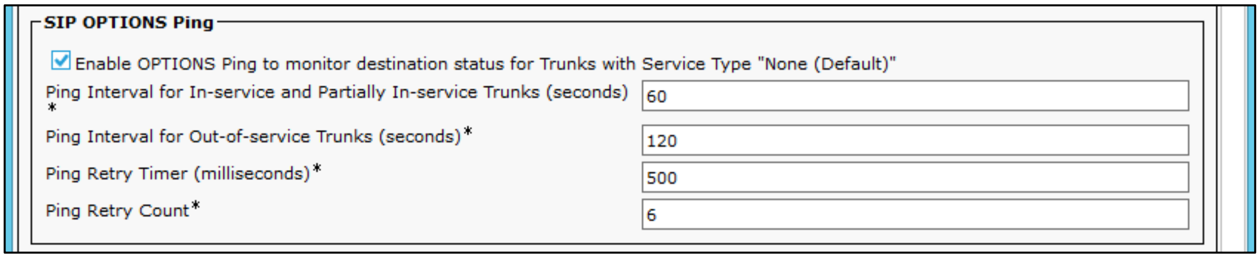
Note: Testing was conducted in tekVizion labs

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

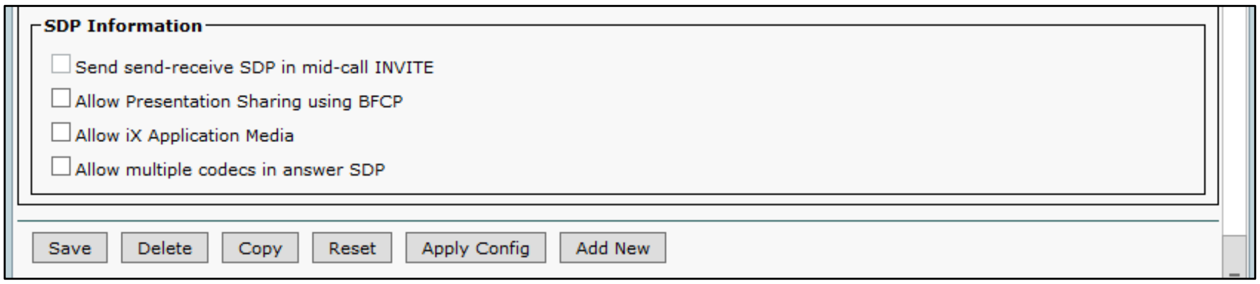








SIP Profile Configuration used by SIP trunk to Cisco UBE (Continued…)



SIP Trunk to Cisco UBE Configuration

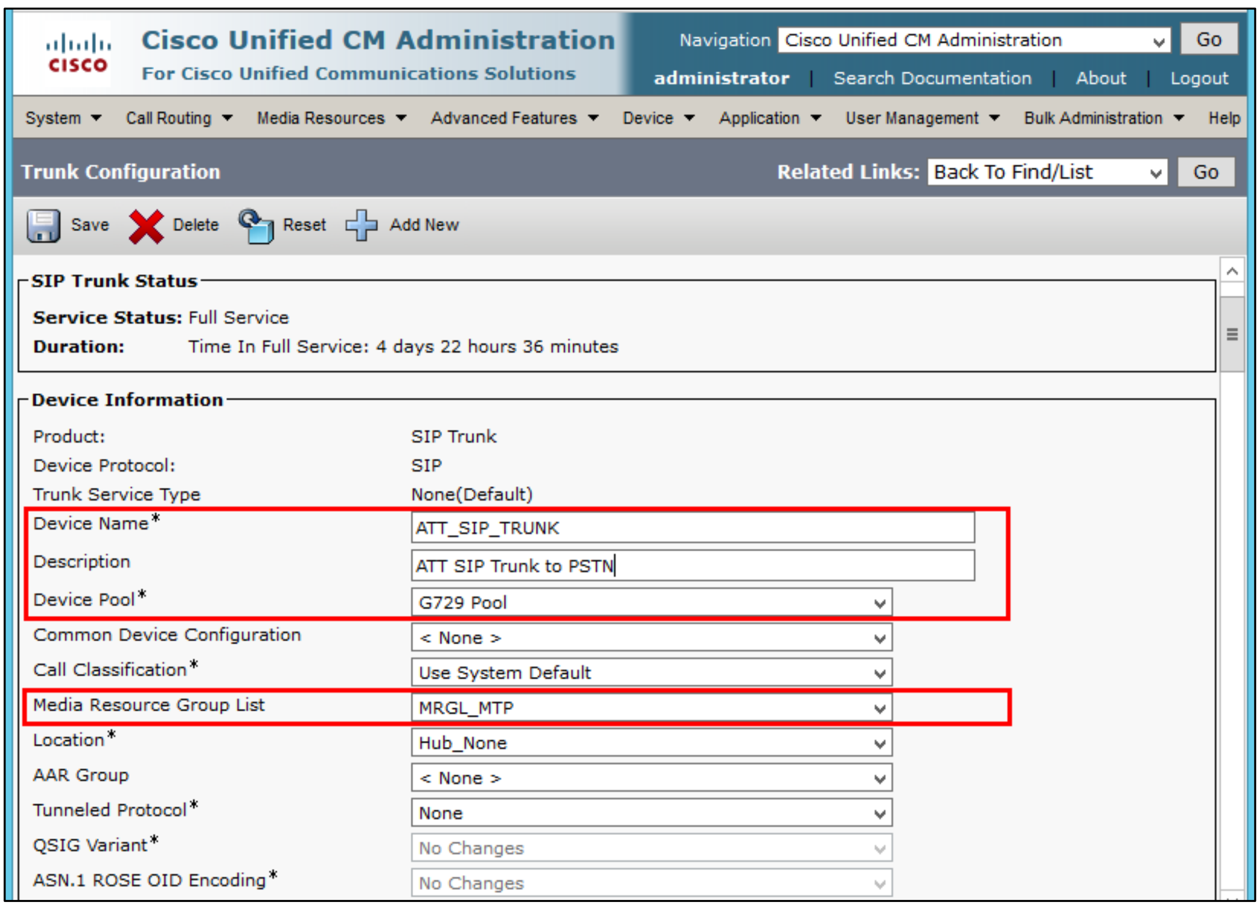
Navigation: Device → Trunk

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

Set Description = ATT SIP Trunk to PSTN. This is used for this example

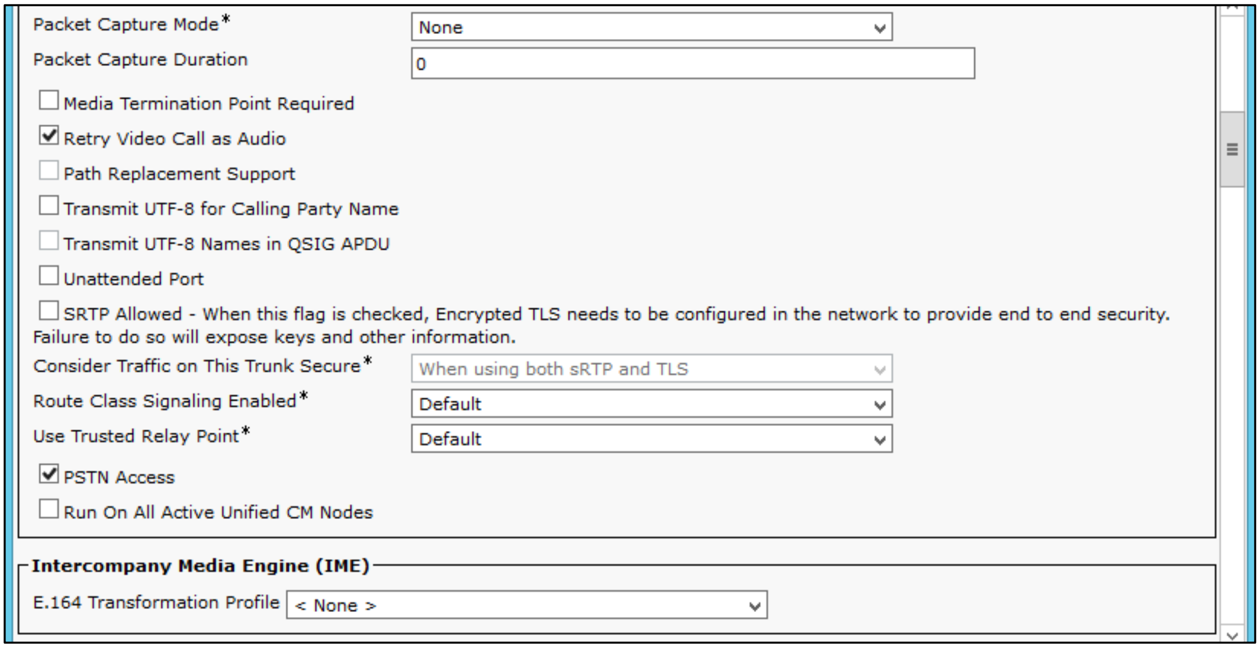
Set Device Pool* = G729_pool. This is used for this example

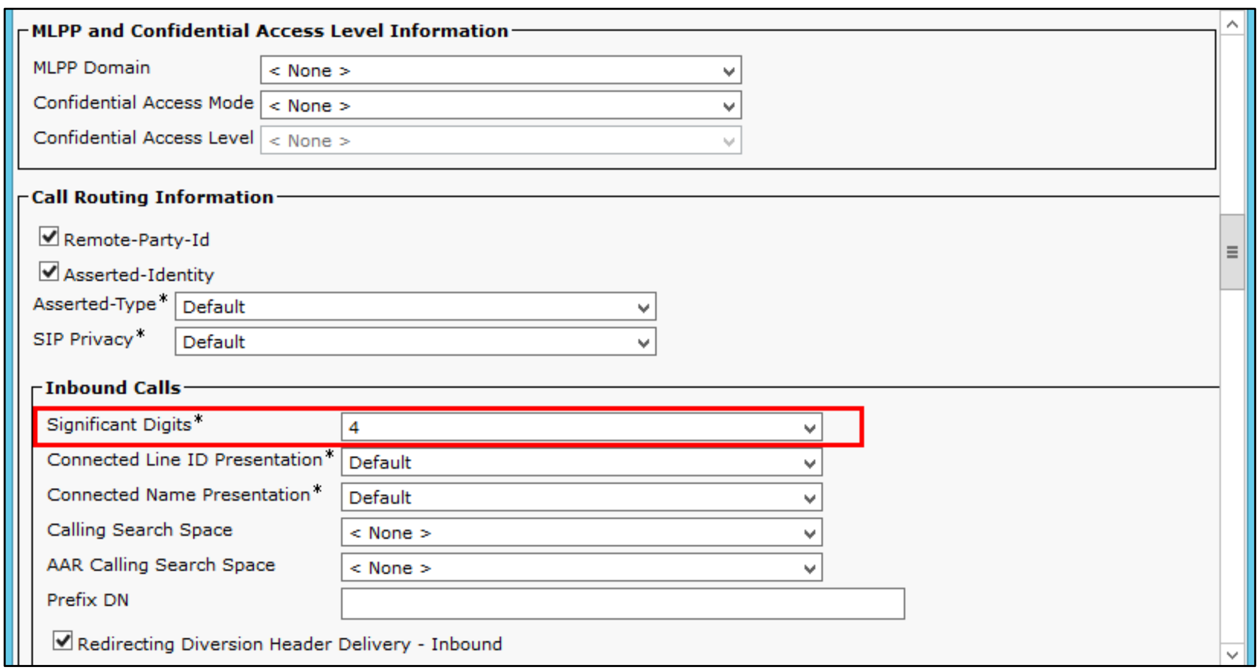
Set Media Resource Group List = MRGL_MTP.



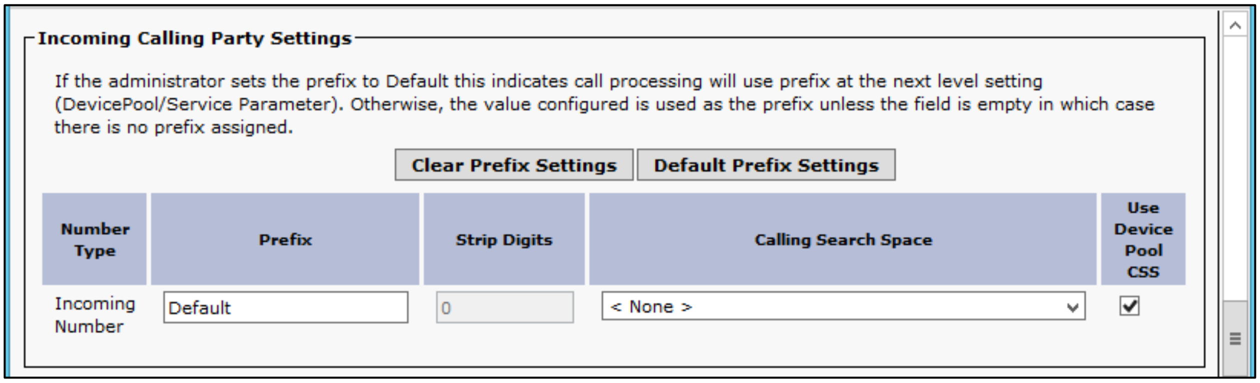
SIP Trunk to Cisco UBE Configuration (Continued…)

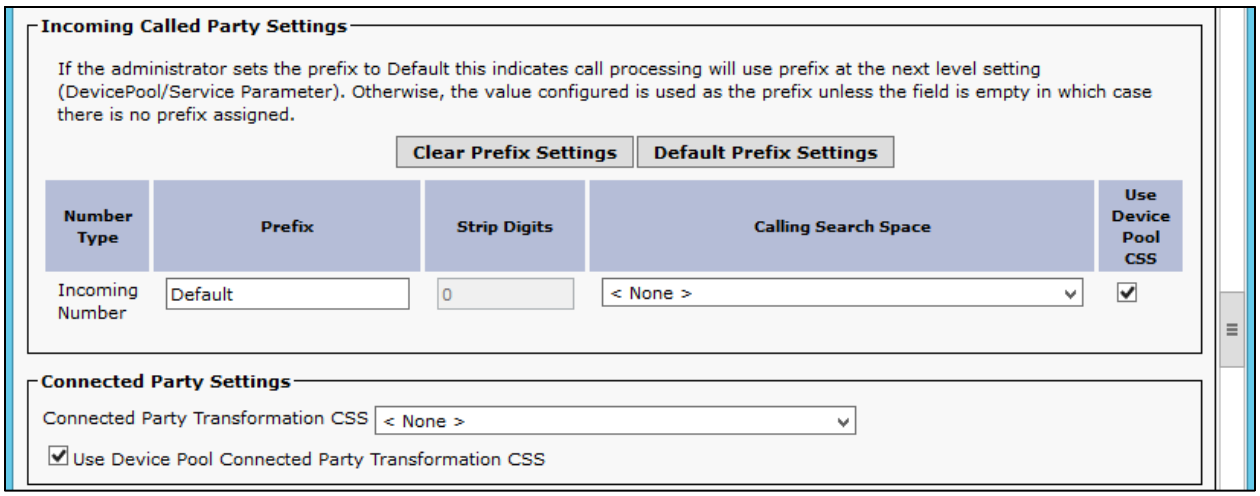
Set Significant Digits* = 4. This is used in this example.

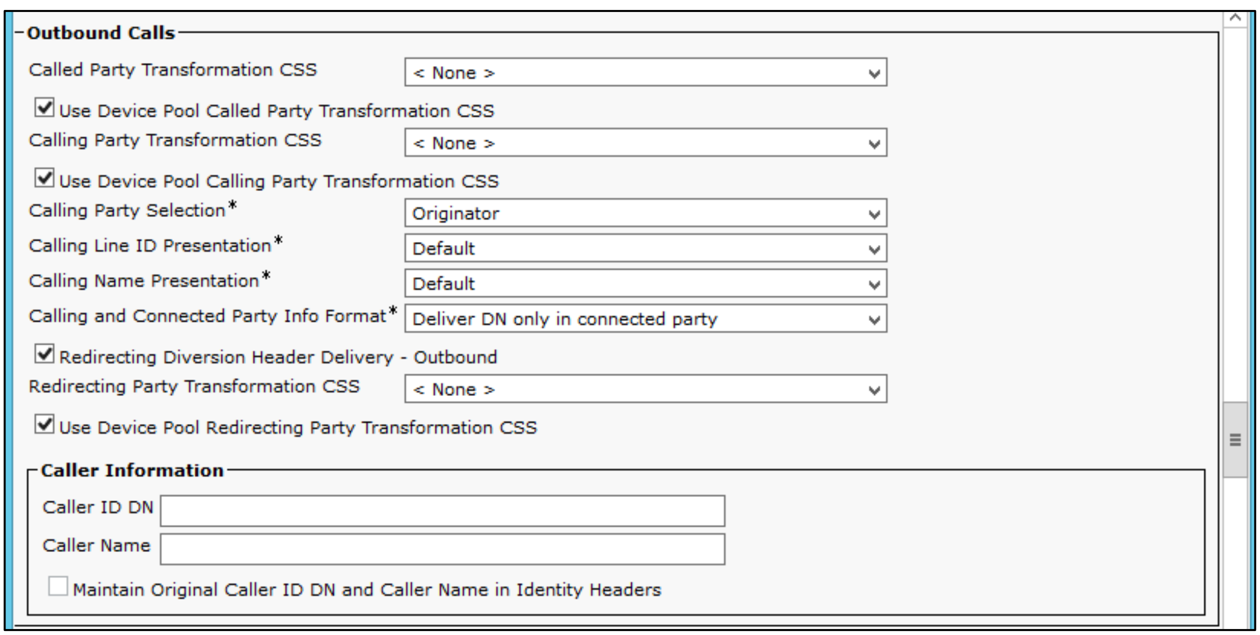




SIP Trunk to Cisco UBE Configuration (Continued…)

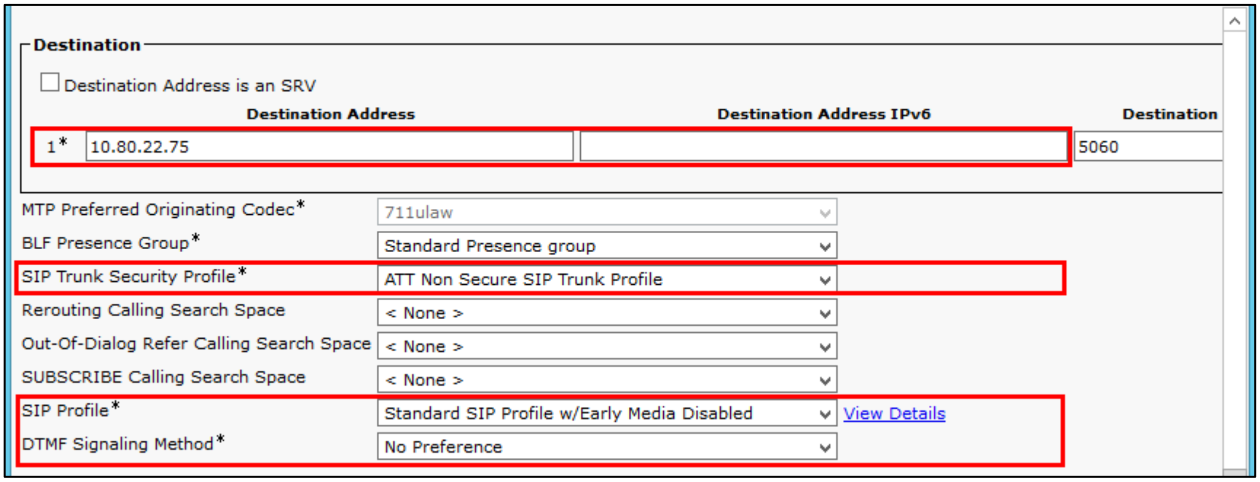

Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
Use Device Pool CSS
Incoming Number
Default
0
< None >

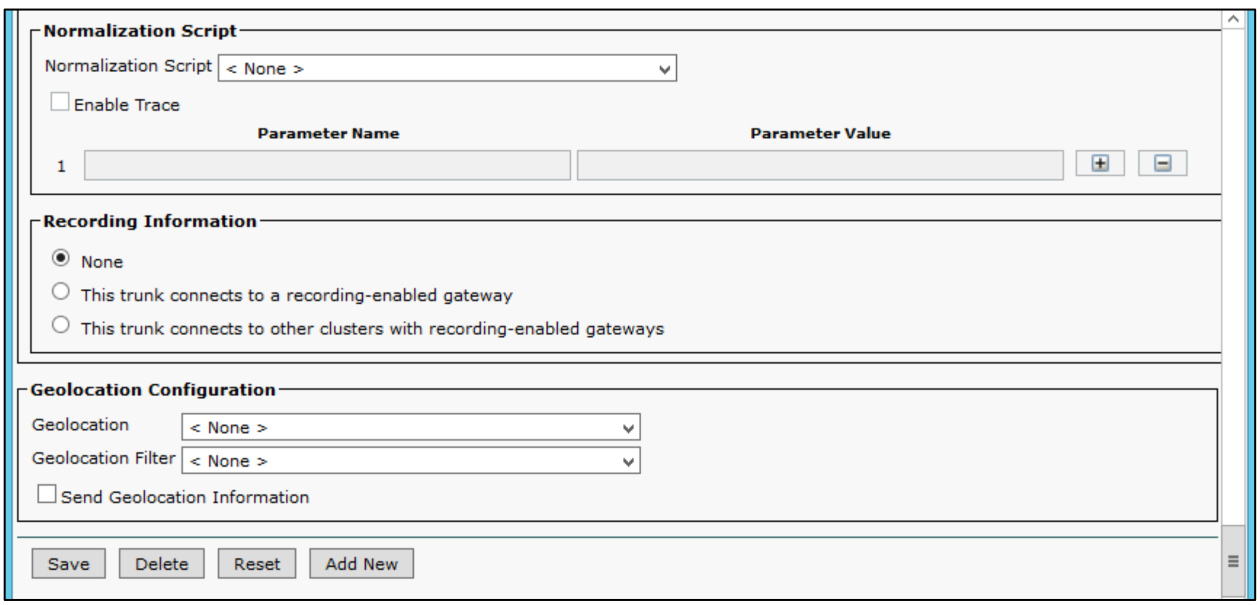

Incoming Called Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
Use Device Pool CSS
Incoming Number
Default
0
< None >
Connected Party Settings
Connected Party Transformation CSS
< None >
Use Device Pool Connected Party Transformation CSS


Outbound Calls
Called Party Transformation CSS
< None >
Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS
< None >
Use Device Pool Calling Party Transformation CSS
Calling Party Selection*
Originator
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Calling and Connected Party Info Format*
Deliver DN only in connected party
Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS
< None >
Use Device Pool Redirecting Party Transformation CSS
Caller Information
Caller ID DN
Caller Name
Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Trunk to Cisco UBE Configuration (Continued…)

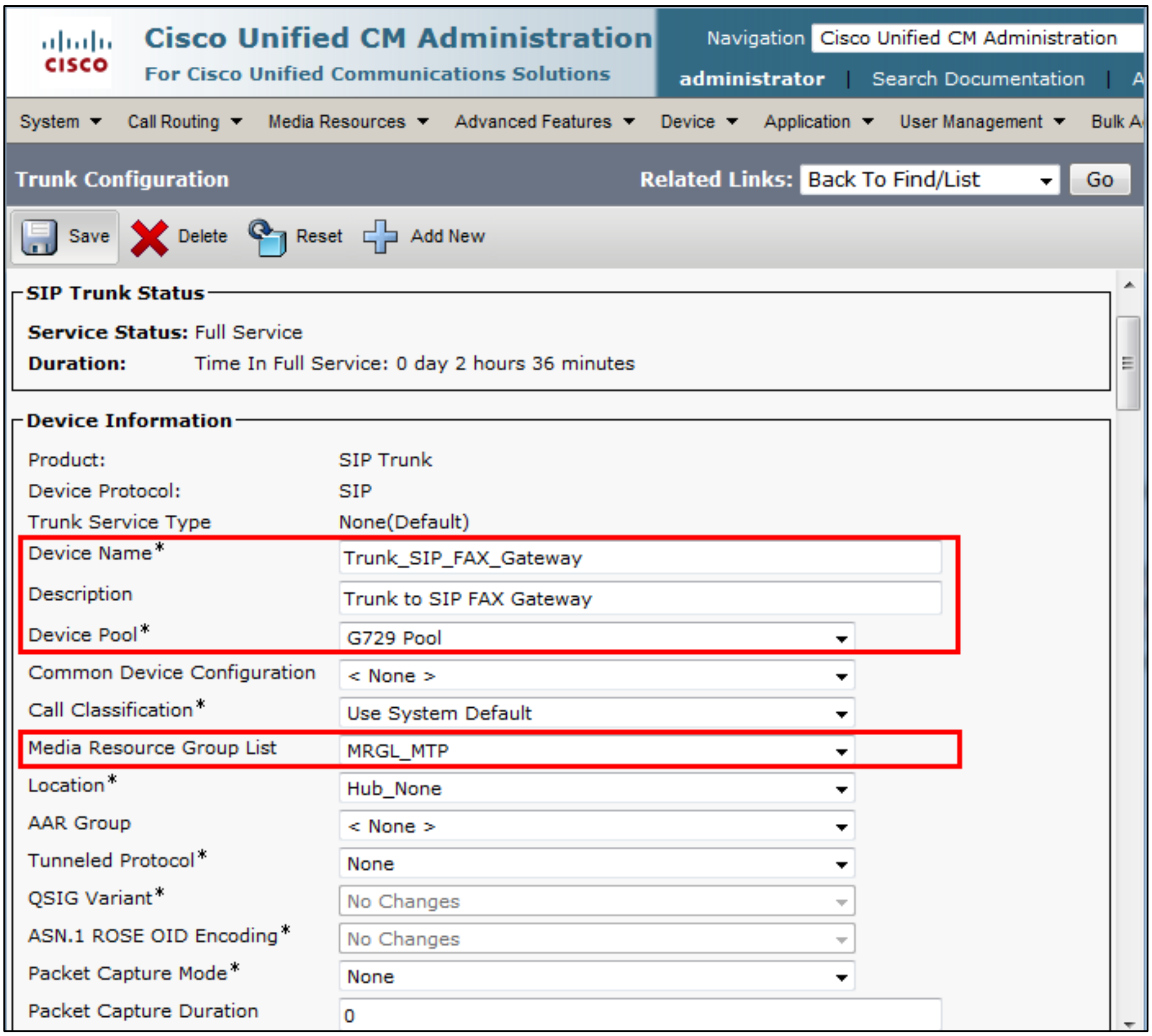
Set Destination Address = Set IP address of ISR-Cisco UBE.
Set SIP Trunk Security Profile* = ATT_Non Secure Sip Trunk Profile.
Set SIP Profile* = ATT_SIP_Profile. This is used in this example.



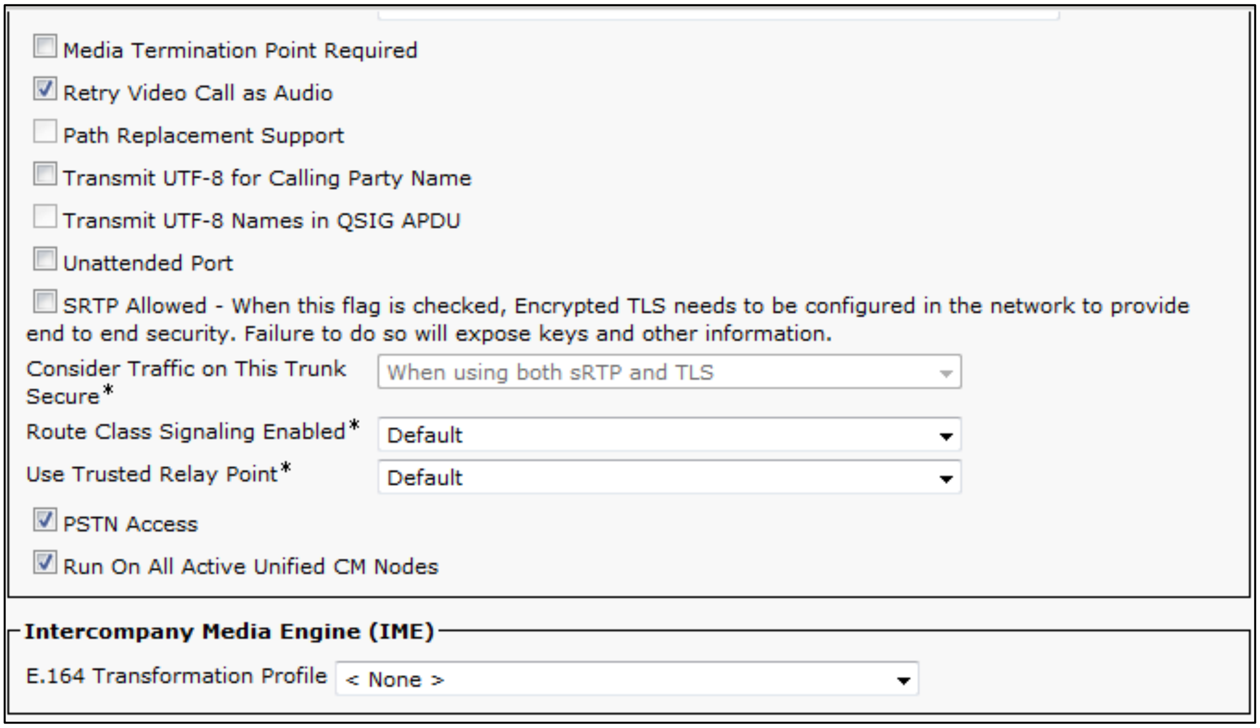


SIP Trunk to Fax Gateway Configuration.

Navigation: Device → Trunk
Set Device Name* = Trunk_SIP_FAX_Gateway. This is used for this example
Set Description = Trunk_SIP_FAX_Gateway. This is used for this example
Set Device Pool* = G729 pool. This is used for this example
Set Media Resource Group List = MRGL_MTP.



SIP Trunk to Fax Gateway Configuration (Continued…)

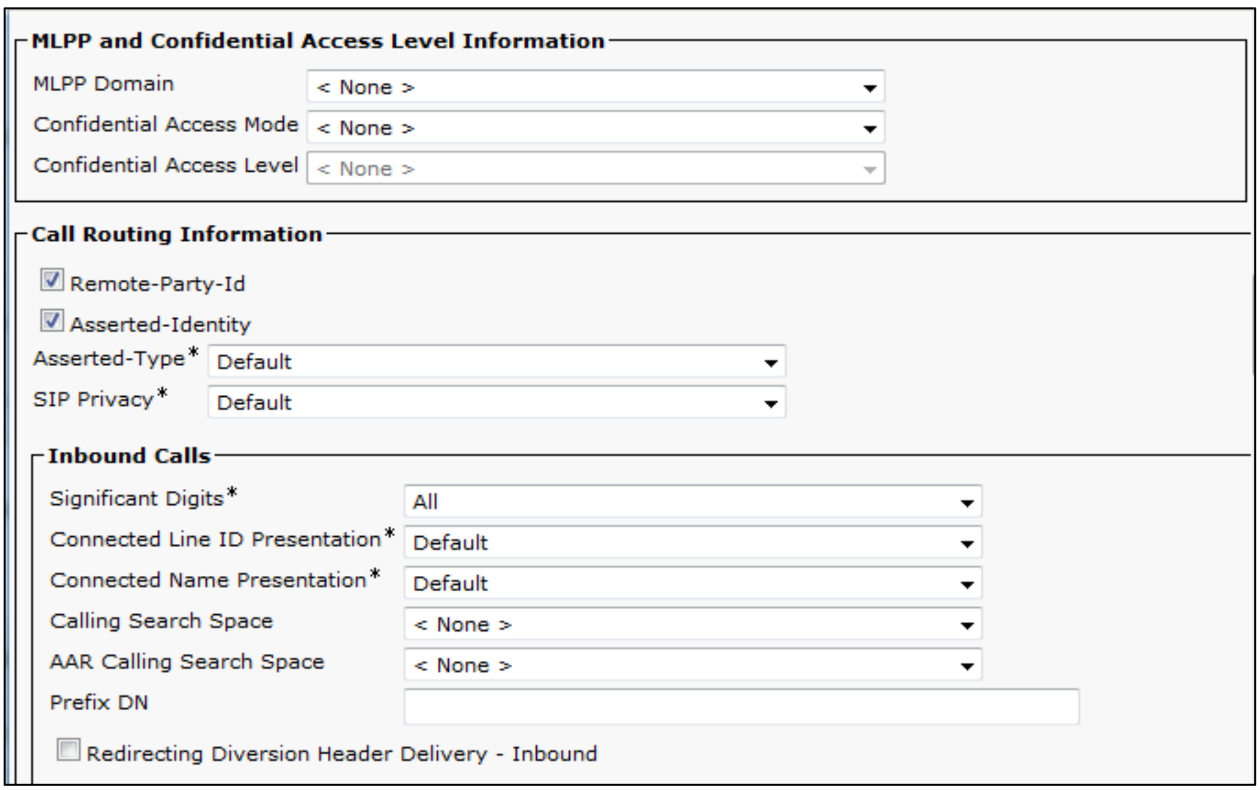

☐ Media Termination Point Required
☑ Retry Video Call as Audio
☐ Path Replacement Support
☐ Transmit UTF-8 for Calling Party Name
☐ Transmit UTF-8 Names in QSIG APDU
☐ Unattended Port
☐ SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end 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

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 (Continued…)

Incoming Calling Party Settings

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

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	☑

Incoming Called Party Settings

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

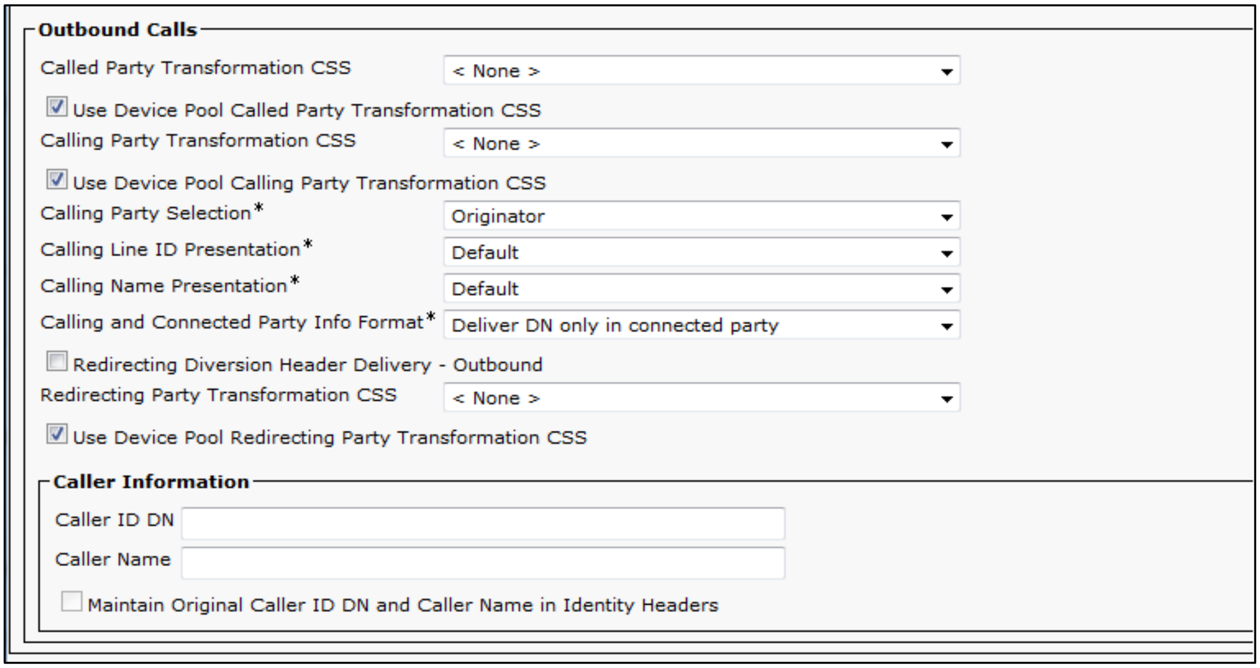
Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	☐

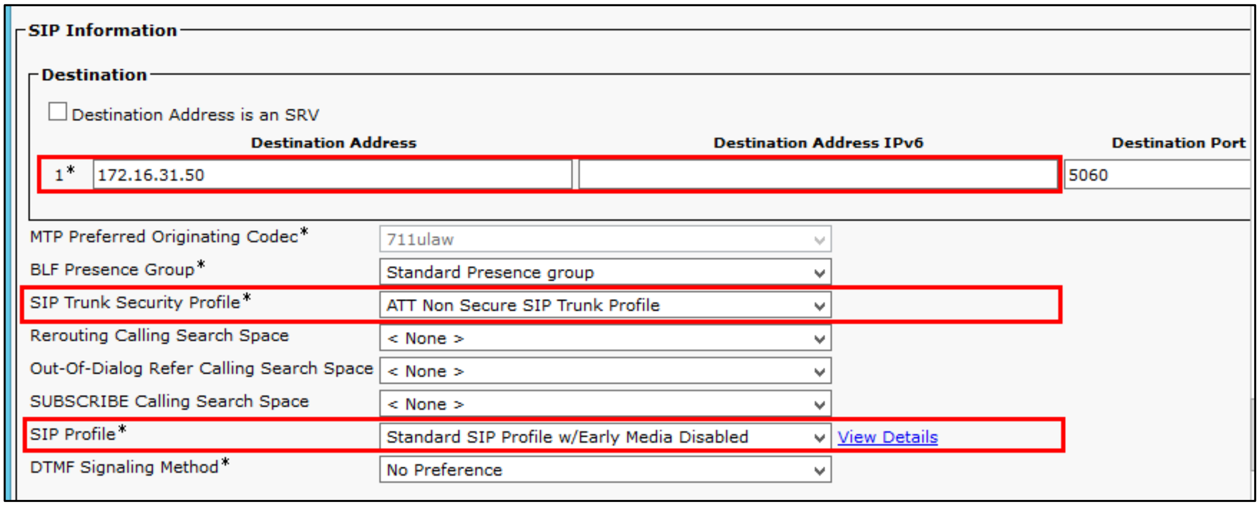
Connected Party Settings

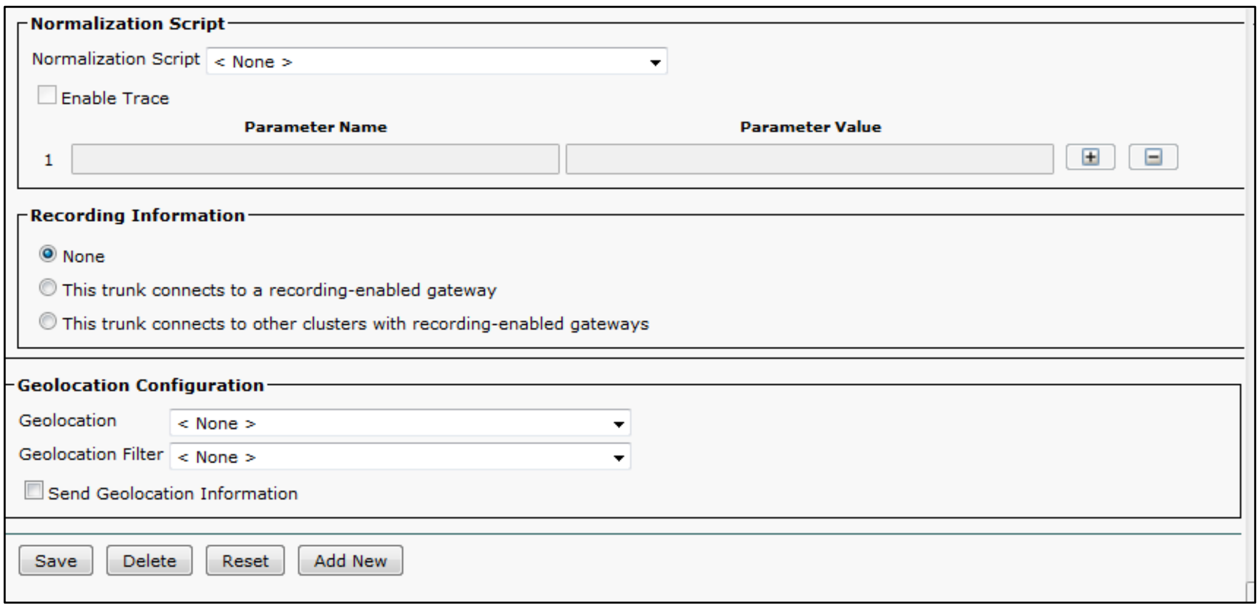
Connected Party Transformation CSS: < None >

☑ Use Device Pool Connected Party Transformation CSS



SIP Trunk to Fax Gateway Configuration (Continued…)

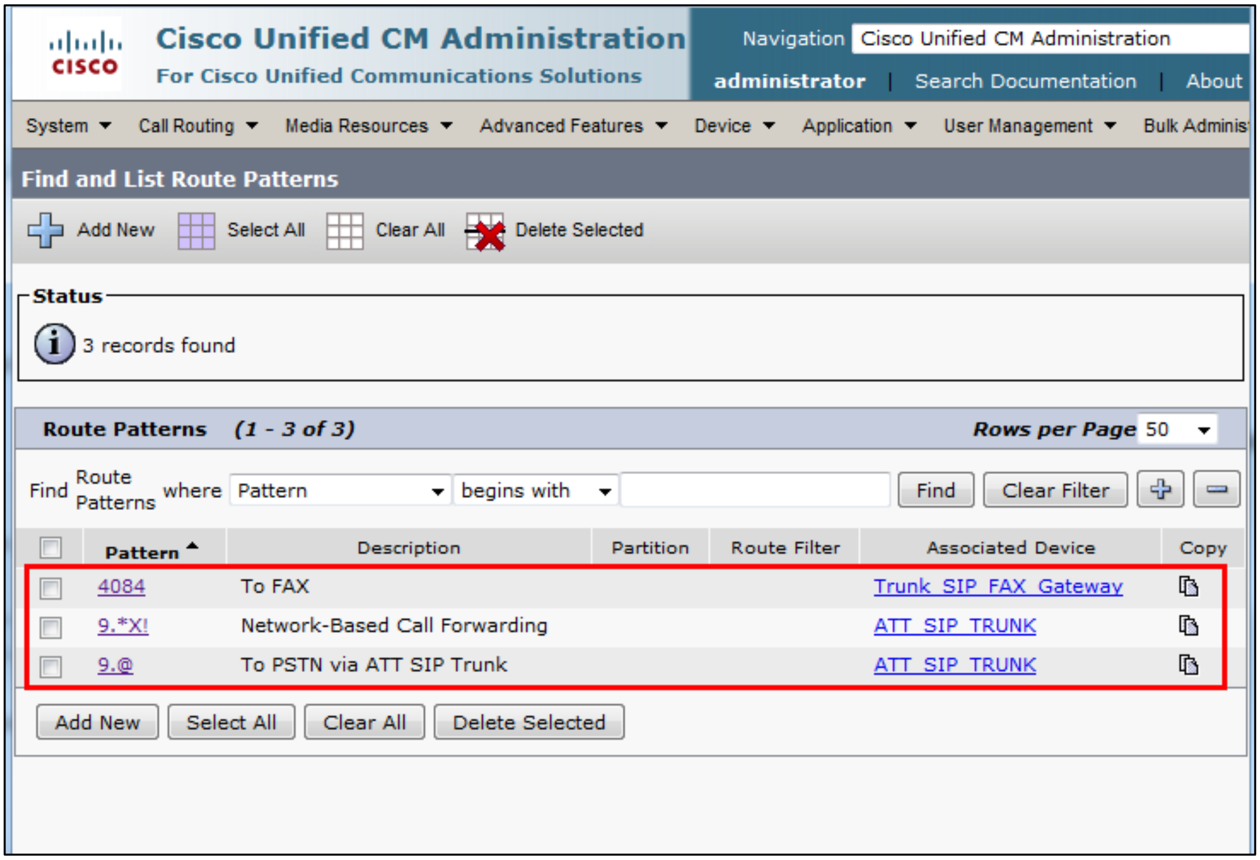

SIP Information
Destination
Destination Address is an SRV
Destination Address
Destination Address IPv6
Destination Port
1*
172.16.31.50
5060
MTP Preferred Originating Codec*
711ulaw
BLF Presence Group*
Standard Presence group
SIP Trunk Security Profile*
ATT Non Secure SIP Trunk Profile
Rerouting Calling Search Space
< None >
Out-Of-Dialog Refer Calling Search Space
< None >
SUBSCRIBE Calling Search Space
< None >
SIP Profile*
Standard SIP Profile w/Early Media Disabled
View Details
DTMF Signaling Method*
No Preference


Normalization Script
Normalization Script
< None >
Enable Trace
Parameter Name
Parameter Value
1
Recording Information
None
This trunk connects to a recording-enabled gateway
This trunk connects to other clusters with recording-enabled gateways
Geolocation Configuration
Geolocation
< None >
Geolocation Filter
< None >
Send Geolocation Information
Save
Delete
Reset
Add New

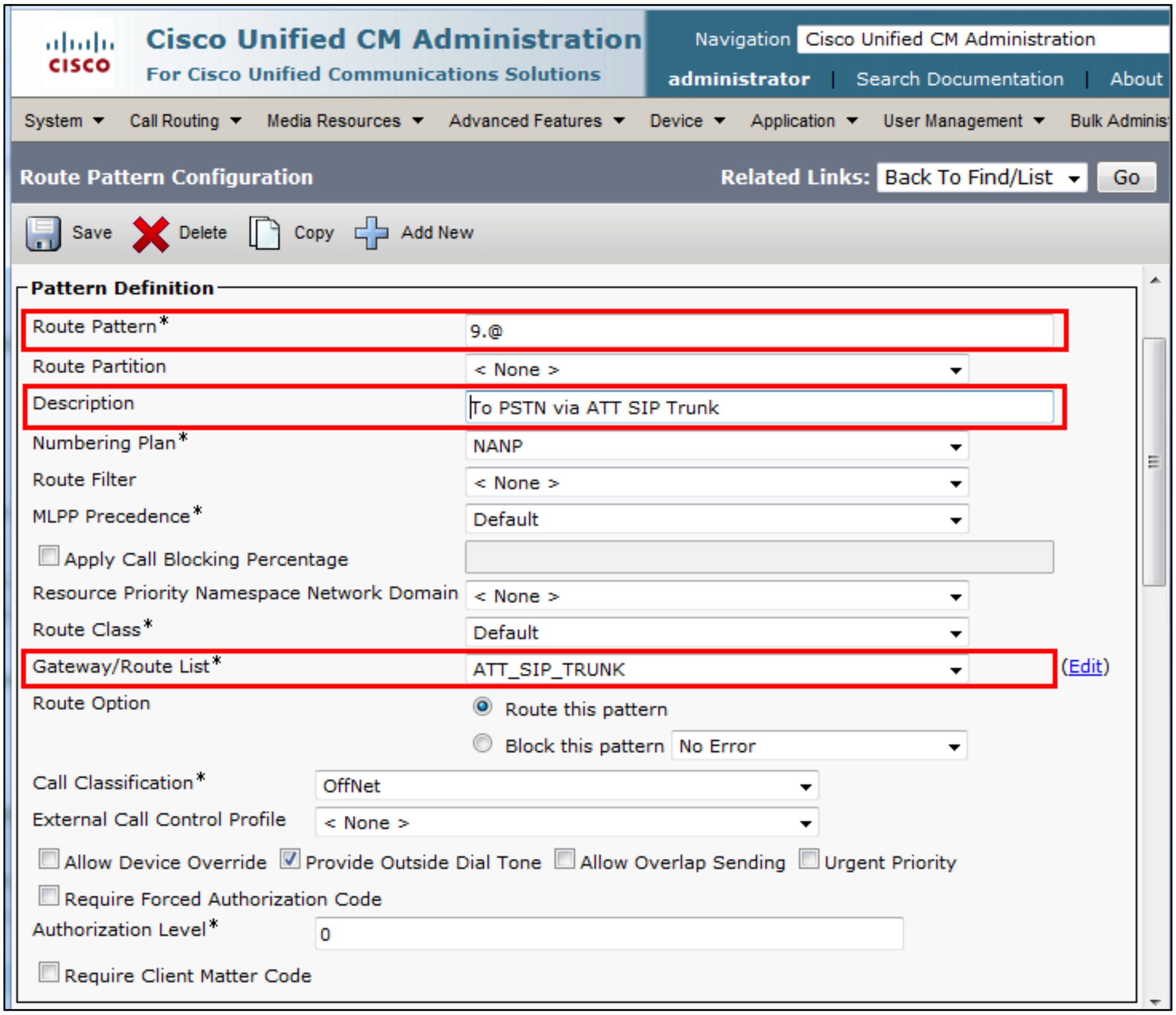
Route Pattern Configuration

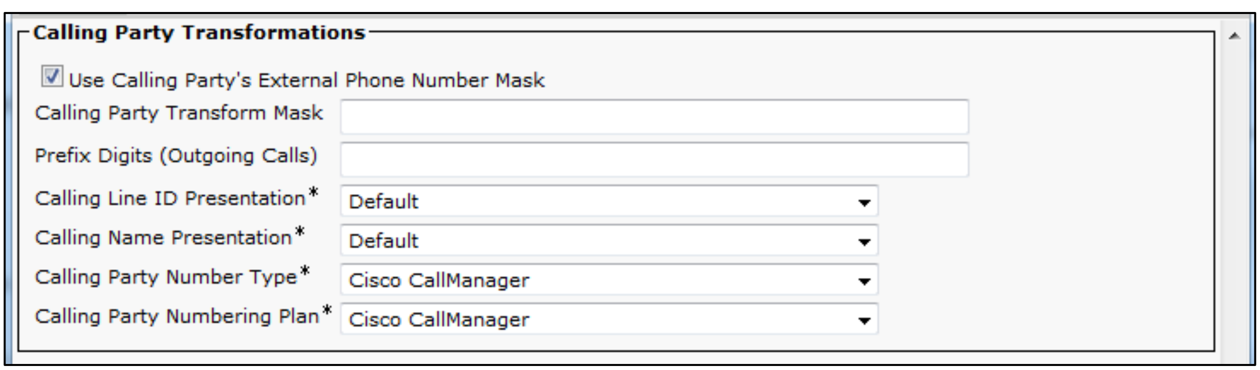
Navigation: Call Routing → Route/Hunt → Route Pattern

Set Route Pattern* = 9. @   This is used to route to AT&T via ISR Cisco UBE.
Set Description = To PSTN via ATT SIP Trunk. This text is used to identify this Route Pattern.
Set Gateway/Route List* = ATT_SIP_TRUNK. This is used for this example.
All other values are default

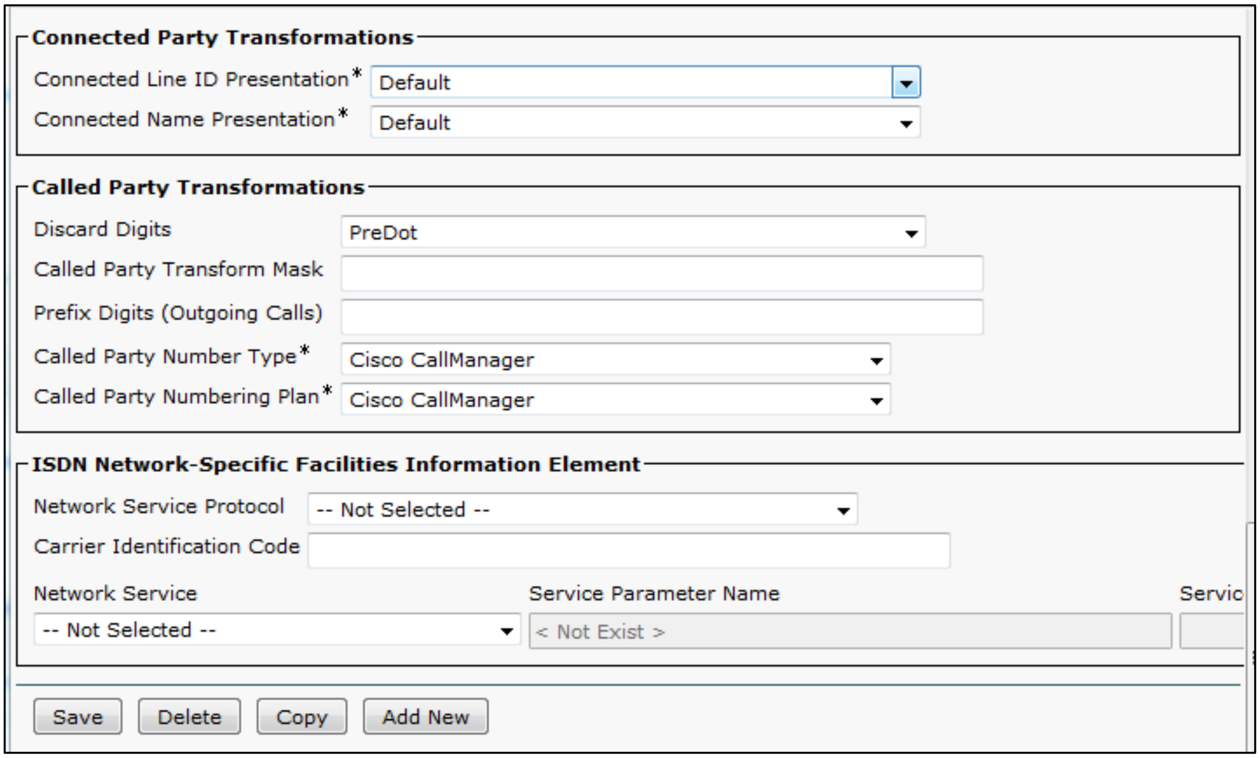


Route Pattern Configuration (Continued…)





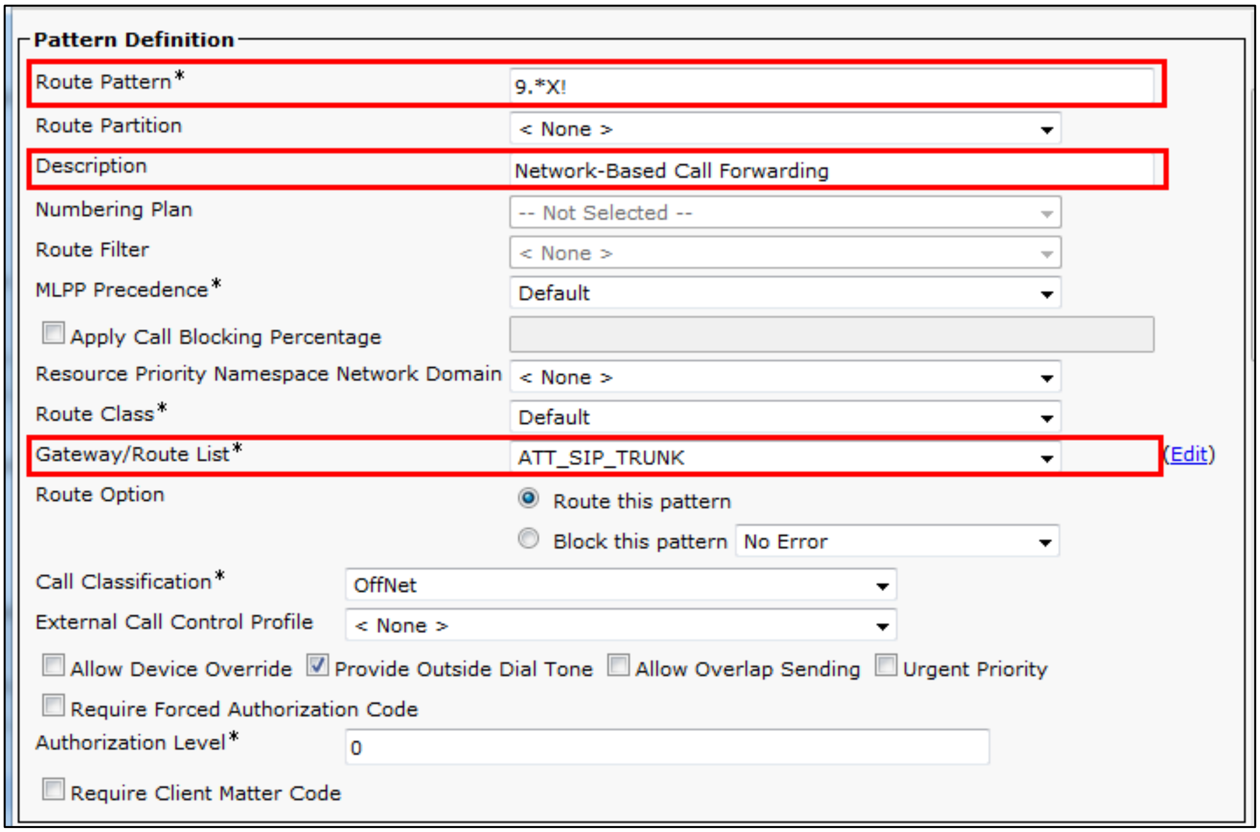
Route Pattern Configuration (Continued…)

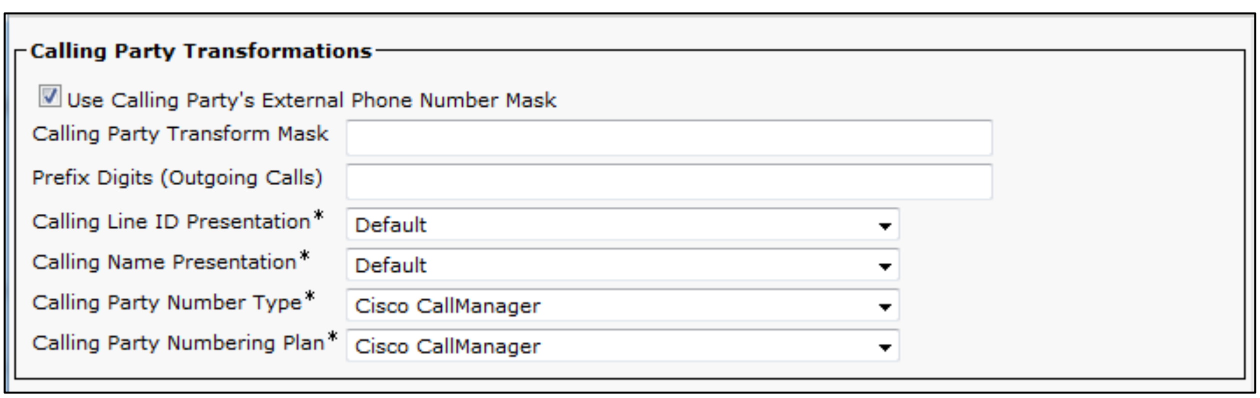


Route Pattern Configuration (Continued…)

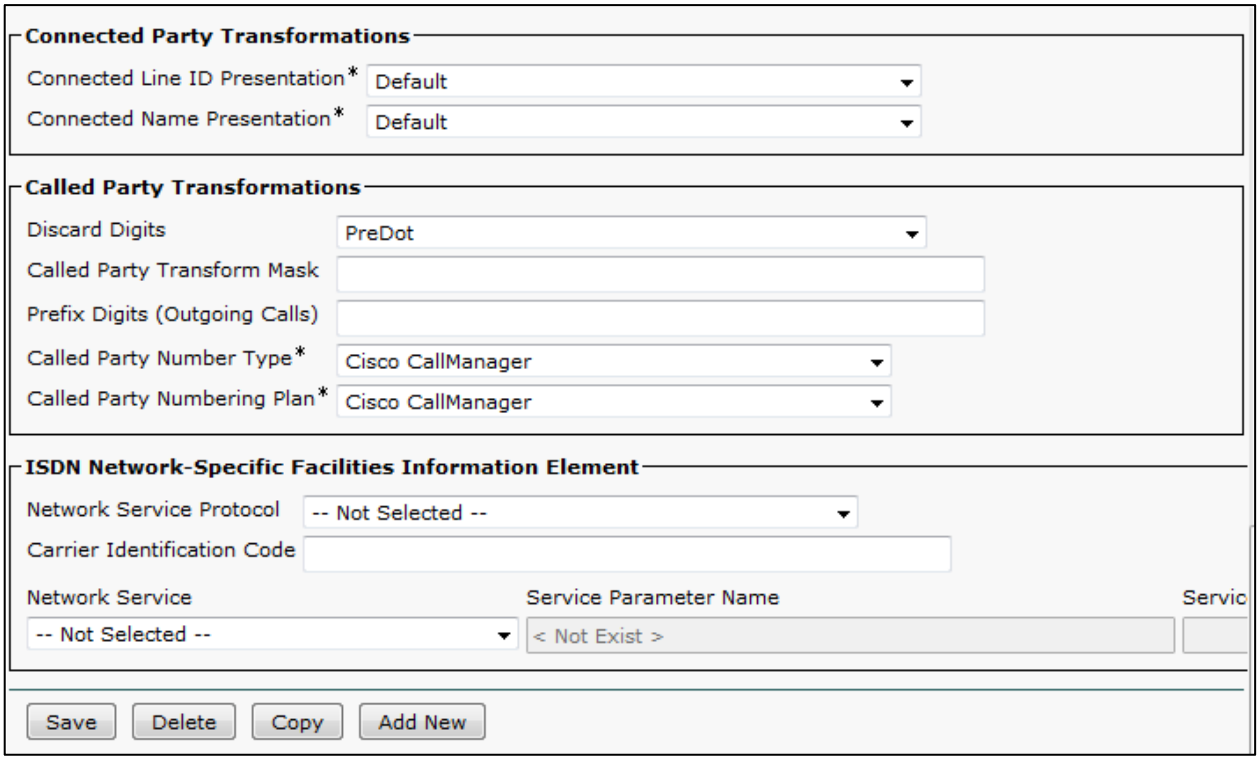
Set Route Pattern* = 9.*X!   This is used to route to AT&T via ISR Cisco UBE.
Set Description = Network-Based Call Forwarding. This text is used to identify this Route Pattern.
Set Gateway/Route List* = ATT_SIP_TRUNK. This is used for this example.
All other values are default

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



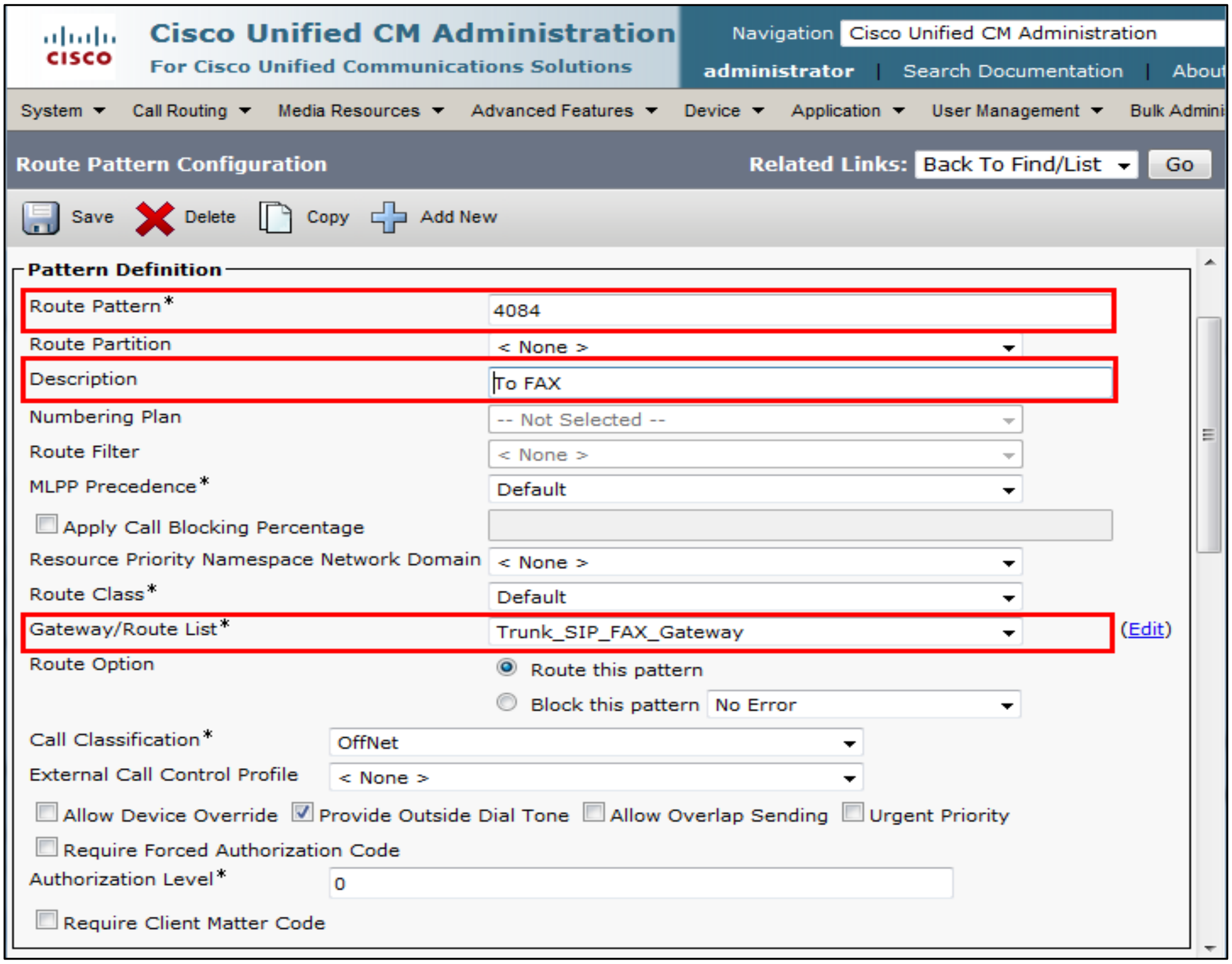


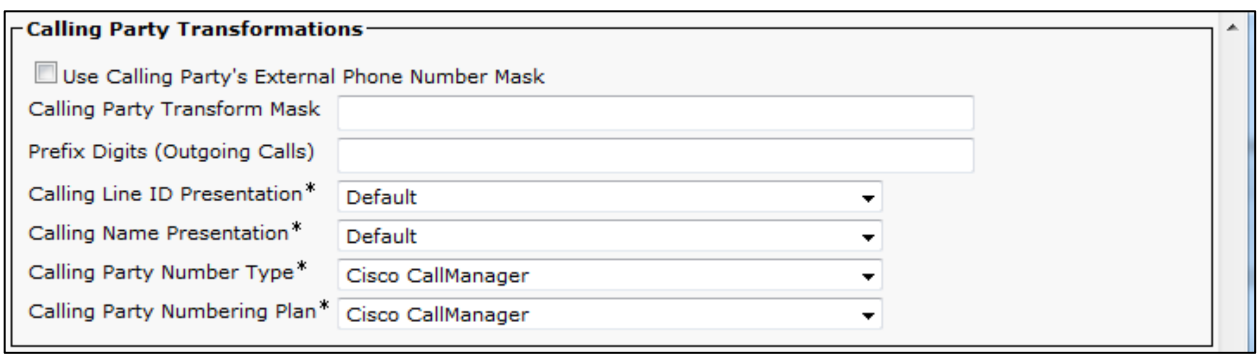
Route Pattern Configuration (Continued…)



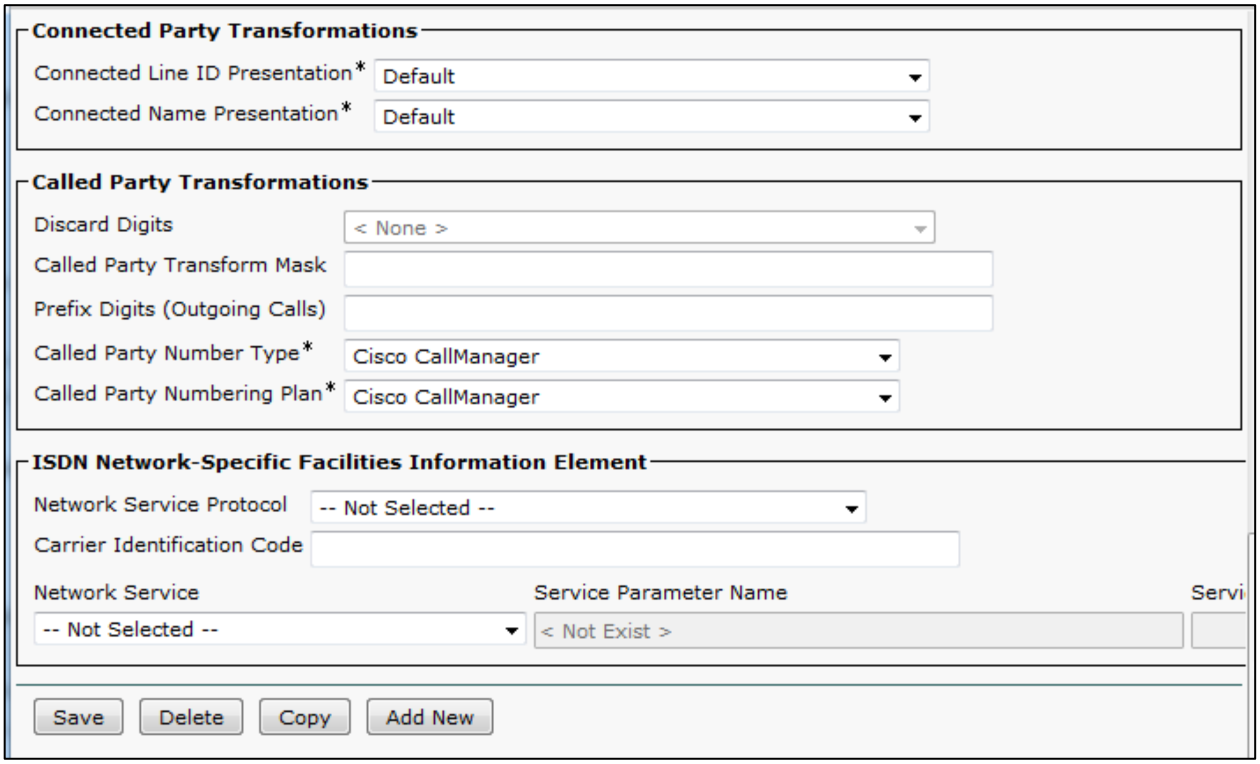
Route Pattern Configuration (Continued…)

Set Route Pattern* = 4084 this is used to route to Fax Client via Fax Gateway.
Set Description = To FAX. This text is used to identify this Route Pattern.
Set Gateway/Route List* = Trunk_SIP_FAX_Gateway. This is used for this example.
All other values are default





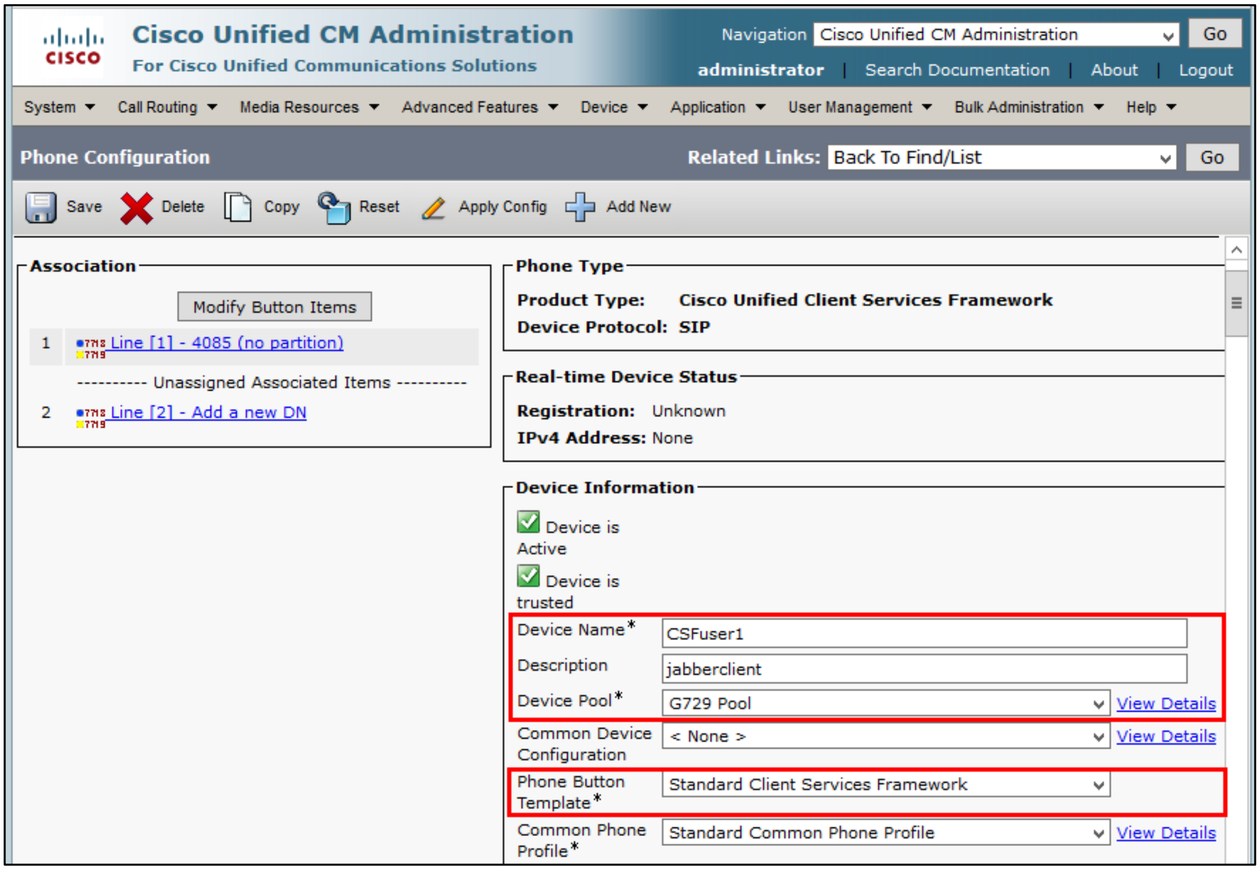
Route Pattern Configuration (Continued…)



Jabber Client Configuration

Navigation: Device → Phone

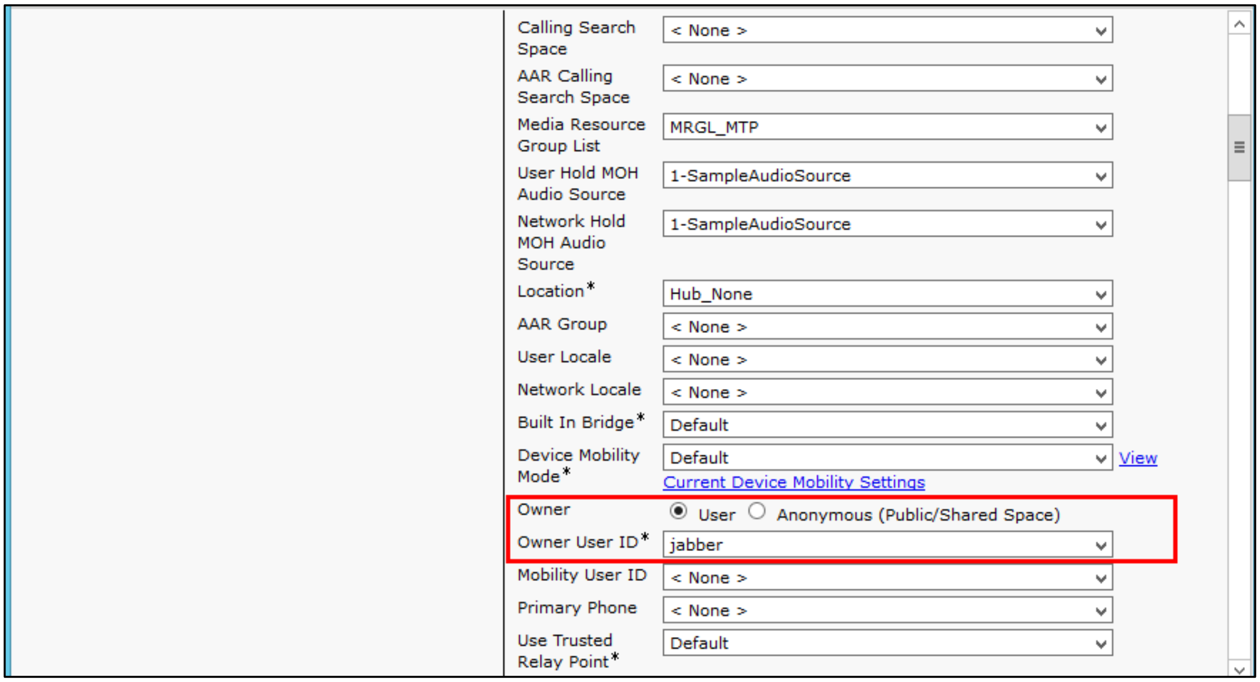
Select Phone Type* = Cisco Unified Client services framework
Set Device Name* = CSFUser1. This is used in this example.
Set Description = jabberclient. This is used in this example.
Select Device Pool = G729 Pool. This is used in this example.
Select Phone Button Template* = Standard Client Services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

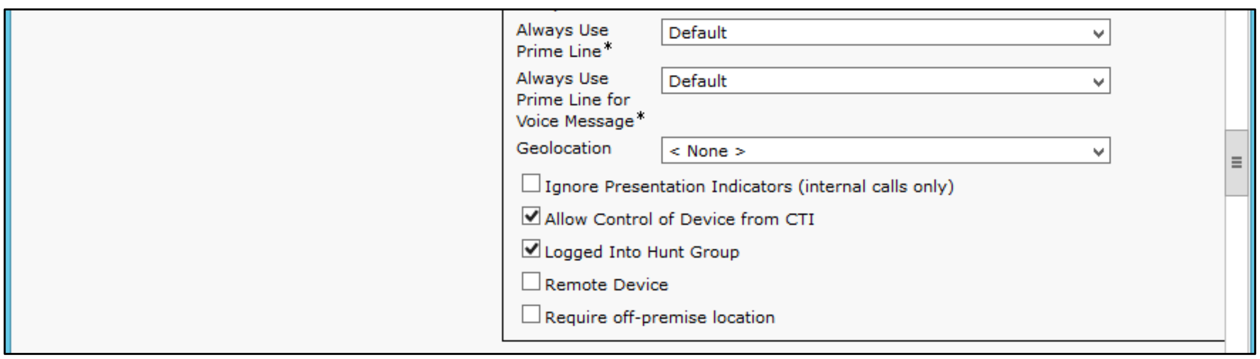


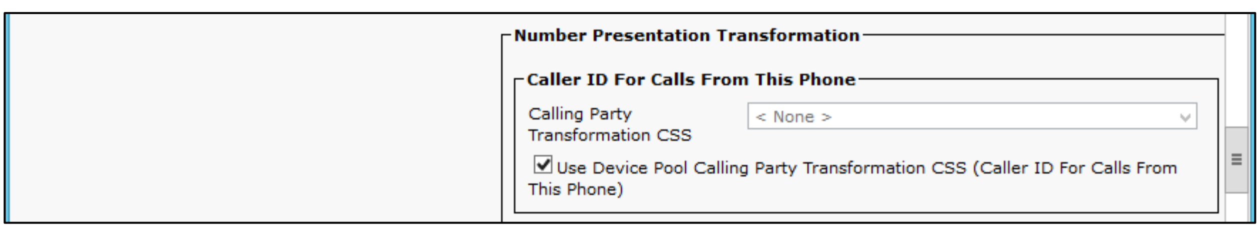
Jabber Client Configuration (Contd…)

Media Resource Group List = MRGL_MTP

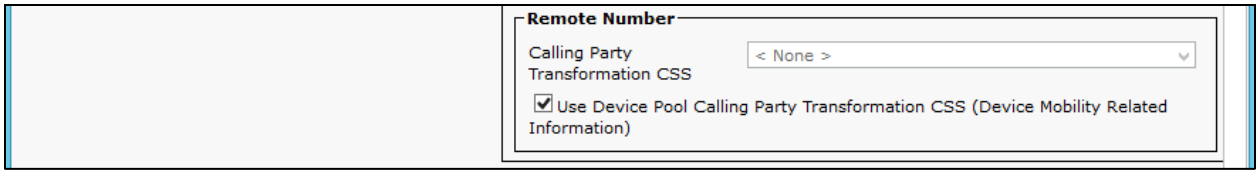
Set Owner check box
Set Owner user ID* = jabber. This is used for this example







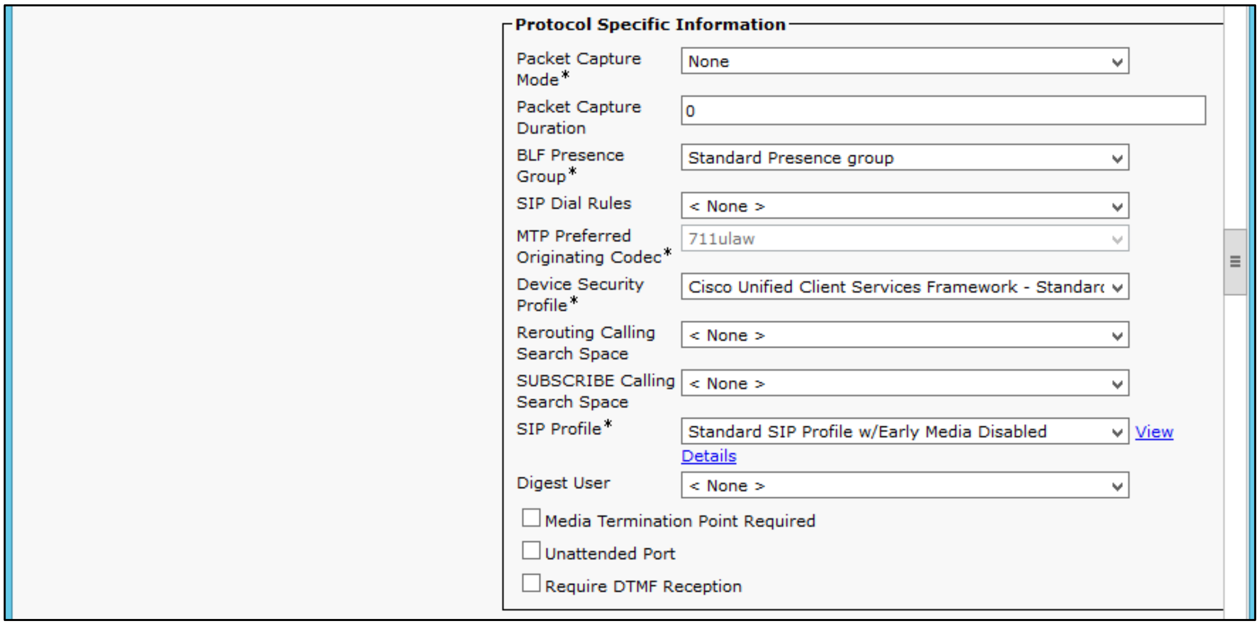
Jabber Client Configuration (Contd…)



Remote Number

Calling Party Transformation CSS: < None >

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



Protocol Specific Information

Packet Capture Mode*: None

Packet Capture Duration: 0

BLF Presence Group*: Standard Presence group

SIP Dial Rules: < None >

MTP Preferred Originating Codec*: 711ulaw

Device Security Profile*: Cisco Unified Client Services Framework - Standarc

Rerouting Calling Search Space: < None >

SUBSCRIBE Calling Search Space: < None >

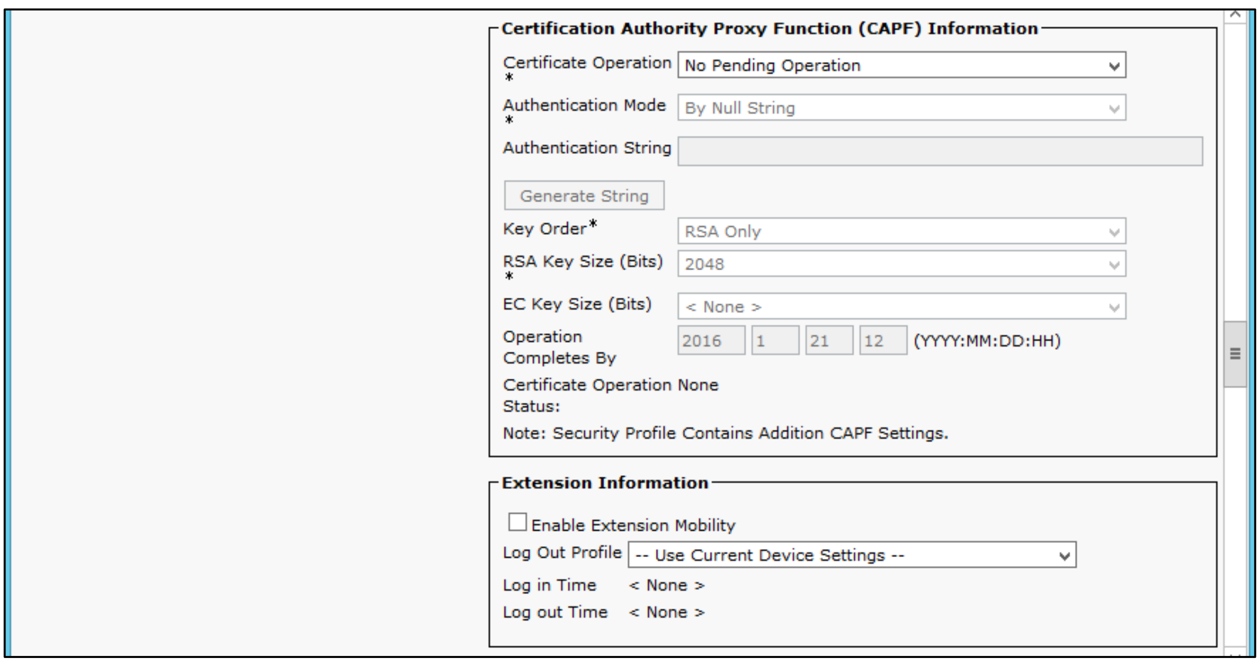
SIP Profile*: Standard SIP Profile w/Early Media Disabled View Details

Digest User: < None >

☐ Media Termination Point Required

☐ Unattended Port

☐ Require DTMF Reception



Certification Authority Proxy Function (CAPF) Information

Certificate Operation*: No Pending Operation

Authentication Mode*: By Null String

Authentication String:

Generate String

Key Order*: RSA Only

RSA Key Size (Bits)*: 2048

EC Key Size (Bits): < None >

Operation Completes By: 2016 1 21 12 (YYYY:MM:DD:HH)

Certificate Operation Status: None

Note: Security Profile Contains Addition CAPF Settings.

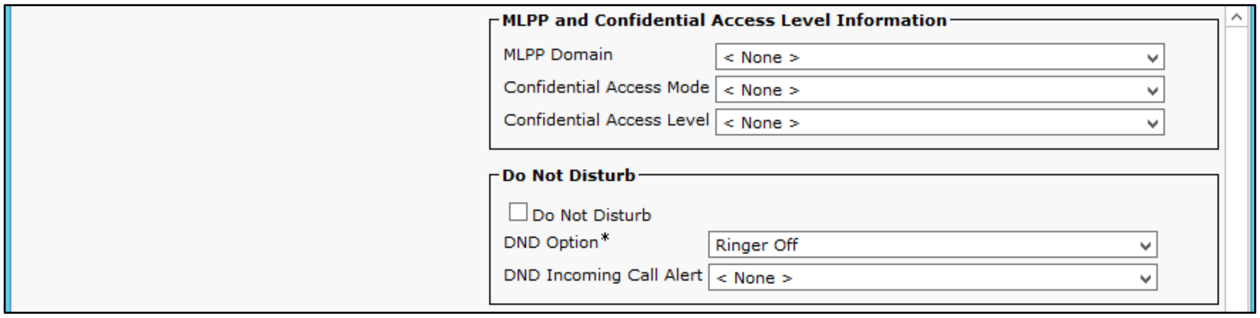
Extension Information

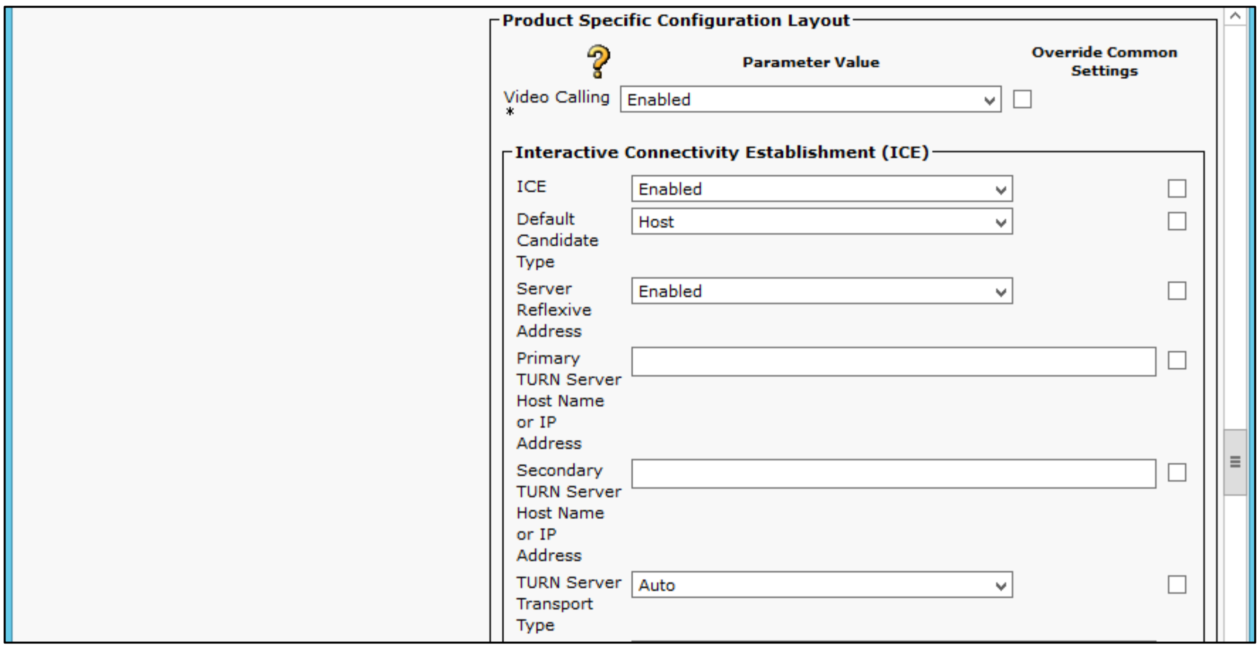
☐ Enable Extension Mobility

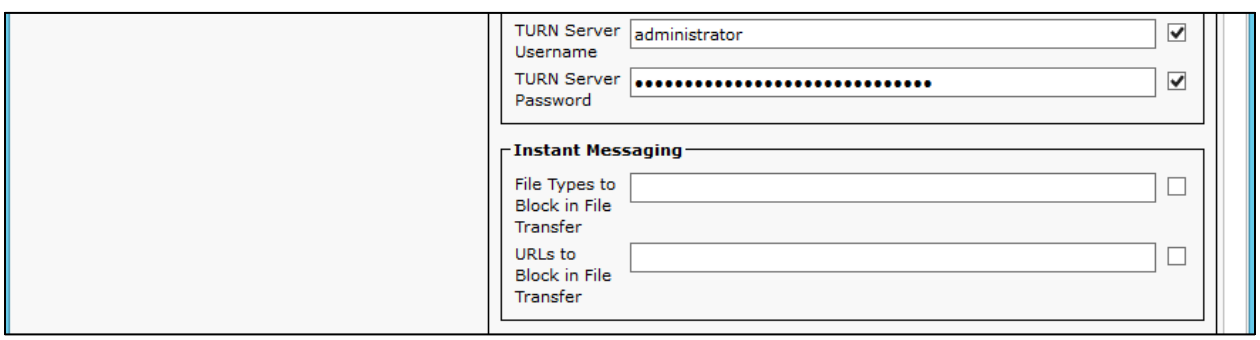
Log Out Profile: -- Use Current Device Settings --

Log in Time: < None >

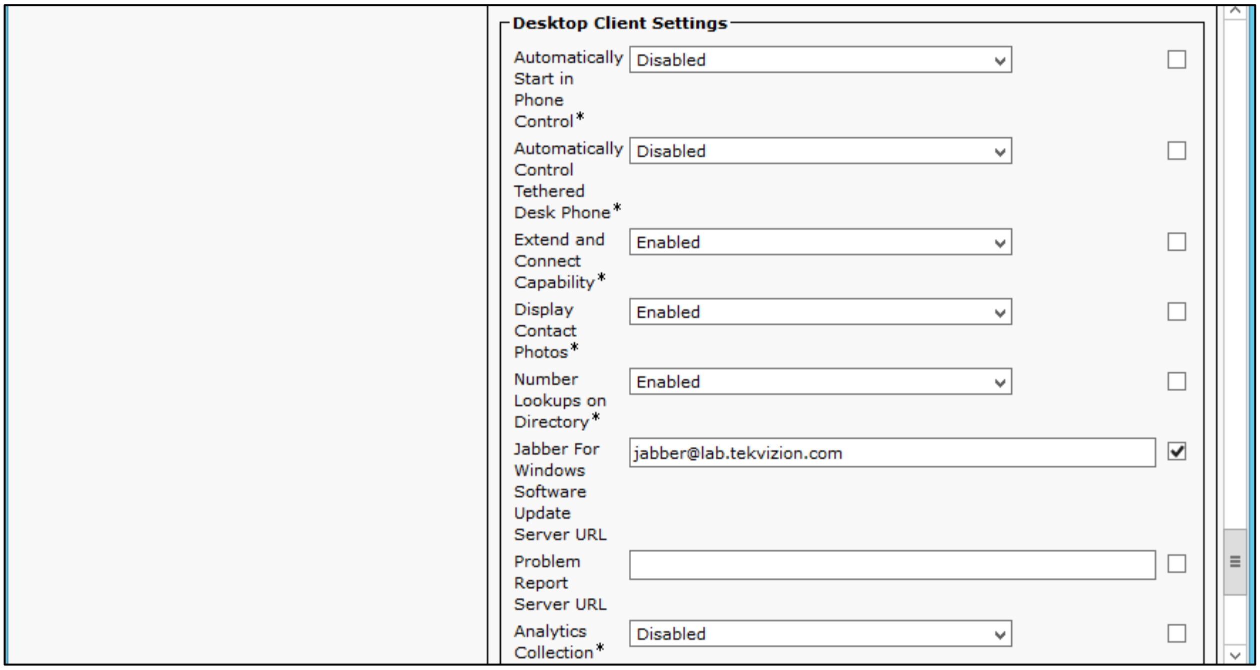
Log out Time: < None >

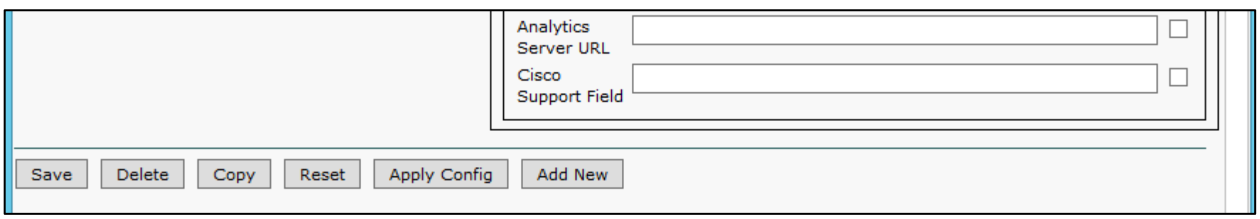

MLPP and Confidential Access Level Information
MLPP Domain
< None >
Confidential Access Mode
< None >
Confidential Access Level
< None >
Do Not Disturb
Do Not Disturb
DND Option*
Ringer Off
DND Incoming Call Alert
< None >


Product Specific Configuration Layout
Parameter Value
Override Common Settings
Video Calling *
Enabled
Interactive Connectivity Establishment (ICE)
ICE
Enabled
Default Candidate Type
Host
Server Reflexive Address
Enabled
Primary TURN Server Host Name or IP Address
Secondary TURN Server Host Name or IP Address
TURN Server Transport Type
Auto


TURN Server Username
administrator
TURN Server Password
Instant Messaging
File Types to Block in File Transfer
URLs to Block in File Transfer

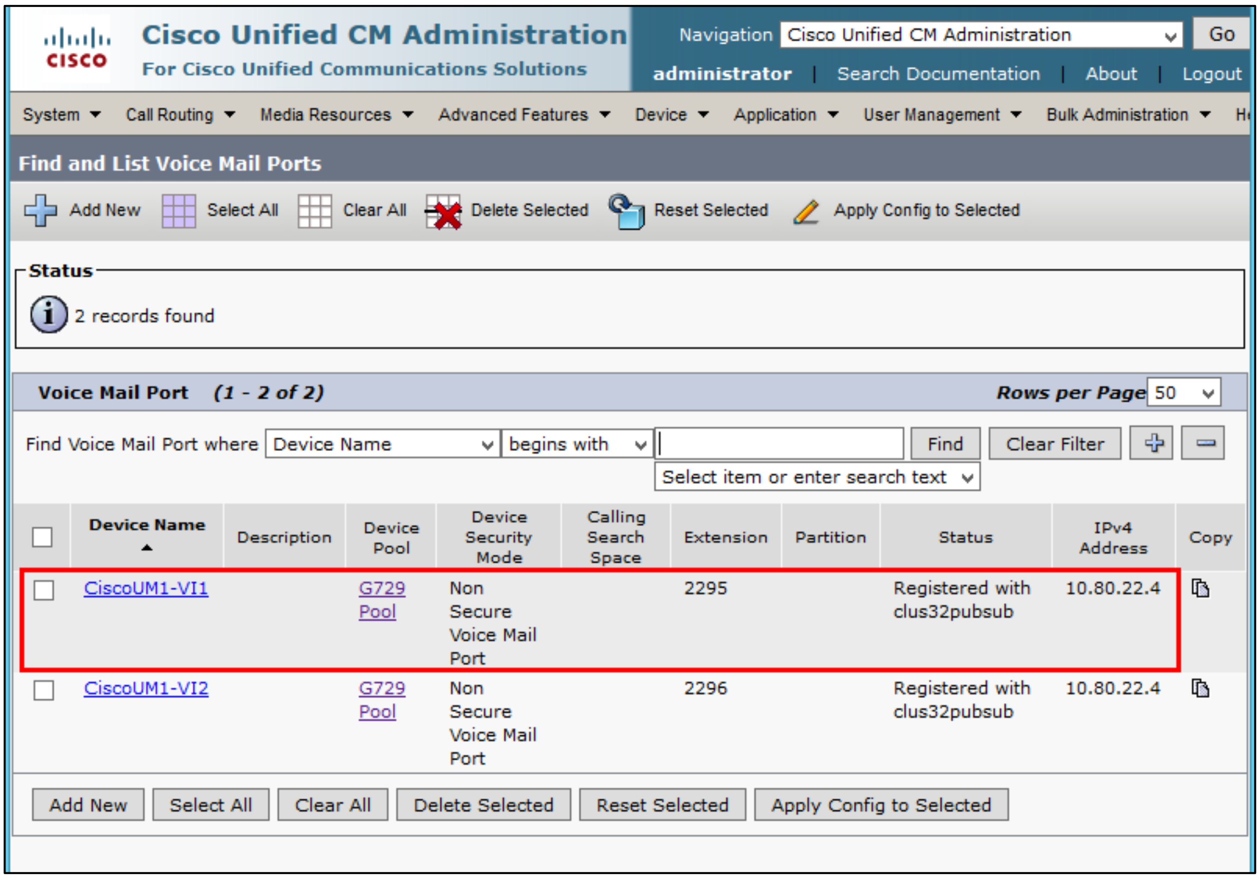
Jabber Client Configuration (Contd…)





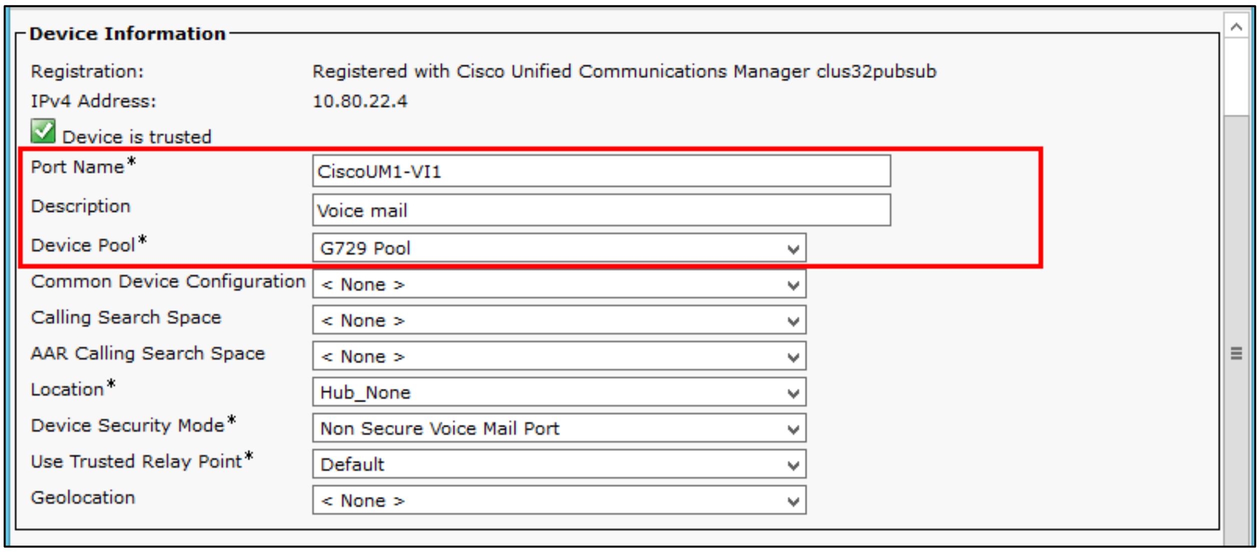
Voicemail Port Configuration

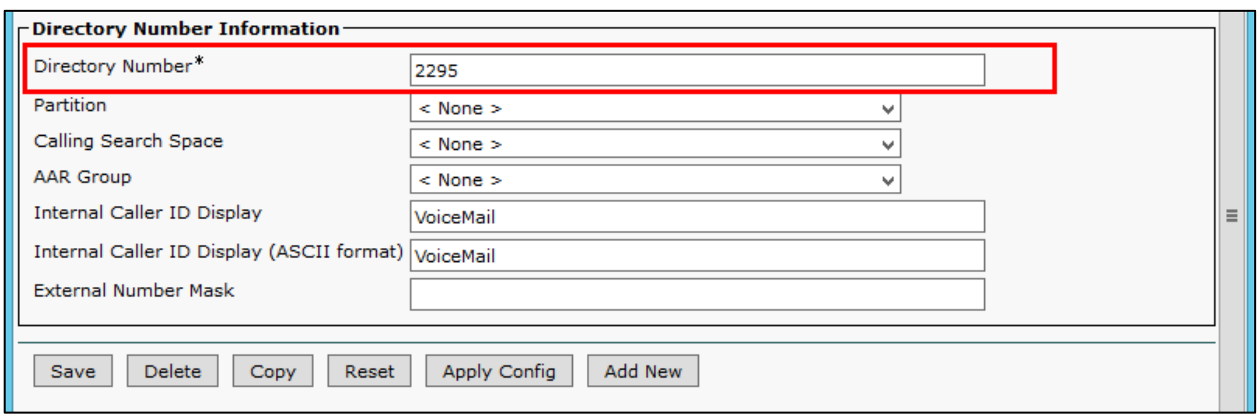
Navigation: Advanced Feature→ Voice Mail → Cisco Voice Mail Port



Voicemail Port Configuration (Continued…)

Set Port Name = CiscoUM1-VI1. This is used for this example.
Set Description = VoiceMail. This is used for this example.
Set Device Pool = G729 Pool
Set Directory Number* = 2295. This is used in this example.

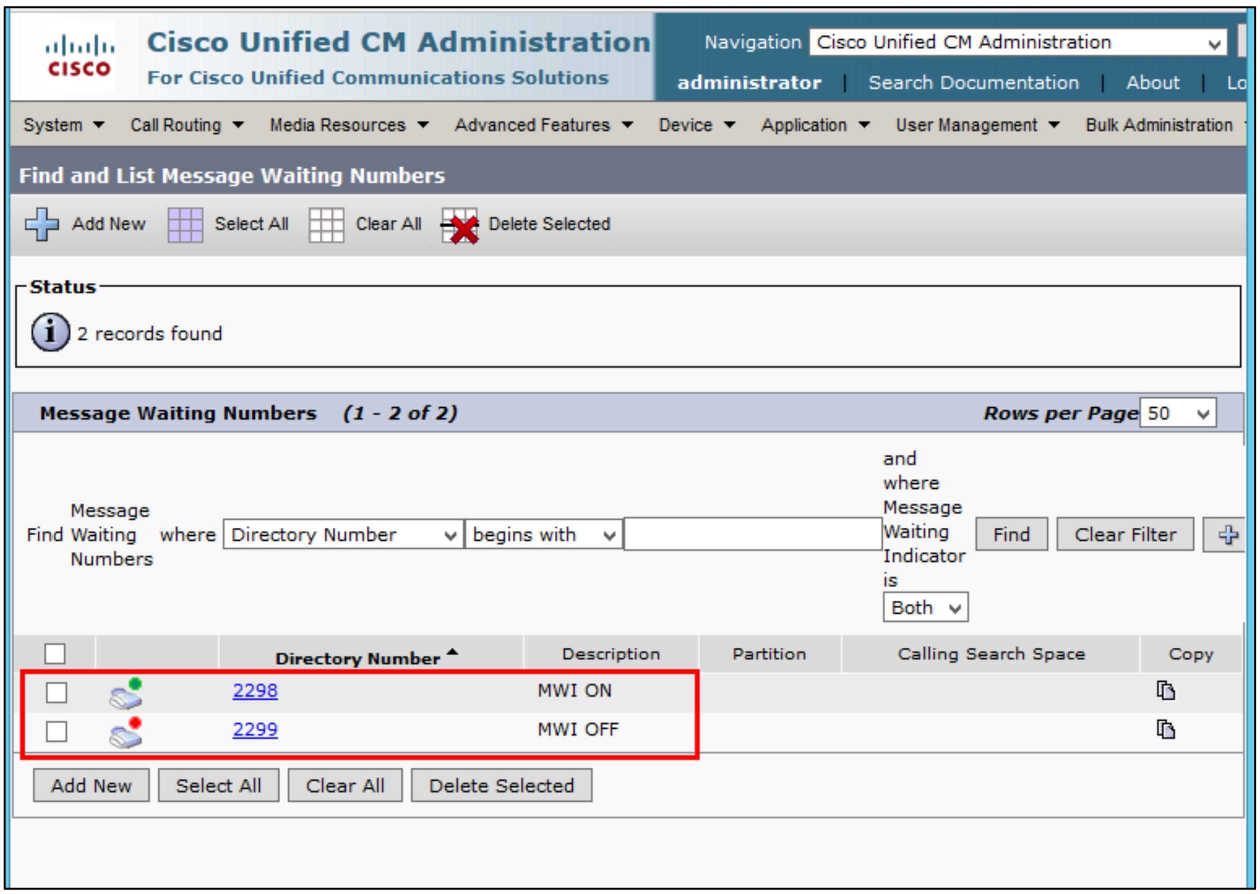




Message Waiting Numbers Configurations

Navigation: Advanced Features → Voice Mail→Message Waiting

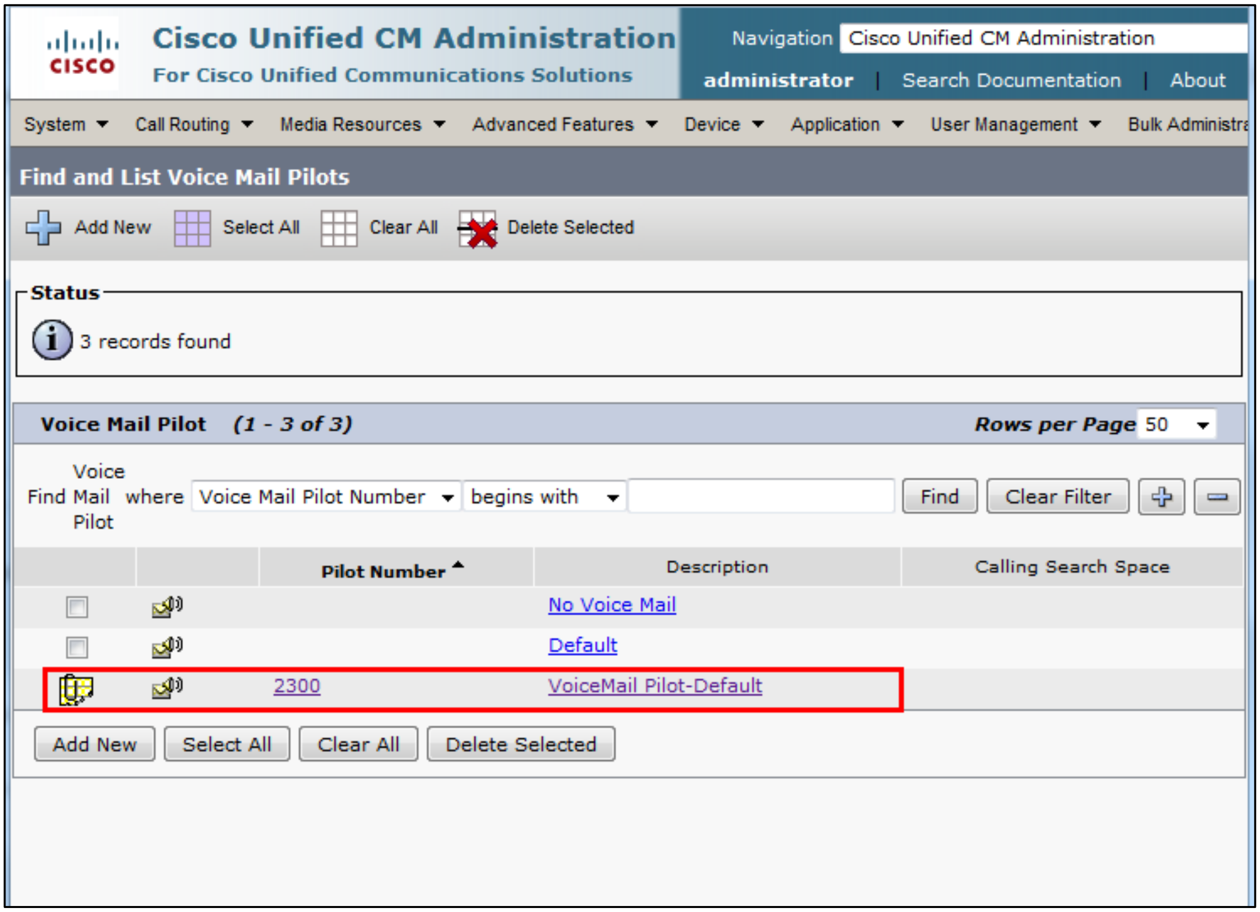
Set Message Waiting Number* = 2298
Set Message Waiting Indicator* = On
Set Message Waiting Number* = 2399
Set Message Waiting Indicator* = Off

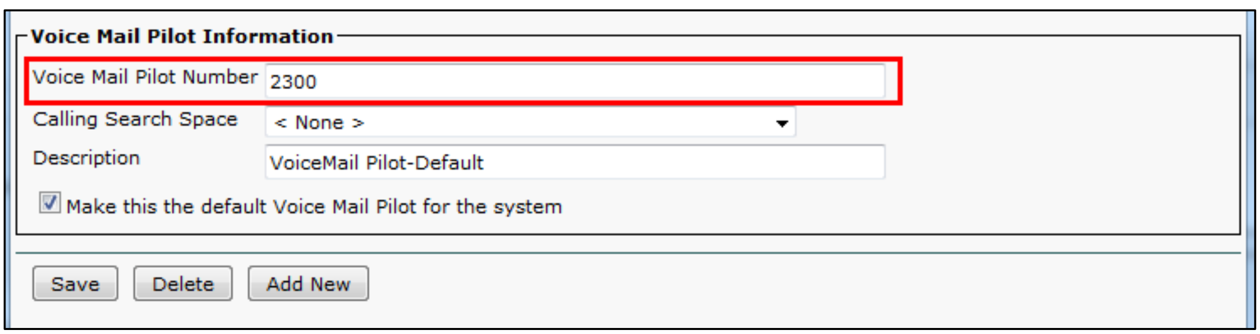


Voicemail Pilot Configuration

Navigation: Advanced Features → Voice Mail → Voice Mail Pilot

Set Voice mail Pilot Number = 2300. This is used for this example
Set Description = VoiceMail Pilot-Default





FAX Gateway Configuration

cme.in.tekvizion.com#**sh version**

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

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

Copyright (c) 1986-2012 by Cisco Systems, Inc.

Compiled Tue 04-Sep-12 15:56 by prod_rel_team

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

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

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

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

Last reload type: Normal Reload

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

Processor board ID FHK1137F4LY

2 Gigabit Ethernet interfaces

62 Serial interfaces

2 terminal lines

2 Channelized E1/PRI ports

4 Voice FXS interfaces

2 cisco service engine(s)

DRAM configuration is 64 bits wide with parity enabled.

239K bytes of non-volatile configuration memory.

62720K bytes of ATA CompactFlash (Read/Write)

License Info:

License UDI:

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

Configuration register is 0x2102

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

Building configuration...

Current configuration : 11391 bytes

!

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

version 15.1

service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

!

hostname cme.in.tekvizion.com

!

boot-start-marker

boot-end-marker

!

!

enable password tekV1z10n

!

aaa new-model

!

!

```
aaa authentication login local_auth local
!
!
!
!
!
aaa session-id common
clock timezone IST 5 30
network-clock-participate wic 2
network-clock-participate wic 3
!
dot11 syslog
ip source-route
!
!
ip cef
!
!
!
ip host Clus1-862-Pub 172.16.26.2
no ipv6 cef
multilink bundle-name authenticated
!
!
!
!
```

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

```
 request ANY sip-header Allow-Header modify "UPDATE," ""
 response ANY sip-header Allow-Header modify "UPDATE," ""
 response ANY sip-header Allow-Header modify "UPDATE," ""
!
!
!
!
!
voice-card 0
!
crypto pki token default removal timeout 0
!
!
!
!
license udi pid CISCO2851 sn FHK1137F4LY
username cisco password 0 tekV1z10n
!
!
controller E1 0/2/0
 pri-group timeslots 1-31 service mgcp
!
controller E1 0/3/0
 clock source internal
 pri-group timeslots 1-31
!
```

```
ip tftp source-interface GigabitEthernet0/0
!
!
!
!
!
!
interface GigabitEthernet0/0
 ip address 172.16.31.50 255.255.255.0
 duplex auto
 speed auto
!
interface Service-Engine0/0
 no ip address
 shutdown
!
interface GigabitEthernet0/1
 no ip address
 ip nat outside
 ip virtual-reassembly in
 shutdown
 duplex auto
 speed auto
!
interface Serial0/2/0:15
 no ip address
```


Note: Testing was conducted in tekVizion labs

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

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


Note: Testing was conducted in tekVizion labs

```
voice-port 0/2/0:15
!
voice-port 0/1/0
 no vad
shutdown
!
voice-port 0/1/1
 cptone IN
 station-id number 7323204084
 caller-id enable
!
no mgcp timer receive-rtcp
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
!
!
!
dial-peer voice 777 pots
 huntstop
 service session
```

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

```
 voice-class codec 3
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!


!
!
gateway
 timer receive-rtp 1200
!
sip-ua
 credentials username +19728522671 password 7 11584B5643475D realm bsrhelas.lab.tekvizion.com
 no remote-party-id
 retry register 5
 timers connection aging 30
 timers update 1000
 no timers hold
 timers register 1000
!
!
telephony-service
```

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

```
ephone  1
 mac-address 001D.A21A.3577
 type 7961
 button  1:1
!
!
!
ephone  2
 mac-address 001D.A21A.291D
 type 7961
 button  1:2
!
!
!
ephone  5
 mac-address 0008.21DF.CEC5
 type 7940
 ssh userid cisco password cisco
 button  1:5
!
!
!
ephone  6
!
!
!
```


Note: Testing was conducted in tekVizion labs

```
ephone  10
 mac-address 0C27.2431.5FB9
 button  1:10
!
!
!
banner login ^CC
==================================================
      WELCOME to CISCO CME 8.6
==================================================

Cisco IOS Software (C2800NM-IPVOICEK9-M)
Version 15.1(4)M5, RELEASE SOFTWARE (fc1)
Cisco 2851 with 249856K/12288K bytes of memory
Processor board ID FHK1137F4LY
2 Gigabit Ethernet interfaces
2 terminal lines
2 Voice FXS interfaces
2 cisco service engine(s)
239K bytes of non-volatile configuration memory
62720K bytes of ATA CompactFlash (Read/Write)

Warning: Access is restricted.
All user activity is logged!
For support, contact 'kkumaraguru@tekvizion.com'
```

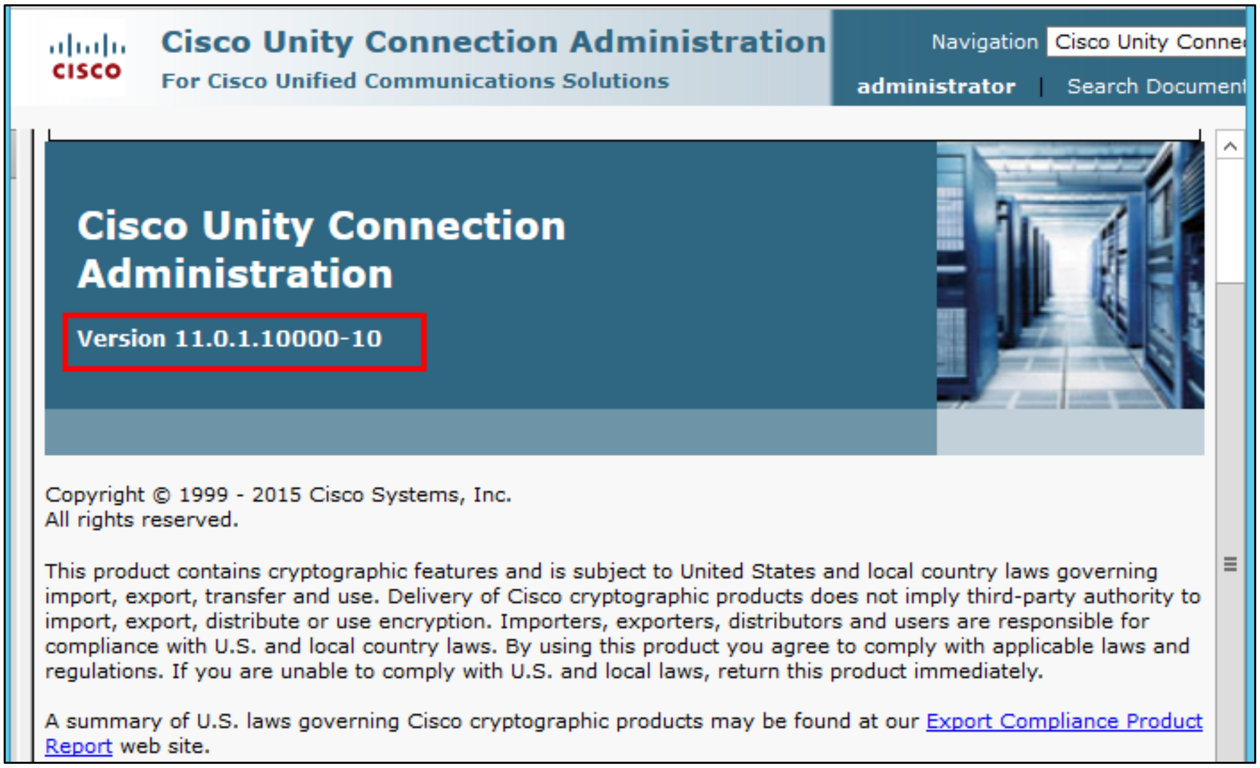
```
==================================================
^C
!
line con 0
line aux 0
line 66
 no activation-character
 no exec
 transport preferred none
 transport input all
 transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
line 194
 no activation-character
 no exec
 transport preferred none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

end

cme.in.tekvizion.com#

Cisco UCM SCCP Integration with Cisco Unity Connection (CUC)

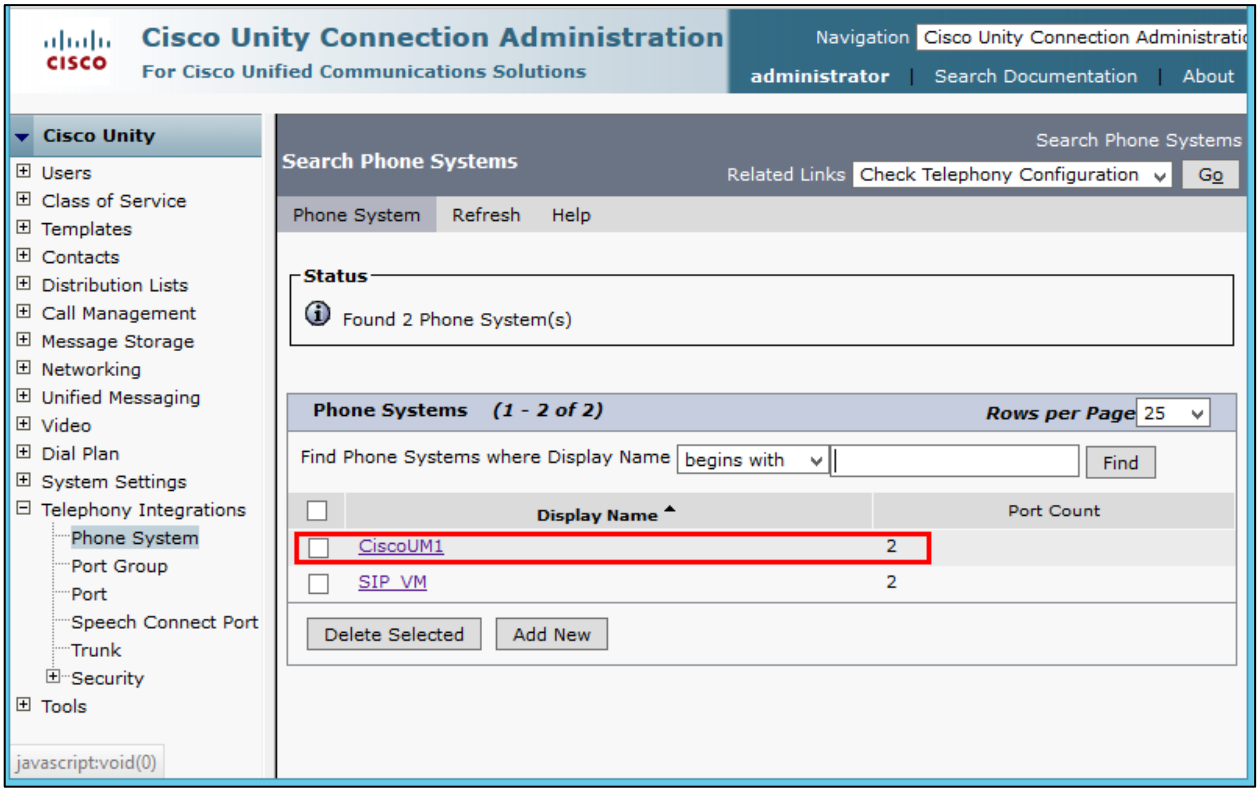
CUC Version



CUC Telephony Integration with Cisco UCM

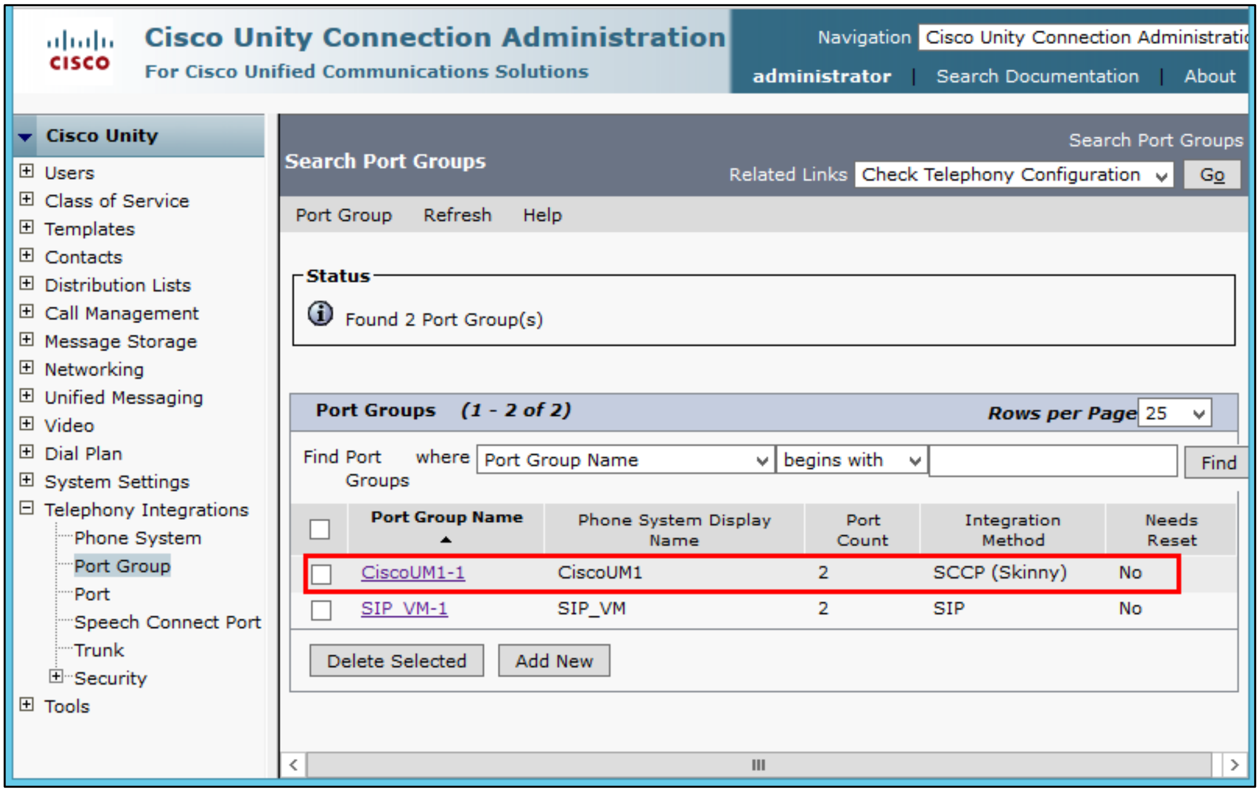
Navigation: Telephony Integrations → Phone system

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



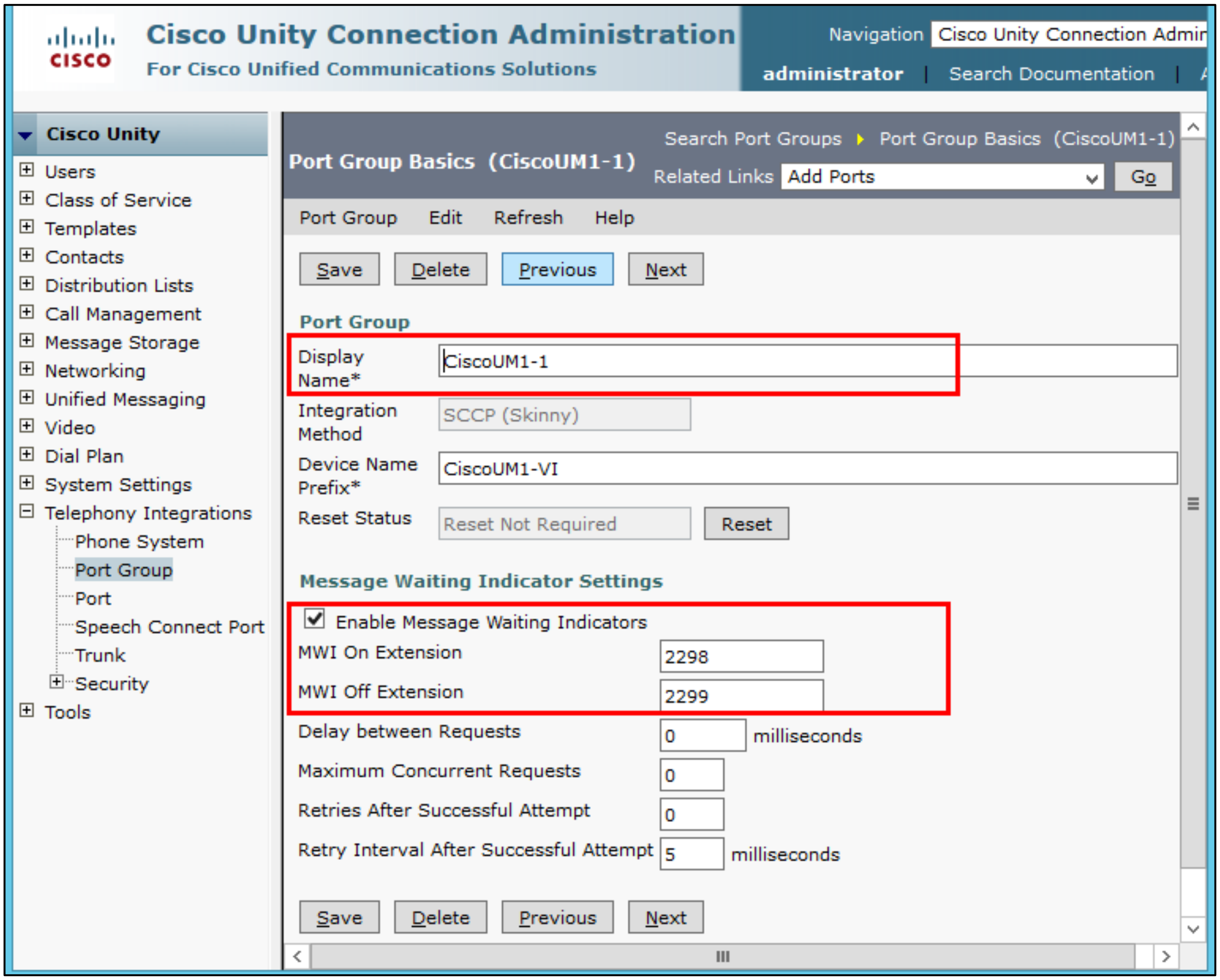
CUC Port Group

Navigation: Telephony Integration → Port Group

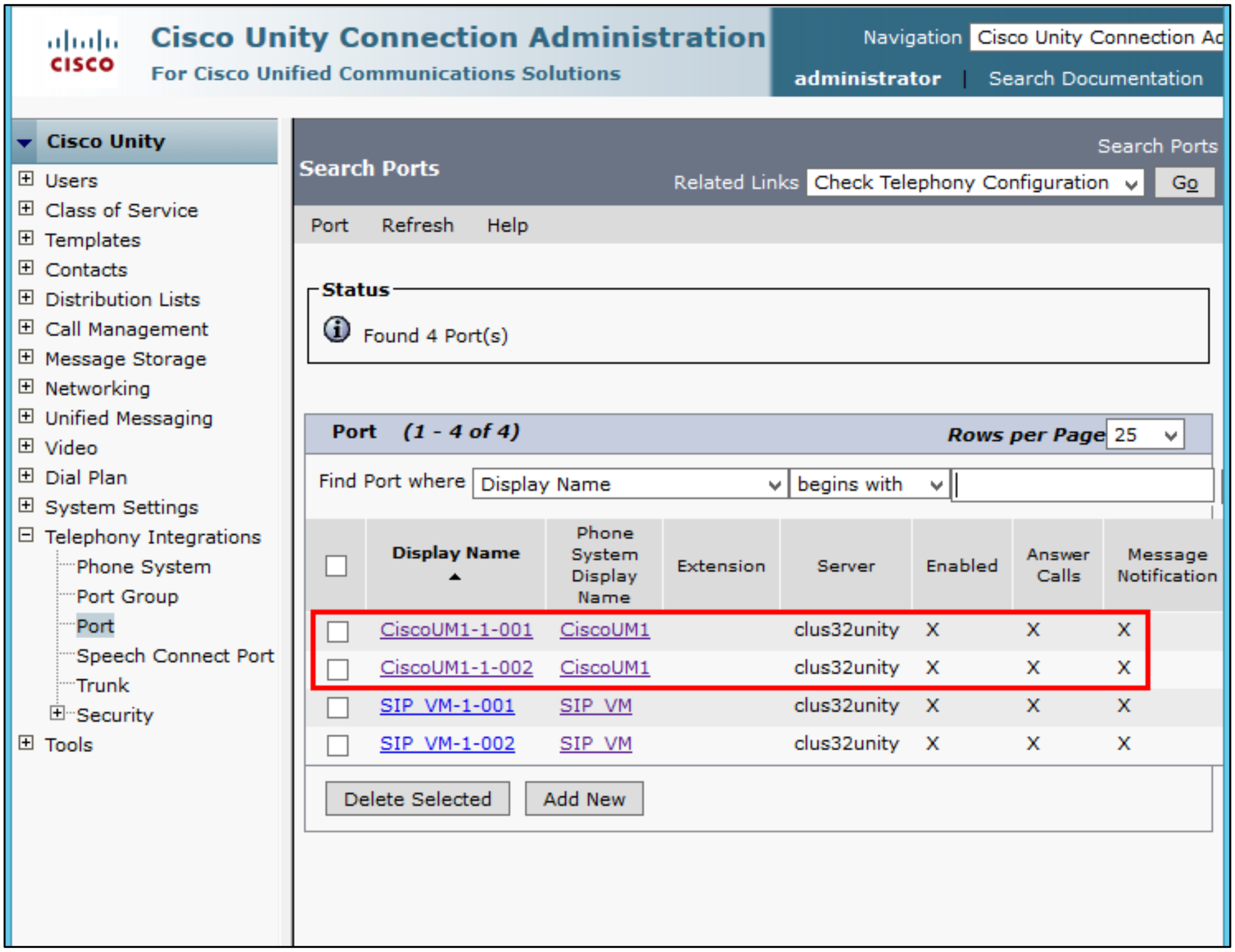


CUC Port Group(continued…)

Set Display Name* = cucmUM-1. This is used in this example.
Check Enable Message waiting indicators.
Set MWI on Extension = 2298. This is used in this example.
Set MWI off Extension= 299. This is used in this example.



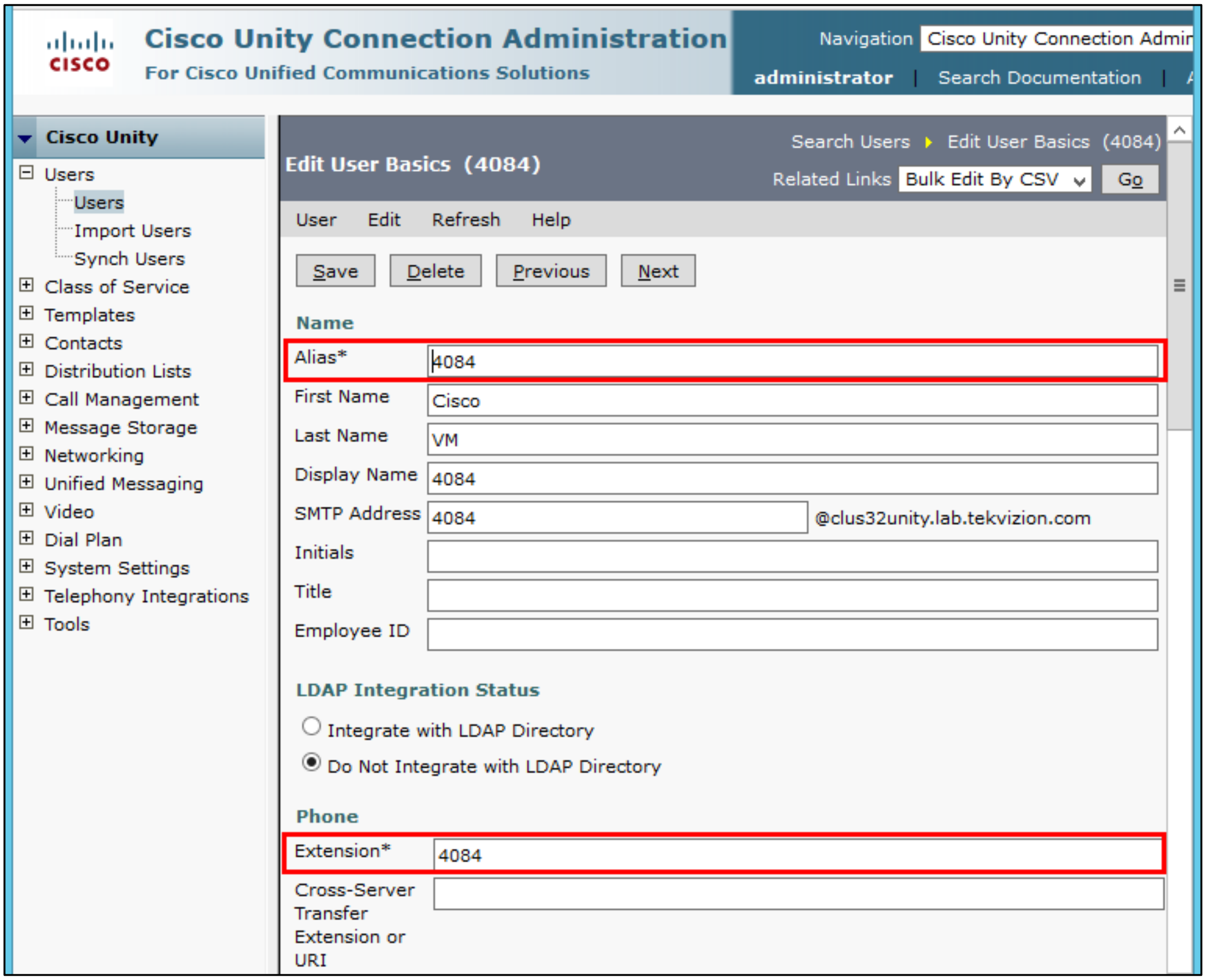
CUC Port Settings



CUC Sample User Basic Settings

Navigation: Cisco Unity connection → Users → Users

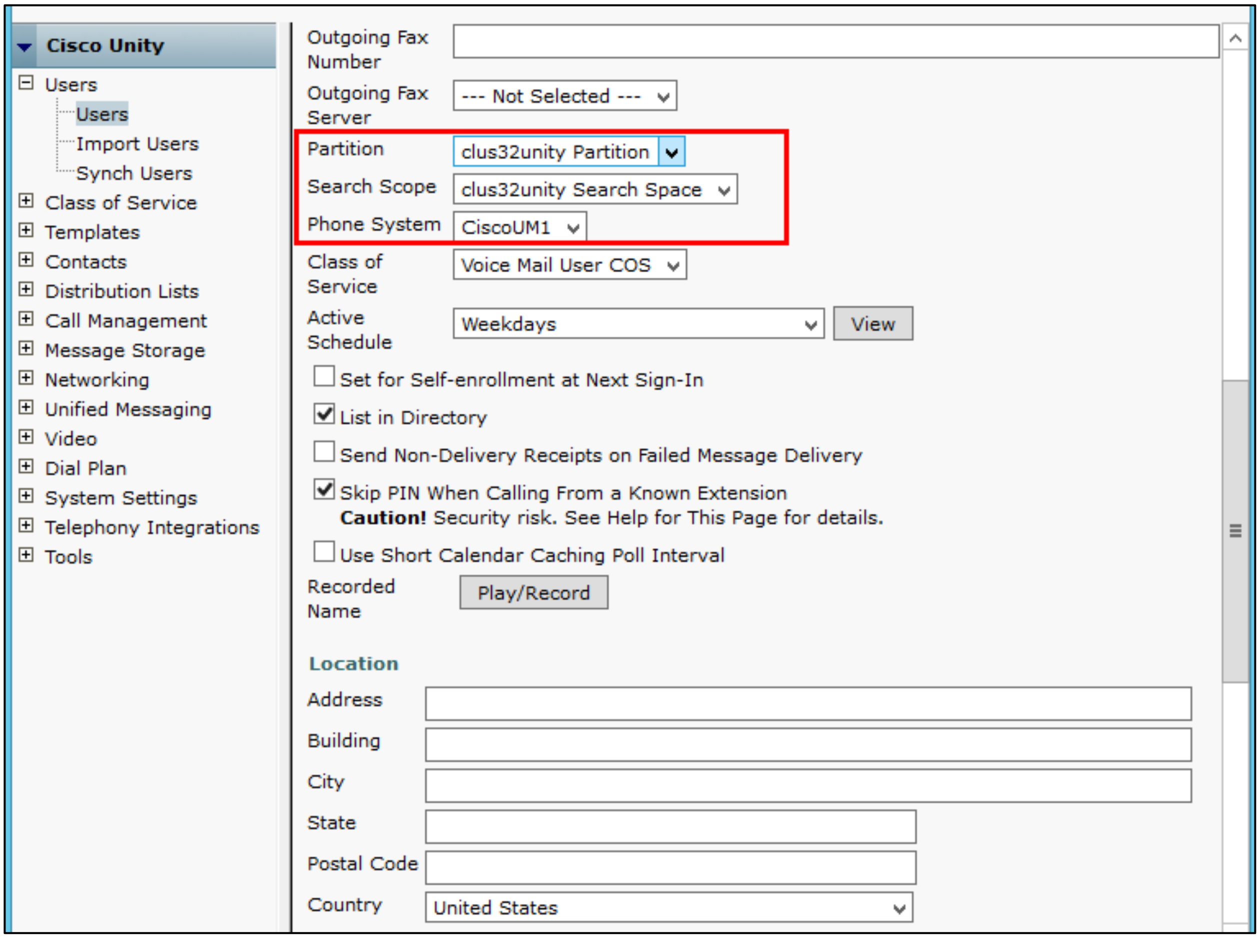
Set Alias = 4084.This is one of the extension used for this testing.
Set Extension = 4084. This is used for this example.



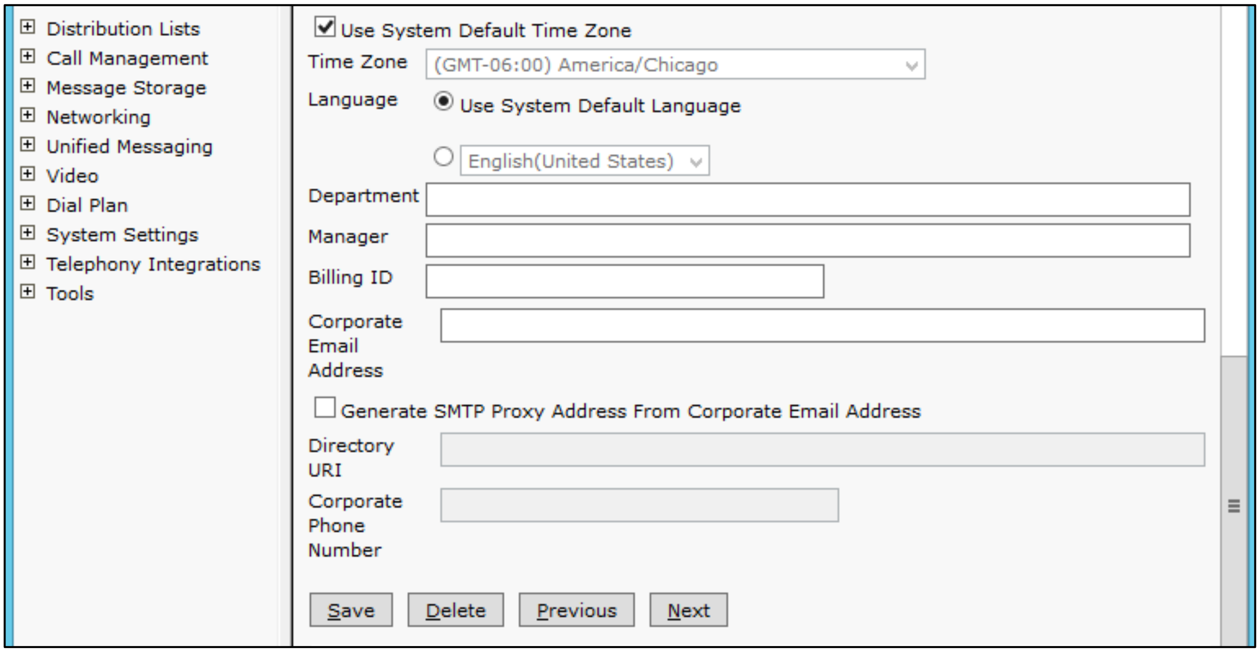
CUC Sample User Basic Settings (Continued…)

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

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



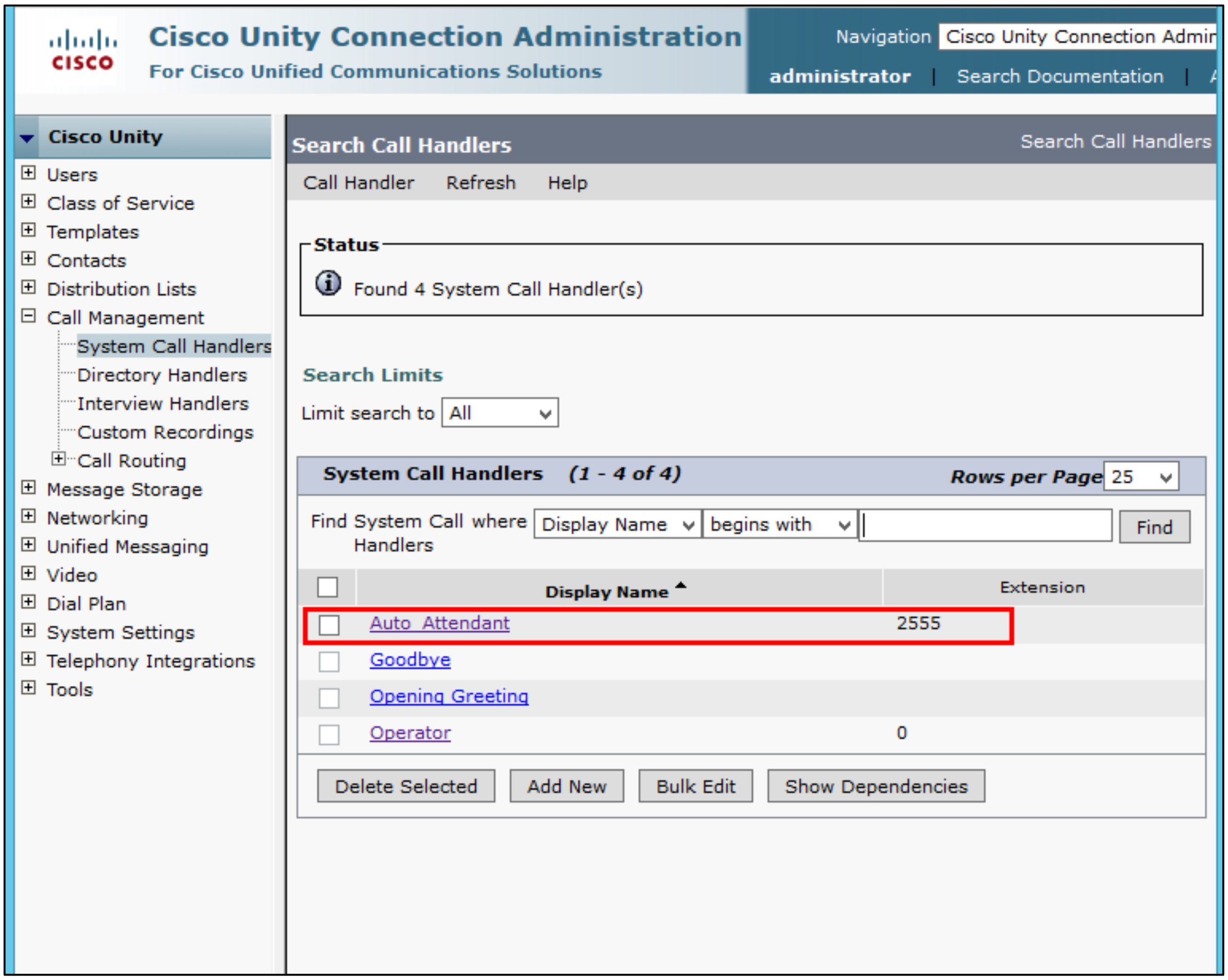
CUC Sample User Basic Settings (Continued…)


Distribution Lists
Call Management
Message Storage
Networking
Unified Messaging
Video
Dial Plan
System Settings
Telephony Integrations
Tools
Use System Default Time Zone
Time Zone
(GMT-06:00) America/Chicago
Language
Use System Default Language
English(United States)
Department
Manager
Billing ID
Corporate Email Address
Generate SMTP Proxy Address From Corporate Email Address
Directory URI
Corporate Phone Number
Save
Delete
Previous
Next

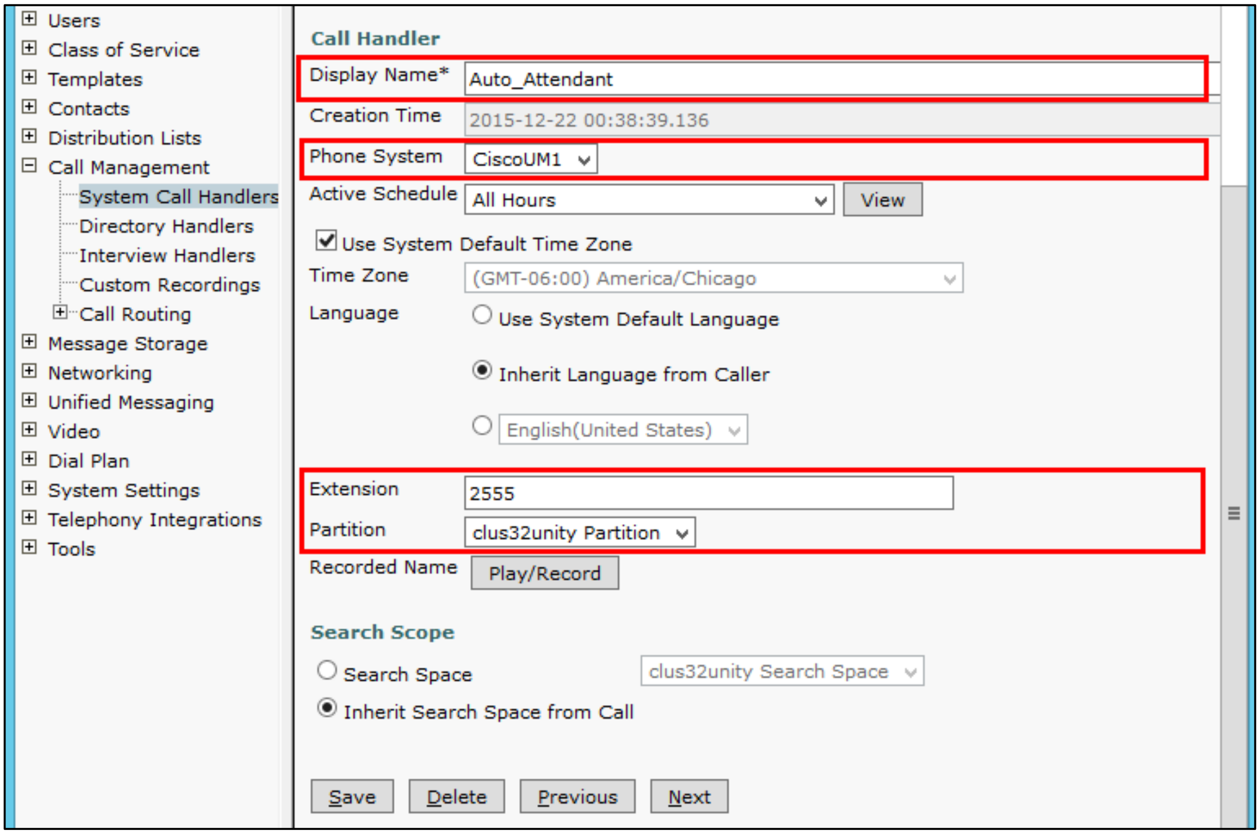
Auto Attendant

Navigation: Call Management → System Call Handlers

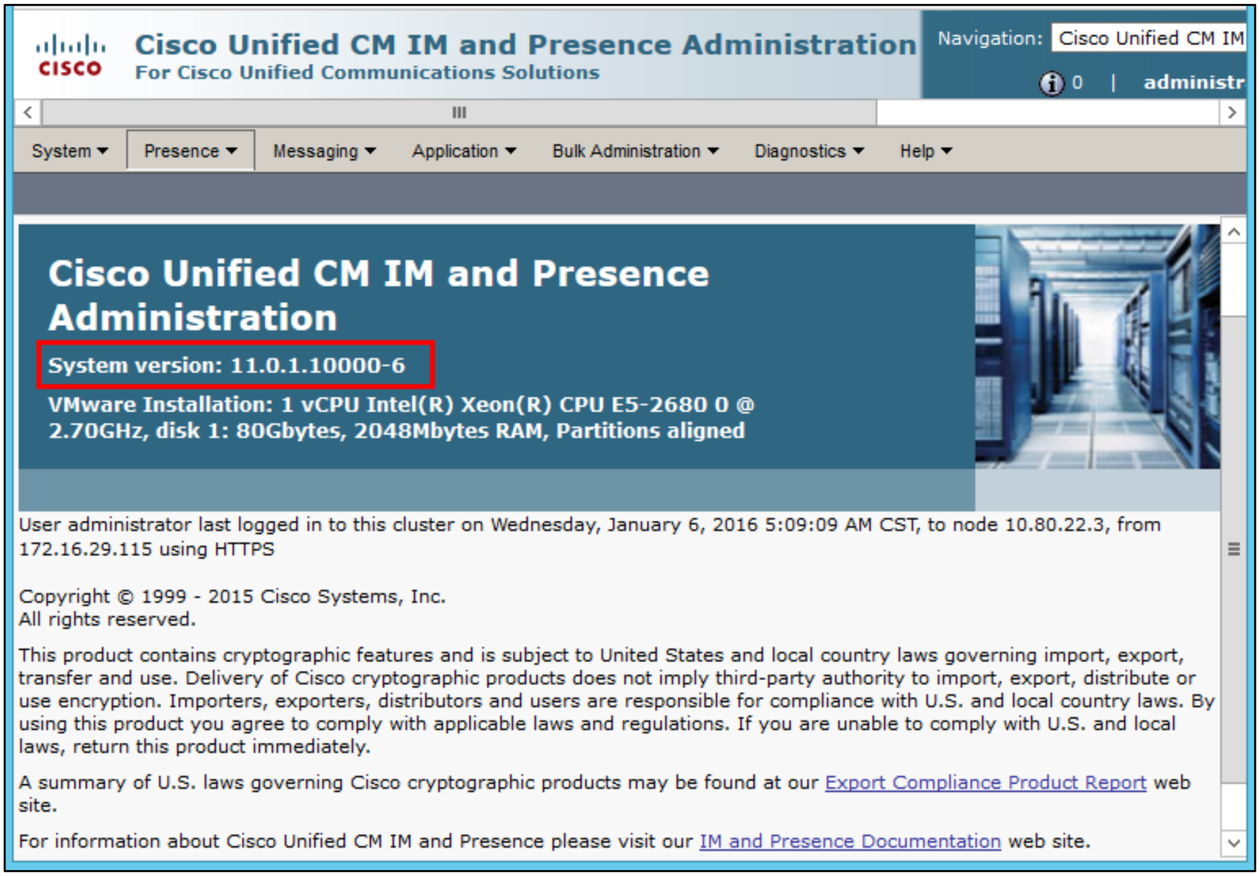
Set Display Name = Demo auto attend. This is used for this example.
Set Phone System = CUCM
Set Extension=2999. This number is used as Auto attendant on this set up.
Set Partition = Clus32unity Partition. This is used for this example.



Auto Attendant (Continued…)

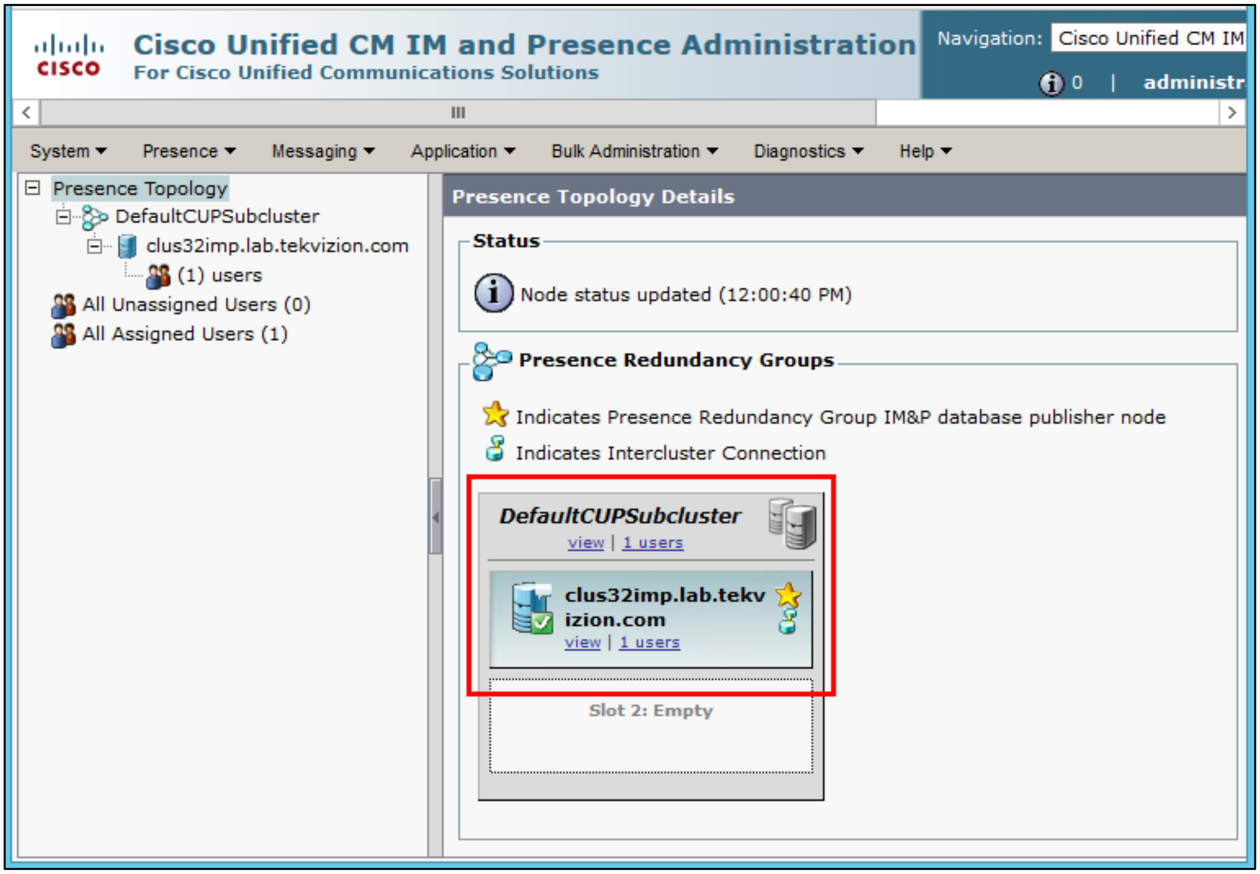


CUP/IMP Version



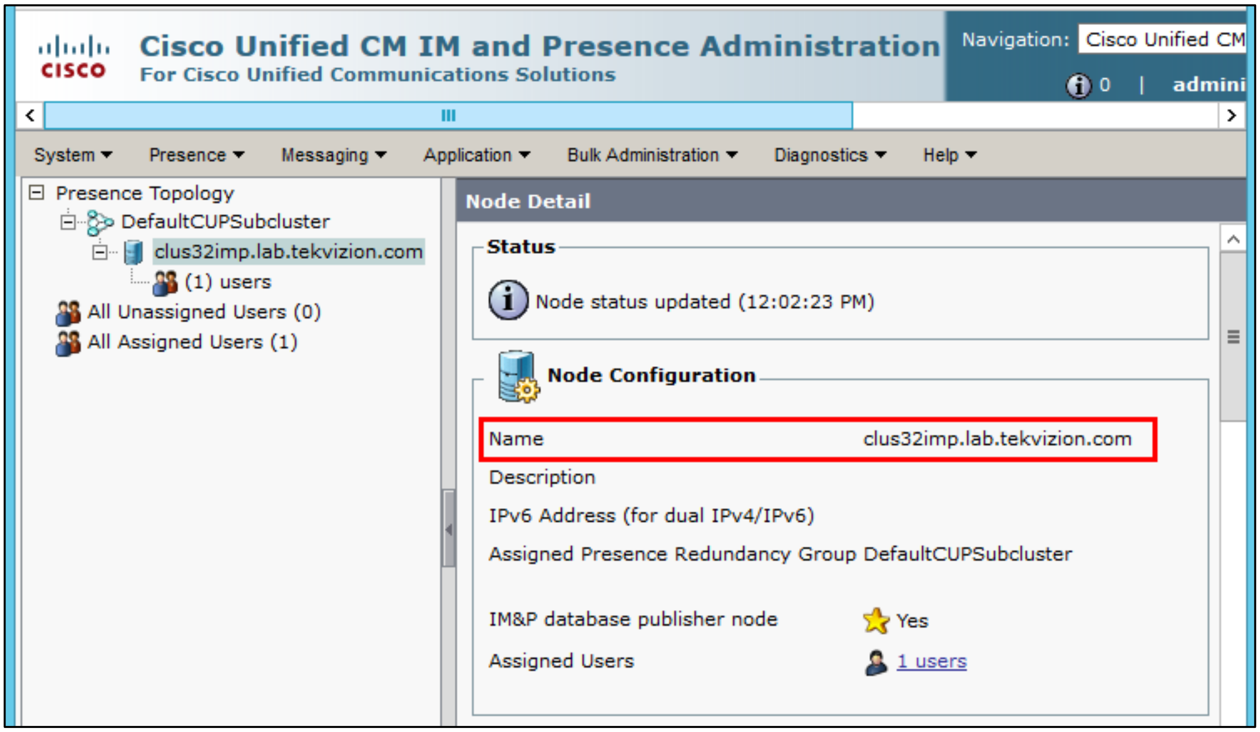
Presence Topology

Navigation: System → Presence Topology



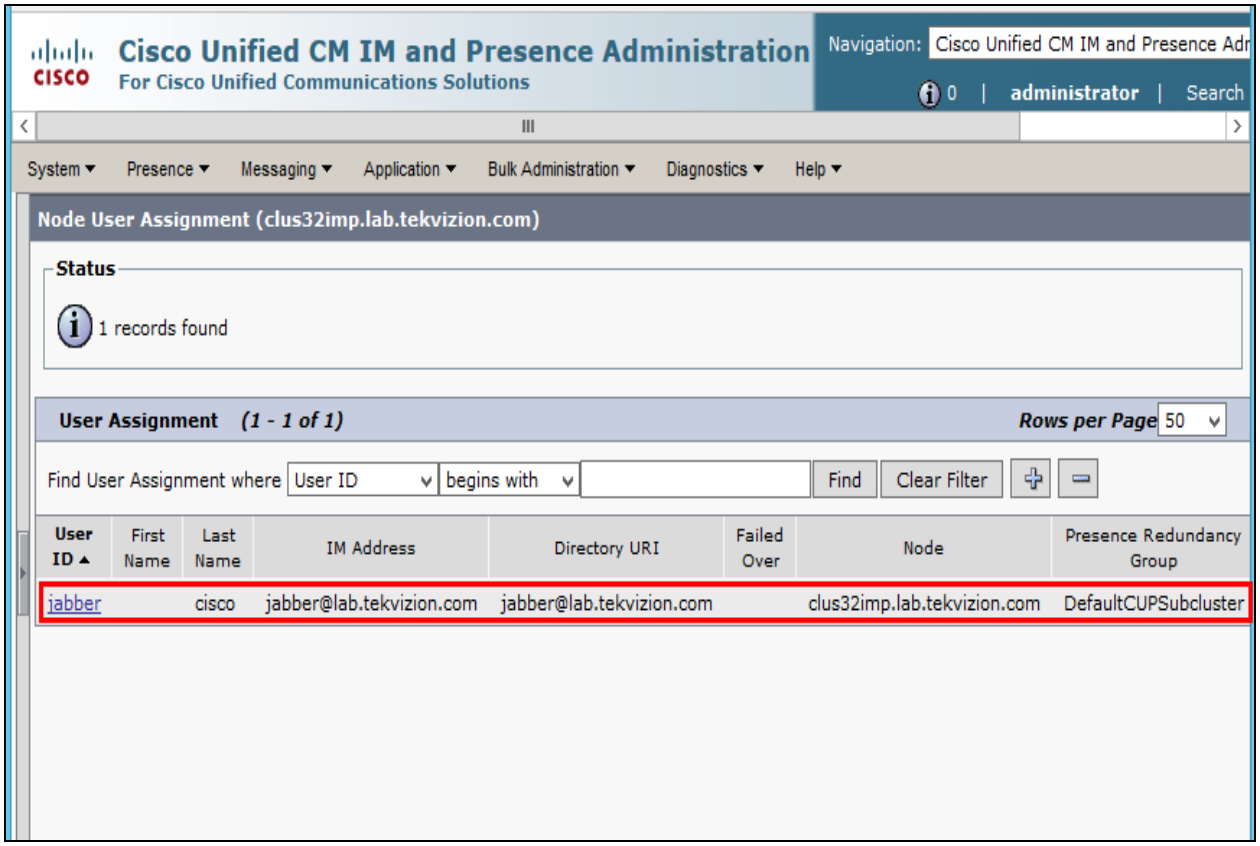
Node Configuration

Navigation: System → Cluster Topology → Fully Qualified Domain Name



Users

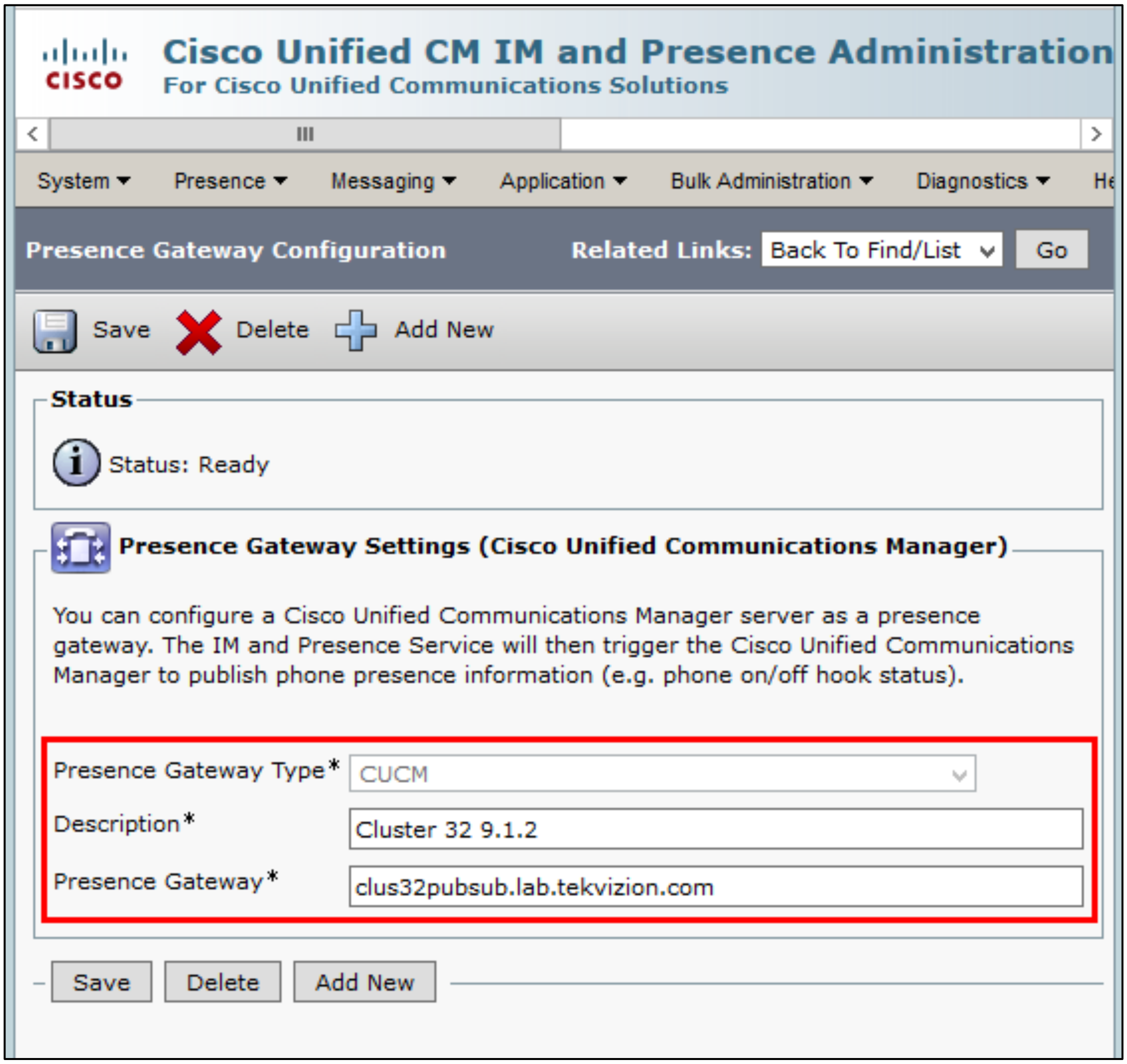
Navigation: System → Cluster Topology → clus32imp.lab.tekvizion.com → Users



Presence gateway configuration

Navigation: Presence → Gateways

Set Presence Gateway Type *= CUCM
Set Description *= Cluster 32 9.1.2. This is used for this example.
Presence Gateway * =clus23pubsub.lab.tekvizion.com



Acronyms

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

Important Information

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS. IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.

Note: Testing was conducted in tekVizion labs

Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters
CiscoSystems International BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

Americas Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

AsiaPacific Headquarters
Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
www.cisco.com
Tel: +65 317 7777
Fax: +65 317 7799

Cisco Systems has more than 200 offices in the following countries and regions. Addresses, phone numbers, and fax numbers are listed on the Cisco Web site at http://www.cisco.com/go/offices.

Argentina • Australia • Austria • Belgium • Brazil • Bulgaria • Canada • Chile • China PRC • Colombia • Costa Rica • Croatia • Czech Republic • Denmark • Dubai, UAE • Finland • France • Germany • Greece • Hong Kong SAR • Hungary • India • Indonesia • Ireland • Israel • Italy • Japan • Korea • Luxembourg • Malaysia • Mexico • The Netherlands • New Zealand • Norway • Peru • Philippines • Poland • Portugal • Puerto Rico • Romania • Russia • Saudi Arabia • Scotland • Singapore • Slovakia • Slovenia • South Africa • Spain • Sweden • Switzerland • Taiwan • Thailand • Turkey Ukraine • United Kingdom • United States • Venezuela • Vietnam • Zimbabwe