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

Note: Testing was conducted in tekVizion labs

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Introduction

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

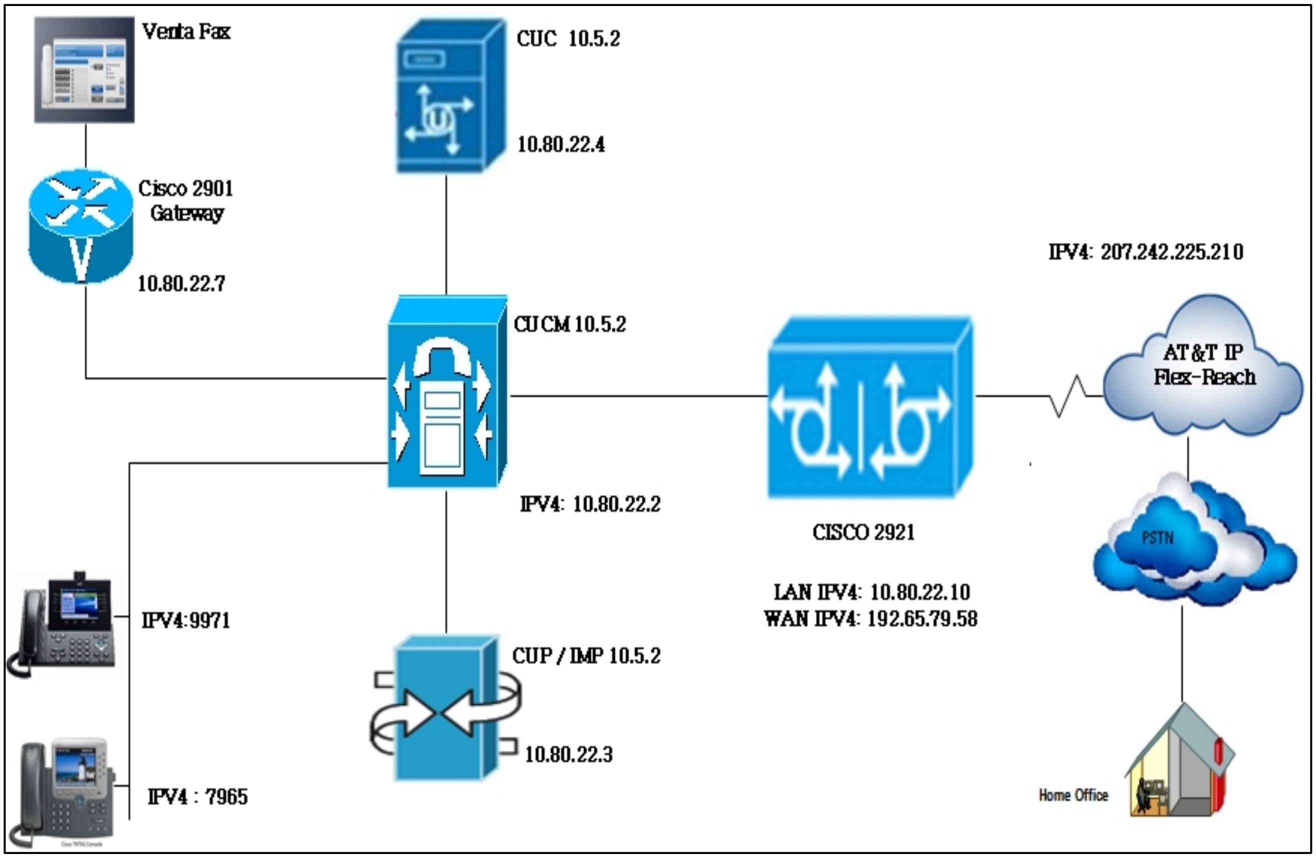
- This application note describes the necessary steps and configurations of Cisco Unified Communications Manager (Cisco UCM) 10.5.2, Cisco Unity Connection 10.5.2, Cisco Unified CM IM and Presence 10.5.2, Cisco Integrated Services Routers (ISR) Version 15.4(3) M1 with connectivity to AT&T's IP Flex-Reach SIP trunk service. It also covers support and configuration example of Cisco Unity Connection (CUC) messaging integrated with Cisco Unified Communications Manager (Cisco UCM). The deployment model covered in this application note is Cisco Integrated Services Routers (ISR) to PSTN (AT&T IP Flexible Reach SIP). AT&T IP Flexible Reach provides inbound and outbound call service.

- Testing was performed in accordance to AT&T's IP Flexible Reach test plan and all features were verified. Key features verified are: inbound and outbound basic call (including international calls), calling name delivery, calling number and name restriction, CODEC negotiation, intra-site transfers, intra-site conferencing, call hold and resume, call forward (forward all, busy and no answer), leaving and retrieving voicemail (Cisco Unity Connection), CISCO auto-attendant (BACD), fax G.711 and T38 (G3 and SG3 speeds), teleconferencing, failover of unresponsive SIP network to PSTN and outbound/inbound calls to/from TDM networks.

The Cisco Unified Border Element function configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between AT&T SIP network and Cisco Integrated services router. The configurations described in this document details the important commands for successful interoperability. Care must be taken by the network administrator deploying Cisco ISR to ensure these commands are set per each dial-peer required, to interoperate done AT&T SIP network.

- Consult your Cisco representative for the correct IOS image and for the specific application and Device Unit License and Feature License requirements for all your Cisco Unified Communication Manager with Cisco Unified Border Element components.

Network Topology



Hardware Components

- UCS-C240 VMWare server running ESXi 5.5
- Cisco IP Phones. This solution was tested with Cisco 7965 & Cisco 9971 phones
- Cisco integrated Service Router G2 - Cisco CISCO2921/K9 (revision 1.0) with 483328K/40960K bytes of memory
- Processor board ID FTX1746AJCC 3 Gigabit Ethernet interfaces, 1 terminal line, 1 Virtual Private Network (VPN) Module, DRAM configuration is 64 bits wide with parity enabled, 255K bytes of non-volatile configuration memory

Software Requirements

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

Features

Features – Supported

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

Network Based Features - Supported

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

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

Features - Not Supported

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

Caveats

Fax

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

Auto-Attendant

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

Hold/Resume & Music on Hold (MOH)

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

Ringback Tone on Early Unattended Transfer

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

PBX Based Call Forward Unconditional

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

SIP Provisional Acknowledgement/Early media

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

AT&T IP Teleconferencing (IPTC)

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

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

Configuration Considerations

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

Emergency 911/E911 Services Limitations and Restrictions

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

ISR Configuration

topcube#**sh version**

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

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

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

Compiled Sat 25-Oct-14 03:34 by prod_rel_team

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

topcube uptime is 3 days, 8 hours, 0 minutes

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

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

Last reload type: Normal Reload

Last reload reason: Reload Command

This product contains cryptographic features and is subject to United

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

use. Delivery of Cisco cryptographic products does not imply

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

Importers, exporters, distributors and users are responsible for

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

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

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

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

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

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

export@cisco.com.

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

Processor board ID FTX1746AJCC

3 Gigabit Ethernet interfaces

1 terminal line

1 Virtual Private Network (VPN) Module

DRAM configuration is 64 bits wide with parity enabled.

255K bytes of non-volatile configuration memory.

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

License Info:

License UDI:

Device#   PID   SN

----

*1   CISCO2921/K9   FTX1746AJCC

Technology Package License Information for Module:'c2900'

Technology   Technology-package   Technology-package

Current   Type   Next reboot

----

ipbase   ipbasek9   Permanent   ipbasek9

security   securityk9   Permanent   securityk9

uc   uck9   Permanent   uck9

data   None   None   None

NtwkEss   None   None   None

CollabPro   None   None   None

Configuration register is 0x2102

topcube#**sh running-configuration**

Building configuration...

Current configuration : 10923 bytes

!

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

!

version 15.4

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

```
!
!
!
!
!
!
no ip domain lookup
ip cef
no ipv6 cef
!
multilink bundle-name authenticated
!
!
!
!
!
!
cts logging verbose
!
!
voice-card 0
 dspfarm
 dsp services dspfarm
!
!
!
```


Note: Testing was conducted in tekVizion labs

```
voice service voip
 no ip address trusted authenticate
 address-hiding [1]
 mode border-element [2]
 allow-connections sip to sip [3]
 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
```

---

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

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

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

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

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

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

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

[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.

!

voice class codec 1 [9]

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.

```
!
!
!
!
!
voice translation-rule 1 [13]
 rule 1 /^.*\(40..\)/ /732320\1/
!
voice translation-rule 2
 rule 2 /^\+\(1\)\(7.........\)/ /\2/
!
!
voice translation-profile NPA
 translate calling 1
!
voice translation-profile test+1
 translate called 2
!
!
!
license udi pid CISCO2921/K9 sn FTX1746AJCC
hw-module pvdm 0/0
!
```

---

[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.

[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.

```
!
!
username cisco privilege 15 password 0 cisco
!
redundancy
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
interface Embedded-Service-Engine0/0
 no ip address
 shutdown
!
interface GigabitEthernet0/0 [14]
```

[14] WAN interface to AT&T

```
 ip address 192.65.79.58 255.255.255.224
 duplex auto
 speed auto
!
interface GigabitEthernet0/1 [15]
 ip address 10.80.22.10 255.255.255.0 [16]
 duplex auto
 speed auto
!
interface GigabitEthernet0/2
 no ip address
 shutdown
 duplex auto
 speed auto
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
!
ip route 0.0.0.0 0.0.0.0 192.65.79.33
ip route 10.0.0.0 255.0.0.0 10.80.22.1
ip route 10.64.0.0 255.255.0.0 10.80.22.1
ip route 172.16.0.0 255.255.0.0 10.80.22.1
```

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

```
!
!
!
!
control-plane
!
 !
 !
 !
 !
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
!
!
dial-peer voice 100 voip
```
[17]

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

[17] Dial peer for AT&T facing network

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 [23]

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

dtmf-relay rtp-nte [24]

fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none [25]

no vad

!

dial-peer voice 200 voip

description "Outgoing To AT&T .IP PBX facing side"

no modem passthrough

---

[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.

[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

```
 session protocol sipv2
 incoming called-number [27]T
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/1
 voice-class sip bind media source-interface GigabitEthernet0/1
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 300 voip
 description " Int'l calls to AT&T - AT&T facing side "
 destination-pattern 011T
 no modem passthrough
 session protocol sipv2
 session target ipv4:207.242.225.210
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
```

```
 voice-class sip bind control source-interface GigabitEthernet0/0
 voice-class sip bind media source-interface GigabitEthernet0/0
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 400 voip
 description " Int'l calls to AT&T - IP-PBX facing side "
 no modem passthrough
 session protocol sipv2
 incoming called-number 011T
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/1
 voice-class sip bind media source-interface GigabitEthernet0/1
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 500 voip
 description " N11 Calls to AT&T - AT&T facing side "
```

```
 destination-pattern .11
 no modem passthrough
 session protocol sipv2
 session target ipv4:207.242.225.210
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0
 voice-class sip bind media source-interface GigabitEthernet0/0
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 600 voip
 description " N11 Calls to AT&T - IP-PBX facing side "
 no modem passthrough
 session protocol sipv2
 incoming called-number .11
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
```

```
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/1
 voice-class sip bind media source-interface GigabitEthernet0/1
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 800 voip
 description " Incoming AT&T to IP-PBX . AT&T facing side "
 translation-profile incoming test+1
 huntstop
 no modem passthrough
 session protocol sipv2
 incoming called-number +1[37][13][24]32040..
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0
 voice-class sip bind media source-interface GigabitEthernet0/0
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
```


Note: Testing was conducted in tekVizion labs

```
!
dial-peer voice 214 voip
 description "Outgoing To AT&T"-AT&T facing side
 destination-pattern [2-9]T
 no modem passthrough
 session protocol sipv2
 session target ipv4:207.242.225.210
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0
 voice-class sip bind media source-interface GigabitEthernet0/0
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 122 voip
 description "OPERATOR TESTING"
 destination-pattern 0
 no modem passthrough
 session protocol sipv2
```

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

```
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0
 voice-class sip bind media source-interface GigabitEthernet0/0
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 2151 voip
 description "Outgoing To AT&T"-AT&T facing side
 destination-pattern 7323204292
 no modem passthrough
 session protocol sipv2
 session target ipv4:207.242.225.210
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0
 voice-class sip bind media source-interface GigabitEthernet0/0
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
```

```
!
dial-peer voice 700 voip
 description " Incoming AT&T to IP-PBX - IP-PBX facing side "
 huntstop
 destination-pattern [37][13][24].......
 no modem passthrough
 session protocol sipv2
 session target ipv4:10.80.22.2:5060
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/1
 voice-class sip bind media source-interface GigabitEthernet0/1
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
!
gateway
 media-inactivity-criteria all
 timer receive-rtcp 5
```


Note: Testing was conducted in tekVizion labs

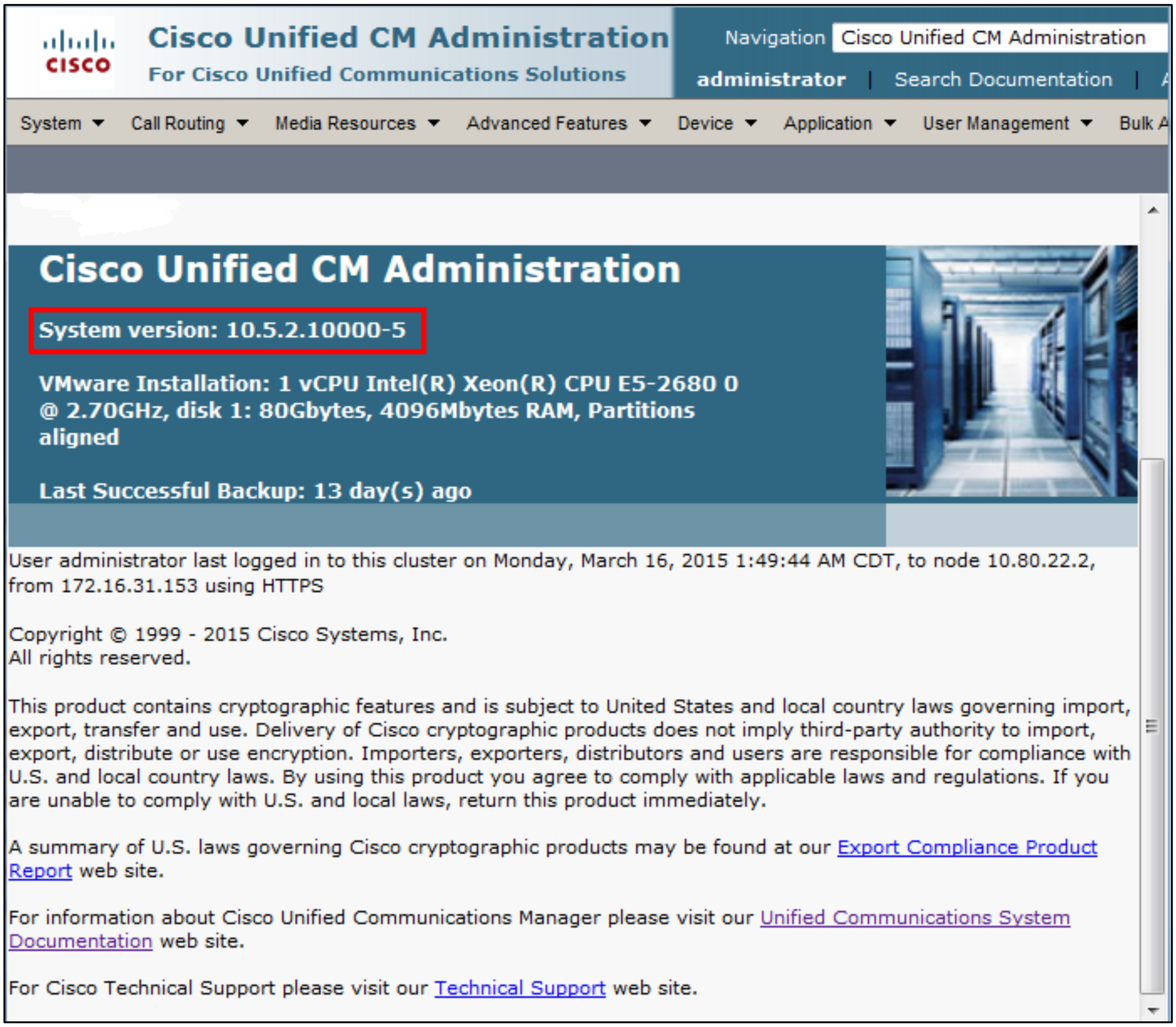
```
 timer receive-rtp 86400
!
sip-ua
 no remote-party-id
 retry invite 2
 timers expires 1800000
 connection-reuse
 protocol mode ipv4
!
!
!
gatekeeper
 shutdown
!
!
!
line con 0
 logging synchronous
line aux 0
line 2
 session-timeout 90
 no activation-character
 no exec
 transport preferred none
 transport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

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

Cisco UCM Configuration

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

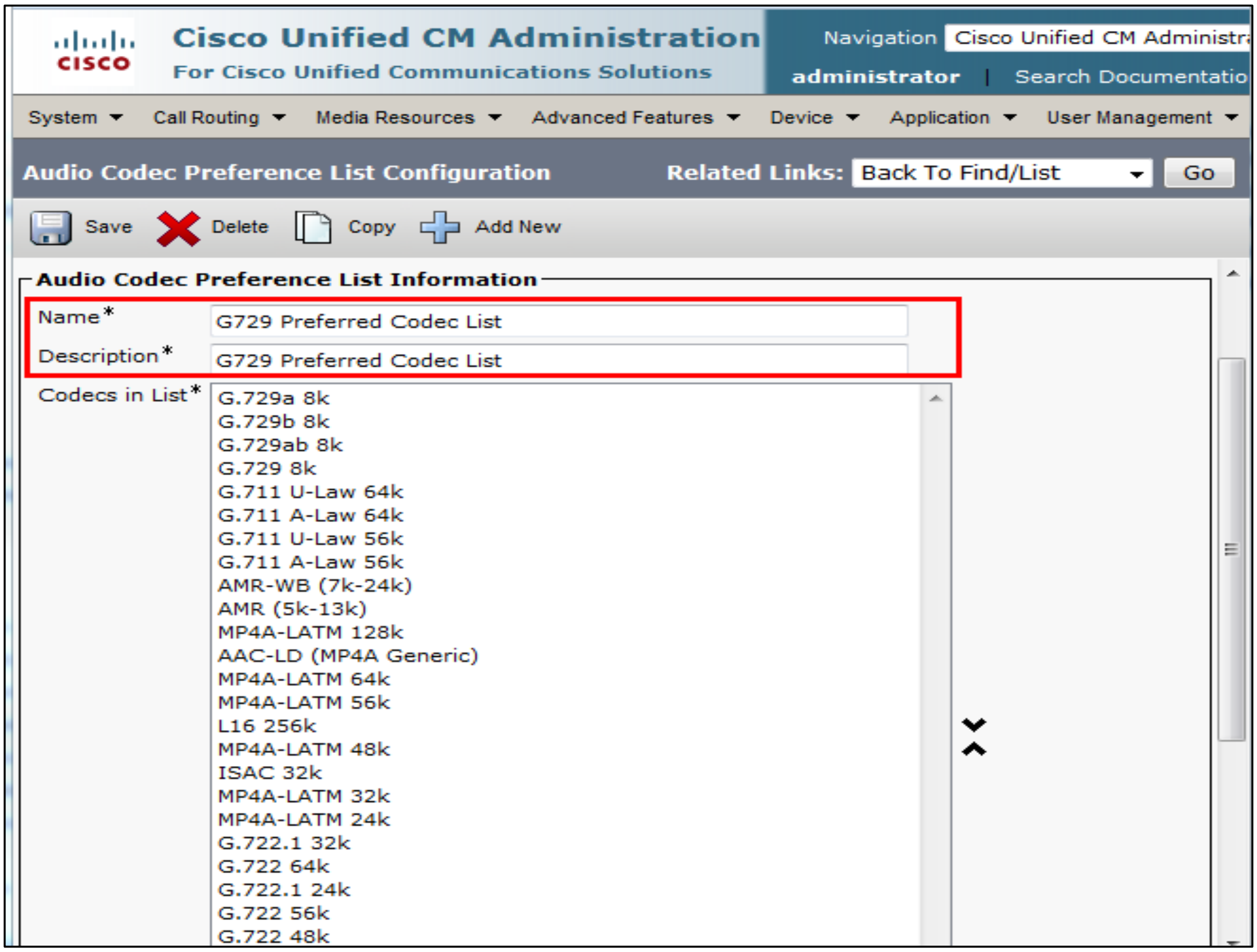
Cisco UCM Version

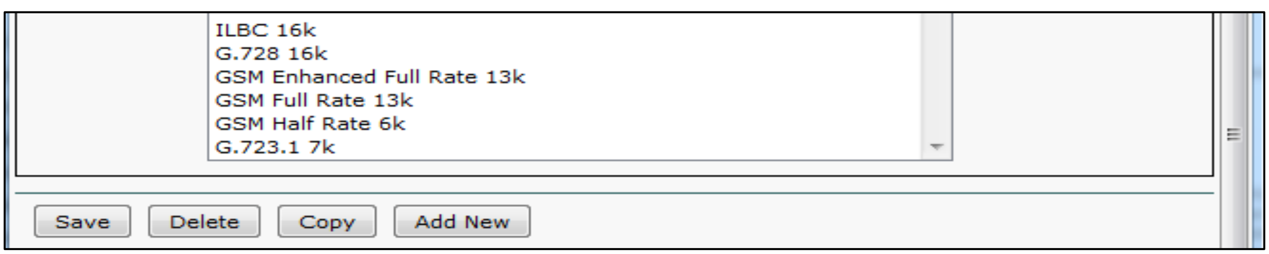


Cisco UCM Audio Codec Preference List

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

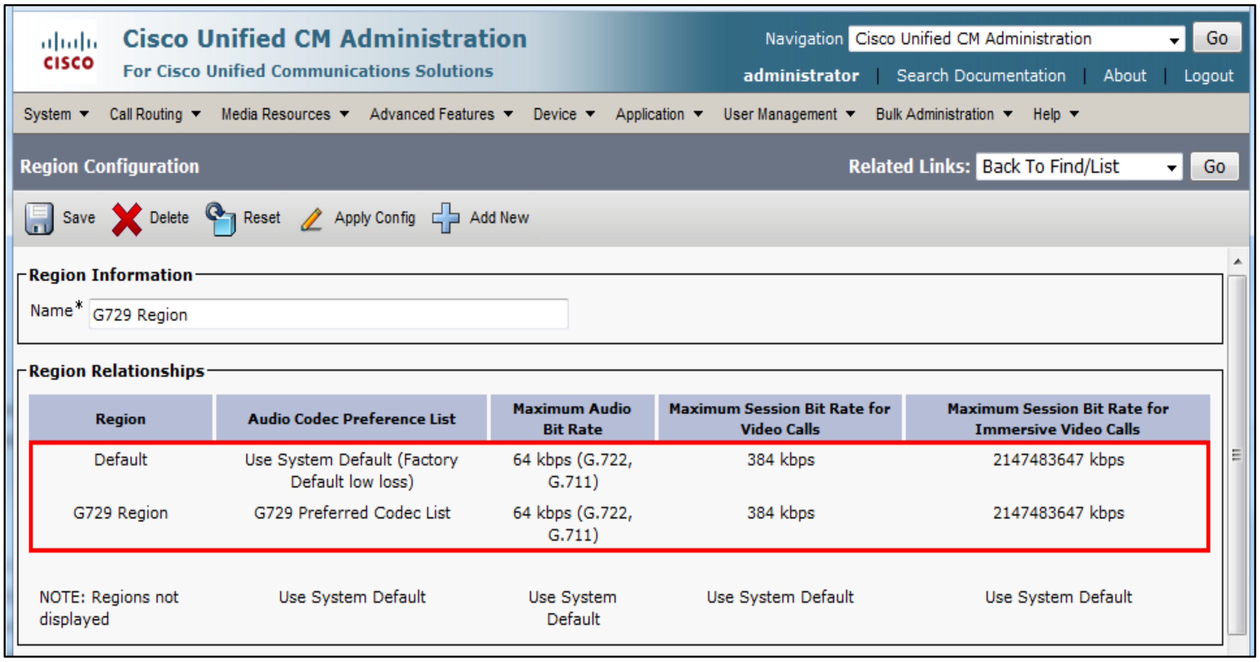
Cisco UCM 10.5.2 has a feature called Audio Codec Preference List. This feature allows to configure the order of audio codec preference both for Inter and Intra Region calls. Audio Codec Preference list is assigned to the Region used by the Device Pool for Phones and by Conference Bridges. Based on user requirement, different codec regions can be assigned as their first choice codec with this configuration for inbound calls as well as conferences initiated by Cisco IP phones. Audio codec preference for outbound calls is determined by Cisco UBE (via configuration of voice-class codec)

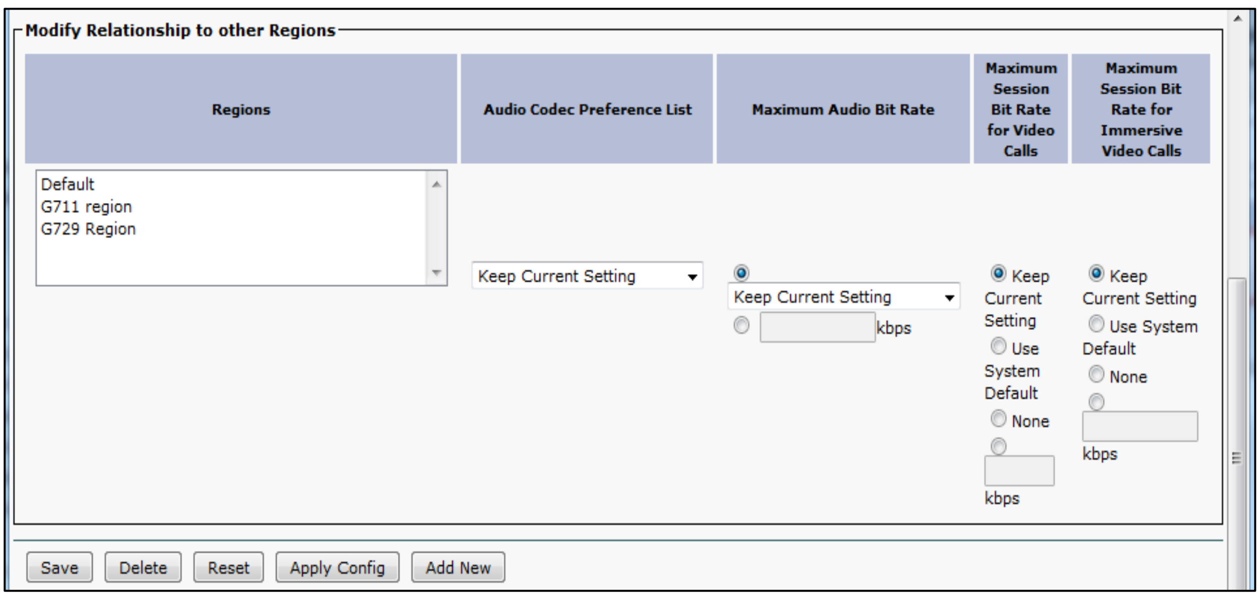




Cisco UCM Region Configuration

Navigation Path: System → Region Information → Region

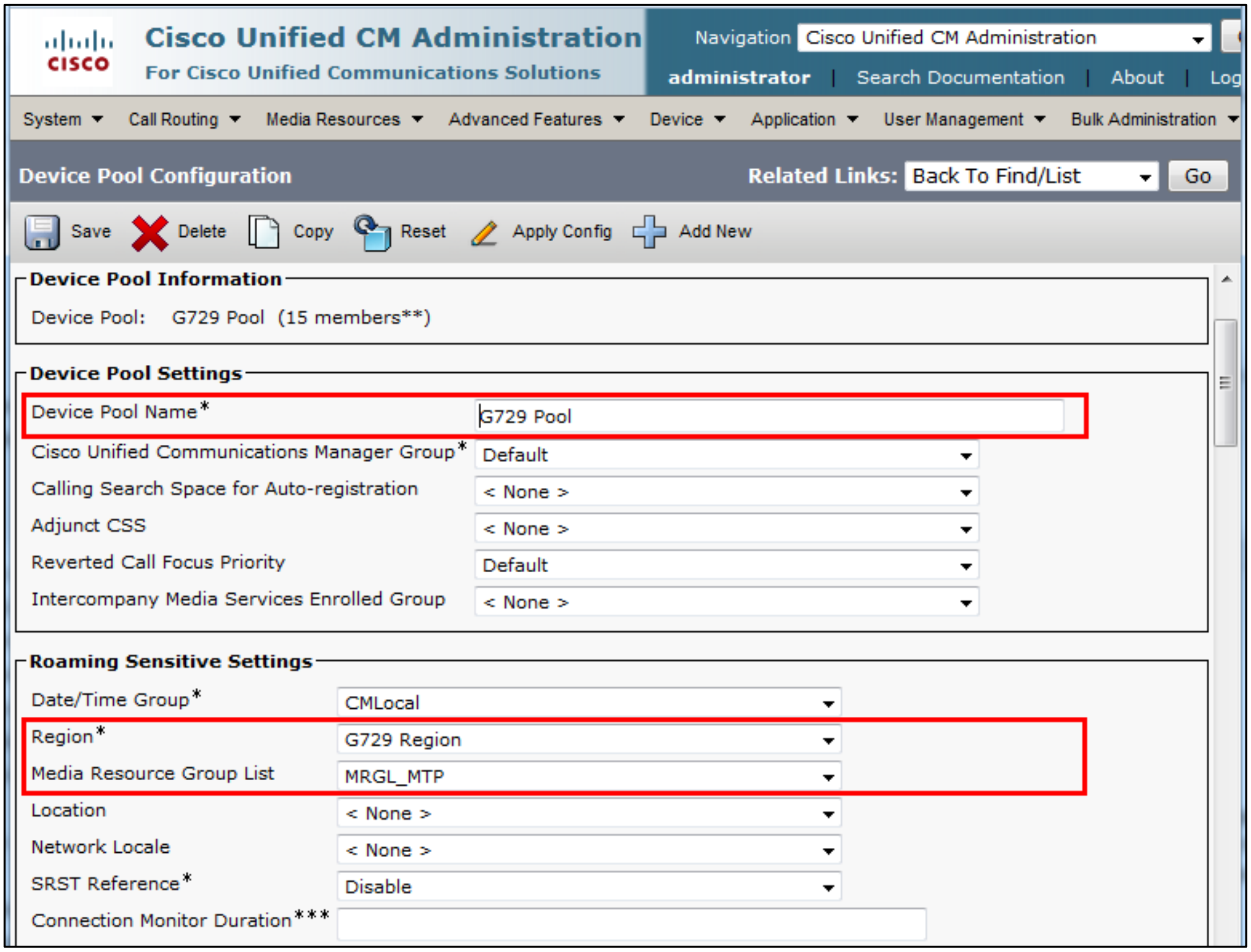


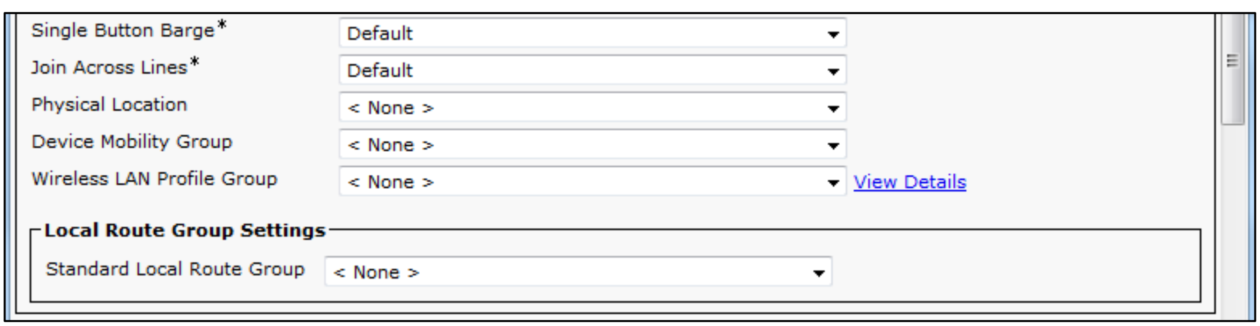


Device Pool Configuration

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…)

Device Mobility Related Information****

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

Geolocation Configuration

Geolocation	< None >
Geolocation Filter	< None >

Call Routing Information

Incoming Calling Party Settings

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

Clear Prefix Settings | Default Prefix Settings

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

Incoming Called Party Settings

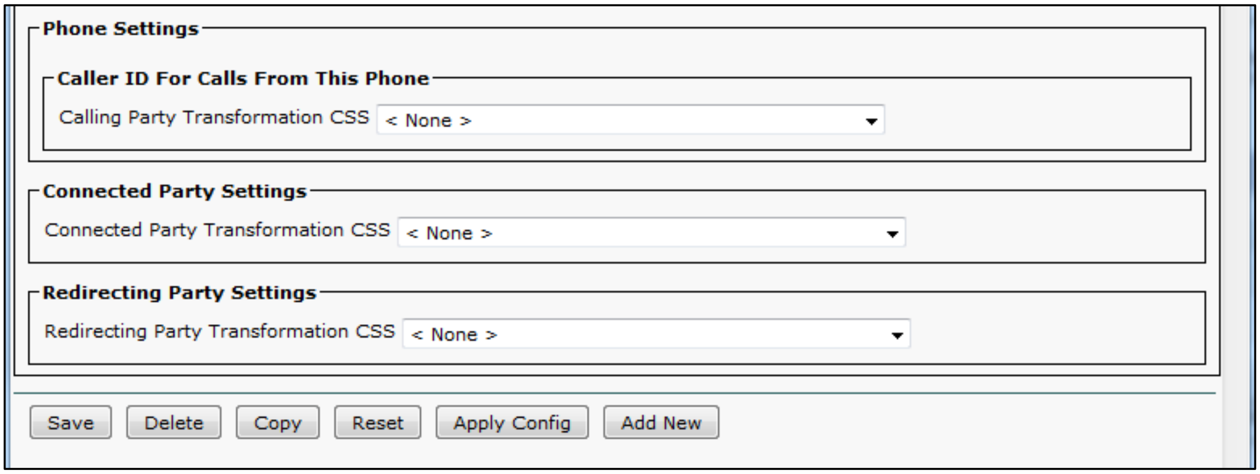
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

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

Note: Testing was conducted in tekVizion labs

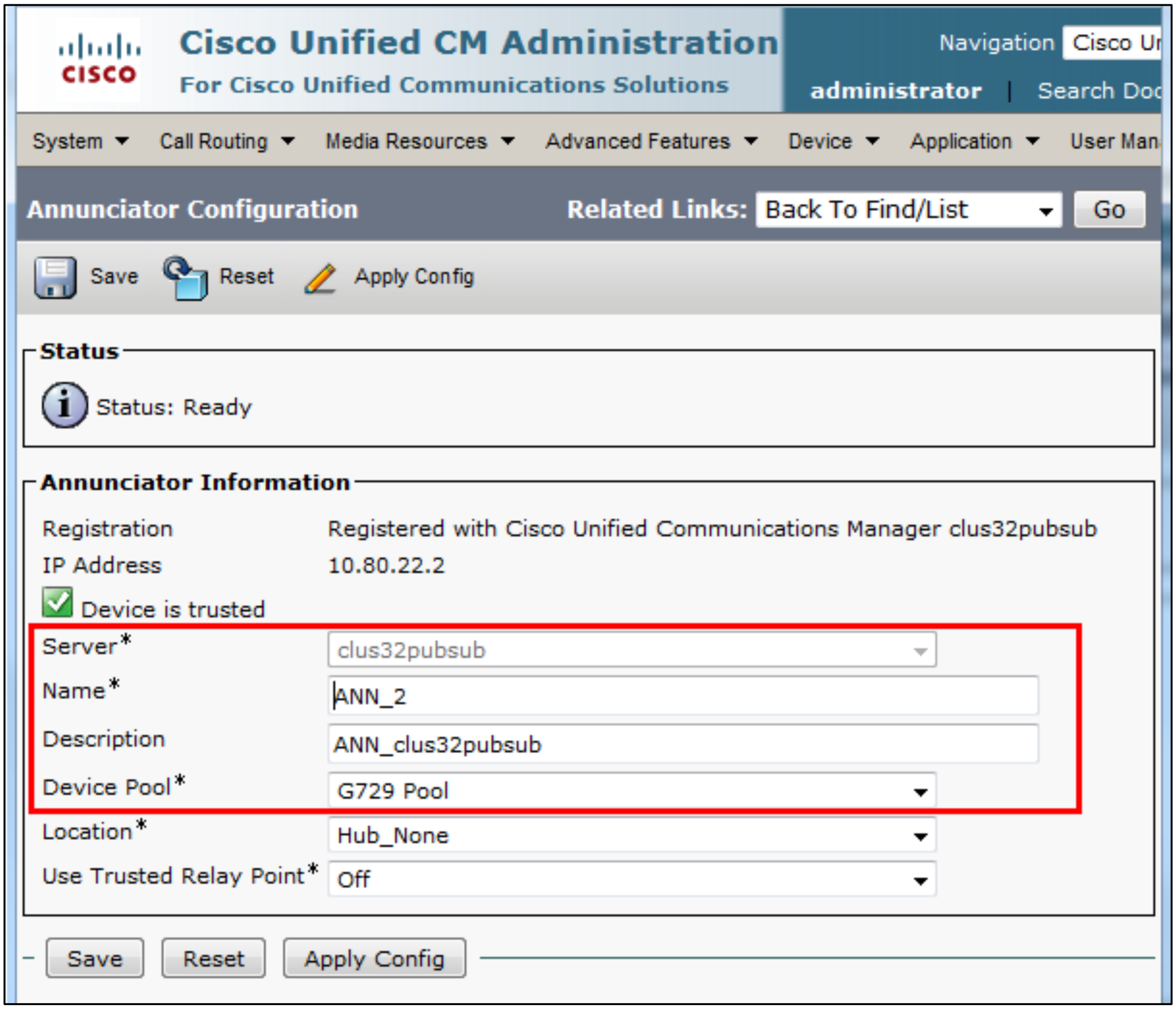
Device Pool Configuration (continued…)



Annunciator Configuration

Navigation: Media Resource → Annunciator

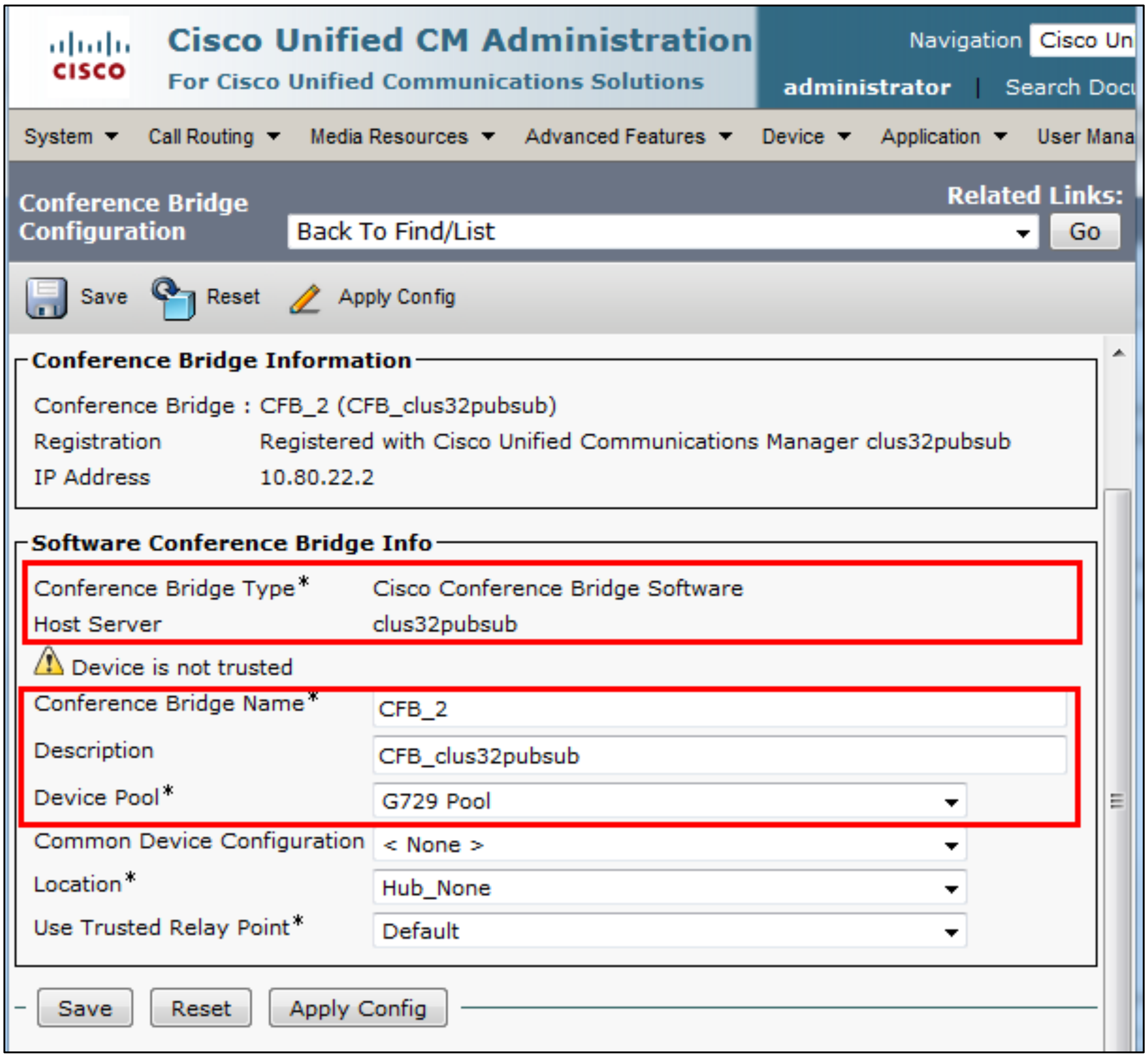
Set Name* = ANN_2.
Set Description = ANN_clus32pubsub. This is used for this example
Set Device Pool* = G729_pool.



Conference Bridge Configuration

Navigation: Media Resources → Conference Bridge

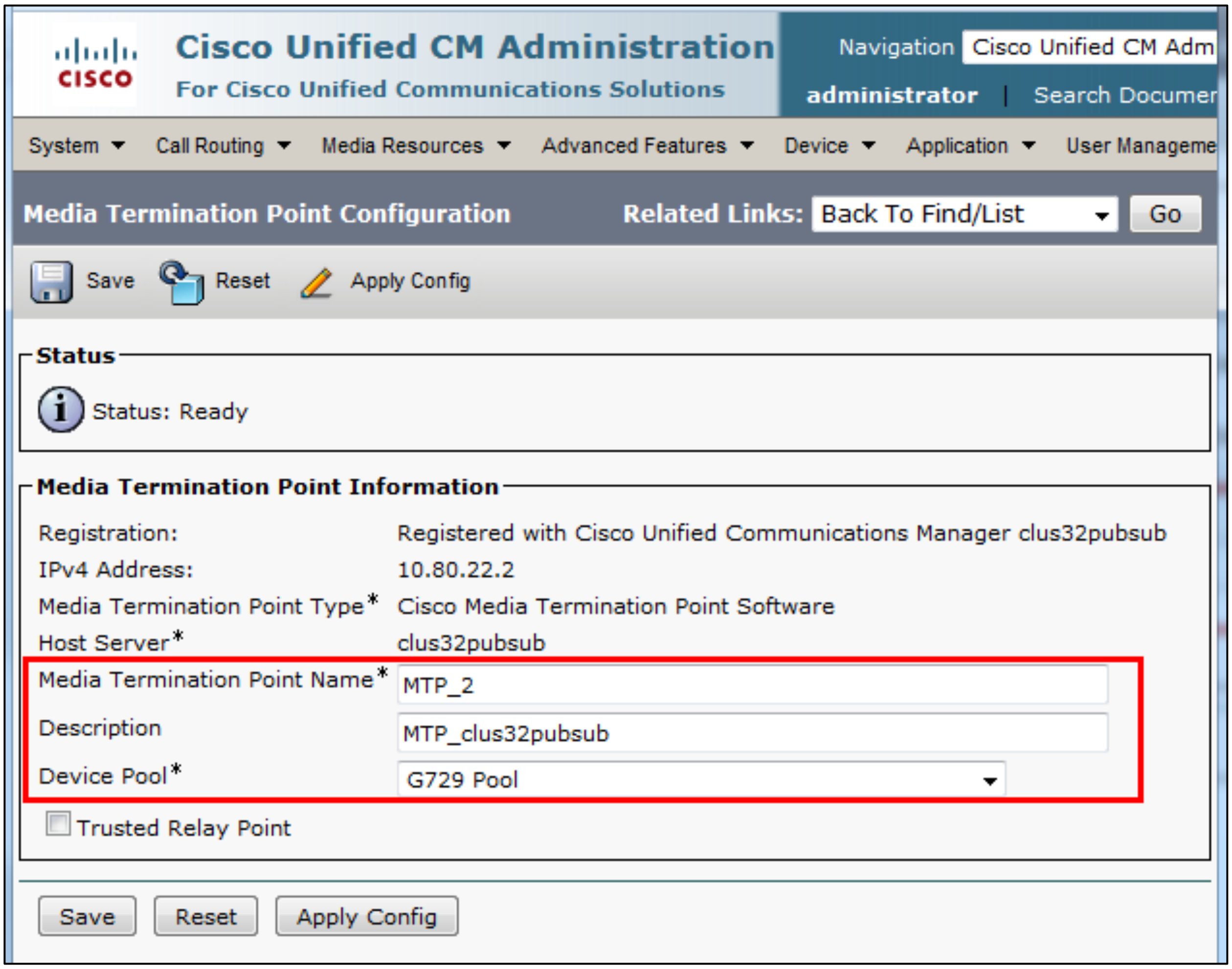
Set Conference Bridge Type* = Cisco Conference Bridge Software.
Set Host Server = clus32pubsub. This is used for this example.
Set Conference Bridge Name* = CFB_2.
Set Description = CFB_clus32pubsub. This is used in this example.
Set Device Pool* = G729_pool.



Media Termination Point Configuration

Navigation: Media Resource→Media Termination Point

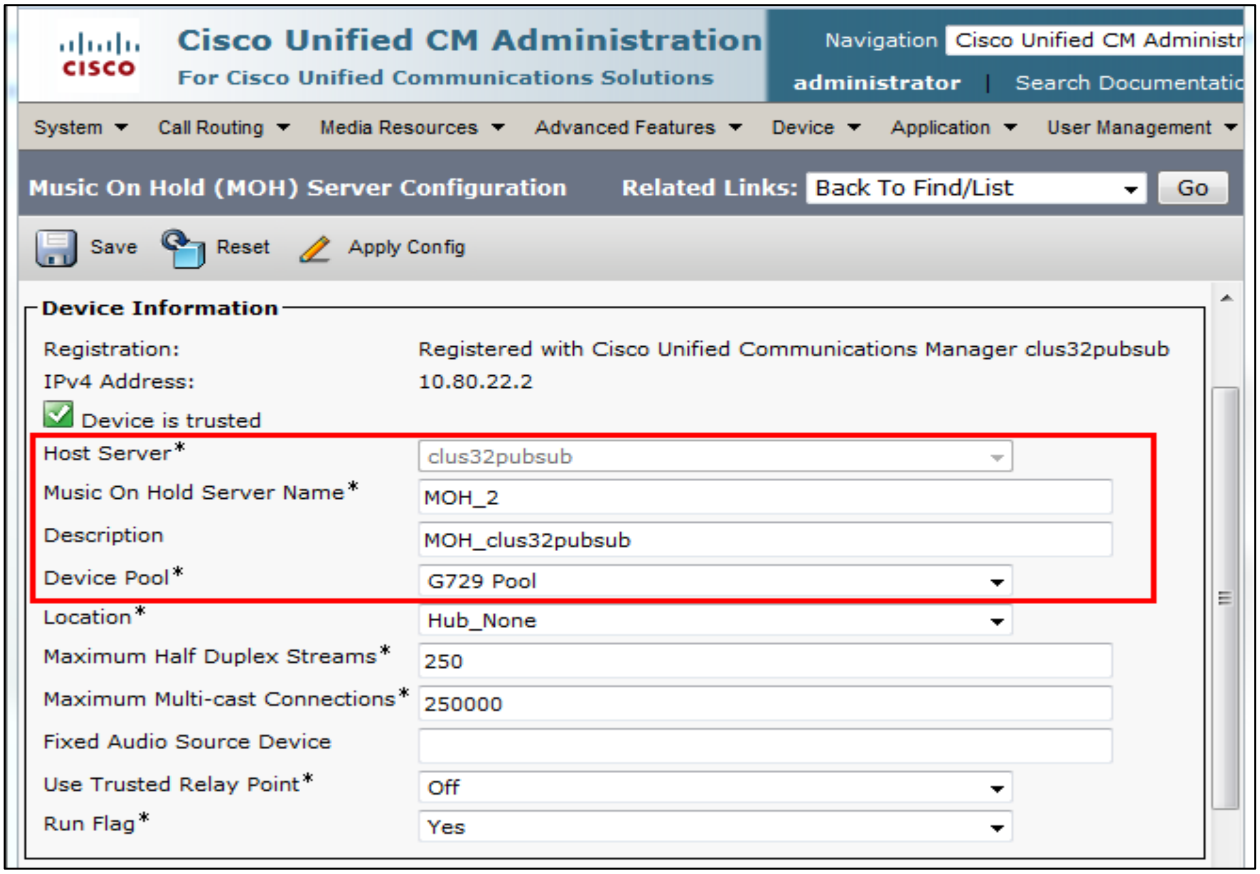
Set Media Termination Point Name* = MTP_2
Set Description = MTP_clus32pubsub. This is used for this example
Set Device pool* = G729 Pool

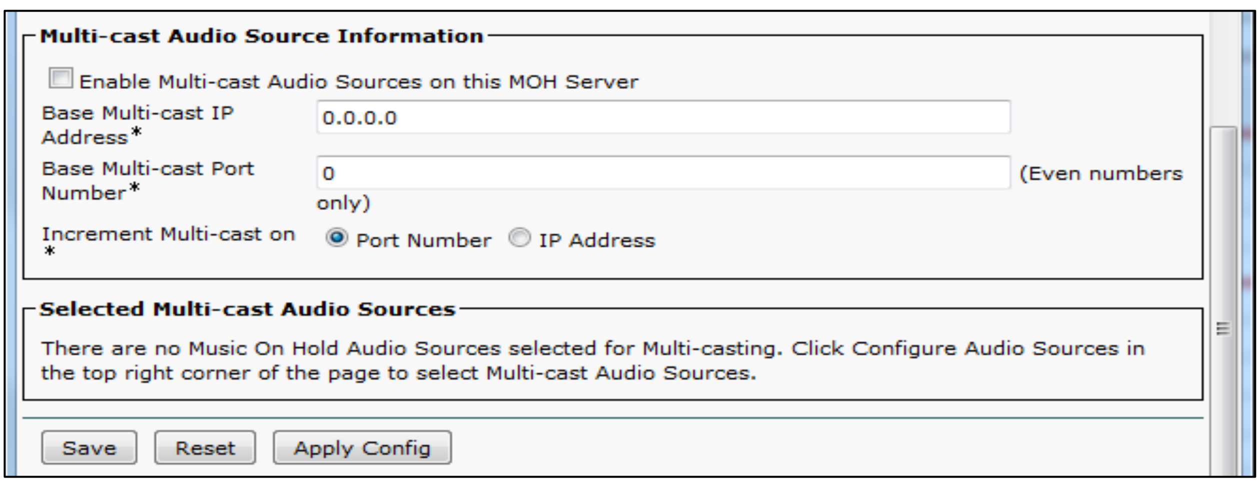


Music on Hold Server Configuration

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.



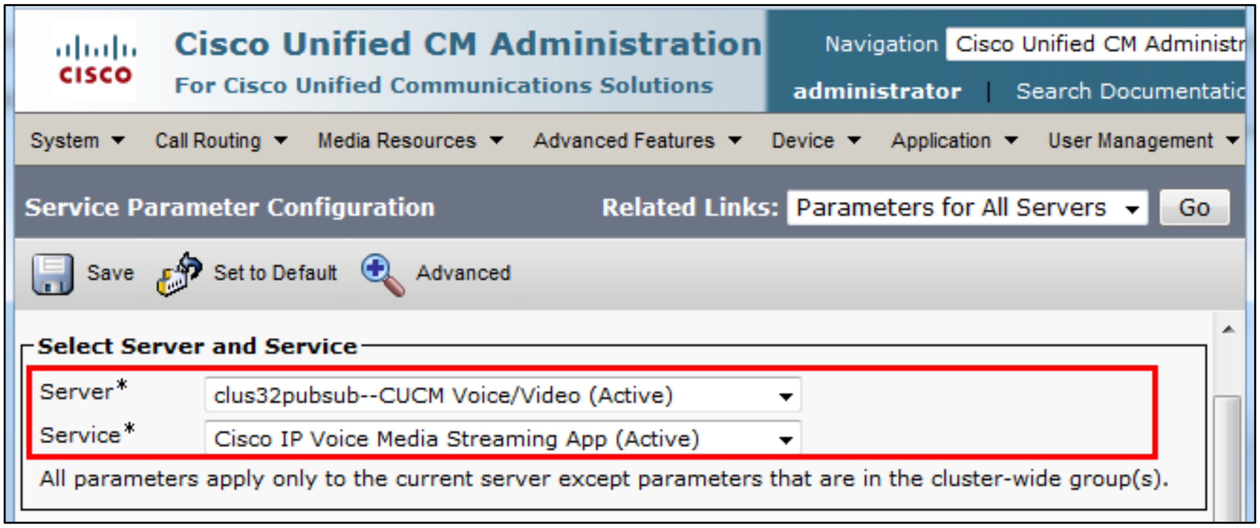


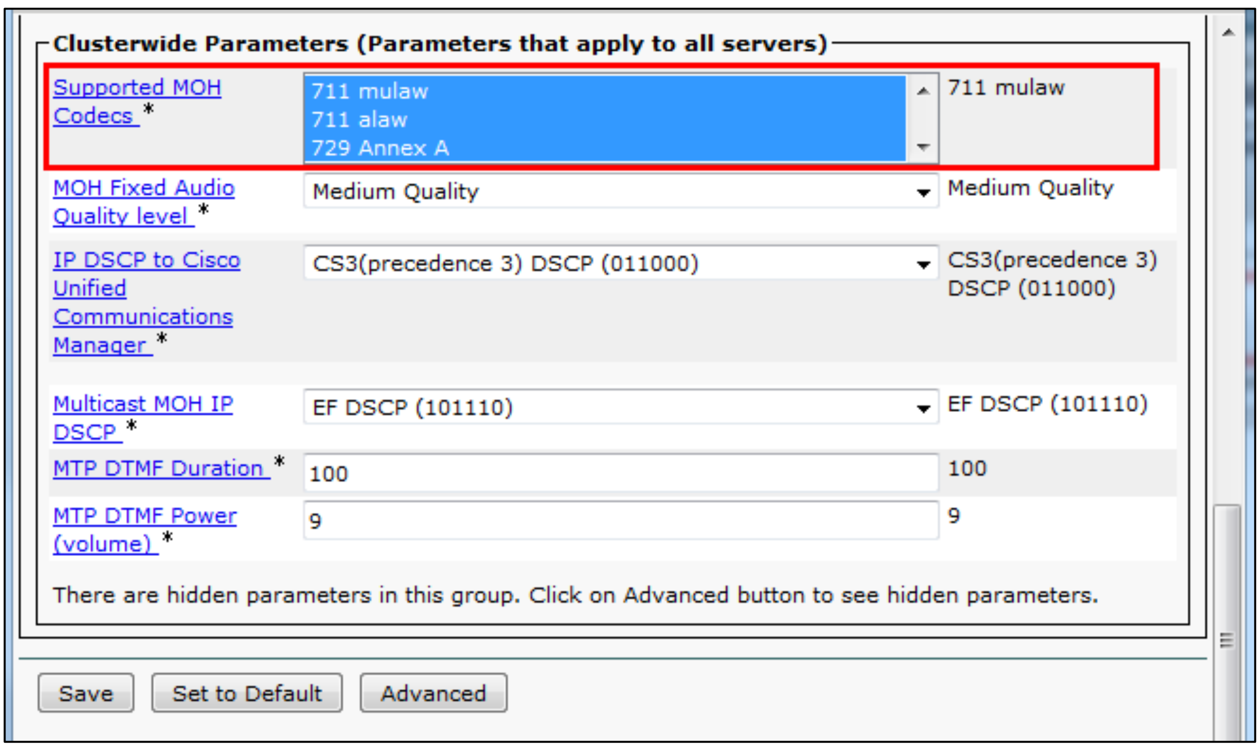
Music on Hold Service (IP Voice Media Streaming App) Parameter Settings

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).



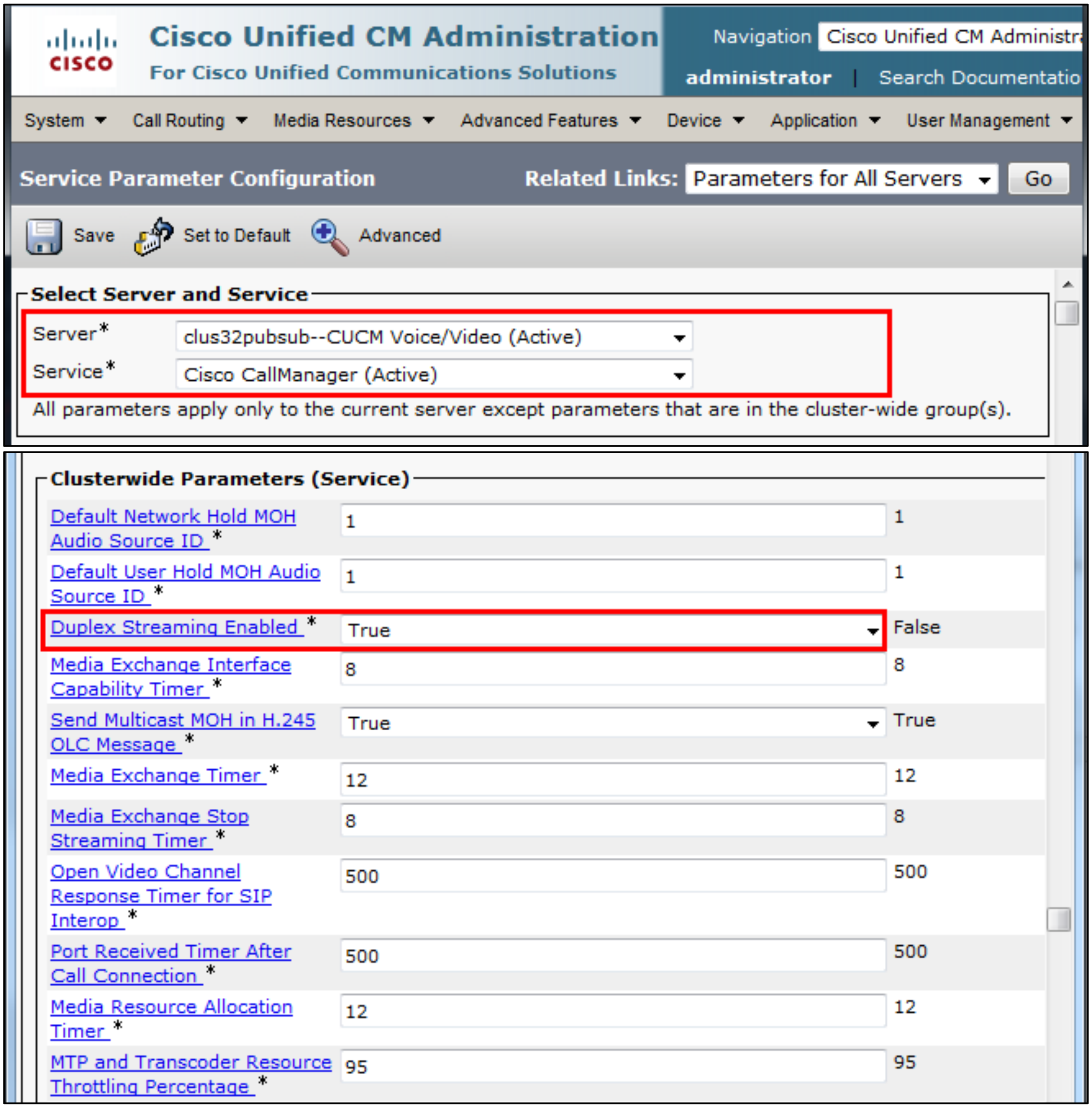


Music on Hold Service (Duplex Streaming) Parameter Settings

Navigation: System → Service Parameter

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

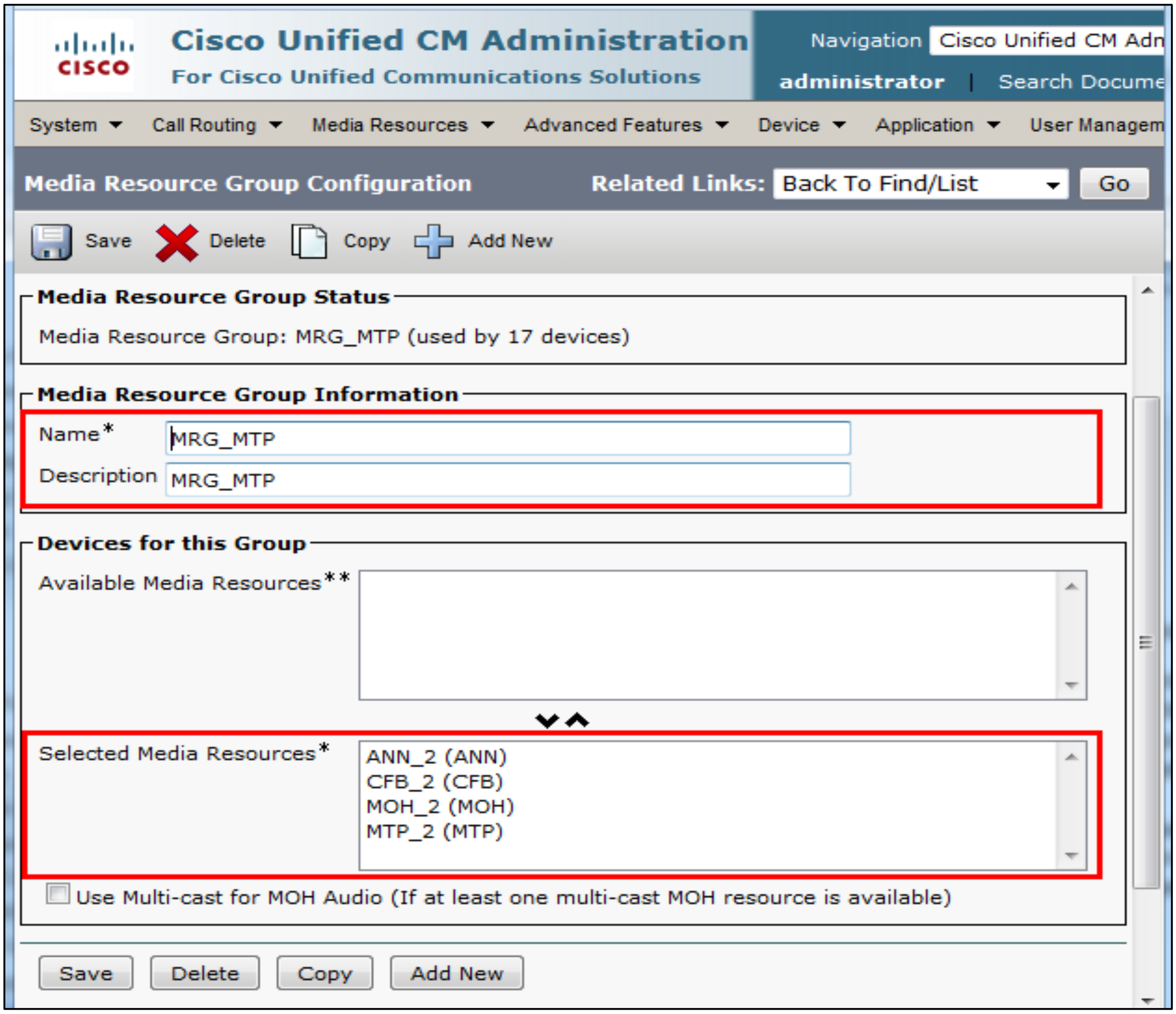
Select Service* = Cisco CallManager (Active).
Select Duplex Streaming Enabled * = True



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.

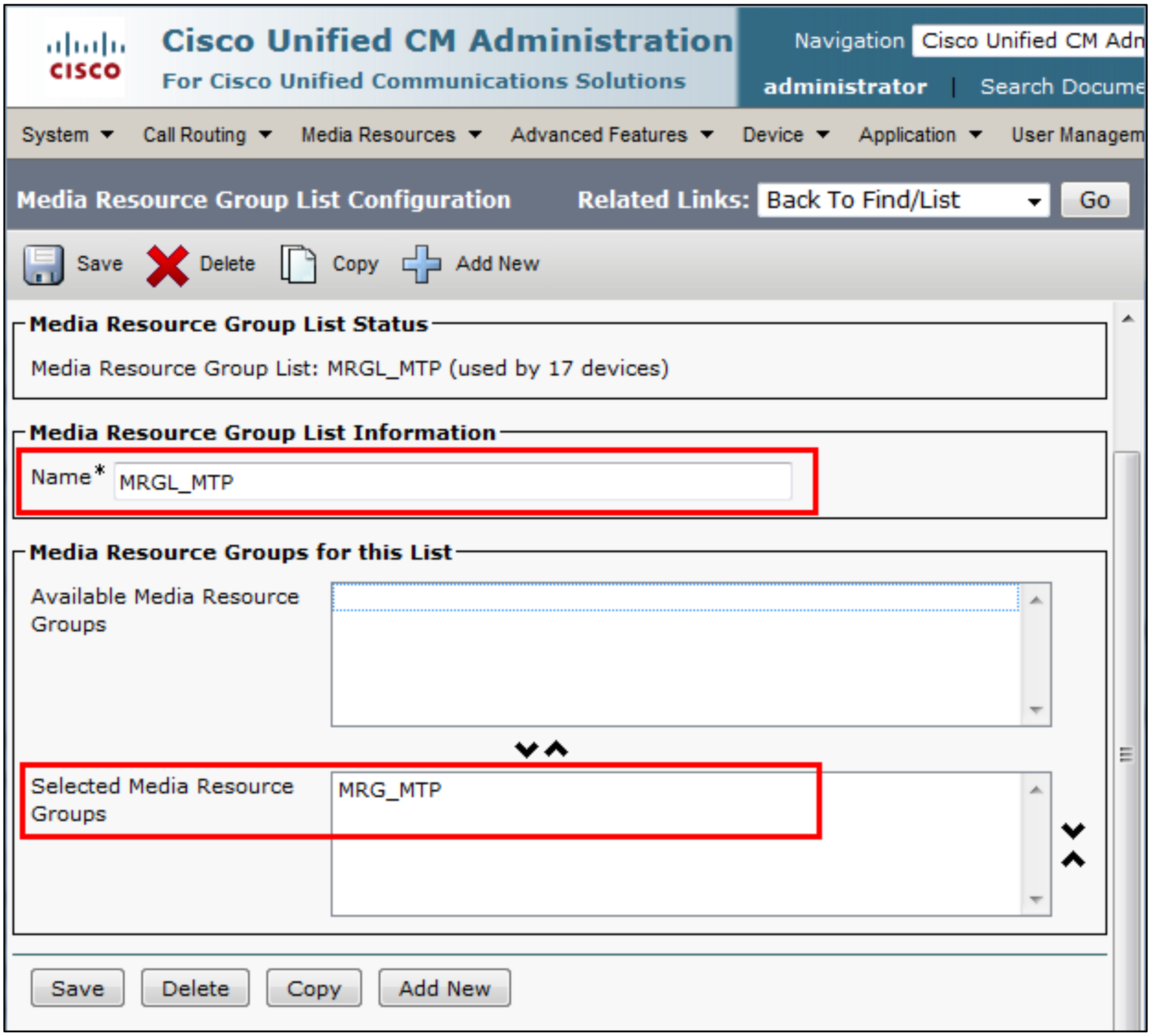


Media Resource Group List Configuration

Navigation Path: Media Resources → Media Resource Group List

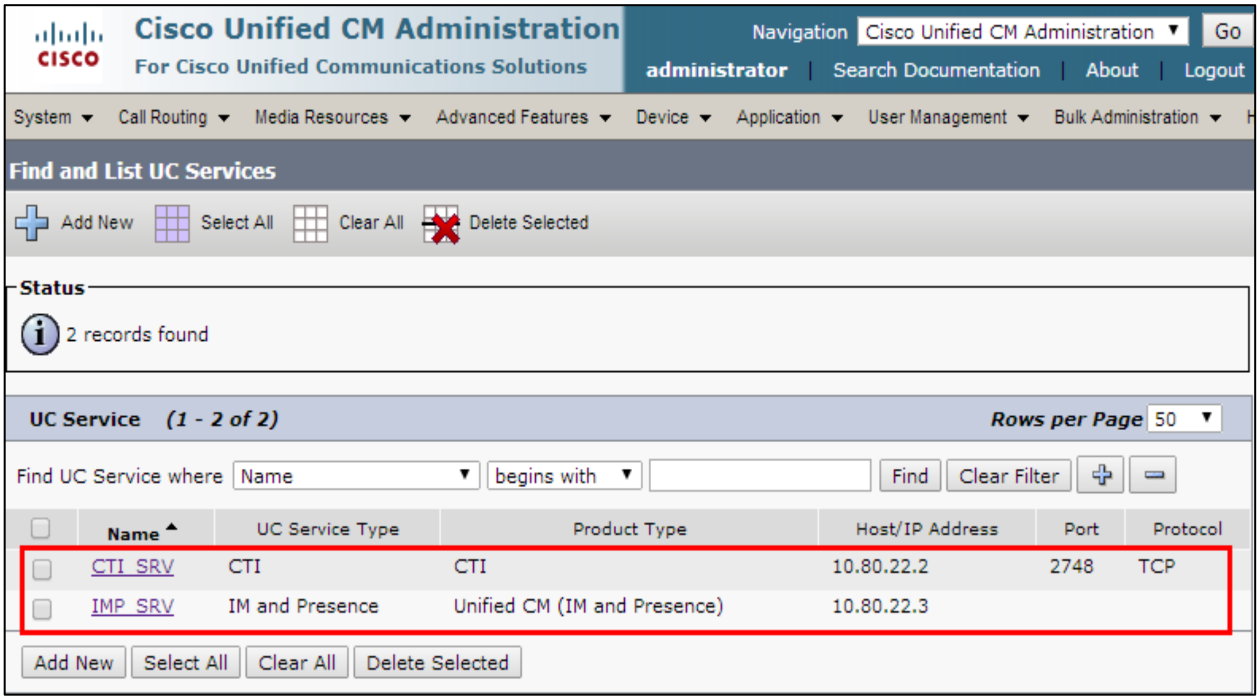
Set Name = MRGL_MTP.

Set selected Media Resource Groups = MRG_MTP.



UC Service Configuration

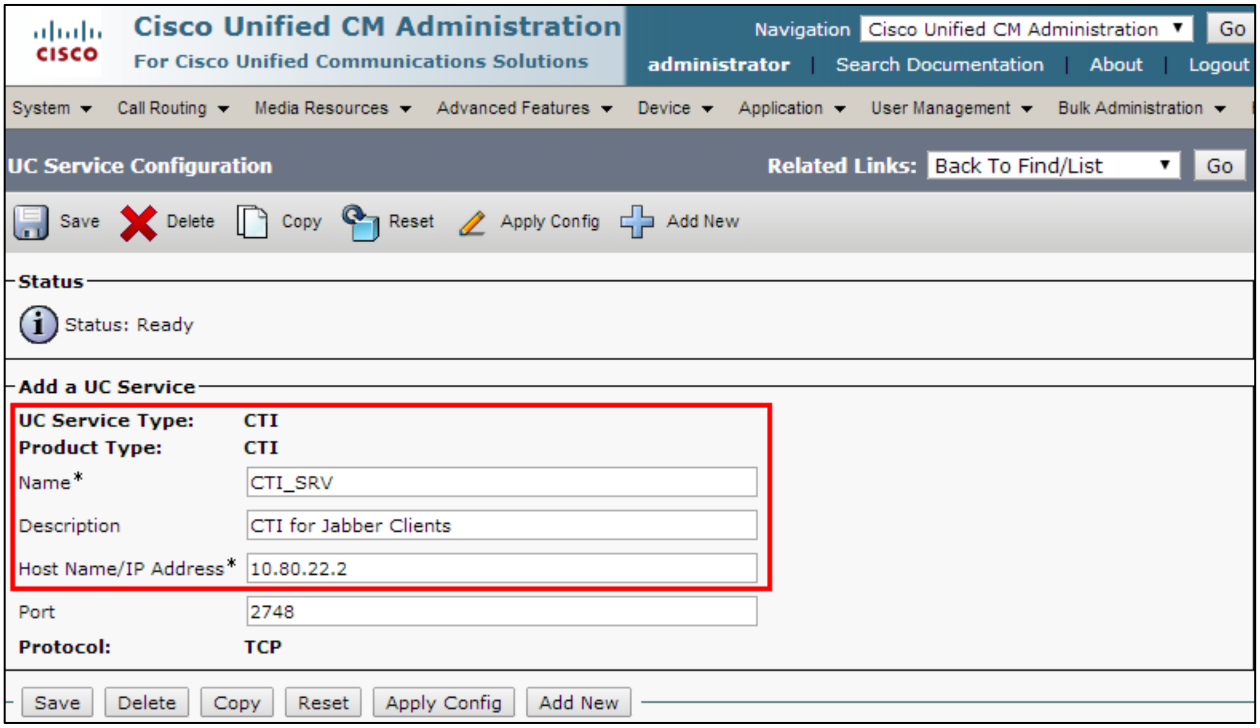
Navigation: User Management → User Settings → UC Service



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)

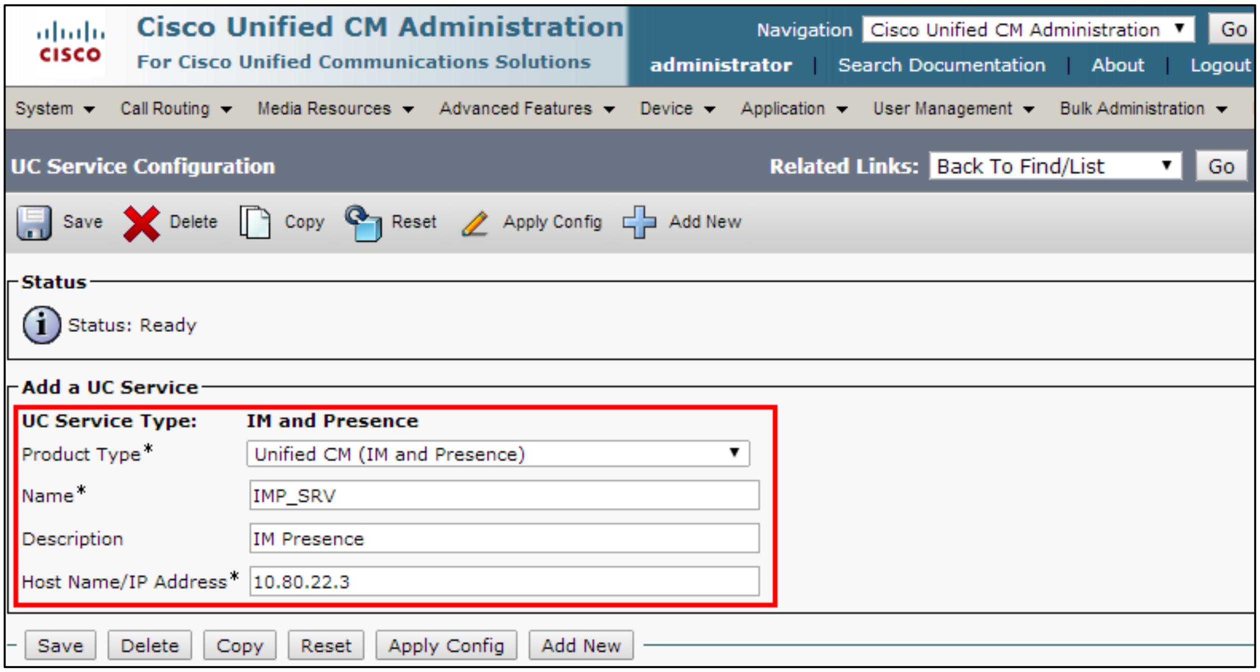


UC Service Configuration (Contd…)

Select UC Service Type: = IM and Presence
Set Name* = IMP_SRV. This is used in this example.

Note: Testing was conducted in tekVizion labs

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)



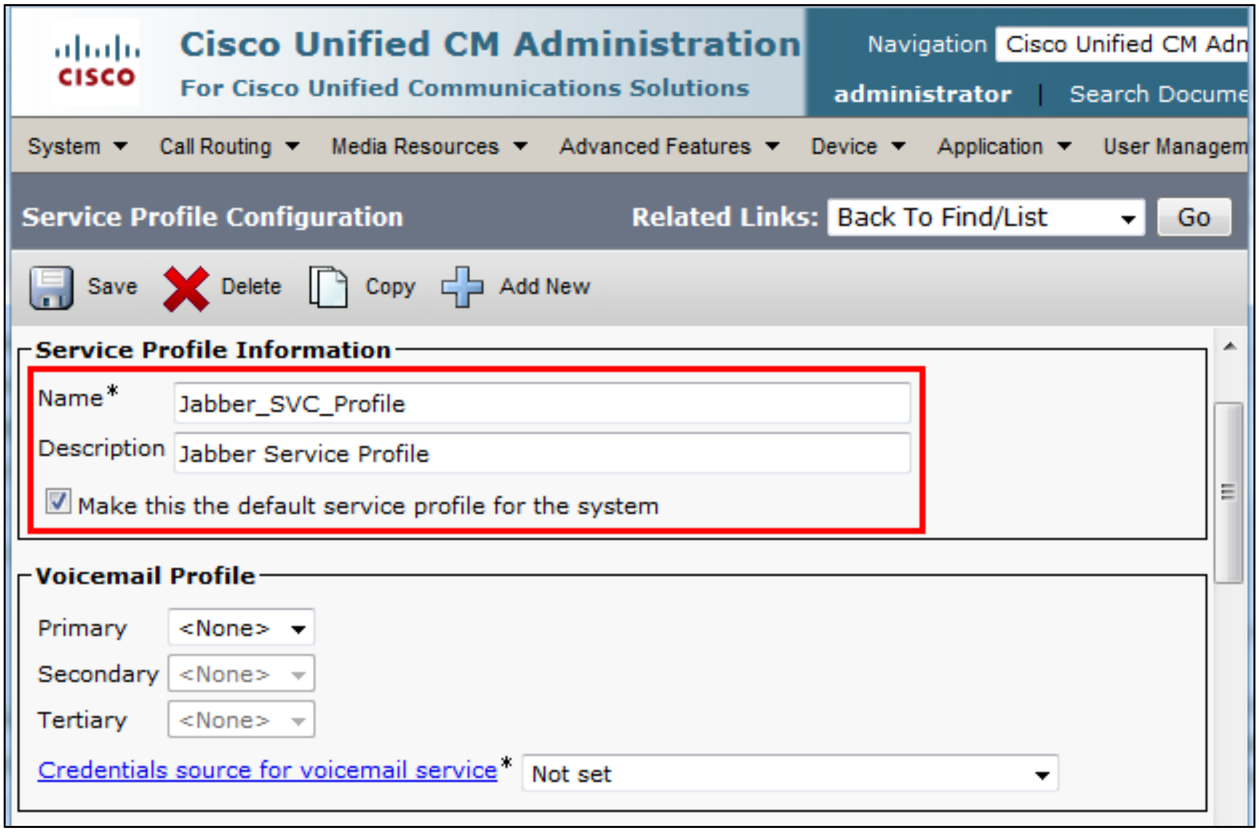
Service Profile Configuration

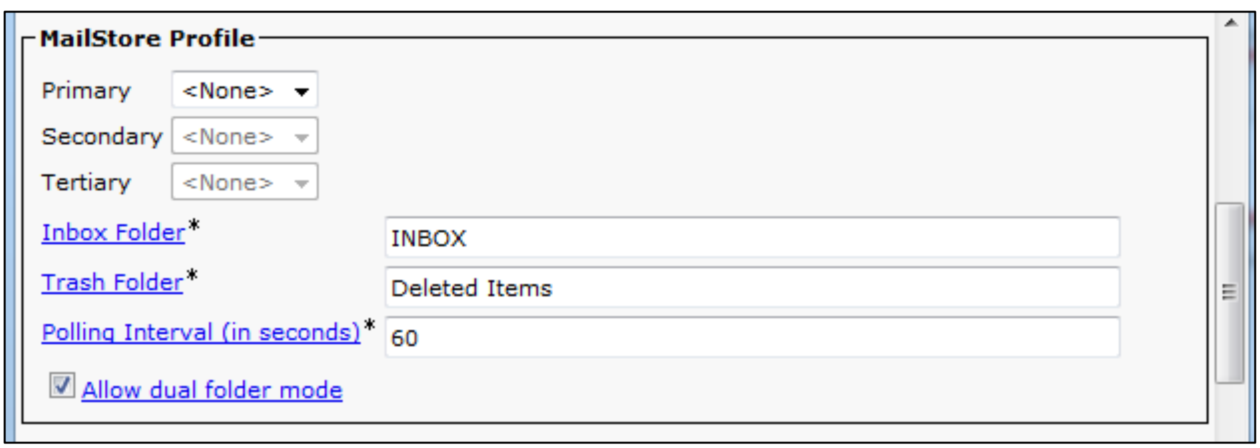
Navigation: User Management → User Settings → Service Profile

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

Note: Testing was conducted in tekVizion labs

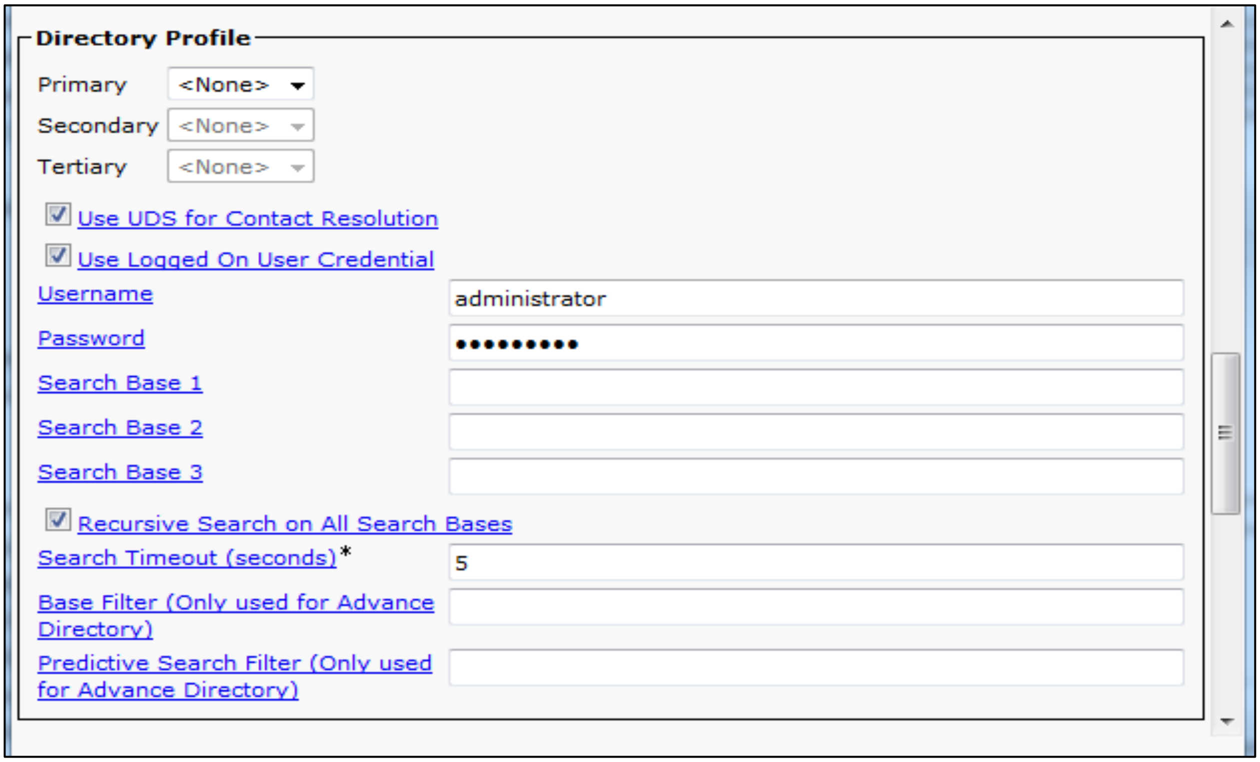
Set Description = Jabber Service Profile. This is used in this example.
Check - Make this the default service profile for the system.

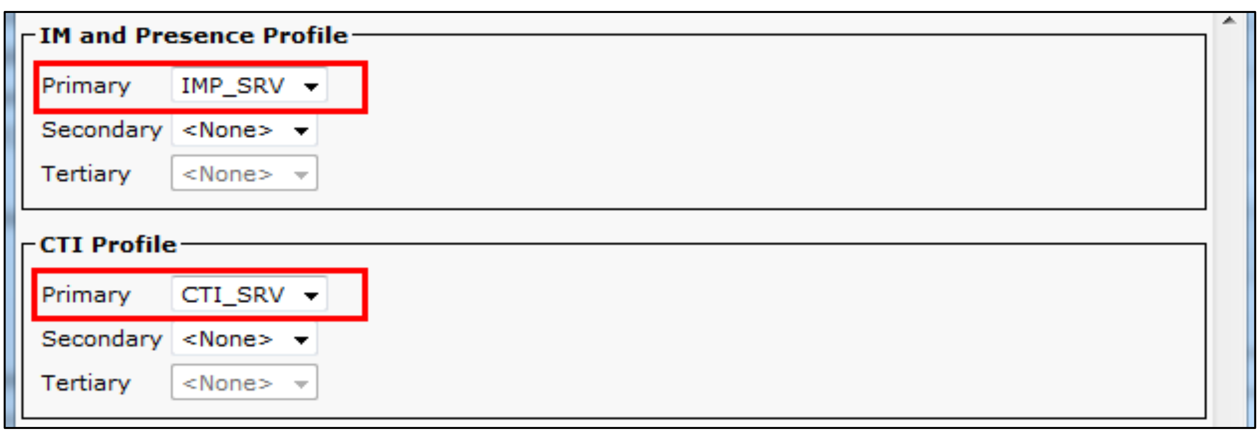




Service Profile Configuration (Contd…)


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

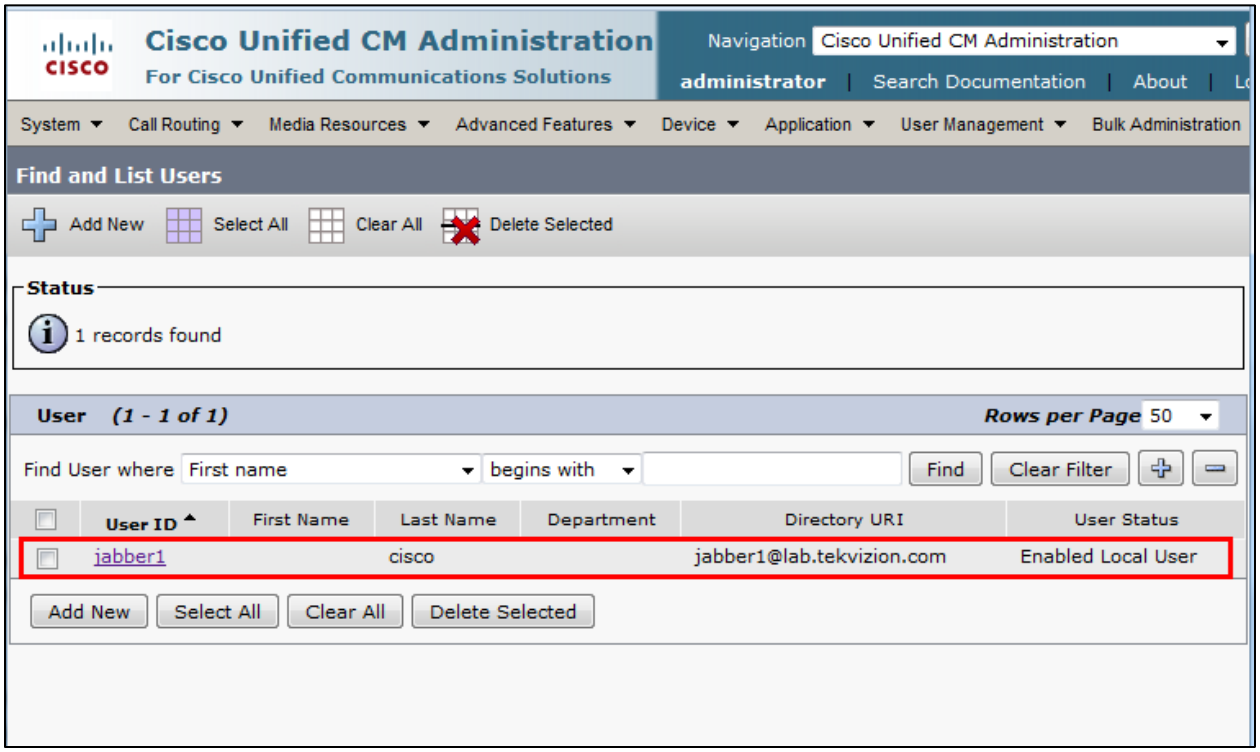

Directory Profile
Primary <None>
Secondary <None>
Tertiary <None>
Use UDS for Contact Resolution
Use Logged On User Credential
Username administrator
Password •••••••••
Search Base 1
Search Base 2
Search Base 3
Recursive Search on All Search Bases
Search Timeout (seconds)* 5
Base Filter (Only used for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

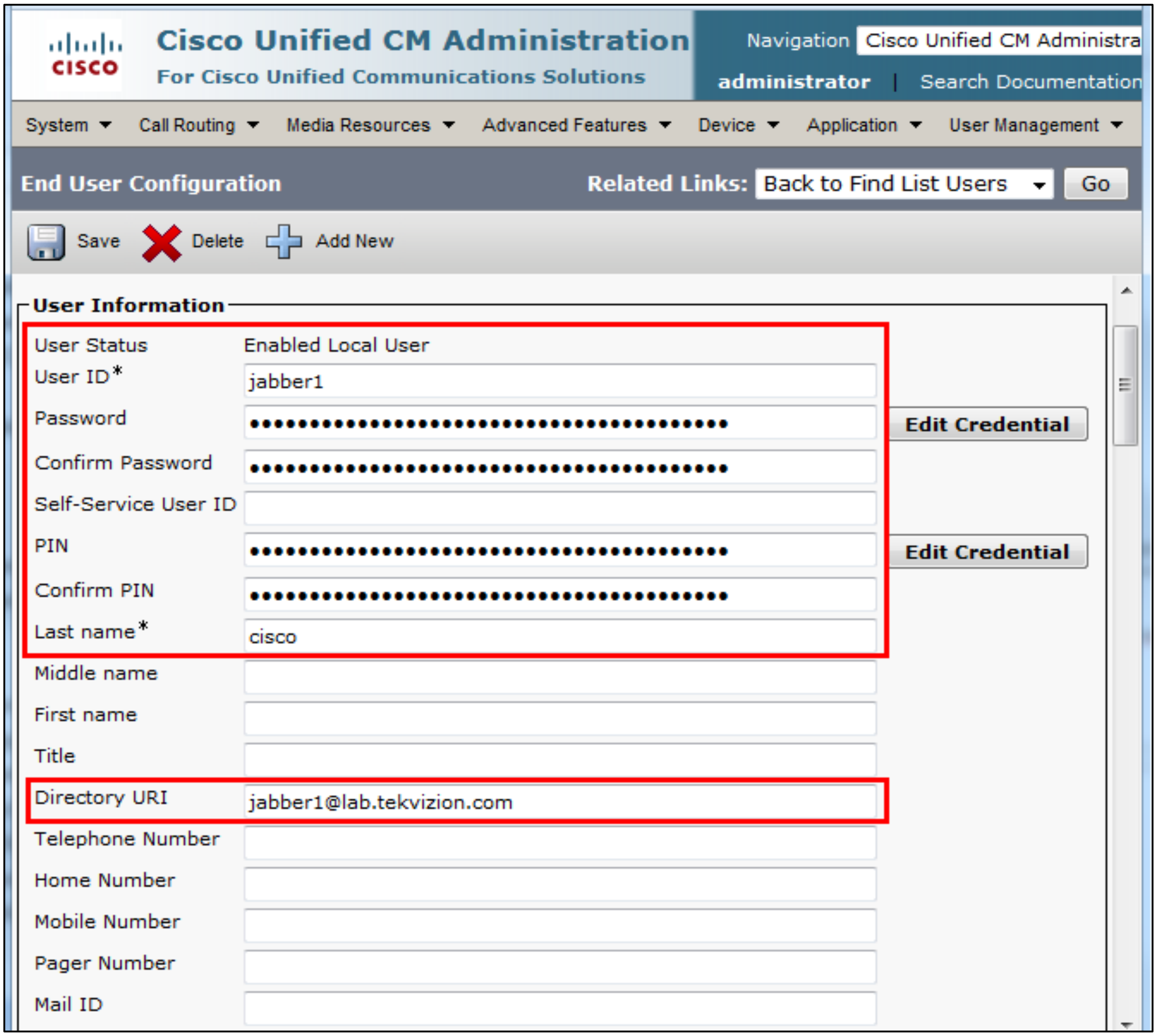
End User Configuration

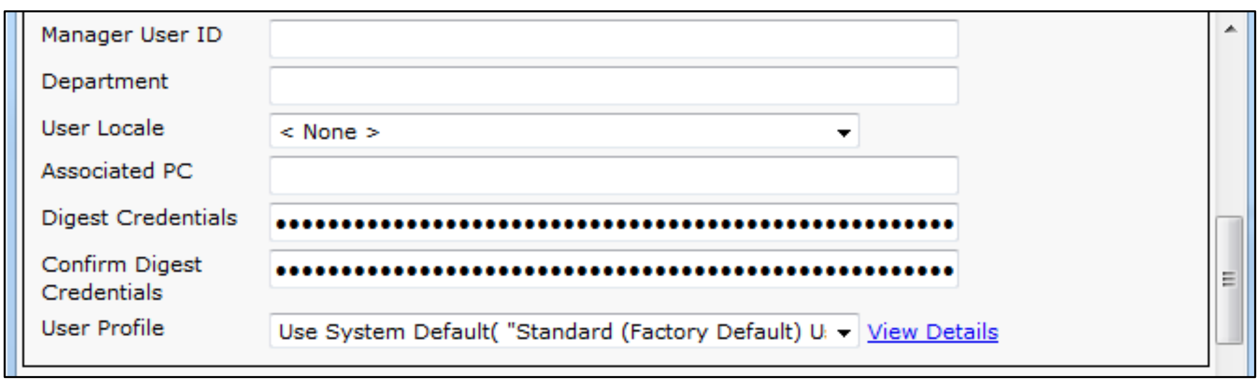
Navigation: User Management → End User

Set User ID* = jabber1 – This is used in this example.
Set Password = Password for profile.
Set Directory URI = jabber1@lab.tekvizion.com.

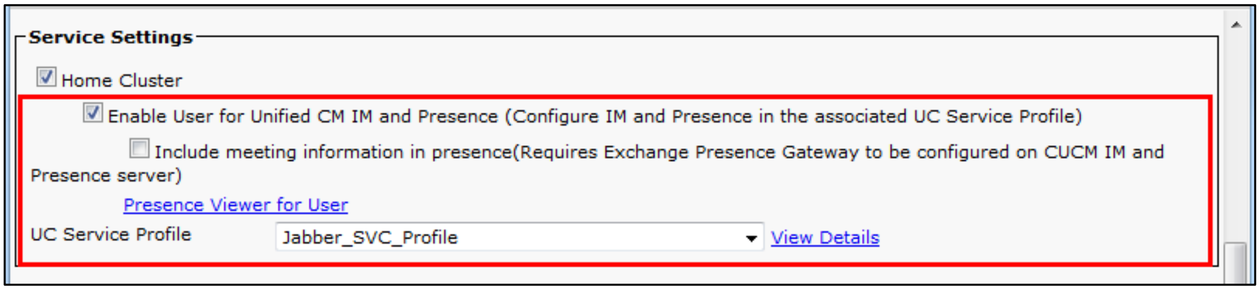


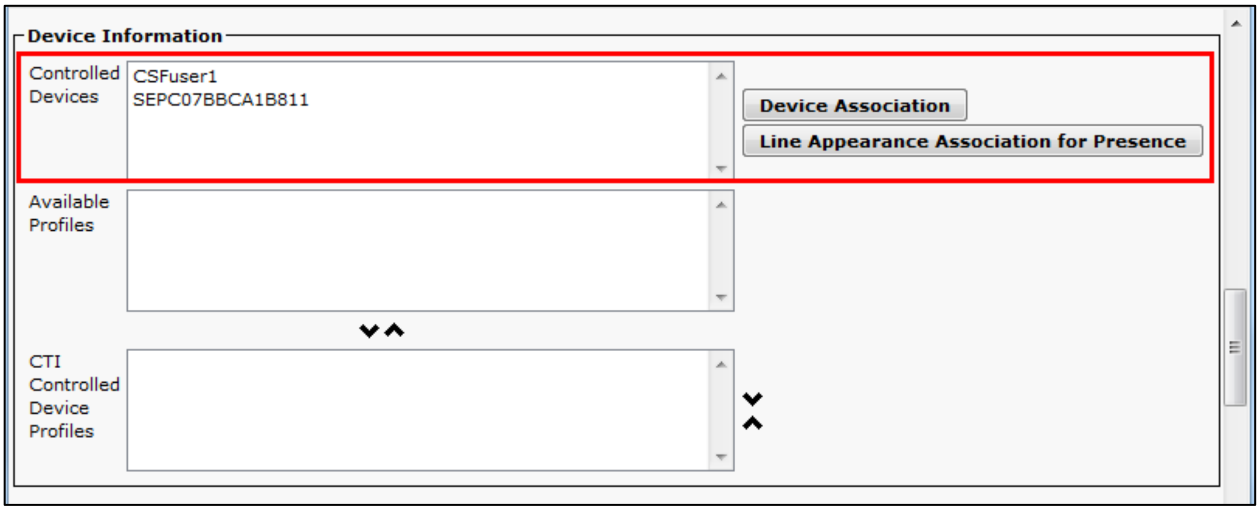
End User Configuration (continued…)

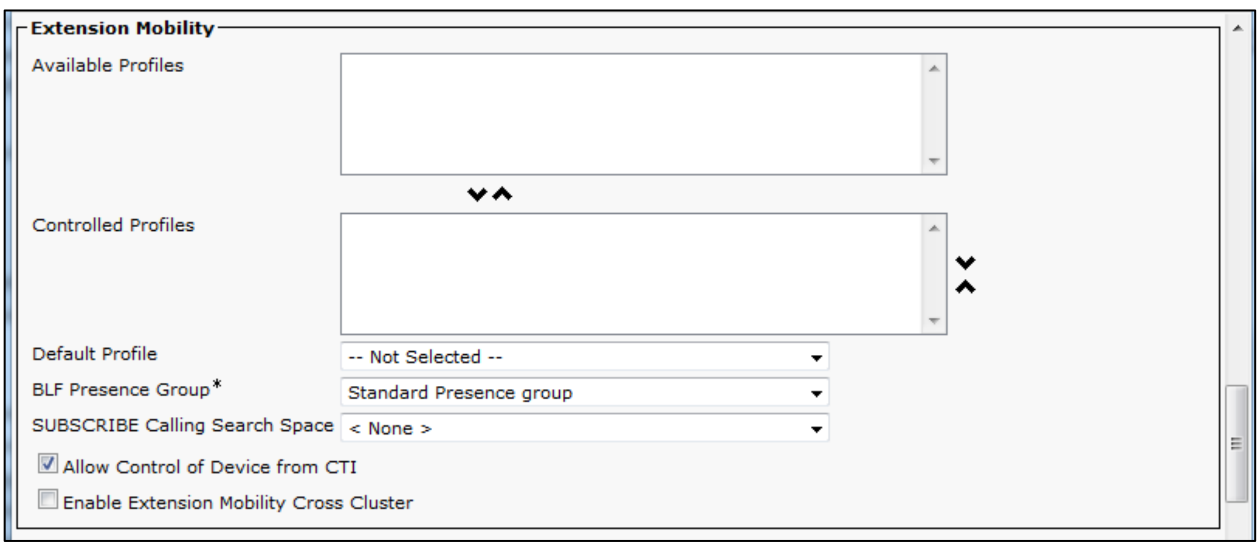

Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administra
administrator | Search Documentation
System Call Routing Media Resources Advanced Features Device Application User Management
End User Configuration
Related Links: Back to Find List Users Go
Save Delete Add New
User Information
User Status Enabled Local User
User ID* jabber1
Password
Edit Credential
Confirm Password
Self-Service User ID
PIN
Edit Credential
Confirm PIN
Last name* cisco
Middle name
First name
Title
Directory URI jabber1@lab.tekvizion.com
Telephone Number
Home Number
Mobile Number
Pager Number
Mail ID


Manager User ID
Department
User Locale < None >
Associated PC
Digest Credentials
Confirm Digest Credentials
User Profile Use System Default("Standard (Factory Default) U View Details

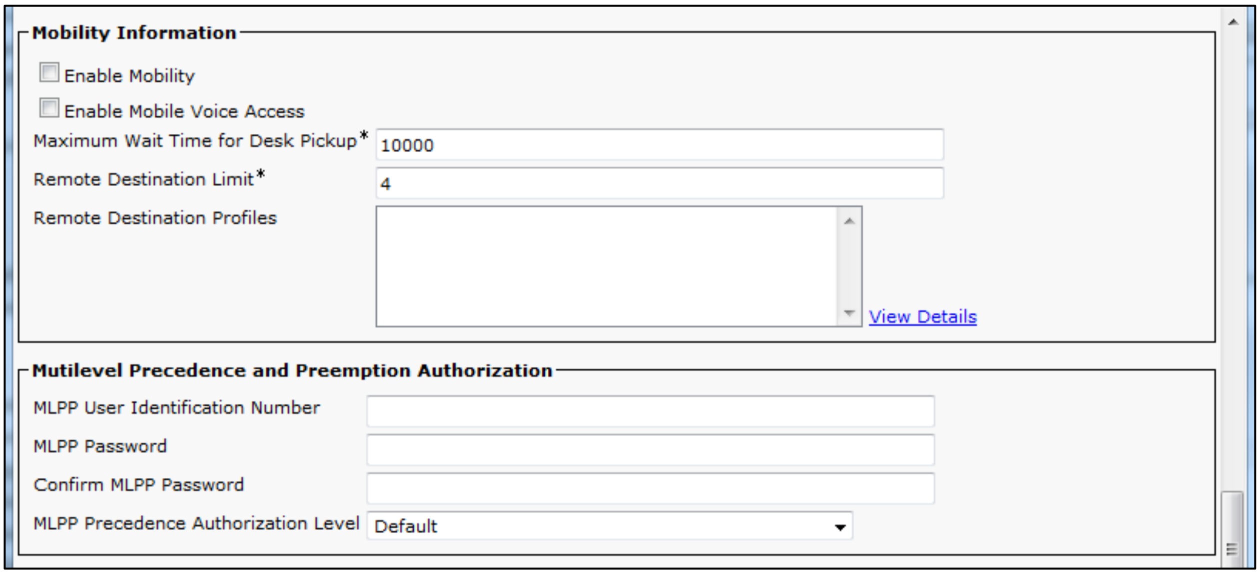
End User Configuration(continued…)

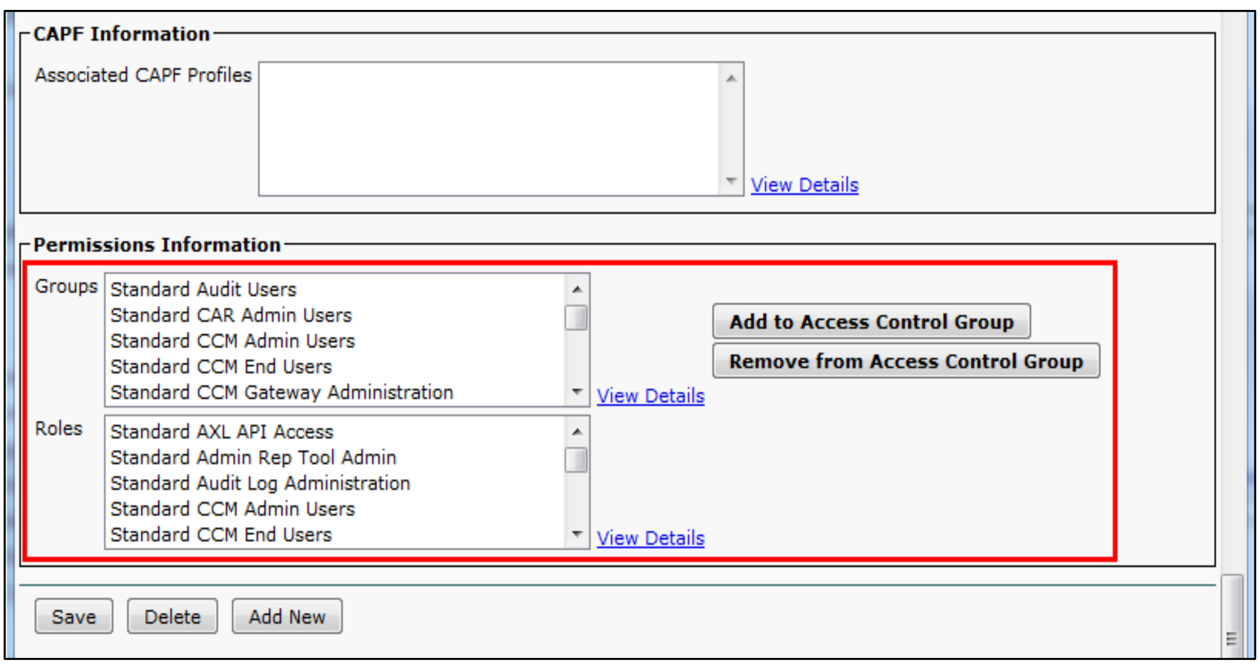






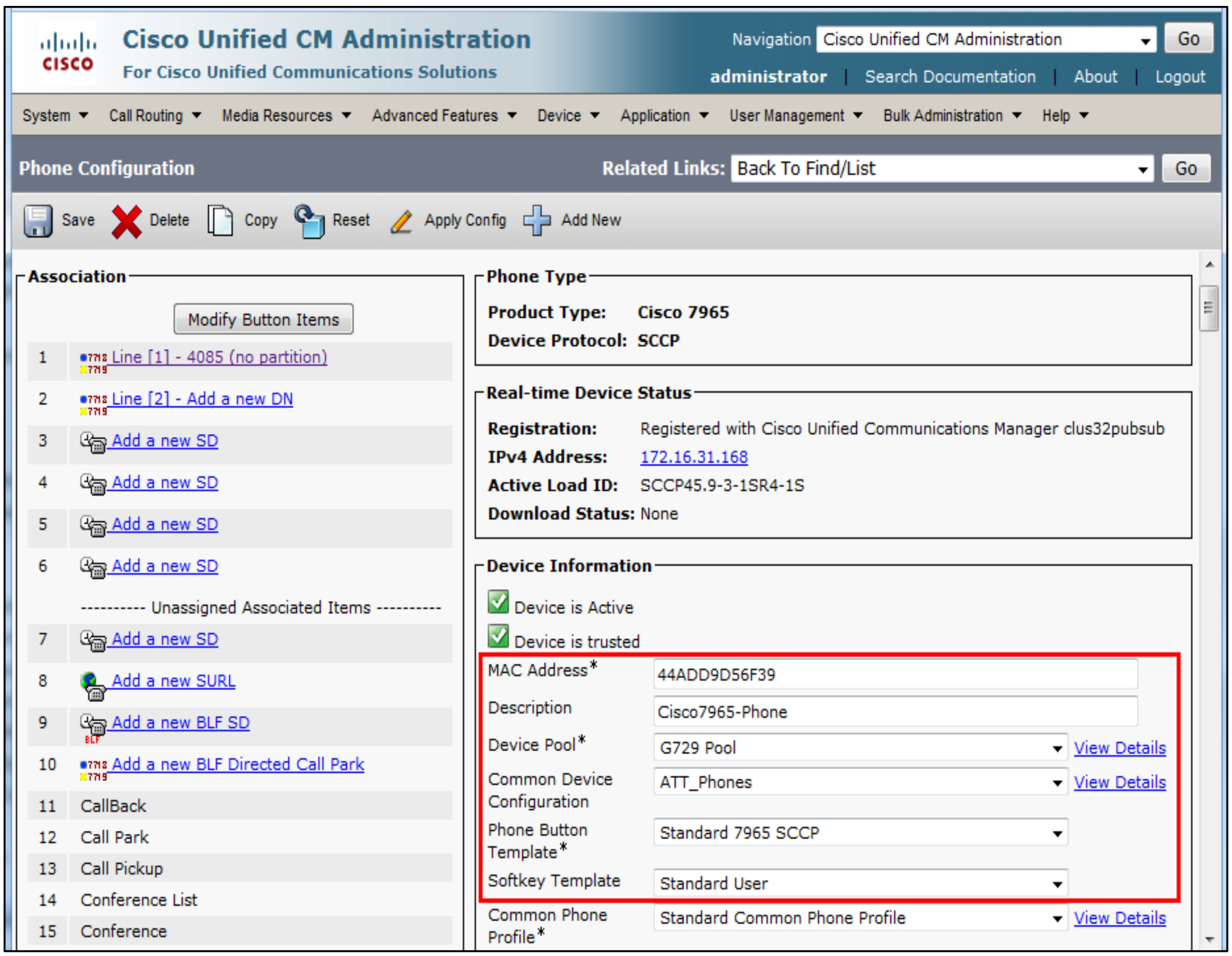
End User Configuration(continued…)





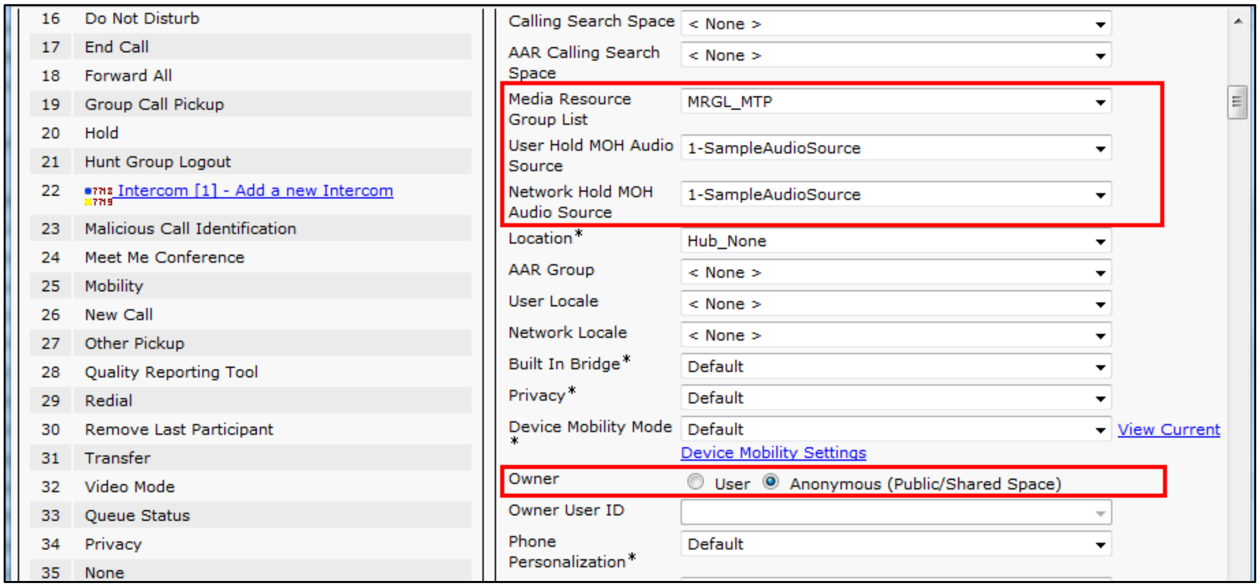
Cisco IP Phone 7965 SCCP Configuration

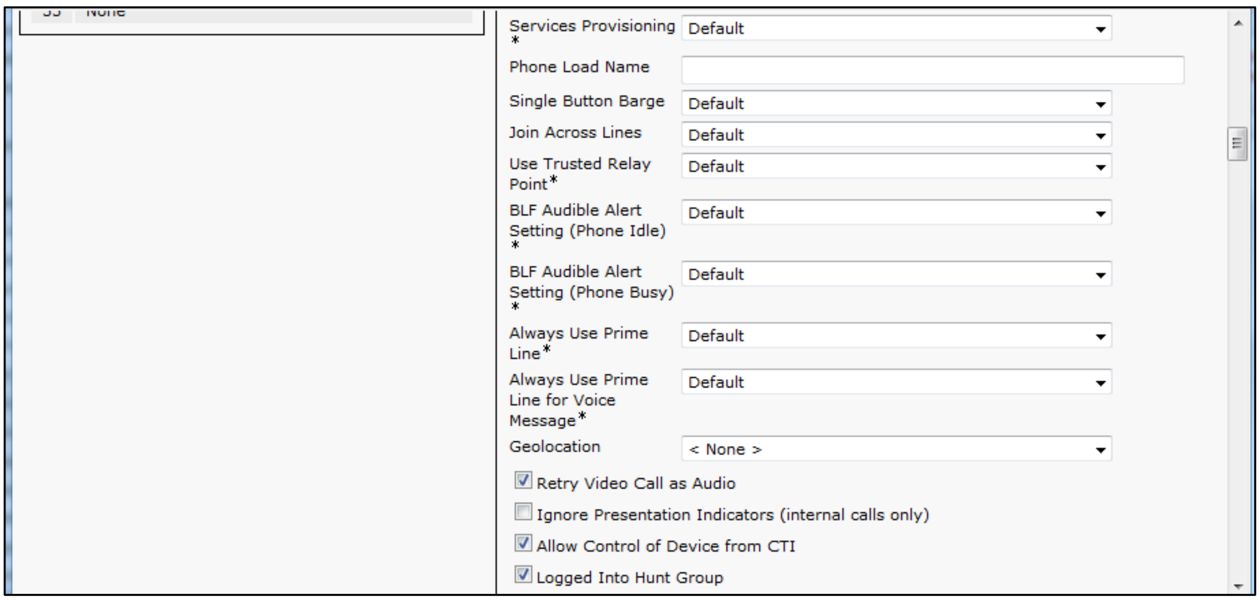
Set MAC Address* = the below mac is used in this example.
Set Description = Cisco7965_Phone. this text is used to identify this Phone.
Set Device Pool*= G729 pool. This is used in this example.
Set Phone Button Template*= Standard 7965 SCCP. This is used in this example.
Set Softkey Template = Standard User. This is used in this example.



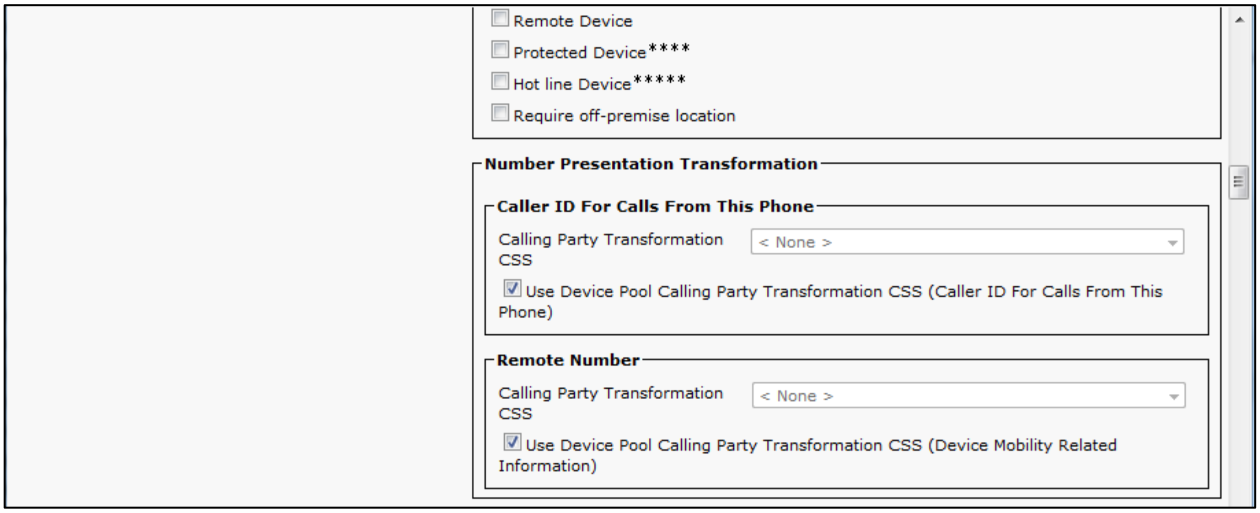
Cisco IP Phone 7965 SCCP Configuration (Continued…)

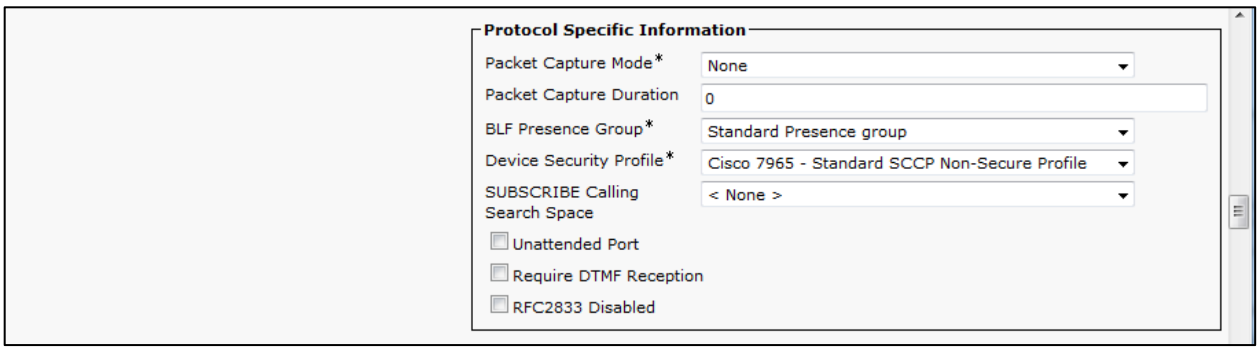
Set Media Resource Group List = MRGL_MTP. This is used in this example.
Set User Hold MOH Audio Source = 1-SampleAudioSource.
Set Network Hold MOH Audio Source = 1-SampleAudioSource.
Check Owner = Anonymous (Public/Shared Space). This is used in this example.

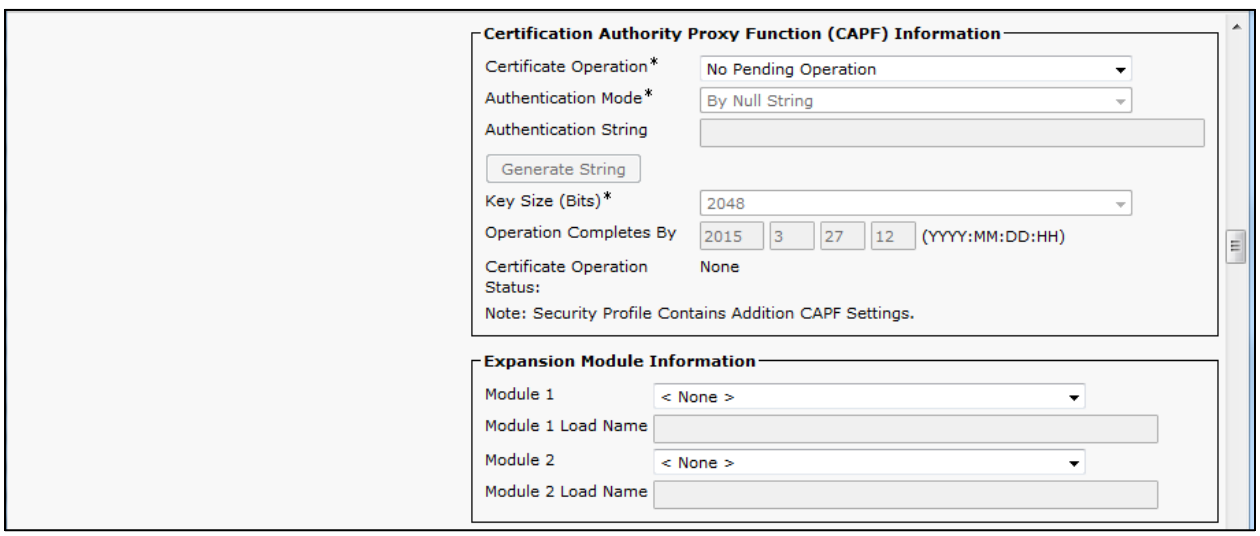




Cisco IP Phone 7965 SCCP Configuration (Continued…)

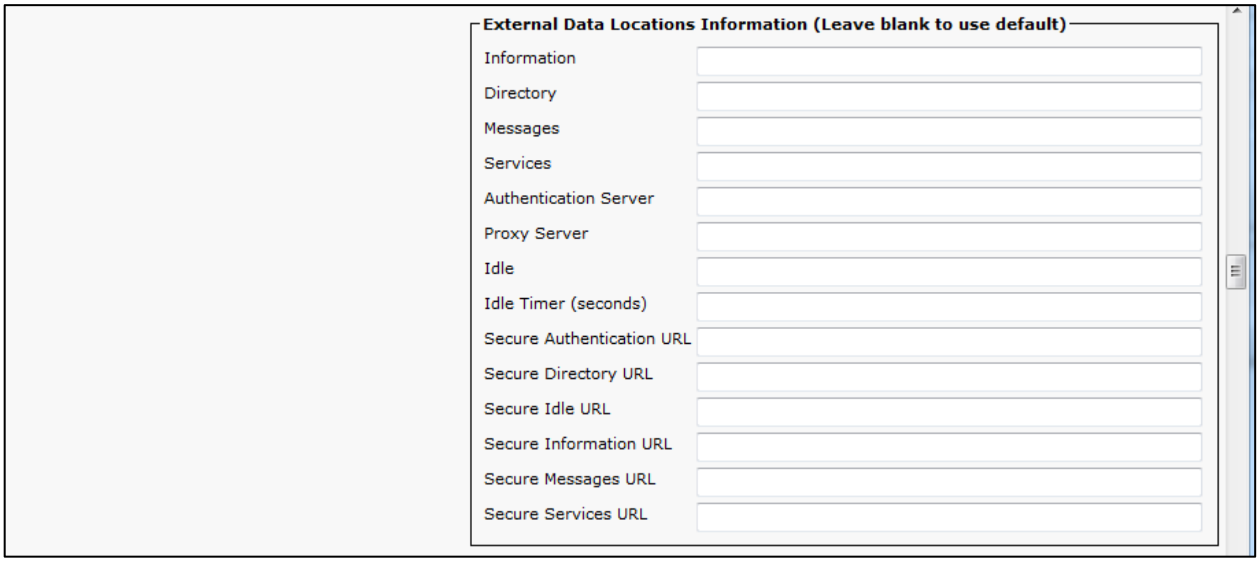


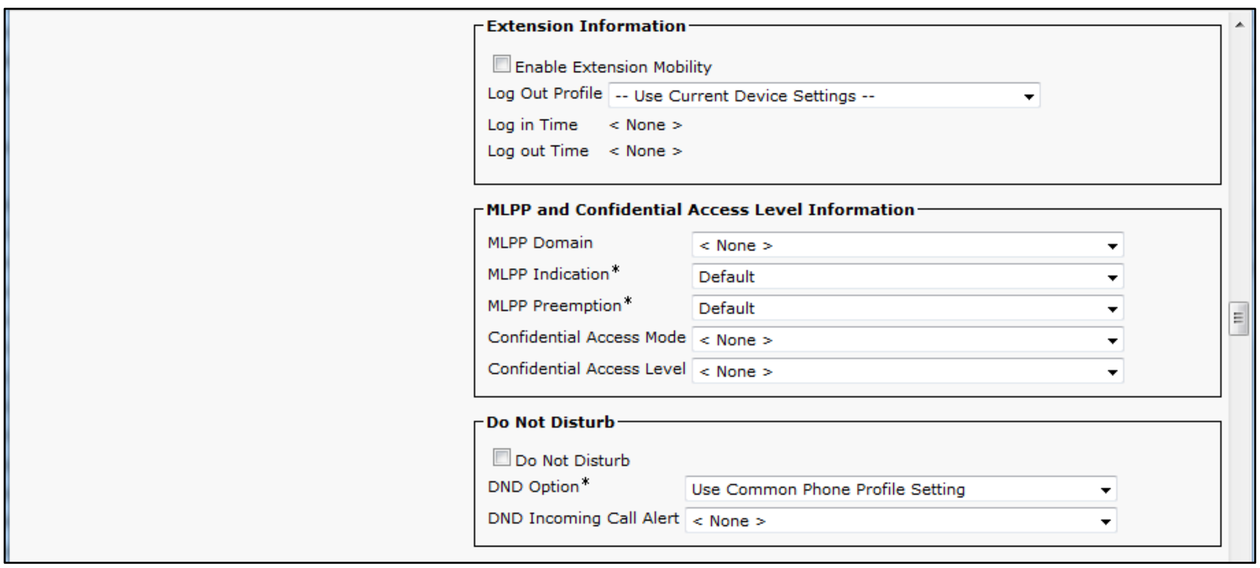


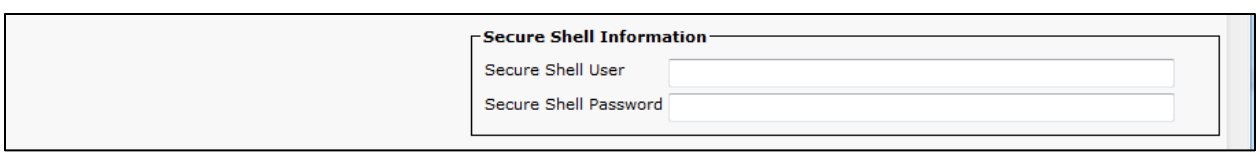


Note: Testing was conducted in tekVizion labs

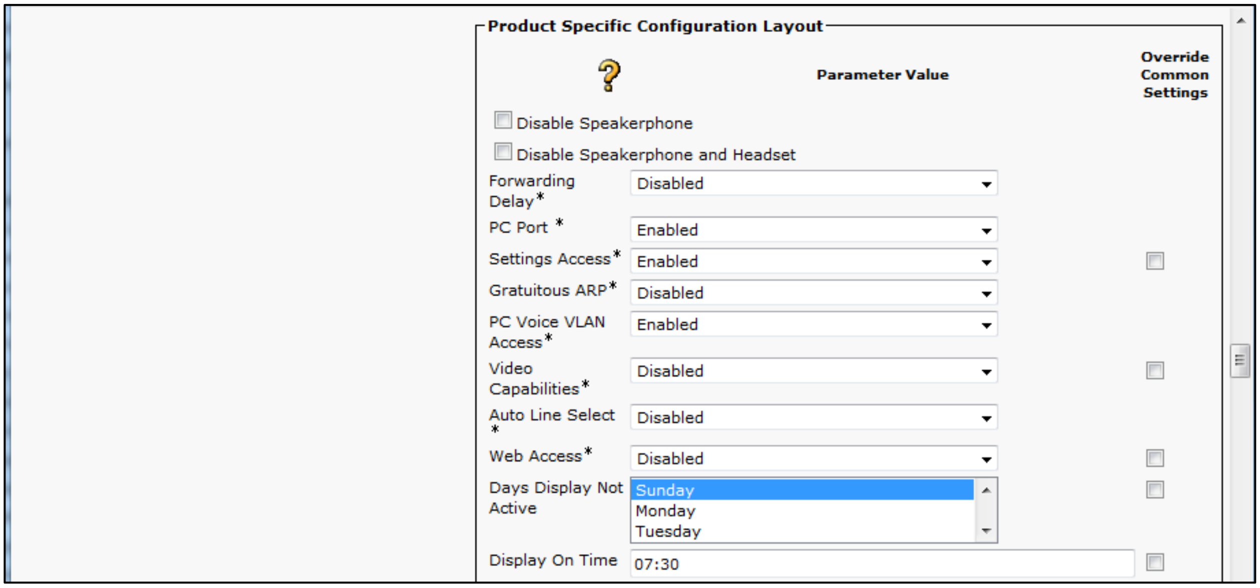
Cisco IP Phone 7965 SCCP Configuration (Continued…)

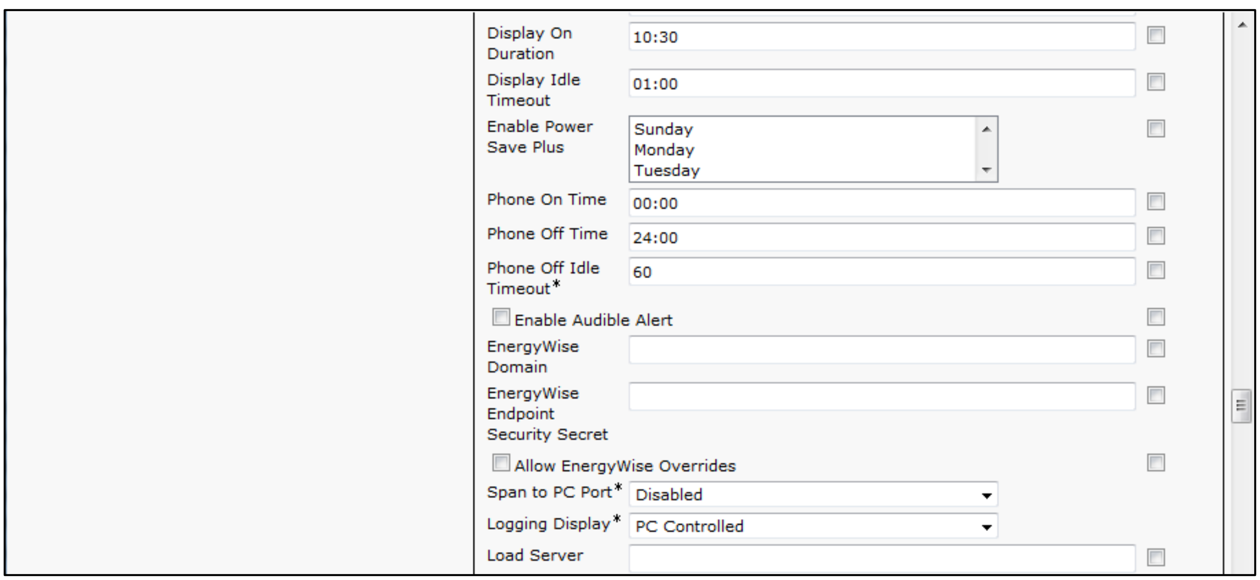


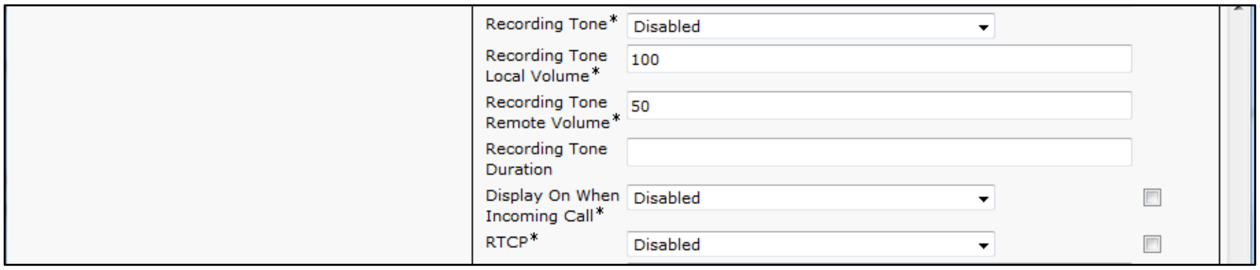




Cisco IP Phone 7965 SCCP Configuration (Continued…)







Cisco IP Phone 7965 SCCP Configuration (Continued…)

"more" Soft Key Timer	5
Auto Call Select*	Enabled
Log Server	
Advertise G.722 Codec*	Use System Default
Wideband Headset UI Control*	Enabled
Wideband Headset*	Enabled
Peer Firmware Sharing*	Enabled
Cisco Discovery Protocol (CDP): Switch Port*	Enabled
Cisco Discovery Protocol (CDP): PC Port*	Enabled
Link Layer Discovery Protocol - Media Endpoint	Enabled

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

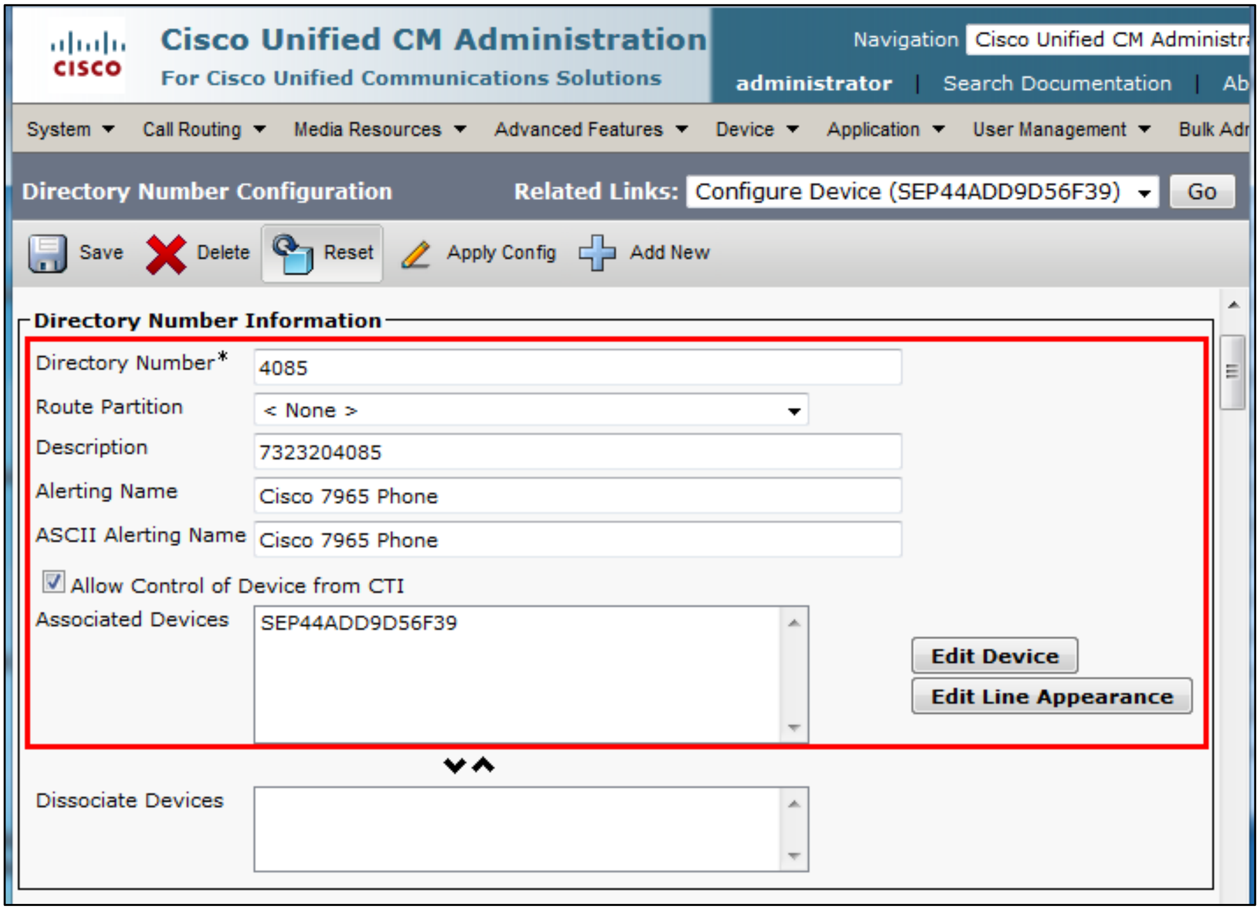
Cisco IP Phone 7965 SCCP Configuration (Continued…)

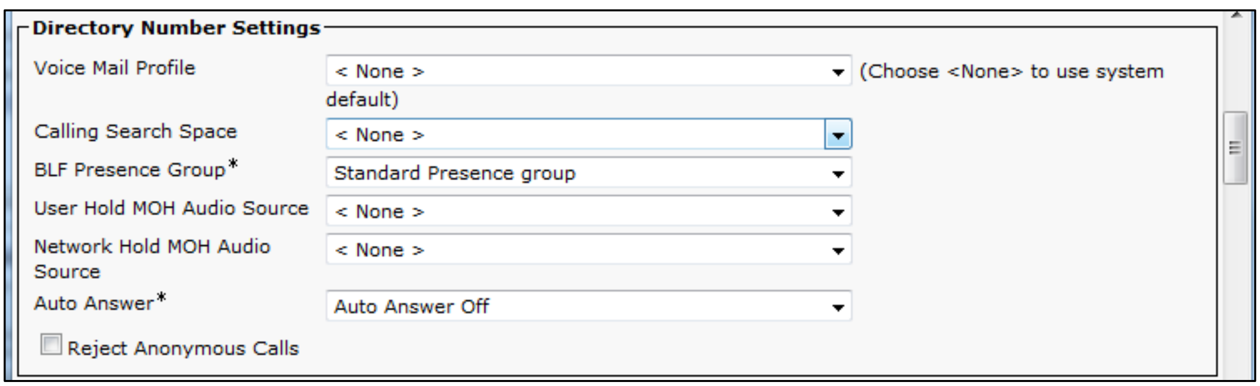
Headset Sidetone Level*	Default
Headset Send Gain*	Default
HTTPS Server*	http and https Enabled
Handset/Headset Monitor*	Enabled
Headset Recording*	Disabled
Enbloc Dialing*	Enabled
Switch Port Remote Configuration*	Disabled
PC Port Remote Configuration*	Disabled
Automatic Port Synchronization*	Disabled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

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

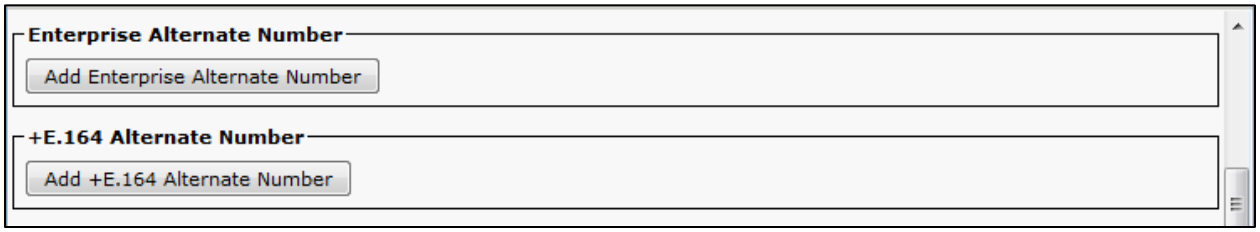
Cisco IP Phone 7965 SCCP Configuration (Continued…)

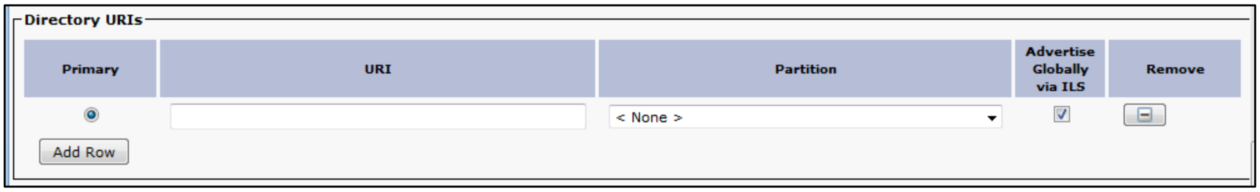
Set Directory Number* = 4085. This is used in this example.
Set Description = 7323204085. This is used in this example.
Set Alerting Name = Cisco 7965 Phone. This is used in this example.
Set ASCII Alerting Name = Cisco 7965 Phone. This is used in this example.

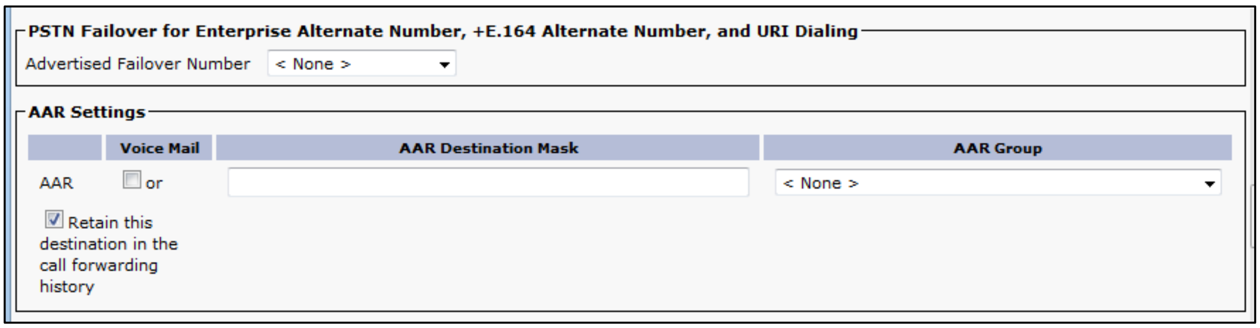


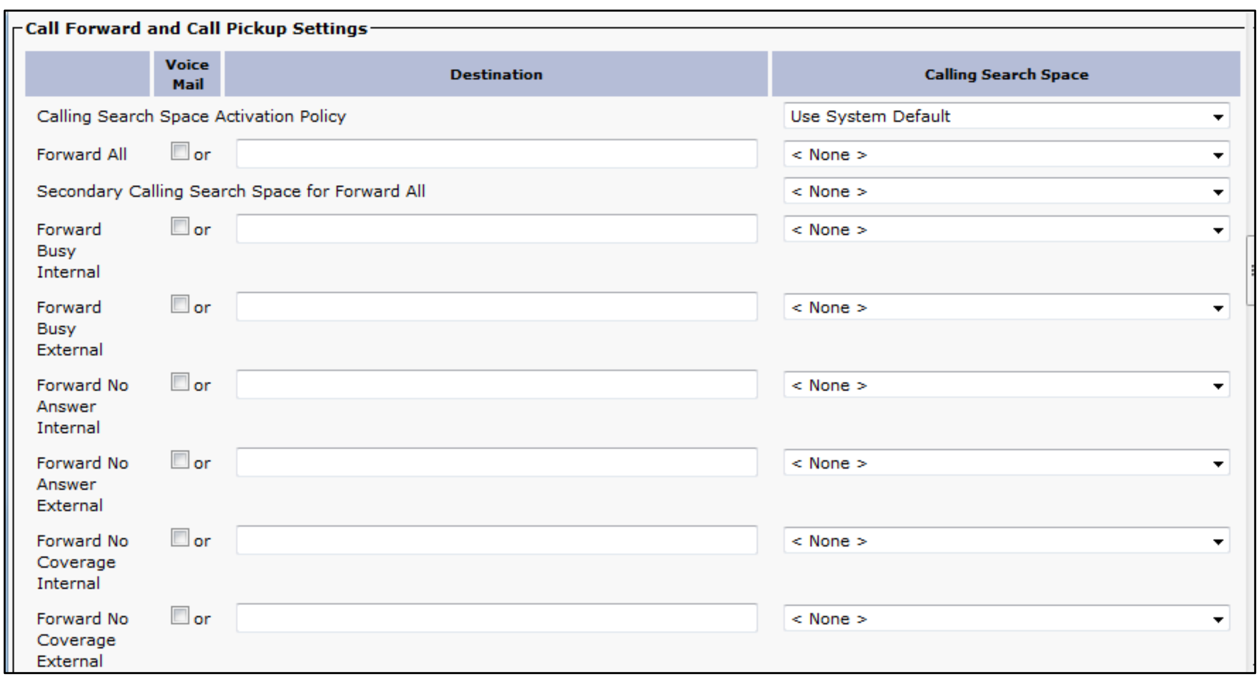


Cisco IP Phone 7965 SCCP Configuration (Continued…)

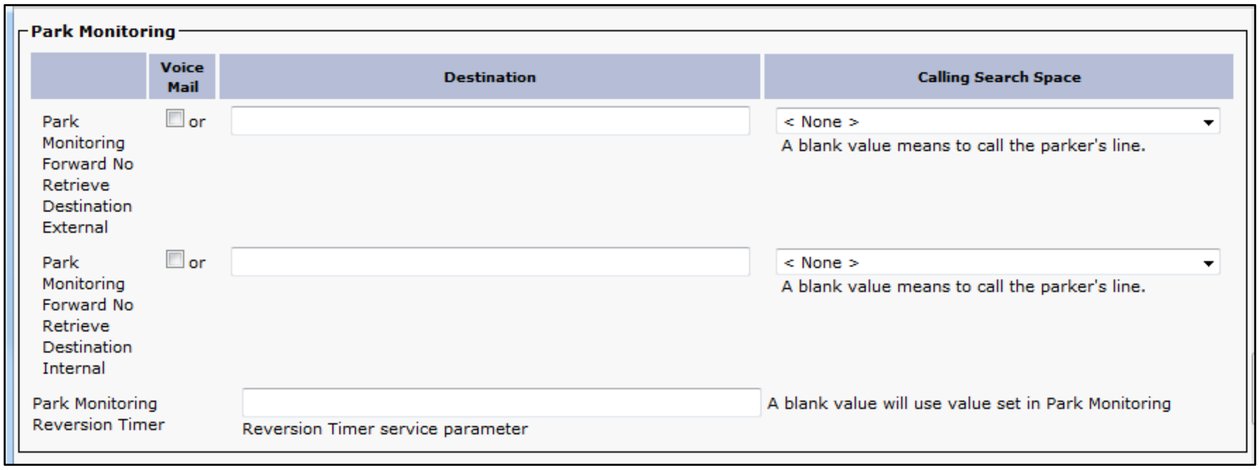

Enterprise Alternate Number
Add Enterprise Alternate Number
+E.164 Alternate Number
Add +E.164 Alternate Number

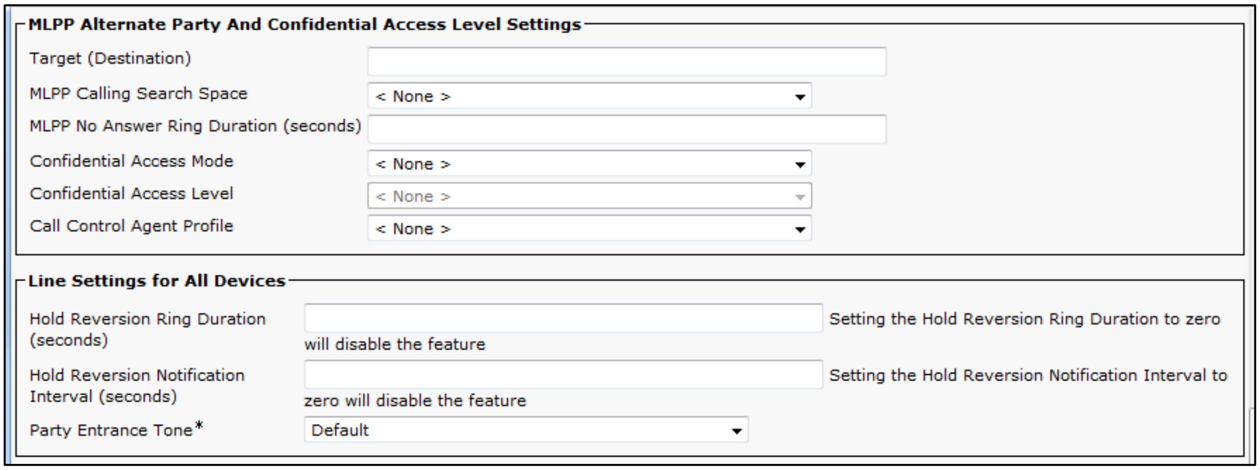

Directory URIs
Primary
URI
Partition
Advertise Globally via ILS
Remove
< None >
Add Row


PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing
Advertised Failover Number
< None >
AAR Settings
Voice Mail
AAR Destination Mask
AAR Group
AAR
or
< None >
Retain this destination in the call forwarding history

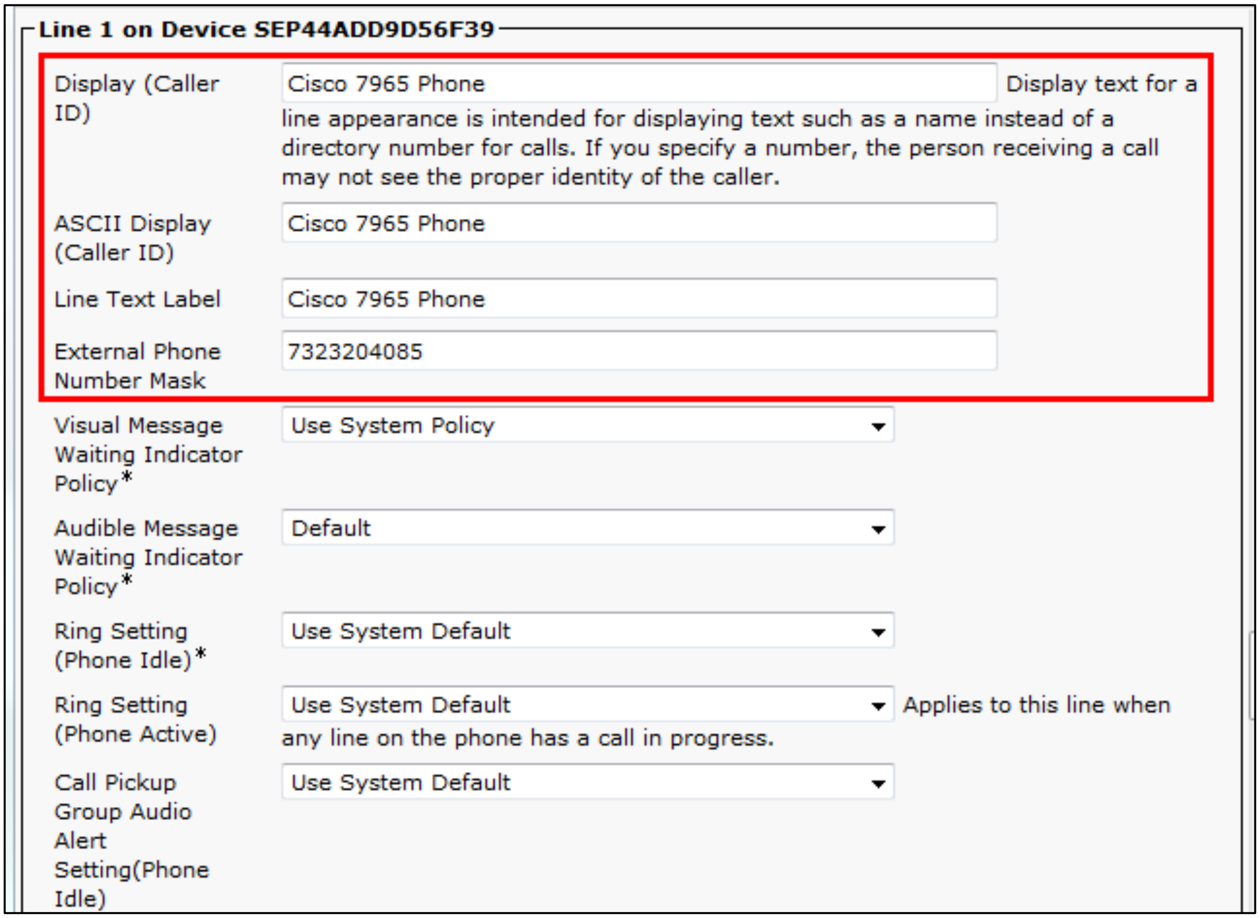

Call Forward and Call Pickup Settings
Voice Mail
Destination
Calling Search Space
Calling Search Space Activation Policy
Use System Default
Forward All
or
< None >
Secondary Calling Search Space for Forward All
< None >
Forward Busy Internal
or
< None >
Forward Busy External
or
< None >
Forward No Answer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

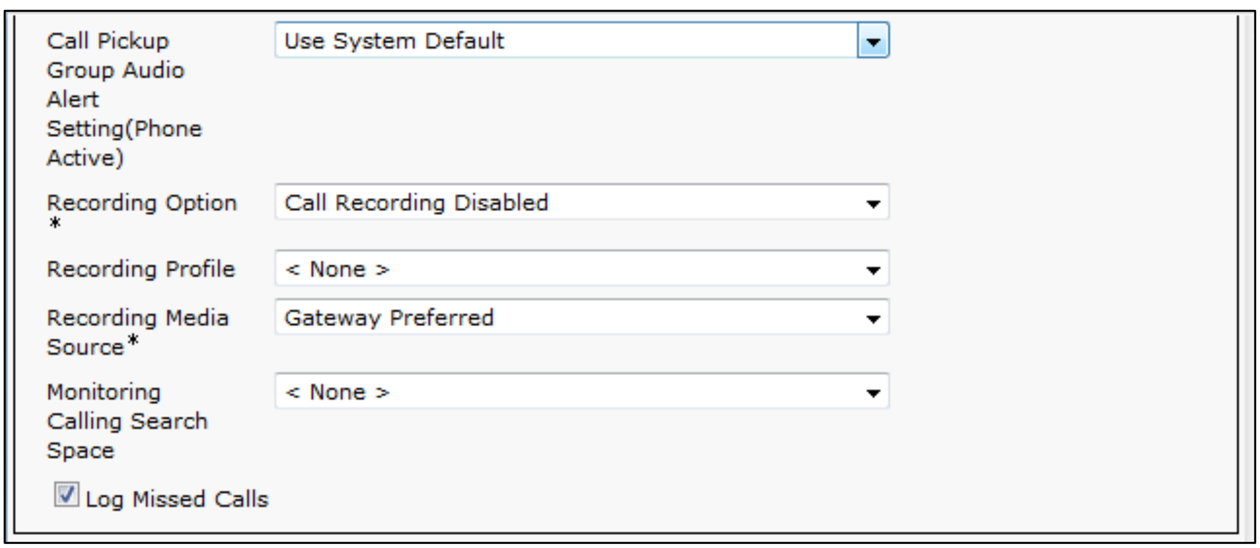
Cisco IP Phone 7965 SCCP Configuration (Continued…)



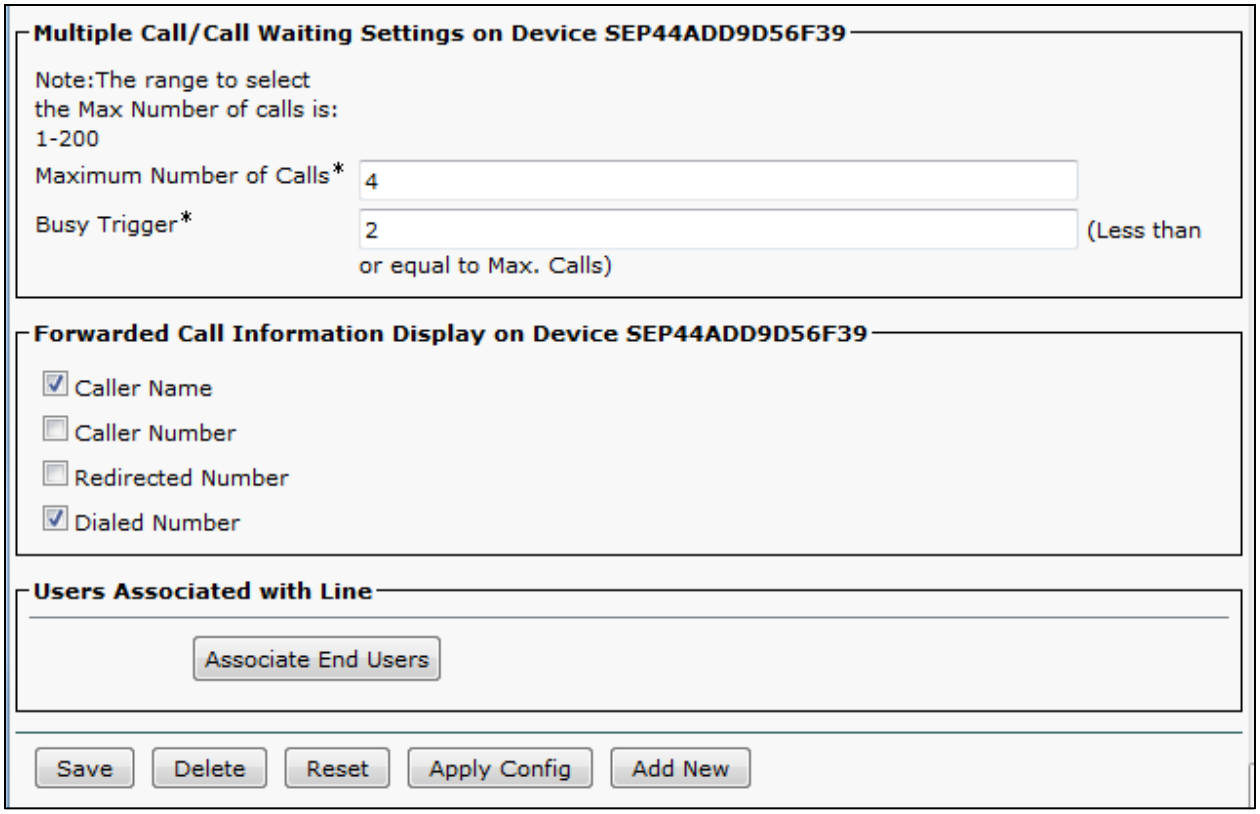


Cisco IP Phone 7965 SCCP Configuration (Continued…)



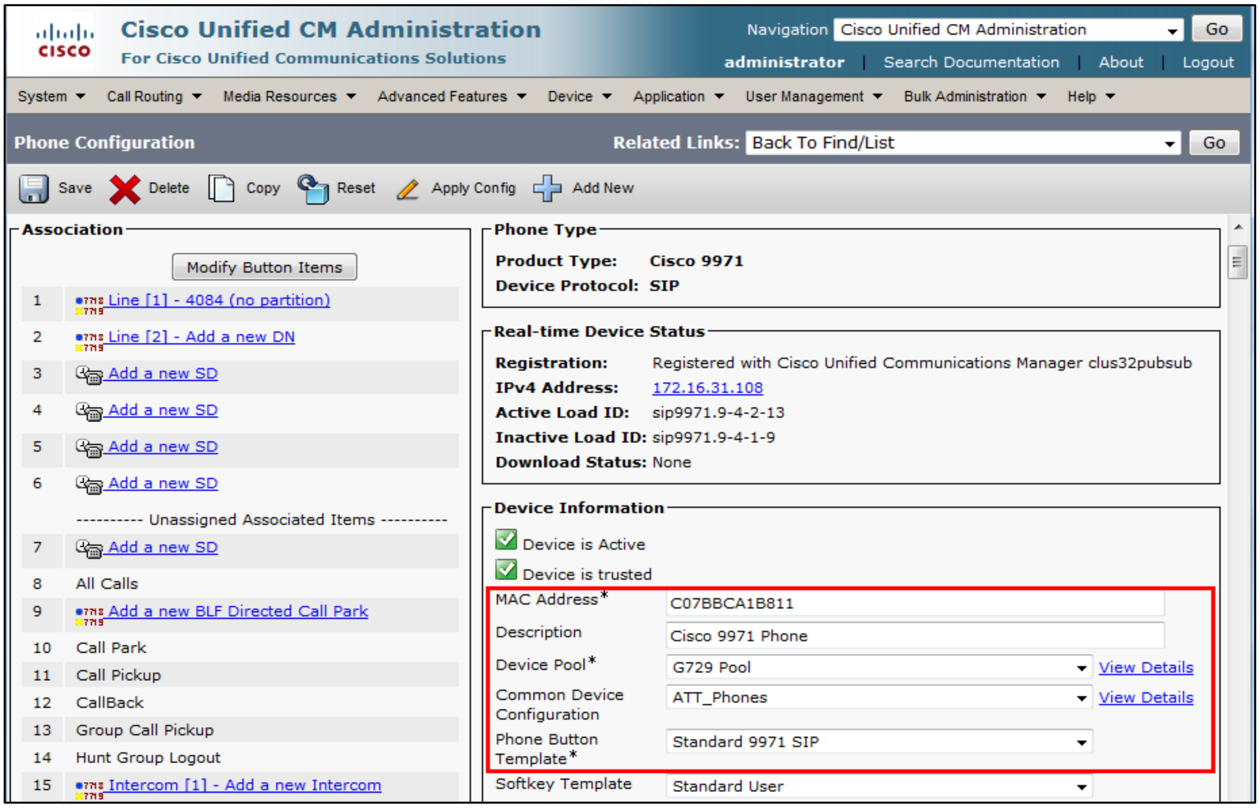


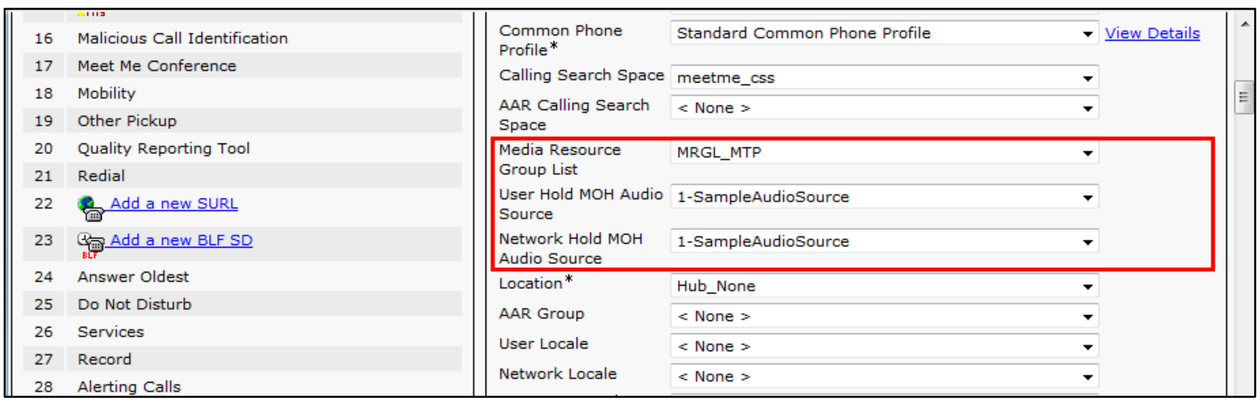
Cisco IP Phone 7965 SCCP Configuration (Continued…)


Multiple Call/Call Waiting Settings on Device SEP44ADD9D56F39
Note:The range to select the Max Number of calls is: 1-200
Maximum Number of Calls*
4
Busy Trigger*
2
(Less than or equal to Max. Calls)
Forwarded Call Information Display on Device SEP44ADD9D56F39
Caller Name
Caller Number
Redirected Number
Dialed Number
Users Associated with Line
Associate End Users
Save
Delete
Reset
Apply Config
Add New

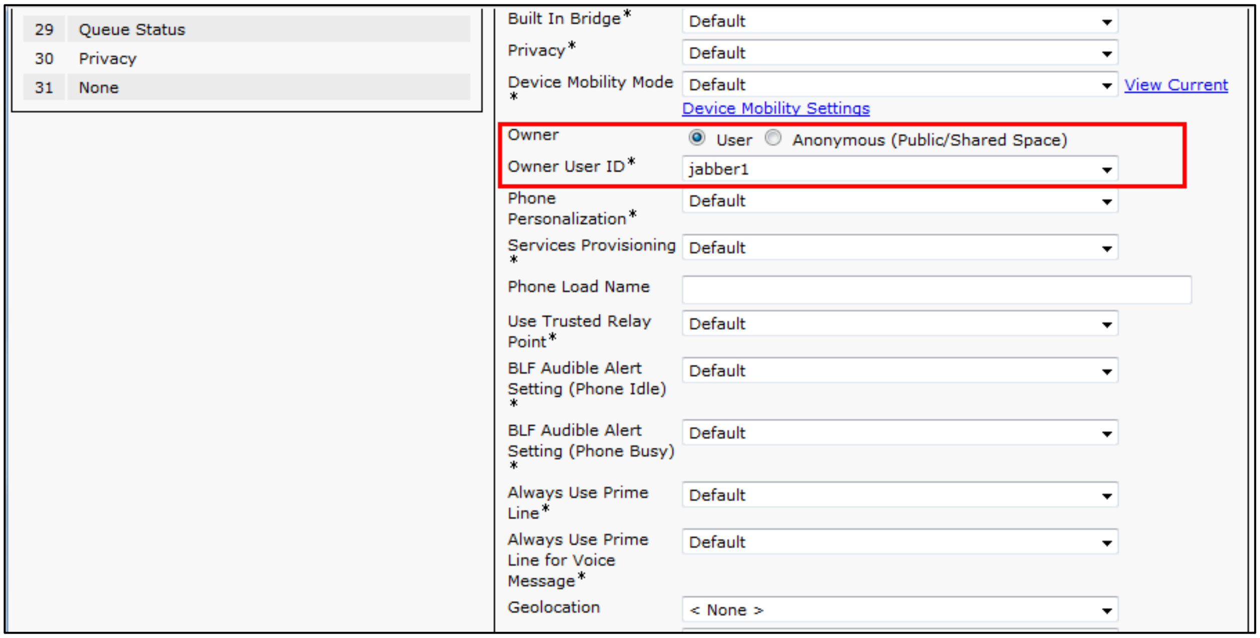
Cisco IP Phone 9971 SIP Configuration

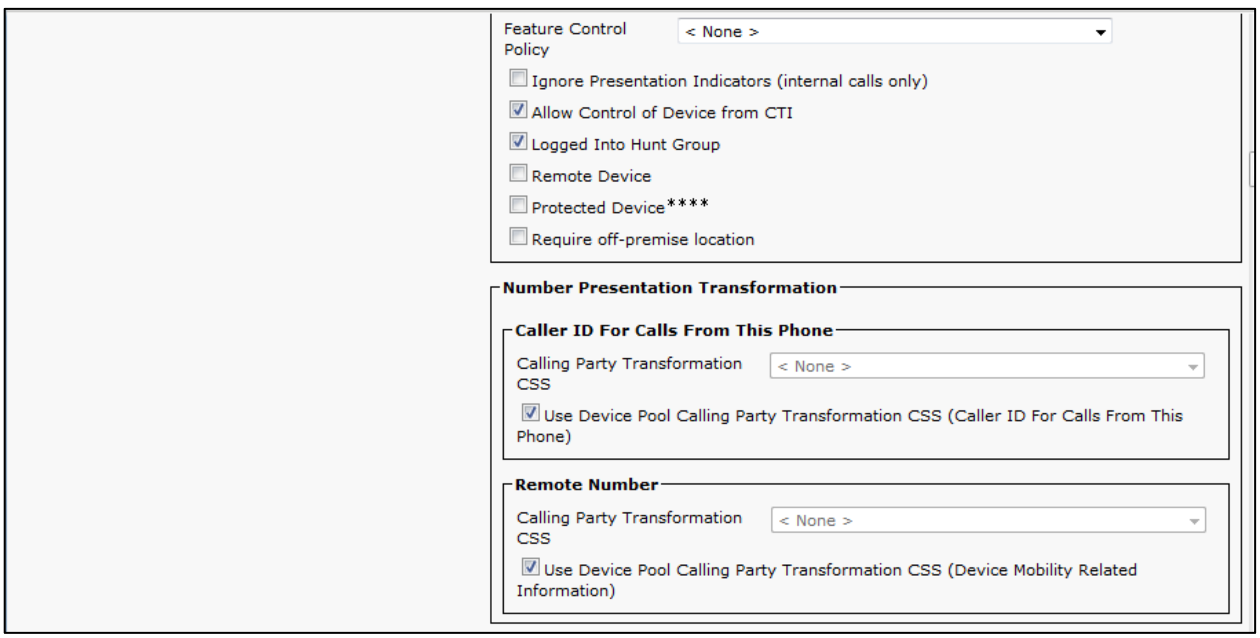
Set MAC Address* = the below mac is used in this example.
Set Description = Cisco 9971 Phone. this text is used to identify this Phone.
Set Device Pool*= G729 Pool. This is used in this example.
Set Phone Button Template*= Standard 9971 SIP. This is used in this example.
Set Media Resource Group List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





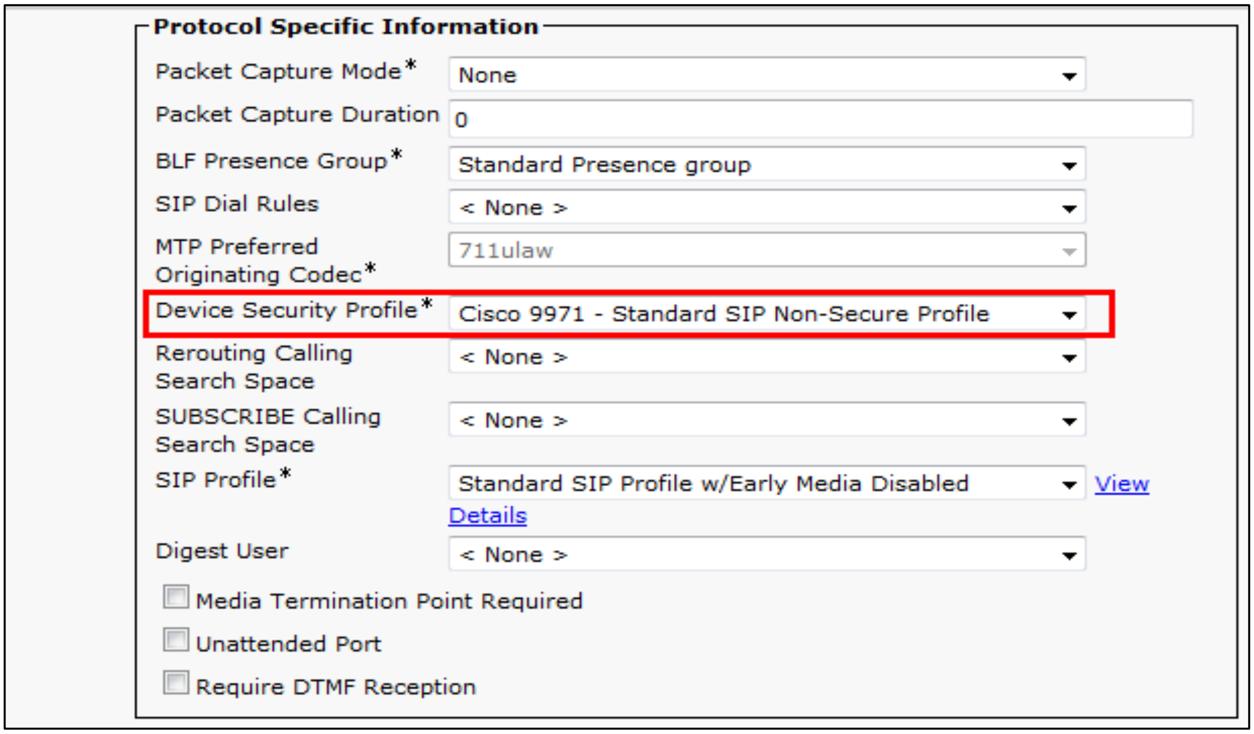
Cisco IP Phone 9971 SIP Configuration(Continued…)

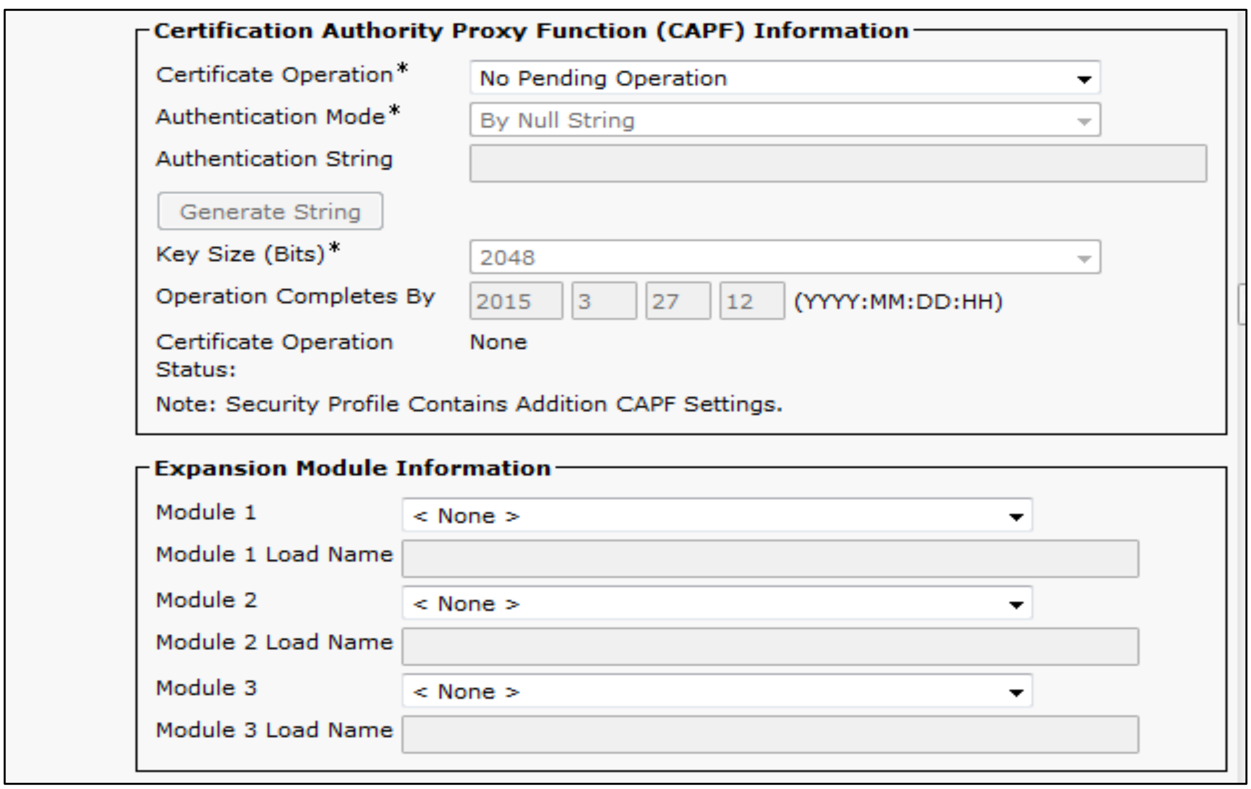




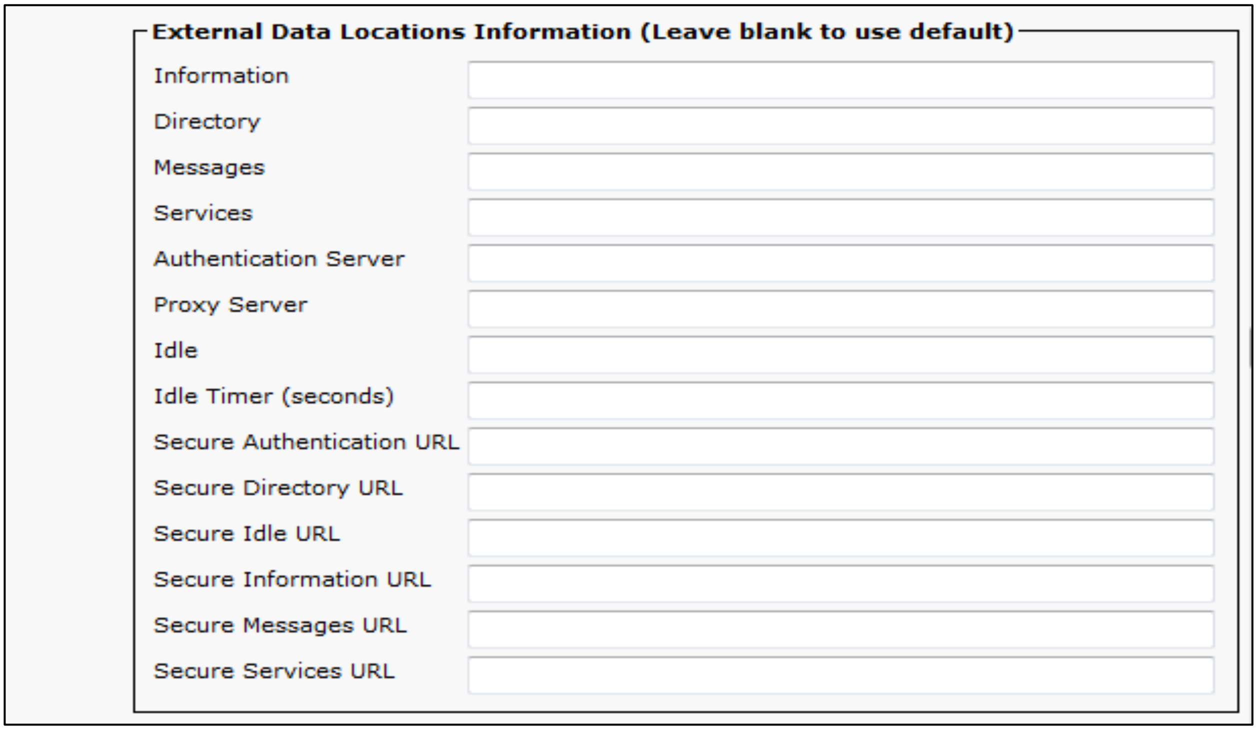
Note: Testing was conducted in tekVizion labs

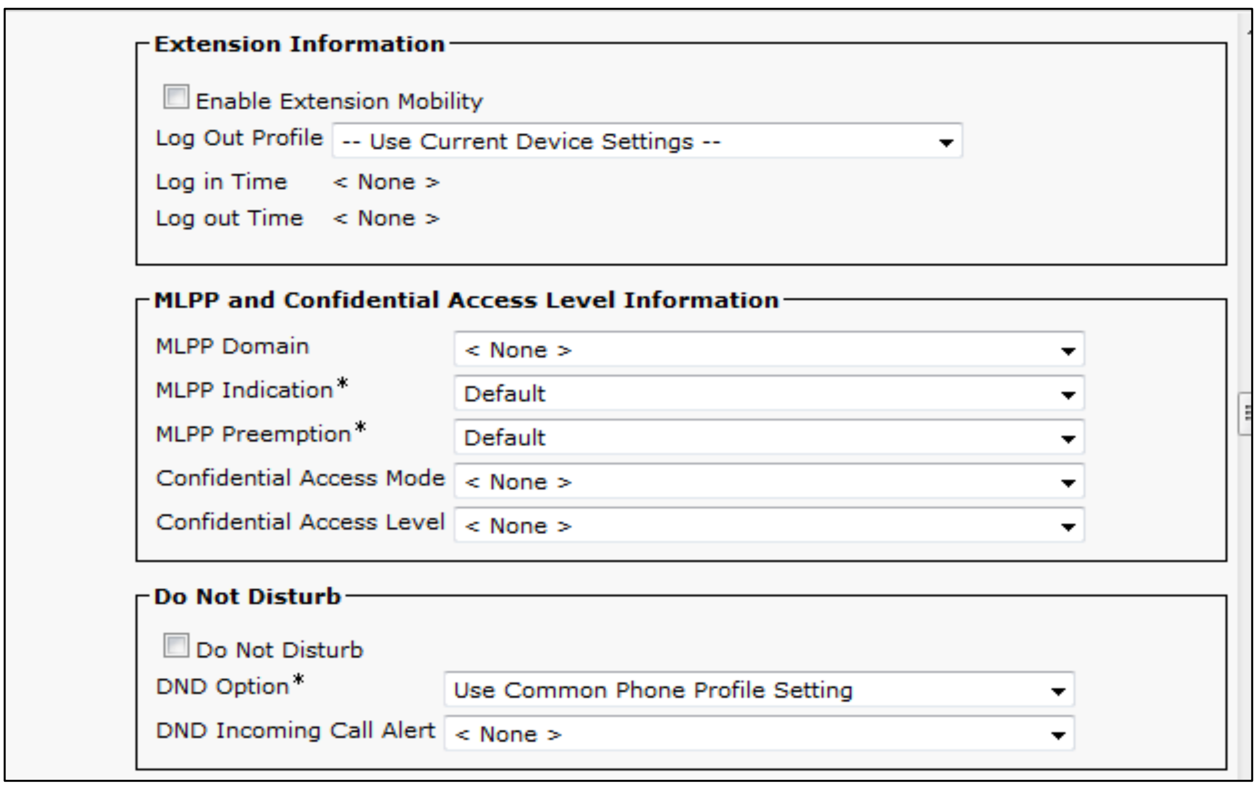
Cisco IP Phone 9971 SIP Configuration(Continued…)


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

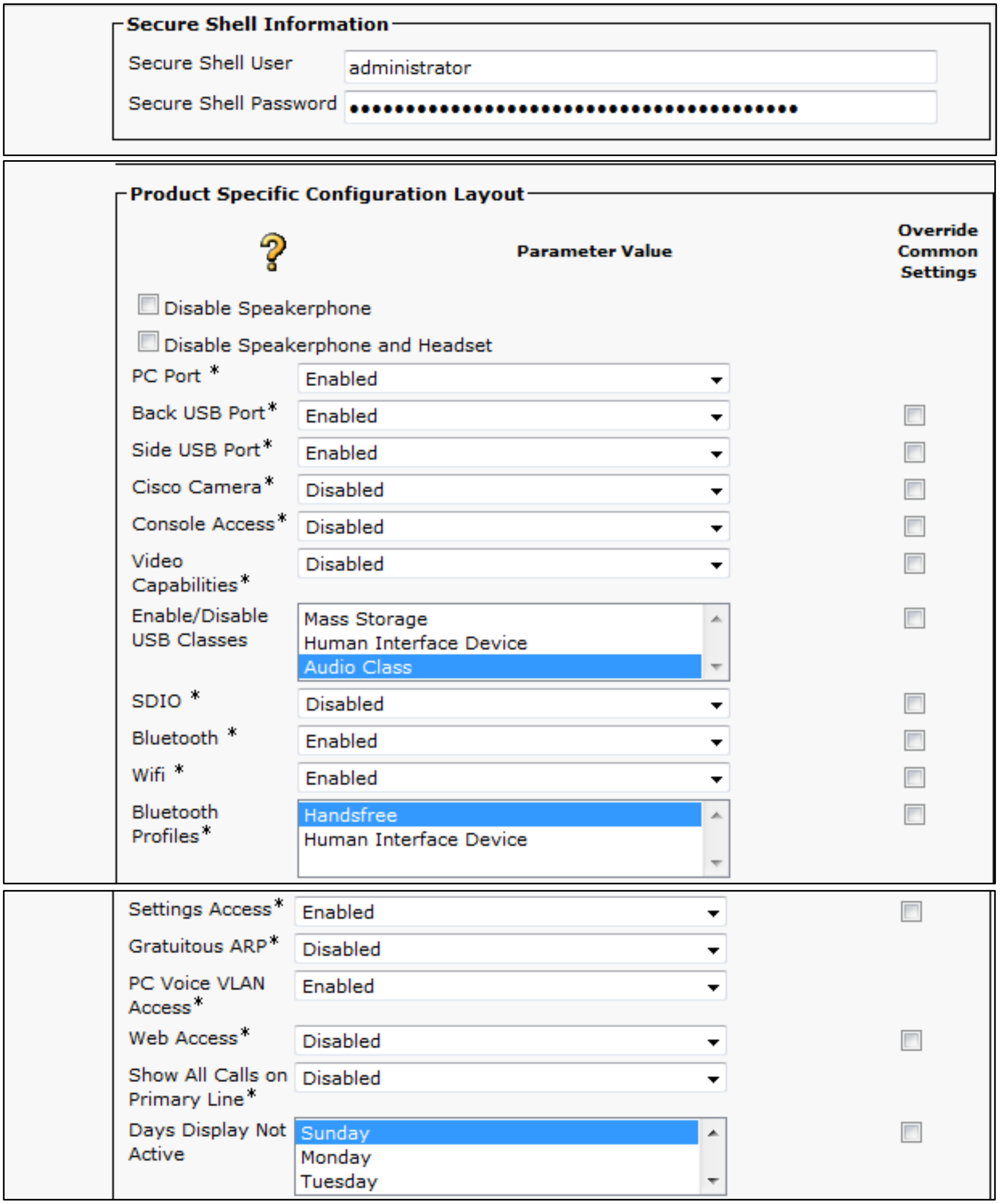

Certification Authority Proxy Function (CAPF) Information
Certificate Operation*
No Pending Operation
Authentication Mode*
By Null String
Authentication String
Generate String
Key Size (Bits)*
2048
Operation Completes By
2015 3 27 12 (YYYY:MM:DD:HH)
Certificate Operation Status:
None
Note: Security Profile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

Cisco IP Phone 9971 SIP Configuration(Continued…)

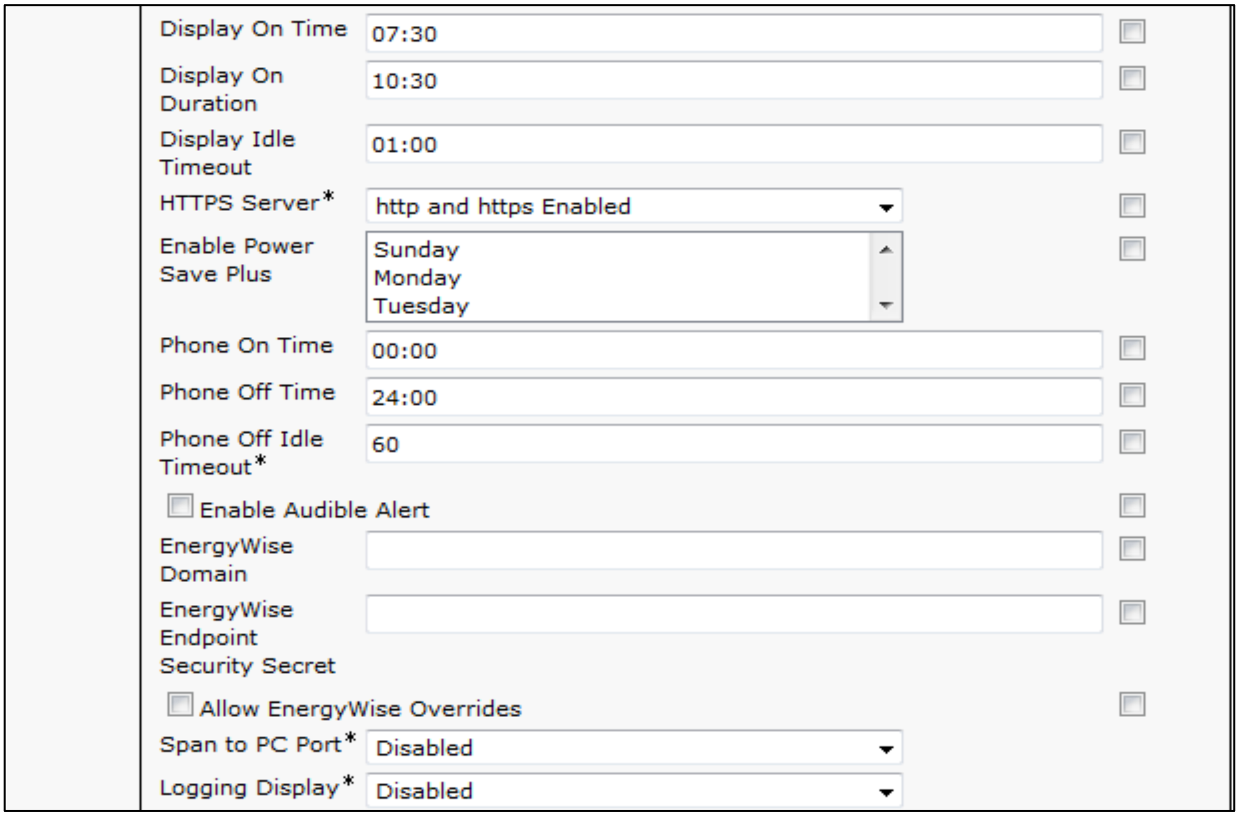

External Data Locations Information (Leave blank to use default)
Information
Directory
Messages
Services
Authentication Server
Proxy Server
Idle
Idle Timer (seconds)
Secure Authentication URL
Secure Directory URL
Secure Idle URL
Secure Information URL
Secure Messages URL
Secure Services URL


Extension Information
Enable Extension Mobility
Log Out Profile
-- Use Current Device Settings --
Log in Time
< None >
Log out Time
< None >
MLPP and Confidential Access Level Information
MLPP Domain
< None >
MLPP Indication*
Default
MLPP Preemption*
Default
Confidential Access Mode
< None >
Confidential Access Level
< None >
Do Not Disturb
Do Not Disturb
DND Option*
Use Common Phone Profile Setting
DND Incoming Call Alert
< None >

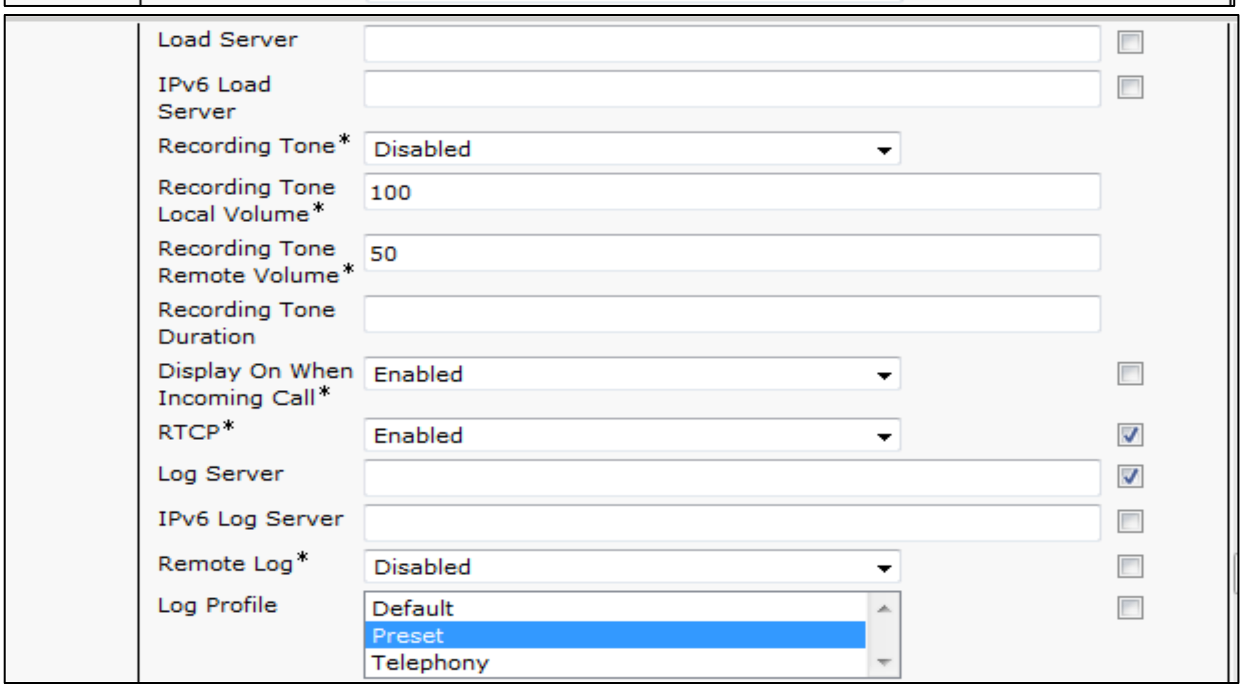
Cisco IP Phone 9971 SIP Configuration(Continued…)



Cisco IP Phone 9971 SIP Configuration(Continued…)



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	
Allow EnergyWise Overrides	
Span to PC Port*	Disabled
Logging Display*	Disabled



Load Server	
IPv6 Load Server	
Recording Tone*	Disabled
Recording Tone Local Volume*	100
Recording Tone Remote Volume*	50
Recording Tone Duration	
Display On When Incoming Call*	Enabled
RTCP*	Enabled
Log Server	
IPv6 Log Server	
Remote Log*	Disabled
Log Profile	Default Preset Telephony

Cisco IP Phone 9971 SIP Configuration(Continued…)

Advertise G.722 and iSAC Codecs *	Use System Default	
Wideband Headset UI Control*	Enabled	
Wideband Headset*	Enabled	
Peer Firmware Sharing*	Enabled	☐
Cisco Discovery Protocol (CDP): Switch Port*	Enabled	☐
Cisco Discovery Protocol (CDP): PC Port*	Enabled	☐
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	
802.1x Authentication*	User Controlled	☐
FIPS Mode*	Disabled	☐
Detect Unified CM Connection Failure*	Normal	☐
Switch Port Remote Configuration*	Disabled	☐
PC Port Remote Configuration*	Disabled	☐
Automatic Port Synchronization*	Disabled	☐
Power Negotiation*	Enabled	☐
Restrict Data Rates*	Disabled	
SSH Access*	Disabled	☐
Incoming Call Toast Timer*	5	☐
Provide Dial Tone from Release Button*	Disabled	☐

Cisco IP Phone 9971 SIP Configuration(Continued…)

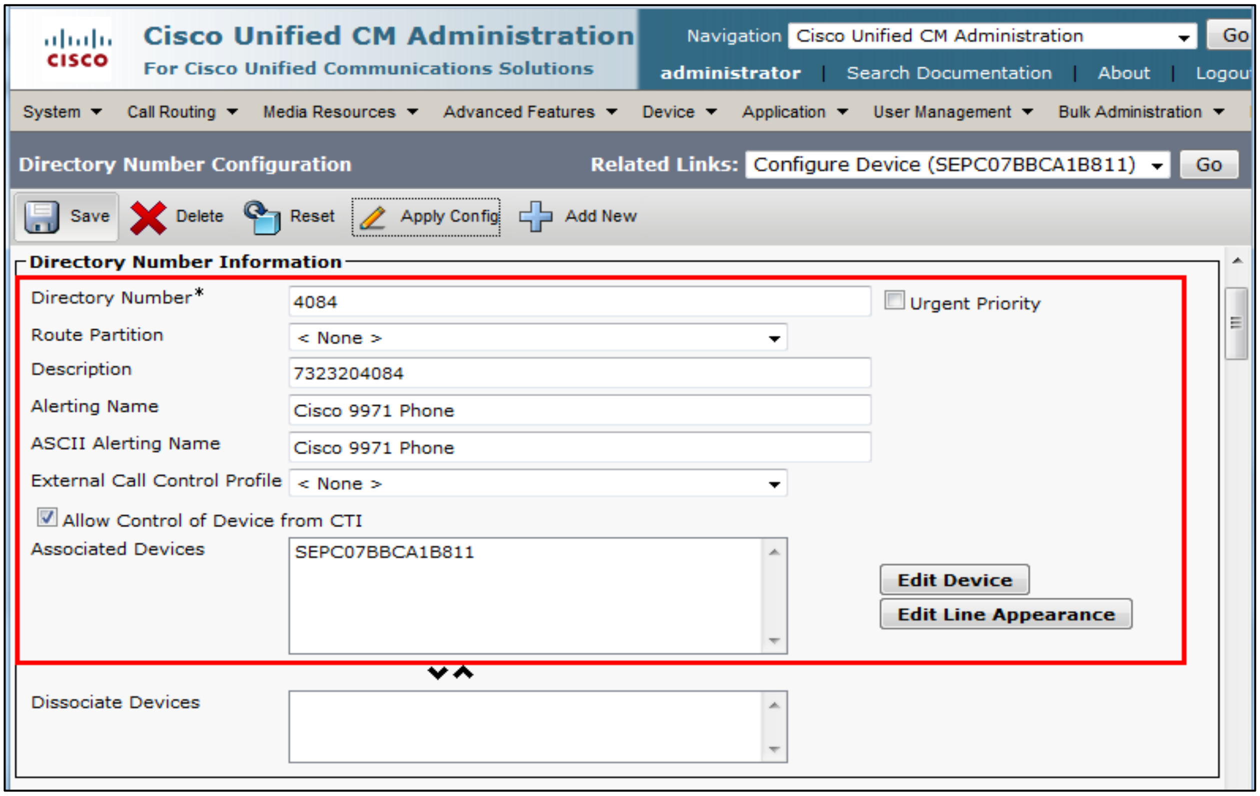
Hide Video By Default*	Disabled
Background Image	
Simplified New Call UI*	Disabled
Enable VXC VPN for MAC	
VXC VPN Option*	Dual Tunnel
VXC Challenge*	Challenge
VXC-M Servers	
Revert to All Calls*	Disabled
RTCP for Video*	Enabled
Record Call Log from Shared Line*	Disabled
Show Remote Private Calls*	Disabled
Record Call Log For Remote Private Calls*	Enabled
Show Call History for Selected Line Only.*	Disabled
Actionable Incoming Call Alert*	Disabled
DF bit*	0
Default Line Filter	
Separate Audio and Video Mute*	Disabled
Softkey Control*	Feature Control Policy
Start Video Port	
Stop Video Port	
Lowest Alerting Line State Priority*	Disabled
TLS Resumption Timer*	3600
Audio EQ*	Default : Default

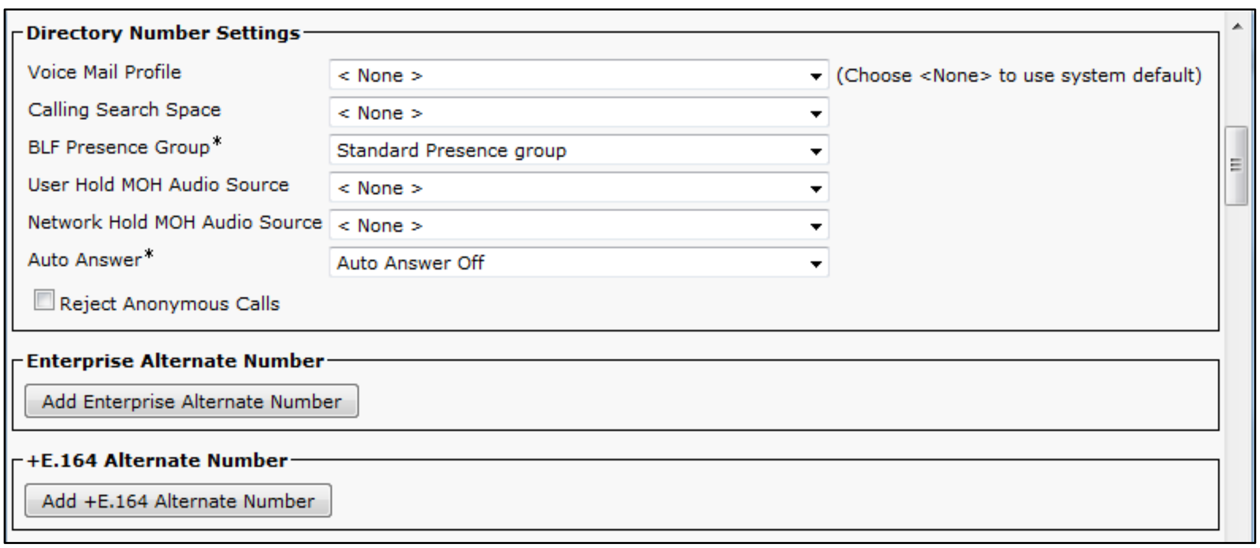
Save | Delete | Copy | Reset | Apply Config | Add New

Note: Testing was conducted in tekVizion labs

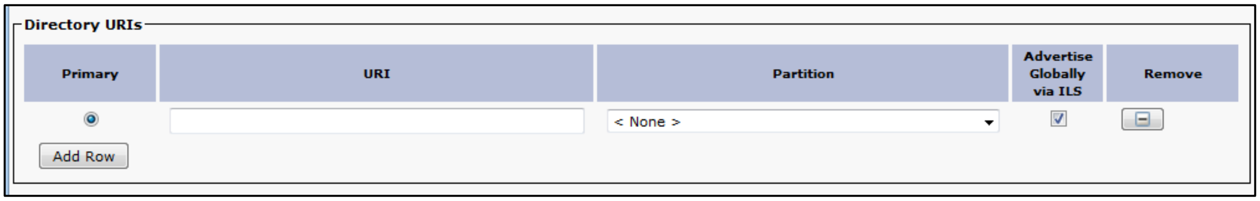
Cisco IP Phone 9971 SIP Configuration (Continued…)

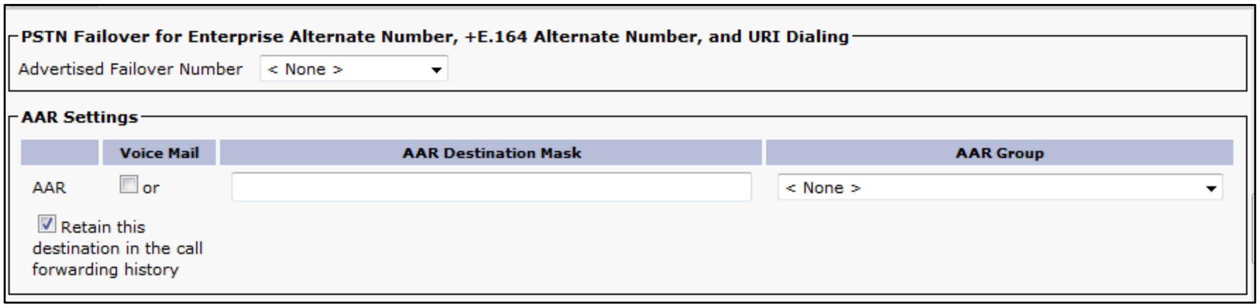
Set Directory Number* = 4084. This is used in this example.
Set Description = 7323204084. This is used in this example.
Set Alerting Name = Cisco 9971 Phone. This is used in this example.
Set ASCII Alerting Name = Cisco 9971 Phone. This is used in this example.

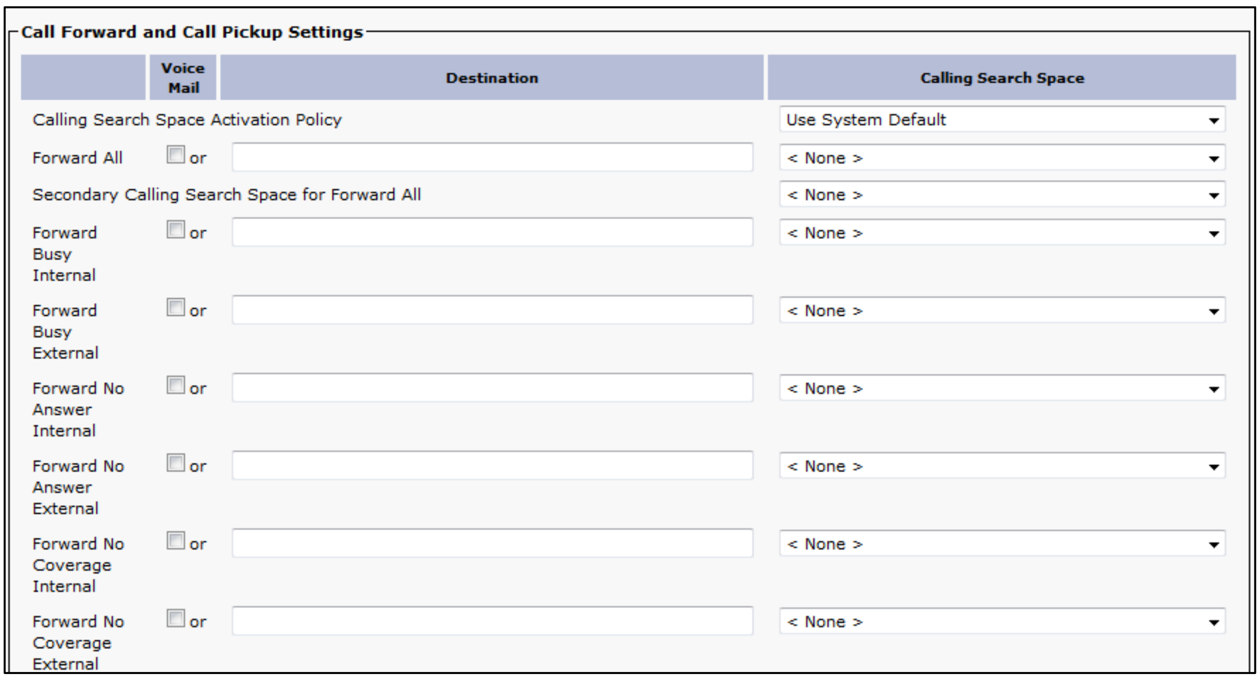




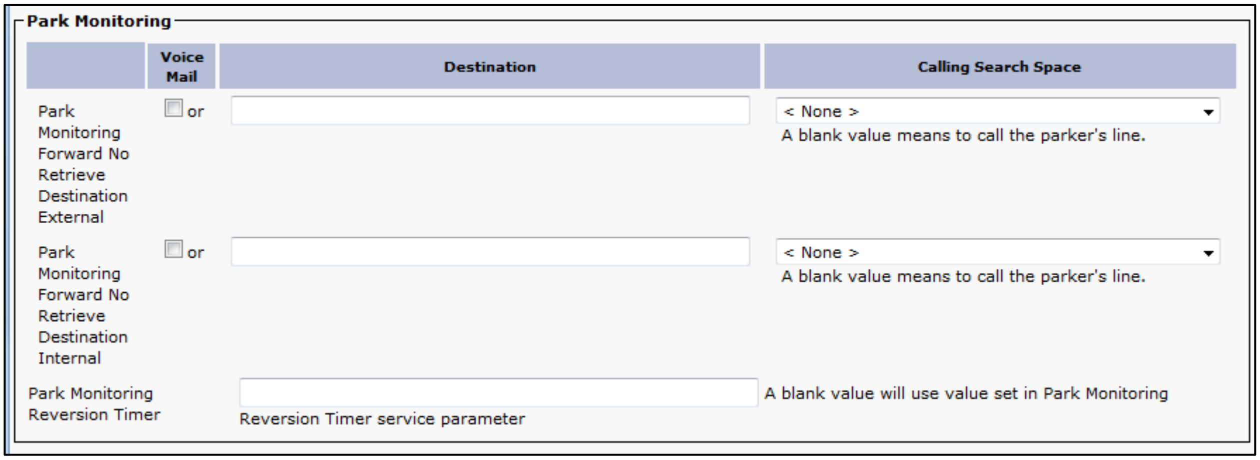
Cisco IP Phone 9971 SIP Configuration (Continued…)

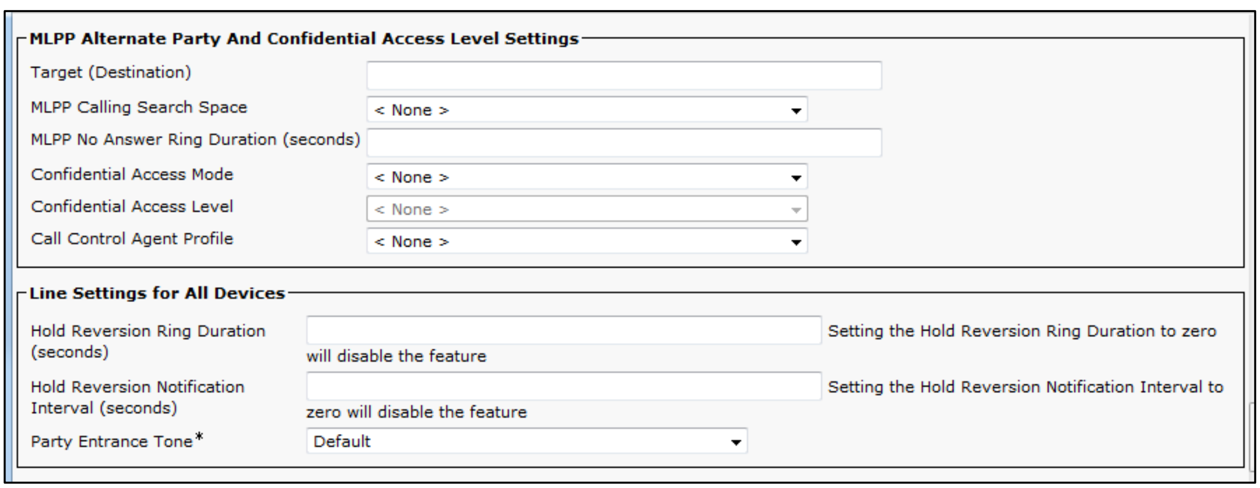

Directory URIs
Primary
URI
Partition
Advertise Globally via ILS
Remove
< None >
Add Row


PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing
Advertised Failover Number
< None >
AAR Settings
Voice Mail
AAR Destination Mask
AAR Group
AAR
or
< None >
Retain this destination in the call forwarding history


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

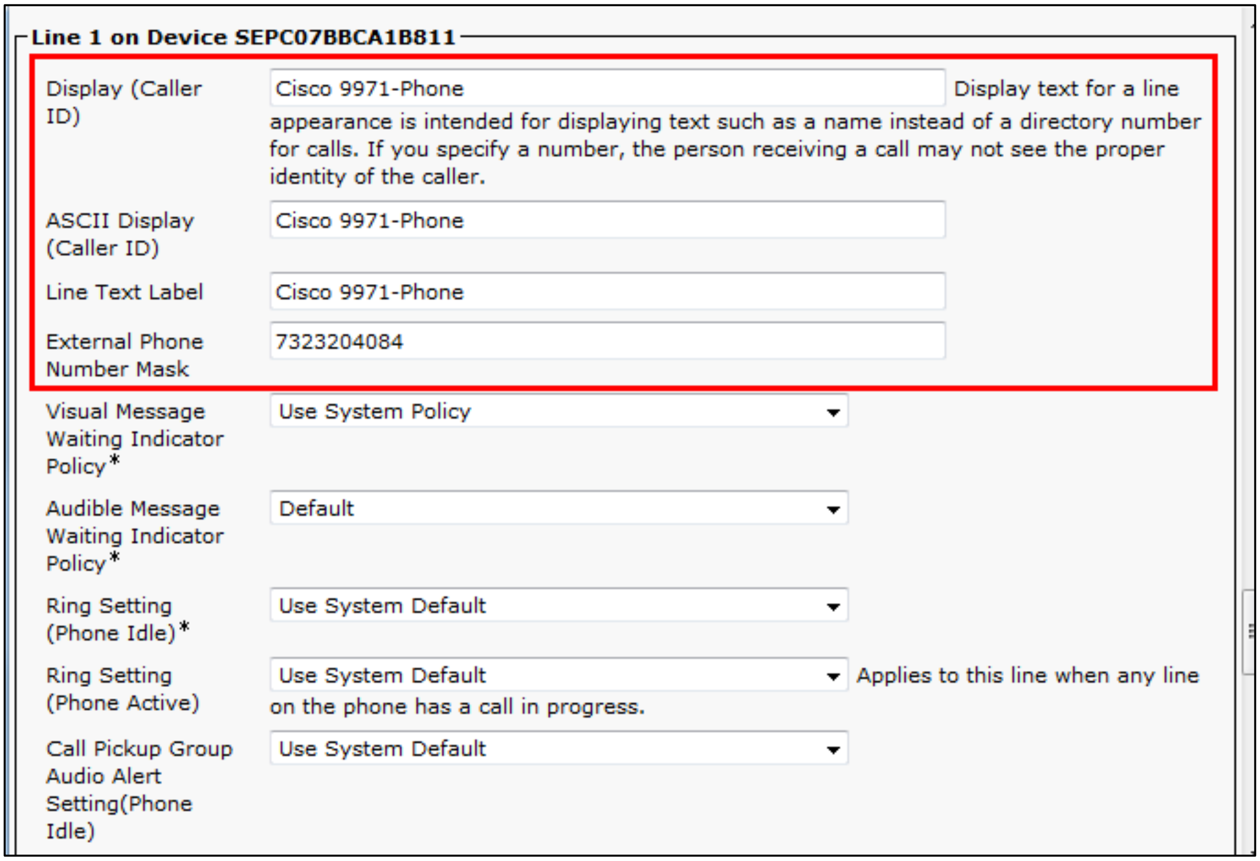
Cisco IP Phone 9971 SIP Configuration (Continued…)

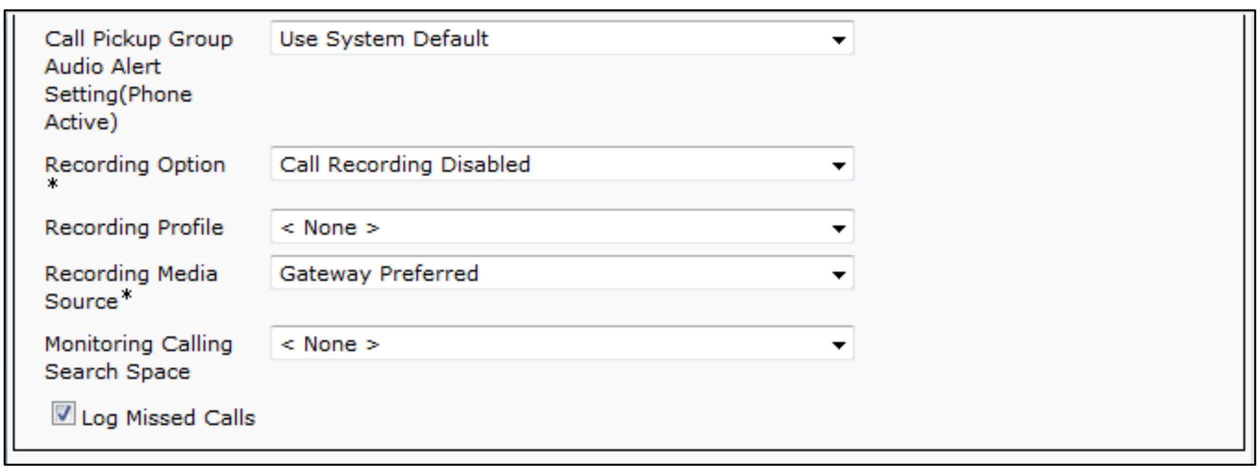




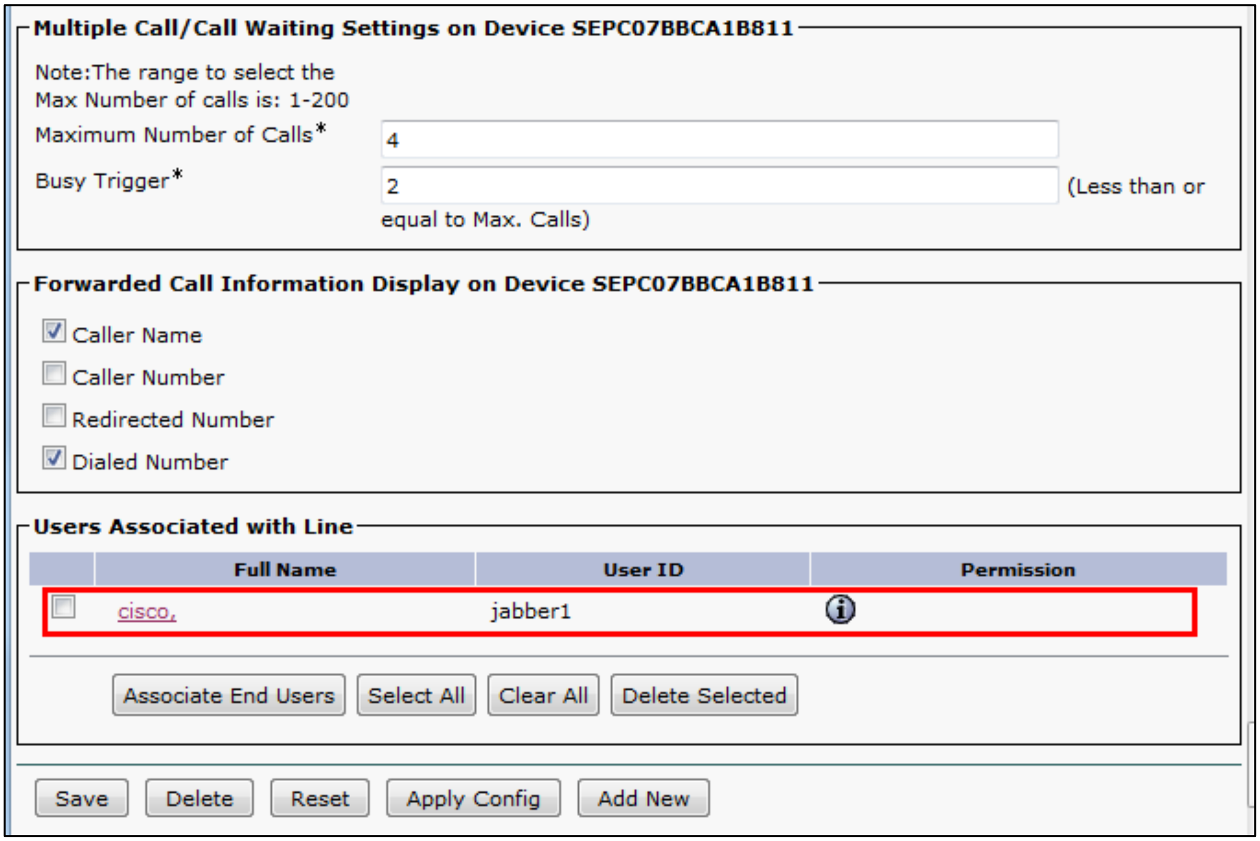
Cisco IP Phone 9971 SIP Configuration (Continued…)

Set Display (caller ID) = Cisco9971-Phone. This is used in this example.
Set ASCII Display (caller ID) = Cisco9971-Phone. This is used in this example.
Set Line Text Label = Cisco9971-Phone. This is used in this example.
Set External Phone Number Mask = 7323204084. This is used in this example.





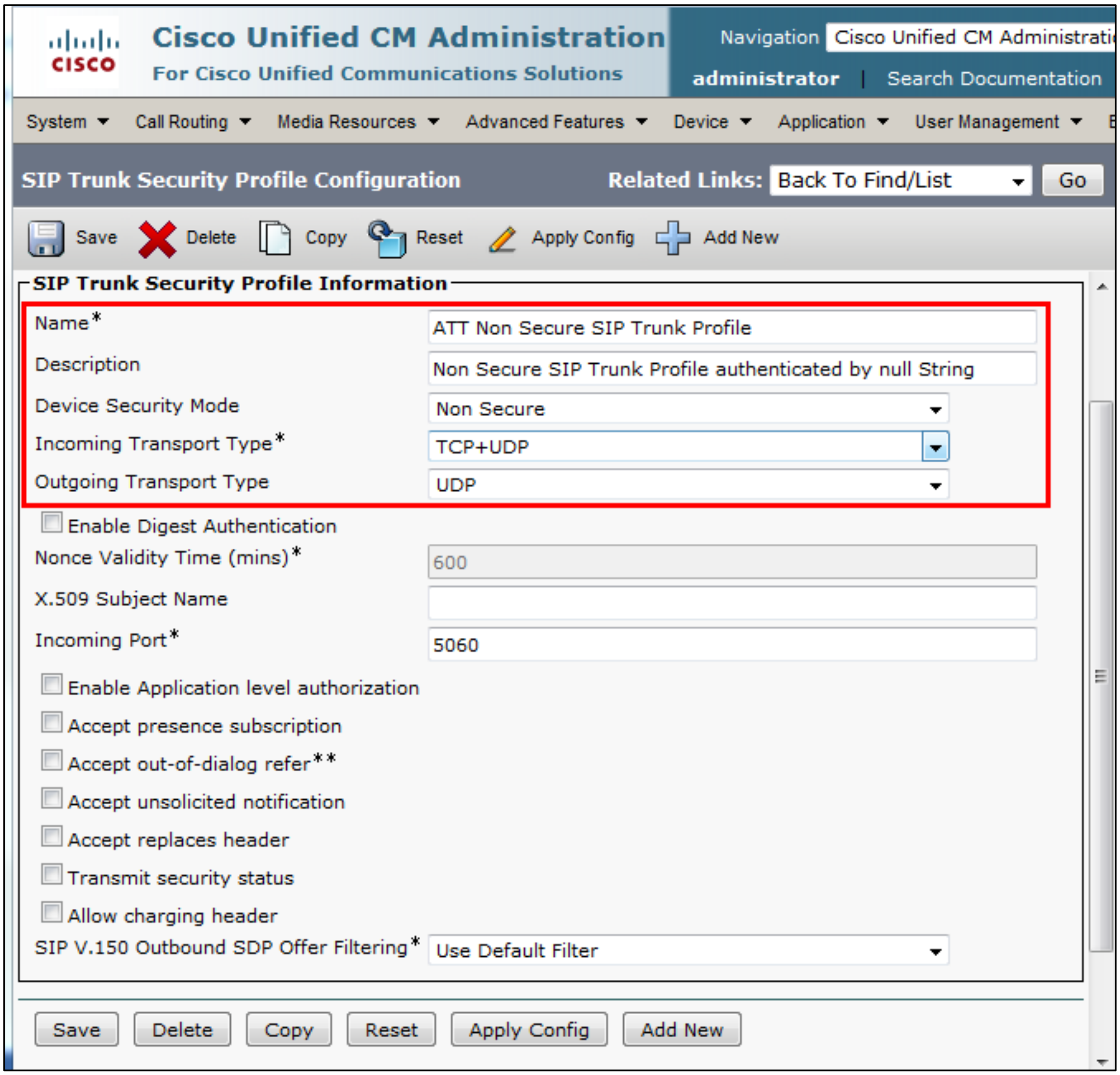
Cisco IP Phone 9971 SIP Configuration (Continued…)


Multiple Call/Call Waiting Settings on Device SEPC07BBCA1B811
Note:The range to select the Max Number of calls is: 1-200
Maximum Number of Calls*
4
Busy Trigger*
2
(Less than or equal to Max. Calls)
Forwarded Call Information Display on Device SEPC07BBCA1B811
Caller Name
Caller Number
Redirected Number
Dialed Number
Users Associated with Line
Full Name
User ID
Permission
cisco,
jabber1
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

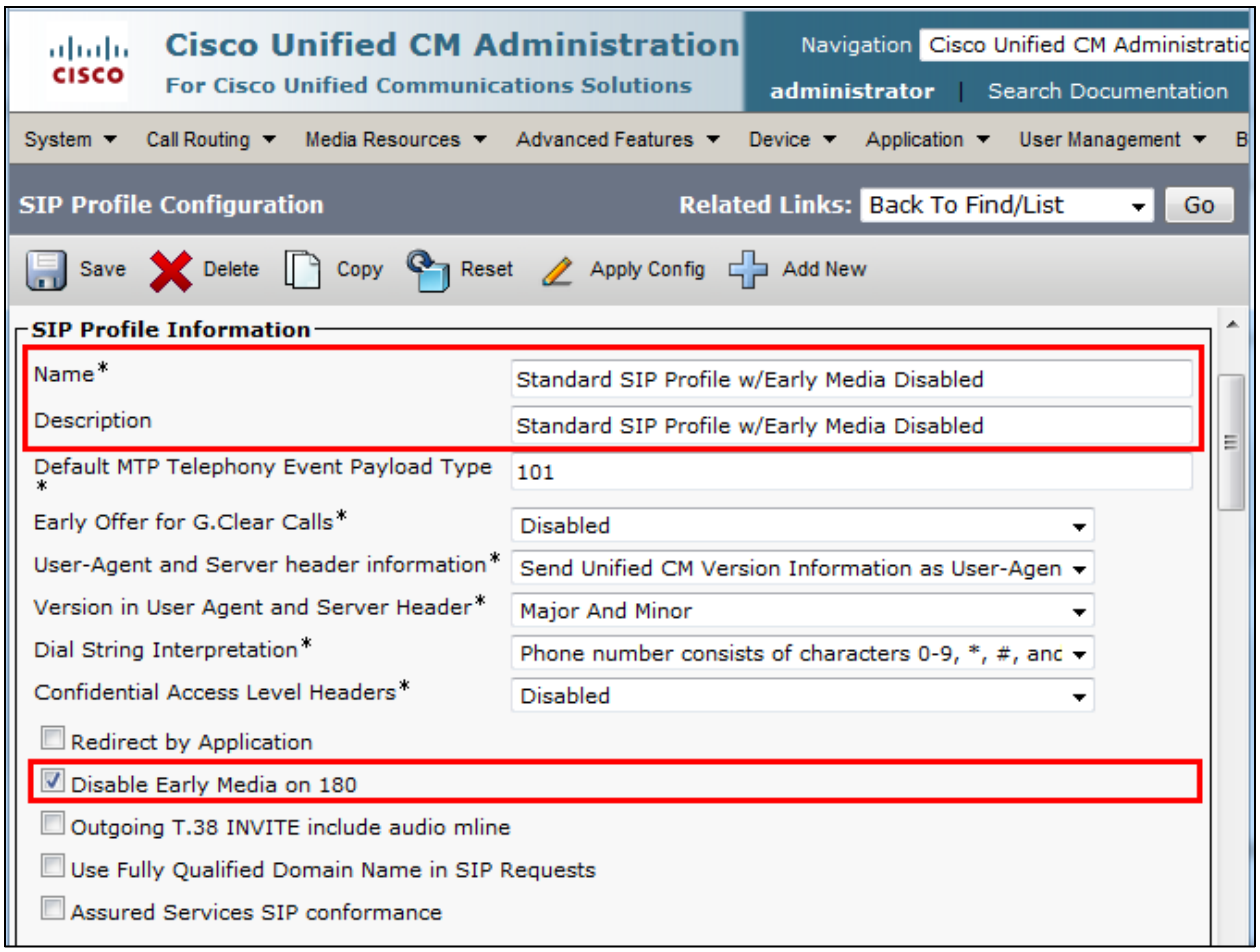
Set Name* = ATT Non Secure SIP Trunk Profile. This is used in this example.
Set Description = Non Secure SIP Trunk Profile authenticated by null String. This is used in this example.
Set Device Security Mode = Non Secure.
Set Incoming Transport Type* = TCP+UDP.
Set Outgoing Transport Type = UDP.



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

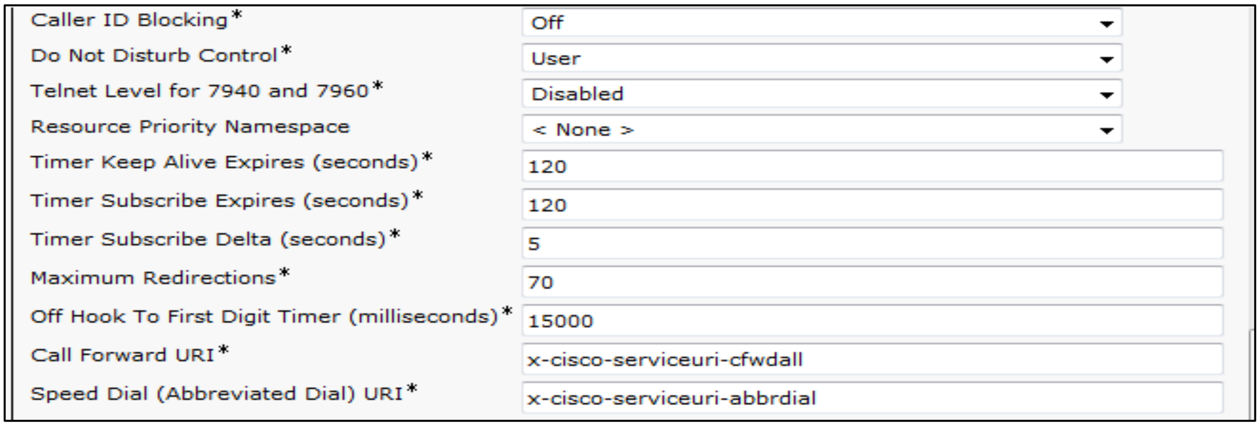
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer *	Default

☐ Require SDP Inactive Exchange for Mid-Call Media Change

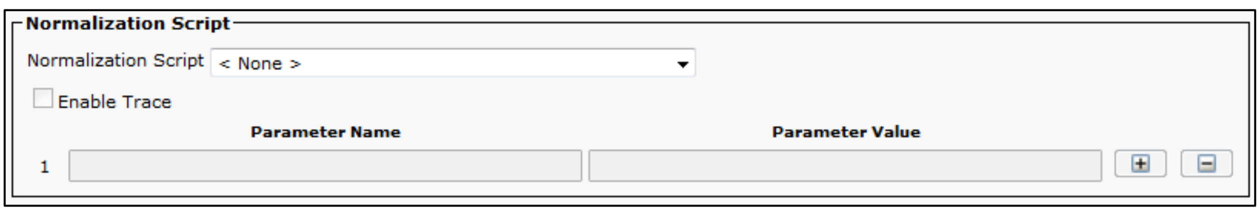
☐ Allow RR/RS bandwidth modifier (RFC 3556)

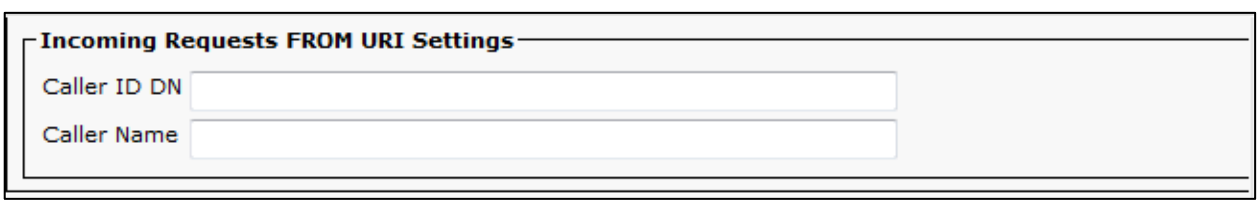
Parameters used in Phone

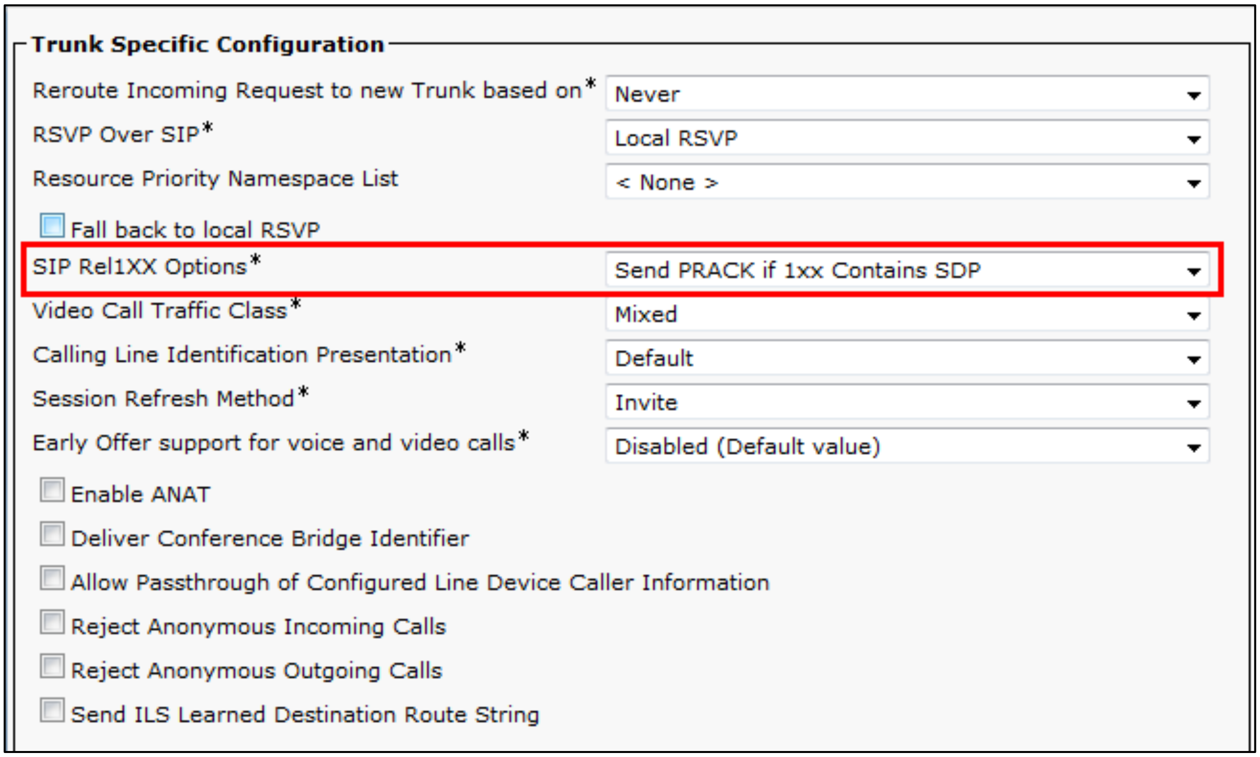
Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other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



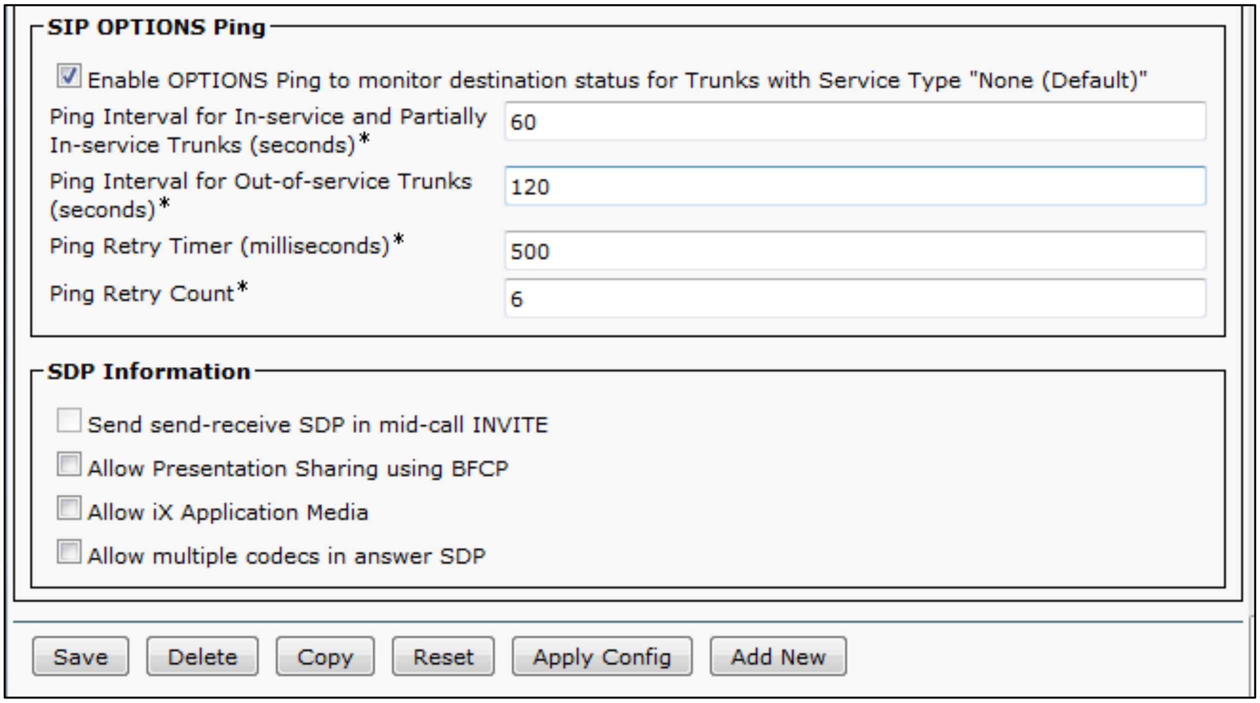
SIP Profile Configuration used by SIP trunk to Cisco UBE (Continued…)







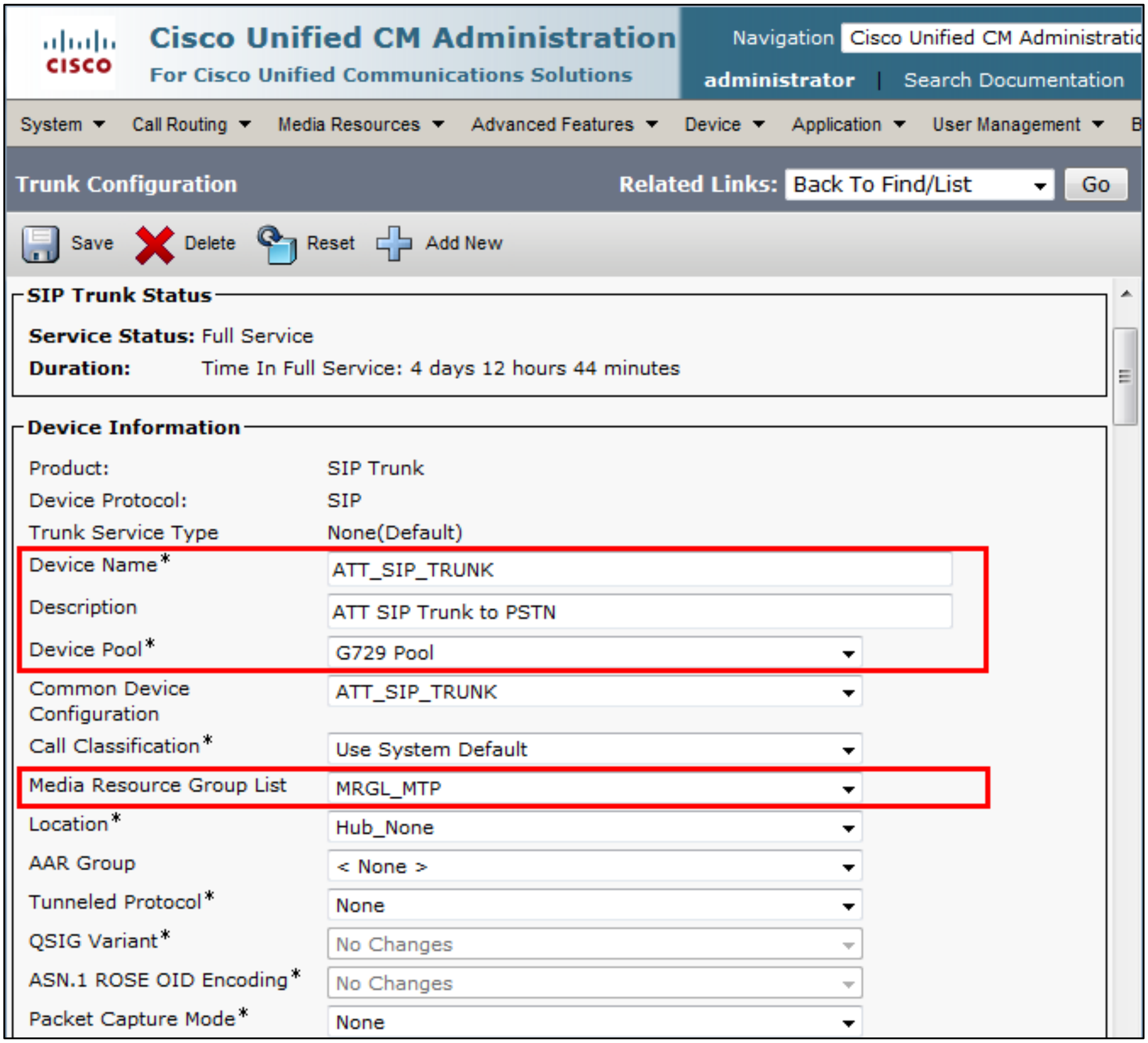
SIP Profile Configuration used by SIP trunk to Cisco UBE (Continued…)



SIP Trunk to Cisco UBE Configuration

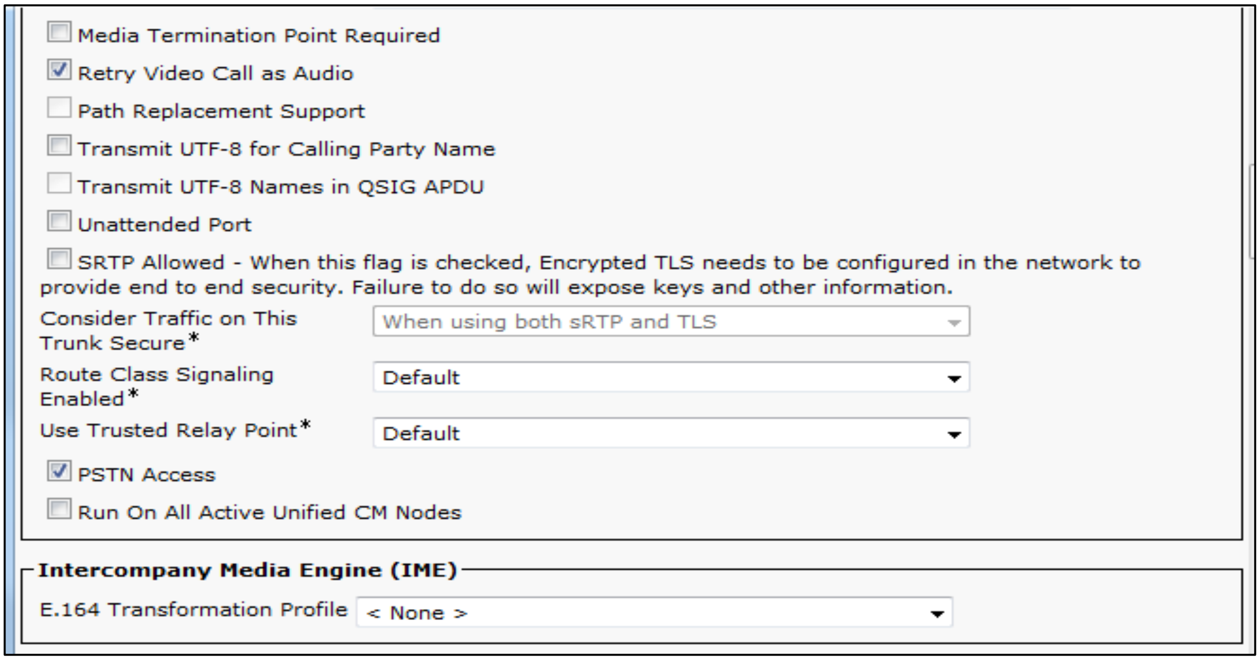
Navigation: Device → Trunk

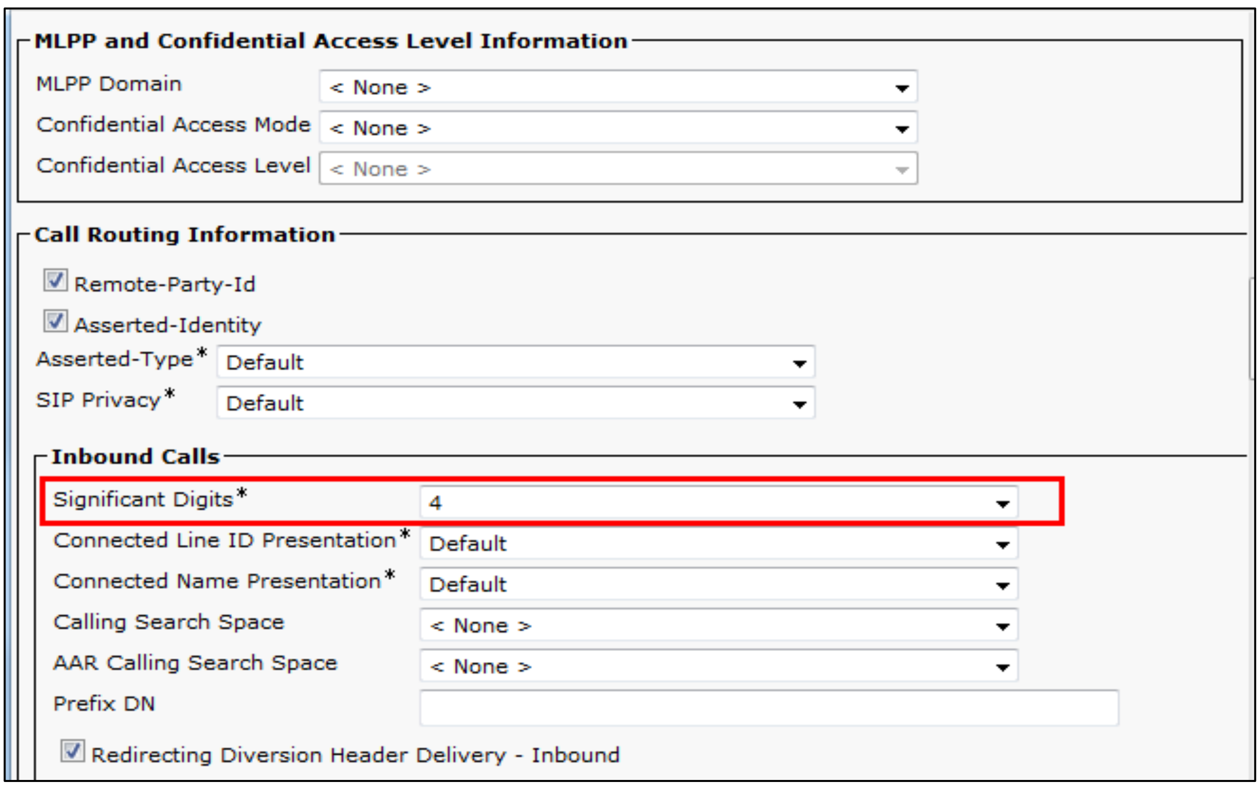
Set Device Name* = ATT_SIP_TRUNK. This is used for this example
Set Description = ATT SIP Trunk to PSTN. This is used for this example
Set Device Pool* = G729_pool. This is used for this example
Set Media Resource Group List = MRGL_MTP.

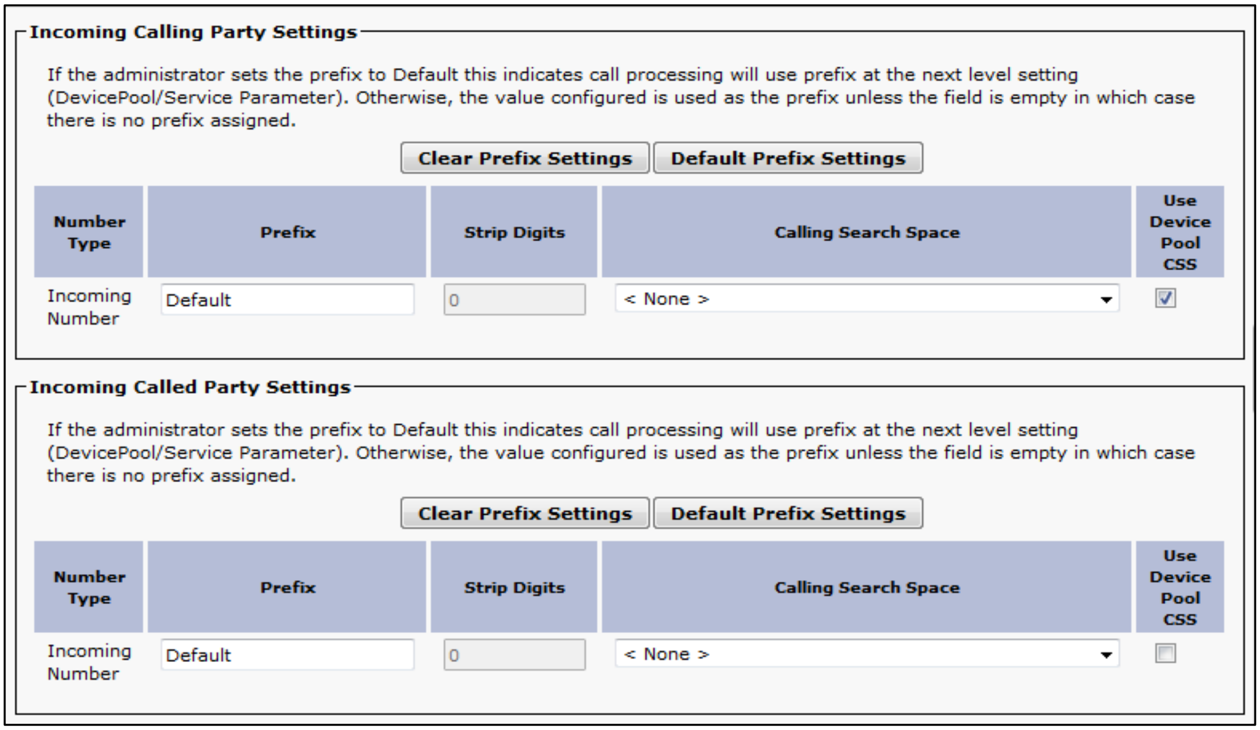


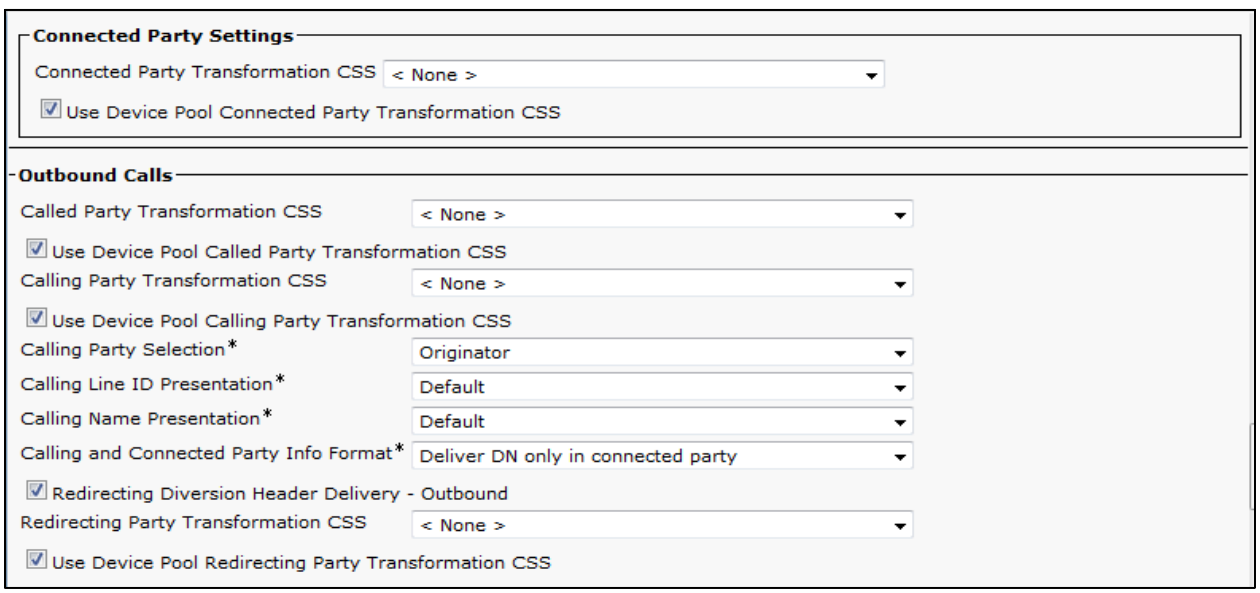
SIP Trunk to Cisco UBE Configuration (Continued…)

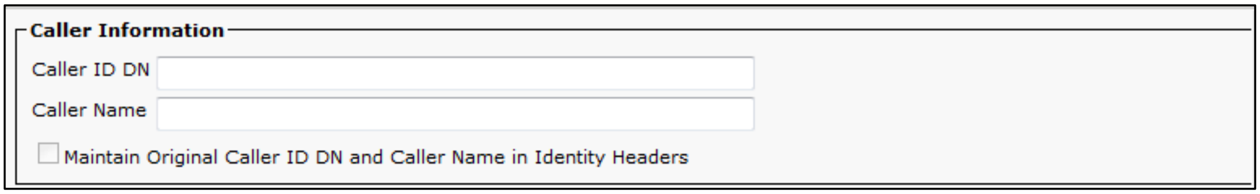
Set Significant Digits* = 4. This is used in this example.





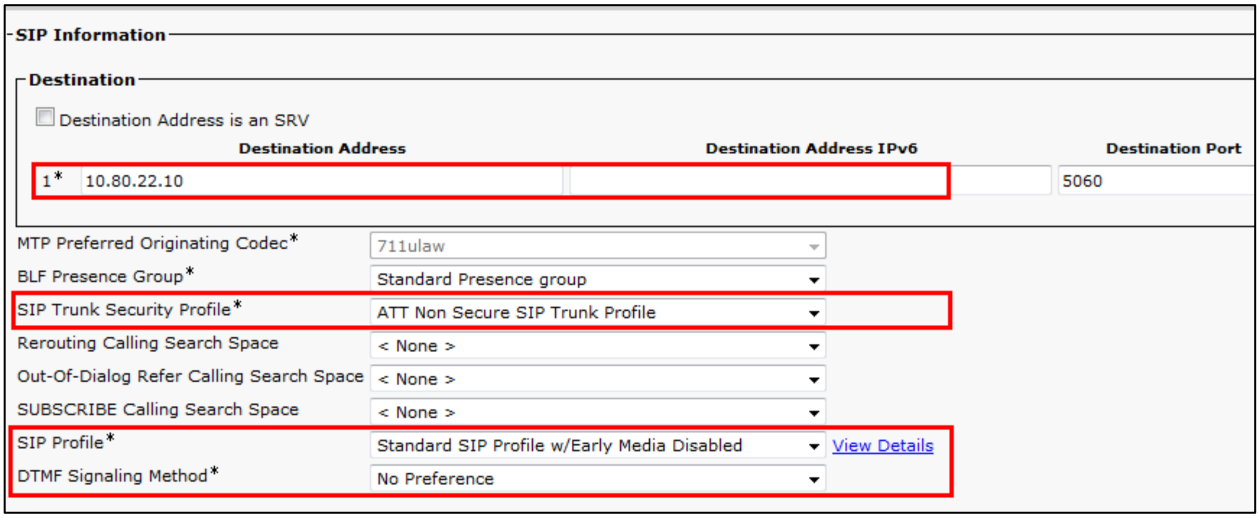

Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.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

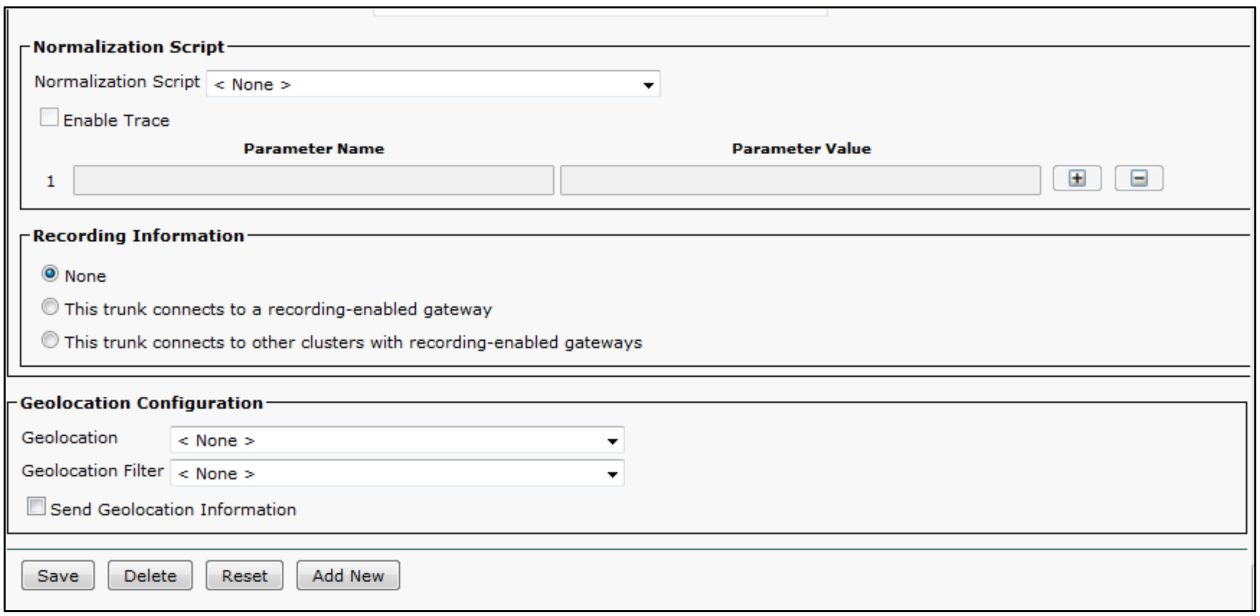

Connected Party Settings
Connected Party Transformation CSS
< None >
Use Device Pool Connected Party Transformation CSS
Outbound Calls
Called Party Transformation CSS
< None >
Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS
< None >
Use Device Pool Calling Party Transformation 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



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.

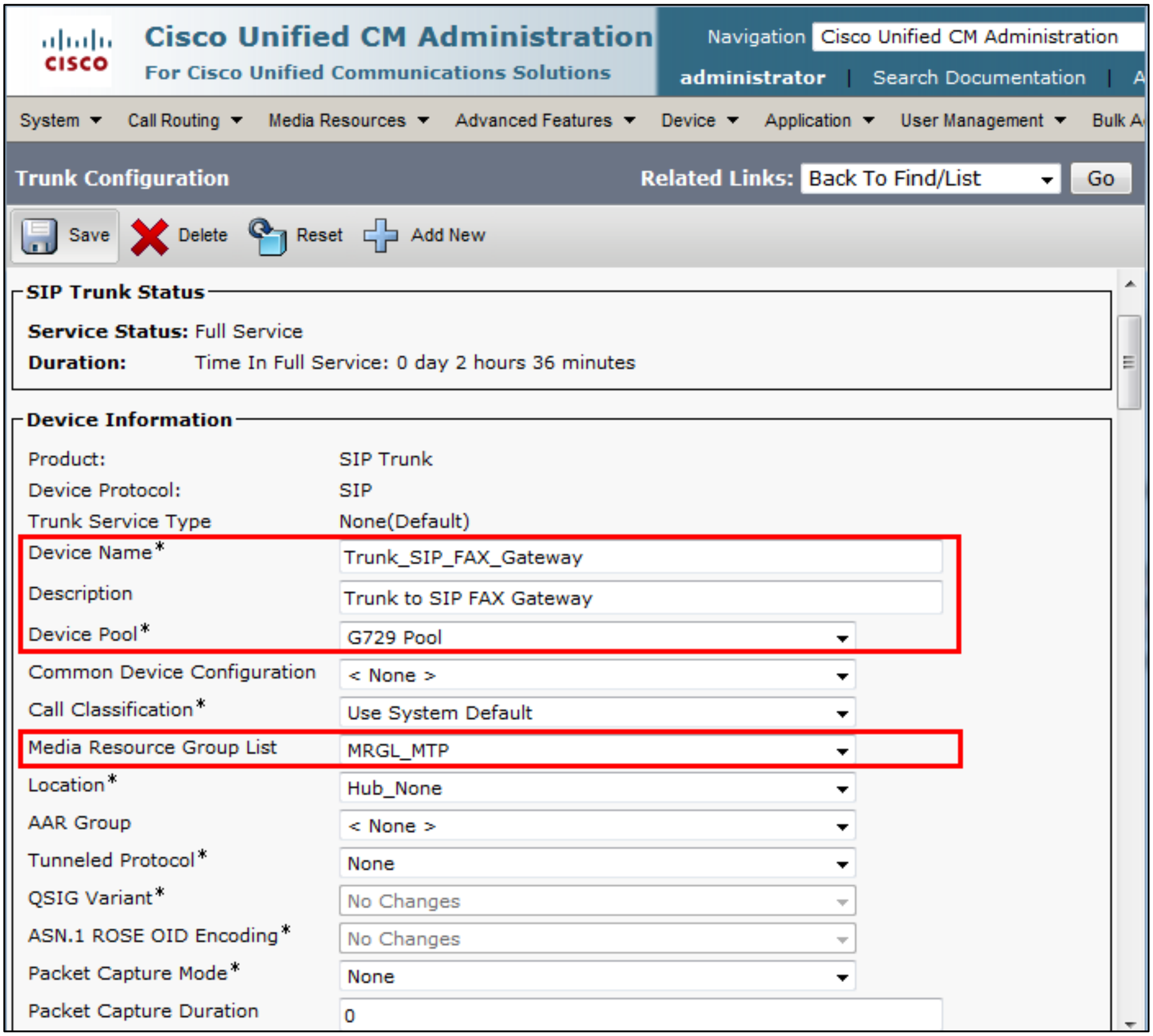




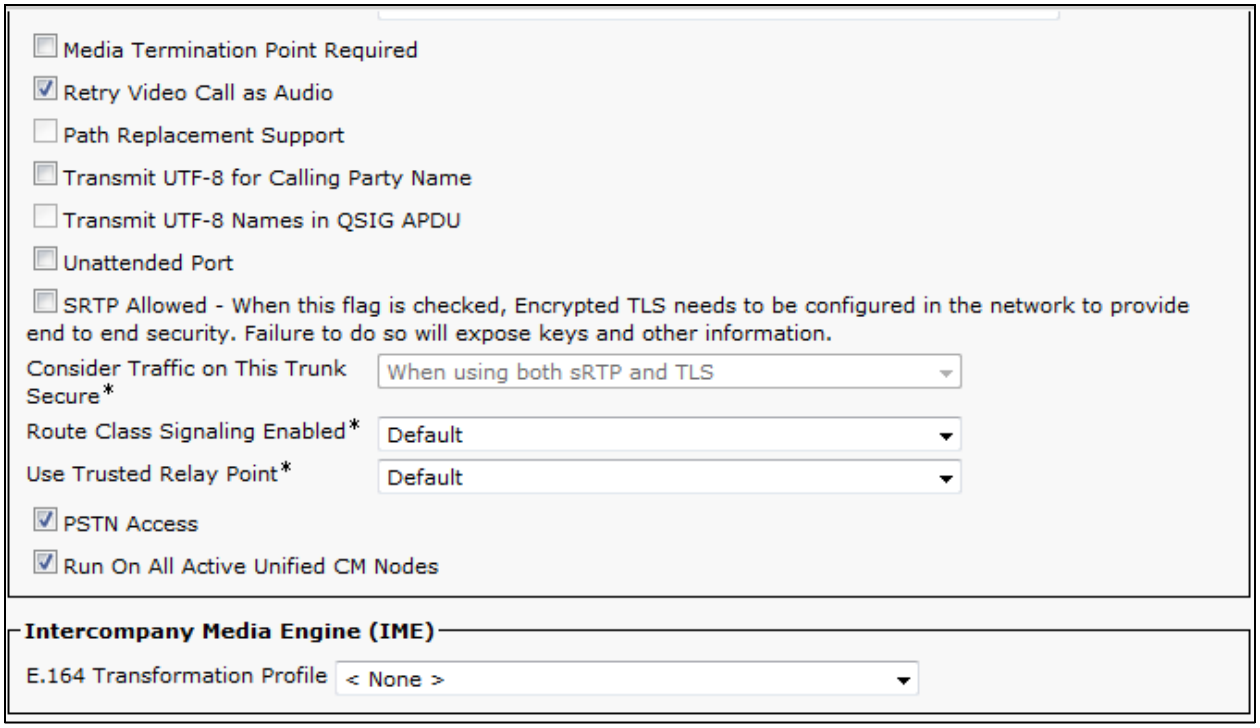
SIP Trunk to Fax Gateway Configuration.

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…)



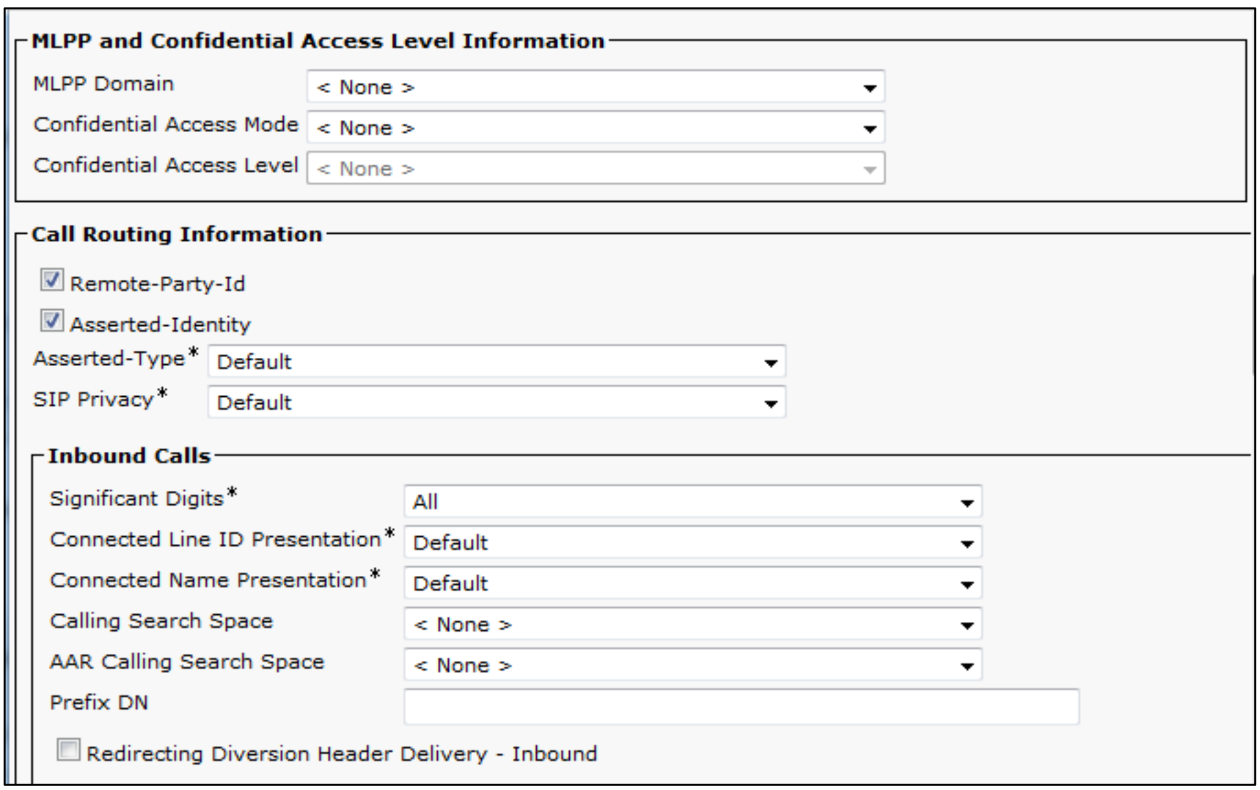
- [ ] Media Termination Point Required
- [x] Retry Video Call as Audio
- [ ] Path Replacement Support
- [ ] Transmit UTF-8 for Calling Party Name
- [ ] Transmit UTF-8 Names in QSIG APDU
- [ ] Unattended Port
- [ ] SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network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

- [x] PSTN Access
- [x] 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

- [x] Remote-Party-Id
- [x] Asserted-Identity

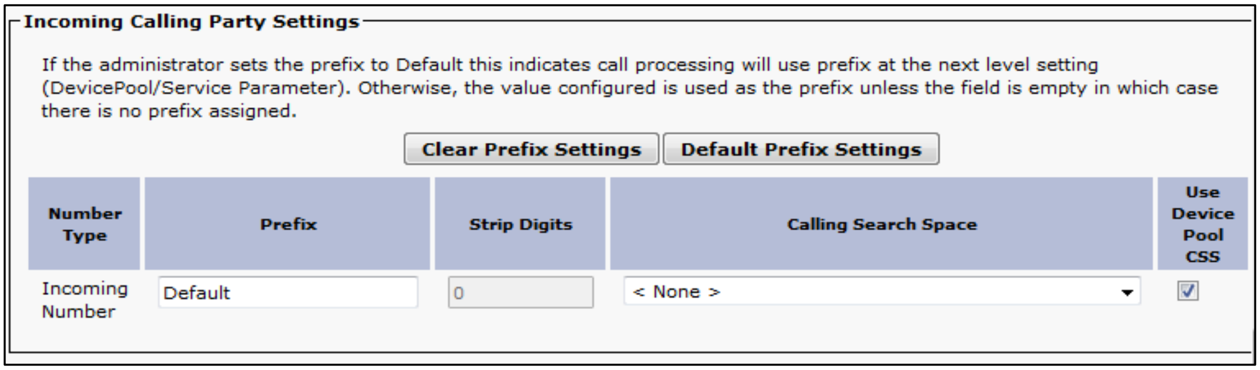
Asserted-Type*	Default
SIP Privacy*	Default

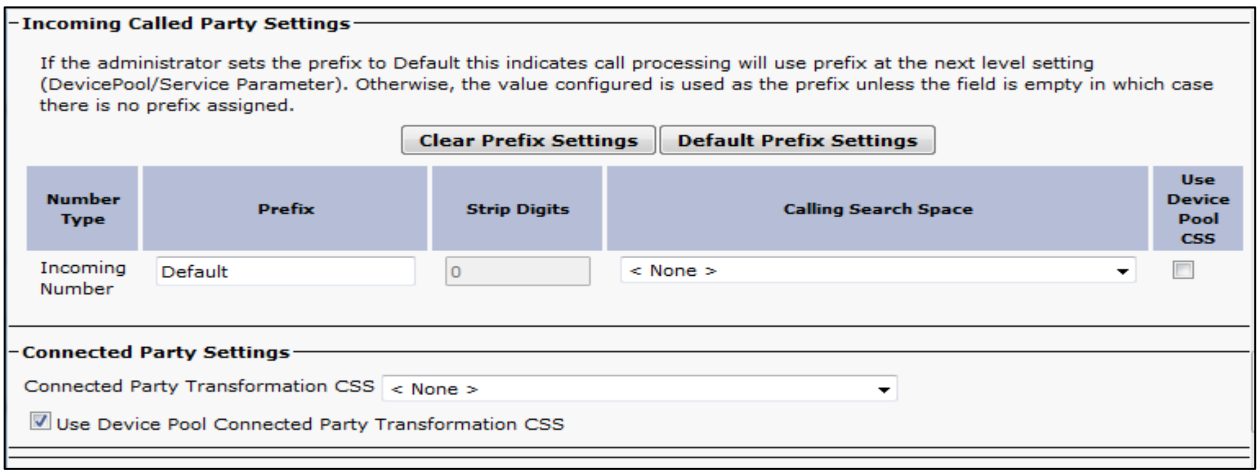
Inbound Calls

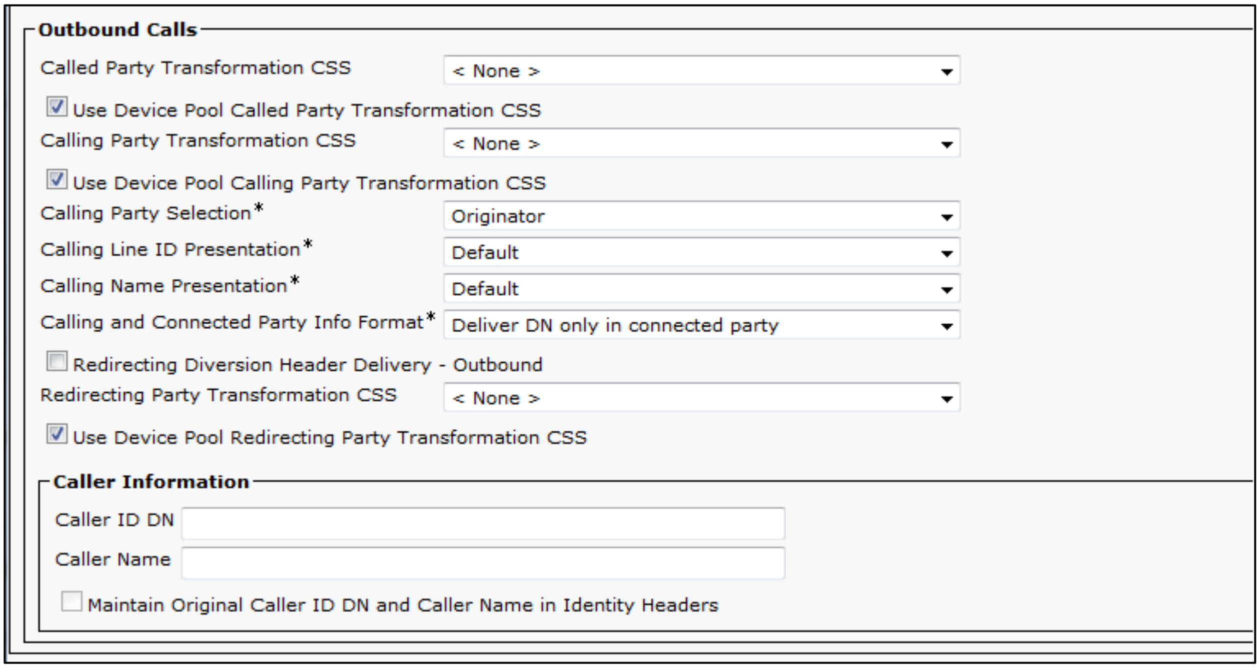
Significant Digits*	All
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	

- [ ] Redirecting Diversion Header Delivery - Inbound

SIP Trunk to Fax Gateway Configuration (Continued…)







Outbound Calls

Called Party Transformation CSS: < None >

[x] Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS: < None >

[x] 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 >

[x] Use Device Pool Redirecting Party Transformation CSS

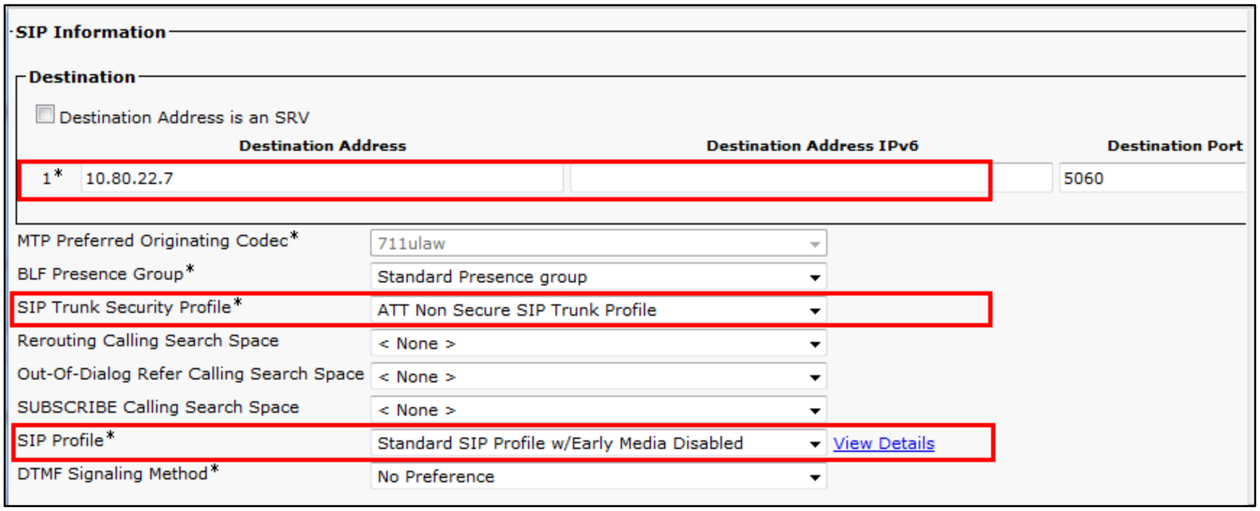
Caller Information

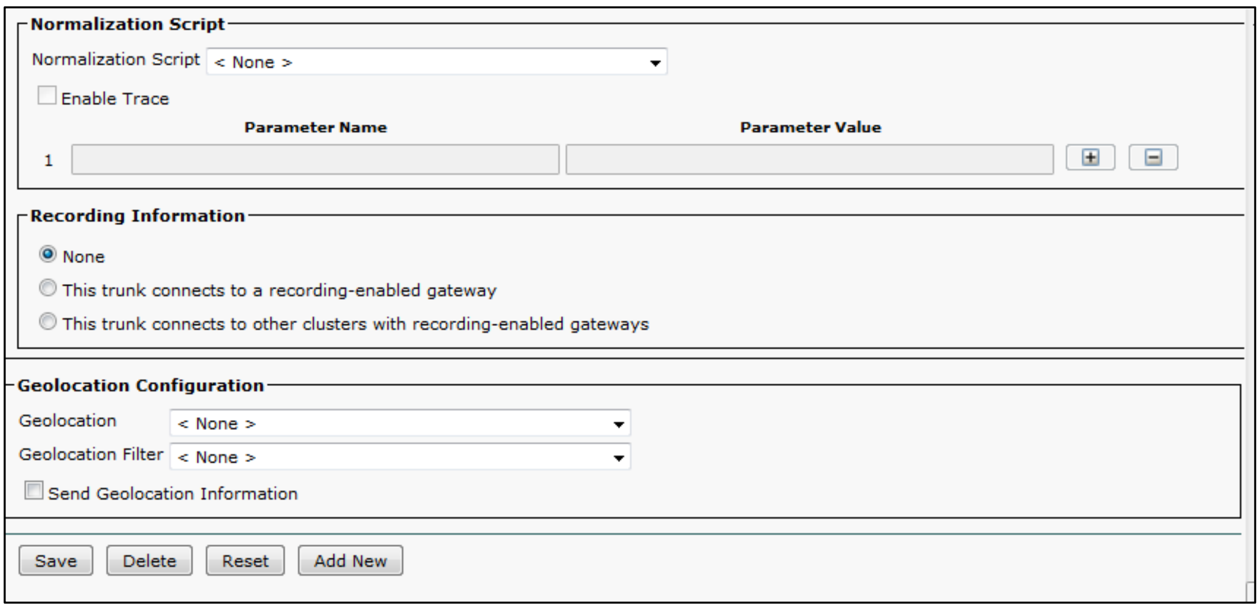
Caller ID DN:

Caller Name:

[ ] Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Trunk to Fax Gateway Configuration (Continued…)



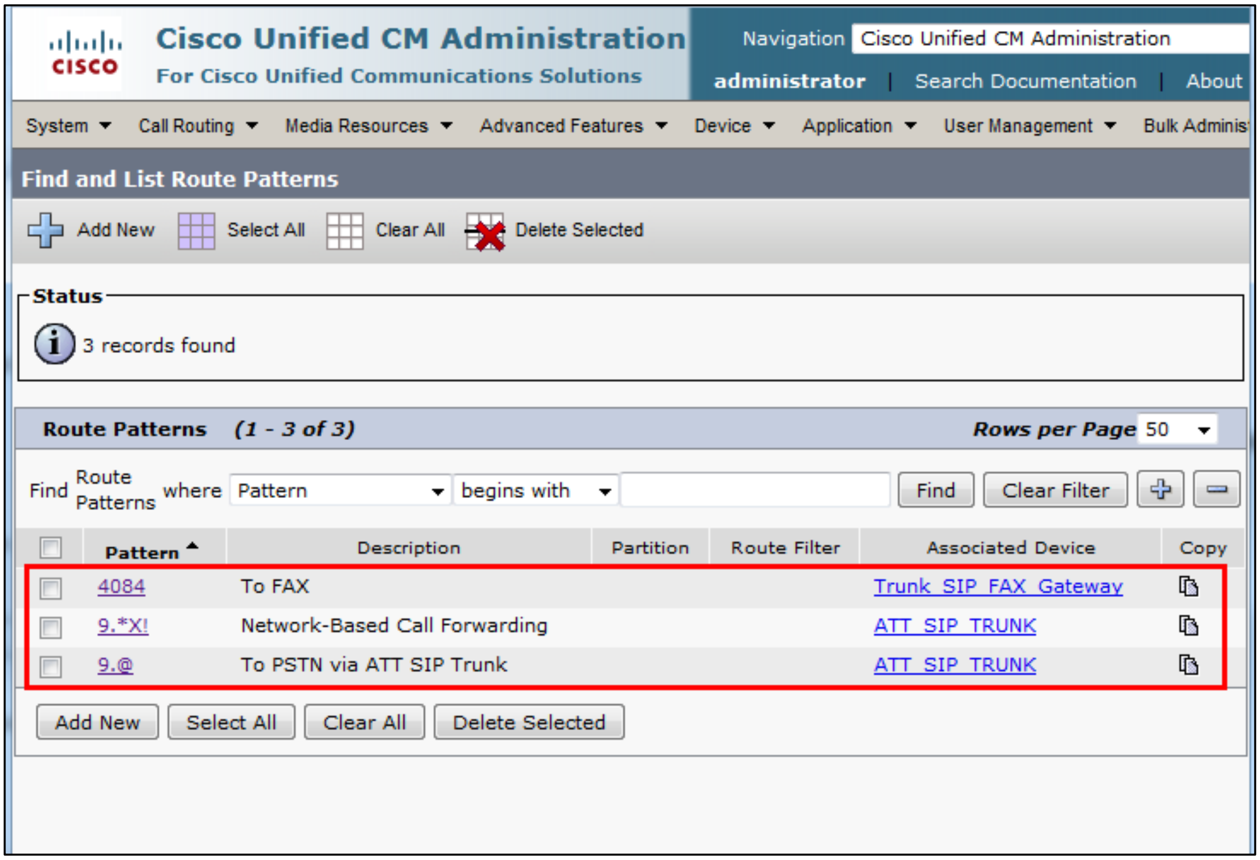


Note: Testing was conducted in tekVizion labs

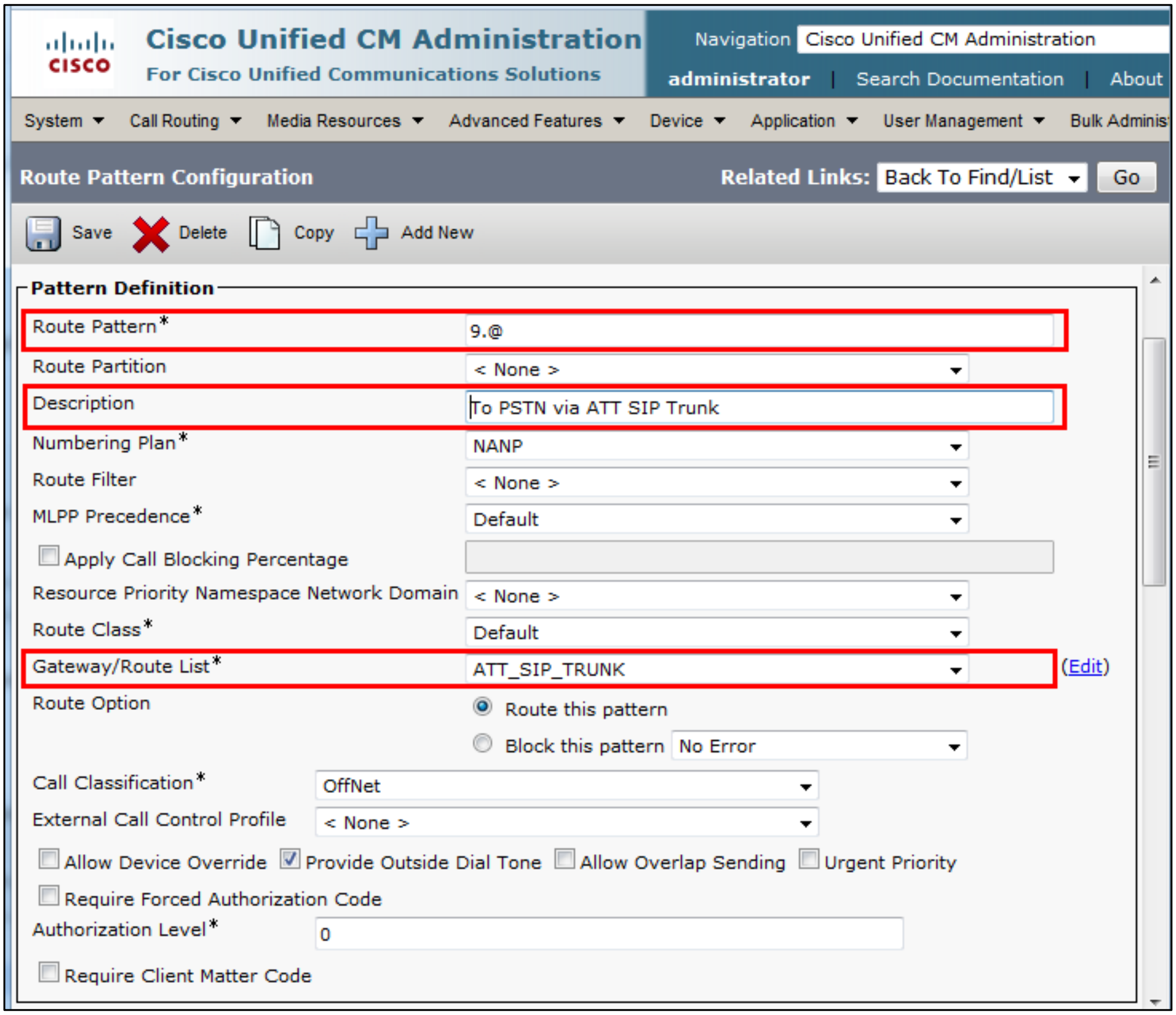
Route Pattern Configuration

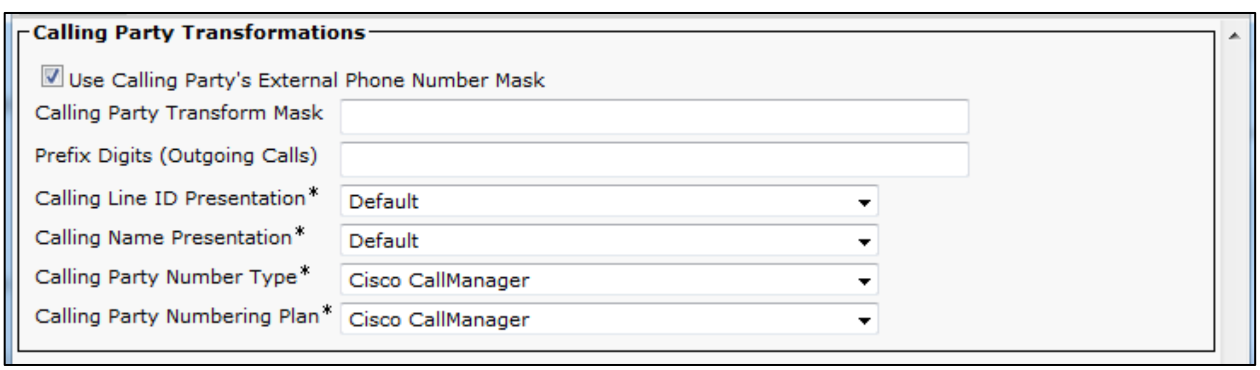
Navigation: Call Routing → Route/Hunt → Route Pattern

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

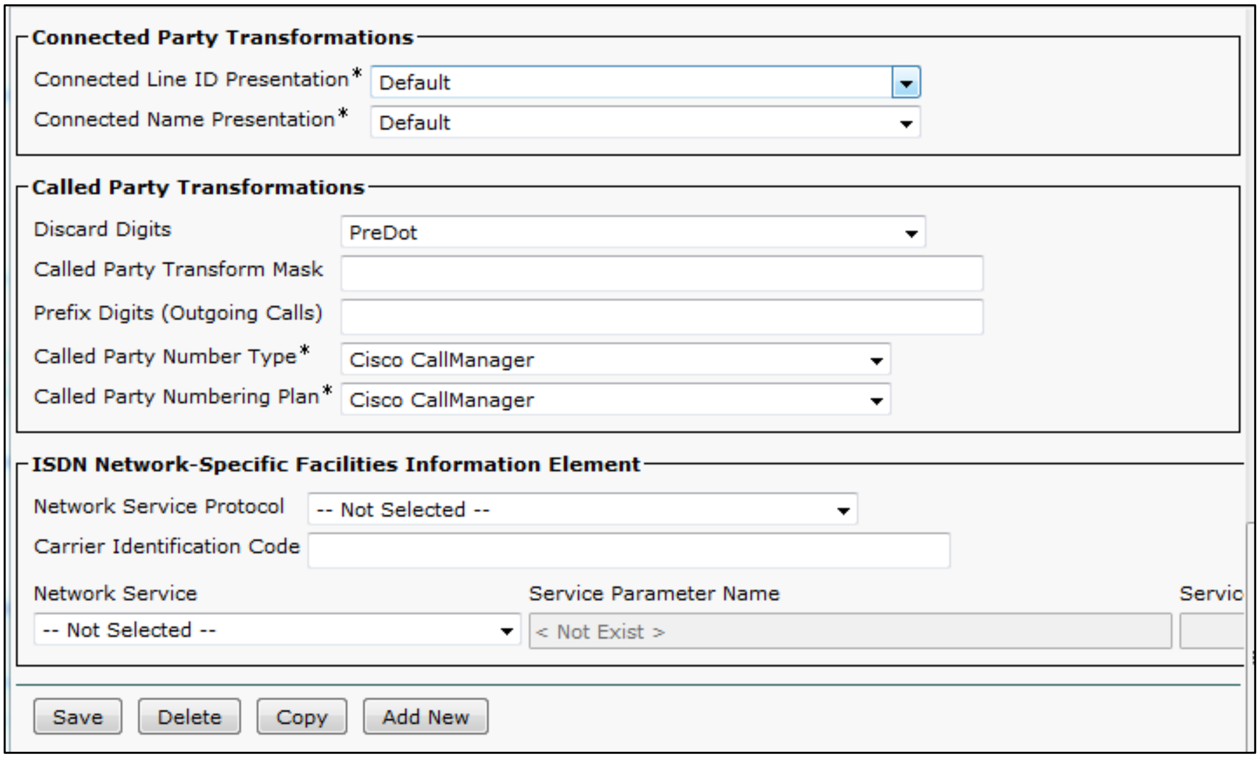


Route Pattern Configuration (Continued…)





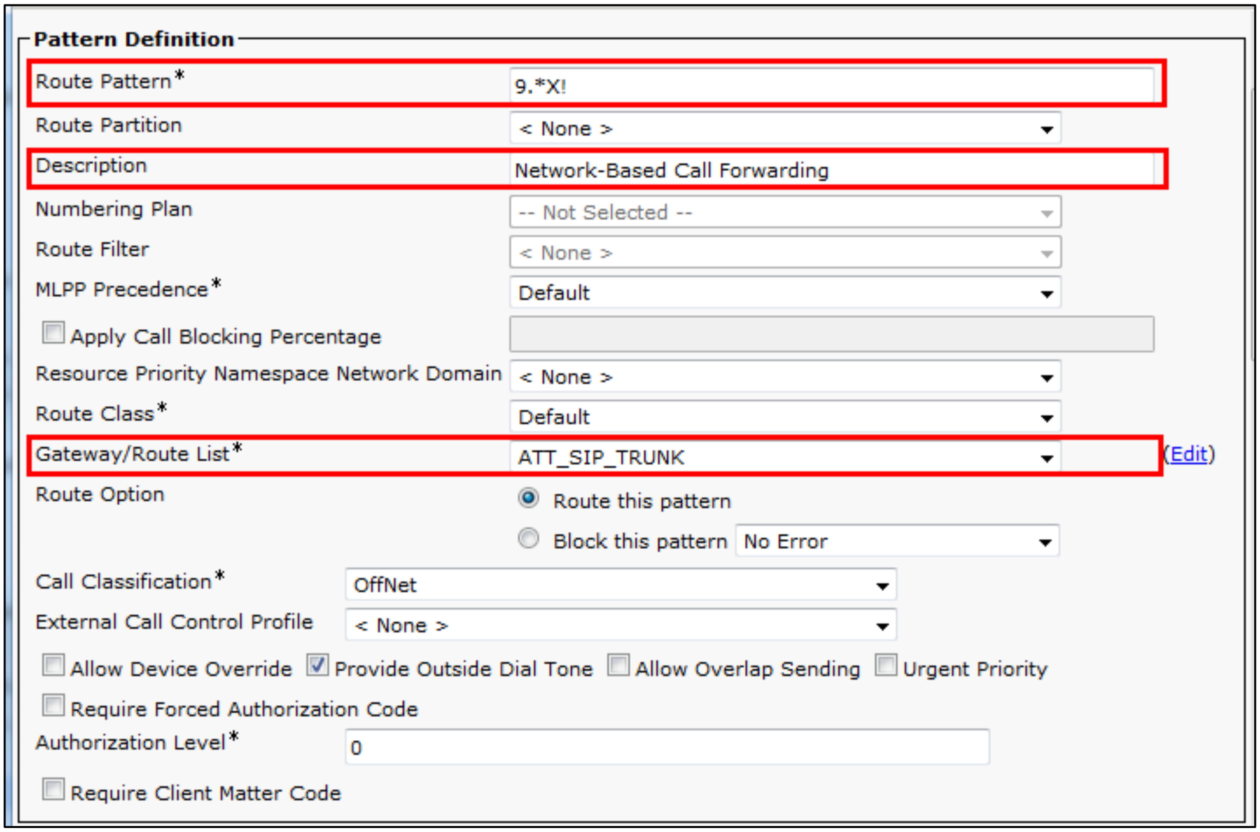
Route Pattern Configuration (Continued…)

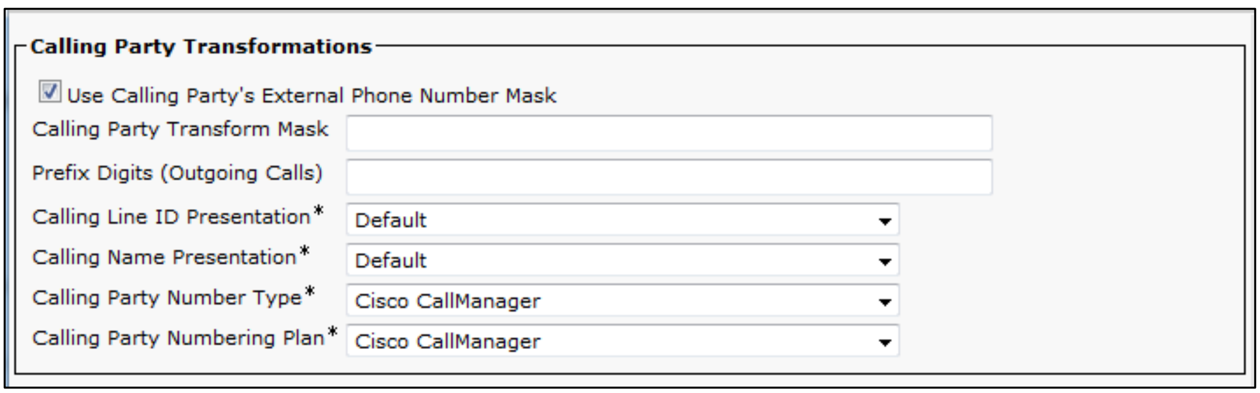


Route Pattern Configuration (Continued…)

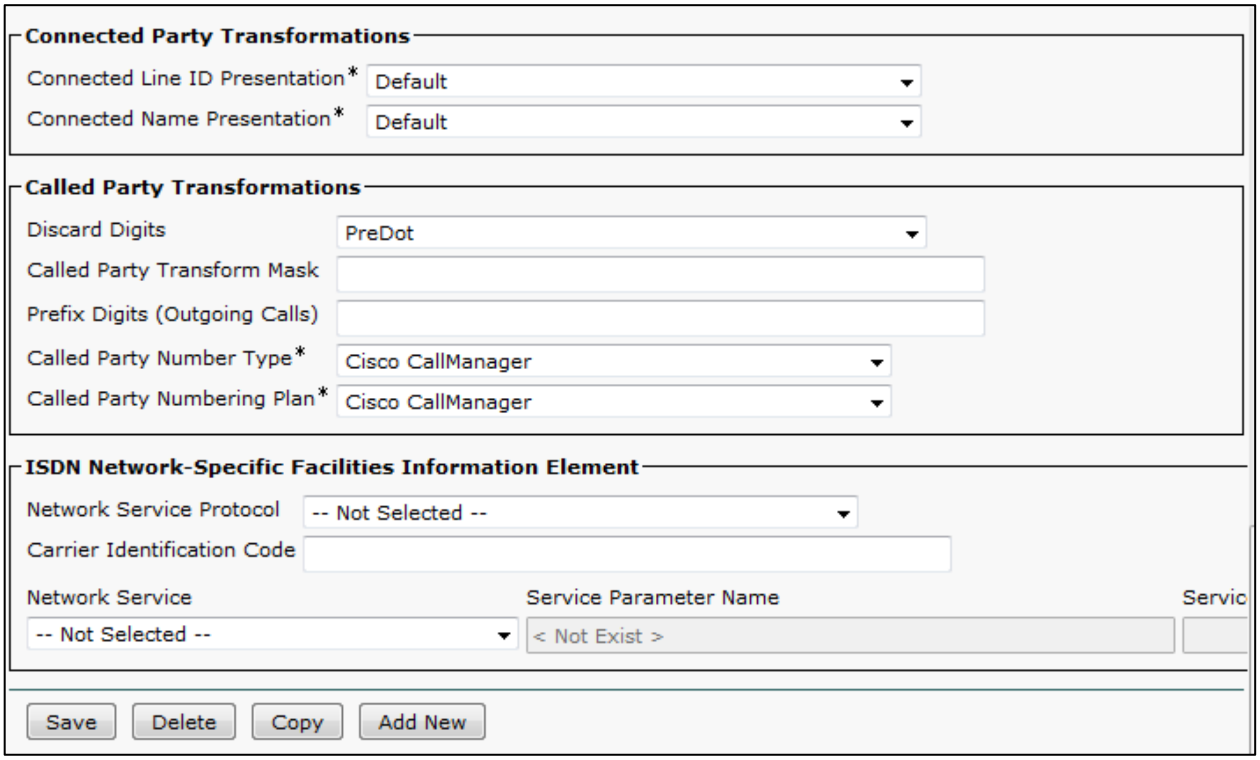
Set Route Pattern* = 9.*X!   This is used to route to AT&T via ISR Cisco UBE.
Set Description = Network-Based Call Forwarding. This text is used to identify this Route Pattern.
Set Gateway/Route List* = ATT_SIP_TRUNK. This is used for this example.
All other values are default

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



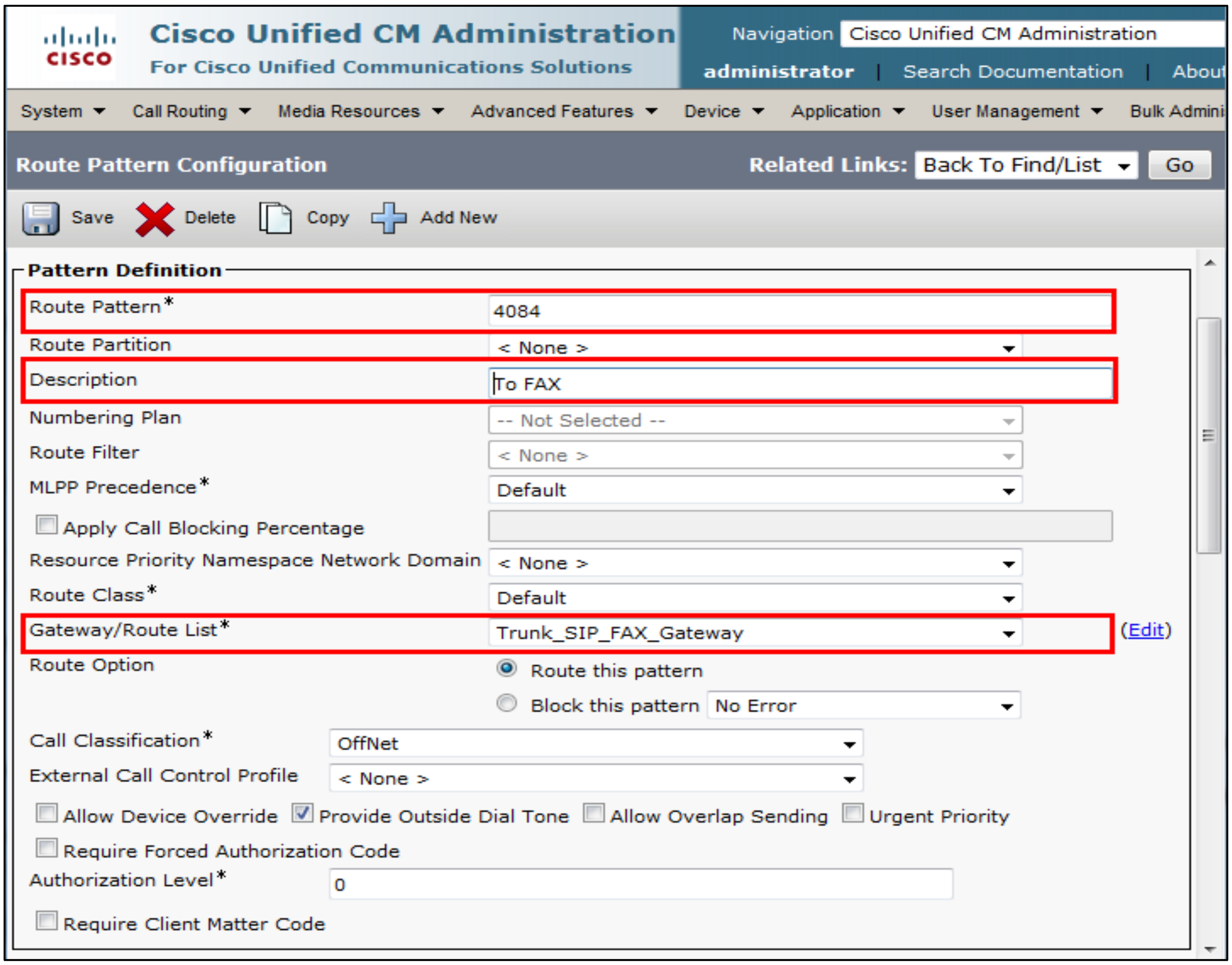


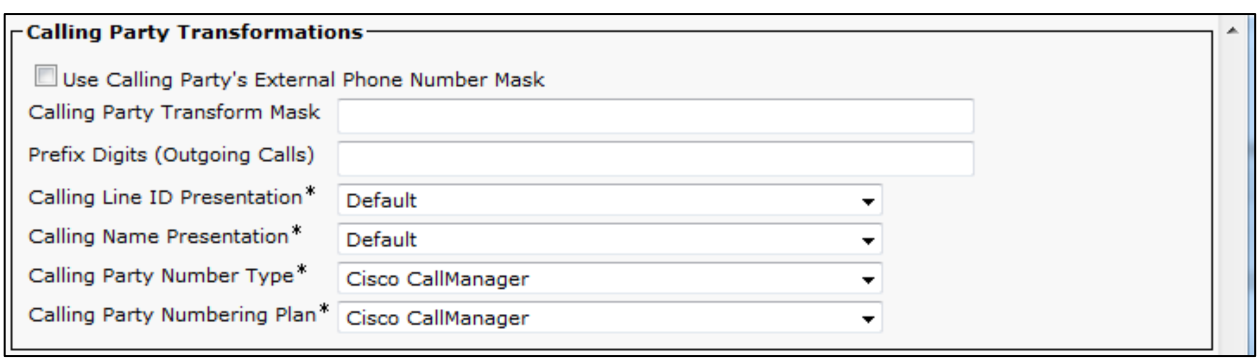
Route Pattern Configuration (Continued…)



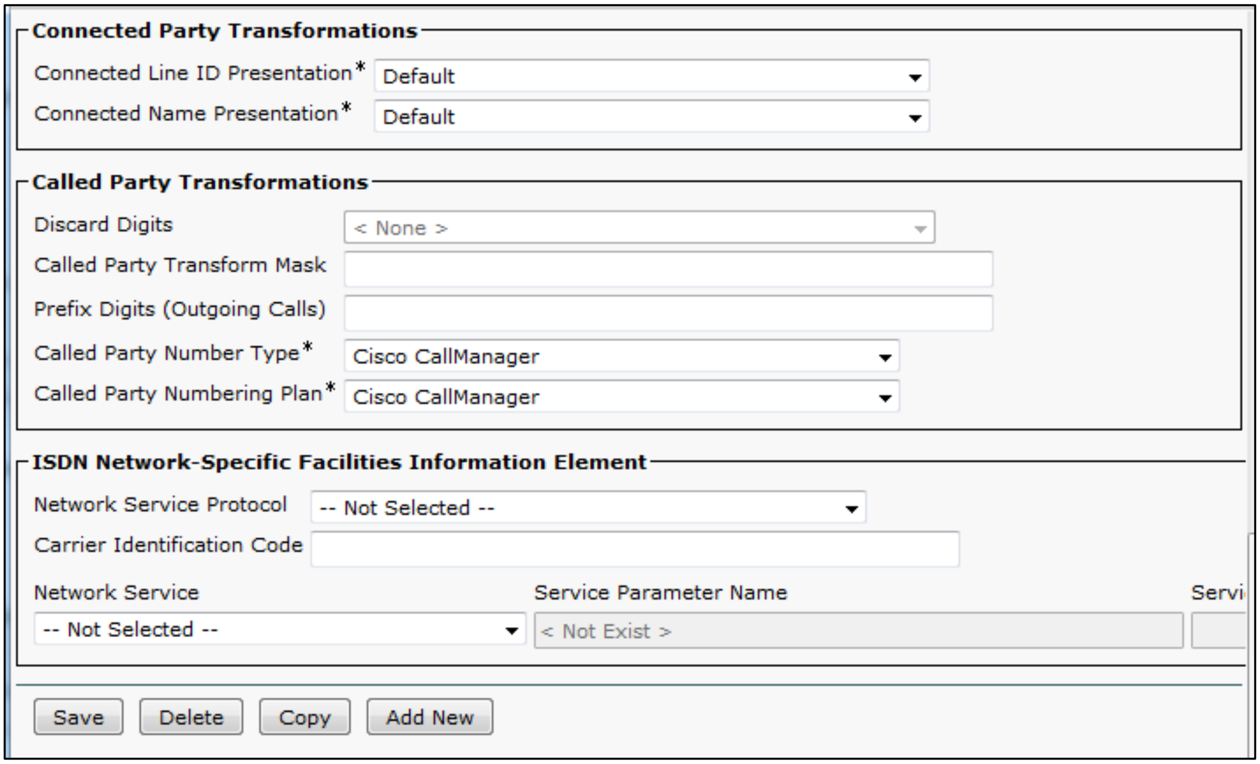
Route Pattern Configuration (Continued…)

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





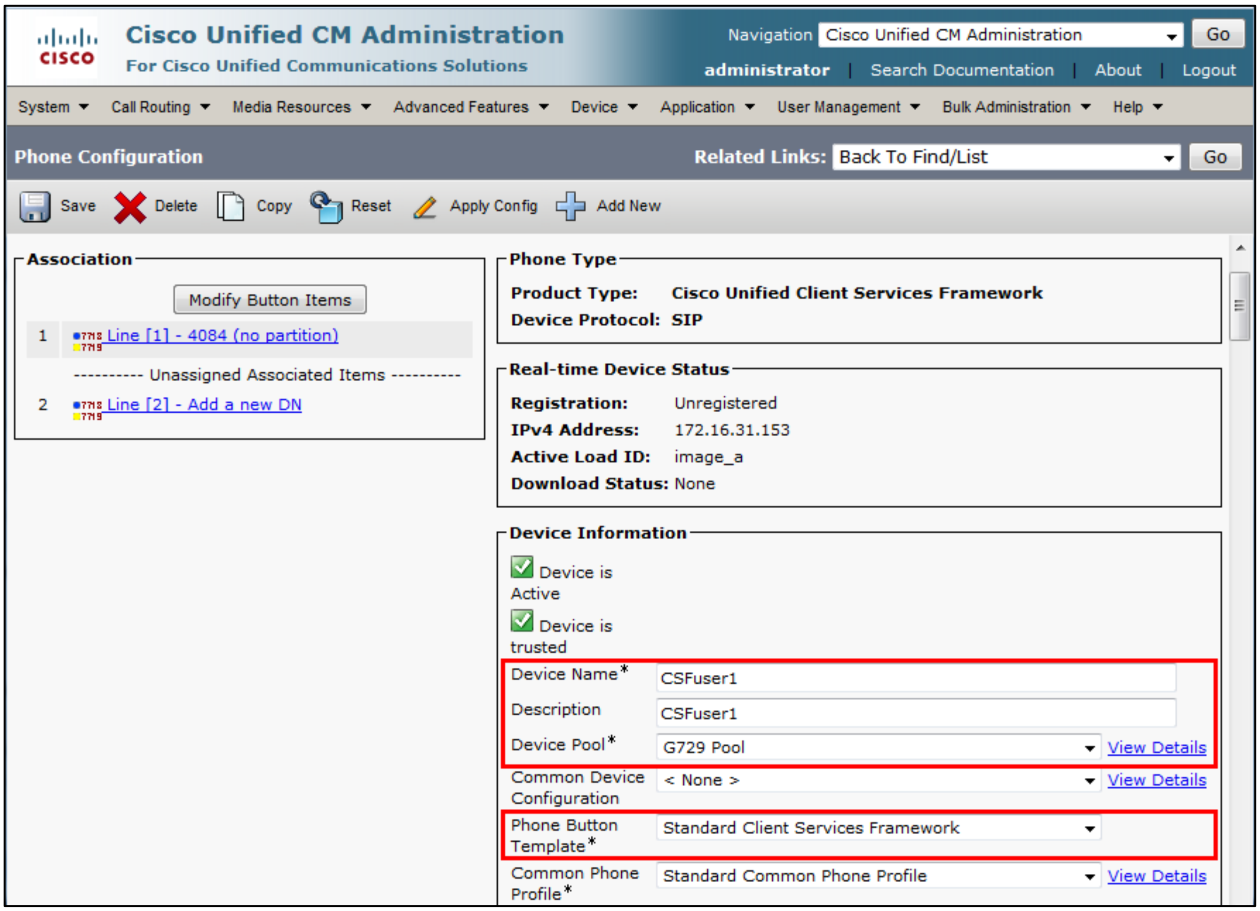
Route Pattern Configuration (Continued…)



Jabber Client Configuration

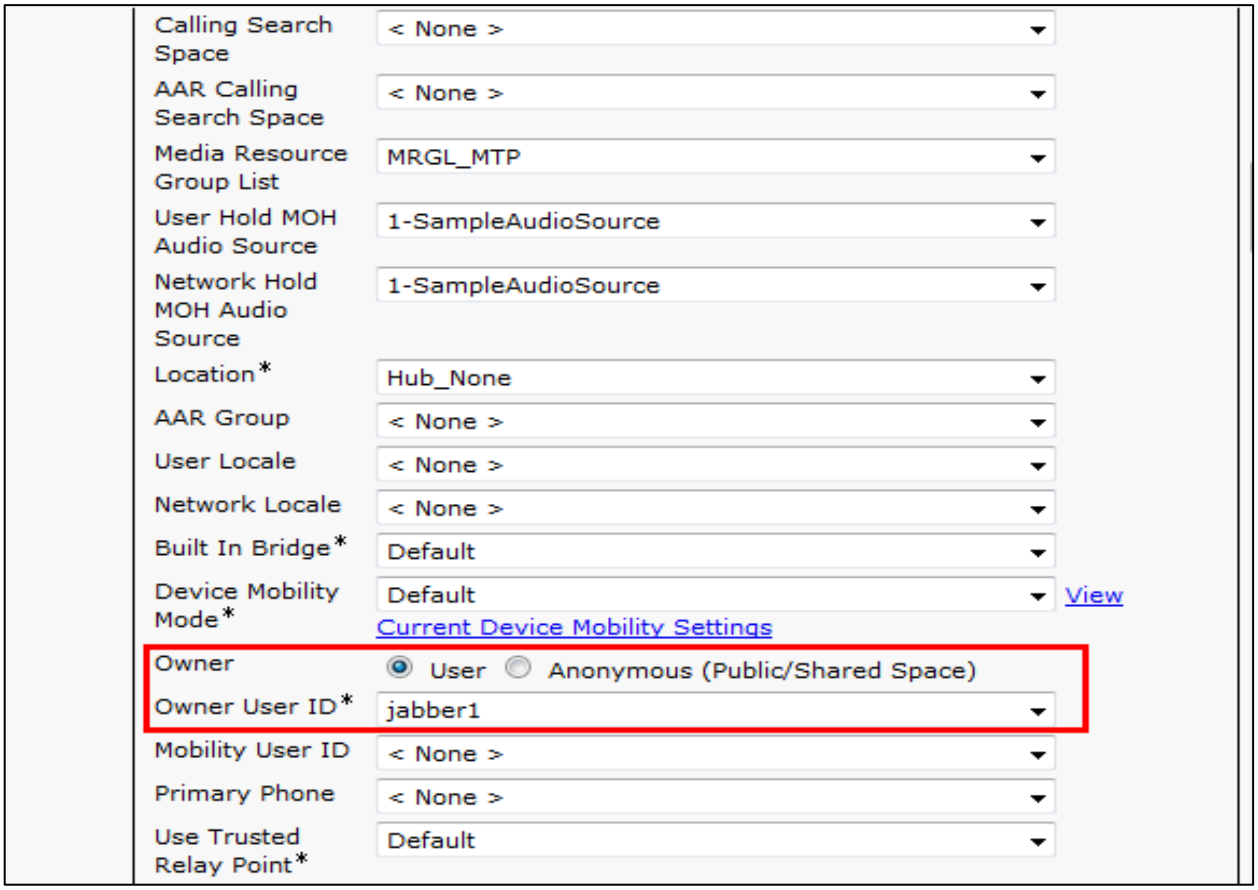
Navigation: Device → Phone

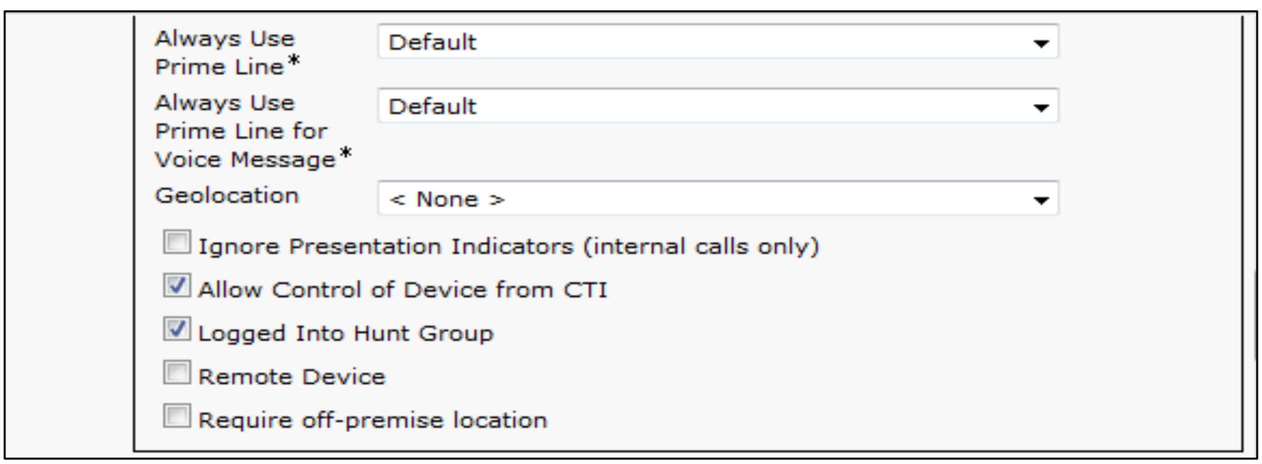
Select Phone Type* = Cisco Unified Client services framework
Set Device Name* = CSFUser1. This is used in this example.
Set Description = CSFUser1. This is used in this example.
Select Device Pool = G729 Pool. This is used in this example.
Select Phone Button Template* = Standard Client Services Framework.



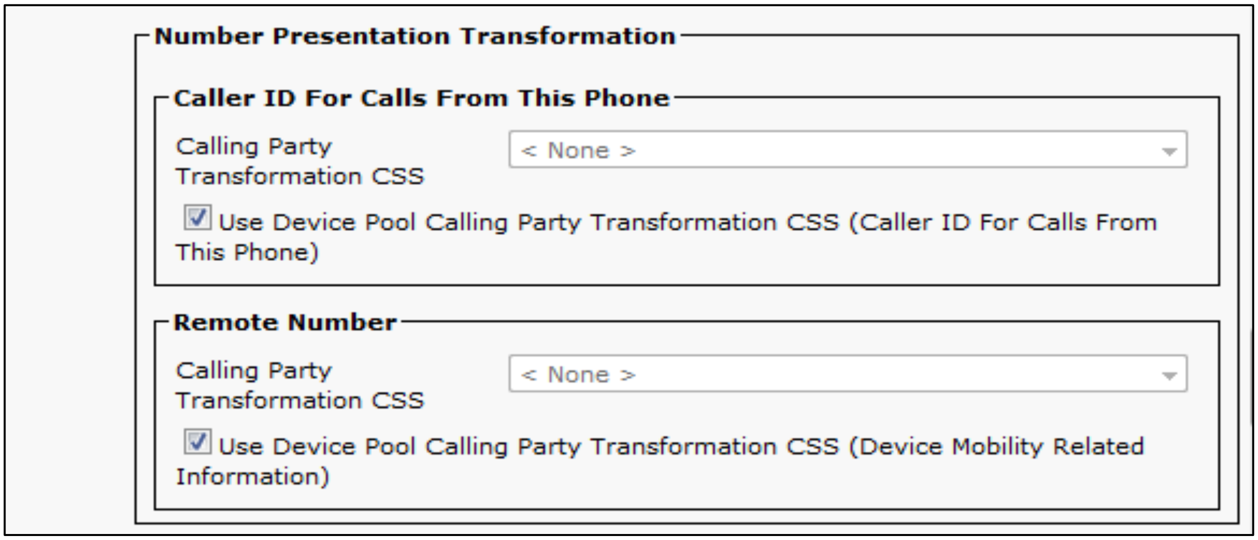
Jabber Client Configuration (Contd…)

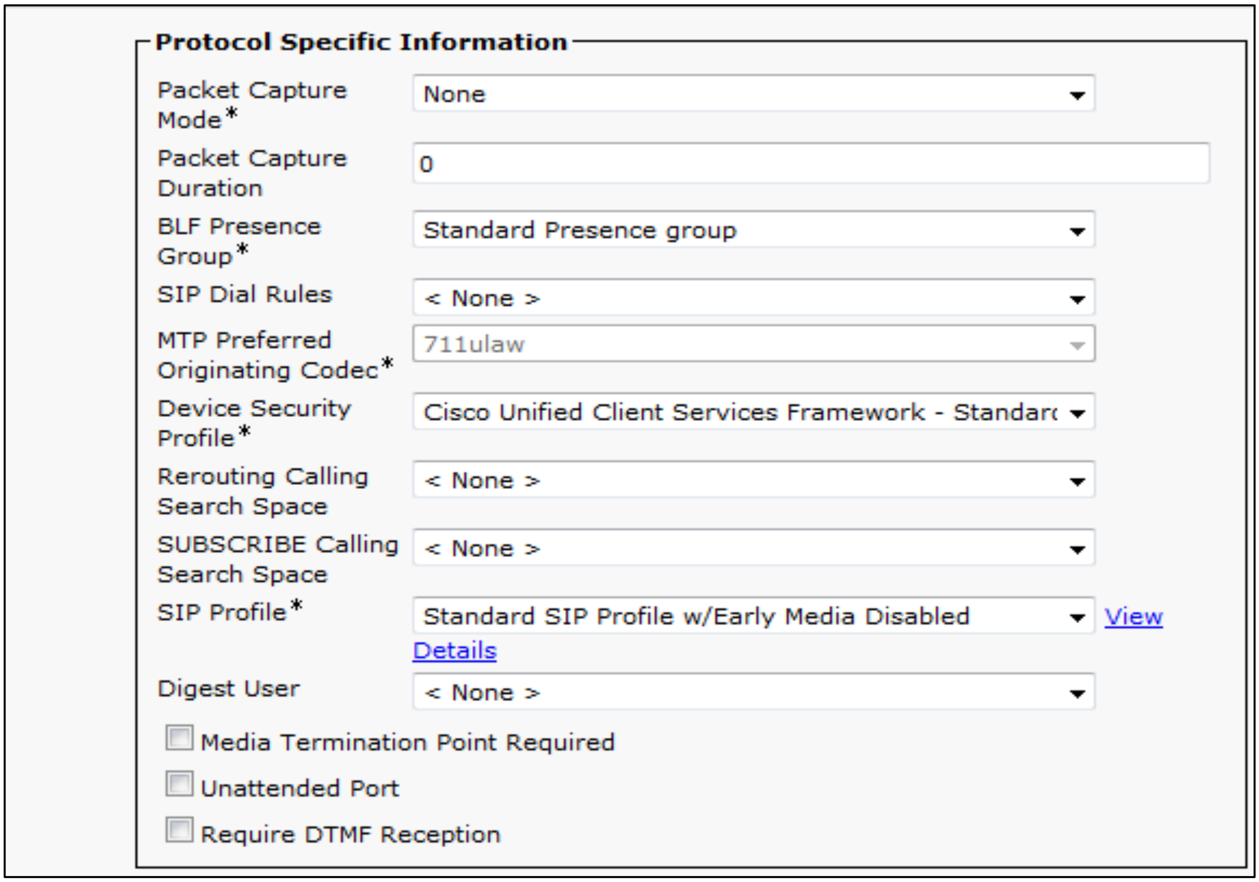
Media Resource Group List = MRGL_MTP
Set Owner check box
Set Owner user ID* = jabber1. This is used for this example



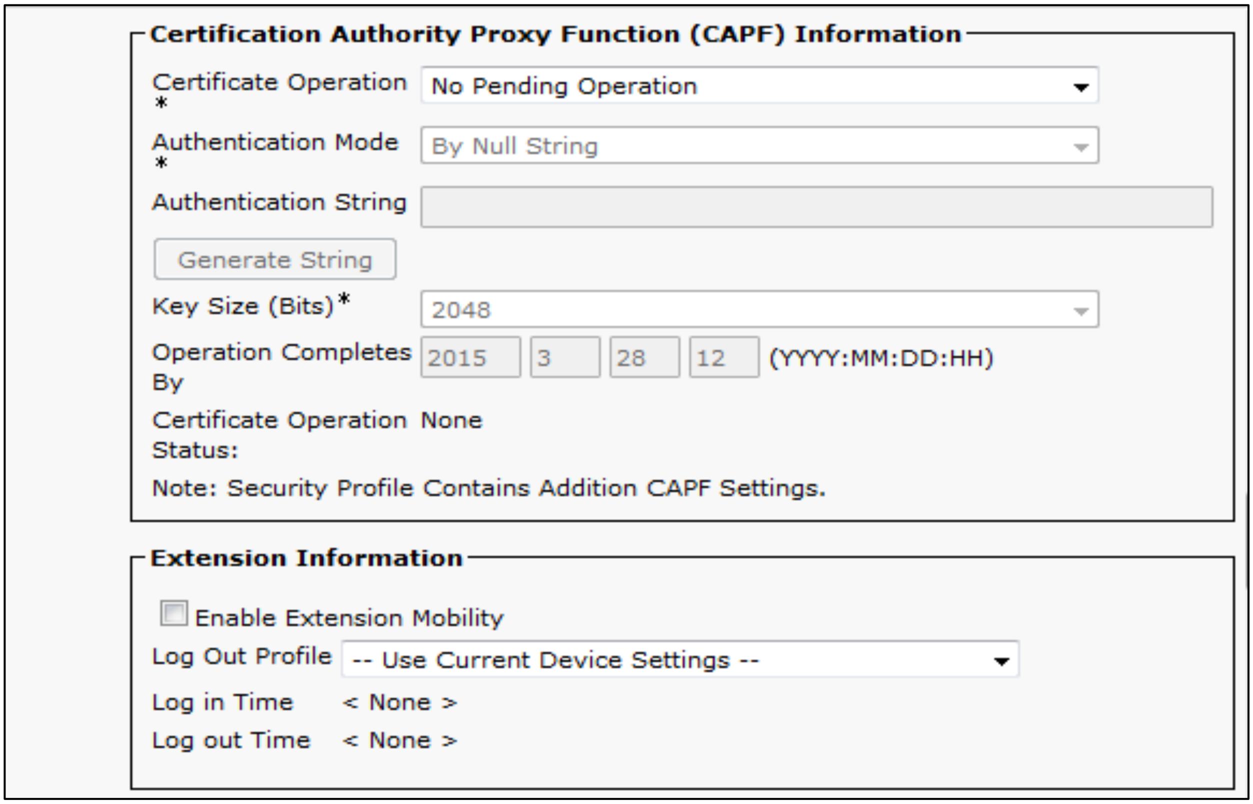


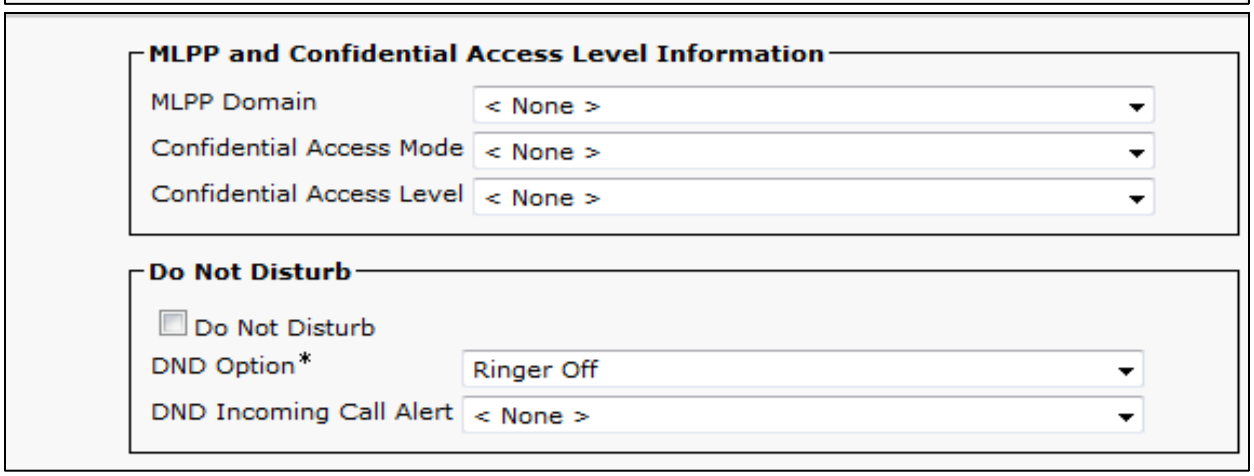
Jabber Client Configuration (Contd…)



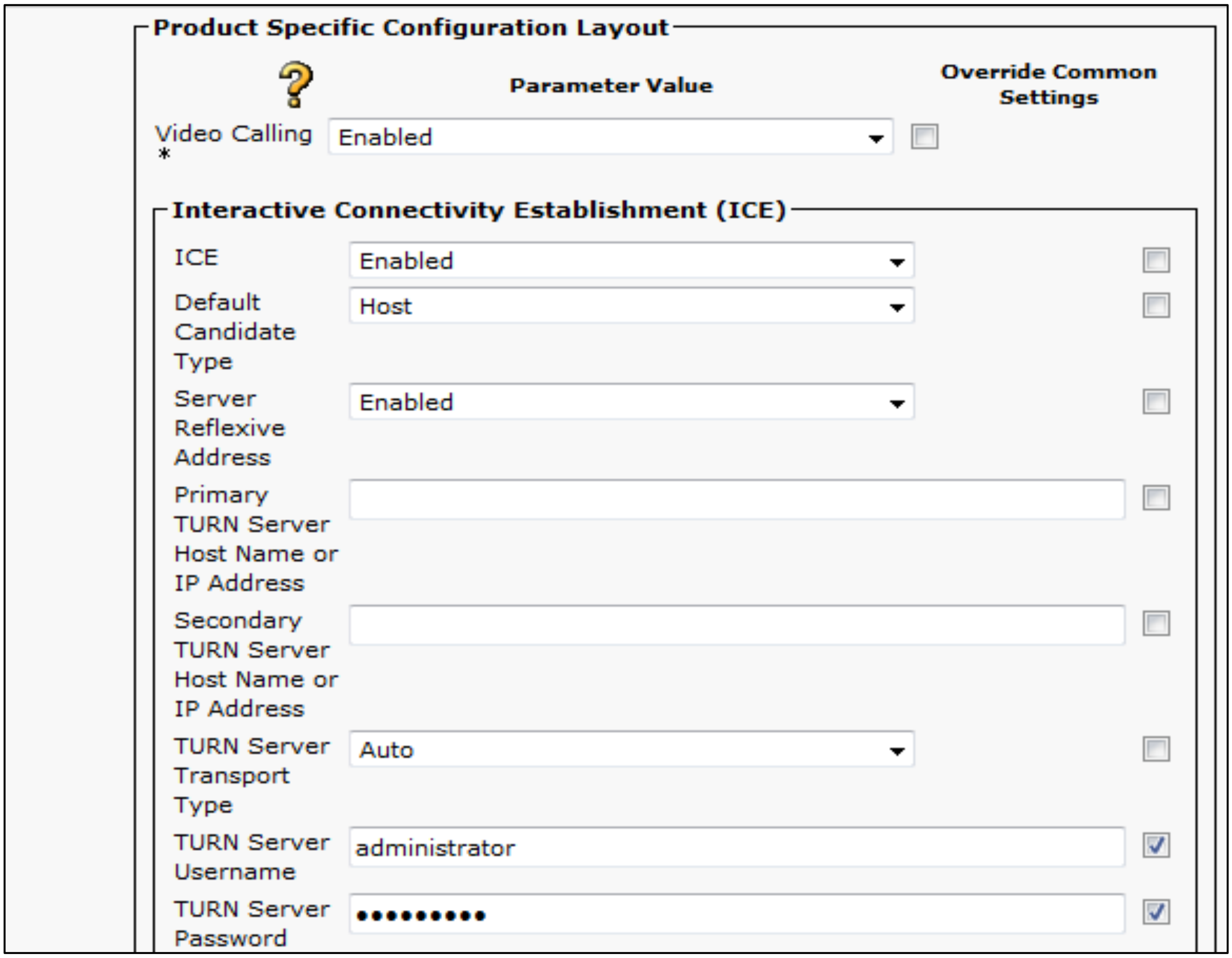


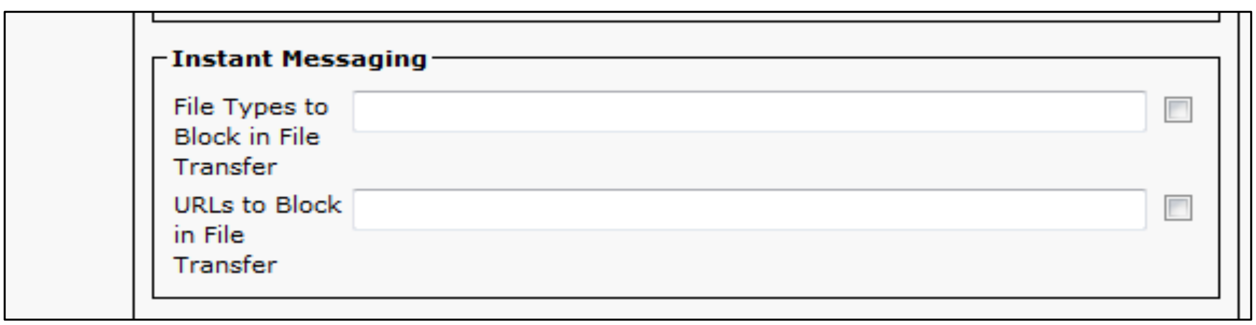
Jabber Client Configuration (Contd…)



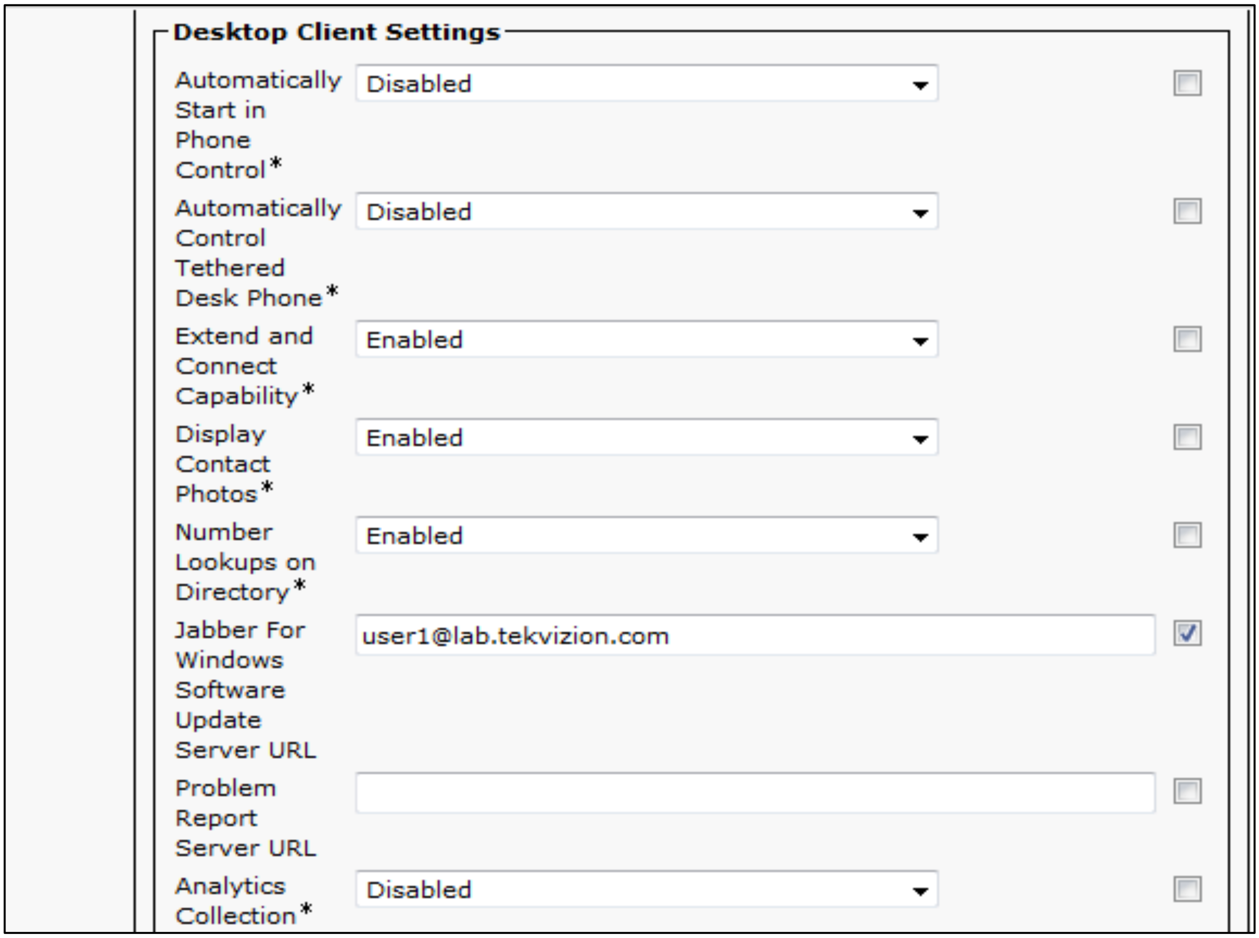


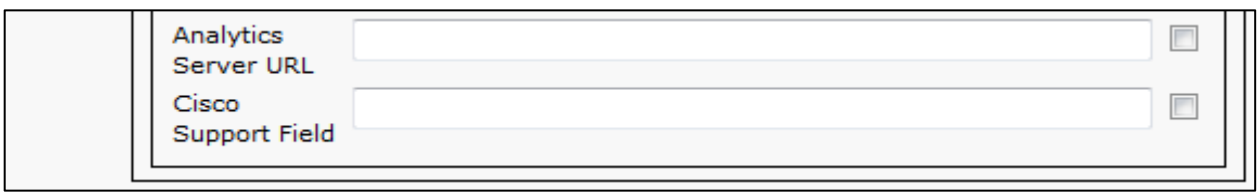
Jabber Client Configuration (Contd…)

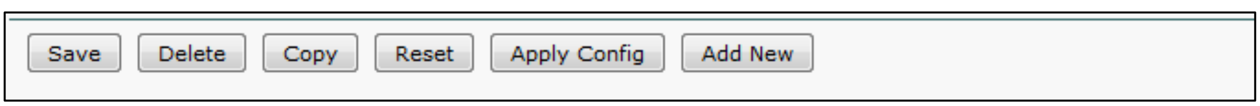




Jabber Client Configuration (Contd…)



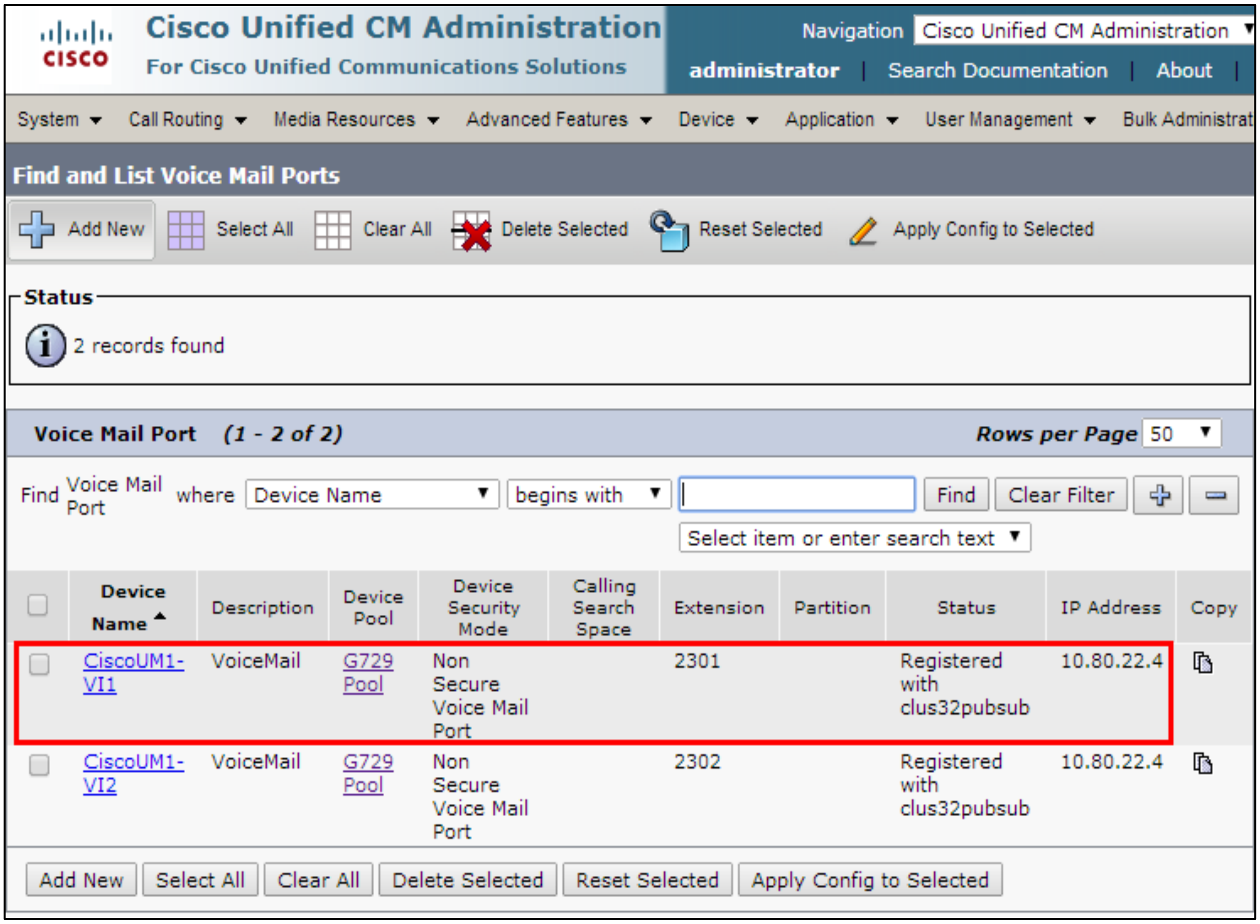




Note: Testing was conducted in tekVizion labs

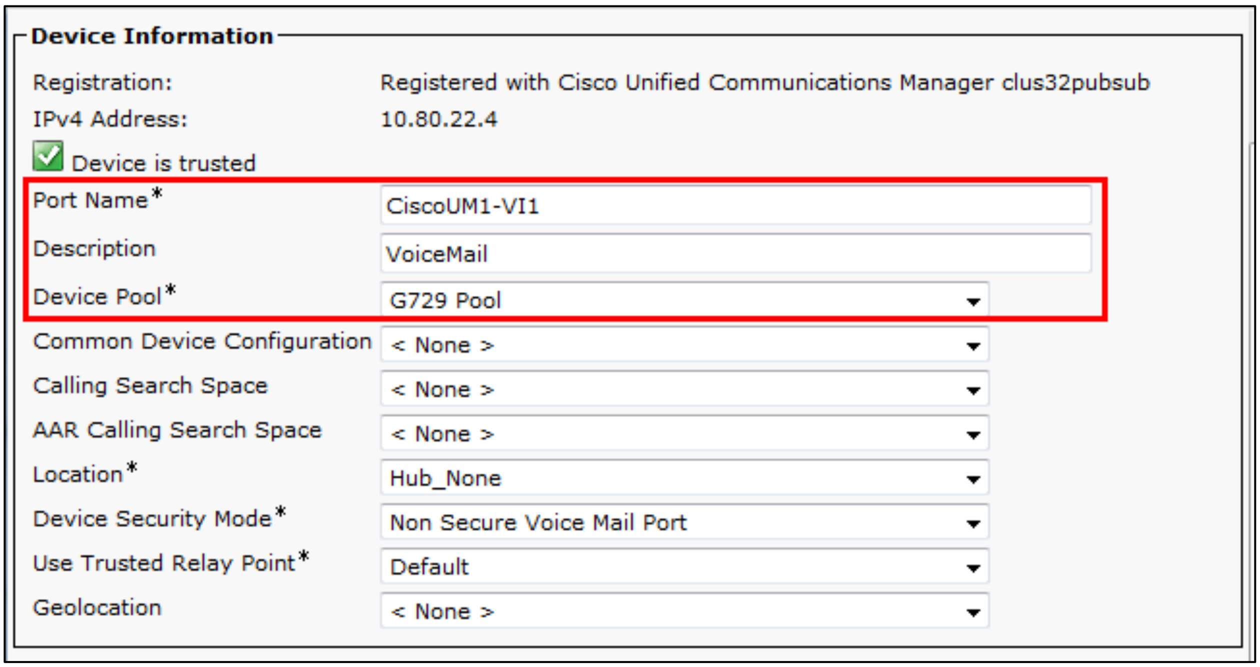
Voicemail Port Configuration

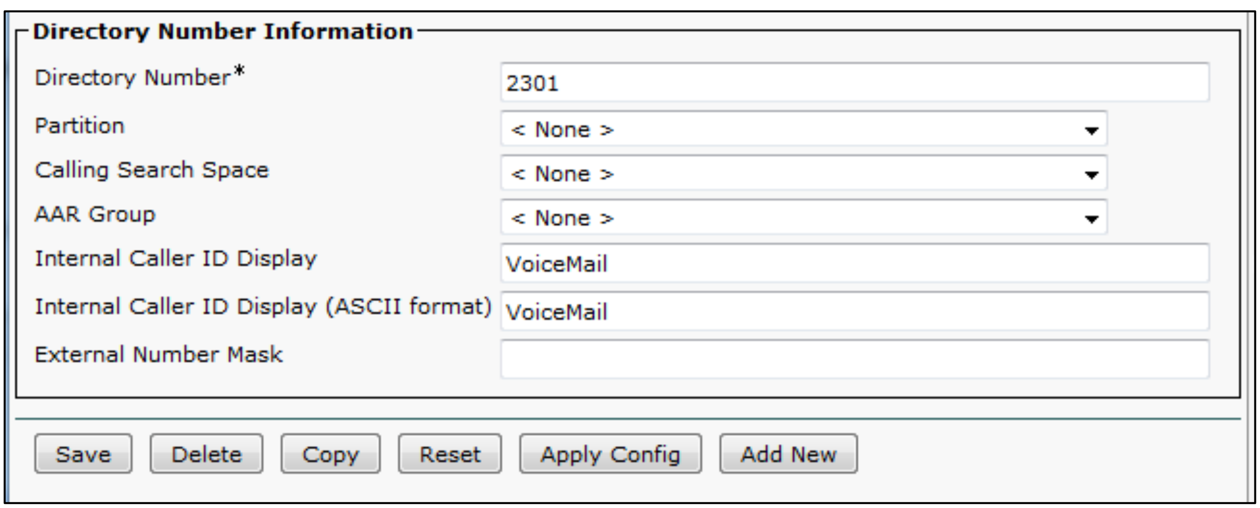
Navigation: Advanced Feature→ Voice Mail → Cisco Voice Mail Port



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* = 2301. This is used in this example.



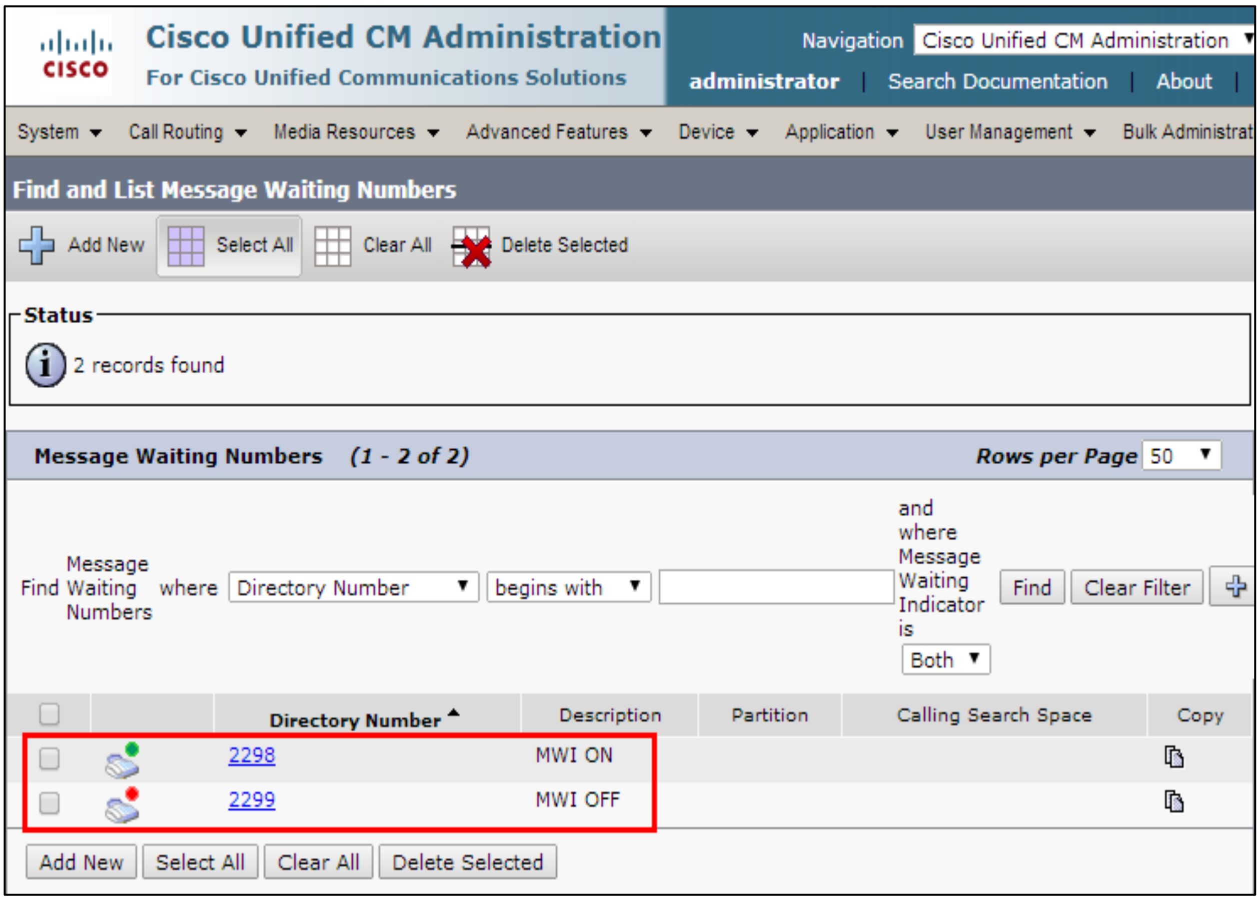


Note: Testing was conducted in tekVizion labs

Message Waiting Numbers Configurations

Navigation: Advanced Features → Voice Mail→Message Waiting

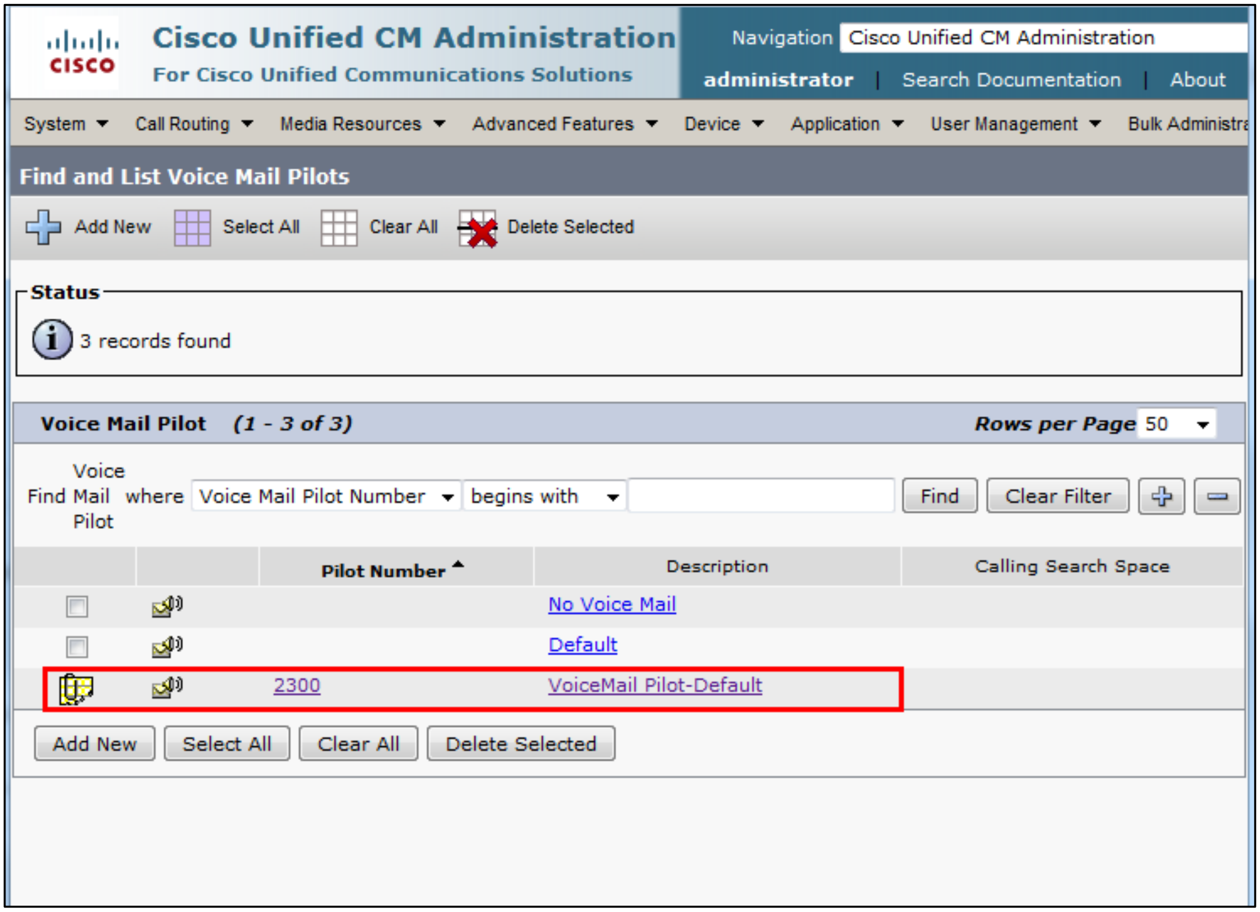
Set Message Waiting Number* = 2298
Set Message Waiting Indicator* = On
Set Message Waiting Number* = 2399
Set Message Waiting Indicator* = Off

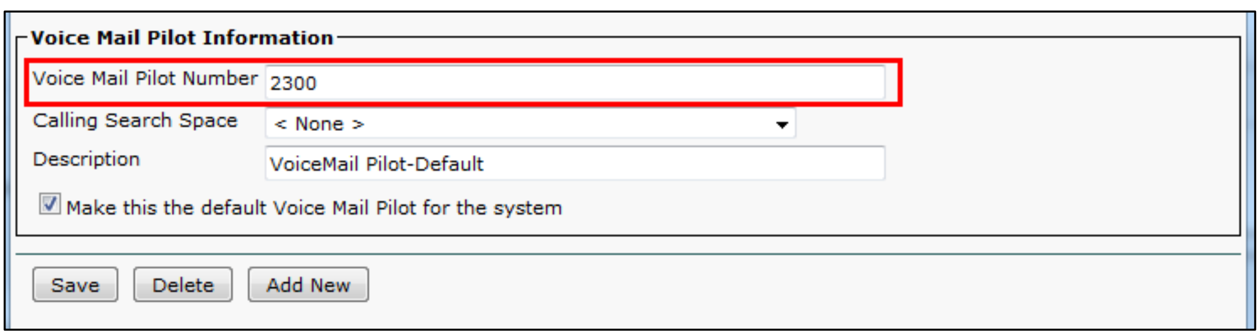


Voicemail Pilot Configuration

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

FAX-GATEWAY2#**sh version**

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

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

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

Compiled Sat 25-Oct-14 03:34 by prod_rel_team

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

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

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

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

Last reload type: Normal Reload

Last reload reason: Reload Command

This product contains cryptographic features and is subject to United

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

use. Delivery of Cisco cryptographic products does not imply

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

Importers, exporters, distributors and users are responsible for

compliance with U.S. and local country laws. By using this product you 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 CISCO2901/K9 (revision 1.0) with 483328K/40960K bytes of memory.

Processor board ID FTX174081SJ

2 Gigabit Ethernet interfaces

1 terminal line

2 Voice FXS interfaces

DRAM configuration is 64 bits wide with parity enabled.

255K bytes of non-volatile configuration memory.

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

License Info:

License UDI:

Device#   PID   SN

----

*1 CISCO2901/K9 FTX174081SJ

Technology Package License Information for Module:'c2900'

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

Configuration register is 0x2102

FAX-GATEWAY2#sh run

Building configuration...

Current configuration : 7131 bytes

!

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

!

version 15.4

service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

!

hostname FAX-GATEWAY2

!

boot-start-marker

boot-end-marker

!

aqm-register-fnf

!

logging queue-limit 1000000000

logging buffered 10000000

logging rate-limit 10000

no logging console

no logging monitor

enable secret 4 iR3uUX3Bo6oYbT6ajhFwJe39FR4g.1QCmm7yYduKGZI

!

no aaa new-model

!

```
!
!
!
!
!
!
!
!
!
!
!
!
!
ip domain name lab.tekvizion.com
ip name-server 10.64.1.3
ip cef
no ipv6 cef
multilink bundle-name authenticated
!
!
!
!
stcapp feature access-code
!
stcapp feature speed-dial
!
```

```
!
!
!
!
cts logging verbose
!
crypto pki trustpoint TP-self-signed-2189441908
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-2189441908
 revocation-check none
 rsakeypair TP-self-signed-2189441908
!
!
crypto pki certificate chain TP-self-signed-2189441908
 certificate self-signed 01
  3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
  31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
  69666963 6174652D 32313839 34343139 3038301E 170D3133 31303031 32303234
  30325A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D32 31383934
  34313930 3830819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
  810092C7 1982BC36 792DA64E 8FB4D8BC 1DDD4D7A 0882107F B14FCB24 699A35A9
  D521C88A 5B43F4FC D394E945 81A1380A 2E753478 93190ADE 75AA8971 883E9214
  C607CCDF 6FCCDE9C E95CE01A AEE4FCBE 3E91A43C D11C638F FC3E4ED2 57569523
  70A8D7C6 EFAD6688 C6244C79 5B955391 BF75EE61 DC4D0ADE 8D897AE2 CE76A938
  983F0203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603
```

```
  551D2304 18301680 14279B59 09E3EB37 0AE0DCE0 F8075BB6 DF93858A 45301D06
  03551D0E 04160414 279B5909 E3EB370A E0DCE0F8 075BB6DF 93858A45 300D0609
  2A864886 F70D0101 05050003 8181006E CF10B11F 9D8B59A9 AEACDEB8 26649CBB
  0F6C9690 12EAEB70 4BF5703D 98D2665A CD1B27D2 9B29351D 3ADF0B97 3C41F59A
  0DD82FF8 66CE4689 2D089FE8 EF3FFE54 5C85608C EE45908F D1160BDE A9185D58
  D3DA8795 428A7CE7 B9522F7C 60796800 485EDA2F B6C86F7A DF66B562 74942705
  C81F1883 7D4E29FC 8E999F7E EAE070
        quit
voice-card 0
 dsp services dspfarm
!
!
!
voice service voip
 allow-connections sip to sip
 no supplementary-service sip handle-replaces
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
voice class codec 2
 codec preference 1 g711ulaw
```

```
 codec preference 2 g729r8
!
voice class sip-profiles 1
 response ANY sip-header Allow-Header modify "UPDATE," ""
 request ANY sip-header Allow-Header modify "UPDATE," ""
 response ANY sip-header Allow-Header modify "UPDATE," ""
 response ANY sip-header Allow-Header modify "UPDATE," ""
!
!
!
!
!
!
!
!
license udi pid CISCO2901/K9 sn FTX174081SJ
hw-module pvdm 0/0
!
!
!
username cisco privilege 15 secret 4 tnhtc92DXBhelxjYk8LWJrPV36S2i4ntXrpb4RFmfqY
!
redundancy
!
!
!
```

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

```
ip route 10.64.0.0 255.255.0.0 10.80.22.1
ip route 10.80.0.0 255.255.0.0 10.80.22.1
ip route 172.16.0.0 255.255.0.0 10.80.22.1
!
!
!
!
control-plane
!
!
voice-port 0/0/0
 no vad
 shutdown
!
voice-port 0/0/1
 no echo-cancel enable
 no vad
 station-id name fax test
 station-id number 7323204084
 caller-id enable
 !
 !
 !
 !
!
no mgcp timer receive-rtcp
```

```
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
ccm-manager music-on-hold
!
no ccm-manager fax protocol cisco
!
dial-peer voice 110 pots
 service session
 destination-pattern 4084
 port 0/0/1
!
dial-peer voice 200 voip
 description CUCM to Gateway
 service session
 session protocol sipv2
 session transport udp
 incoming called-number 4084
 voice-class codec 1
 voice-class sip profiles 1
 dtmf-relay rtp-nte
```

```
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 201 voip
 description Gateway to CUCM
 service session
 destination-pattern [2-9]T
 session protocol sipv2
 session target ipv4:10.80.22.2
 session transport udp
 voice-class codec 1
 voice-class sip profiles 1
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
!
gateway
 timer receive-rtp 1200
!
!
!
gatekeeper
 shutdown
```

!

!

banner exec ^C

% Password expiration warning.

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

and it provides the default username "cisco" for one-time use. If you have

already used the username "cisco" to login to the router and your IOS image

supports the "one-time" user option, then this username has already expired.

You will not be able to login to the router with this username after you exit

this session.

It is strongly suggested that you create a new username with a privilege level

of 15 using the following command.

username <myuser> privilege 15 secret 0 <mypassword>

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

use.

----

^C

banner login ^C

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

This feature requires the one-time use of the username "cisco" with the

password "cisco". These default credentials have a privilege level of 15.

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

CREDENTIALS

Here are the Cisco IOS commands.

username <myuser> privilege 15 secret 0 <mypassword>

no username cisco

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

to use.

IF YOU DO NOT CHANGE THE PUBLICLY-KNOWN CREDENTIALS, YOU WILL NOT BE ABLE

TO LOG INTO THE DEVICE AGAIN AFTER YOU HAVE LOGGED OFF.

For more information about Cisco CP please follow the instructions in the

QUICK START GUIDE for your router or go to http://www.cisco.com/go/ciscocp

----

^C

!

line con 0

 login local

line aux 0

line 2

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

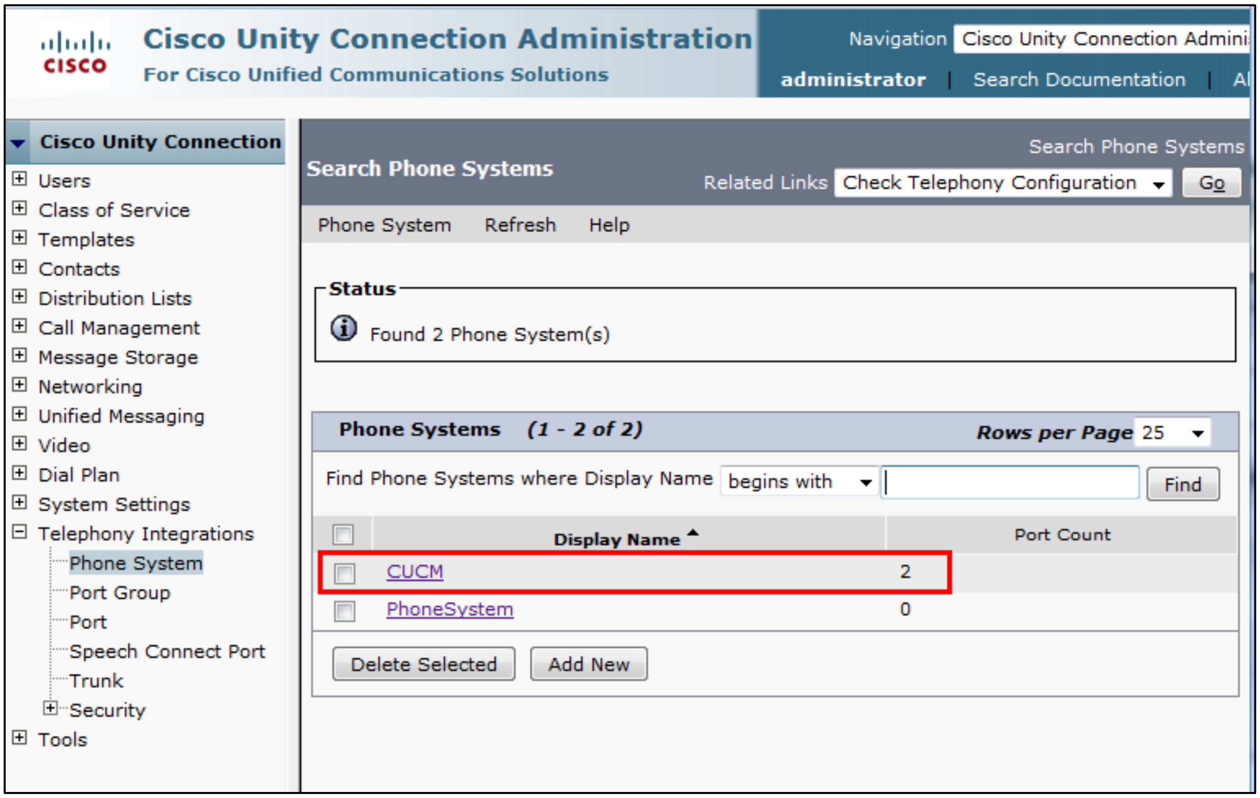
Cisco UCM SCCP Integration with Cisco Unity Connection (CUC)

CUC Version

CUC Telephony Integration with Cisco UCM

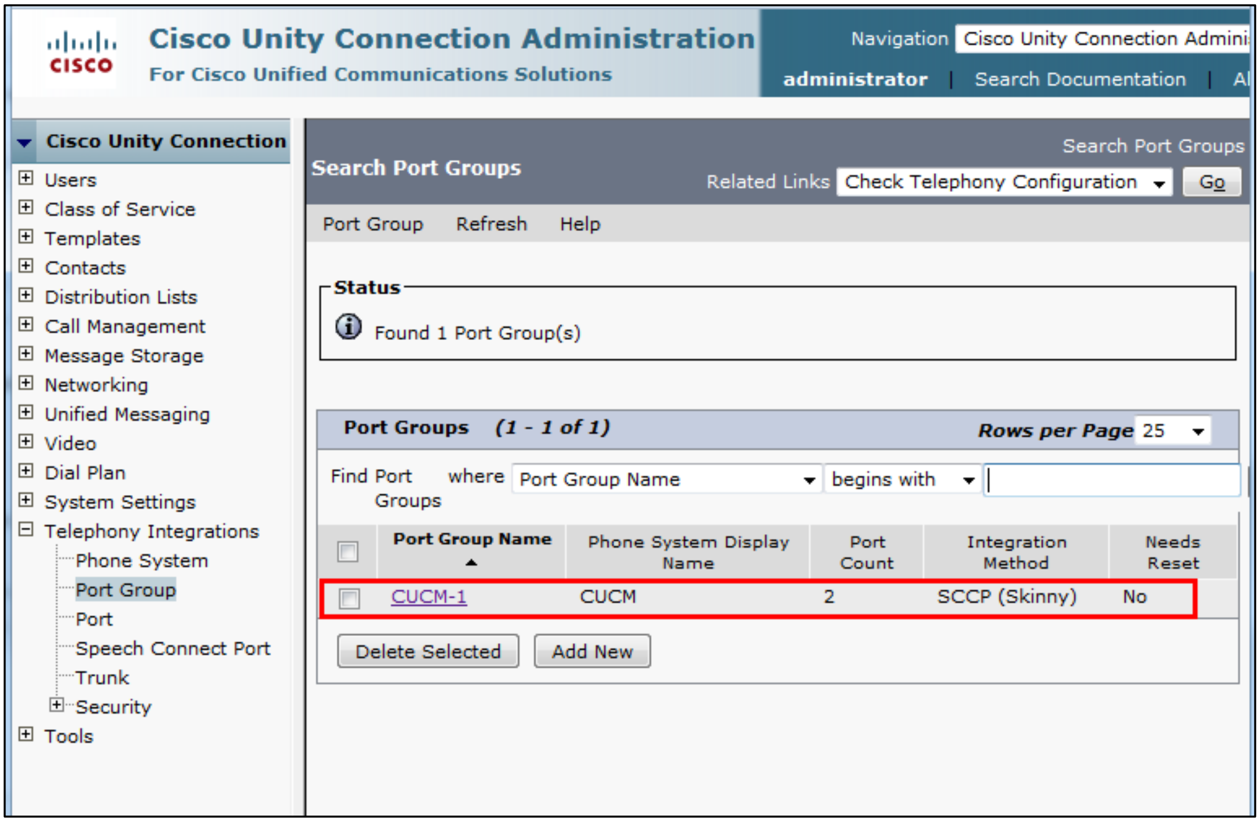
Navigation: Telephony Integrations → Phone system

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



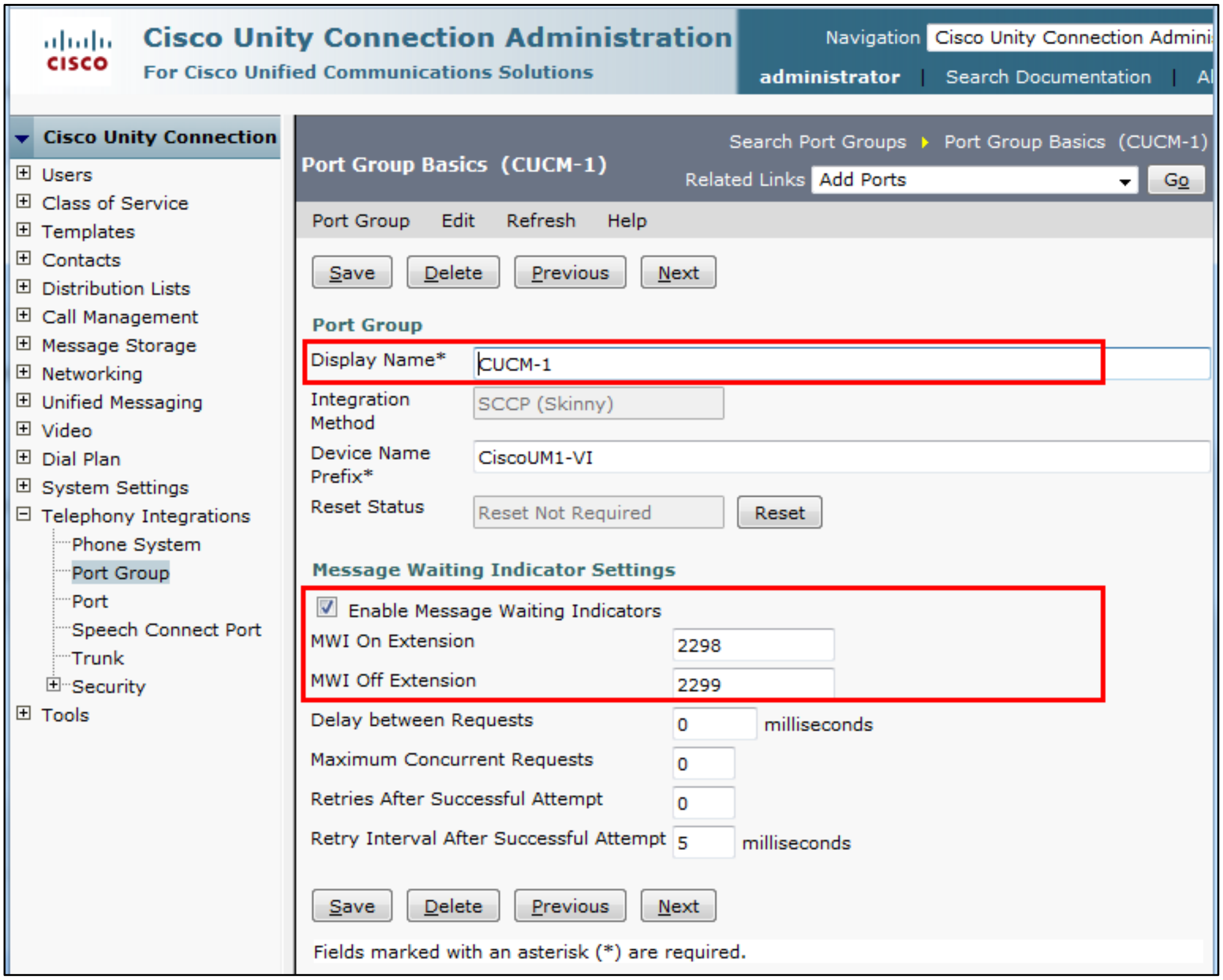
CUC Port Group

Navigation: Telephony Integration → Port Group



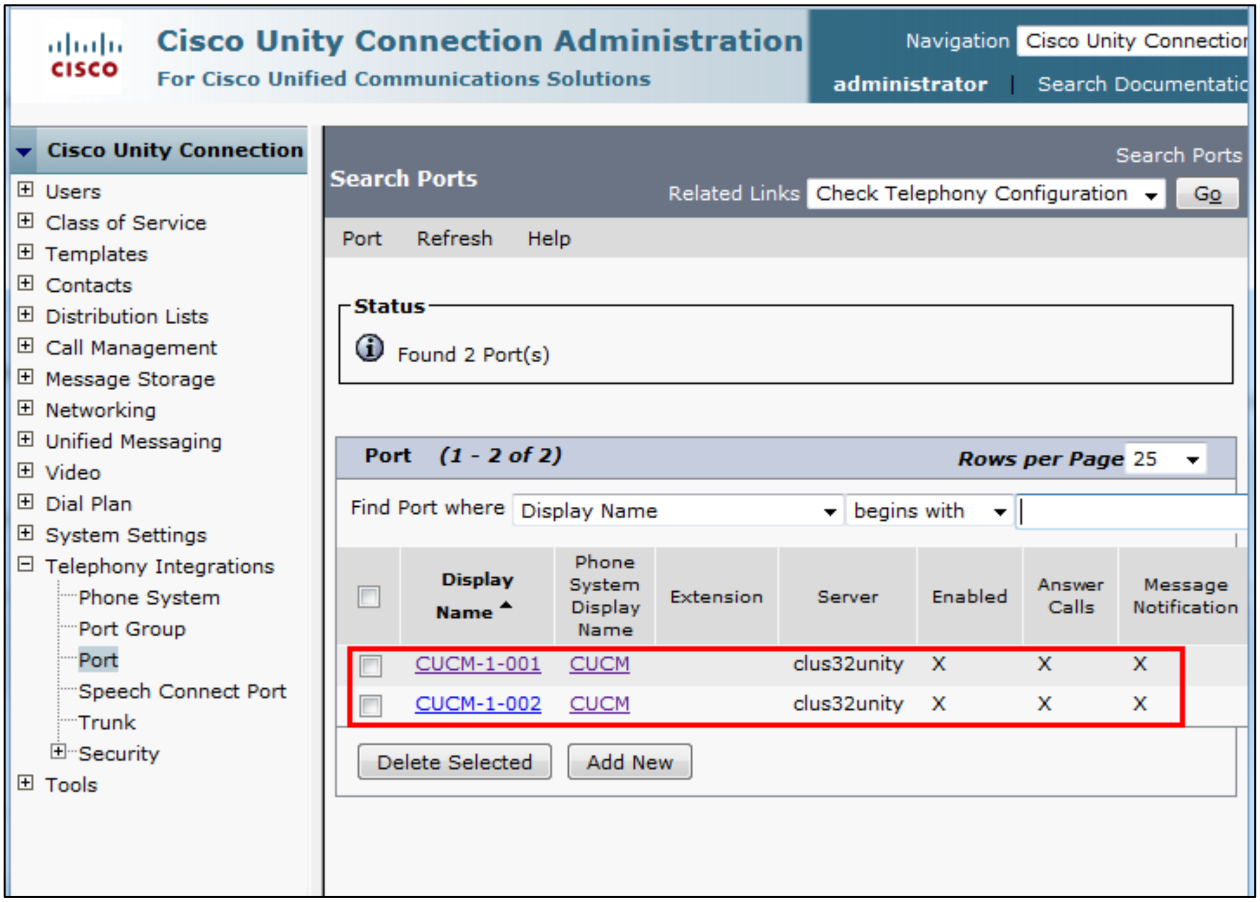
CUC Port Group(continued…)

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



Note: Testing was conducted in tekVizion labs

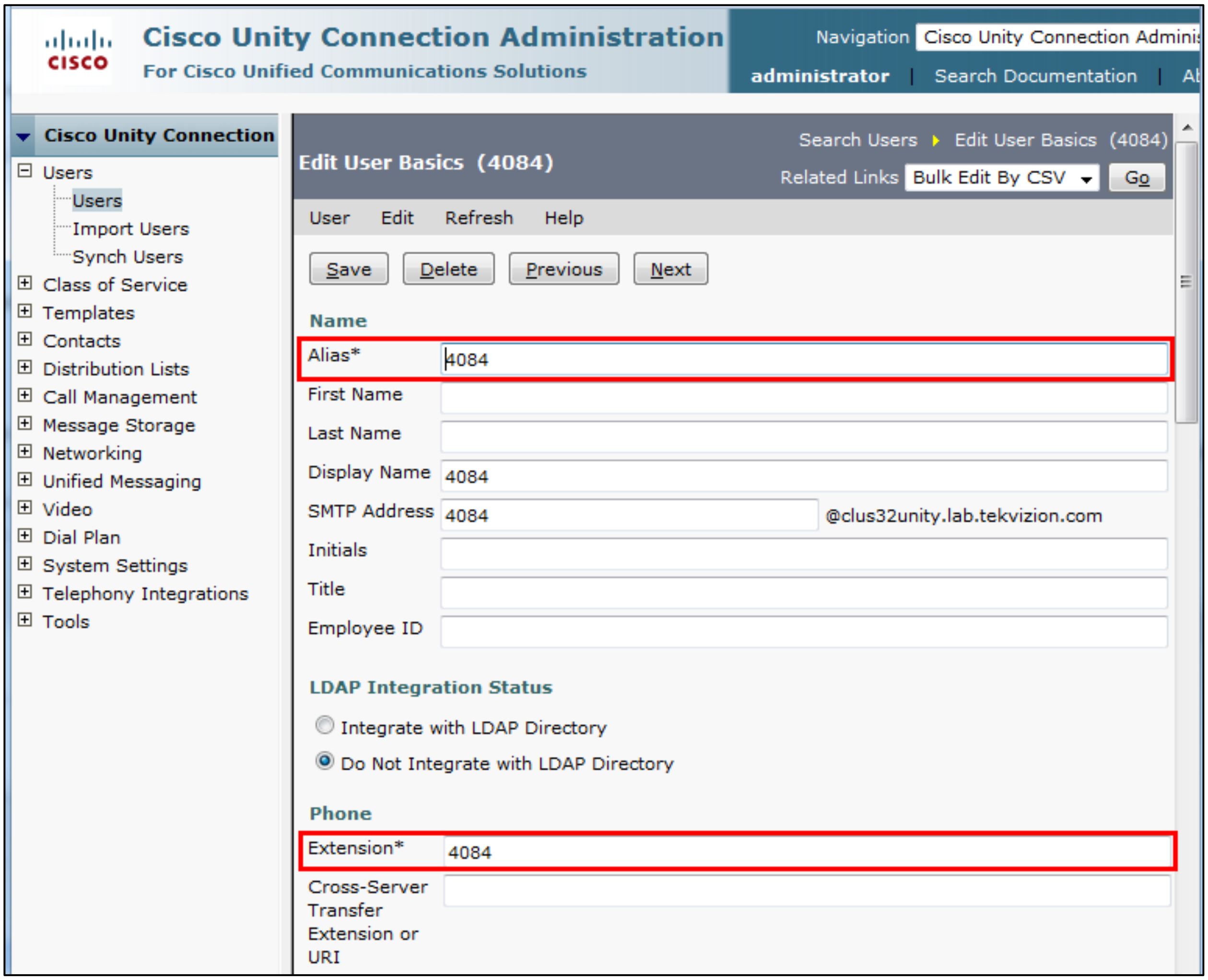
CUC Port Settings



CUC Sample User Basic Settings

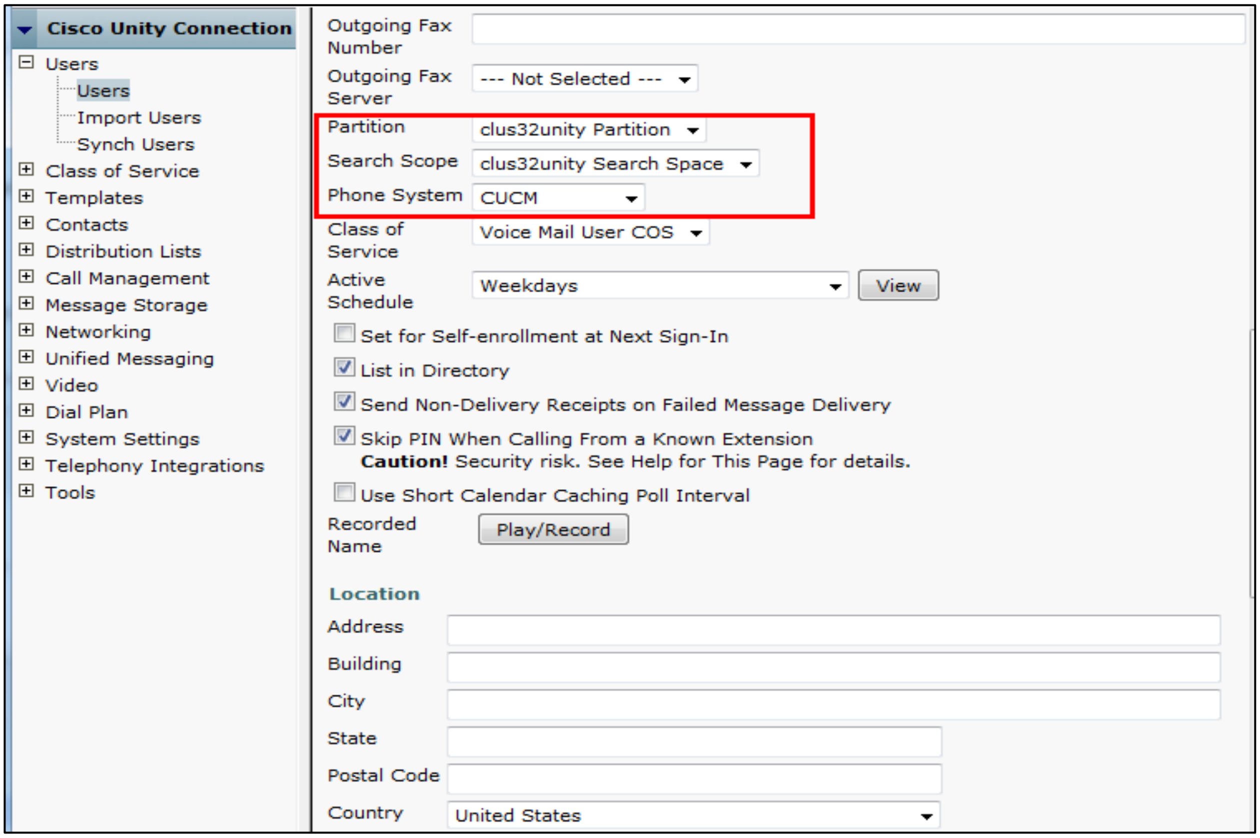
Navigation: Cisco Unity connection → Users → Users

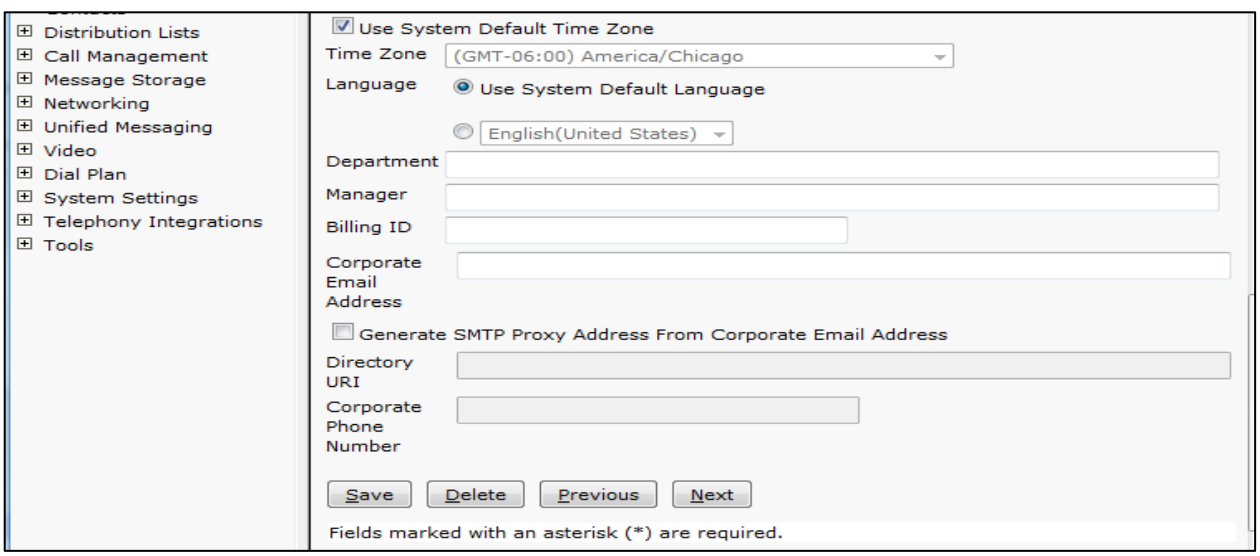
Set Alias = 4084.This is one of the extension used for this testing.
Set Extension = 4084. This is used for this example.



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 = CUCM.

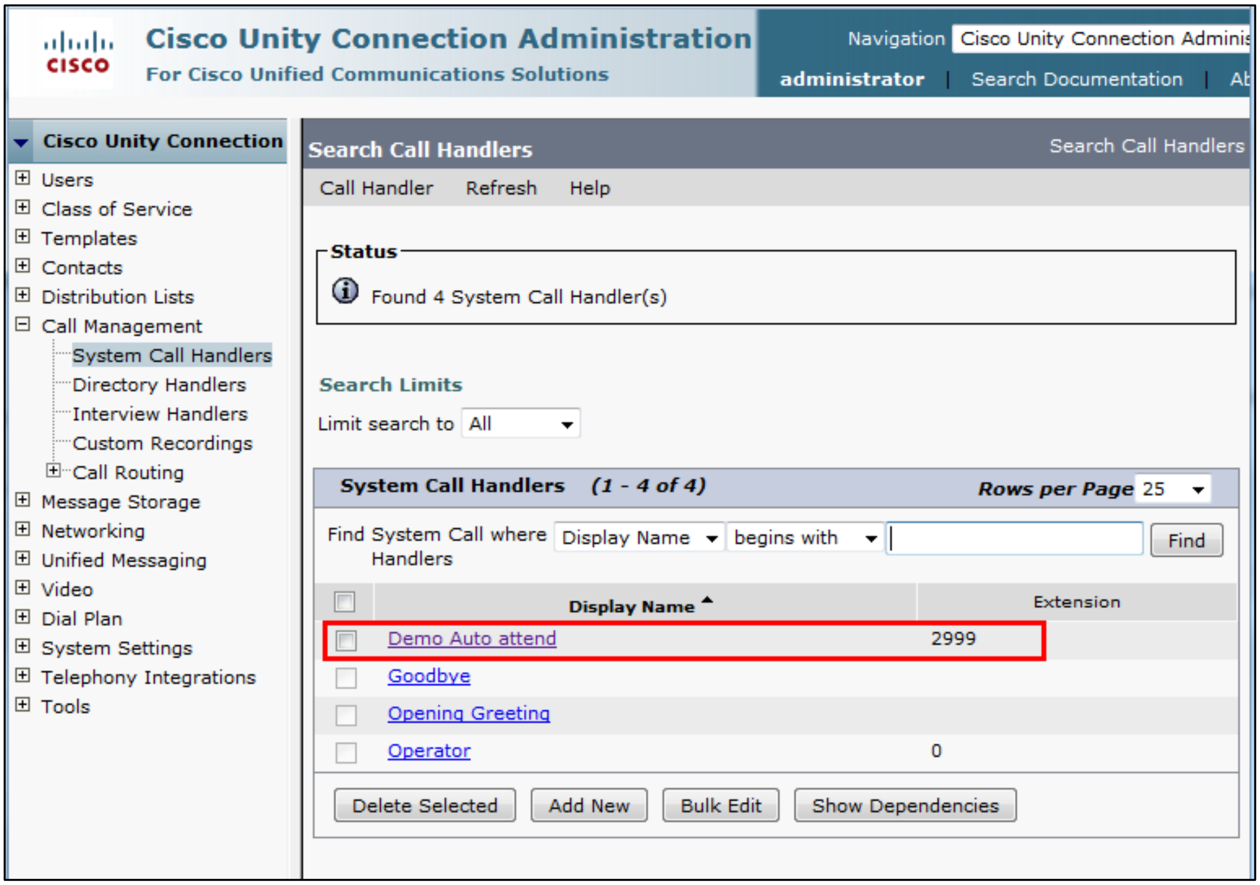




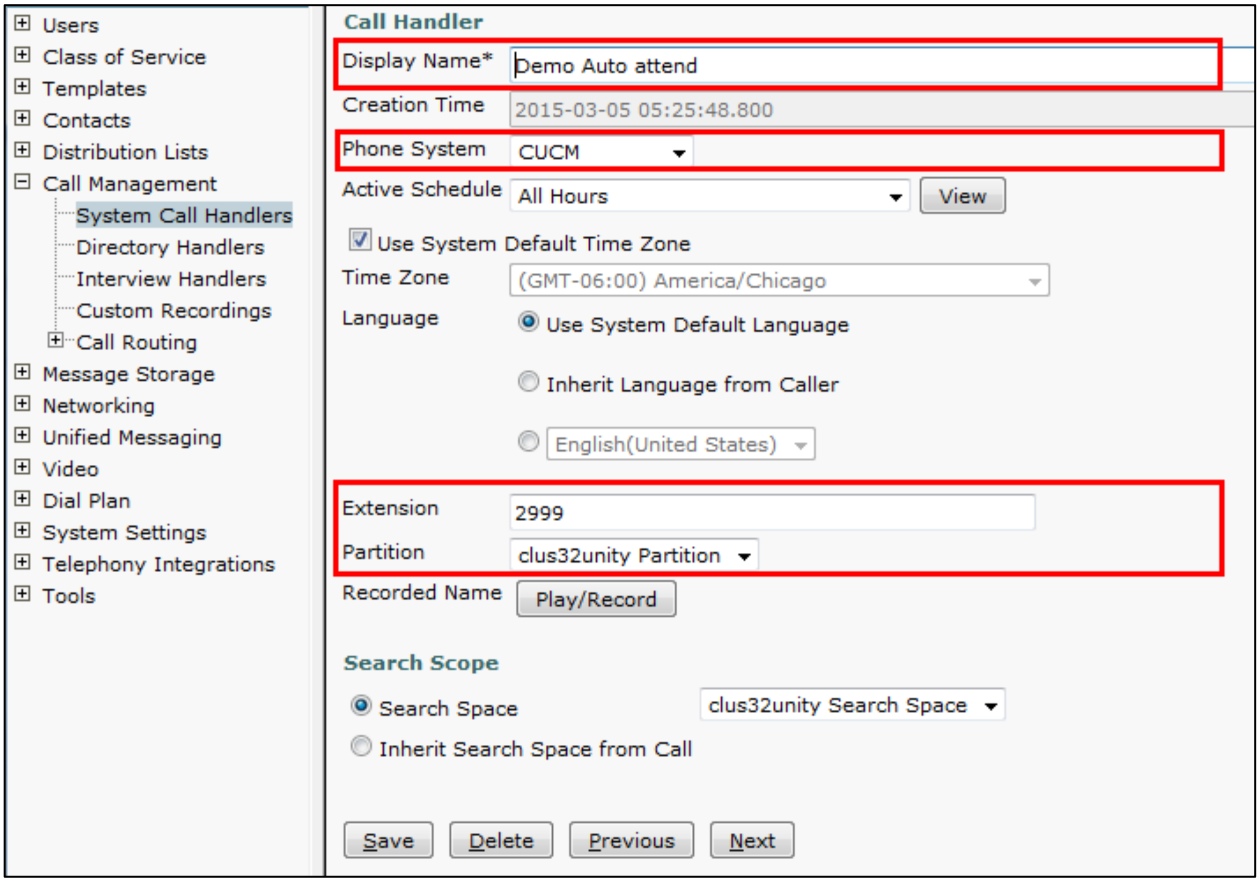
Auto Attendant

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.



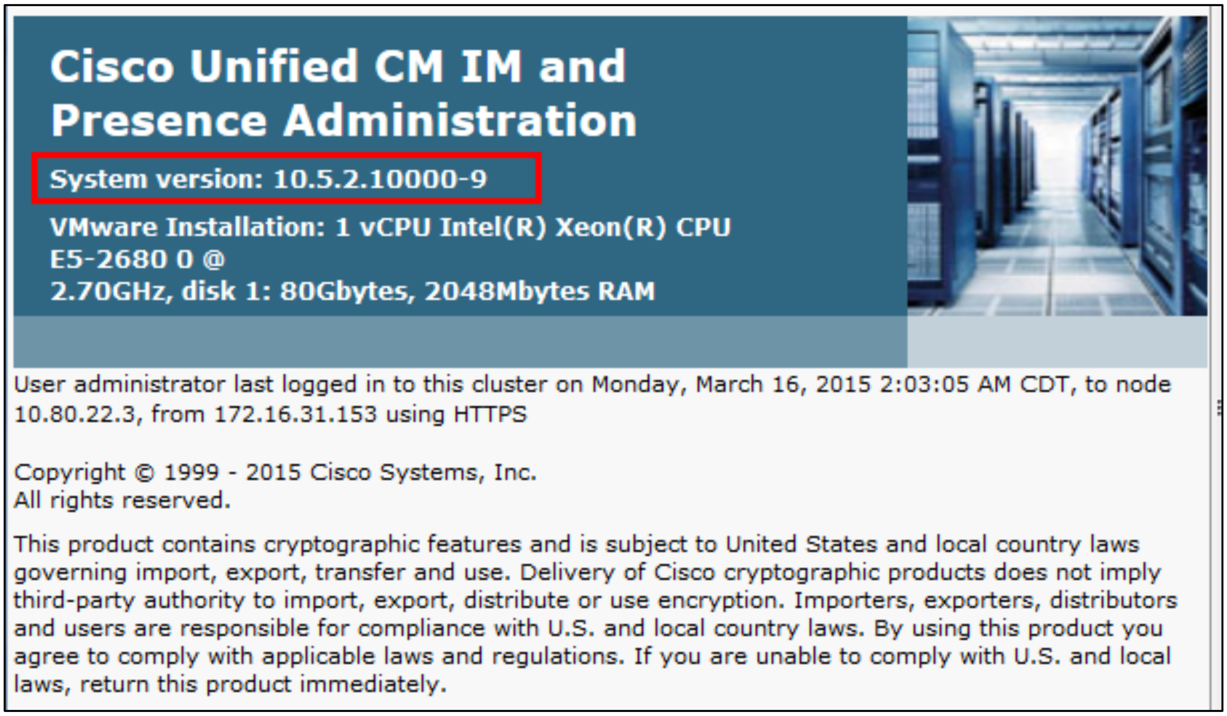
Auto Attendant (Continued…)



Users
Class of Service
Templates
Contacts
Distribution Lists
Call Management
System Call Handlers
Directory Handlers
Interview Handlers
Custom Recordings
Call Routing
Message Storage
Networking
Unified Messaging
Video
Dial Plan
System Settings
Telephony Integrations
Tools
Call Handler
Display Name*
Demo Auto attend
Creation Time
2015-03-05 05:25:48.800
Phone System
CUCM
Active Schedule
All Hours
View
Use System Default Time Zone
Time Zone
(GMT-06:00) America/Chicago
Language
Use System Default Language
Inherit Language from Caller
English(United States)
Extension
2999
Partition
clus32unity Partition
Recorded Name
Play/Record
Search Scope
Search Space
clus32unity Search Space
Inherit Search Space from Call
Save
Delete
Previous
Next

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

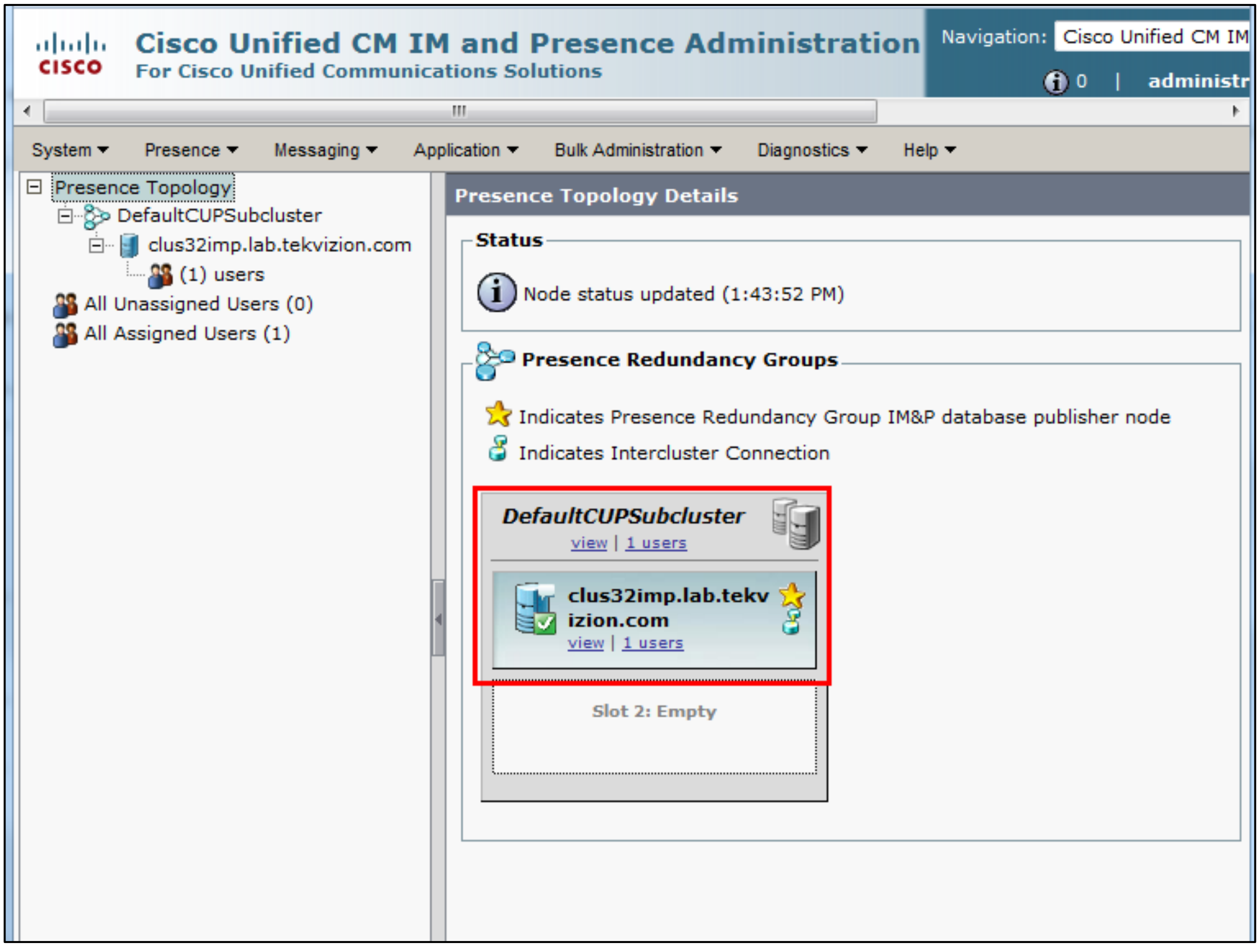
CUP/IMP Version



Note: Testing was conducted in tekVizion labs

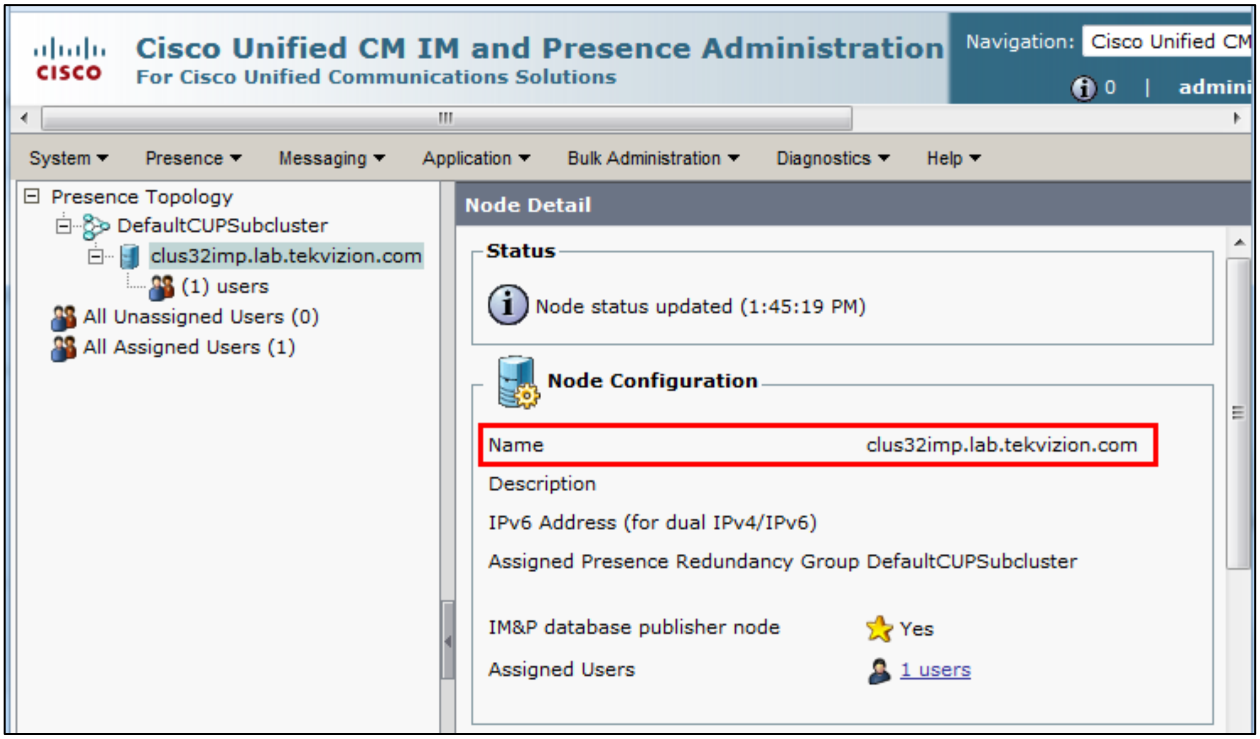
Presence Topology

Navigation: System → Presence Topology



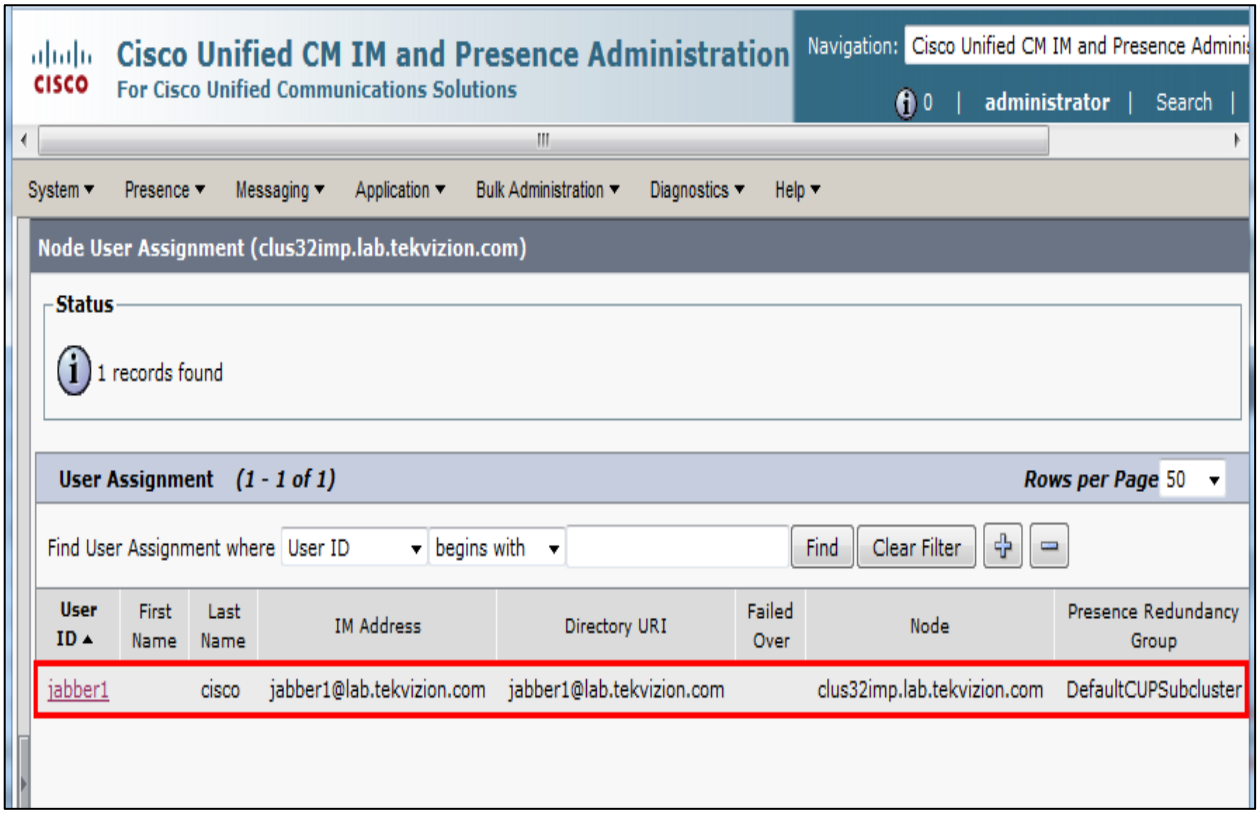
Node Configuration

Navigation: System → Cluster Topology → Fully Qualified Domain Name



Users

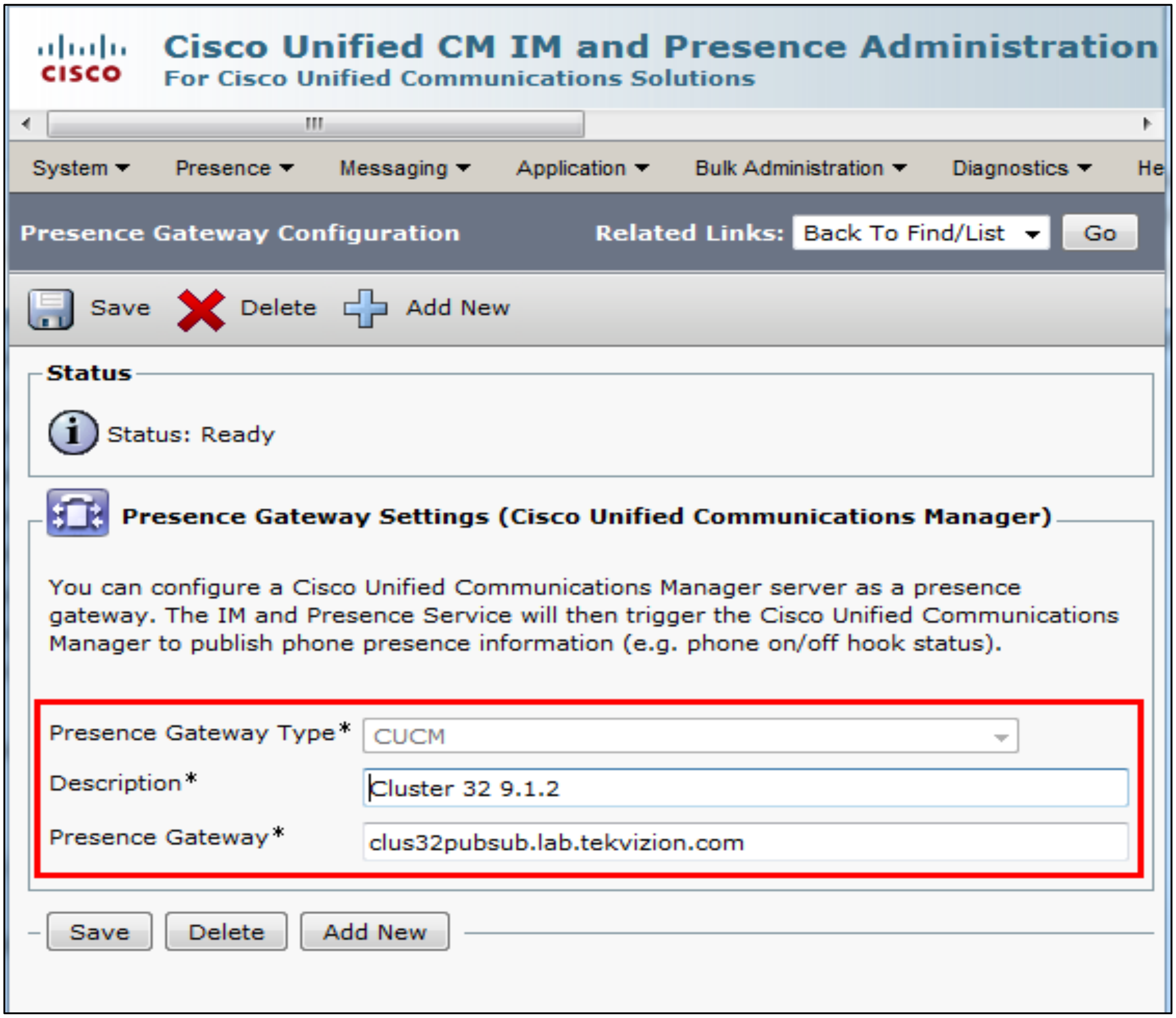
Navigation: System → Cluster Topology → clus32imp.lab.tekvizion.com → Users



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

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