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
Sep 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.

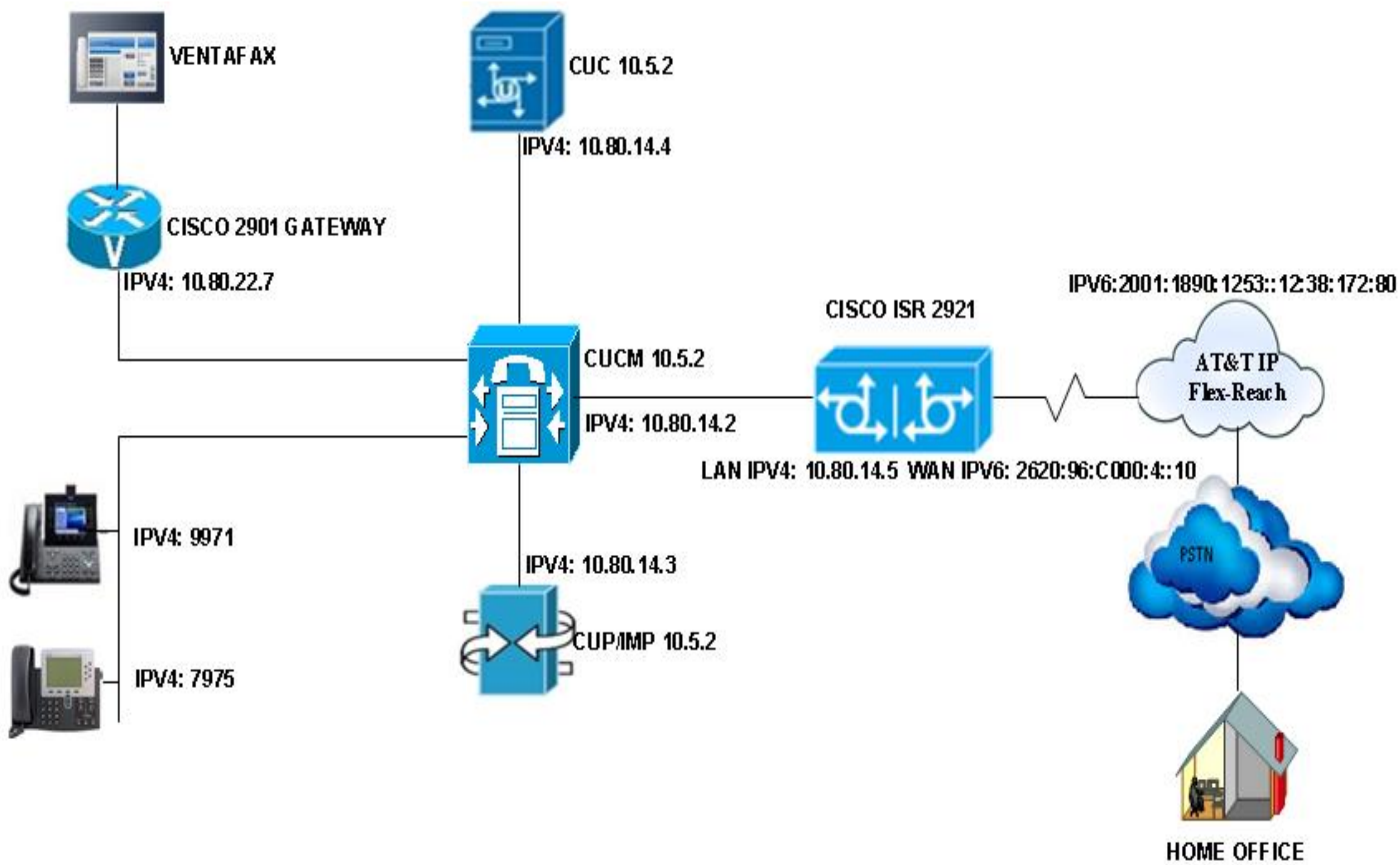
- This application note describes the necessary steps and configurations of Cisco Unified Communications Manager (Cisco UCM) 10.5.2, Cisco Unity Connection 10.5.2, Cisco Unified CM IM and Presence 10.5.2, Cisco Integrated Services Routers (ISR) Version 15.4(3) M1 with connectivity to AT&T's IP Flex-Reach 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. Network Topology between Cisco ISR and AT&T Flexible reach SIP is deployed using IPV6.

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

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

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

Network Topology



Hardware Components

- Cisco UCS-C240 VMWare server running ESXi 5.5
- Cisco IP Phones. This solution was tested with Cisco 7965, Cisco 7975 and Cisco 9971 phones
- Cisco Integrated Services Router - Cisco CISCO2921/K9 (revision 1.0) with 483328K/40960K bytes of memory.
- Processor board ID FTX1746AJCB
  3 Gigabit Ethernet interfaces
  1 terminal line
  1 Virtual Private Network (VPN) Module
  DRAM configuration is 64 bits wide with parity enabled.
  255K bytes of non-volatile configuration memory.
  250880K bytes of ATA System CompactFlash 0 (Read/Write)

Software Requirements

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

Features

Features – Supported

- Basic Call using G.729 and G711
- Calling Party Number Presentation and Restriction
- Calling Name Presentation
- AT&T Advanced 8YY Call Prompter (8YY)
- Cisco UBE Delayed-Offer-to-Early-Offer conversion of an initial SIP INVITE without SDP
- Intra-site Call Transfer
- Intra-site Conference
- Call Hold and Resume
- Call Forward All, Busy and No Answer
- AT&T IP Teleconferencing
- Fax over T.38 and G.711
- Incoming DNIS Translation and Routing
- Outbound calls to AT&T's IP and TDM networks
- Inbound calls from AT&T's IP and TDM networks
- CPE voicemail managed service, leave and retrieve voice messages via incoming AT&T SIP trunk (Cisco Unity Connection)
- Auto-attendant transfer-to service (See Caveat section for details)
- Failover (From non-responsive SIP network to ATT SIP network)
- Inbound & Outbound Calls using Cisco Jabber
- Emergency and 911 calls were terminated to a voicemail platform in lab environment within AT&T for test

Network Based Features - Supported

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

NOTE: Using the AT&T IP Flexible Reach Portal, provision TN(s) on the CPE with the Sequential Ringing and Simultaneous Ringing features. Provisioning is self-explanatory. AT&T representative can be contacted for any help required in this regard.

Features - Not Supported

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

Caveats

Fax

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

Auto-Attendant

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

Hold/Resume & Music on Hold (MOH)

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

Ring back Tone on Early Unattended Transfer

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

PBX Based Call Forward Unconditional

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

SIP Provisional Acknowledgement/Early media

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

AT&T IP Teleconferencing (IPTC)

Following scenarios were executed with limitations on AT&T network

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

Configuration Considerations

- To enable conference using G729 on AT&T IP Flexible Reach and Cisco UCM SIP trunk, it is required to configure a conference bridge (CFB) resource to initiate a three-way conference between end-points. See configuration section for details.
- Forwarded calls from Cisco UCM user to PSTN (out to AT&T's IP Flexible Reach service), AT&T serviced areas require that the SIP Diversion header contain the full 10-digit DID number of the forwarding party. In this application note the assumption has been made that a typical customer will utilize extension numbers (4-digit assignments in this example) and map 10-digit DID number using Cisco UBE translation profile. This is because the Cisco UCM uses 4-digit extensions on Cisco UCM IP phones and it is necessary to expand the 4-digit extension included in the Diversion header of a forwarding INVITE message to its full 10-digit DID number when the IP phone is set to call-forward. The requirement to expand the Diversion-Header has been achieved by the use of a SIP profile in Cisco UBE. See configuration section for details.
- Upon receiving inbound calls, AT&T SIP network will always have the first choice codec presented in the initial SIP INVITE (unless the end-device does not support the listed preferred codec), and processes calls accordingly. Customers wishing to place/receive G.711-only calls must configure separate voice class codec on Cisco UBE with G.711 as the first choice.
- SIP Profiles may also be employed to advertise desired RTP payload packet size.
- "voice-class sip privacy id" needs to configure in Cisco UBE dial peer to make call From a CPE Phone to PSTN phone.
- This test environment is not configured with Cisco UBE High Availability (HA).
- Cisco UCM sends a SIP UPDATE message to Cisco UBE for a call transfer. AT&T network does not support SIP UPDATE message. This causes the Cisco UBE to timeout and hence the call transfer is not completed. As a workaround, SIP profile has been applied on the Cisco UBE to remove UPDATE from the allowed headers. See configuration section for details.

Emergency 911/E911 Services Limitations and Restrictions

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

ISR Configuration

ATT-IPV6#sh version

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

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

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

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

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

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

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

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

Last reload type: Normal Reload

Last reload reason: Reload Command

This product contains cryptographic features and is subject to United

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

use. Delivery of Cisco cryptographic products does not imply

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

Importers, exporters, distributors and users are responsible for

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

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

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

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

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

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

export@cisco.com.

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

Processor board ID FTX1746AJCB

3 Gigabit Ethernet interfaces

1 terminal line

1 Virtual Private Network (VPN) Module

DRAM configuration is 64 bits wide with parity enabled.

255K bytes of non-volatile configuration memory.

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

License Info:

License UDI:

Device#   PID   SN

----

*1   CISCO2921/K9   FTX1746AJCB

Technology Package License Information for Module:'c2900'

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

Configuration register is 0x21024 Gigabit Ethernet interfaces

32768K bytes of non-volatile configuration memory.

4194304K bytes of physical memory.

7741439K bytes of eUSB flash at bootflash:.

Configuration register is 0x2102

```
ATT-IPV6#sh run

Building configuration...


Current configuration : 11243 bytes

!

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

!

version 15.4

service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

service sequence-numbers

!

hostname ATT-IPV6

!

boot-start-marker

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

boot-end-marker

!

aqm-register-fnf

!

logging queue-limit 1000000000

logging buffered 30000000

logging rate-limit 10000

no logging console

no logging monitor
```

enable secret 4 Pe0NhiWw5IXZpE.k5VhTSCoGPcuVeRyrer9kEPz20Z6!

no aaa new-model

!

!

!

!

!

!

!

!

!

no ip domain lookup

ip cef

ipv6 unicast-routing

ipv6 cef

!

multilink bundle-name authenticated

!

!

!

!

!

!

cts logging verbose

!

!

```
voice-card 0
 dspfarm
 dsp services dspfarm!
!
!
!
voice service voip
 no ip address trusted authenticate
 address-hiding[1]
 mode border-element[2]
 media disable-detailed-stats
 allow-connections sip to sip[3]
 no supplementary-service sip handle-replaces
 redirect ip2ip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  header-passing
  error-passthru[4]
  asserted-id pai[5]
  early-offer forced[6]
```

---

[1] Hide signaling and media peer addresses from endpoints other than gateway.
[2] If the mode border-element command is not entered, border-element-related commands are not available for Cisco Unified Border Element voice connections on the cisco 2900 and Cisco 3900 series platforms with a universal feature set. The mode border-element command is not available on any other platforms.
[3] This command enables Cisco UBE basic IP-to-IP voice communication feature.
[4] This command allows SIP error messages to pass-through end-to-end without modification through Cisco UBE.
[5] This command enables router to send P-Asserted ID within the SIP Message Header. Alternatively, this command can also be applied to individual dial-peers (voice-class sip asserted-id pai).
[6] This command enables delay offer-to-early offer conversion of initial SIP INVITE message to calls matched to this dial-peer level.

```
  no silent-discard untrusted
  midcall-signaling passthru[7]
  privacy-policy passthru[8]
  g729 annexb-all
!
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 sip-profiles 1
 response ANY sip-header Allow-Header modify "UPDATE," ""
  request INVITE sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"[10]
 response ANY sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"
 request INVITE sdp-header Audio-Attribute add "a=ptime:30"[11]
!
```

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

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

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

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

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

```
!
!
!
!
license udi pid CISCO2921/K9 sn FTX1746AJCB
hw-module pvdm 0/0
!
!
!
username cisco privilege 15 password 0 cisco
!
redundancy
!
!
!
!
!
!
!
!
interface Embedded-Service-Engine0/0
 no ip address
 shutdown
!
```

```
interface GigabitEthernet0/0[12]
 description Wan Interface
 no ip address
 negotiation auto
 ipv6 address 2620:96:C000:4::10/64[13]
 ipv6 enable
 cdp enable
!
interface GigabitEthernet0/1[14]
 description Lan Interface
 ip address 10.80.14.5 255.255.255.0[15]
 negotiation auto
 cdp enable
!
interface GigabitEthernet0/2
 no ip address
 shutdown
 duplex auto
 speed auto
!
ip forward-protocol nd
!
no ip http server
```

[12] Cisco UBE WAN interface to AT&T
[13] Cisco UBE WAN interface IPv6 Address
[14] Cisco UBE LAN interface to Cisco UCM
[15] Cisco UBE LAN interface IPv4 Address

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

```
!
dial-peer voice 300 voip
 description " Int'l calls to AT&T - AT&T facing side "
 destination-pattern 011T
 session protocol sipv2
 session target ipv6:[2001:1890:1253:0:12:38:172:80]
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 800 voip
 description " Incoming AT&T to IP-PBX . AT&T facing side "
 huntstop
 session protocol sipv2
 incoming called-number [27][37][27].......
 voice-class codec 1
 voice-class sip asymmetric payload full
```

```
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip bind control source-interface GigabitEthernet0/0
 voice-class sip bind media source-interface GigabitEthernet0/0
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 214 voip
 description "Outgoing To AT&T"-AT&T facing side
 destination-pattern [2-9]T
 session protocol sipv2
 session target ipv6:[2001:1890:1253:0:12:38:172:80]
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0[16]
 voice-class sip bind media source-interface GigabitEthernet0/0
 dtmf-relay rtp-nte
```

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

```
fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 122 voip
 description "OPERATOR TESTING"
 destination-pattern 0
 session protocol sipv2
 session target ipv6:[2001:1890:1253:0:12:38:172:80]
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
 voice-class sip bind media source-interface GigabitEthernet0/0
 dtmf-relay rtp-nte
no vad
!
dial-peer voice 141 voip
 description "Network Feature"
 destination-pattern *..
 session protocol sipv2
 session target ipv6:[2001:1890:1253:0:12:38:172:80]
 voice-class codec 1
```

```
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
 voice-class sip bind media source-interface GigabitEthernet0/0
 dtmf-relay rtp-nte
no vad
!
dial-peer voice 2151 voip
 description " Incoming AT&T to IP-PBX - IP-PBX facing side "
 destination-pattern 7322162...
 session protocol sipv2
 session target ipv4:10.80.14.2:5060
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/1
 voice-class sip bind media source-interface GigabitEthernet0/1
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
```

!

dial-peer voice 700 voip[17]

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

 huntstop

 destination-pattern 7323680...

 session protocol sipv2[18]

 session target ipv4:10.80.14.2:5060

 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 early-offer forced

 voice-class sip bind control source-interface GigabitEthernet0/1[22]

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

 dtmf-relay rtp-nte[23]

 fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none[24]

 no vad

!

[17] Dial-peer facing AT&T Network

[18] Session protocol SIPv2 is used for this testing.

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

[20] Configures the dynamic SIP asymmetric payload support.

[21] This command allows for privacy settings to be transparently passed between AT&T network and Cisco UCM. In this example, the command is set at the dial-peer level, you can also set the command at a global level to affect all dial-peers without necessarily setting the command on each dial-peer.

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

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

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

```
dial-peer voice 400 voip
 description " Int'l calls to AT&T - IP-PBX facing side "
 session protocol sipv2
 incoming called-number 011T
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 500 voip
 description " N11 Calls to AT&T - AT&T facing side "
 destination-pattern .11
 session protocol sipv2
 session target ipv6:[2001:1890:1253:0:12:38:172:80]
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
```

```
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
 dtmf-relay rtp-nte
no vad
!
dial-peer voice 600 voip
 description " N11 Calls to AT&T - IP-PBX facing side "
 session protocol sipv2
 incoming called-number .11
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
 dtmf-relay rtp-nte
no vad
!
dial-peer voice 2152 voip
 description " Incoming AT&T to IP-PBX - AT&T facing side "
 session protocol sipv2
 incoming called-number 7322162...
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
```

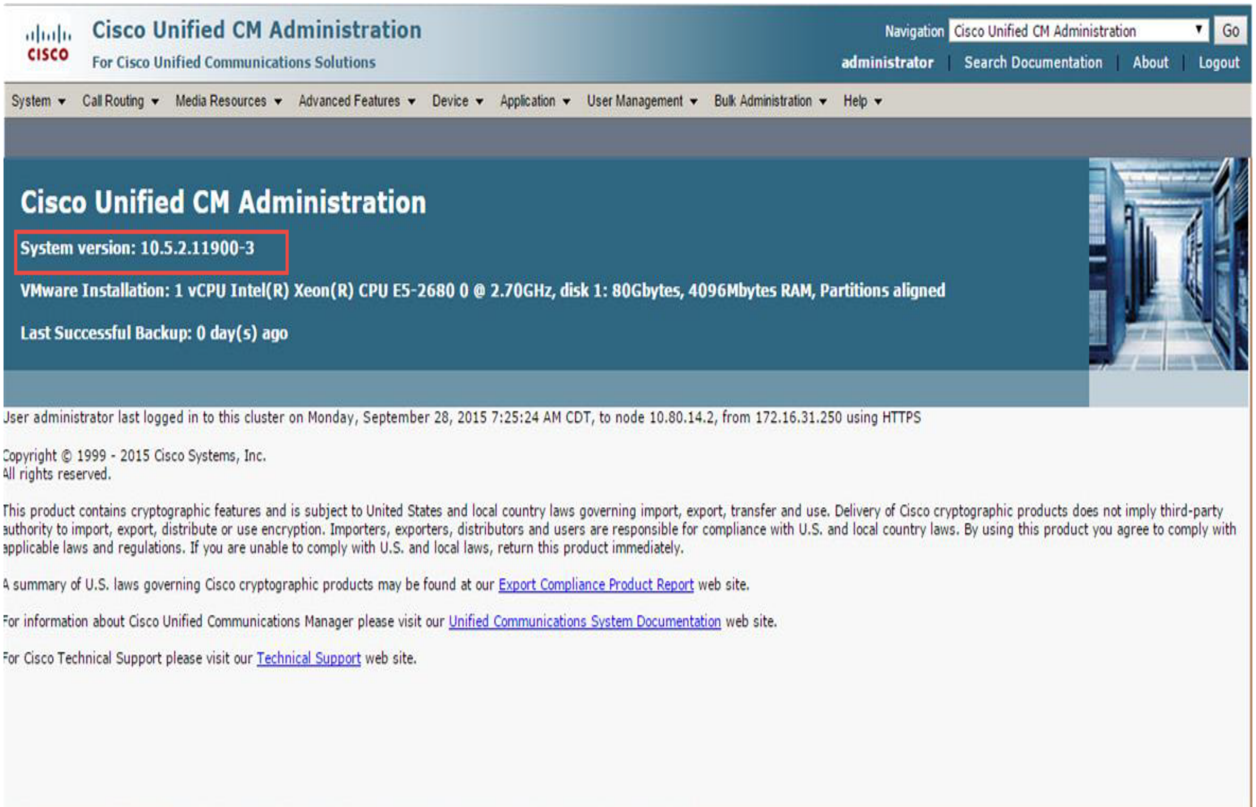
```
 voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
!
gateway
 media-inactivity-criteria all
 timer receive-rtcp 5
 timer receive-rtp 86400
!
sip-ua
 no remote-party-id
 timers expires 1800000
 connection-reuse
 protocol mode dual-stack
!
!
!
gatekeeper
 shutdown
!
!
```

```
!
line con 0
 logging synchronous
line aux 0
line 2
 no activation-character
 no exec
 transport preferred none
 transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
 stopbits 1
line vty 0 4
 exec-timeout 960 0
 logging synchronous
 login local
 transport input all
!
scheduler allocate 20000 1000
!
end
```

Cisco UCM Configuration

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

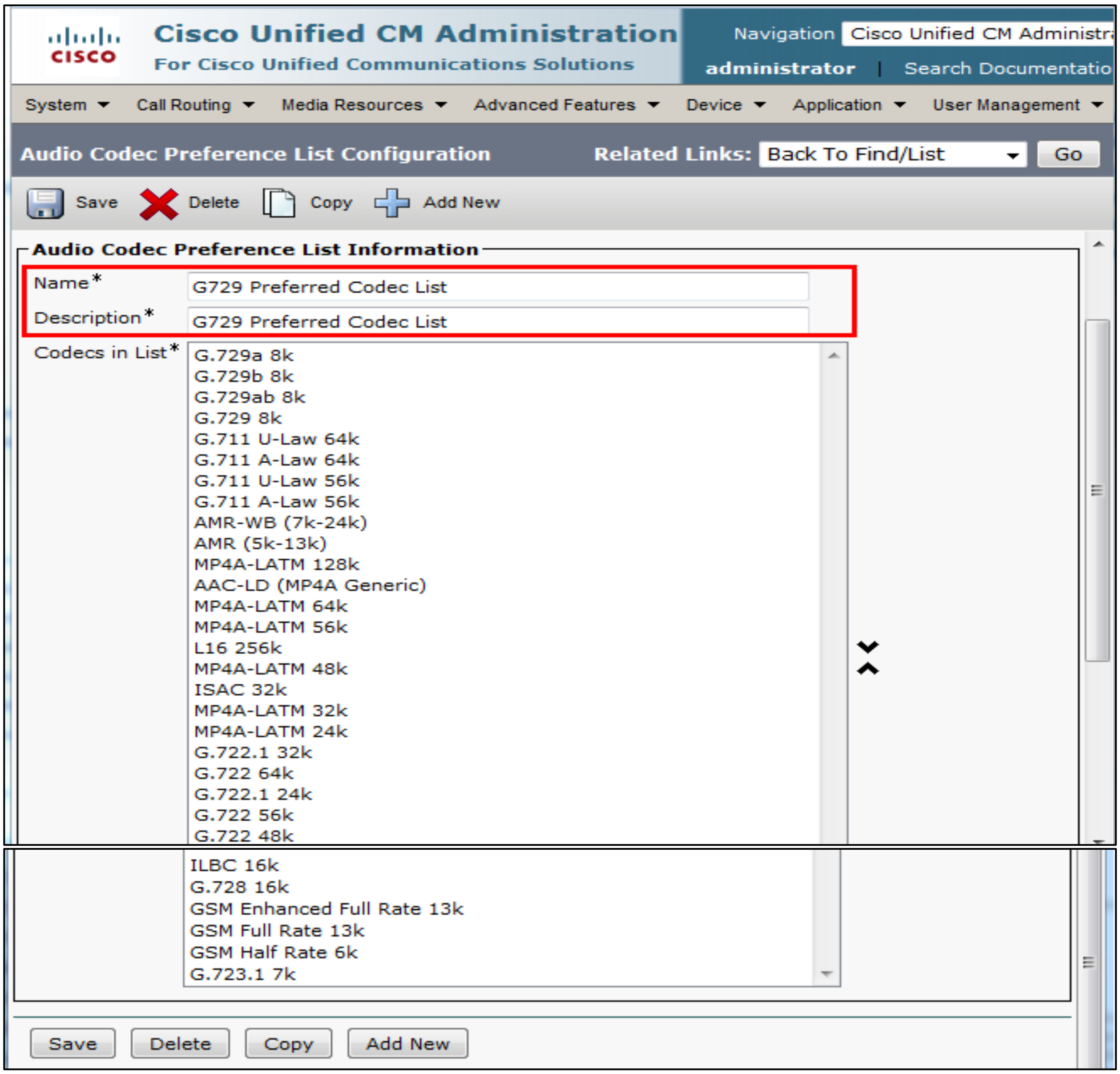
Cisco UCM Version



Cisco UCM Audio Codec Preference List

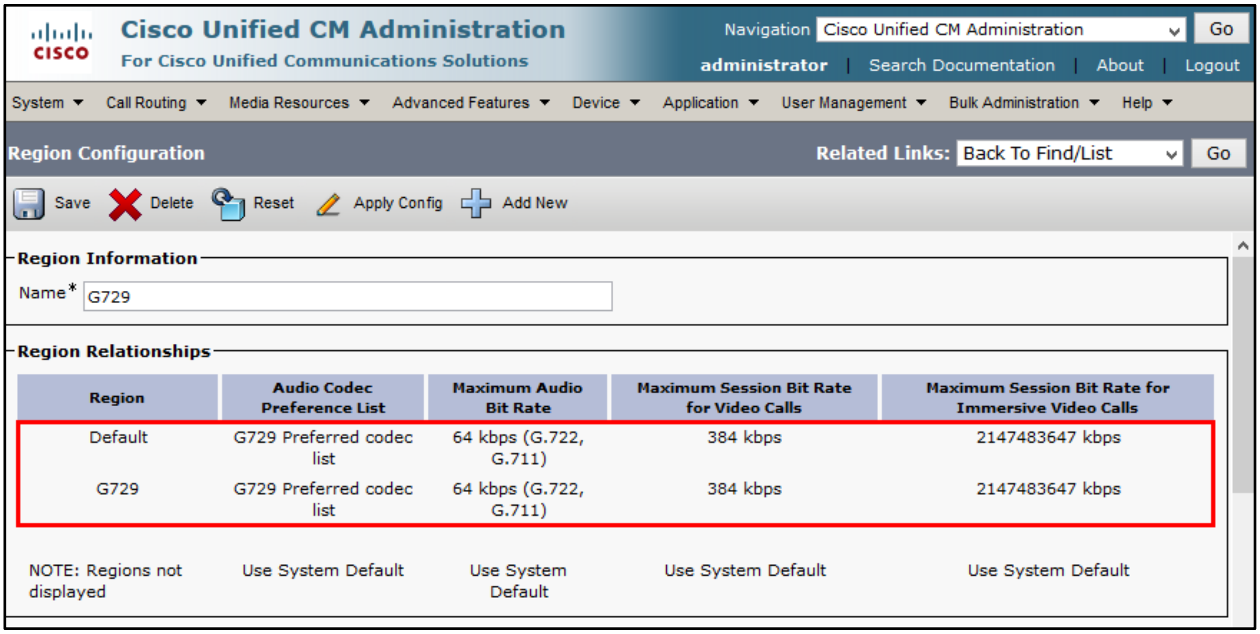
Navigation Path: System → Region Information → Audio codec preference list

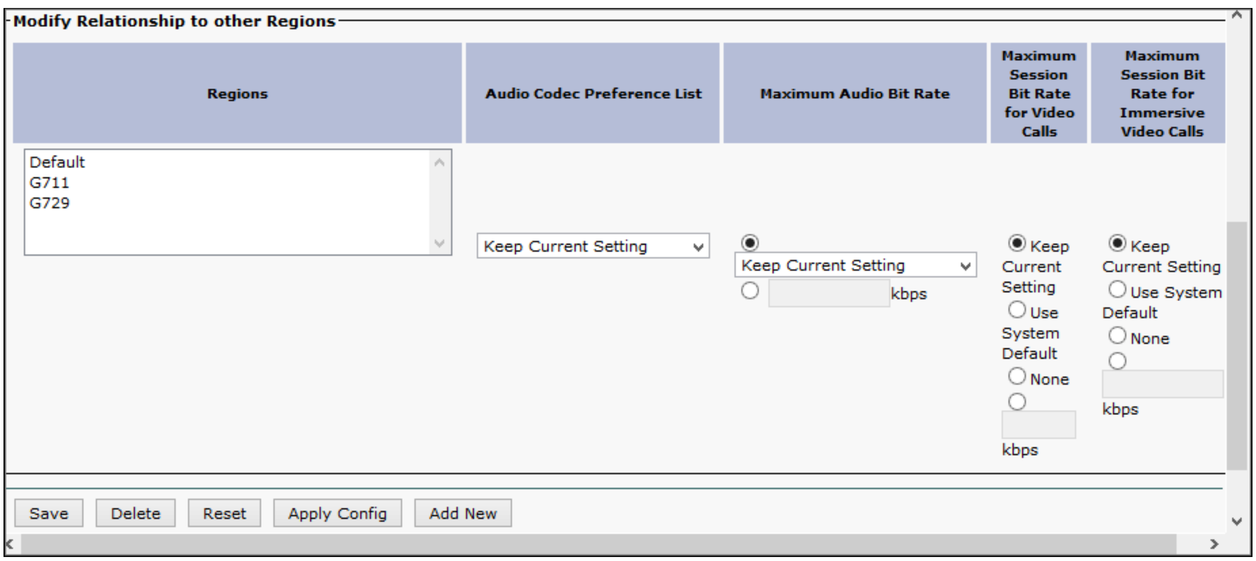
Cisco UCM 10.5.2 has a feature called Audio Codec Preference List. This feature allows to configure the order of audio codec preference both for Inter and Intra Region calls. Audio Codec Preference list is assigned to the Region used by the Device Pool for Phones and by Conference Bridges. Based on user 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

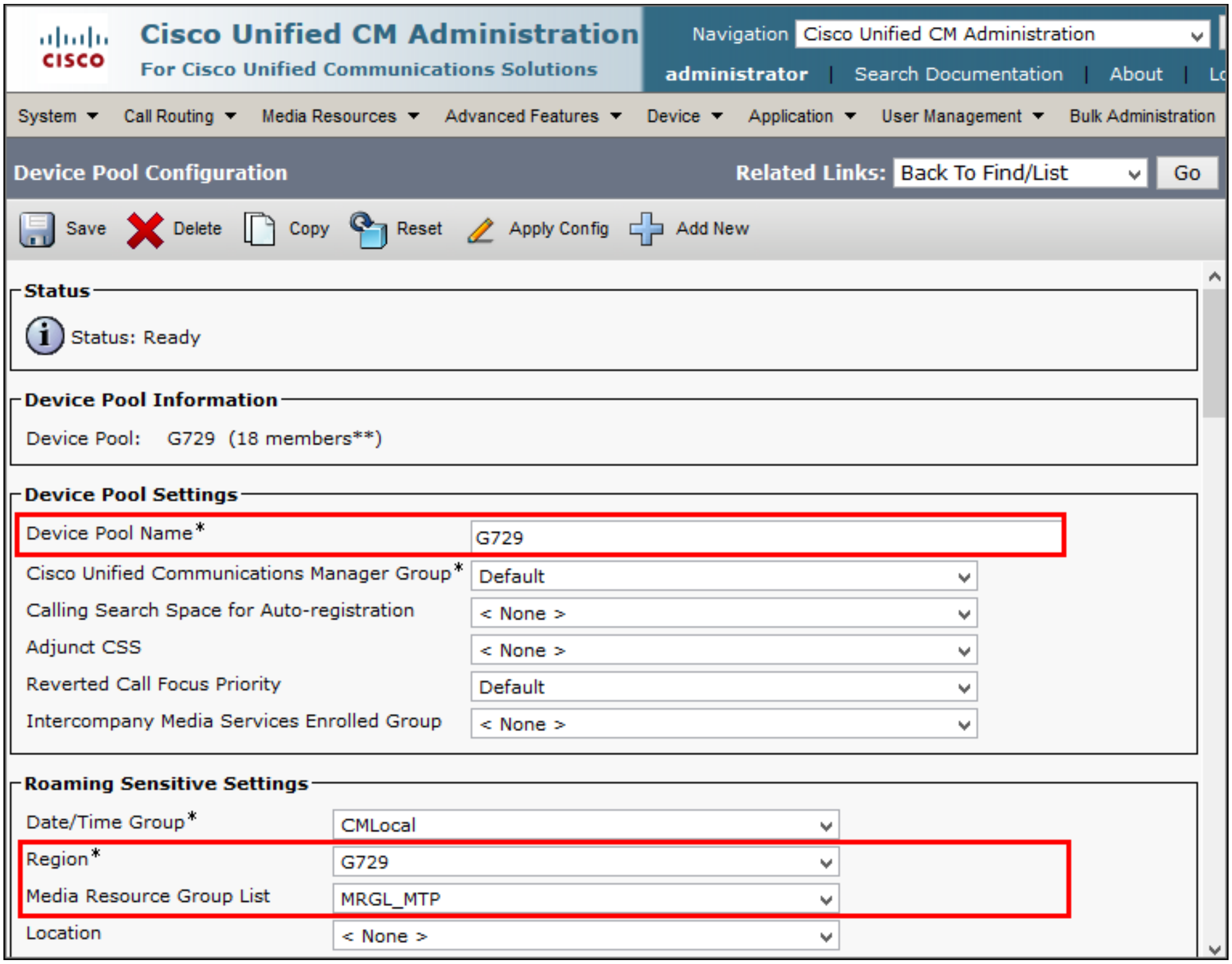


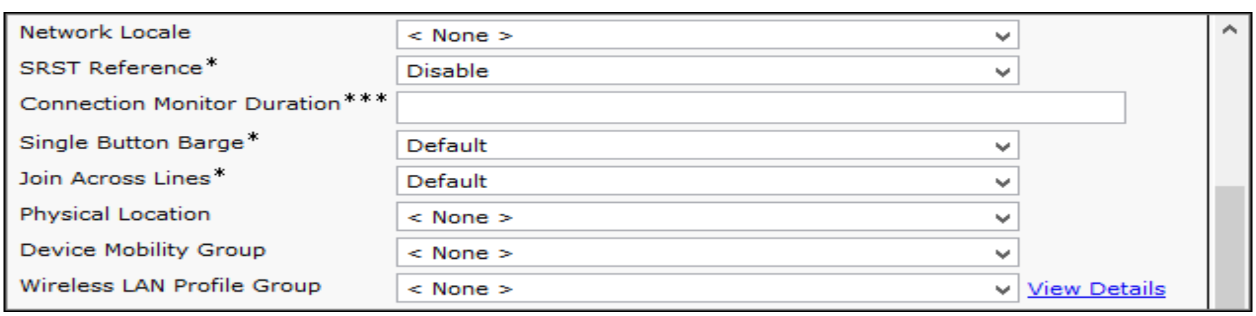


Device Pool Configuration

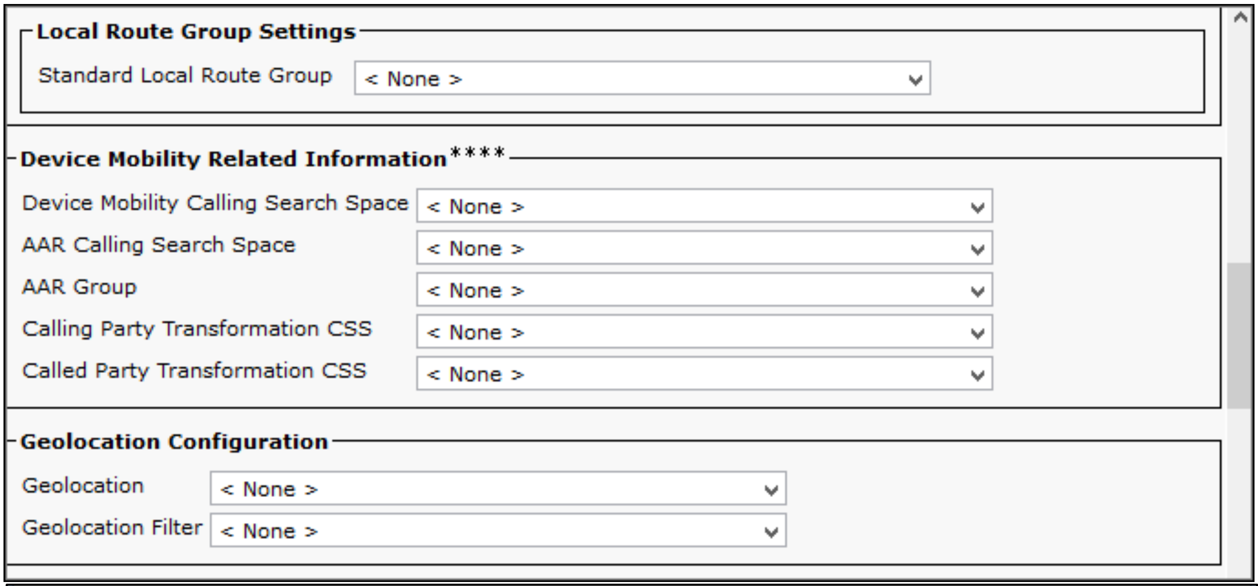
Navigation Path: System → Device Pool

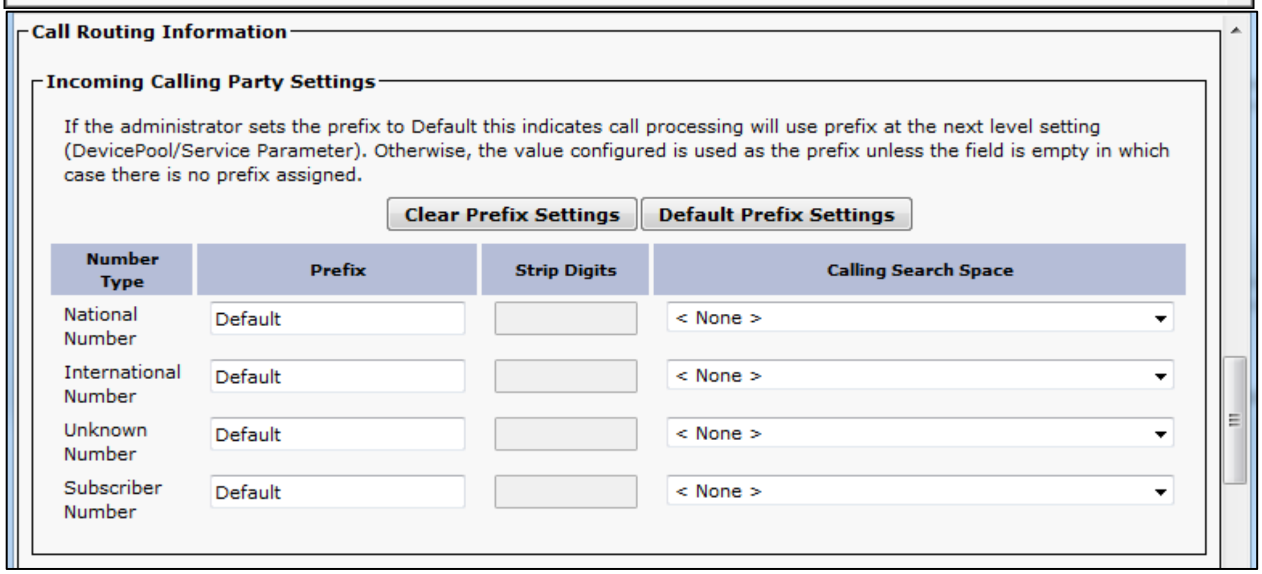
"G729_pool" Device Pool is configured for testing the interoperability. No special consideration needs to be taken when configuring the Device Pools. Optionally, a Media Resource Group List can be added to the Device Pools, if needed, to assign selected Media Resources (Conference Bridges, Transcoders,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



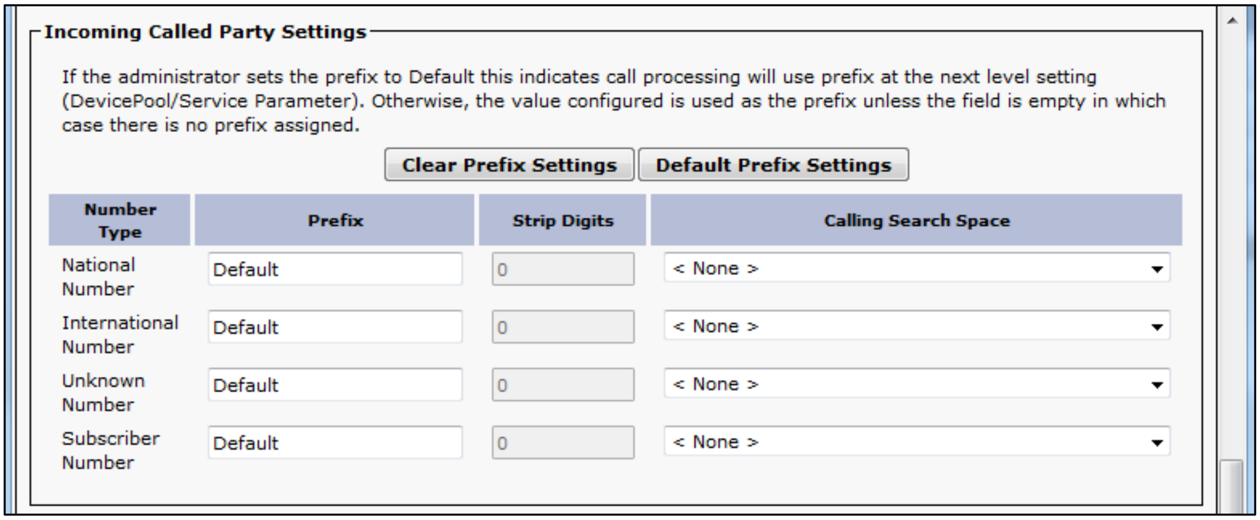


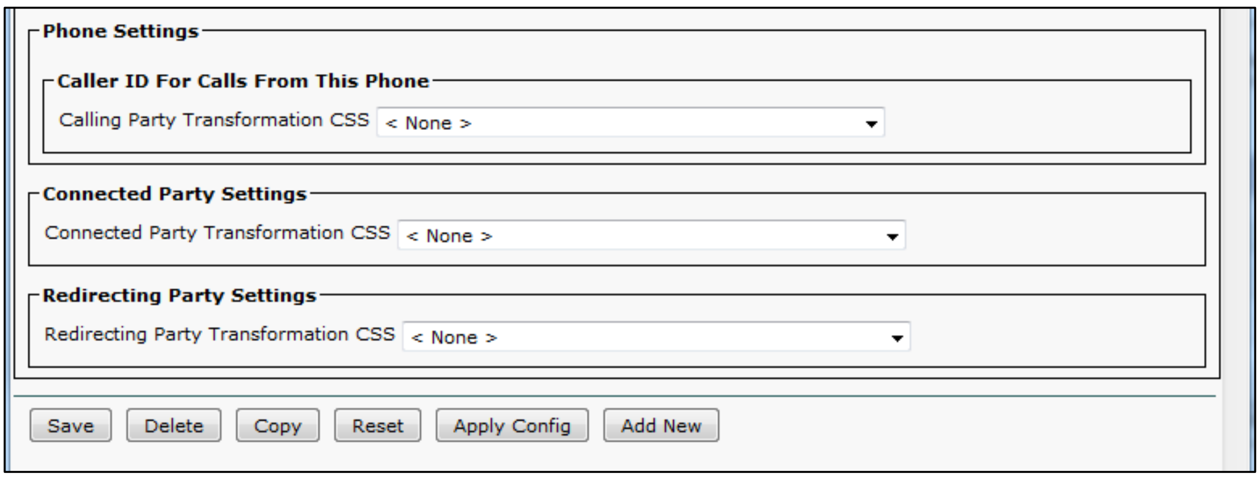
Device Pool Configuration (Contd.)





Device Pool Configuration (Contd.)

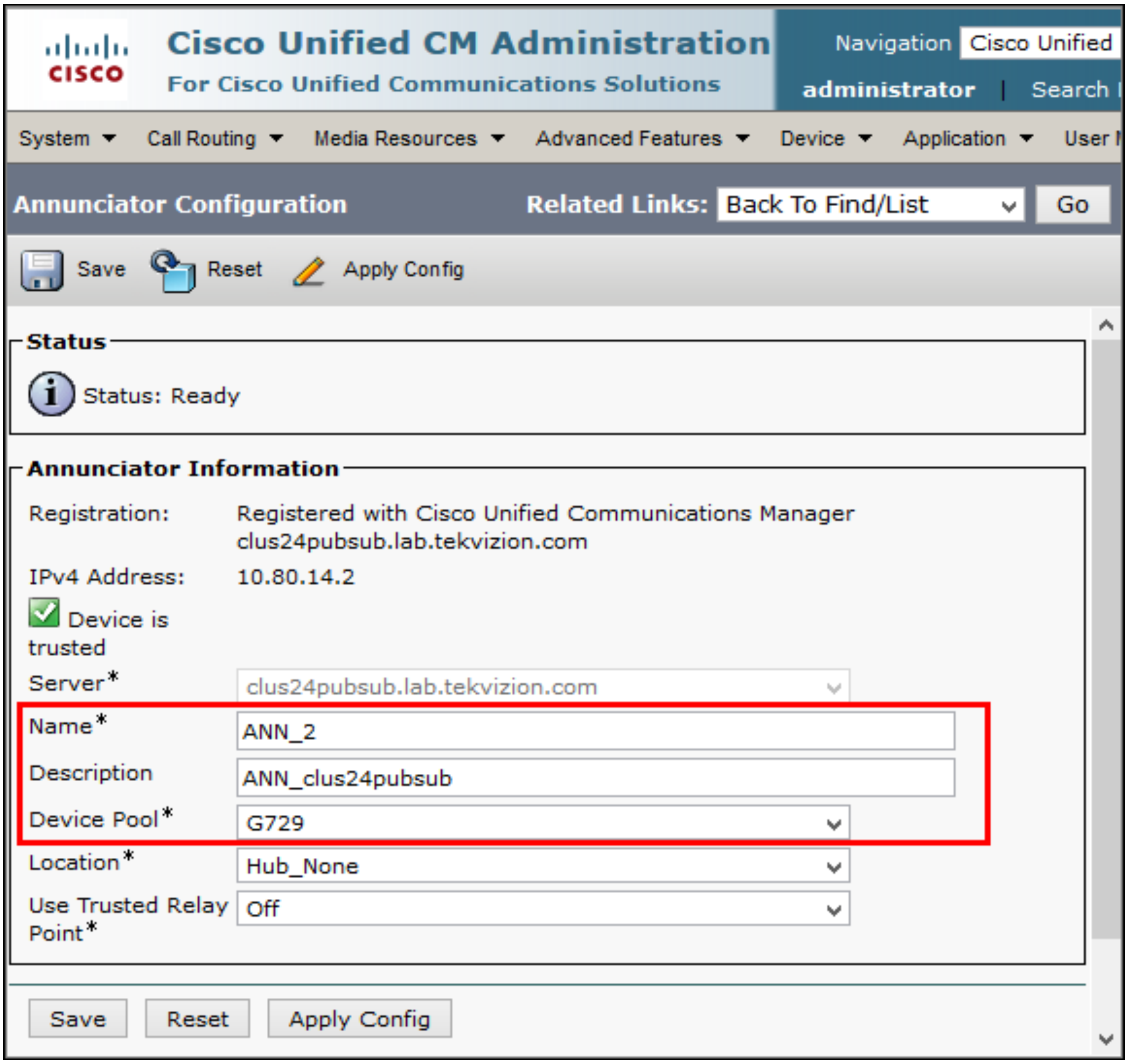




Annunciator Configuration

Navigation: Media Resource → Annunciator

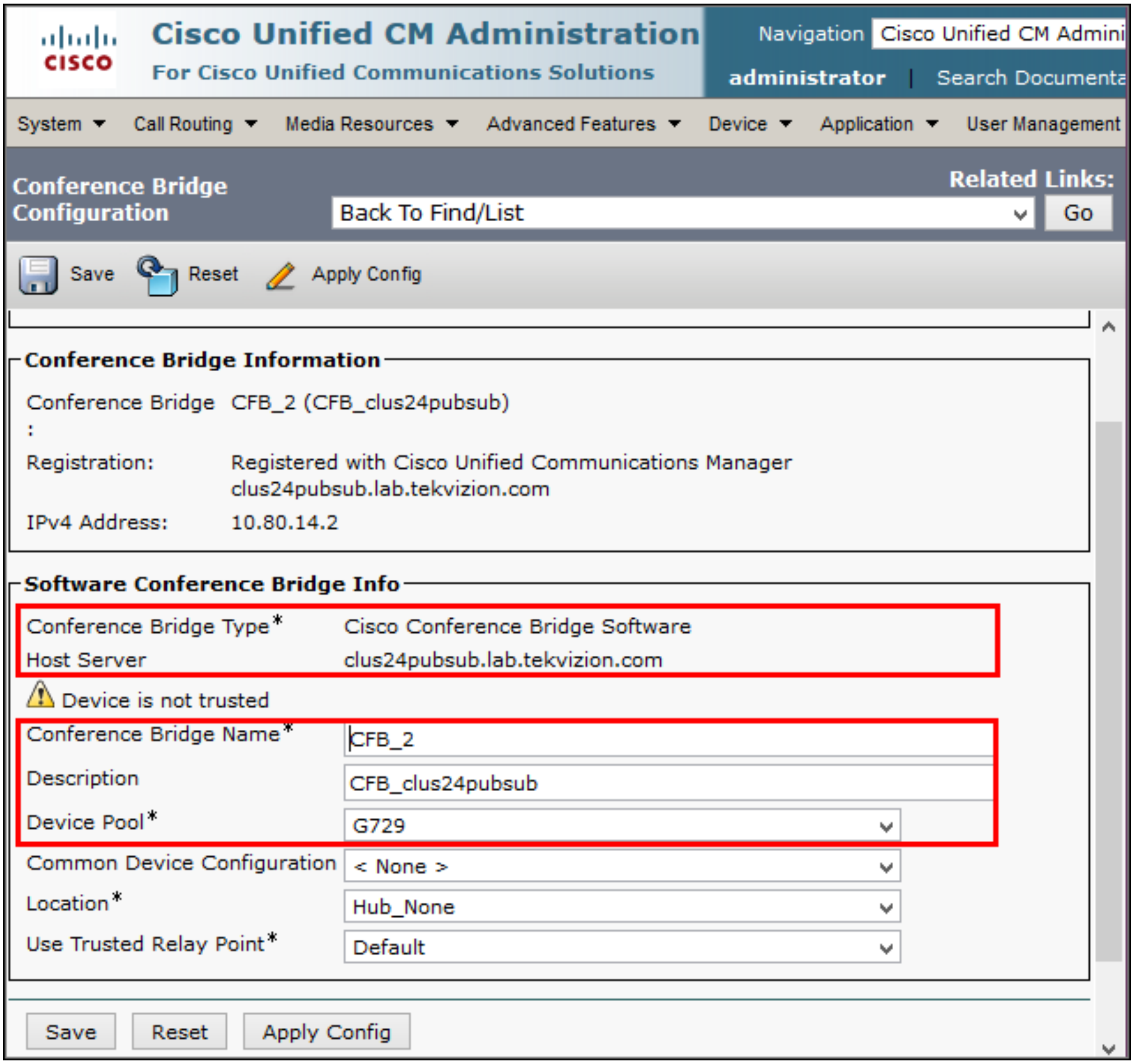
Set Name* = ANN_2.
Set Description = ANN_clus24pubsub. This is used for this example
Set Device Pool* = G729



Conference Bridge Configuration

Navigation: Media Resources → Conference Bridge

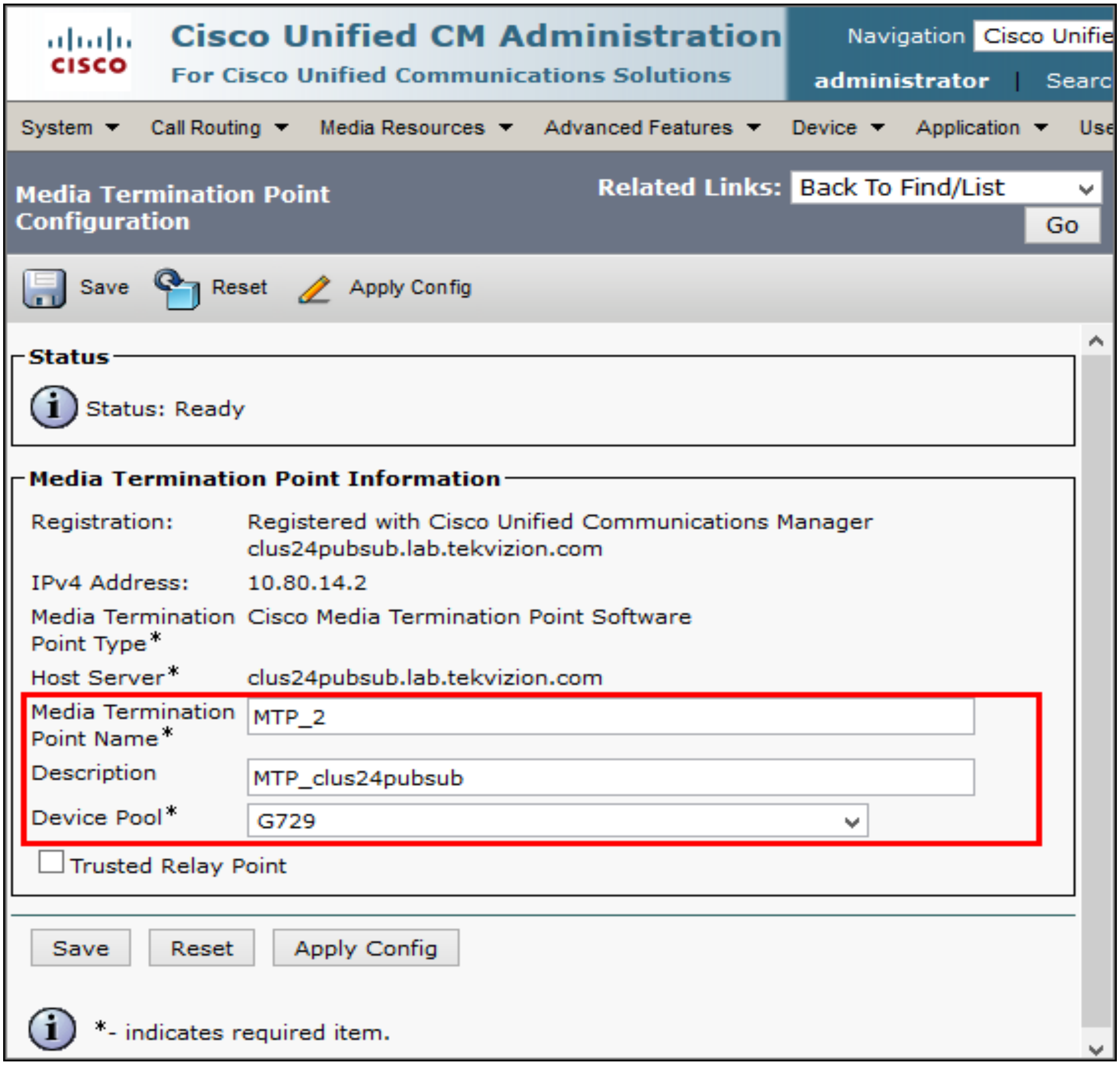
Set Conference Bridge Type* = Cisco Conference Bridge Software.
Set Host Server = clus24pubsub.lab.tekvizion.com. This is used for this example.
Set Conference Bridge Name* = CFB_2.
Set Description = CFB_clus24pubsub. This is used in this example.
Set Device Pool* = G729



Media Termination Point Configuration

Navigation: Media Resource→Media Termination Point

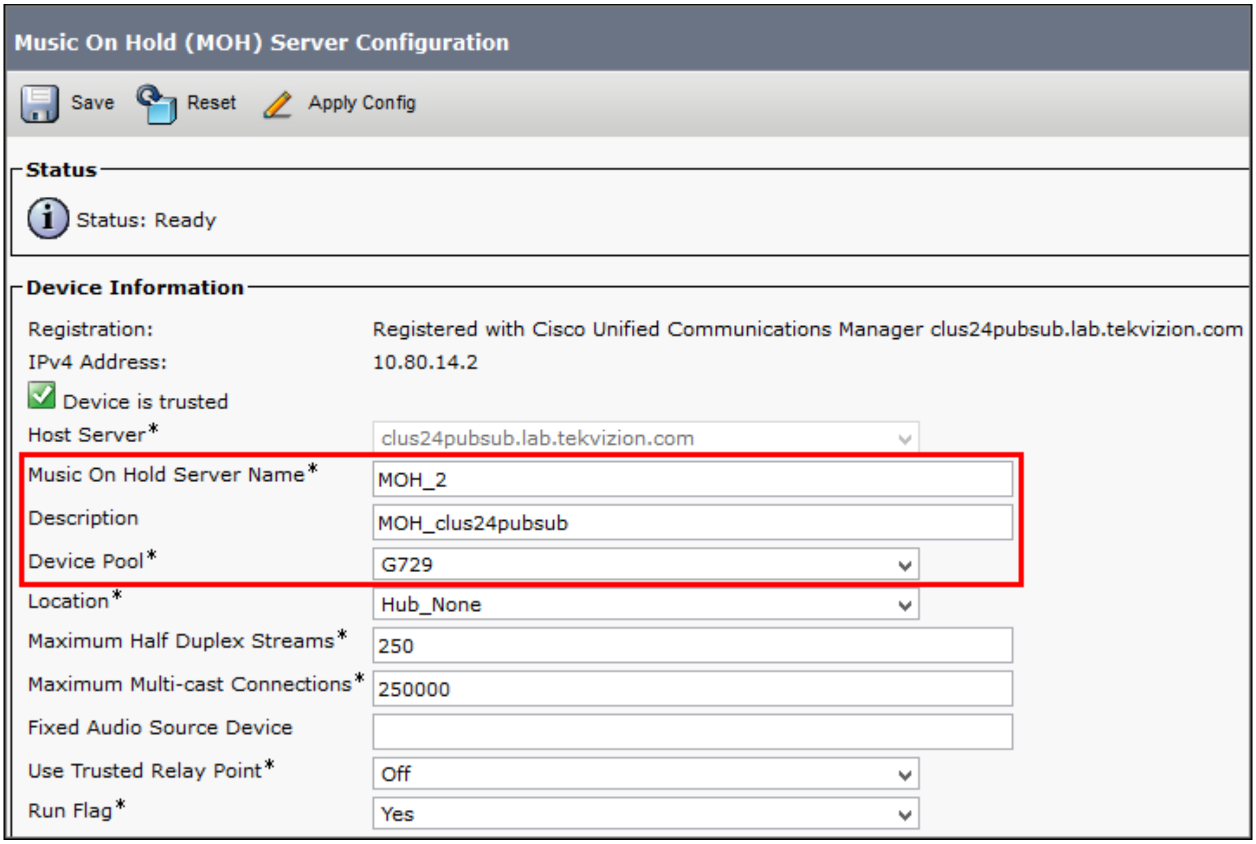
Set Media Termination Point Name* = MTP_2
Set Description = MTP_clus24pubsub. This is used for this example
Set Device pool* = G729

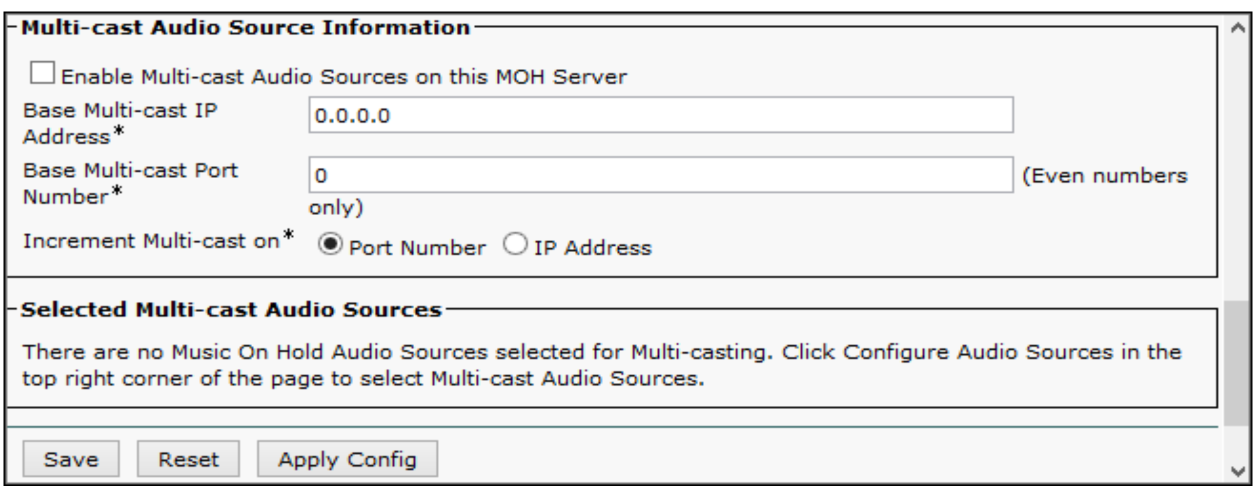


Music on Hold Server Configuration

Navigation: Media Resources → Music on Hold Server

Set Music on Hold Server Name* = MOH_2.
Set Description = MOH_clus24pubsub. This is used for this example.
Set Device Pool* = G729

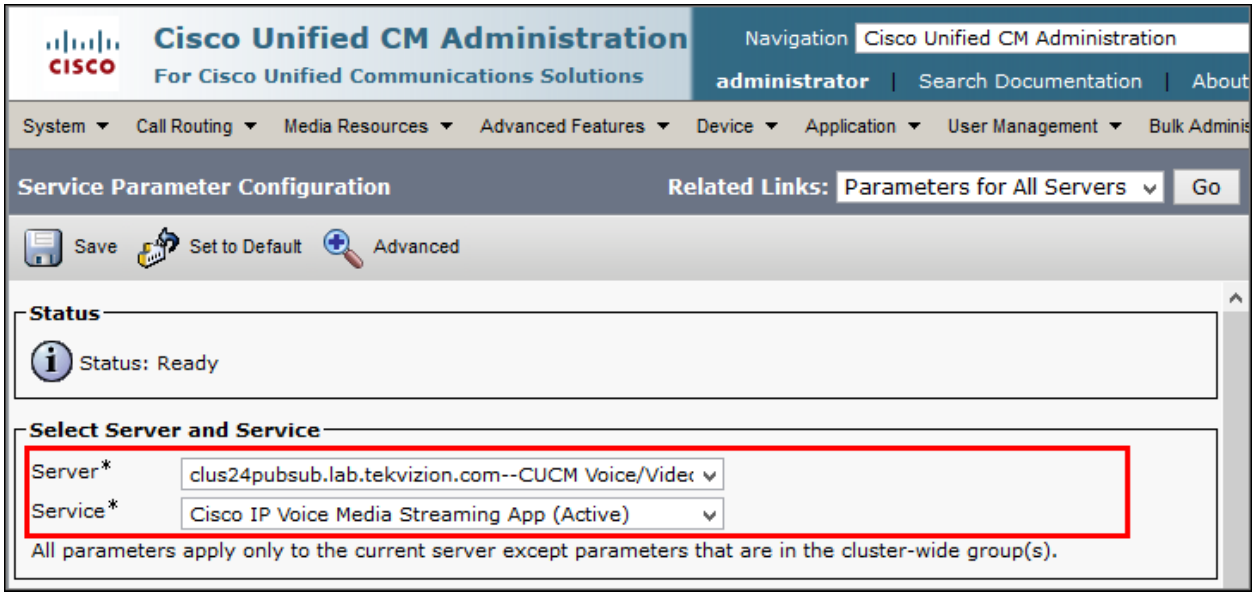


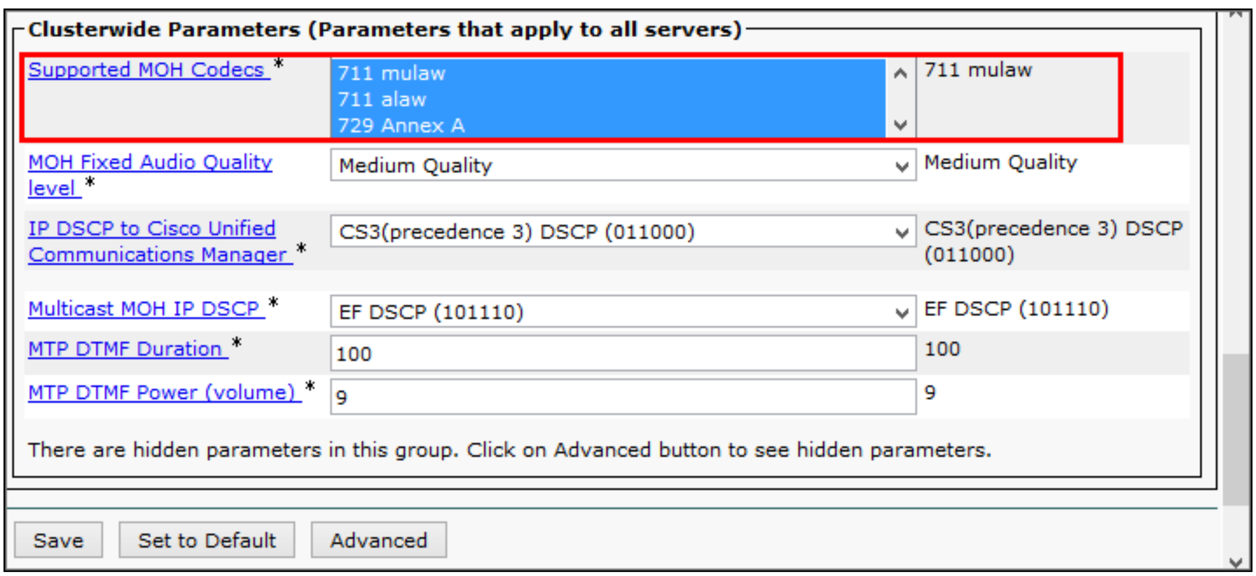


Music on Hold Service (IP Voice Media Streaming App) Parameter Settings

Navigation: System → Service Parameter

Note: Make sure codecs G.729 Annex A and G.711 mu-law are configured in parameter Supported MOH Codecs.
Select Server* = clus24pubsub.lab.tekvizion.com (Active). This is used in this example.
Select Service* = Cisco IP Voice Media Streaming App (Active).

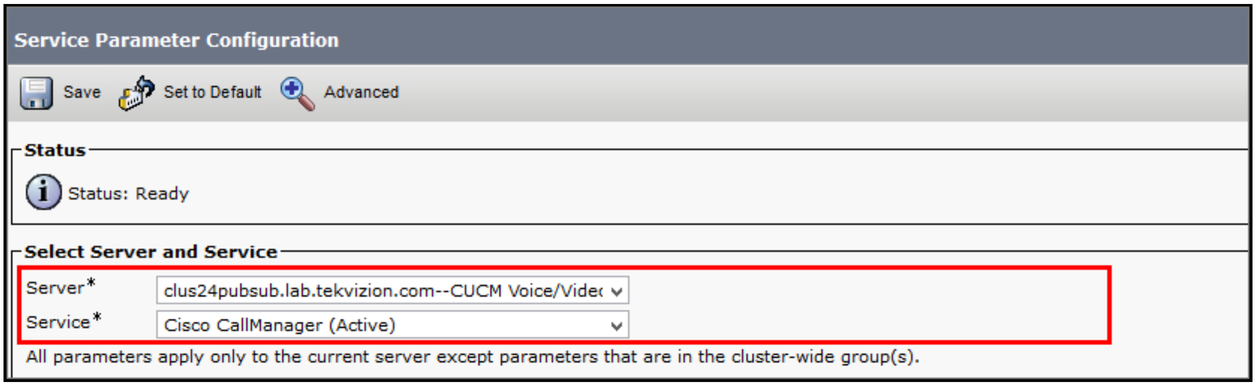


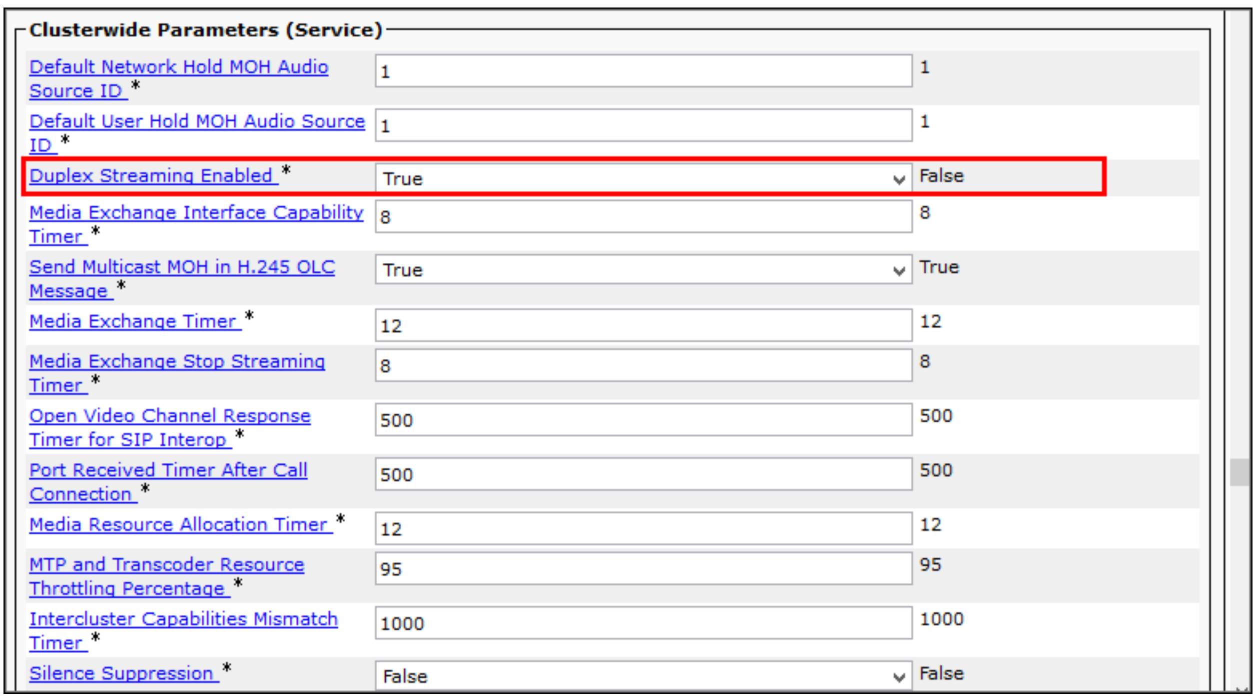


Music on Hold Service (Duplex Streaming) Parameter Settings

Navigation: System → Service Parameter

Select Server* = clus24pubsub.lab.tekvizion.com (Active). This is used in this example.
Select Service* = Cisco Call Manager (Active).
Select Duplex Streaming Enabled * = True



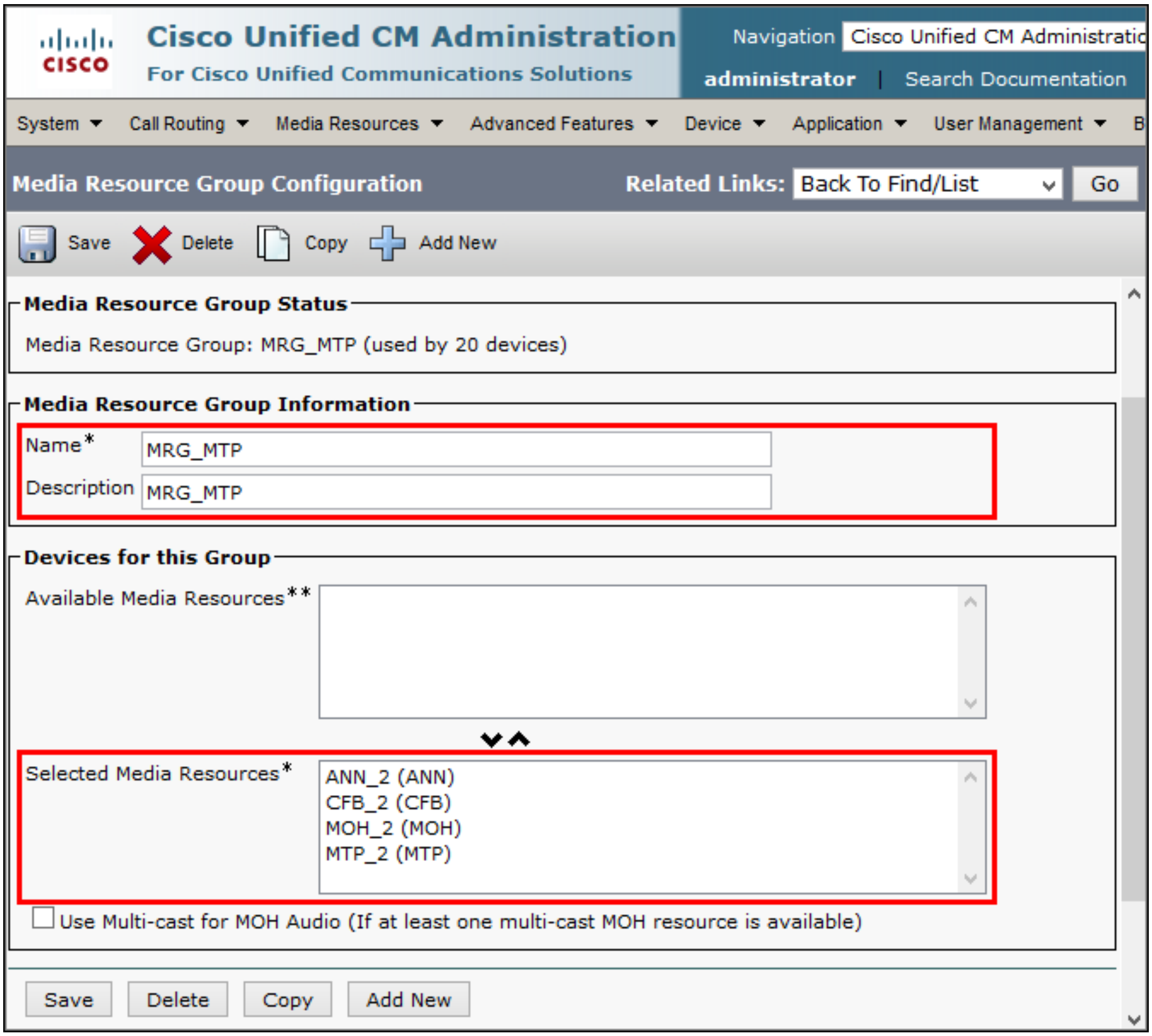


Media Resource Group Configuration

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.

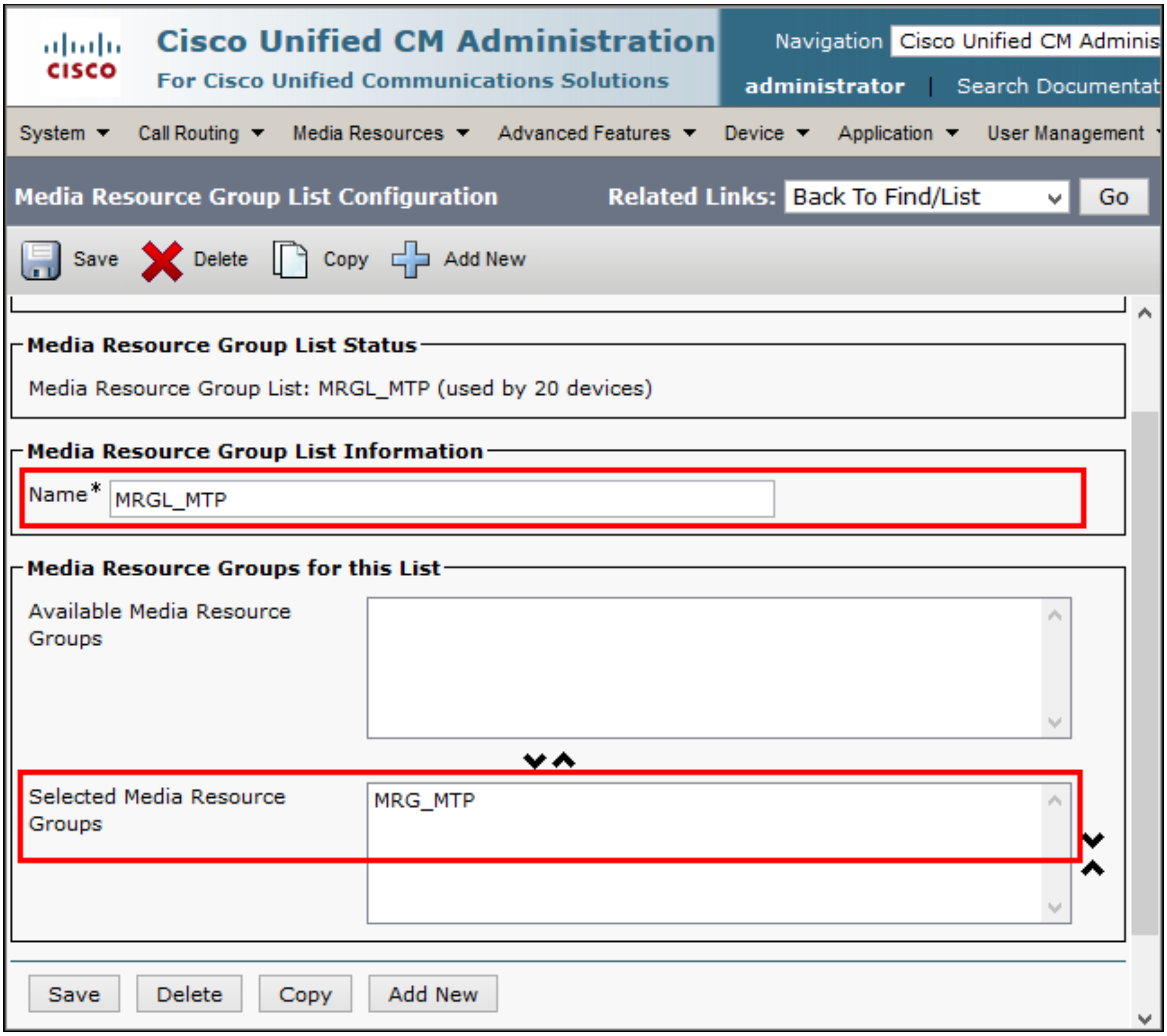
Set Name*= MRG_MTP. This is used for this example.
Set Description = MRG_MTP. This text is used to define this Media Resource Group List.
Set all Resources in the selected Media Resources Box.



Media Resource Group List Configuration

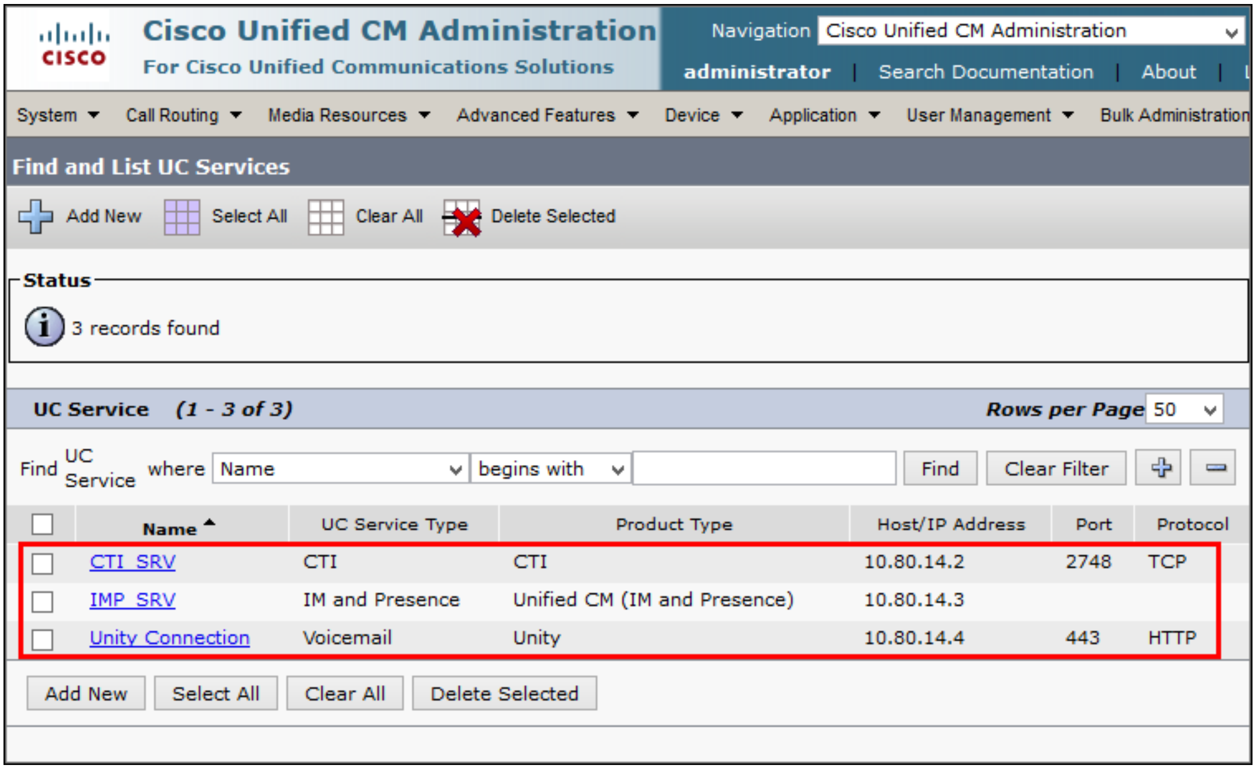
Navigation Path: Media Resources → Media Resource Group List

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



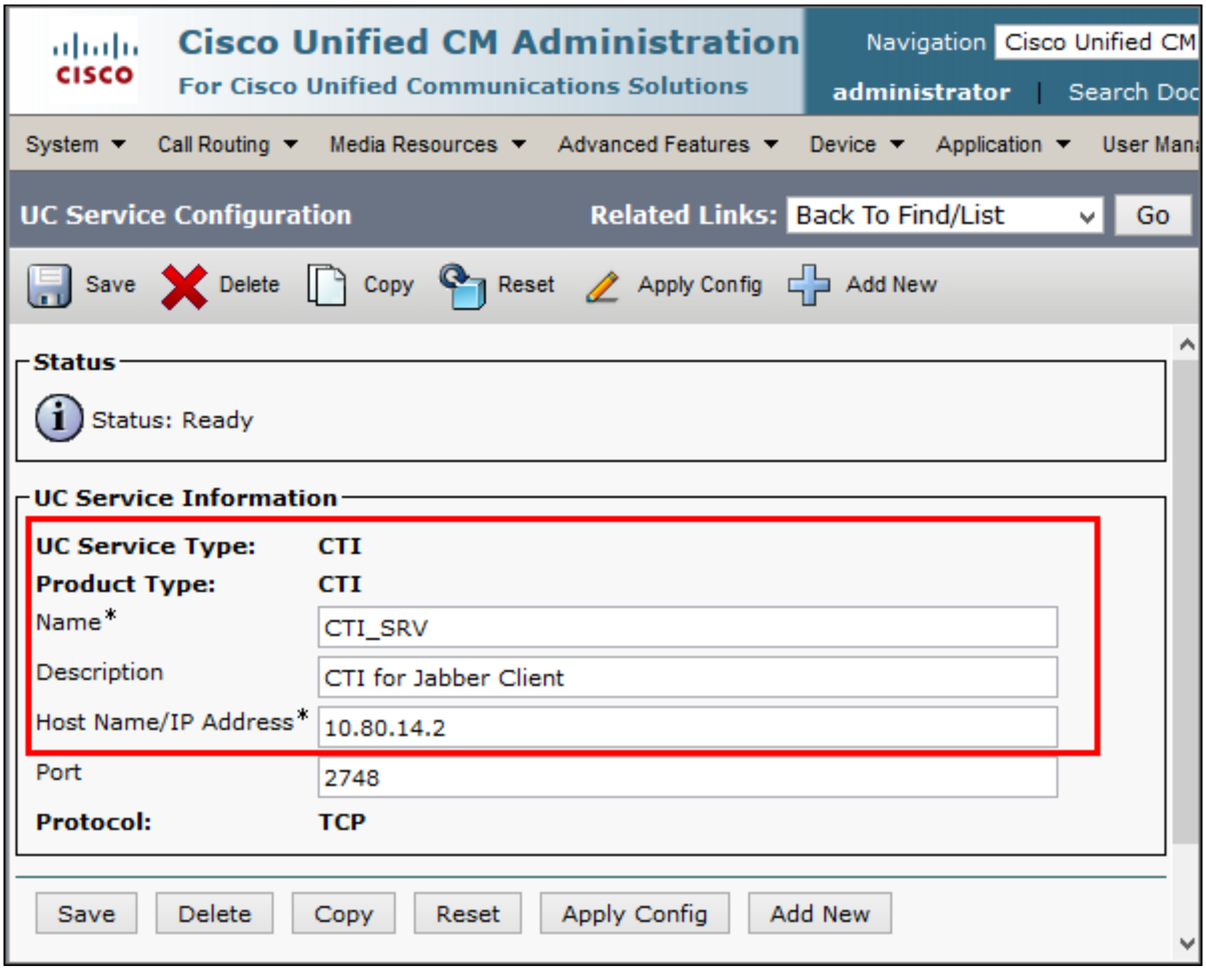
UC Service Configuration

Navigation: User Management → User Settings → UC Service



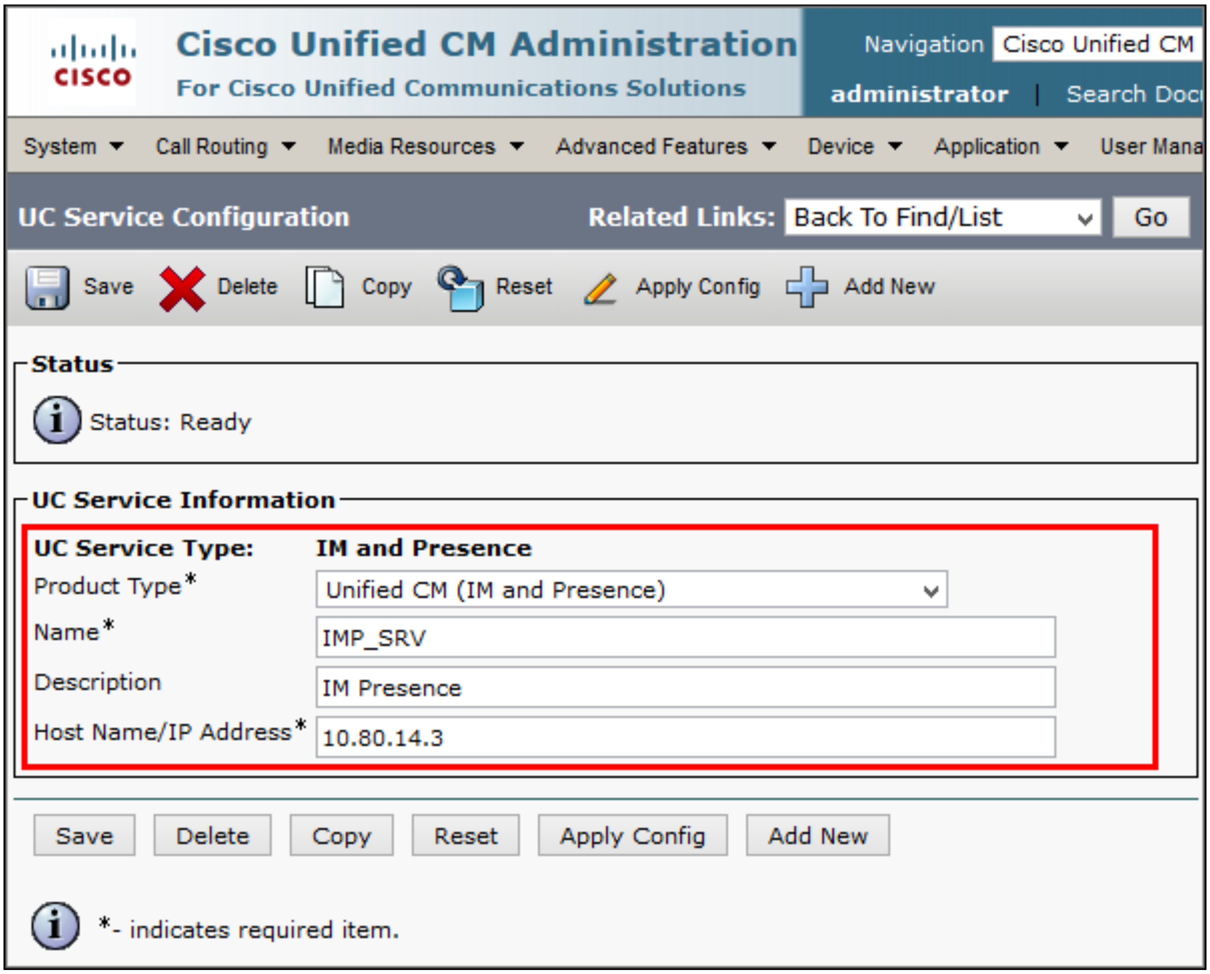
UC Service Configuration (Contd.)

Select UC Service Type: = CTI
Set Name* = CTI_SRV. This is used in this example.
Set Description = CTI for Jabber Client. This is used in this example.
Set Host Name/IP Address* = 10.80.14.2 (Cisco UCM Address)



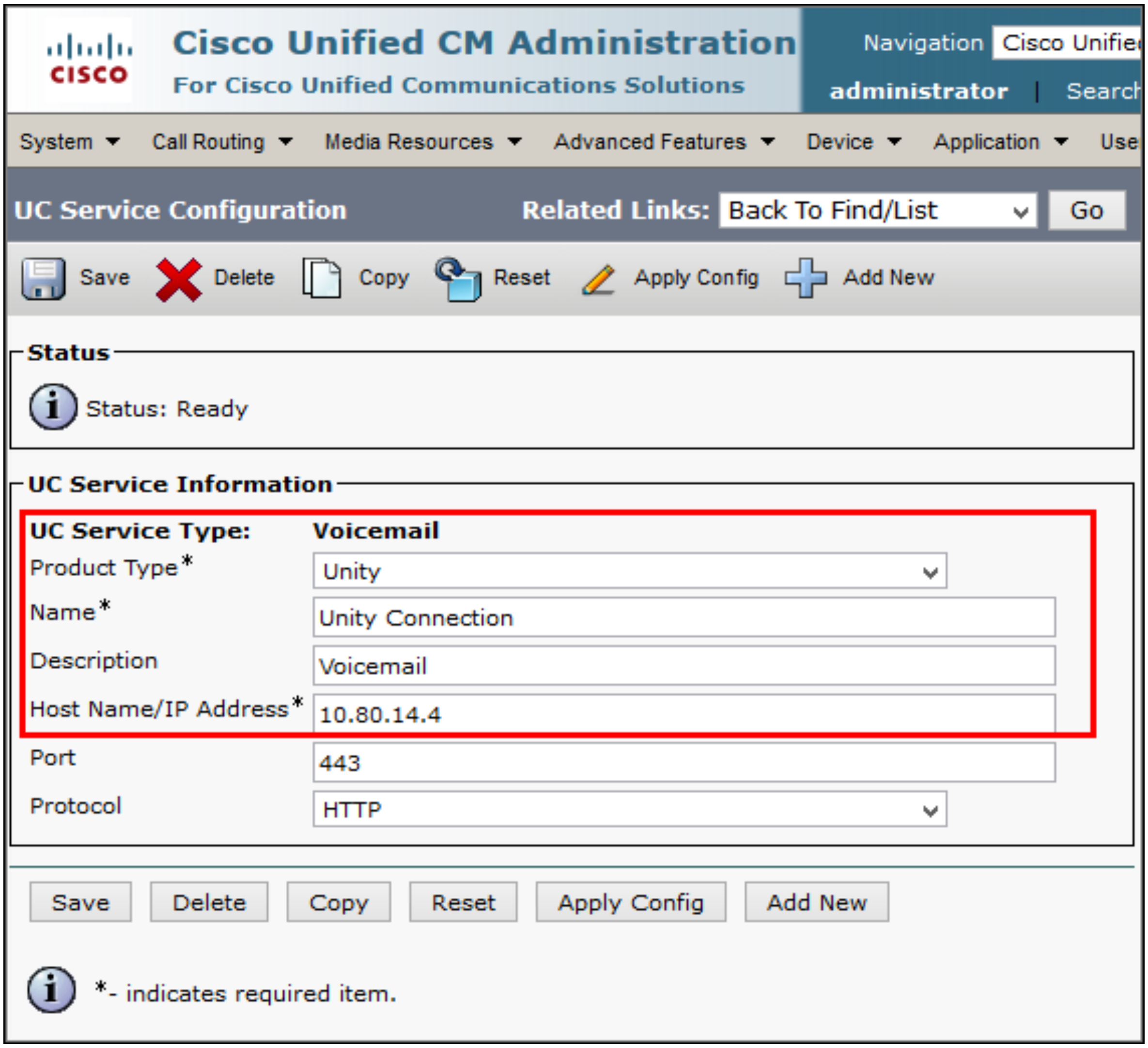
UC Service Configuration (Contd.)

Select UC Service Type: = IM and Presence
Set Name* = IMP_SRV. This is used in this example.
Set Description = IM Presence. This is used in this example.
Set Host Name/IP Address* = 10.80.14.3 (Cisco UCM IM & Presence IP Address)



UC Service Configuration (Contd.)

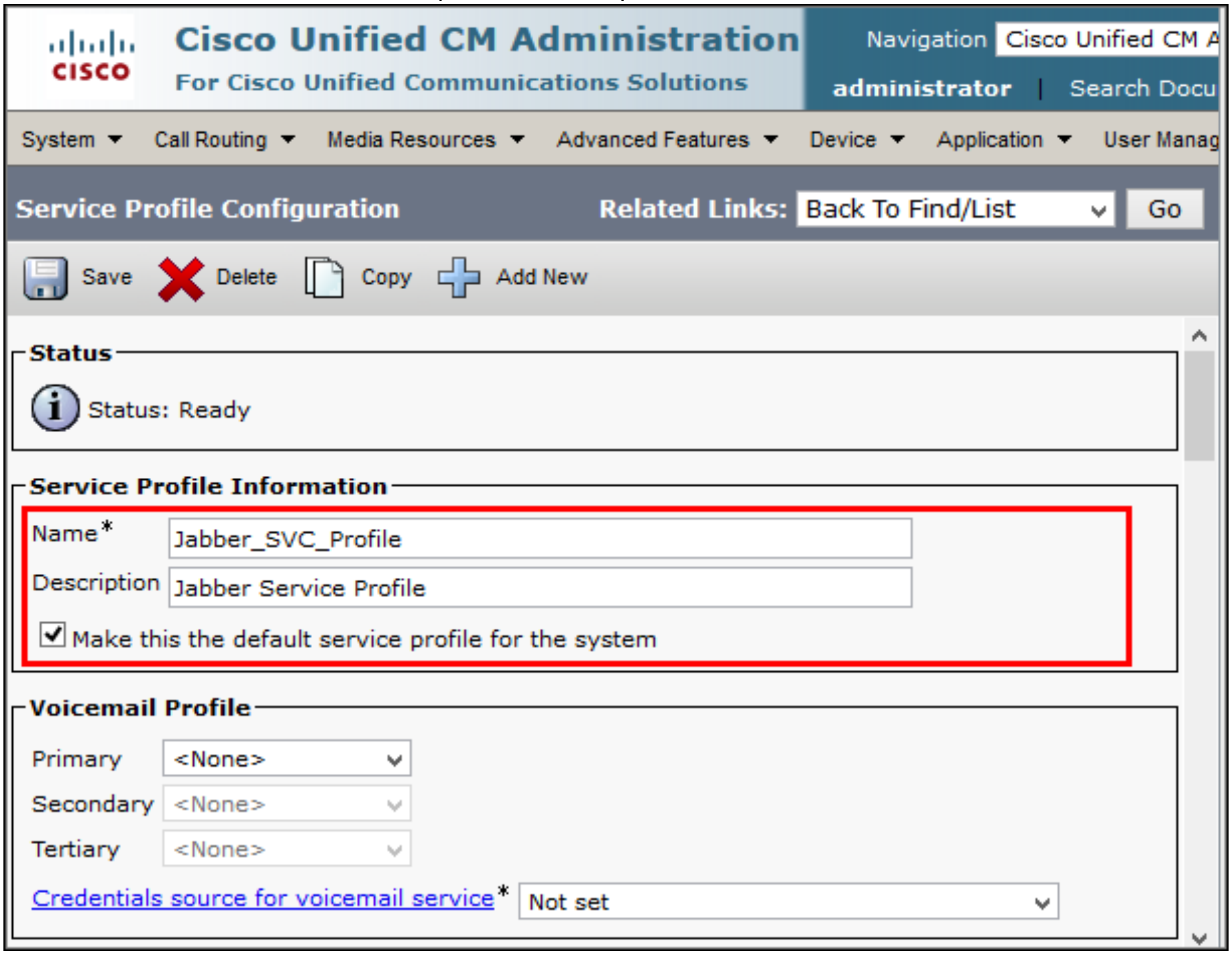
Select UC Service Type: = Voicemail
Set Name* = Unity Connection. This is used in this example.
Set Description = Voicemail. This is used in this example.
Set Host Name/IP Address* = 10.80.14.4 (Cisco Unity Connection Administration)

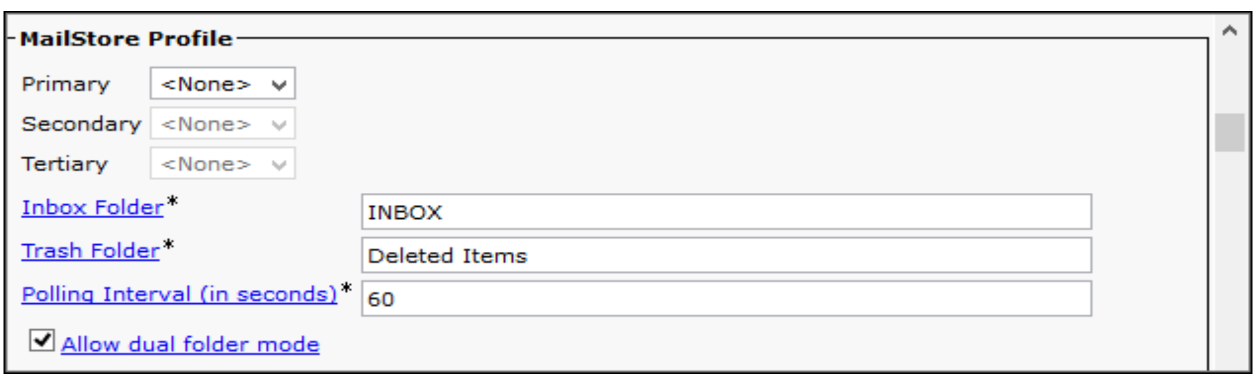


Service Profile Configuration

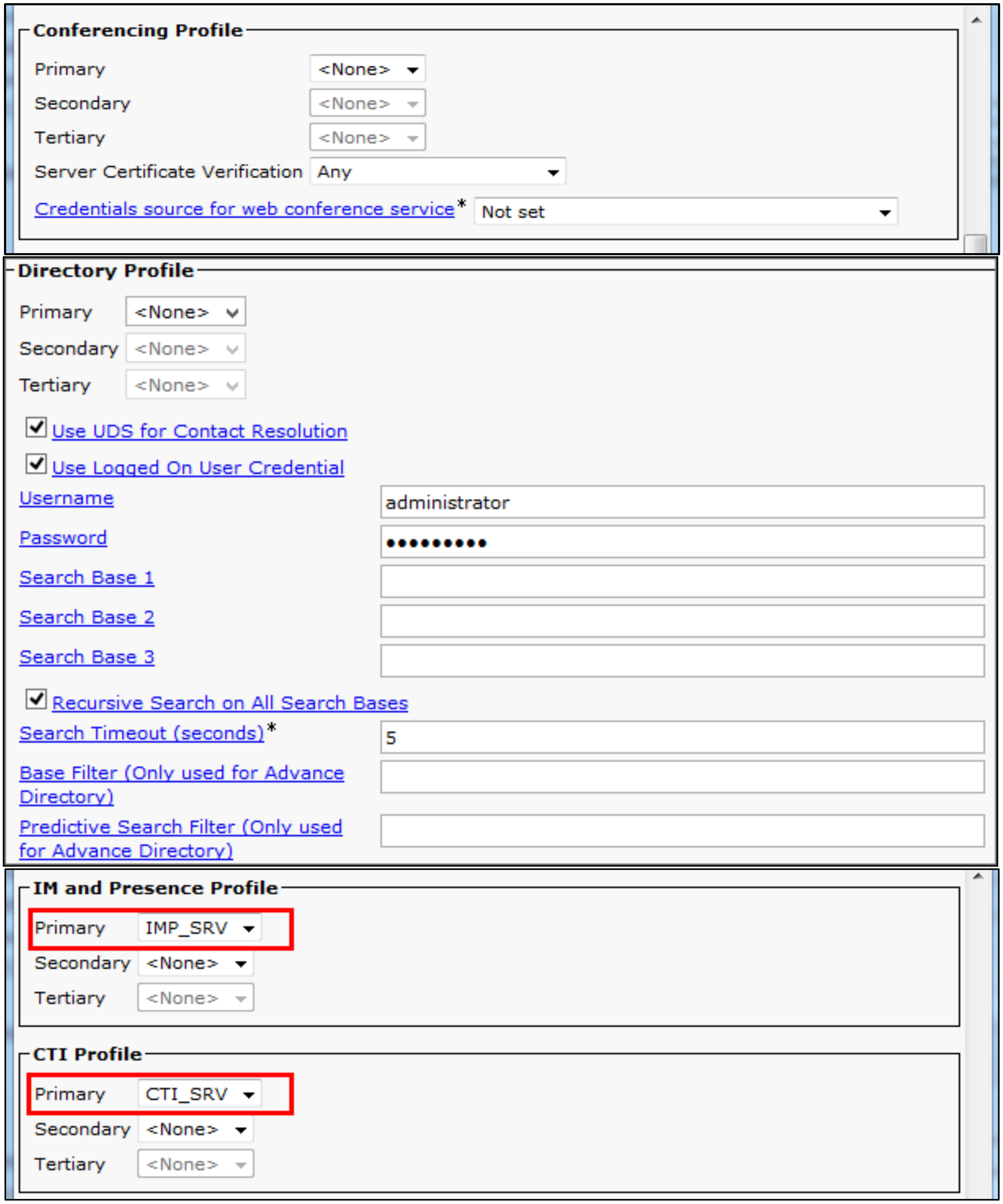
Navigation: User Management → User Settings → Service Profile

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

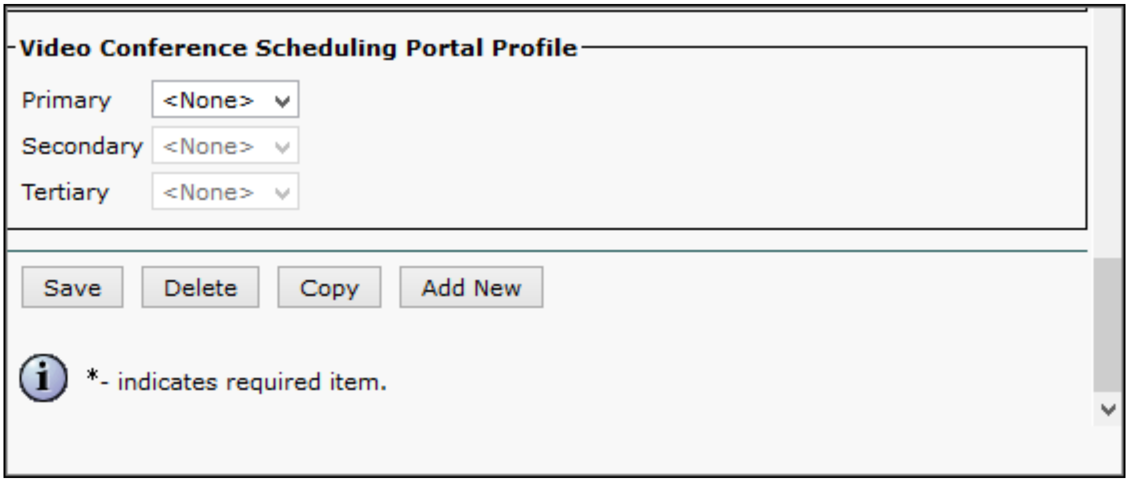




Service Profile Configuration (Contd.)



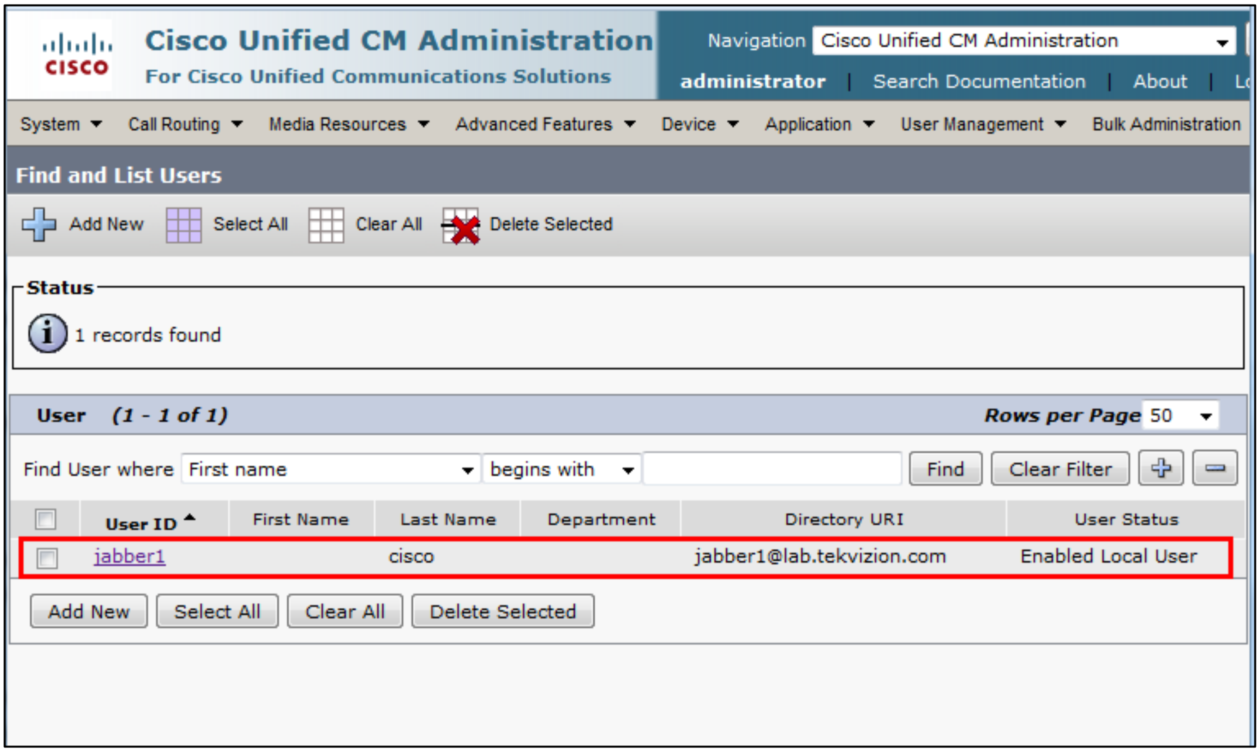
Service Profile Configuration (Contd.)



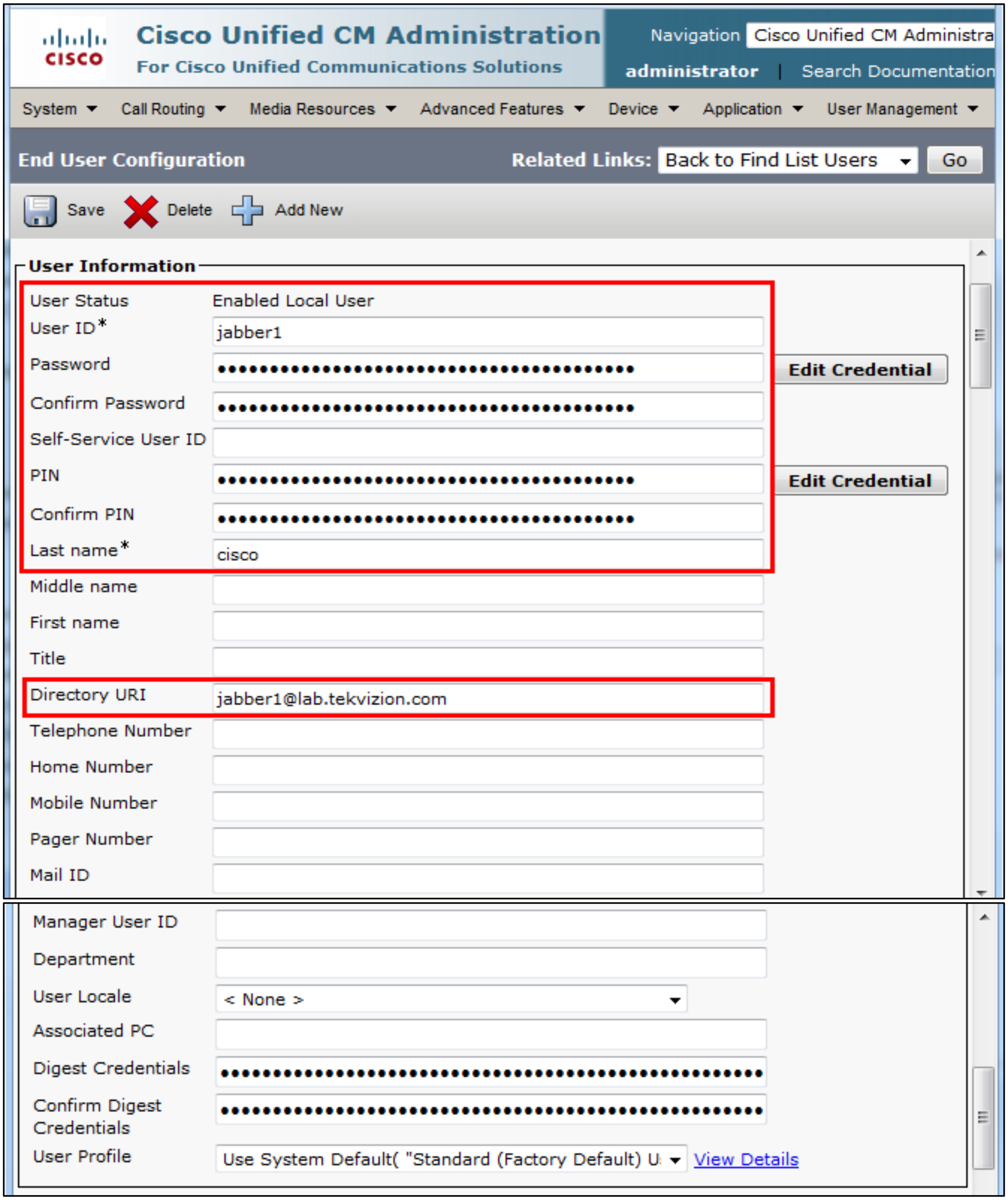
End User Configuration

Navigation: User Management → End User

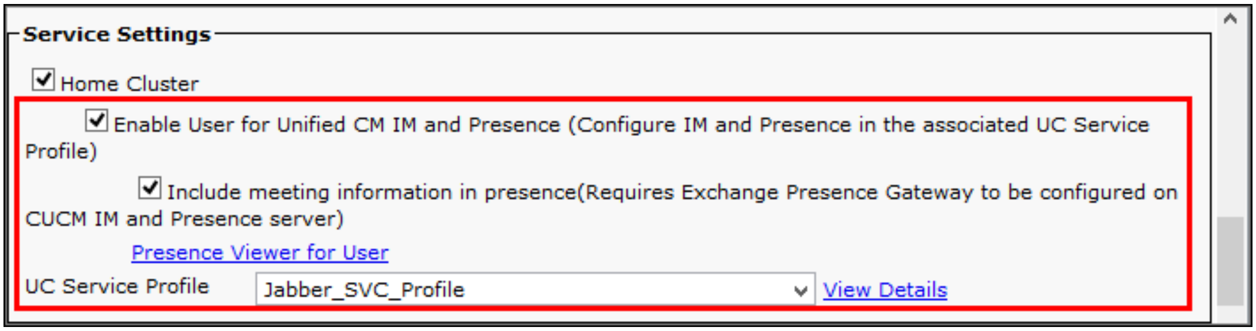
Set User ID* = jabber1. This is used in this example.
Set Password = Password for profile.
Set Directory URI = jabber1@lab.tekvizion.com

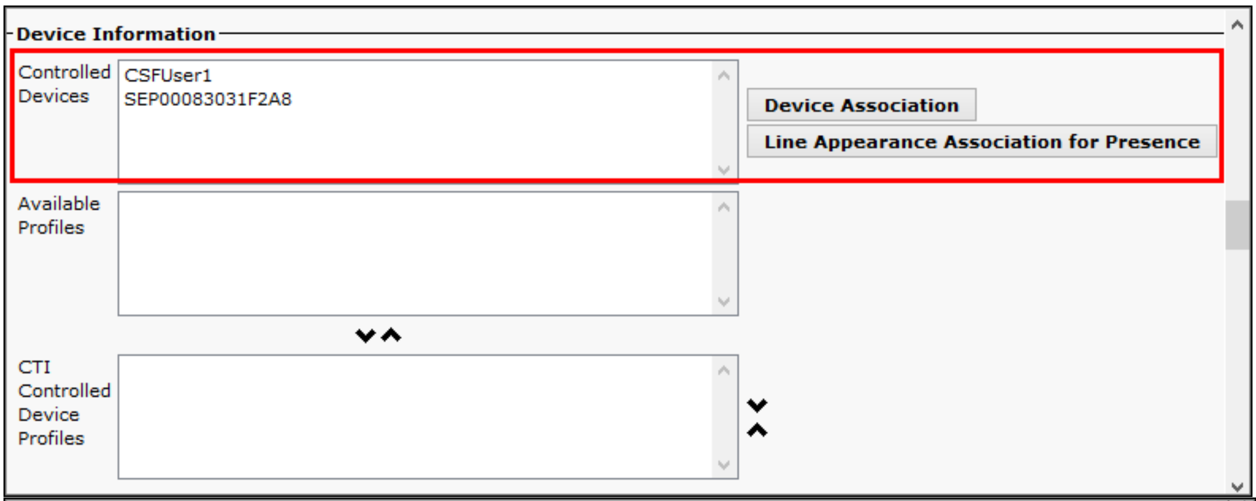


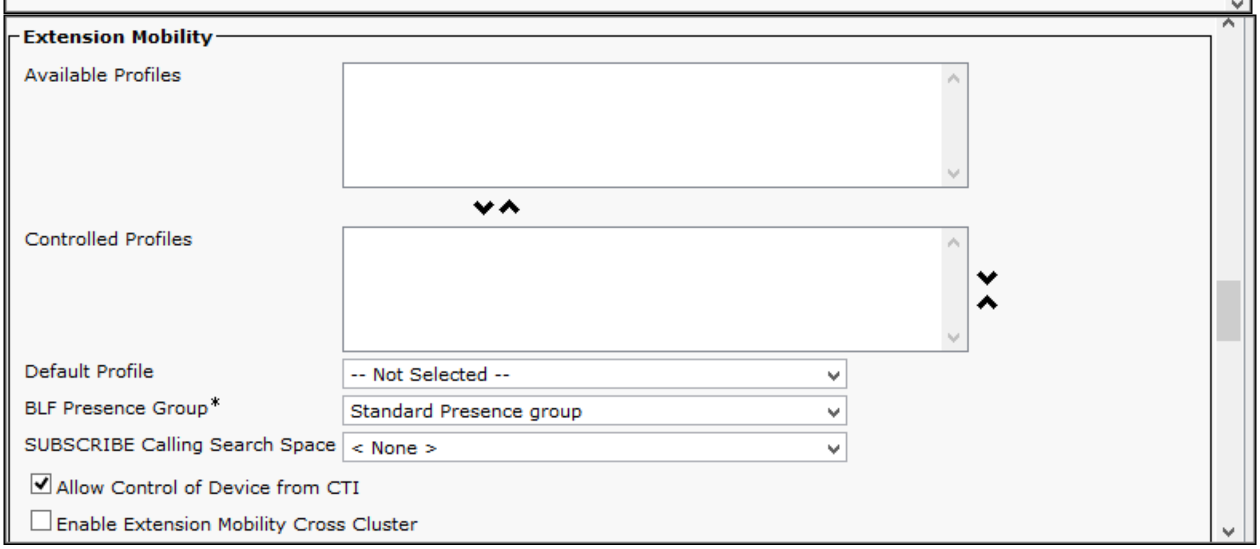
End User Configuration (Contd.)



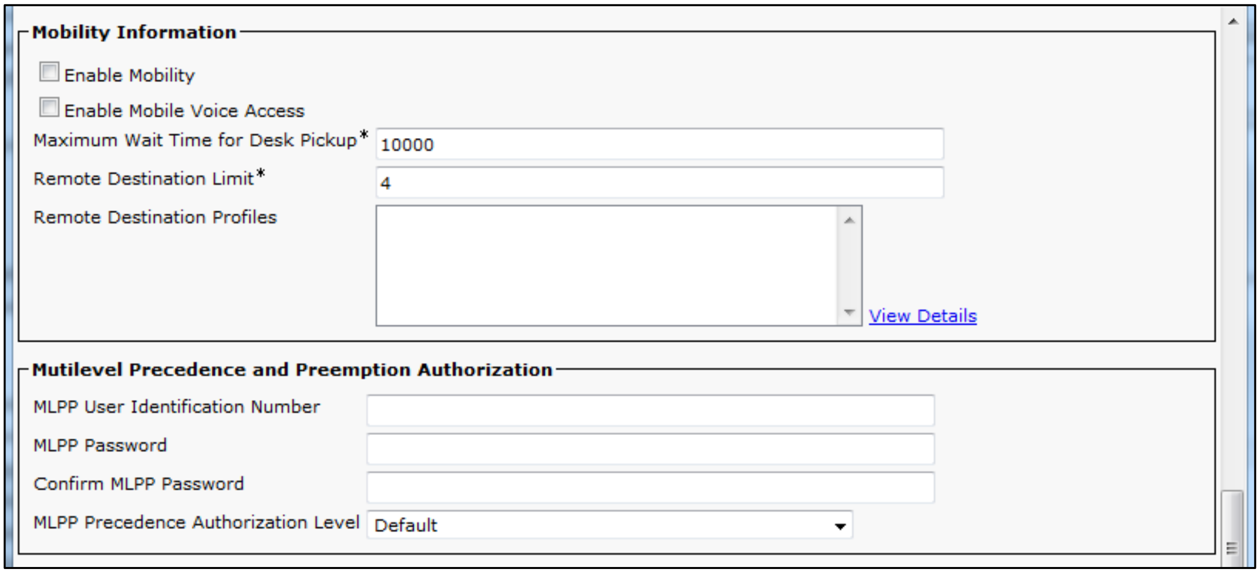
End User Configuration (Contd.)

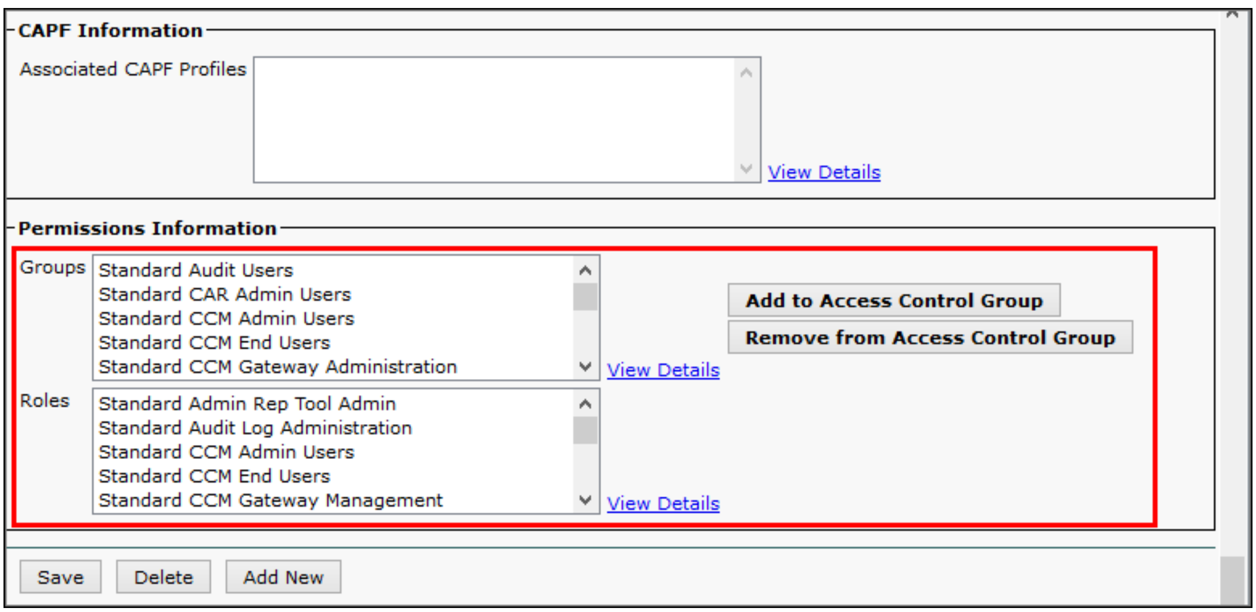






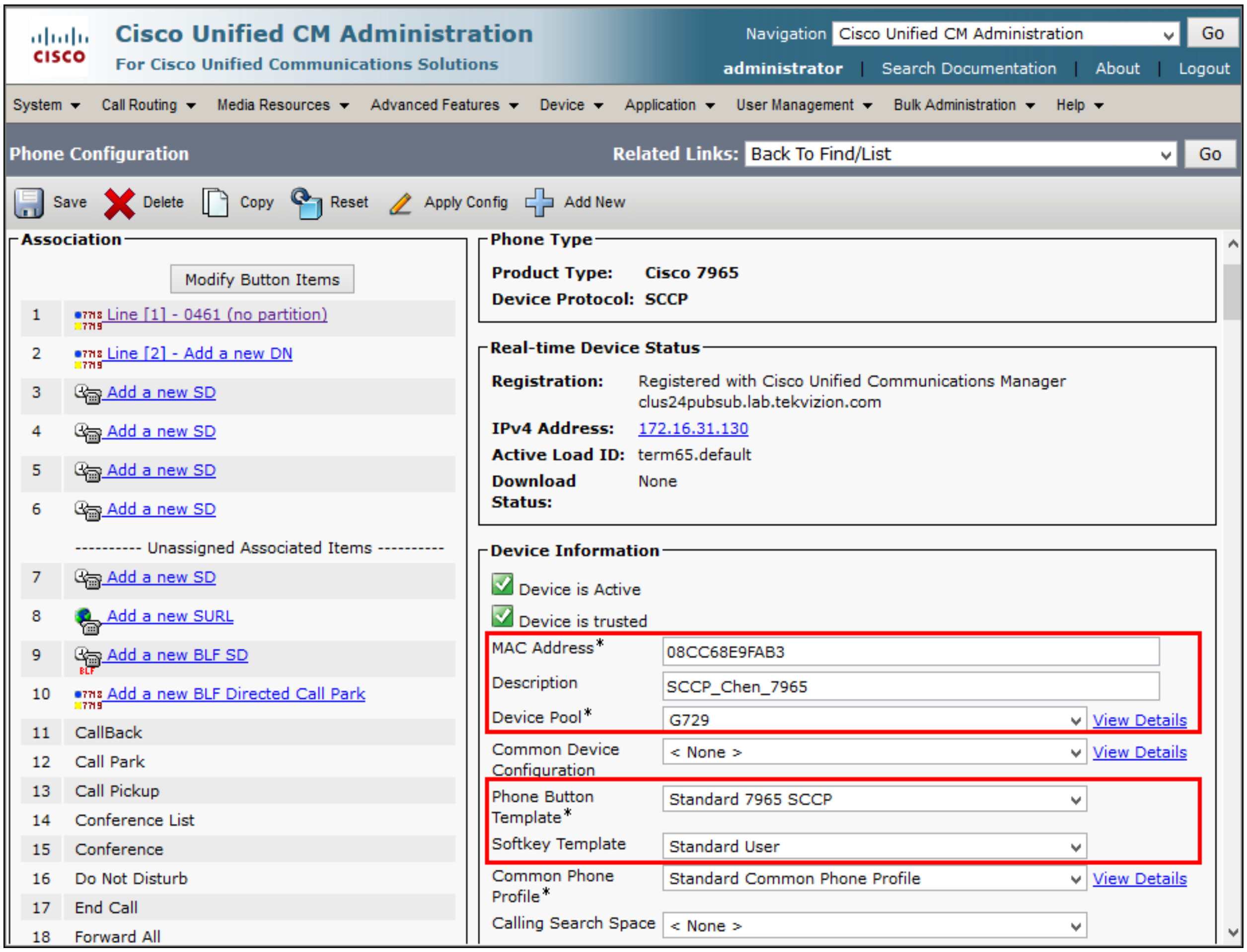
End User Configuration (Contd.)





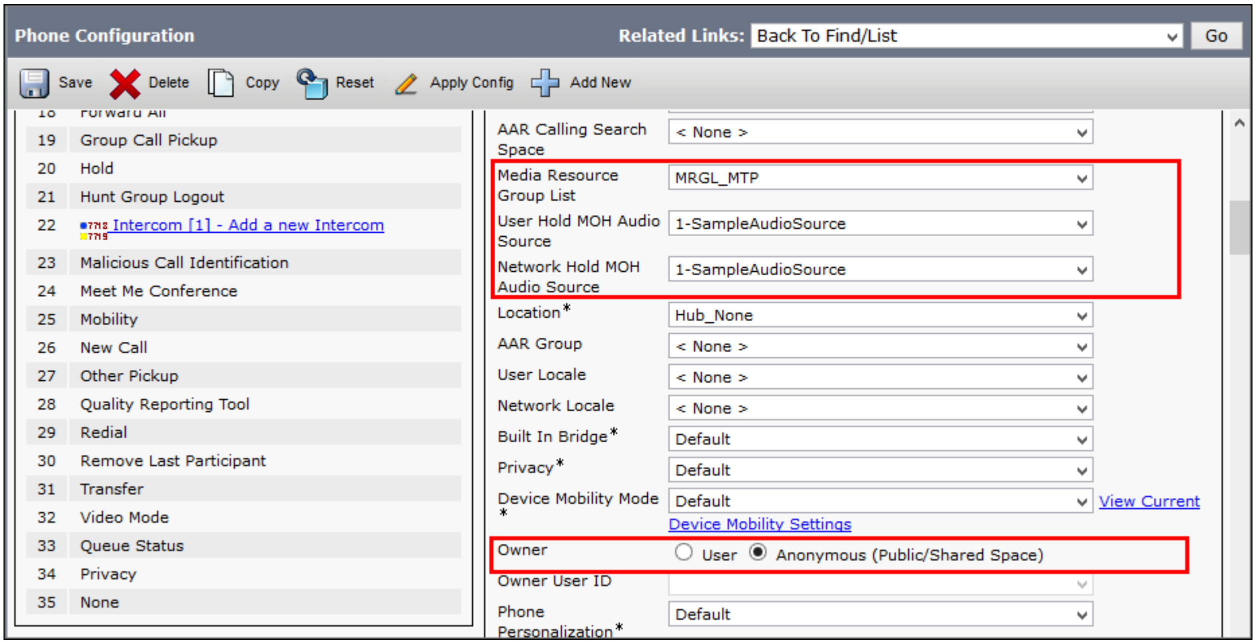
Cisco IP Phone 7965 SCCP Configuration

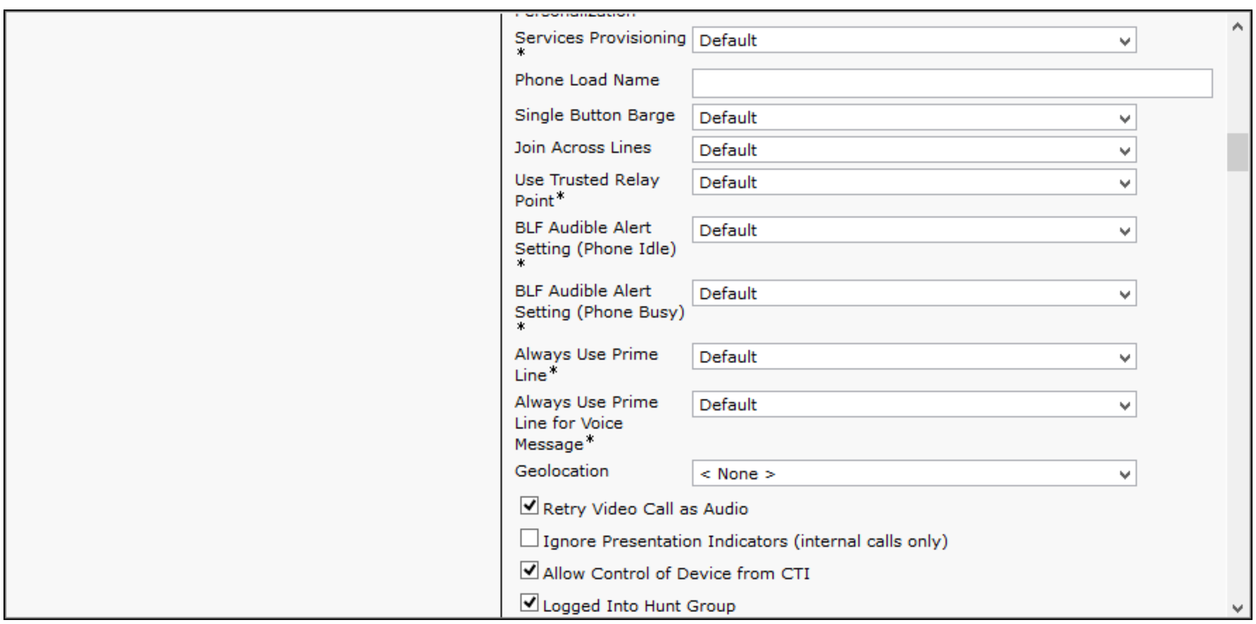
Set MAC Address* = the below mac is used in this example.
Set Description = SCCP_Chen_7965. This text is used to identify this Phone.
Set Device Pool*= G729. This is used in this example.
Set Phone Button Template*= Standard 7965 SCCP. This is used in this example.
Set Soft key Template = Standard User. This is used in this example.



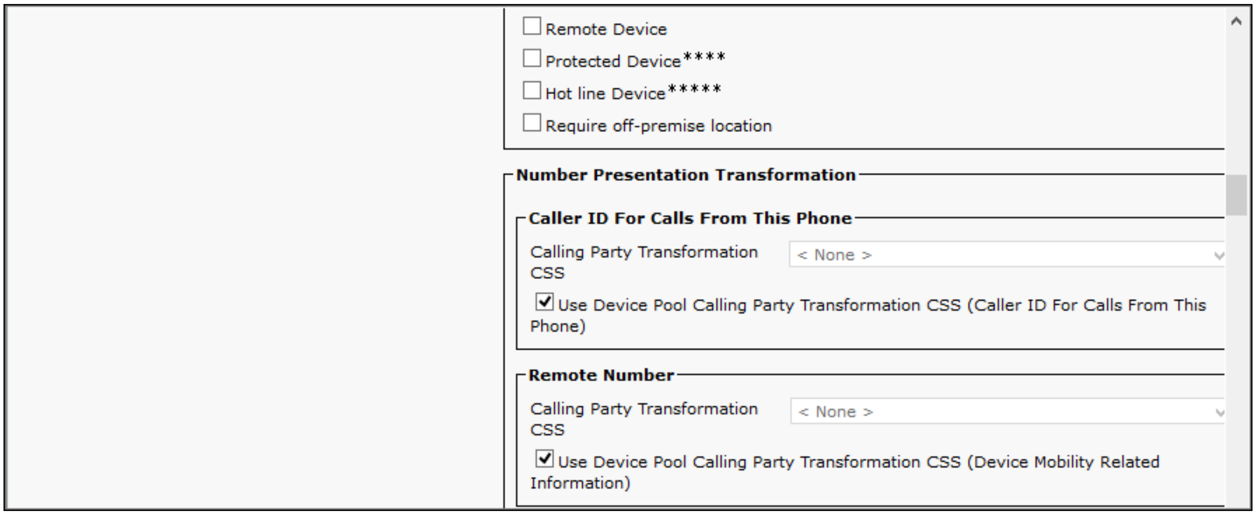
Cisco IP Phone 7965 SCCP Configuration (Contd.)

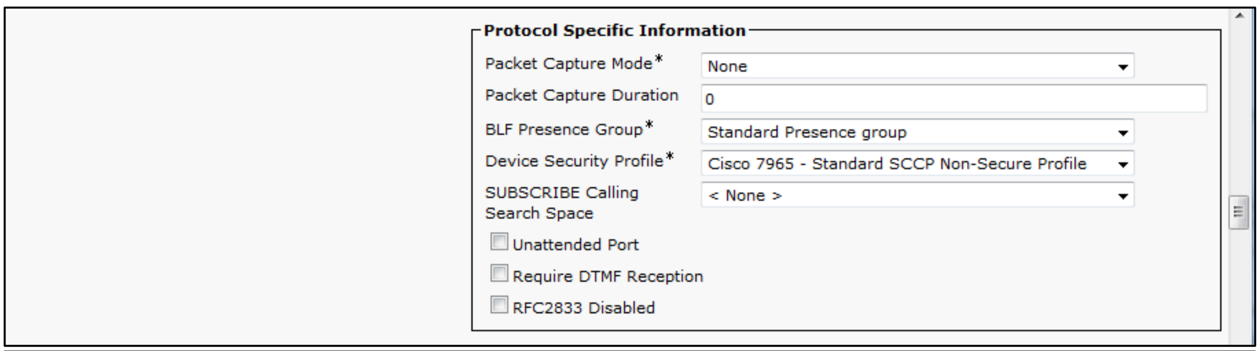
Set Media Resource Group List = MRGL_MTP. This is used in this example.
Set User Hold MOH Audio Source = 1-SampleAudioSource.
Set Network Hold MOH Audio Source = 1-SampleAudioSource.
Check Owner = Anonymous (Public/Shared Space). This is used in this example.

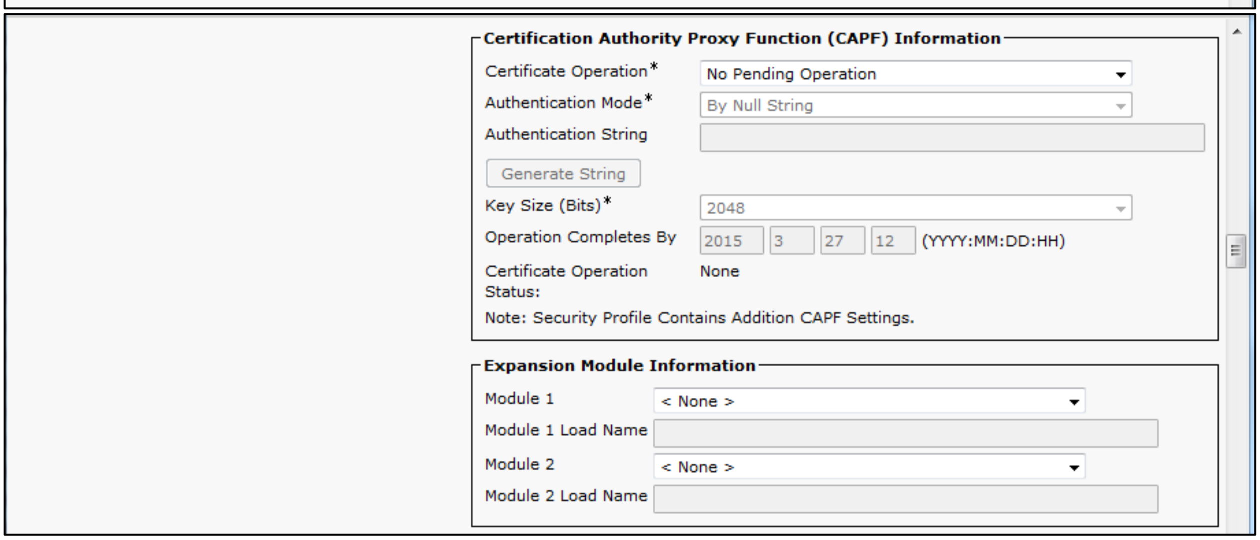


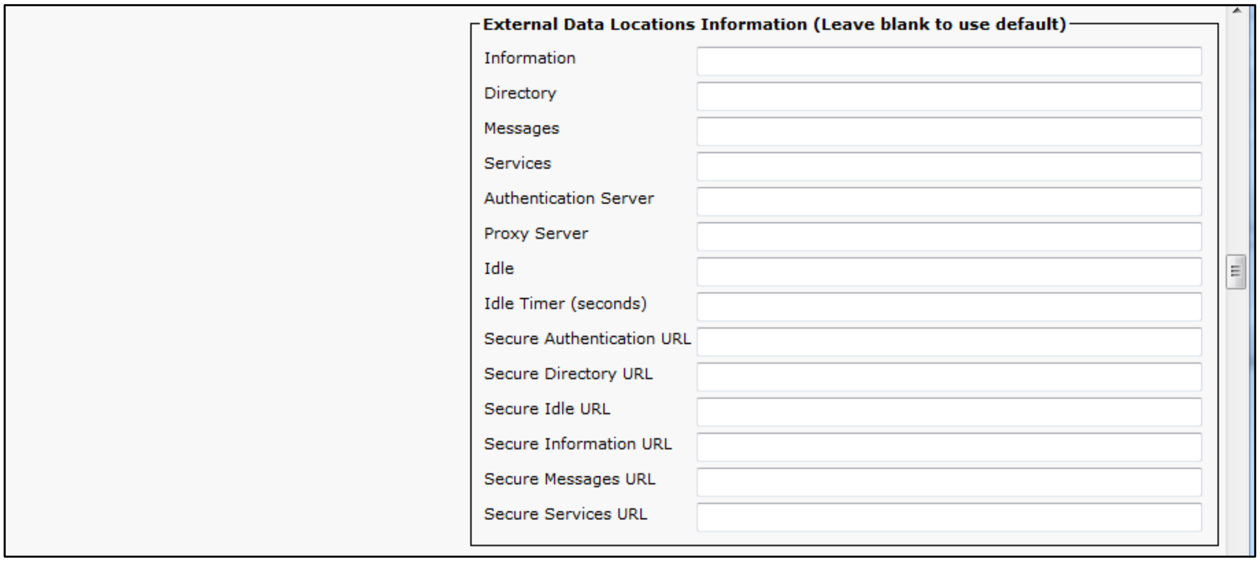


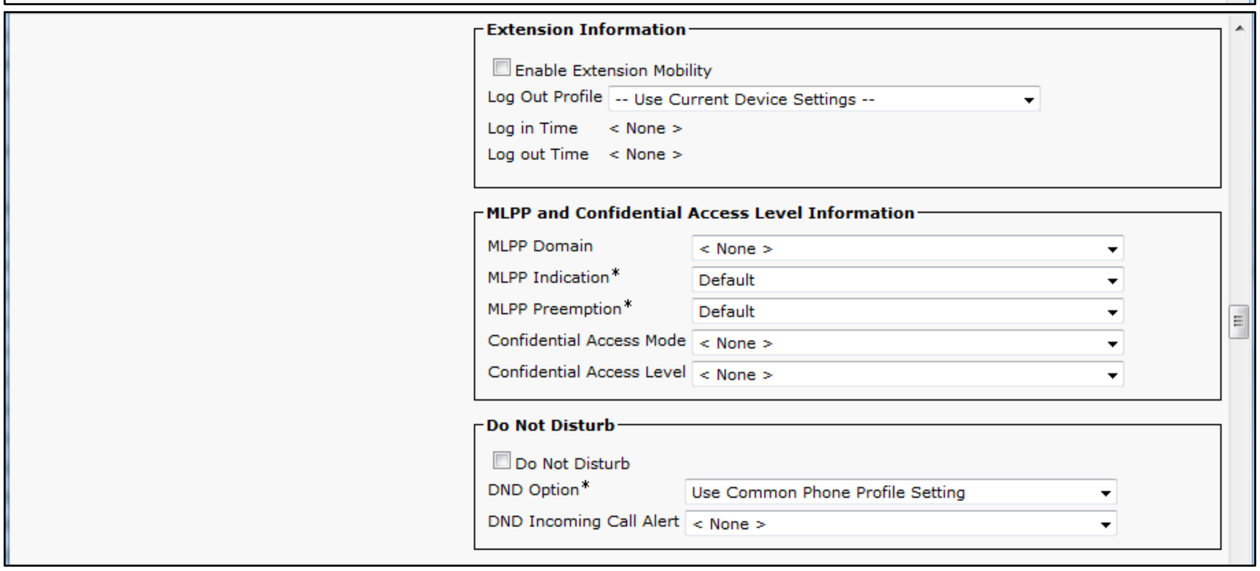
Cisco IP Phone 7965 SCCP Configuration (Contd.)

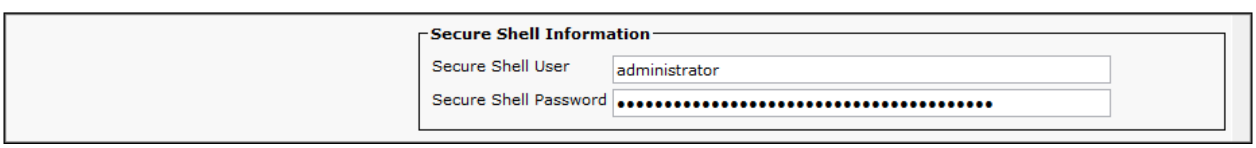




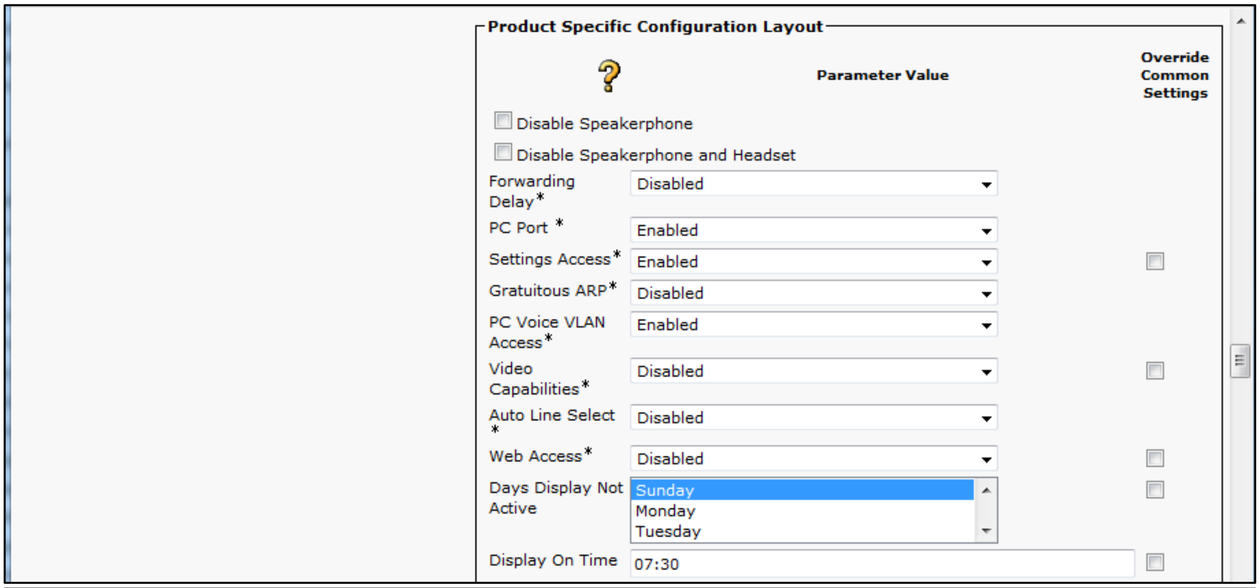


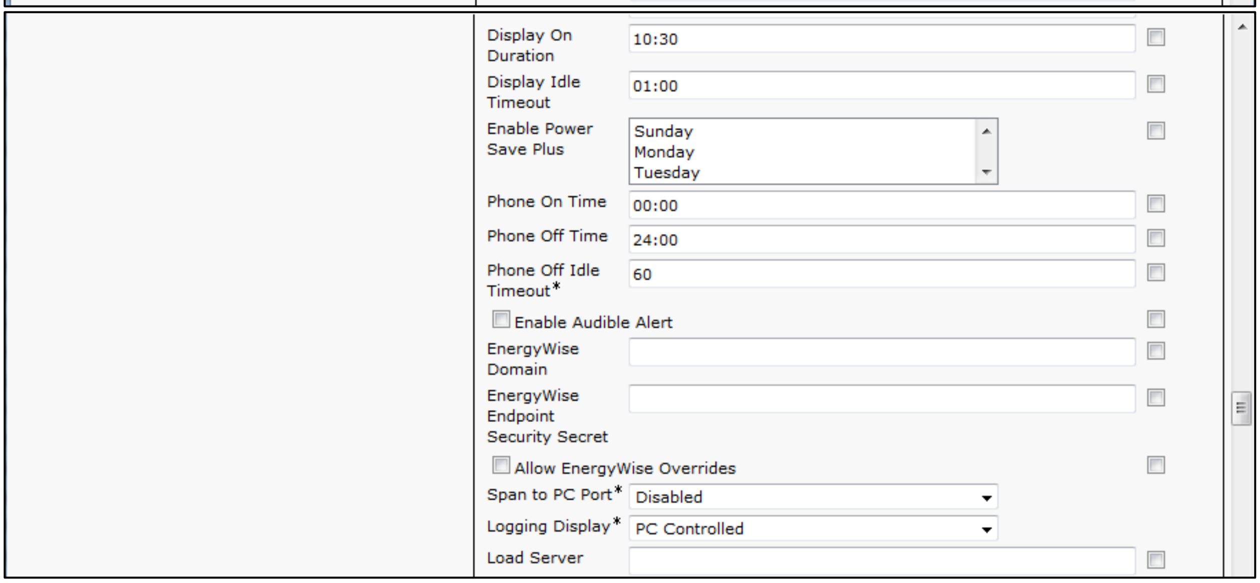

External Data Locations Information (Leave blank to use default)
Information
Directory
Messages
Services
Authentication Server
Proxy Server
Idle
Idle Timer (seconds)
Secure Authentication URL
Secure Directory URL
Secure Idle URL
Secure Information URL
Secure Messages URL
Secure Services URL

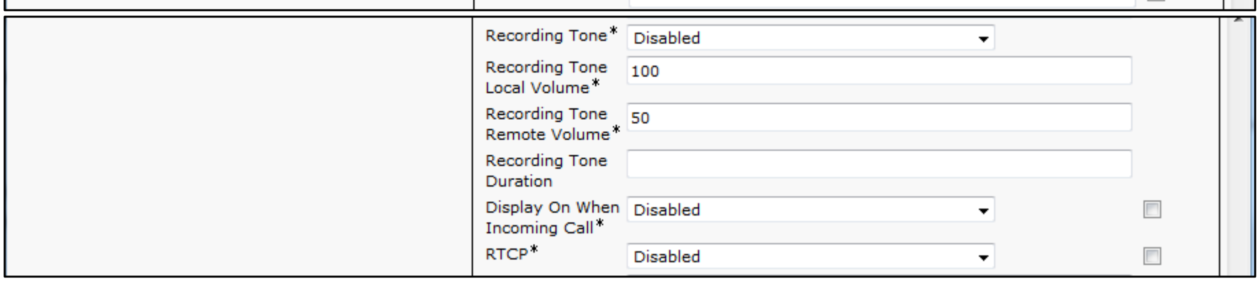

Extension Information
Enable Extension Mobility
Log Out Profile -- Use Current Device Settings --
Log in Time < None >
Log out Time < None >
MLPP and Confidential Access Level Information
MLPP Domain < None >
MLPP Indication* Default
MLPP Preemption* Default
Confidential Access Mode < None >
Confidential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


Secure Shell Information
Secure Shell User administrator
Secure Shell Password

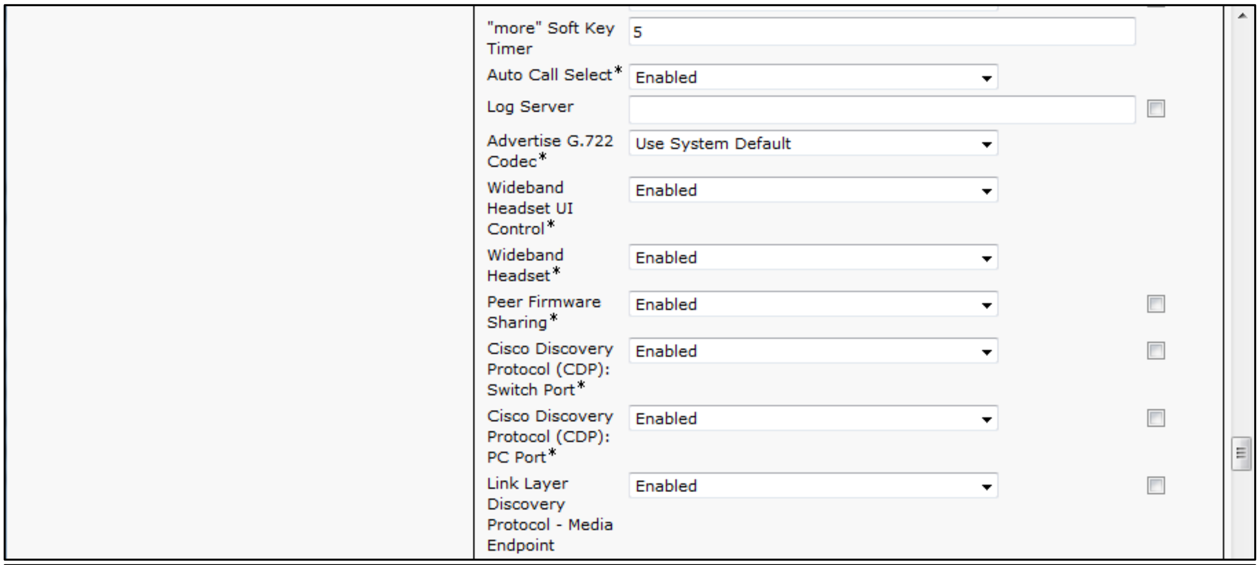
Cisco IP Phone 7965 SCCP Configuration (Contd.)

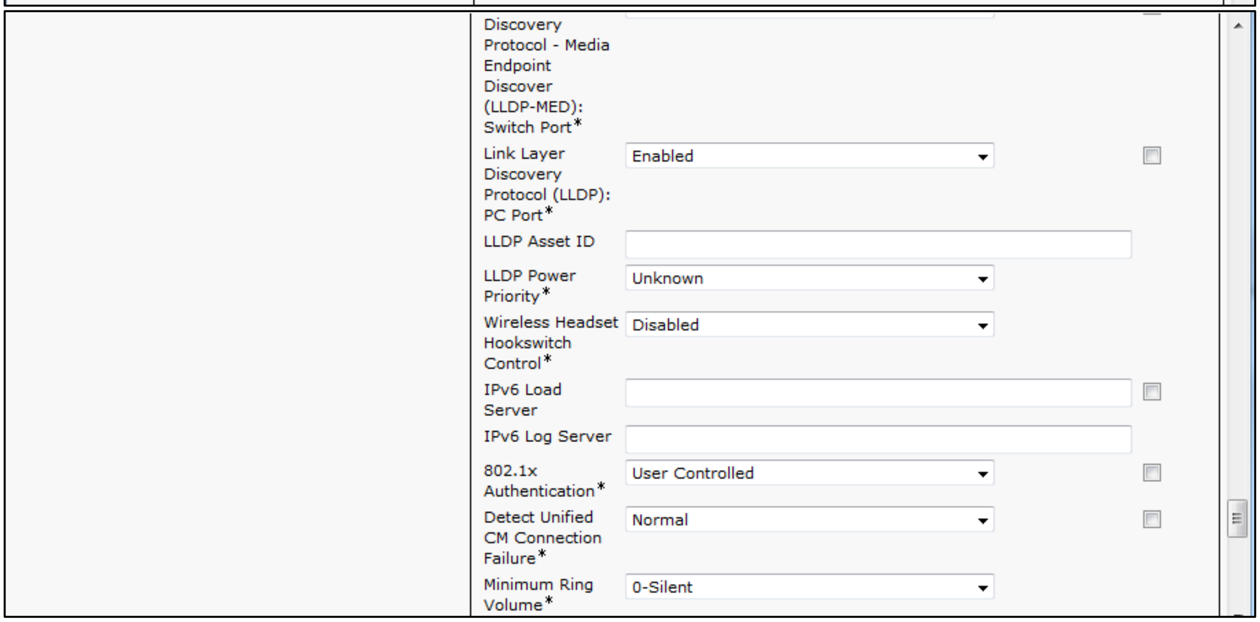




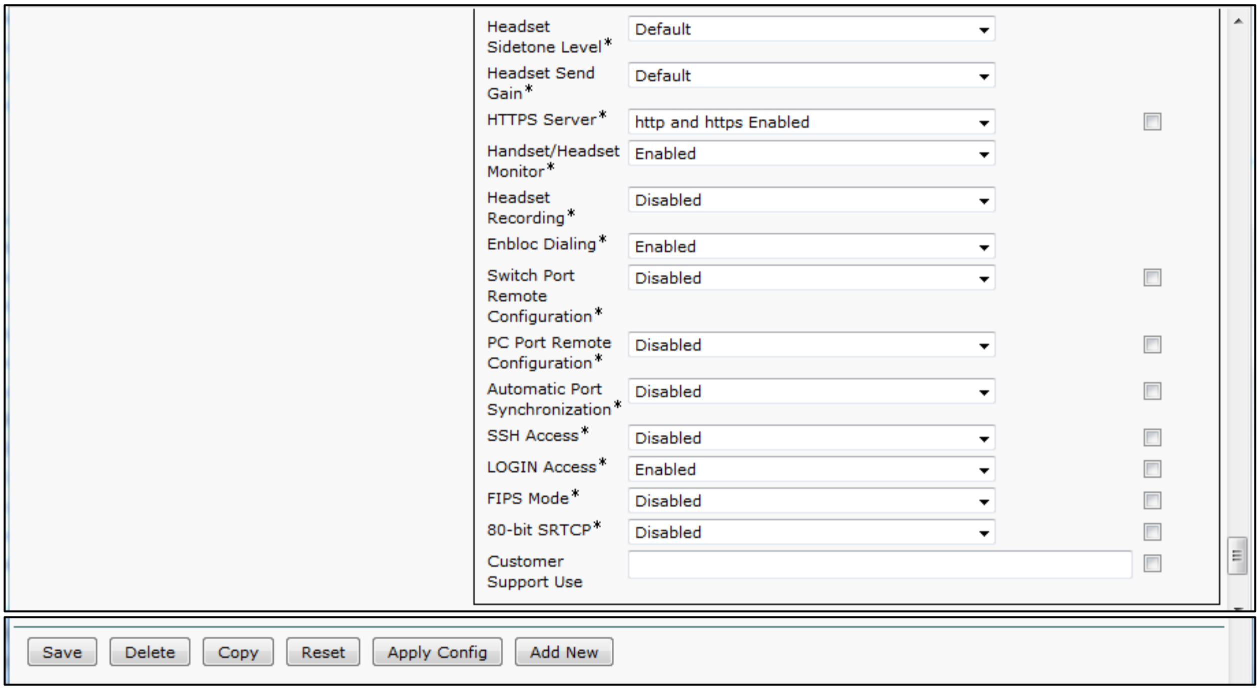


Cisco IP Phone 7965 SCCP Configuration (Contd.)



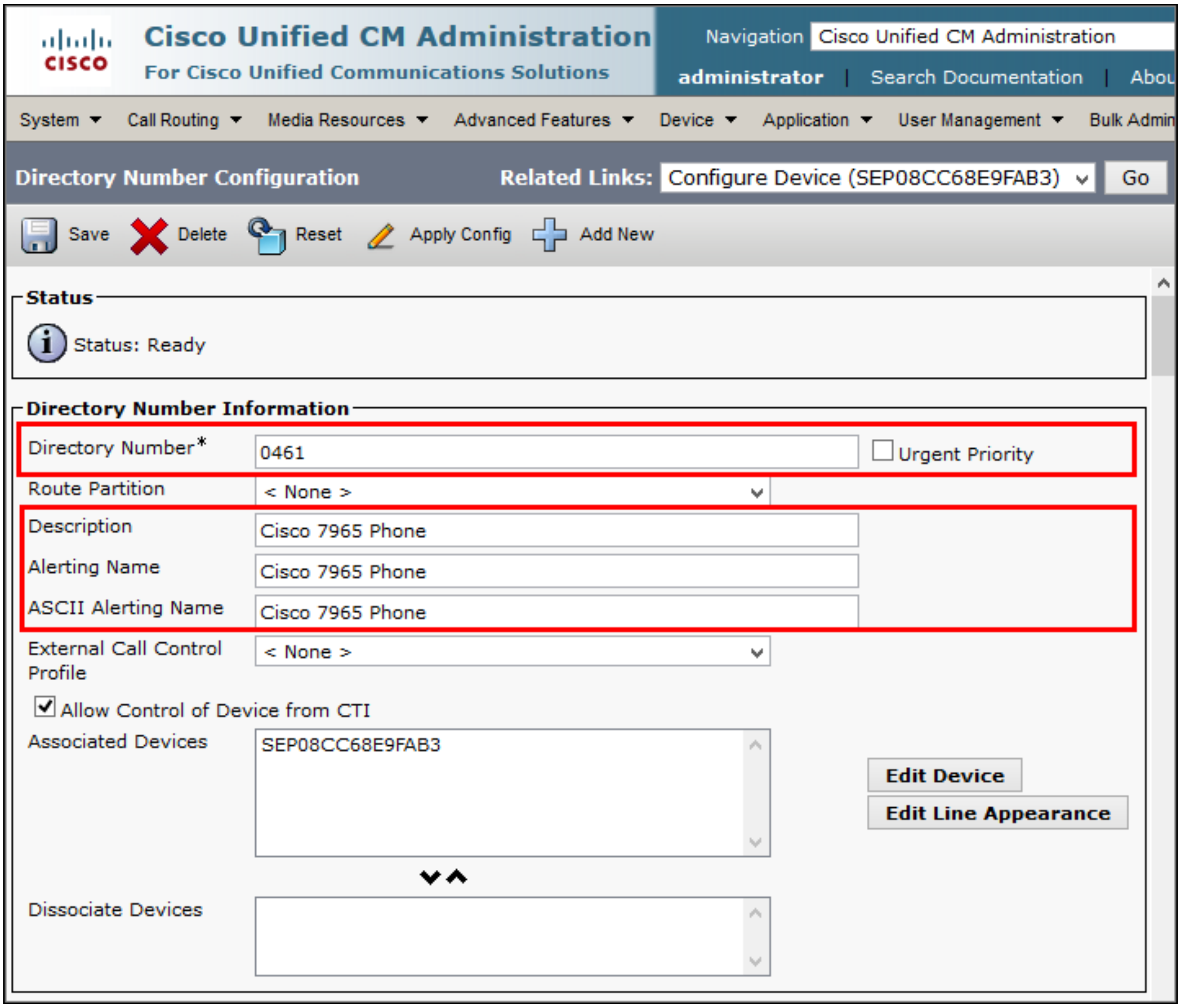


Cisco IP Phone 7965 SCCP Configuration (Contd.)

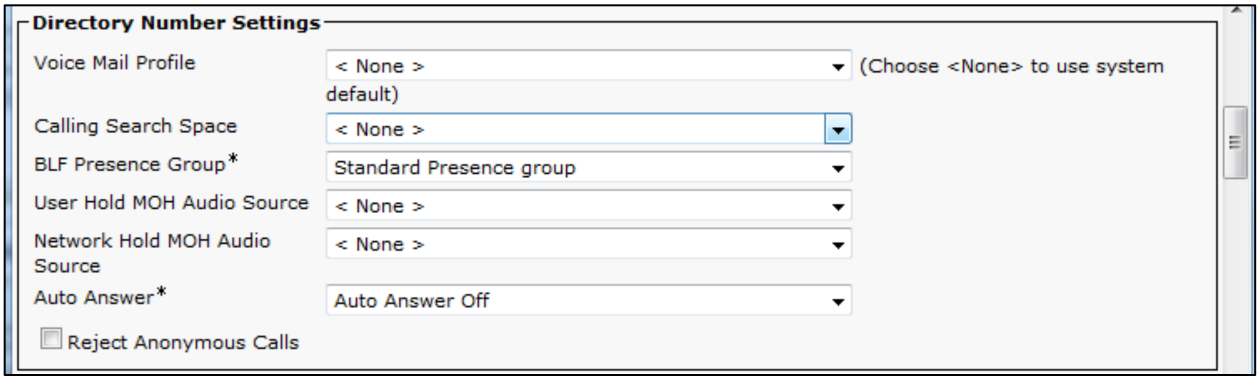


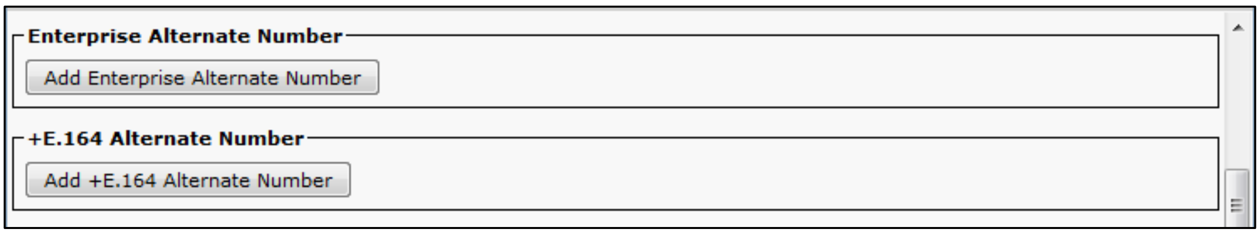
Cisco IP Phone 7965 SCCP Configuration (Contd.)

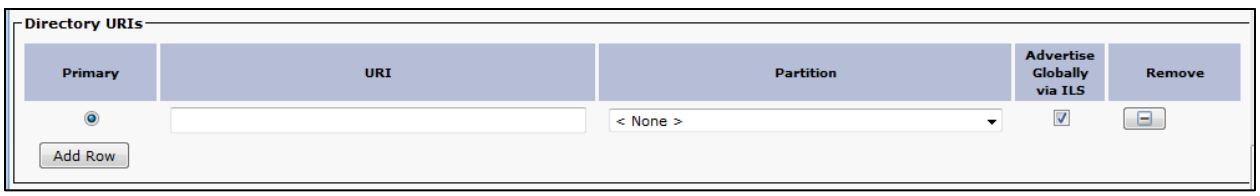
Set Directory Number* = 0461. This is used in this example.
Set Description = Cisco 7965 Phone. This is used in this example.
Set Alerting Name = Cisco 7965 Phone. This is used in this example.
Set ASCII Alerting Name = Cisco 7965 Phone. This is used in this example.

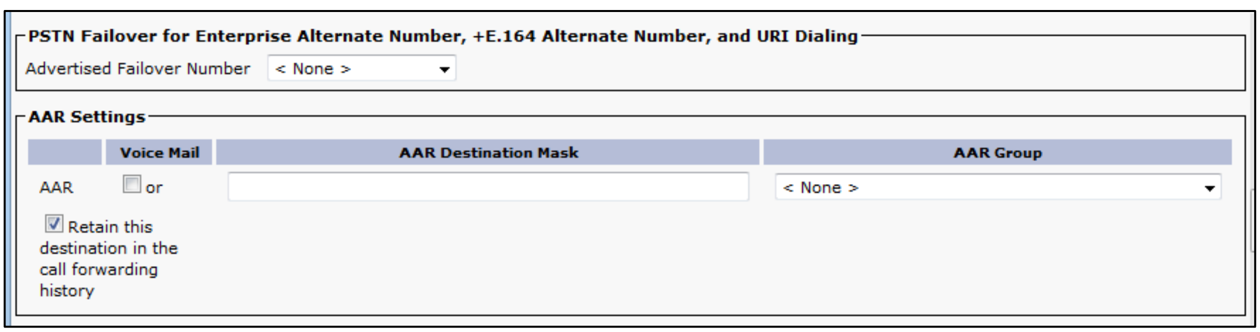


Cisco IP Phone 7965 SCCP Configuration (Contd.)

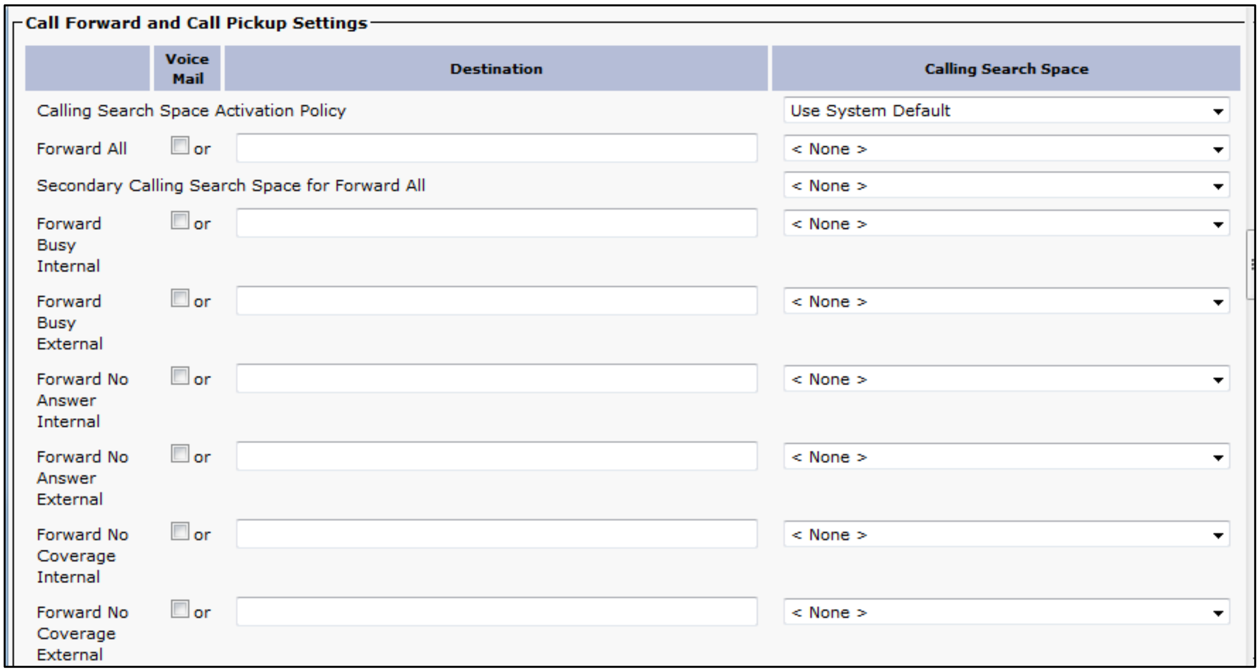


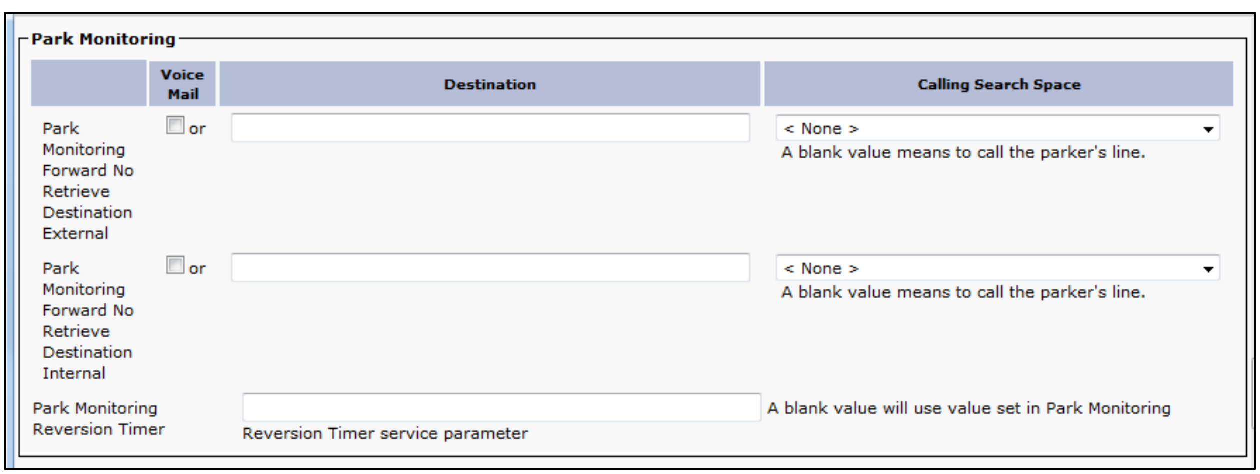






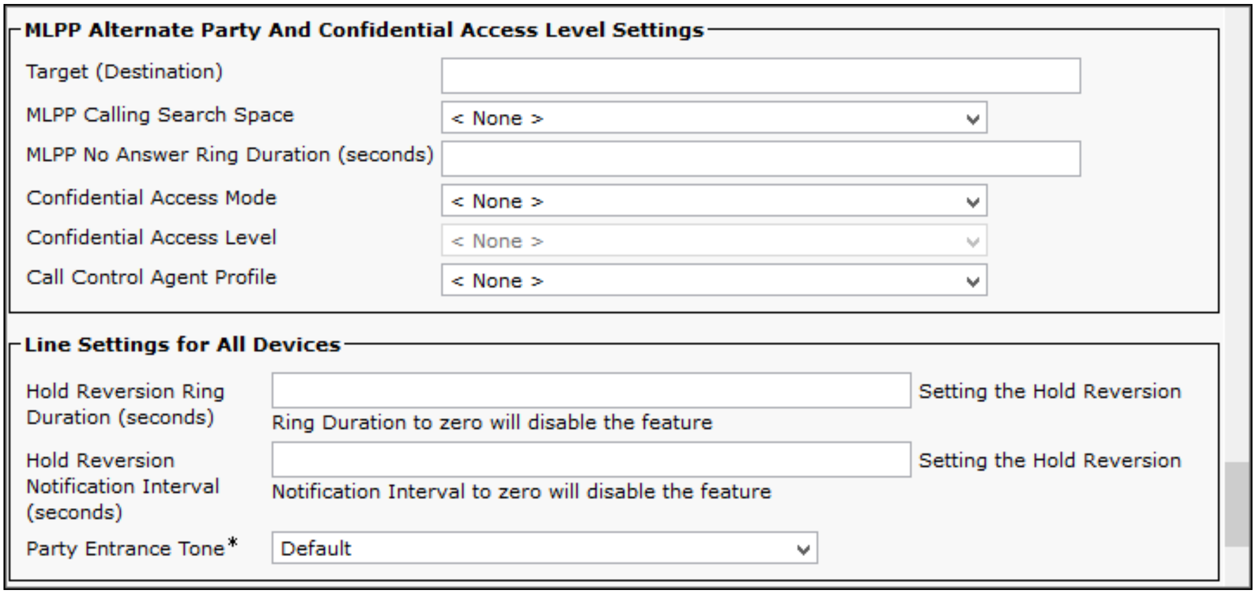
Cisco IP Phone 7965 SCCP Configuration (Contd.)





Cisco IP Phone 7965 SCCP Configuration (Contd.)

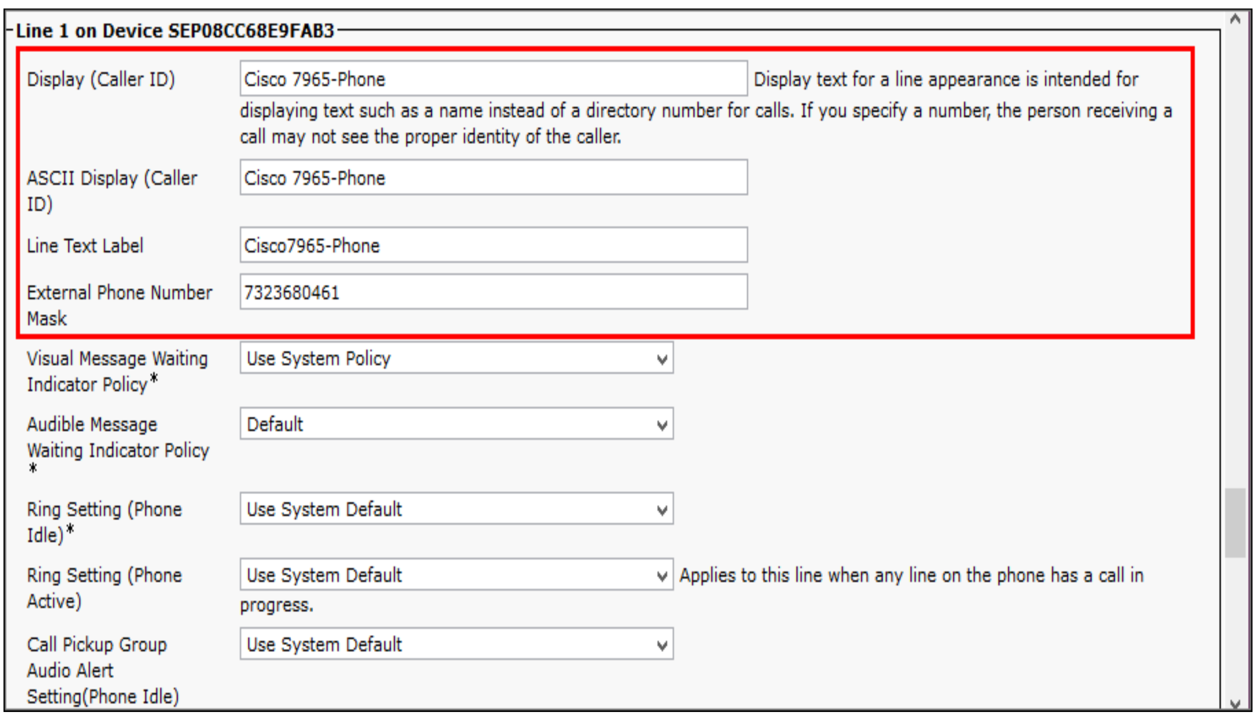
Set Display (Caller ID) = Cisco 7965-Phone. This is used in this example.
Set ASCII Display (Caller ID) = Cisco 7965-Phone. This is used in this example.
Set Line Text Label = Cisco 7965-Phone. This is used in this example.
Set External Phone Number Mask = 7323680461. This is used in this example.


MLPP Alternate Party And Confidential Access Level Settings

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

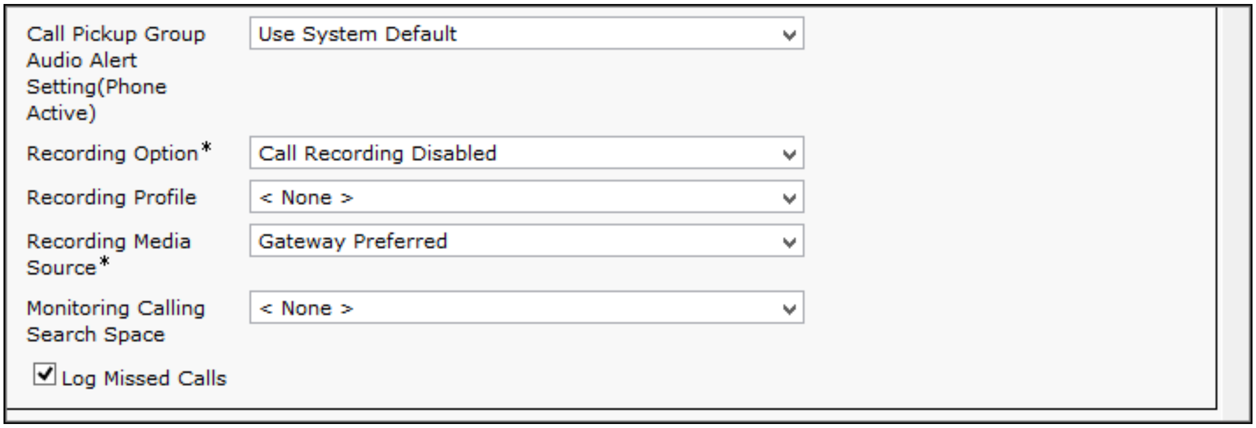
Line Settings for All Devices

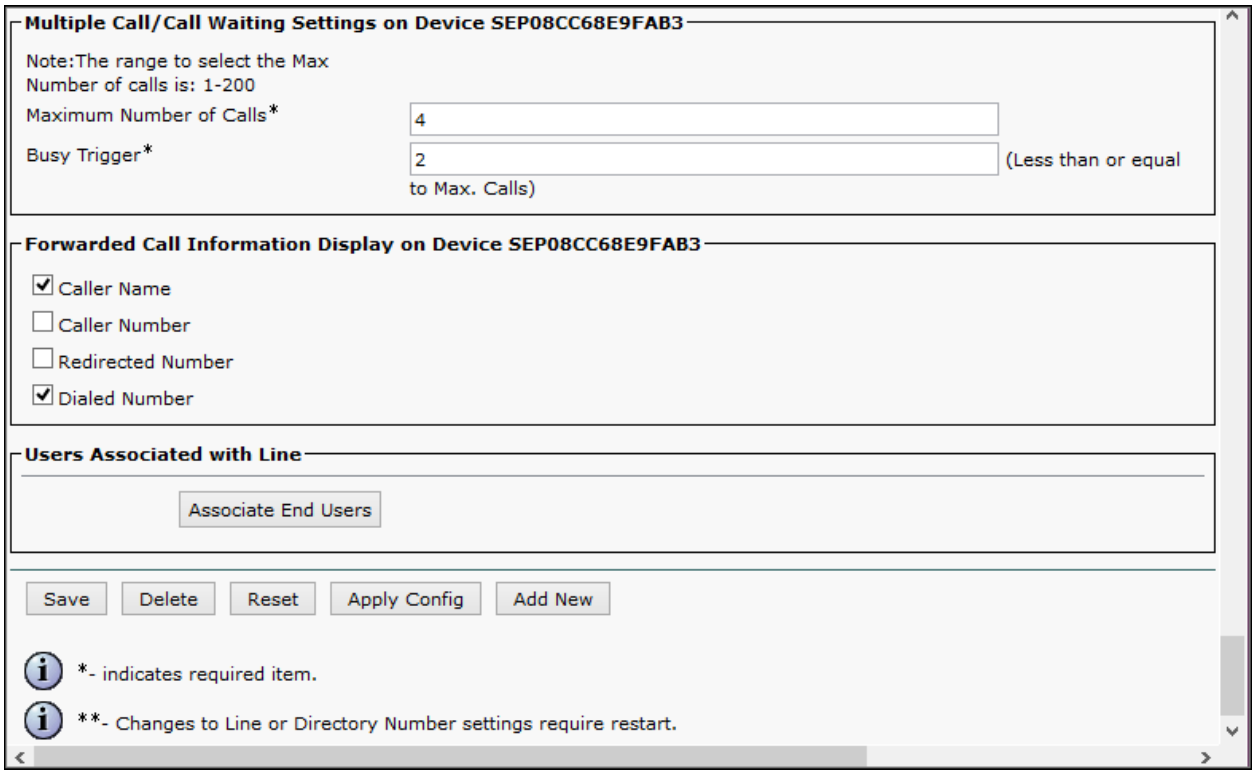
Hold Reversion Ring Duration (seconds)	Setting the Hold Reversion Ring Duration to zero will disable the feature
Hold Reversion Notification Interval (seconds)	Setting the Hold Reversion Notification Interval to zero will disable the feature
Party Entrance Tone*	Default


Line 1 on Device SEP08CC68E9FAB3

Display (Caller ID)	Cisco 7965-Phone Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Caller ID)	Cisco 7965-Phone
Line Text Label	Cisco7965-Phone
External Phone Number Mask	7323680461
Visual Message Waiting Indicator Policy*	Use System Policy
Audible Message Waiting Indicator Policy*	Default
Ring Setting (Phone Idle)*	Use System Default
Ring Setting (Phone Active)	Use System Default Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)	Use System Default

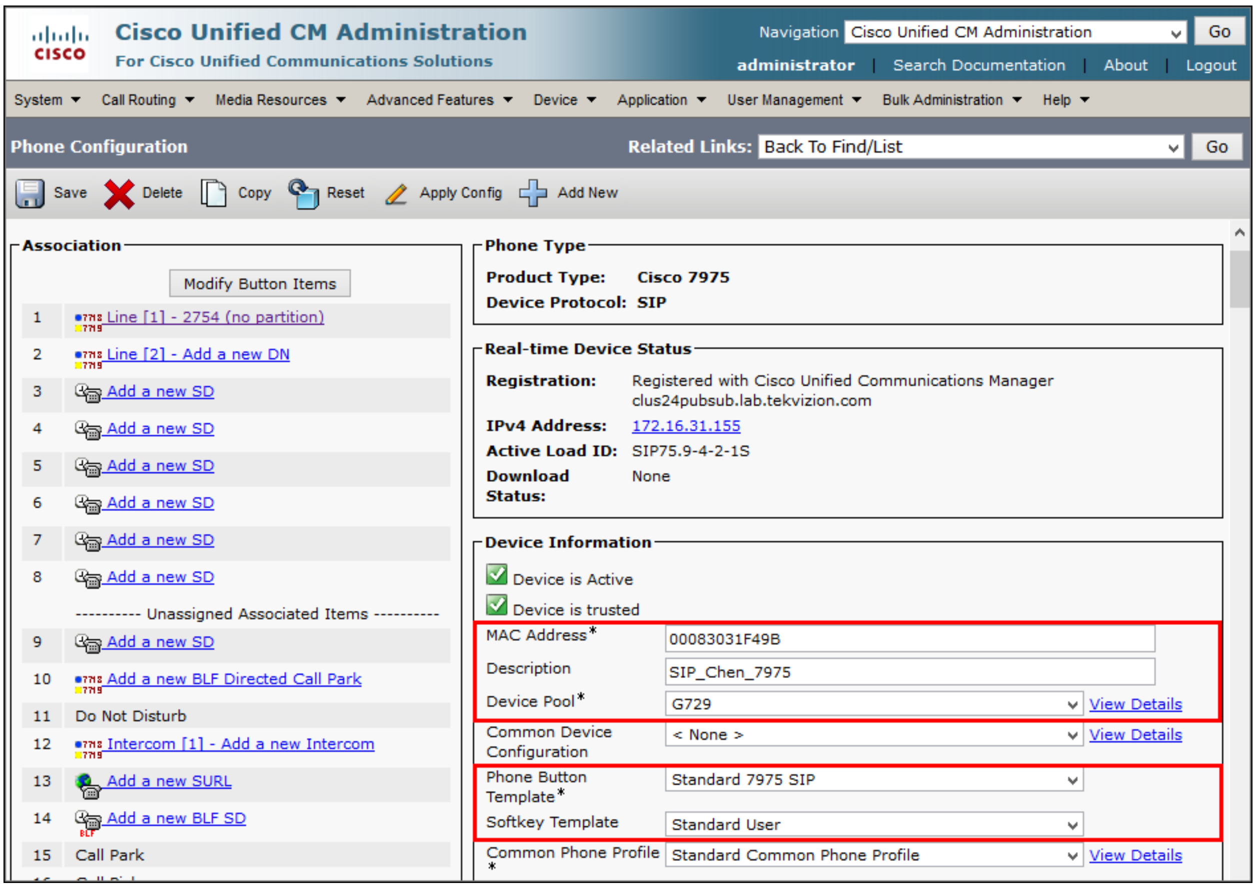
Cisco IP Phone 7965 SCCP Configuration (Contd.)

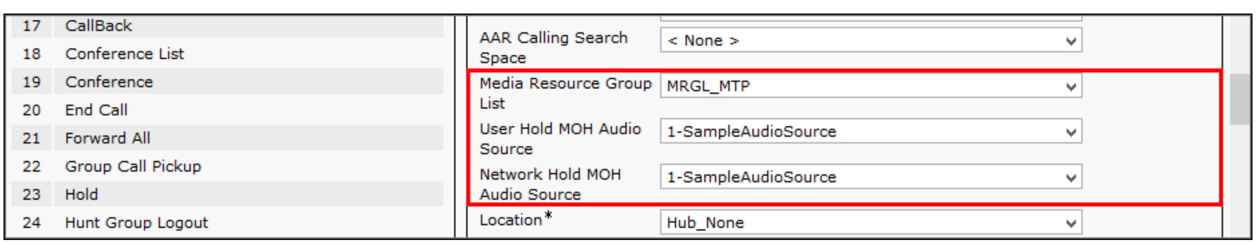




Cisco IP Phone 7975 SIP Configuration

Set MAC Address* = the below mac is used in this example.
Set Description = SIP_Chen_ 7975. This text is used to identify this Phone.
Set Device Pool*= G729. This is used in this example.
Set Phone Button Template*= Standard 7975 SIP. This is used in this example.
Set Soft key Template = Standard User. This is used in this example.
Set Media Resource Group List = MRGL_MTP. This is used in this example.
Set User Hold MOH Audio Source = 1-SampleAudioSource
Set Network Hold MOH Audio Source = 1-SampleAudioSource

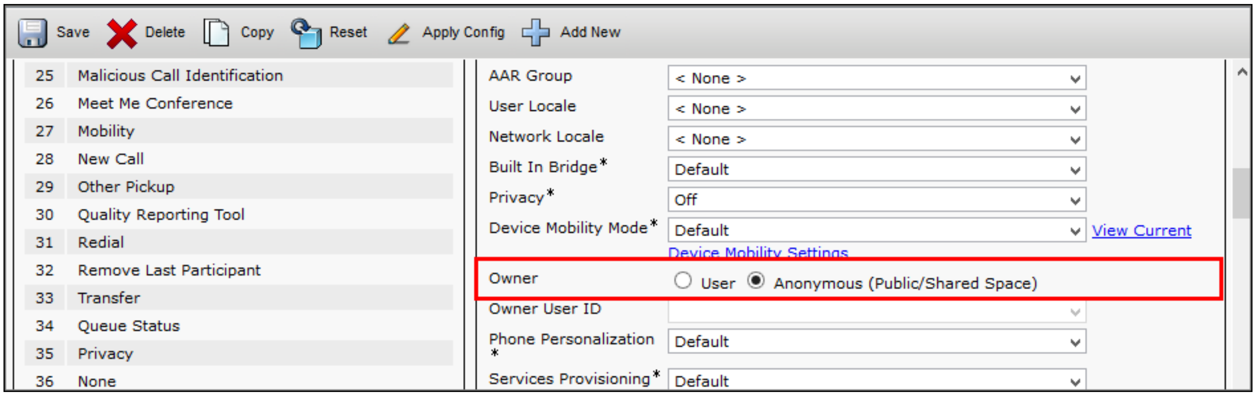


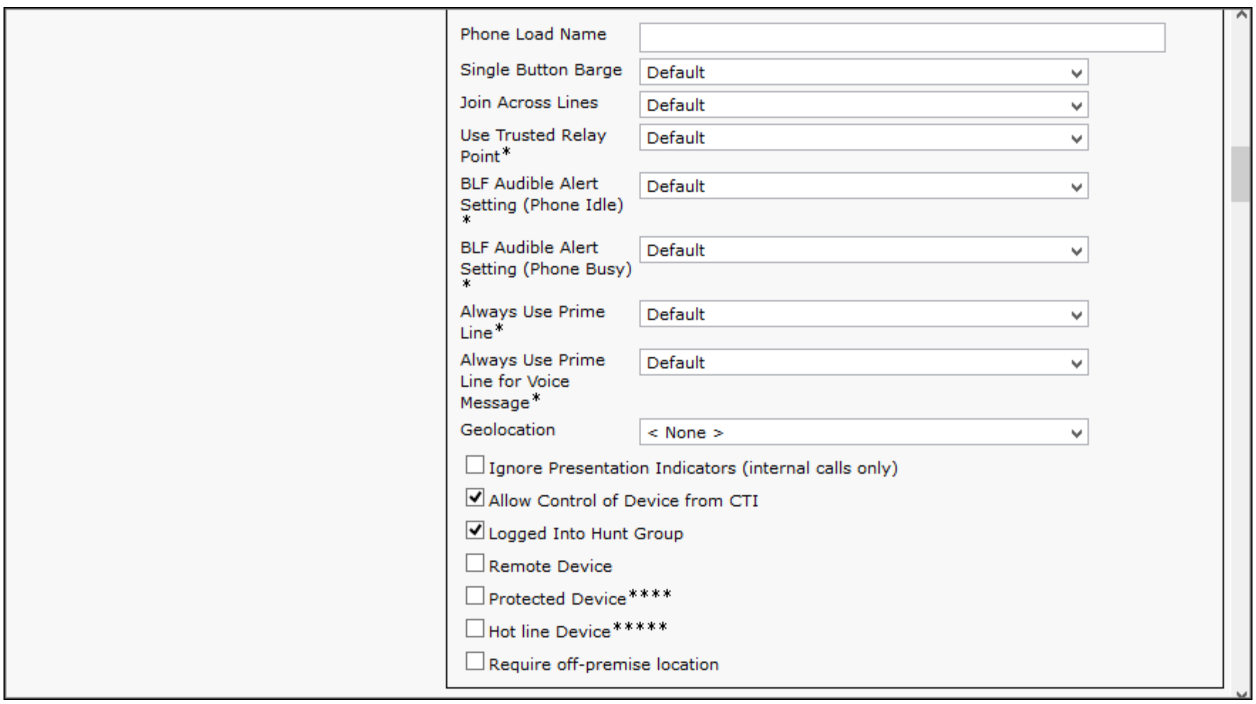


Note: Testing was conducted in tekVizion labs

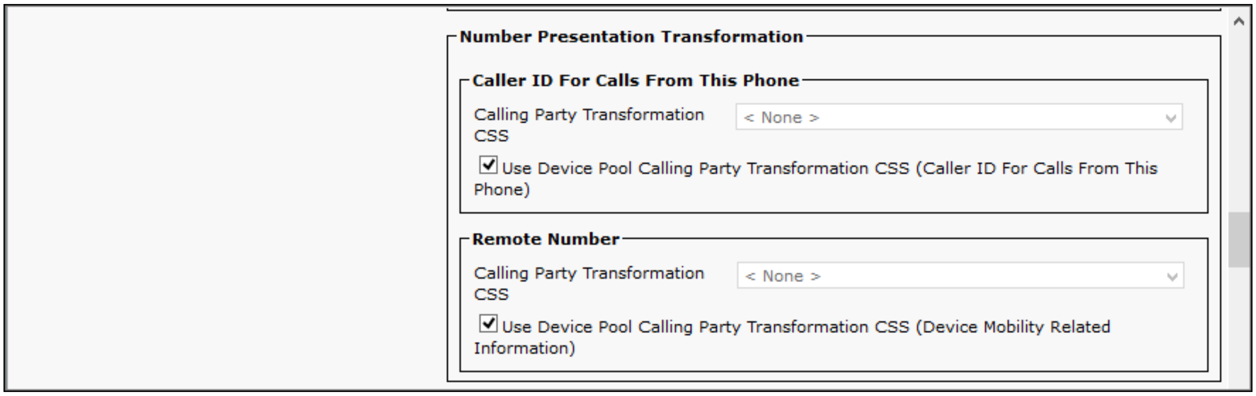
Cisco IP Phone 7975 SIP Configuration (Contd.)

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





Cisco IP Phone 7975 SIP Configuration (Contd.)



Number Presentation Transformation

Caller ID For Calls From This Phone

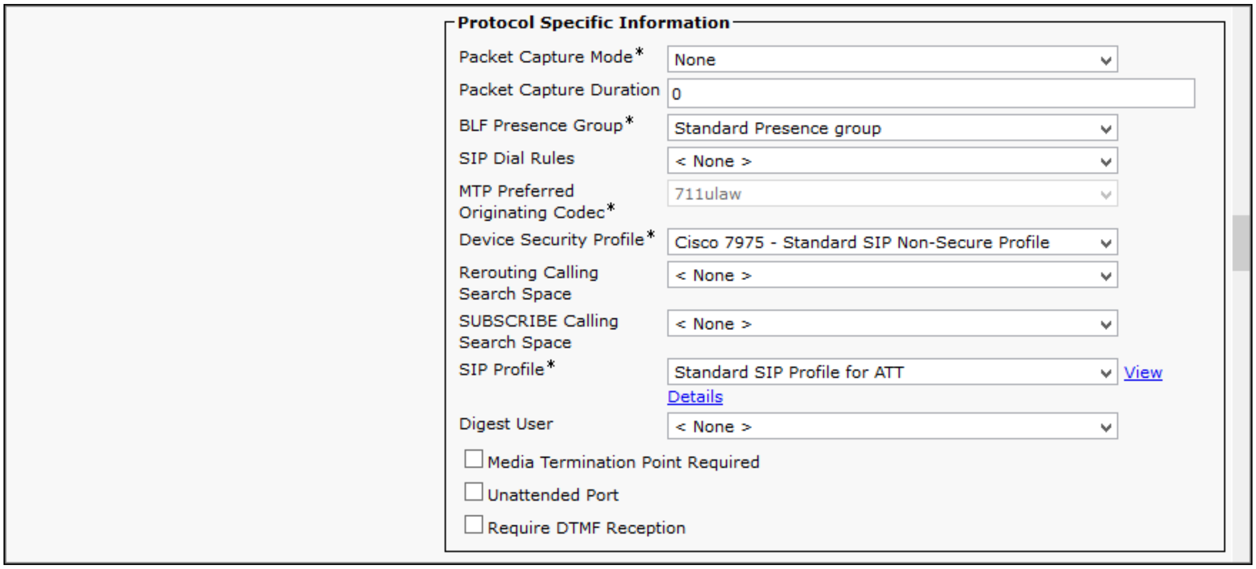
Calling Party Transformation CSS	< None >

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

Remote Number

Calling Party Transformation CSS	< None >

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



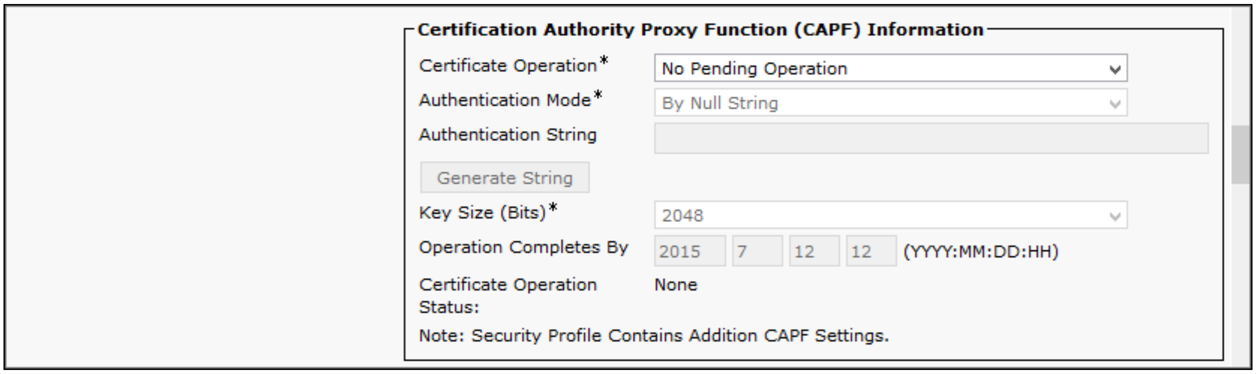
Protocol Specific Information

Packet Capture Mode*	None
Packet Capture Duration	0
BLF Presence Group*	Standard Presence group
SIP Dial Rules	< None >
MTP Preferred Originating Codec*	711ulaw
Device Security Profile*	Cisco 7975 - Standard SIP Non-Secure Profile
Rerouting Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile for ATT View Details
Digest User	< None >

☐ Media Termination Point Required

☐ Unattended Port

☐ Require DTMF Reception



Certification Authority Proxy Function (CAPF) Information

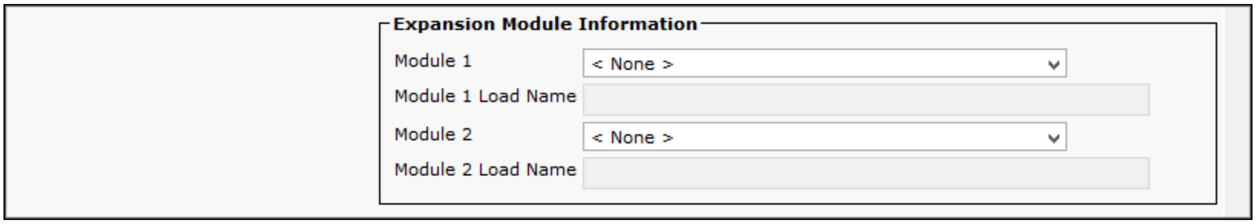
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	

Generate String

Key Size (Bits)*	2048
Operation Completes By	2015 7 12 12 (YYYY:MM:DD:HH)
Certificate Operation Status:	None

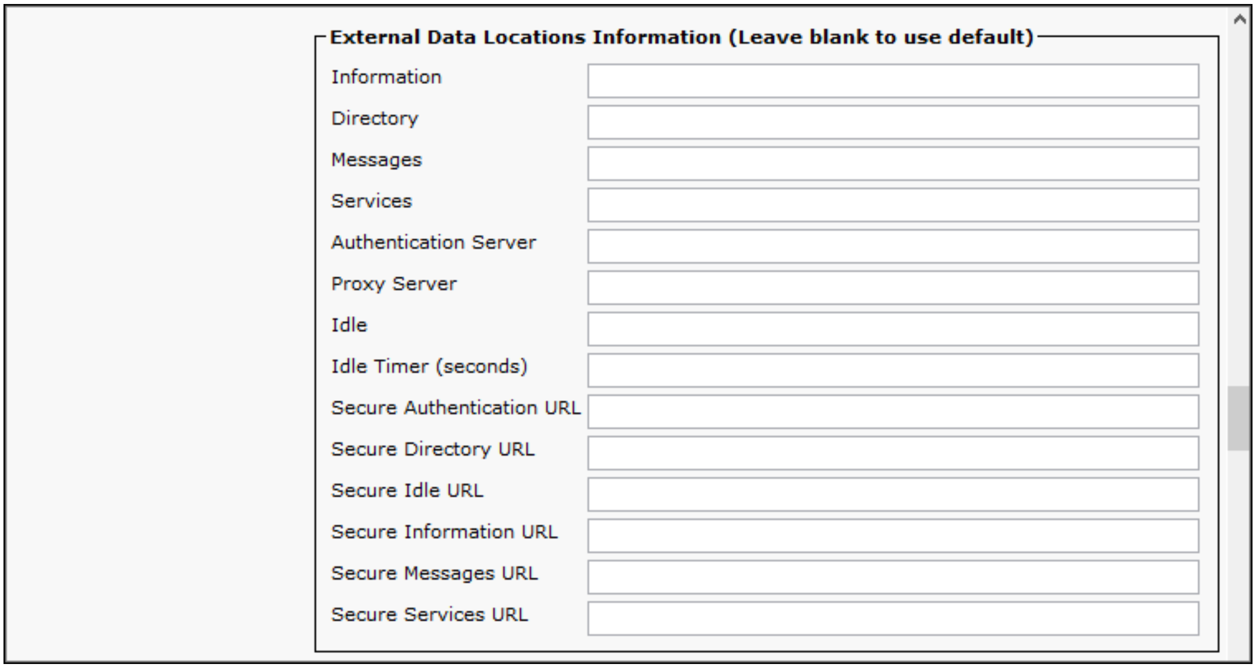
Note: Security Profile Contains Addition CAPF Settings.

Cisco IP Phone 7975 SIP Configuration (Contd.)



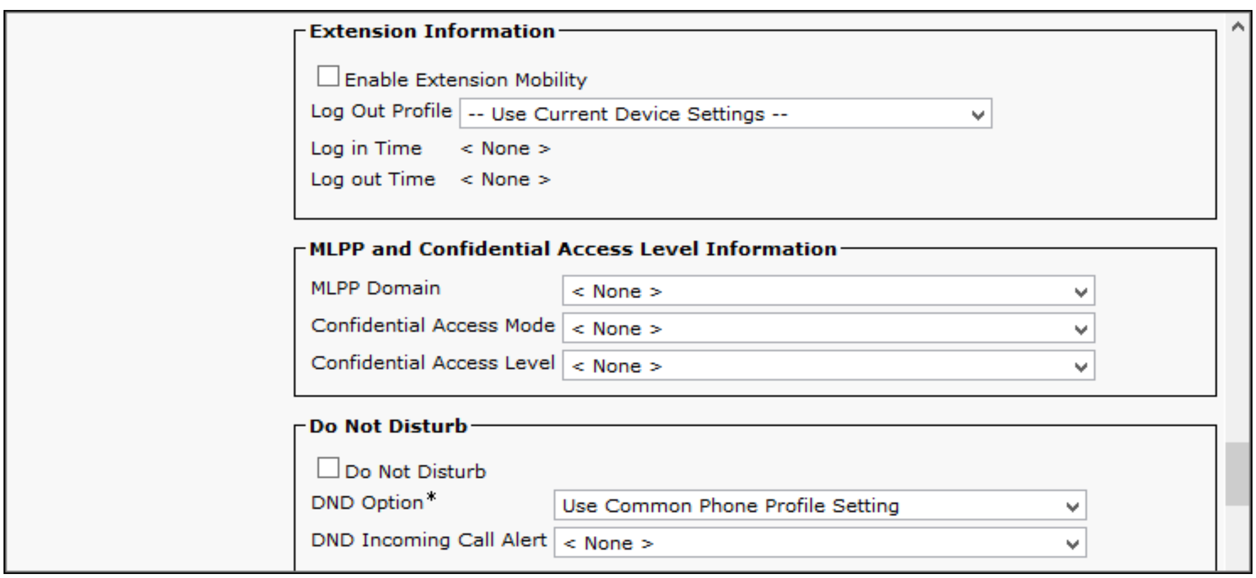
Expansion Module Information

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



External Data Locations Information (Leave blank to use default)

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



Extension Information

☐ Enable Extension Mobility

Log Out Profile	-- Use Current Device Settings --
Log in Time	< None >
Log out Time	< None >

MLPP and Confidential Access Level Information

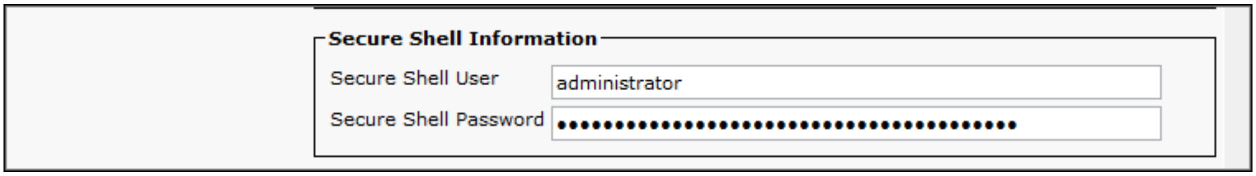
MLPP Domain	< None >
Confidential Access Mode	< None >
Confidential Access Level	< None >

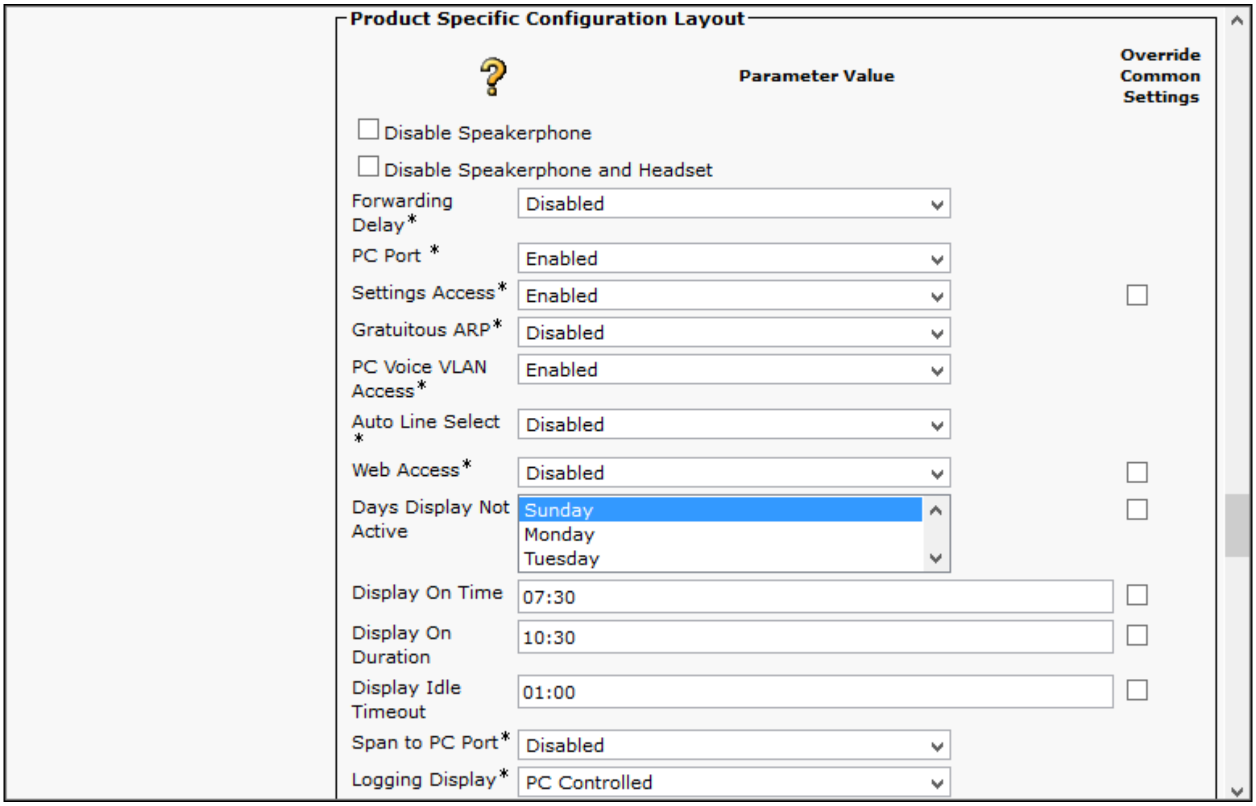
Do Not Disturb

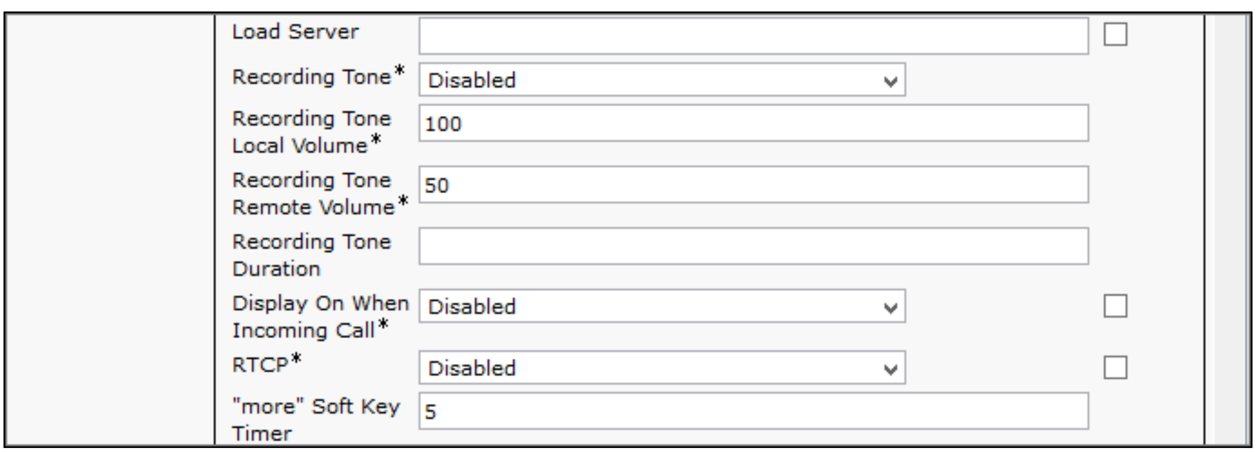
☐ Do Not Disturb

DND Option*	Use Common Phone Profile Setting
DND Incoming Call Alert	< None >

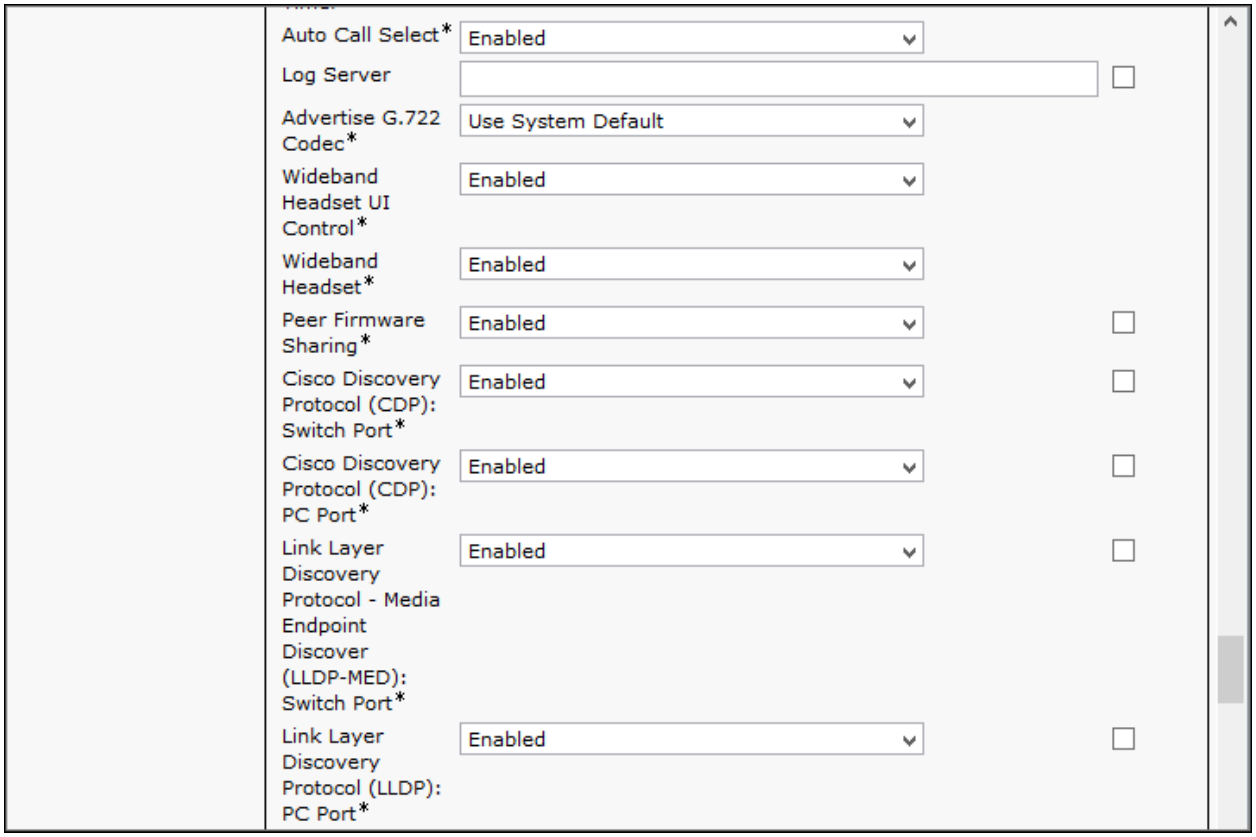
Cisco IP Phone 7975 SIP Configuration (Contd.)

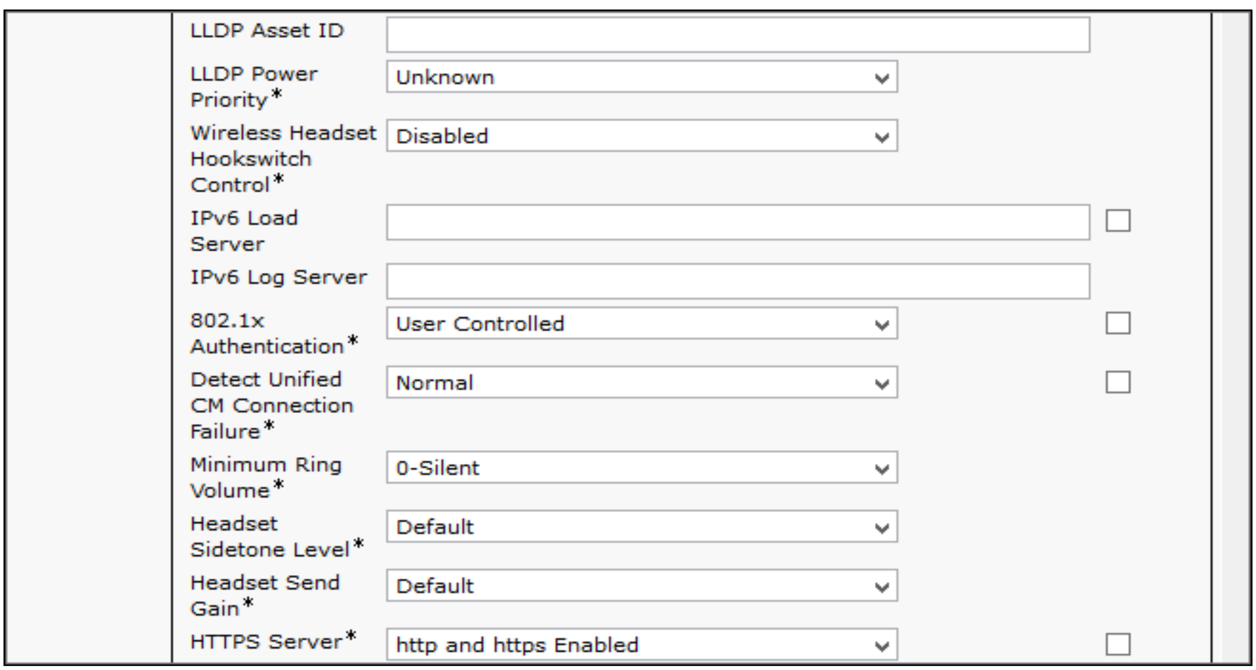






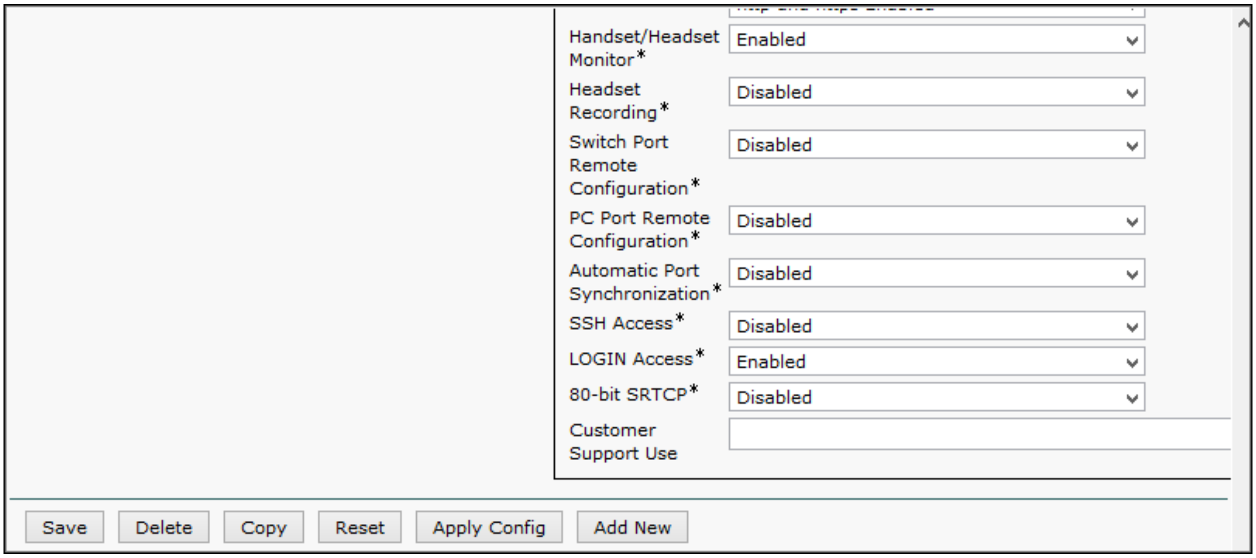
Cisco IP Phone 7975 SIP Configuration (Contd.)





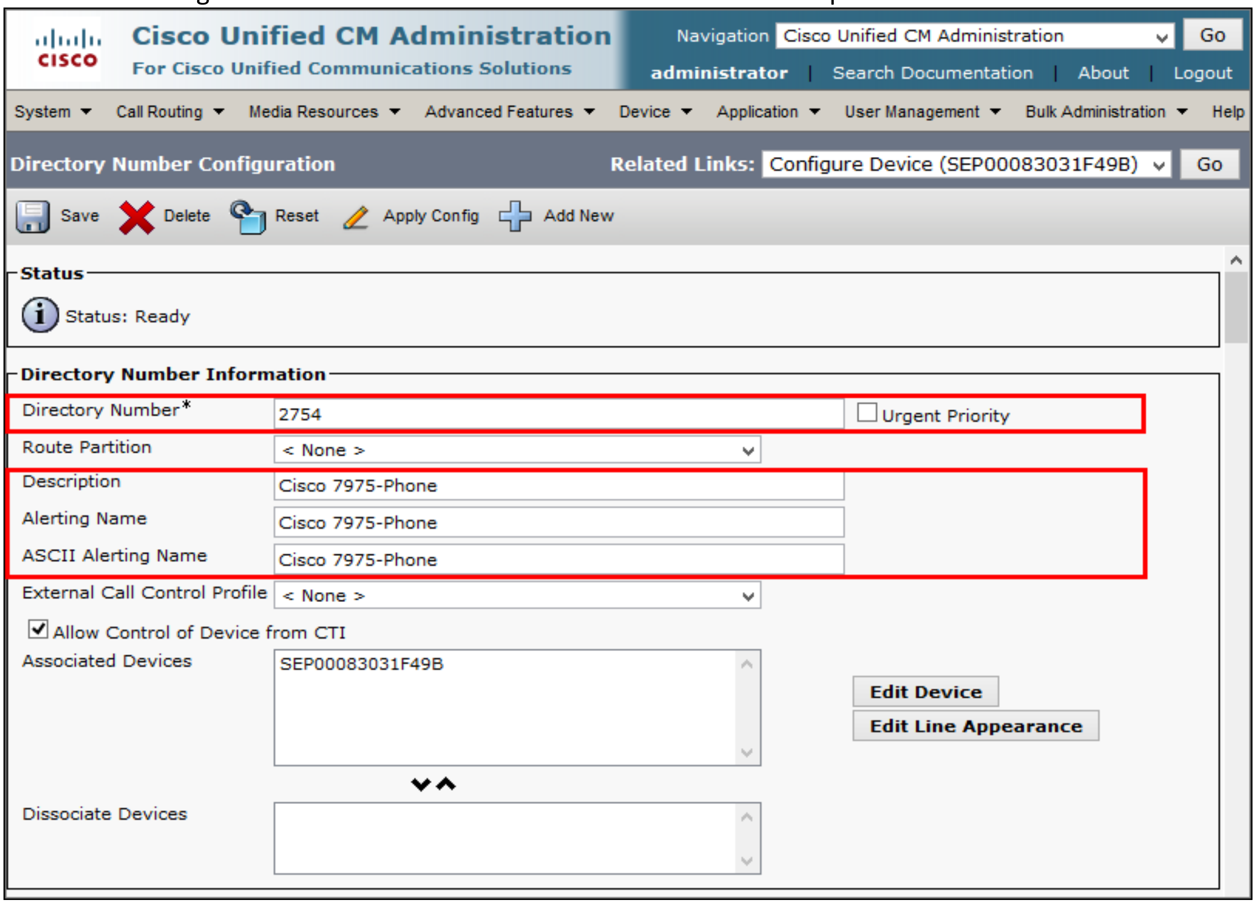
Note: Testing was conducted in tekVizion labs

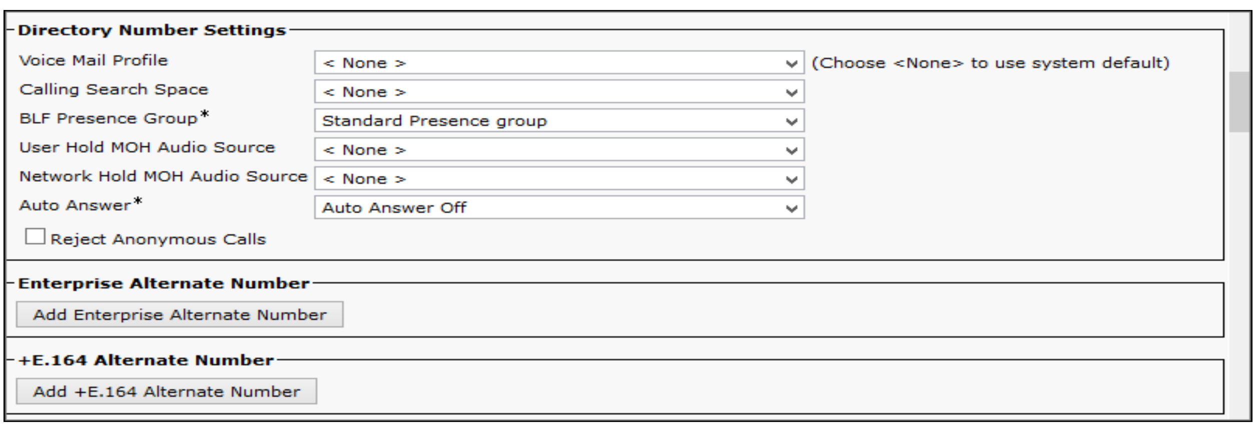
Cisco IP Phone 7975 SIP Configuration (Contd.)



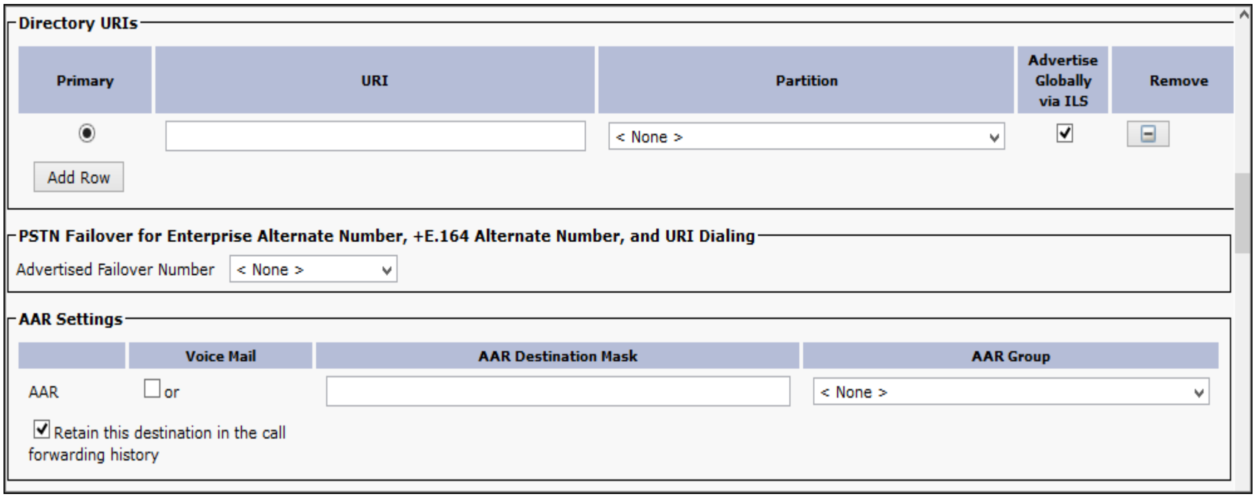
Cisco IP Phone 7975 SIP Configuration (Contd.)

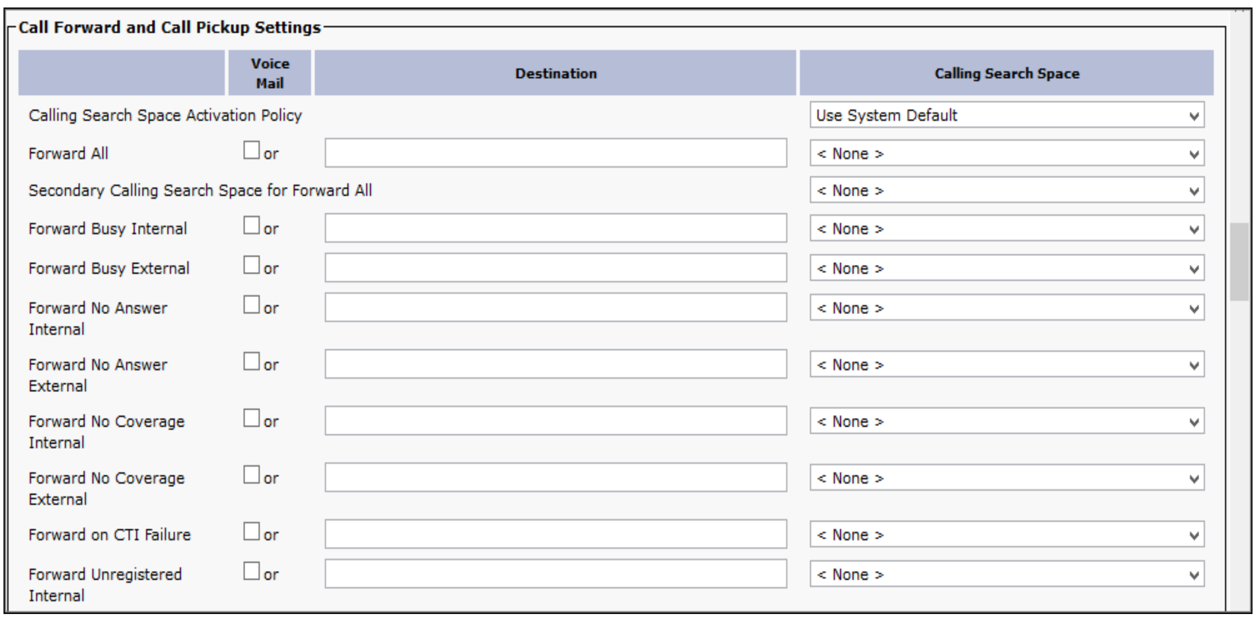
Set Directory Number* = 2754. This is used in this example.
Set Description = Cisco 7975-Phone. This is used in this example.
Set Alerting Name = Cisco 7975-Phone. This is used in this example.
Set ASCII Alerting Name = Cisco 7975-Phone. This is used in this example.

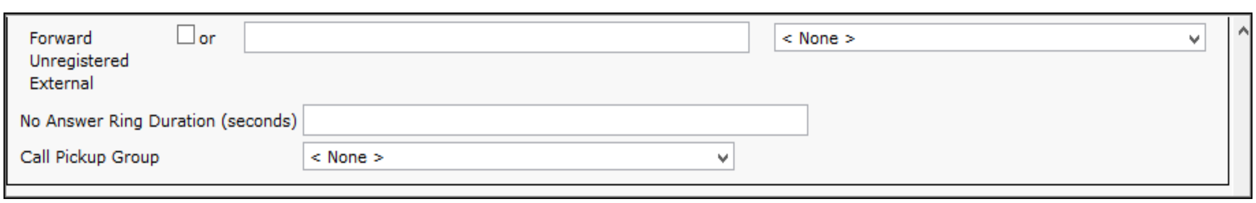




Cisco IP Phone 7975 SIP Configuration (Contd.)

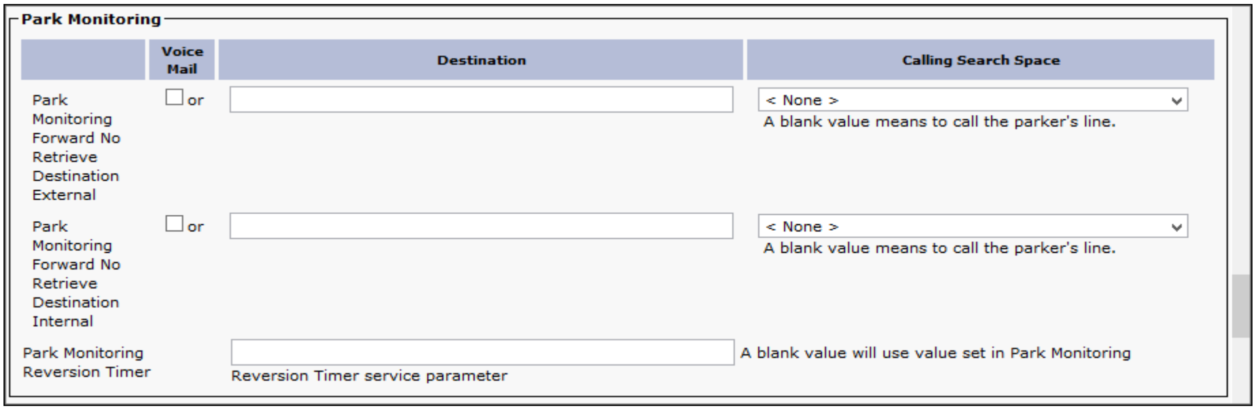






Cisco IP Phone 7975 SIP Configuration (Contd.)

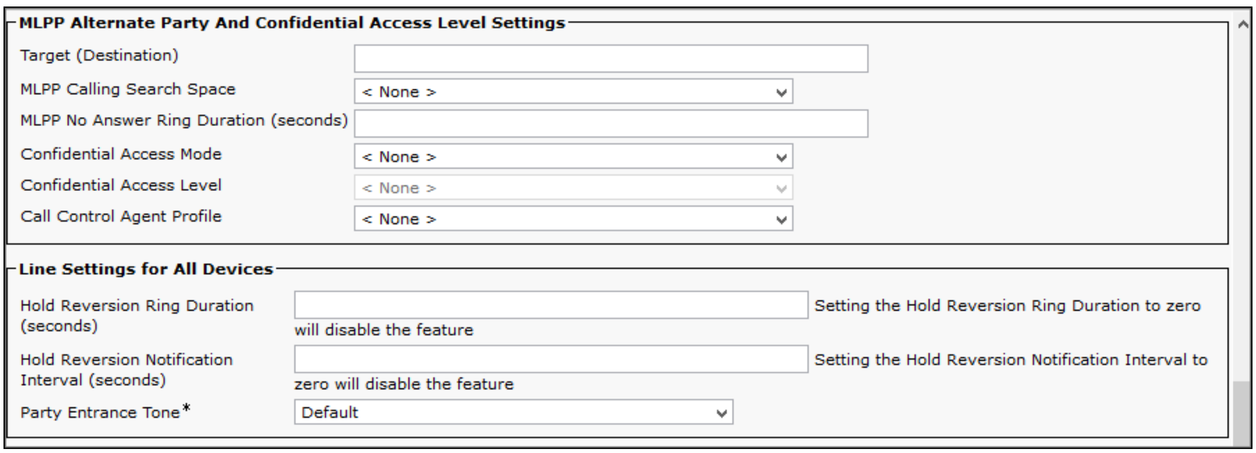
Set Display (Caller ID) = Cisco 7975-Phone. This is used in this example.
Set ASCII Display (Caller ID) = Cisco 7975-Phone. This is used in this example.
Set Line Text Label = Cisco 7975-Phone. This is used in this example.
Set External Phone Number Mask = 7322162754. This is used in this example.



Park Monitoring

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

Park Monitoring Reversion Timer — A blank value will use value set in Park Monitoring Reversion Timer service parameter

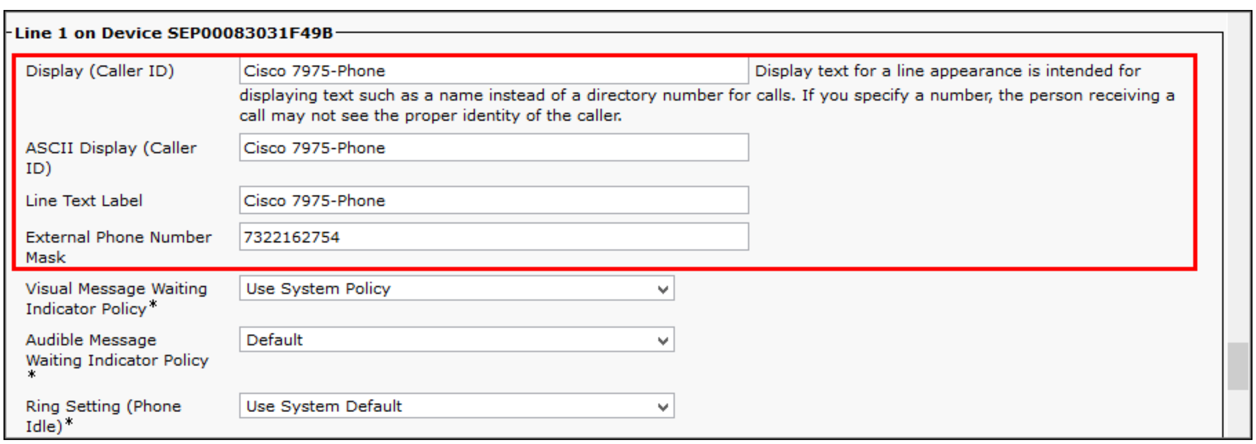


MLPP Alternate Party And Confidential Access Level Settings

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

Line Settings for All Devices

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



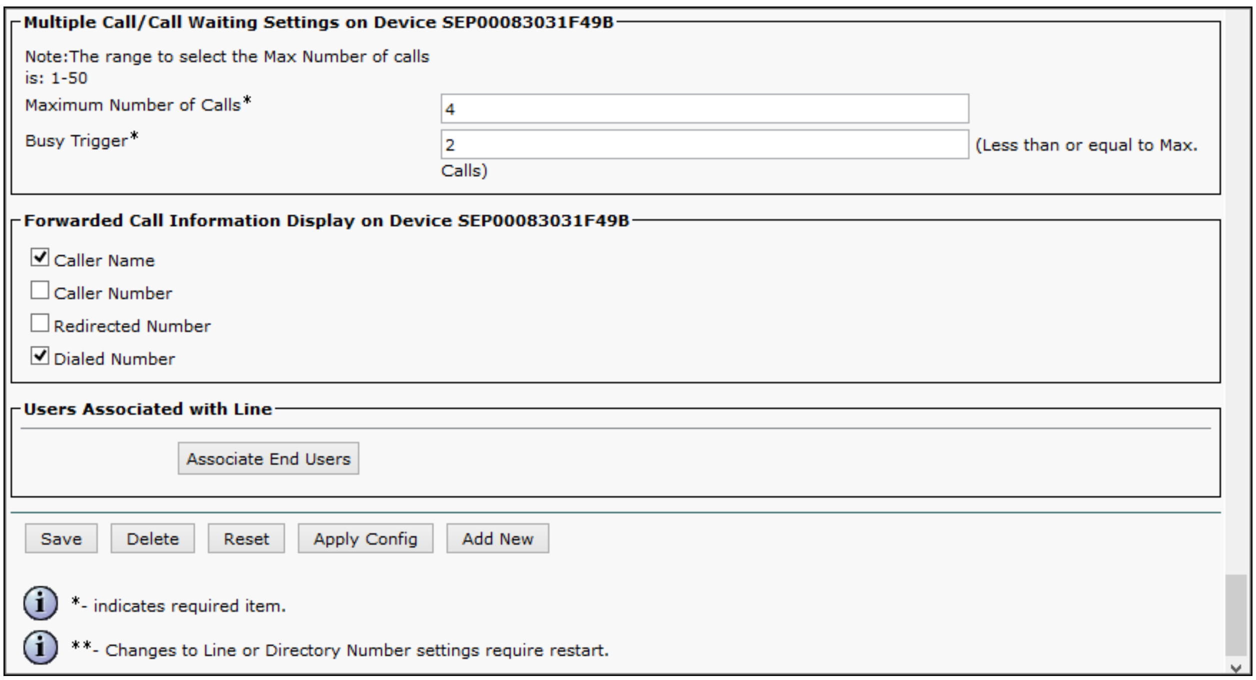
Line 1 on Device SEP00083031F49B

Display (Caller ID)	Cisco 7975-Phone — Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Caller ID)	Cisco 7975-Phone
Line Text Label	Cisco 7975-Phone
External Phone Number Mask	7322162754
Visual Message Waiting Indicator Policy*	Use System Policy
Audible Message Waiting Indicator Policy*	Default
Ring Setting (Phone Idle)*	Use System Default

Cisco IP Phone 7975 SIP Configuration (Continued…)

Ring Setting (Phone Active)	Use System Default — Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)	Use System Default
Call Pickup Group Audio Alert Setting(Phone Active)	Use System Default
Recording Option*	Call Recording Disabled
Recording Profile	< None >
Recording Media Source*	Gateway Preferred
Monitoring Calling Search Space	< None >

☑ Log Missed Calls



Multiple Call/Call Waiting Settings on Device SEP00083031F49B

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

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

Forwarded Call Information Display on Device SEP00083031F49B

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

Users Associated with Line

Associate End Users

Save | Delete | Reset | Apply Config | Add New

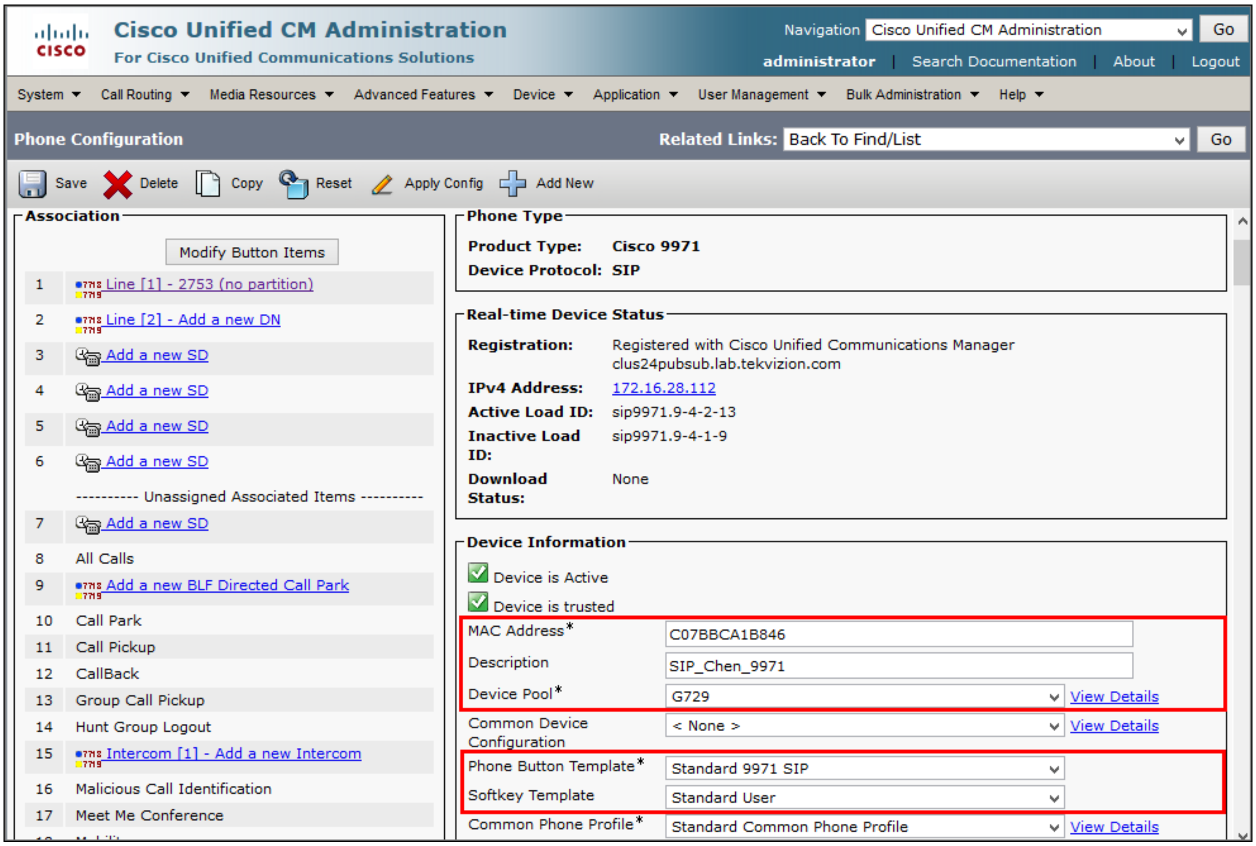
*- indicates required item.

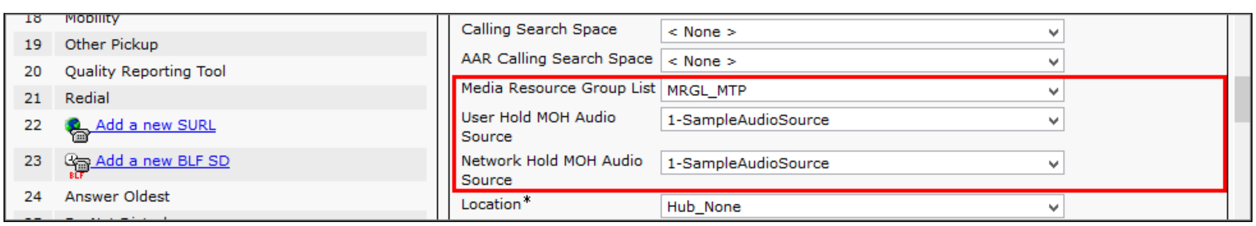
**- Changes to Line or Directory Number settings require restart.

Note: Testing was conducted in tekVizion labs

Cisco IP Phone 9971 SIP Configuration

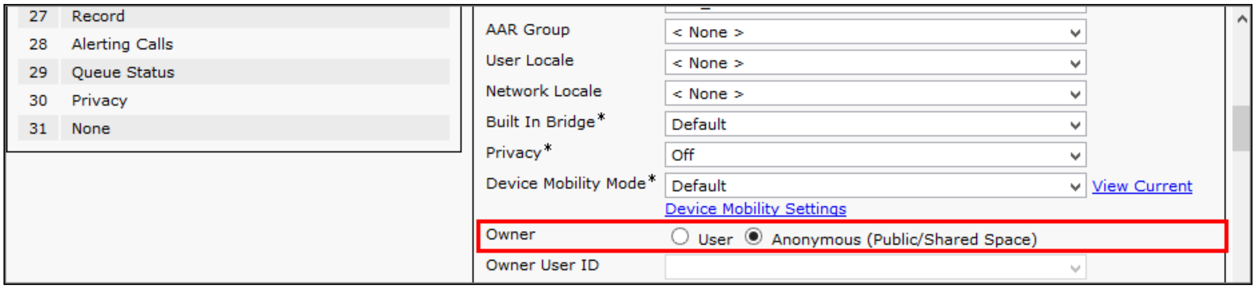
Set MAC Address* = the below mac is used in this example.
Set Description = SIP_Chen_ 9971. This text is used to identify this Phone.
Set Device Pool*= G729. This is used in this example.
Set Phone Button Template*= Standard 9971 SIP. This is used in this example.
Set Soft key Template = Standard User.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

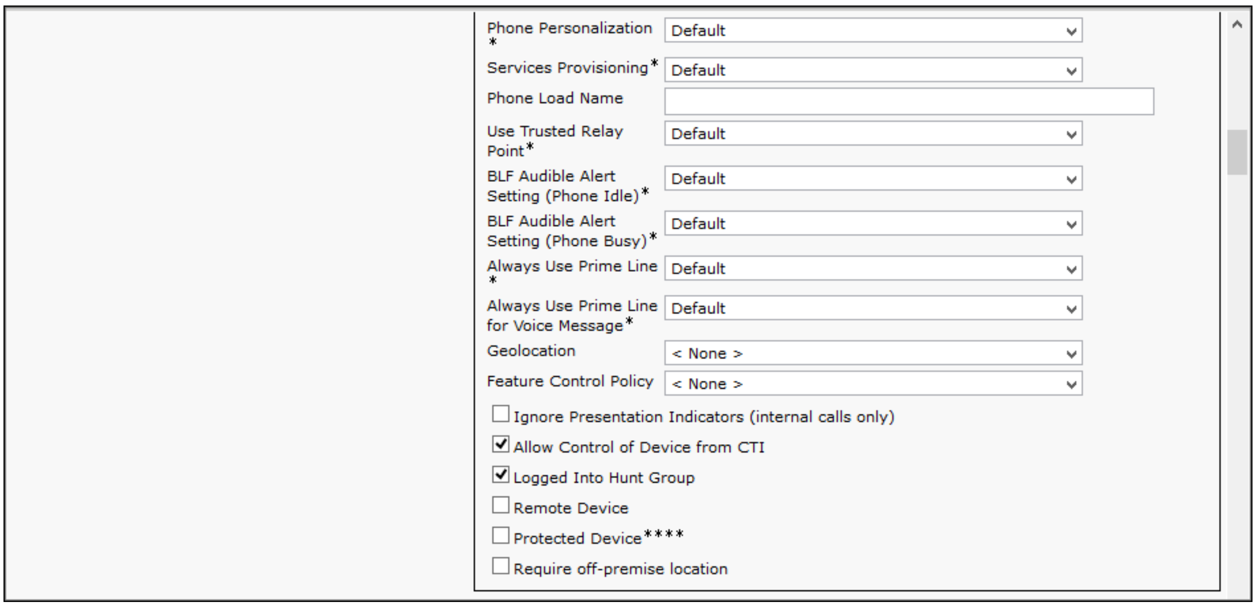




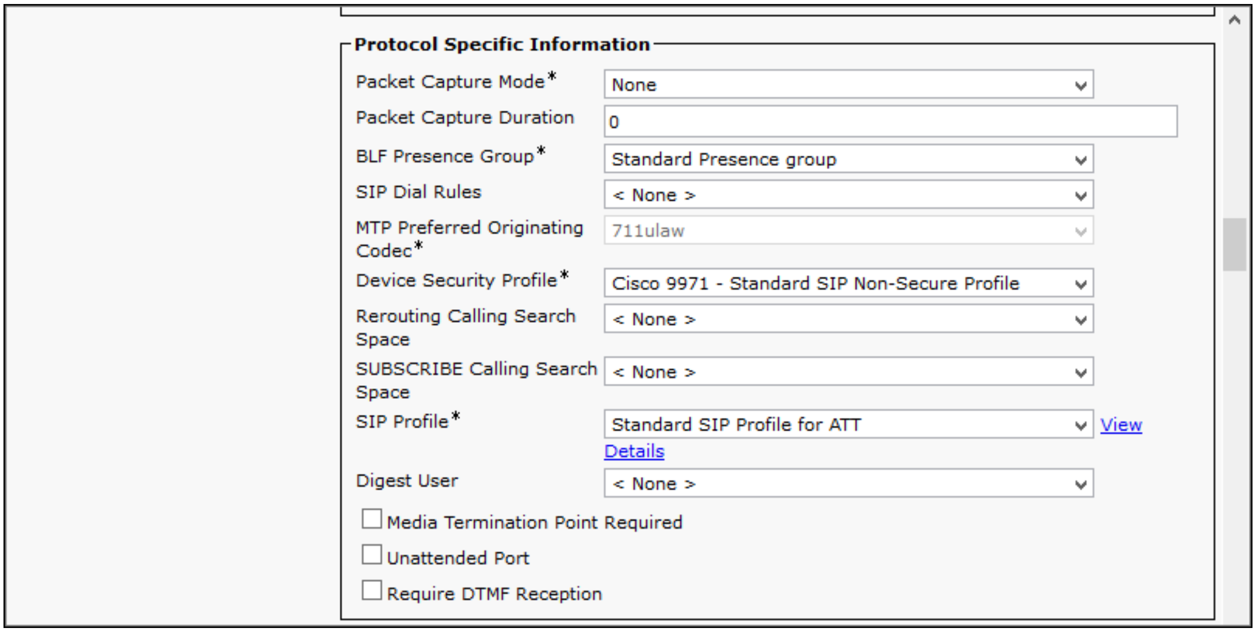
Cisco IP Phone 9971 SIP Configuration (Contd.)

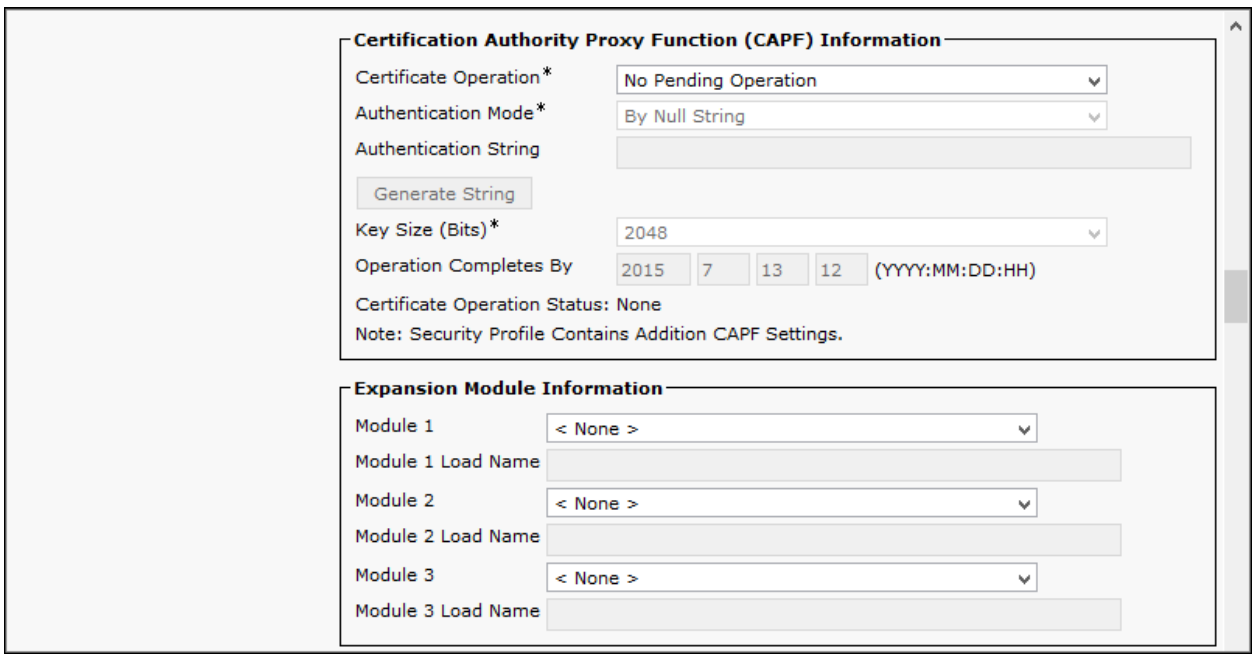
Check “Owner = Anonymous (Public/Shared Space)” This is used in this example.

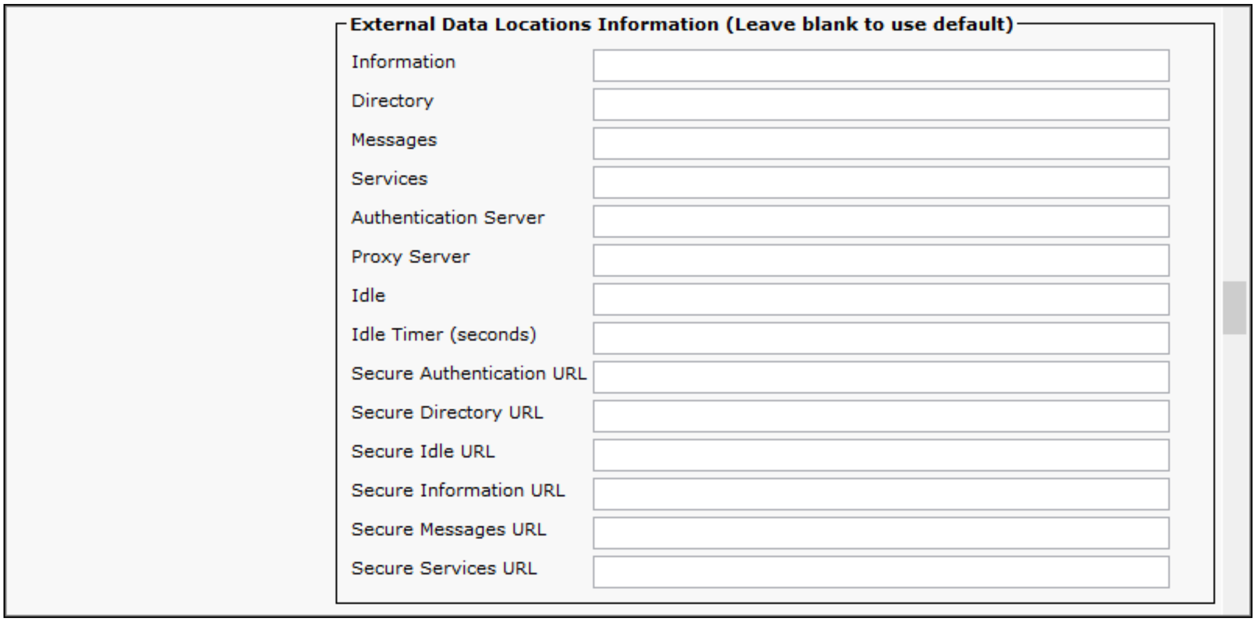


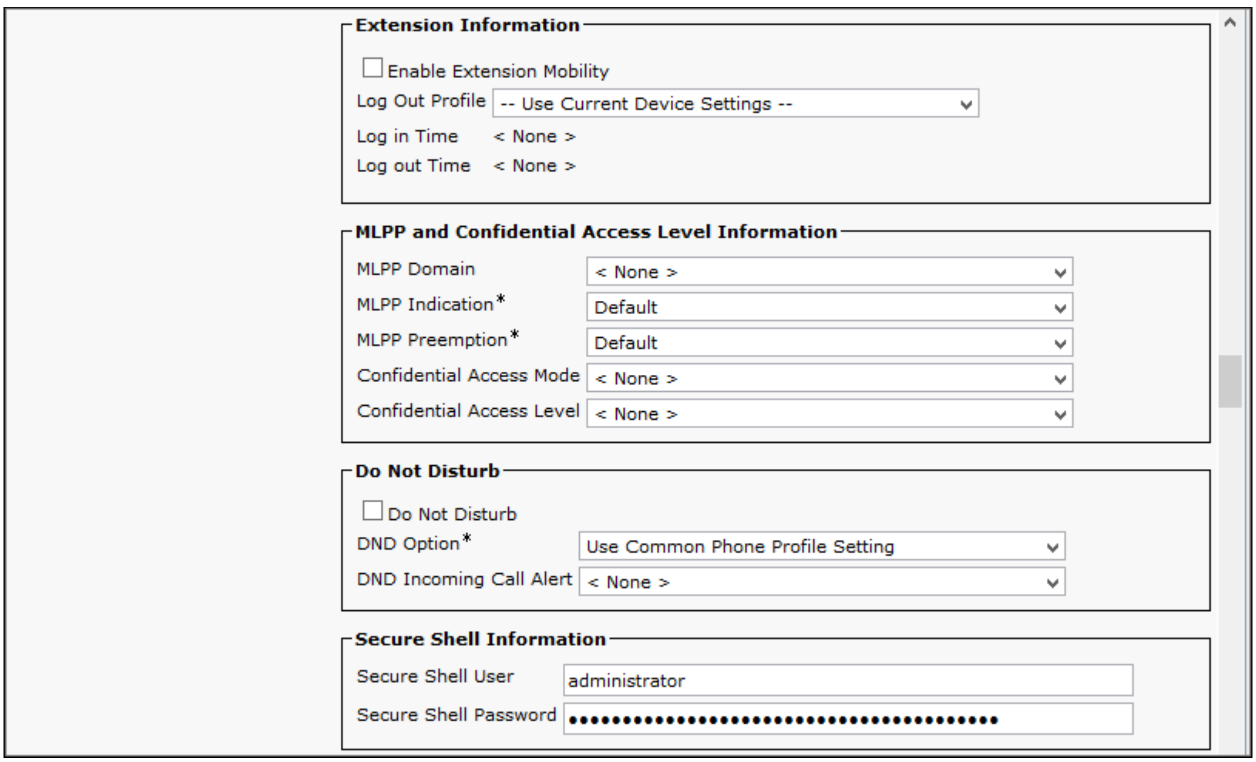


Cisco IP Phone 9971 SIP Configuration (Contd.)



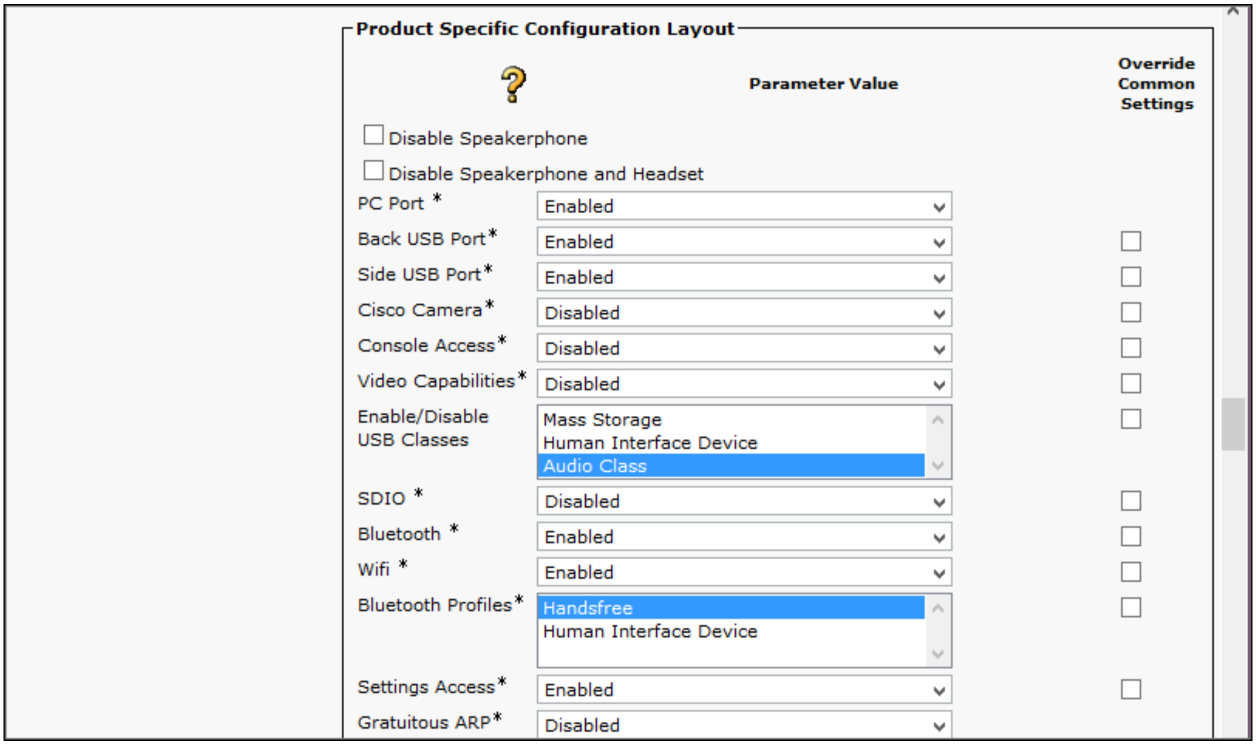


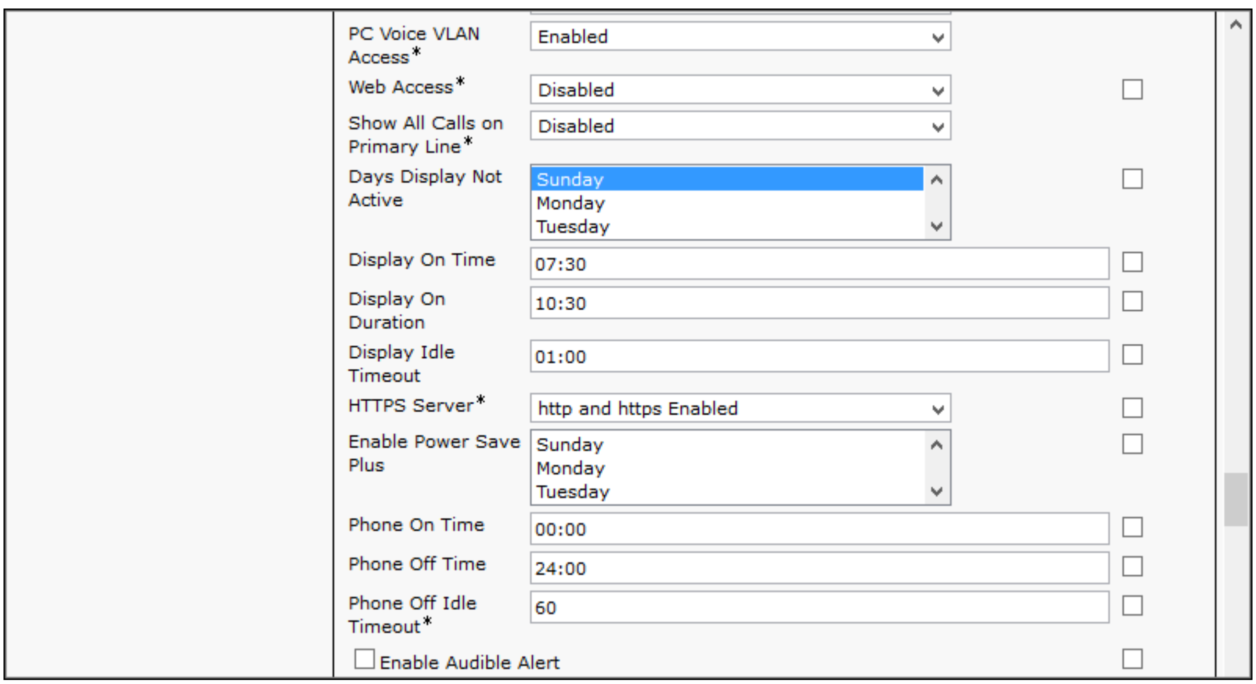




Note: Testing was conducted in tekVizion labs

Cisco IP Phone 9971 SIP Configuration (Contd.)



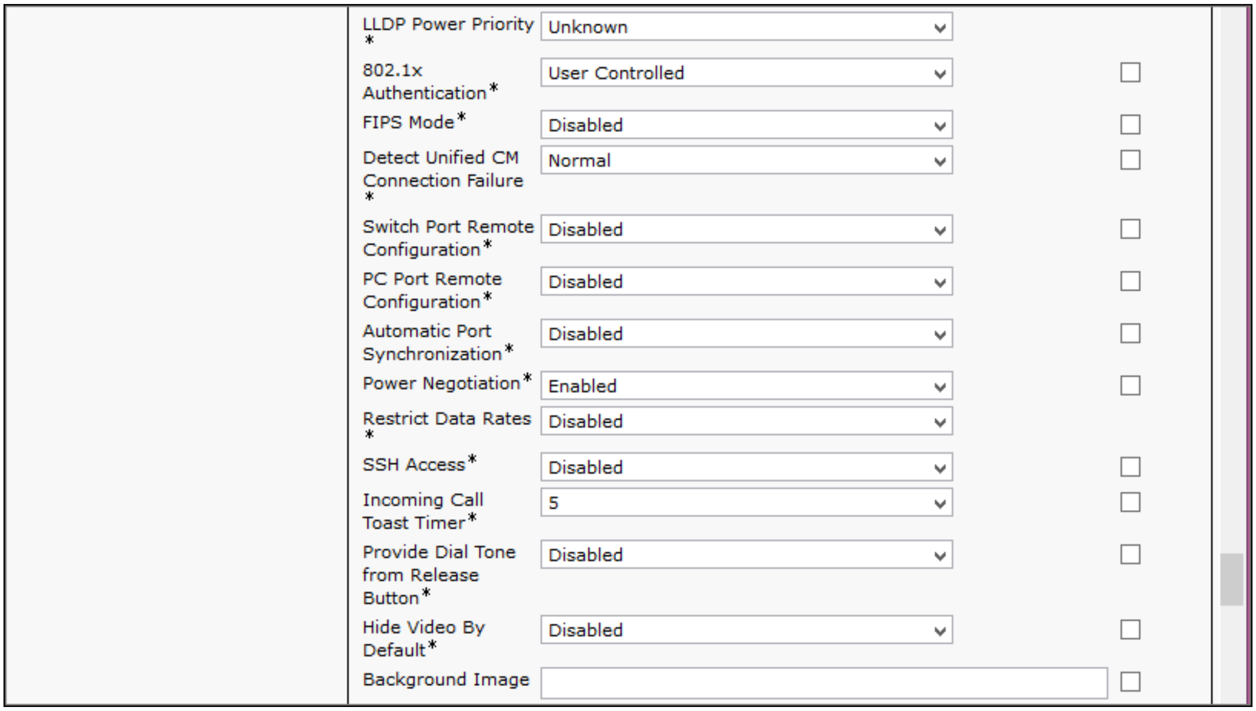


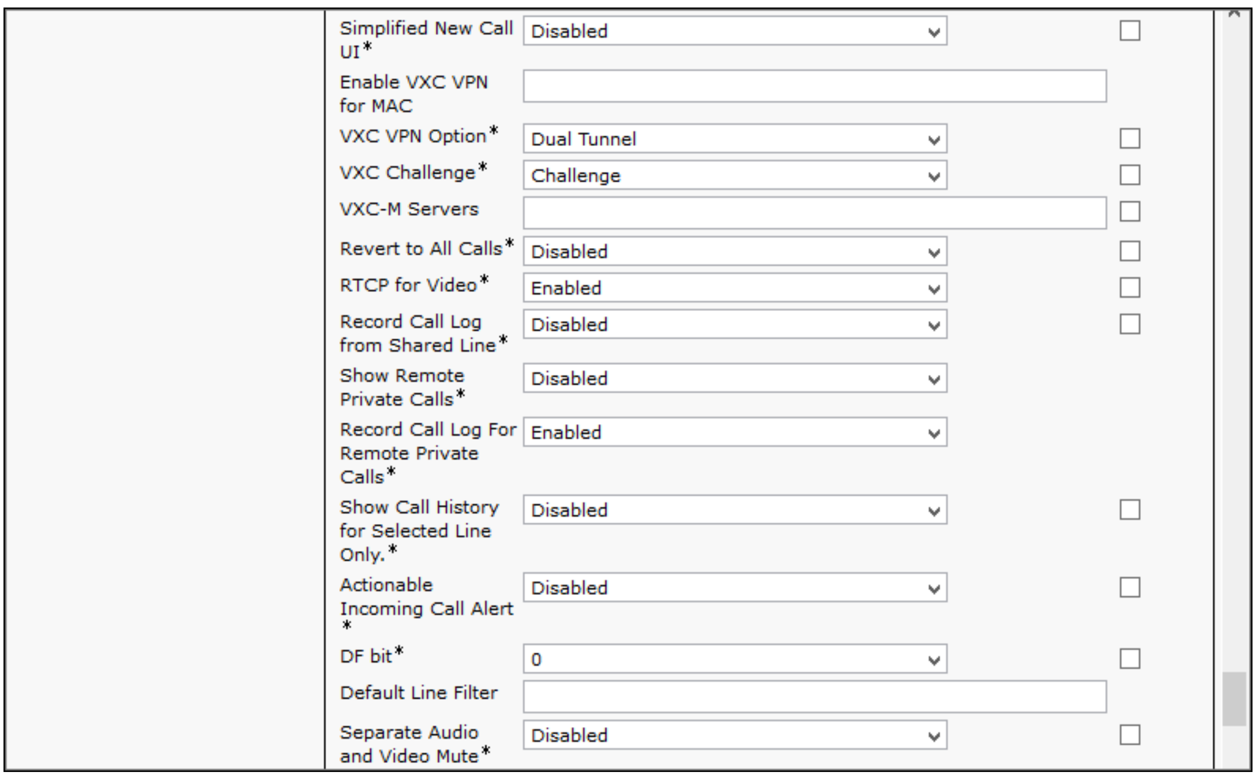
Cisco IP Phone 9971 SIP Configuration (Contd.)

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*	Disabled
Log Server	
IPv6 Log Server	
Remote Log*	Disabled

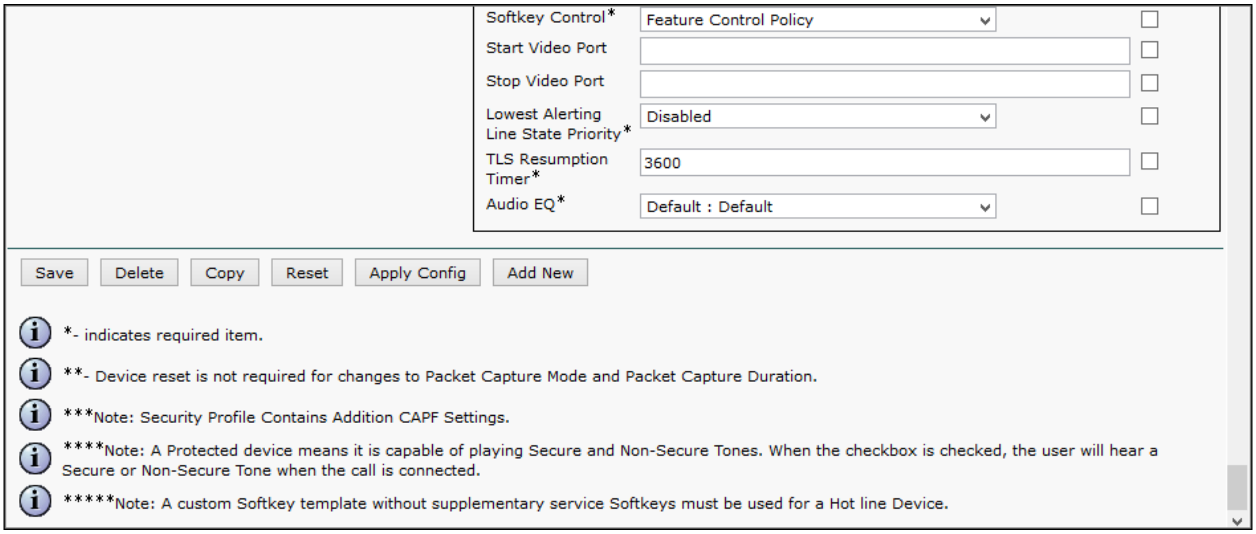
Log Profile	Default / Preset / Telephony (Preset selected)
Advertise G.722 and iSAC Codecs *	Use System Default
Wideband Headset UI Control*	Enabled
Wideband Headset *	Enabled
Peer Firmware Sharing*	Enabled
Cisco Discovery Protocol (CDP): Switch Port*	Enabled
Cisco Discovery Protocol (CDP): PC Port*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

Cisco IP Phone 9971 SIP Configuration (Contd.)



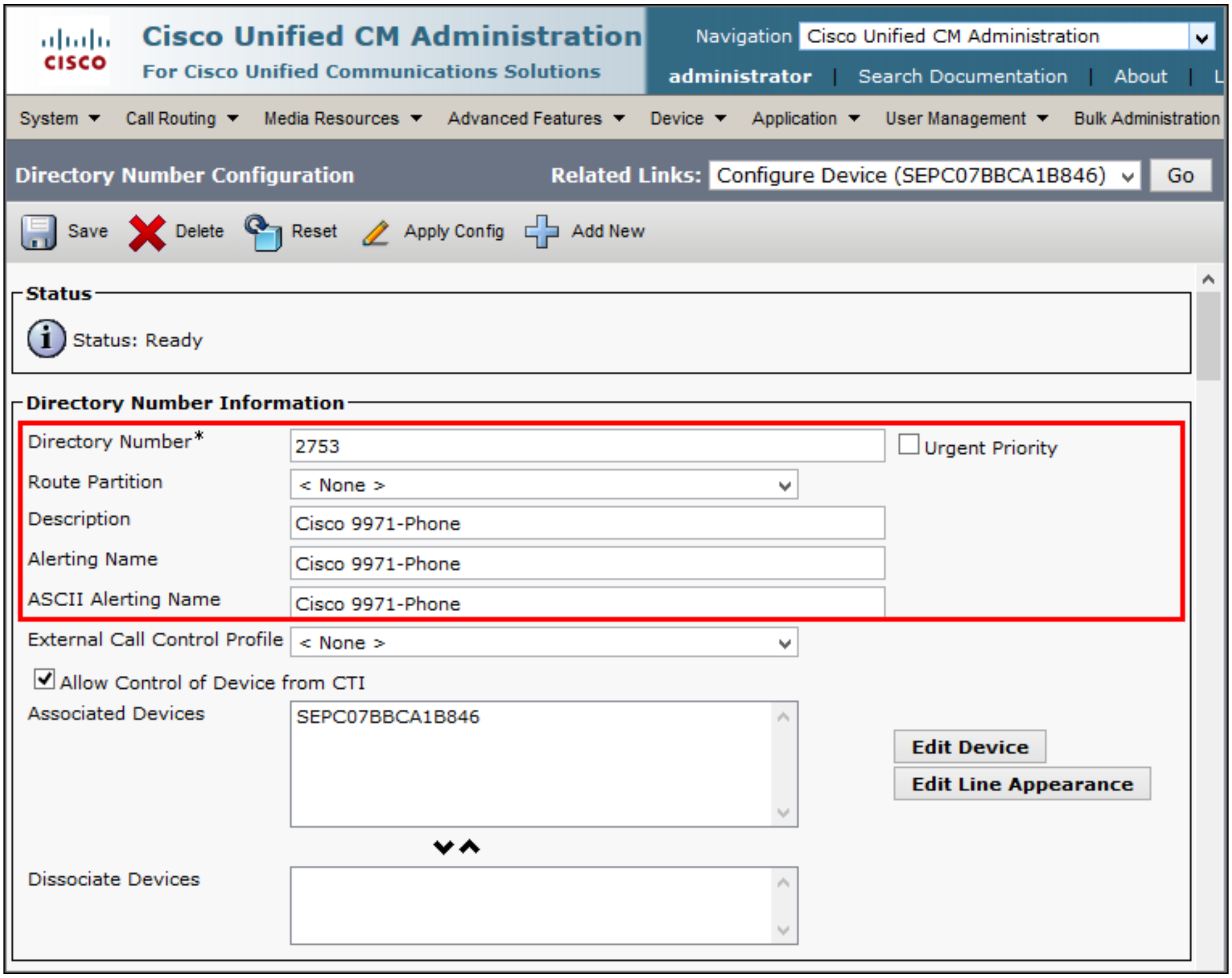


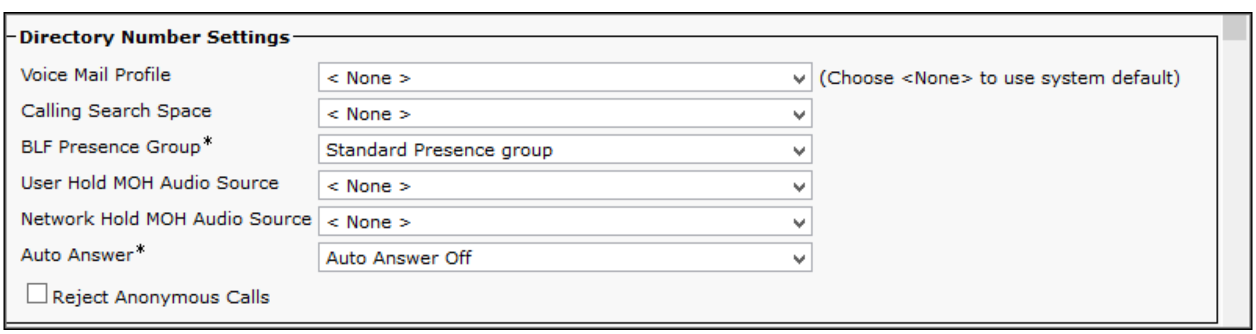
Cisco IP Phone 9971 SIP Configuration (Contd.)



Cisco IP Phone 9971 SIP Configuration (Contd.)

Set Directory Number* = 2753. This is used in this example.
Set Description = Cisco 9971-Phone. This is used in this example.
Set Alerting Name = Cisco 9971-Phone. This is used in this example.
Set ASCII Alerting Name = Cisco 9971-Phone. This is used in this example.





Cisco IP Phone 9971 SIP Configuration (Contd.)

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

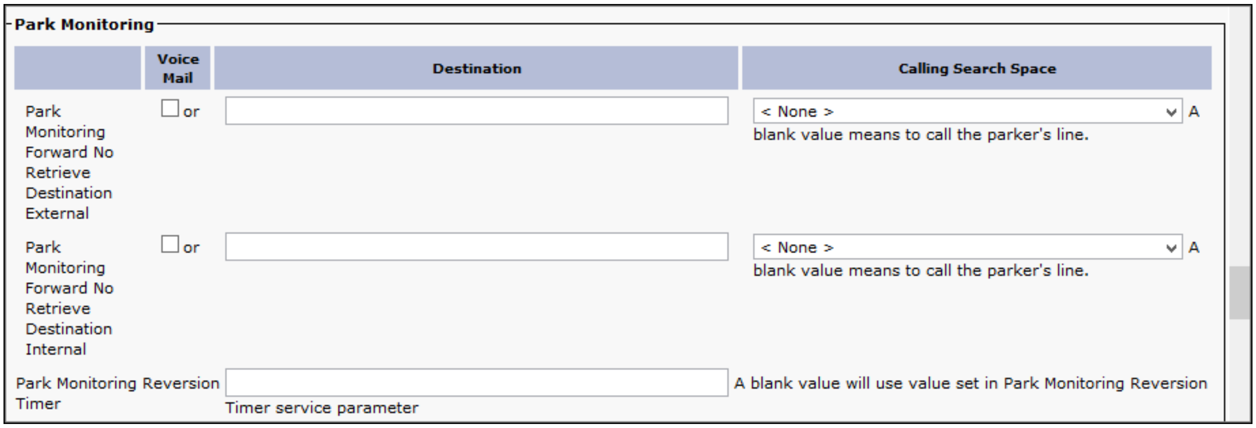
	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			Use System Default
Forward All	☐ or		< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	☐ or		< None >
Forward Busy External	☐ or		< None >
Forward No Answer Internal	☐ or		< None >
Forward No Answer External	☐ or		< None >
Forward No Coverage Internal	☐ or		< None >
Forward No Coverage External	☐ or		< None >
Forward on CTI Failure	☐ or		< None >
Forward Unregistered Internal	☐ or		< None >
Forward Unregistered External	☐ or		< None >

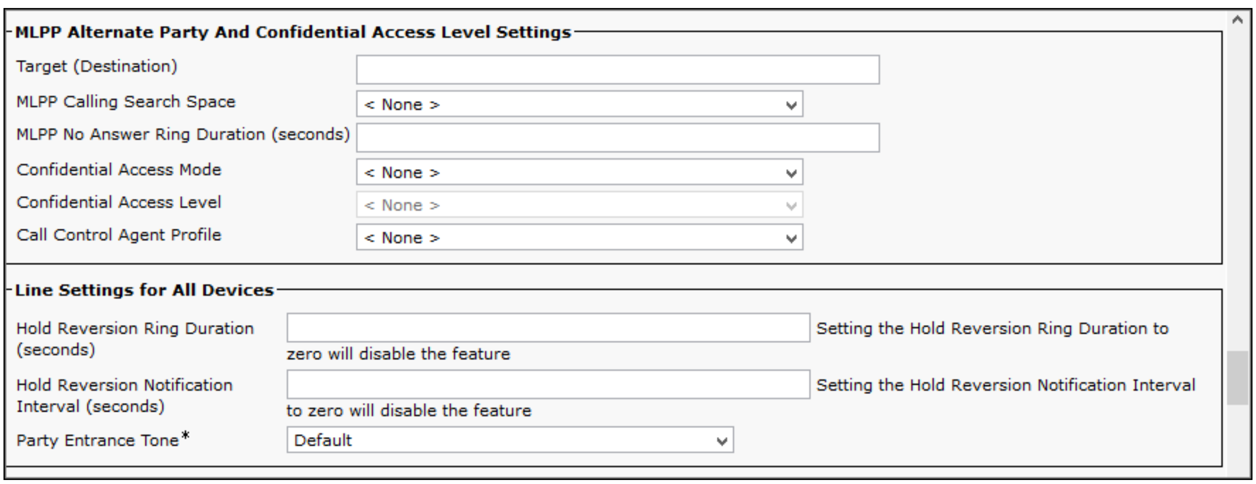
No Answer Ring Duration (seconds):

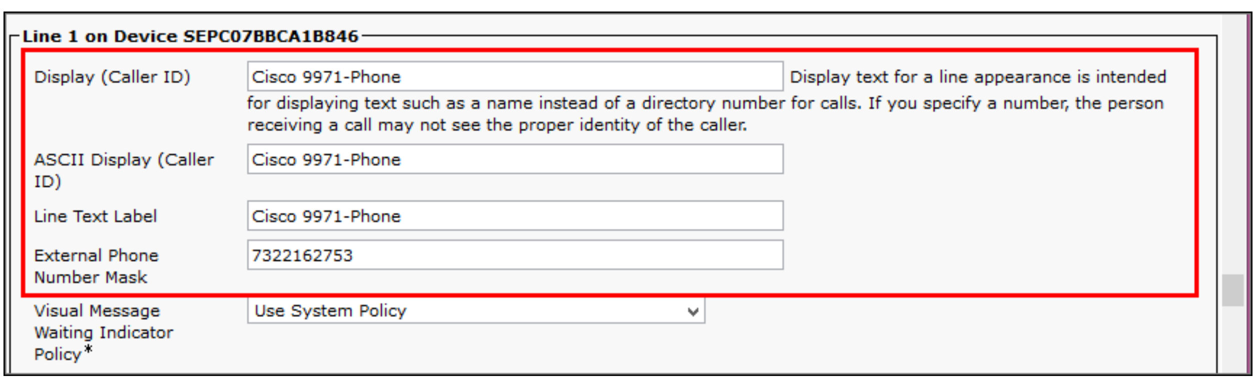
Call Pickup Group: < None >

Cisco IP Phone 9971 SIP Configuration (Contd.)

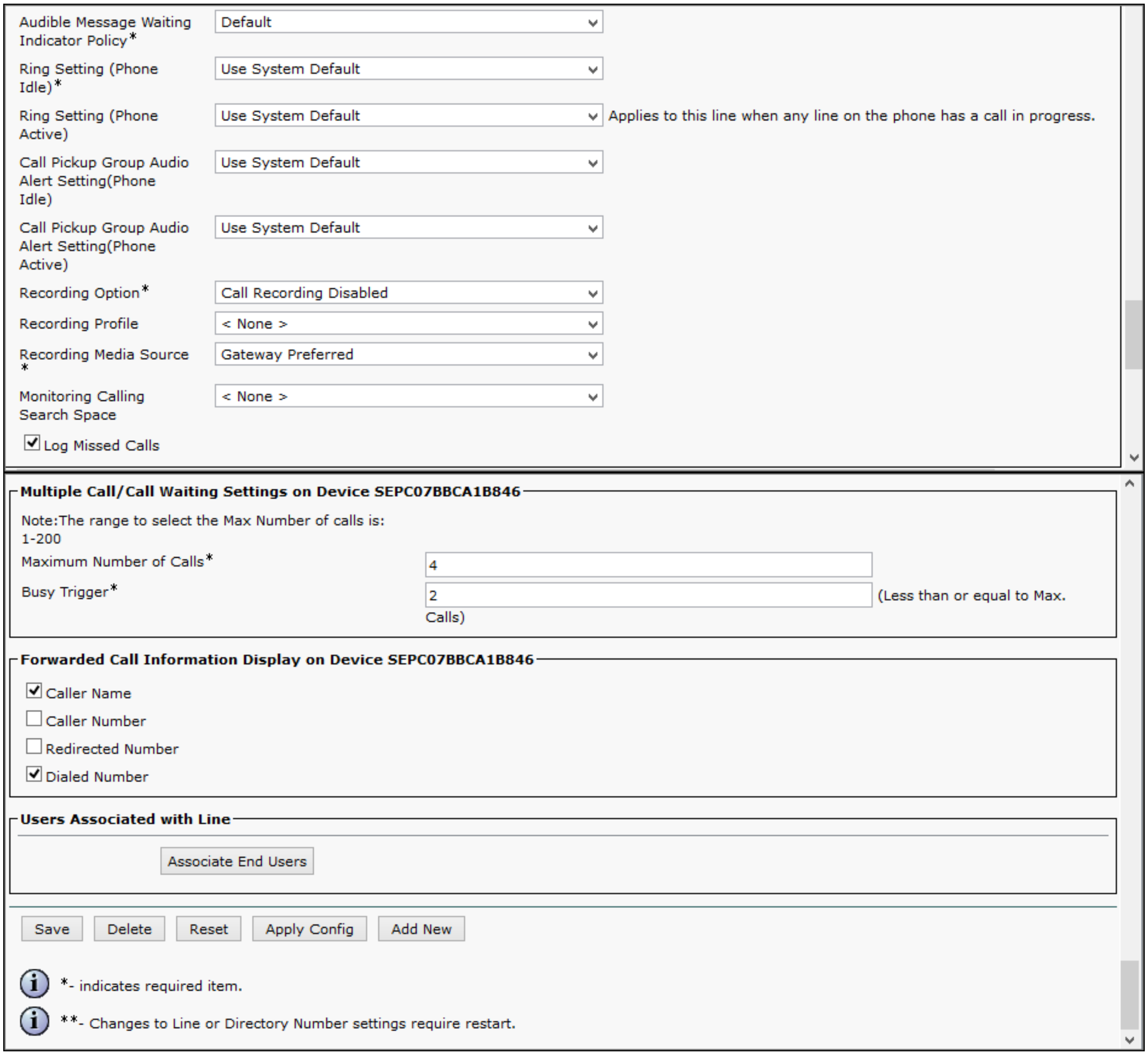
Set Display (Caller ID) = Cisco 9971-Phone. This is used in this example.
Set ASCII Display (Caller ID) = Cisco 9971-Phone. This is used in this example.
Set Line Text Label = Cisco 9971-Phone. This is used in this example.
Set External Phone Number Mask = 7322162753. This is used in this example.







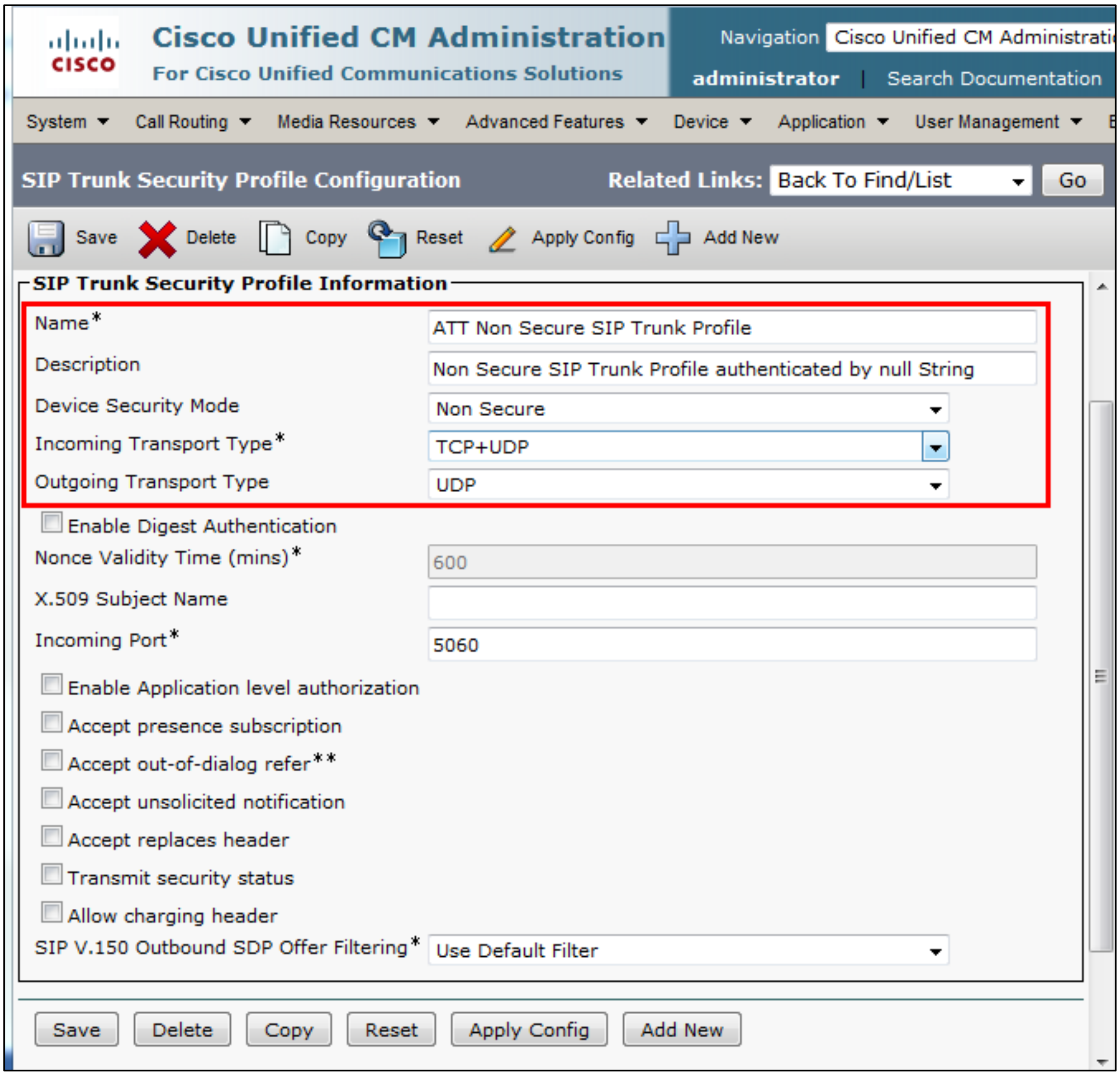
Cisco IP Phone 9971 SIP Configuration (Contd.)



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

Navigation: System → Security → SIP Trunk Security Profile

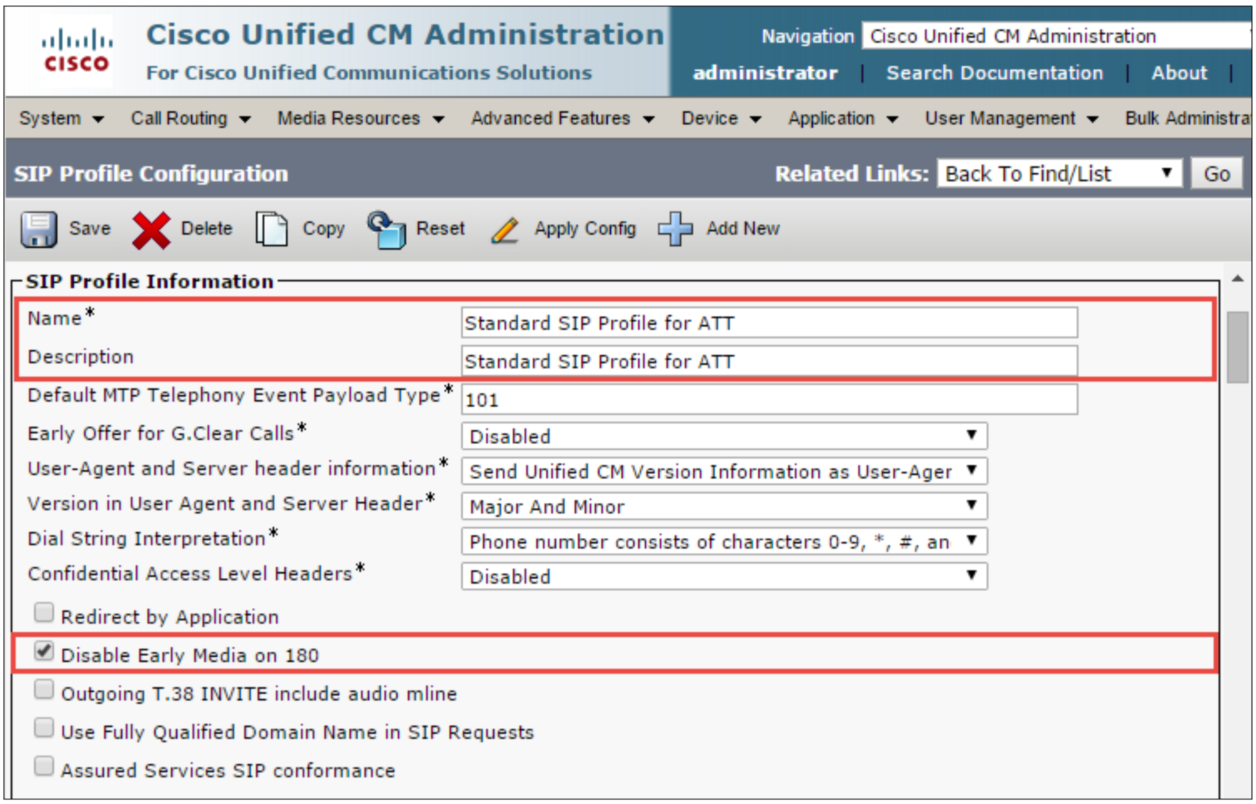
Set Name* = ATT Non Secure SIP Trunk Profile. This is used in this example.
Set Description = Non Secure SIP Trunk Profile authenticated by null String. This is used in this example.
Set Device Security Mode = Non Secure
Set Incoming Transport Type* = TCP+UDP
Set Outgoing Transport Type = UDP



SIP Profile Configuration used by SIP trunk to Cisco UBE

Navigation: Device → Device Settings → SIP Profile

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



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

SDP Information

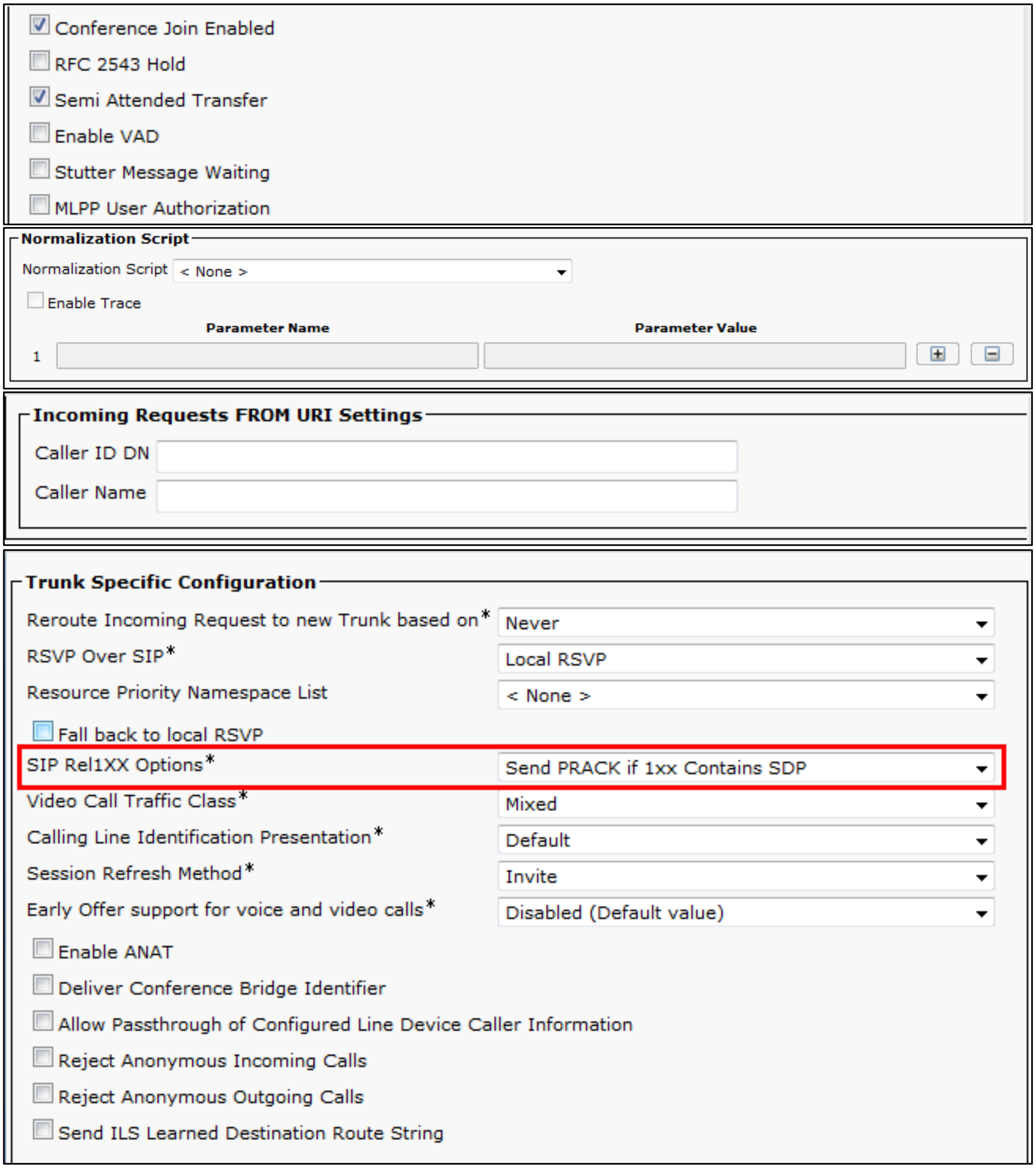
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer *	Default

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

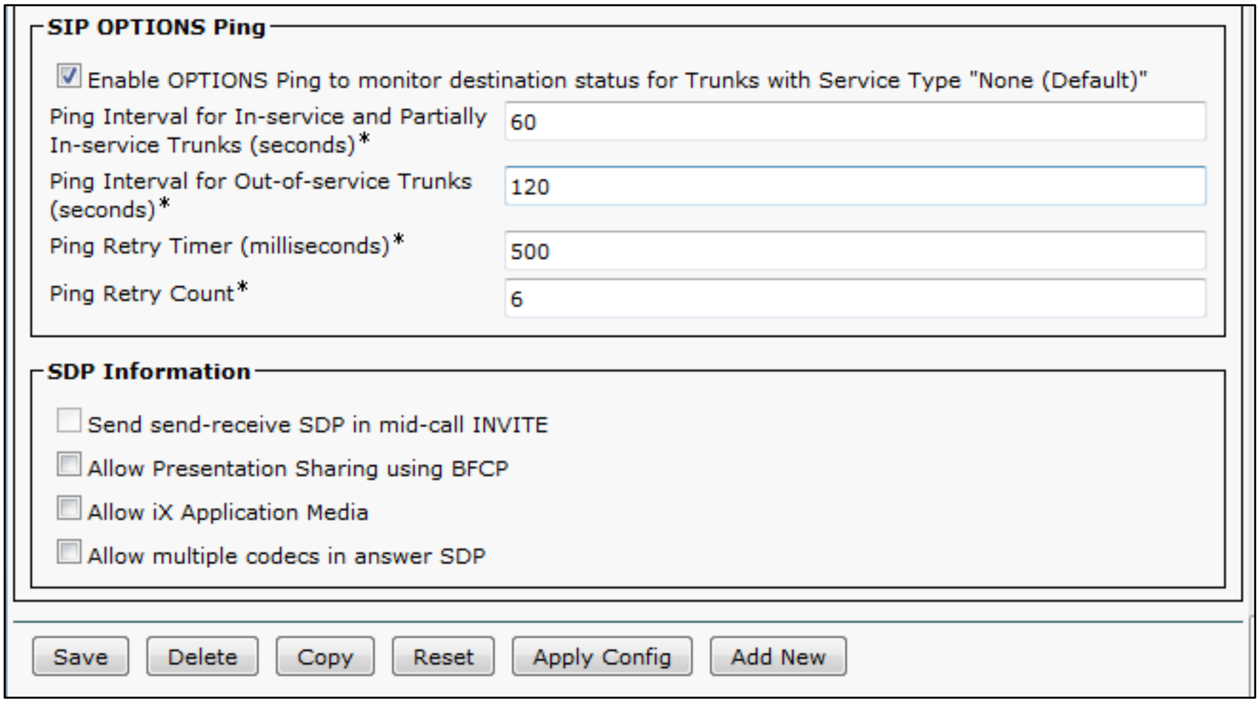
Parameters used in Phone

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

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

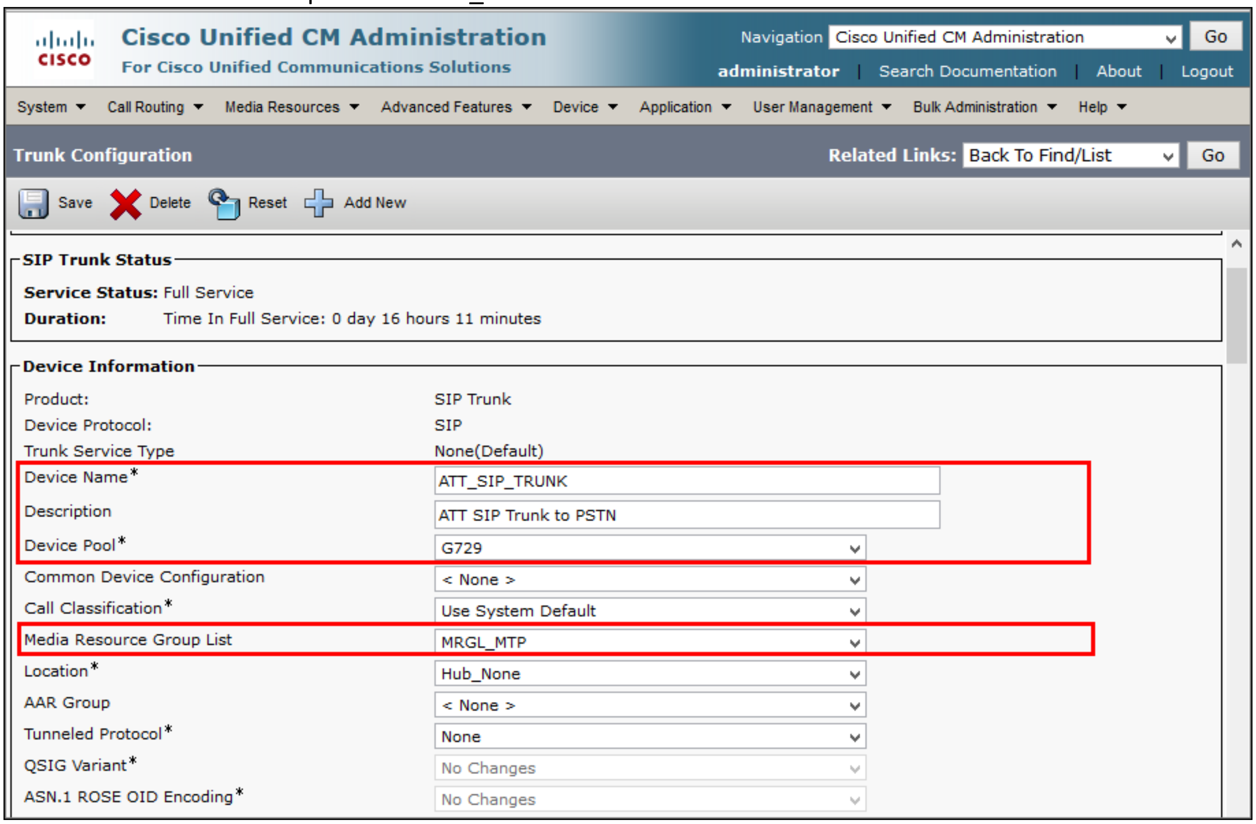


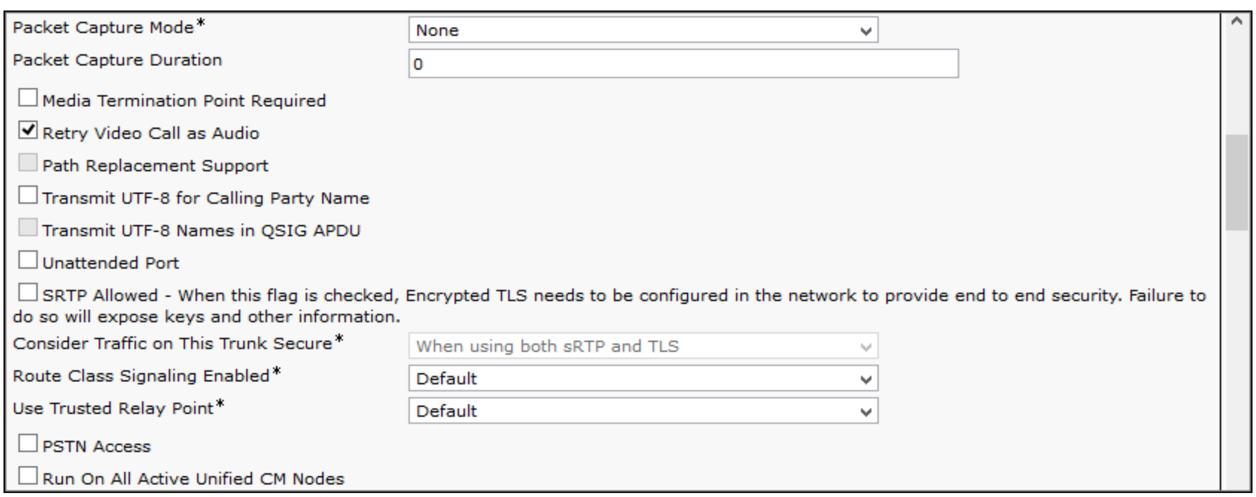
SIP Profile Configuration used by SIP trunk to Cisco UBE (Contd.)



SIP Trunk to Cisco UBE Configuration

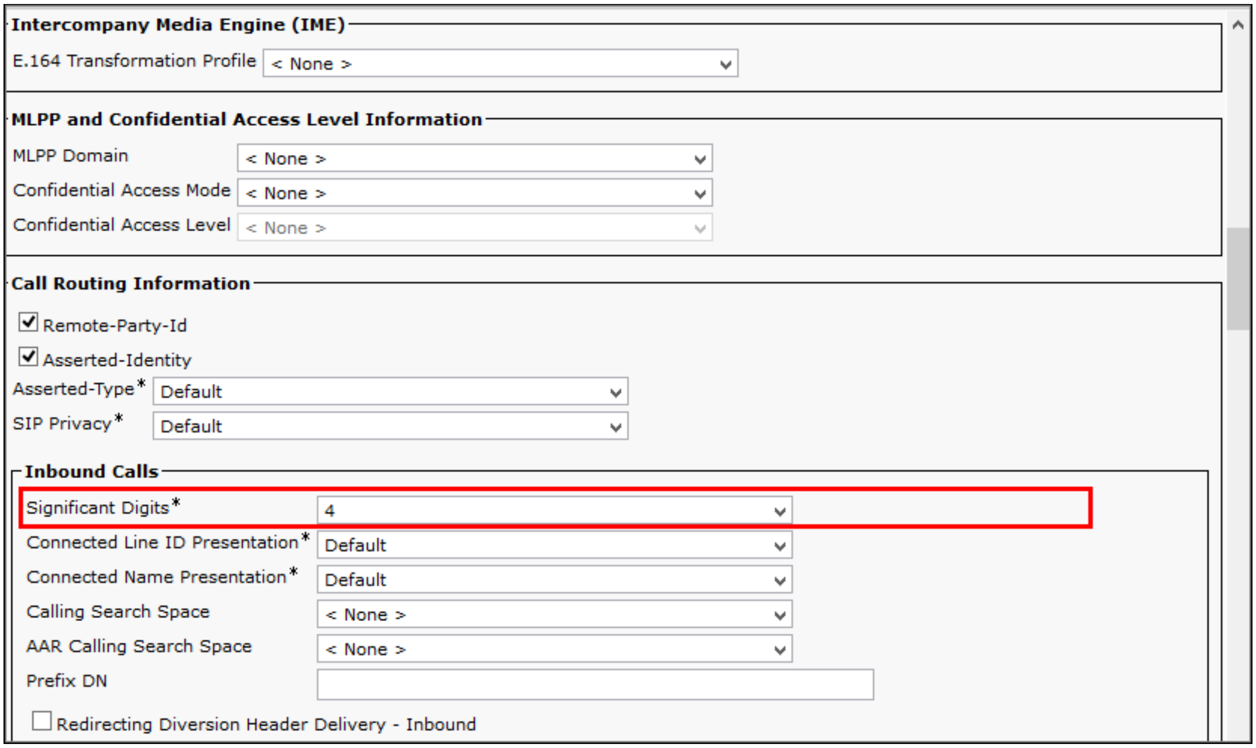
Navigation: Device → Trunk
Set Device Name* = ATT_SIP_TRUNK. This is used for this example
Set Description = ATT SIP Trunk to PSTN. This is used for this example
Set Device Pool* = G729. This is used for this example
Set Media Resource Group List = MRGL_MTP.

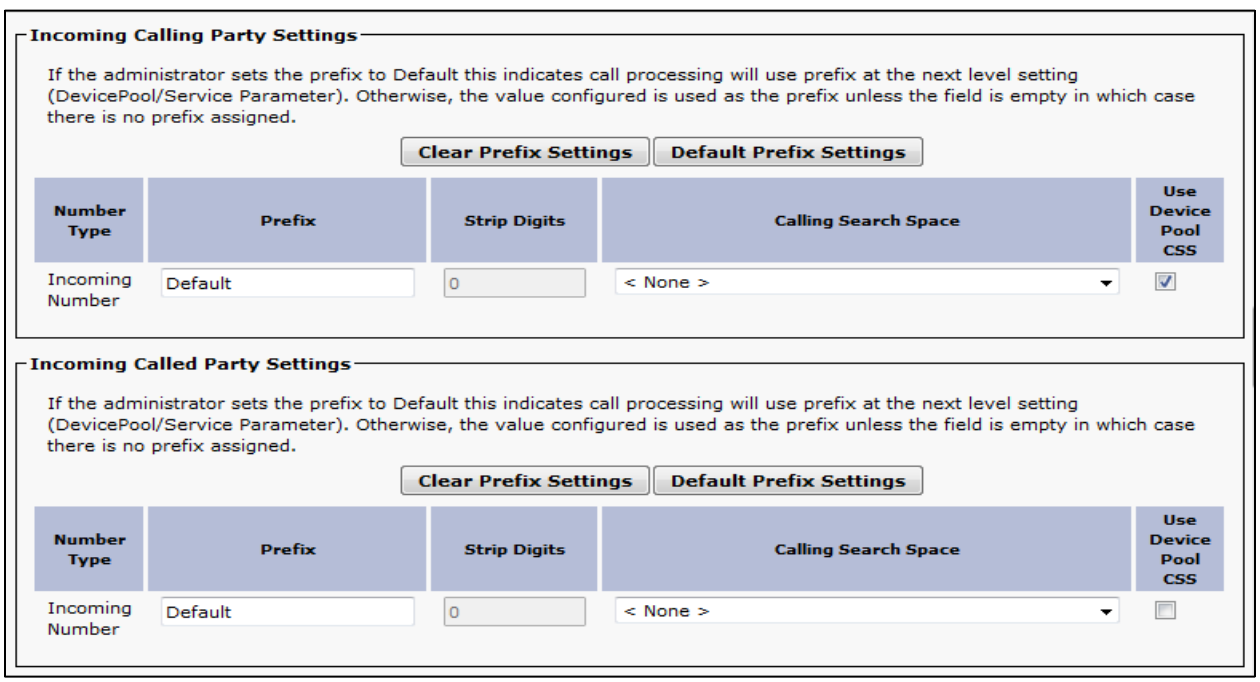




SIP Trunk to Cisco UBE Configuration (Contd.)

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

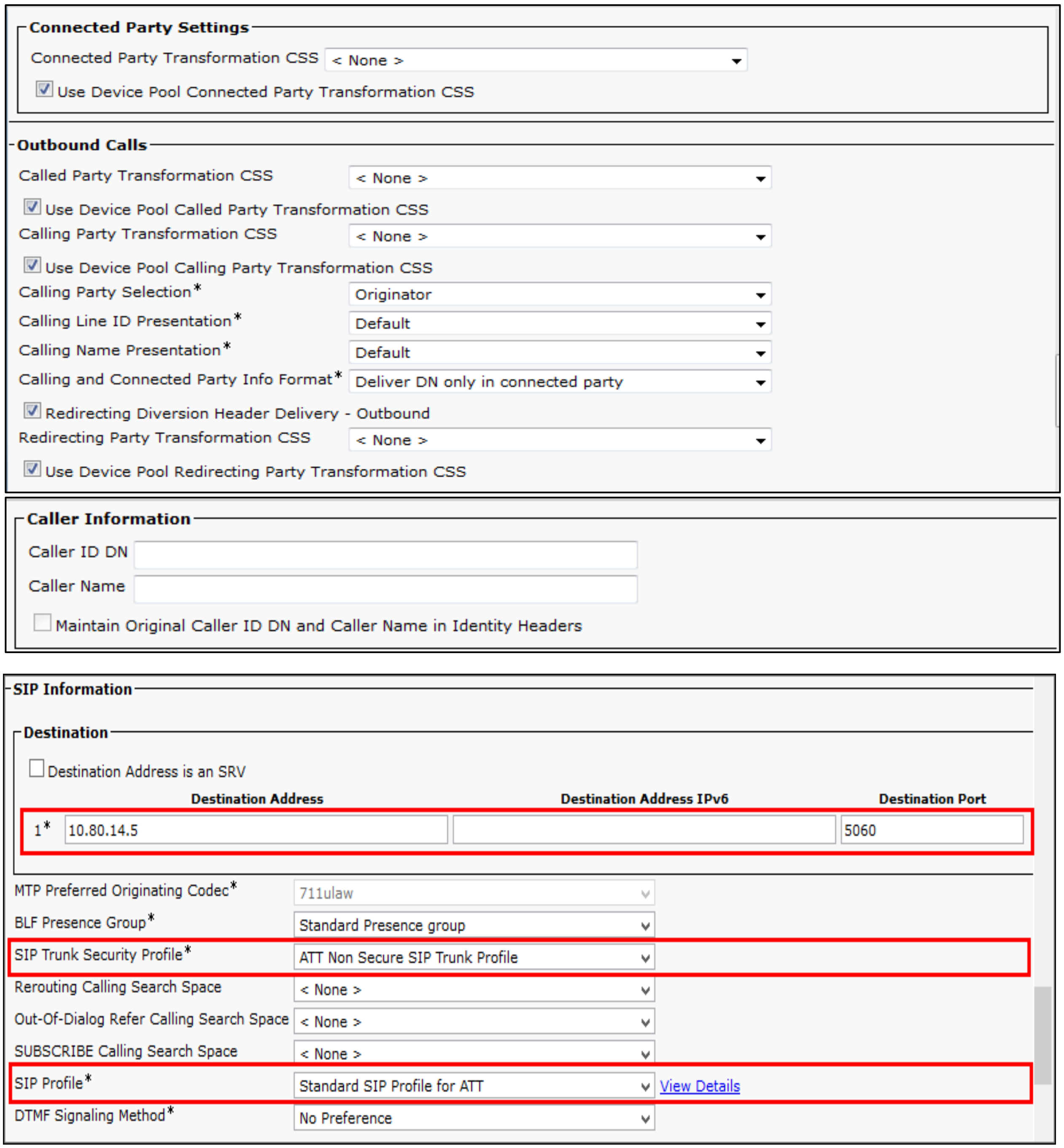




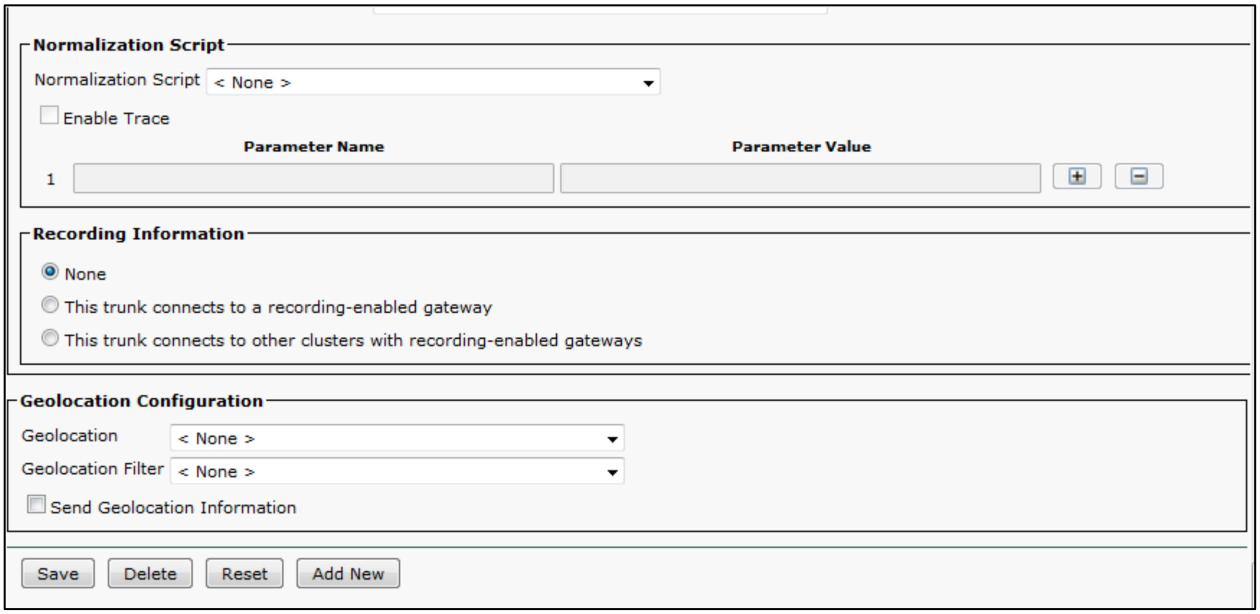
Note: Testing was conducted in tekVizion labs

SIP Trunk to Cisco UBE Configuration (Contd.)

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



SIP Trunk to Cisco UBE Configuration (Contd.)



SIP Trunk to Fax Gateway Configuration

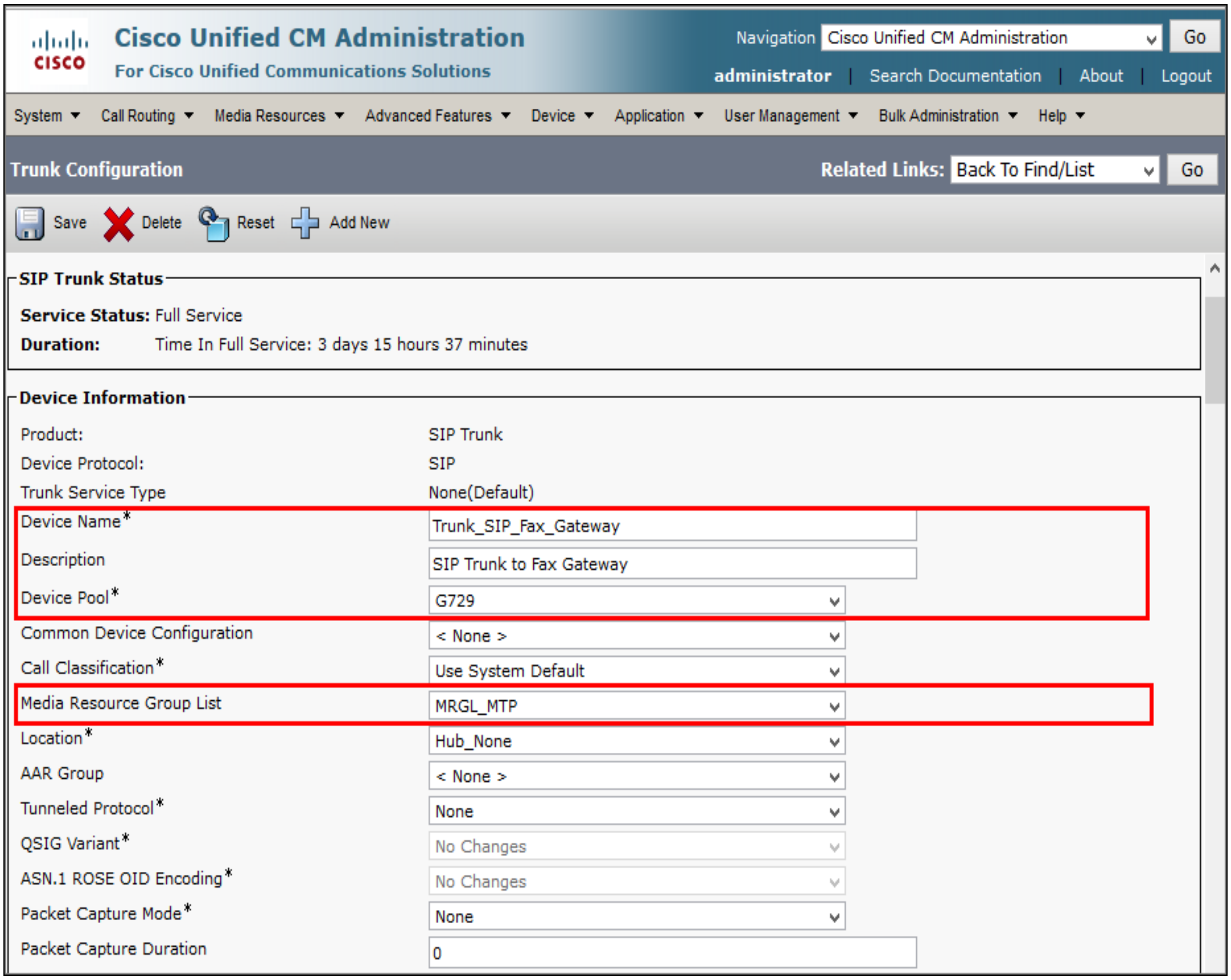
Navigation: Device → Trunk

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

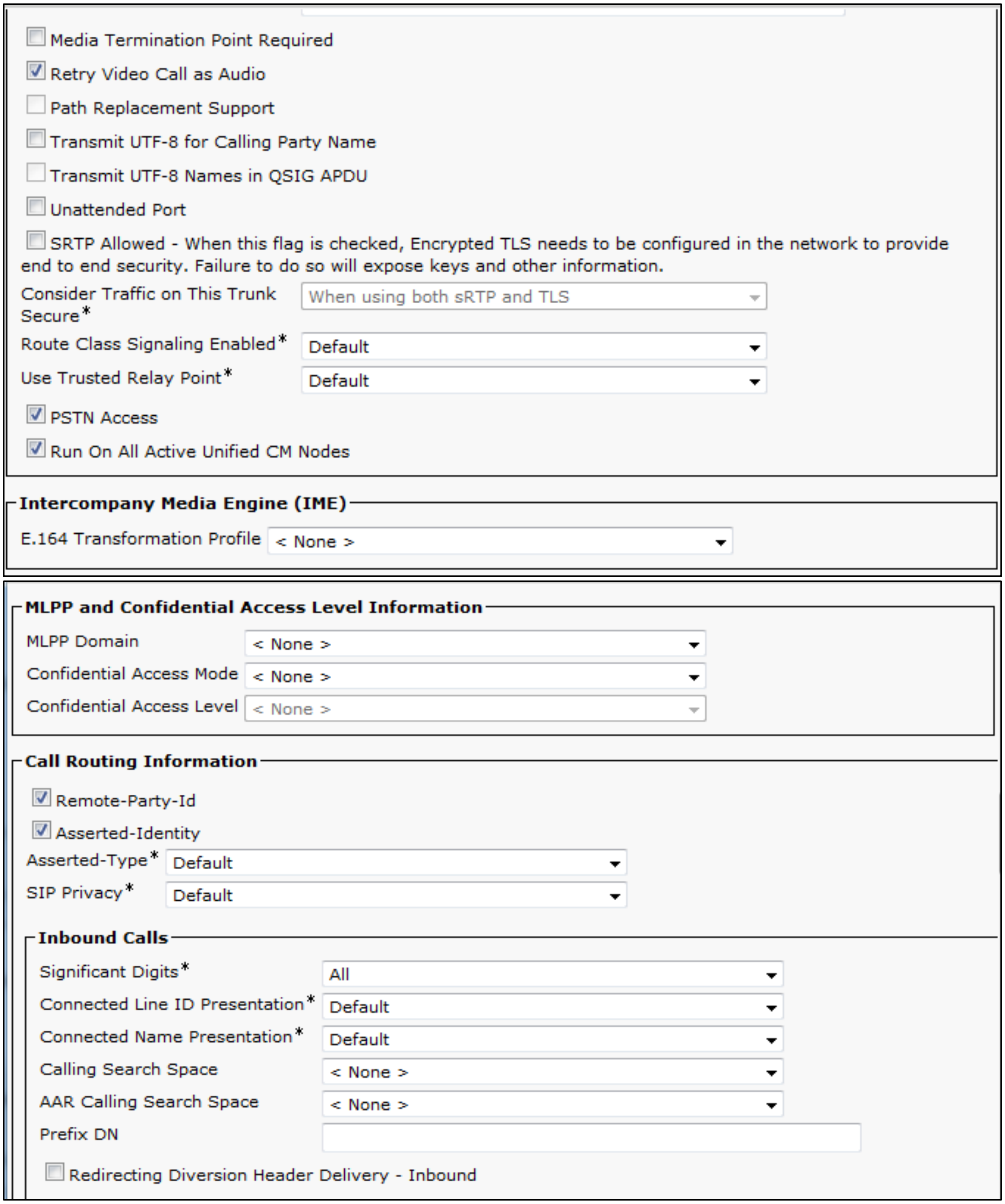
Set Description = SIP Trunk to FAX Gateway. This is used for this example

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

Set Media Resource Group List = MRGL_MTP.



SIP Trunk to Fax Gateway Configuration (Contd.)



SIP Trunk to Fax Gateway Configuration (Contd.)

Incoming Calling Party Settings

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

Clear Prefix Settings | Default Prefix Settings

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

Incoming Called Party Settings

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

Clear Prefix Settings | Default Prefix Settings

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

Connected Party Settings

Connected Party Transformation CSS: < None >

☑ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS: < None >

☑ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS: < None >

☑ Use Device Pool Calling Party Transformation CSS

Calling Party Selection*: Originator

Calling Line ID Presentation*: Default

Calling Name Presentation*: Default

Calling and Connected Party Info Format*: Deliver DN only in connected party

☐ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS: < None >

☑ Use Device Pool Redirecting Party Transformation CSS

Caller Information

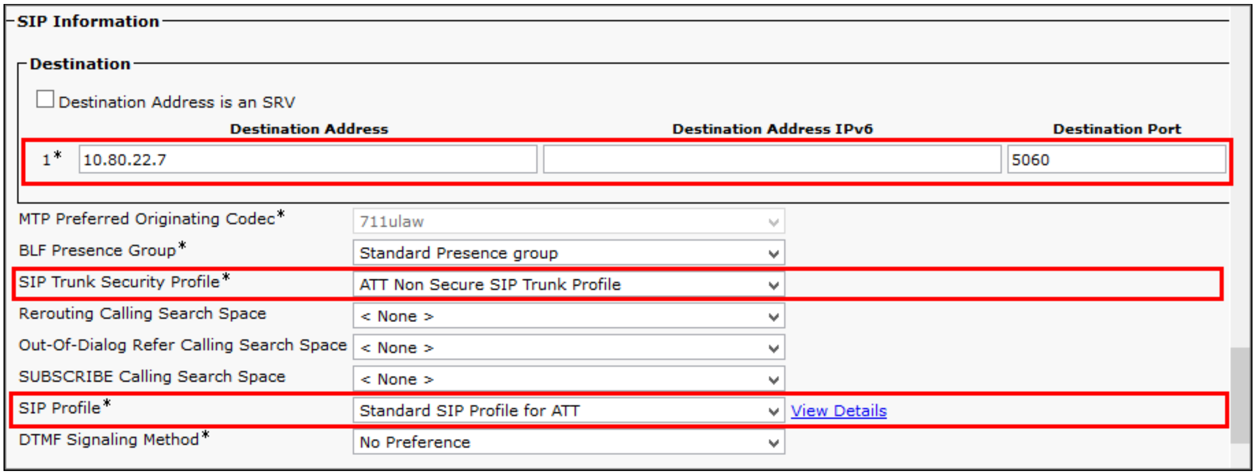
Caller ID DN:

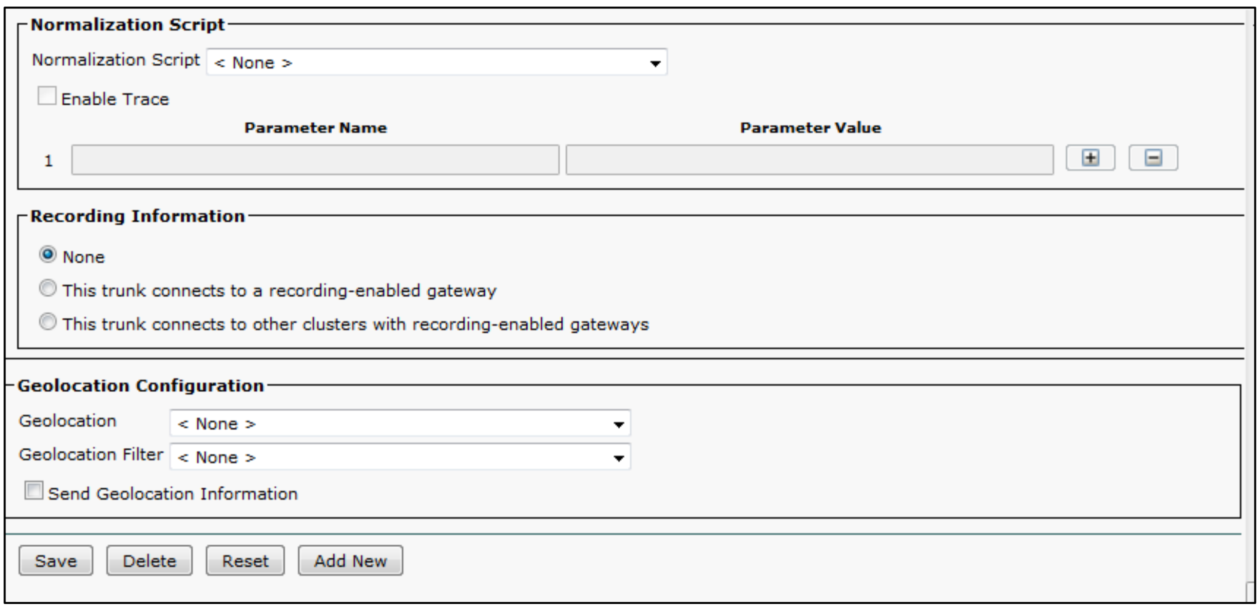
Caller Name:

☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Trunk to Fax Gateway Configuration (Contd.)

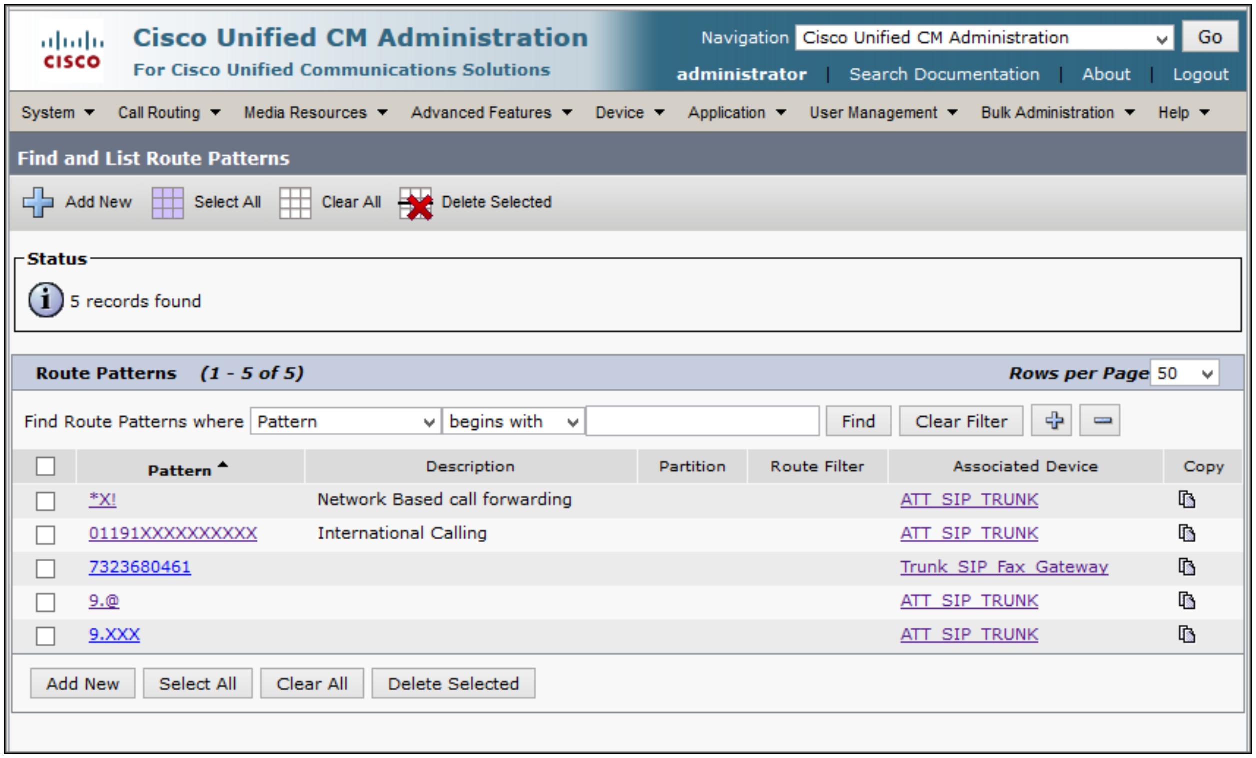
Set Destination Address = Set IP address of Fax Gateway.
Set SIP Trunk Security Profile* = ATT Non Secure Sip Trunk Profile.
Set SIP Profile* = Standard SIP Profile for ATT. This is used in this example.





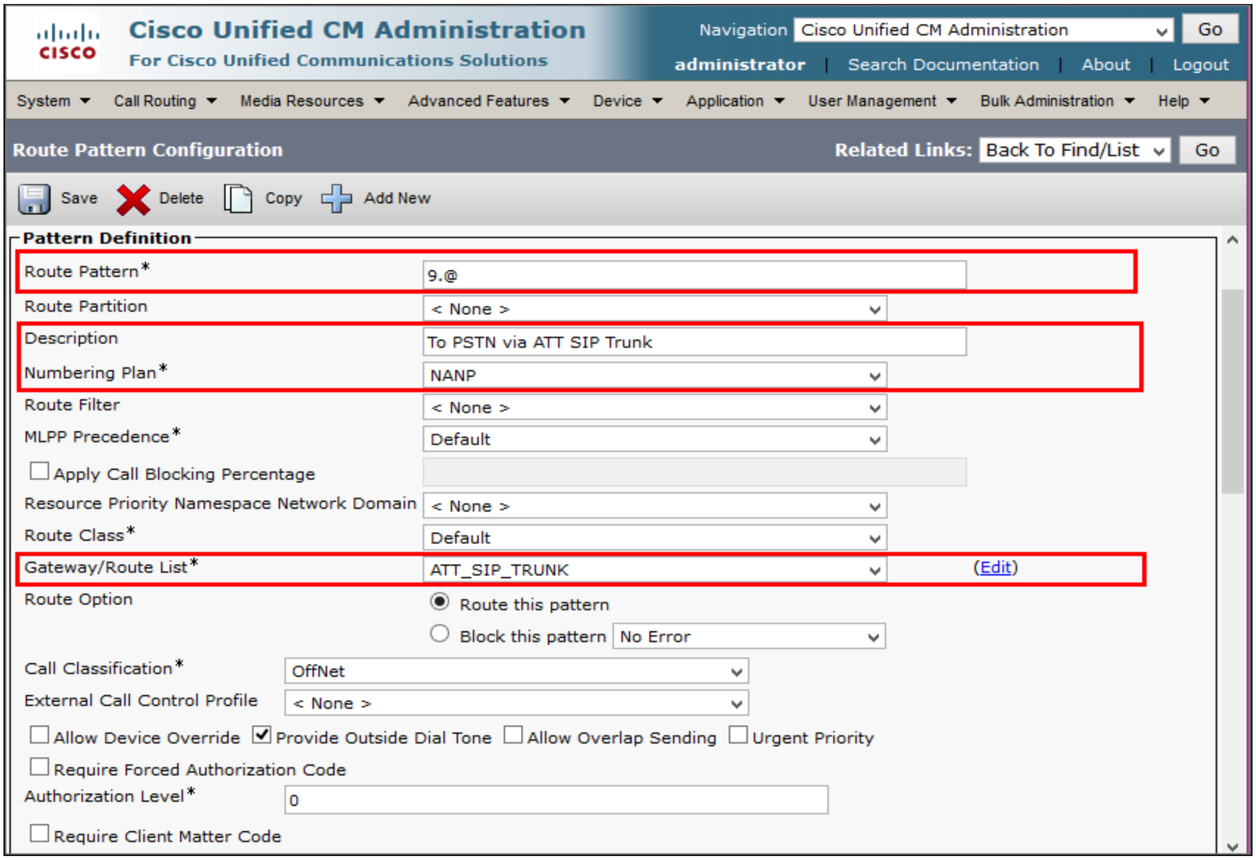
Route Pattern Configuration

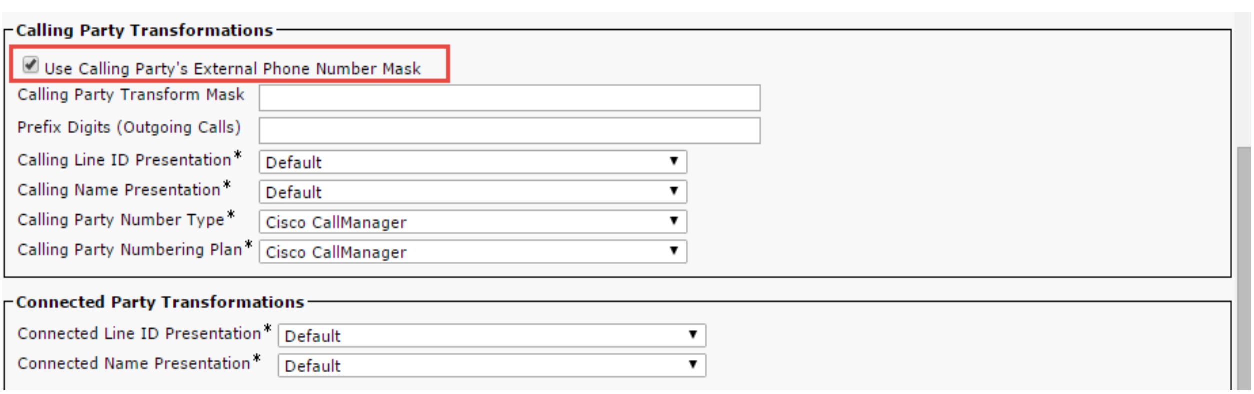
Navigation: Call Routing → Route/Hunt → Route Pattern



Route Pattern Configuration (Contd.)

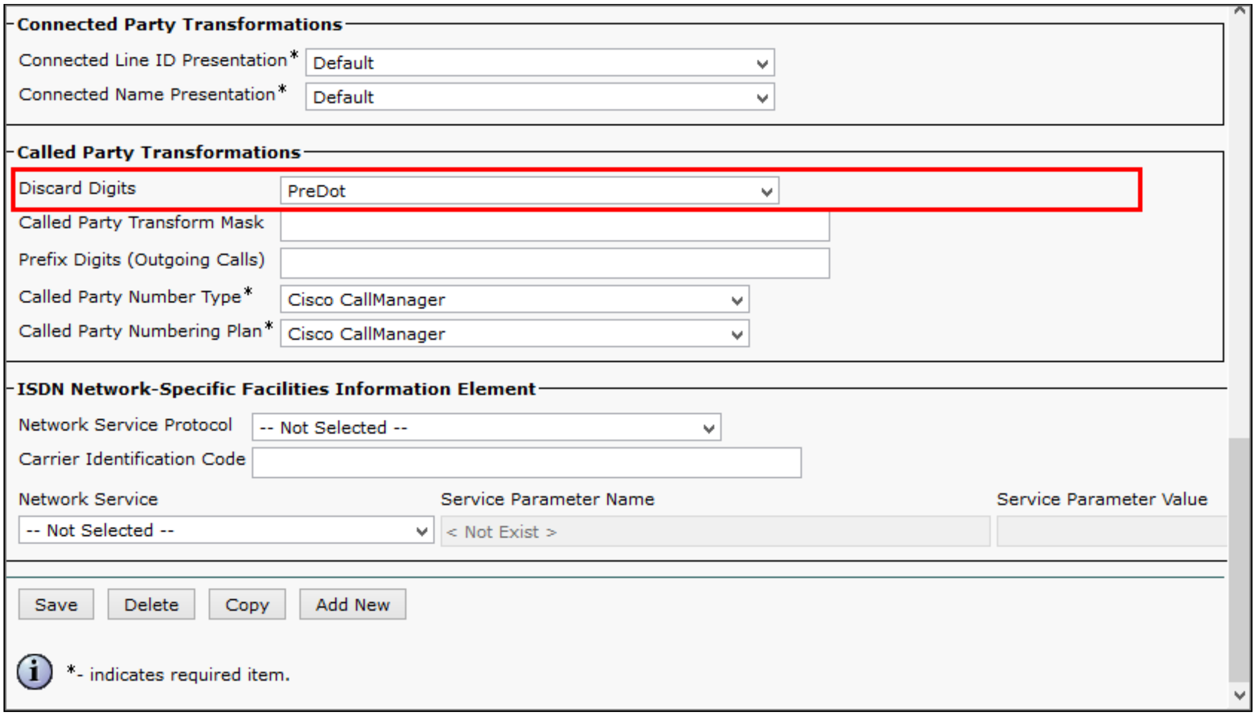
Set Route Pattern* = 9. @   This is used to route to AT&T via ISR Cisco UBE.
Set Description = To PSTN via ATT SIP Trunk. This text is used to identify this Route Pattern.
Set Numbering Plan* = NANP. This text is used to identify the Numbering Plan.
Set Gateway/Route List* = ATT_SIP_TRUNK. This is used for this example.
Check the Use Calling party's External Phone Number Mask check box, for external phone number to be used for calling line identification (CLID) on outgoing calls
All other values are default.





Route Pattern Configuration (Contd.)

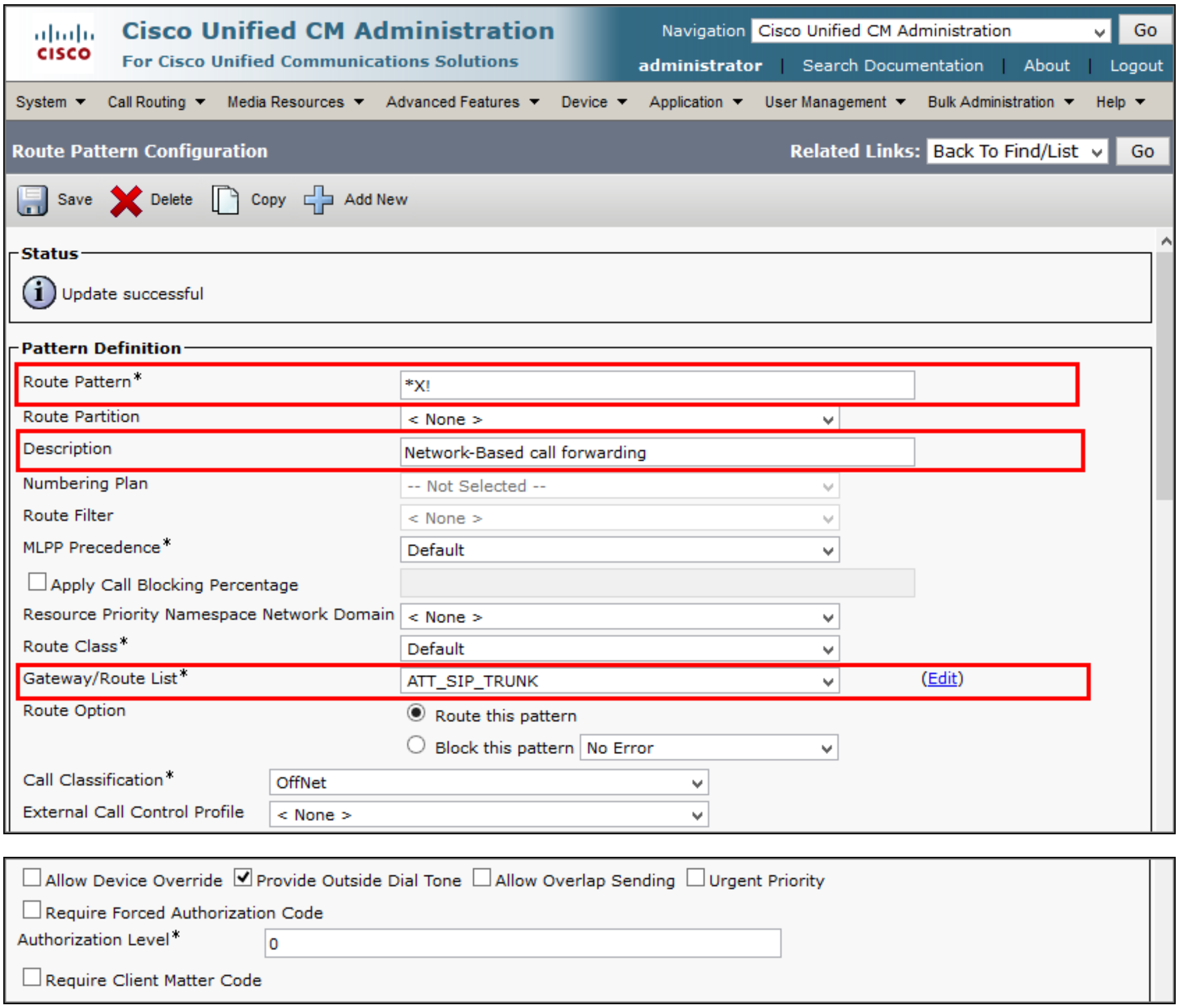
Set Discard Digits = PreDot.



Route Pattern Configuration (Contd.)

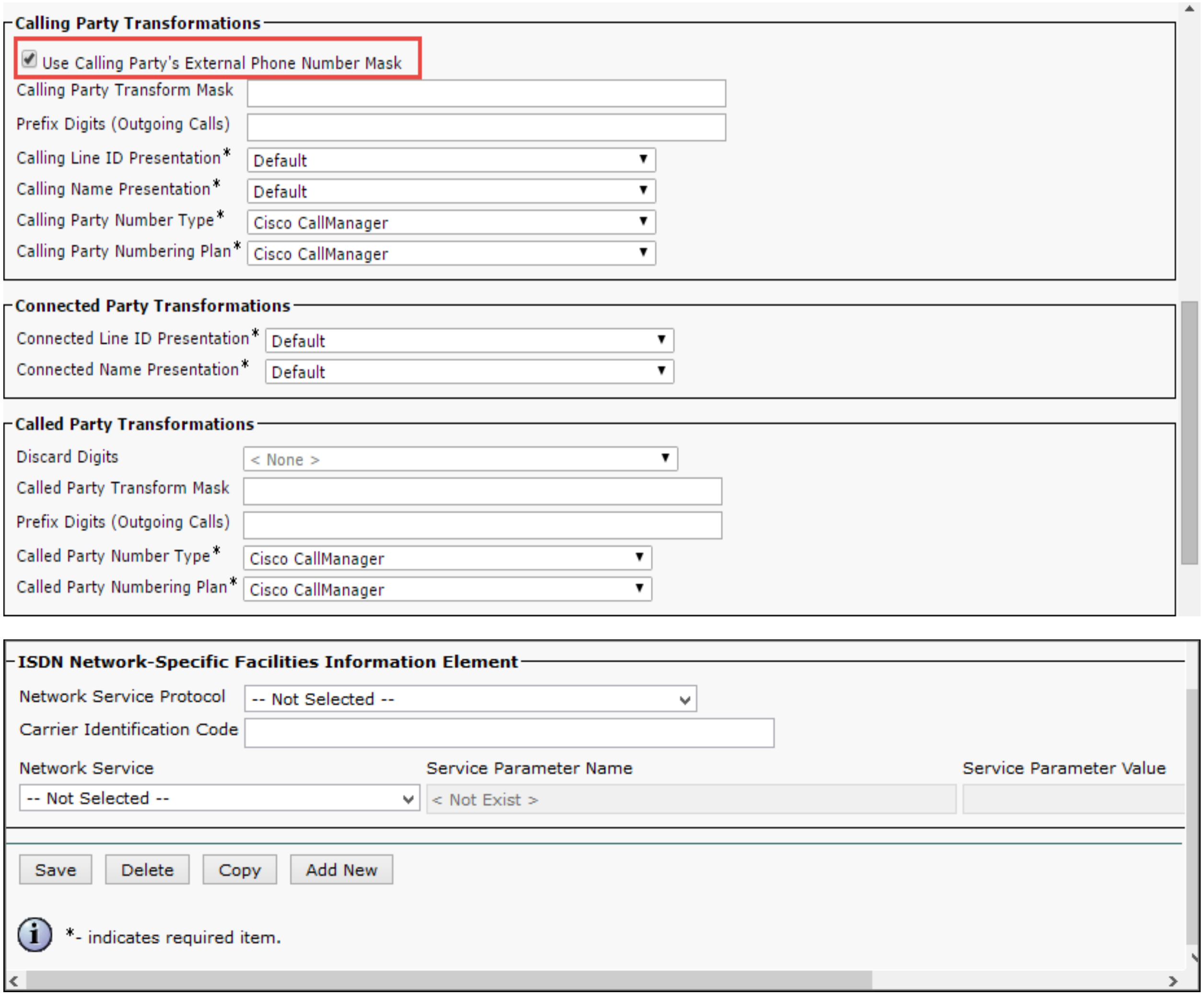
Set Route Pattern* = *X!   This is used to route to AT&T via ISR Cisco UBE.
Set Description = Network-Based Call Forwarding. This text is used to identify this Route Pattern.
Set Gateway/Route List* = ATT_SIP_TRUNK. This is used for this example.
All other values are default.

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



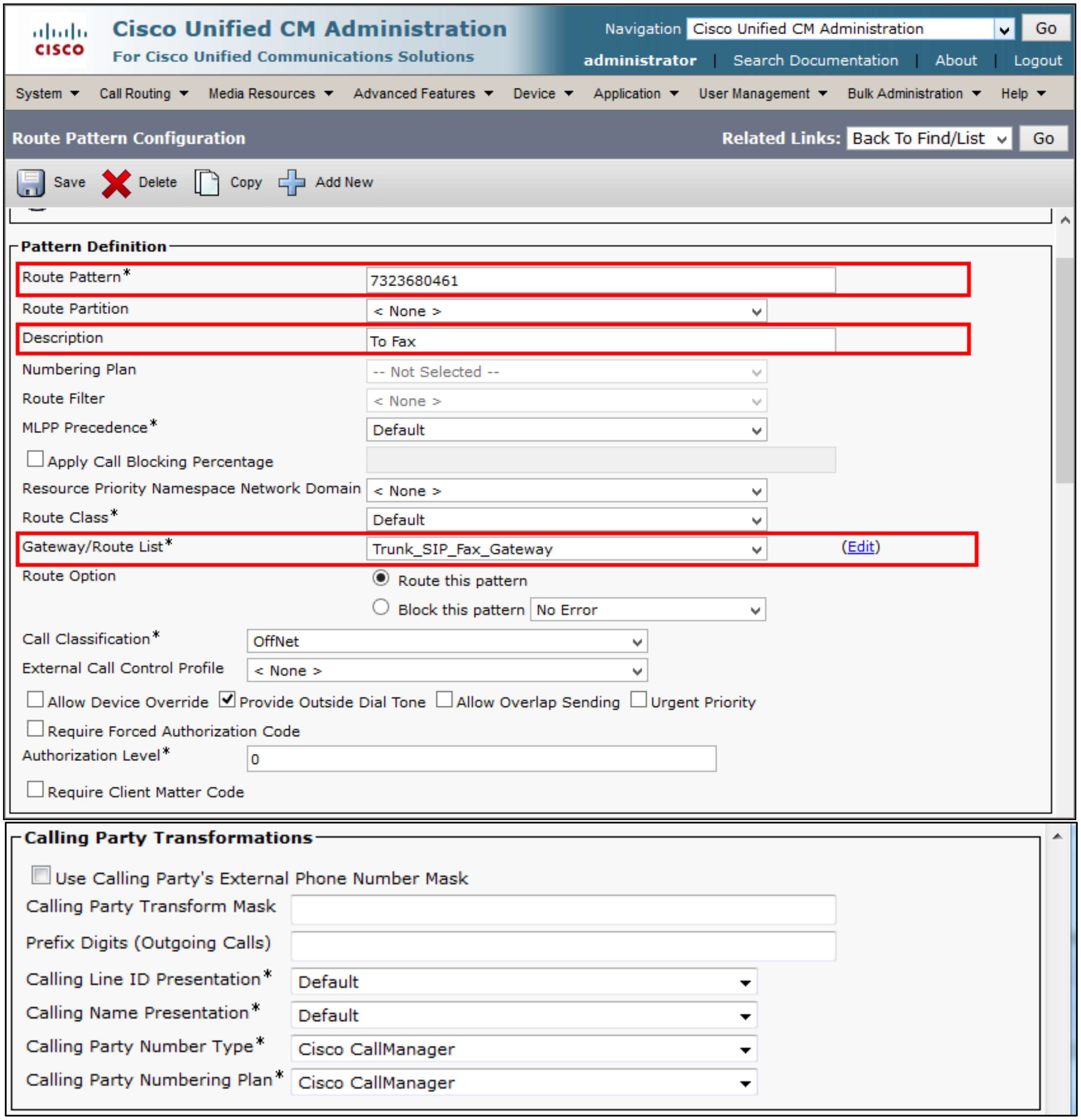
Route Pattern Configuration (Contd.)

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

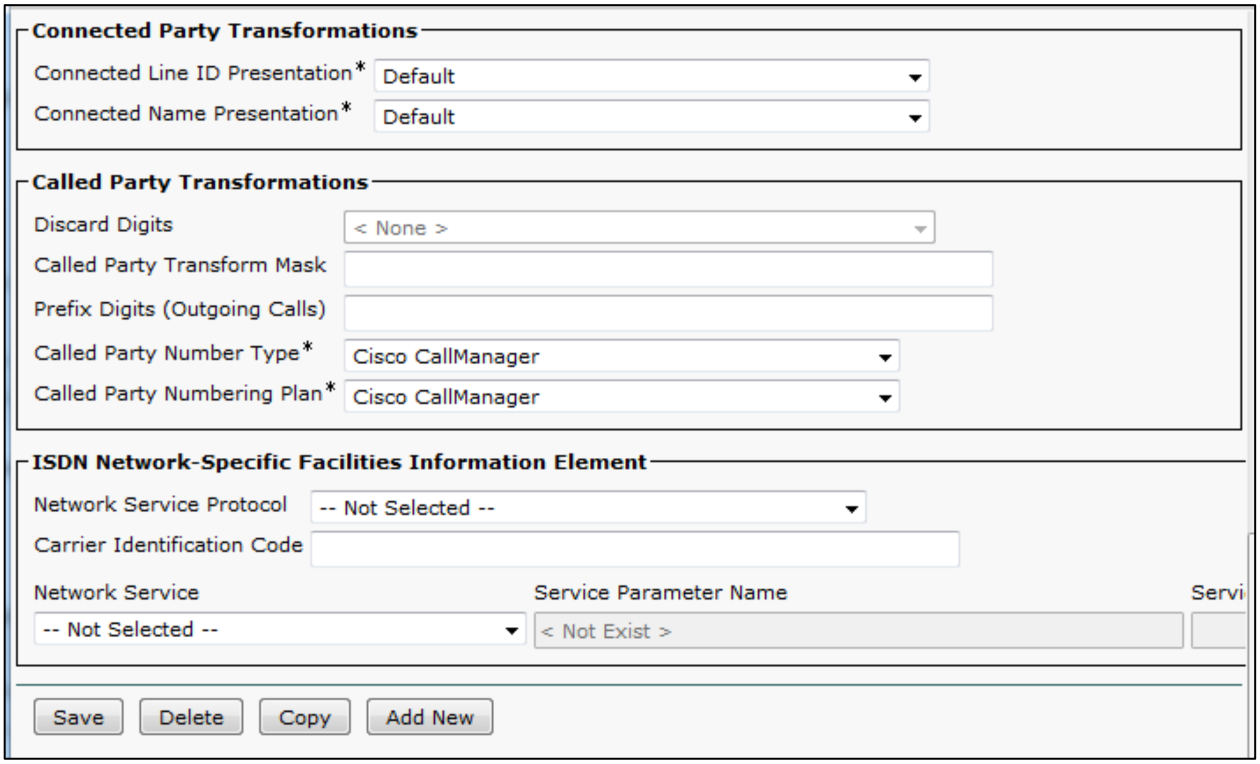


Route Pattern Configuration (Contd.)

Set Route Pattern* = 7323680461. This is used to route to Fax Client via Fax Gateway.
Set Description = To FAX. This text is used to identify this Route Pattern.
Set Gateway/Route List* = Trunk_SIP_FAX_Gateway. This is used for this example.
All other values are default.



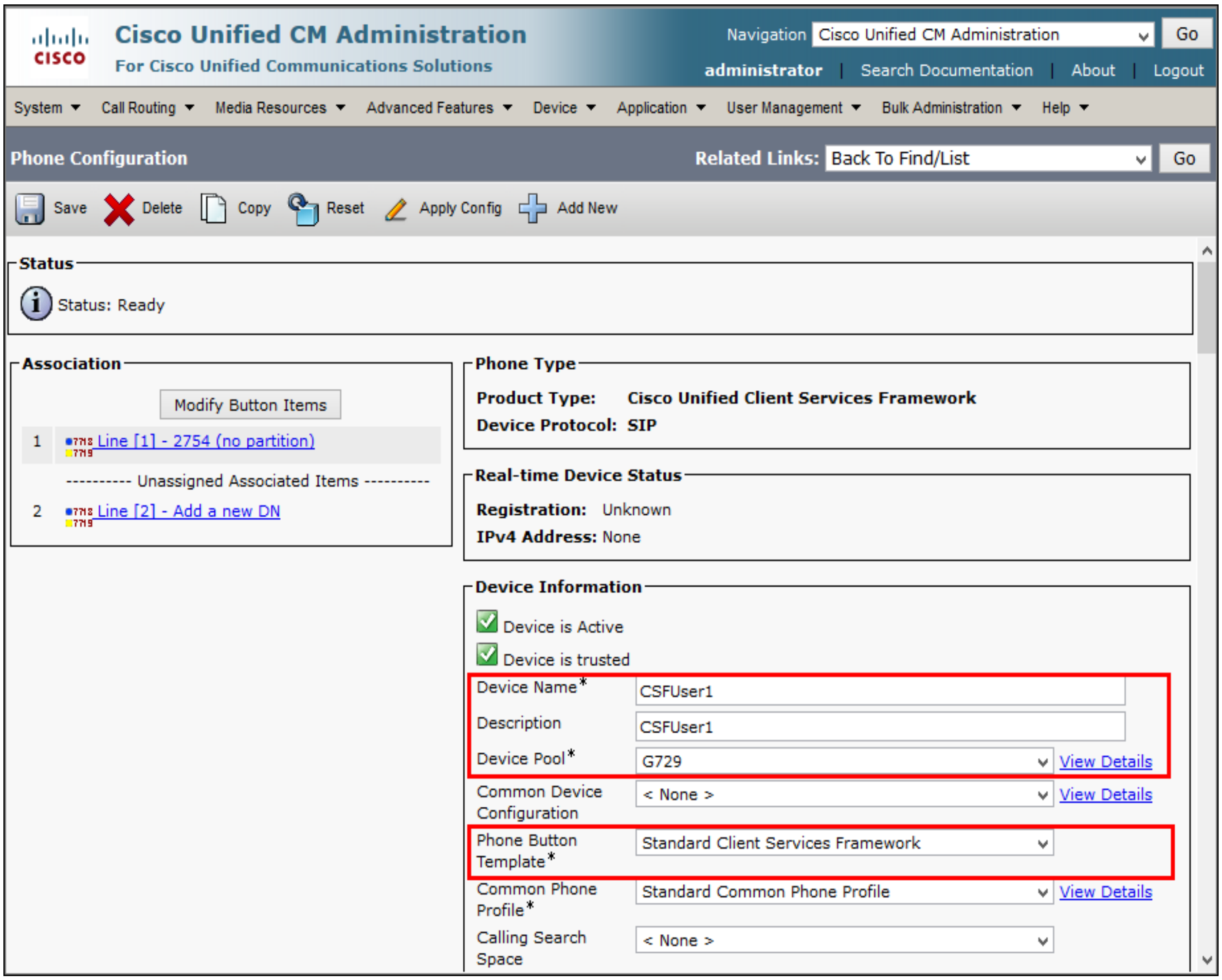
Route Pattern Configuration (Contd.)



Jabber Client Configuration

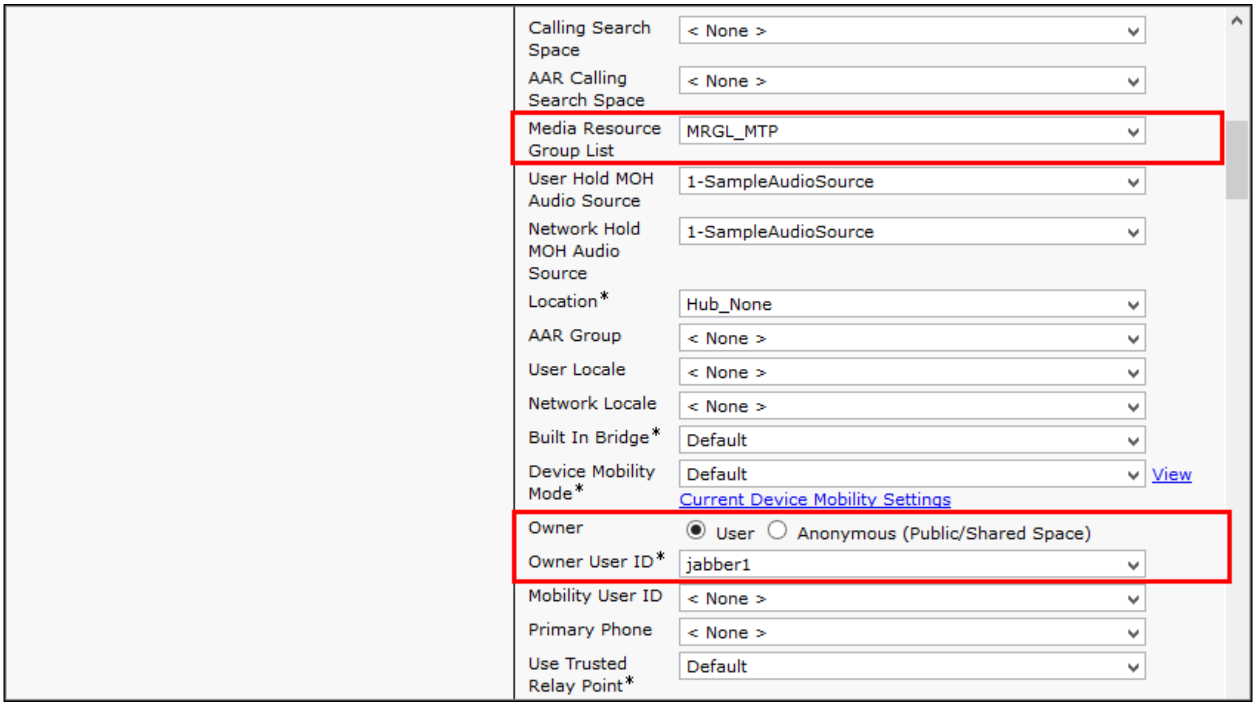
Navigation: Device → Phone

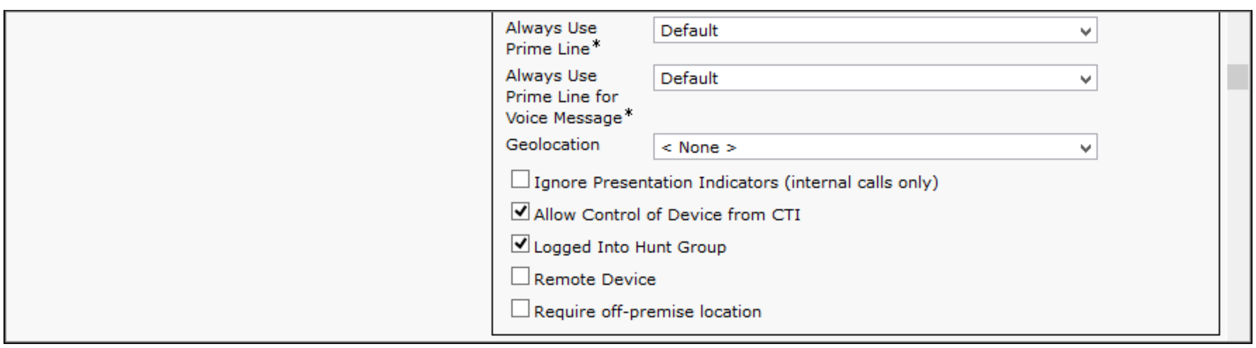
Select Phone Type* = Cisco Unified Client Services Framework
Set Device Name* = CSFUser1. This is used in this example.
Set Description = CSFUser1. This is used in this example.
Select Device Pool = G729. This is used in this example.
Select Phone Button Template* = Standard Client Services Framework.



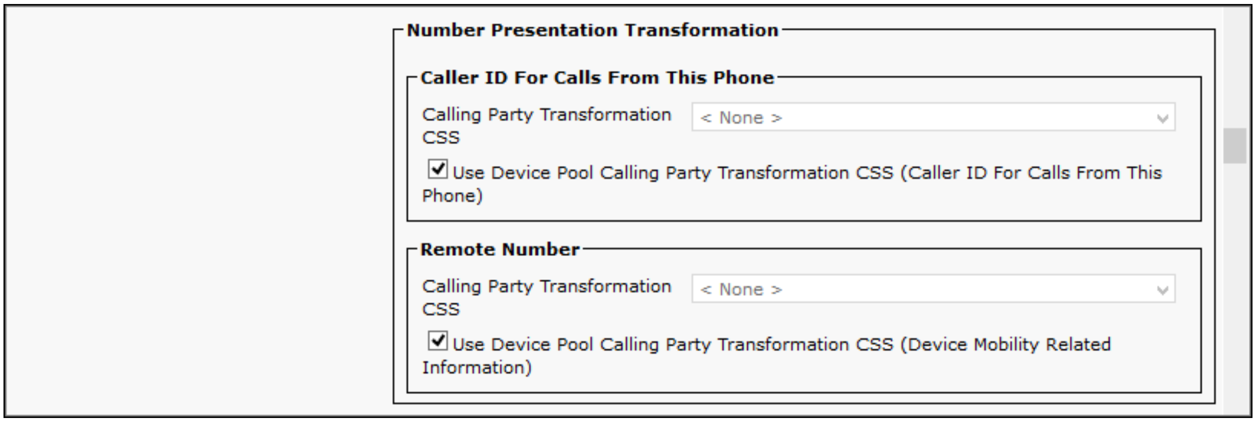
Jabber Client Configuration (Contd.)

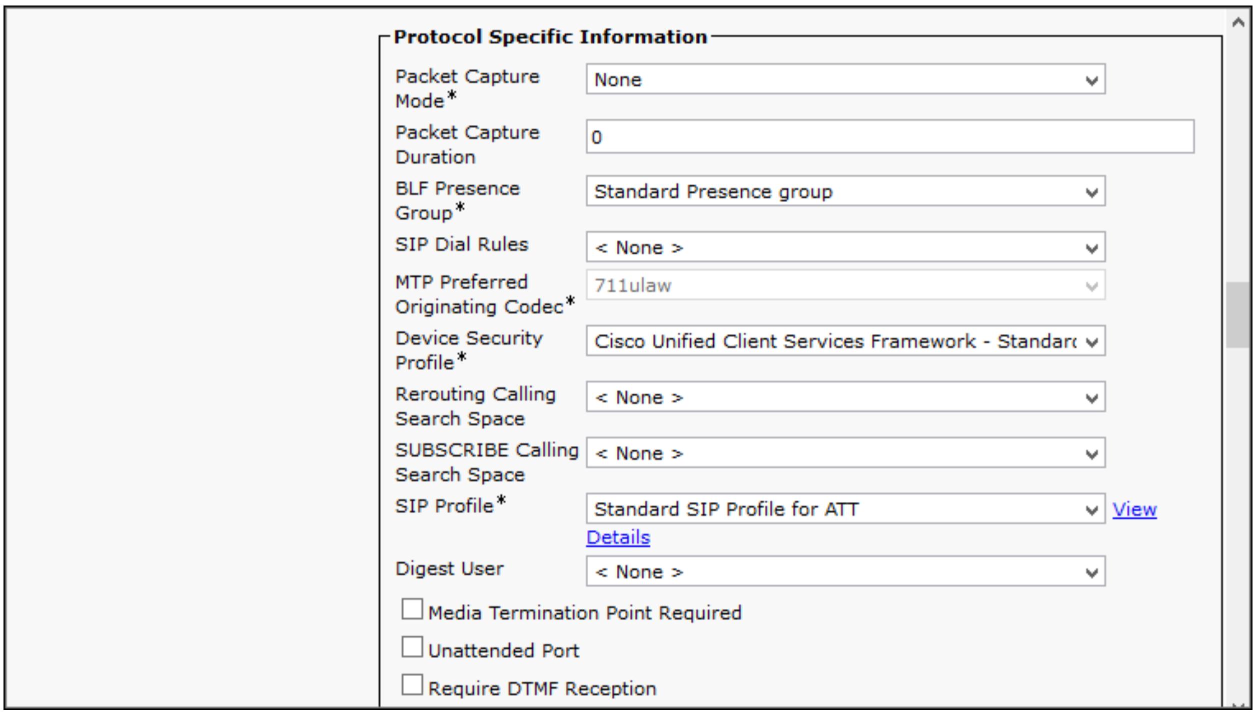
Media Resource Group List = MRGL_MTP
Check Owner = User. This is used in this example.
Set Owner user ID* = jabber1. This is used for this example



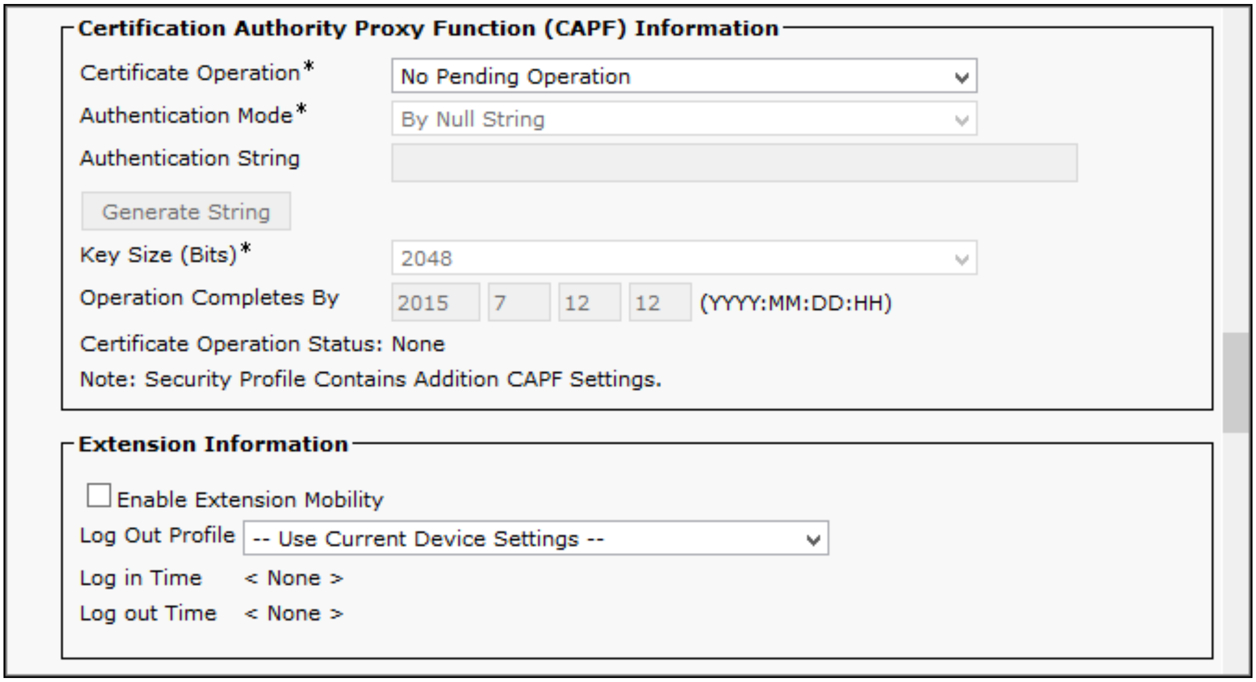


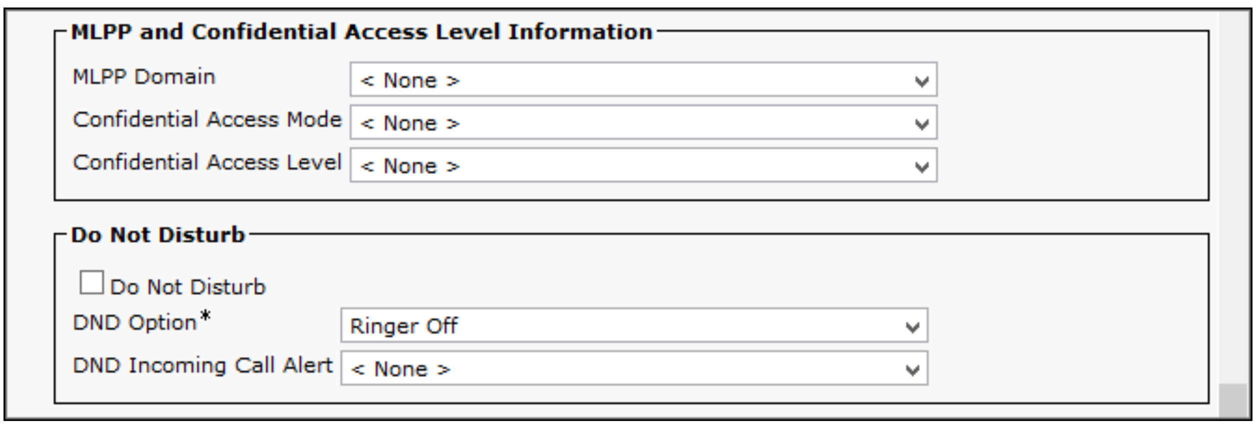
Jabber Client Configuration (Contd.)



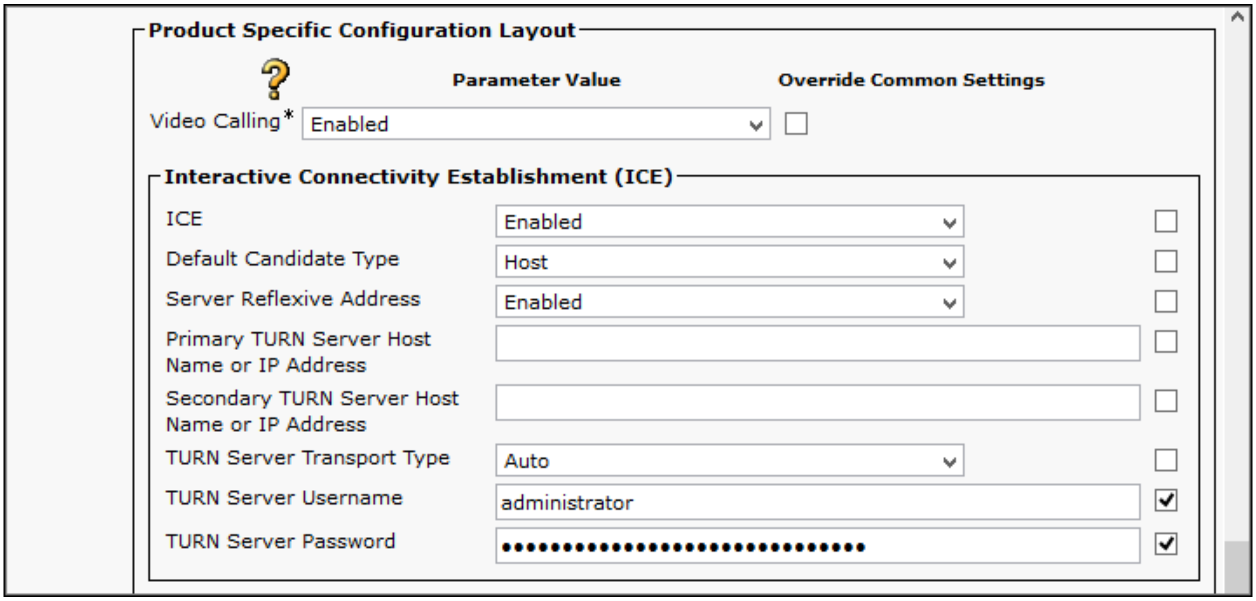


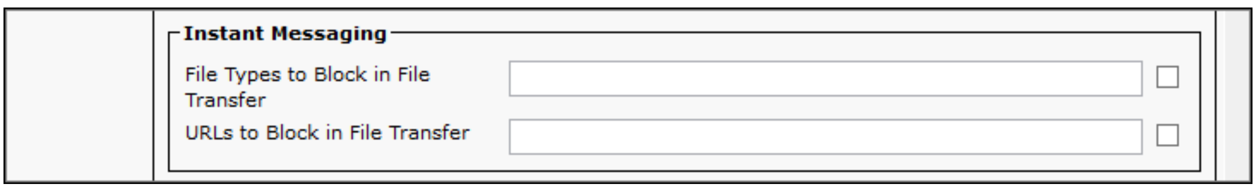
Jabber Client Configuration (Contd.)



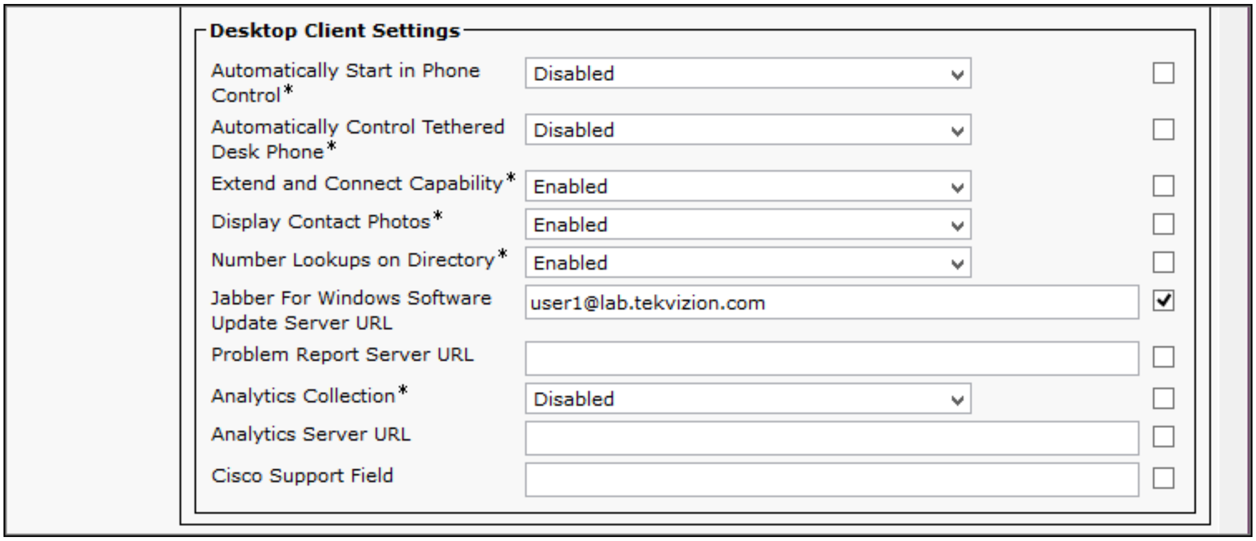


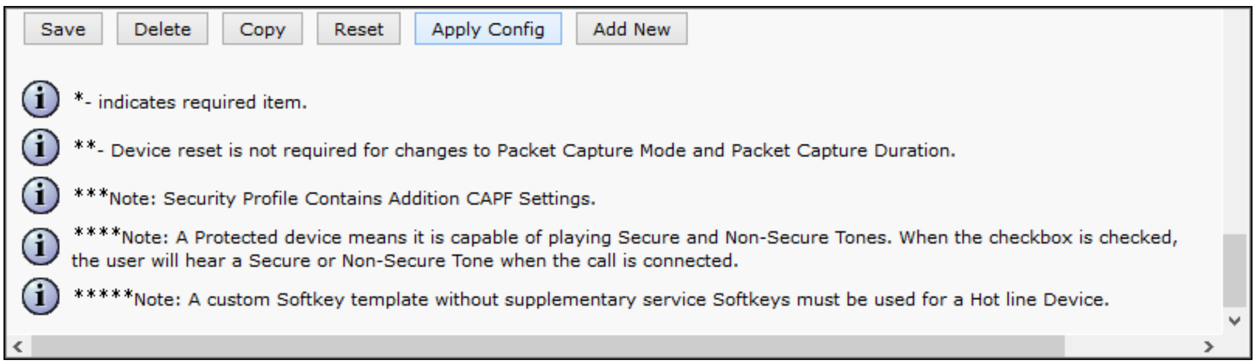
Jabber Client Configuration (Contd.)





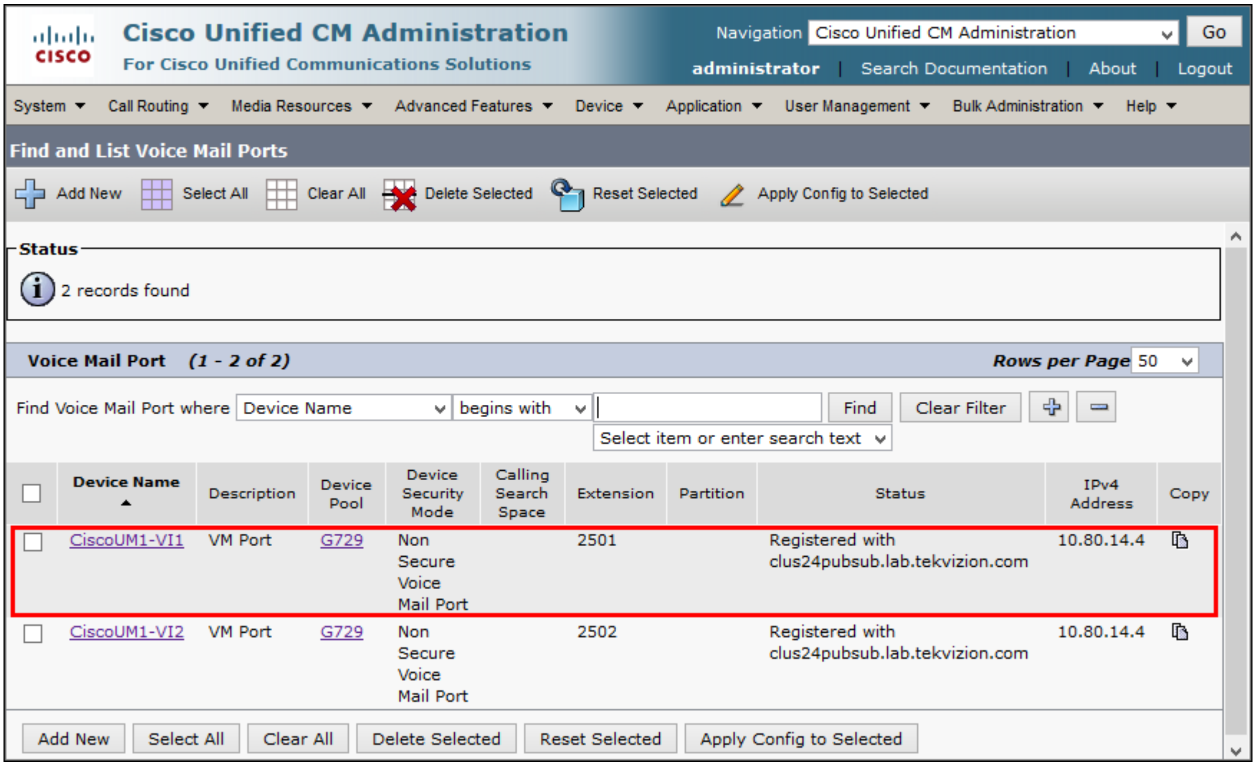
Jabber Client Configuration (Contd.)





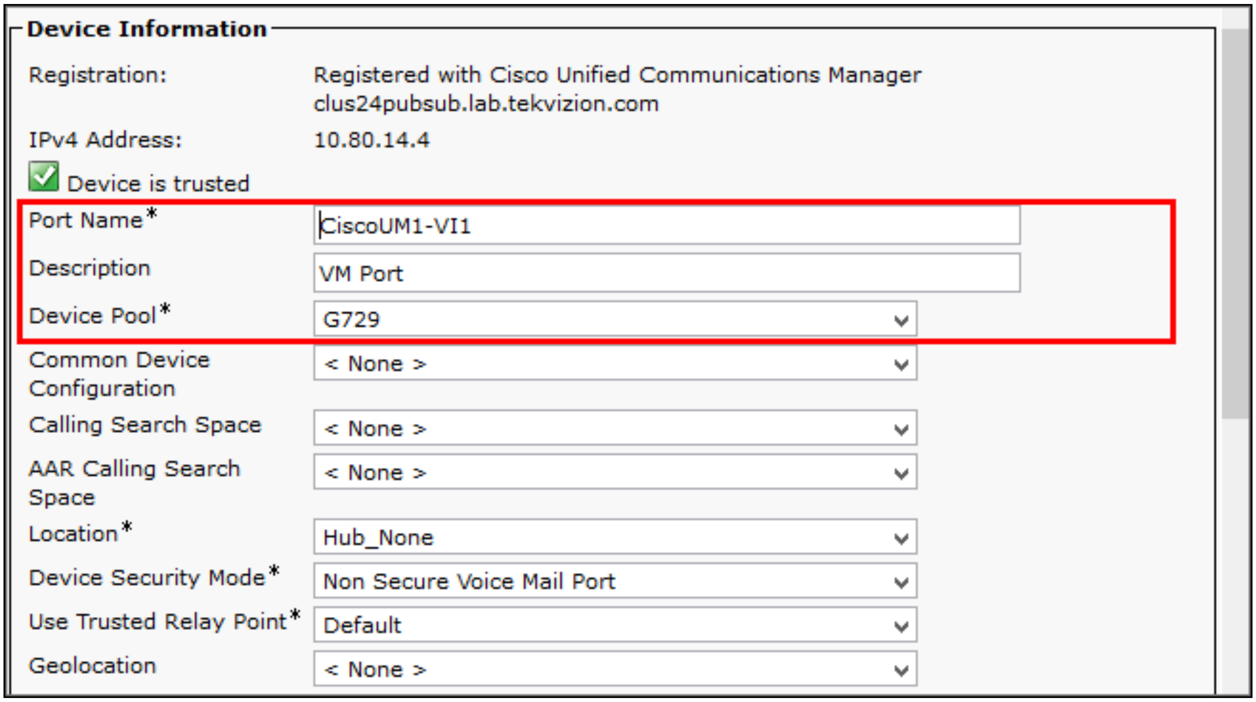
Voicemail Port Configuration

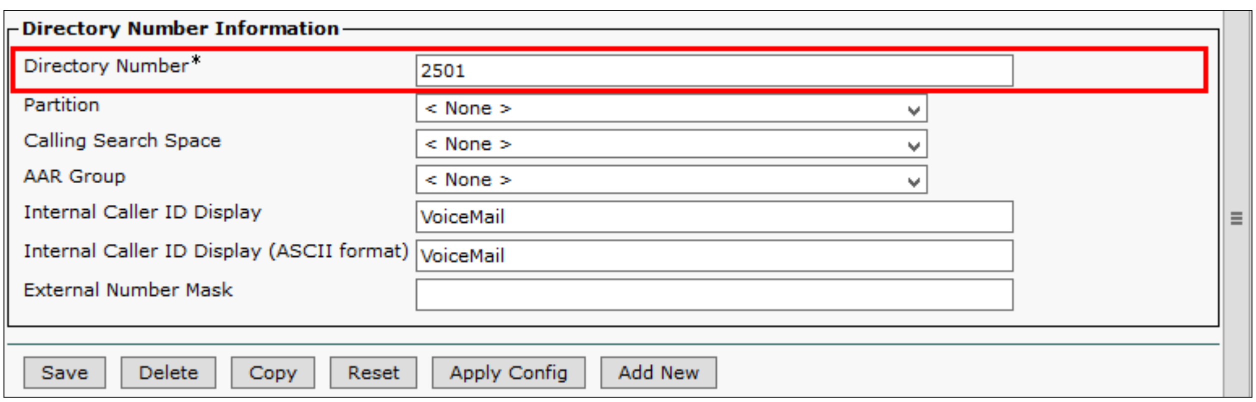
Navigation: Advanced Feature→ Voice Mail → Cisco Voice Mail Port



Voicemail Port Configuration (Contd.)

Set Port Name = CiscoUM1-VI1. This is used for this example.
Set Description = VM Port. This is used for this example.
Set Device Pool = G729
Set Directory Number* = 2501. This is used in this example.

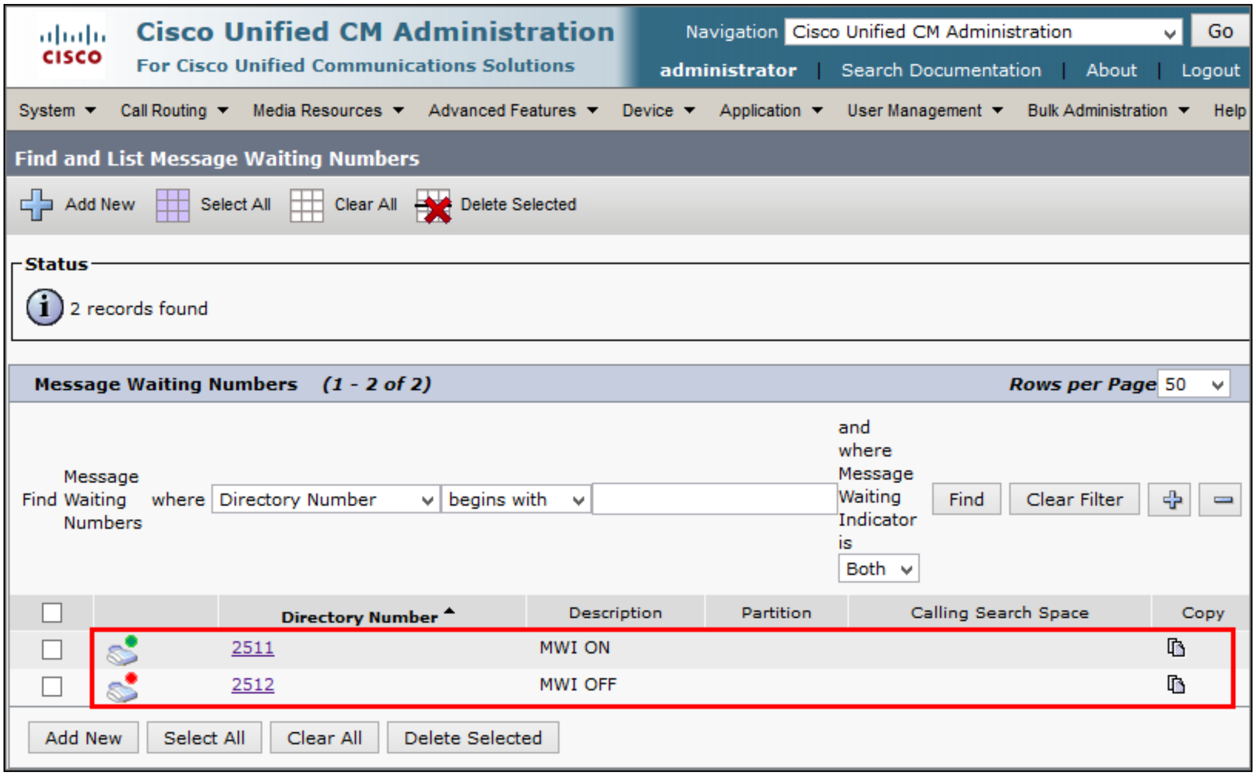




Message Waiting Numbers Configurations

Navigation: Advanced Features → Voice Mail→Message Waiting

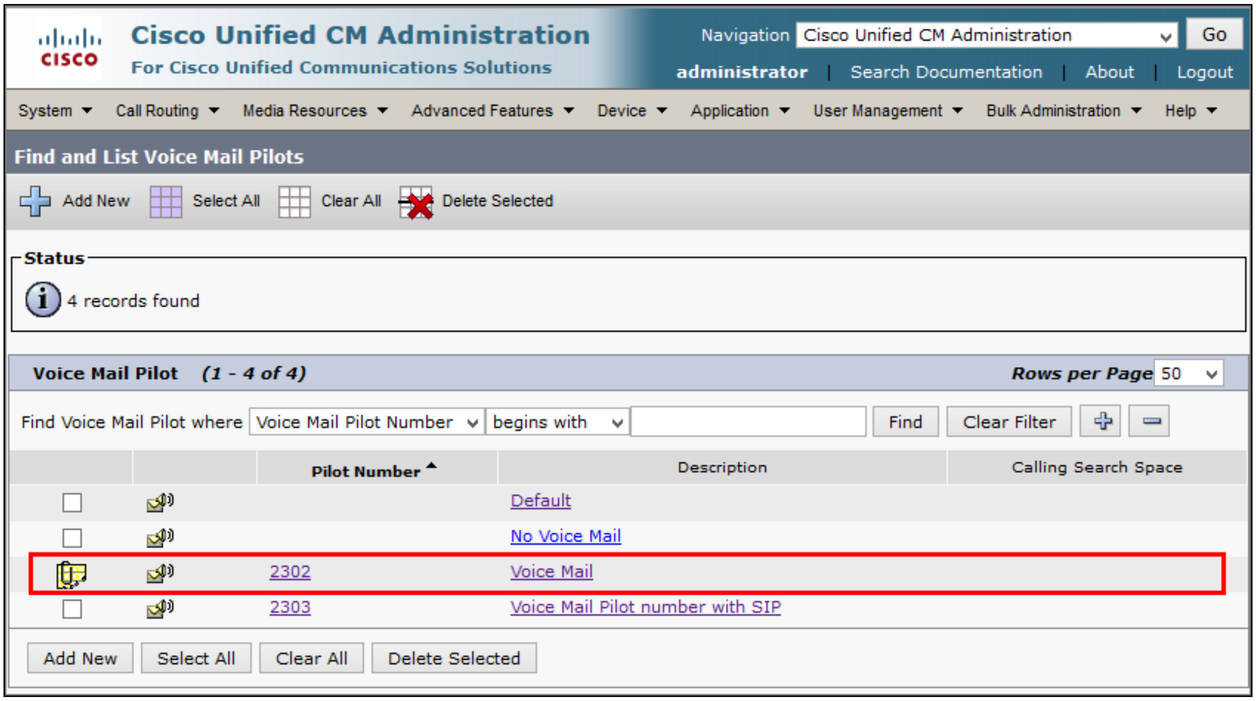
Set Message Waiting Number* = 2511
Set Message Waiting Indicator* = On
Set Message Waiting Number* = 2512
Set Message Waiting Indicator* = Off

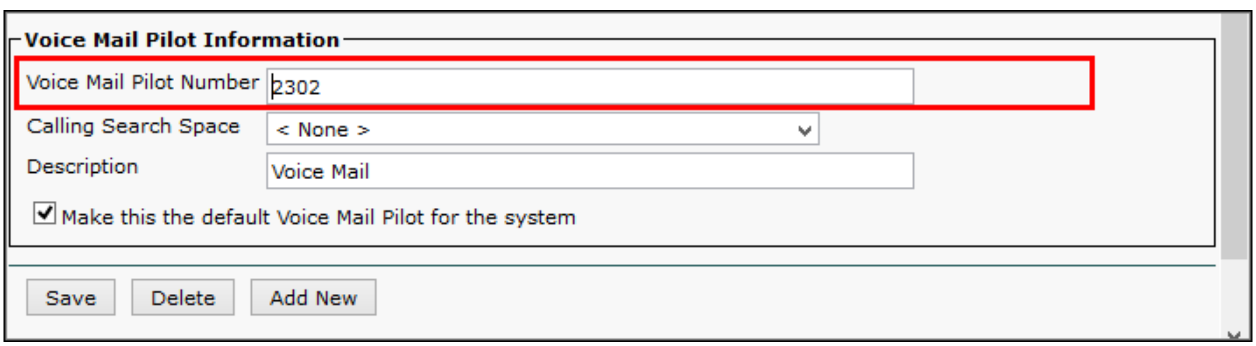


Voicemail Pilot Configuration

Navigation: Advanced Features → Voice Mail → Voice Mail Pilot

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





FAX Gateway Configuration

FAX-GATEWAY2# **show version**

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

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

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

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

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

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

System returned to ROM by power-on

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

Last reload type: Normal Reload

Last reload reason: power-on

This product contains cryptographic features and is subject to United

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

use. Delivery of Cisco cryptographic products does not imply

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

Importers, exporters, distributors and users are responsible for

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

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

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

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

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

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

export@cisco.com.

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

Processor board ID FTX174081SJ

2 Gigabit Ethernet interfaces

1 terminal line

2 Voice FXS interfaces

DRAM configuration is 64 bits wide with parity enabled.

255K bytes of non-volatile configuration memory.

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

License Info:

License UDI:

Device#   PID   SN

----

*1   CISCO2901/K9   FTX174081SJ

Technology Package License Information for Module:'c2900'

Technology   Technology-package   Technology-package

           Current   Type   Next reboot

----

ipbase   ipbasek9   Permanent   ipbasek9

security   None   None   None

uc   uck9   Permanent   uck9

data   None   None   None

NtwkEss   None   None   None

CollabPro   None   None   None

Configuration register is 0x2102

FAX-GATEWAY2# **sh running-config**

Building configuration...

Current configuration : 8536 bytes

!

! Last configuration change at 11:58:36 UTC Wed Jul 1 2015 by cisco

!

version 15.4

```
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname FAX-GATEWAY2
!
boot-start-marker
boot-end-marker
!
aqm-register-fnf
!
logging queue-limit 1000000000
logging buffered 10000000
logging rate-limit 10000
no logging console
no logging monitor
enable secret 4 iR3uUX3Bo6oYbT6ajhFwJe39FR4g.1QCmm7yYduKGZI
!
no aaa new-model
!
!
!
!
!
!
!
```

!

!

!

!

!

!

!

ip domain name lab.tekvizion.com

ip name-server 10.64.1.3

ip cef

no ipv6 cef

multilink bundle-name authenticated

!

!

!

!

stcapp feature access-code

!

stcapp feature speed-dial

!

!

!

!

!

cts logging verbose

!

```
crypto pki trustpoint TP-self-signed-2189441908
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-2189441908
 revocation-check none
 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
 no ip address trusted authenticate
 address-hiding
 mode border-element
 allow-connections sip to sip
 no supplementary-service sip handle-replaces
 redirect ip2ip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  midcall-signaling passthru
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
!
voice class sip-profiles 1
 response ANY sip-header Allow-Header modify "UPDATE," ""
```


Note: Testing was conducted in tekVizion labs

```
 request 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
 no vad
 shutdown
!
voice-port 0/1
 no echo-cancel enable
 no vad
 cptone IN
 station-id name fax test
 station-id number 7323680461
 caller-id enable
 !
 !
 !
 !
!
no mgcp timer receive-rtcp
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
```

```
!
!
ccm-manager music-on-hold
!
no ccm-manager fax protocol cisco
!
dial-peer voice 110 pots
 service session
 destination-pattern 7323680461
 no digit-strip
 port 0/1
 forward-digits all
!
dial-peer voice 200 voip
 description CUCM to Gateway
 service session
 session protocol sipv2
 session transport udp
 incoming called-number 7323680461
 voice-class codec 1
 voice-class sip profiles 1
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
```

```
dial-peer voice 201 voip
 description Gateway to CUCM
 service session
 destination-pattern 9T
 session protocol sipv2
 session target ipv4:10.80.14.2
 session transport udp
 voice-class codec 1
 voice-class sip profiles 1
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
!
gateway
 timer receive-rtp 1200
!
sip-ua
!
!
!
gatekeeper
 shutdown
!
!
```

banner exec ^C

% Password expiration warning.

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

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

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

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

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

this session.

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

of 15 using the following command.

username <myuser> privilege 15 secret 0 <mypassword>

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

use.

----

^C

banner login ^C

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

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

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

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

Here are the Cisco IOS commands.

username <myuser> privilege 15 secret 0 <mypassword>

no username cisco

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

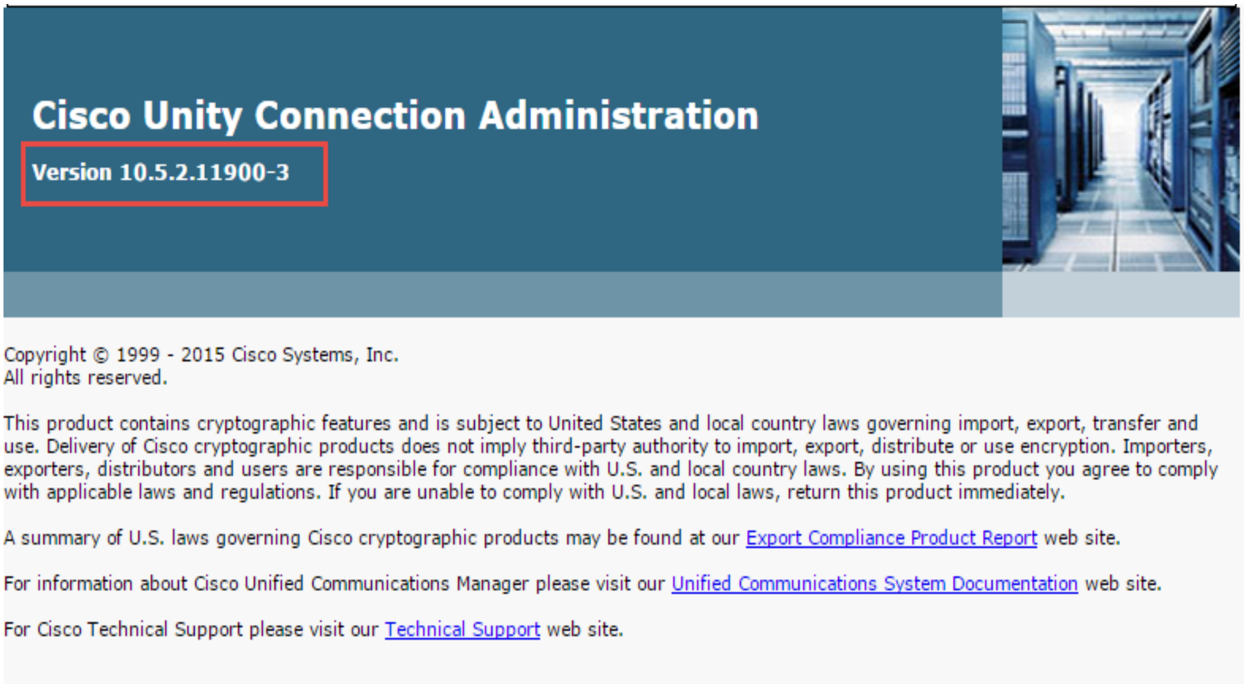
IF YOU DO NOT CHANGE THE PUBLICLY-KNOWN CREDENTIALS, YOU WILL NOT BE ABLE TO LOG INTO THE DEVICE AGAIN AFTER YOU HAVE LOGGED OFF.

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

```
-----------------------------------------------------------------------
^C
!
line con 0
 login local
line aux 0
line 2
 no activation-character
 no exec
```

```
 transport preferred none
 transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 login local
 transport input telnet ssh
line vty 5 15
 login local
 transport input telnet ssh
!
scheduler allocate 20000 1000
!
end
```

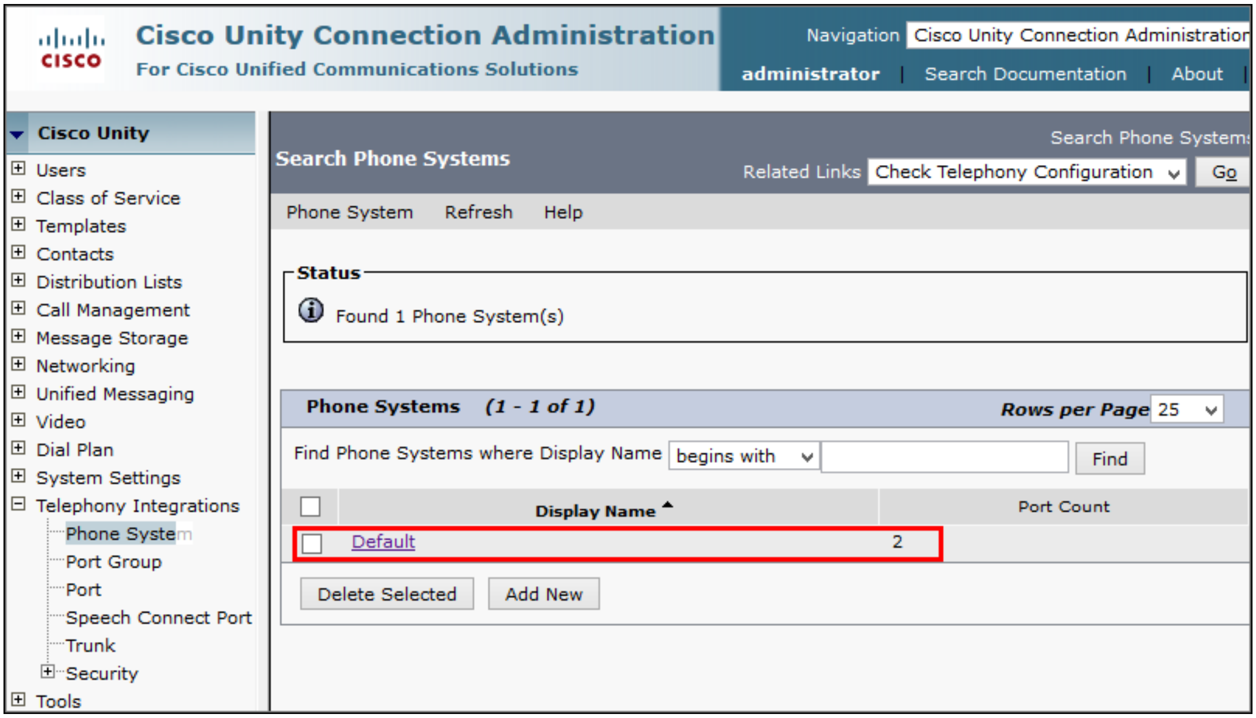
CUC Version


Cisco Unity Connection Administration
Version 10.5.2.11900-3
Copyright © 1999 - 2015 Cisco Systems, Inc.
All rights reserved.
This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.
A summary of U.S. laws governing Cisco cryptographic products may be found at our Export Compliance Product Report web site.
For information about Cisco Unified Communications Manager please visit our Unified Communications System Documentation web site.
For Cisco Technical Support please visit our Technical Support web site.

CUC Telephony Integration with Cisco UCM

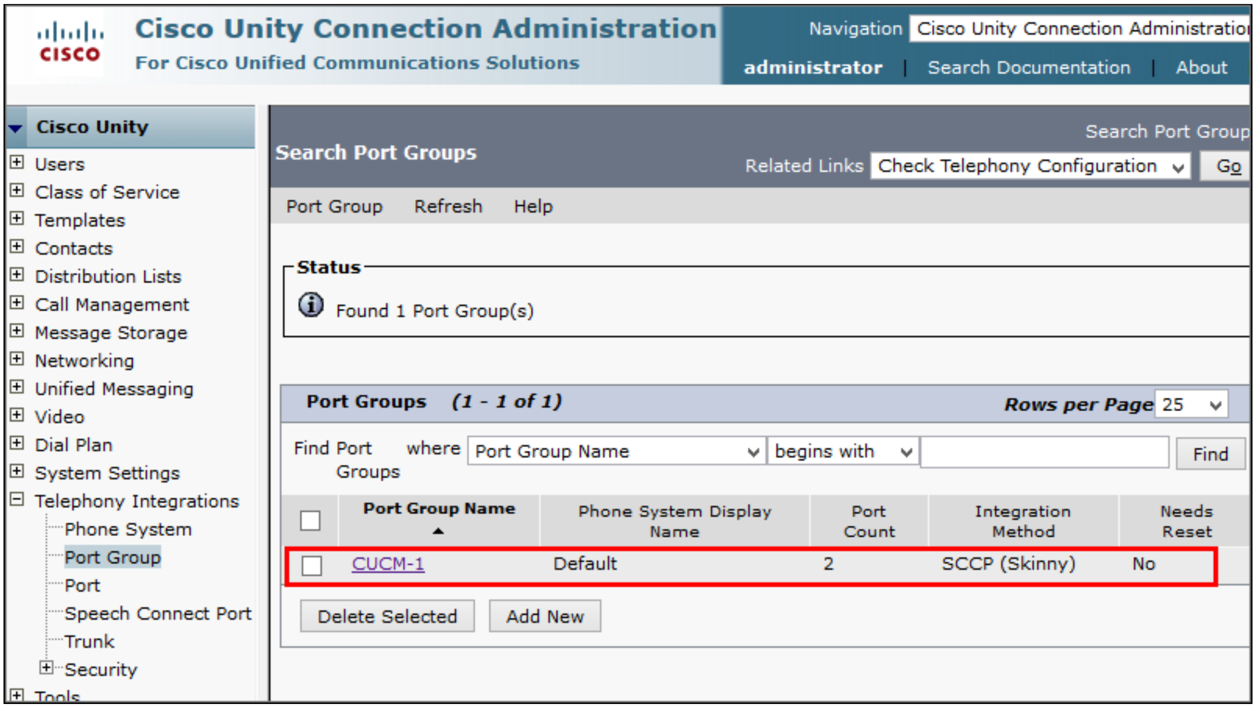
Navigation: Telephony Integrations → Phone system

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



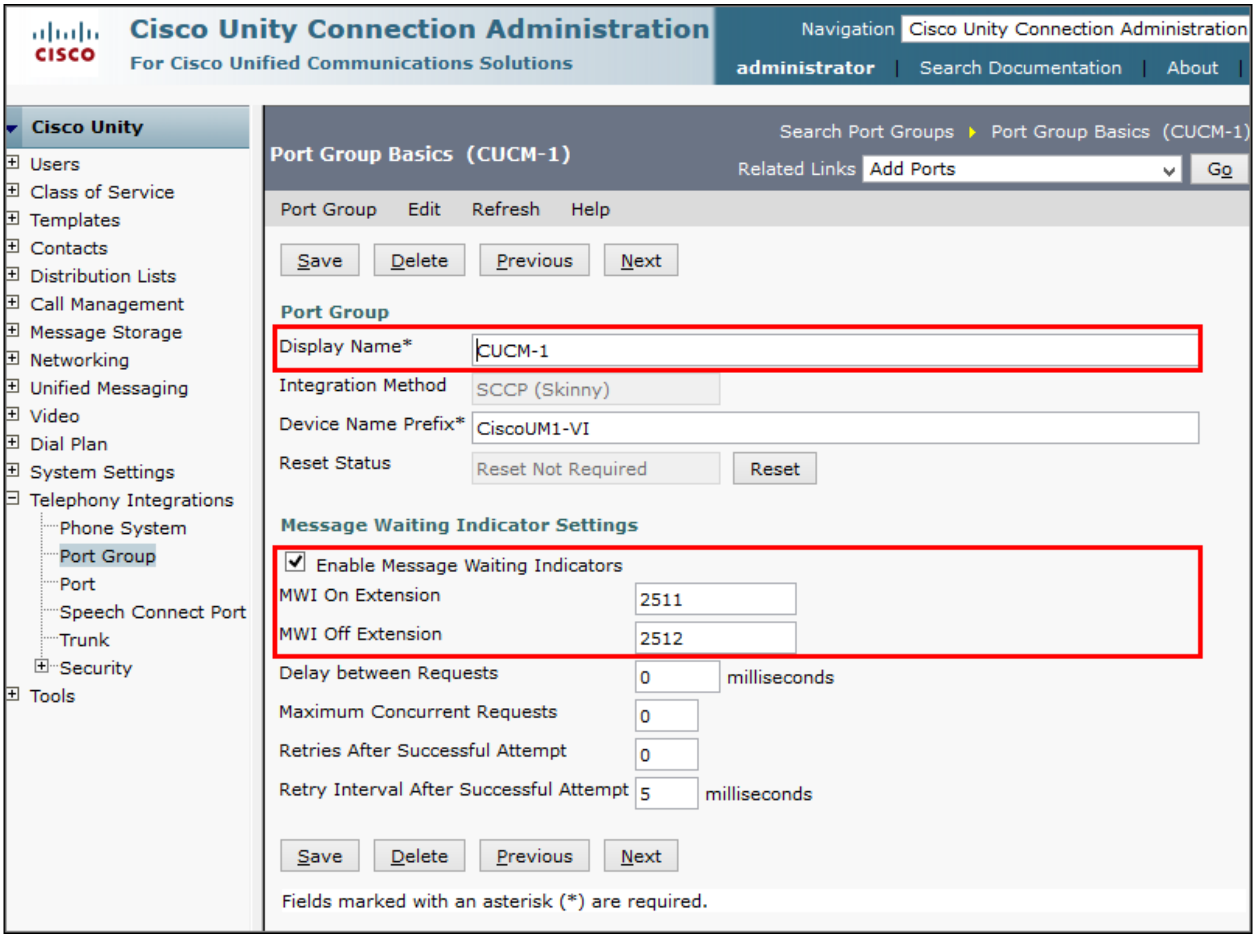
CUC Port Group

Navigation: Telephony Integration → Port Group



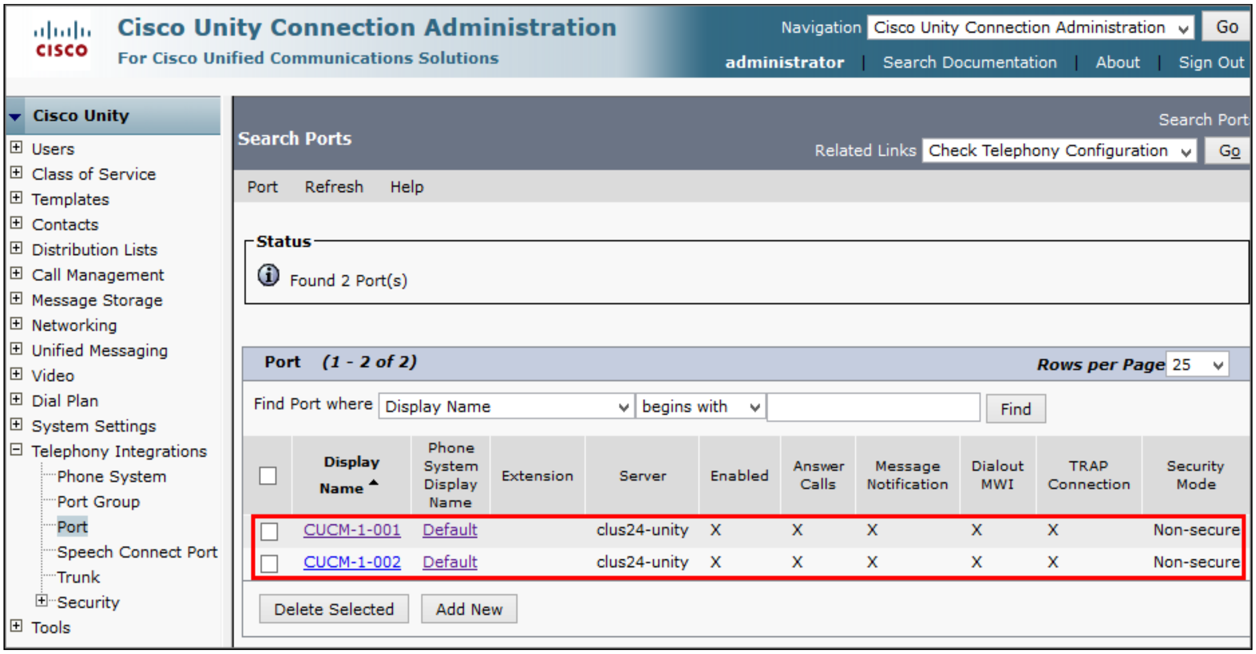
CUC Port Group (Contd.)

Set Display Name* = CUCM-1. This is used in this example.
Check Enable Message waiting indicators.
Set MWI On Extension = 2511. This is used in this example.
Set MWI Off Extension= 2512. This is used in this example.



CUC Port Settings

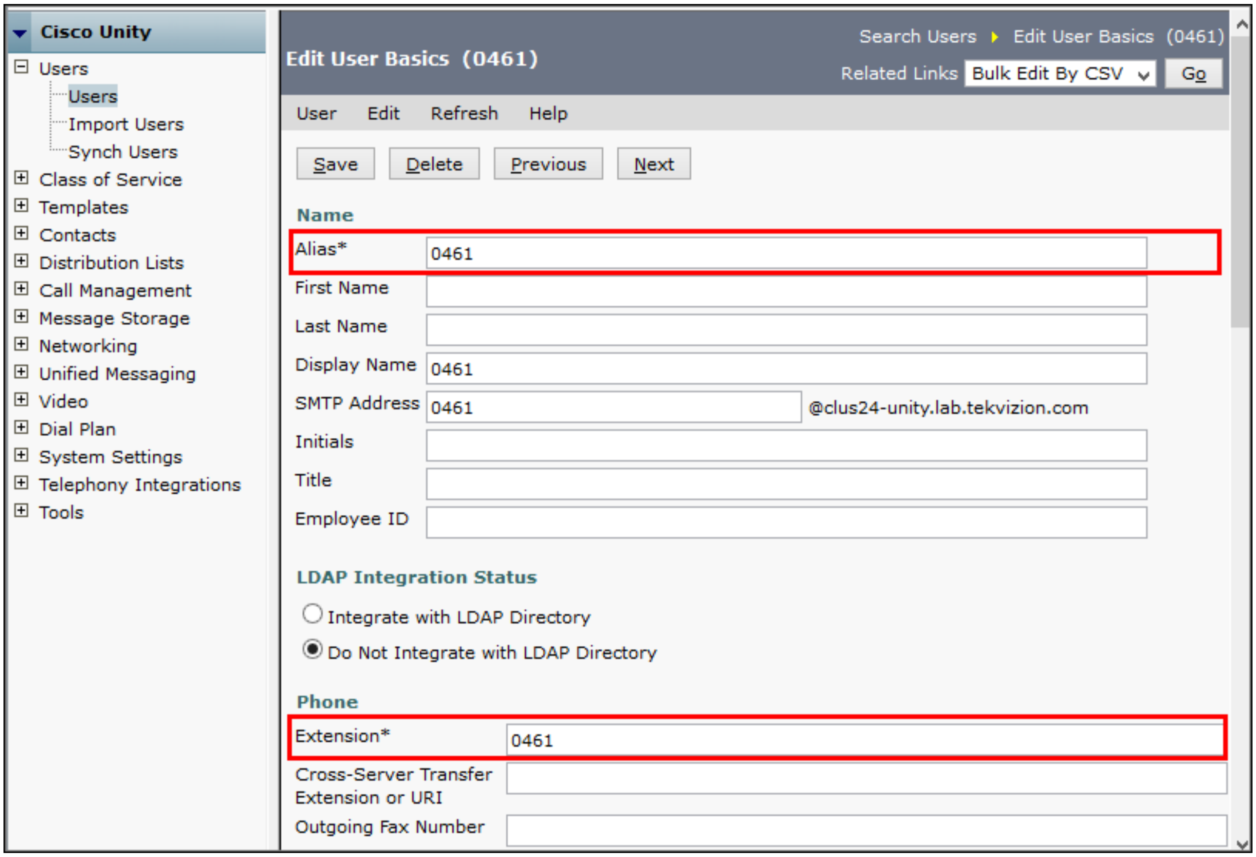
Navigation: Telephony Integration → Port



CUC Sample User Basic Settings

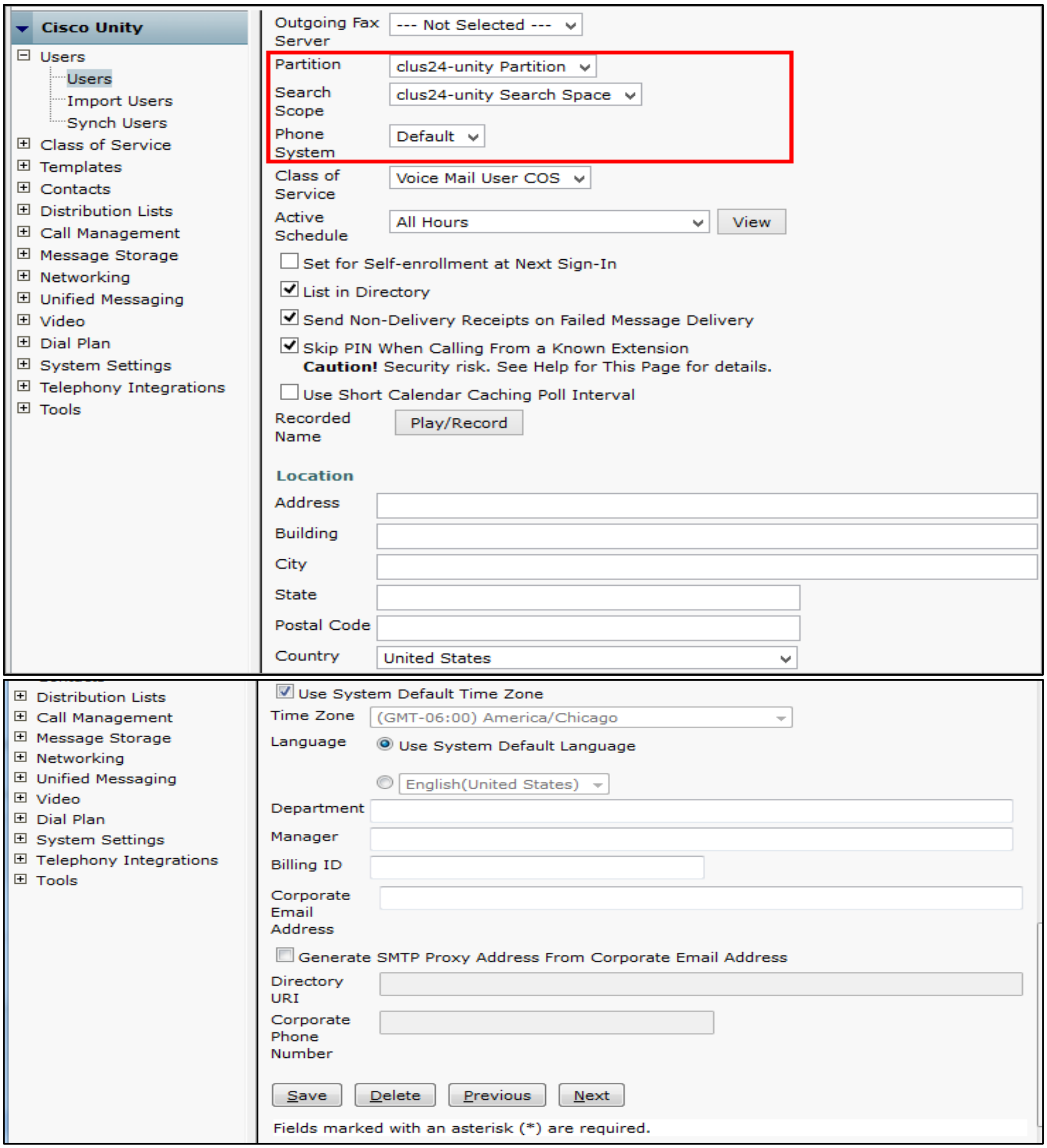
Navigation: Cisco Unity Connection → Users → Users

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



CUC Sample User Basic Settings (Contd.)

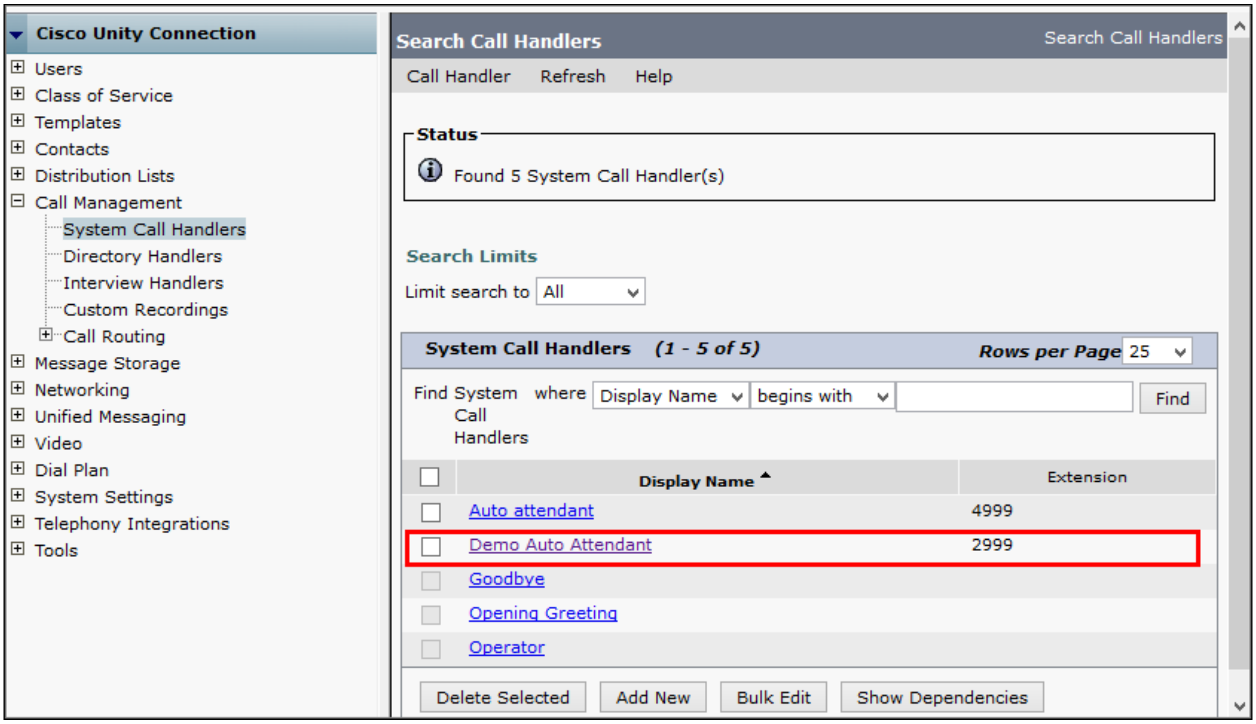
Set Partition = clus24-unity Partition. This is used for this example.
Select Search Space = clus24-unity Search Space.
Select Phone System = Default.



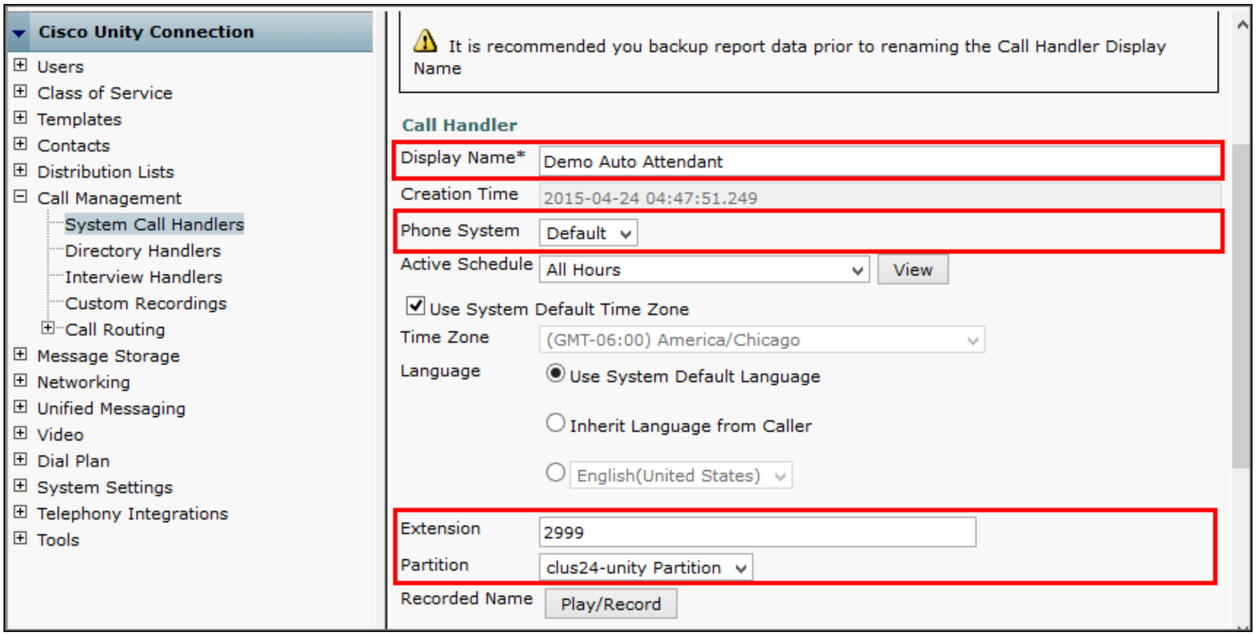
Auto Attendant

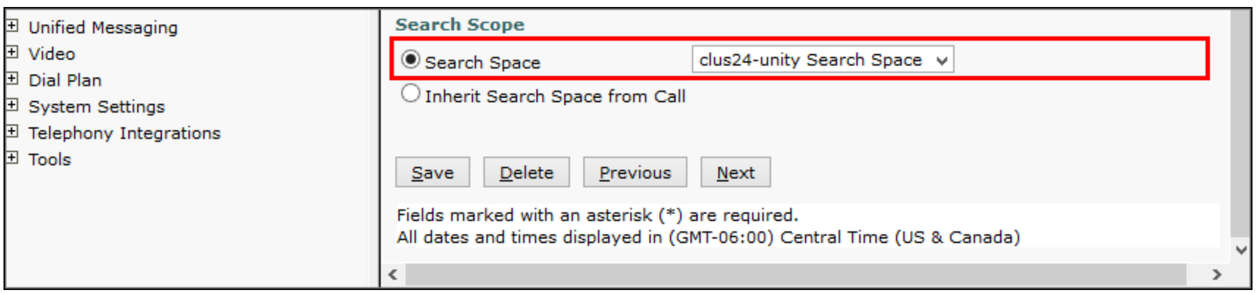
Navigation: Call Management → System Call Handlers

Set Display Name = Demo Auto Attendant. This is used for this example.
Set Phone System = Default
Set Extension=2999. This number is used as Auto attendant on this set up.
Set Partition = clus24-unity Partition. This is used for this example.
Set Search Space = clus24-unity Search Space.



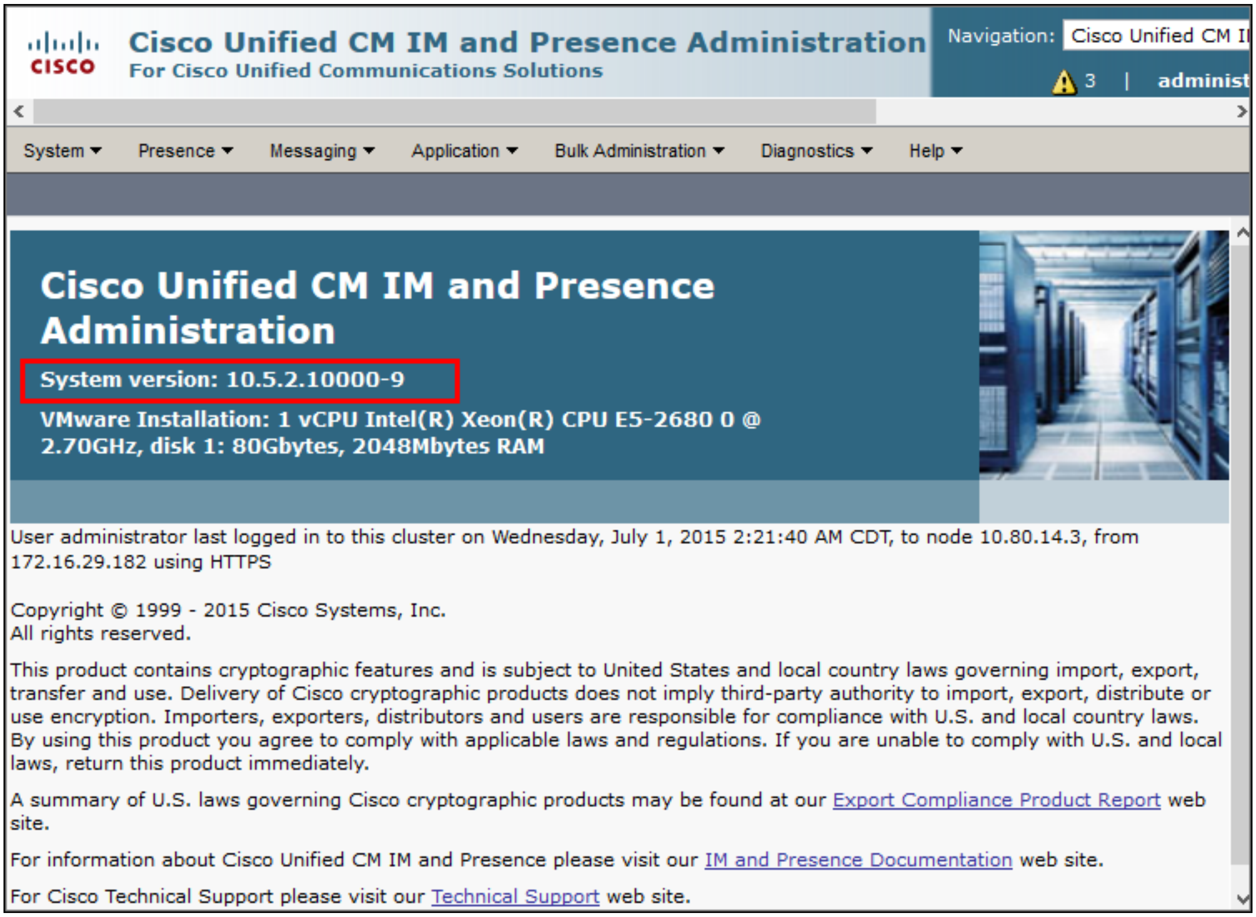
Auto Attendant (Contd.)





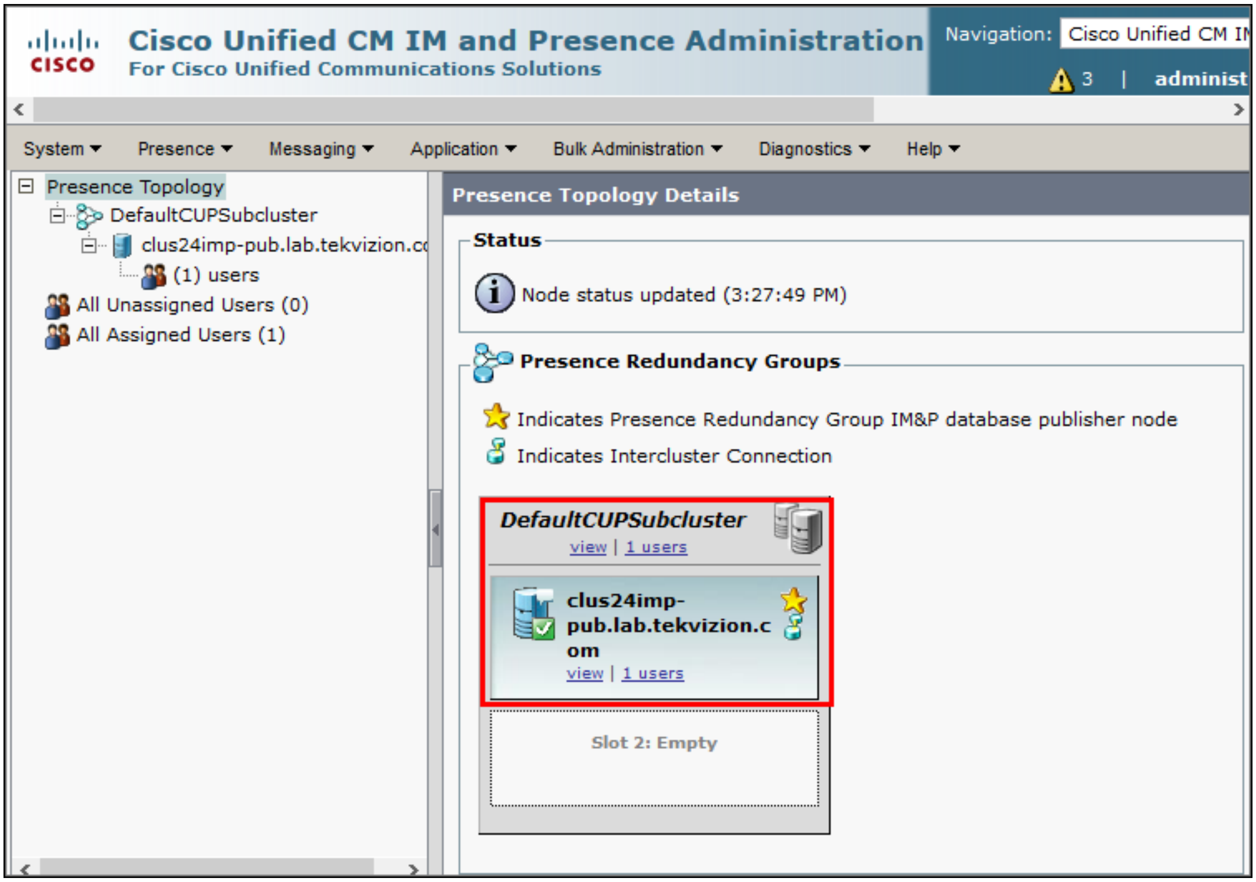
Cisco UCM Integration with Cisco Unified CM IM and Presence (CUP/IMP)

CUP/IMP Version



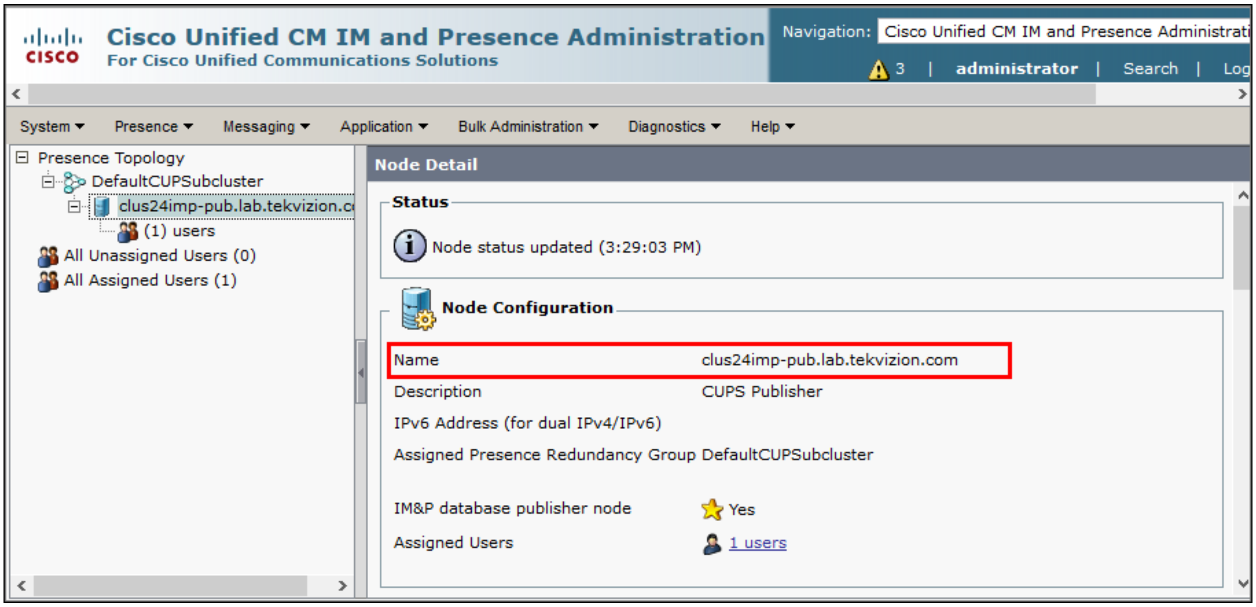
Presence Topology

Navigation: System → Presence Topology



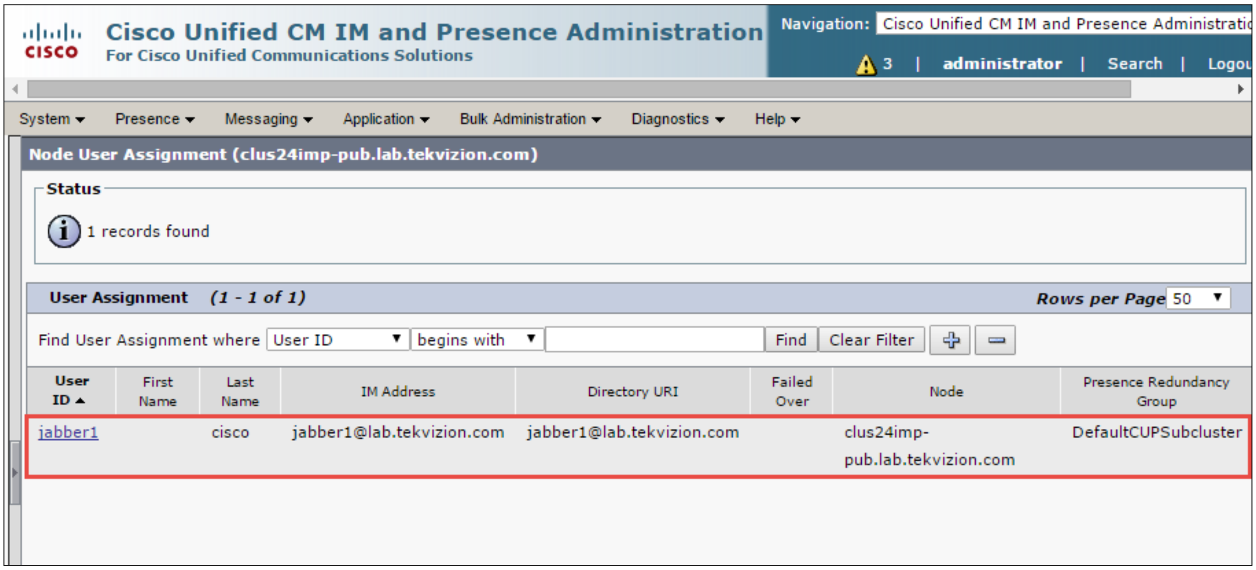
Node Configuration

Navigation: System → Presence Topology → Fully Qualified Domain Name



Users

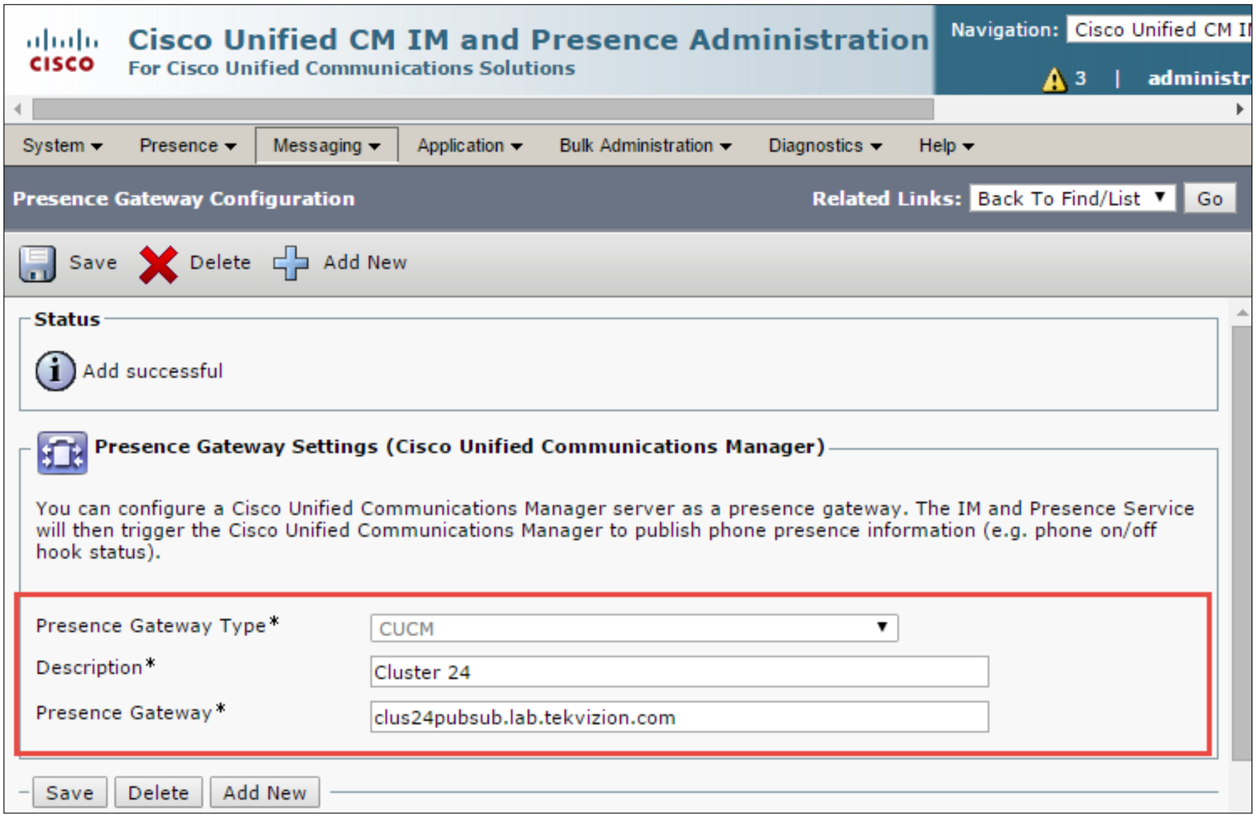
Navigation: System → Presence Topology → clus24imp-pub.lab.tekvizion.com → Users



Presence gateway configuration

Navigation: Presence → Gateways

Set Presence Gateway Type *= CUCM
Set Description *= Cluster 24. This is used for this example.
Presence Gateway * = clus24pubsub.lab.tekvizion.com



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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