Application Note

AT&T IP Flexible Reach Service with Enhanced Features Using MIS / PNT or AT&T Virtual Private Network Transport with Cisco Unified Communications Manager v. 11.0 and Cisco UBE v. 11.1.0 on an ISR 4431 Router with IPv6 SIP Interface

MAR 2016

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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) 11.0, Cisco Unity Connection 11.0, Cisco Unified CM IM and Presence 11.0, Cisco Integrated Services Routers (ISR) Version 15.5(3) S1 with connectivity to AT&T's IP Flex-Reach SIP trunk service. It also covers support and configuration example of Cisco Unity Connection (CUC) messaging integrated with Cisco Unified Communications Manager (Cisco UCM). The deployment model covered in this application note is Cisco Integrated Services Routers (ISR) to PSTN (AT&T IP Flexible Reach SIP). AT&T IP Flexible Reach provides inbound and outbound call service.

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

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

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

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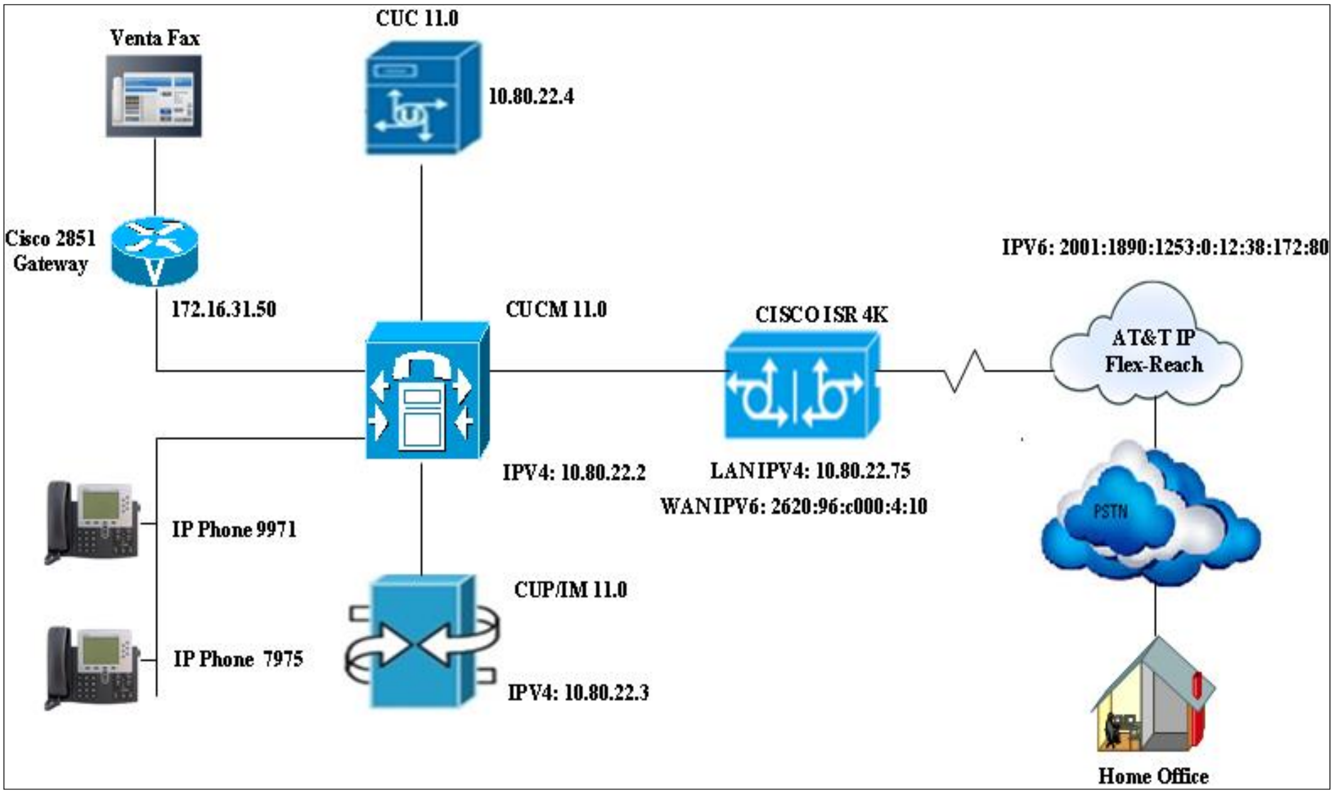
Network Topology



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Hardware Components

- UCS-C240 VMWare server running ESXi 5.5
- Cisco IP Phones. This solution was tested with Cisco 7975 & Cisco 9971 phones
- Cisco ISR4431/K9 (1RU) processor with 1659383K/6147K bytes of memory
- Processor board ID FTX1850ALVU
  4 Gigabit Ethernet interfaces,
  32768K bytes of non-volatile configuration memory,
  4194304K bytes of physical memory,
  7057407K bytes of flash memory at bootflash.

Software Requirements

- Cisco UCM: System version: 11.0.1.10000-10, including Business Edition 6000 and Business Edition 7000.
- ISR: ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.5(3) S1a, RELEASE SOFTWARE (fc1).
- Cisco UBE Software Release 11.1.0
- System image file is “isr4400-universalk9.03.16.01a.S.155-3.S1a-ext.SPA.bin".
- Cisco Unity Connection version: System version: 11.0.1.10000-10.
- Cisco Unified CM IM and Presence: System version: 11.0.1.10000-6.
- Cisco Jabber client version:11.0.0 Build 65527
- VentaFax client version: 7.6.244.598 I

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Features

Features – Supported

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

Network Based Features - Supported

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

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

Features - Not Supported

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

Caveats

Auto-Attendant

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

Hold/Resume & Music on Hold (MOH)

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

PBX Based Call Forward Unconditional

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

SIP Provisional Acknowledgement/Early media

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

AT&T IP Teleconferencing (IPTC)

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

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

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Configuration Considerations

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

Emergency 911/E911 Services Limitations and Restrictions

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

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ISR Configuration

CISCO_4K_ROUTER2#sh version

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

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

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

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

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

software.

ROM: IOS-XE ROMMON

CISCO_4K_ROUTER2 uptime is 2 weeks, 22 hours, 50 minutes

Uptime for this control processor is 2 weeks, 22 hours, 51 minutes

System returned to ROM by reload

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

Last reload reason: Reload Command

This product contains cryptographic features and is subject to United

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

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

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

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

export@cisco.com.

Suite License Information for Module:'esg'

Suite Suite Current Type Suite Next reboot

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```
--------------------------------------------------------------------------------
FoundationSuiteK9     None              None         None
securityk9
appxk9

AdvUCSuiteK9        None              None         None
uck9
cme-srst
cube



Technology Package License Information:


-----------------------------------------------------------------
Technology    Technology-package         Technology-package
           Current       Type          Next reboot
------------------------------------------------------------------
appxk9         appxk9         RightToUse      appxk9
uck9           uck9           Evaluation      uck9
securityk9      securityk9      EvalRightToUse   securityk9
ipbase         ipbasek9        Permanent       ipbasek9

cisco ISR4431/K9 (1RU) processor with 1659383K/6147K bytes of memory.
Processor board ID FTX1850ALVU
4 Gigabit Ethernet interfaces
32768K bytes of non-volatile configuration memory.
```

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4194304K bytes of physical memory.

7057407K bytes of flash memory at bootflash:.

Configuration register is 0x2102

CISCO_4K_ROUTER2#sh run

Building configuration...

Current configuration : 13184 bytes

!

! Last configuration change at 06:38:10 UTC Tue Mar 1 2016 by cisco

!

version 15.5

service timestamps debug datetime msec

service timestamps log datetime msec

no platform punt-keepalive disable-kernel-core

!

hostname CISCO_4K_ROUTER2

!

boot-start-marker

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

boot-end-marker

!

!

```
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
enable secret 5 $1$zQRB$CCbzfD1aYzk3kPvzAm2KU0
enable password cisco
!
aaa new-model
!
!
!
!
!
!
!
aaa session-id common
!
ipc zone default
 association 1
  no shutdown
!
!
```

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```
ip name-server 8.8.8.8

no ip domain lookup
!
!
!
ipv6 unicast-routing
!
!
!
!
!
!
!
subscriber templating
!
multilink bundle-name authenticated
!
!
!
!
!
!
!
cts logging verbose
!
```

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```
!
voice service voip
 no ip address trusted authenticate
rtp-port range 16384 32766
 address-hiding[1]
 mode border-element[2]
 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
 no fax-relay sg3-to-g3
 sip
  header-passing
  error-passthru[4]
  asserted-id pai[5]
  early-offer forced[6]
  no silent-discard untrusted
  midcall-signaling passthru[7]
  privacy-policy passthru[8]
```

---

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

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

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

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

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

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

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

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```
 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 codec 3
 codec preference 1 g711ulaw
!
!
voice class sip-profiles 1
 response ANY sip-header Allow-Header modify "UPDATE," ""
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:732216\1@\2>"[10]
 request INVITE sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"[11]
 response ANY sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"
 request INVITE sdp-header Audio-Attribute add "a=ptime:30"[12]
```

---

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

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

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```
!
!
!
!
!
!
!
license udi pid ISR4431/K9 sn FOC18232988
license boot level appxk9
license boot level uck9
license boot level securityk9
!
spanning-tree extend system-id
!
username cisco privilege 15 secret 5 $1$AGR7$e7pQx6UI0be3bzRbc0lr81
!
redundancy
 mode none
!
!
!
!
!
vlan internal allocation policy ascending
```

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

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```
!
!
!
interface GigabitEthernet0/0/0
 ip address 10.64.4.20 255.255.0.0
 shutdown
 media-type rj45
 negotiation auto
!
interface GigabitEthernet0/0/1
 no ip address
 shutdown
 media-type rj45
 negotiation auto
!
interface GigabitEthernet0/0/2[13]
 ip address 10.80.22.75 255.255.255.0[14]
 media-type rj45
 negotiation auto
!
interface GigabitEthernet0/0/3[15]
 description Wan Interface
 no ip address
 media-type rj45
```

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

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```
 negotiation auto
 ipv6 address 2620:96:C000:4::10/64
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 shutdown
 negotiation auto
!
interface Vlan1
 no ip address
 shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.97
ip route 10.64.0.0 255.255.0.0 10.80.22.1
ip route 10.80.22.0 255.255.255.0 10.80.22.1
ip route 172.16.0.0 255.255.0.0 10.80.22.1
!
!
ipv6 route ::/0 2620:96:C000:4::1
!
!
control-plane
```

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```
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
!
!
dial-peer voice 200 voip
 description "Outgoing To AT&T .IP PBX facing side"
 session protocol sipv2
 incoming called-number [2-9]T
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 no fax-relay sg3-to-g3
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
```

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```
 no vad
!
dial-peer voice 214 voip[16]
 description "Outgoing To AT&T"-AT&T facing side
 destination-pattern [2-9]T
 session protocol sipv2[17]
 session target ipv6:[2001:1890:1253:0:12:38:172:80]
 voice-class codec 1[18]
 voice-class sip asymmetric payload full[19]
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru[20]
 voice-class sip early-offer forced
 voice-class sip profiles 1[21]
 voice-class sip bind control source-interface GigabitEthernet0/0/3[22]
 voice-class sip bind media source-interface GigabitEthernet0/0/3
 dtmf-relay rtp-nte
 no fax-relay sg3-to-g3
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
```

[16] Dial peer for AT&T facing network
[17] Session protocol SIPv2 is used for this testing.
[18] Assigns voice class codec 1 settings to dial-peer (codec support and filtering).
[19] Configures the dynamic SIP asymmetric payload support.
[20] 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.
[21] This command enables the dial peer to use SIP profile 1
[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.

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```
!
dial-peer voice 800 voip
 description " Incoming AT&T to IP-PBX . AT&T facing side "
 translation-profile incoming test+1
 huntstop
 session protocol sipv2
 incoming called-number [37][13][24][12]......
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/3
 voice-class sip bind media source-interface GigabitEthernet0/0/3
 dtmf-relay rtp-nte[23]
 no fax-relay sg3-to-g3
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none[24]
 no vad
!
dial-peer voice 700 voip
 description " Incoming AT&T to IP-PBX - IP-PBX facing side "
 huntstop
 destination-pattern [37][13][24][12]......
 no modem passthrough
```

[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

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```
 session protocol sipv2
 session target ipv4:10.80.22.2:5060
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 no fax-relay sg3-to-g3
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 300 voip
 description " Int'l calls to AT&T - AT&T facing side "
 destination-pattern 011T
 no modem passthrough
 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
```

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```
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/3
 voice-class sip bind media source-interface GigabitEthernet0/0/3
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 400 voip
 description " Int'l calls to AT&T - IP-PBX facing side "
 no modem passthrough
 session protocol sipv2
 incoming called-number 011T
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
```

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```
dial-peer voice 500 voip
 description " N11 Calls to AT&T - AT&T facing side "
 destination-pattern .11
 no modem passthrough
 session protocol sipv2
 session target 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 GigabitEthernet0/0/3
 voice-class sip bind media source-interface GigabitEthernet0/0/3
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 600 voip
 description " N11 Calls to AT&T - IP-PBX facing side "
 no modem passthrough
 session protocol sipv2
 incoming called-number .11
 voice-class codec 1
 voice-class sip asymmetric payload full
```

```
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 141 voip
description "Network Feature"
destination-pattern *..
no modem passthrough
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 GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax rate 14400
```

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```
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 2151 voip
 description "Outgoing To AT&T"-AT&T facing side
 destination-pattern 7323204...
 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 GigabitEthernet0/0/3
 voice-class sip bind media source-interface GigabitEthernet0/0/3
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 2152 voip
 description "Outgoing To AT&T"-AT&T facing side -BVoip Number
 destination-pattern 8772888362
 session protocol sipv2
 session target ipv6:[2001:1890:1253:0:12:38:172:80]
```

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```
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/3
 voice-class sip bind media source-interface GigabitEthernet0/0/3
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
gateway
 media-inactivity-criteria all
 timer receive-rtcp 5
 timer receive-rtp 86400
!
sip-ua
 no remote-party-id
 timers options 1000
 connection-reuse
 protocol mode dual-stack
!
!
```

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

Cisco UCM Configuration

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

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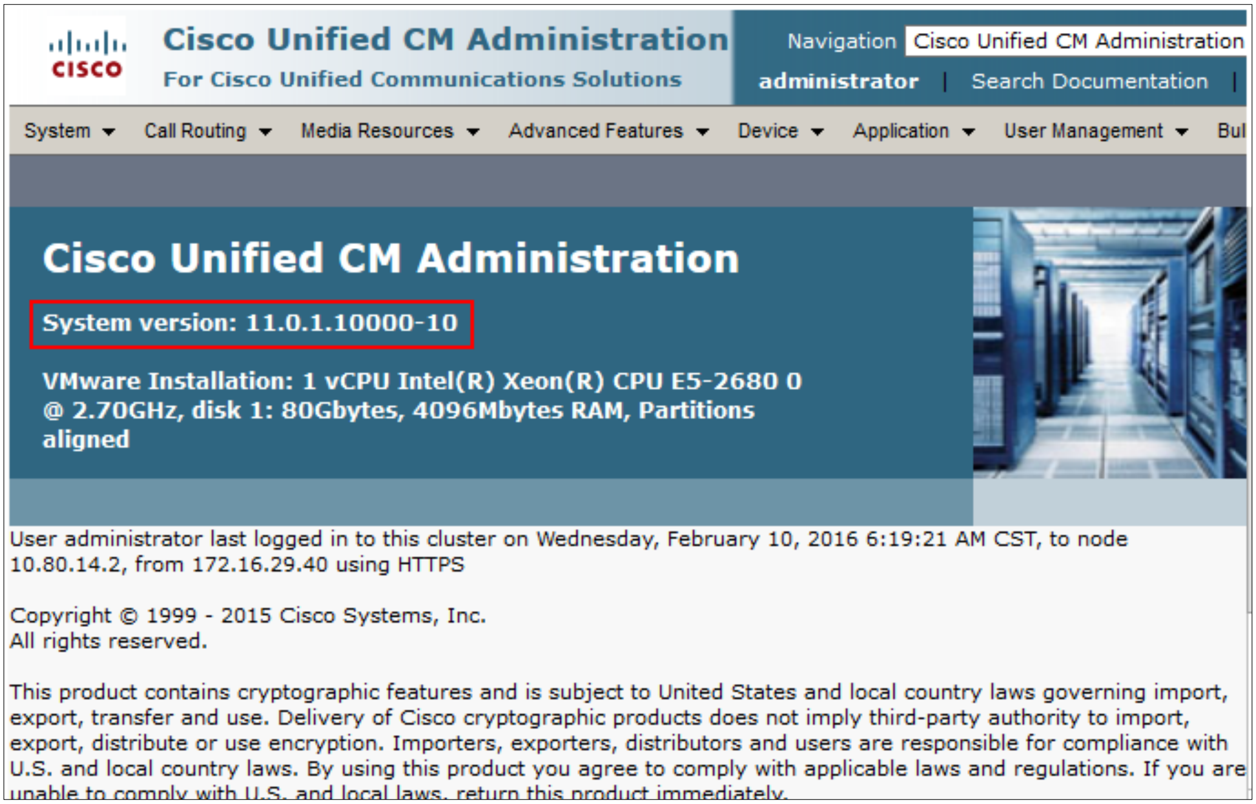
Cisco UCM Version



Cisco UCM Audio Codec Preference List

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

Cisco UCM 11.0 has a feature called Audio Codec Preference List. This feature allows to configure the order of audio codec preference both for Inter and Intra Region calls. Audio Codec Preference list is assigned to the Region used by the Device Pool for Phones and by Conference Bridges. Based on user

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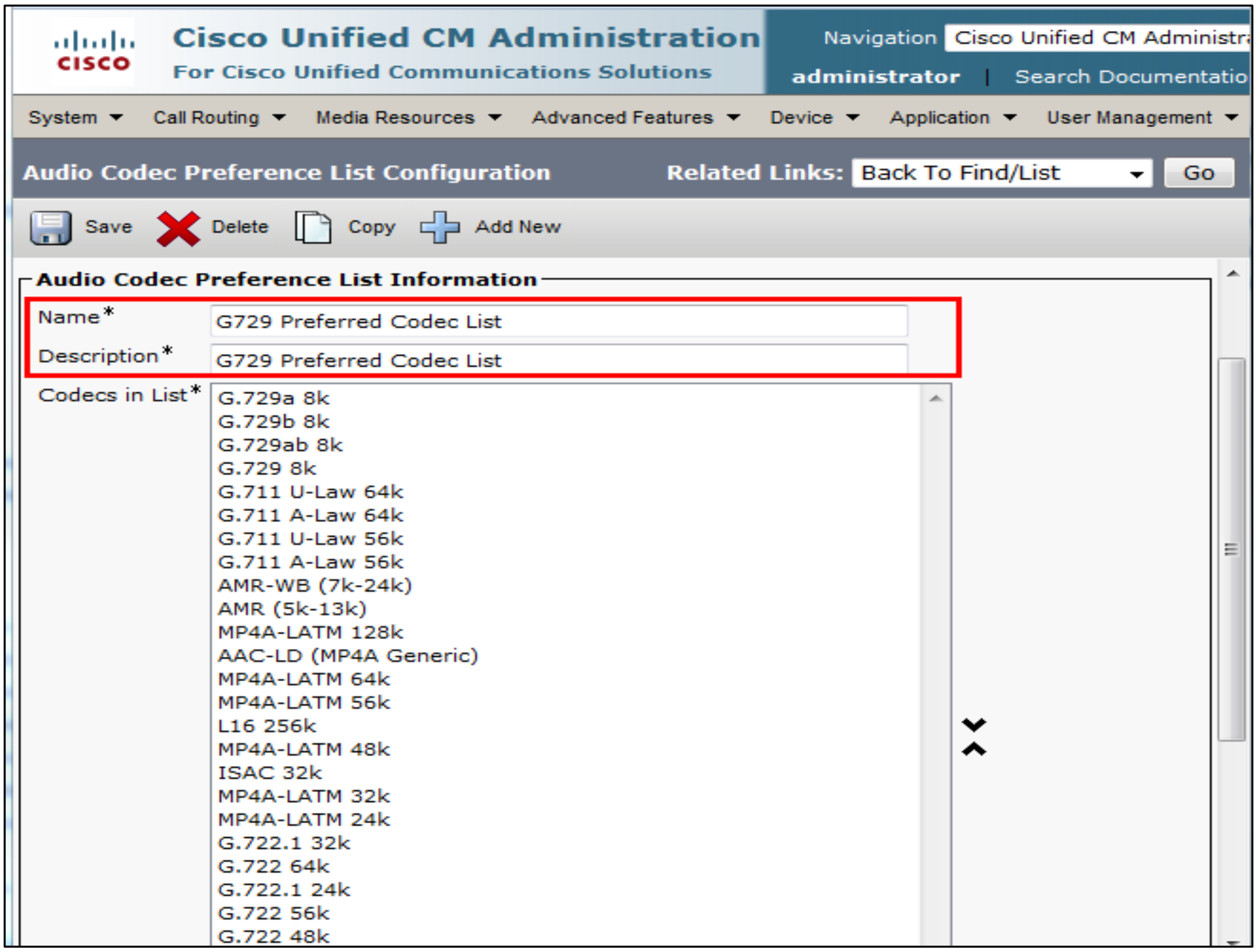
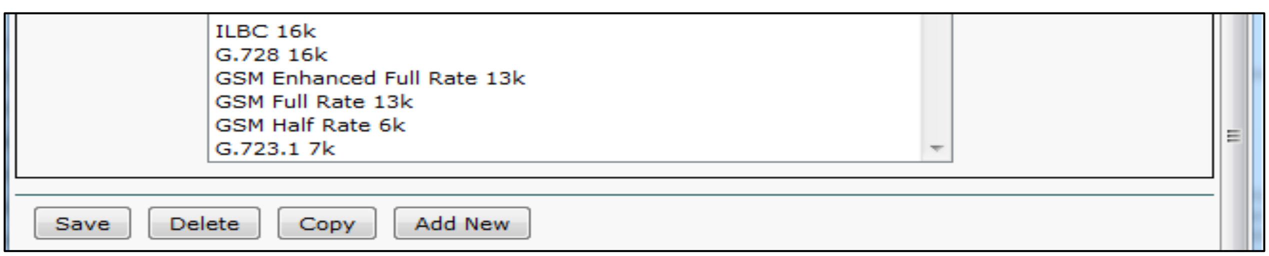
requirement, different codec regions can be assigned as their first choice codec with this configuration for inbound calls as well as conferences initiated by Cisco IP phones. Audio codec preference for outbound calls is determined by Cisco UBE (via configuration of voice-class codec)





Cisco UCM Region Configuration

Navigation Path: System → Region Information → Region

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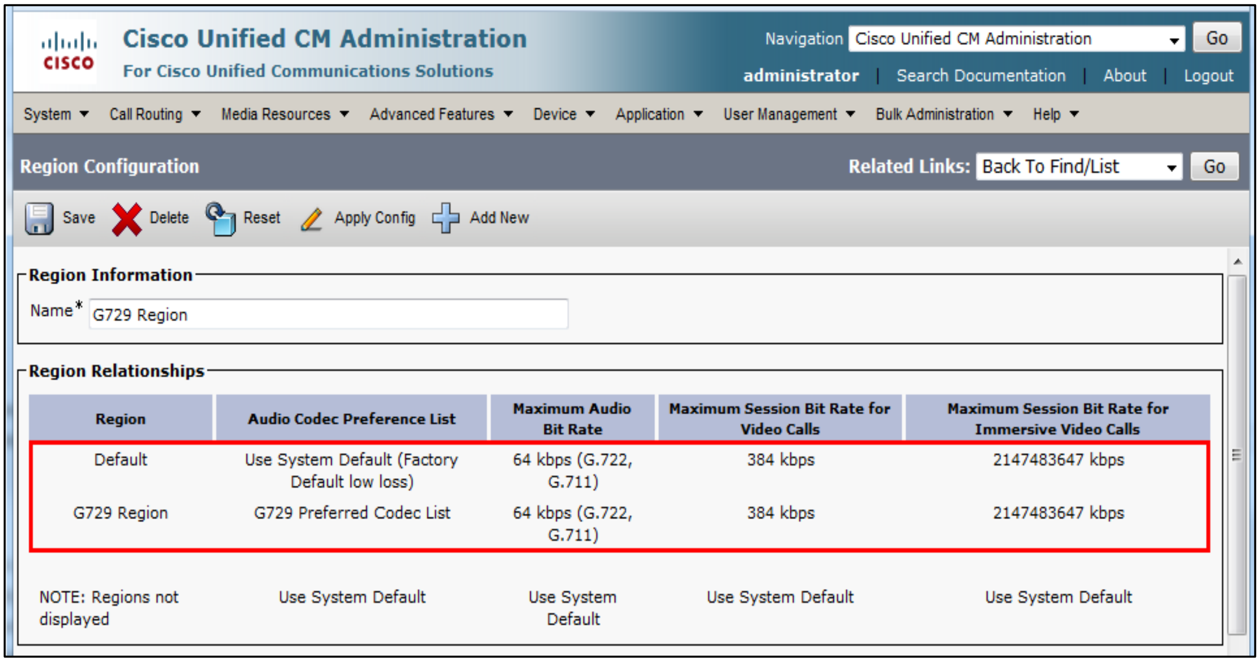

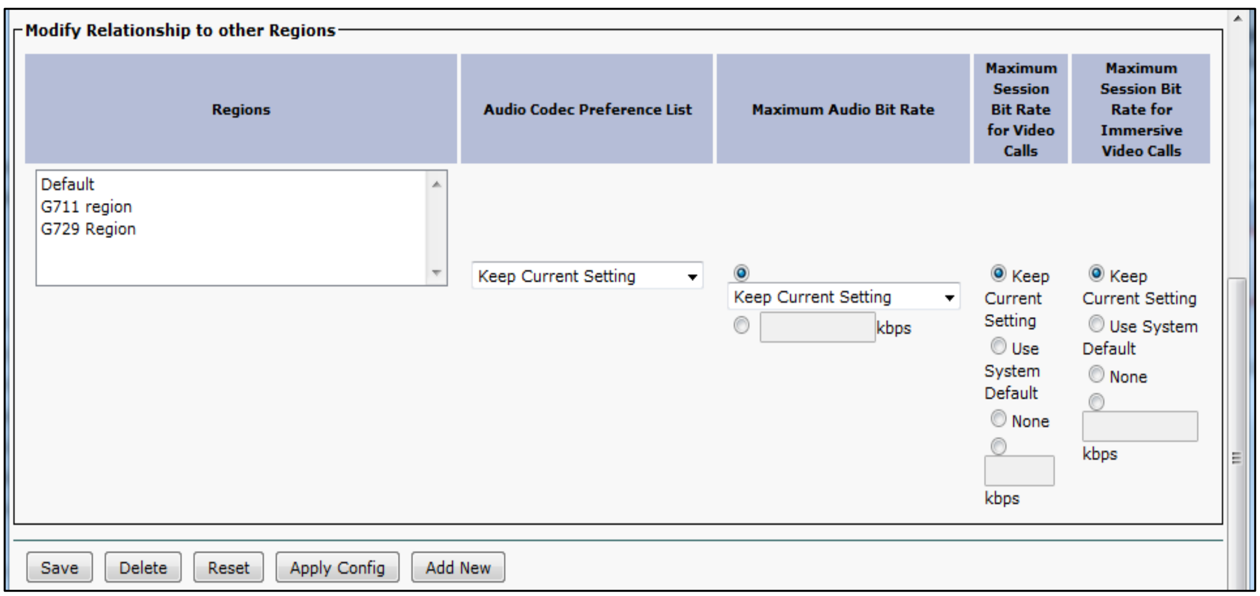

Device Pool Configuration

Navigation Path: System → Device Pool

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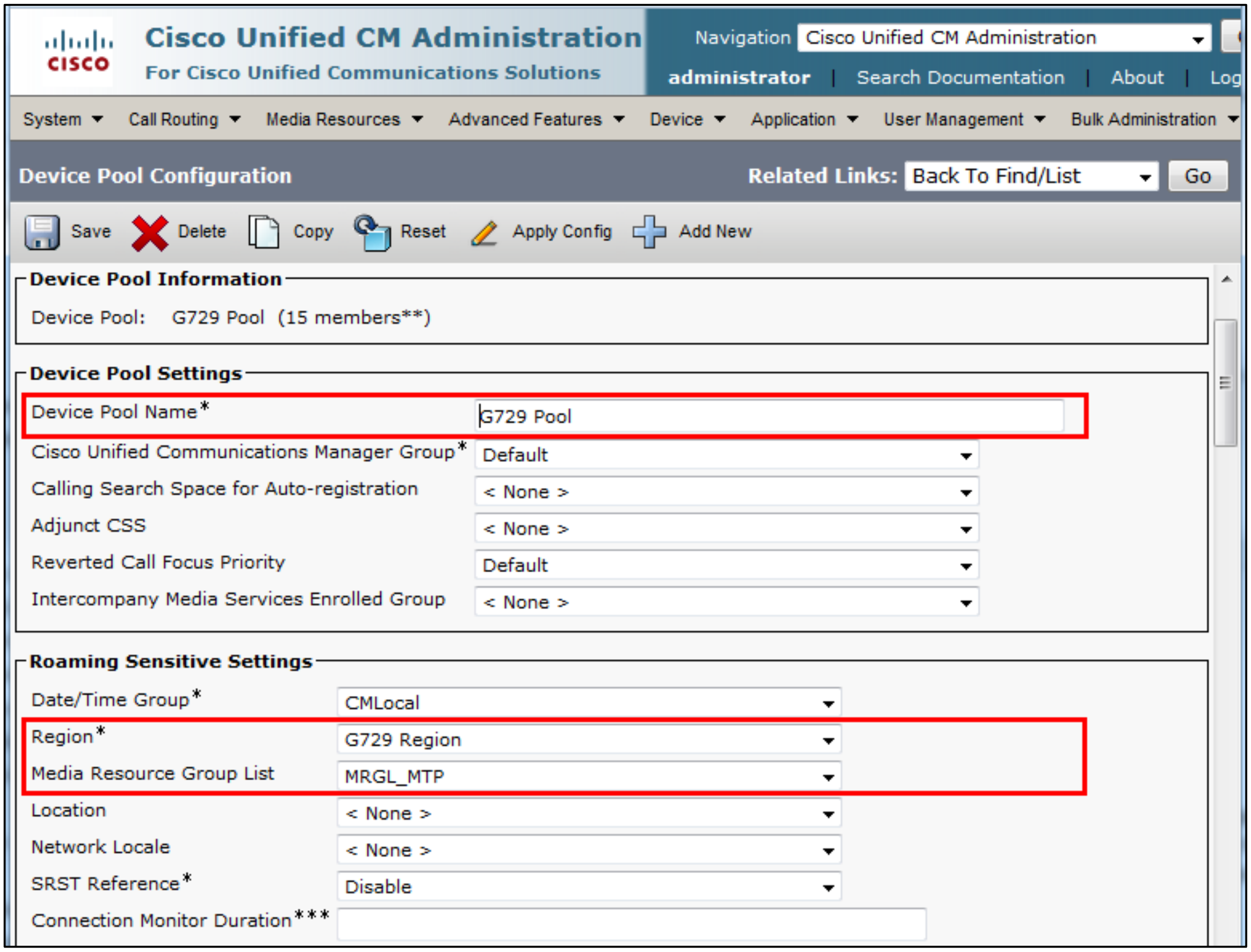
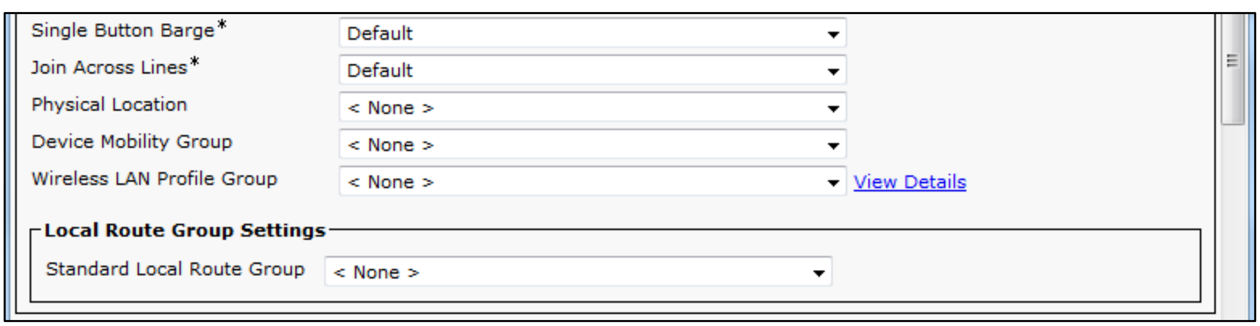
"G729" Device Pool is configured for testing the interoperability. No special consideration needs to be taken when configuring the Device Pools. Optionally, a Media Resource Group List can be added to the Device Pools, if needed, to assign selected Media Resources (Conference Bridges, Transcoders, MoH servers, Annunciators) to devices.





Device Pool Configuration (continued…)

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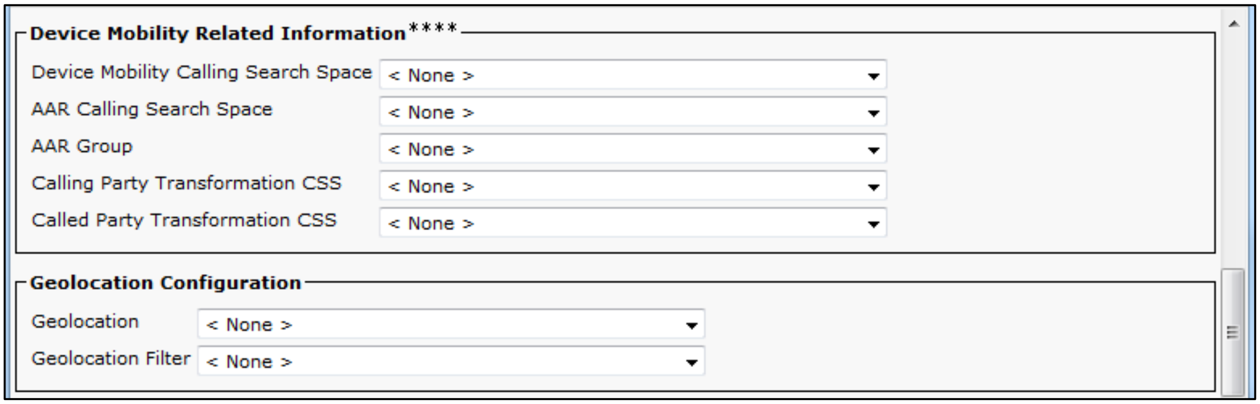

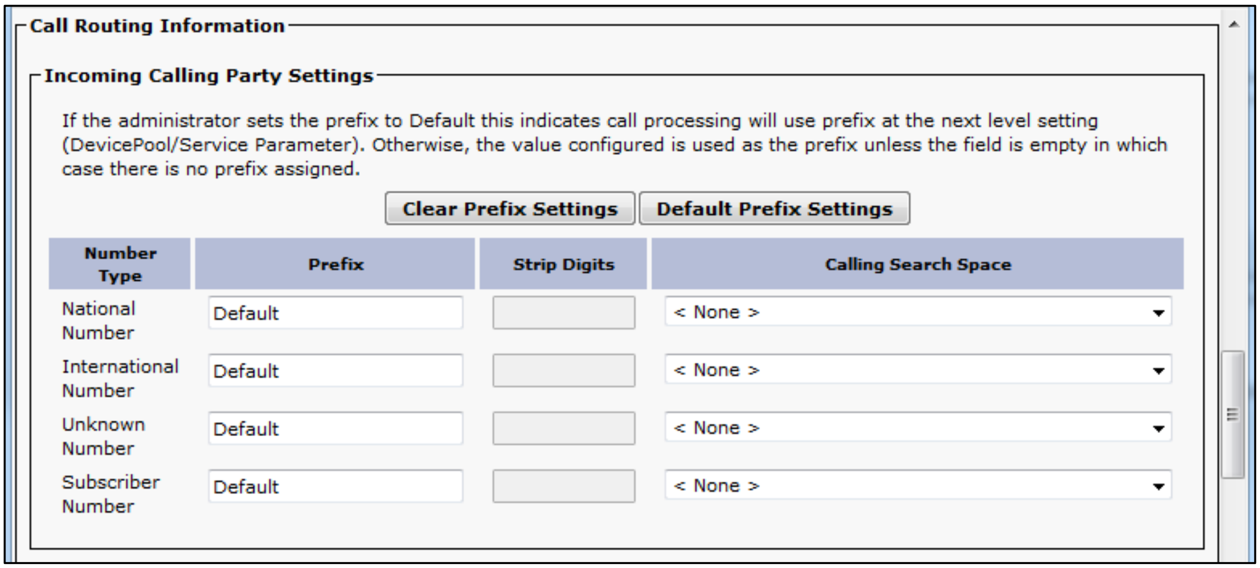

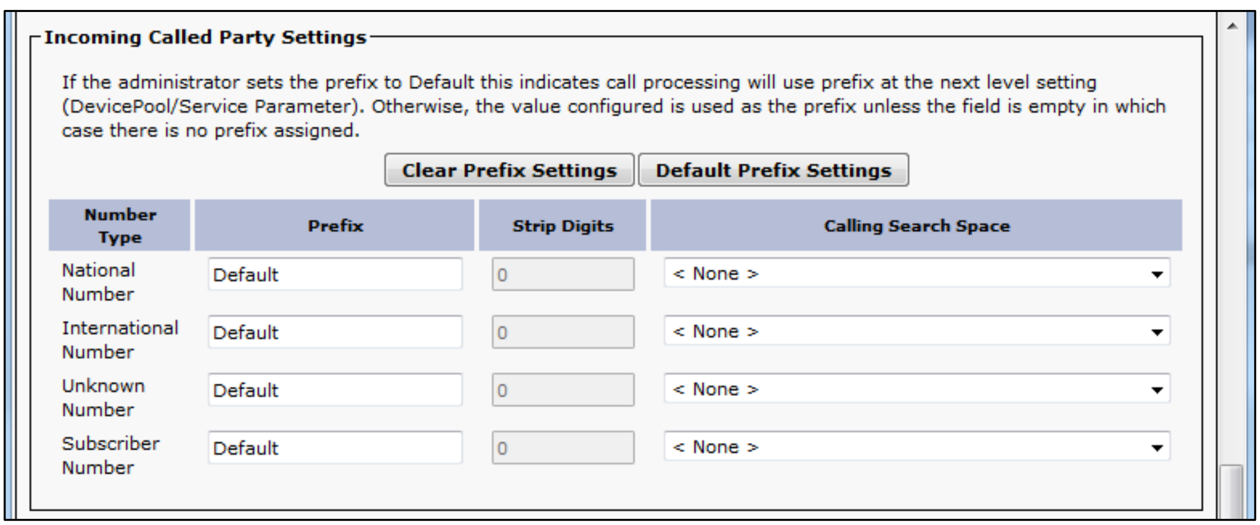

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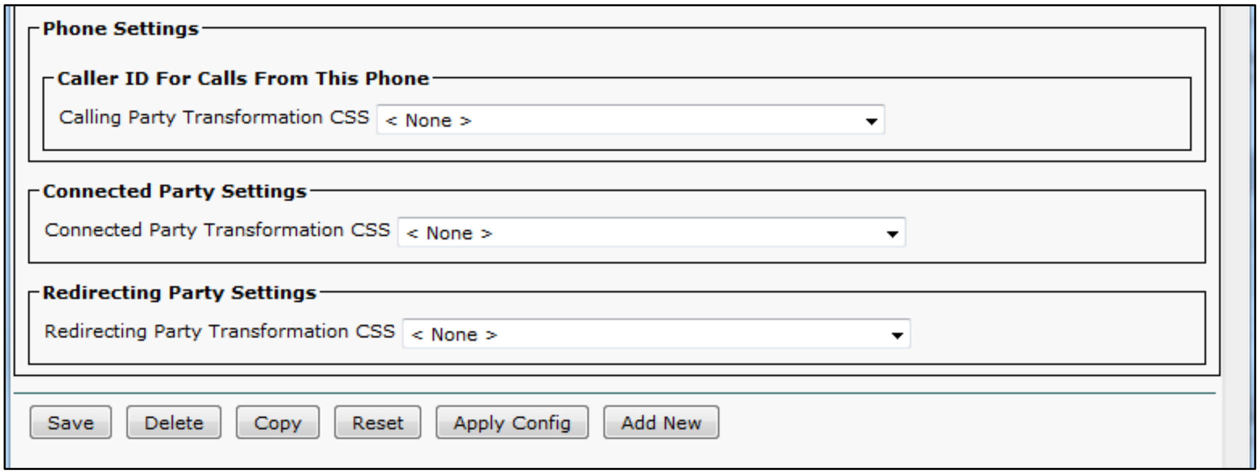
Device Pool Configuration (continued…)



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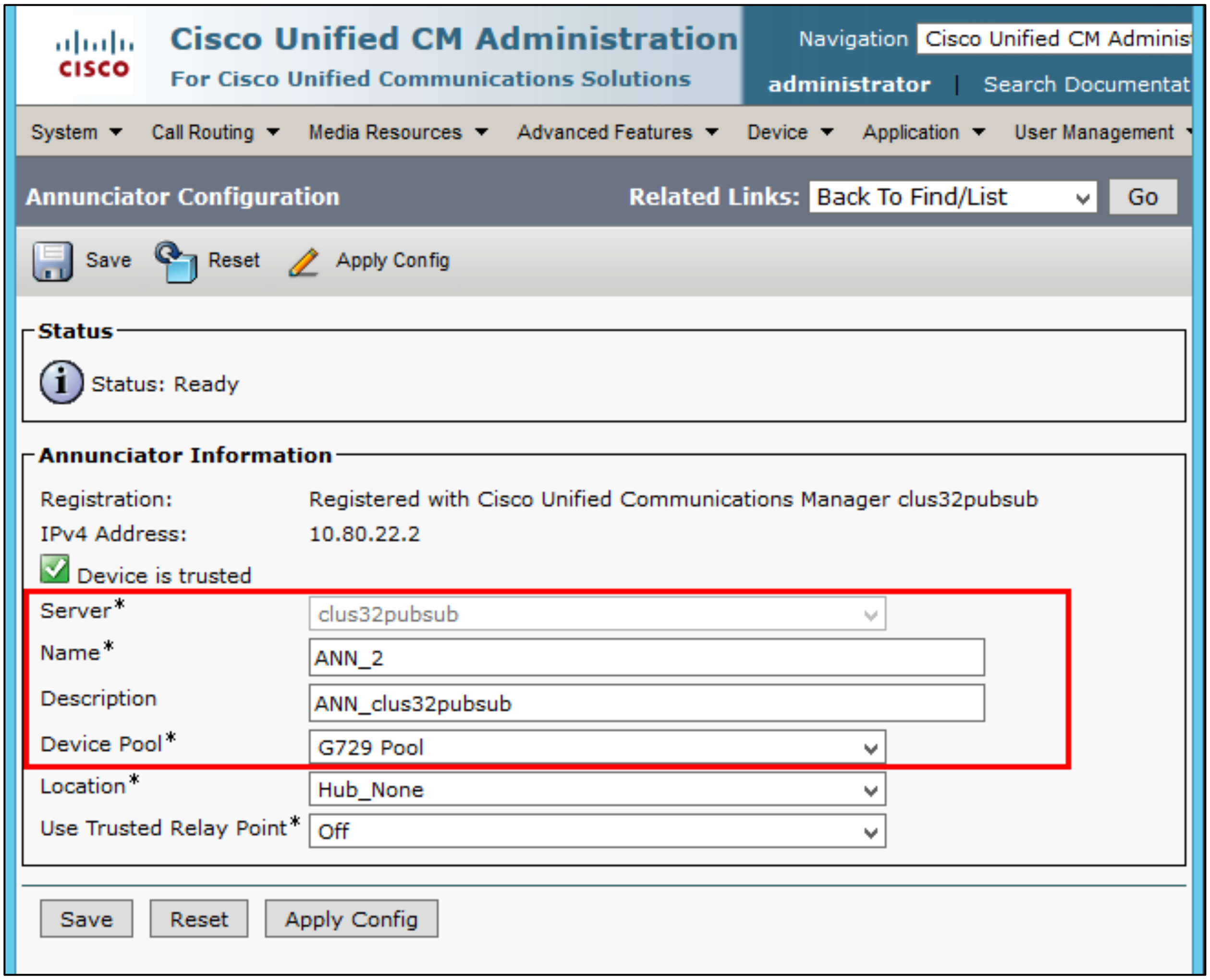
Annunciator Configuration

Navigation: Media Resource → Annunciator

Set Name* = ANN_2.
Set Description = ANN_clus32pubsub. This is used for this example
Set Device Pool* = G729 Pool.



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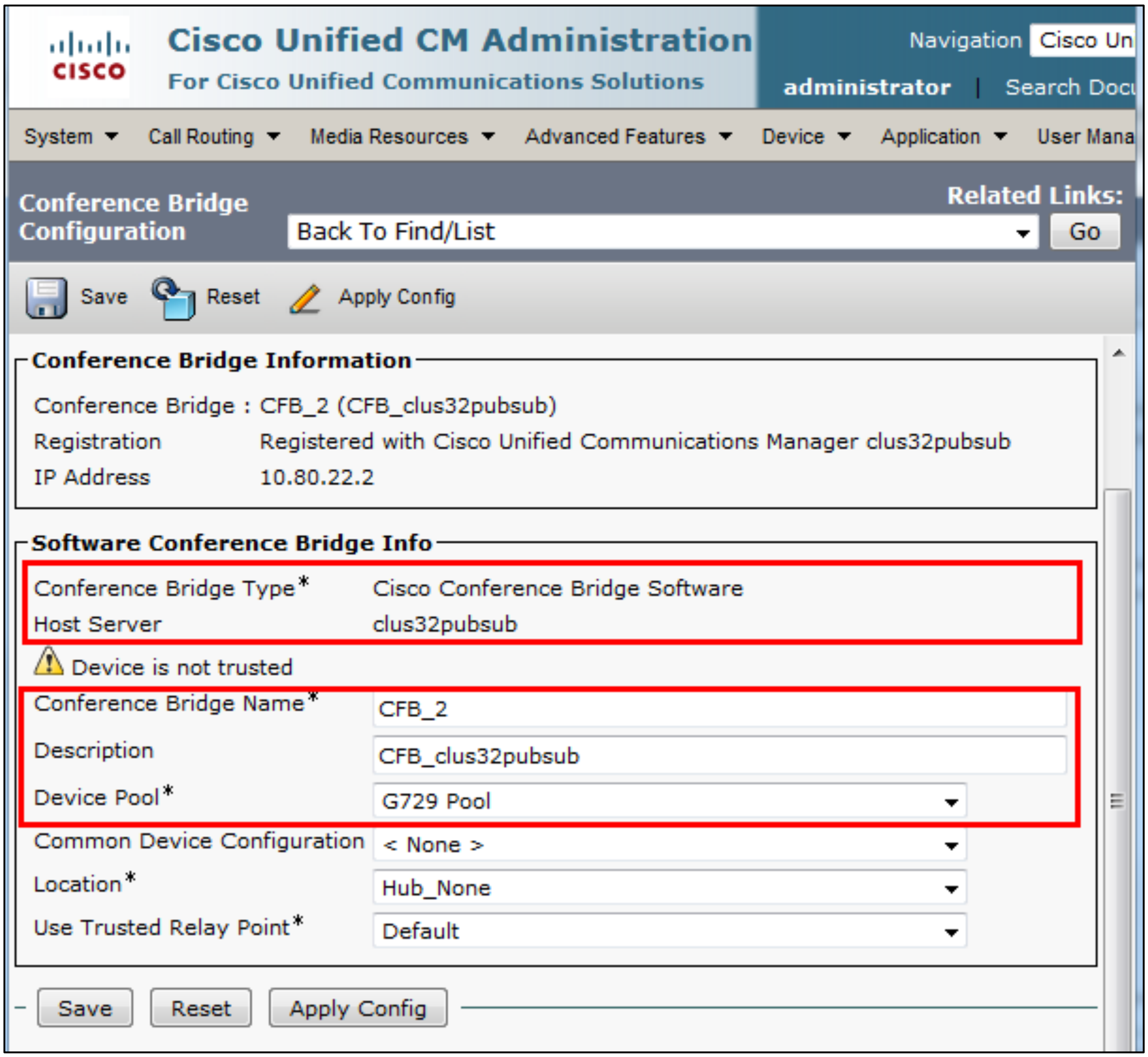
Conference Bridge Configuration

Navigation: Media Resources → Conference Bridge

Set Conference Bridge Type* = Cisco Conference Bridge Software.
Set Host Server = clus32pubsub. This is used for this example.
Set Conference Bridge Name* = CFB_2.
Set Description = CFB_clus32pubsub. This is used in this example.
Set Device Pool* = G729.



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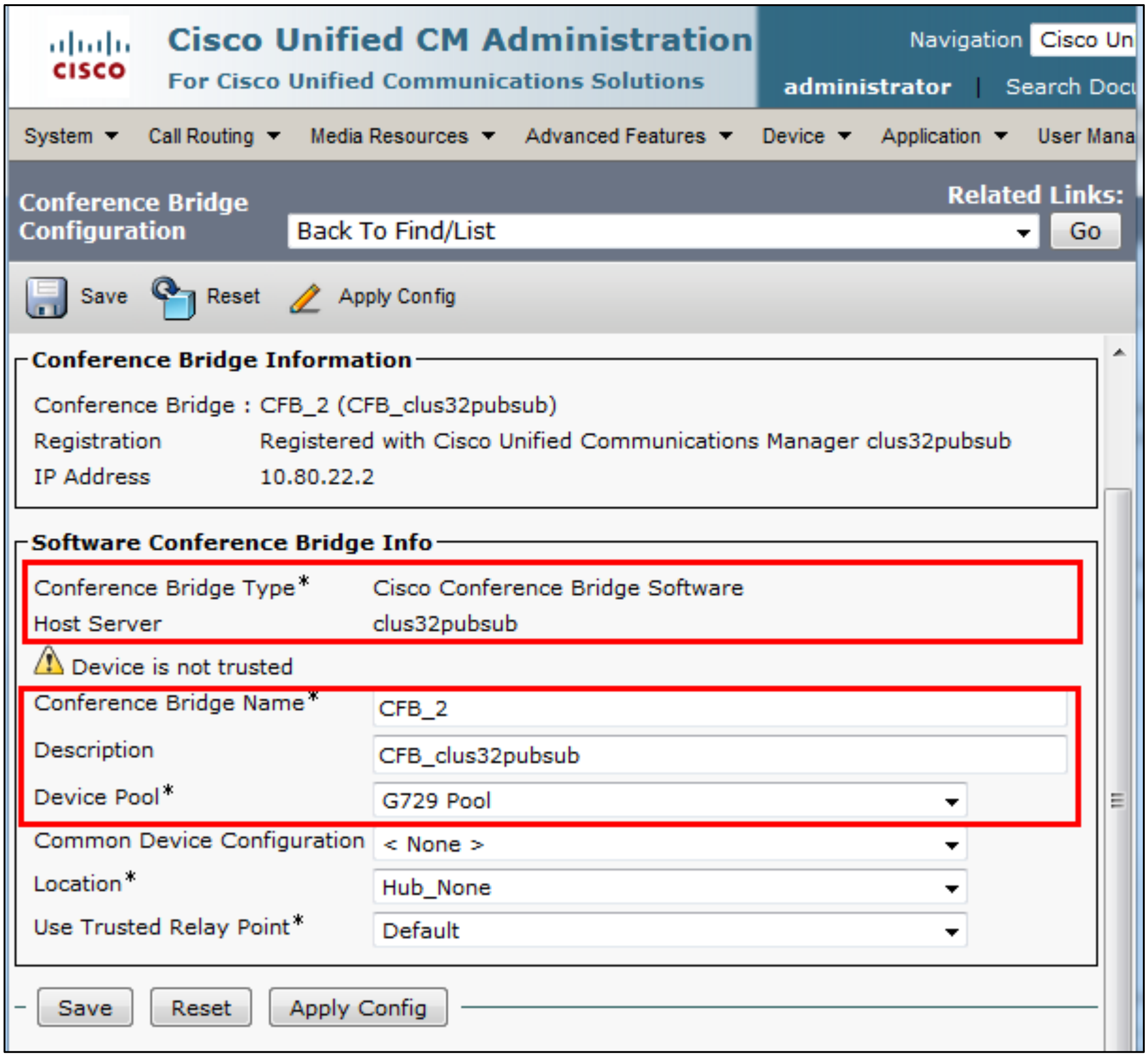
Media Termination Point Configuration

Navigation: Media Resource→Media Termination Point

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



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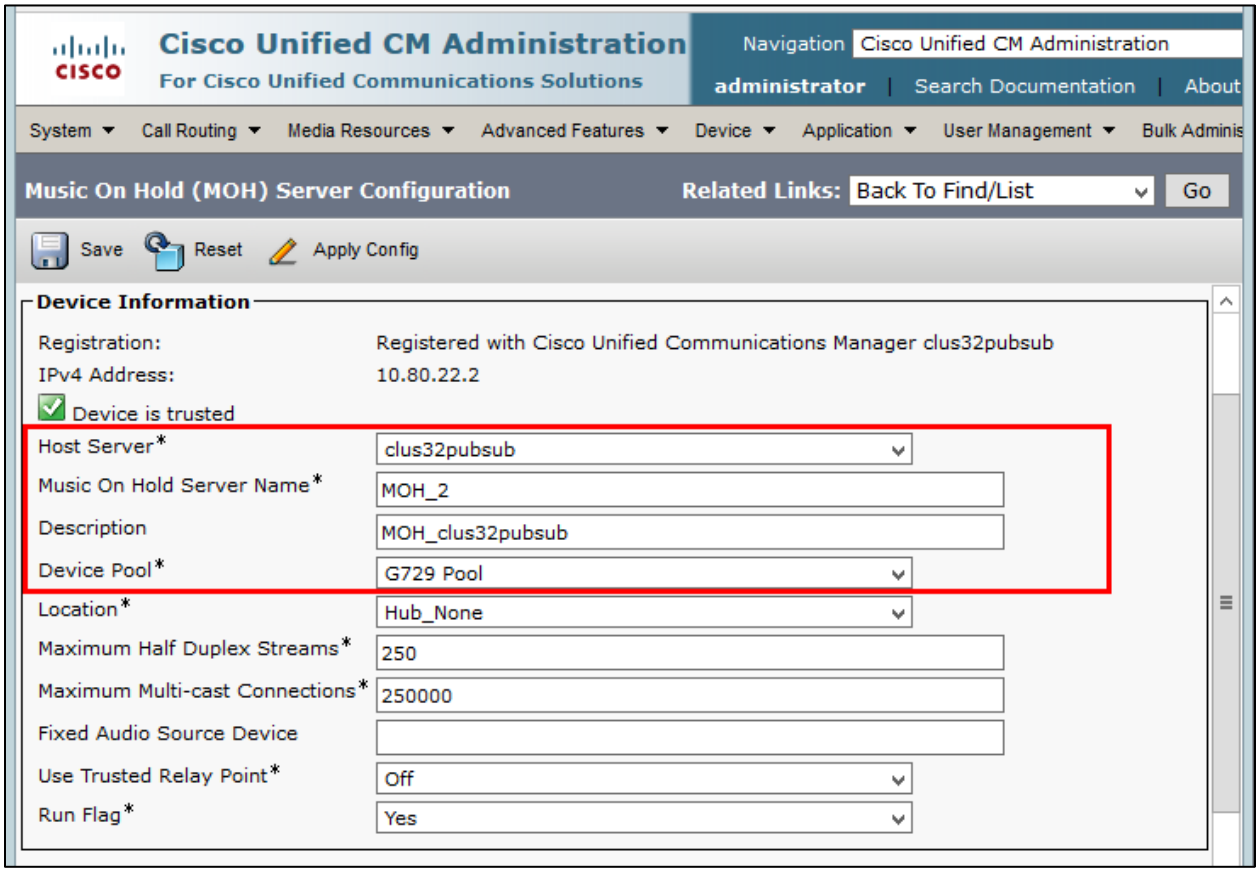
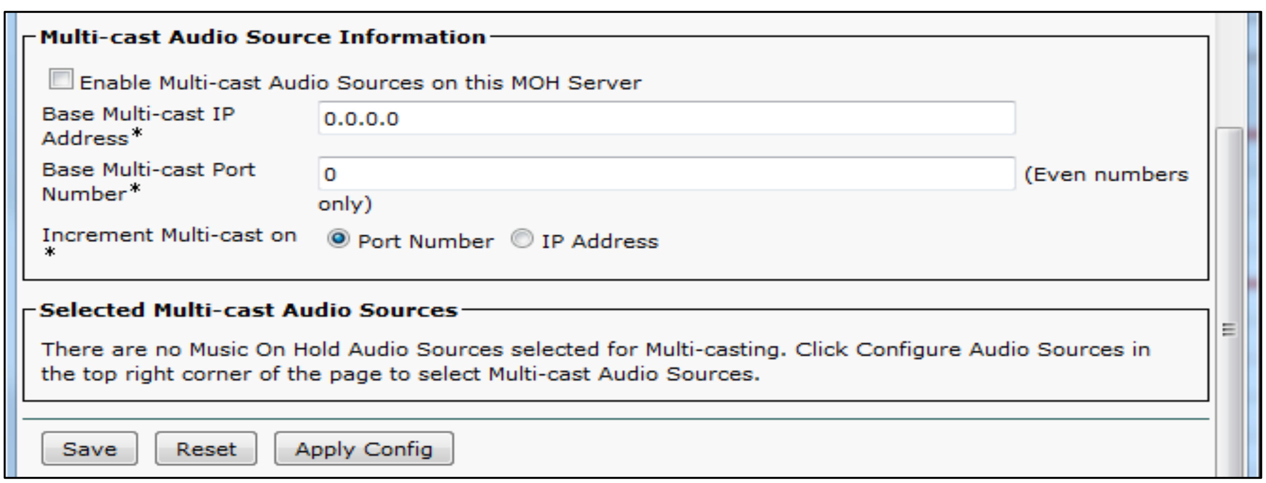
Music on Hold Server Configuration

Navigation: Media Resources → Music on Hold Server

Set Music on Hold Server Name* = MOH_2.
Set Description = MOH_clus32pubsub. This is used for this example.
Set Device Pool* = G729





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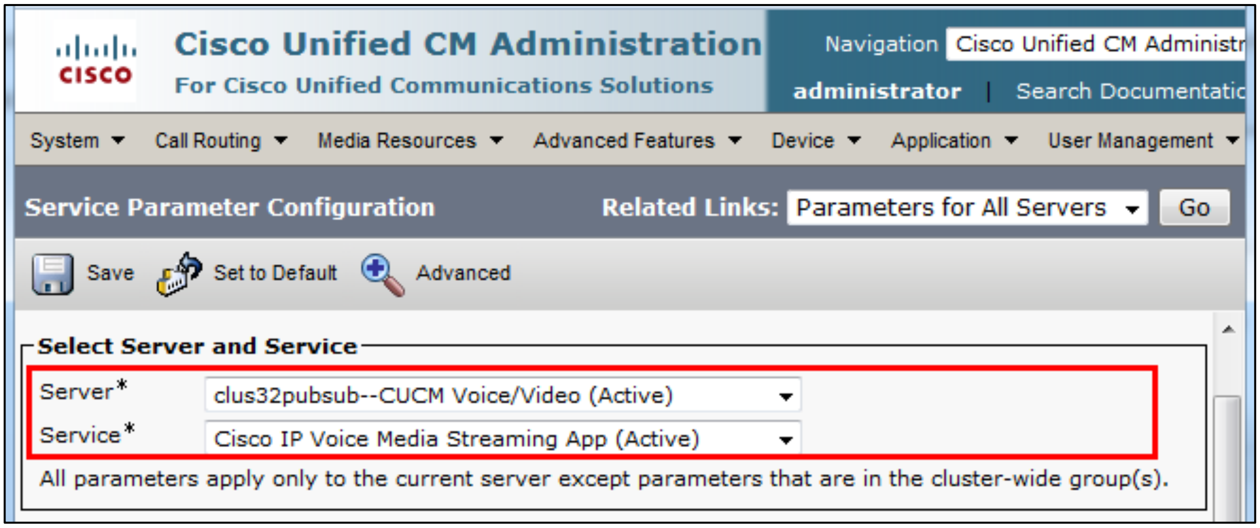
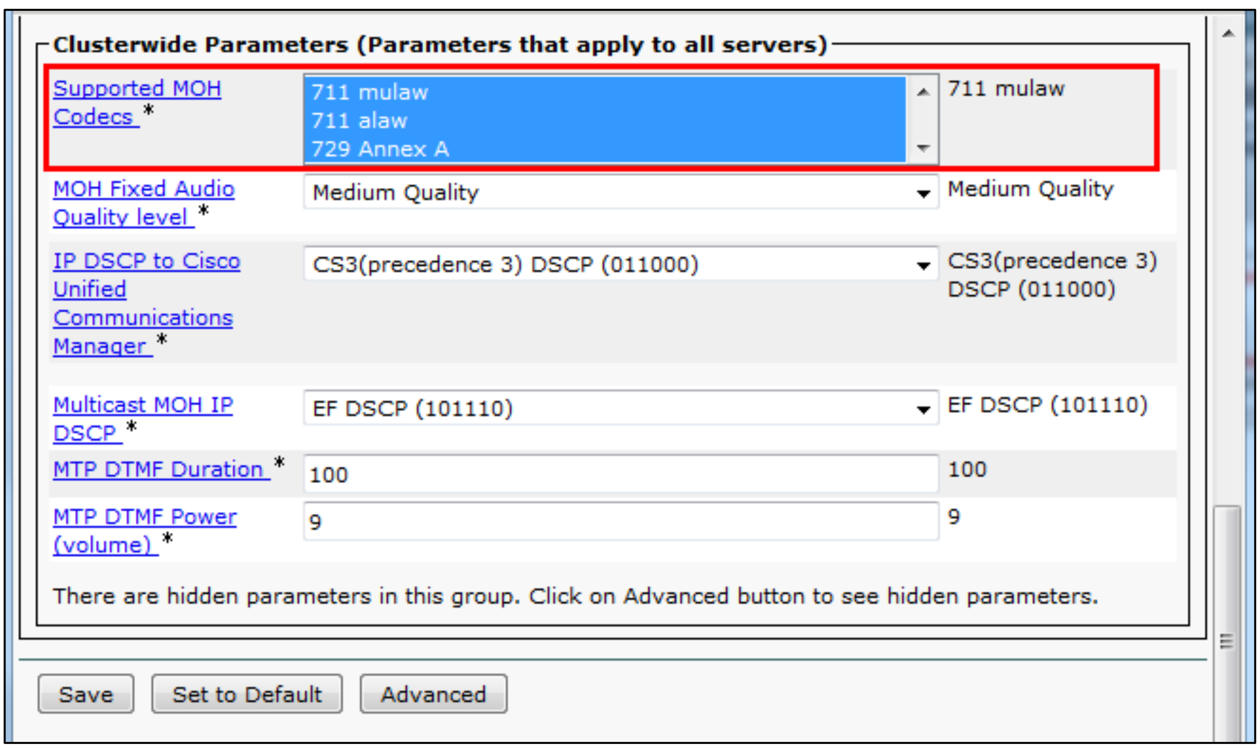
Music on Hold Service (IP Voice Media Streaming App) Parameter Settings

Navigation: System → Service Parameter

Note: Make sure codecs G.729 Annex A and G.711 mulaw are configured in parameter Supported MOH Codecs.
Select Server* = clus32pubsub--CUCM Voice/Video (Active). This is used in this example.
Select Service* = Cisco IP Voice Media Streaming App (Active)





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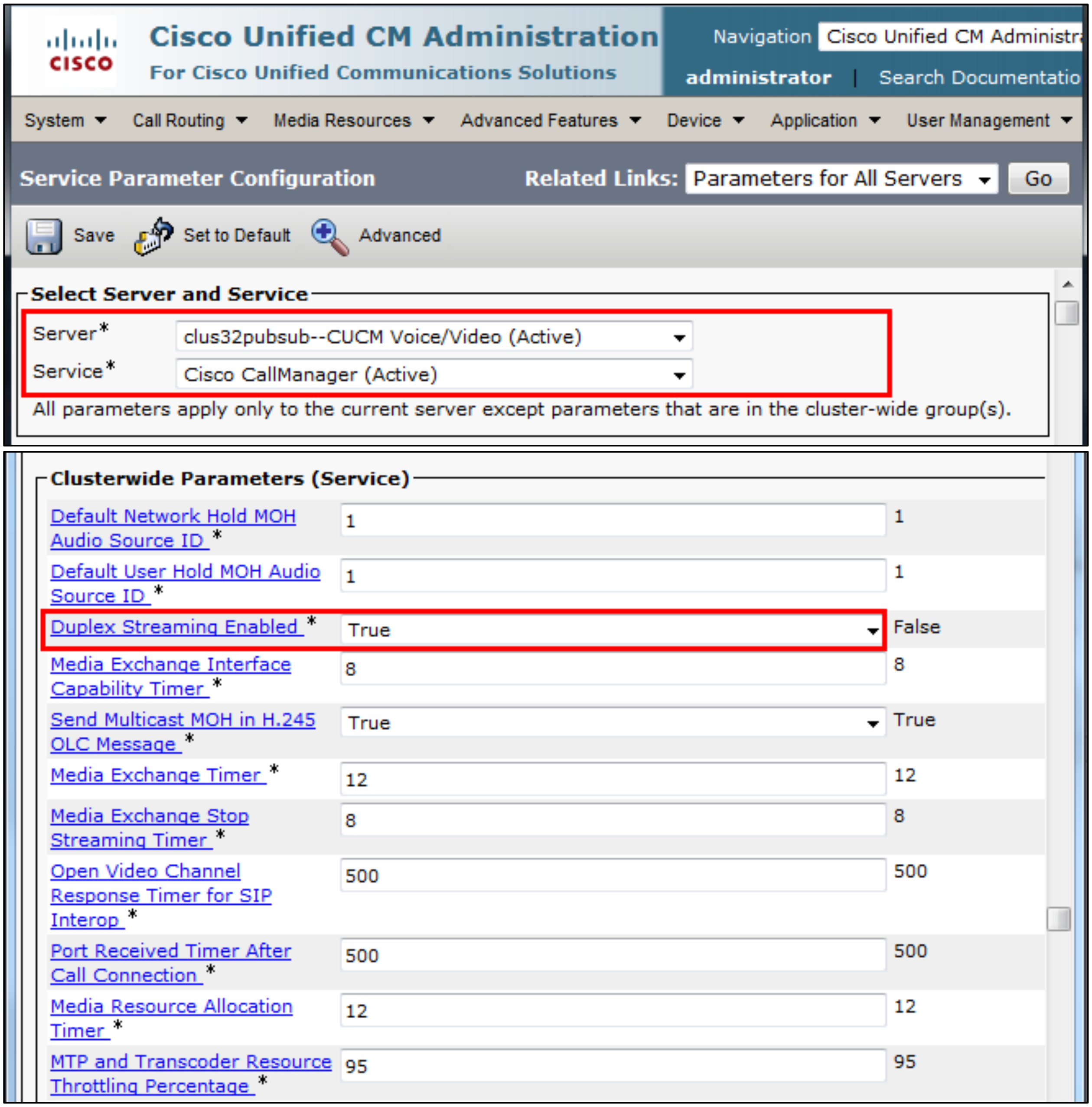
Music on Hold Service (Duplex Streaming) Parameter Settings

Navigation: System → Service Parameter

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



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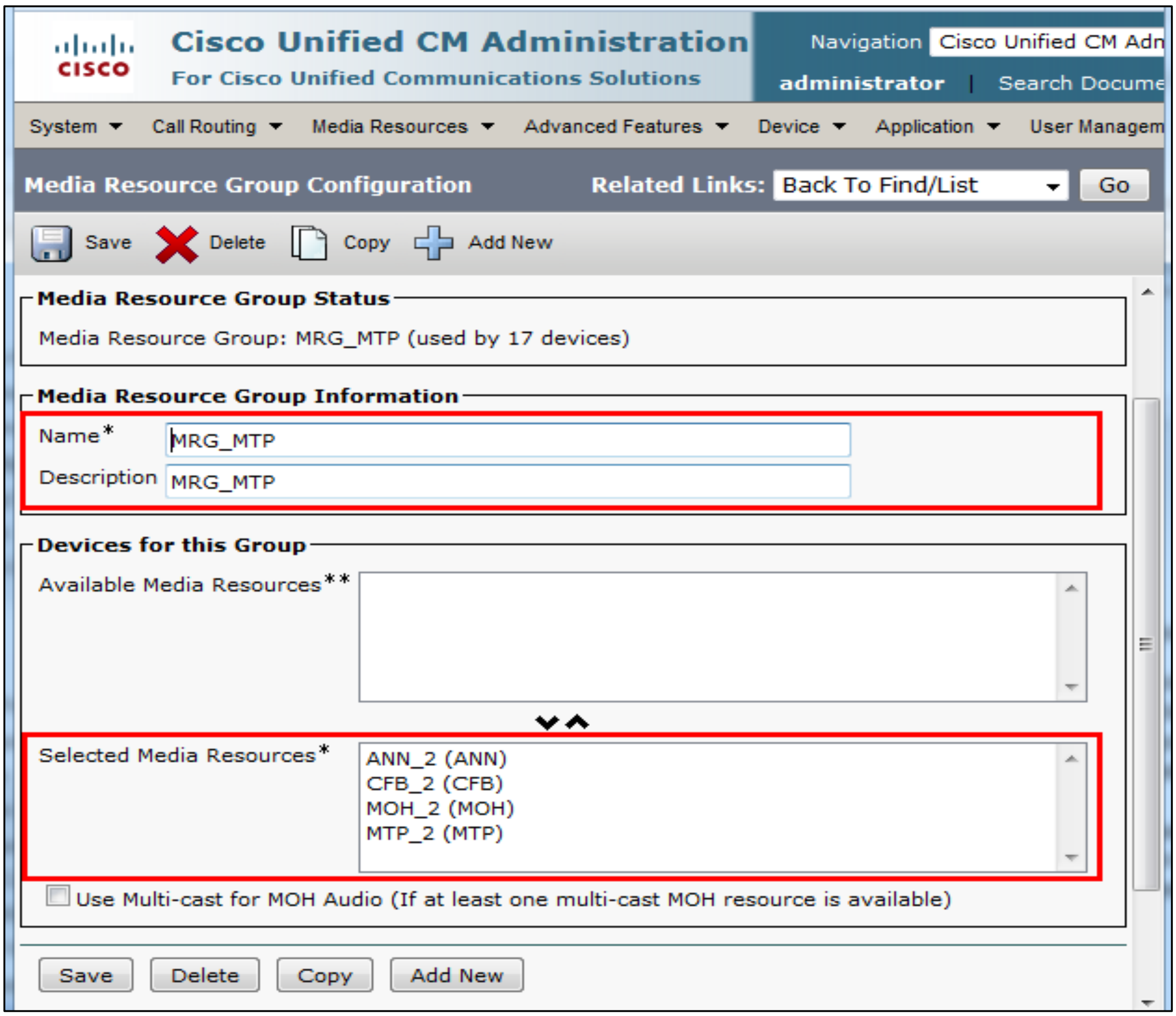
Media Resource Group Configuration

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.



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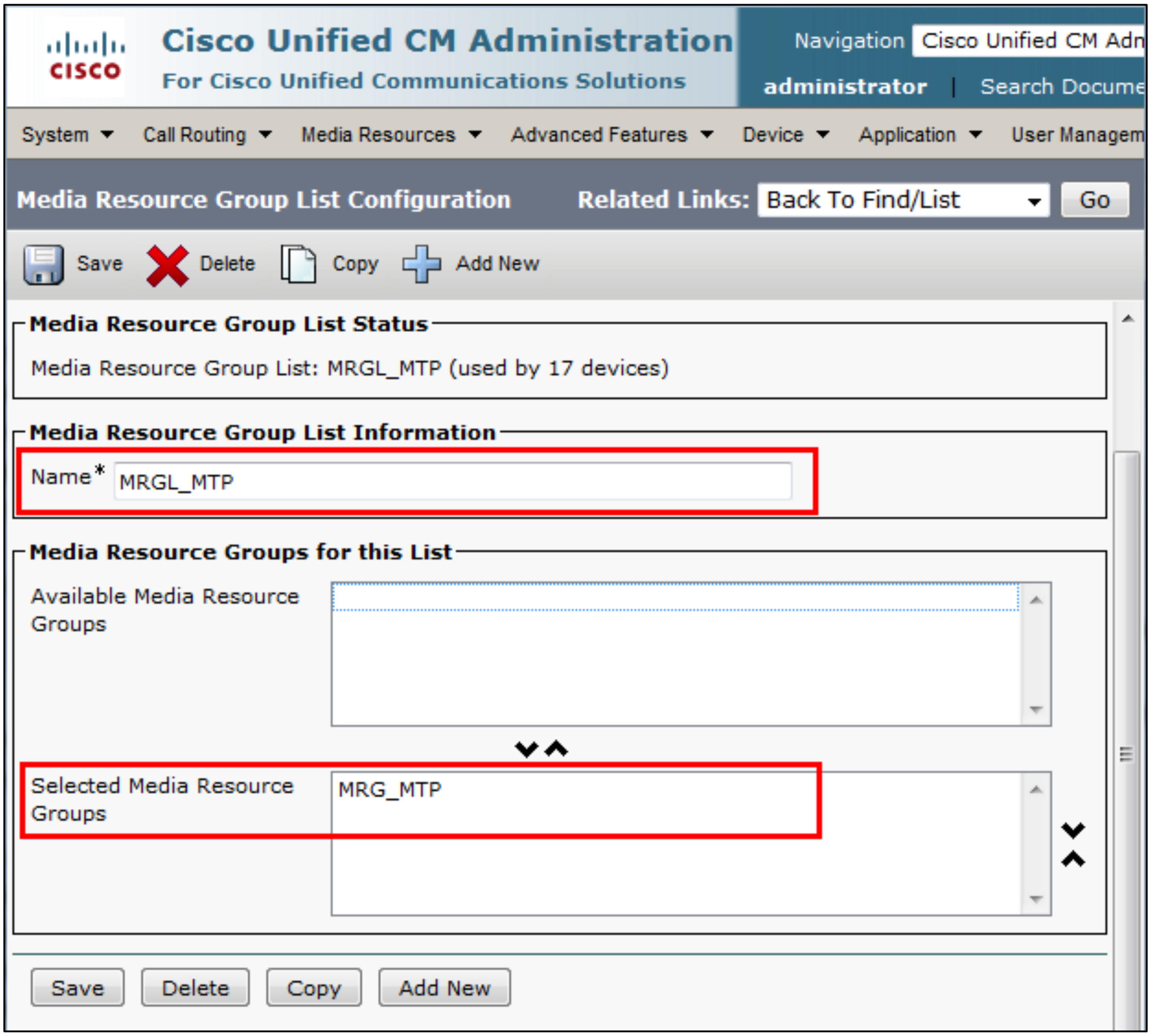
Media Resource Group List Configuration

Navigation Path: Media Resources → Media Resource Group List

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



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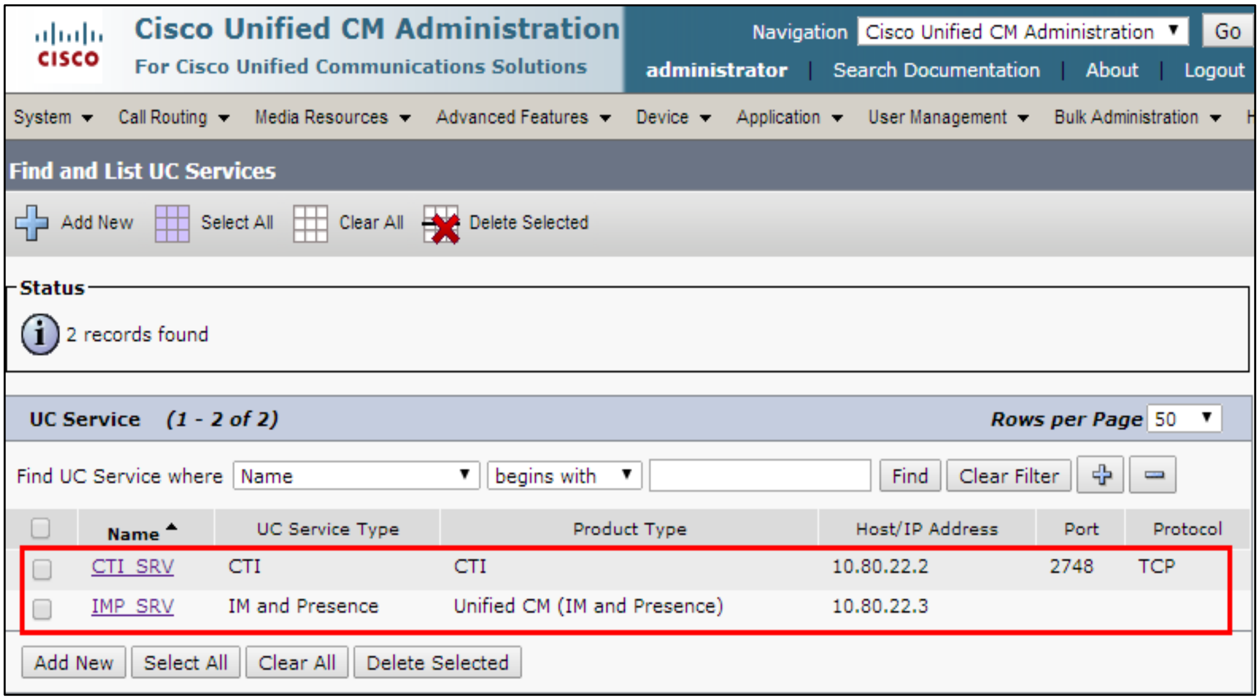
UC Service Configuration

Navigation: User Management → User Settings → UC Service



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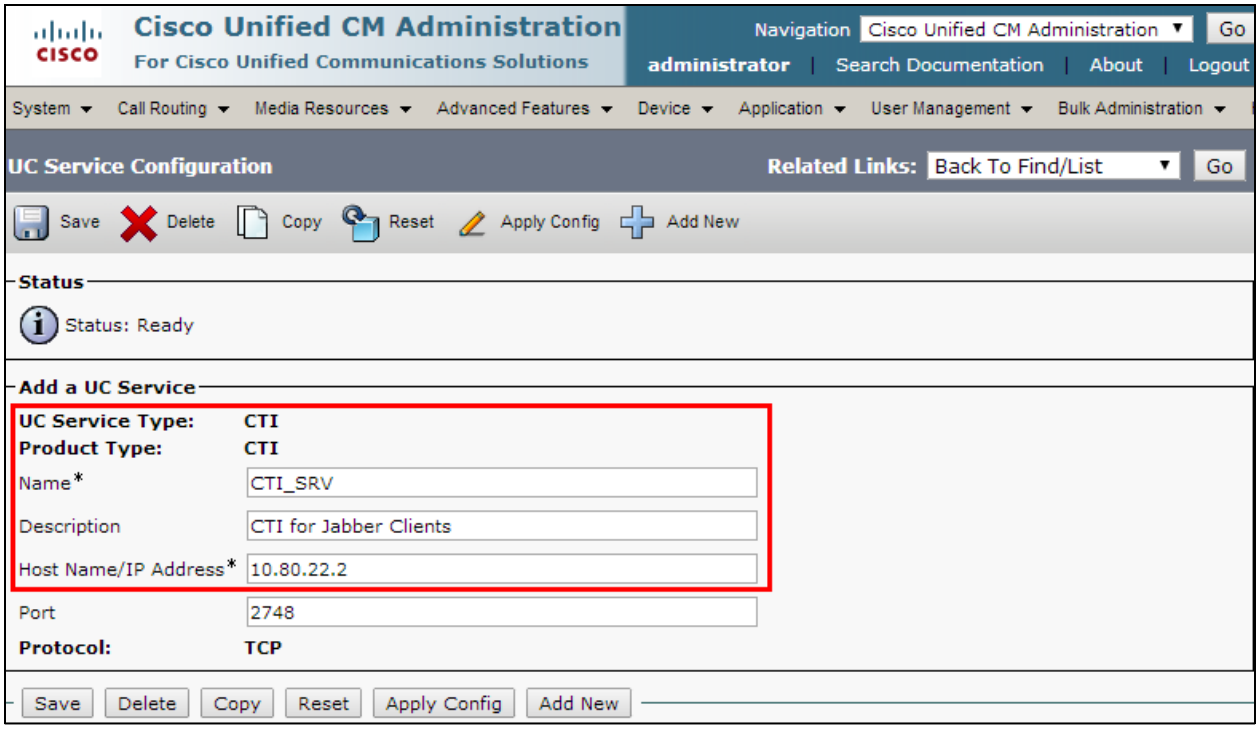
UC Service Configuration (Contd…)

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



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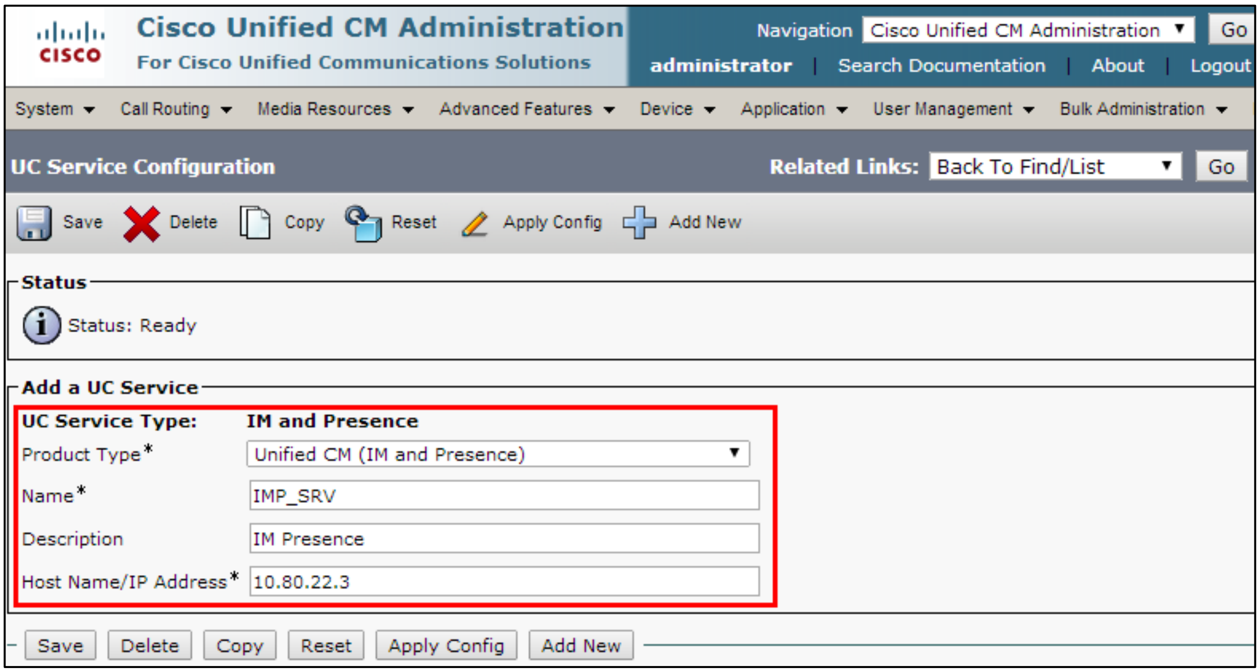
Note: Testing was conducted in tekVizion labs

UC Service Configuration (Contd…)

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



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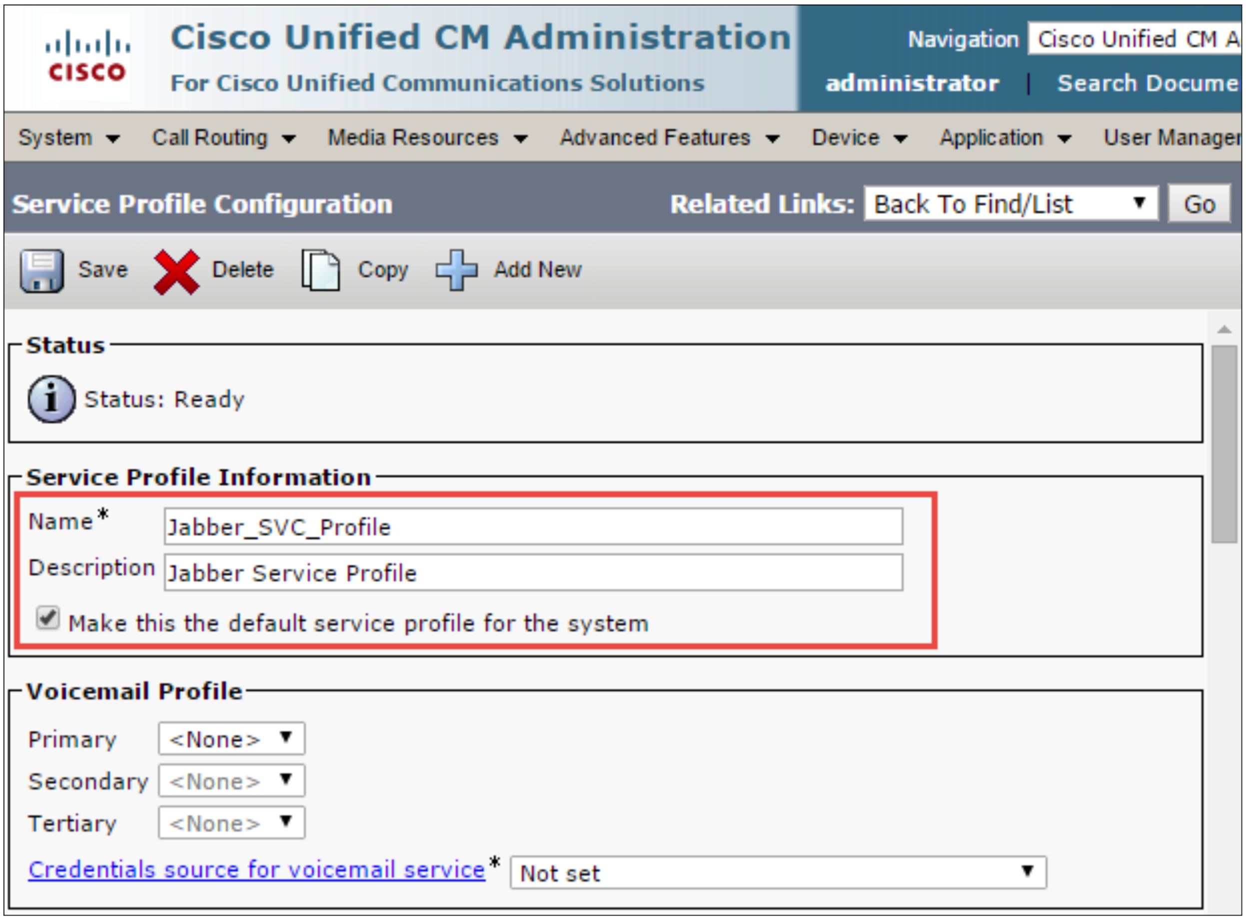
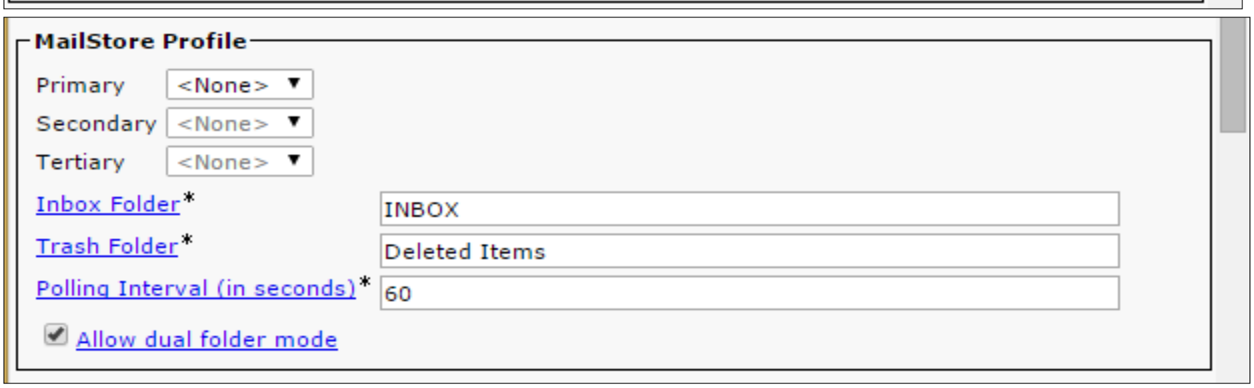
Service Profile Configuration

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.





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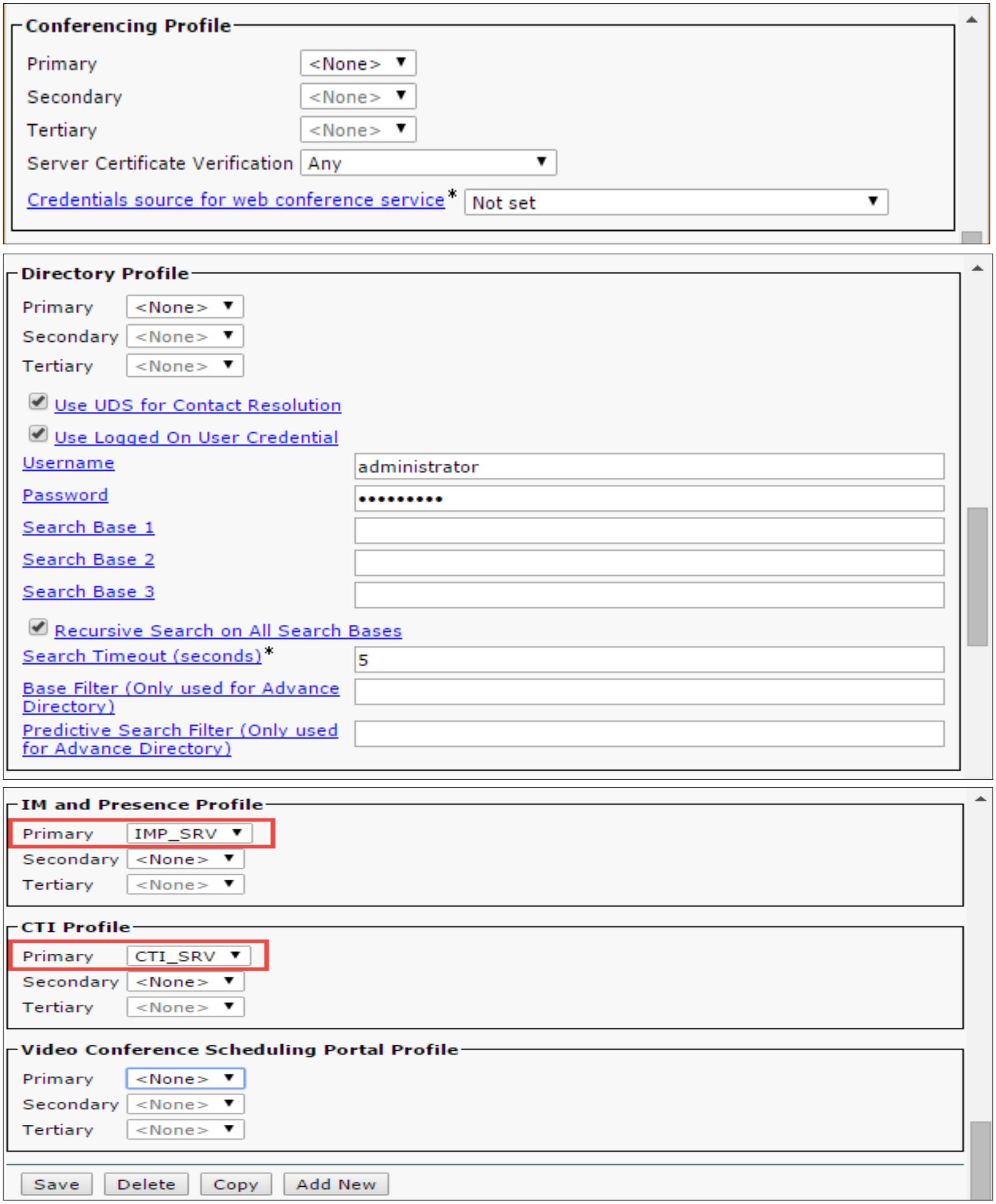
Service Profile Configuration (Contd…)



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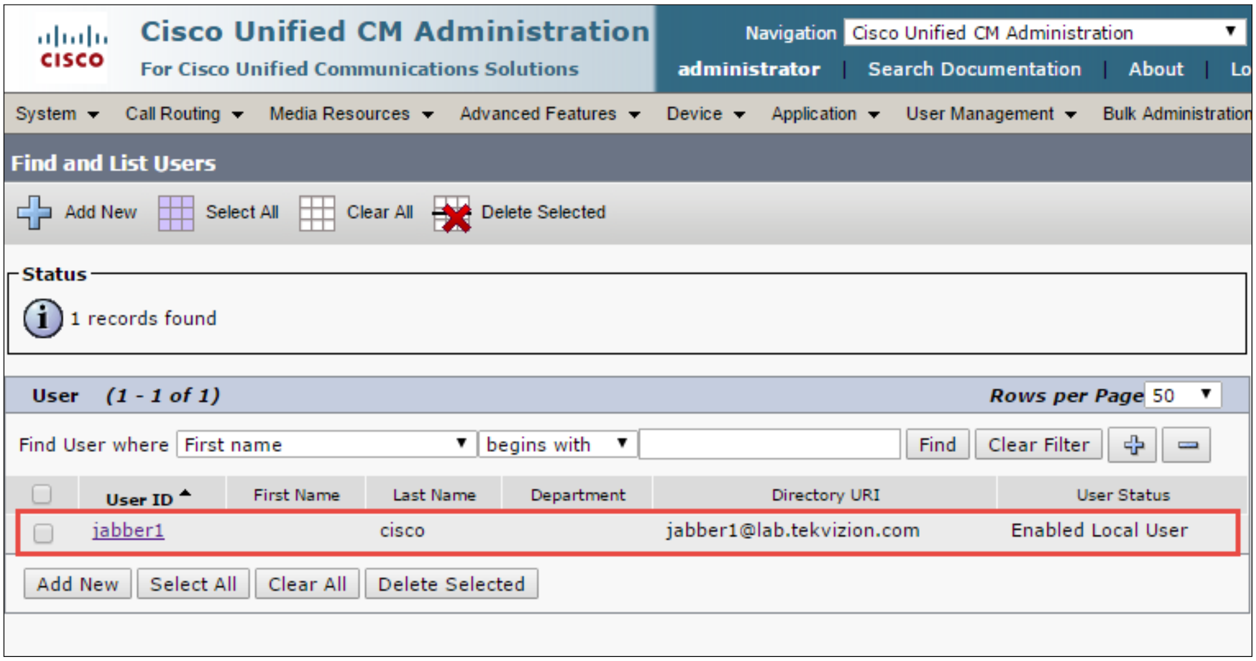
End User Configuration

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.



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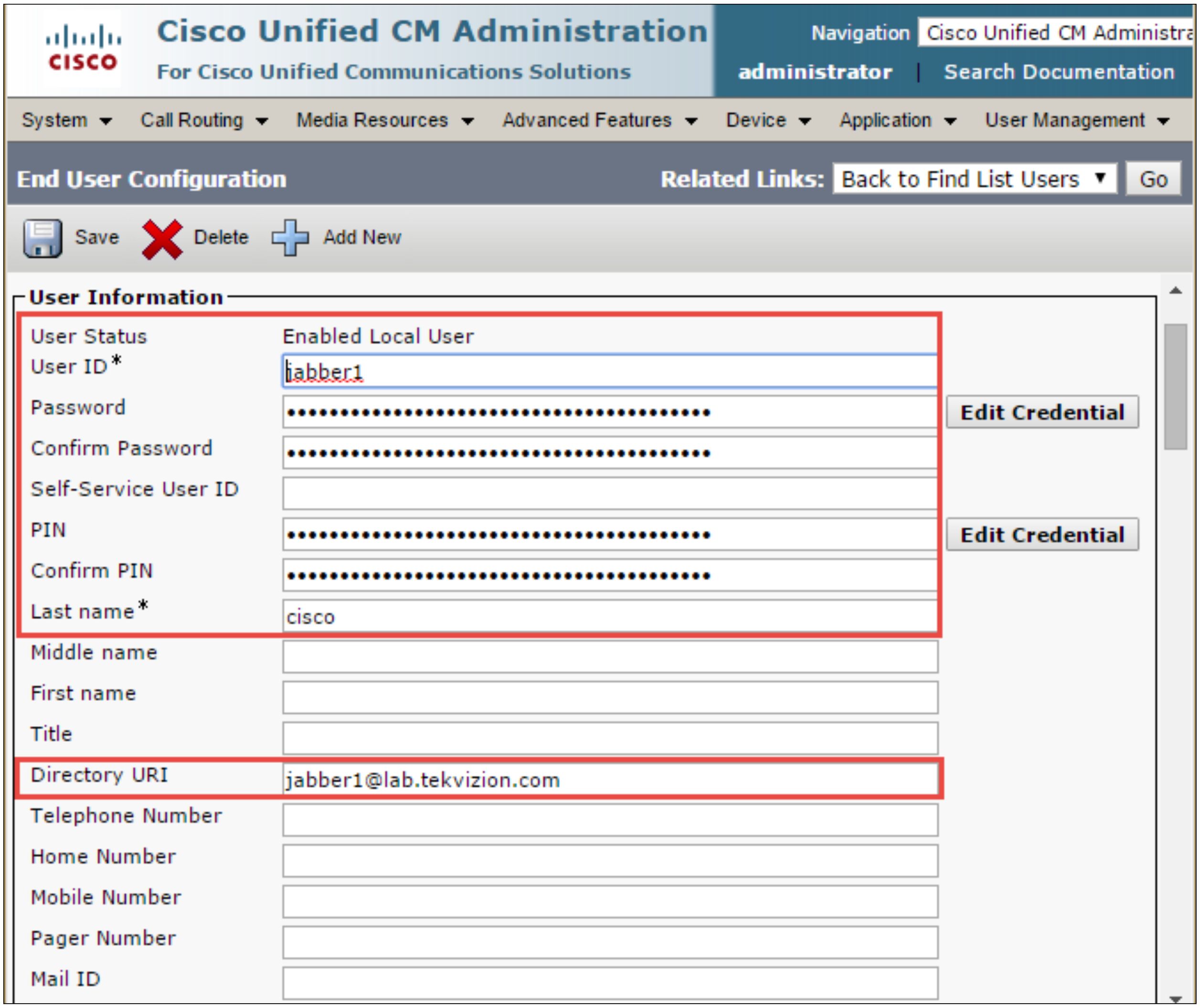
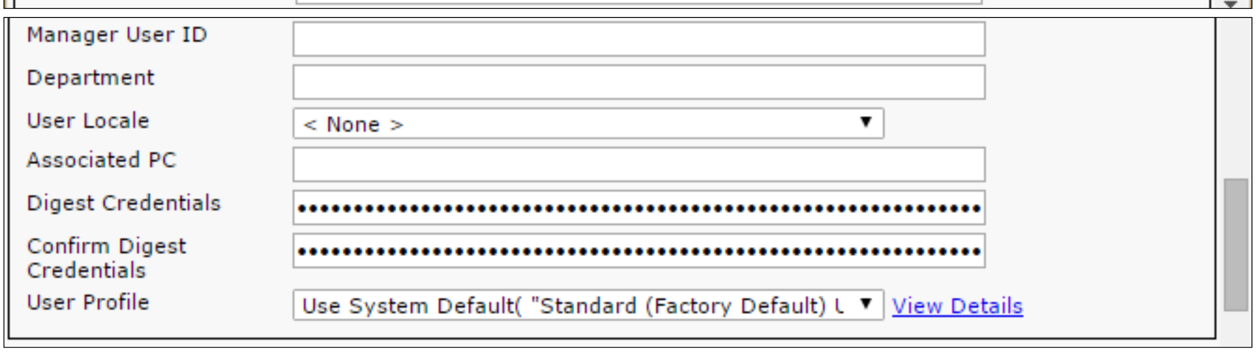
End User Configuration (continued…)





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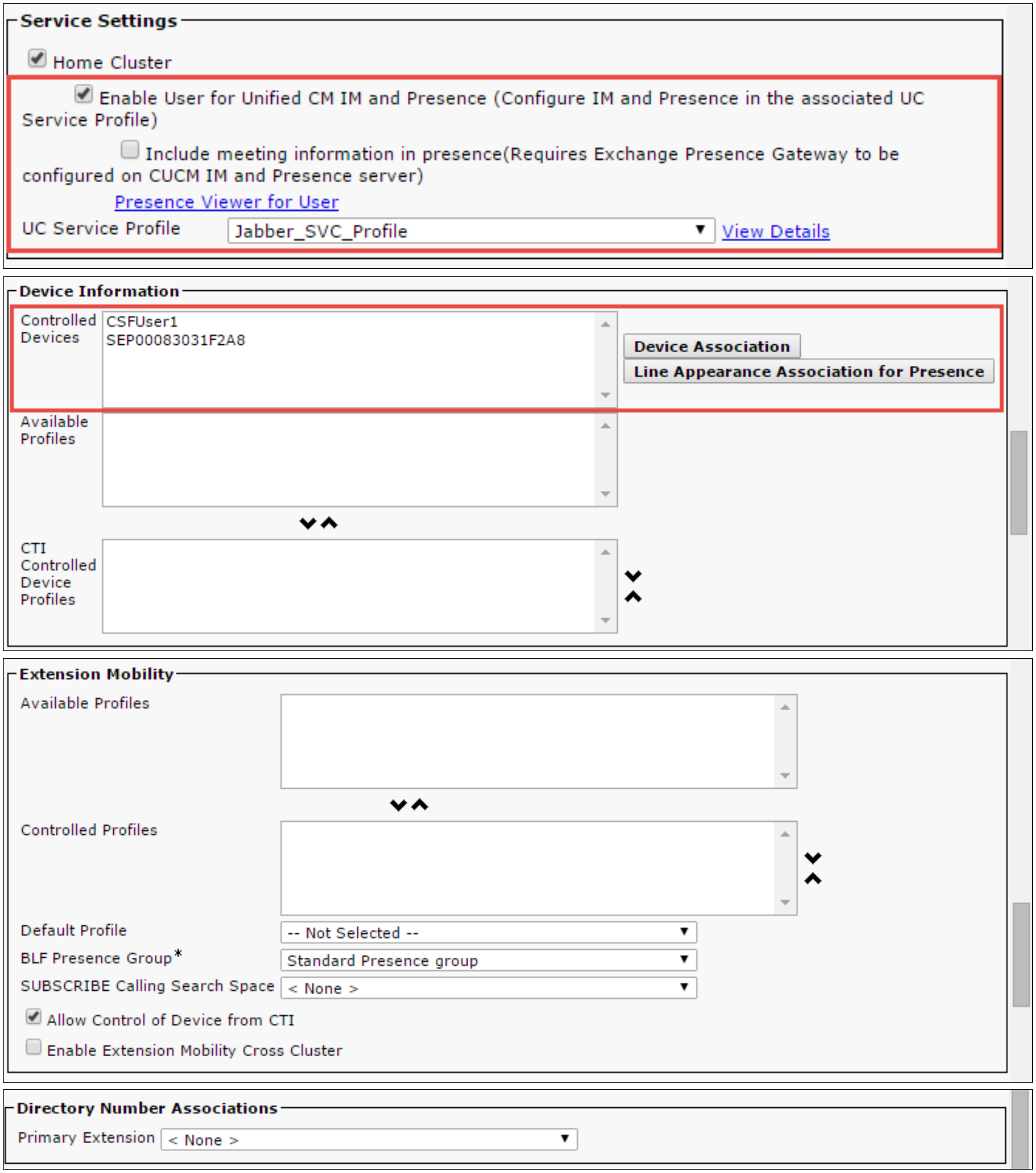
End User Configuration (continued…)



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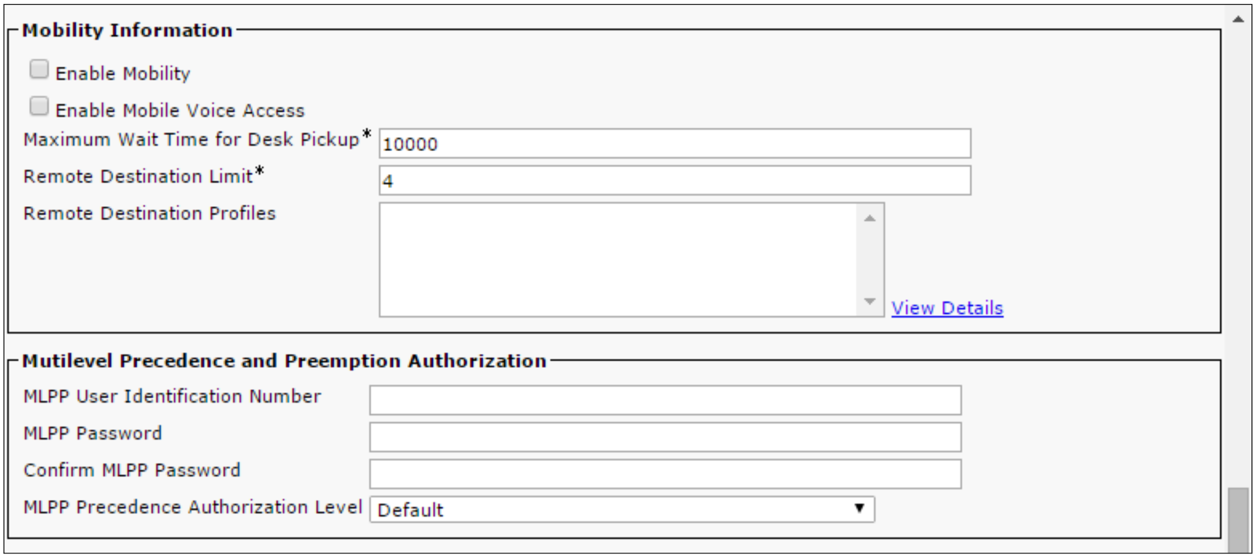
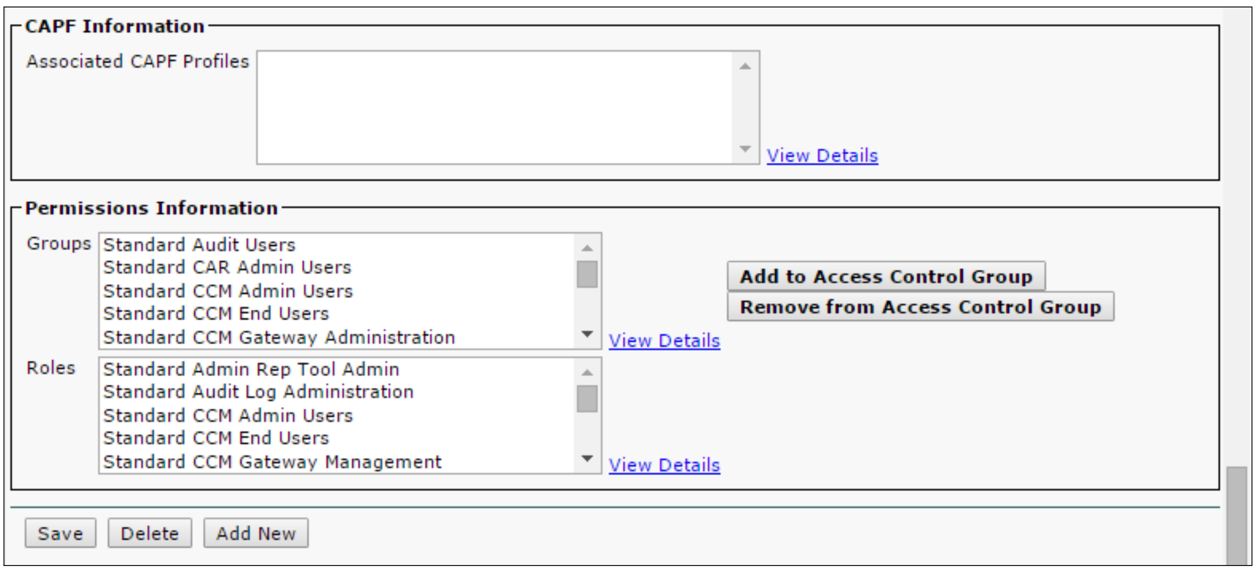
End User Configuration(continued…)





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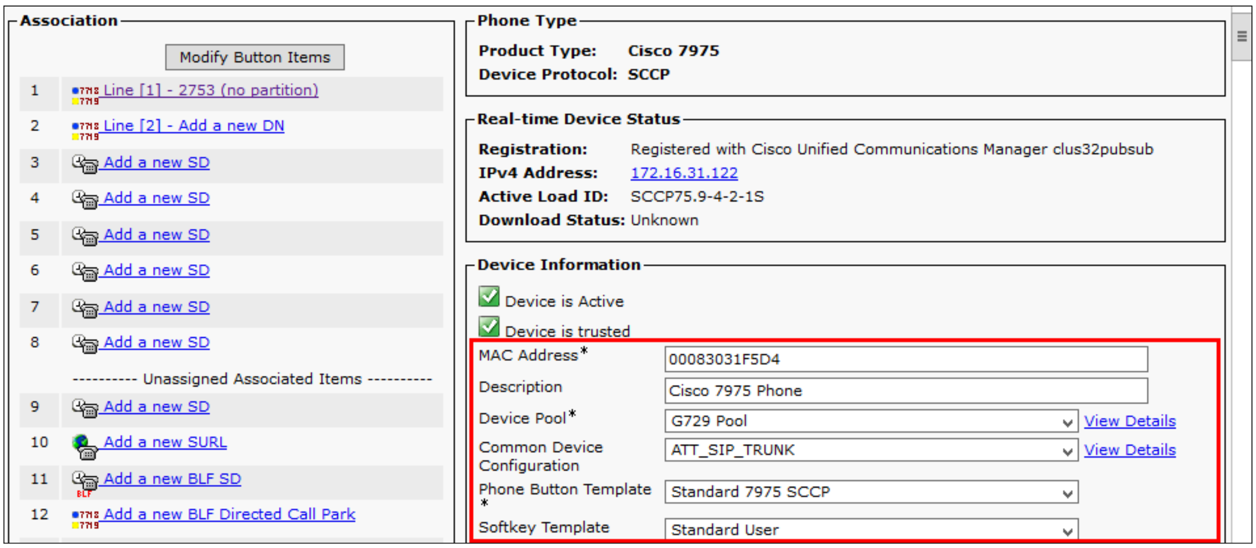
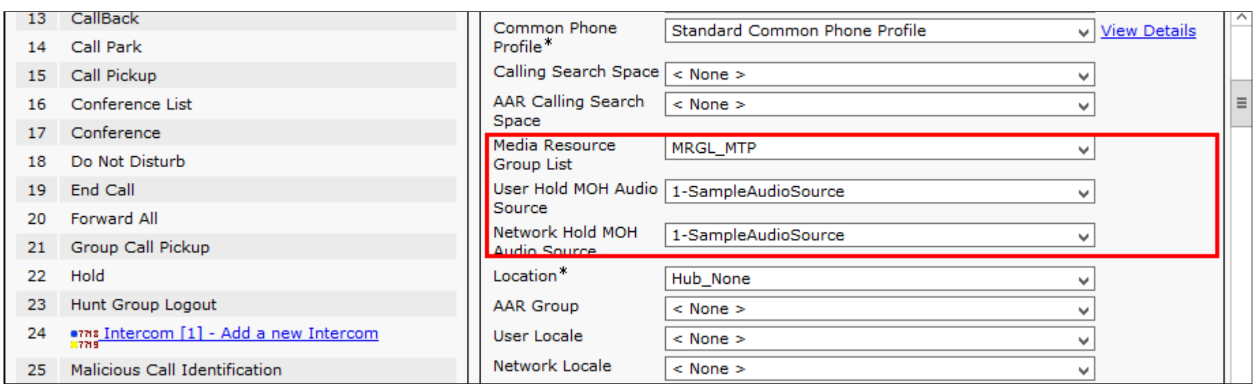
Cisco IP Phone 7975 SCCP Configuration

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





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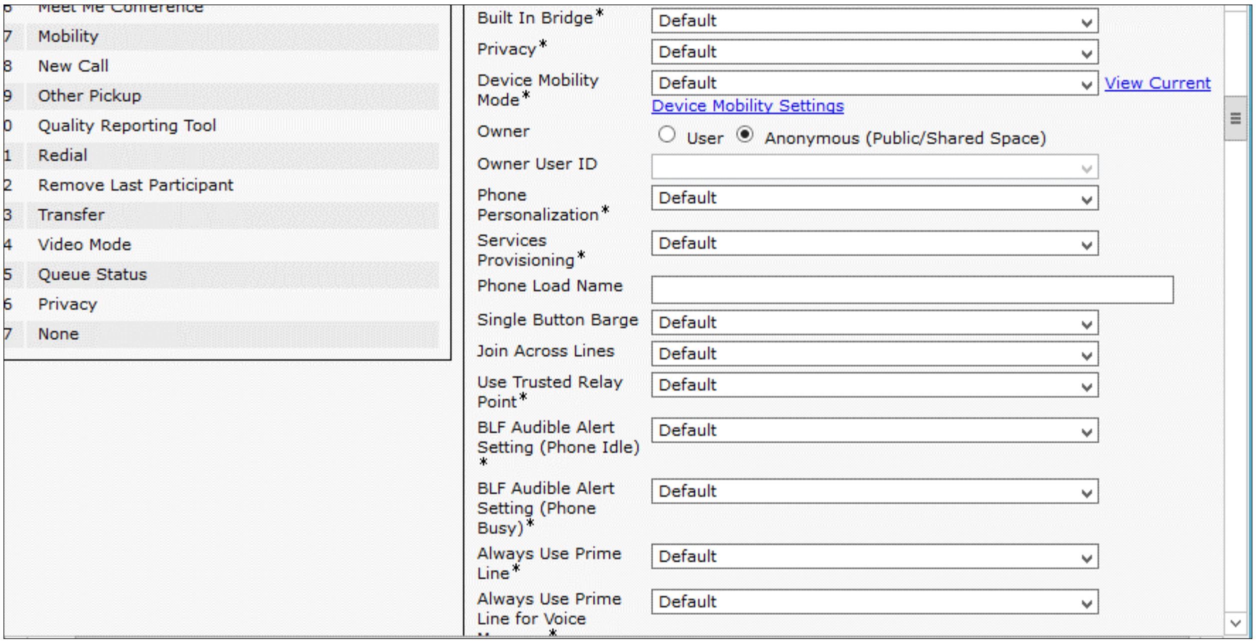
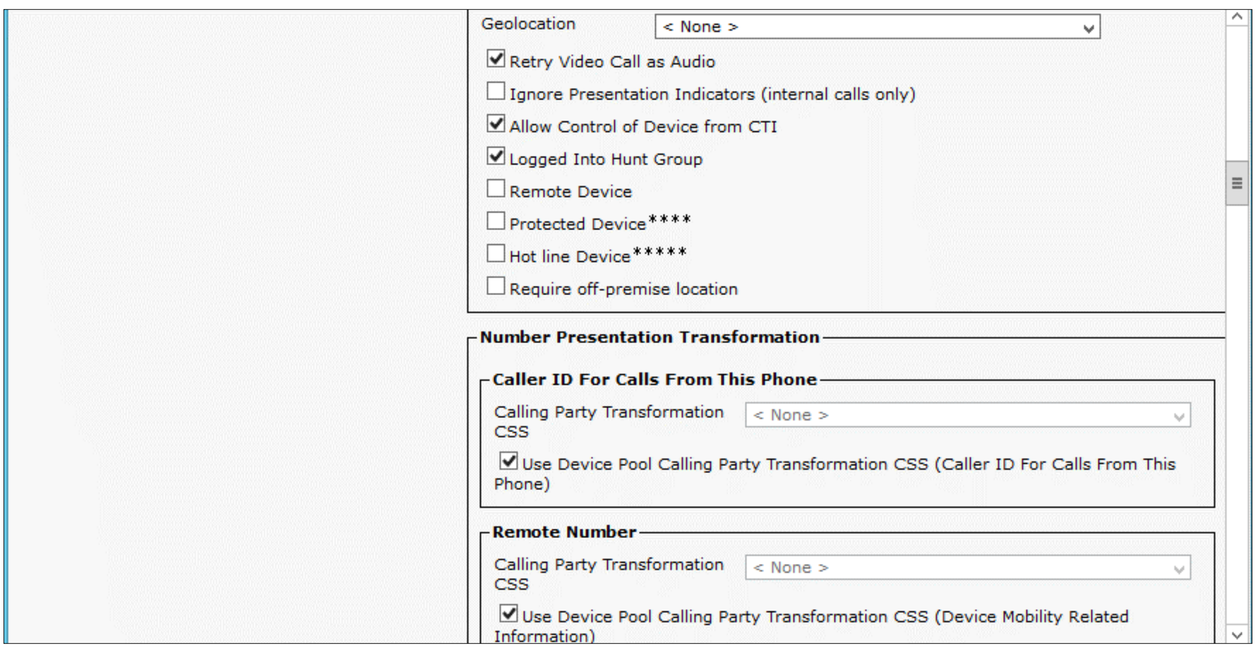
Cisco IP Phone 7975 SCCP Configuration (Continued…)





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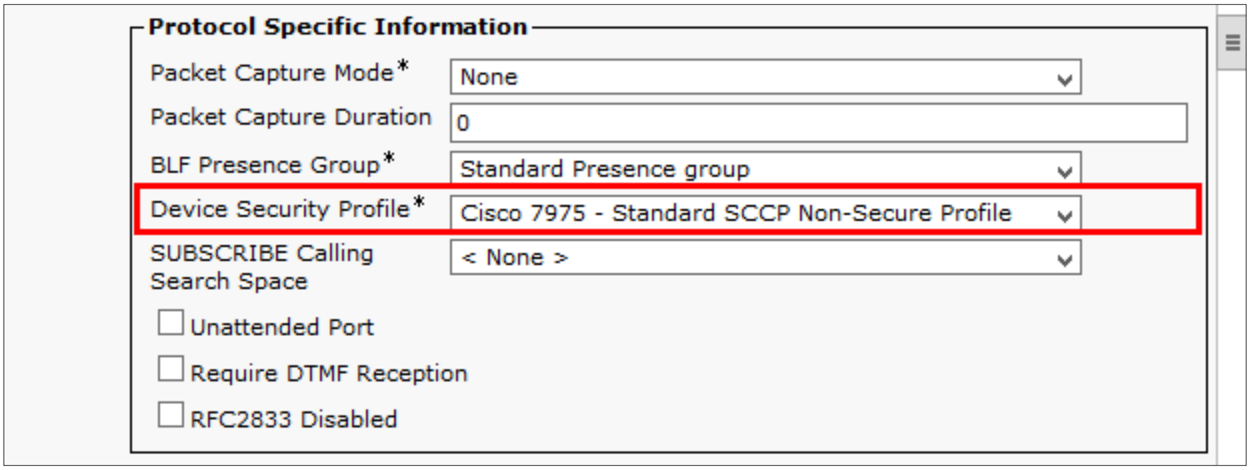
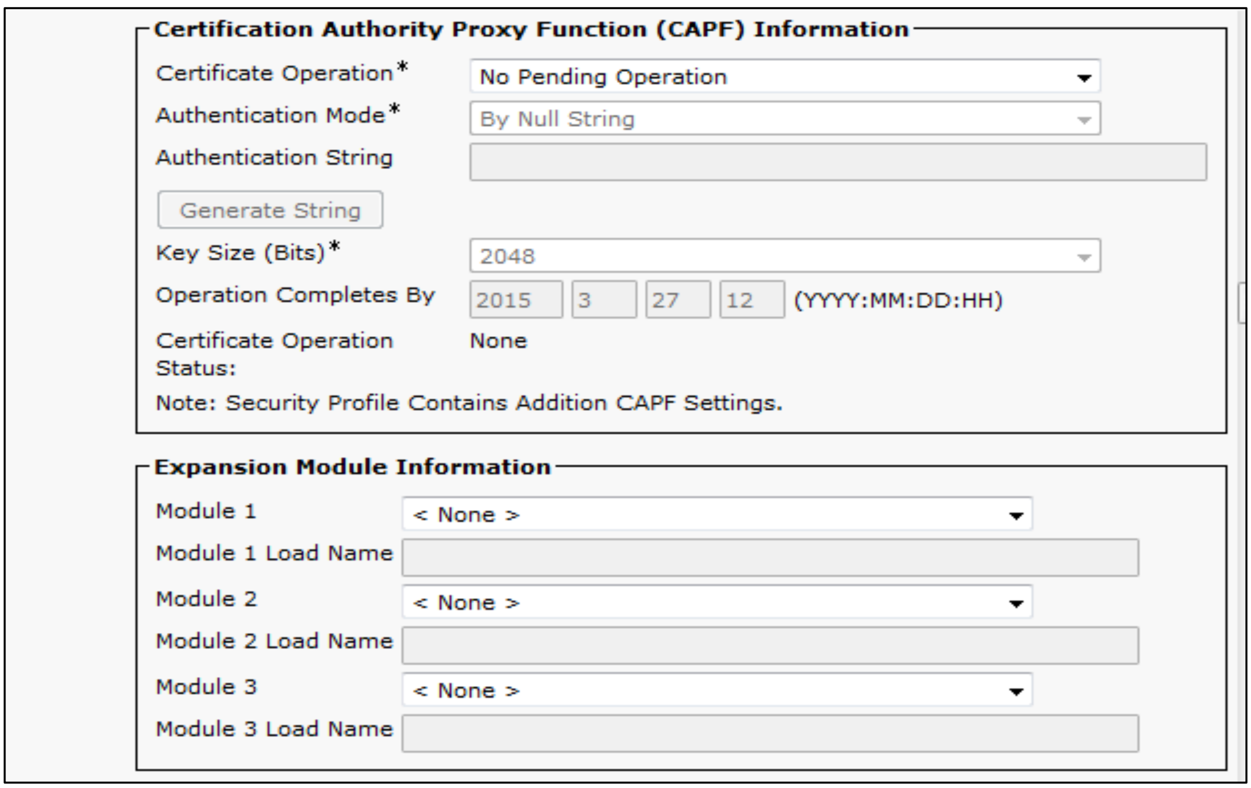
Cisco IP Phone 7975 SCCP Configuration (Continued…)





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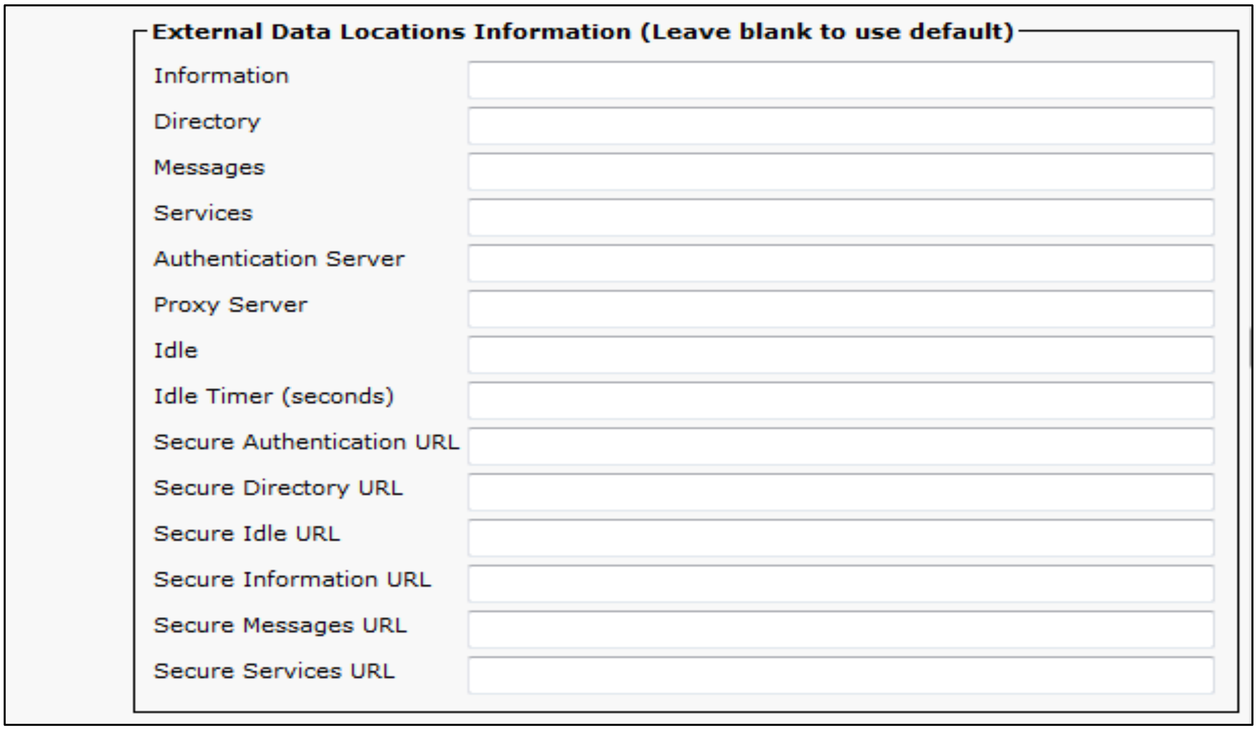
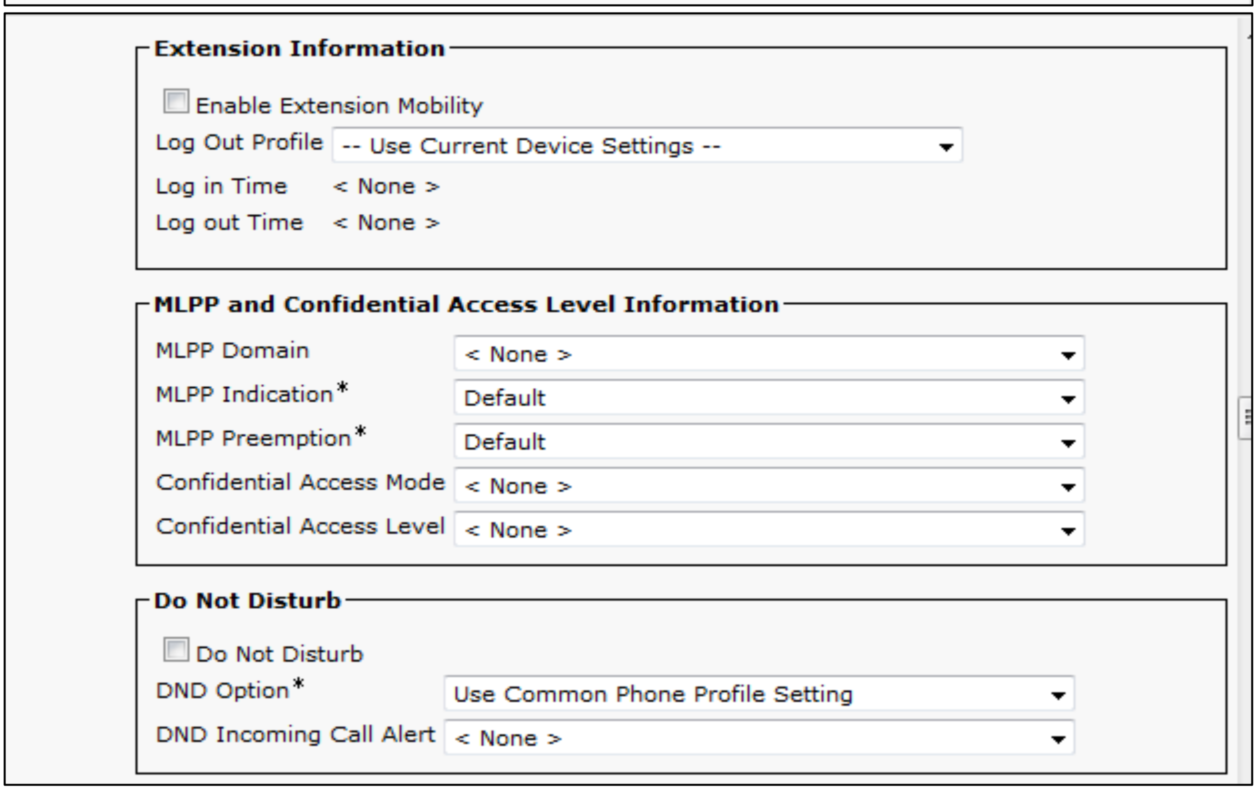
Cisco IP Phone 7975 SCCP Configuration (Continued…)





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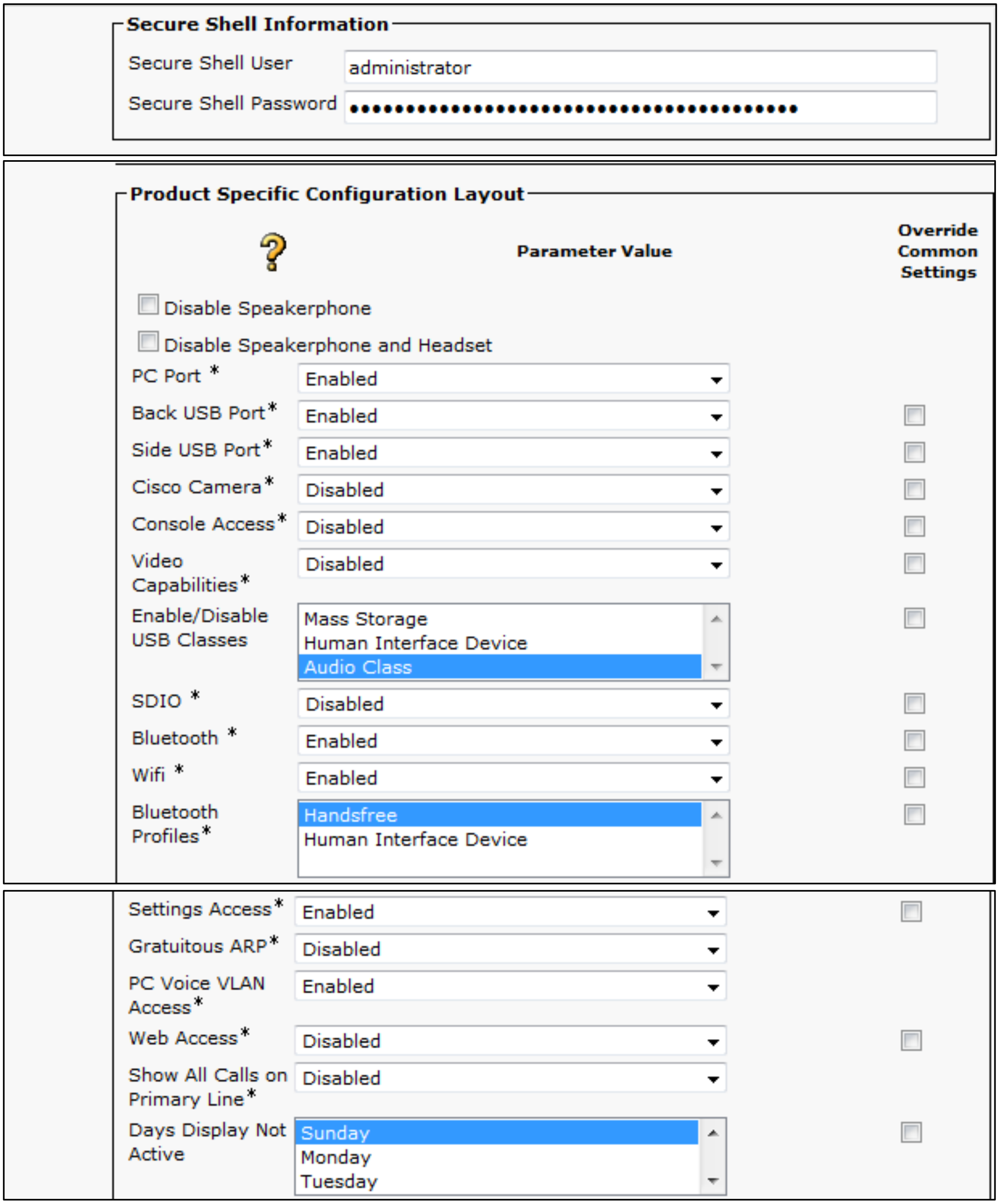
Cisco IP Phone 7975 SCCP Configuration (Continued…)



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Cisco IP Phone 7975 SCCP Configuration (Continued…)

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

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

Cisco IP Phone 7975 SCCP Configuration (Continued…)

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

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

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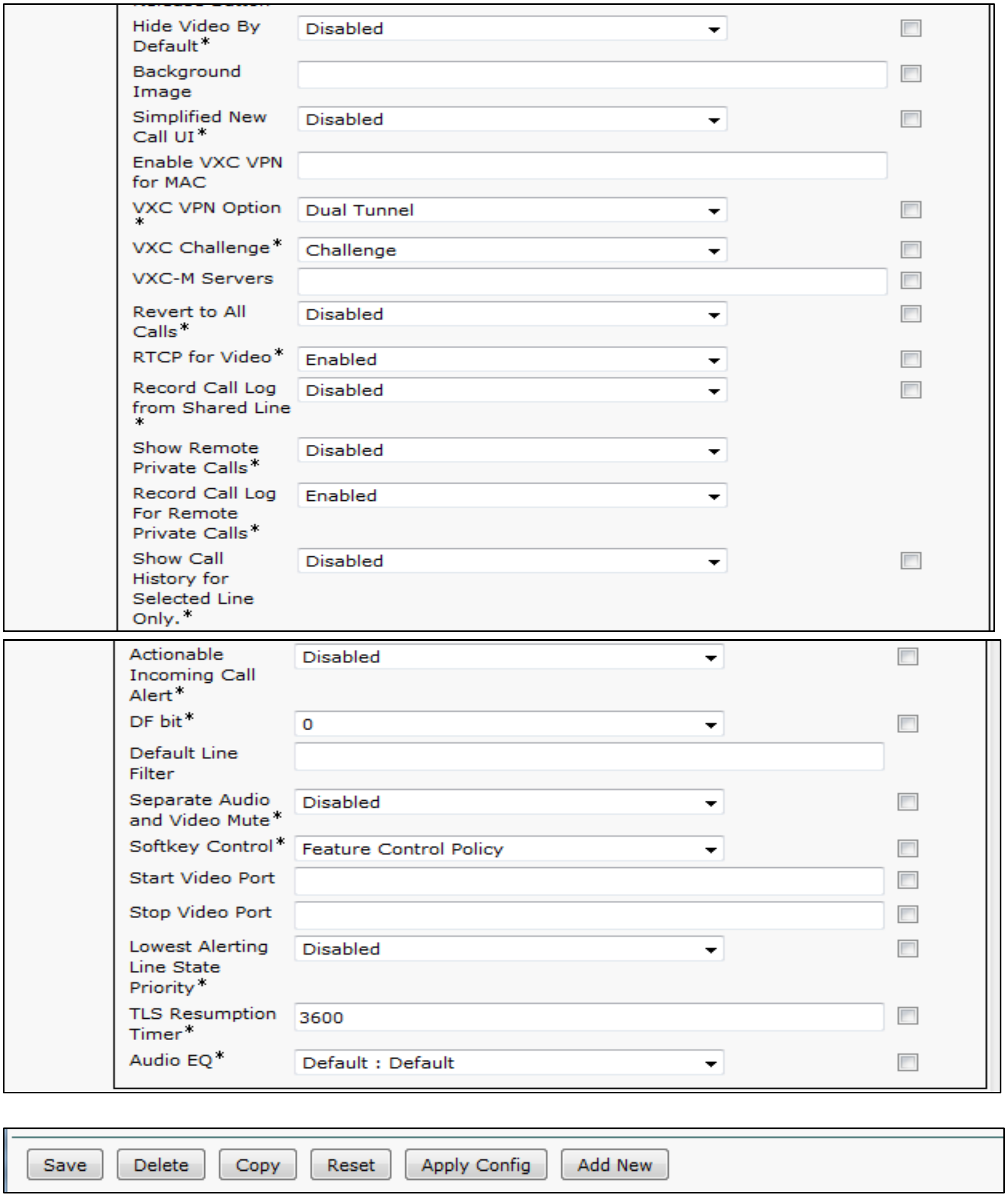
Cisco IP Phone 7975 SCCP Configuration (Continued…)



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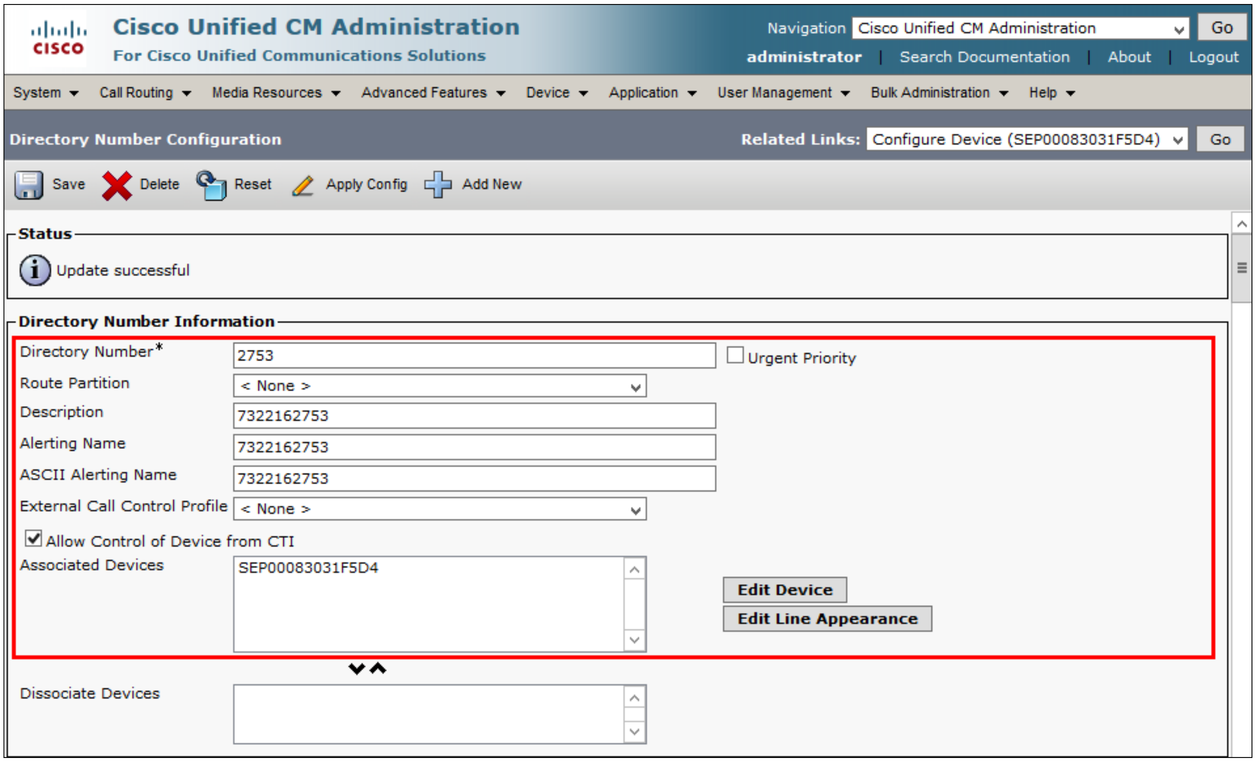
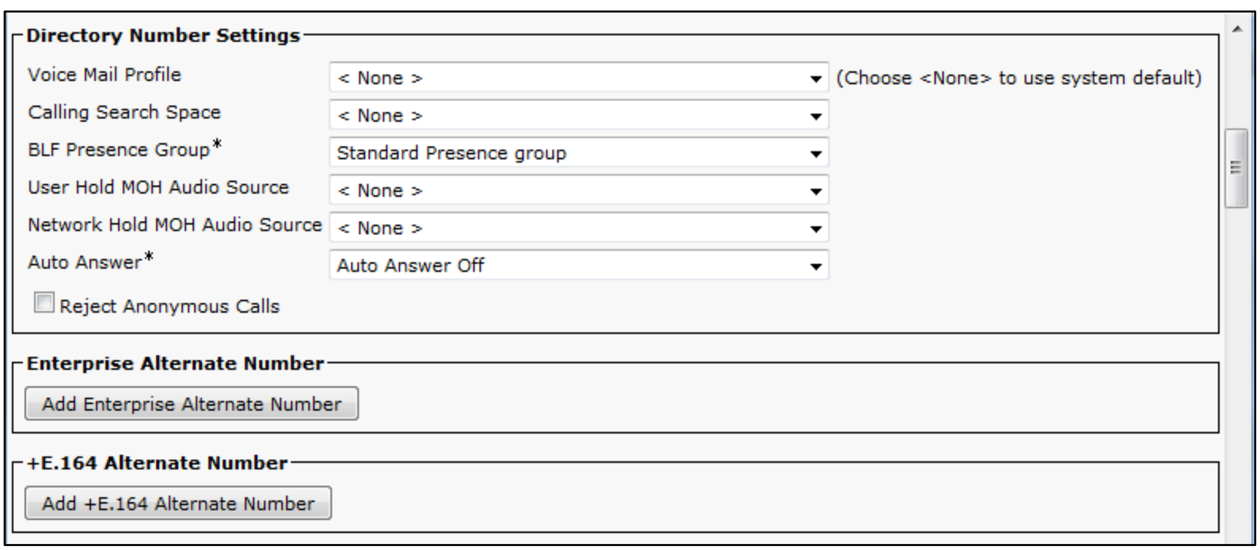
Cisco IP Phone 7975 SCCP Configuration (Continued…)

Set Directory Number* = 2753. This is used in this example.
Set Description = 7322162753. This is used in this example.
Set Alerting Name = 7322162753. This is used in this example.
Set ASCII Alerting Name = 7322162753. This is used in this example.





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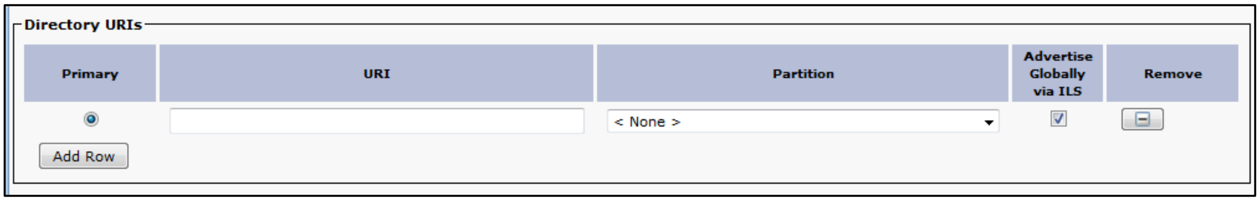
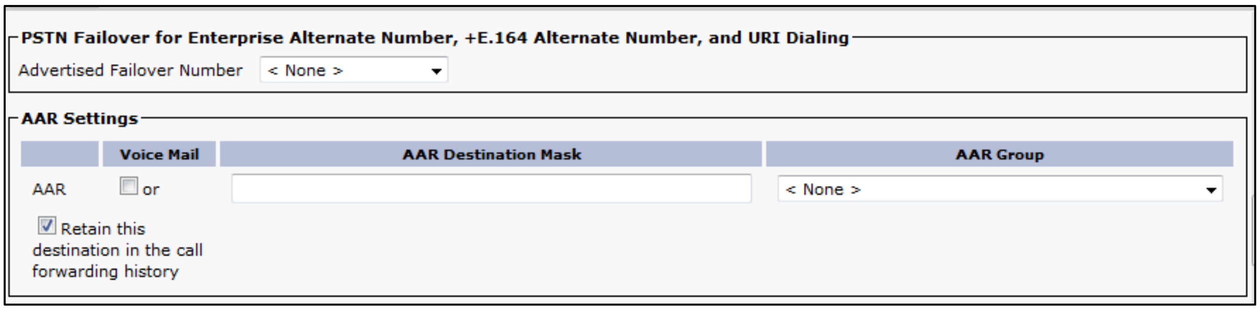
Cisco IP Phone 7975 SCCP Configuration (Continued…)



Directory URIs

Primary	URI	Partition	Advertise Globally via ILS	Remove
◉		< None >	☑	⊟

Add Row



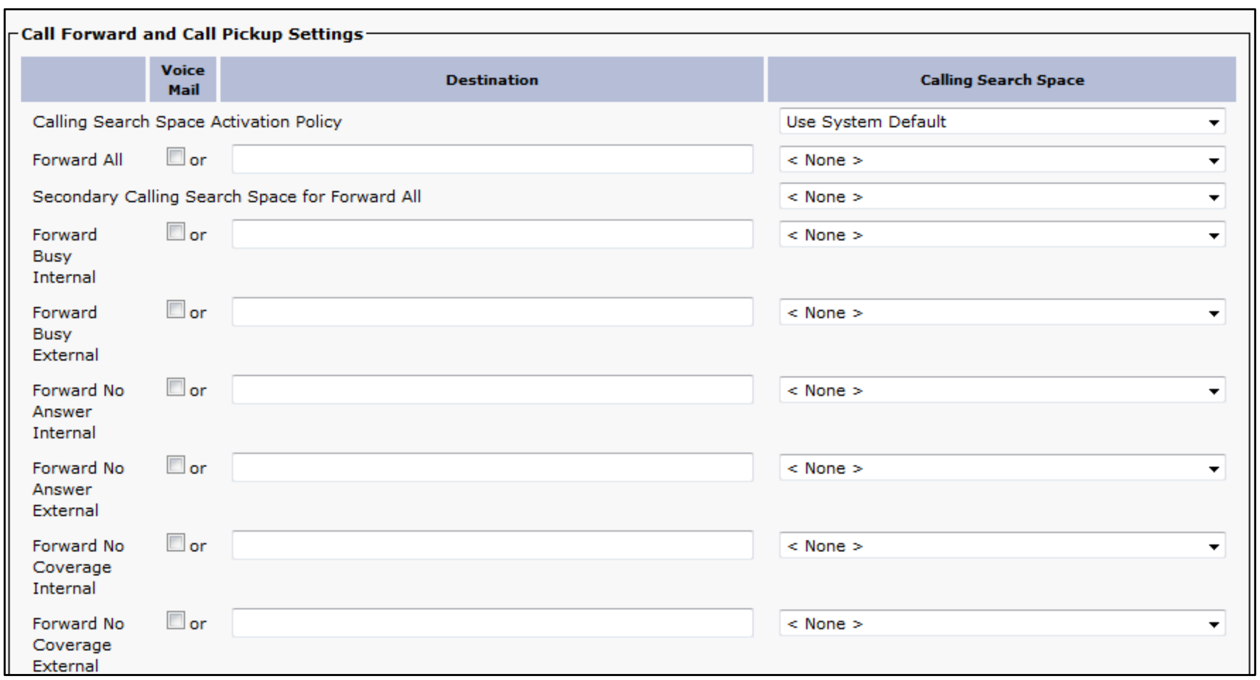
PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing

Advertised Failover Number < None >

AAR Settings

	Voice Mail	AAR Destination Mask	AAR Group
AAR	☐ or		< None >

☑ Retain this destination in the call forwarding history



Call Forward and Call Pickup Settings

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

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Cisco IP Phone 7975 SCCP Configuration (Continued…)

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 >

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

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Cisco IP Phone 7975 SCCP Configuration (Continued…)

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

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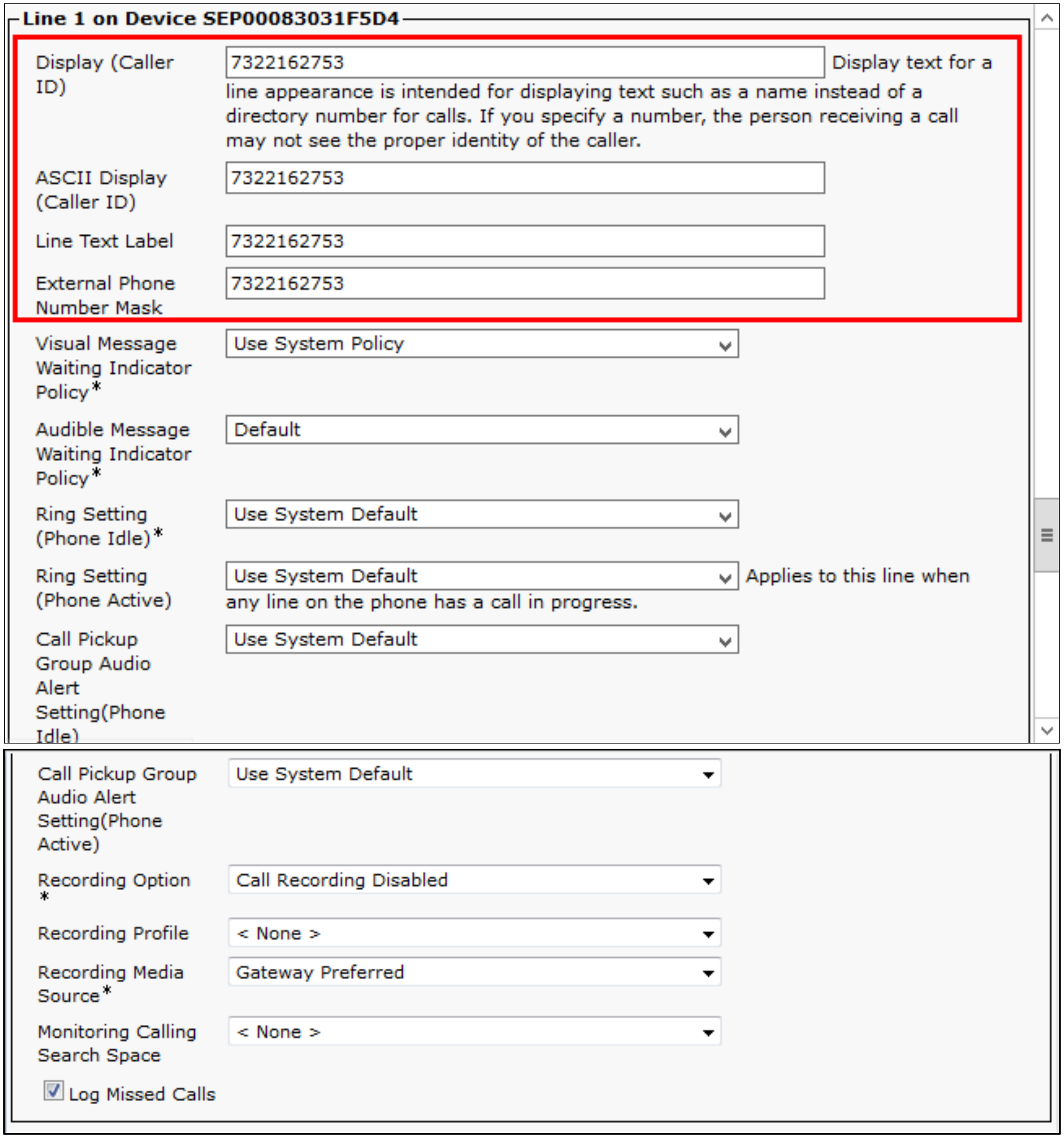

Cisco IP Phone 7975 SCCP Configuration (Continued…)

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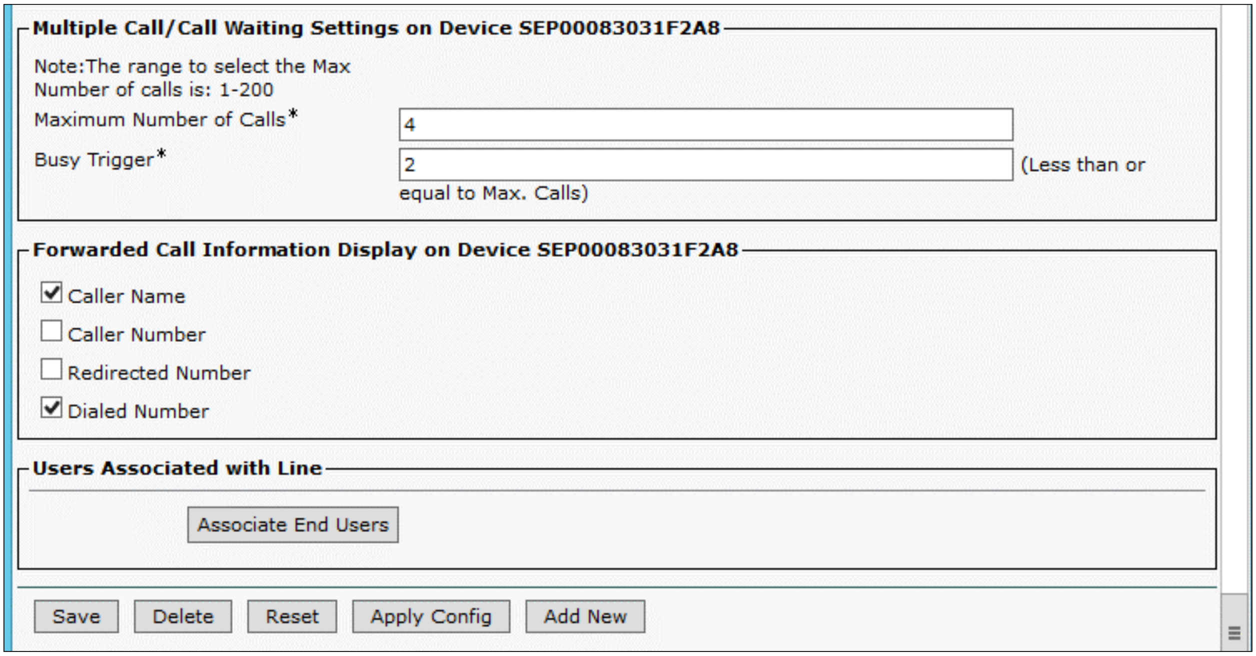
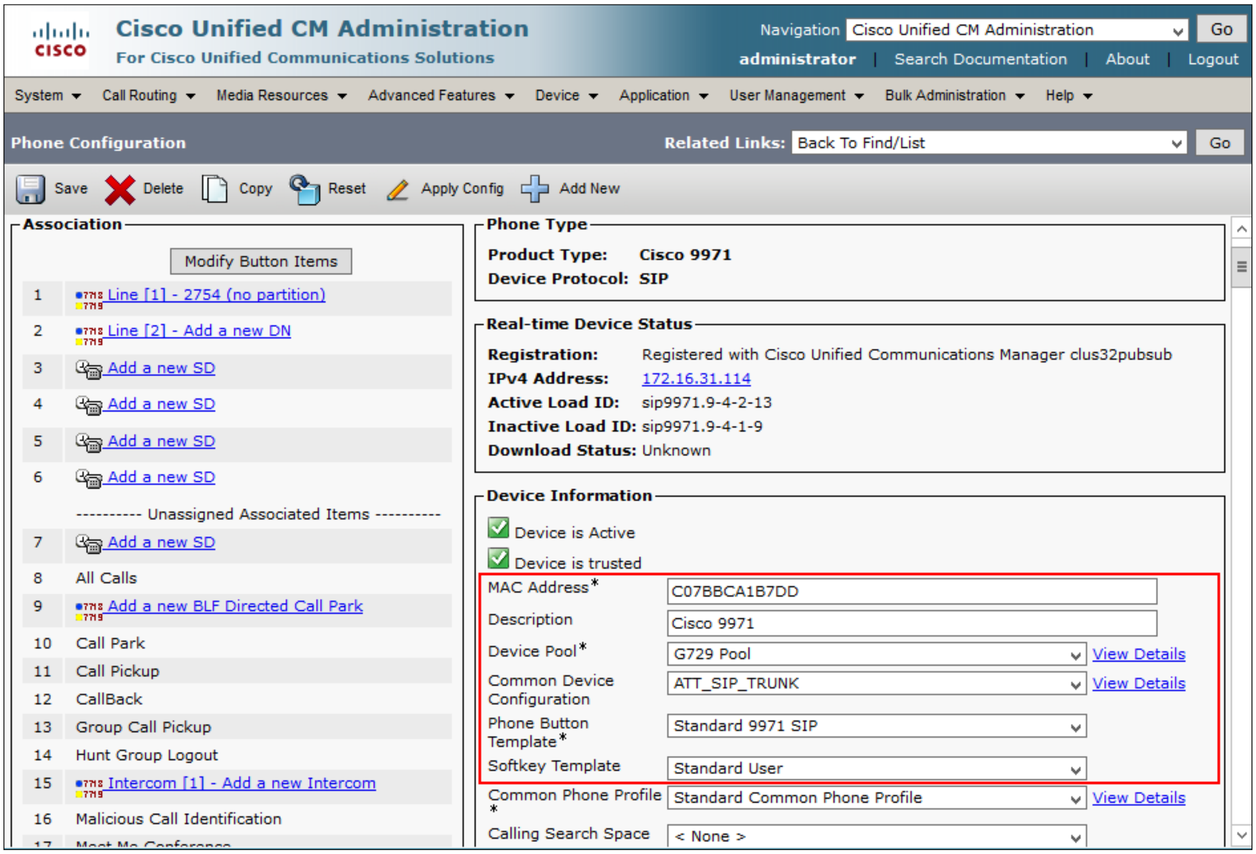
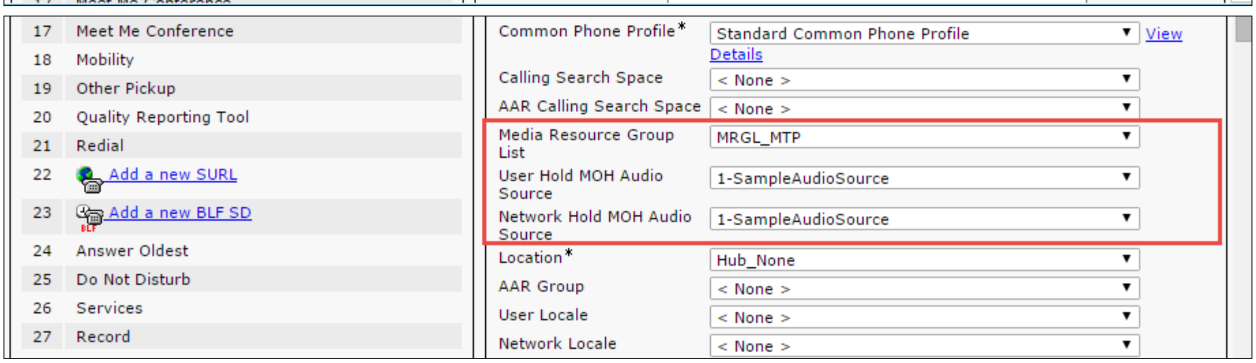


Cisco IP Phone 9971 SIP Configuration

Set MAC Address* = the below mac is used in this example.
Set Description = Cisco 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 Media Resource Group List = MRGL_MTP. This is used in this example.
Set User Hold MOH Audio Source = 1-SampleAudioSource.
Set Network Hold MOH Audio Source = 1-SampleAudioSource





Cisco IP Phone 9971 SIP Configuration (Continued…)

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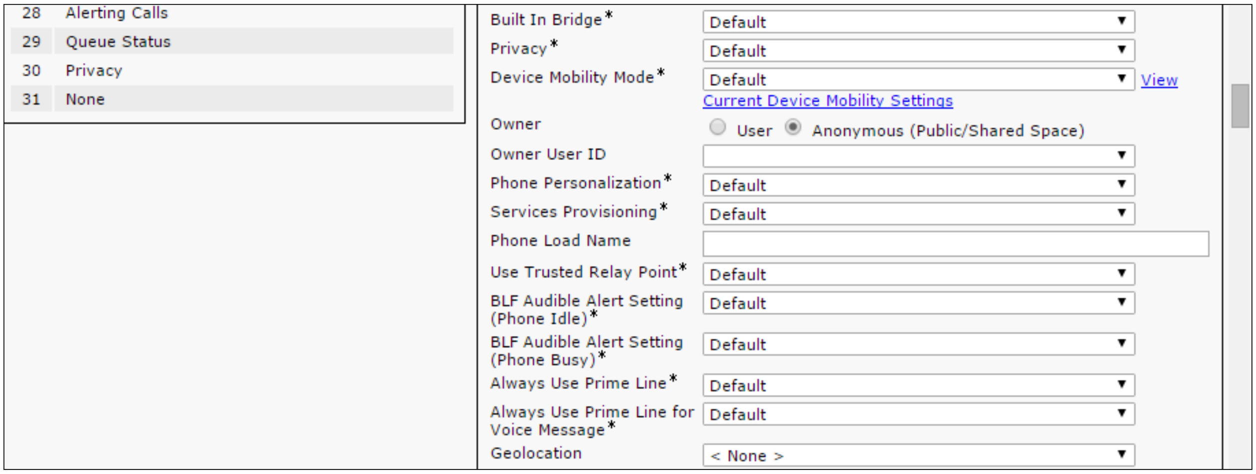

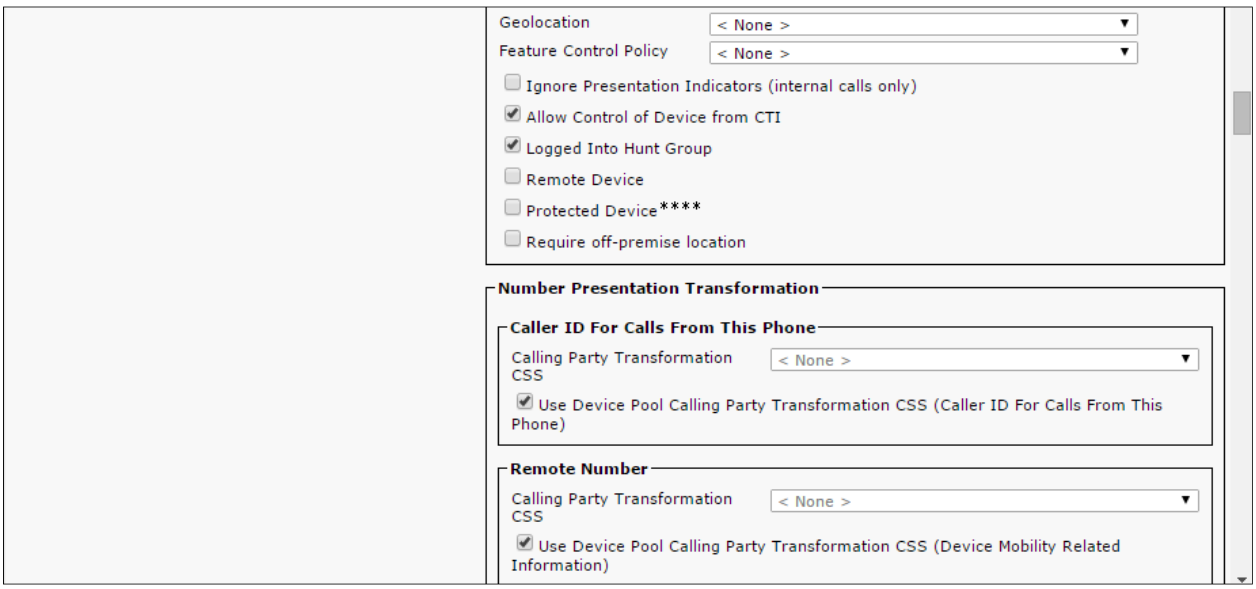

Cisco IP Phone 9971 SIP Configuration (Continued…)

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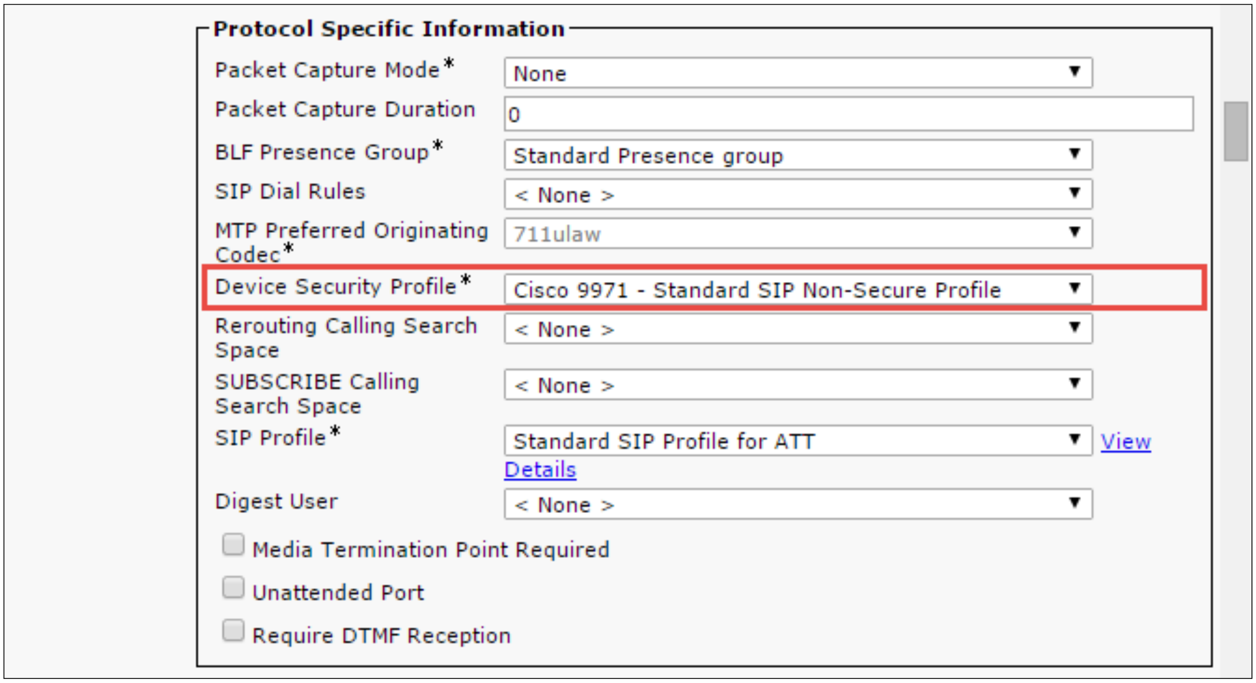

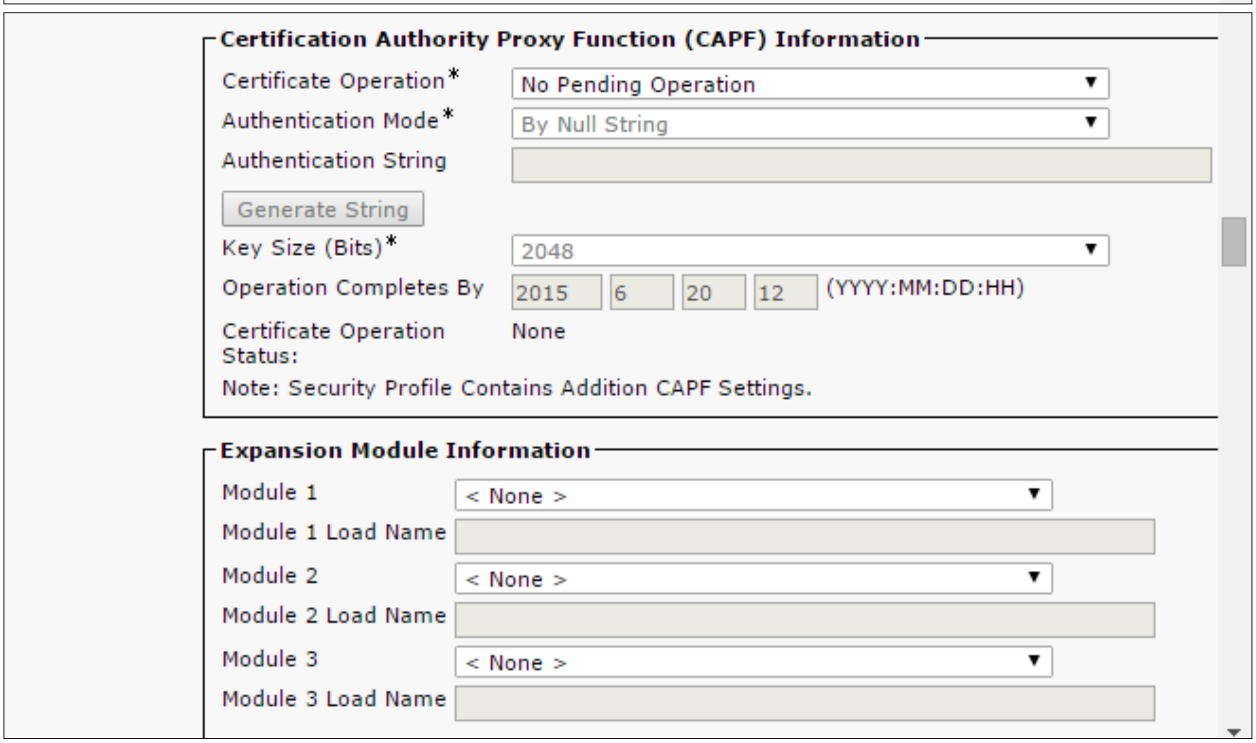

Cisco IP Phone 9971 SIP Configuration (Continued…)

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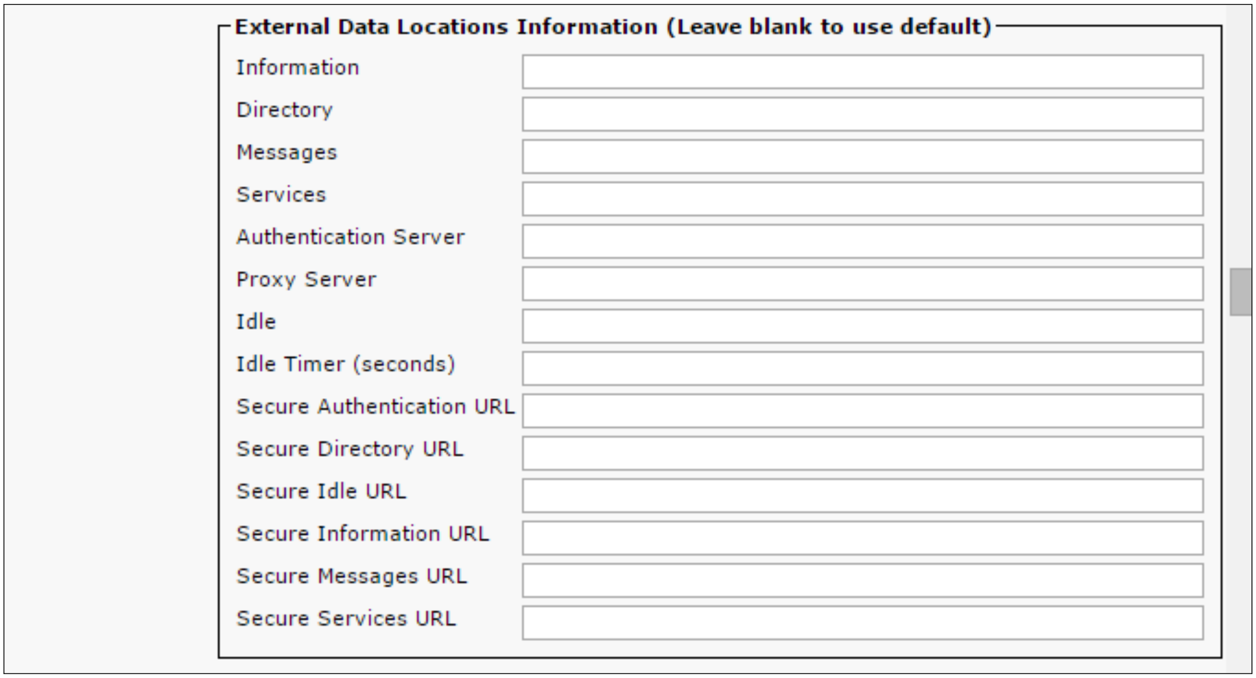

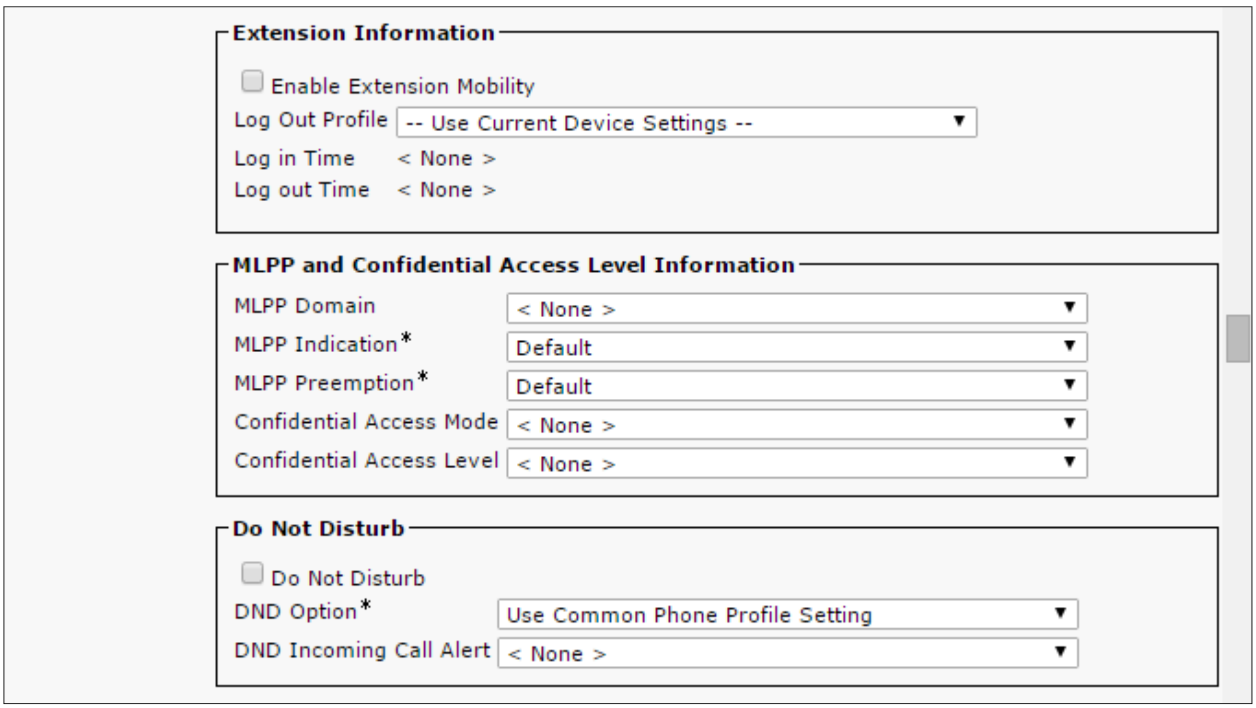

Cisco IP Phone 9971 SIP Configuration (Continued…)

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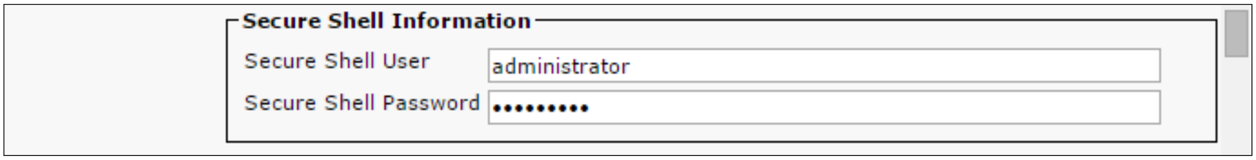
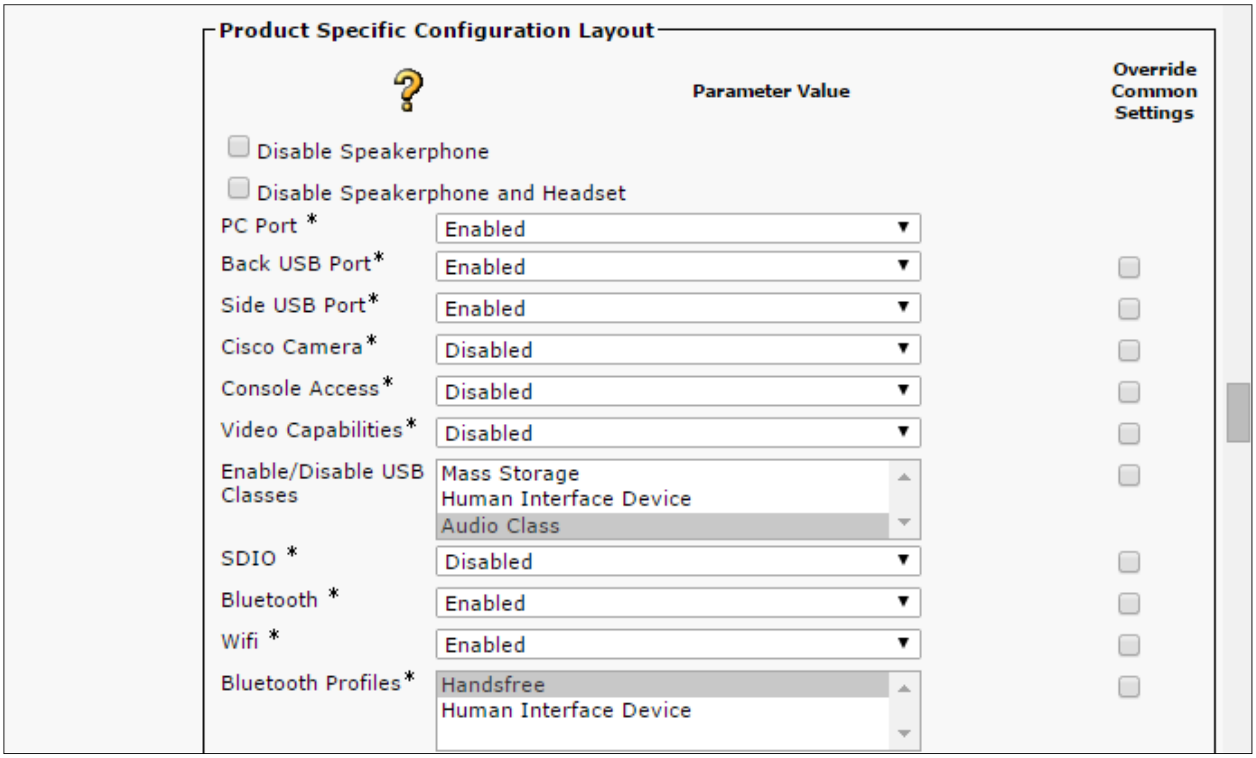


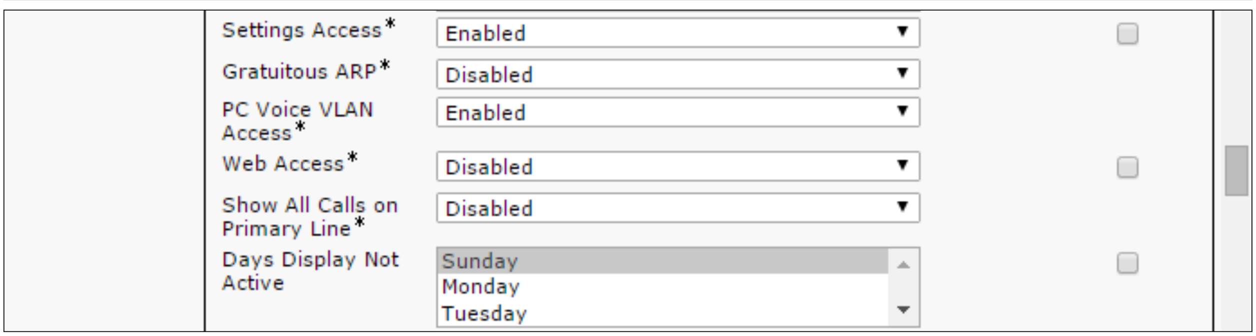
Secure Shell Information

Secure Shell User	administrator
Secure Shell Password	•••••••••



Product Specific Configuration Layout

	Parameter Value	Override Common Settings
Disable Speakerphone		
Disable Speakerphone and Headset		
PC Port *	Enabled	
Back USB Port*	Enabled	
Side USB Port*	Enabled	
Cisco Camera*	Disabled	
Console Access*	Disabled	
Video Capabilities*	Disabled	
Enable/Disable USB Classes	Mass Storage Human Interface Device Audio Class	
SDIO *	Disabled	
Bluetooth *	Enabled	
Wifi *	Enabled	
Bluetooth Profiles*	Handsfree Human Interface Device	



Settings Access*	Enabled	
Gratuitous ARP*	Disabled	
PC Voice VLAN Access*	Enabled	
Web Access*	Disabled	
Show All Calls on Primary Line*	Disabled	
Days Display Not Active	Sunday Monday Tuesday	

Cisco IP Phone 9971 SIP Configuration (Continued…)

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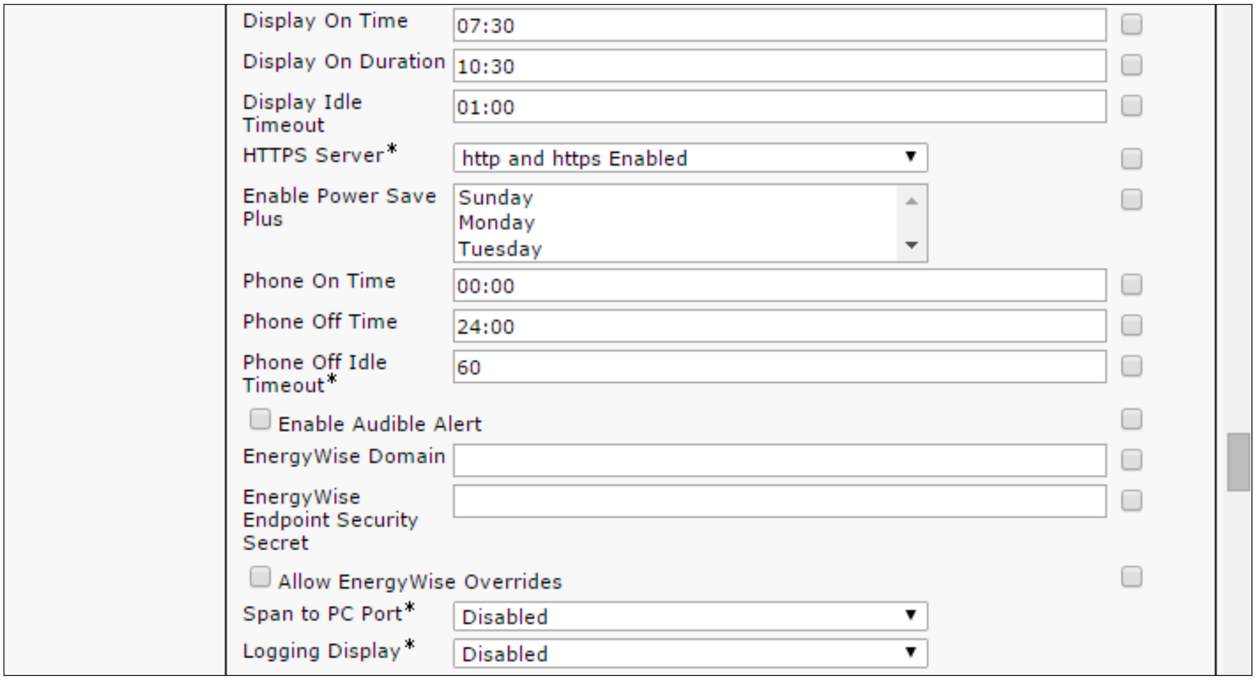

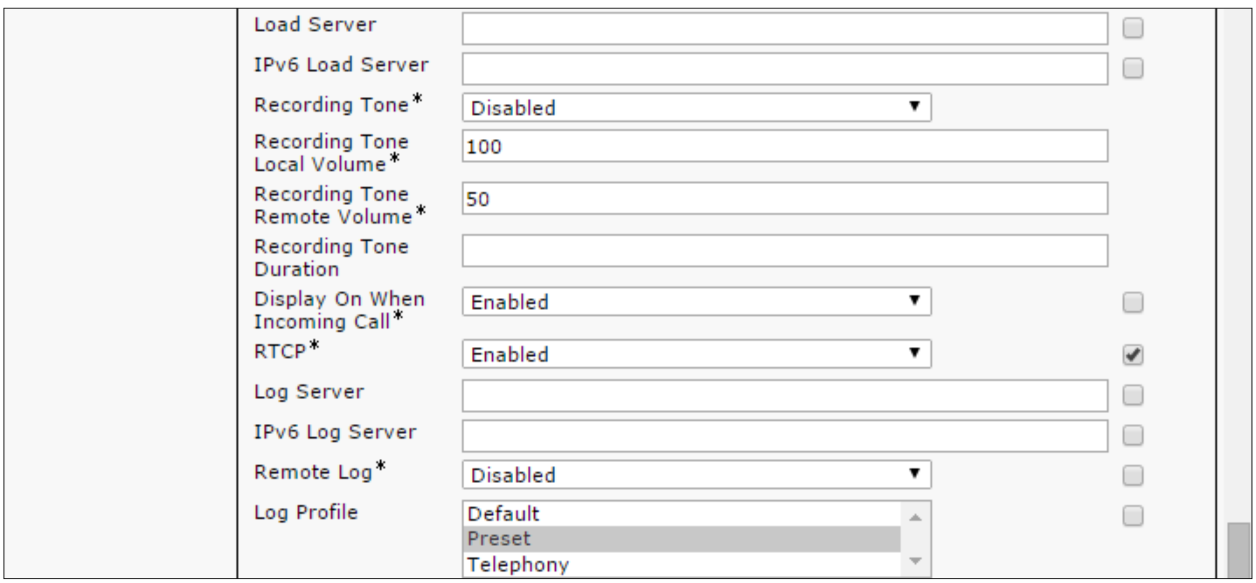

Cisco IP Phone 9971 SIP Configuration (Continued…)

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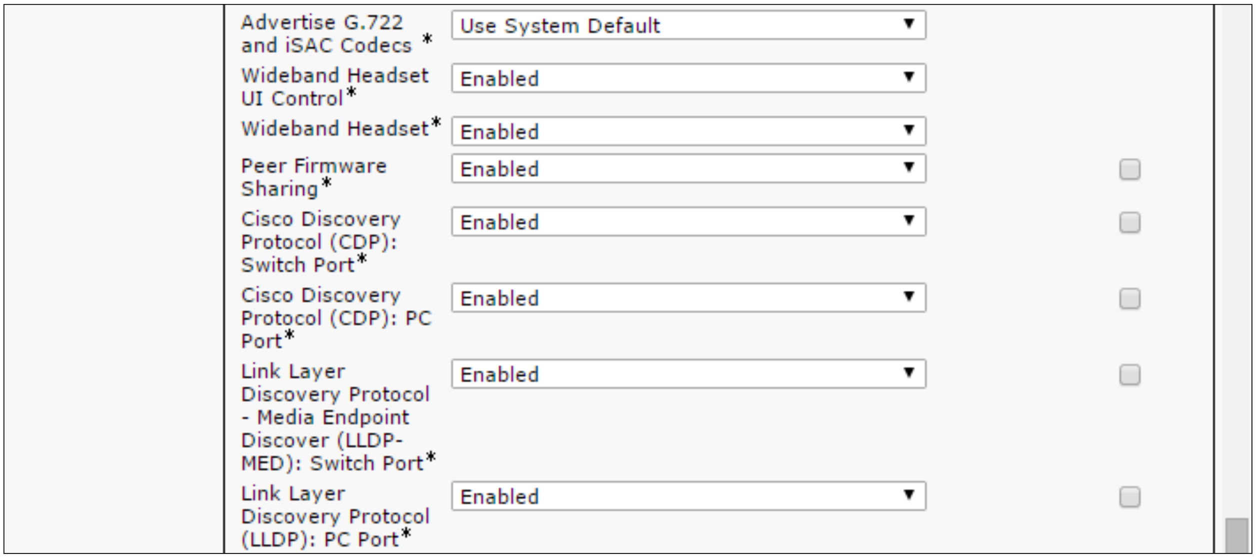

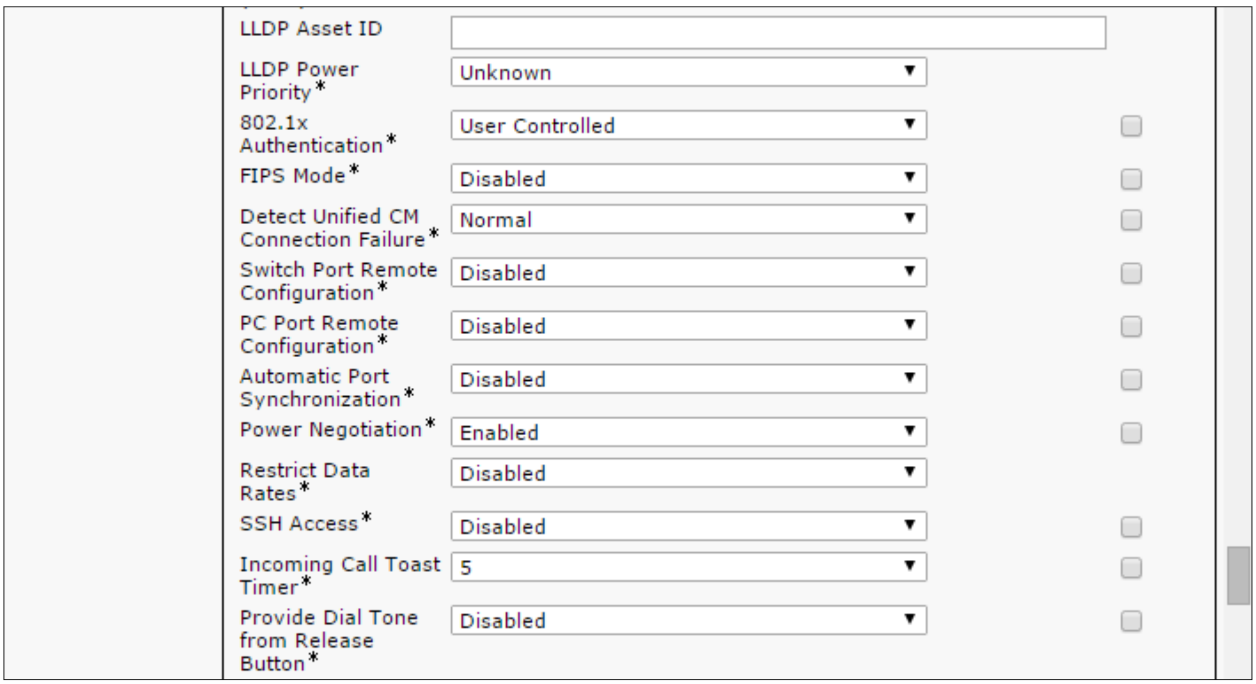

Cisco IP Phone 9971 SIP Configuration (Continued…)

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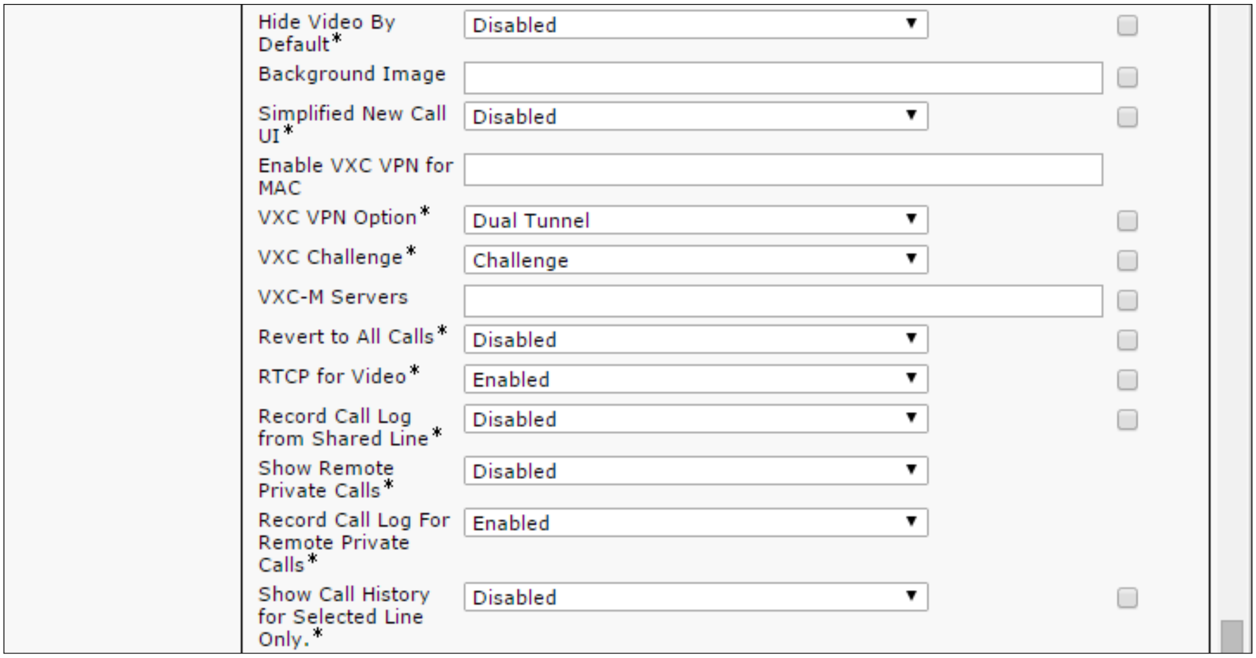
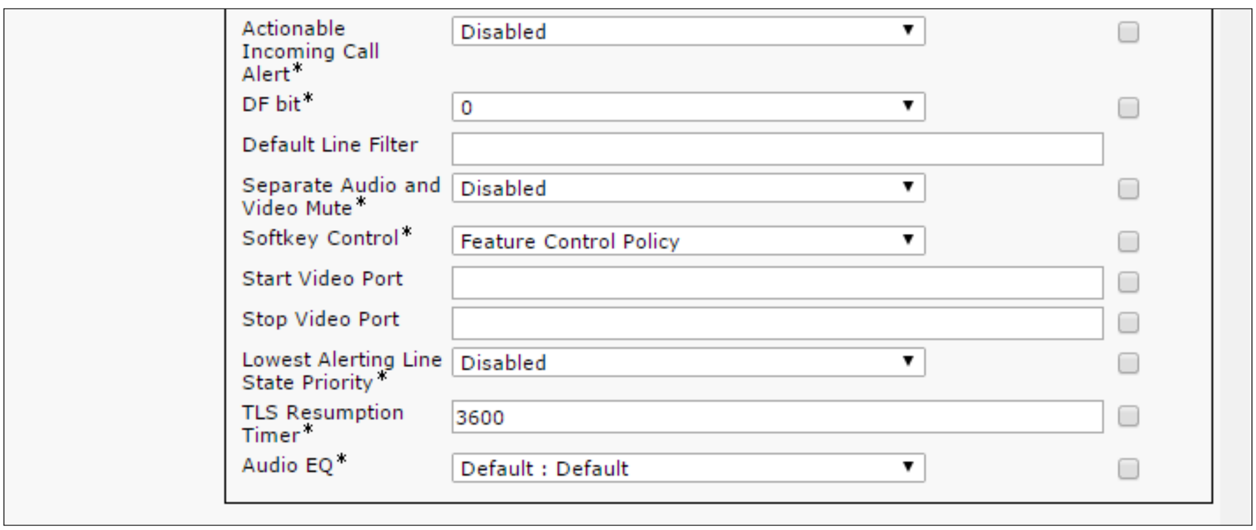
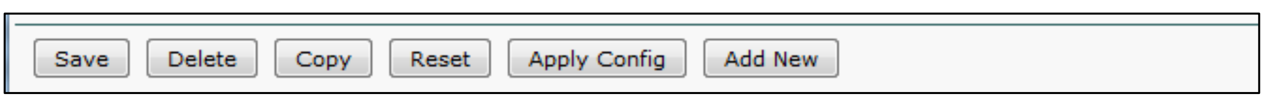
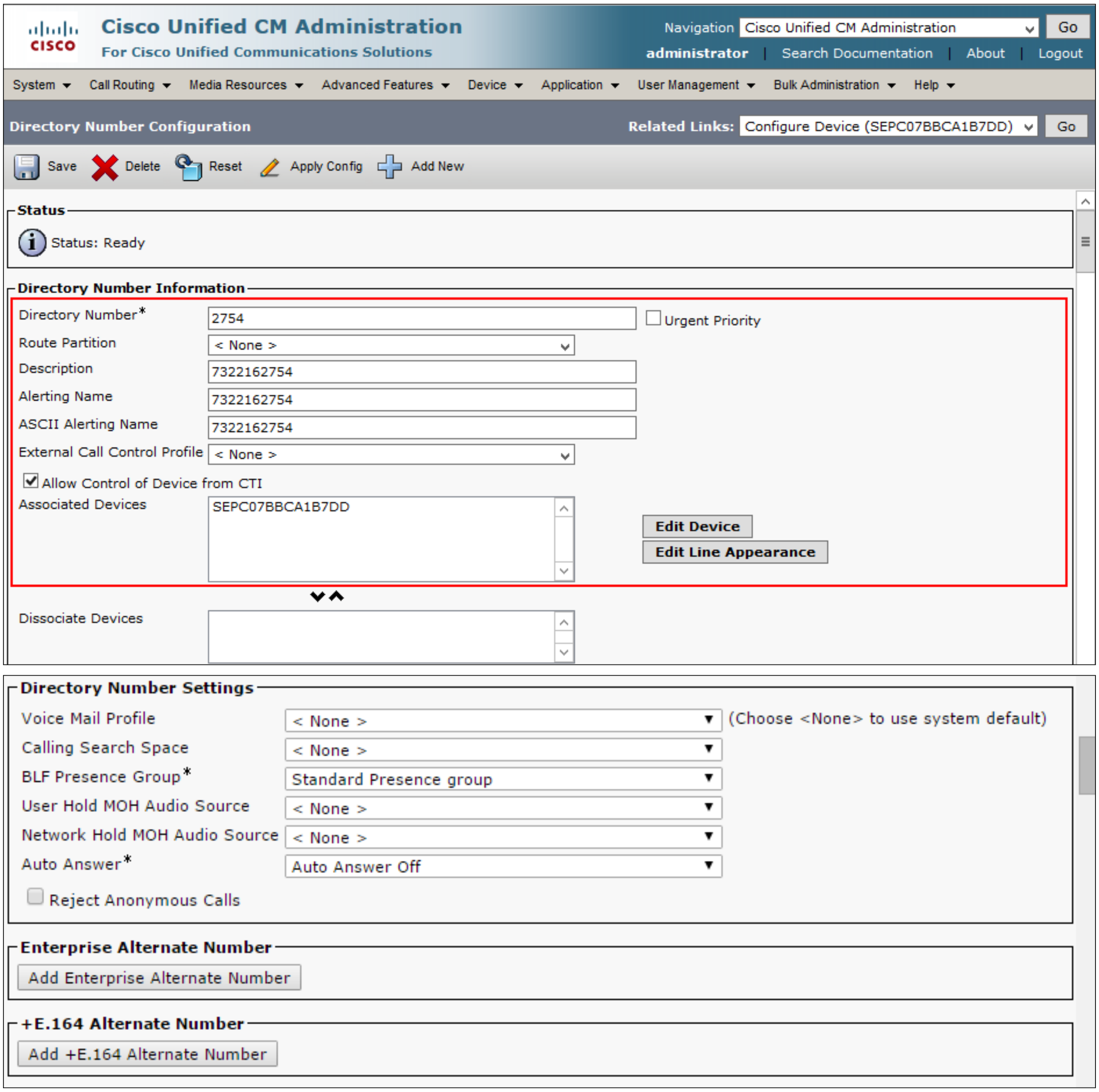






Cisco IP Phone 9971 SIP Configuration (Continued…)

Set Directory Number* = 2754. This is used in this example.
Set Description = 7322162754. This is used in this example.
Set Alerting Name = 7322162754. This is used in this example.
Set ASCII Alerting Name = 7322162754. This is used in this example.



Cisco IP Phone 9971 SIP Configuration (Continued…)

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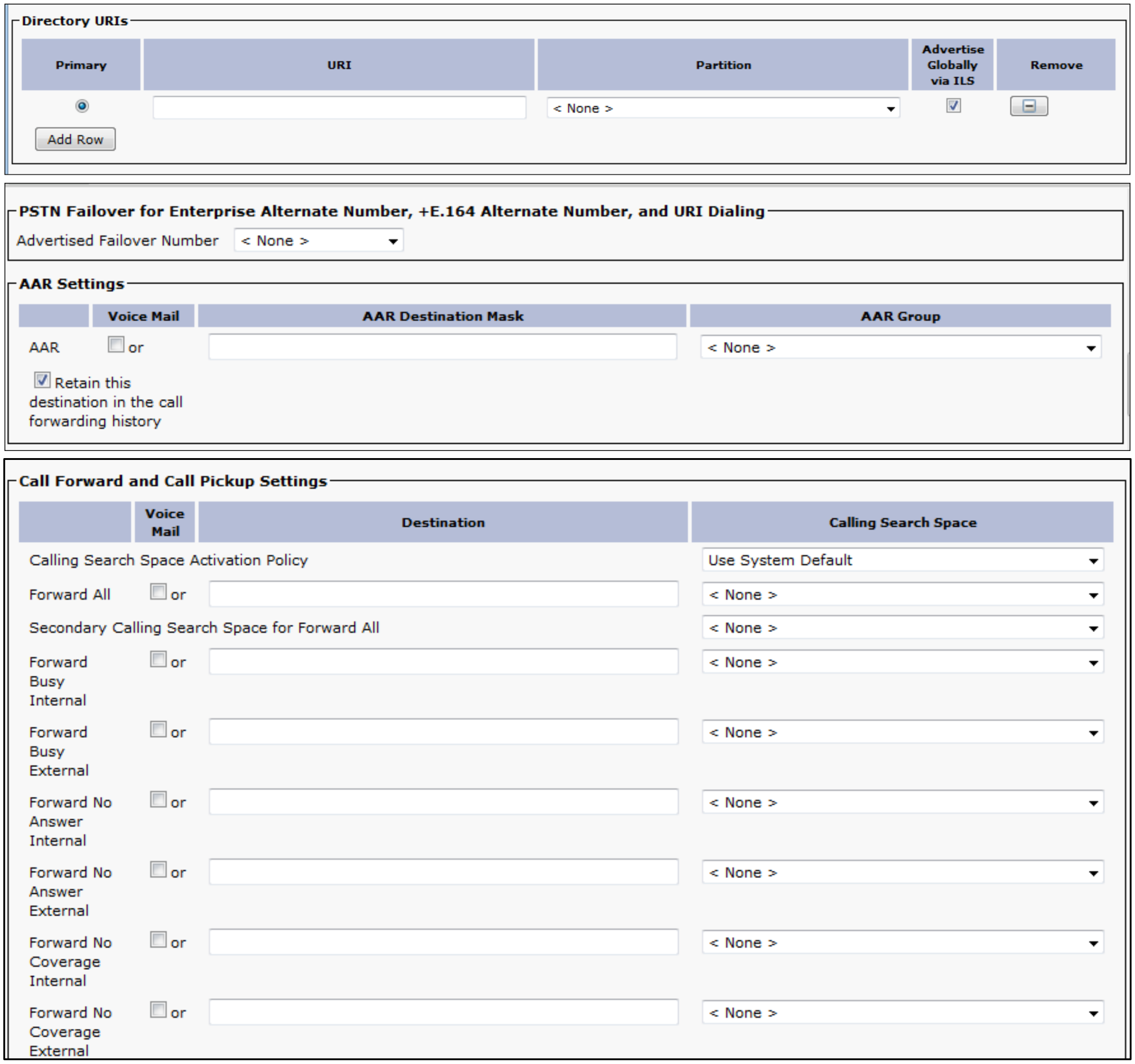

Cisco IP Phone 9971 SIP Configuration (Continued…)

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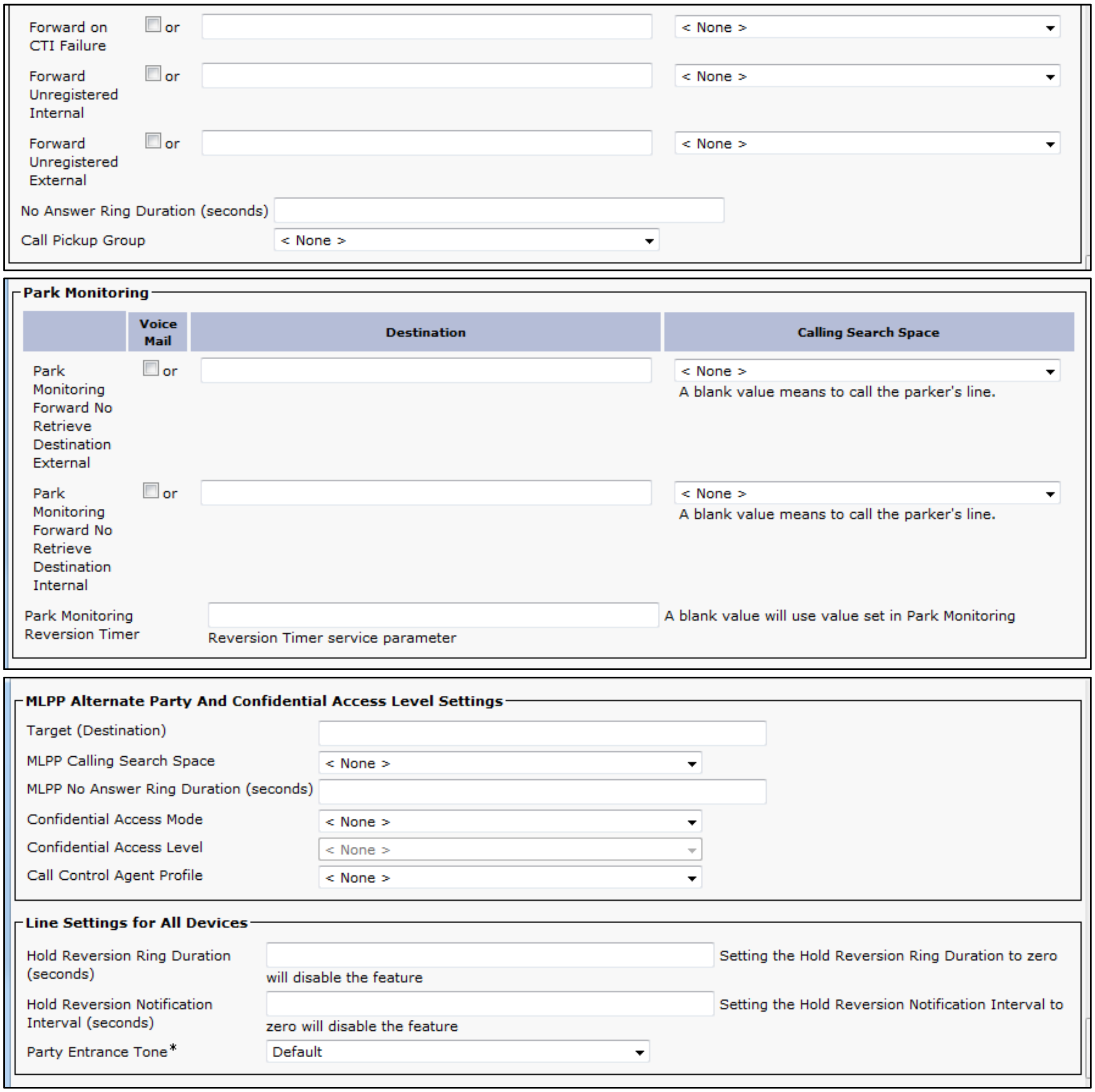

Forward on CTI Failure | or | | < None >

Forward Unregistered Internal | or | | < None >

Forward Unregistered External | or | | < None >

No Answer Ring Duration (seconds)

Call Pickup Group < None >

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

Cisco IP Phone 9971 SIP Configuration (Continued…)

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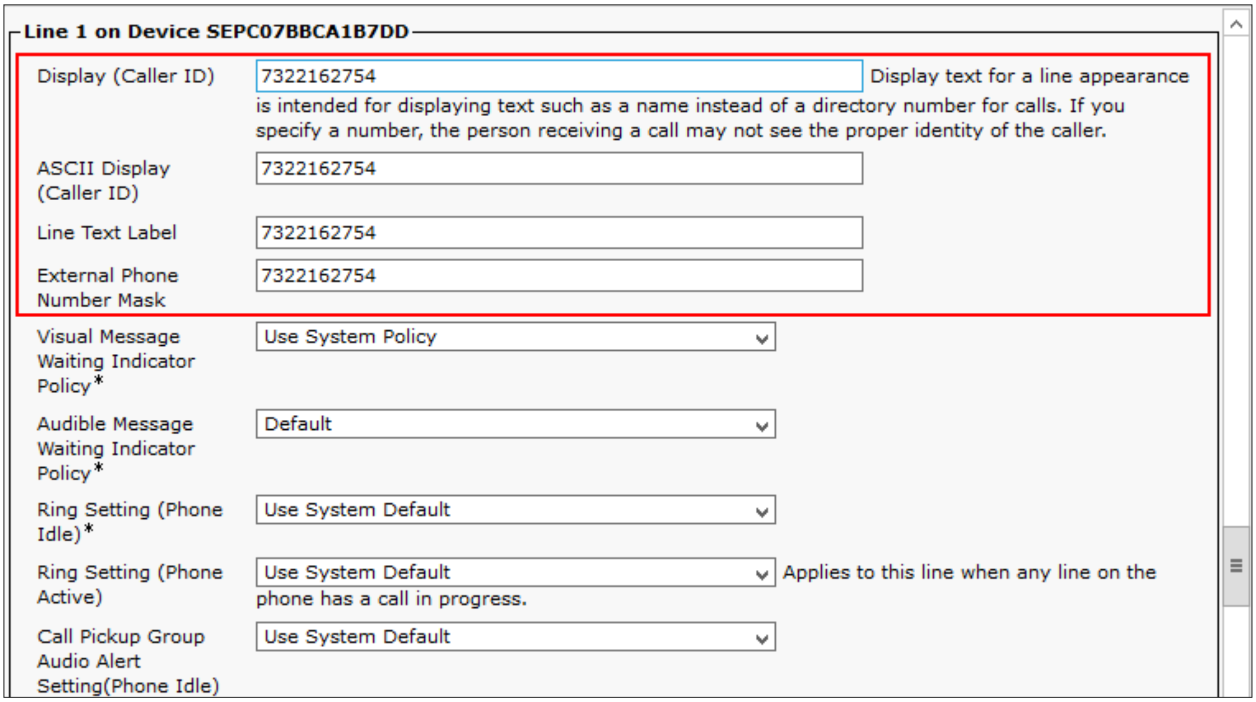
Set Display (caller ID) = 7322162754. This is used in this example.
Set ASCII Display (caller ID) = 7322162754. This is used in this example.
Set Line Text Label = 7322162754. This is used in this example.
Set External Phone Number Mask = 7322162754. This is used in this example.



Line 1 on Device SEPC07BBCA1B7DD

Display (Caller ID): 7322162754 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): 7322162754

Line Text Label: 7322162754

External Phone Number Mask: 7322162754

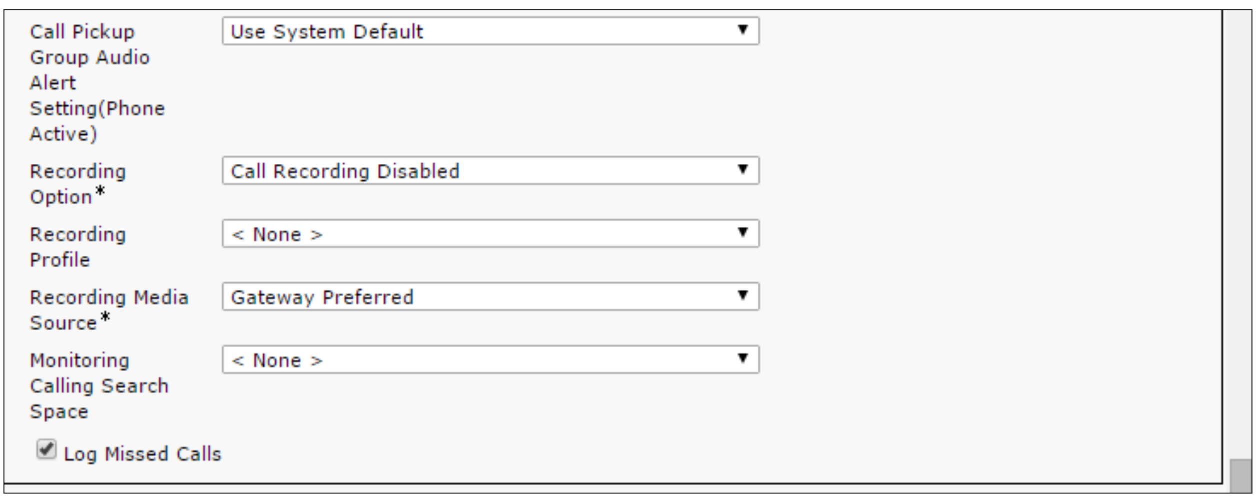
Visual Message Waiting Indicator Policy*: Use System Policy

Audible Message Waiting Indicator Policy*: Default

Ring Setting (Phone Idle)*: Use System Default

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

Cisco IP Phone 9971 SIP Configuration (Continued…)

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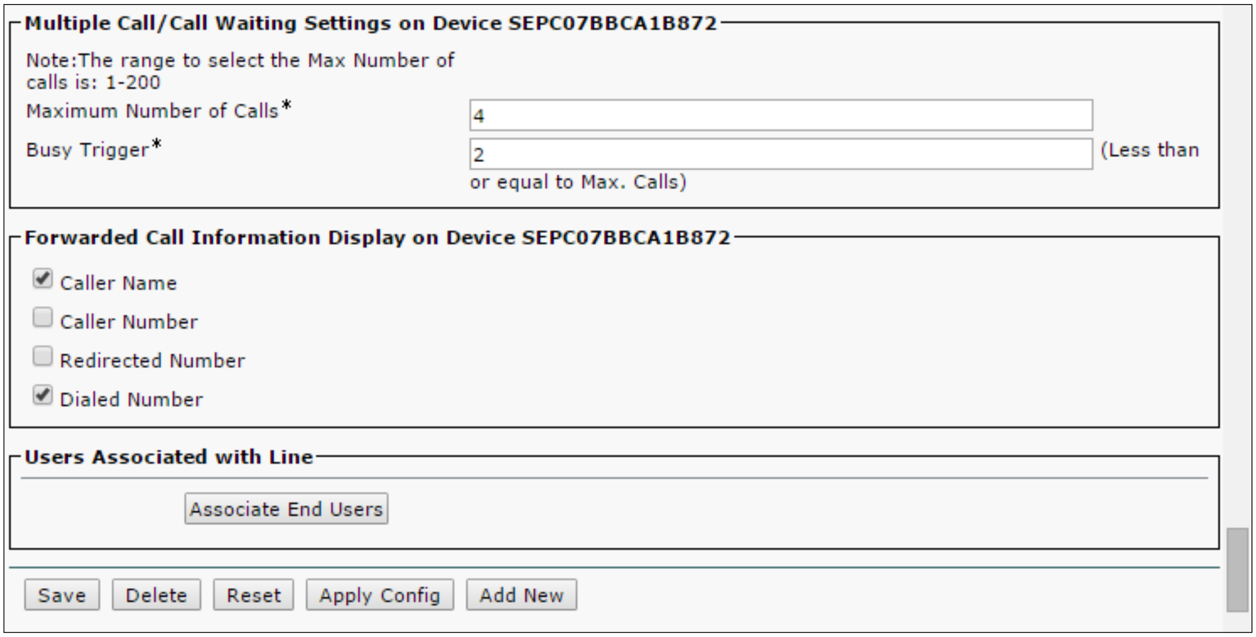
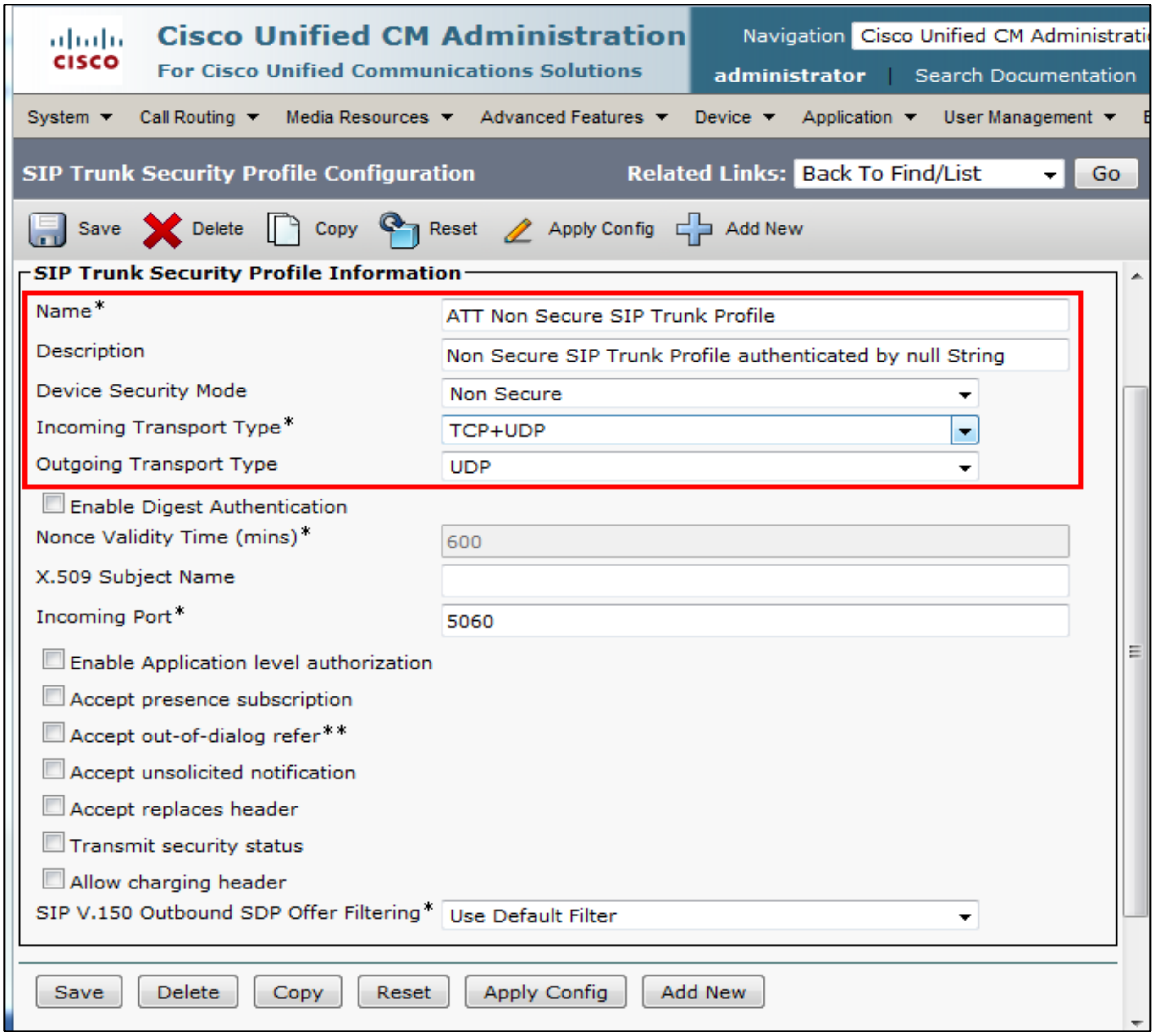


SIP Trunk Security Profile Configuration used by SIP trunk to Cisco UBE

Navigation: System → Security → SIP Trunk Security Profile

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



SIP Profile Configuration used by SIP trunk to Cisco UBE

Navigation: Device → Device Settings → SIP Profile

Set SIP profile Name * = Standard SIP Profile w/Early Media Disabled. This is used for this example

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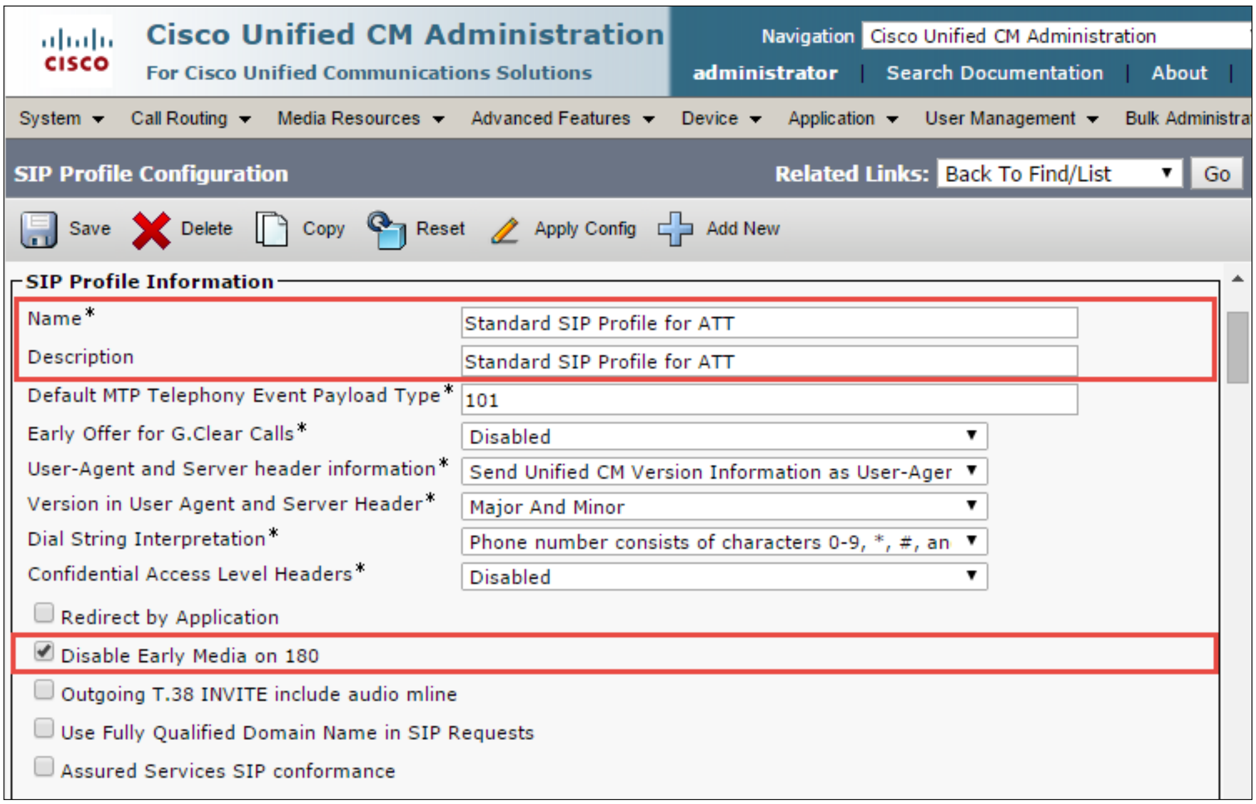
Check Disable Early Media on 180
Set SIP Rel1xx Options* = Send PRACK if 1xx contains SDP
Note*= Some PSTN network call prompters utilize early-media cut-through to offer menu options to the caller (DTMF select menu) before the call is connected. In order for Cisco UCM/Cisco UBE solution to achieve successful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



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

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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

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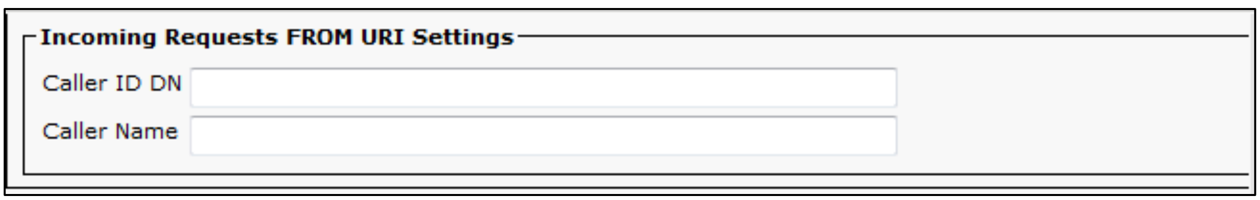
SIP Profile Configuration used by SIP trunk to Cisco UBE (Continued…)



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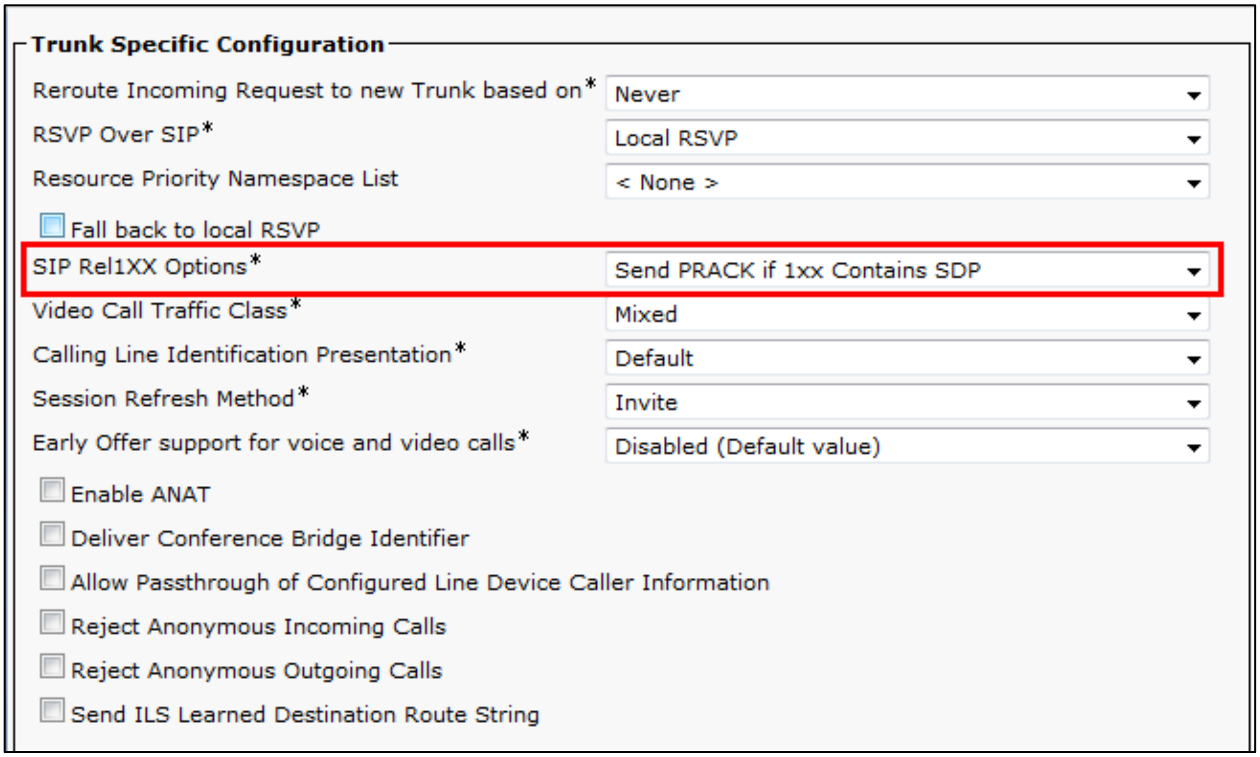

SIP Profile Configuration used by SIP trunk to Cisco UBE (Continued…)

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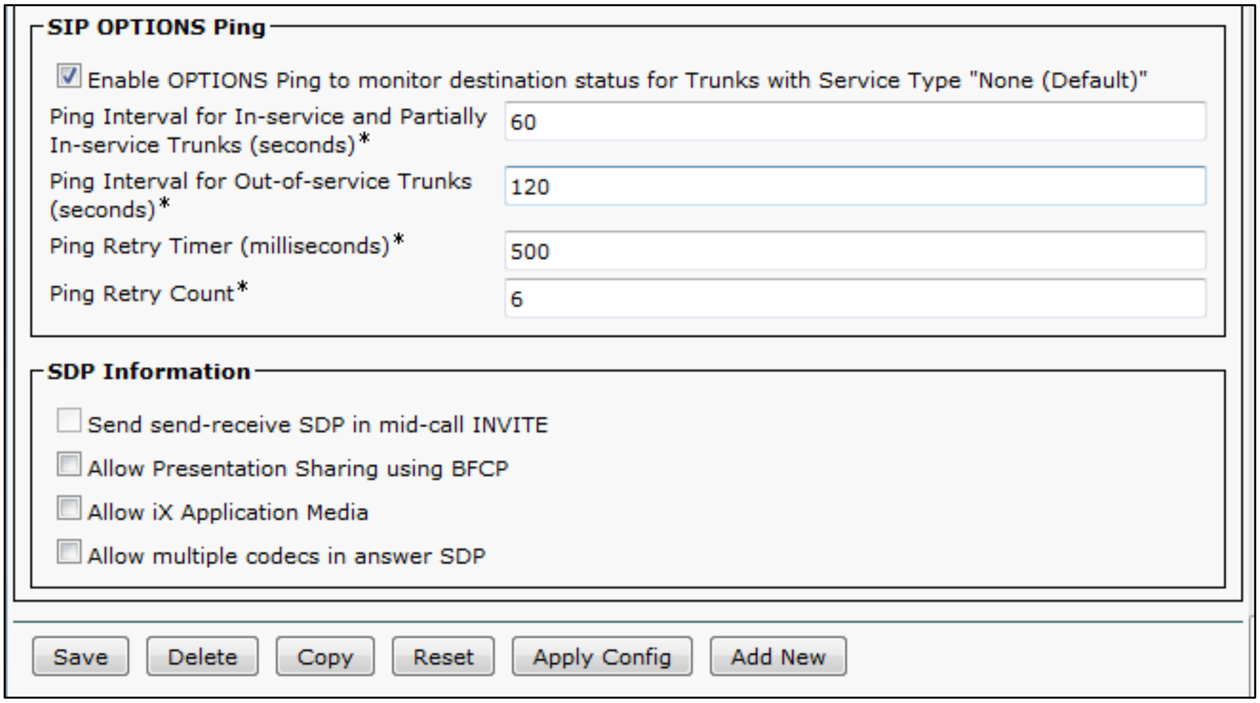

SIP Trunk to Cisco UBE Configuration

Navigation: Device → Trunk

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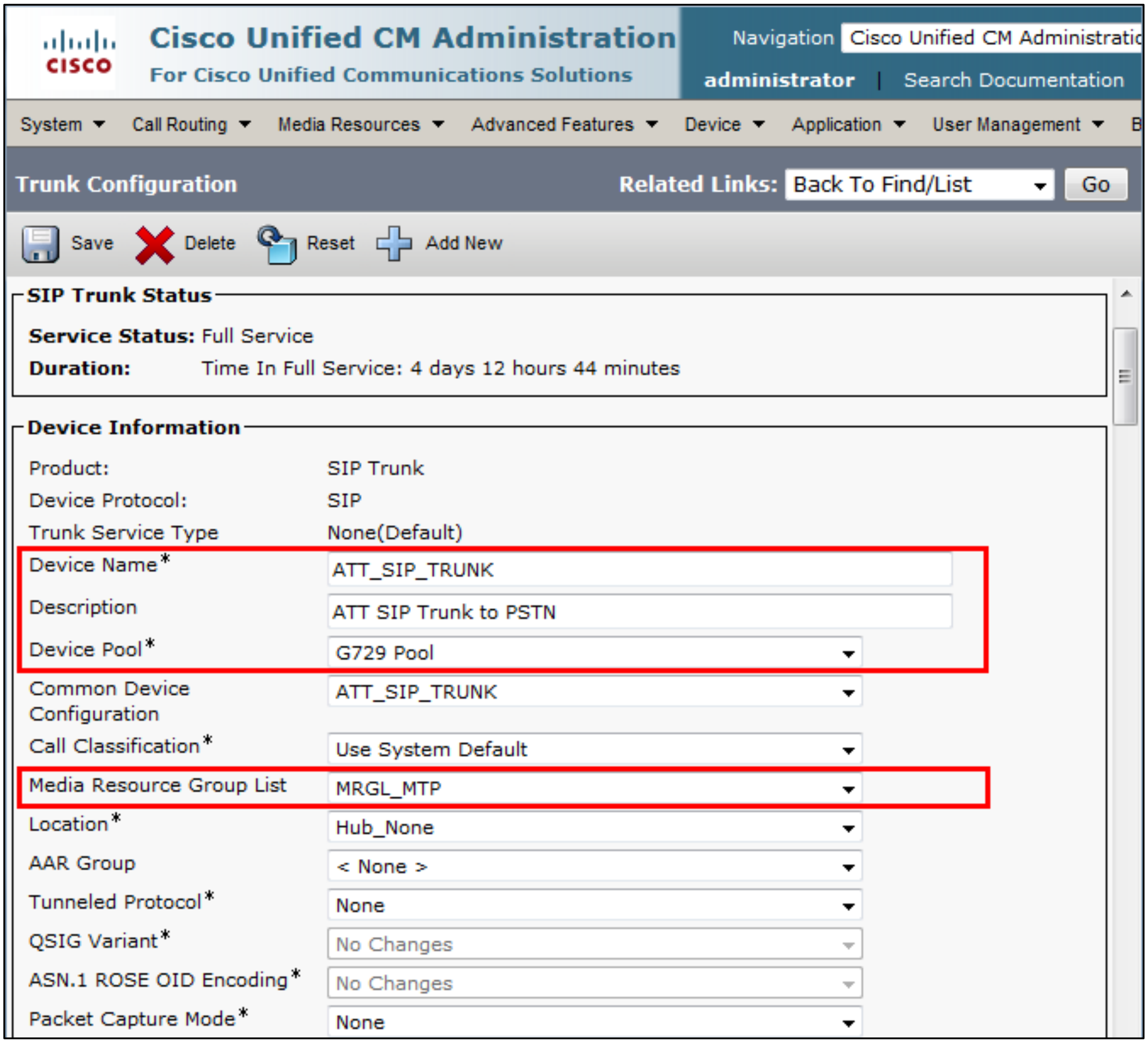
Set Device Name* = ATT_SIP_TRUNK. This is used for this example
Set Description = ATT SIP Trunk to PSTN. This is used for this example
Set Device Pool* = G729_pool. This is used for this example
Set Media Resource Group List = MRGL_MTP.



SIP Trunk to Cisco UBE Configuration (Continued…)

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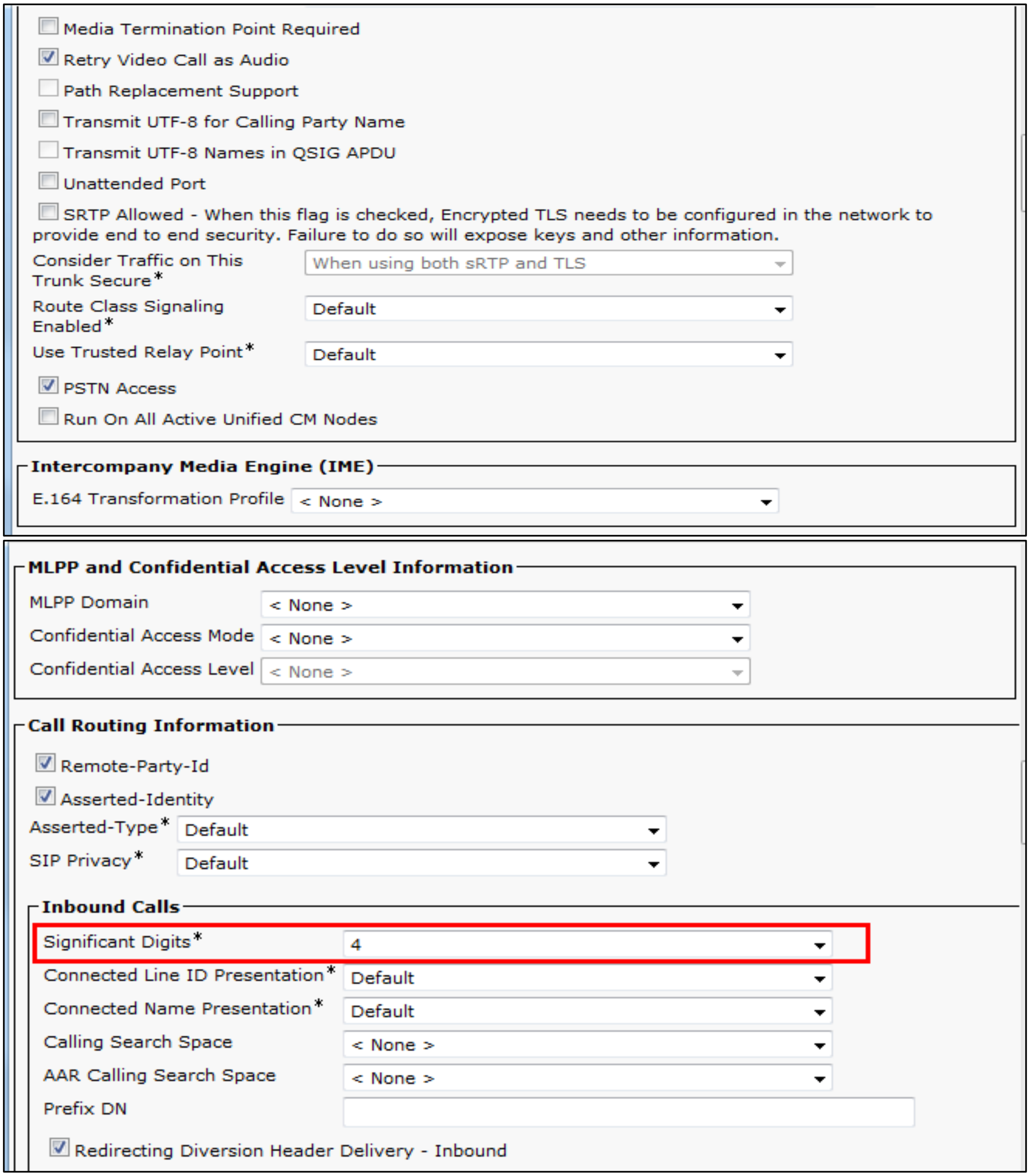
Set Significant Digits* = 4. This is used in this example.



SIP Trunk to Cisco UBE Configuration (Continued…)

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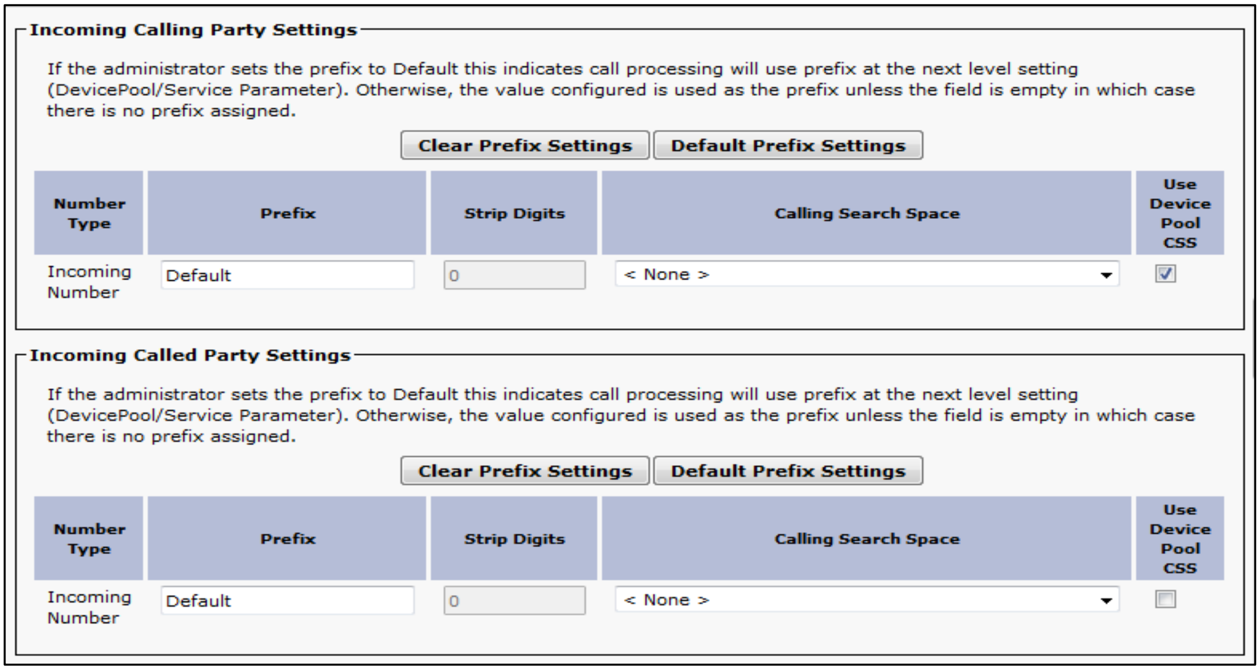

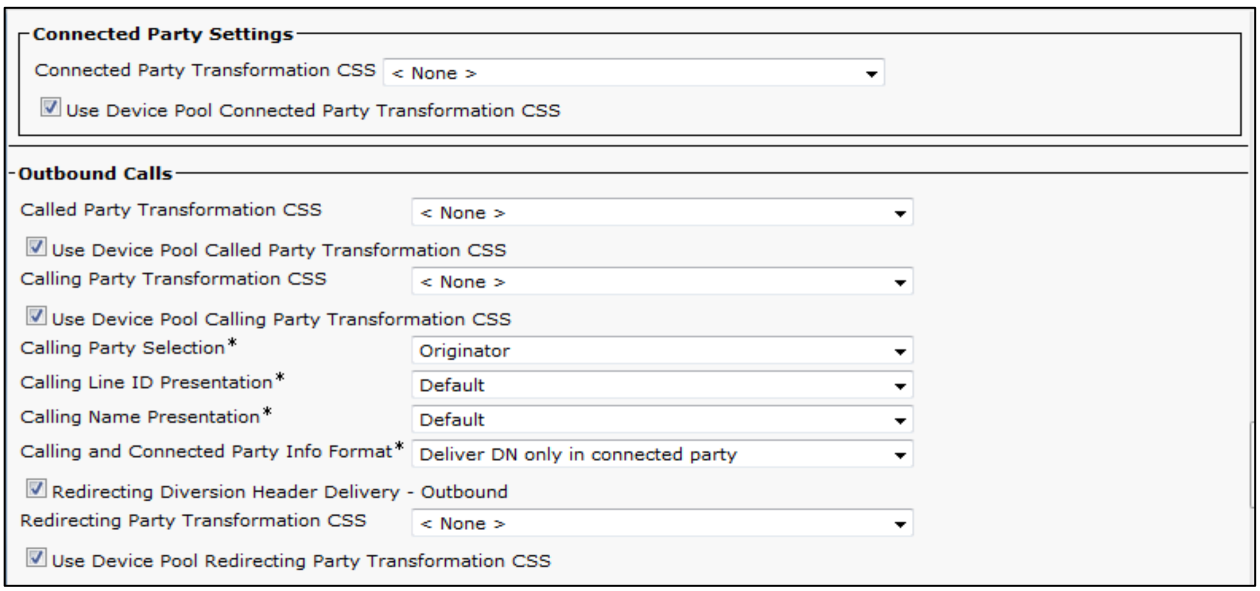

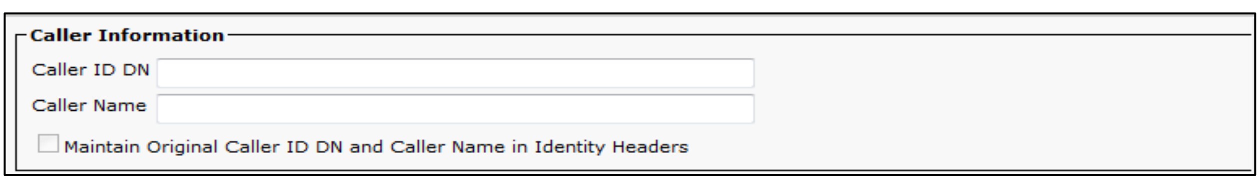

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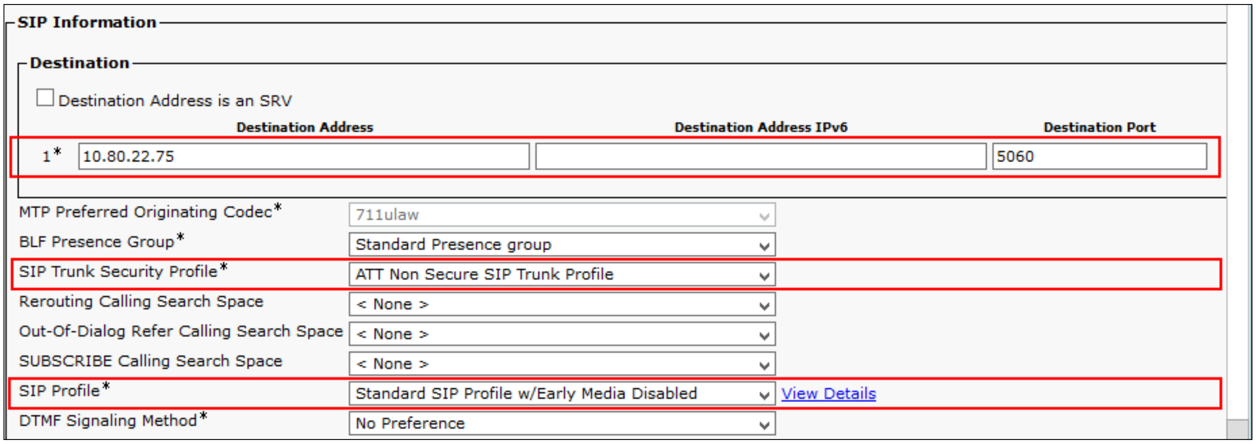
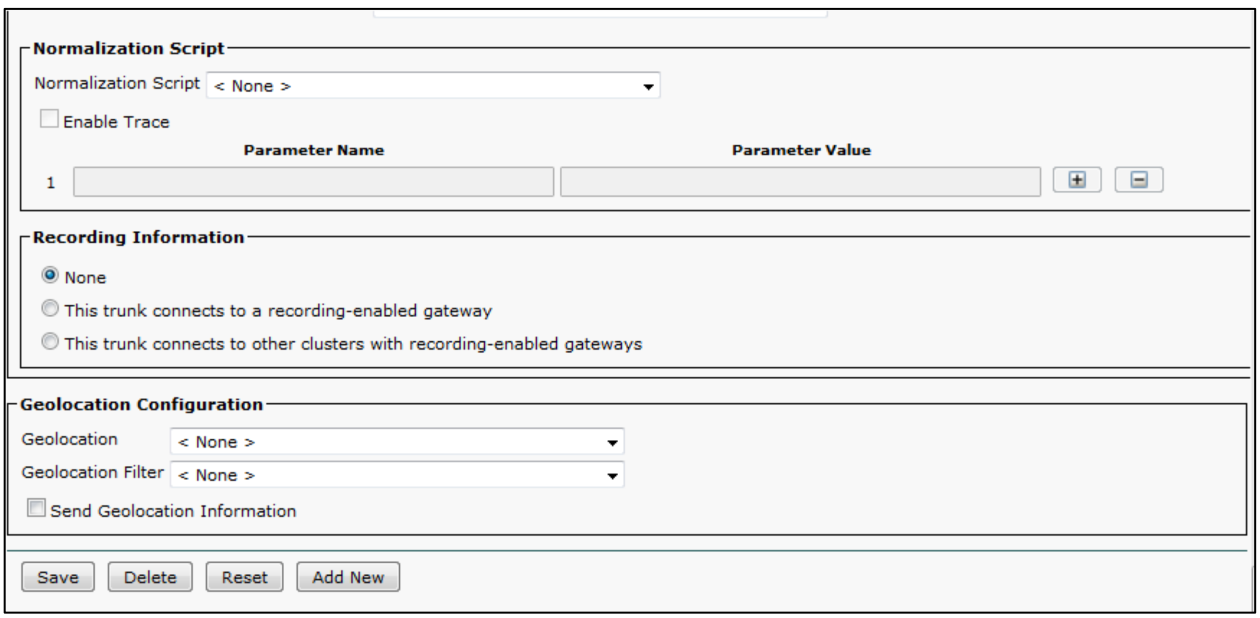
SIP Trunk to Cisco UBE Configuration (Continued…)

Set Destination Address = Set IP address of ISR-Cisco UBE.
Set SIP Trunk Security Profile* = ATT_Non Secure Sip Trunk Profile.
Set SIP Profile* = Standard SIP Profile w/Early Media Disabled. This is used in this example.





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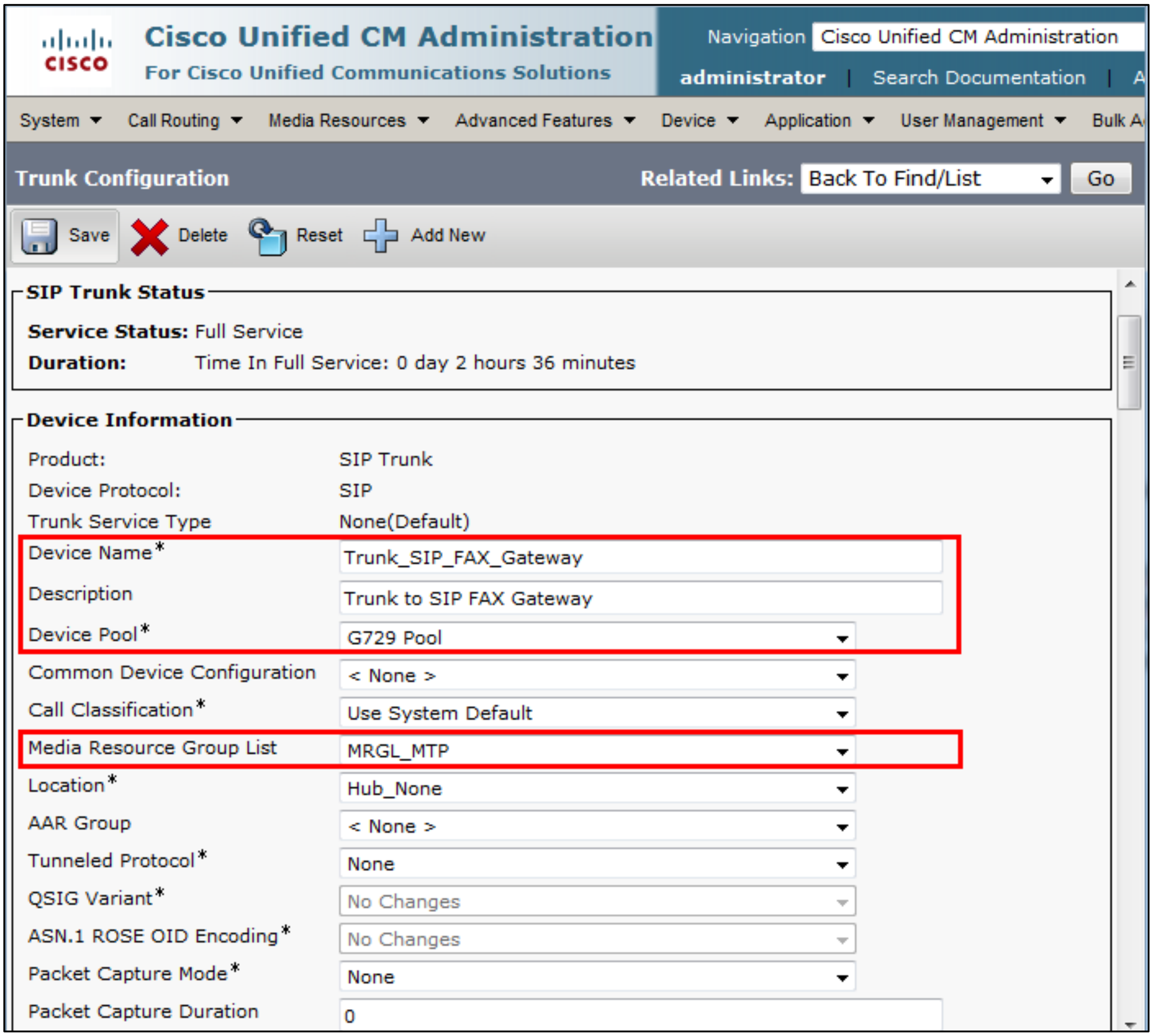
SIP Trunk to Fax Gateway Configuration.

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



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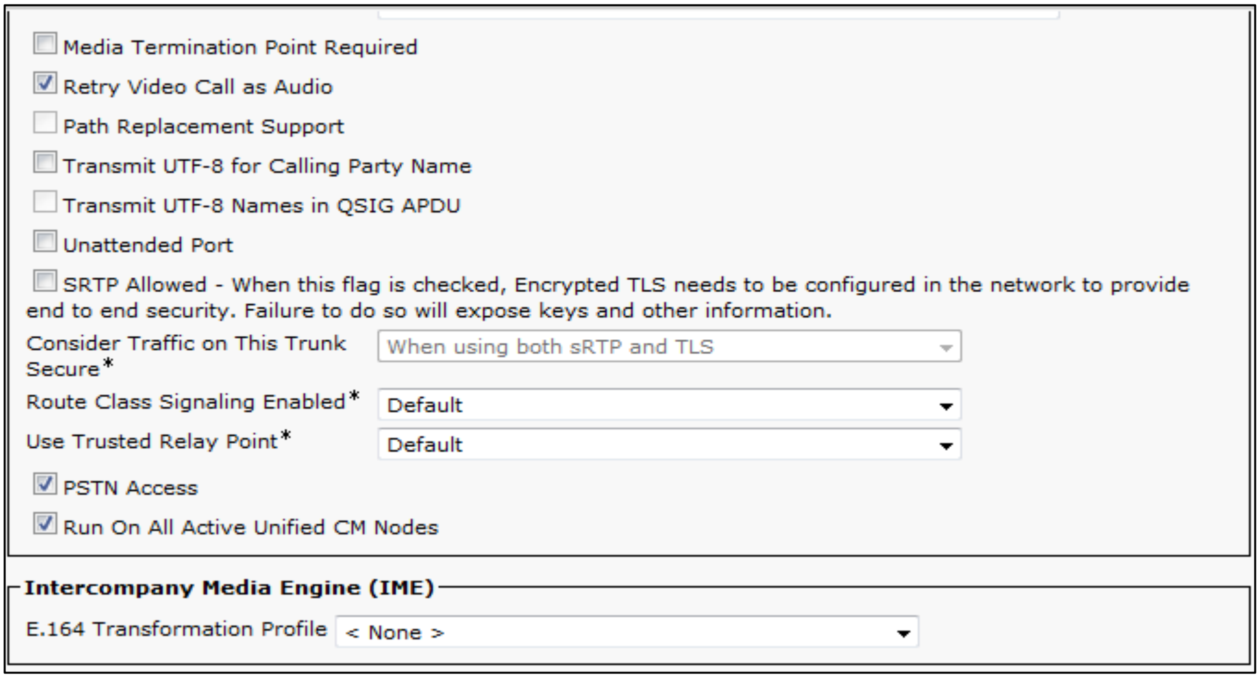


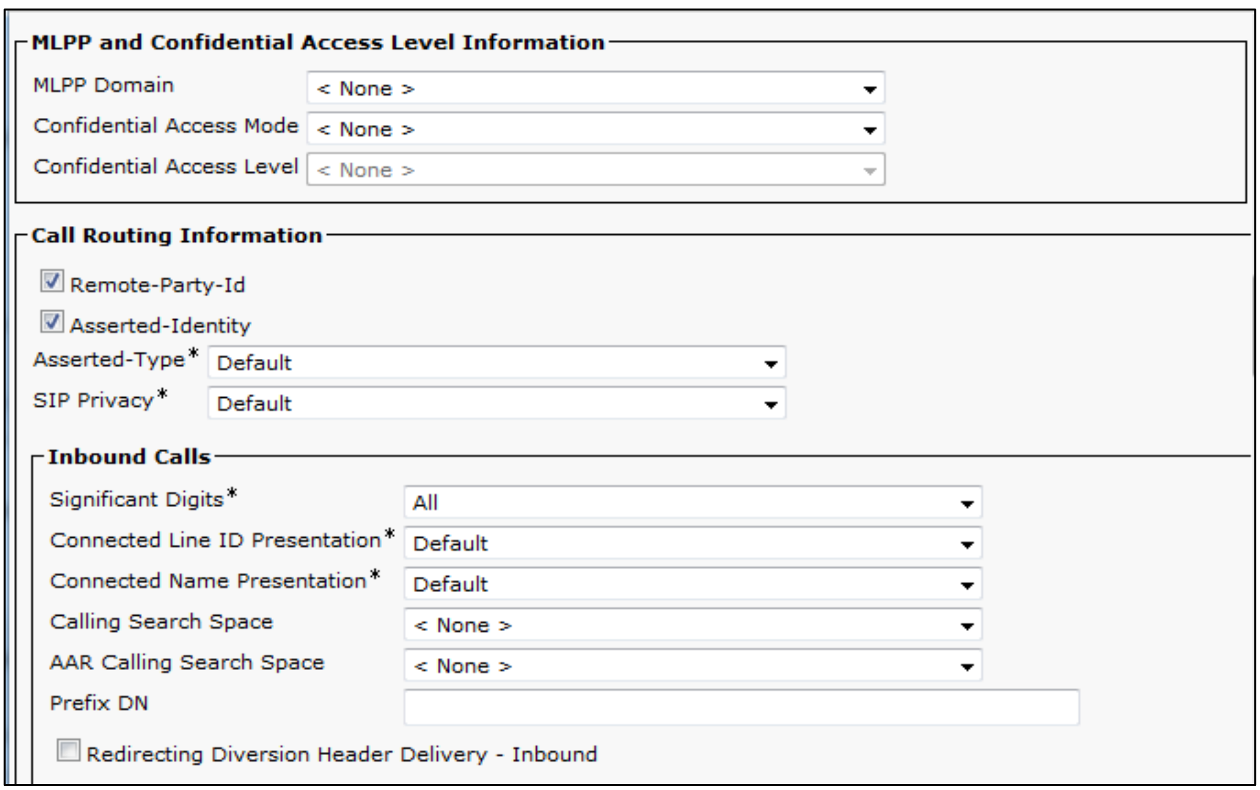
- [ ] Media Termination Point Required
- [x] Retry Video Call as Audio
- [ ] Path Replacement Support
- [ ] Transmit UTF-8 for Calling Party Name
- [ ] Transmit UTF-8 Names in QSIG APDU
- [ ] Unattended Port
- [ ] SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network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

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

- [x] PSTN Access
- [x] Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile	< None >



MLPP and Confidential Access Level Information

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

Call Routing Information

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

Asserted-Type*	Default
SIP Privacy*	Default

Inbound Calls

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

- [ ] Redirecting Diversion Header Delivery - Inbound

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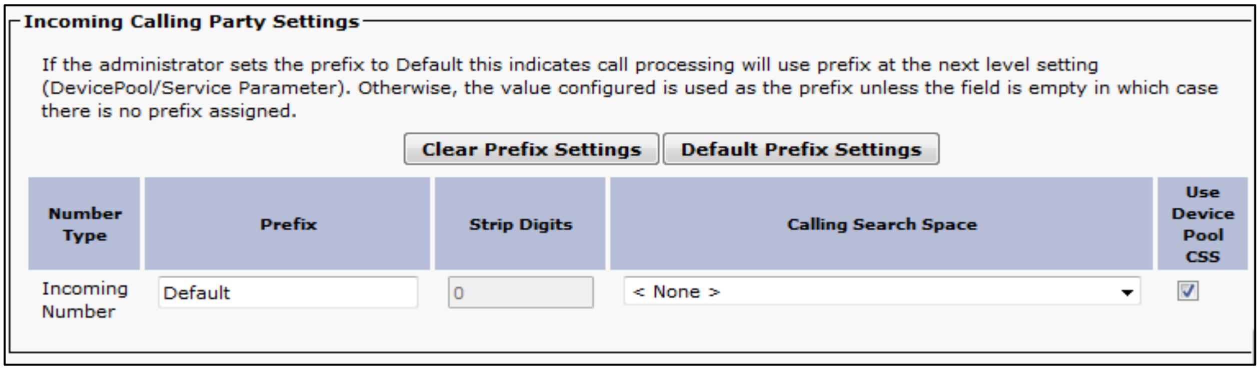
SIP Trunk to Fax Gateway Configuration (Continued…)



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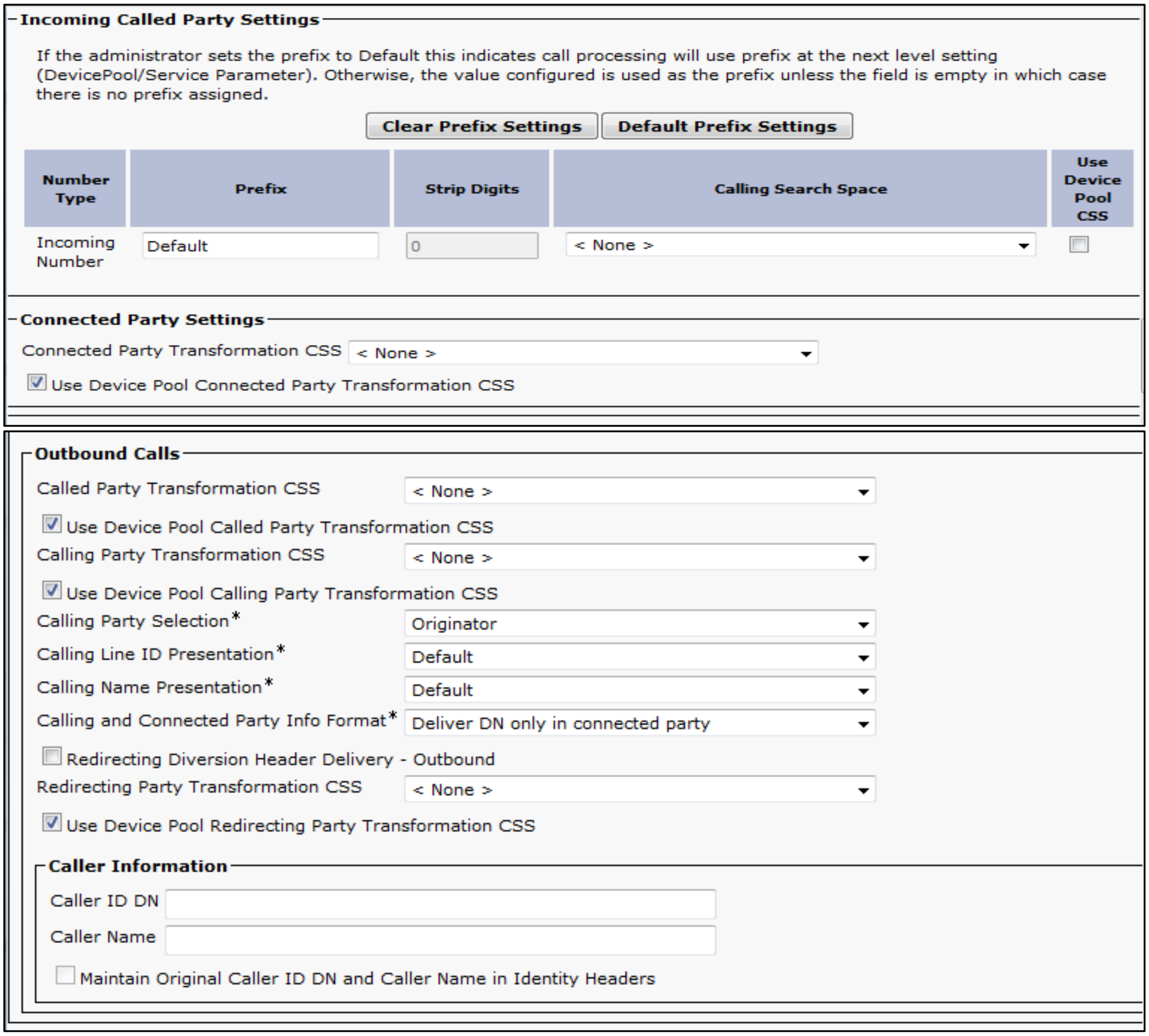

Incoming Called Party Settings

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 (Continued…)

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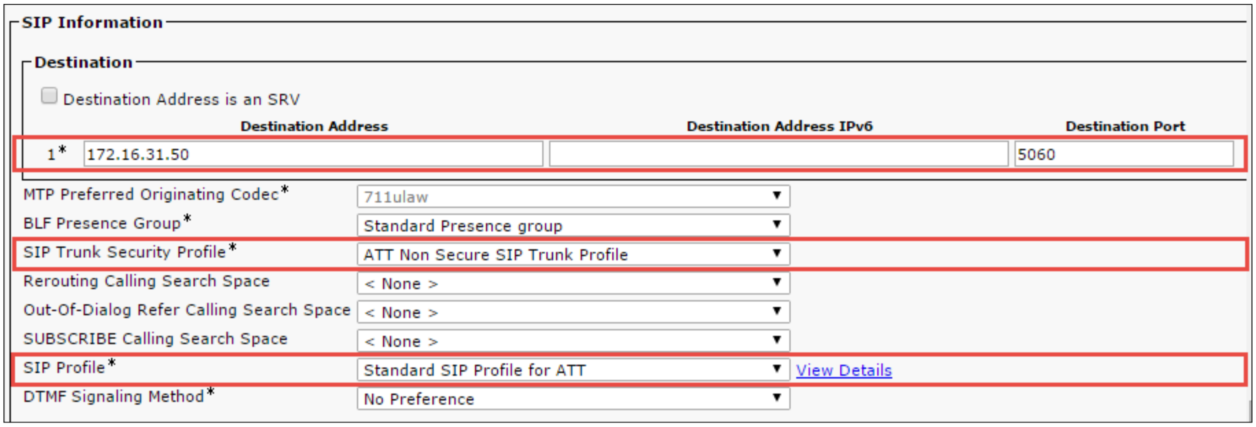

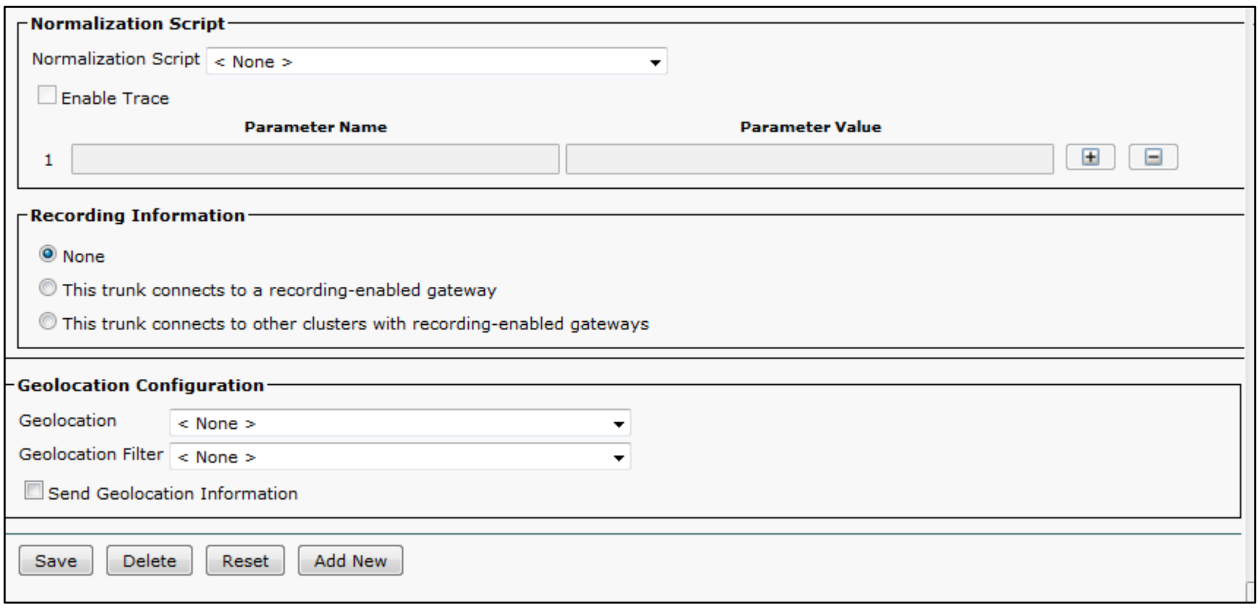

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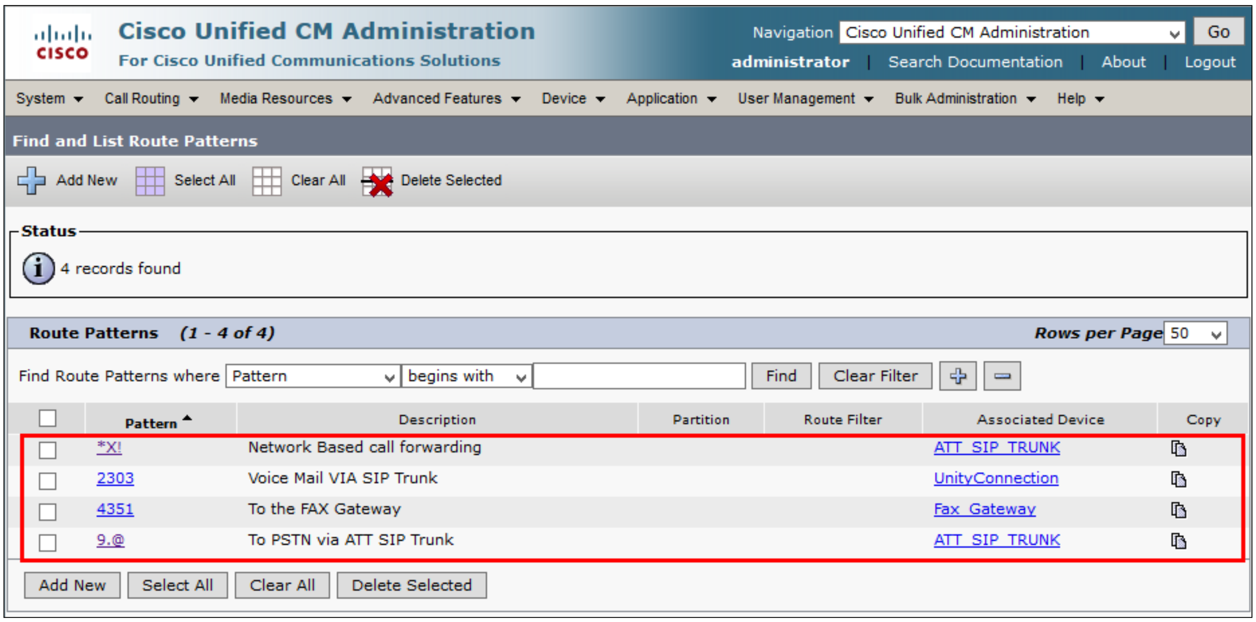
Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

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



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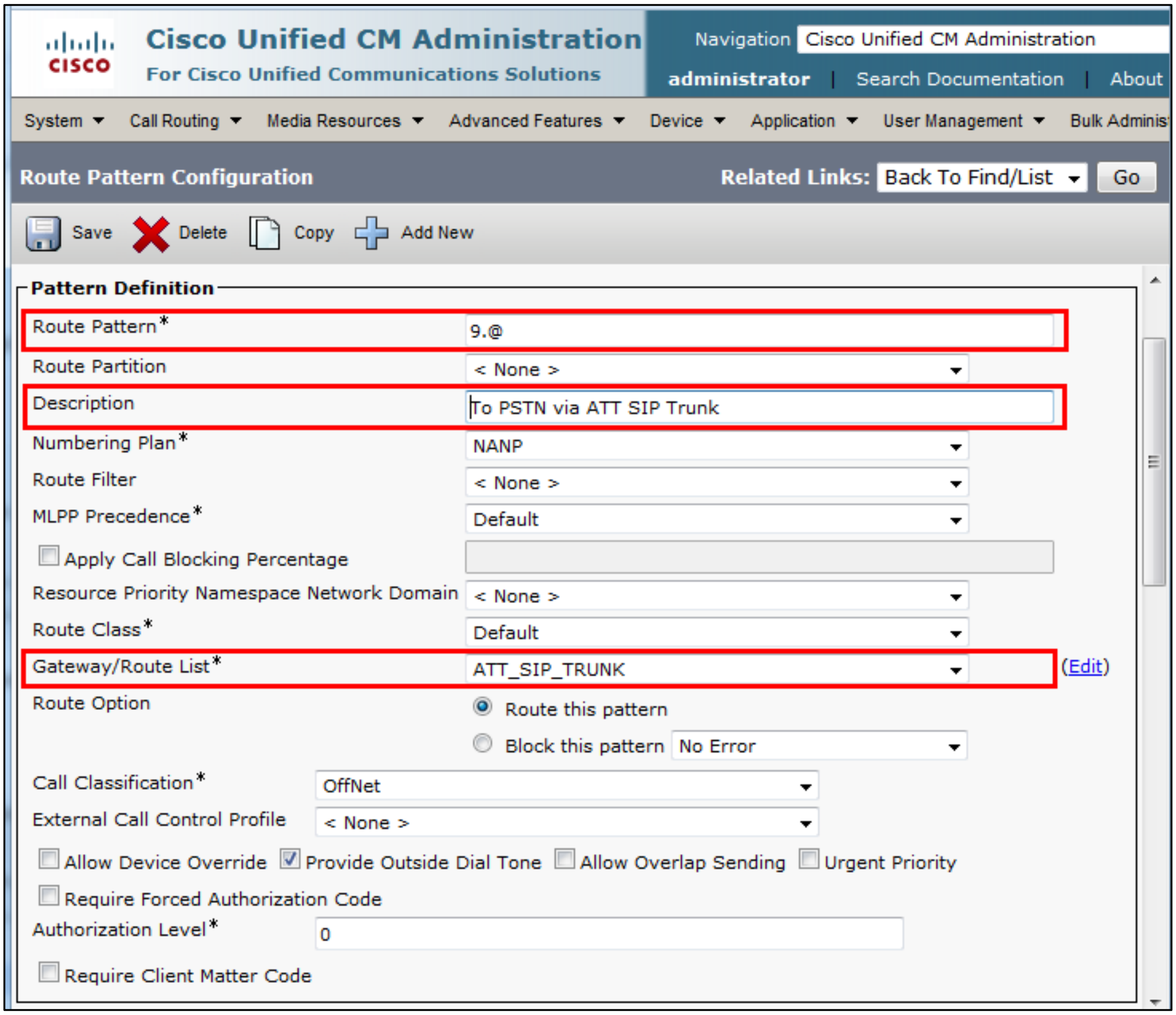
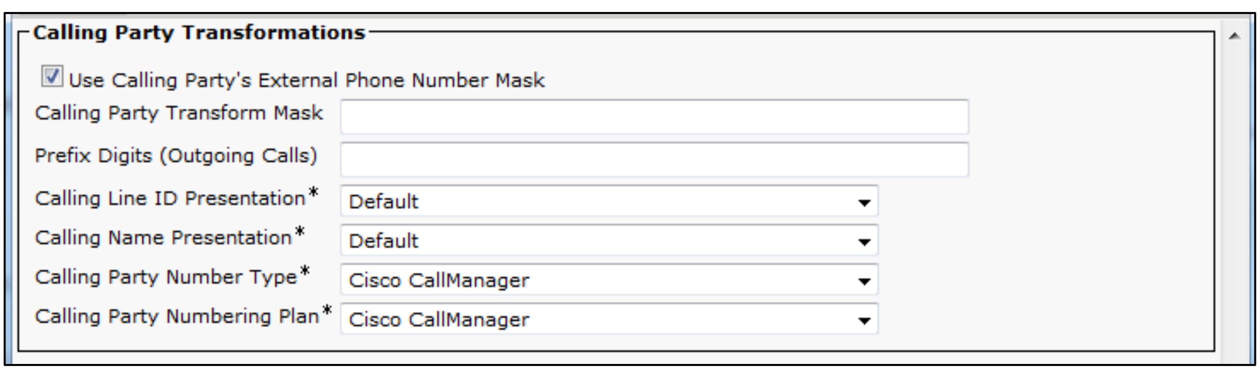
Route Pattern Configuration (Continued…)





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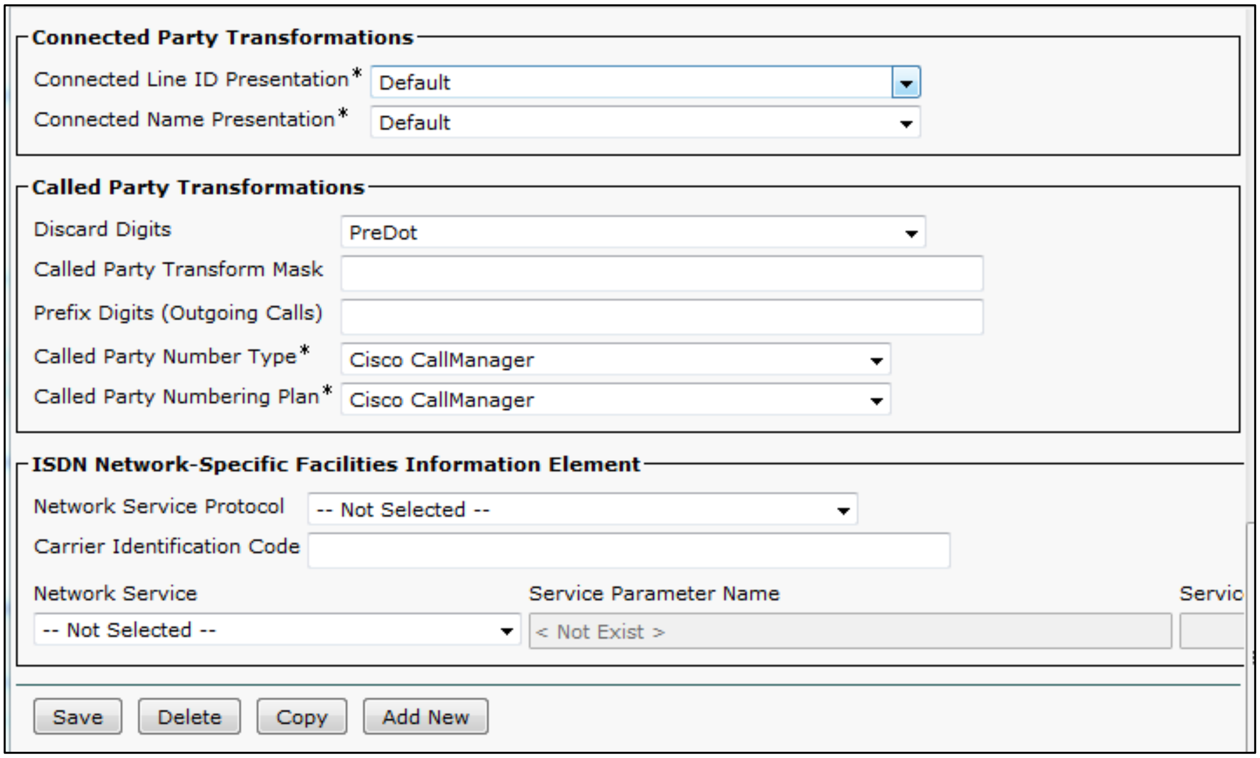
Note: Testing was conducted in tekVizion labs

Route Pattern Configuration (Continued…)



Connected Party Transformations

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Called Party Transformations

Discard Digits PreDot

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type* Cisco CallManager

Called Party Numbering Plan* Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected --

Carrier Identification Code

Network Service | Service Parameter Name | Servic

-- Not Selected -- | < Not Exist >

Save | Delete | Copy | Add New

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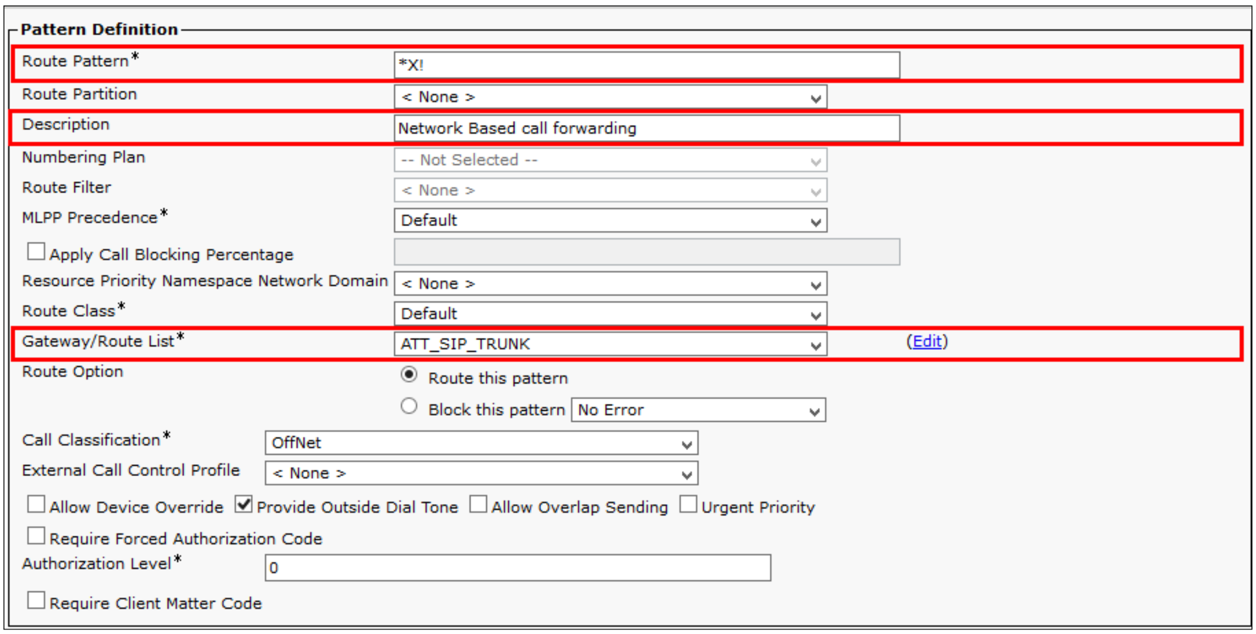
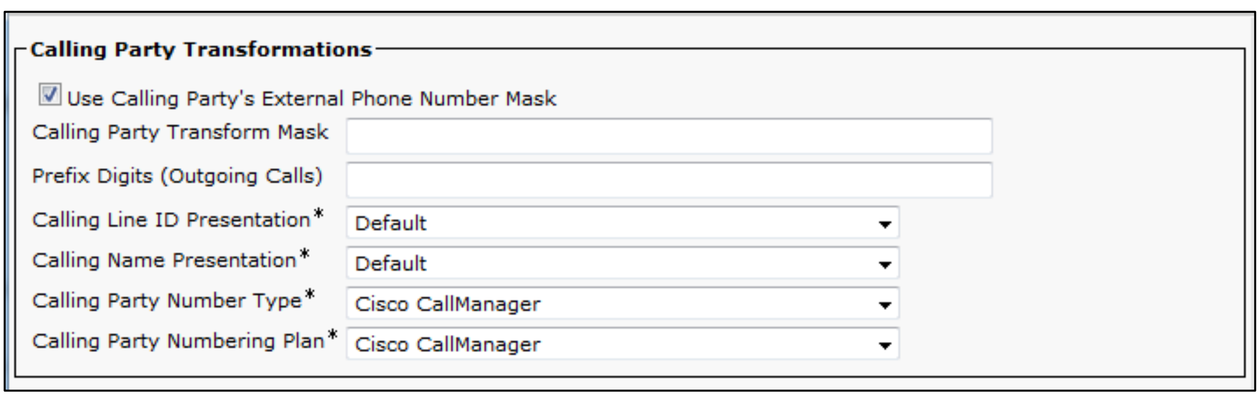
Note: Testing was conducted in tekVizion labs

Route Pattern Configuration (Continued…)

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.





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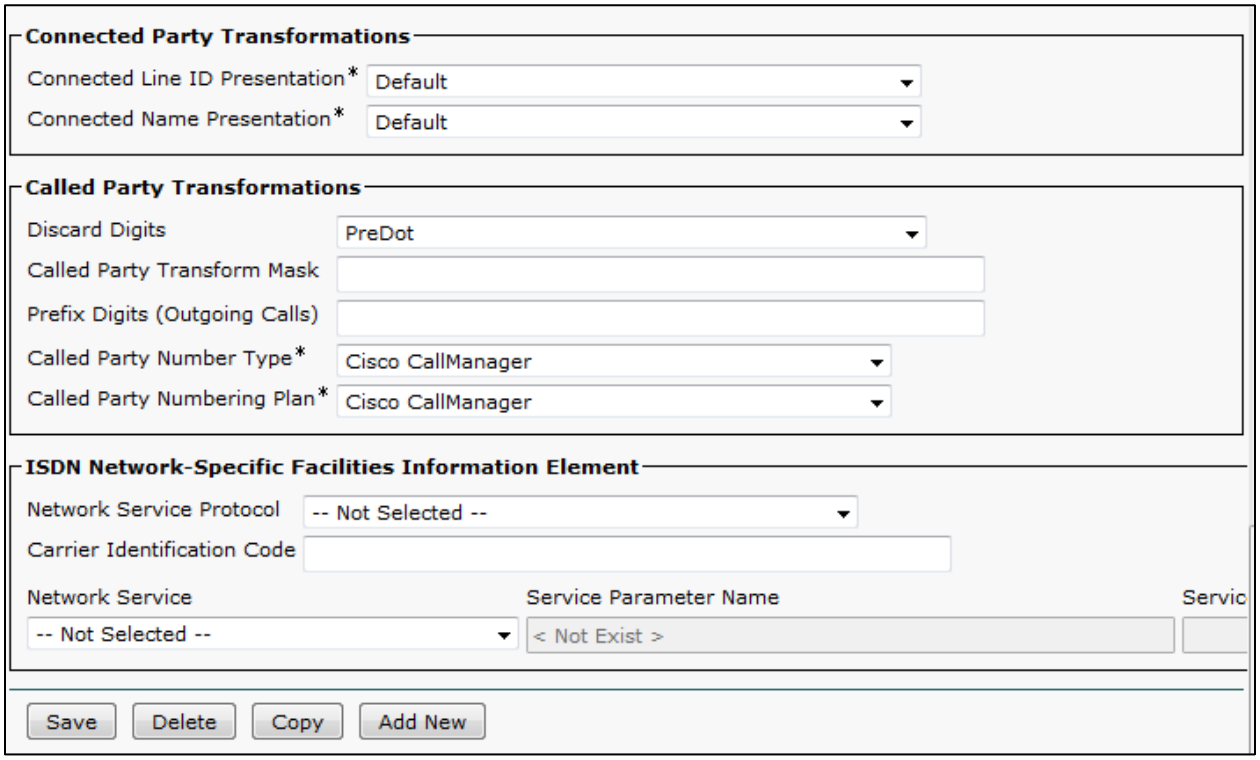
Route Pattern Configuration (Continued…)



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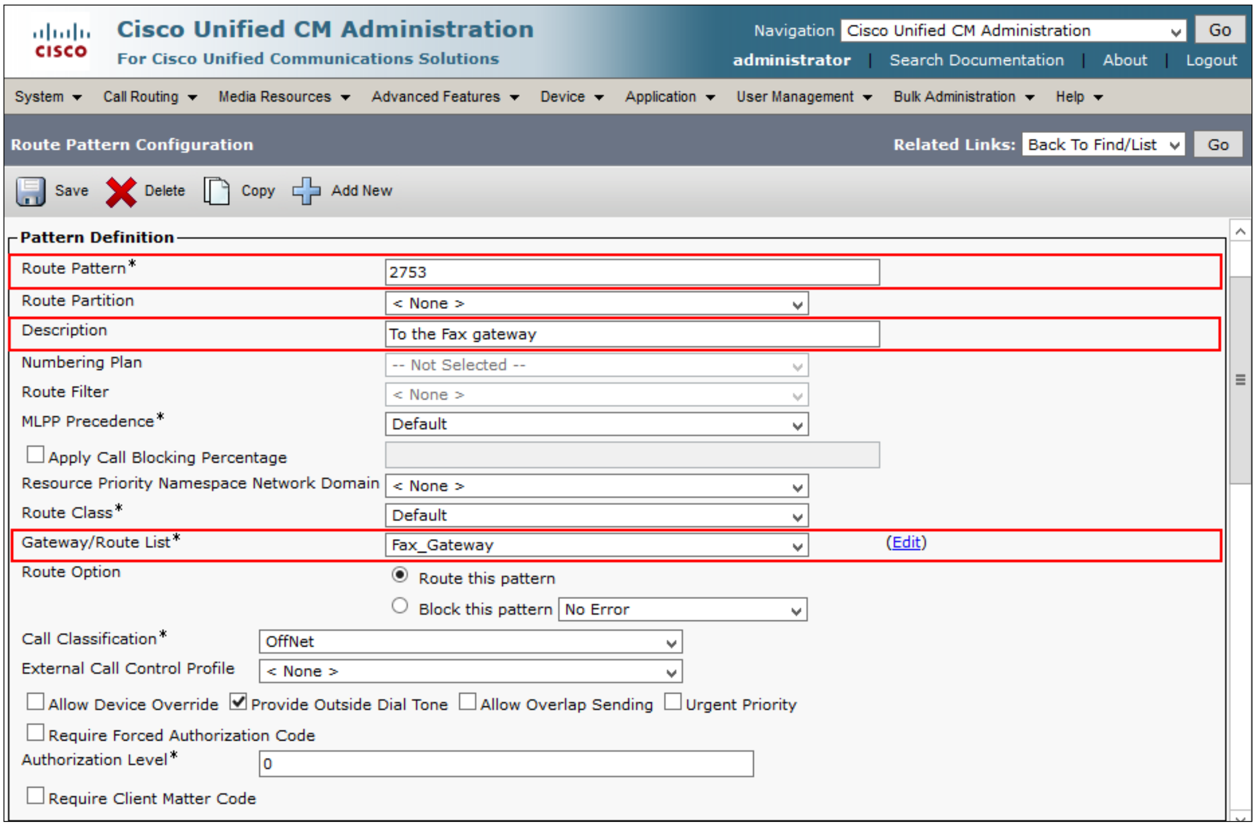
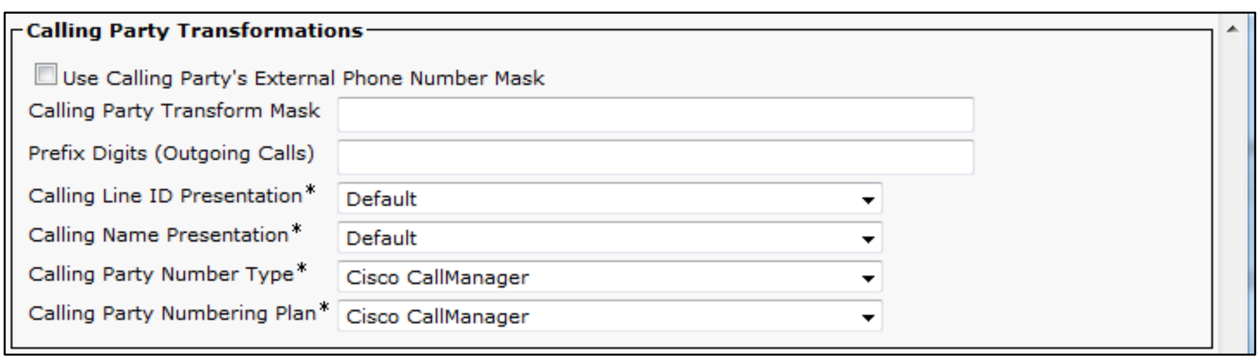
Route Pattern Configuration (Continued…)

Set Route Pattern* = 2753 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* FAX_Gateway. This is used for this example.
All other values are default





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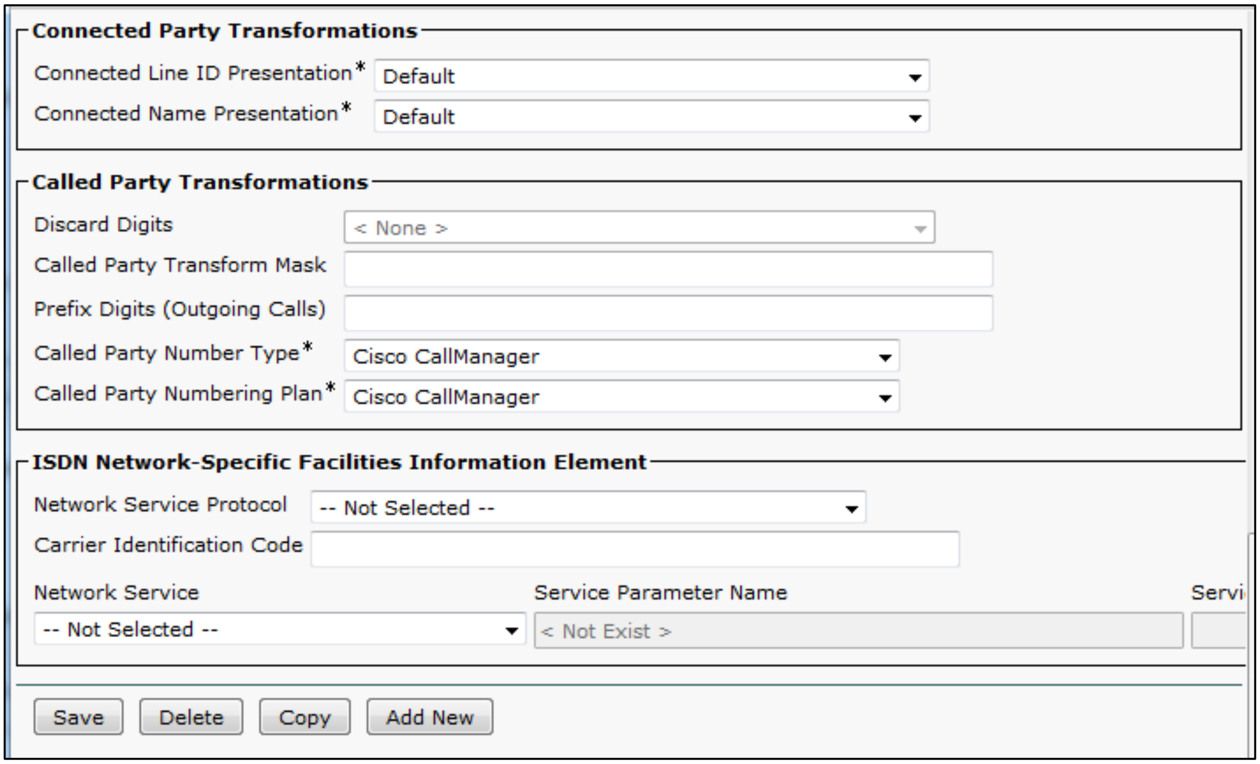
Route Pattern Configuration (Continued…)



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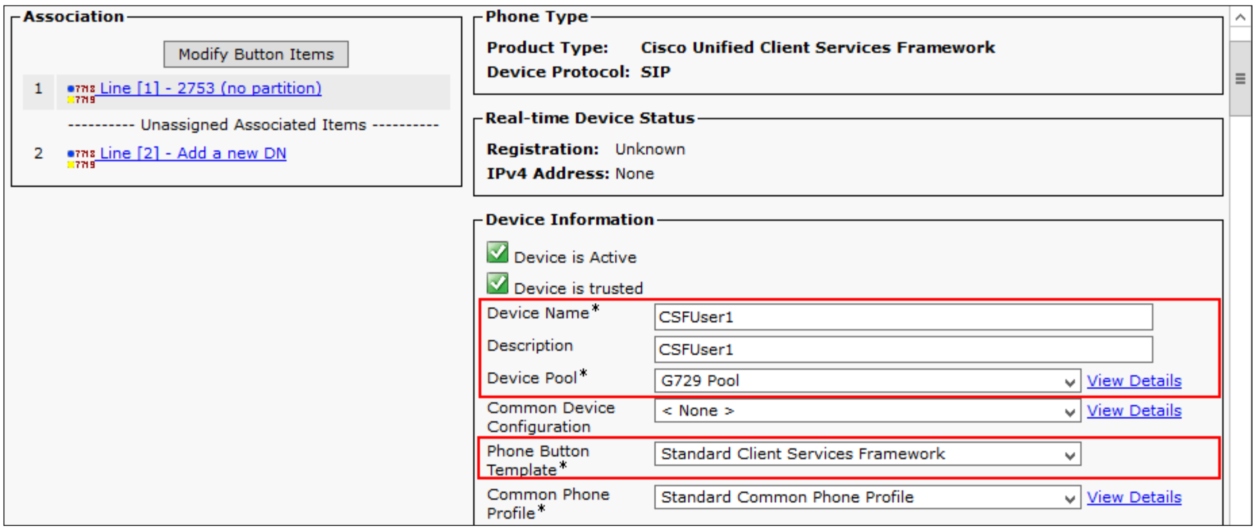
Jabber Client Configuration

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.



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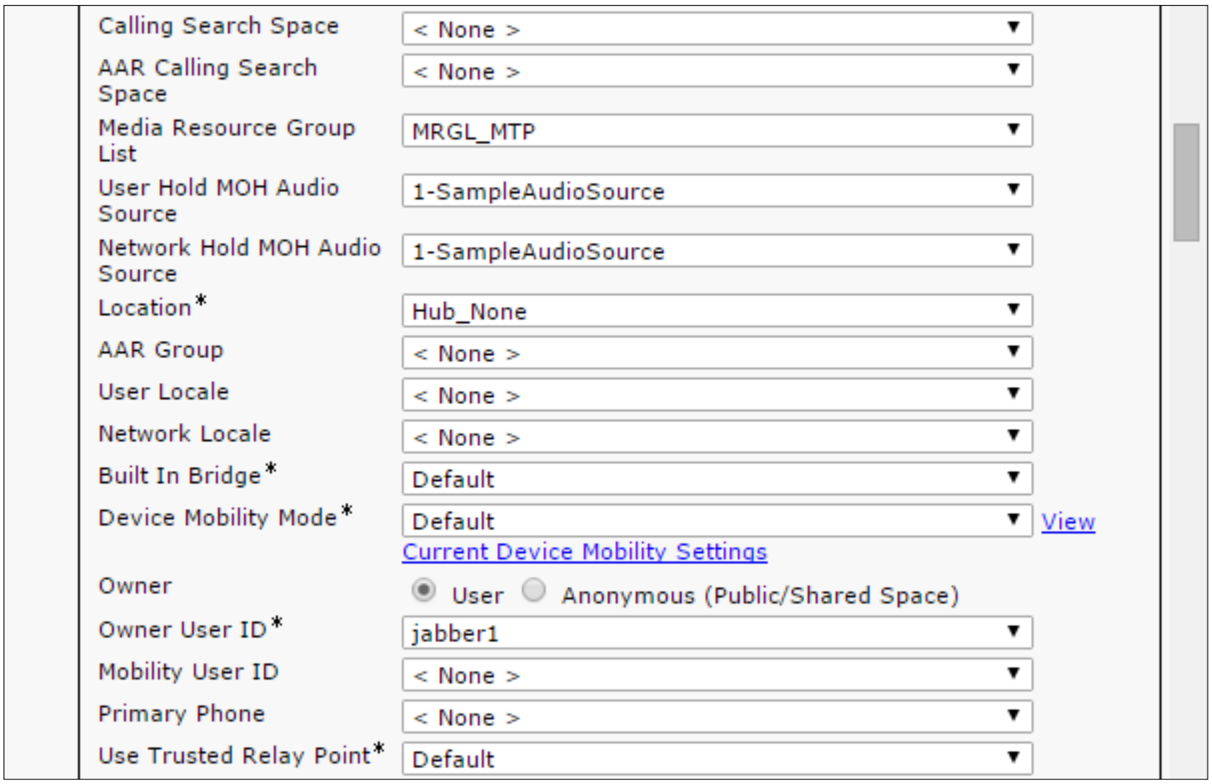
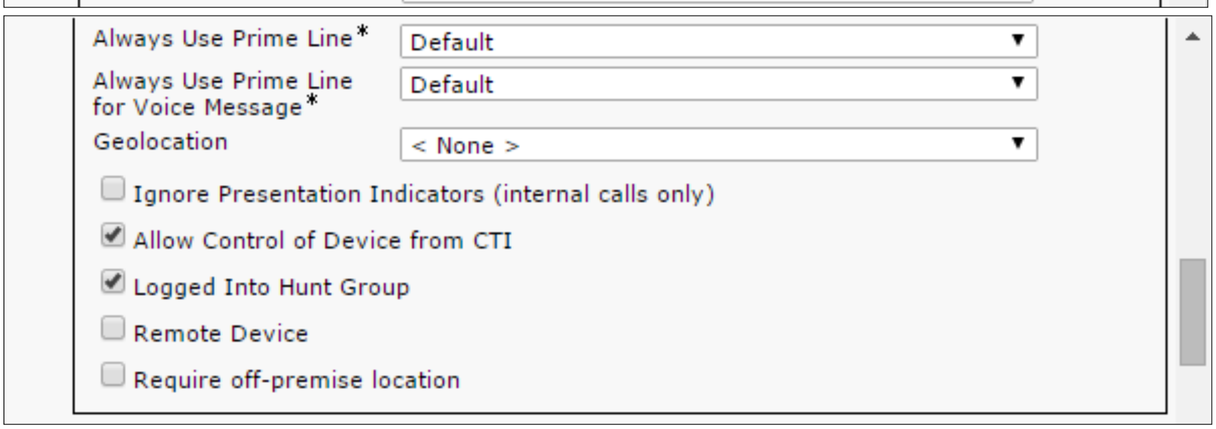
Jabber Client Configuration (Contd…)

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





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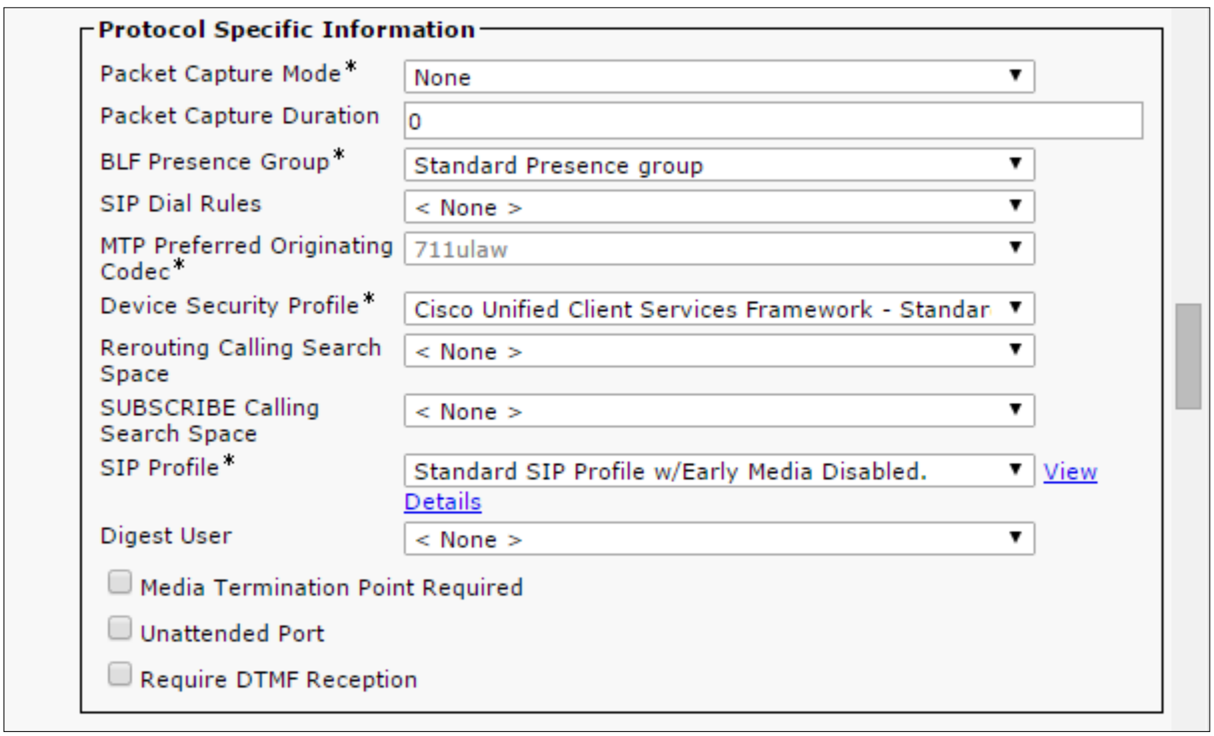
Jabber Client Configuration (Contd…)



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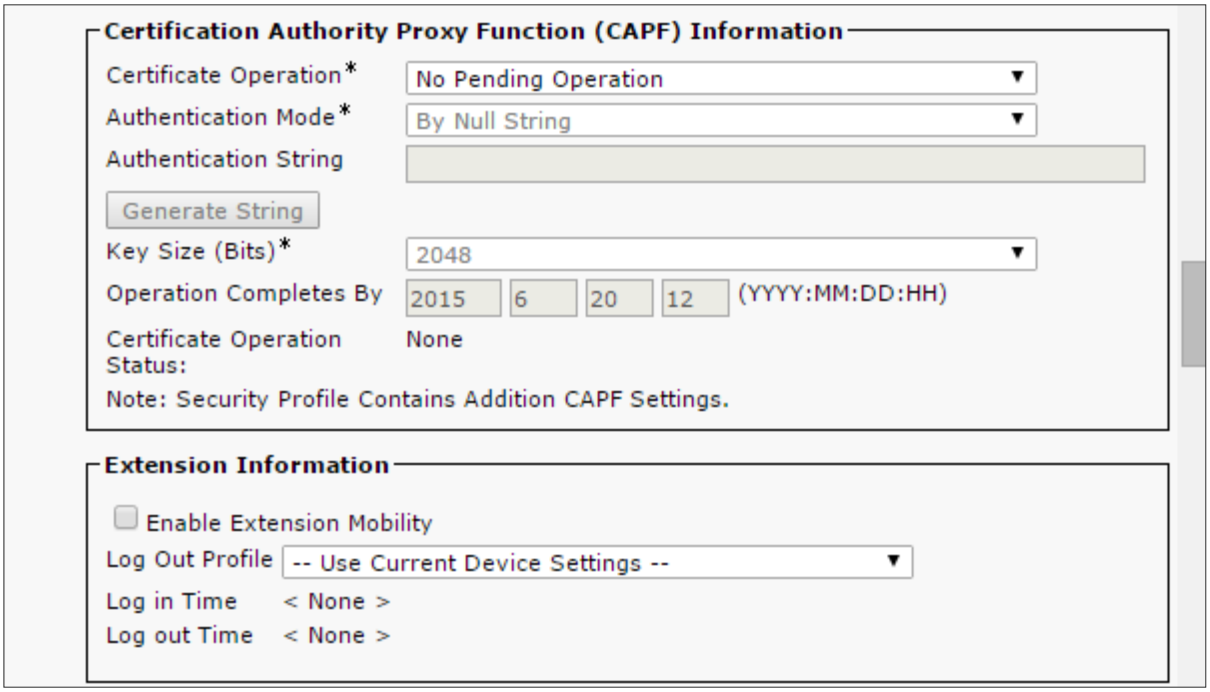
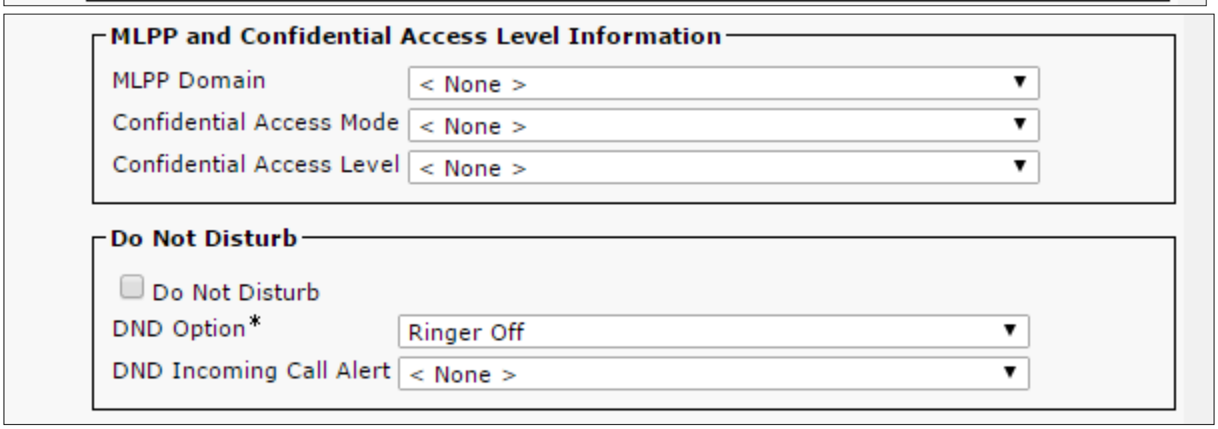
Jabber Client Configuration (Contd…)





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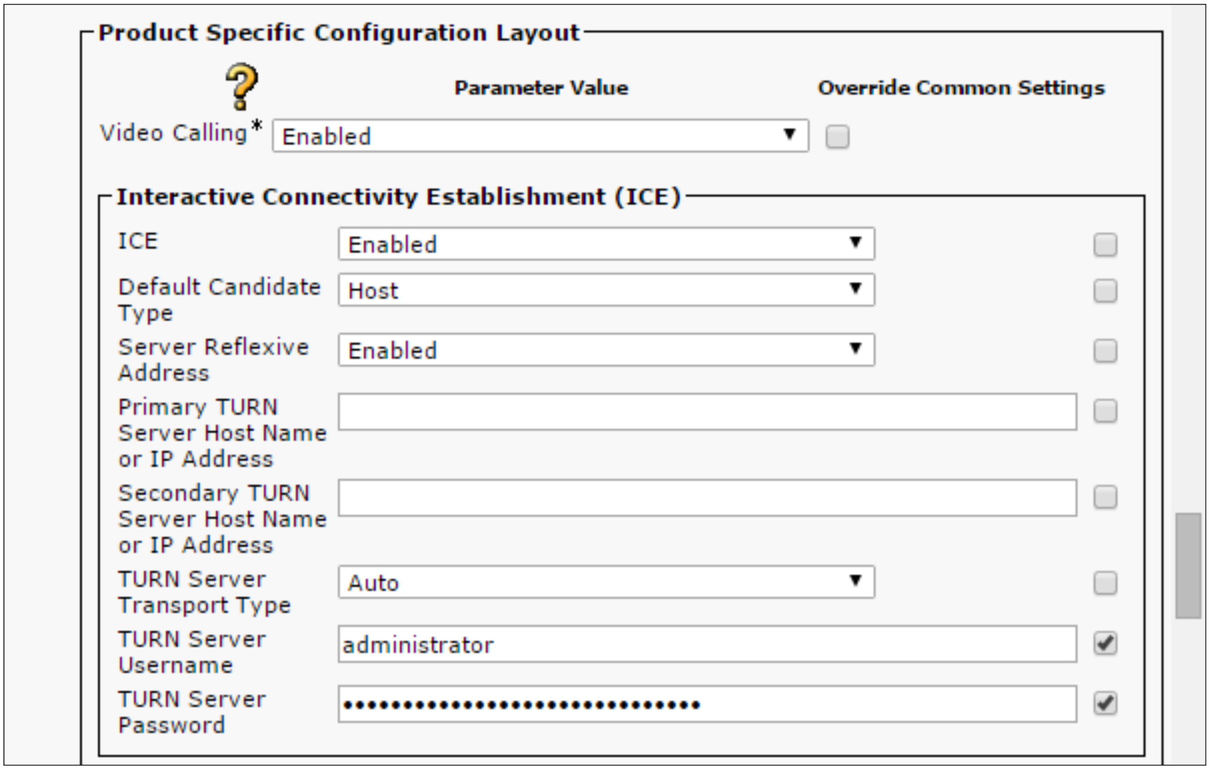
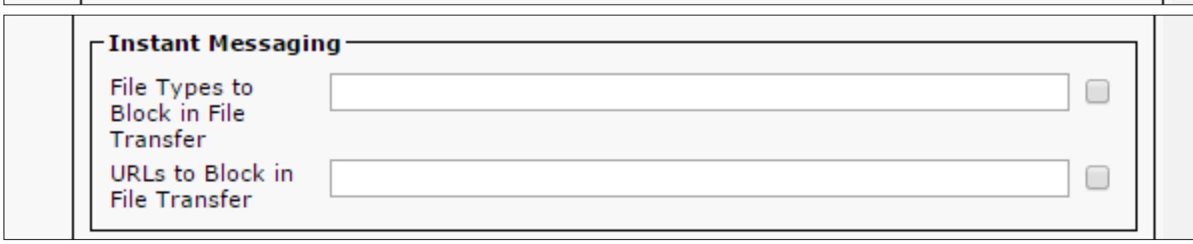
Jabber Client Configuration (Contd…)





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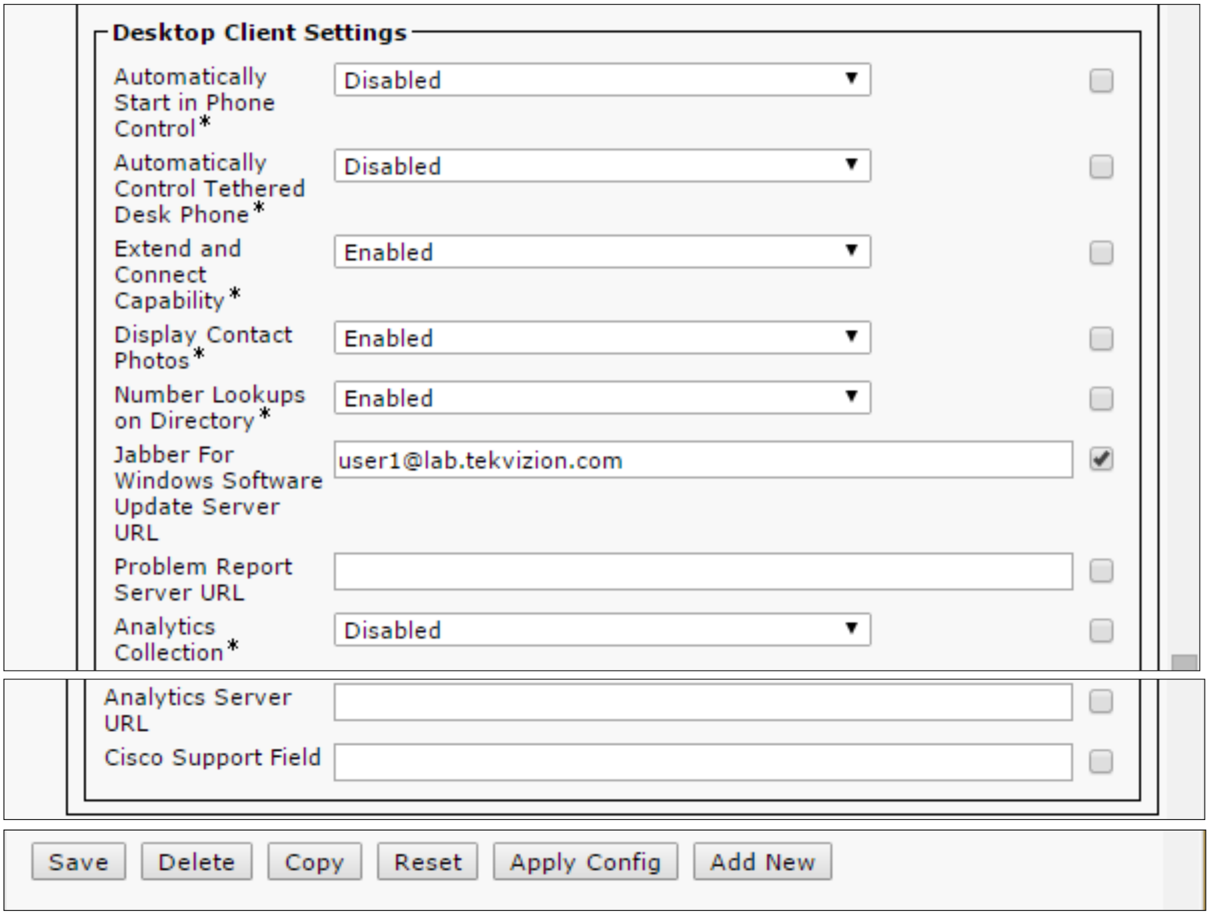
Jabber Client Configuration (Contd…)



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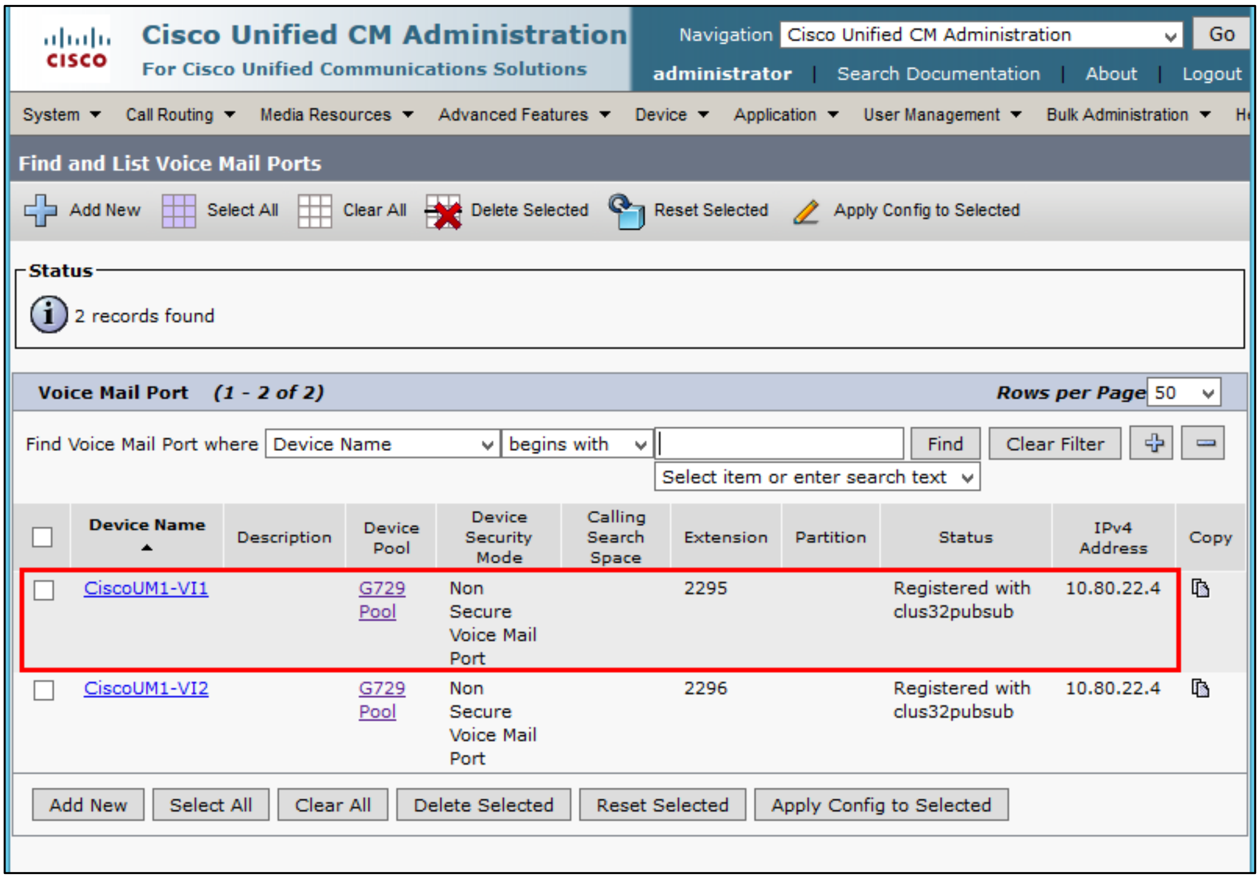
Voicemail Port Configuration

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



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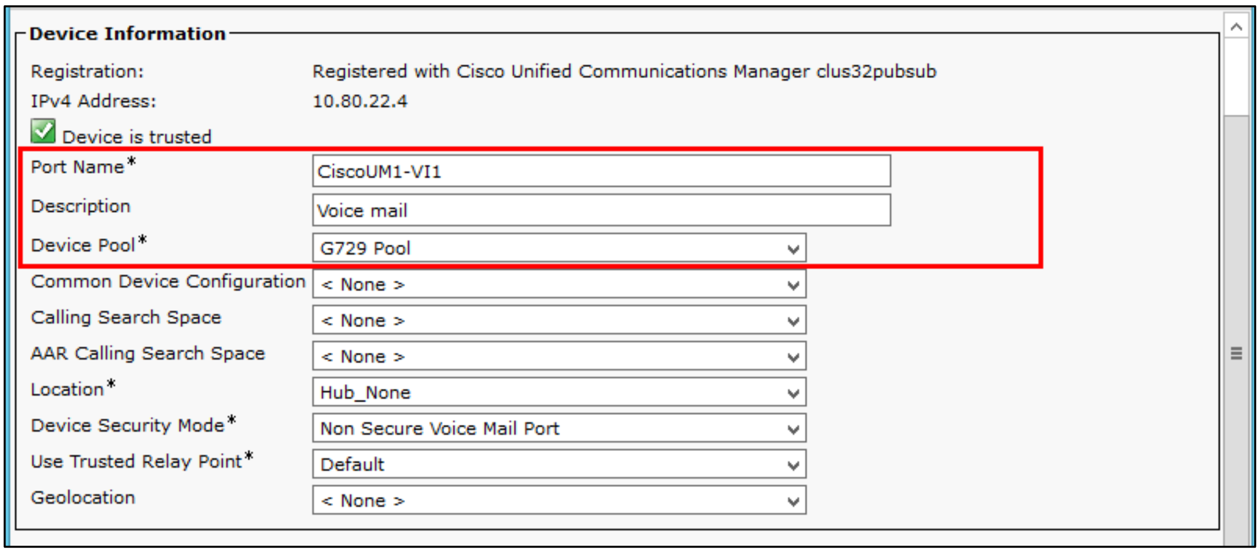
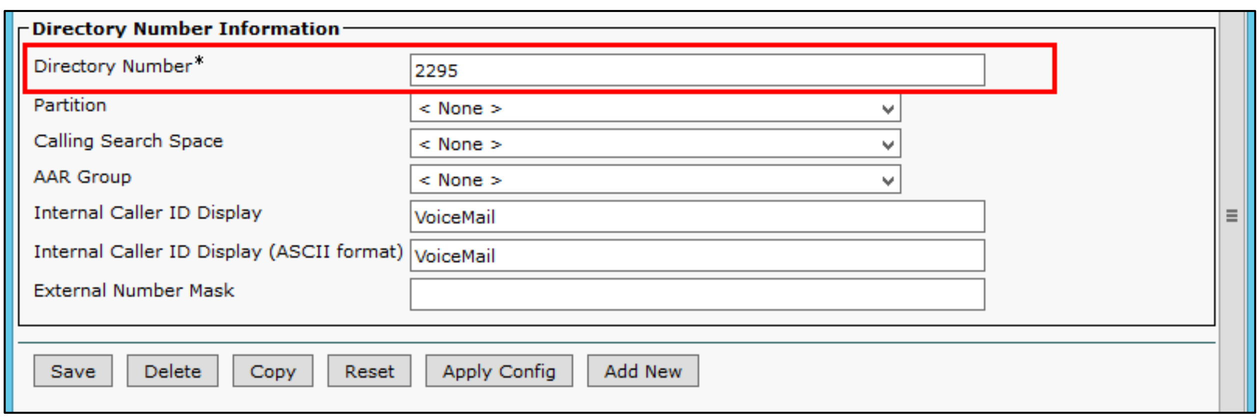
Voicemail Port Configuration (Continued…)

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* = 2295. This is used in this example.





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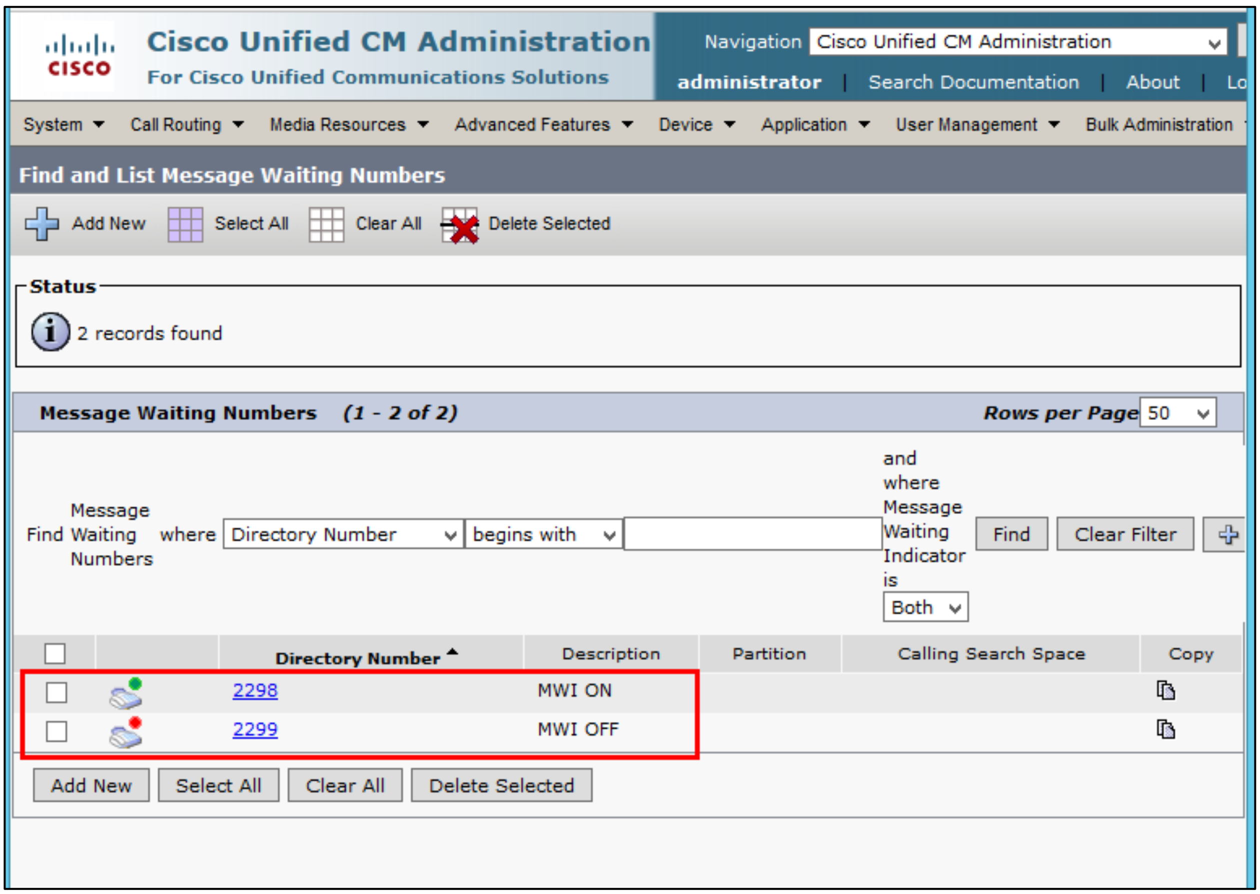
Message Waiting Numbers Configurations

Navigation: Advanced Features → Voice Mail→Message Waiting

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



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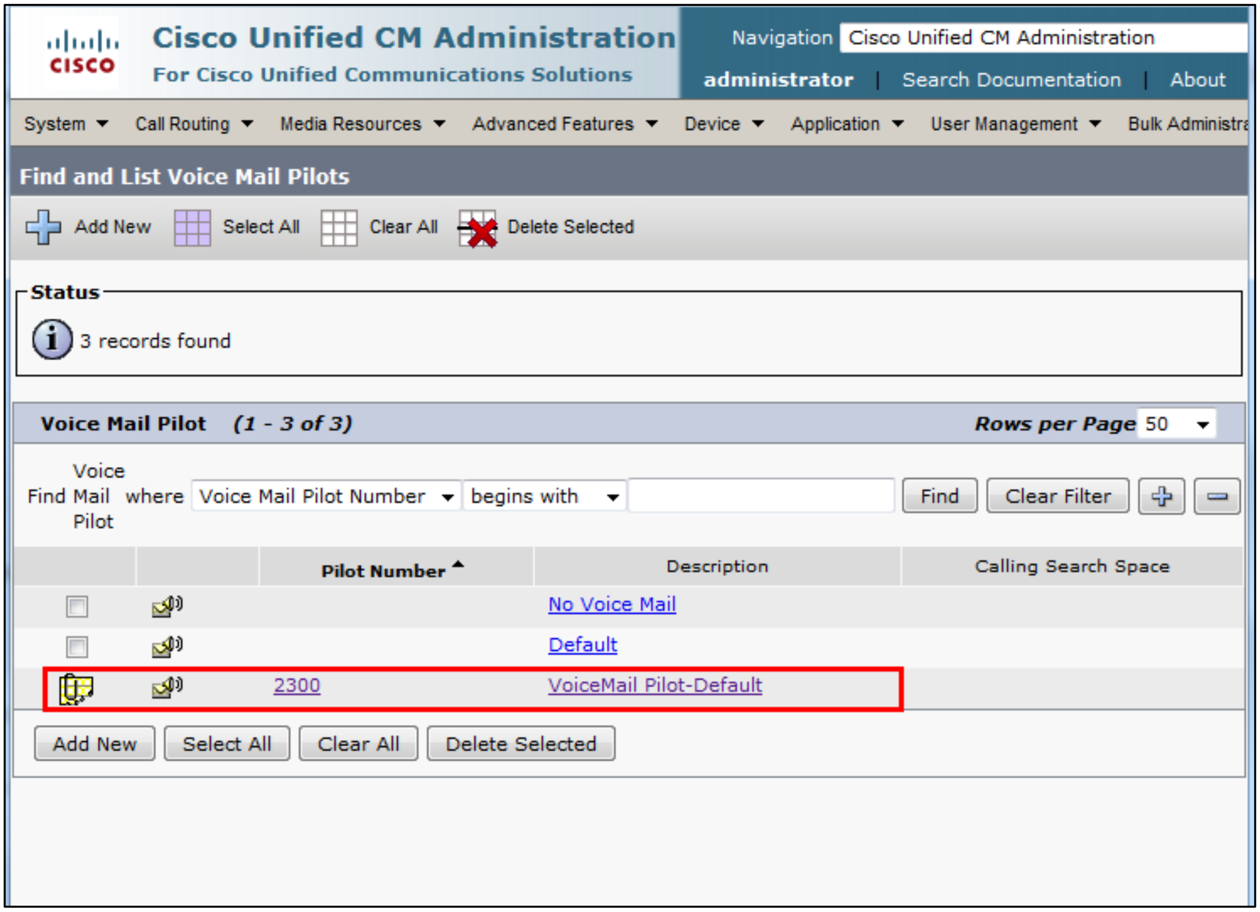
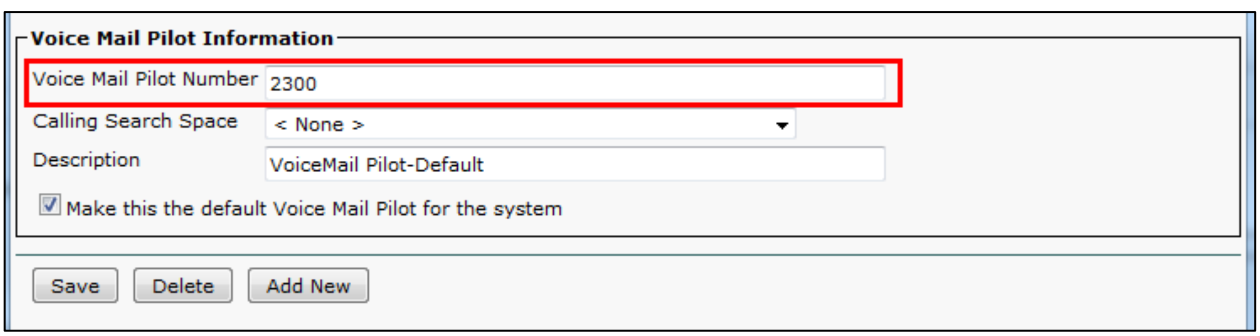
Voicemail Pilot Configuration

Navigation: Advanced Features → Voice Mail → Voice Mail Pilot

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





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FAX Gateway Configuration

```
voice service voip
 no ip address trusted authenticate
 allow-connections sip to sip
 redirect ip2ip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no fax-relay sg3-to-g3
 sip
  midcall-signaling passthru
  g729 annexb-all

voice class codec 1
 codec preference 1 g729br8 bytes 30
 codec preference 2 g711ulaw

voice class sip-profiles 1
 response ANY sip-header Allow-Header modify "UPDATE," ""
 request ANY sip-header Allow-Header modify "UPDATE," ""
 response ANY sip-header Allow-Header modify "UPDATE," ""
 response ANY sip-header Allow-Header modify "UPDATE," ""
```

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```
voice-port 0/1/1
 ring frequency 50
 no echo-cancel enable
 no vad
 cptone IN
 station-id number 2753
 caller-id enable
dial-peer voice 101 pots
 huntstop
 service session
 destination-pattern 2753
 no digit-strip
 port 0/0/1
 forward-digits all

dial-peer voice 200 voip
 description CUCM to Gateway
 service session
 session protocol sipv2
 session transport udp
 incoming called-number 2753
 voice-class codec 1
 voice-class sip profiles 1
```

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```
 dtmf-relay rtp-nte
 no fax-relay sg3-to-g3
 fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad

 dial-peer voice 201 voip
 description Gateway to CUCM
 service session
 destination-pattern [2-9]T
 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
 no fax-relay sg3-to-g3
 fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
```

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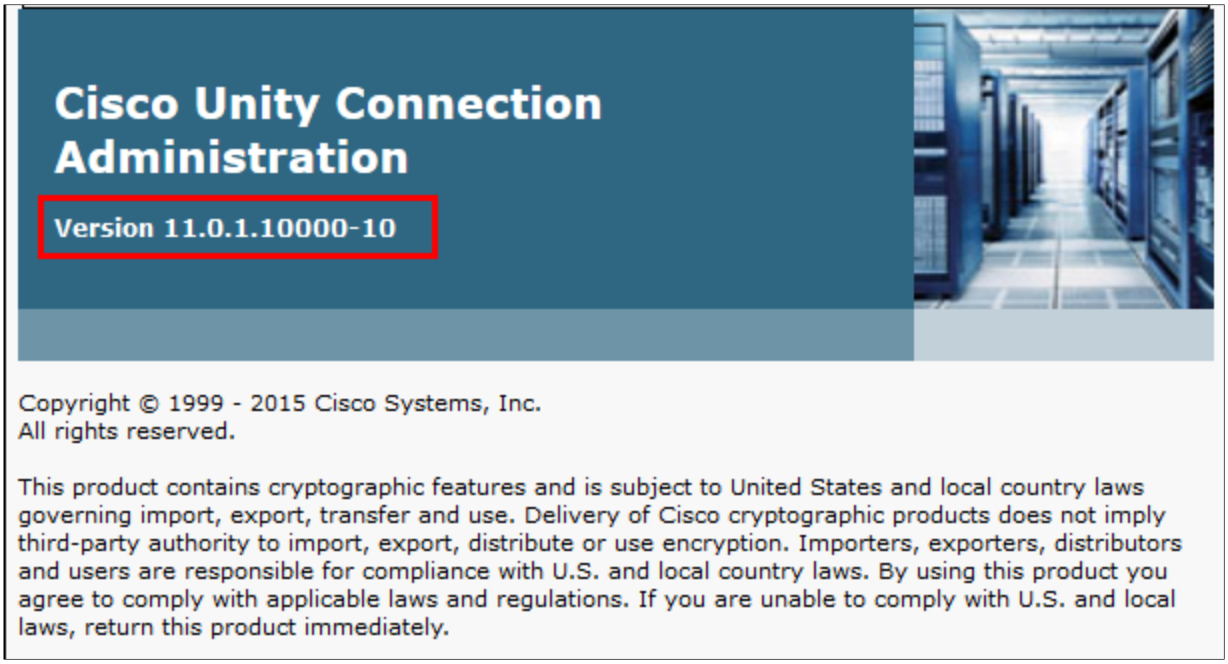
Cisco UCM SIP Integration with Cisco Unity Connection (CUC)

CUC Version



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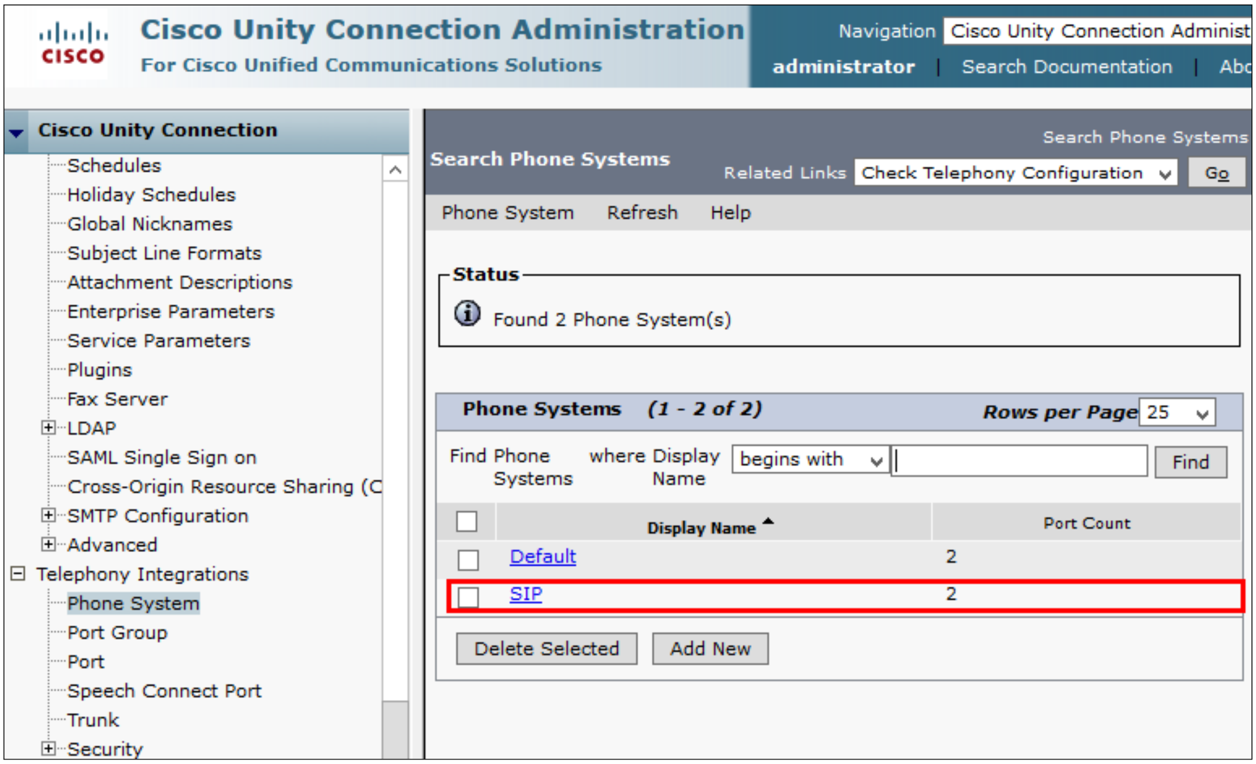
CUC Telephony Integration with Cisco UCM

Navigation: Telephony Integrations → Phone system

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



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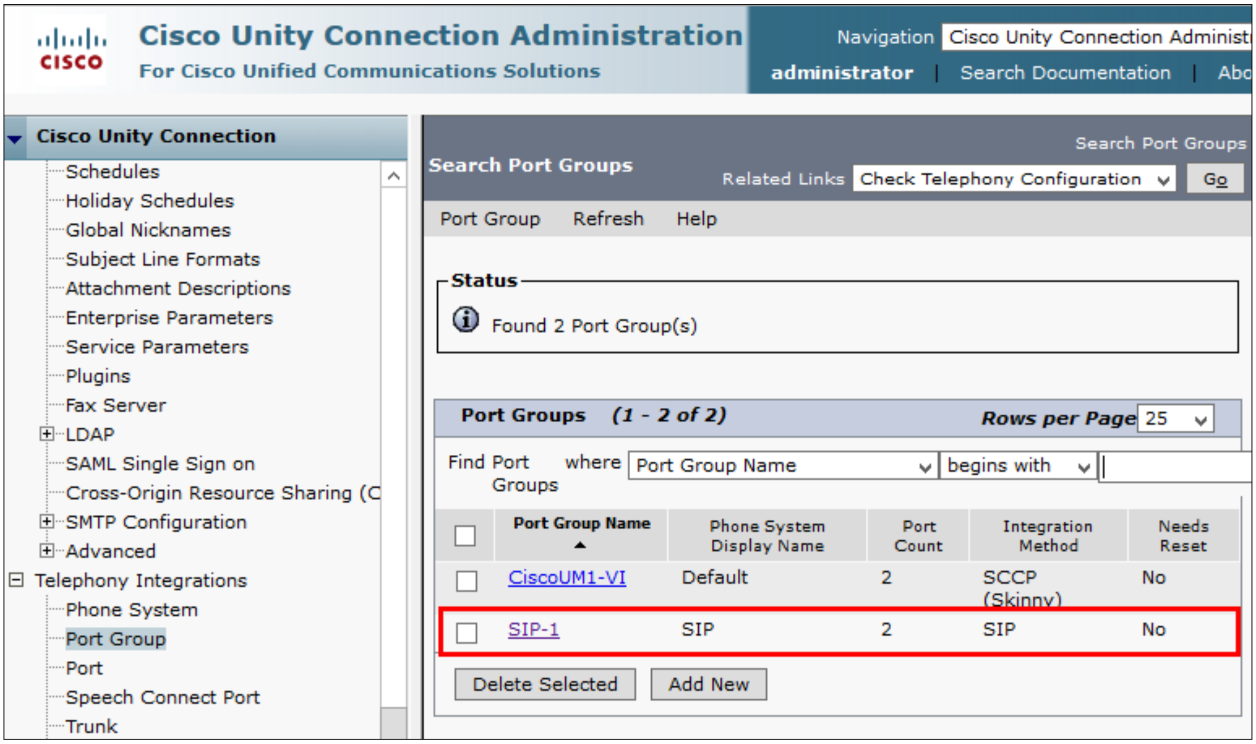
CUC Port Group

Navigation: Telephony Integration → Port Group



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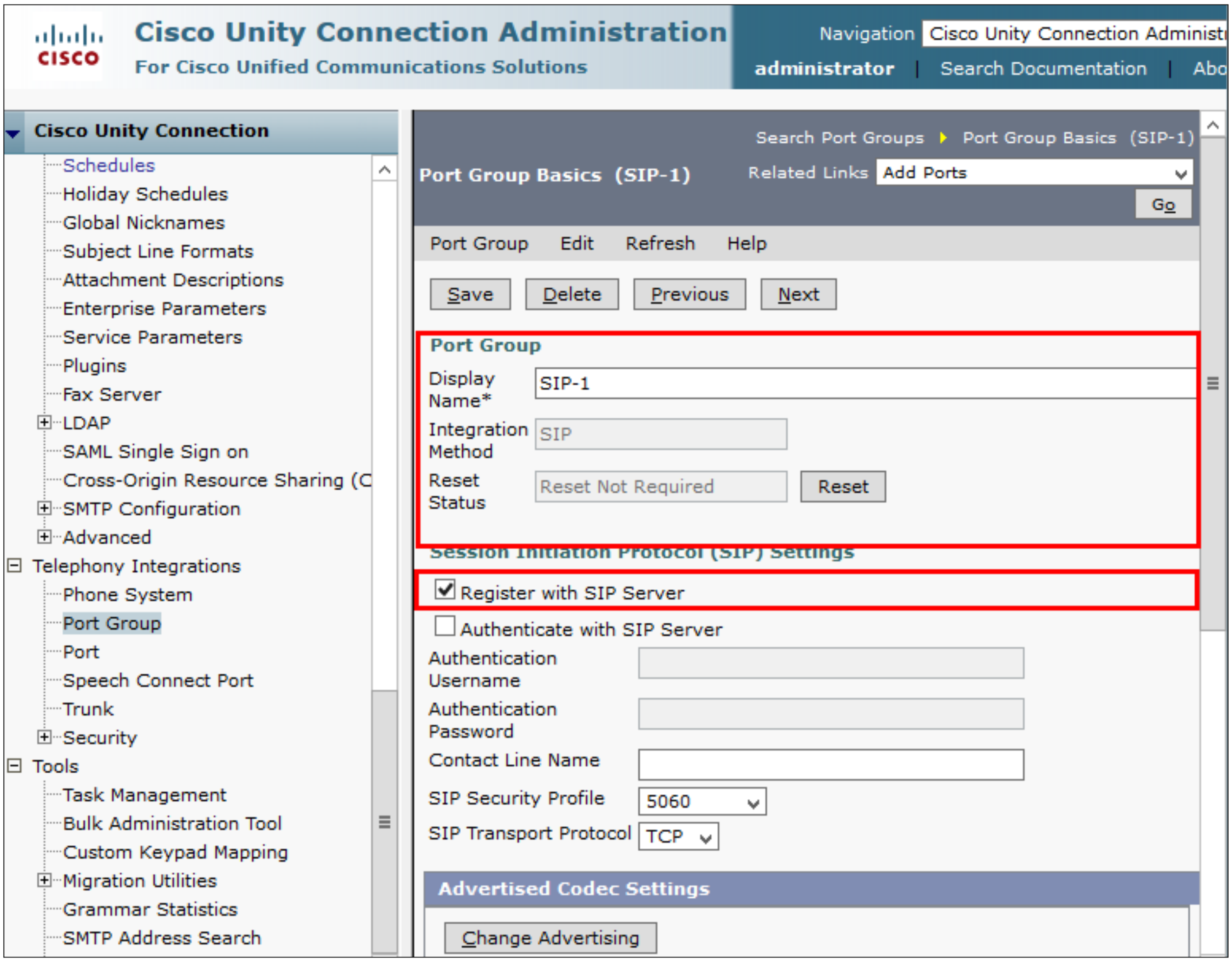
CUC Port Group(continued…)

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

Check Register with SIP Server.



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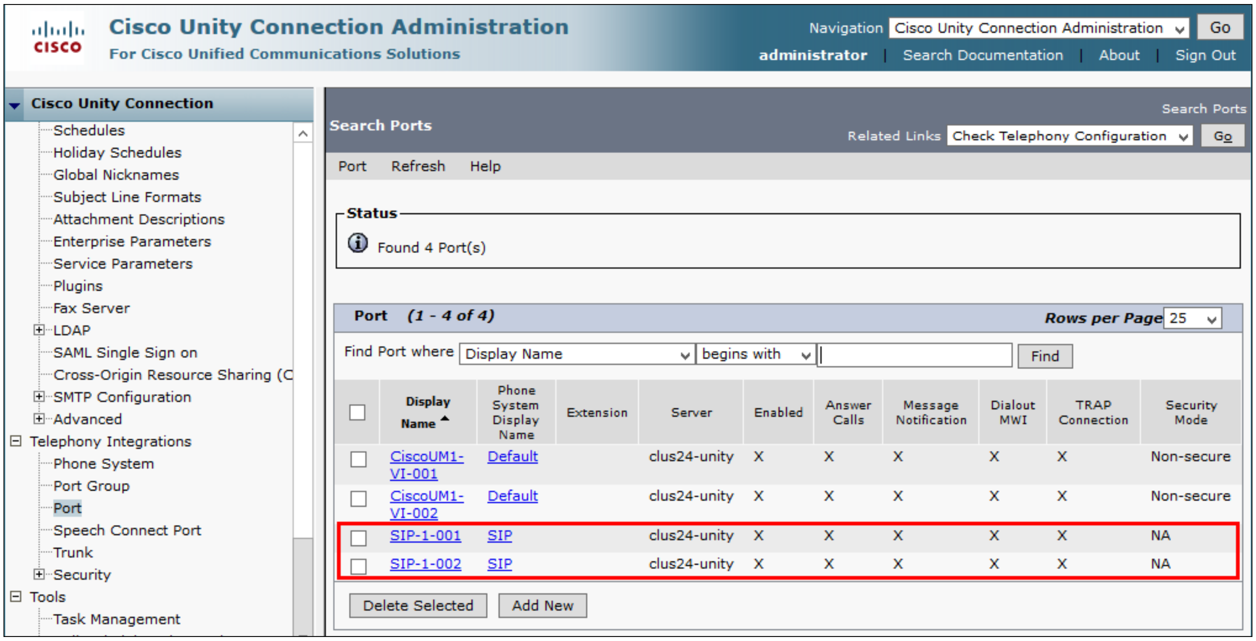
CUC Port Settings



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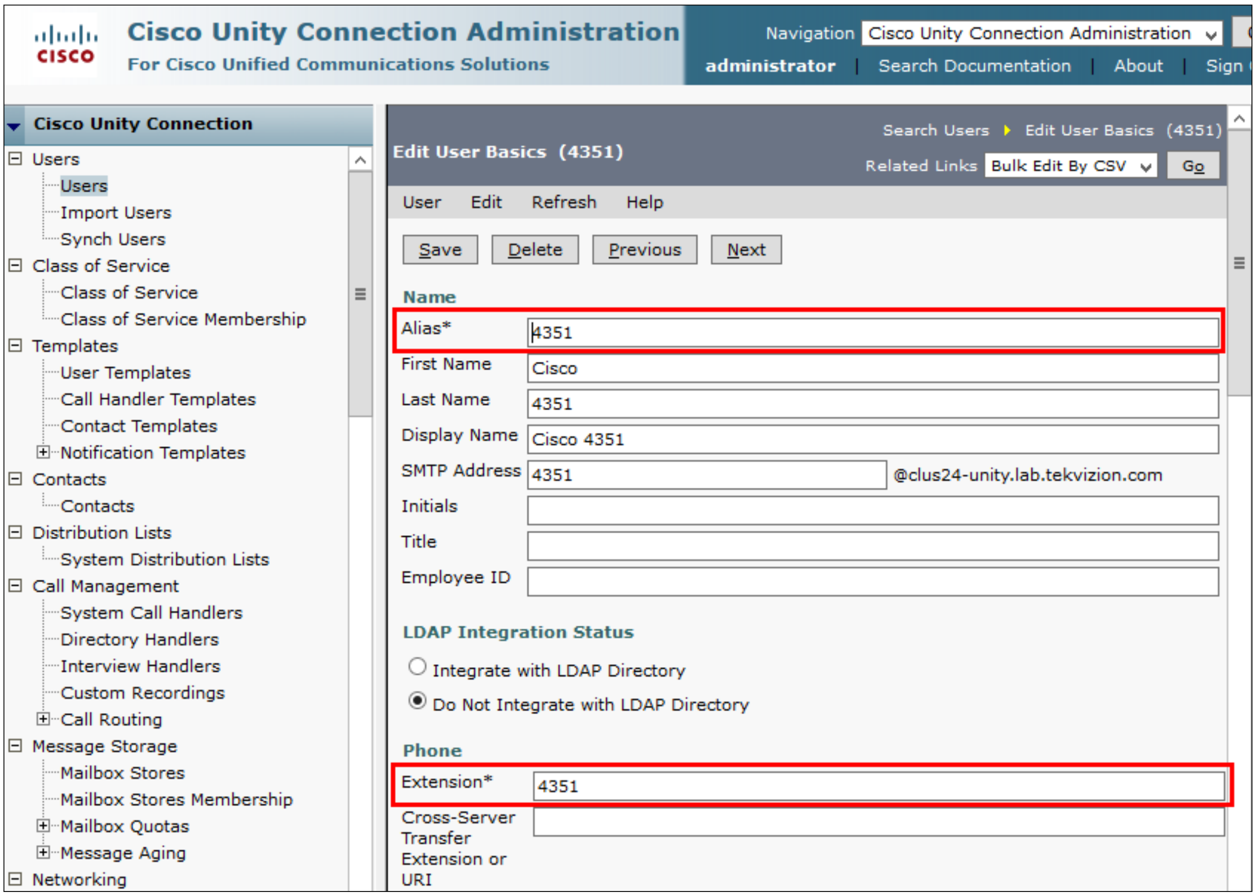
CUC Sample User Basic Settings

Navigation: Cisco Unity connection → Users → Users

Set Alias = 4051.This is one of the extension used for this testing.
Set Extension = 4051. This is used for this example.



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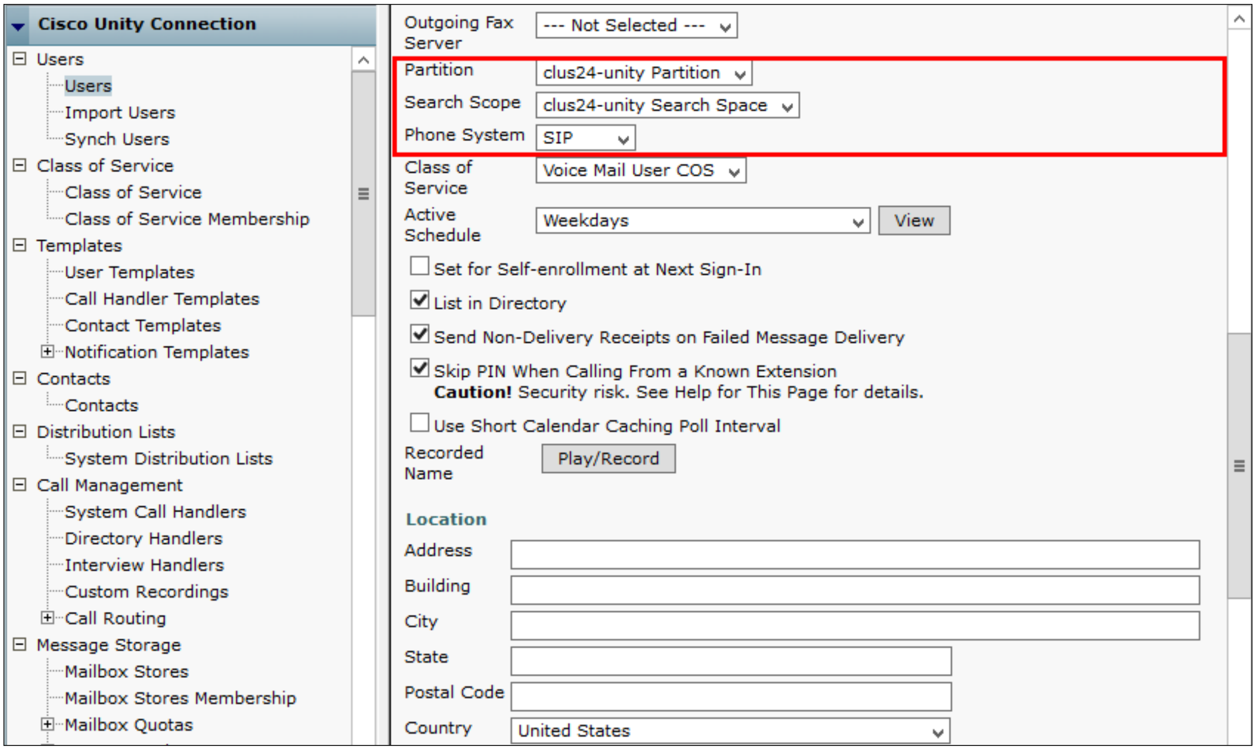
CUC Sample User Basic Settings (Continued…)

Set Partition = clus24-unity partition. This is used for this example.
Select Search Scope = clus24-unity Search Scope.
Select Phone System = SIP.



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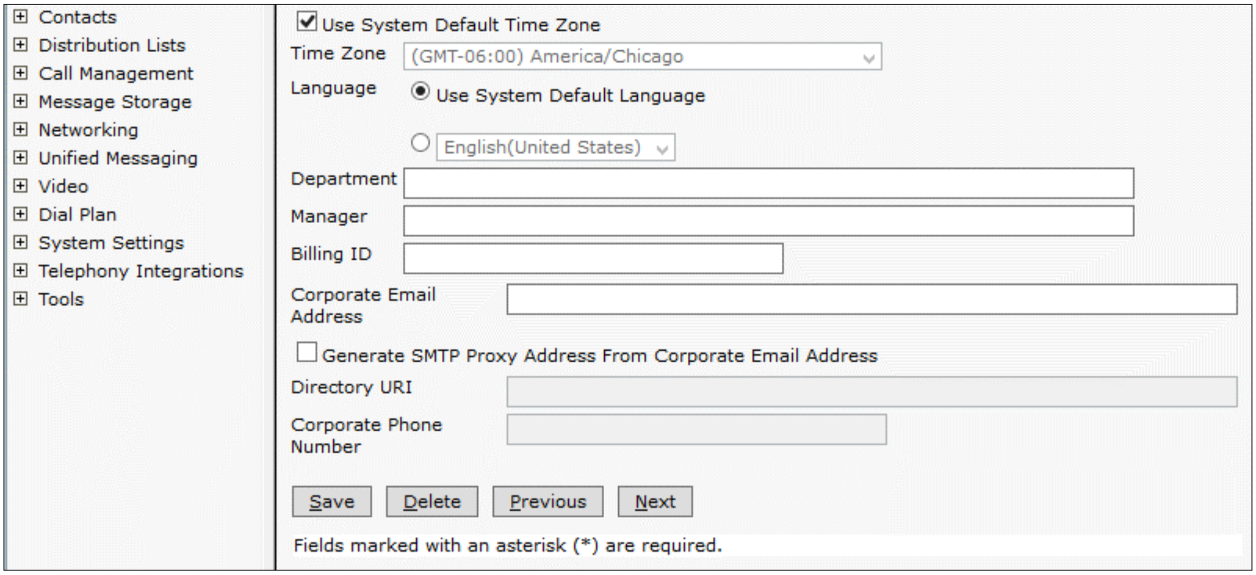

Auto Attendant

Navigation: Call Management → System Call Handlers

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

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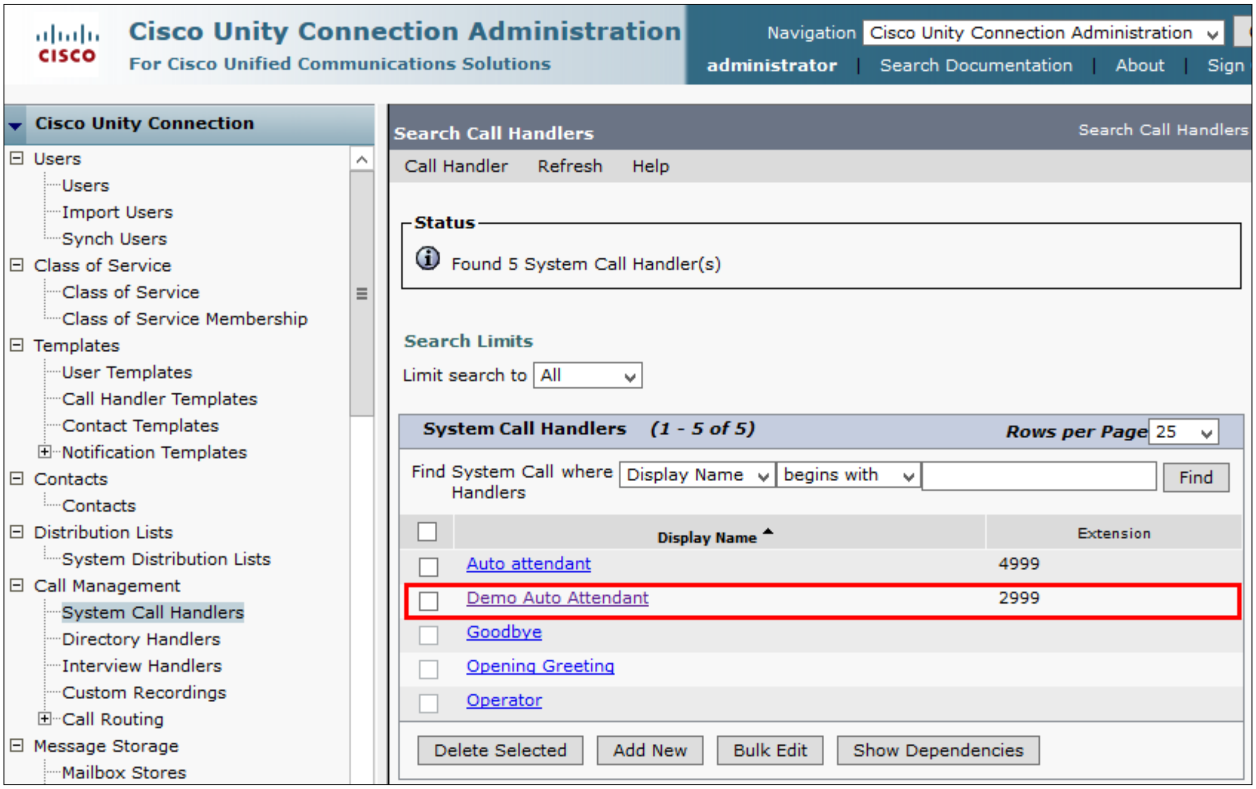

Auto Attendant (Continued…)

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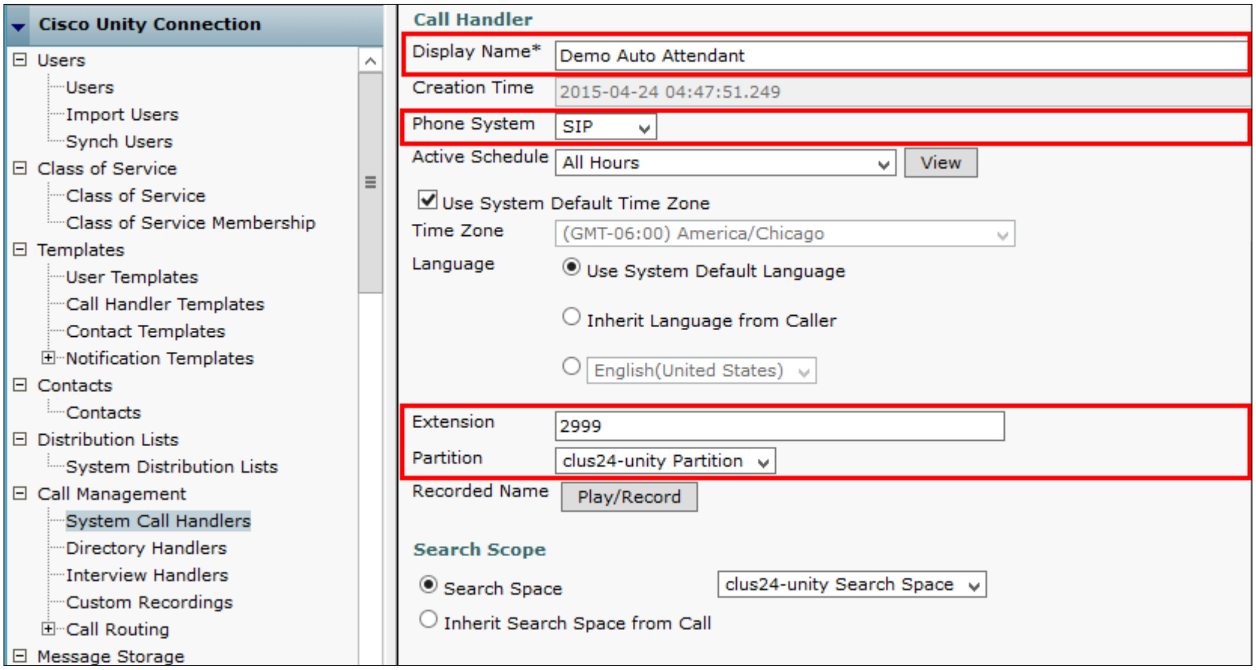

Cisco UCM Integration with Cisco Unified CM IM and Presence (CUP/IMP)

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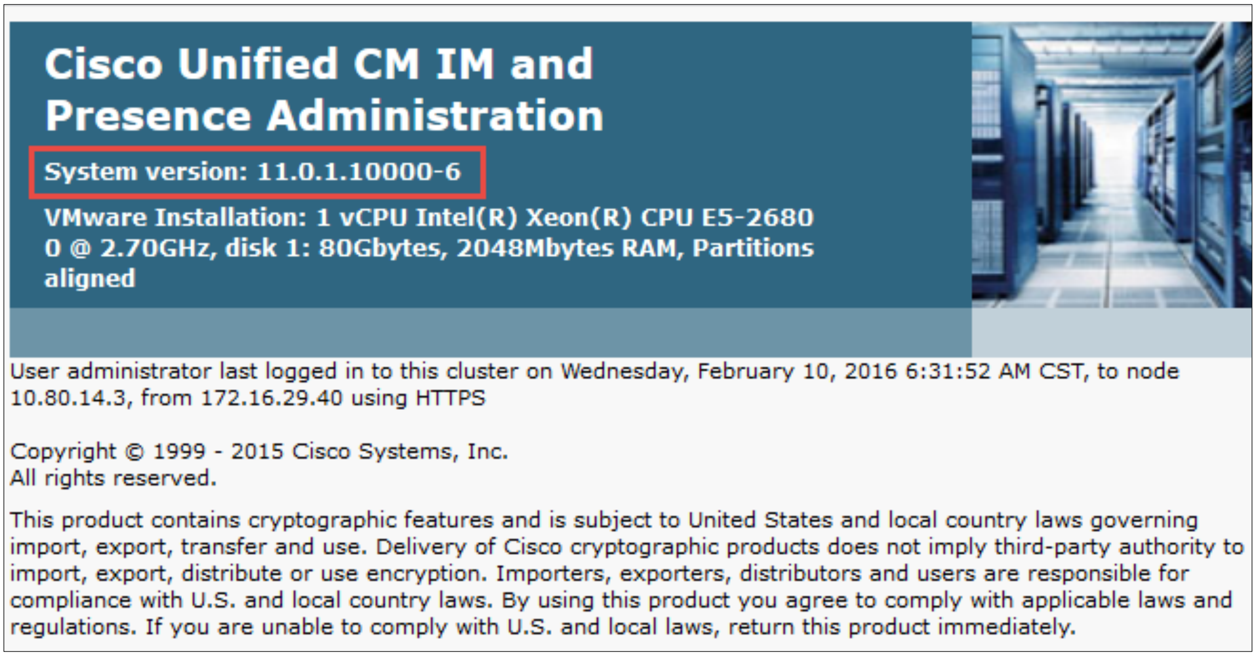
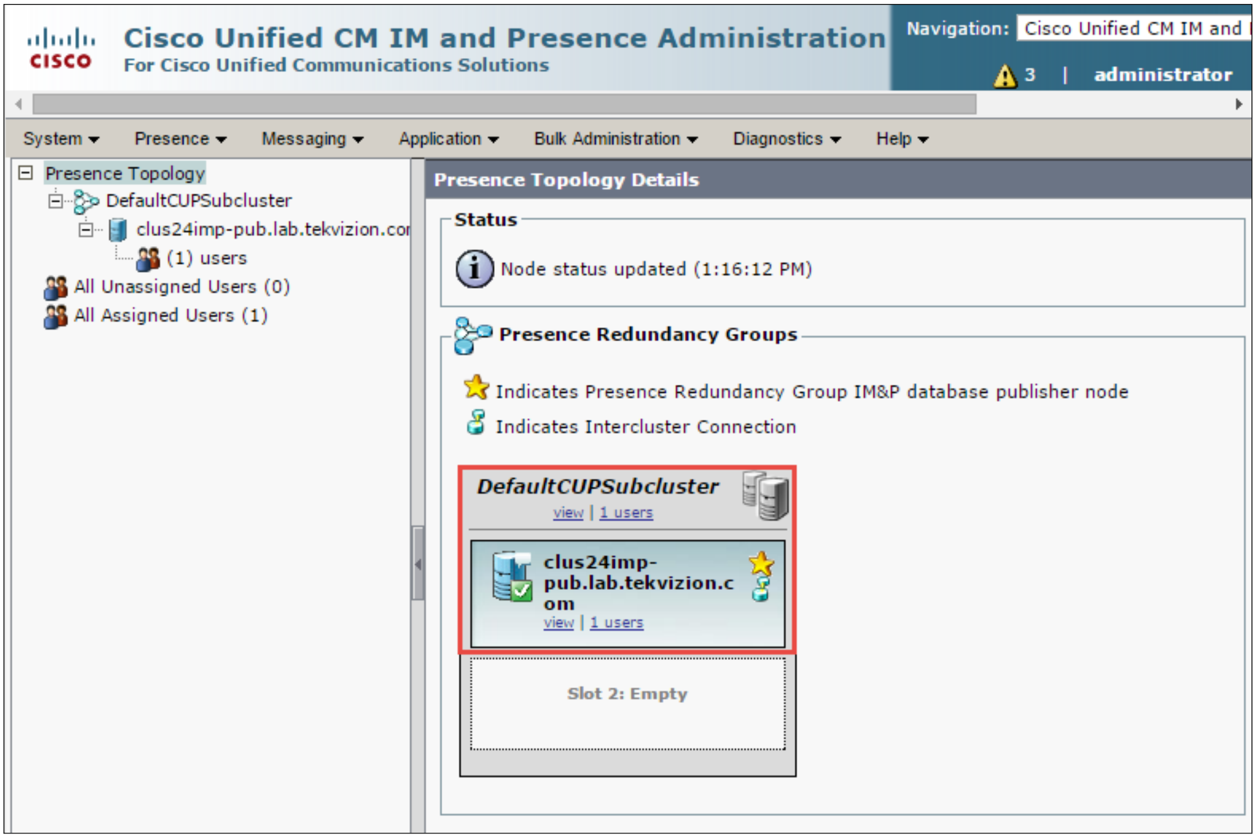
CUP/IMP Version



Presence Topology

Navigation: System → Presence Topology



Node Configuration

Navigation: System → Presence Topology → Fully Qualified Domain Name

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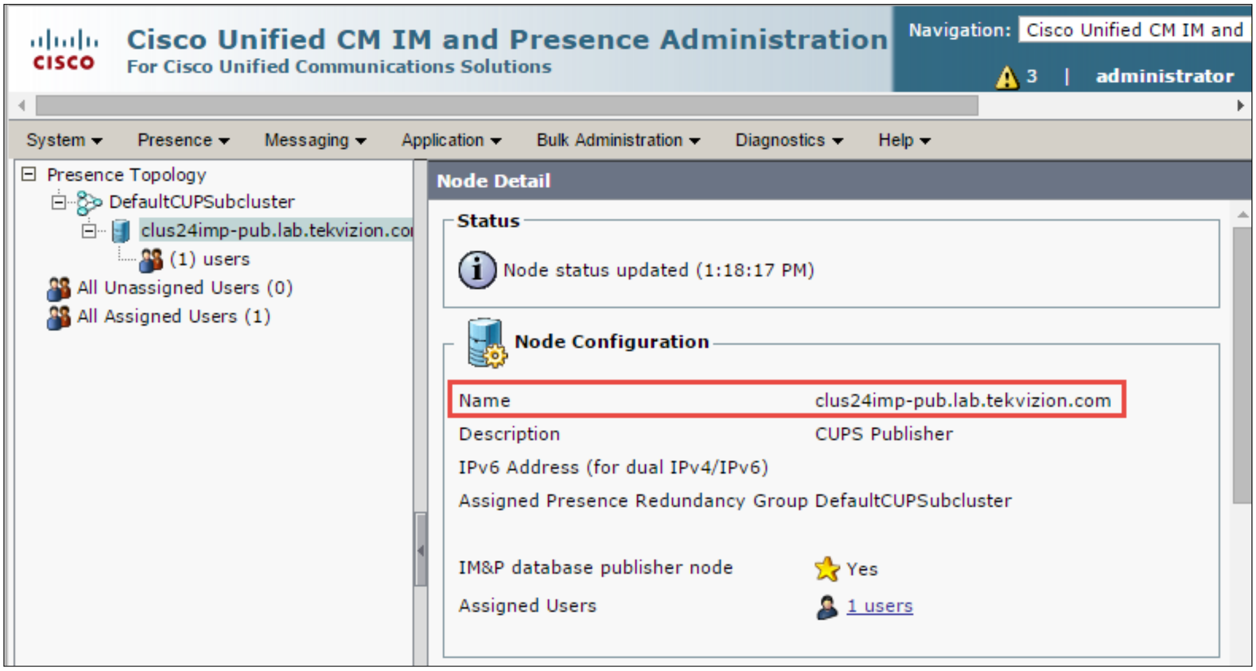

Users
Navigation: System → Cluster Topology → clus24imp.lab.tekvizion.com → Users

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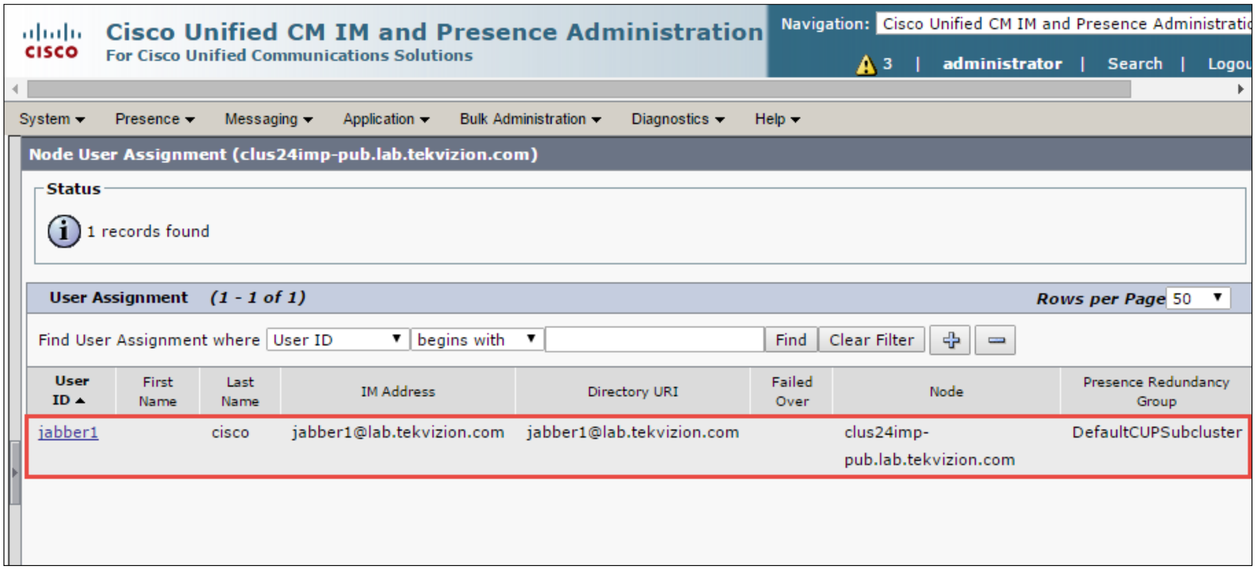

Presence gateway configuration

Navigation: Presence → Gateways

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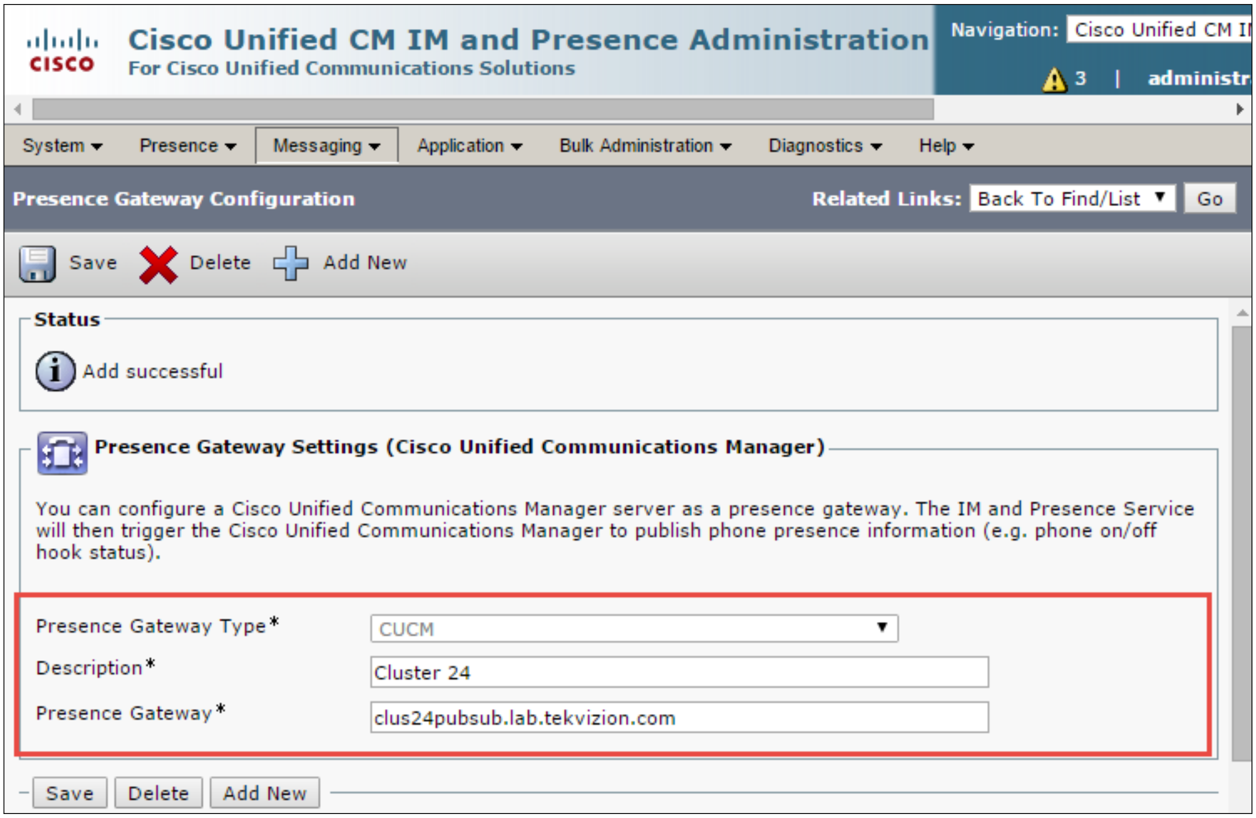
Set Presence Gateway Type *= CUCM
Set Description *= Cluster 24. This is used for this example.
Presence Gateway * =clus24pubsub.lab.tekvizion.com



Acronyms

AVPN	AT&T Virtual Private Network
CODEC	Coder-Decoder (in this document a device used to digitize and undigitize voice signals)

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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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Note: Testing was conducted in tekVizion labs

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