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 ASR Router with SIP Interface

JUN 2015

Note: Testing was conducted in tekVizion labs

Table of Contents

Introduction

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

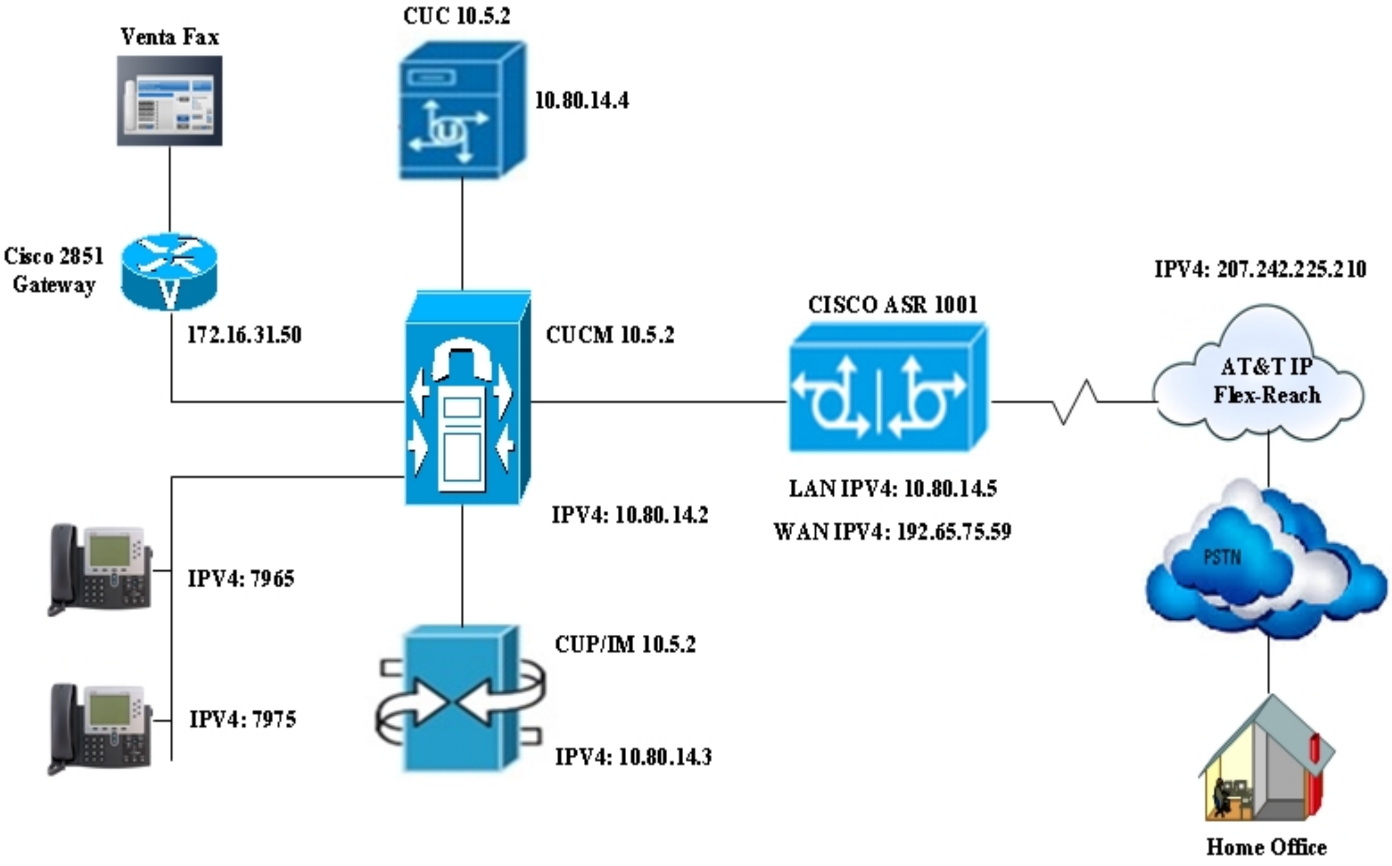
- This application note describes the necessary steps and configurations of Cisco Unified Communications Manager (Cisco UCM) 10.5.2, Cisco Unity Connection 10.5.2, Cisco Unified CM IM and Presence 10.5.2, Cisco Aggregation Services Routers (ASR) Version 15.4(3) S1 with connectivity to AT&T's IP Flex-Reach 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 Aggregation Services Routers (ASR) 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 Aggregation Services Routers (ASR). The configurations described in this document details the important commands for successful interoperability. Care must be taken by the network administrator deploying Cisco ASR to ensure these commands are set per each dial-peer required, to interoperate done AT&T SIP network.

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

Network Topology



Hardware Components

- UCS-C240 VMWare server running ESXi 5.5
- Cisco IP Phones. This solution was tested with Cisco 7965, Cisco 7975 and Cisco 9971 phones
- Cisco Aggregation Services Router - Cisco ASR1001 (revision 1RU) with 1089559K/6147K bytes of memory
- Processor board ID SSI17360FV1 4 Gigabit Ethernet interfaces, 32768K bytes of non-volatile configuration memory

Software Requirements

- Cisco UCM: System version: 10.5.2.10000-5, including Business Edition 6000 and Business Edition 7000.
- ASR: Cisco IOS Software, ASR1000 Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.4(3) S1, RELEASE SOFTWARE (fc3).
- System image file is "bootflash:/asr1001-universalk9.03.13.01.S.154-3.S1-ext.bin"
- Cisco Unity Connection version: System version: 10.5.2.10000-5
- Cisco Unified CM IM and Presence: System version: 10.5.2.10000-9
- Cisco Jabber client version:10.5.0 Build 37889
- VentaFax client version: 7.3.233.582 I

Features

Features – Supported

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

Network Based Features - Supported

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

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

Features - Not Supported

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

Caveats

Fax

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

Auto-Attendant

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

Hold/Resume & Music on Hold (MOH)

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

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 not executed due to limitations on AT&T network

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

Configuration Considerations

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

Emergency 911/E911 Services Limitations and Restrictions

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

ASR Configuration

```
ASR-ATT-IPFR#show version
Cisco IOS XE Software, Version 03.13.01.S - Extended Support Release
Cisco IOS Software, ASR1000 Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.4(3)S1, RELEASE SOFTWARE (fc3)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2014 by Cisco Systems, Inc.
Compiled Fri 31-Oct-14 19:47 by mcpre


Cisco IOS-XE software, Copyright (c) 2005-2014 by cisco Systems, Inc.
All rights reserved.  Certain components of Cisco IOS-XE software are
licensed under the GNU General Public License ("GPL") Version 2.0.  The
software code licensed under GPL Version 2.0 is free software that comes
with ABSOLUTELY NO WARRANTY.  You can redistribute and/or modify such
GPL code under the terms of GPL Version 2.0.  For more details, see the
documentation or "License Notice" file accompanying the IOS-XE software,
or the applicable URL provided on the flyer accompanying the IOS-XE
software.


ROM: IOS-XE ROMMON

ASR-ATT-IPFR uptime is 3 days, 2 hours, 23 minutes
Uptime for this control processor is 3 days, 2 hours, 25 minutes
System returned to ROM by reload
```

```
System image file is "bootflash:/asr1001-universalk9.03.13.01.S.154-3.S1-ext.bin"
Last reload reason: Watchdog



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.

License Level: advipservices
License Type: Permanent
Next reload license Level: advipservices

cisco ASR1001 (1RU) processor (revision 1RU) with 1089559K/6147K bytes of memory.
Processor board ID SSI17360FV1
```

```
4 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



ASR-ATT-IPFR#sh running-config
Building configuration...

Current configuration : 11116 bytes
!
! Last configuration change at 03:49:43 UTC Mon Jun 1 2015 by cisco
!
version 15.4
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname ASR-ATT-IPFR
!
boot-start-marker
boot system flash bootflash:asr1001-universalk9.03.13.01.S.154-3.S1-ext.bin
boot-end-marker
!
```

```
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
logging queue-limit 1000000000
logging buffered 10000000
logging rate-limit 10000
no logging console
no logging monitor
enable secret 4 Pe0NhiWw5IXZpE.k5VhTSCoGPcuVeRyrer9kEPz20Z6
!
no aaa new-model
!
!
subscriber templating
!
multilink bundle-name authenticated
!
!
!
```

```
voice service voip
 no ip address trusted authenticate
 address-hiding[1]
 mode border-element[2]
 allow-connections sip to sip[3]
 redirect ip2ip
 fax protocol pass-through g711ulaw
 no fax-relay sg3-to-g3
 sip
  header-passing
  error-passthru[4]
  asserted-id pai[5]
  no update-callerid
  early-offer forced[6]
  midcall-signaling passthru[7]
  privacy-policy passthru[8]
  g729 annexb-all
```

---

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

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

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

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

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

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

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

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

```
!
voice class codec 1[9]
 codec preference 1 g729r8 bytes 30
 codec preference 2 g711ulaw
!
voice class codec 2
 codec preference 1 g711ulaw
 codec preference 2 g729r8 bytes 30
!
!
voice class sip-profiles 1
 response ANY sip-header Allow-Header modify "UPDATE," ""
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:732320\1@\2>"[10]
 request INVITE sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"[11]
 response ANY sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"
 request INVITE sdp-header Audio-Attribute add "a=ptime:30"[12]

!
!
!
```

---

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

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

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

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

```
voice iec syslog
!
!
voice translation-rule 1[13]
 rule 1 /^\+\(1\)\(7.........\)/ /\2/
 rule 2 /^.*\(435.\)/ /732320\1/
!
!
voice translation-profile AddPlusOne
 translate called 1
!
voice translation-profile NPA
 translate calling 1
!
!
license udi pid ASR1001 sn JAE17430GQ5
license boot level advipservices
spanning-tree extend system-id
!
username cisco privilege 15 password 0 cisco
!
redundancy
 mode none
!
cdp run
```

---

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

```
!
interface GigabitEthernet0/0/0[14]
 description Wan Interface
 ip address 192.65.79.59 255.255.255.224
 negotiation auto
 cdp enable
!
interface GigabitEthernet0/0/1[15]
 description Lan Interface
 ip address 10.80.14.5 255.255.255.0[16]
 negotiation auto
 cdp enable
!
interface GigabitEthernet0/0/2
 no ip address
 shutdown
 negotiation auto
!
interface GigabitEthernet0/0/3
 no ip address
 shutdown
 negotiation auto
!
interface GigabitEthernet0
```

---

[14] WAN interface to AT&T

[15] LAN interface to Cisco UCM

[16] Cisco UBE LAN interface IPv4 Address

```
 vrf forwarding Mgmt-intf
 no ip address
 shutdown
 negotiation auto
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.33
ip route 10.64.0.0 255.255.0.0 10.80.14.1
ip route 172.16.0.0 255.255.0.0 10.80.14.1
!
!
control-plane
!
dial-peer voice 100 voip
 description "Outgoing To AT&T .IP PBX facing side"
 session protocol sipv2
 incoming called-number [27]T
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
```

```
 voice-class sip bind media source-interface GigabitEthernet0/0/1

 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

!

dial-peer voice 101 voip[17]

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

 translation-profile outgoing NPA

 destination-pattern [2-9]T

 session protocol sipv2[18]

 session target ipv4:207.242.225.210

 voice-class codec 1[19]

 voice-class sip asymmetric payload full[20]

 voice-class sip asserted-id pai

 voice-class sip privacy-policy passthru[21]

 voice-class sip early-offer forced

 voice-class sip profiles 1[22]

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

---

[17] Dial peer for AT&T facing network

[18] Session protocol SIPv2 is used for this testing

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

[20] Configures the dynamic SIP asymmetric payload support

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

[22] This command enables the dial peer to use SIP profile 1

[23] Configure the Cisco UBE SIP messaging to use the HSRP virtual address in SIP messaging. Once HSRP is configured under the physical interface and the bind command is issued, calls to the physical IP address will fail. This is because the SIP listening

```
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte[24]
 no fax-relay sg3-to-g3
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none[25]
 no vad
!
dial-peer voice 400 voip
 description " Incoming AT&T to IP-PBX AT&T facing side "
 translation-profile incoming AddPlusOne
 huntstop
 session protocol sipv2
 incoming called-number +1[37][13][24]320435.
 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 GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
no fax-relay sg3-to-g3
 fax rate 14400
```

---

socket is now bound to the virtual IP address but the signaling packets use the physical IP address, and therefore cannot be handled.

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

```
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 401 voip
 description " Incoming AT&T to IP-PBX - IP-PBX facing side "
 huntstop
 destination-pattern [37][13][24].......
 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 profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 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
!
dial-peer voice 500 voip
 description "Outgoing To AT&T"-AT&T facing side - meet me
 destination-pattern 7323204292
```

```
 session protocol sipv2
 session target ipv4:207.242.225.210
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 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 501 voip
 description "Outgoing To AT&T"-AT&T facing side - Teleconferencing
 destination-pattern 7323204607
 session protocol sipv2
 session target ipv4:207.242.225.210
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
```

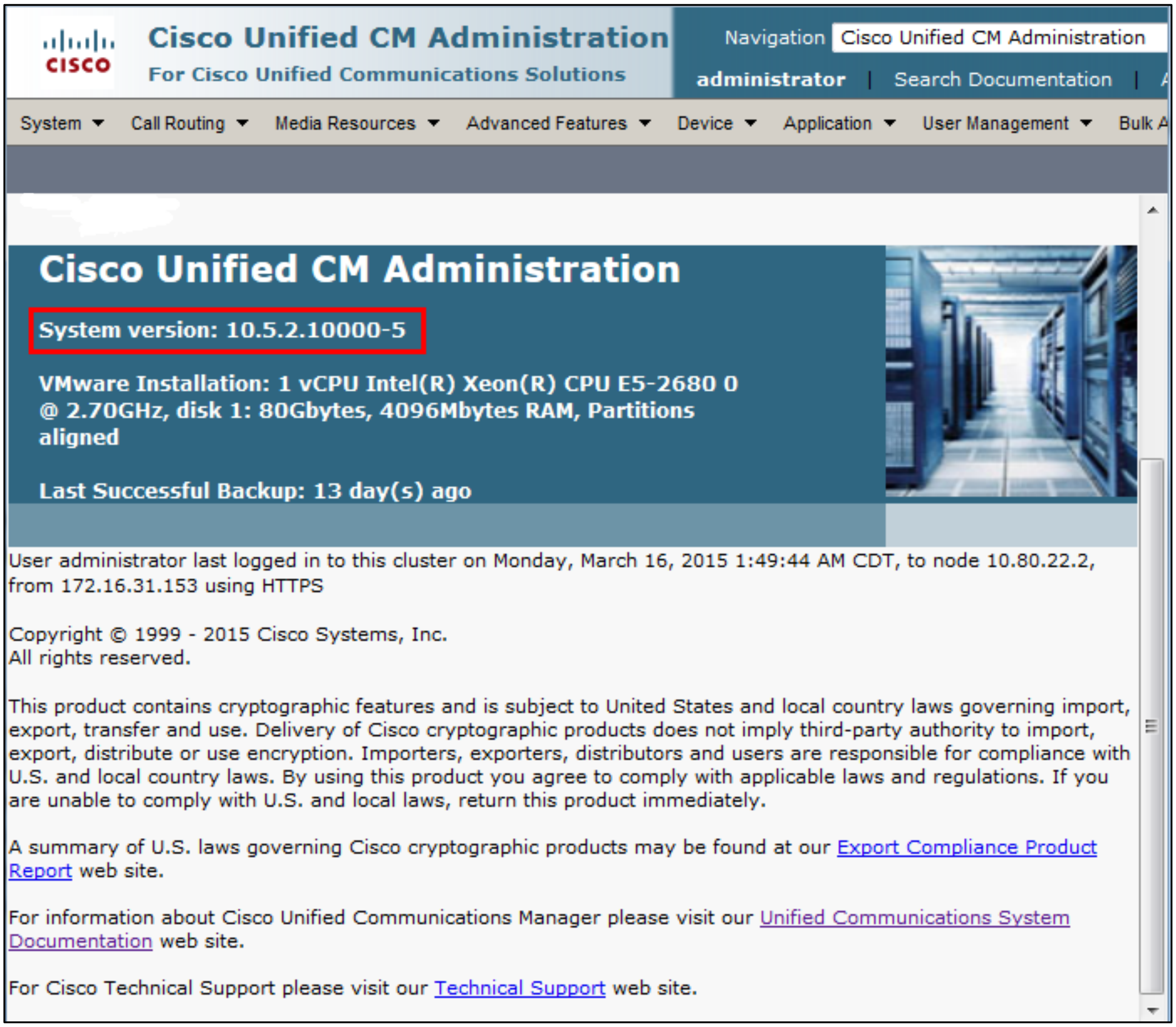
```
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 600 voip
 description "Network based call forwarding Feature"
 translation-profile outgoing NPA
 destination-pattern *..
  session protocol sipv2
 session target ipv4:207.242.225.210
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 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
 retry invite 2
 timers expires 1800000
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 login local
!
End
```

Cisco UCM Configuration

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

Cisco UCM Version



Cisco Unified CM Administration

System version: 10.5.2.10000-5

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

Last Successful Backup: 13 day(s) ago

User administrator last logged in to this cluster on Monday, March 16, 2015 1:49:44 AM CDT, to node 10.80.22.2, from 172.16.31.153 using HTTPS

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.

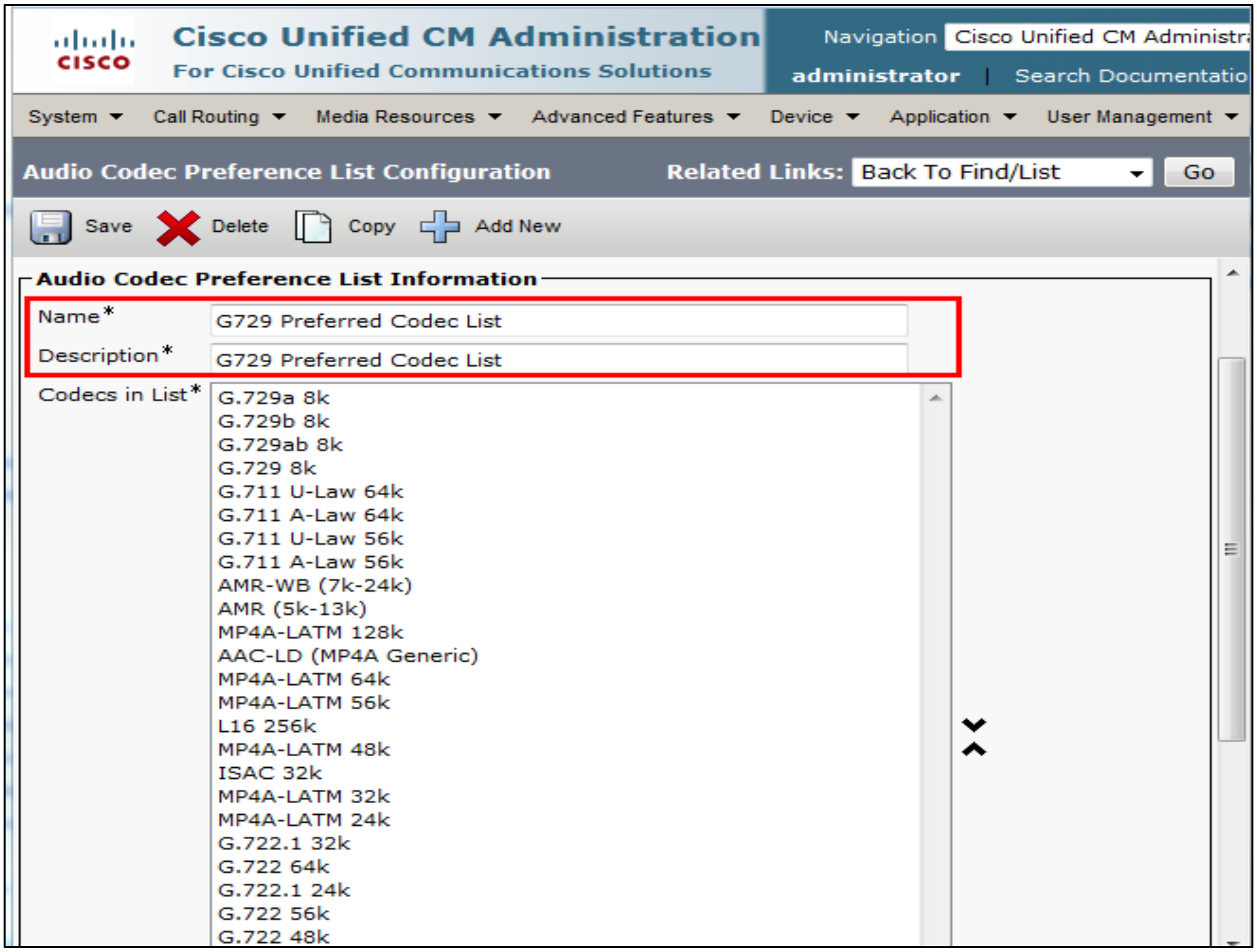
For information about Cisco Unified Communications Manager please visit our Unified Communications System Documentation web site.

For Cisco Technical Support please visit our Technical Support web site.

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)



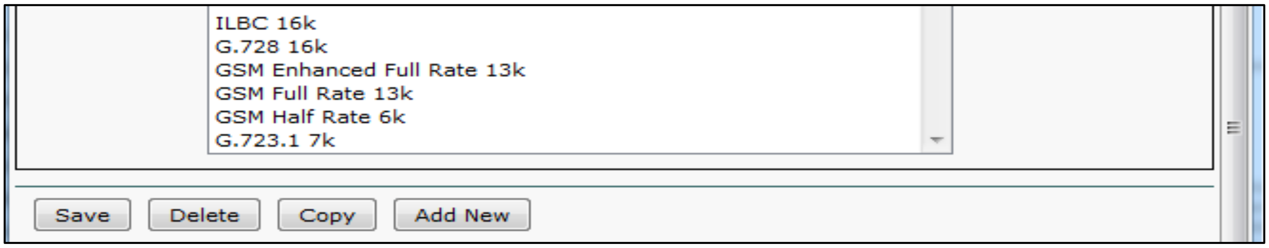
Audio Codec Preference List Information

Name* G729 Preferred Codec List

Description* G729 Preferred Codec List

Codecs in List*

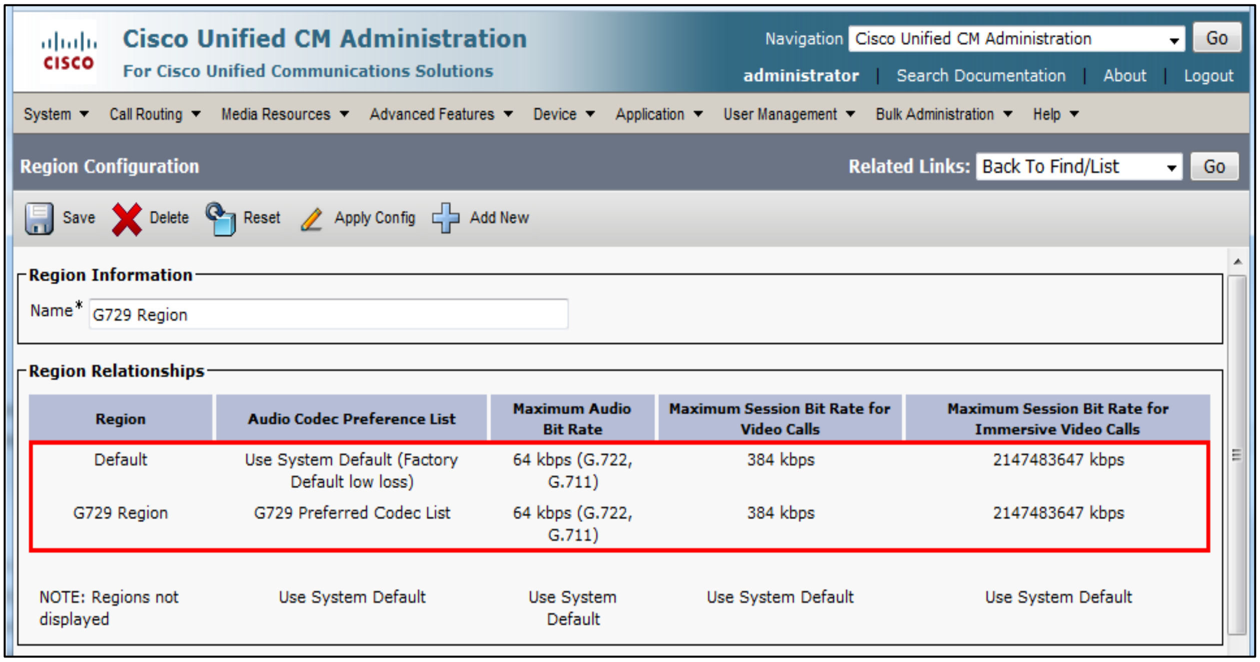
G.729a 8k
G.729b 8k
G.729ab 8k
G.729 8k
G.711 U-Law 64k
G.711 A-Law 64k
G.711 U-Law 56k
G.711 A-Law 56k
AMR-WB (7k-24k)
AMR (5k-13k)
MP4A-LATM 128k
AAC-LD (MP4A Generic)
MP4A-LATM 64k
MP4A-LATM 56k
L16 256k
MP4A-LATM 48k
ISAC 32k
MP4A-LATM 32k
MP4A-LATM 24k
G.722.1 32k
G.722 64k
G.722.1 24k
G.722 56k
G.722 48k



ILBC 16k
G.728 16k
GSM Enhanced Full Rate 13k
GSM Full Rate 13k
GSM Half Rate 6k
G.723.1 7k

Cisco UCM Region Configuration

Navigation Path: System → Region Information → Region



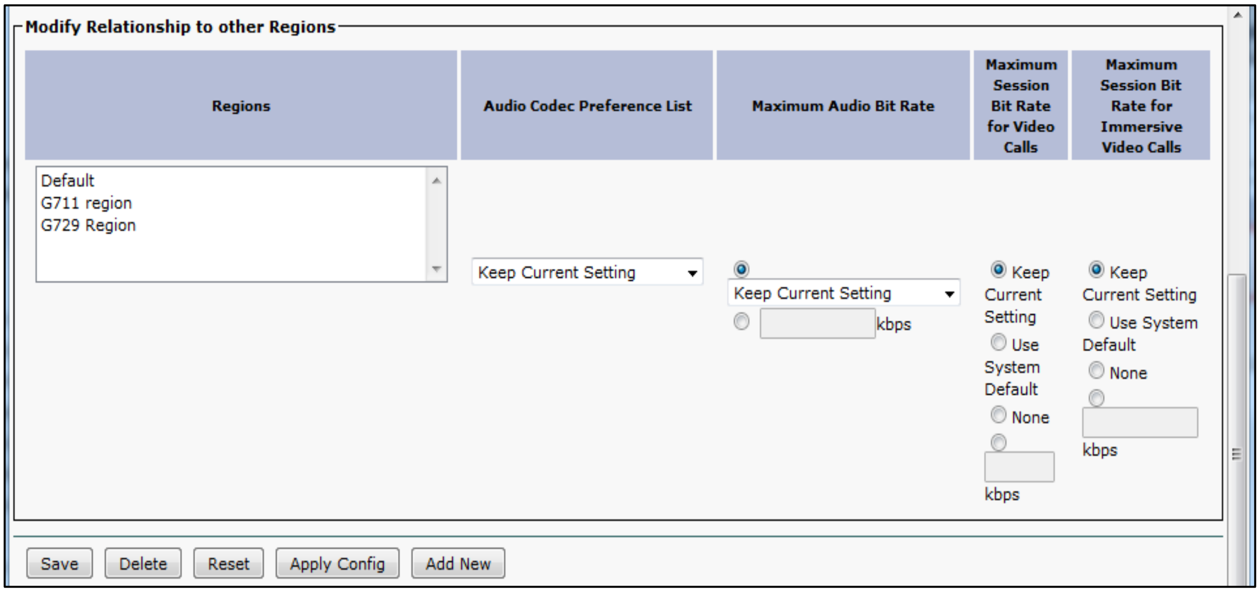
Region Configuration

Region Information

Name* G729 Region

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
G729 Region	G729 Preferred Codec List	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default



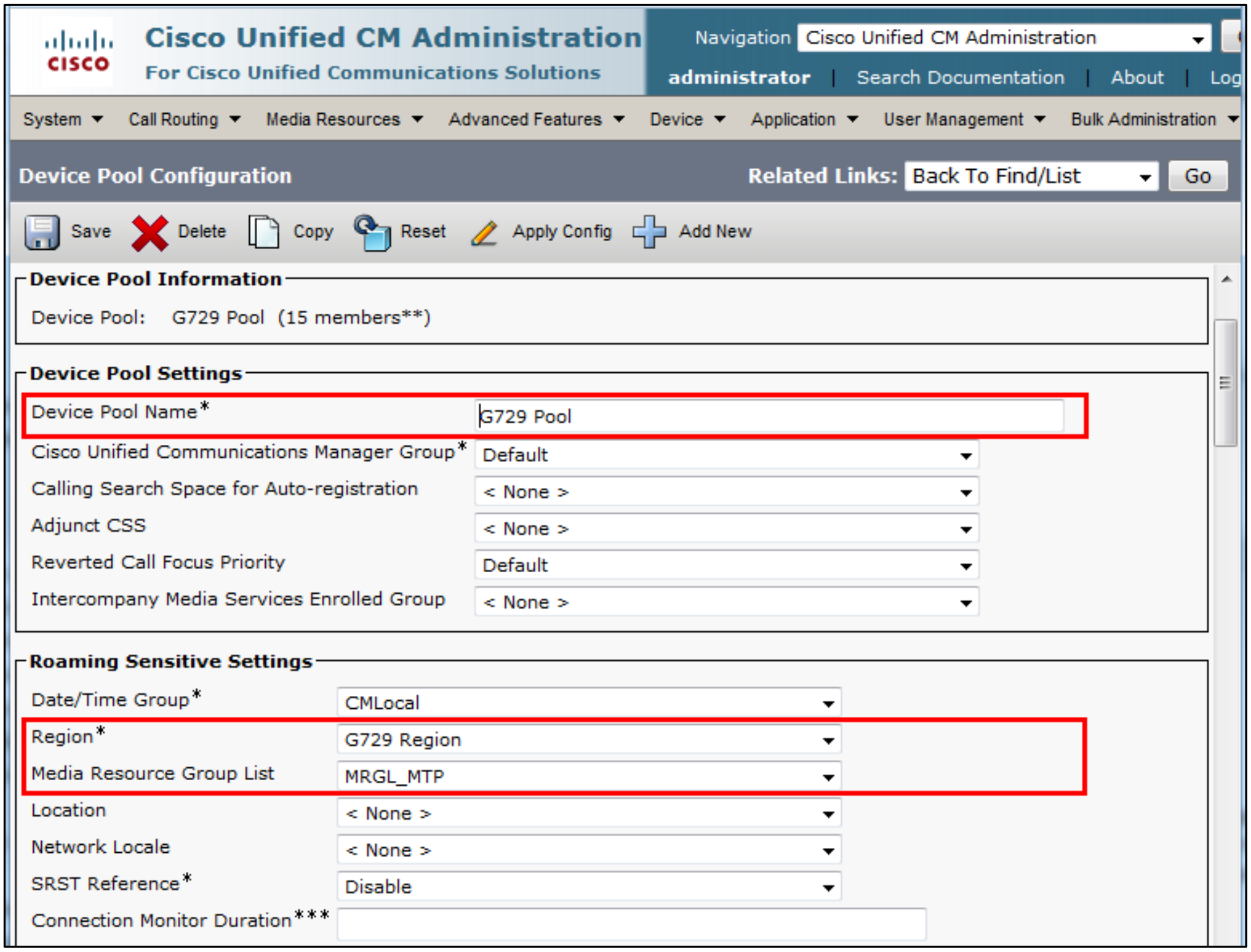
Modify Relationship to other Regions

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default G711 region G729 Region	Keep Current Setting	● Keep Current Setting ○ kbps	● Keep Current Setting ○ Use System Default ○ None ○ kbps	● Keep Current Setting ○ Use System Default ○ None ○ kbps

Device Pool Configuration

Navigation Path: System → Device Pool

“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

Device Pool Information

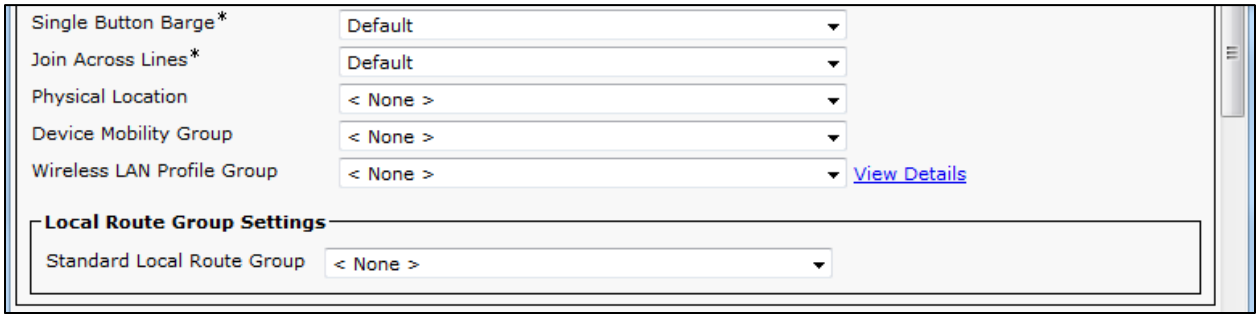
Device Pool: G729 Pool (15 members**)

Device Pool Settings

Device Pool Name* G729 Pool
Cisco Unified Communications Manager Group* Default
Calling Search Space for Auto-registration < None >
Adjunct CSS < None >
Reverted Call Focus Priority Default
Intercompany Media Services Enrolled Group < None >

Roaming Sensitive Settings

Date/Time Group* CMLocal
Region* G729 Region
Media Resource Group List MRGL_MTP
Location < None >
Network Locale < None >
SRST Reference* Disable
Connection Monitor Duration***

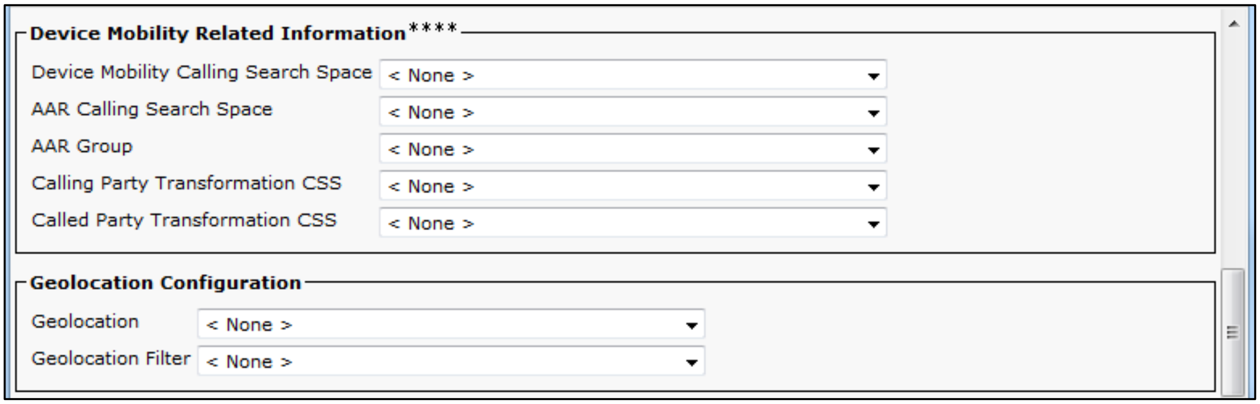


Single Button Barge* Default
Join Across Lines* Default
Physical Location < None >
Device Mobility Group < None >
Wireless LAN Profile Group < None >

Local Route Group Settings

Standard Local Route Group < None >

Device Pool Configuration (continued…)

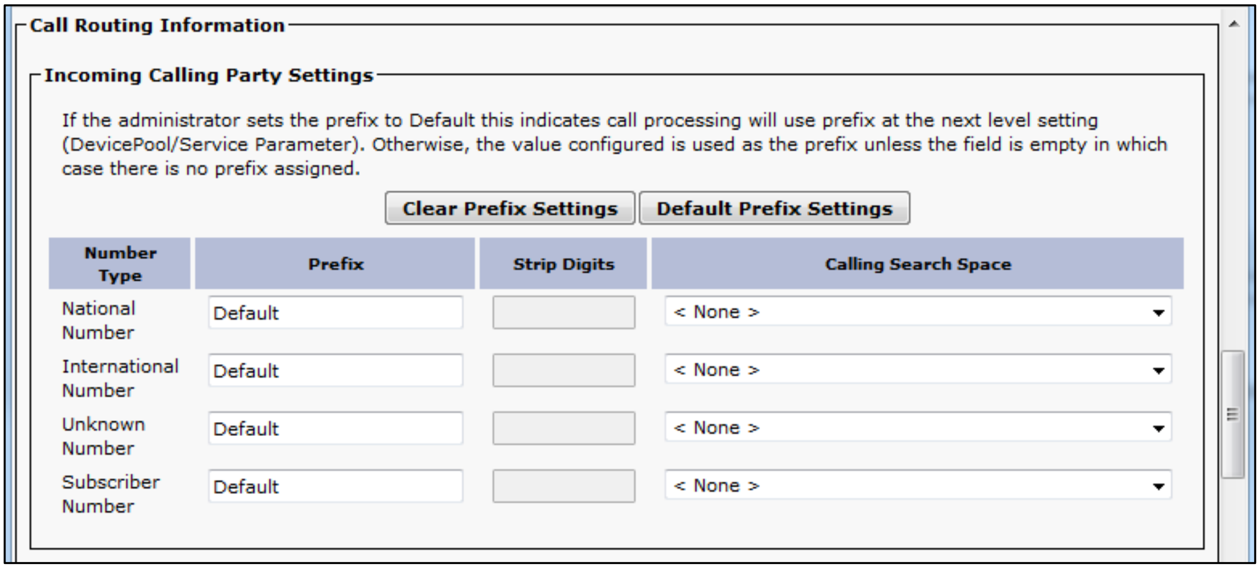


Device Mobility Related Information****

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

Geolocation Configuration

Geolocation	< None >
Geolocation Filter	< None >

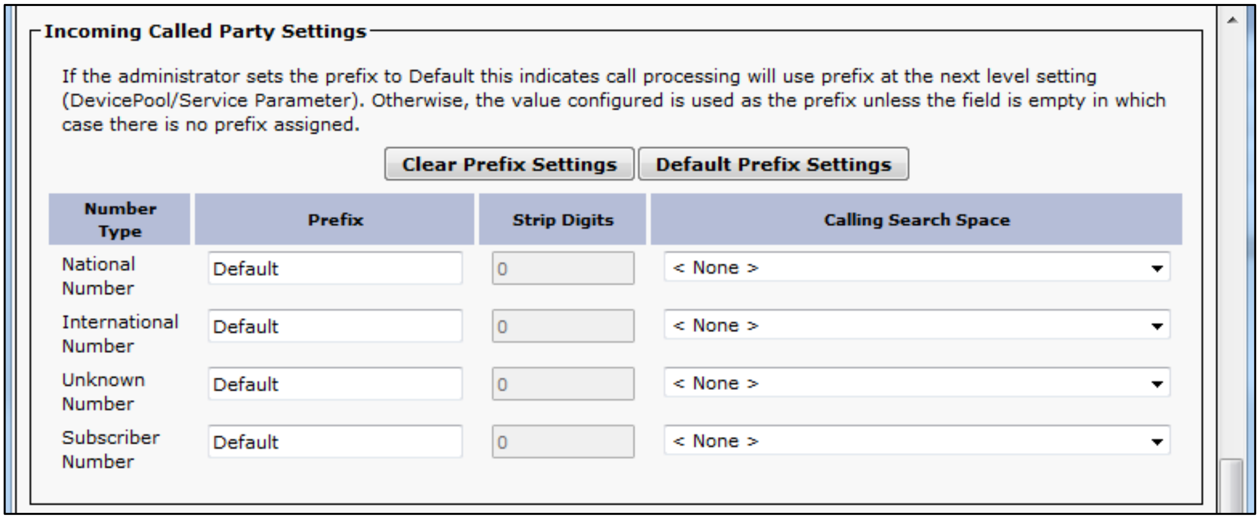


Call Routing Information

Incoming Calling Party Settings

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

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

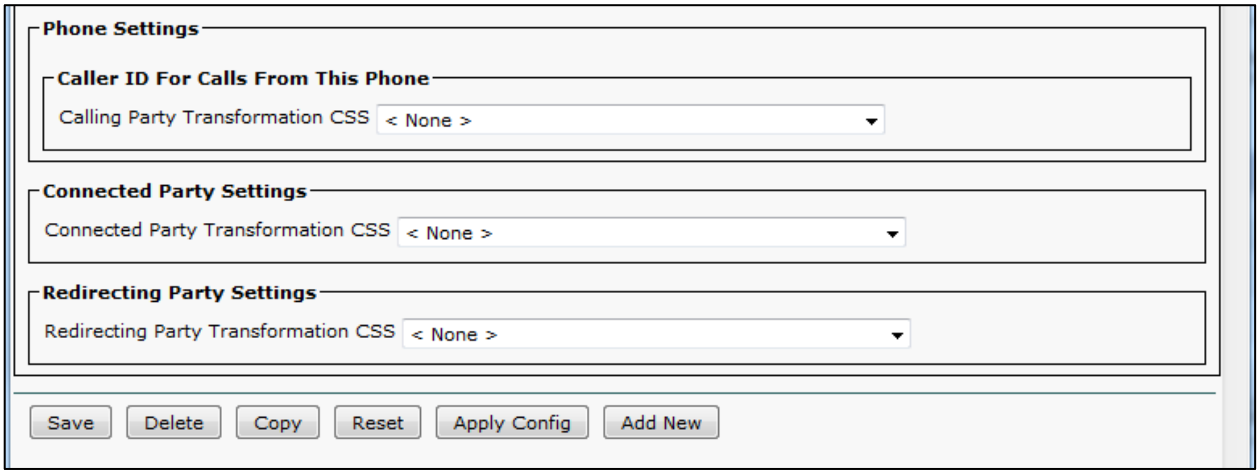


Incoming Called Party Settings

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

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

Device Pool Configuration (continued…)



Phone Settings

Caller ID For Calls From This Phone

Calling Party Transformation CSS < None >

Connected Party Settings

Connected Party Transformation CSS < None >

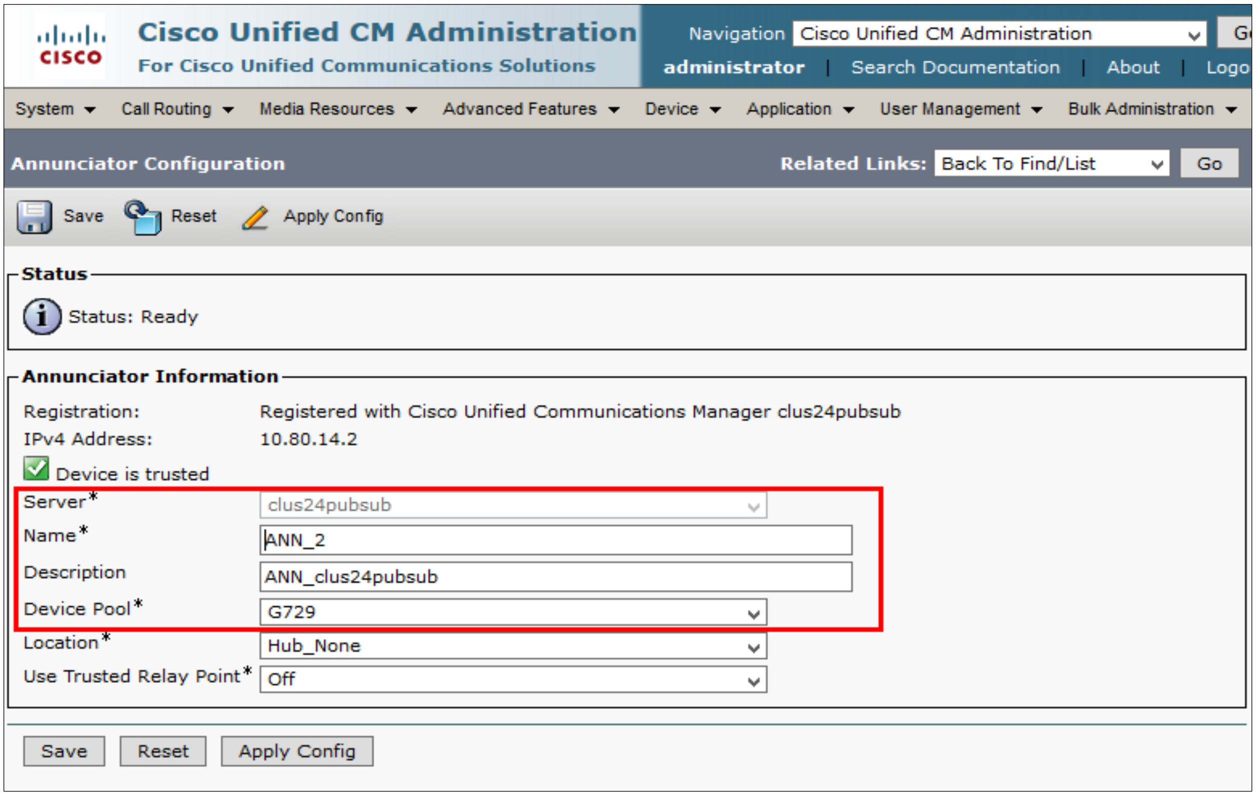
Redirecting Party Settings

Redirecting Party Transformation CSS < None >

Annunciator Configuration

Navigation: Media Resource → Annunciator

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



Status

Status: Ready

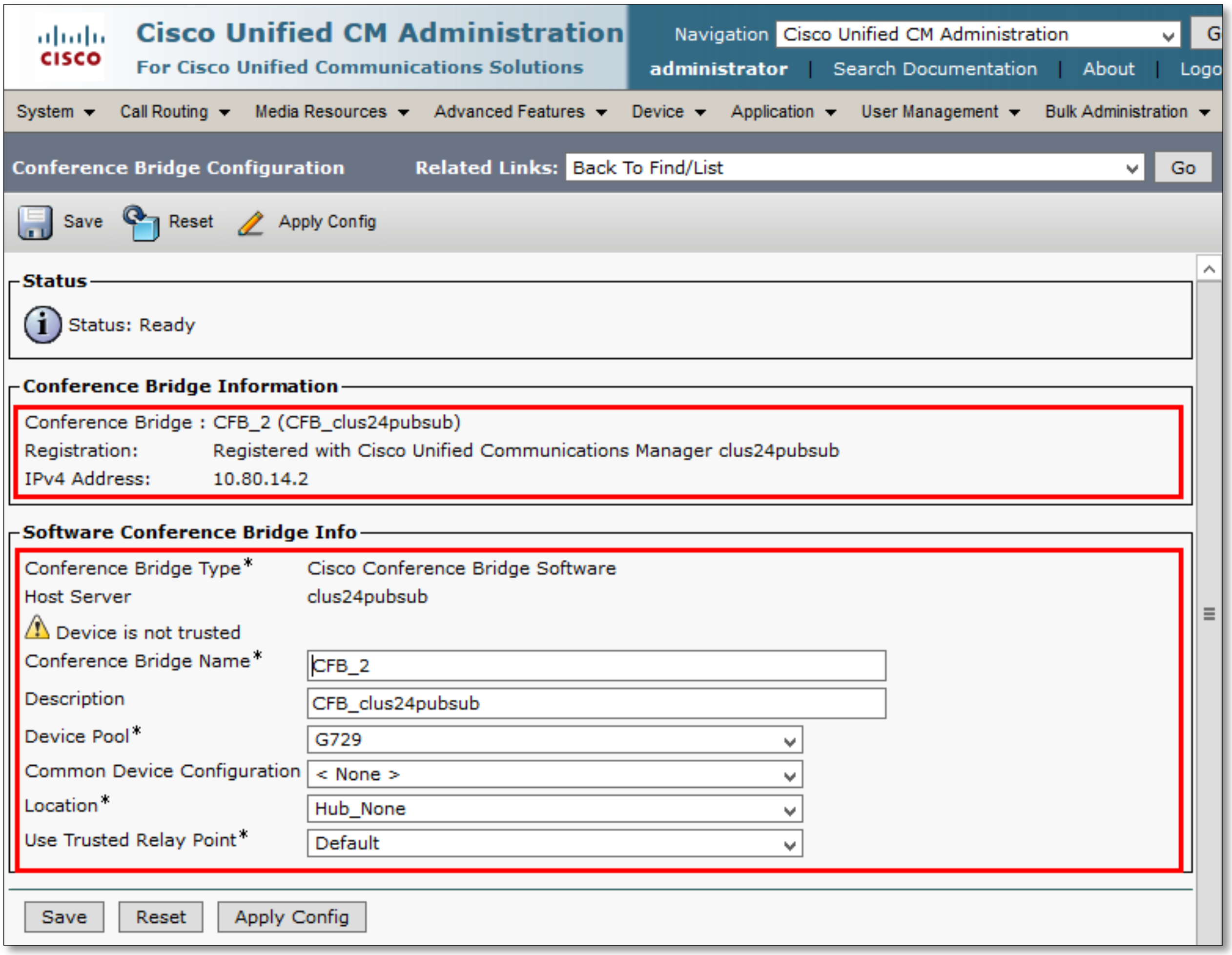
Annunciator Information

Registration: Registered with Cisco Unified Communications Manager clus24pubsub
IPv4 Address: 10.80.14.2
☑ Device is trusted
Server* clus24pubsub
Name* ANN_2
Description ANN_clus24pubsub
Device Pool* G729
Location* Hub_None
Use Trusted Relay Point* Off

Conference Bridge Configuration

Navigation: Media Resources → Conference Bridge

Set Conference Bridge Type* = Cisco Conference Bridge Software.
Set Host Server = clus24pubsub. 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.



Status

Status: Ready

Conference Bridge Information

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

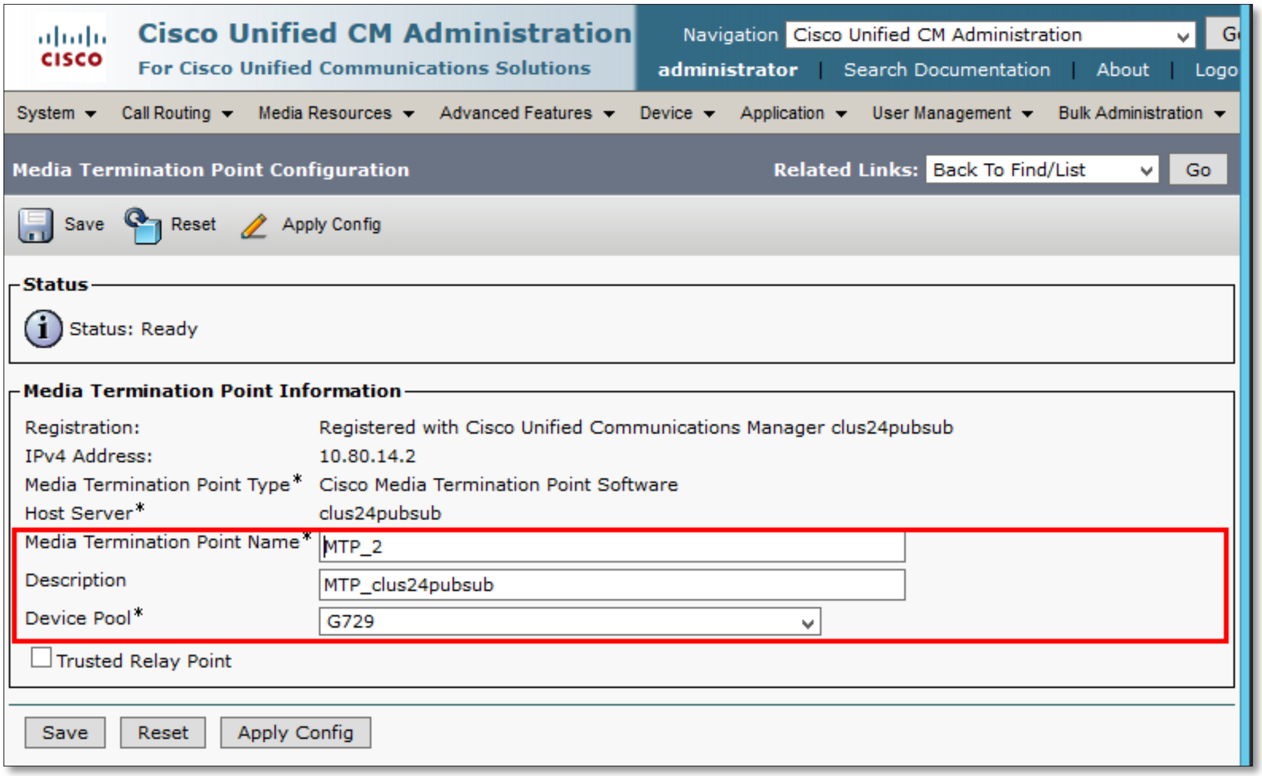
Software Conference Bridge Info

Conference Bridge Type*	Cisco Conference Bridge Software
Host Server	clus24pubsub
⚠ Device is not trusted	
Conference Bridge Name*	CFB_2
Description	CFB_clus24pubsub
Device Pool*	G729
Common Device Configuration	< None >
Location*	Hub_None
Use Trusted Relay Point*	Default

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



Media Termination Point Configuration

Status

Status: Ready

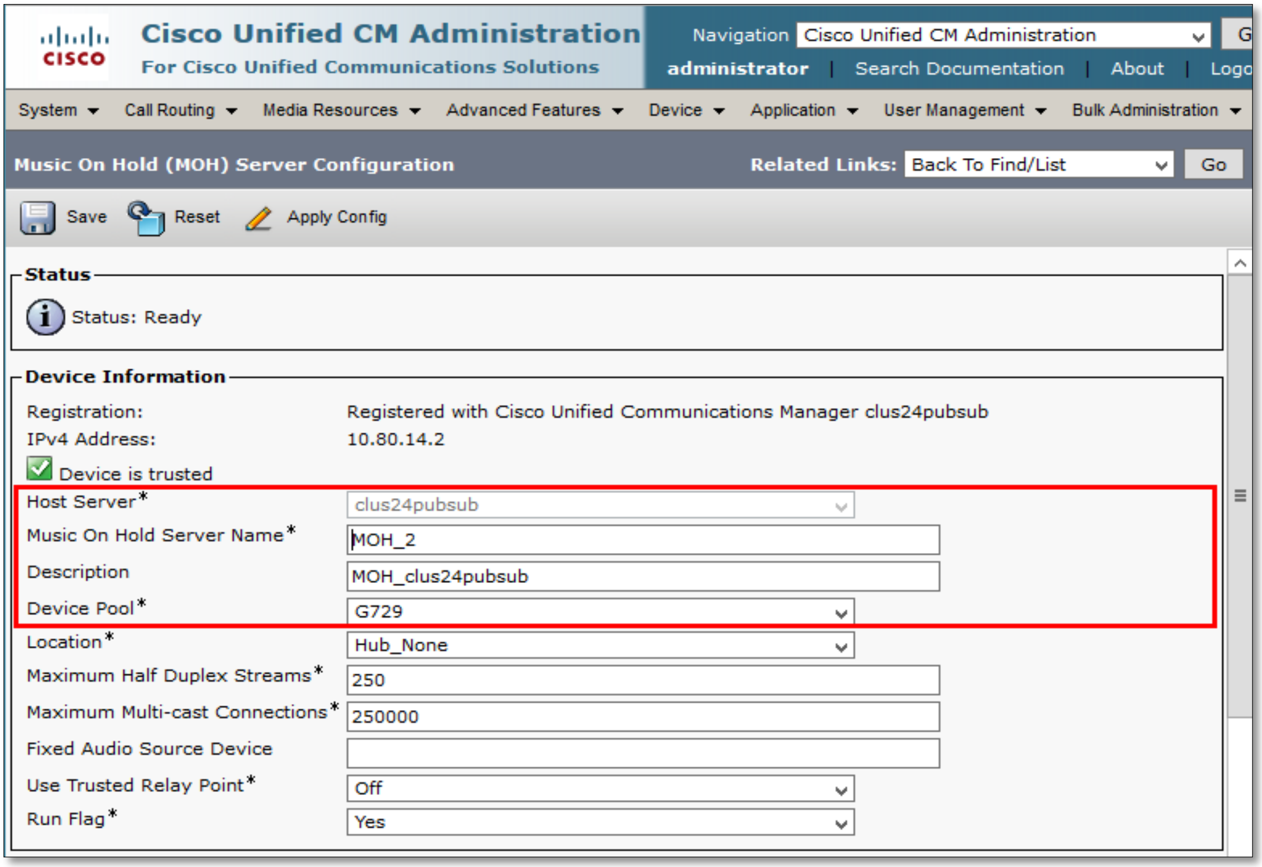
Media Termination Point Information

Registration: Registered with Cisco Unified Communications Manager clus24pubsub
IPv4 Address: 10.80.14.2
Media Termination Point Type* Cisco Media Termination Point Software
Host Server* clus24pubsub
Media Termination Point Name* MTP_2
Description MTP_clus24pubsub
Device Pool* G729
☐ Trusted Relay Point

Music on Hold Server Configuration

Navigation: Media Resources → Music on Hold Server

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



Music On Hold (MOH) Server Configuration

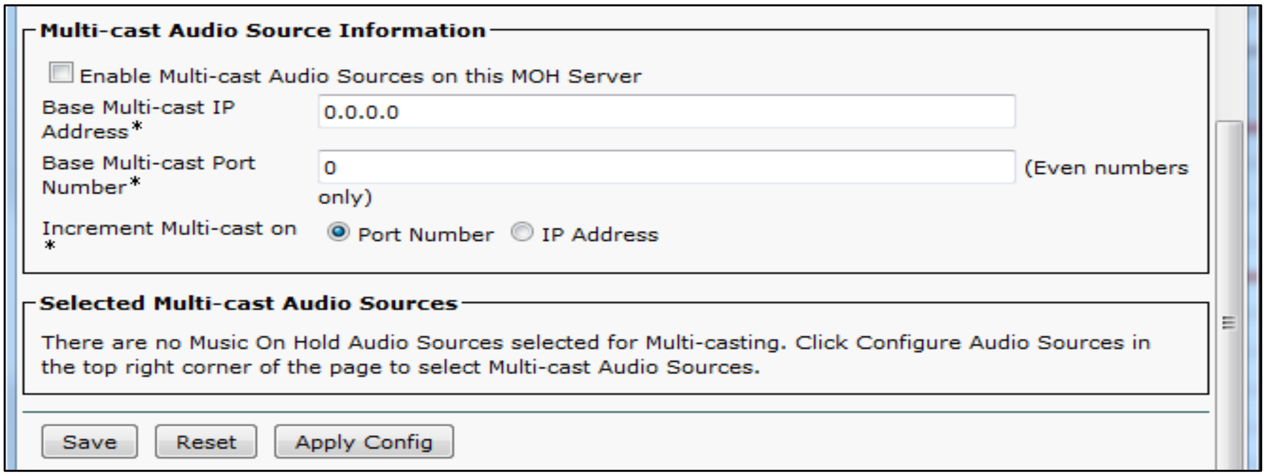
Status

Status: Ready

Device Information

Registration: Registered with Cisco Unified Communications Manager clus24pubsub
IPv4 Address: 10.80.14.2
☑ Device is trusted

Host Server*	clus24pubsub
Music On Hold Server Name*	MOH_2
Description	MOH_clus24pubsub
Device Pool*	G729
Location*	Hub_None
Maximum Half Duplex Streams*	250
Maximum Multi-cast Connections*	250000
Fixed Audio Source Device	
Use Trusted Relay Point*	Off
Run Flag*	Yes



Multi-cast Audio Source Information

☐ Enable Multi-cast Audio Sources on this MOH Server

Base Multi-cast IP Address*	0.0.0.0
Base Multi-cast Port Number*	0 (Even numbers only)
Increment Multi-cast on*	● Port Number ○ IP Address

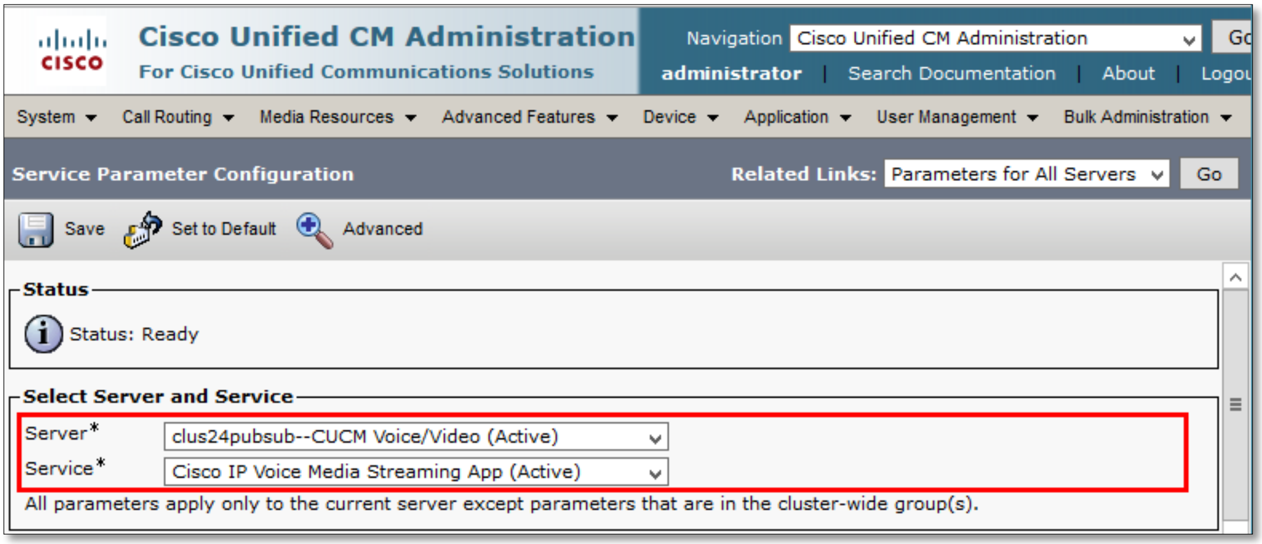
Selected Multi-cast Audio Sources

There are no Music On Hold Audio Sources selected for Multi-casting. Click Configure Audio Sources in the top right corner of the page to select Multi-cast Audio Sources.

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* = clus24pubsub--CUCM Voice/Video (Active). This is used in this example.
Select Service* = Cisco IP Voice Media Streaming App (Active



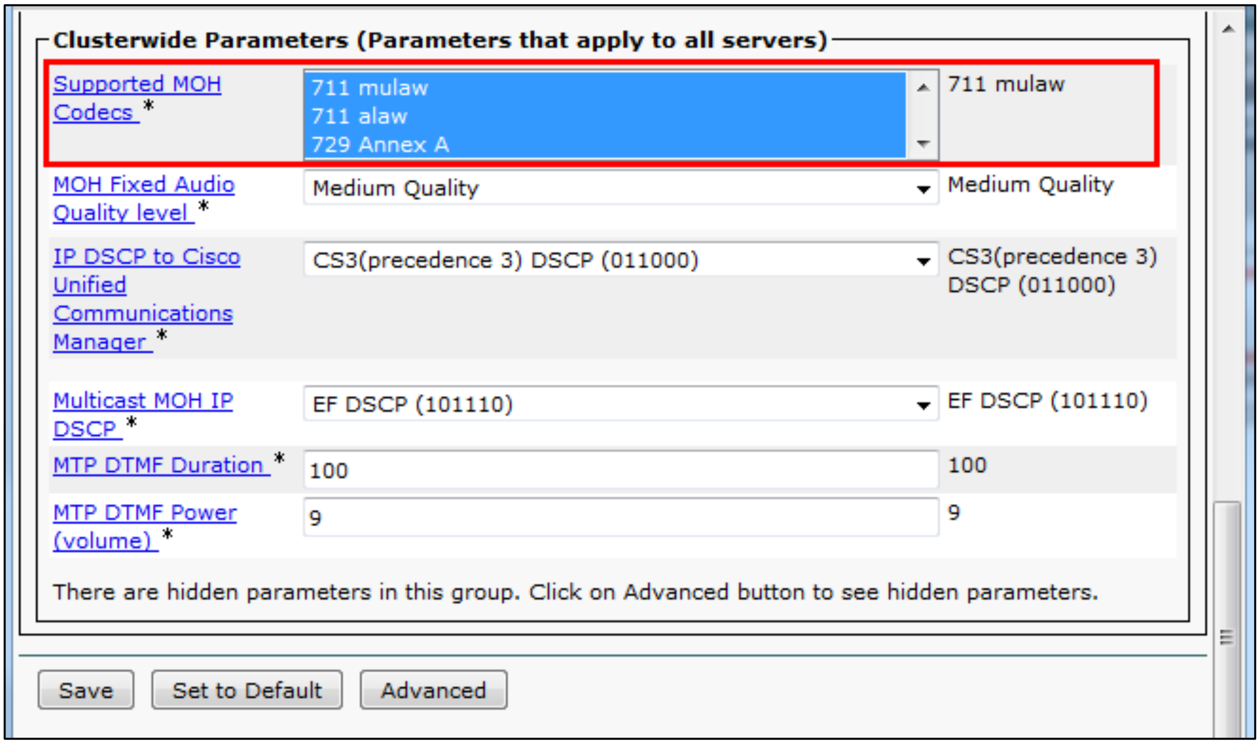
Status

Status: Ready

Select Server and Service

Server* clus24pubsub--CUCM Voice/Video (Active)
Service* Cisco IP Voice Media Streaming App (Active)

All parameters apply only to the current server except parameters that are in the cluster-wide group(s).



Clusterwide Parameters (Parameters that apply to all servers)

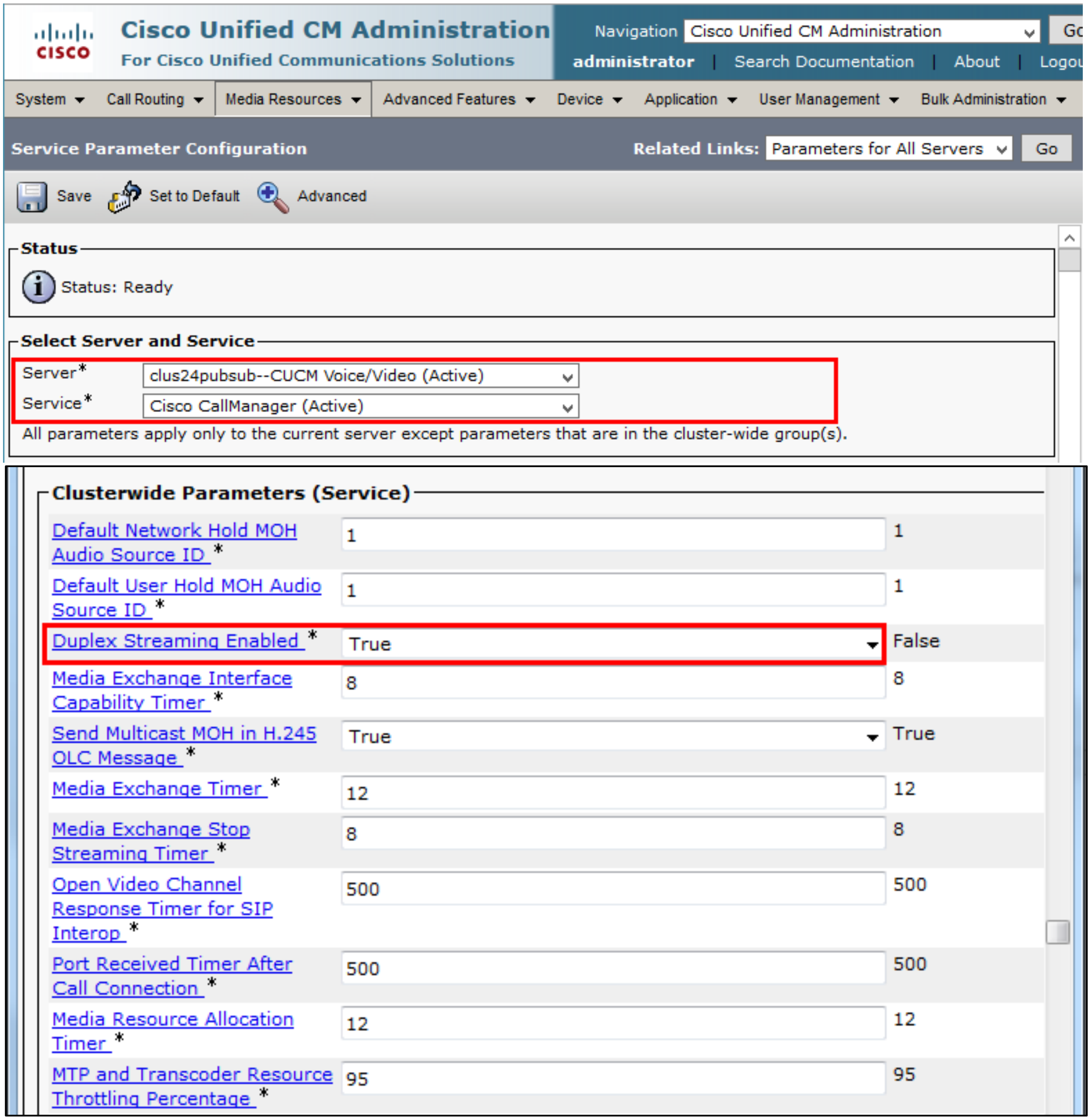
Supported MOH Codecs *	711 mulaw 711 alaw 729 Annex A	711 mulaw
MOH Fixed Audio Quality level *	Medium Quality	Medium Quality
IP DSCP to Cisco Unified Communications Manager *	CS3(precedence 3) DSCP (011000)	CS3(precedence 3) DSCP (011000)
Multicast MOH IP DSCP *	EF DSCP (101110)	EF DSCP (101110)
MTP DTMF Duration *	100	100
MTP DTMF Power (volume) *	9	9

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Music on Hold Service (Duplex Streaming) Parameter Settings

Navigation: System → Service Parameter

Select Server* = clus24pubsub--CUCM Voice/Video (Active). This is used in this example.
Select Service* = Cisco CallManager (Active).
Select Duplex Streaming Enabled * = True



Status

Status: Ready

Select Server and Service

Server* clus24pubsub--CUCM Voice/Video (Active)
Service* Cisco CallManager (Active)

All parameters apply only to the current server except parameters that are in the cluster-wide group(s).

Clusterwide Parameters (Service)

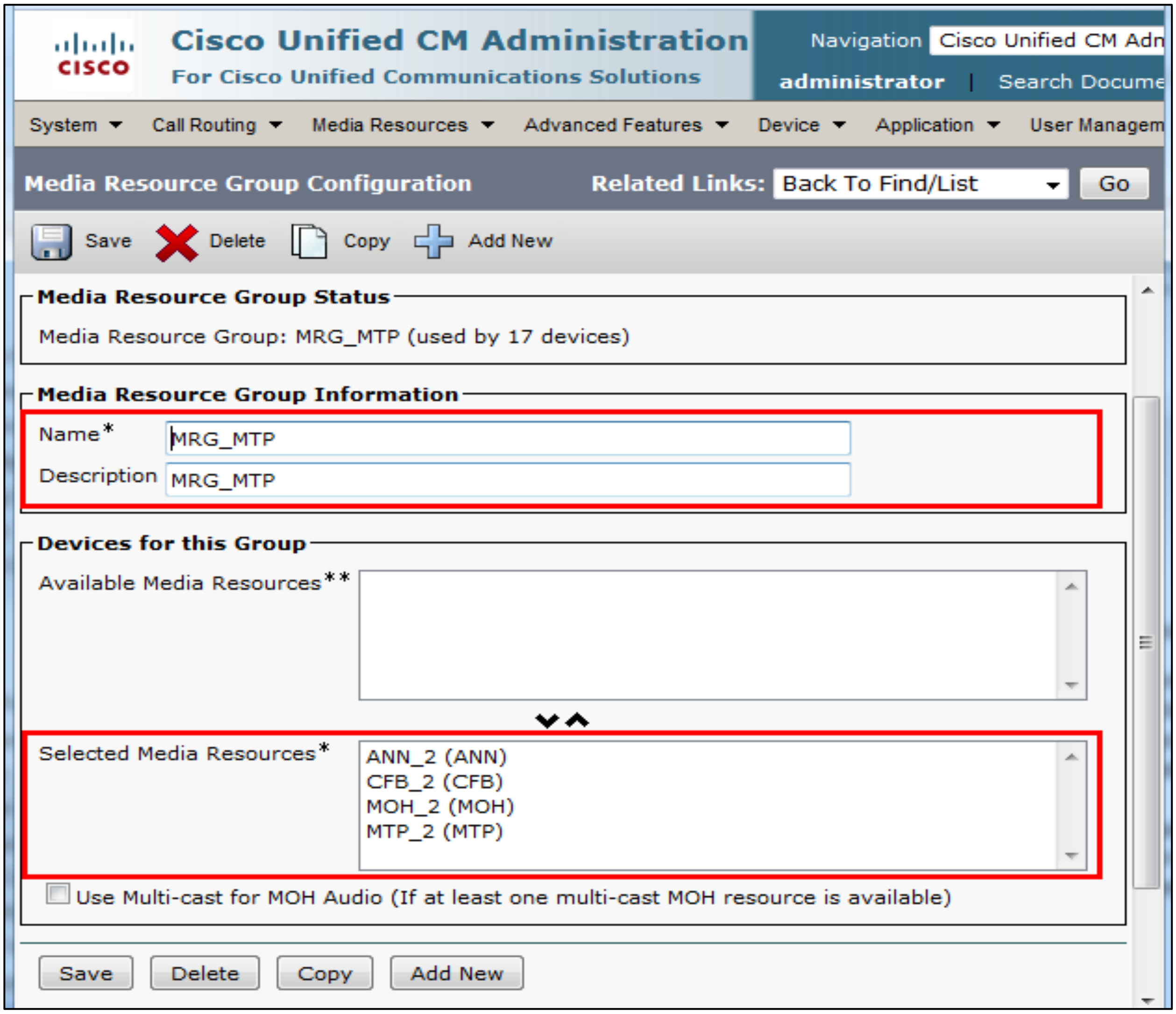
Default Network Hold MOH Audio Source ID *	1	1
Default User Hold MOH Audio Source ID *	1	1
Duplex Streaming Enabled *	True	False
Media Exchange Interface Capability Timer *	8	8
Send Multicast MOH in H.245 OLC Message *	True	True
Media Exchange Timer *	12	12
Media Exchange Stop Streaming Timer *	8	8
Open Video Channel Response Timer for SIP Interop *	500	500
Port Received Timer After Call Connection *	500	500
Media Resource Allocation Timer *	12	12
MTP and Transcoder Resource Throttling Percentage *	95	95

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 Configuration

Media Resource Group Status

Media Resource Group: MRG_MTP (used by 17 devices)

Media Resource Group Information

Name* MRG_MTP
Description MRG_MTP

Devices for this Group

Available Media Resources**

Selected Media Resources*
ANN_2 (ANN)
CFB_2 (CFB)
MOH_2 (MOH)
MTP_2 (MTP)

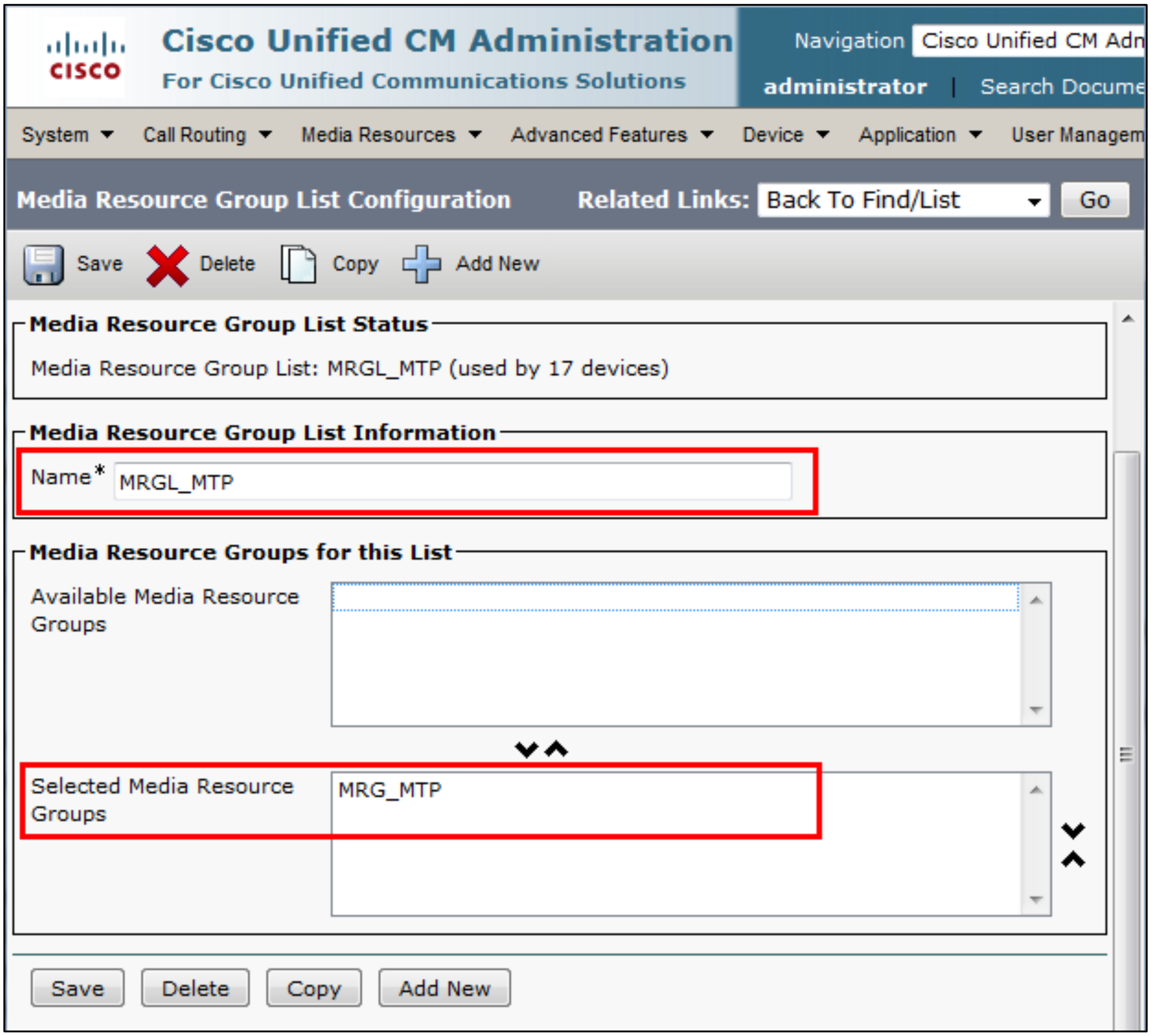
☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

Media Resource Group List Configuration

Navigation Path: Media Resources → Media Resource Group List

Set Name = MRGL_MTP.

Set selected Media Resource Groups = MRG_MTP.



Media Resource Group List Configuration

Media Resource Group List Status

Media Resource Group List: MRGL_MTP (used by 17 devices)

Media Resource Group List Information

Name* MRGL_MTP

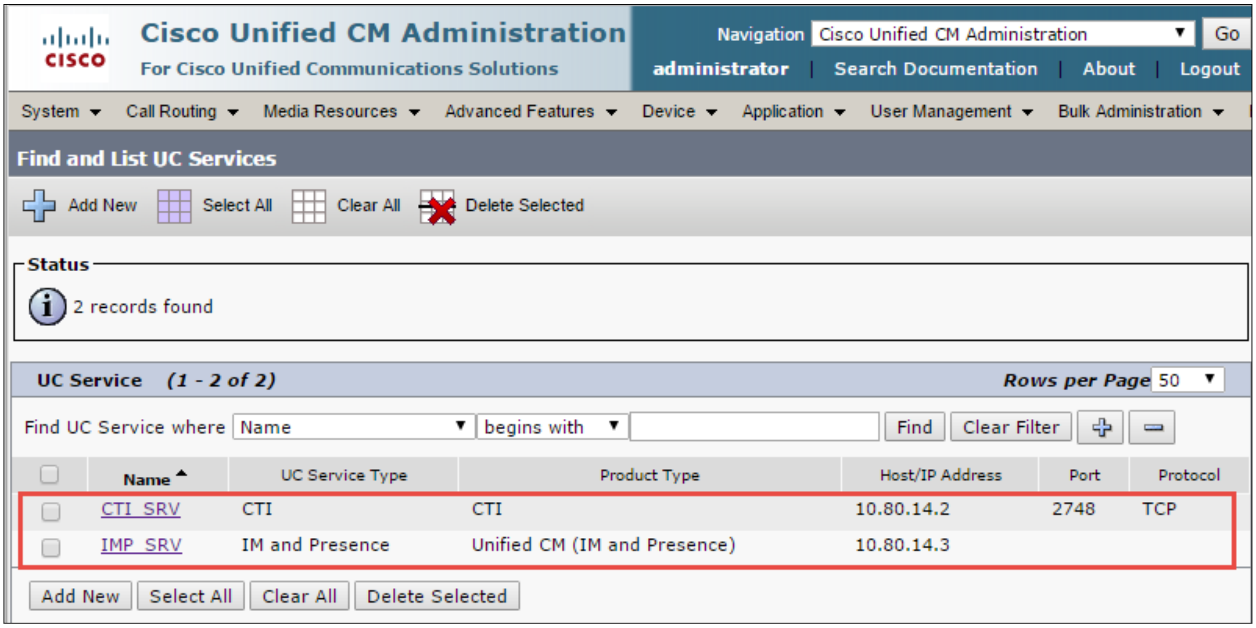
Media Resource Groups for this List

Available Media Resource Groups

Selected Media Resource Groups MRG_MTP

UC Service Configuration

Navigation: User Management → User Settings → UC Service

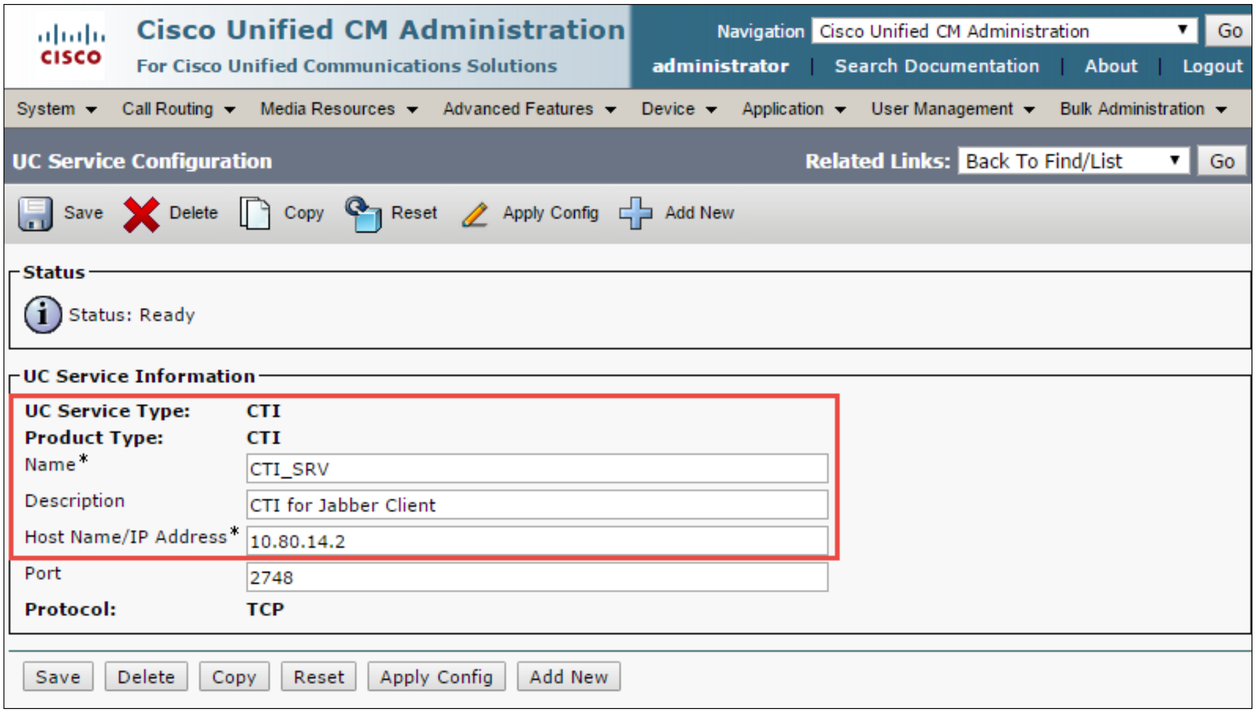


☐	Name	UC Service Type	Product Type	Host/IP Address	Port	Protocol
☐	CTI_SRV	CTI	CTI	10.80.14.2	2748	TCP
☐	IMP_SRV	IM and Presence	Unified CM (IM and Presence)	10.80.14.3		

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* = 10.80.14.2 (Cisco UCM Address)



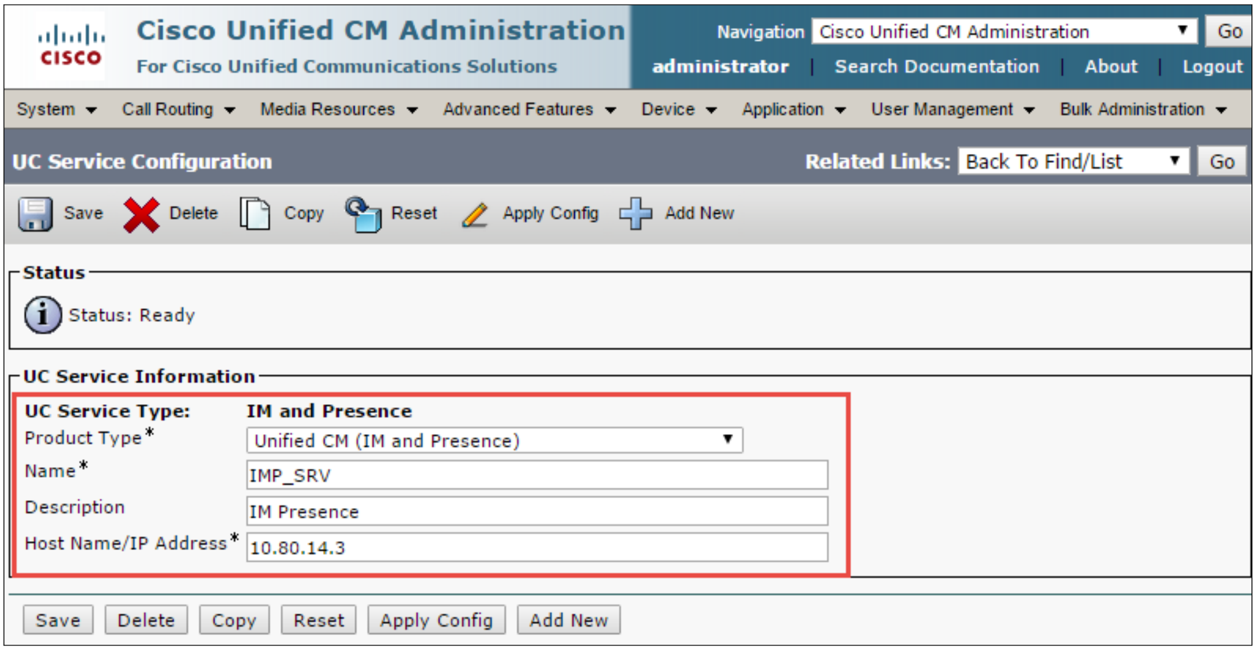
Status: Ready

UC Service Information

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

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)



Status: Ready

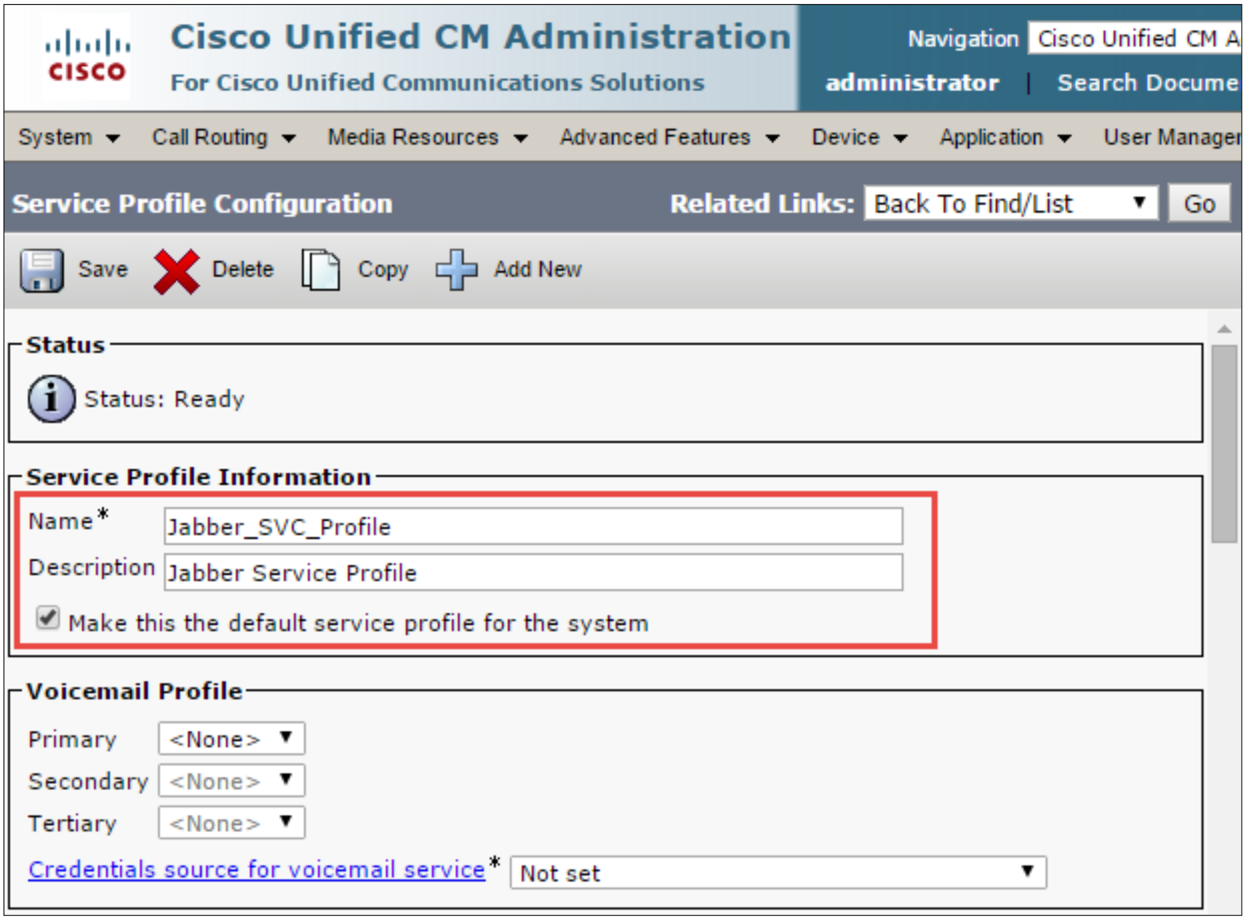
UC Service Type: **IM and Presence**
Product Type* Unified CM (IM and Presence)
Name* IMP_SRV
Description IM Presence
Host Name/IP Address* 10.80.14.3

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.



Status

Status: Ready

Service Profile Information

Name* Jabber_SVC_Profile

Description Jabber Service Profile

☑ Make this the default service profile for the system

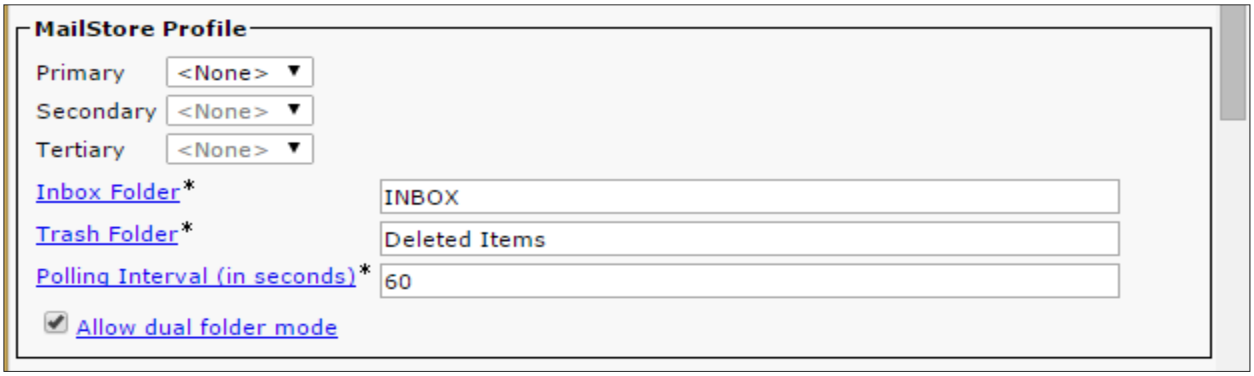
Voicemail Profile

Primary <None>

Secondary <None>

Tertiary <None>

Credentials source for voicemail service* Not set



MailStore Profile

Primary <None>

Secondary <None>

Tertiary <None>

Inbox Folder* INBOX

Trash Folder* Deleted Items

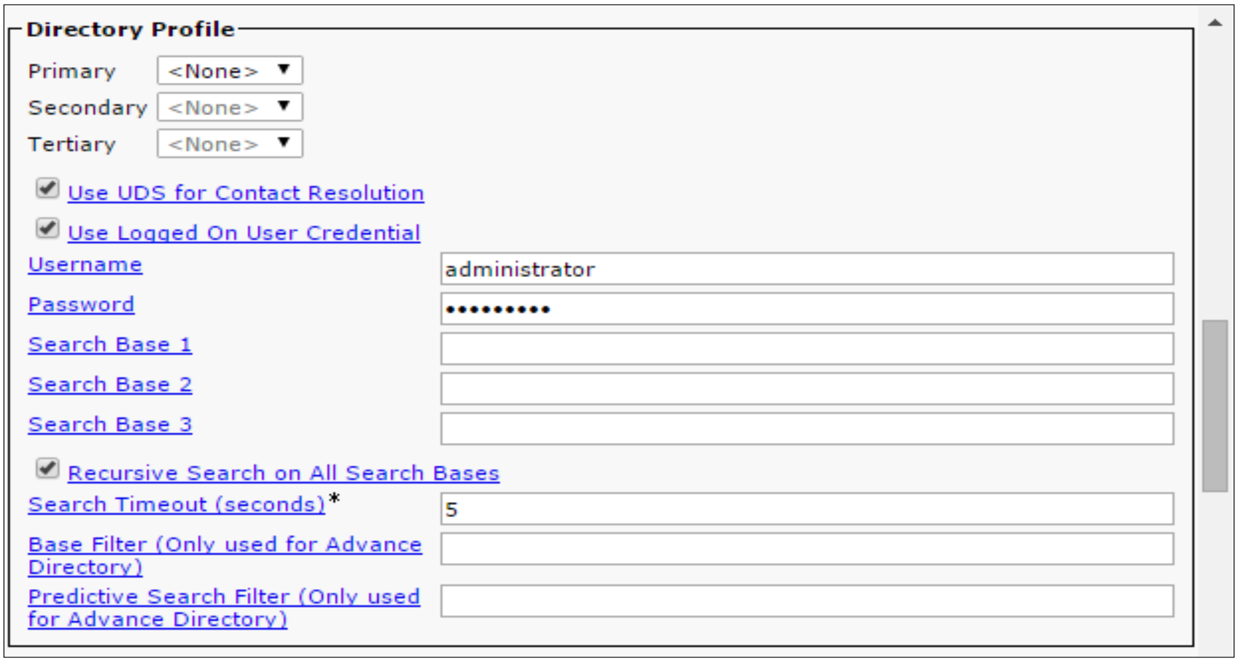
Polling Interval (in seconds)* 60

☑ Allow dual folder mode

Service Profile Configuration (Contd…)

Conferencing Profile

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



Directory Profile

Primary: <None>
Secondary: <None>
Tertiary: <None>

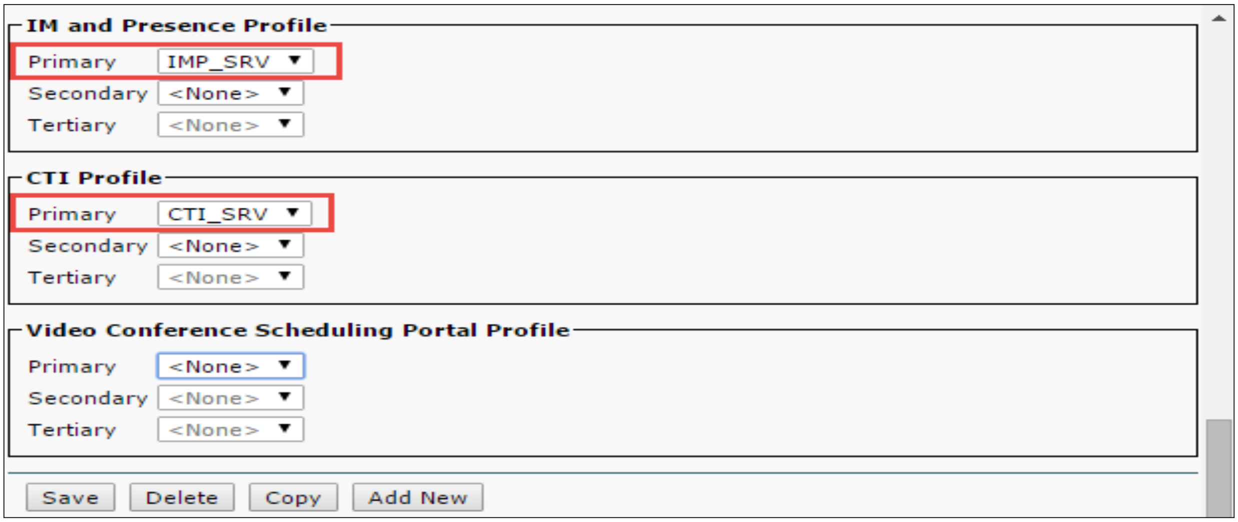
☑ Use UDS for Contact Resolution

☑ Use Logged On User Credential

Username: administrator
Password: •••••••••
Search Base 1:
Search Base 2:
Search Base 3:

☑ Recursive Search on All Search Bases

Search Timeout (seconds)*: 5
Base Filter (Only used for Advance Directory):
Predictive Search Filter (Only used for Advance Directory):



IM and Presence Profile

Primary: IMP_SRV
Secondary: <None>
Tertiary: <None>

CTI Profile

Primary: CTI_SRV
Secondary: <None>
Tertiary: <None>

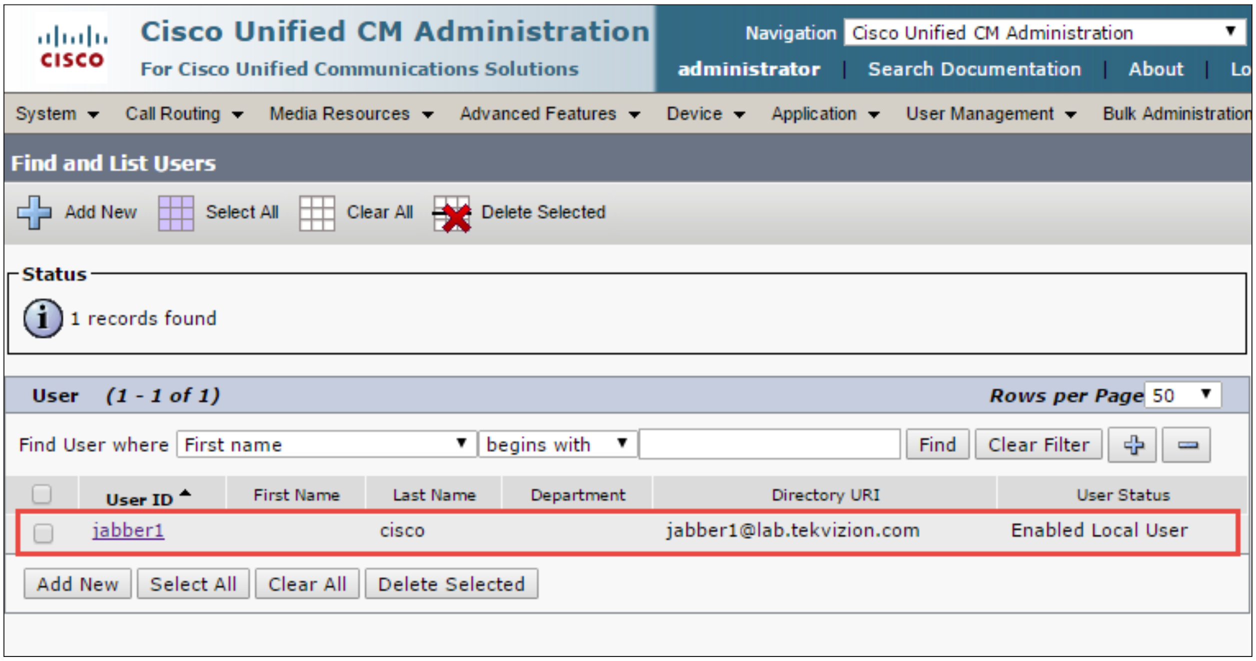
Video Conference Scheduling Portal Profile

Primary: <None>
Secondary: <None>
Tertiary: <None>

End User Configuration

Navigation: User Management → End User

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



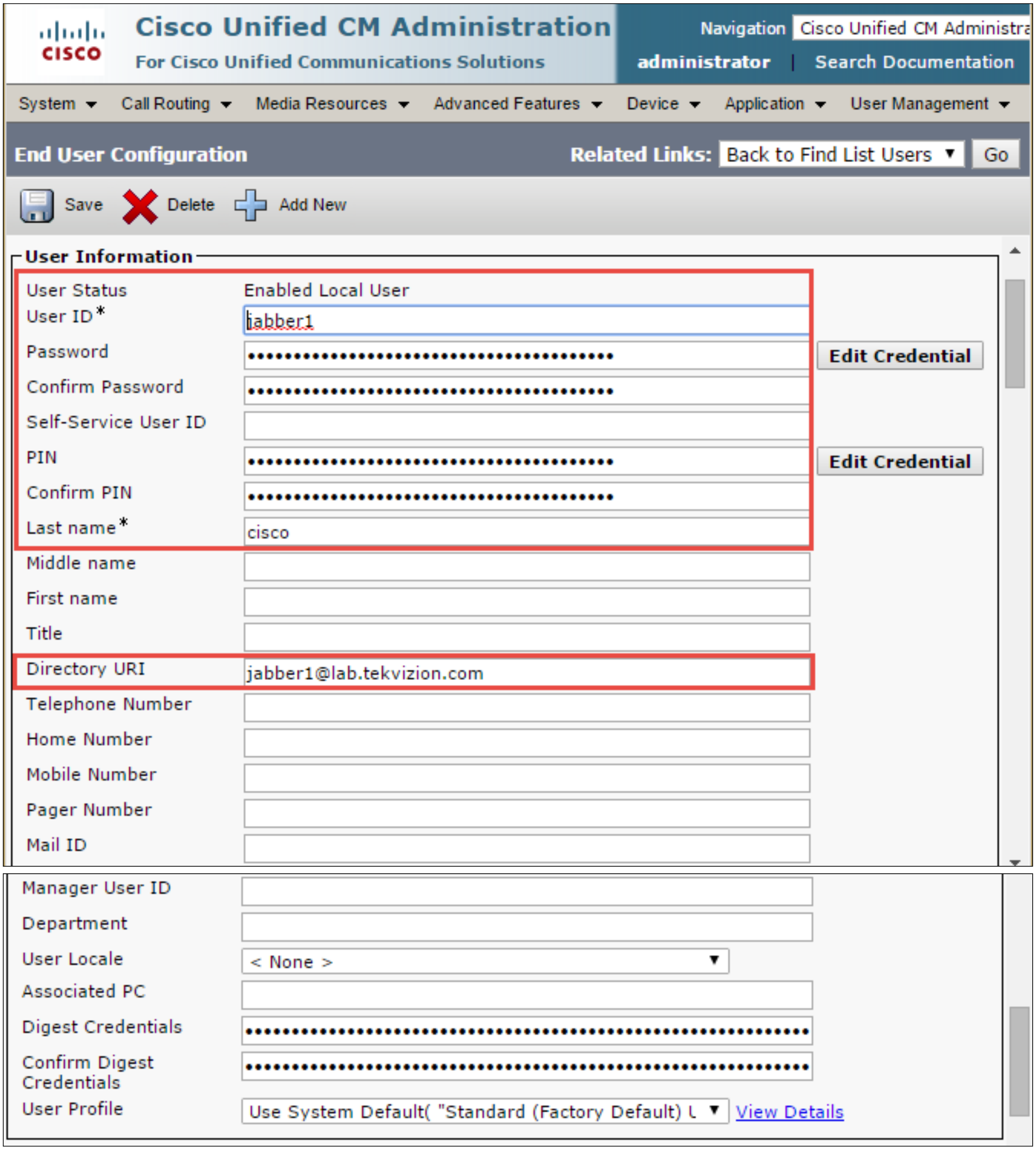
1 records found

User (1 - 1 of 1)

Find User where First name begins with

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

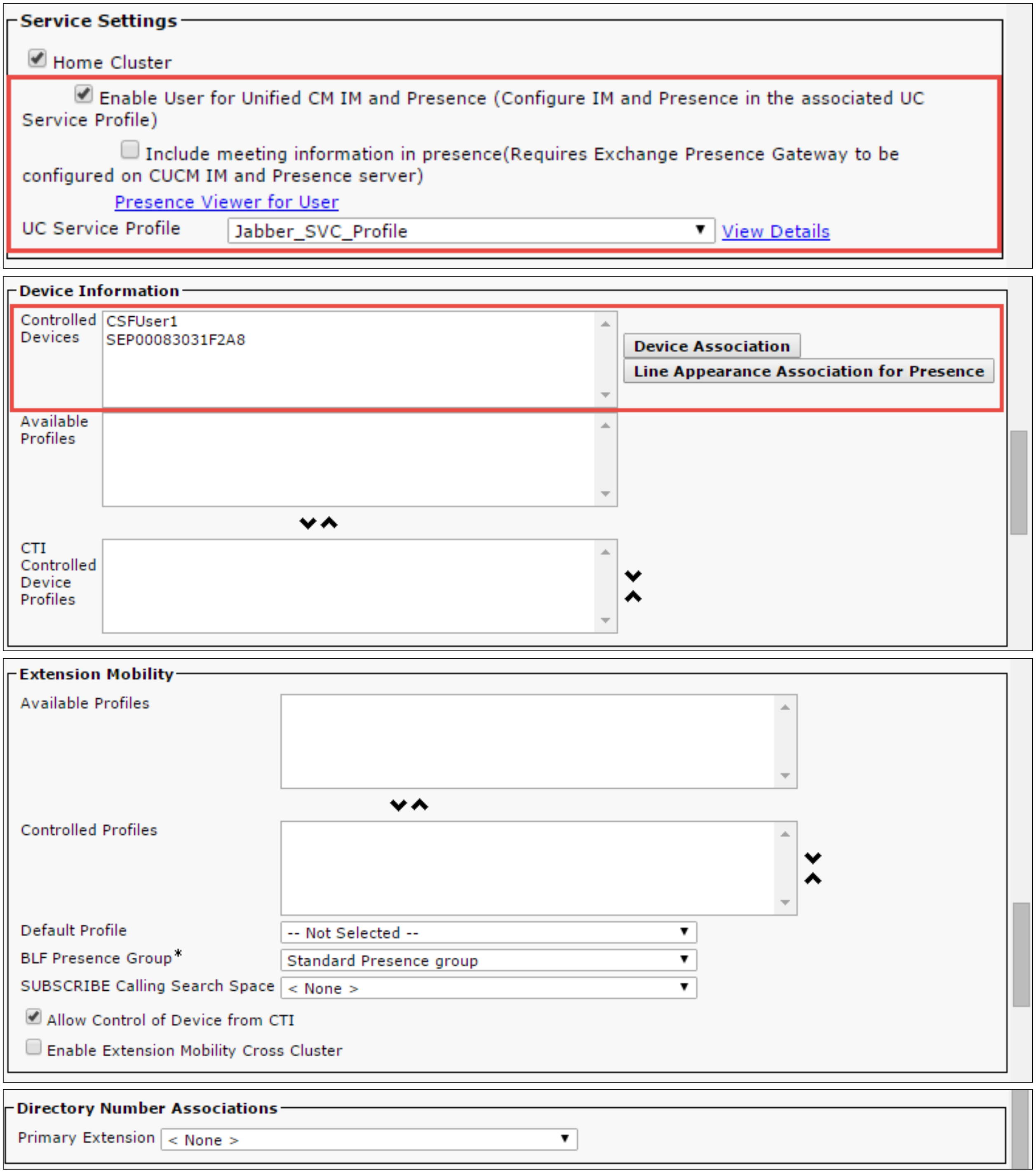
End User Configuration (continued…)



User Information

User Status: Enabled Local User
User ID*: jabber1
Password: ••
Confirm Password: ••
Self-Service User ID:
PIN: ••
Confirm PIN: ••
Last name*: cisco
Middle name:
First name:
Title:
Directory URI: jabber1@lab.tekvizion.com
Telephone Number:
Home Number:
Mobile Number:
Pager Number:
Mail ID:
Manager User ID:
Department:
User Locale: < None >
Associated PC:
Digest Credentials: ••
Confirm Digest Credentials: ••
User Profile: Use System Default("Standard (Factory Default) U

End User Configuration(continued…)



Service Settings

☑ Home Cluster

☑ Enable User for Unified CM IM and Presence (Configure IM and Presence in the associated UC Service Profile)

☐ Include meeting information in presence(Requires Exchange Presence Gateway to be configured on CUCM IM and Presence server)

Presence Viewer for User

UC Service Profile: Jabber_SVC_Profile ▼ View Details

Device Information

Controlled Devices: CSFUser1
SEP00083031F2A8

Available Profiles:

CTI Controlled Device Profiles:

Extension Mobility

Available Profiles:

Controlled Profiles:

Default Profile: -- Not Selected -- ▼

BLF Presence Group*: Standard Presence group ▼

SUBSCRIBE Calling Search Space: < None > ▼

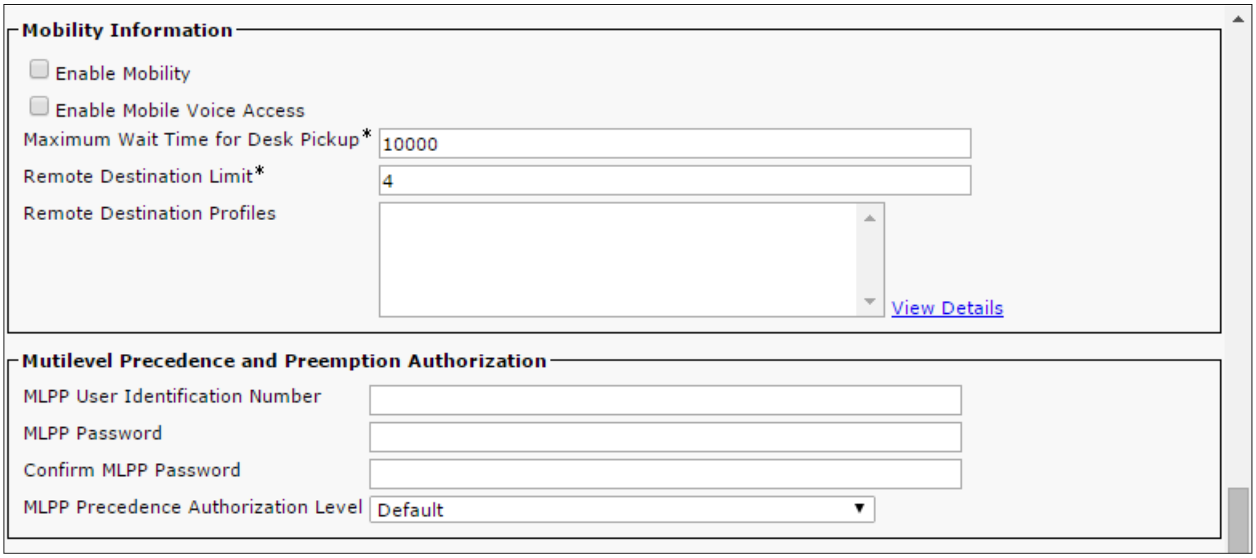
☑ Allow Control of Device from CTI

☐ Enable Extension Mobility Cross Cluster

Directory Number Associations

Primary Extension: < None > ▼

End User Configuration(continued…)


Mobility Information

☐ Enable Mobility

☐ Enable Mobile Voice Access

Maximum Wait Time for Desk Pickup* 10000

Remote Destination Limit* 4

Remote Destination Profiles

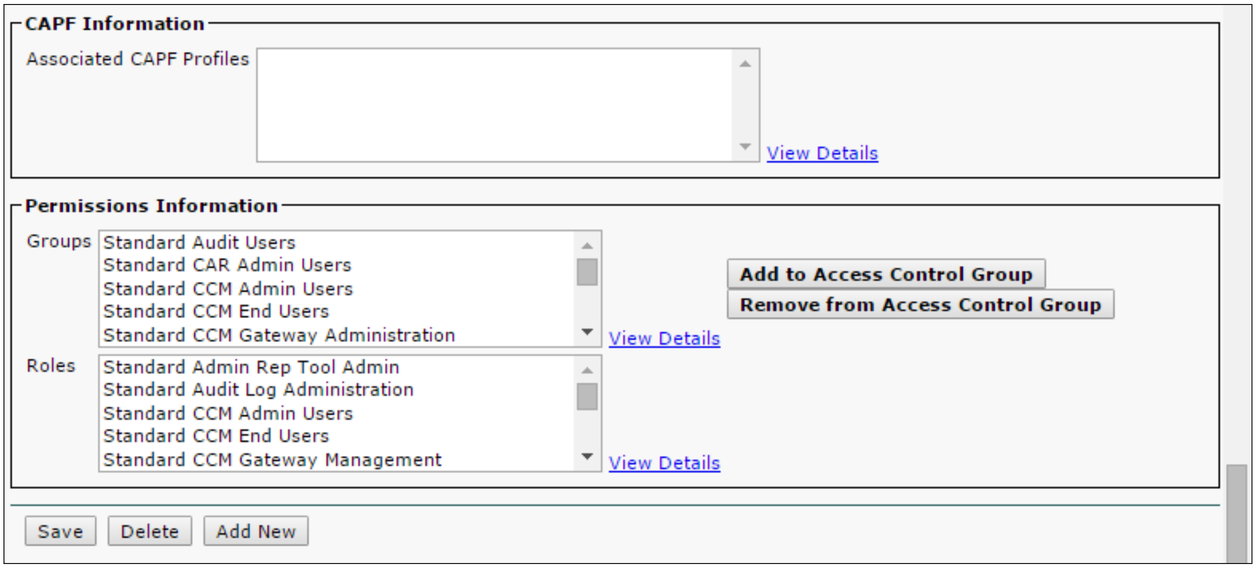
Mutilevel Precedence and Preemption Authorization

MLPP User Identification Number

MLPP Password

Confirm MLPP Password

MLPP Precedence Authorization Level Default


CAPF Information

Associated CAPF Profiles

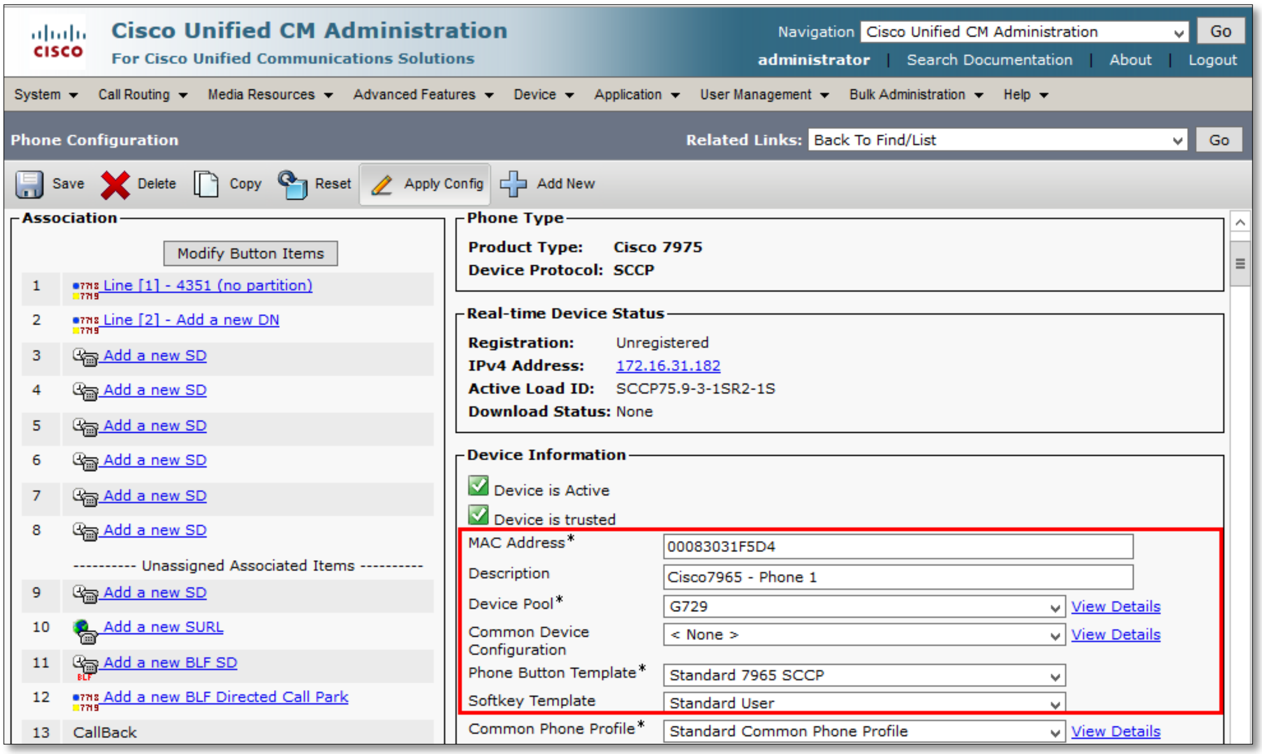
Permissions Information

Groups
- Standard Audit Users
- Standard CAR Admin Users
- Standard CCM Admin Users
- Standard CCM End Users
- Standard CCM Gateway Administration

Roles
- Standard Admin Rep Tool Admin
- Standard Audit Log Administration
- Standard CCM Admin Users
- Standard CCM End Users
- Standard CCM Gateway Management

Cisco IP Phone 7965 SCCP Configuration

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



Association

1	Line [1] - 4351 (no partition)
2	Line [2] - Add a new DN
3	Add a new SD
4	Add a new SD
5	Add a new SD
6	Add a new SD
7	Add a new SD
8	Add a new SD
	---------- Unassigned Associated Items ----------
9	Add a new SD
10	Add a new SURL
11	Add a new BLF SD
12	Add a new BLF Directed Call Park
13	CallBack

Phone Type

Product Type: **Cisco 7975**
Device Protocol: **SCCP**

Real-time Device Status

Registration: Unregistered
IPv4 Address: 172.16.31.182
Active Load ID: SCCP75.9-3-1SR2-1S
Download Status: None

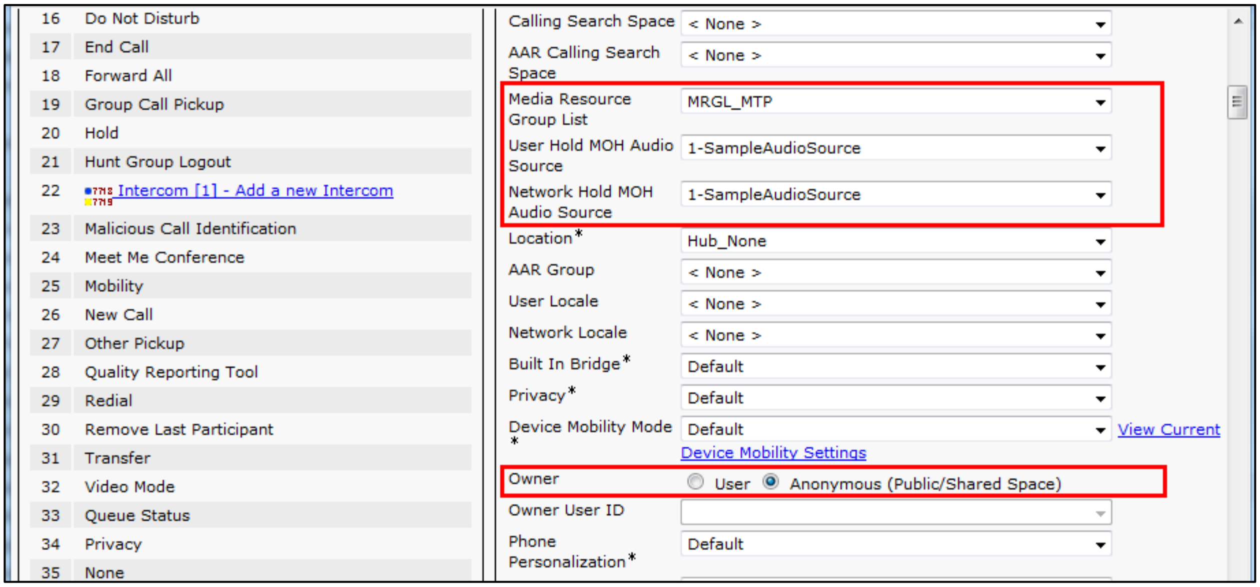
Device Information

☑ Device is Active
☑ Device is trusted

MAC Address*	00083031F5D4
Description	Cisco7965 - Phone 1
Device Pool*	G729 View Details
Common Device Configuration	< None > View Details
Phone Button Template*	Standard 7965 SCCP
Softkey Template	Standard User
Common Phone Profile*	Standard Common Phone Profile View Details

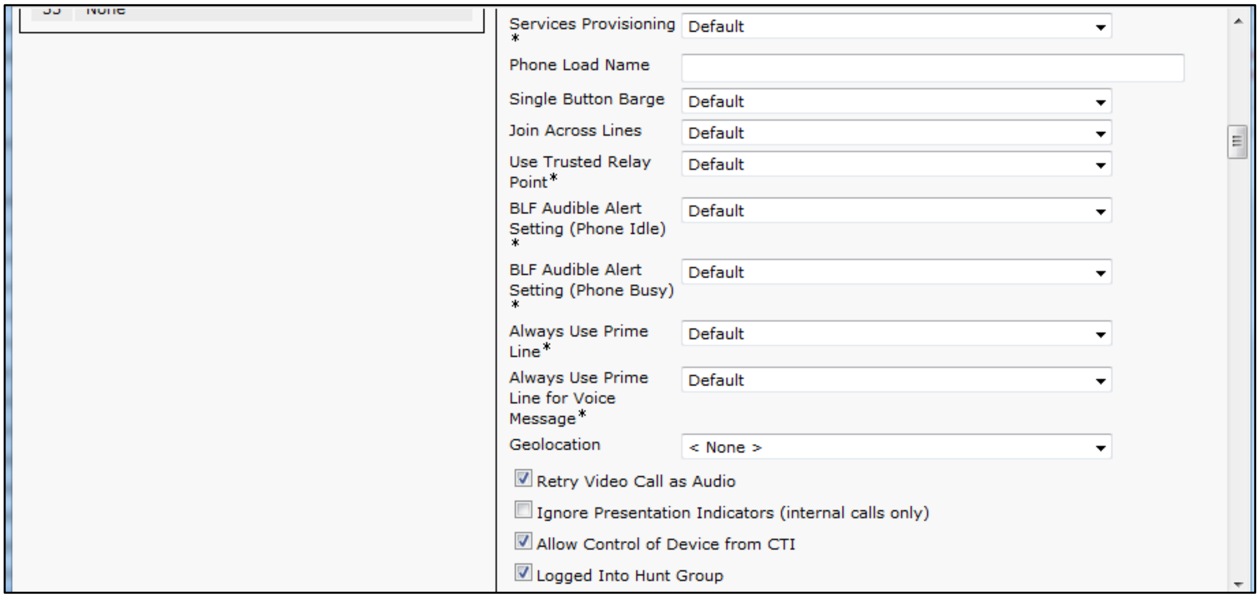
Cisco IP Phone 7965 SCCP Configuration (Continued…)

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



16	Do Not Disturb
17	End Call
18	Forward All
19	Group Call Pickup
20	Hold
21	Hunt Group Logout
22	Intercom [1] - Add a new Intercom
23	Malicious Call Identification
24	Meet Me Conference
25	Mobility
26	New Call
27	Other Pickup
28	Quality Reporting Tool
29	Redial
30	Remove Last Participant
31	Transfer
32	Video Mode
33	Queue Status
34	Privacy
35	None

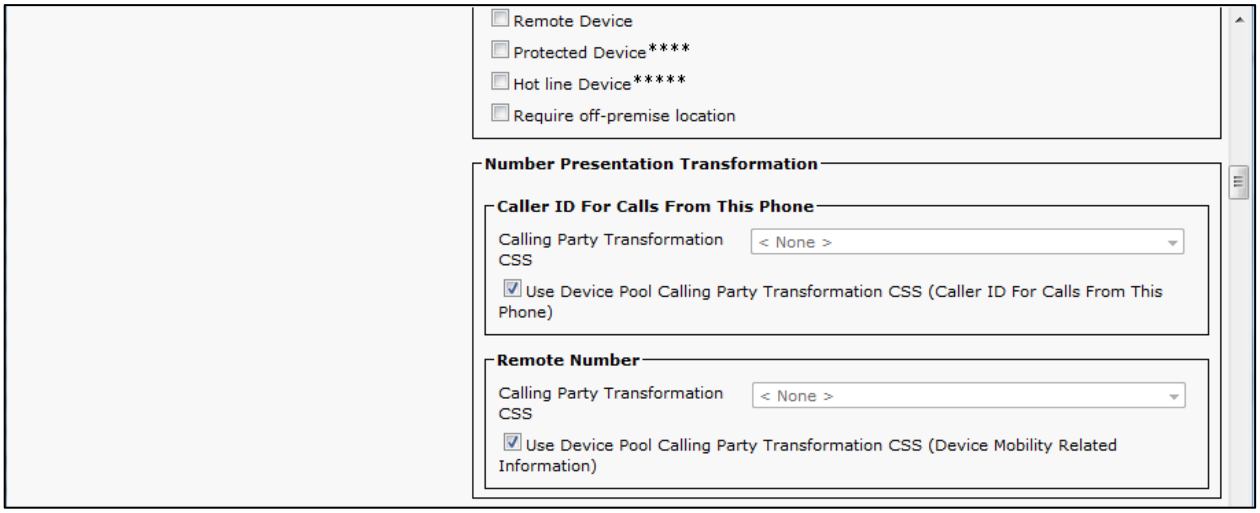
Calling Search Space	< None >
AAR Calling Search Space	< None >
Media Resource Group List	MRGL_MTP
User Hold MOH Audio Source	1-SampleAudioSource
Network Hold MOH Audio Source	1-SampleAudioSource
Location*	Hub_None
AAR Group	< None >
User Locale	< None >
Network Locale	< None >
Built In Bridge*	Default
Privacy*	Default
Device Mobility Mode*	Default View Current Device Mobility Settings
Owner	○ User ● Anonymous (Public/Shared Space)
Owner User ID	
Phone Personalization*	Default



Services Provisioning*	Default
Phone Load Name	
Single Button Barge	Default
Join Across Lines	Default
Use Trusted Relay Point*	Default
BLF Audible Alert Setting (Phone Idle)*	Default
BLF Audible Alert Setting (Phone Busy)*	Default
Always Use Prime Line*	Default
Always Use Prime Line for Voice Message*	Default
Geolocation	< None >

☑ Retry Video Call as Audio
☐ Ignore Presentation Indicators (internal calls only)
☑ Allow Control of Device from CTI
☑ Logged Into Hunt Group

Cisco IP Phone 7965 SCCP Configuration (Continued…)



☐ Remote Device
☐ Protected Device****
☐ Hot line Device*****
☐ Require off-premise location

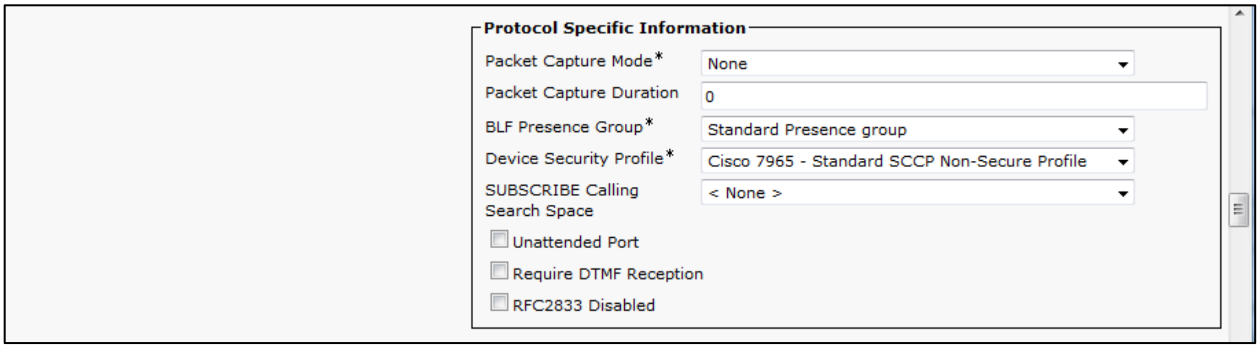
Number Presentation Transformation

Caller ID For Calls From This Phone

Calling Party Transformation CSS: < None >
☑ Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)

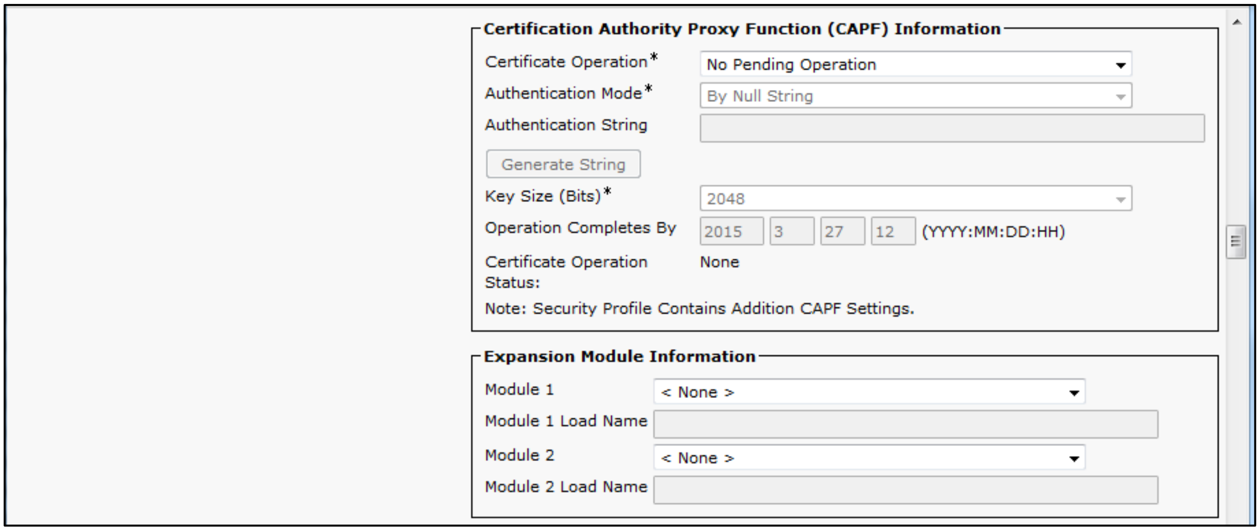
Remote Number

Calling Party Transformation CSS: < None >
☑ Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)



Protocol Specific Information

Packet Capture Mode*: None
Packet Capture Duration: 0
BLF Presence Group*: Standard Presence group
Device Security Profile*: Cisco 7965 - Standard SCCP Non-Secure Profile
SUBSCRIBE Calling Search Space: < None >
☐ Unattended Port
☐ Require DTMF Reception
☐ RFC2833 Disabled



Certification Authority Proxy Function (CAPF) Information

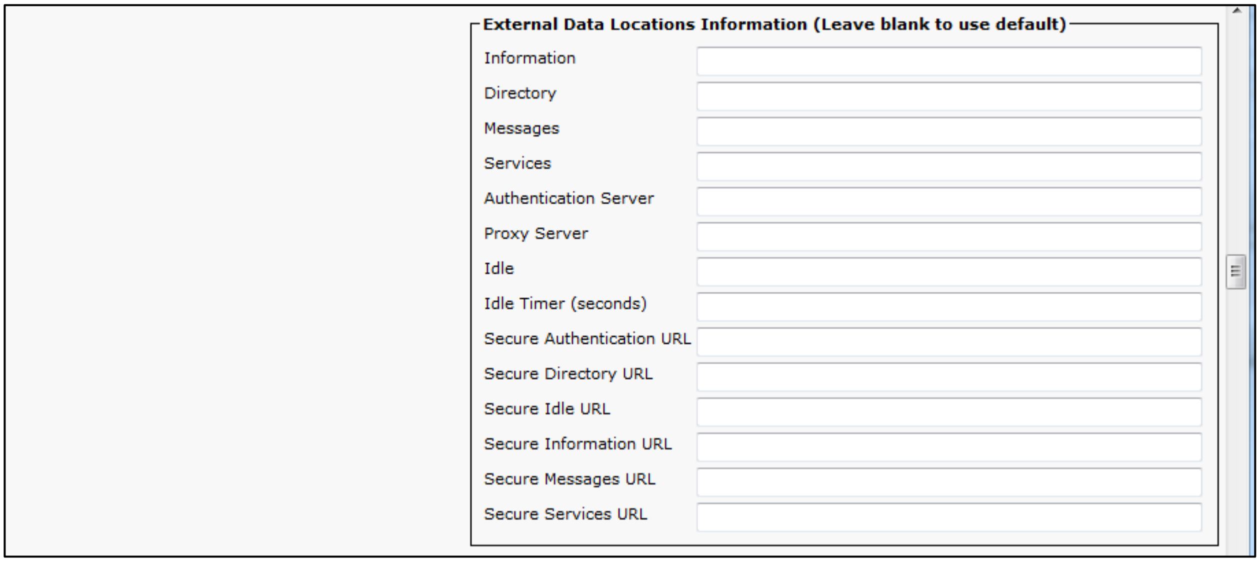
Certificate Operation*: No Pending Operation
Authentication Mode*: By Null String
Authentication String:

Key Size (Bits)*: 2048
Operation Completes By: 2015 3 27 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None
Note: Security Profile Contains Addition CAPF Settings.

Expansion Module Information

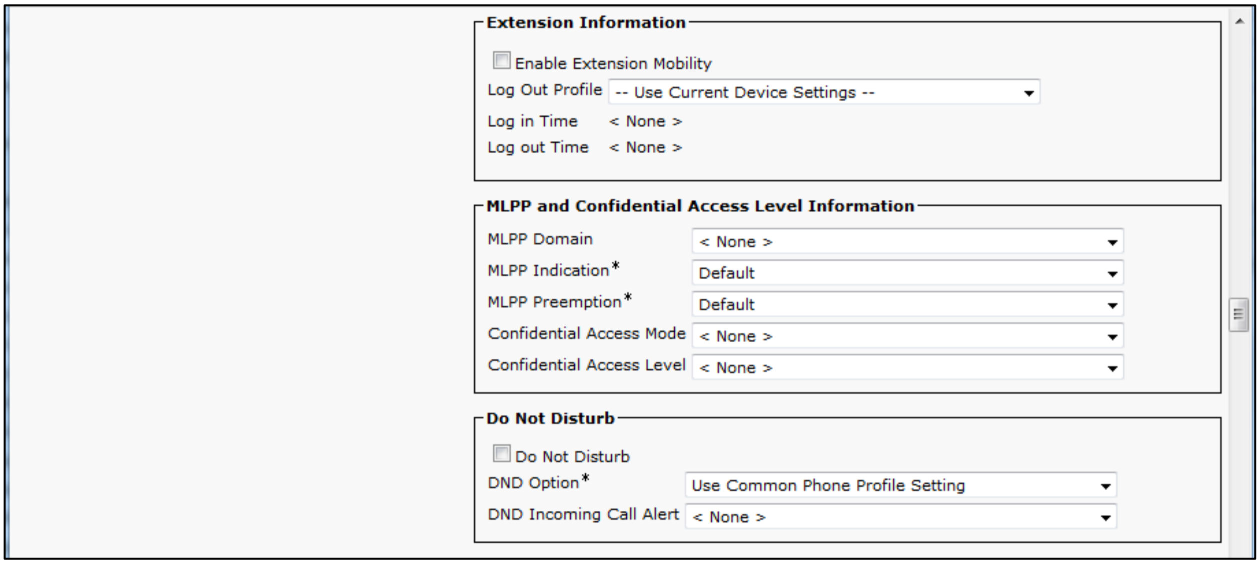
Module 1: < None >
Module 1 Load Name:
Module 2: < None >
Module 2 Load Name:

Cisco IP Phone 7965 SCCP Configuration (Continued…)



External Data Locations Information (Leave blank to use default)

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



Extension Information

☐ Enable Extension Mobility

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

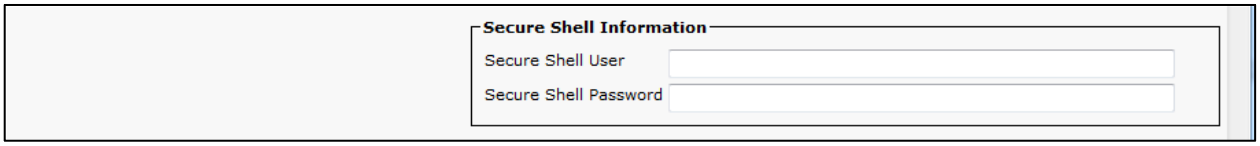
MLPP and Confidential Access Level Information

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

Do Not Disturb

☐ Do Not Disturb

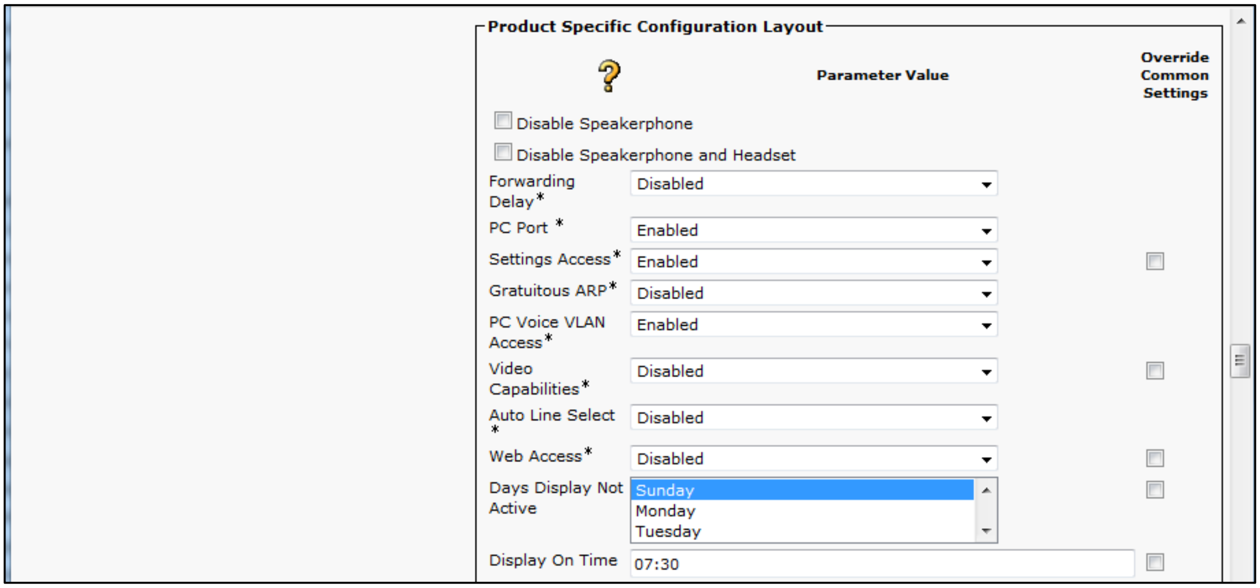
DND Option*	Use Common Phone Profile Setting
DND Incoming Call Alert	< None >



Secure Shell Information

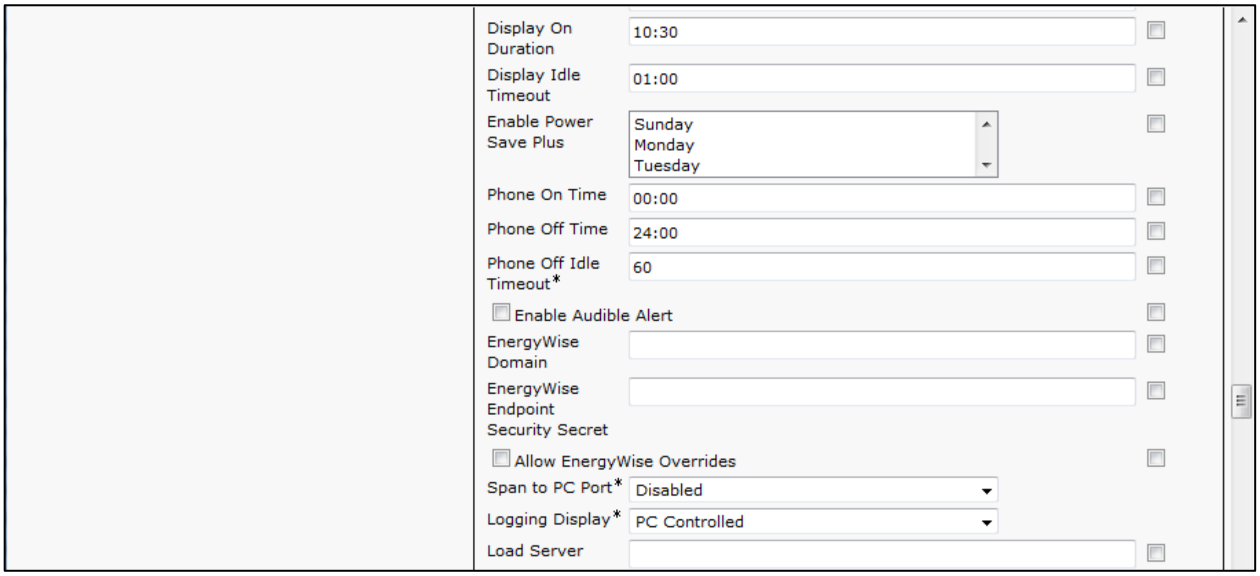
Secure Shell User	
Secure Shell Password	

Cisco IP Phone 7965 SCCP Configuration (Continued…)

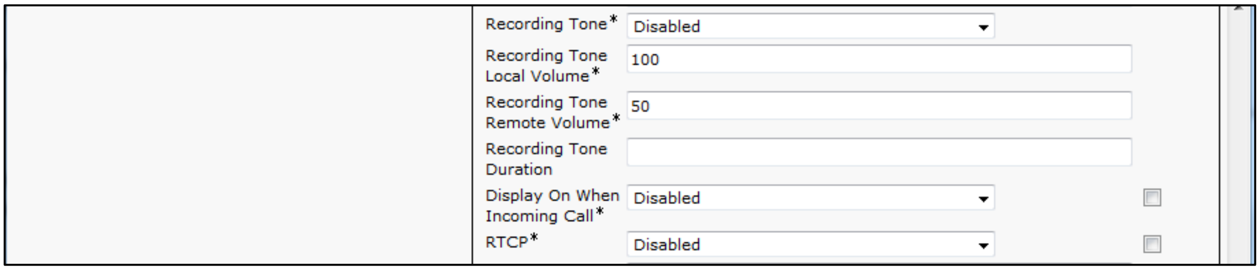


Product Specific Configuration Layout

	Parameter Value	Override Common Settings
☐ Disable Speakerphone		
☐ Disable Speakerphone and Headset		
Forwarding Delay*	Disabled	
PC Port *	Enabled	
Settings Access*	Enabled	☐
Gratuitous ARP*	Disabled	
PC Voice VLAN Access*	Enabled	
Video Capabilities*	Disabled	☐
Auto Line Select *	Disabled	
Web Access*	Disabled	☐
Days Display Not Active	Sunday Monday Tuesday	☐
Display On Time	07:30	☐

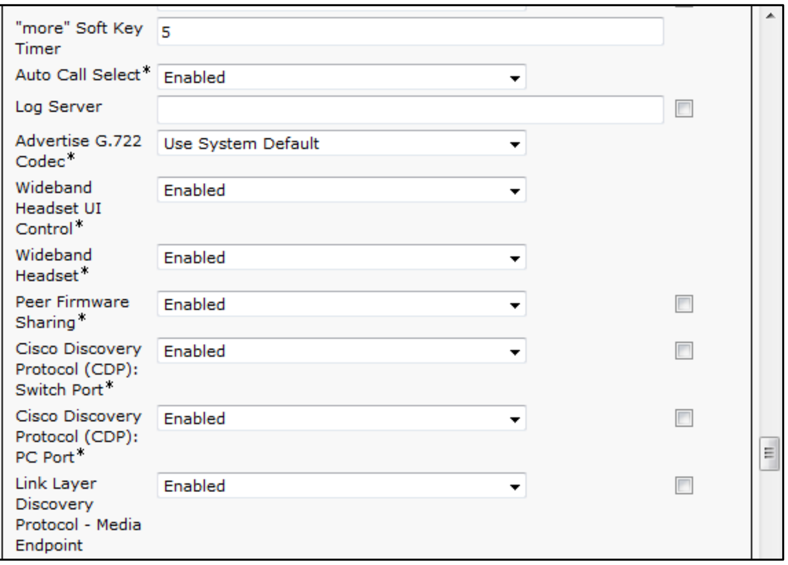


Display On Duration	10:30	☐
Display Idle Timeout	01:00	☐
Enable Power Save Plus	Sunday Monday Tuesday	☐
Phone On Time	00:00	☐
Phone Off Time	24:00	☐
Phone Off Idle Timeout*	60	☐
☐ Enable Audible Alert		☐
EnergyWise Domain		☐
EnergyWise Endpoint Security Secret		☐
☐ Allow EnergyWise Overrides		☐
Span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



Recording Tone*	Disabled	
Recording Tone Local Volume*	100	
Recording Tone Remote Volume*	50	
Recording Tone Duration		
Display On When Incoming Call*	Disabled	☐
RTCP*	Disabled	☐

Cisco IP Phone 7965 SCCP Configuration (Continued…)



"more" Soft Key Timer: 5

Auto Call Select*: Enabled

Log Server: ☐

Advertise G.722 Codec*: Use System Default

Wideband Headset UI Control*: Enabled

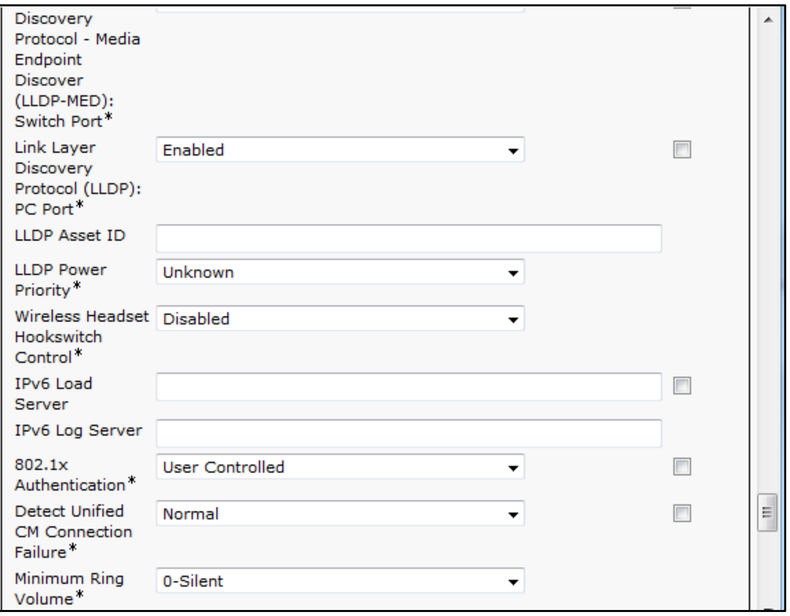
Wideband Headset*: Enabled

Peer Firmware Sharing*: Enabled ☐

Cisco Discovery Protocol (CDP): Switch Port*: Enabled ☐

Cisco Discovery Protocol (CDP): PC Port*: Enabled ☐

Link Layer Discovery Protocol - Media Endpoint: Enabled ☐



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

Link Layer Discovery Protocol (LLDP): PC Port*: Enabled ☐

LLDP Asset ID:

LLDP Power Priority*: Unknown

Wireless Headset Hookswitch Control*: Disabled

IPv6 Load Server: ☐

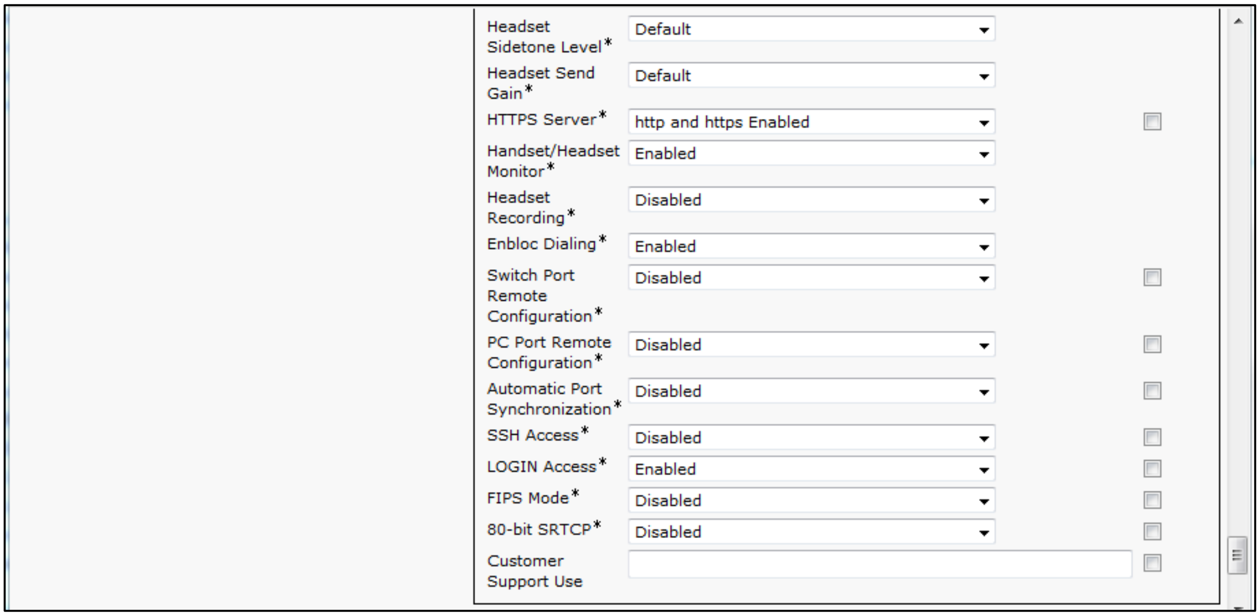
IPv6 Log Server:

802.1x Authentication*: User Controlled ☐

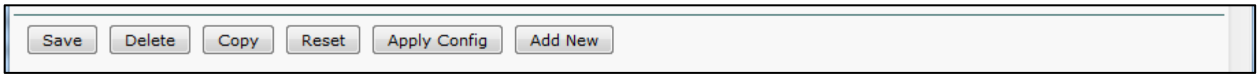
Detect Unified CM Connection Failure*: Normal ☐

Minimum Ring Volume*: 0-Silent

Cisco IP Phone 7965 SCCP Configuration (Continued…)

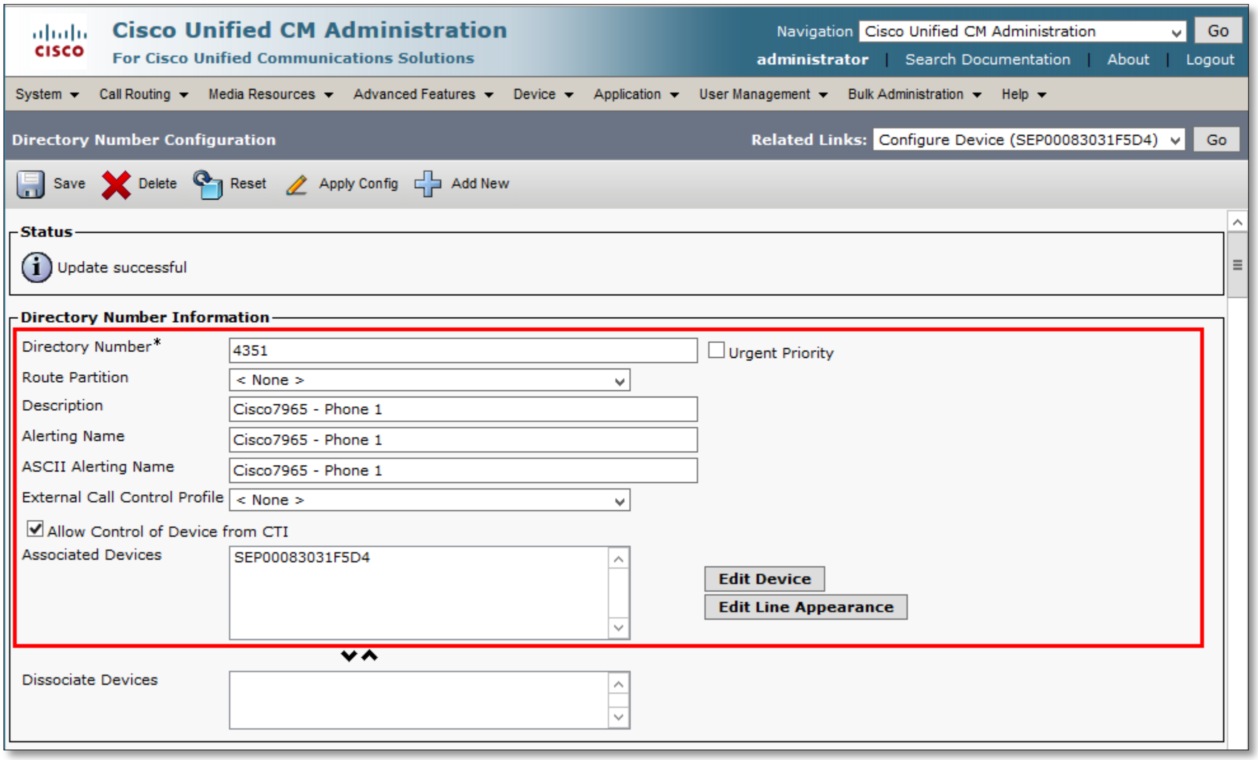


Headset Sidetone Level*	Default	
Headset Send Gain*	Default	
HTTPS Server*	http and https Enabled	☐
Handset/Headset Monitor*	Enabled	
Headset Recording*	Disabled	
Enbloc Dialing*	Enabled	
Switch Port Remote Configuration*	Disabled	☐
PC Port Remote Configuration*	Disabled	☐
Automatic Port Synchronization*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



Cisco IP Phone 7965 SCCP Configuration (Continued…)

Set Directory Number* = 4351. This is used in this example.
Set Description = 7323204351. 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.



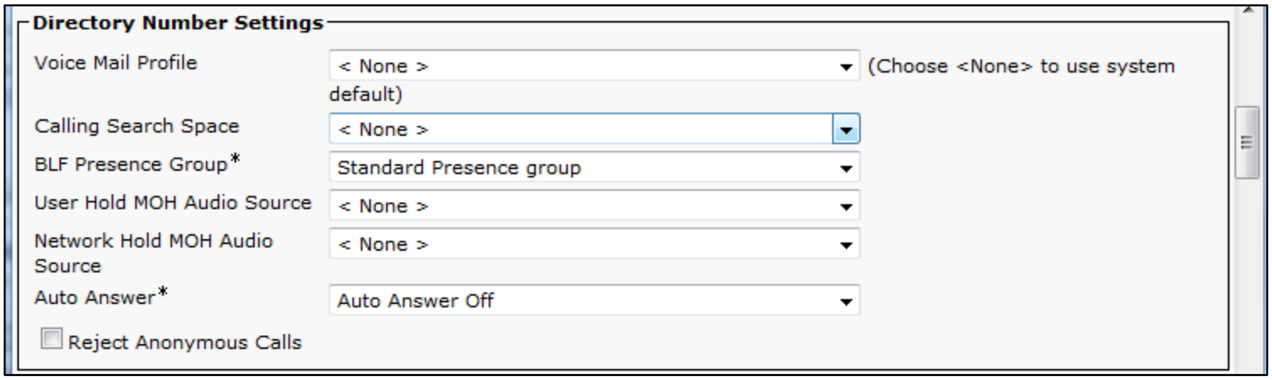
Directory Number Configuration

Status

Update successful

Directory Number Information

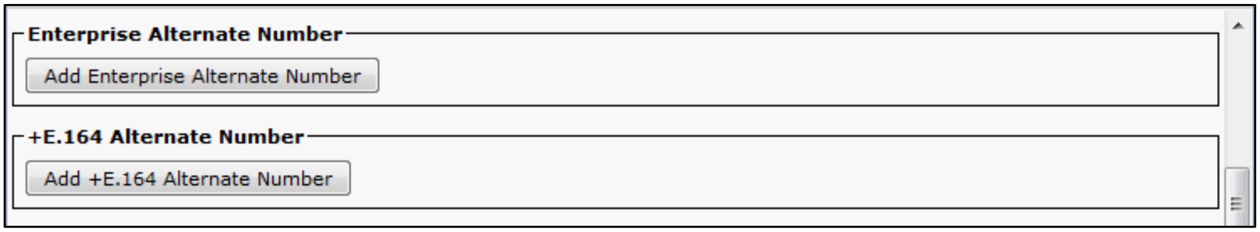
Directory Number* 4351 ☐ Urgent Priority
Route Partition < None >
Description Cisco7965 - Phone 1
Alerting Name Cisco7965 - Phone 1
ASCII Alerting Name Cisco7965 - Phone 1
External Call Control Profile < None >
☑ Allow Control of Device from CTI
Associated Devices SEP00083031F5D4
Dissociate Devices

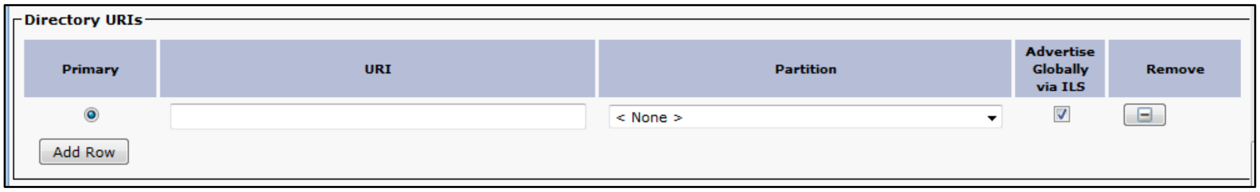


Directory Number Settings

Voice Mail Profile < None > (Choose <None> to use system default)
Calling Search Space < None >
BLF Presence Group* Standard Presence group
User Hold MOH Audio Source < None >
Network Hold MOH Audio Source < None >
Auto Answer* Auto Answer Off
☐ Reject Anonymous Calls

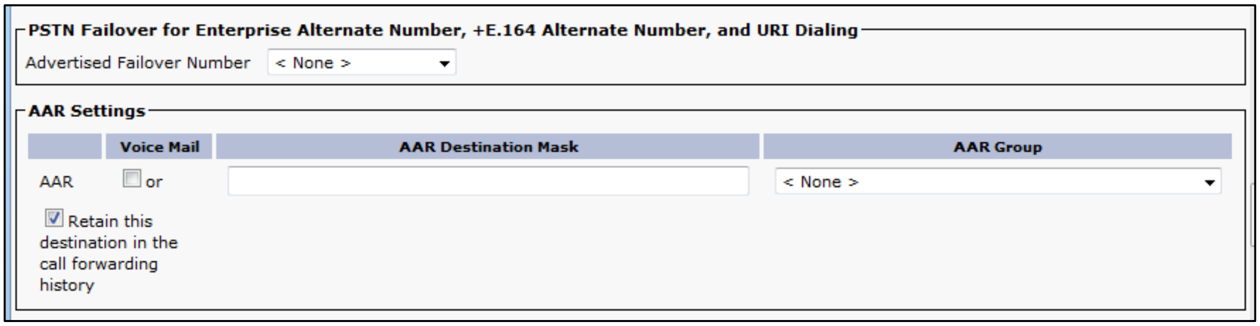
Cisco IP Phone 7965 SCCP Configuration (Continued…)





Directory URIs

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



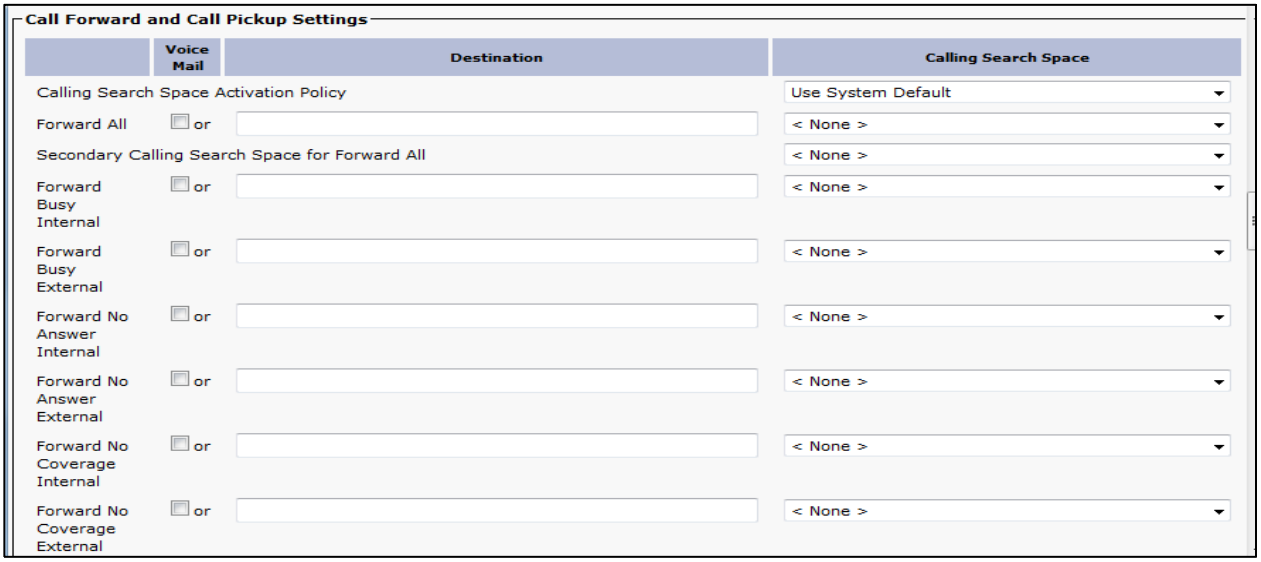
PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing

Advertised Failover Number: < None >

AAR Settings

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

☑ Retain this destination in the call forwarding history



Call Forward and Call Pickup Settings

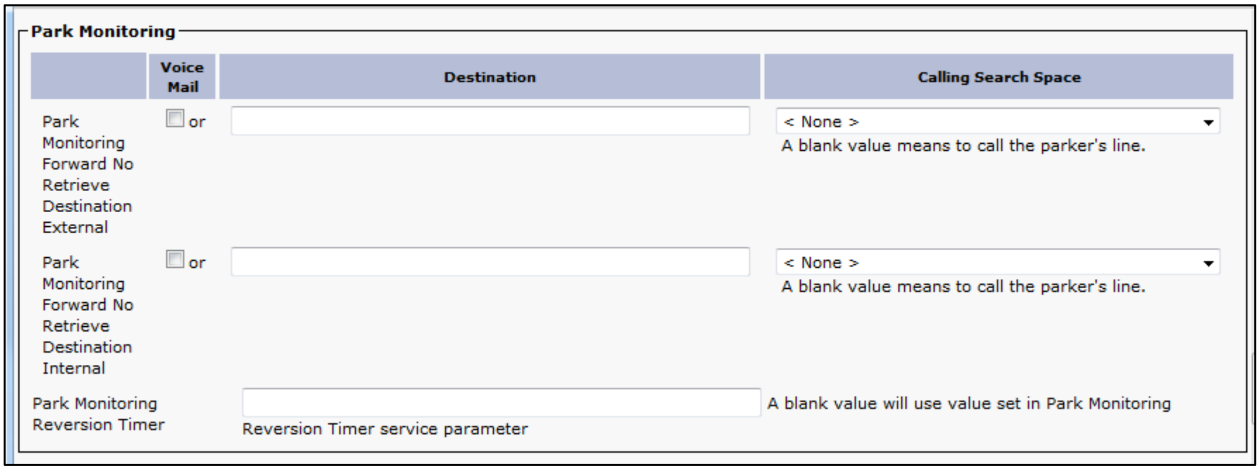
	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			Use System Default
Forward All	☐ or		< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	☐ or		< None >
Forward Busy External	☐ or		< None >
Forward No Answer Internal	☐ or		< None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

Cisco IP Phone 7965 SCCP Configuration (Continued…)

Forward on CTI Failure	☐ or		< None >
Forward Unregistered Internal	☐ or		< None >
Forward Unregistered External	☐ or		< None >

No Answer Ring Duration (seconds)

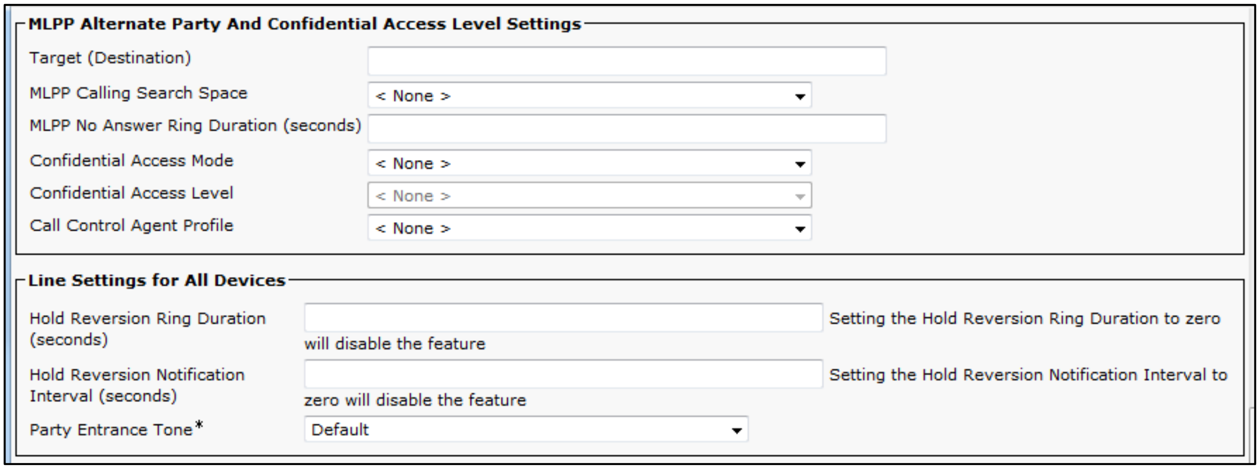
Call Pickup Group: < None >



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 >

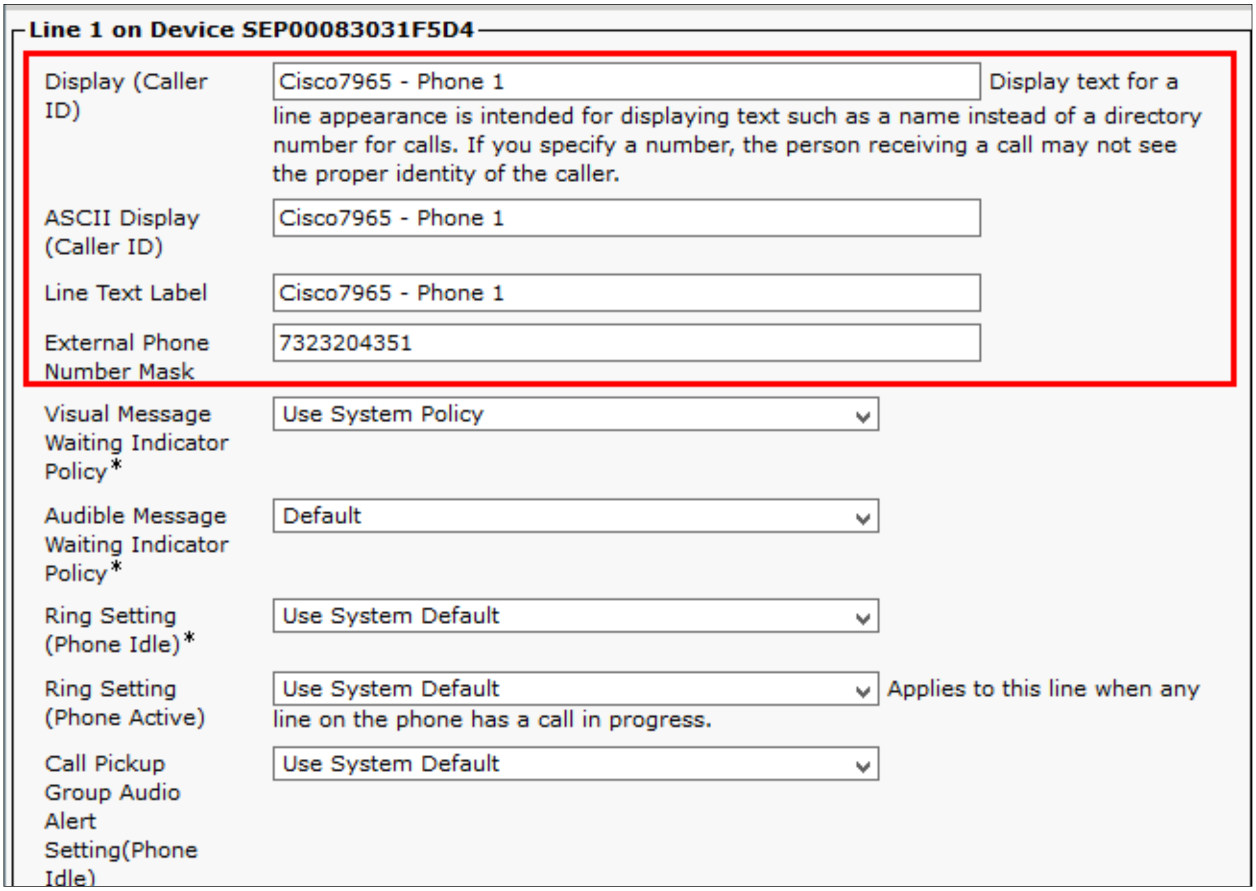
Line Settings for All Devices

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 7965 SCCP Configuration (Continued…)



Line 1 on Device SEP00083031F5D4

Display (Caller ID): Cisco7965 - Phone 1 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): Cisco7965 - Phone 1

Line Text Label: Cisco7965 - Phone 1

External Phone Number Mask: 7323204351

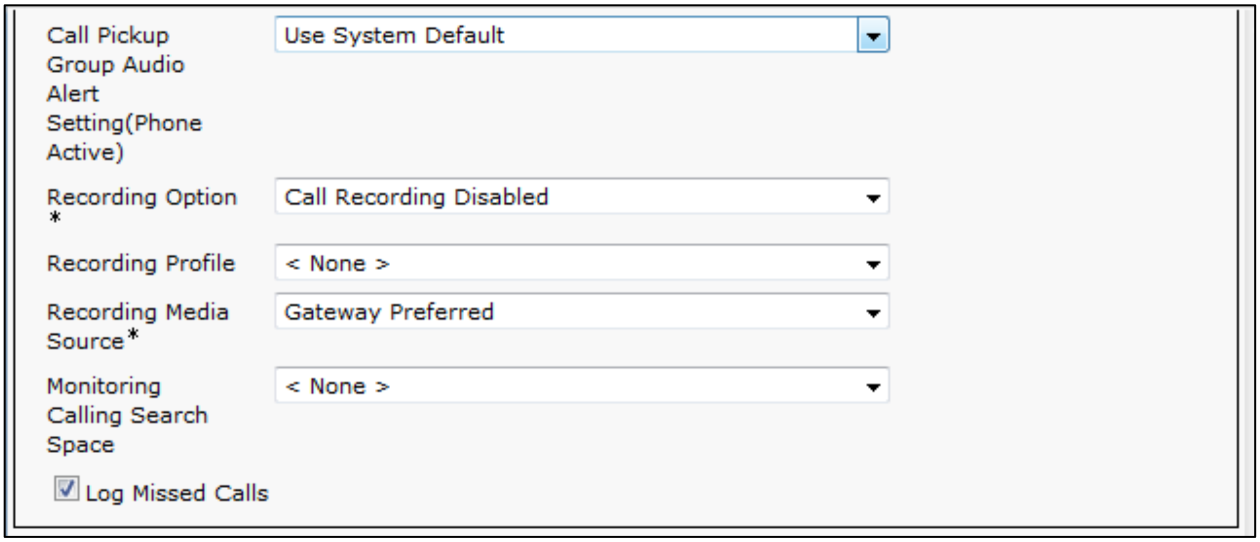
Visual Message Waiting Indicator Policy*: Use System Policy

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

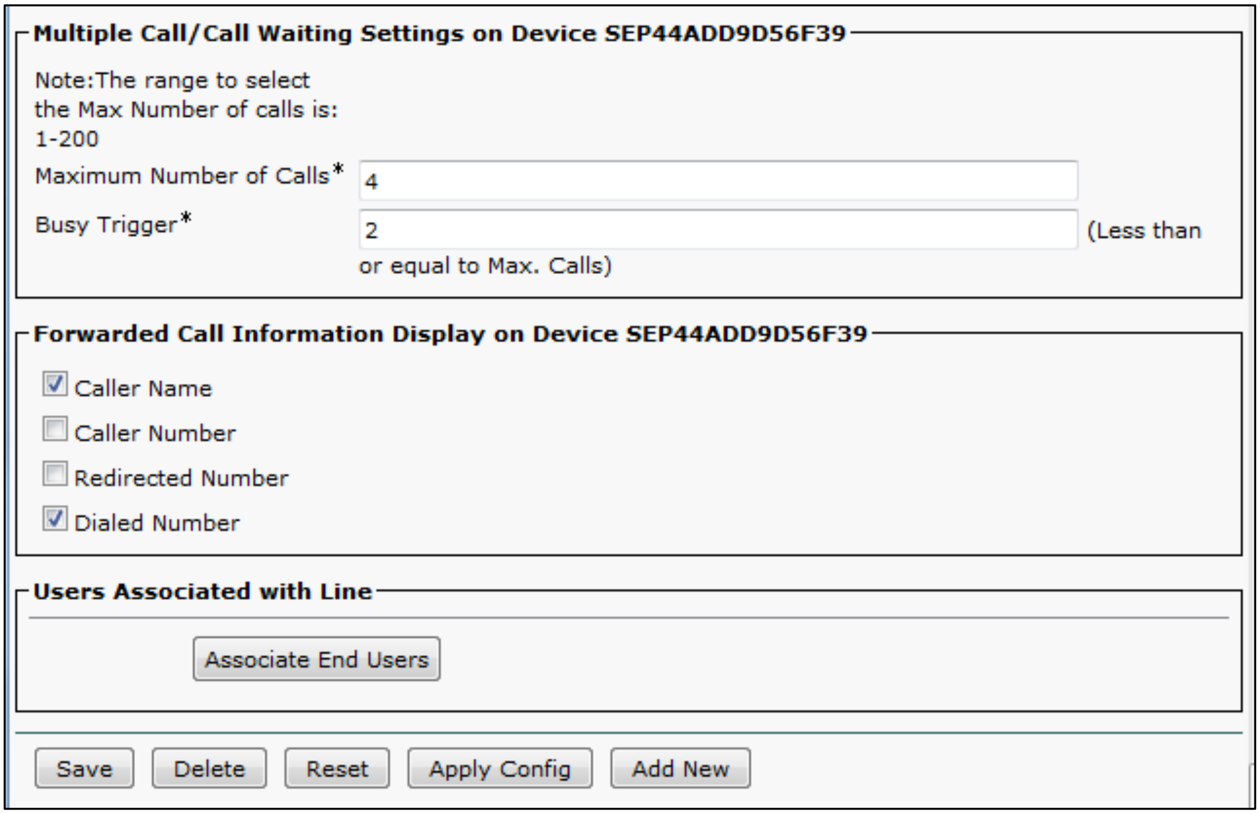
Recording Profile: < None >

Recording Media Source*: Gateway Preferred

Monitoring Calling Search Space: < None >

☑ Log Missed Calls

Cisco IP Phone 7965 SCCP Configuration (Continued…)



Multiple Call/Call Waiting Settings on Device SEP44ADD9D56F39

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

Maximum Number of Calls* 4

Busy Trigger* 2 (Less than or equal to Max. Calls)

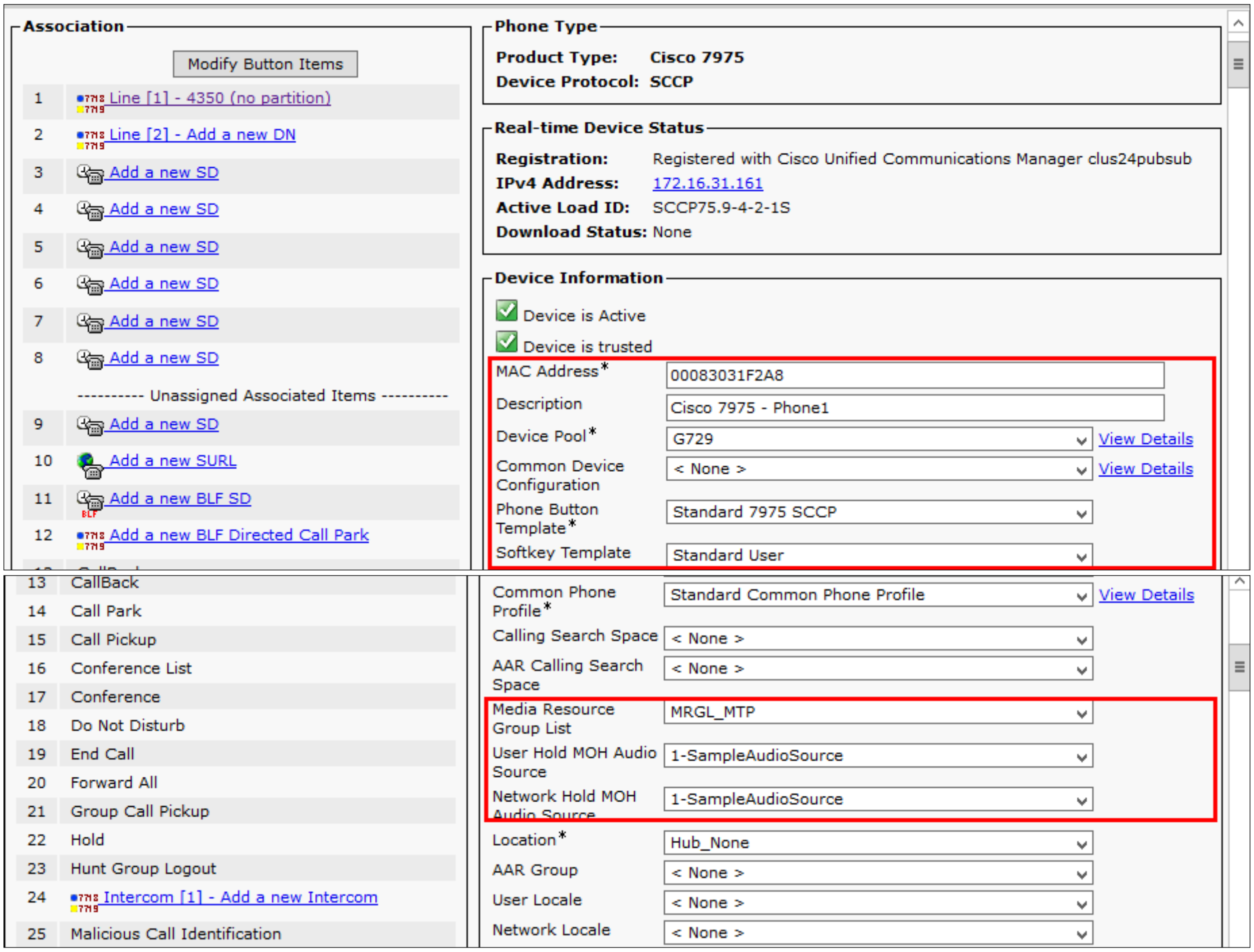
Forwarded Call Information Display on Device SEP44ADD9D56F39

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

Users Associated with Line

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



Association

Modify Button Items

1	Line [1] - 4350 (no partition)
2	Line [2] - Add a new DN
3	Add a new SD
4	Add a new SD
5	Add a new SD
6	Add a new SD
7	Add a new SD
8	Add a new SD
	--------- Unassigned Associated Items ---------
9	Add a new SD
10	Add a new SURL
11	Add a new BLF SD
12	Add a new BLF Directed Call Park
13	CallBack
14	Call Park
15	Call Pickup
16	Conference List
17	Conference
18	Do Not Disturb
19	End Call
20	Forward All
21	Group Call Pickup
22	Hold
23	Hunt Group Logout
24	Intercom [1] - Add a new Intercom
25	Malicious Call Identification

Phone Type

Product Type: **Cisco 7975**
Device Protocol: **SCCP**

Real-time Device Status

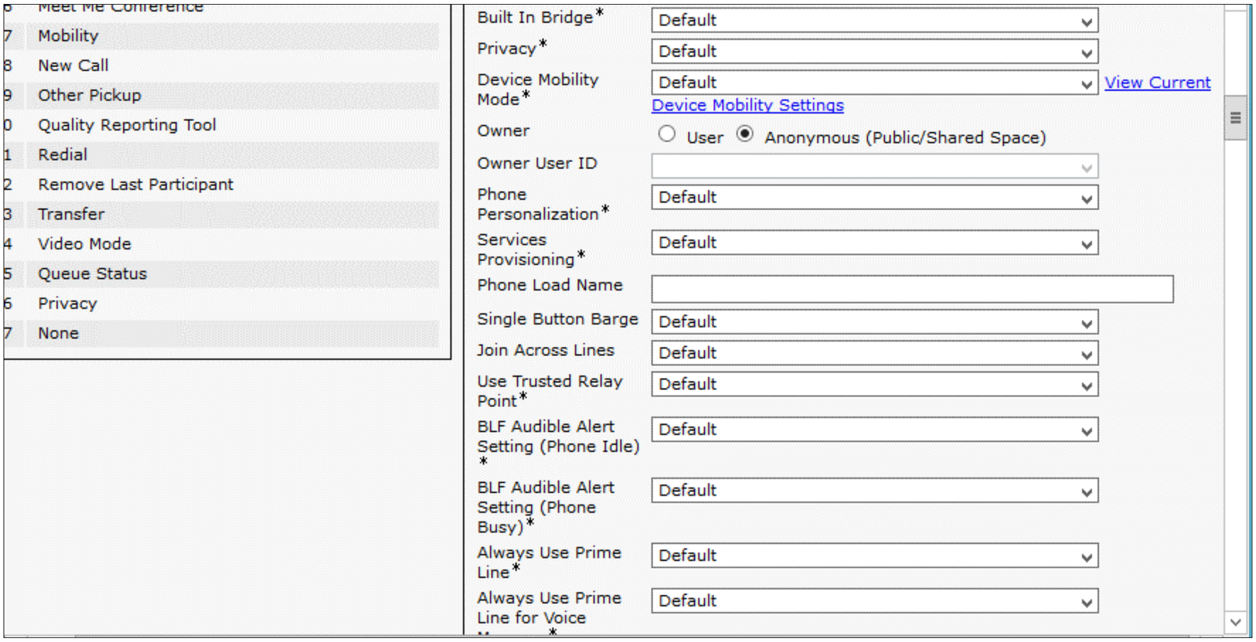
Registration: Registered with Cisco Unified Communications Manager clus24pubsub
IPv4 Address: 172.16.31.161
Active Load ID: SCCP75.9-4-2-1S
Download Status: None

Device Information

☑ Device is Active
☑ Device is trusted

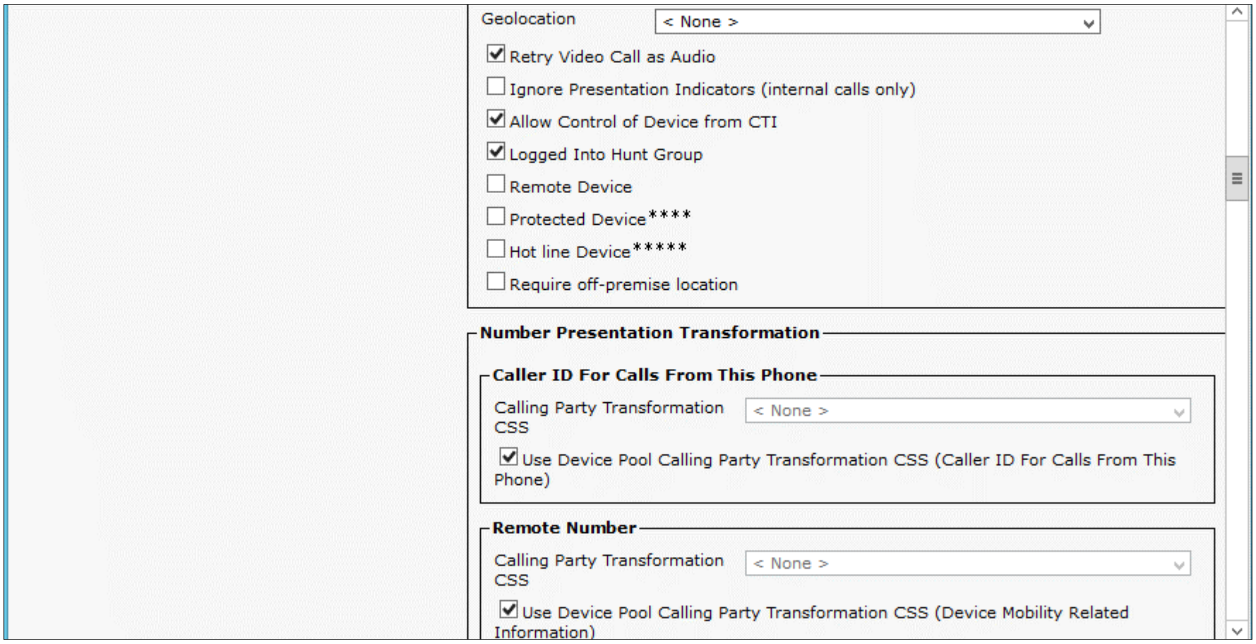
MAC Address*	00083031F2A8
Description	Cisco 7975 - Phone1
Device Pool*	G729
Common Device Configuration	< None >
Phone Button Template*	Standard 7975 SCCP
Softkey Template	Standard User
Common Phone Profile*	Standard Common Phone Profile
Calling Search Space	< None >
AAR Calling Search Space	< None >
Media Resource Group List	MRGL_MTP
User Hold MOH Audio Source	1-SampleAudioSource
Network Hold MOH Audio Source	1-SampleAudioSource
Location*	Hub_None
AAR Group	< None >
User Locale	< None >
Network Locale	< None >

Cisco IP Phone 7975 SCCP Configuration (Continued…)



6 Meet Me Conference
7 Mobility
8 New Call
9 Other Pickup
0 Quality Reporting Tool
1 Redial
2 Remove Last Participant
3 Transfer
4 Video Mode
5 Queue Status
6 Privacy
7 None

Built In Bridge*	Default
Privacy*	Default
Device Mobility Mode*	Default View Current Device Mobility Settings
Owner	○ User ● Anonymous (Public/Shared Space)
Owner User ID	
Phone Personalization*	Default
Services Provisioning*	Default
Phone Load Name	
Single Button Barge	Default
Join Across Lines	Default
Use Trusted Relay Point*	Default
BLF Audible Alert Setting (Phone Idle)*	Default
BLF Audible Alert Setting (Phone Busy)*	Default
Always Use Prime Line*	Default
Always Use Prime Line for Voice	Default



Geolocation: < None >

☑ Retry Video Call as Audio
☐ Ignore Presentation Indicators (internal calls only)
☑ Allow Control of Device from CTI
☑ Logged Into Hunt Group
☐ Remote Device
☐ Protected Device****
☐ Hot line Device*****
☐ Require off-premise location

Number Presentation Transformation

Caller ID For Calls From This Phone

Calling Party Transformation CSS: < None >

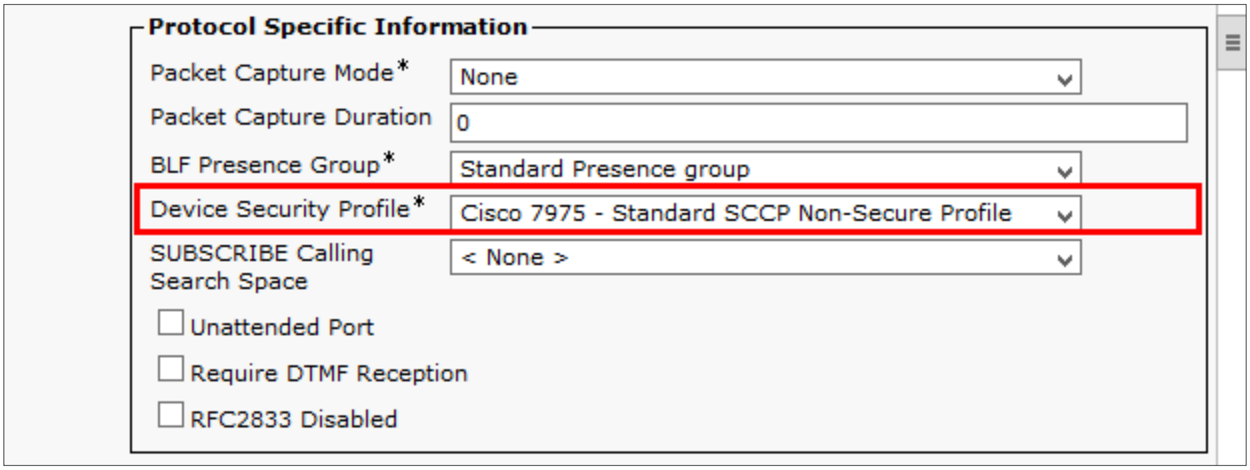
☑ Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)

Remote Number

Calling Party Transformation CSS: < None >

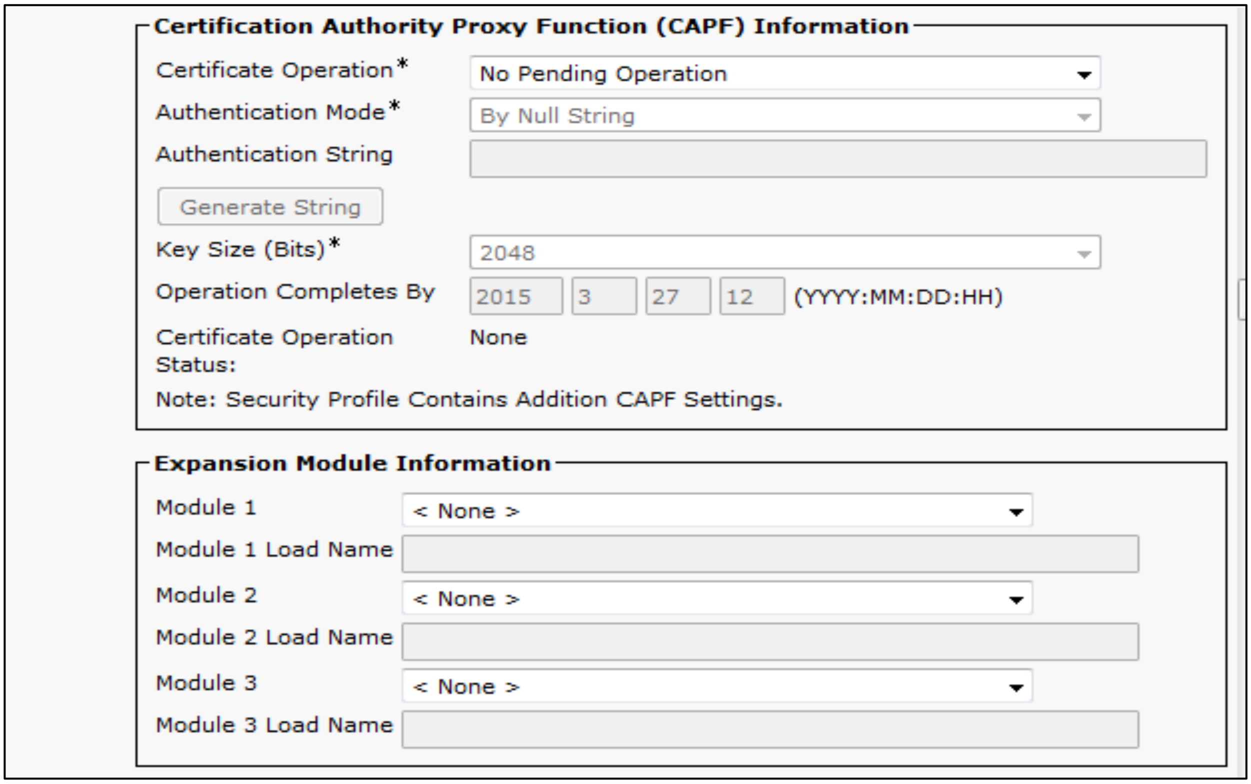
☑ Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

Cisco IP Phone 7975 SCCP Configuration (Continued…)



Protocol Specific Information

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



Certification Authority Proxy Function (CAPF) Information

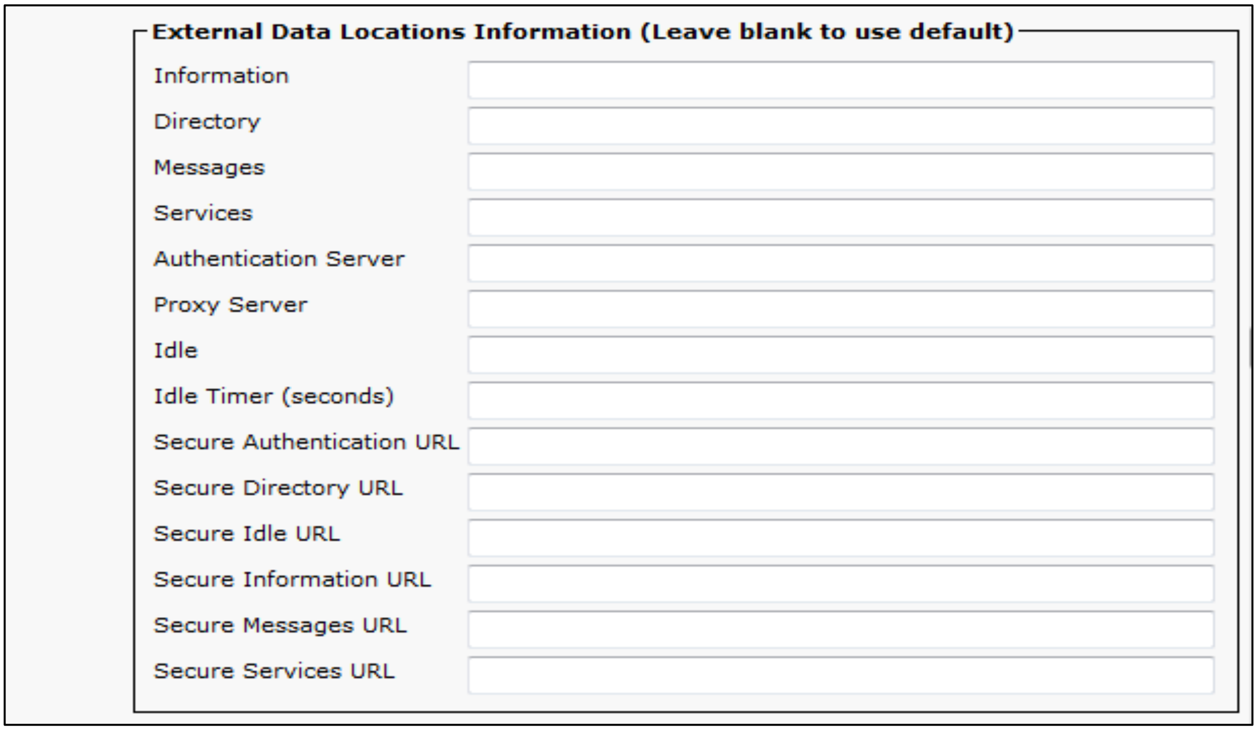
Certificate Operation*: No Pending Operation
Authentication Mode*: By Null String
Authentication String:

Key Size (Bits)*: 2048
Operation Completes By: 2015 3 27 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None
Note: Security Profile Contains Addition CAPF Settings.

Expansion Module Information

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

Cisco IP Phone 7975 SCCP Configuration (Continued…)



External Data Locations Information (Leave blank to use default)

Information

Directory

Messages

Services

Authentication Server

Proxy Server

Idle

Idle Timer (seconds)

Secure Authentication URL

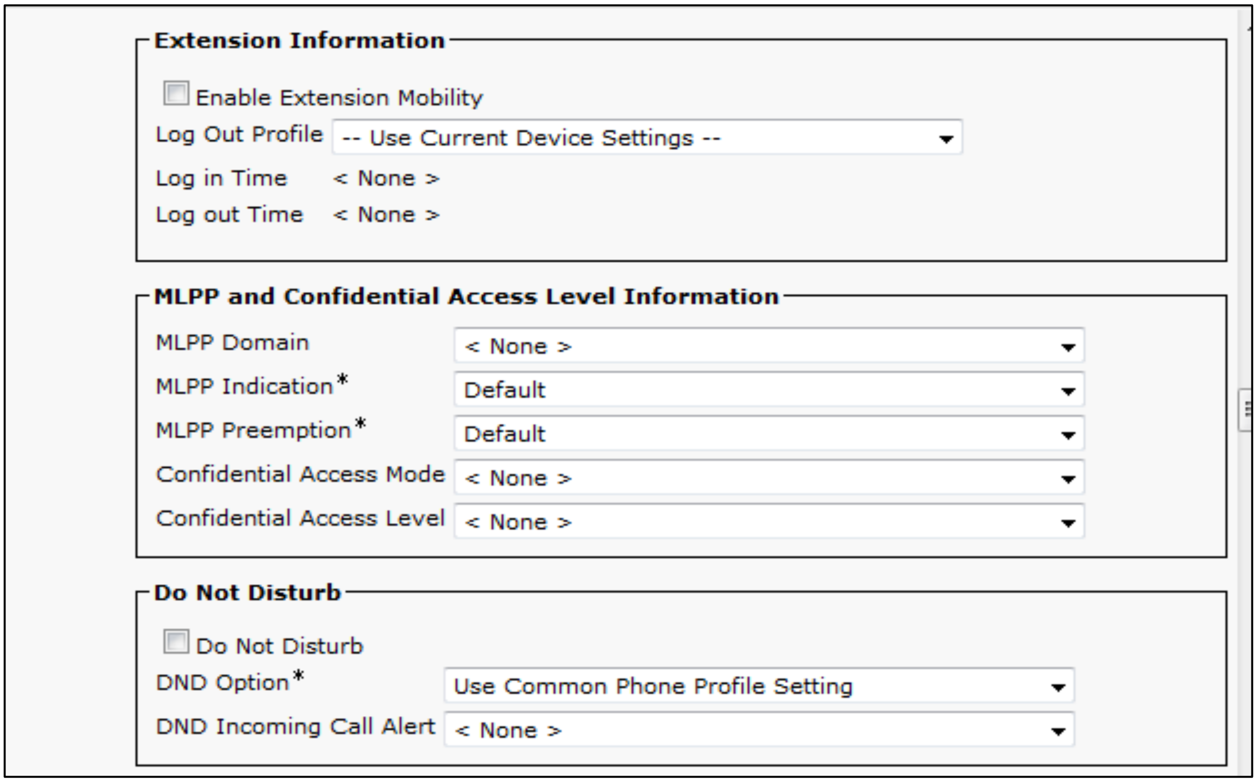
Secure Directory URL

Secure Idle URL

Secure Information URL

Secure Messages URL

Secure Services URL



Extension Information

☐ Enable Extension Mobility

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

Log in Time: < None >

Log out Time: < None >

MLPP and Confidential Access Level Information

MLPP Domain: < None >

MLPP Indication*: Default

MLPP Preemption*: Default

Confidential Access Mode: < None >

Confidential Access Level: < None >

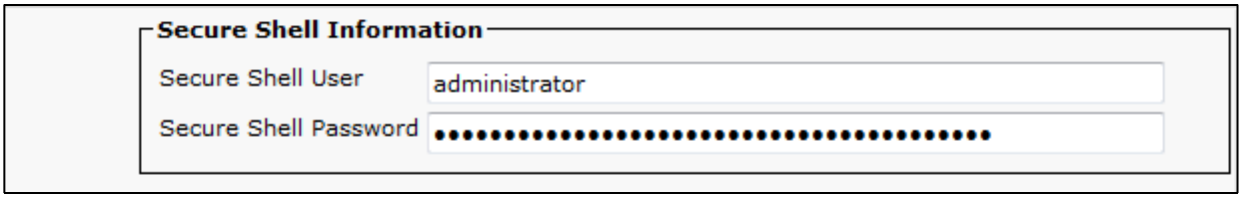
Do Not Disturb

☐ Do Not Disturb

DND Option*: Use Common Phone Profile Setting

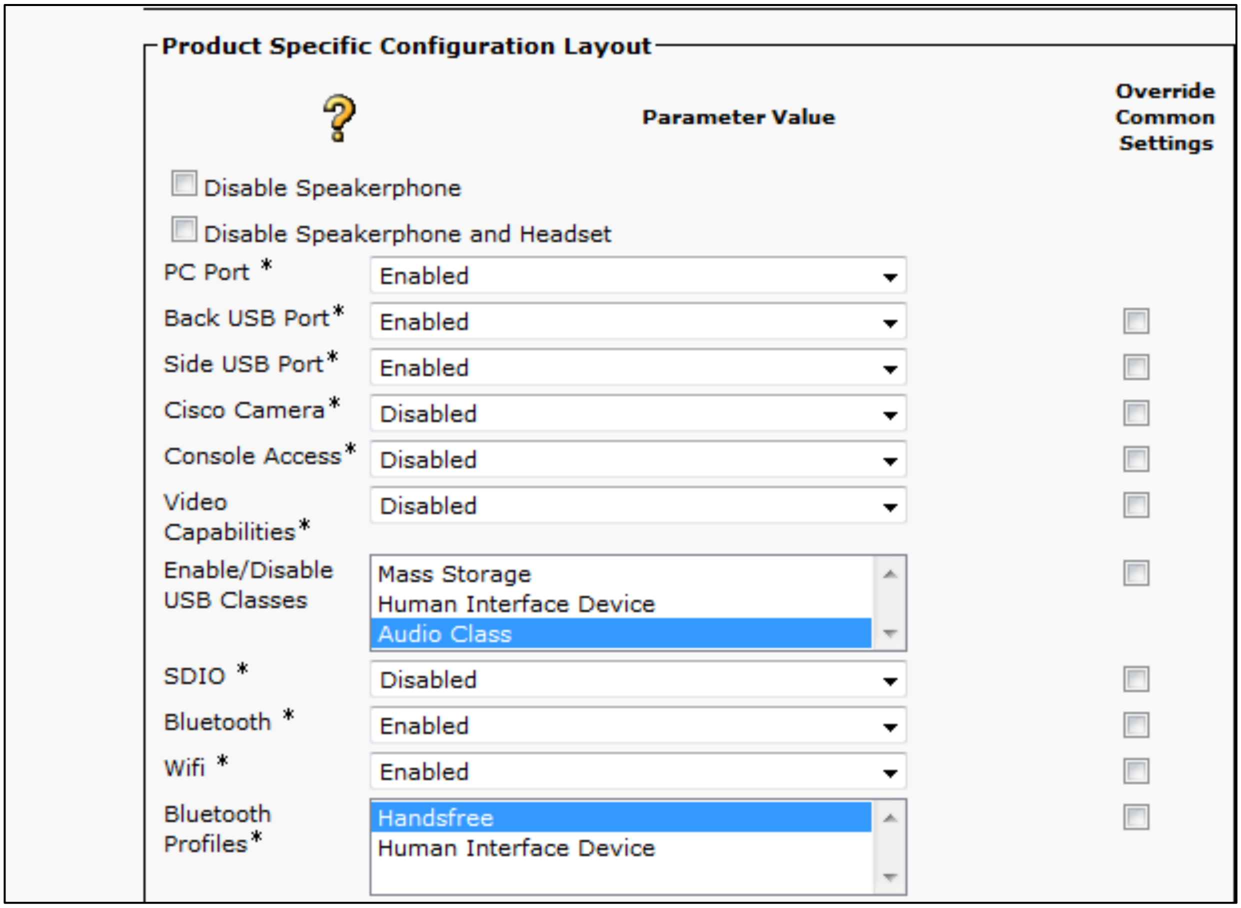
DND Incoming Call Alert: < None >

Cisco IP Phone 7975 SCCP Configuration (Continued…)



Secure Shell Information

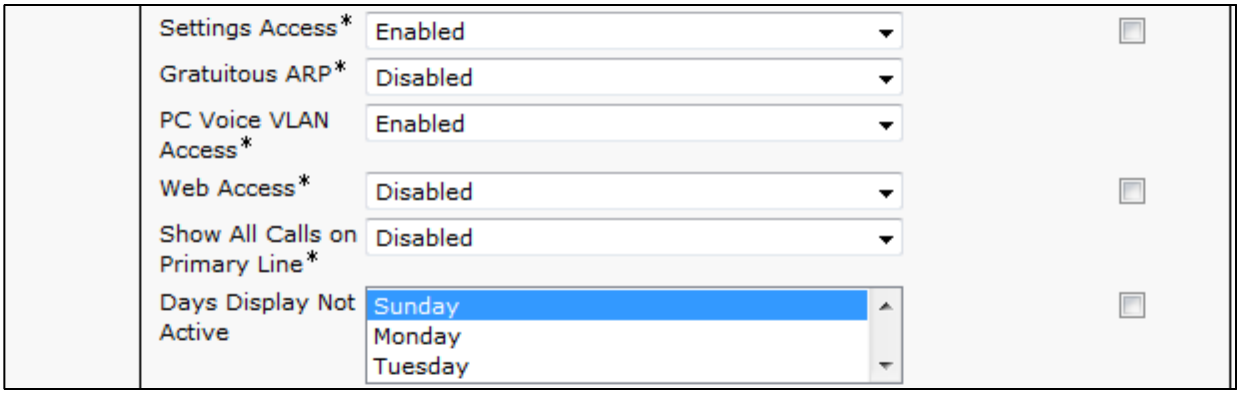
Secure Shell User: administrator
Secure Shell Password: •••



Product Specific Configuration Layout

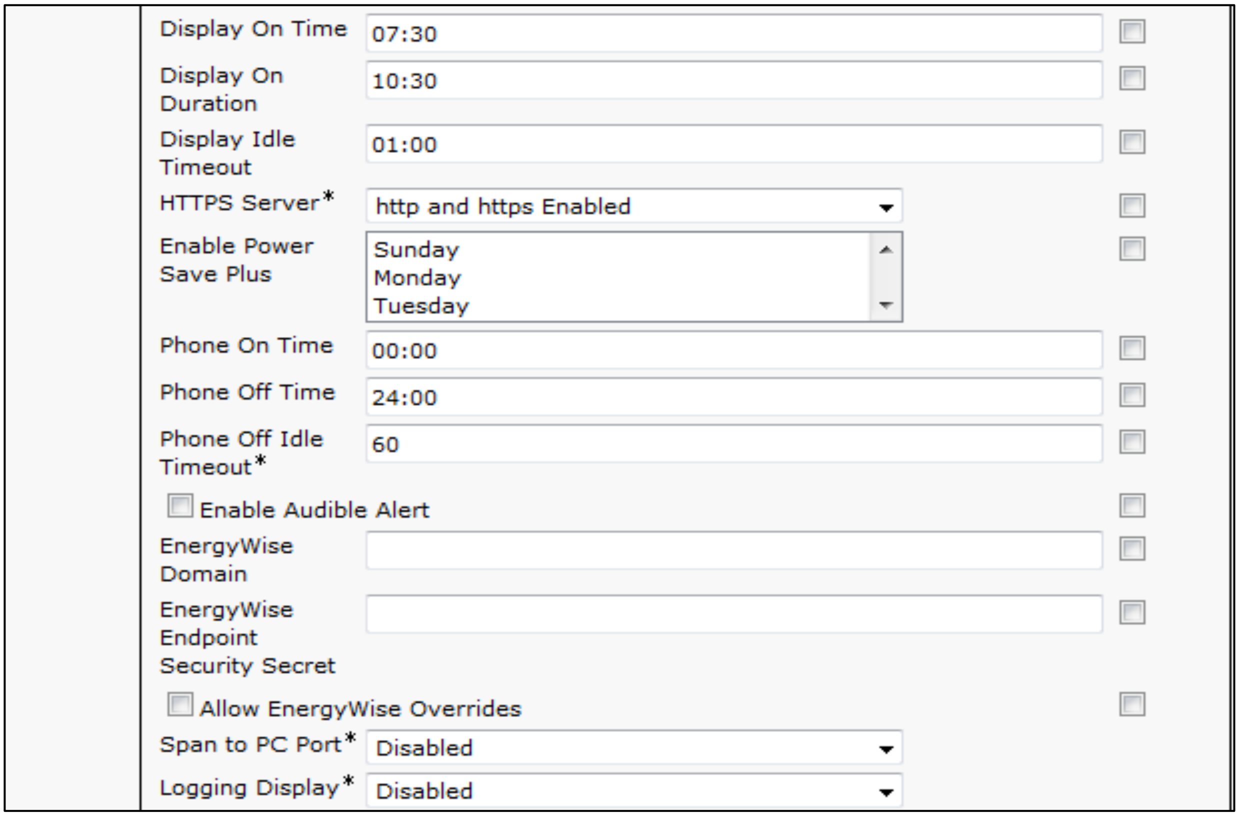
☐ Disable Speakerphone
☐ Disable Speakerphone and Headset

	Parameter Value	Override Common Settings
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	☐

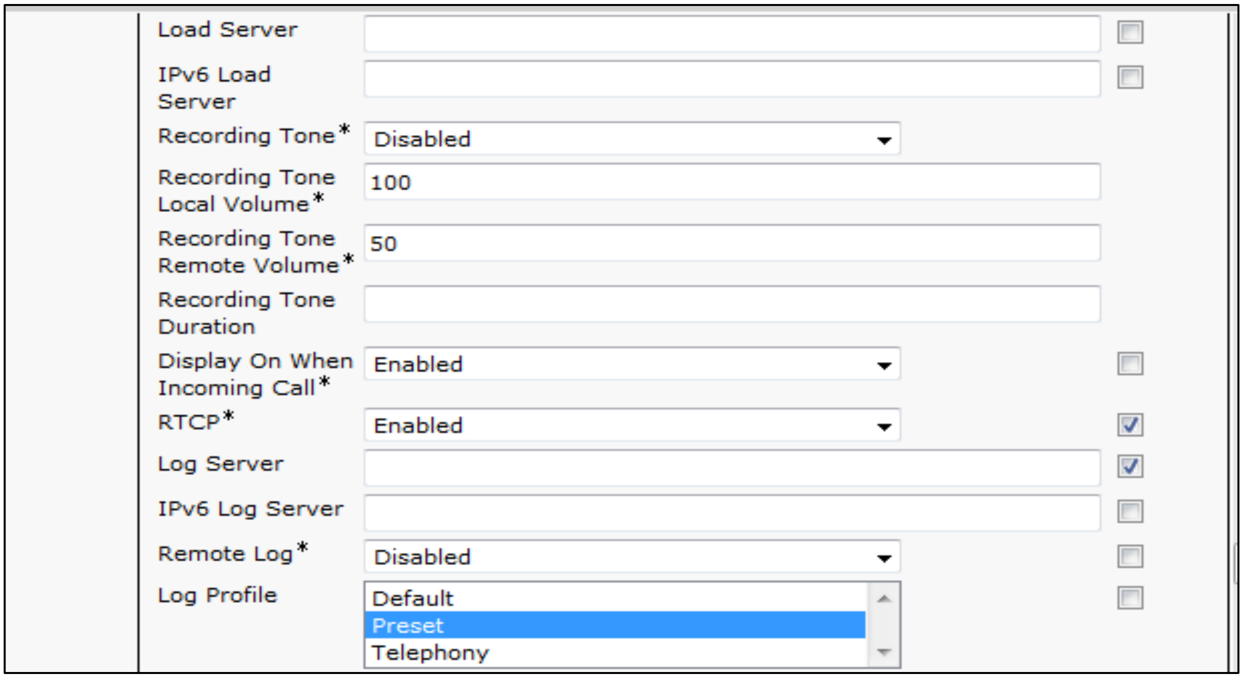


Settings Access*	Enabled	☐
Gratuitous ARP*	Disabled	
PC Voice VLAN Access*	Enabled	
Web Access*	Disabled	☐
Show All Calls on Primary Line*	Disabled	
Days Display Not Active	Sunday Monday Tuesday	☐

Cisco IP Phone 7975 SCCP Configuration (Continued…)

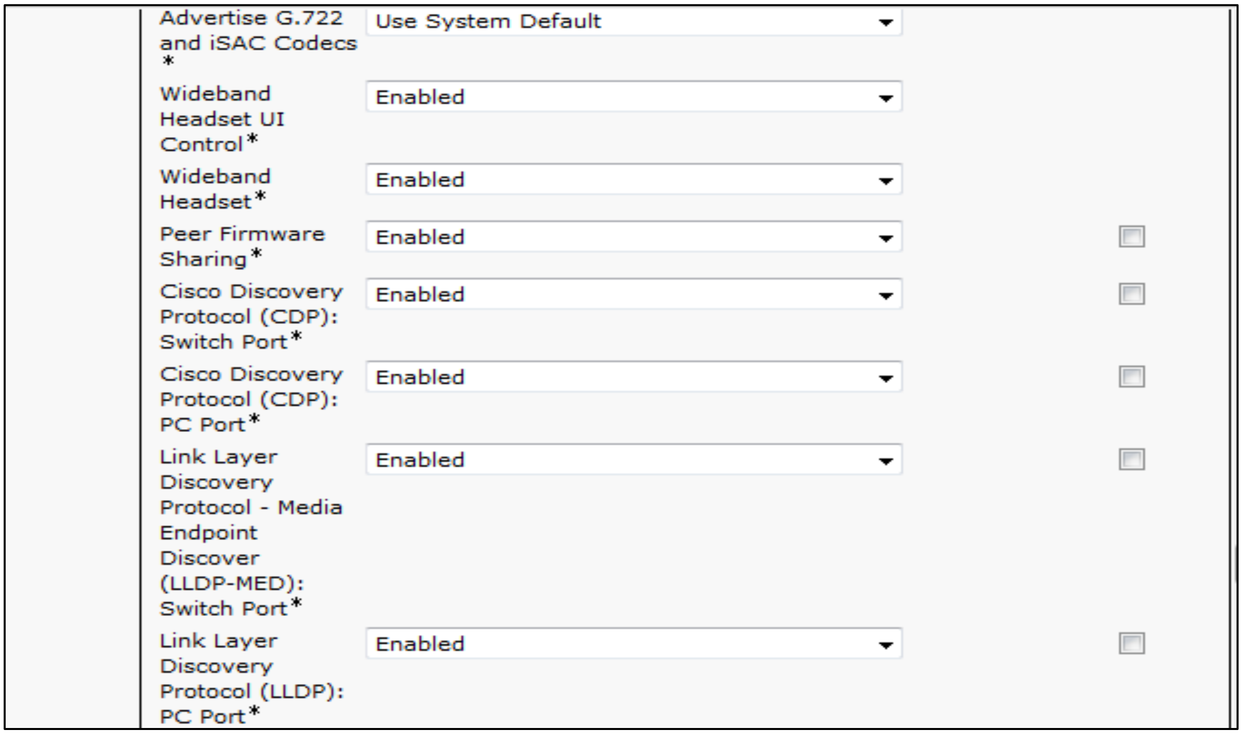


Display On Time	07:30	☐
Display On Duration	10:30	☐
Display Idle Timeout	01:00	☐
HTTPS Server*	http and https Enabled	☐
Enable Power Save Plus	Sunday Monday Tuesday	☐
Phone On Time	00:00	☐
Phone Off Time	24:00	☐
Phone Off Idle Timeout*	60	☐
☐ Enable Audible Alert		☐
EnergyWise Domain		☐
EnergyWise Endpoint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

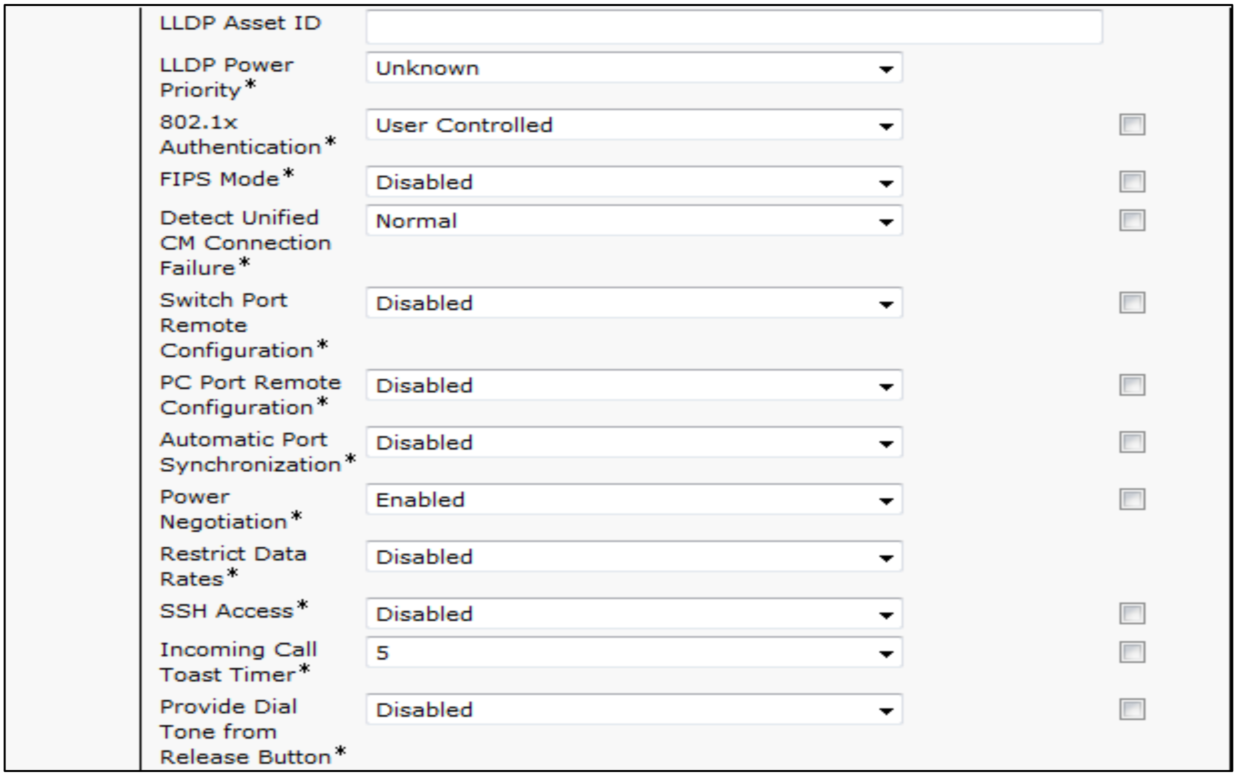


Load Server		☐
IPv6 Load Server		☐
Recording Tone*	Disabled	
Recording Tone Local Volume*	100	
Recording Tone Remote Volume*	50	
Recording Tone Duration		
Display On When Incoming Call*	Enabled	☐
RTCP*	Enabled	☑
Log Server		☑
IPv6 Log Server		☐
Remote Log*	Disabled	☐
Log Profile	Default Preset Telephony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

Cisco IP Phone 7975 SCCP Configuration (Continued…)

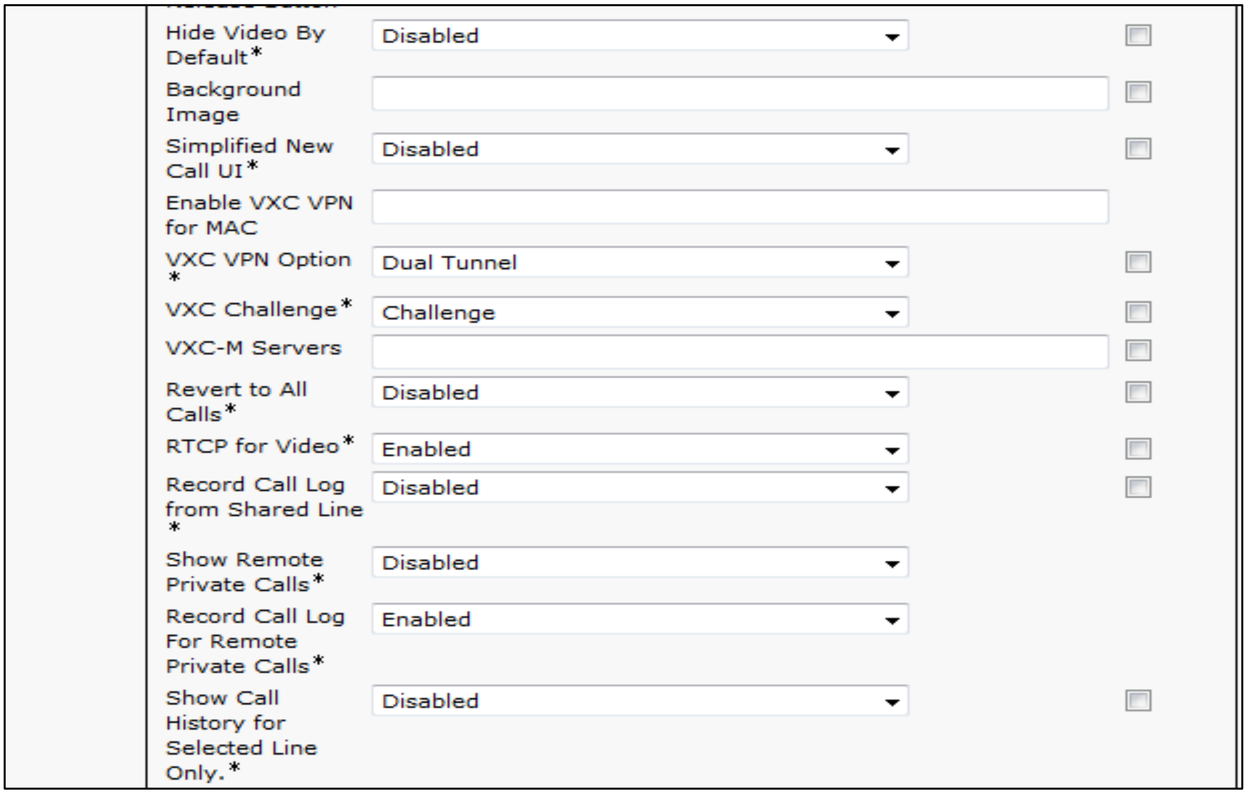


Advertise G.722 and iSAC Codecs *	Use System Default	
Wideband Headset UI Control*	Enabled	
Wideband Headset*	Enabled	
Peer Firmware Sharing*	Enabled	☐
Cisco Discovery Protocol (CDP): Switch Port*	Enabled	☐
Cisco Discovery Protocol (CDP): PC Port*	Enabled	☐
Link Layer Discovery Protocol - Media Endpoint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

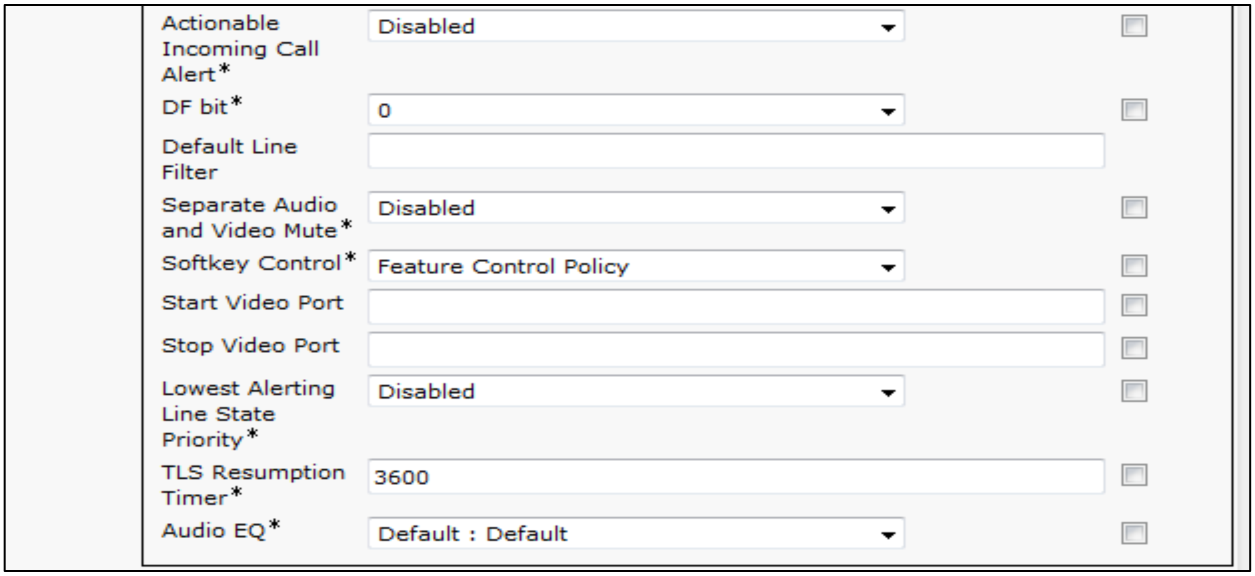


LLDP Asset ID		
LLDP Power Priority*	Unknown	
802.1x Authentication*	User Controlled	☐
FIPS Mode*	Disabled	☐
Detect Unified CM Connection Failure*	Normal	☐
Switch Port Remote Configuration*	Disabled	☐
PC Port Remote Configuration*	Disabled	☐
Automatic Port Synchronization*	Disabled	☐
Power Negotiation*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

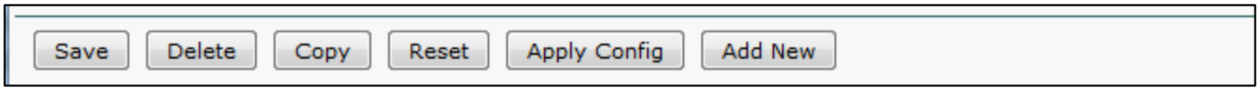
Cisco IP Phone 7975 SCCP Configuration (Continued…)



Hide Video By Default*	Disabled	☐
Background Image		☐
Simplified New Call UI*	Disabled	☐
Enable VXC VPN for MAC		
VXC VPN Option *	Dual Tunnel	☐
VXC Challenge*	Challenge	☐
VXC-M Servers		☐
Revert to All Calls*	Disabled	☐
RTCP for Video*	Enabled	☐
Record Call Log from Shared Line *	Disabled	☐
Show Remote Private Calls*	Disabled	
Record Call Log For Remote Private Calls*	Enabled	
Show Call History for Selected Line Only.*	Disabled	☐

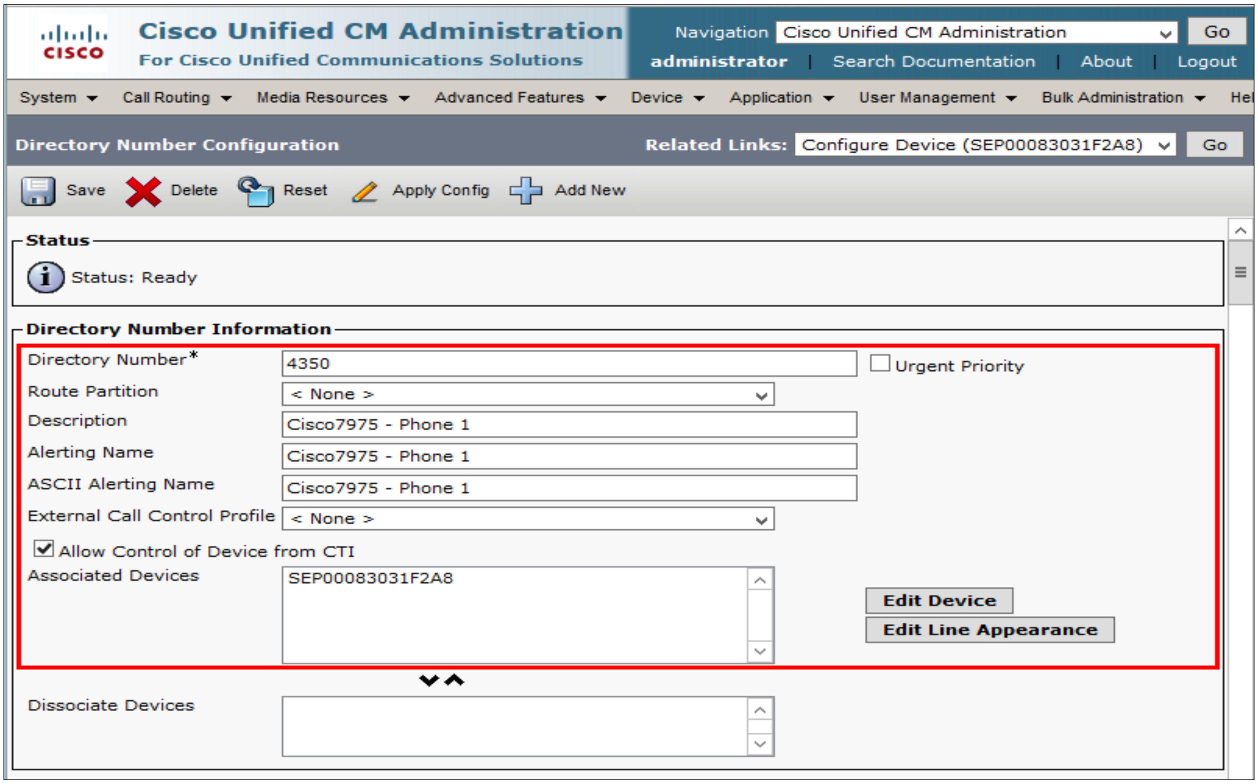


Actionable Incoming Call Alert*	Disabled	☐
DF bit*	0	☐
Default Line Filter		
Separate Audio and Video Mute*	Disabled	☐
Softkey Control*	Feature Control Policy	☐
Start Video Port		☐
Stop Video Port		☐
Lowest Alerting Line State Priority*	Disabled	☐
TLS Resumption Timer*	3600	☐
Audio EQ*	Default : Default	☐



Cisco IP Phone 7975 SCCP Configuration (Continued…)

Set Directory Number* = 4350. This is used in this example.
Set Description = 7323204350. This is used in this example.
Set Alerting Name = Cisco7975 Phone. This is used in this example.
Set ASCII Alerting Name = Cisco7975 Phone. This is used in this example.

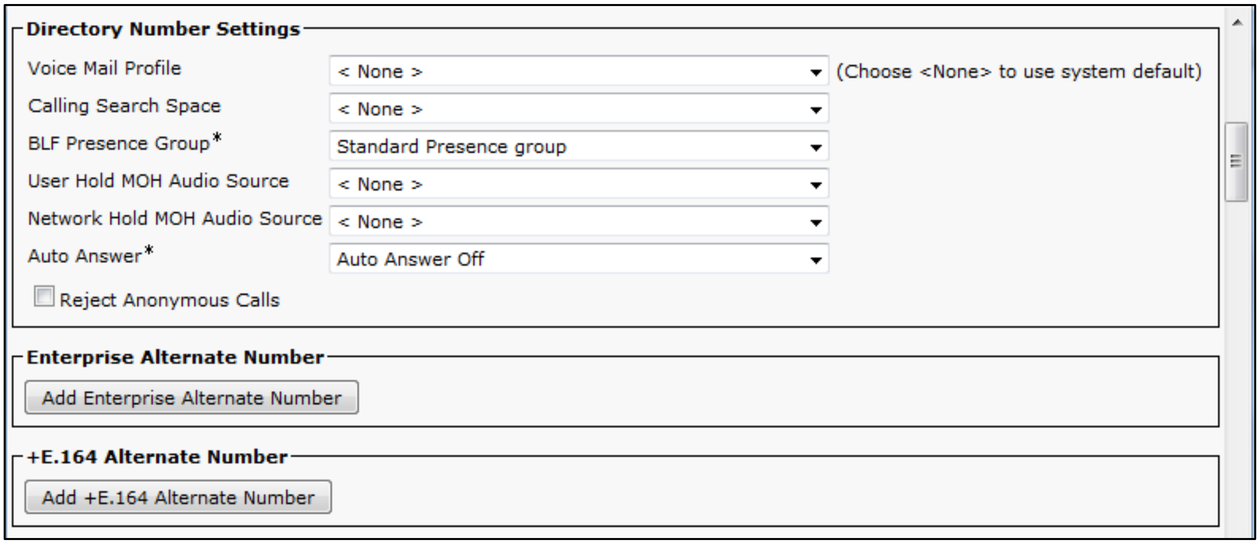


Status

Status: Ready

Directory Number Information

Directory Number* 4350
☐ Urgent Priority
Route Partition < None >
Description Cisco7975 - Phone 1
Alerting Name Cisco7975 - Phone 1
ASCII Alerting Name Cisco7975 - Phone 1
External Call Control Profile < None >
☑ Allow Control of Device from CTI
Associated Devices SEP00083031F2A8
Dissociate Devices



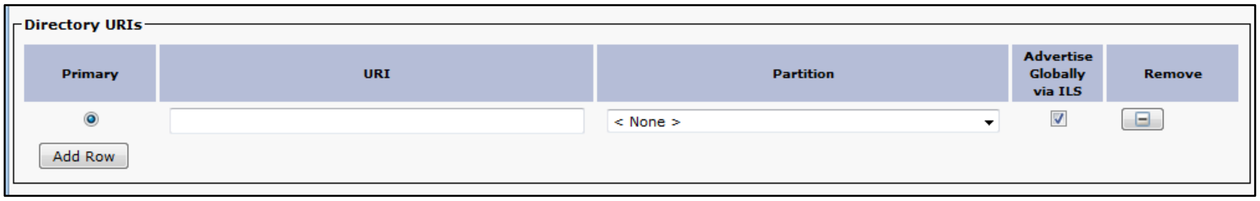
Directory Number Settings

Voice Mail Profile < None > (Choose <None> to use system default)
Calling Search Space < None >
BLF Presence Group* Standard Presence group
User Hold MOH Audio Source < None >
Network Hold MOH Audio Source < None >
Auto Answer* Auto Answer Off
☐ Reject Anonymous Calls

Enterprise Alternate Number

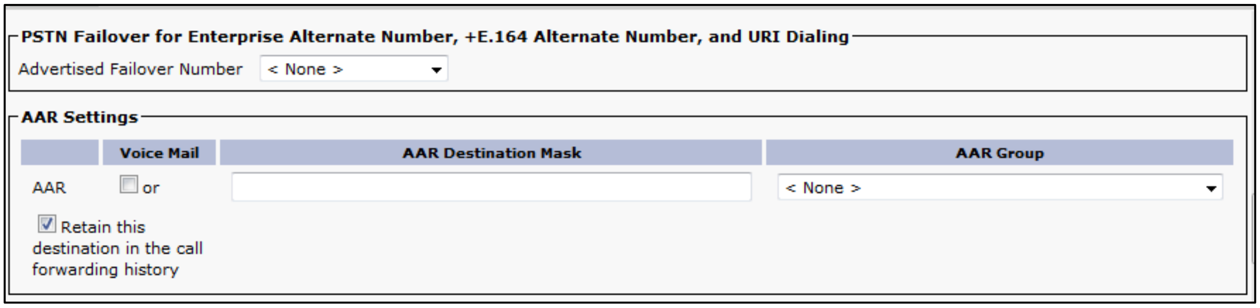
+E.164 Alternate Number

Cisco IP Phone 7975 SCCP Configuration (Continued…)



Directory URIs

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



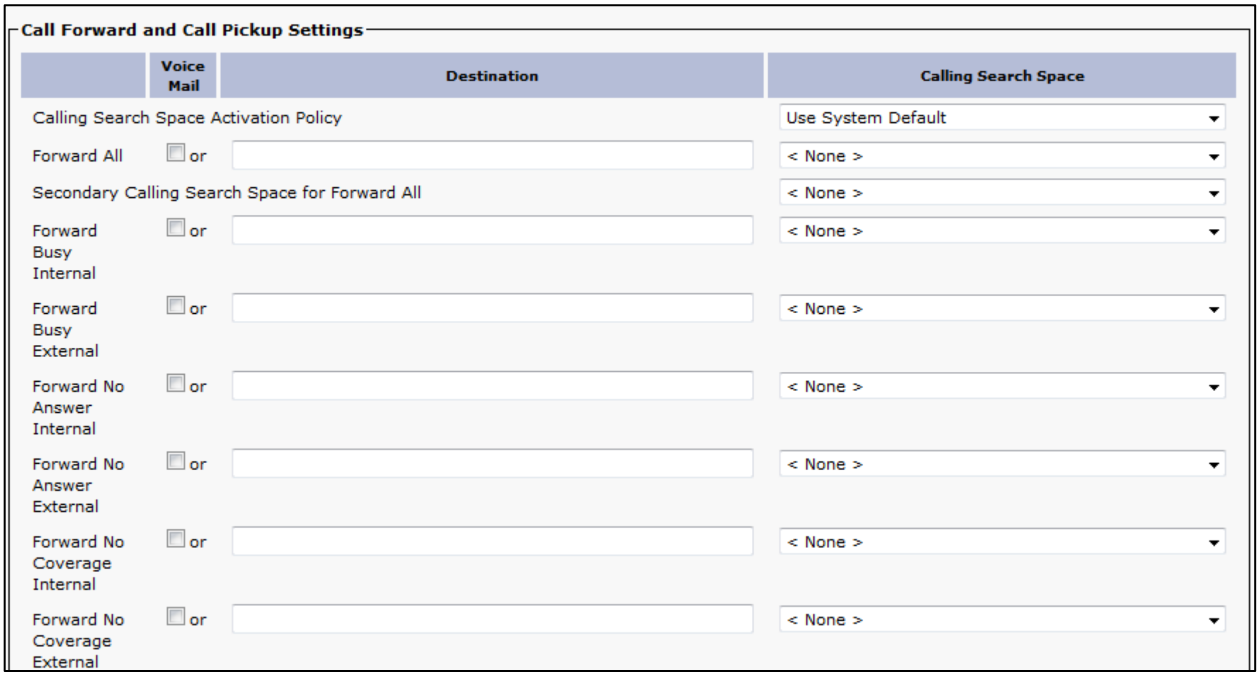
PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing

Advertised Failover Number: < None >

AAR Settings

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

☑ Retain this destination in the call forwarding history



Call Forward and Call Pickup Settings

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

Cisco IP Phone 7975 SCCP Configuration (Continued…)

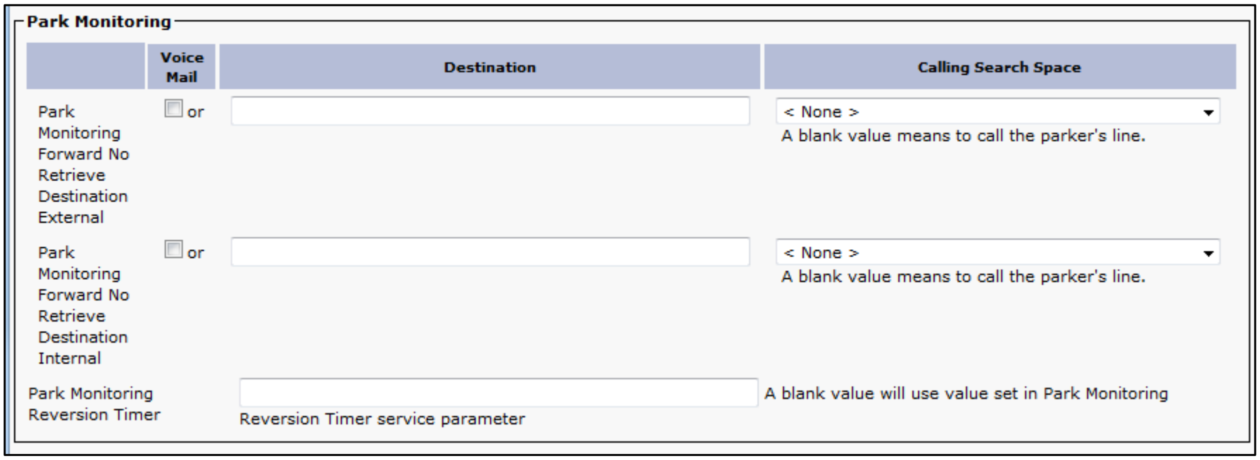
Forward on CTI Failure ☐ or < None >

Forward Unregistered Internal ☐ or < None >

Forward Unregistered External ☐ or < None >

No Answer Ring Duration (seconds)

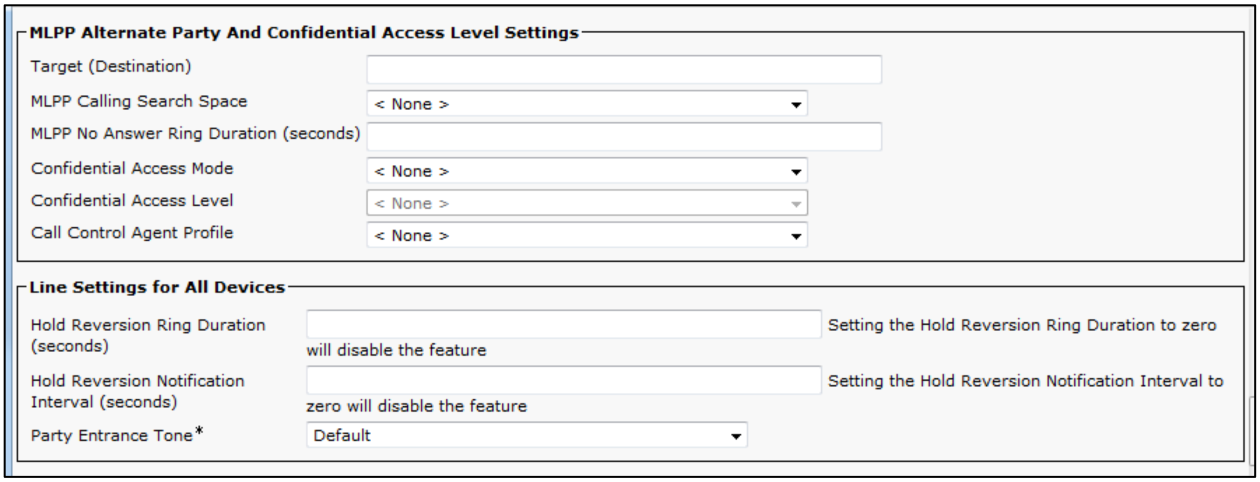
Call Pickup Group < None >



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

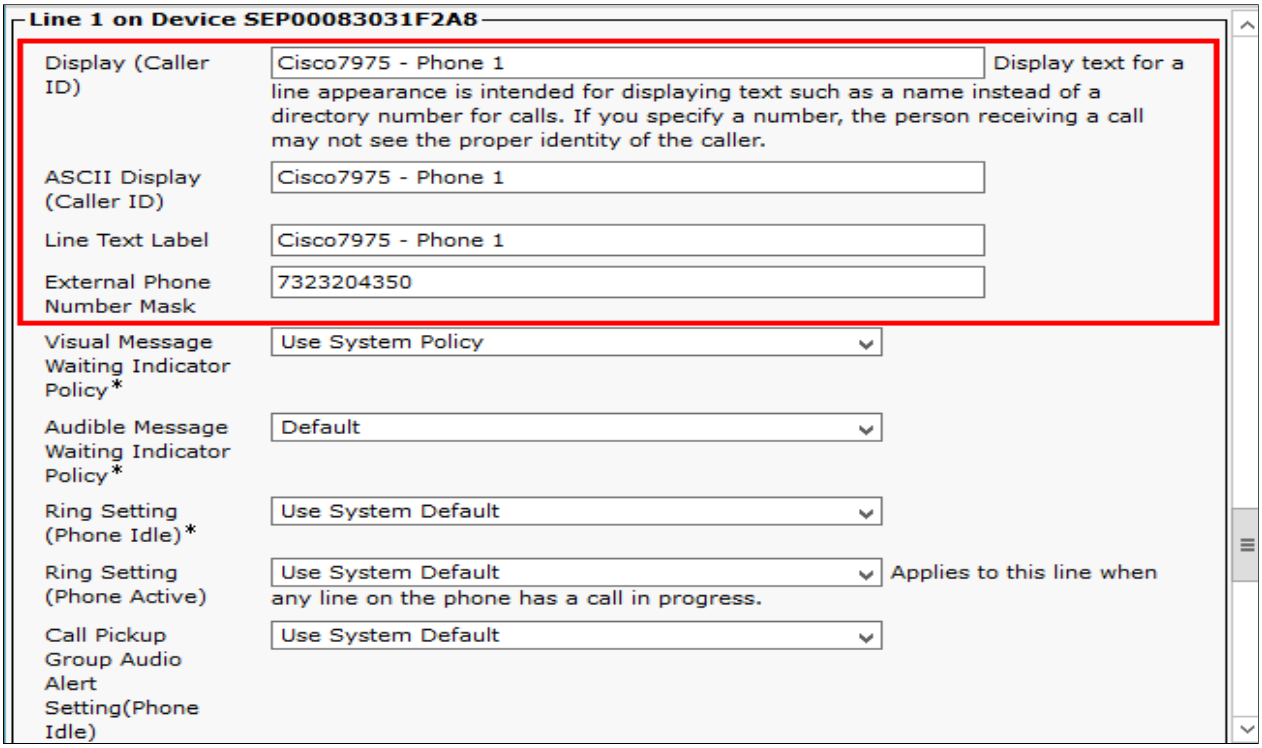
Hold Reversion Ring Duration (seconds) Setting the Hold Reversion Ring Duration to zero will disable the feature

Hold Reversion Notification Interval (seconds) Setting the Hold Reversion Notification Interval to zero will disable the feature

Party Entrance Tone* Default

Cisco IP Phone 7975 SCCP Configuration (Continued…)

Set Display (caller ID) = Cisco7975-Phone 1. This is used in this example.
Set ASCII Display (caller ID) = Cisco7975-Phone 1. This is used in this example.
Set Line Text Label = Cisco7975-Phone 1. This is used in this example.
Set External Phone Number Mask = 7323204350. This is used in this example.



Line 1 on Device SEP00083031F2A8

Display (Caller ID): Cisco7975 - Phone 1 — 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): Cisco7975 - Phone 1

Line Text Label: Cisco7975 - Phone 1

External Phone Number Mask: 7323204350

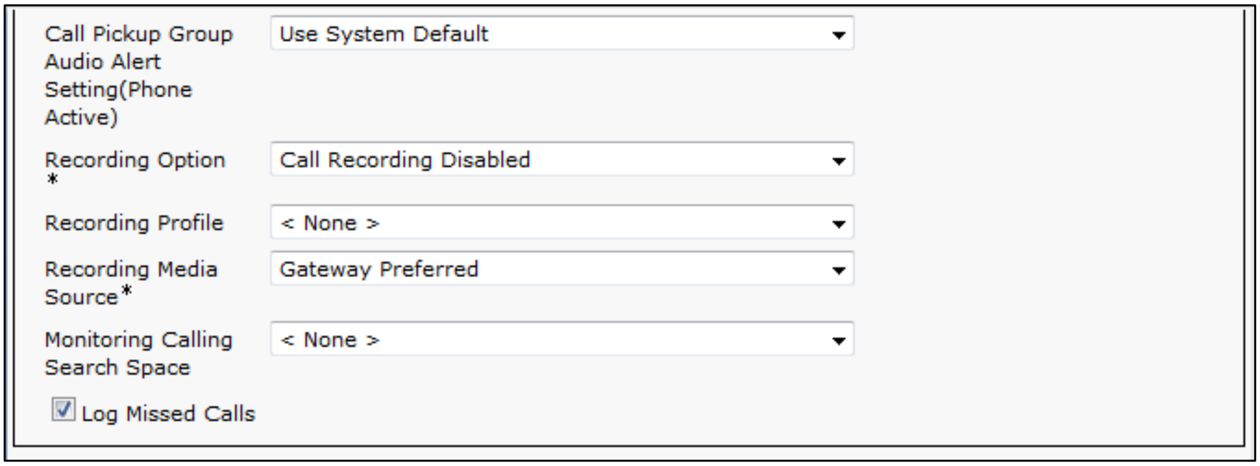
Visual Message Waiting Indicator Policy*: Use System Policy

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

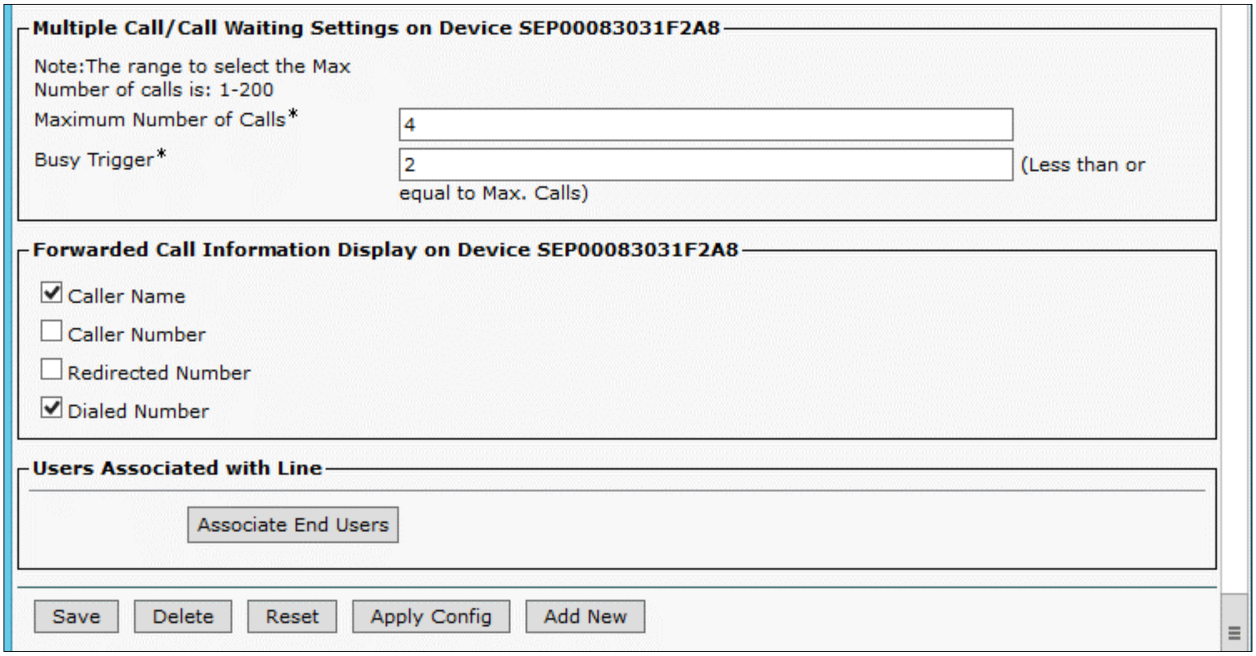
Recording Profile: < None >

Recording Media Source*: Gateway Preferred

Monitoring Calling Search Space: < None >

☑ Log Missed Calls

Cisco IP Phone 7975 SCCP Configuration (Continued…)



Multiple Call/Call Waiting Settings on Device SEP00083031F2A8

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

Maximum Number of Calls* 4

Busy Trigger* 2 (Less than or equal to Max. Calls)

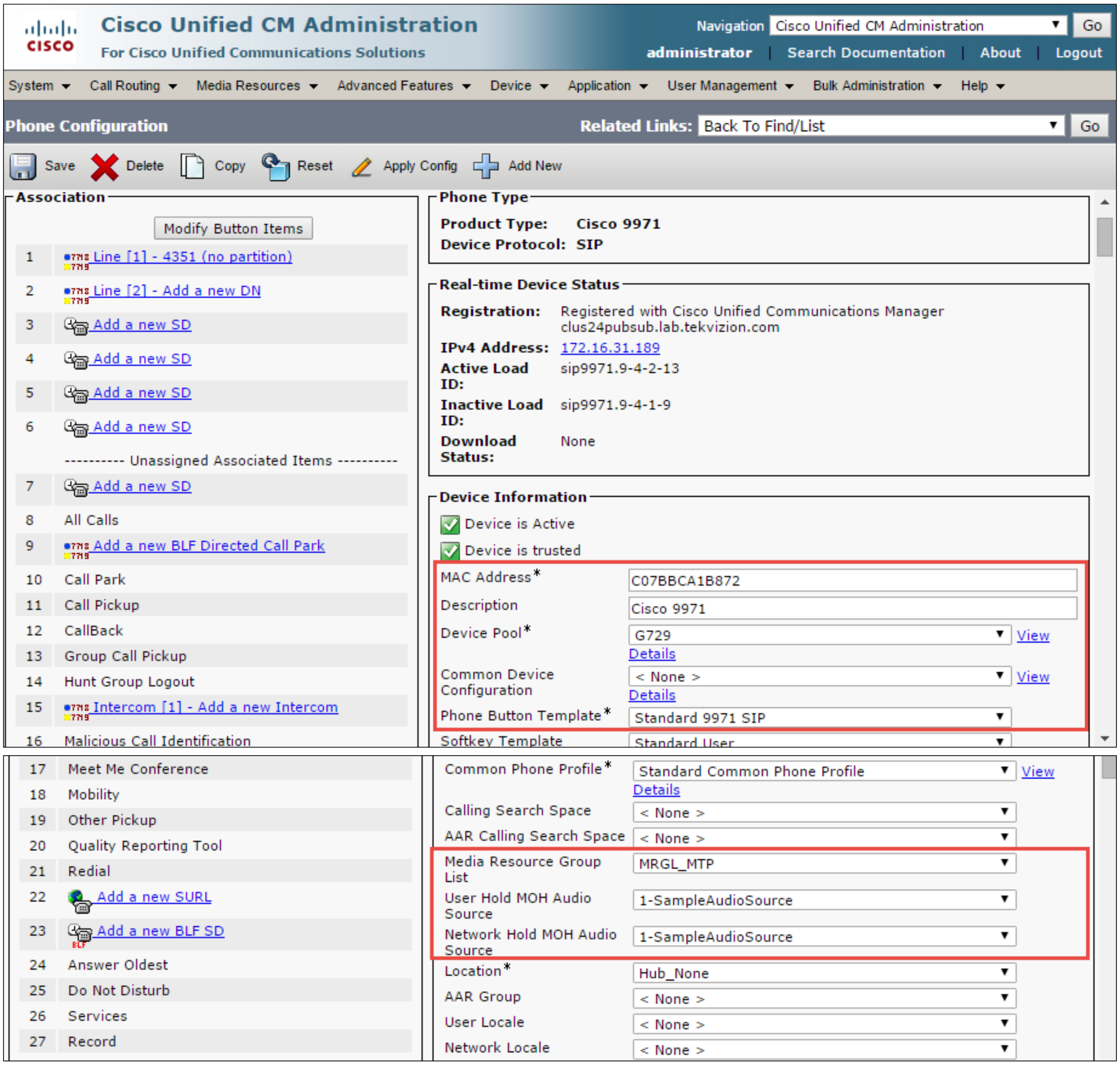
Forwarded Call Information Display on Device SEP00083031F2A8

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

Users Associated with Line

Cisco IP Phone 9971 SIP Configuration

Set MAC Address* = the below mac is used in this example.
Set Description = Cisco 9971 Phone 2. 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



Association

1	Line [1] - 4351 (no partition)
2	Line [2] - Add a new DN
3	Add a new SD
4	Add a new SD
5	Add a new SD
6	Add a new SD
	---------- Unassigned Associated Items ----------
7	Add a new SD
8	All Calls
9	Add a new BLF Directed Call Park
10	Call Park
11	Call Pickup
12	CallBack
13	Group Call Pickup
14	Hunt Group Logout
15	Intercom [1] - Add a new Intercom
16	Malicious Call Identification
17	Meet Me Conference
18	Mobility
19	Other Pickup
20	Quality Reporting Tool
21	Redial
22	Add a new SURL
23	Add a new BLF SD
24	Answer Oldest
25	Do Not Disturb
26	Services
27	Record

Phone Type

Product Type: Cisco 9971
Device Protocol: SIP

Real-time Device Status

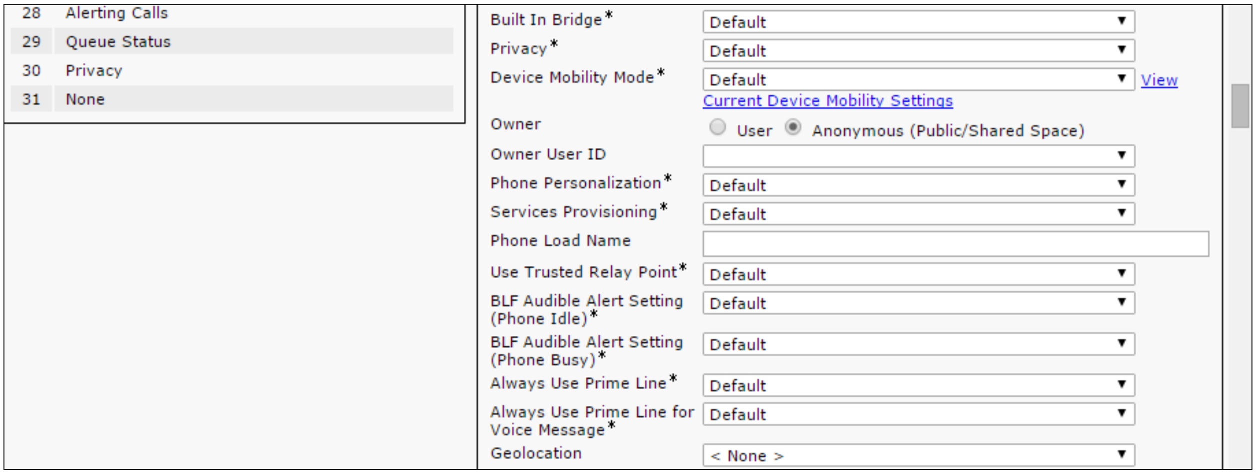
Registration: Registered with Cisco Unified Communications Manager clus24pubsub.lab.tekvizion.com
IPv4 Address: 172.16.31.189
Active Load ID: sip9971.9-4-2-13
Inactive Load ID: sip9971.9-4-1-9
Download Status: None

Device Information

☑ Device is Active
☑ Device is trusted

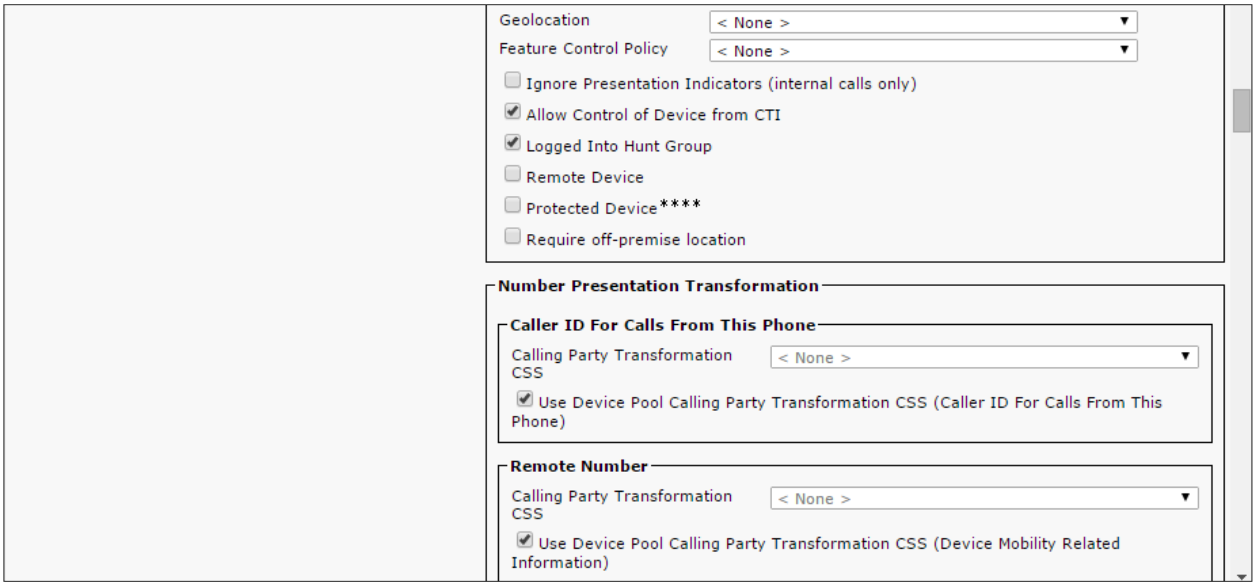
MAC Address*	C07BBCA1B872
Description	Cisco 9971
Device Pool*	G729
Common Device Configuration	< None >
Phone Button Template*	Standard 9971 SIP
Softkey Template	Standard User
Common Phone Profile*	Standard Common Phone Profile
Calling Search Space	< None >
AAR Calling Search Space	< None >
Media Resource Group List	MRGL_MTP
User Hold MOH Audio Source	1-SampleAudioSource
Network Hold MOH Audio Source	1-SampleAudioSource
Location*	Hub_None
AAR Group	< None >
User Locale	< None >
Network Locale	< None >

Cisco IP Phone 9971 SIP Configuration(Continued…)



28	Alerting Calls
29	Queue Status
30	Privacy
31	None

Built In Bridge* Default
Privacy* Default
Device Mobility Mode* Default View Current Device Mobility Settings
Owner ○ User ● Anonymous (Public/Shared Space)
Owner User ID
Phone Personalization* Default
Services Provisioning* Default
Phone Load Name
Use Trusted Relay Point* Default
BLF Audible Alert Setting (Phone Idle)* Default
BLF Audible Alert Setting (Phone Busy)* Default
Always Use Prime Line* Default
Always Use Prime Line for Voice Message* Default
Geolocation < None >



Geolocation < None >
Feature Control Policy < None >
☐ Ignore Presentation Indicators (internal calls only)
☑ Allow Control of Device from CTI
☑ Logged Into Hunt Group
☐ Remote Device
☐ Protected Device****
☐ Require off-premise location

Number Presentation Transformation

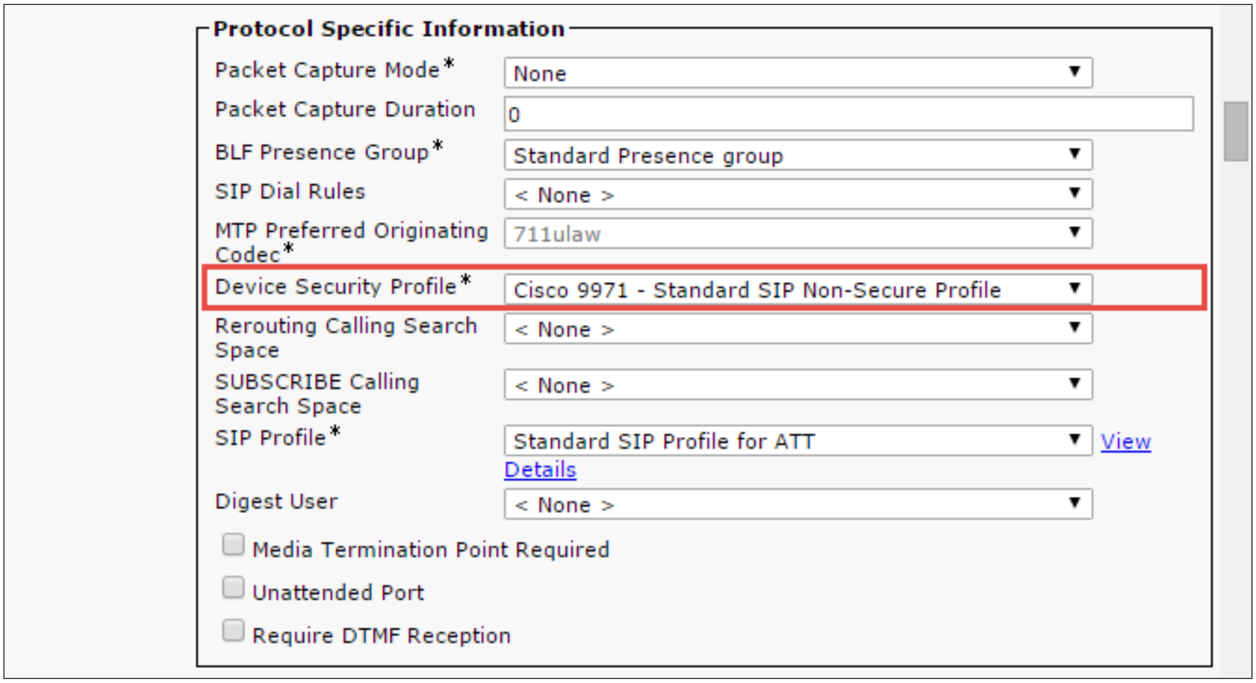
Caller ID For Calls From This Phone

Calling Party Transformation CSS < None >
☑ Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)

Remote Number

Calling Party Transformation CSS < None >
☑ Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

Cisco IP Phone 9971 SIP Configuration(Continued…)



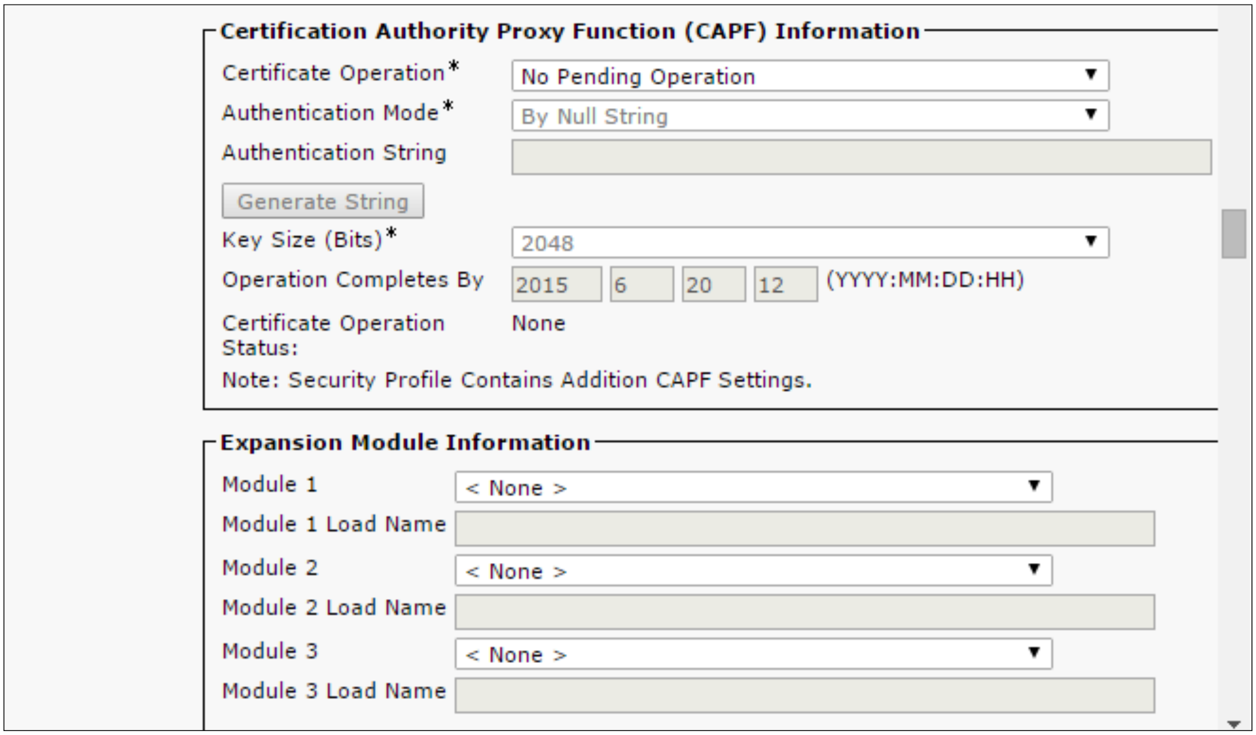
Protocol Specific Information

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

☐ Media Termination Point Required

☐ Unattended Port

☐ Require DTMF Reception



Certification Authority Proxy Function (CAPF) Information

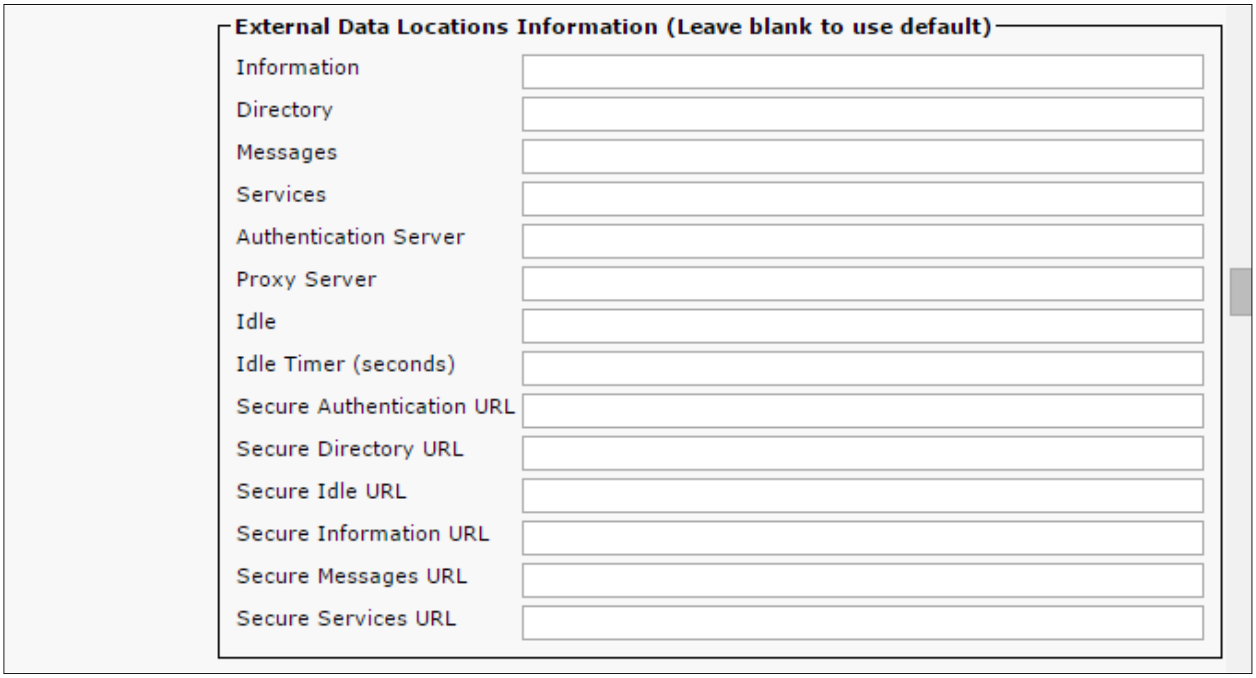
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
Generate String	
Key Size (Bits)*	2048
Operation Completes By	2015 6 20 12 (YYYY:MM:DD:HH)
Certificate Operation Status:	None

Note: Security Profile Contains Addition CAPF Settings.

Expansion Module Information

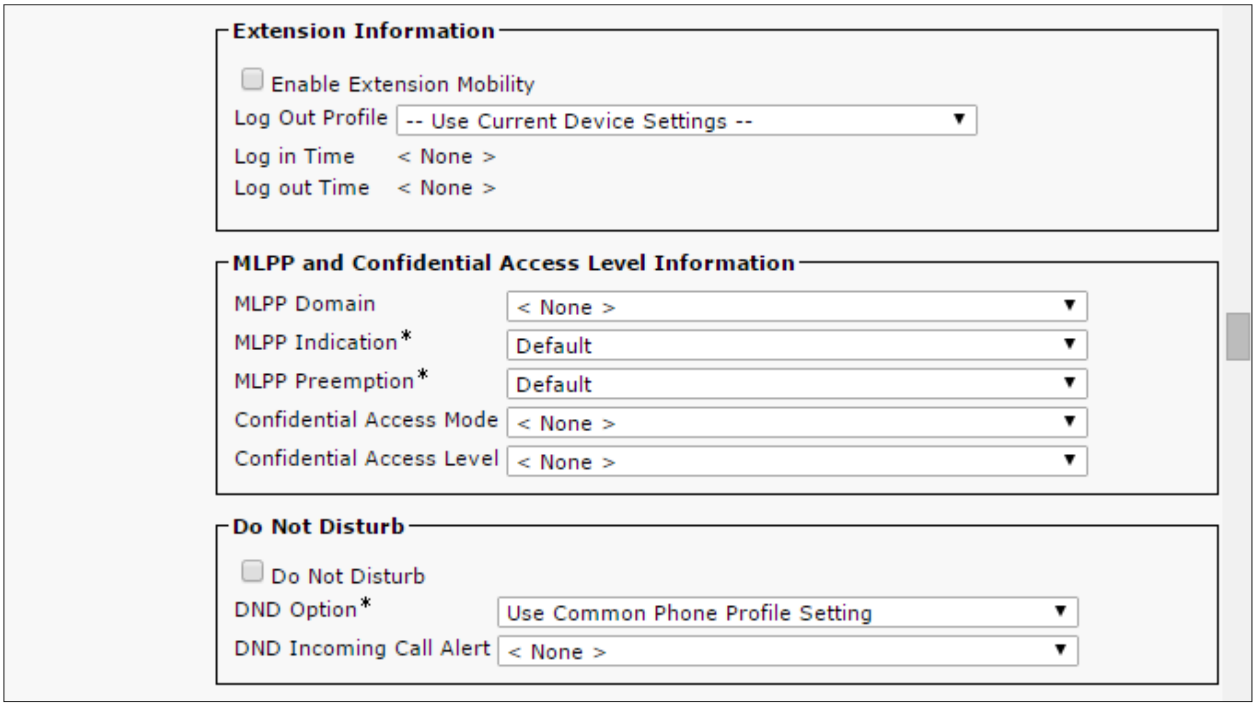
Module 1	< None >
Module 1 Load Name	
Module 2	< None >
Module 2 Load Name	
Module 3	< None >
Module 3 Load Name	

Cisco IP Phone 9971 SIP Configuration(Continued…)



External Data Locations Information (Leave blank to use default)

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



Extension Information

☐ Enable Extension Mobility
Log Out Profile: -- Use Current Device Settings --
Log in Time: < None >
Log out Time: < None >

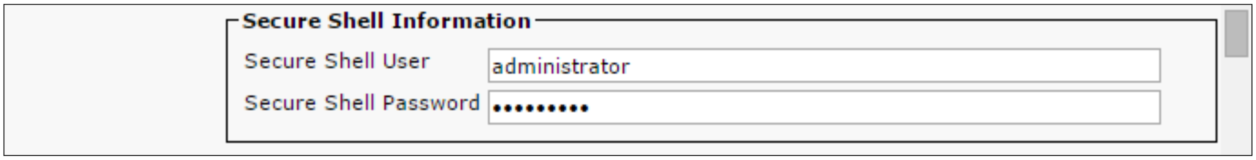
MLPP and Confidential Access Level Information

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

Do Not Disturb

☐ Do Not Disturb
DND Option*: Use Common Phone Profile Setting
DND Incoming Call Alert: < None >

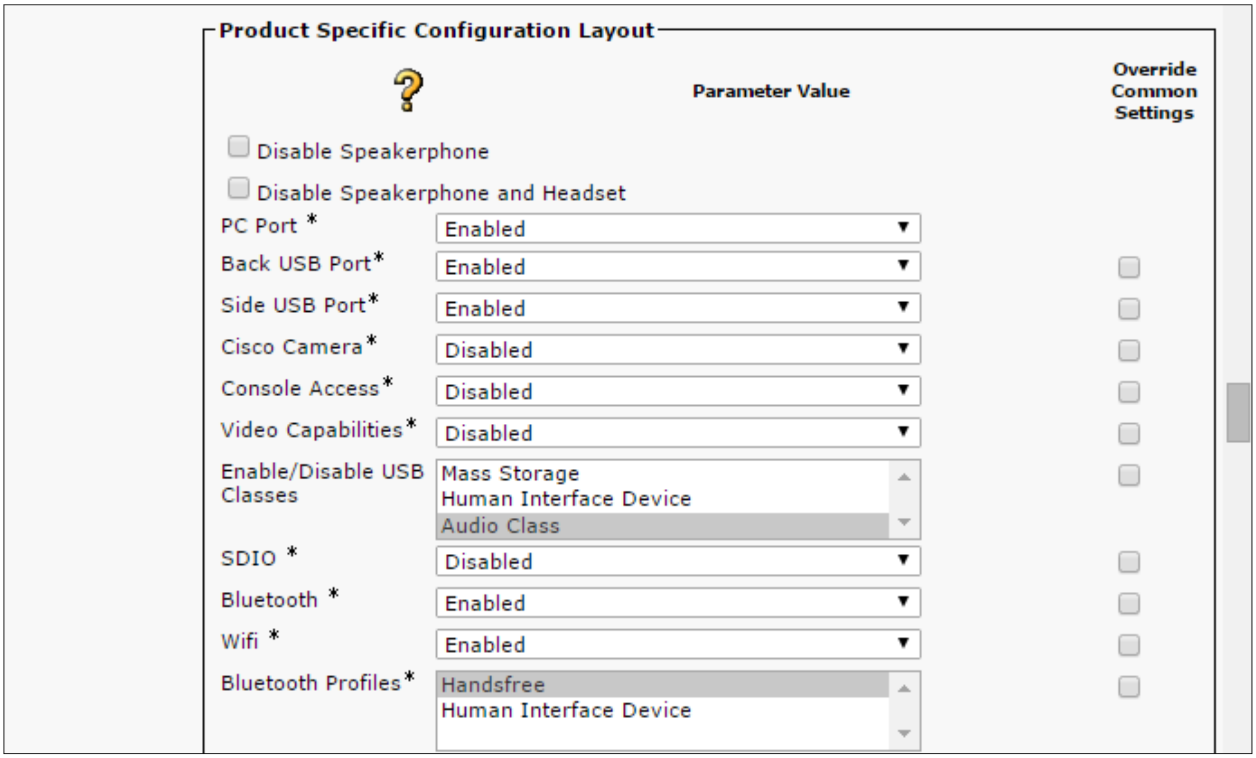
Cisco IP Phone 9971 SIP Configuration(Continued…)



Secure Shell Information

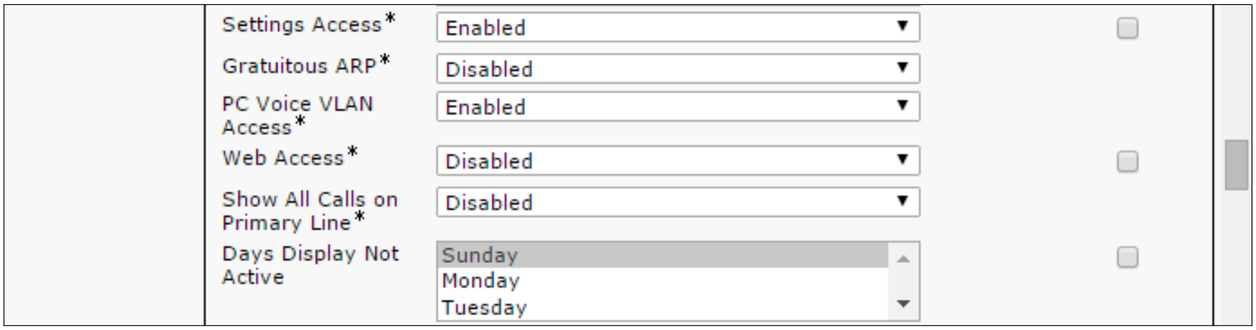
Secure Shell User: administrator

Secure Shell Password: ●●●●●●●●●



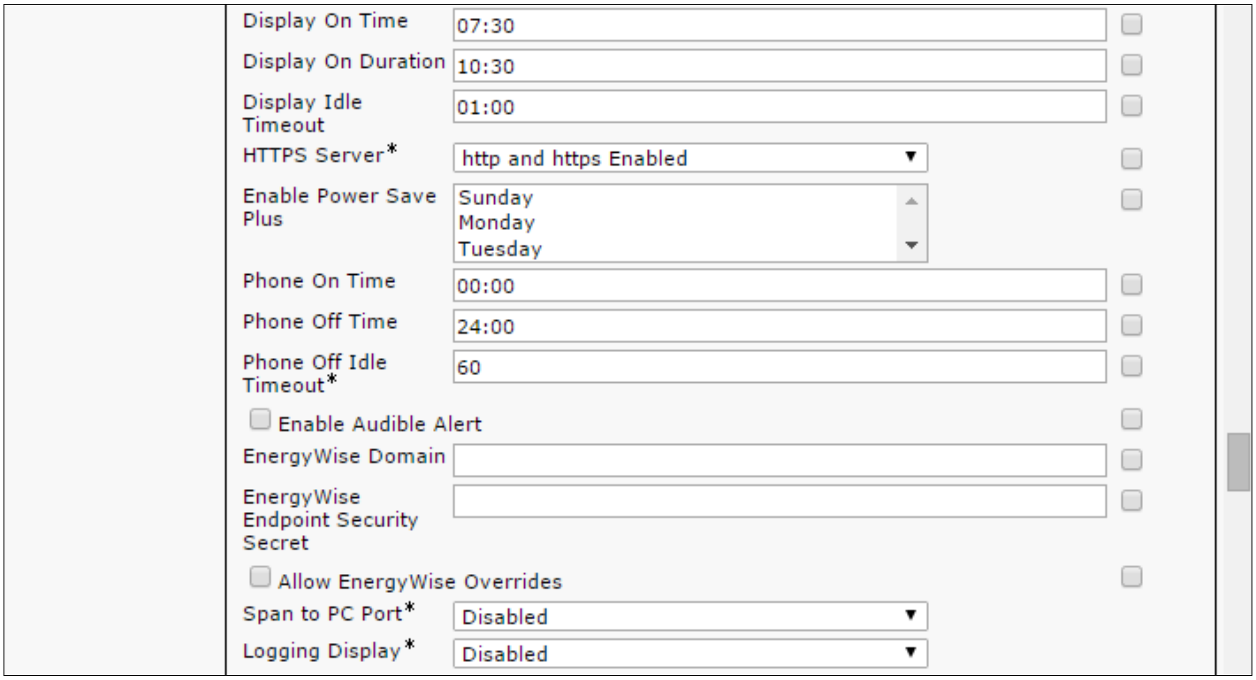
Product Specific Configuration Layout

	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	☐

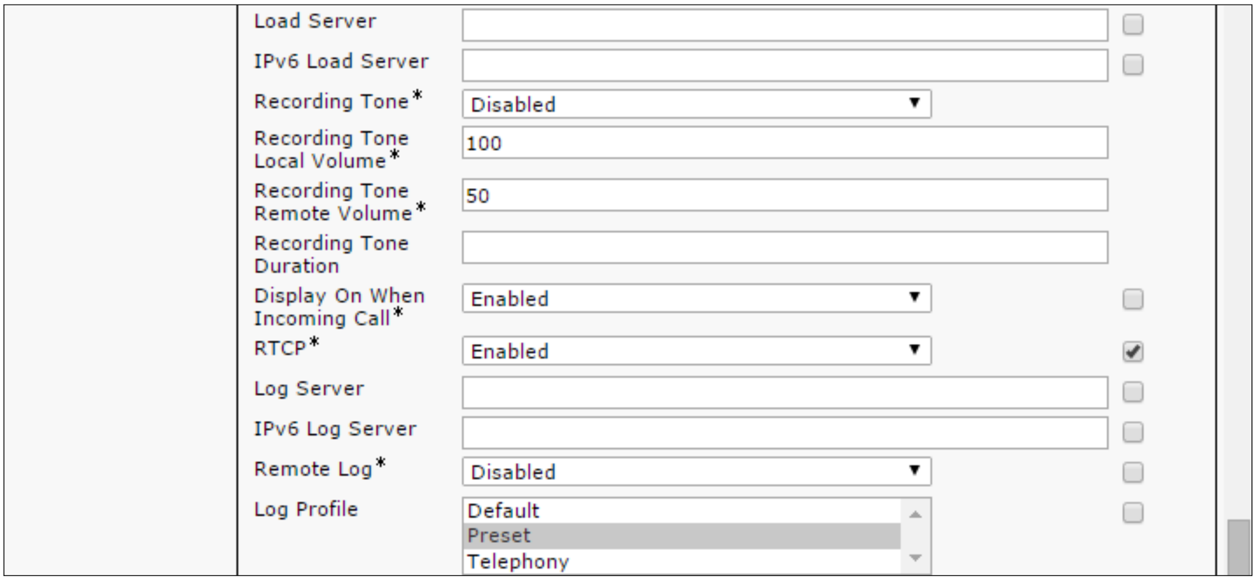


Settings Access*	Enabled	☐
Gratuitous ARP*	Disabled	
PC Voice VLAN Access*	Enabled	
Web Access*	Disabled	☐
Show All Calls on Primary Line*	Disabled	
Days Display Not Active	Sunday Monday Tuesday	☐

Cisco IP Phone 9971 SIP Configuration(Continued…)

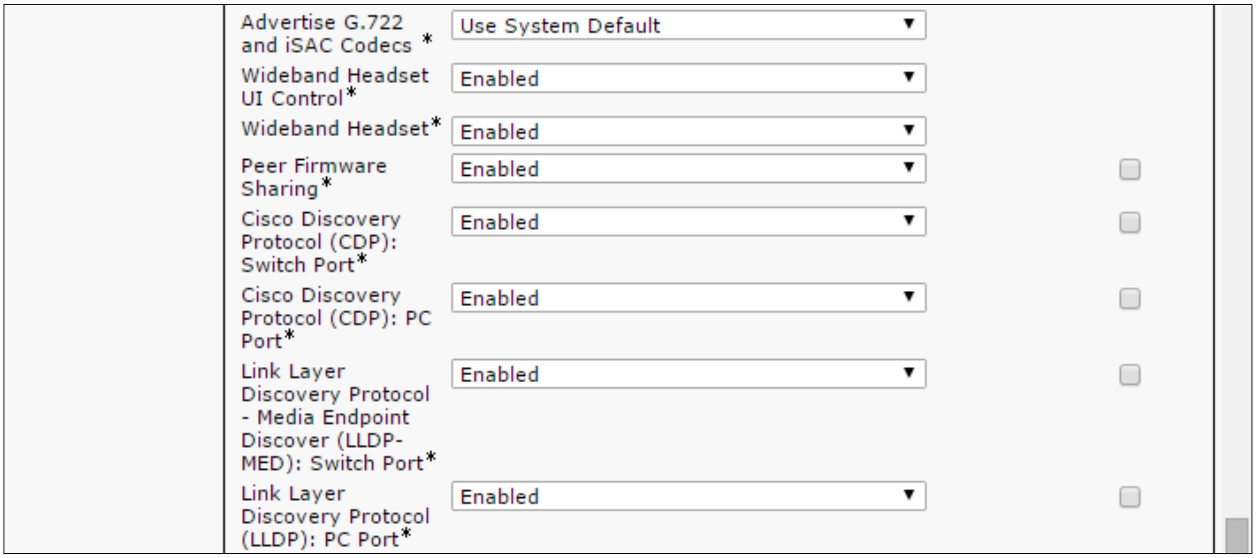


Display On Time	07:30	☐
Display On Duration	10:30	☐
Display Idle Timeout	01:00	☐
HTTPS Server*	http and https Enabled ▼	☐
Enable Power Save Plus	Sunday Monday Tuesday	☐
Phone On Time	00:00	☐
Phone Off Time	24:00	☐
Phone Off Idle Timeout*	60	☐
☐ Enable Audible Alert		☐
EnergyWise Domain		☐
EnergyWise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

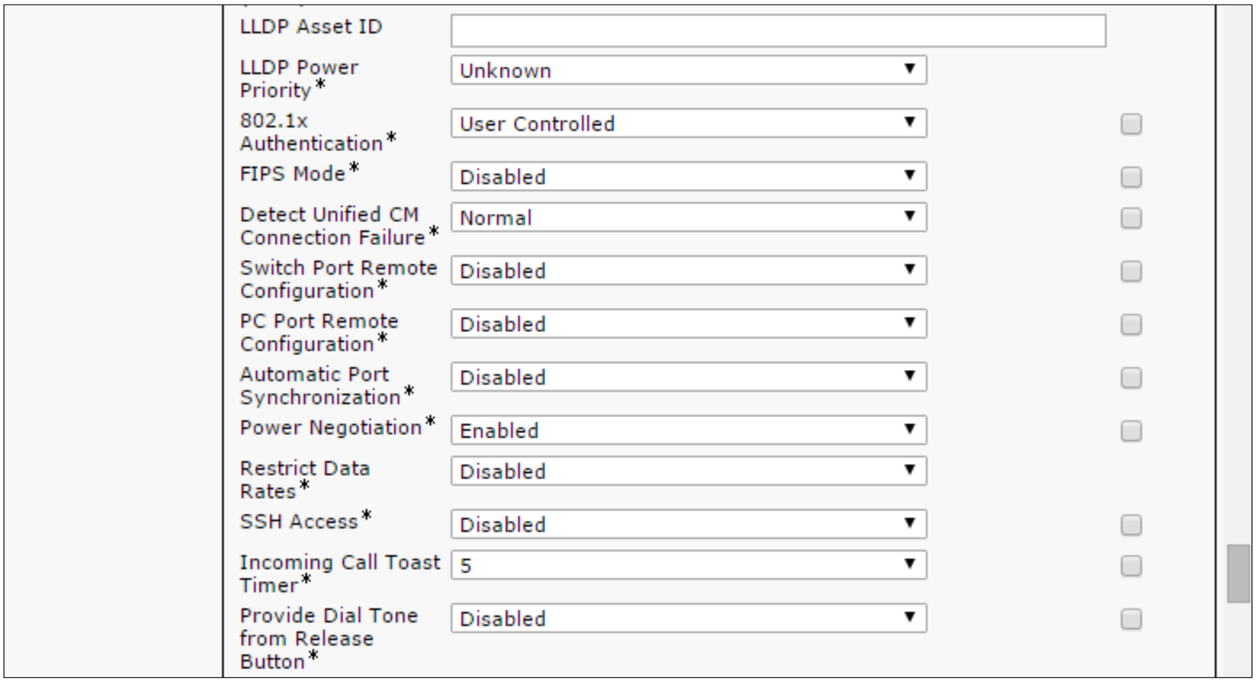


Load Server		☐
IPv6 Load Server		☐
Recording Tone*	Disabled ▼	
Recording Tone Local Volume*	100	
Recording Tone Remote Volume*	50	
Recording Tone Duration		
Display On When Incoming Call*	Enabled ▼	☐
RTCP*	Enabled ▼	☑
Log Server		☐
IPv6 Log Server		☐
Remote Log*	Disabled ▼	☐
Log Profile	Default Preset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

Cisco IP Phone 9971 SIP Configuration(Continued…)

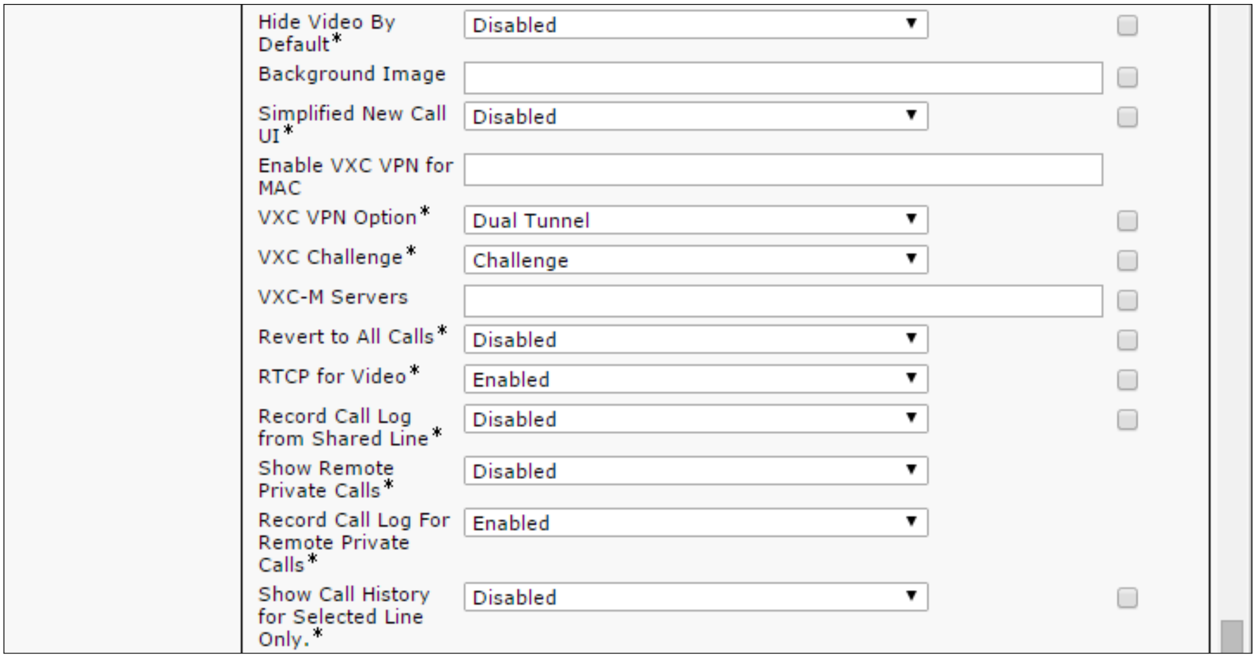


Advertise G.722 and iSAC Codecs *	Use System Default	
Wideband Headset UI Control*	Enabled	
Wideband Headset*	Enabled	
Peer Firmware Sharing*	Enabled	☐
Cisco Discovery Protocol (CDP): Switch Port*	Enabled	☐
Cisco Discovery Protocol (CDP): PC Port*	Enabled	☐
Link Layer Discovery Protocol - Media Endpoint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

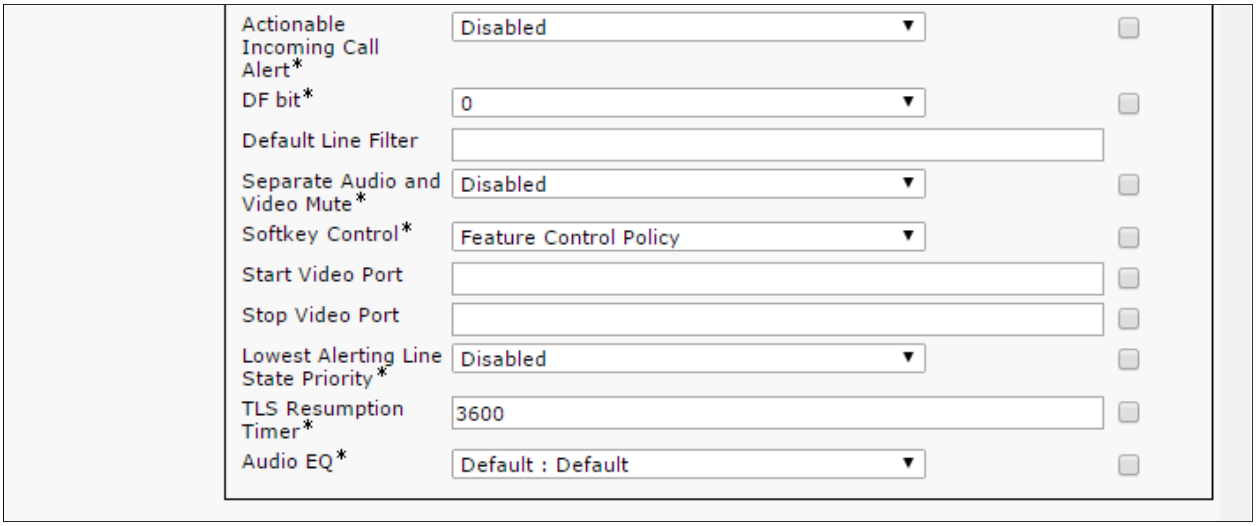


LLDP Asset ID		
LLDP Power Priority*	Unknown	
802.1x Authentication*	User Controlled	☐
FIPS Mode*	Disabled	☐
Detect Unified CM Connection Failure*	Normal	☐
Switch Port Remote Configuration*	Disabled	☐
PC Port Remote Configuration*	Disabled	☐
Automatic Port Synchronization*	Disabled	☐
Power Negotiation*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

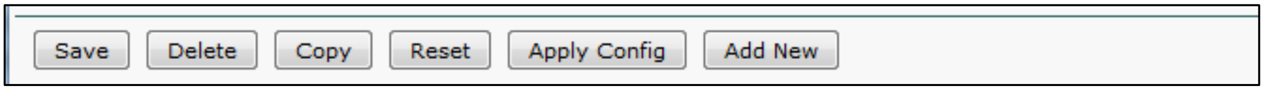
Cisco IP Phone 9971 SIP Configuration(Continued…)



Hide Video By Default*	Disabled	☐
Background Image		☐
Simplified New Call UI*	Disabled	☐
Enable VXC VPN for MAC		
VXC VPN Option*	Dual Tunnel	☐
VXC Challenge*	Challenge	☐
VXC-M Servers		☐
Revert to All Calls*	Disabled	☐
RTCP for Video*	Enabled	☐
Record Call Log from Shared Line*	Disabled	☐
Show Remote Private Calls*	Disabled	
Record Call Log For Remote Private Calls*	Enabled	
Show Call History for Selected Line Only.*	Disabled	☐

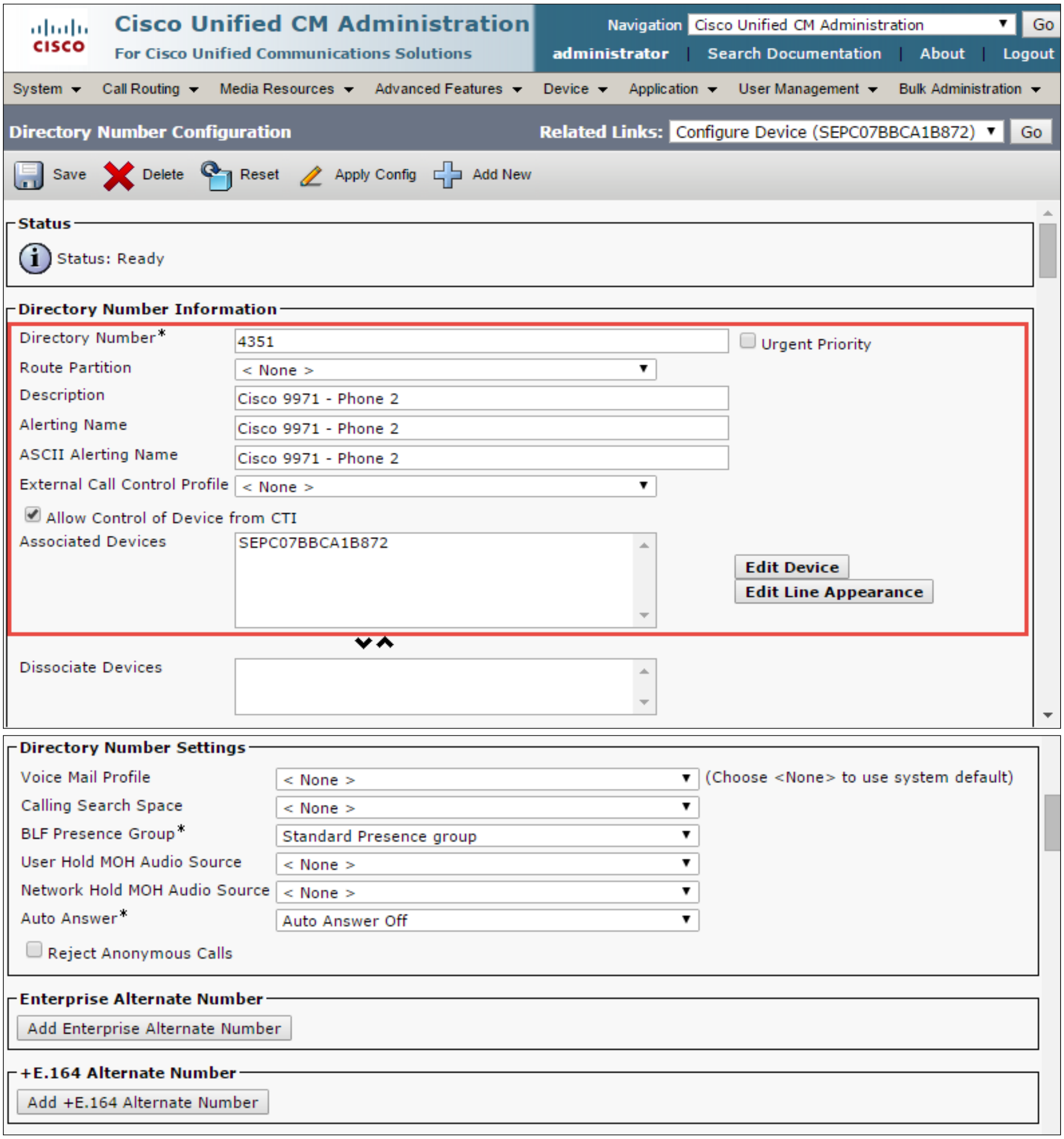


Actionable Incoming Call Alert*	Disabled	☐
DF bit*	0	☐
Default Line Filter		
Separate Audio and Video Mute*	Disabled	☐
Softkey Control*	Feature Control Policy	☐
Start Video Port		☐
Stop Video Port		☐
Lowest Alerting Line State Priority*	Disabled	☐
TLS Resumption Timer*	3600	☐
Audio EQ*	Default : Default	☐



Cisco IP Phone 9971 SIP Configuration(Continued…)

Set Directory Number* = 4351. This is used in this example.
Set Description = 7323204351. This is used in this example.
Set Alerting Name = Cisco 9971 Phone 2. This is used in this example.
Set ASCII Alerting Name = Cisco 9971 Phone 2. This is used in this example.



Status

Status: Ready

Directory Number Information

Directory Number* 4351 ☐ Urgent Priority
Route Partition < None >
Description Cisco 9971 - Phone 2
Alerting Name Cisco 9971 - Phone 2
ASCII Alerting Name Cisco 9971 - Phone 2
External Call Control Profile < None >
☑ Allow Control of Device from CTI
Associated Devices SEPC07BBCA1B872
Dissociate Devices

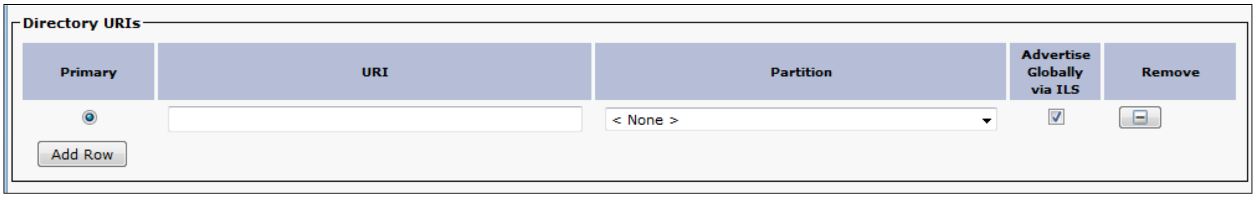
Directory Number Settings

Voice Mail Profile < None > (Choose <None> to use system default)
Calling Search Space < None >
BLF Presence Group* Standard Presence group
User Hold MOH Audio Source < None >
Network Hold MOH Audio Source < None >
Auto Answer* Auto Answer Off
☐ Reject Anonymous Calls

Enterprise Alternate Number

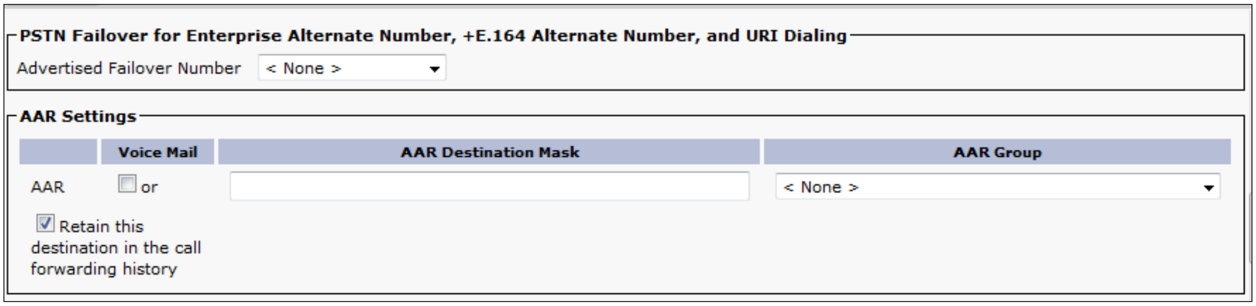
+E.164 Alternate Number

Cisco IP Phone 9971 SIP Configuration (Continued…)



Directory URIs

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



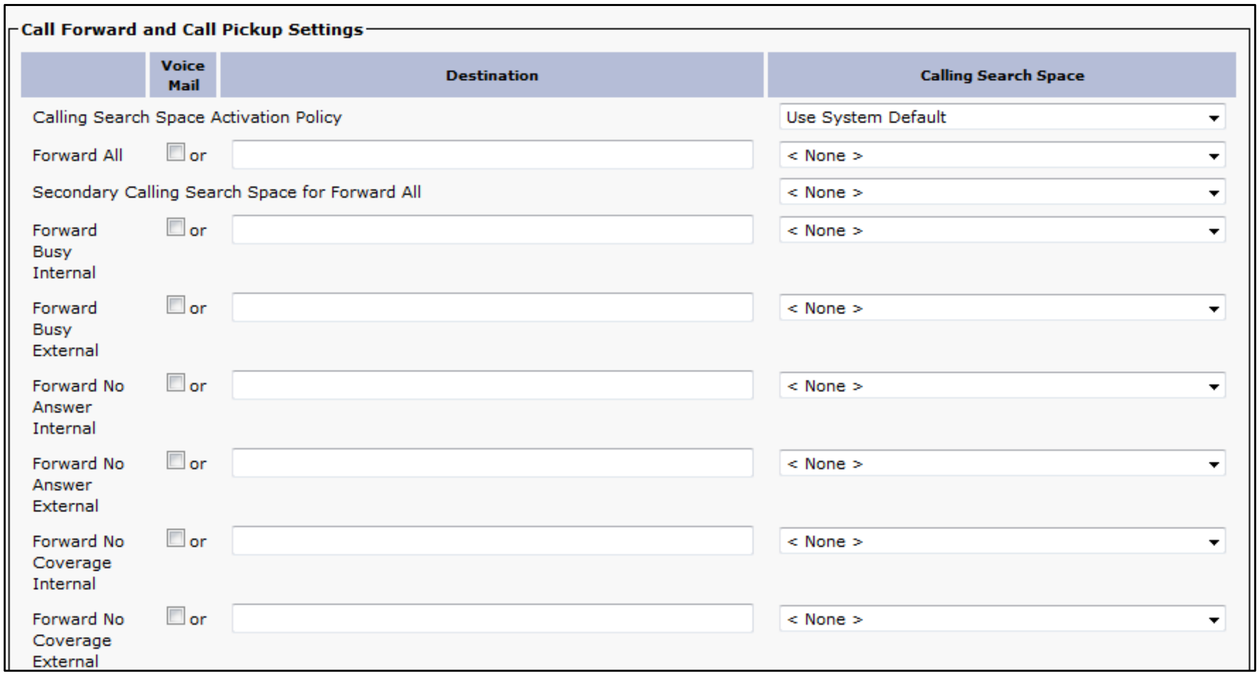
PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing

Advertised Failover Number: < None >

AAR Settings

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

☑ Retain this destination in the call forwarding history



Call Forward and Call Pickup Settings

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

Cisco IP Phone 9971 SIP Configuration (Continued…)

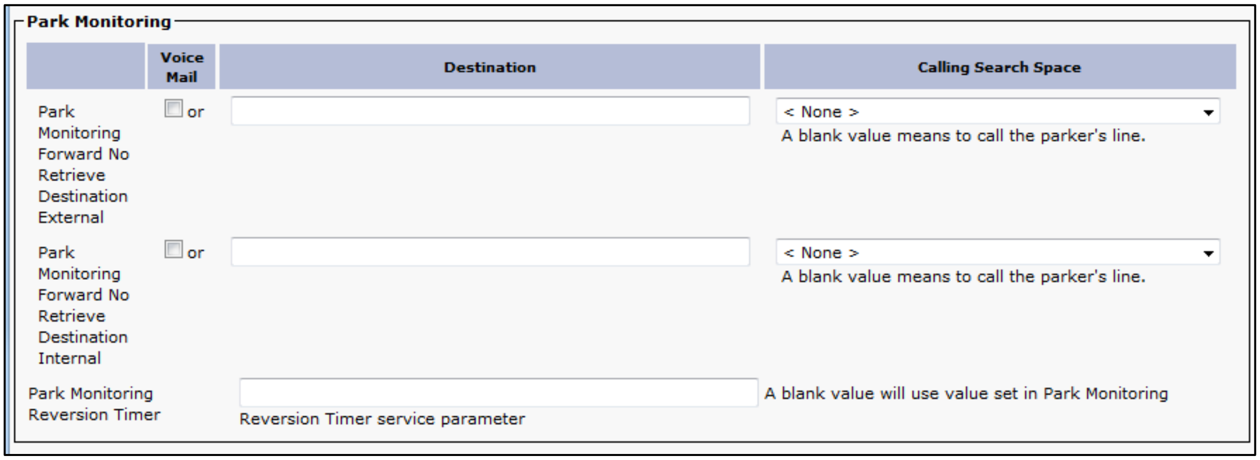
Forward on CTI Failure ☐ or < None >

Forward Unregistered Internal ☐ or < None >

Forward Unregistered External ☐ or < None >

No Answer Ring Duration (seconds)

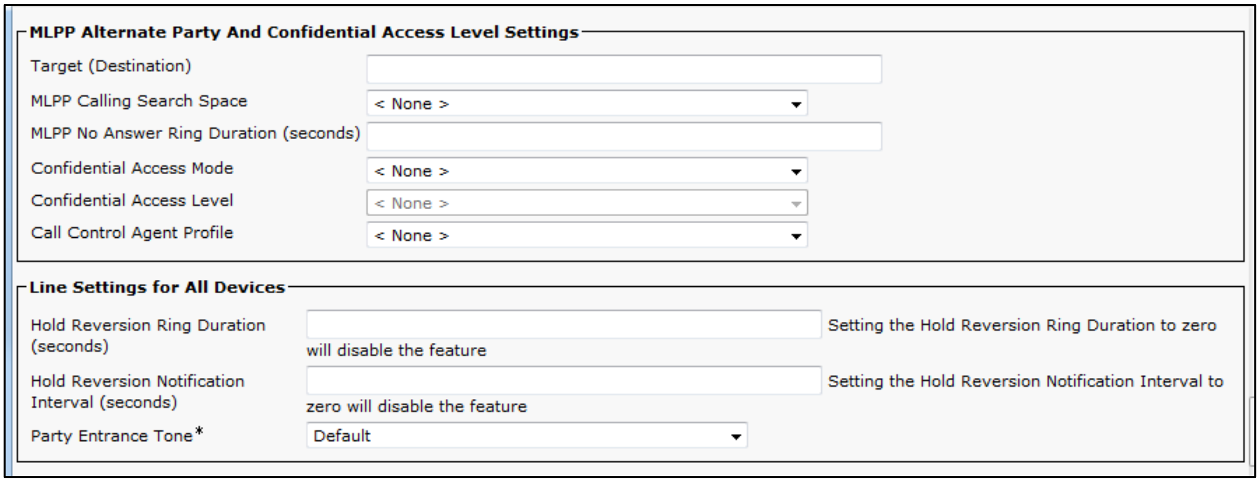
Call Pickup Group < None >



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

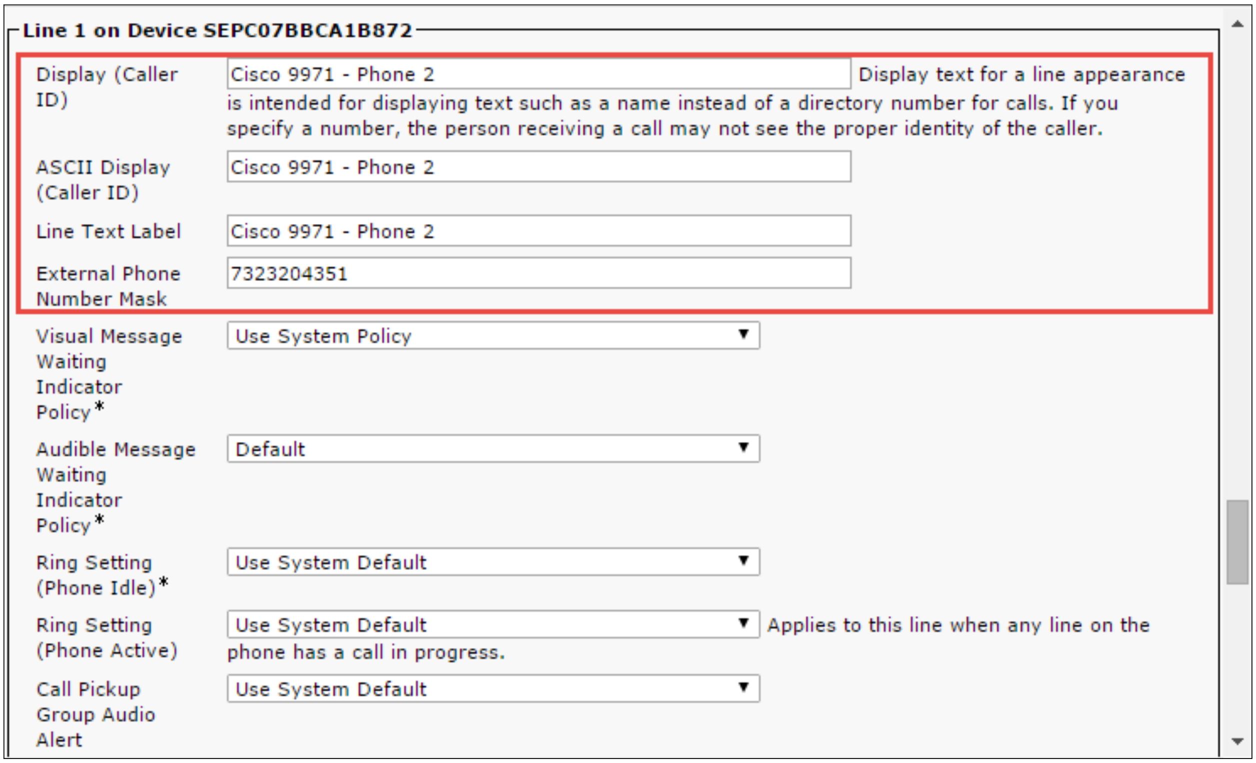
Hold Reversion Ring Duration (seconds) — Setting the Hold Reversion Ring Duration to zero will disable the feature

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…)

Set Display (caller ID) = Cisco9971-Phone 2. This is used in this example.
Set ASCII Display (caller ID) = Cisco9971-Phone 2. This is used in this example.
Set Line Text Label = Cisco9971-Phone 2. This is used in this example.
Set External Phone Number Mask = 7323204351. This is used in this example.



Line 1 on Device SEPC07BBCA1B872

Display (Caller ID): Cisco 9971 - Phone 2 — Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.

ASCII Display (Caller ID): Cisco 9971 - Phone 2

Line Text Label: Cisco 9971 - Phone 2

External Phone Number Mask: 7323204351

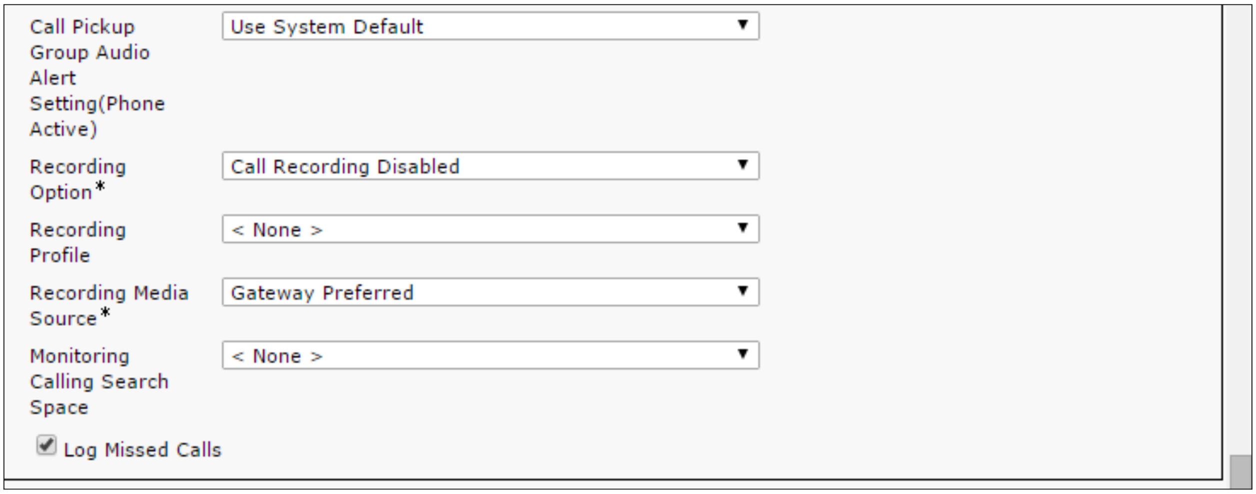
Visual Message Waiting Indicator Policy*: Use System Policy

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: Use System Default



Call Pickup Group Audio Alert Setting(Phone Active): Use System Default

Recording Option*: Call Recording Disabled

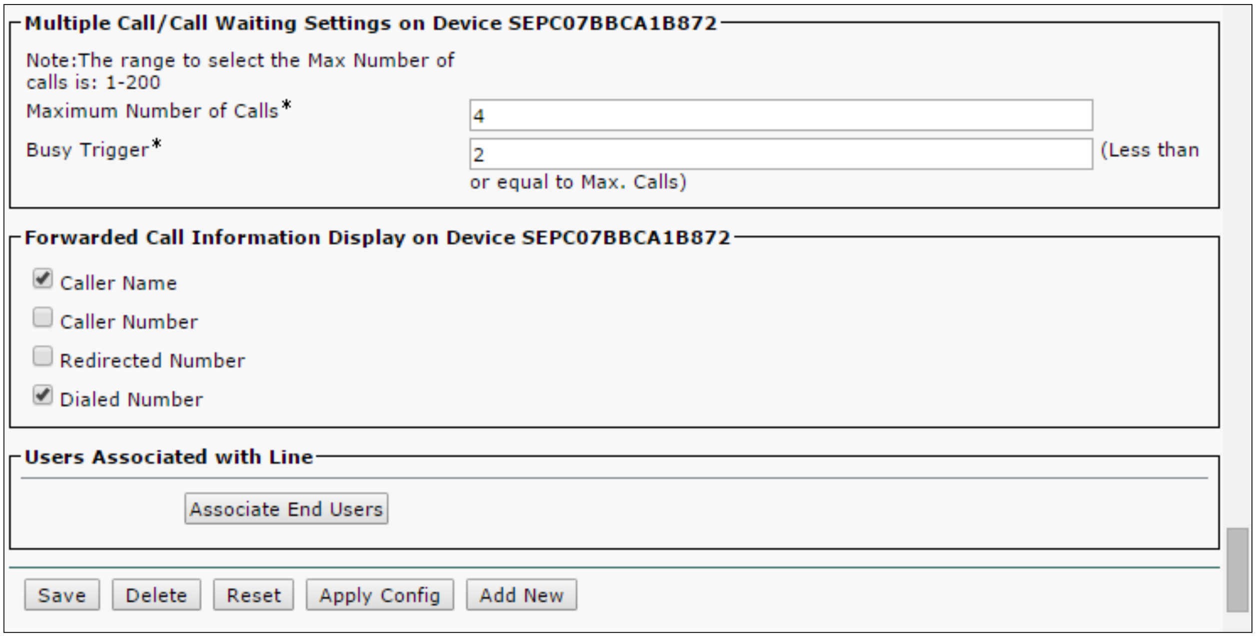
Recording Profile: < None >

Recording Media Source*: Gateway Preferred

Monitoring Calling Search Space: < None >

☑ Log Missed Calls

Cisco IP Phone 9971 SIP Configuration (Continued…)


Multiple Call/Call Waiting Settings on Device SEPC07BBCA1B872

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

Maximum Number of Calls* 4

Busy Trigger* 2 (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEPC07BBCA1B872

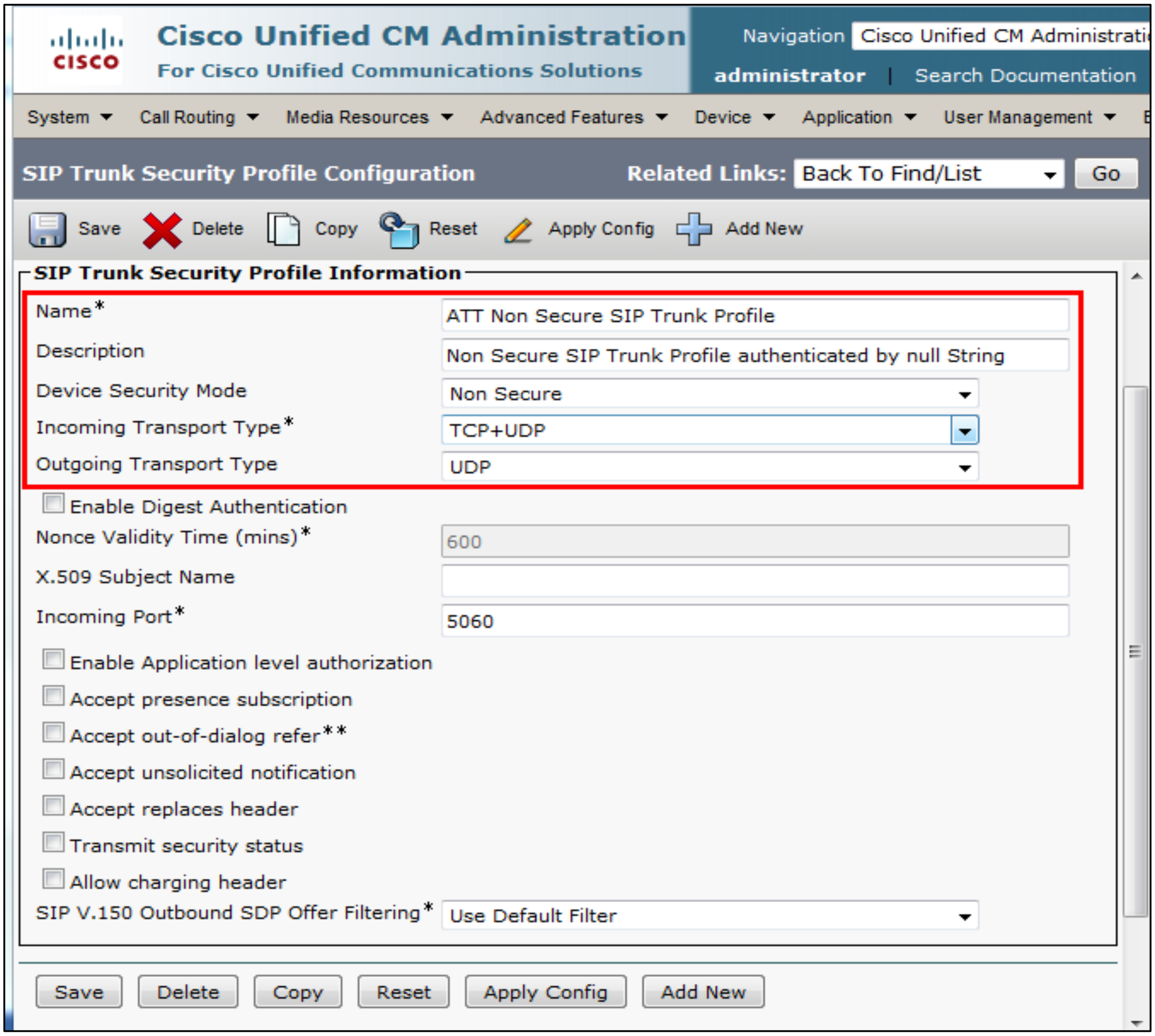
- ☑ Caller Name
- ☐ Caller Number
- ☐ Redirected Number
- ☑ Dialed Number

Users Associated with Line

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 Trunk Security Profile Configuration

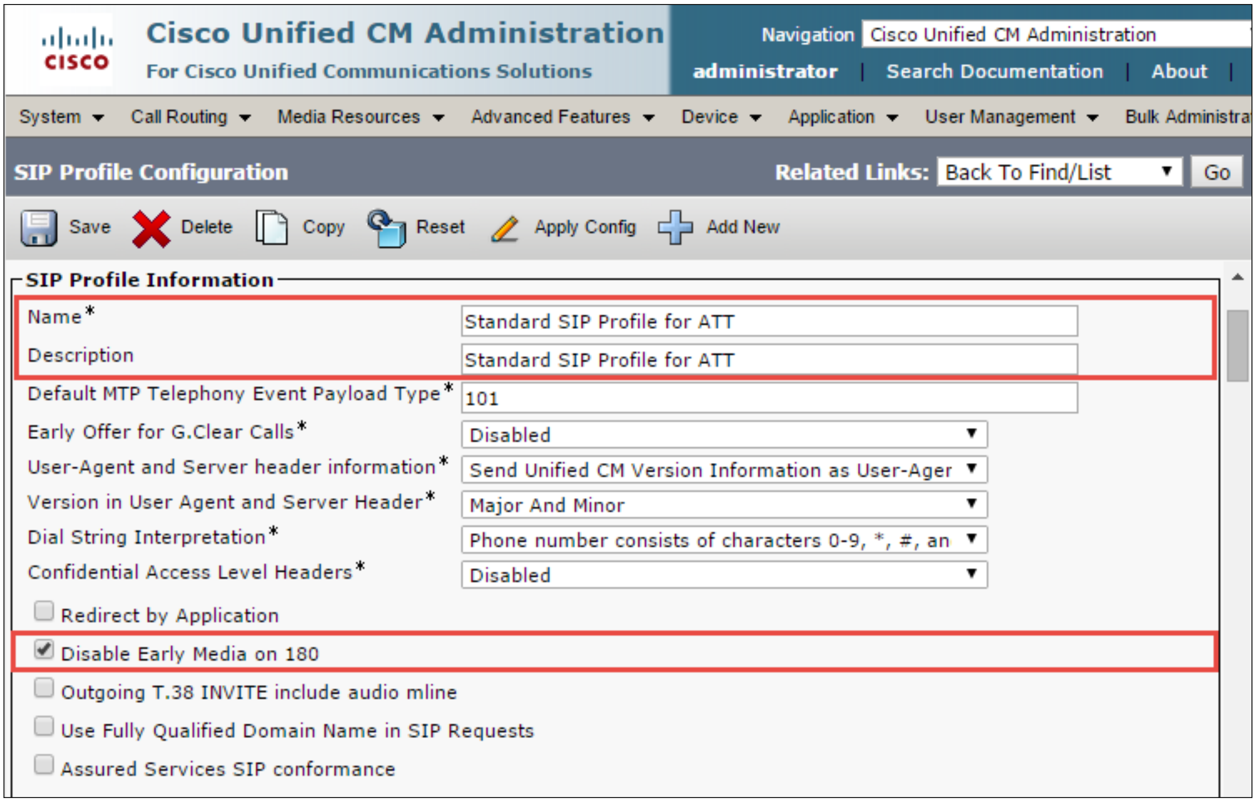
SIP Trunk Security Profile Information

Name* ATT Non Secure SIP Trunk Profile
Description Non Secure SIP Trunk Profile authenticated by null String
Device Security Mode Non Secure
Incoming Transport Type* TCP+UDP
Outgoing Transport Type UDP
☐ Enable Digest Authentication
Nonce Validity Time (mins)* 600
X.509 Subject Name
Incoming Port* 5060
☐ Enable Application level authorization
☐ Accept presence subscription
☐ Accept out-of-dialog refer**
☐ Accept unsolicited notification
☐ Accept replaces header
☐ Transmit security status
☐ Allow charging header
SIP V.150 Outbound SDP Offer Filtering* Use Default Filter

SIP Profile Configuration used by SIP trunk to Cisco UBE
Navigation: Device → Device Settings → SIP Profile

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

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

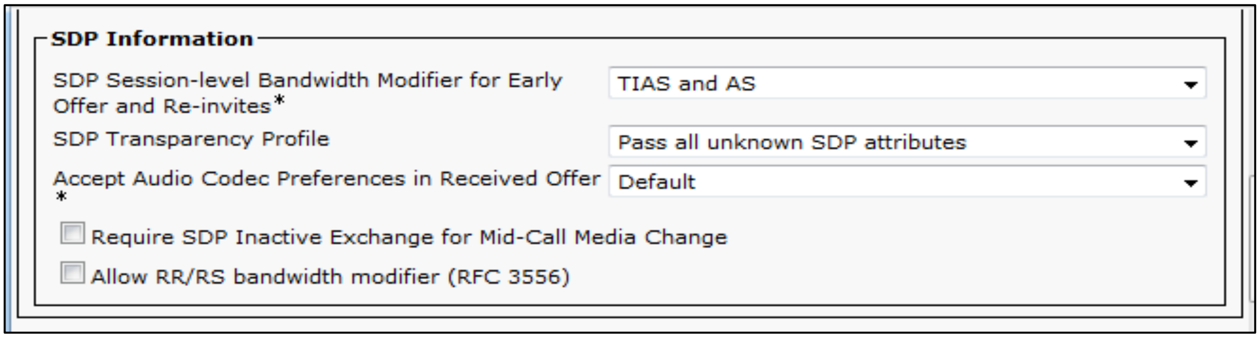


SIP Profile Information

Name* Standard SIP Profile for ATT
Description Standard SIP Profile for ATT
Default MTP Telephony Event Payload Type* 101
Early Offer for G.Clear Calls* Disabled
User-Agent and Server header information* Send Unified CM Version Information as User-Ager
Version in User Agent and Server Header* Major And Minor
Dial String Interpretation* Phone number consists of characters 0-9, *, #, an
Confidential Access Level Headers* Disabled

☐ Redirect by Application
☑ Disable Early Media on 180
☐ Outgoing T.38 INVITE include audio mline
☐ Use Fully Qualified Domain Name in SIP Requests
☐ Assured Services SIP conformance

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

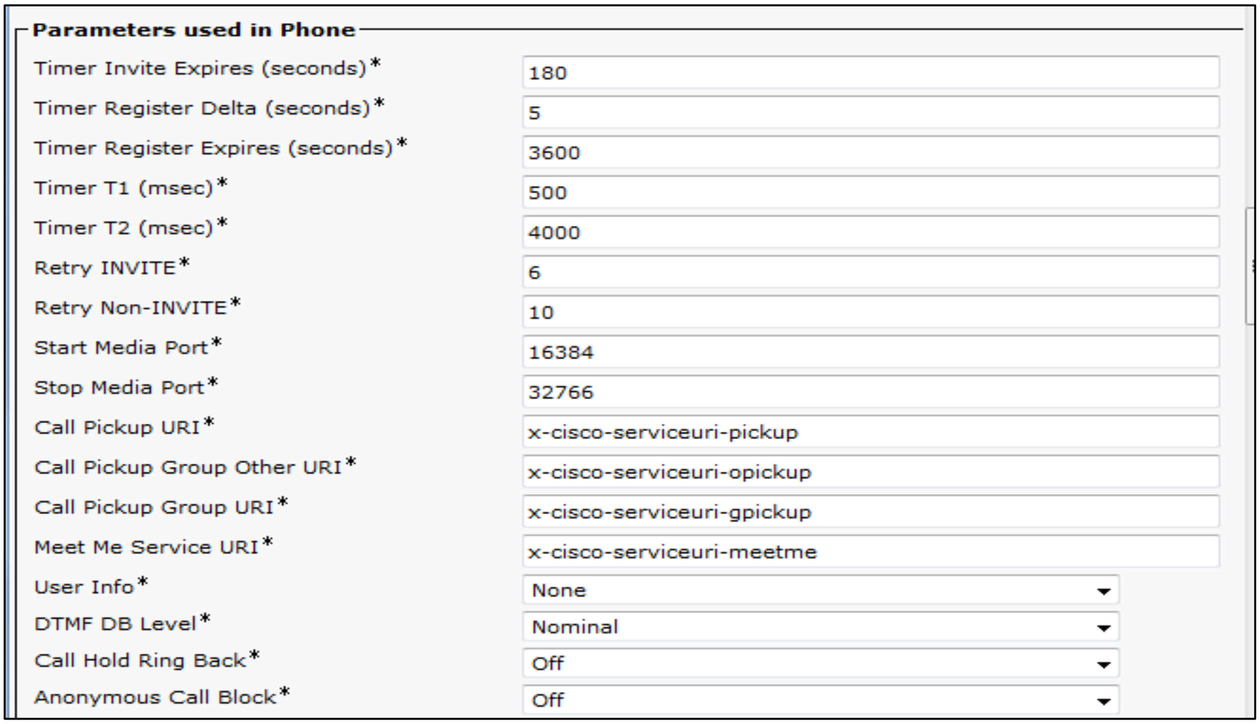


SDP Information

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

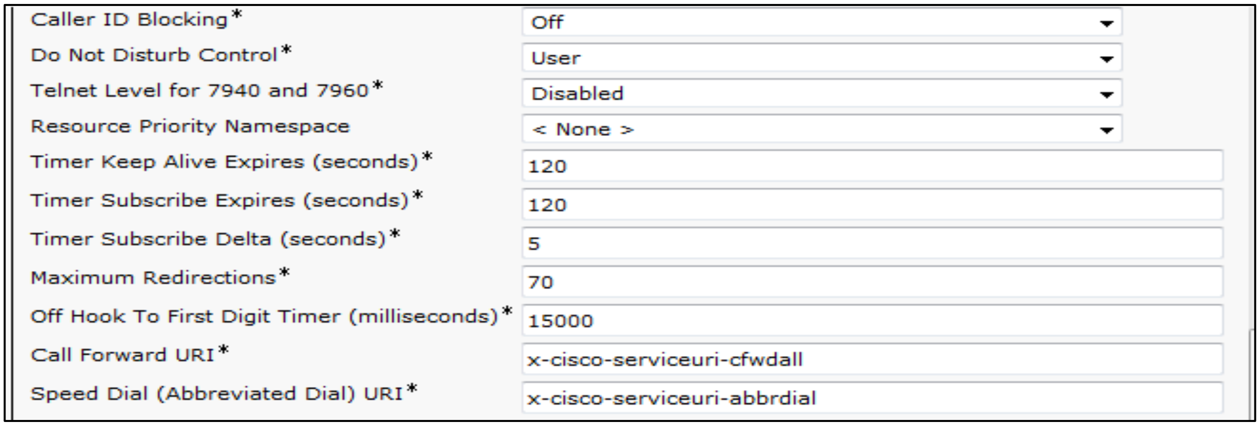
☐ Require SDP Inactive Exchange for Mid-Call Media Change

☐ Allow RR/RS bandwidth modifier (RFC 3556)



Parameters used in Phone

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



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 (Continued…)

☑ Conference Join Enabled

☐ RFC 2543 Hold

☑ Semi Attended Transfer

☐ Enable VAD

☐ Stutter Message Waiting

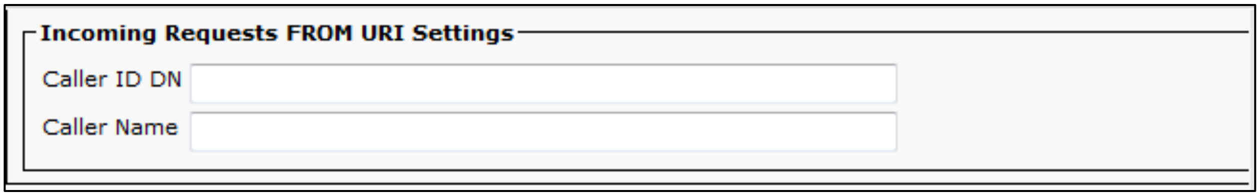
☐ MLPP User Authorization

Normalization Script

Normalization Script: < None >

☐ Enable Trace

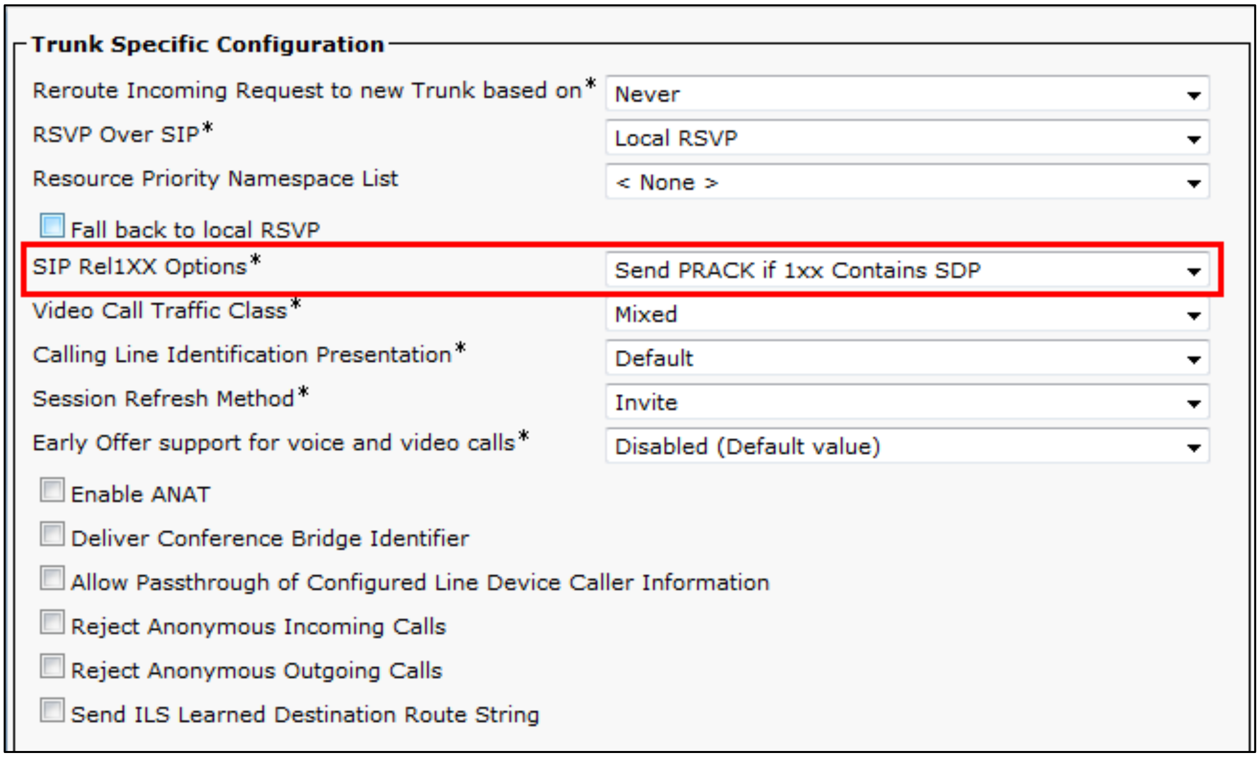
	Parameter Name	Parameter Value
1		



Incoming Requests FROM URI Settings

Caller ID DN:

Caller Name:



Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on* Never

RSVP Over SIP* Local RSVP

Resource Priority Namespace List < None >

☐ Fall back to local RSVP

SIP Rel1XX Options* Send PRACK if 1xx Contains SDP

Video Call Traffic Class* Mixed

Calling Line Identification Presentation* Default

Session Refresh Method* Invite

Early Offer support for voice and video calls* Disabled (Default value)

☐ Enable ANAT

☐ Deliver Conference Bridge Identifier

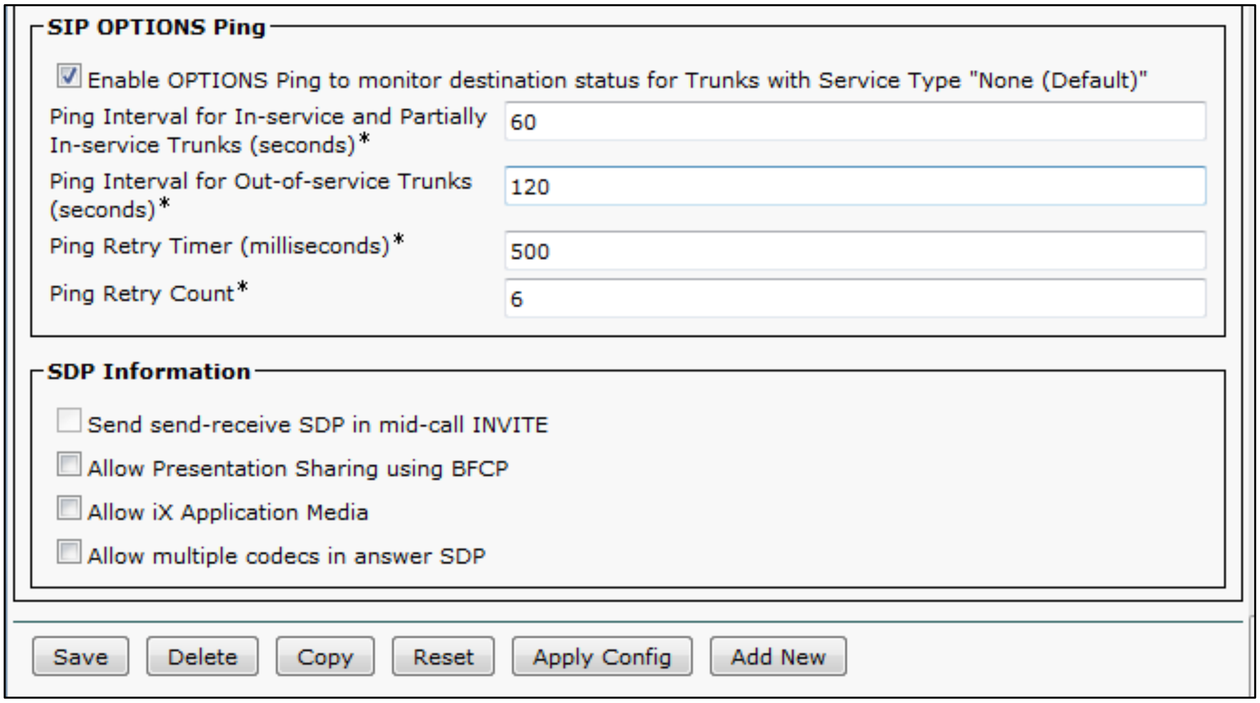
☐ Allow Passthrough of Configured Line Device Caller Information

☐ Reject Anonymous Incoming Calls

☐ Reject Anonymous Outgoing Calls

☐ Send ILS Learned Destination Route String

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



SIP OPTIONS Ping

☑ Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)* 60

Ping Interval for Out-of-service Trunks (seconds)* 120

Ping Retry Timer (milliseconds)* 500

Ping Retry Count* 6

SDP Information

☐ Send send-receive SDP in mid-call INVITE

☐ Allow Presentation Sharing using BFCP

☐ Allow iX Application Media

☐ Allow multiple codecs in answer SDP

SIP Trunk to Cisco UBE Configuration

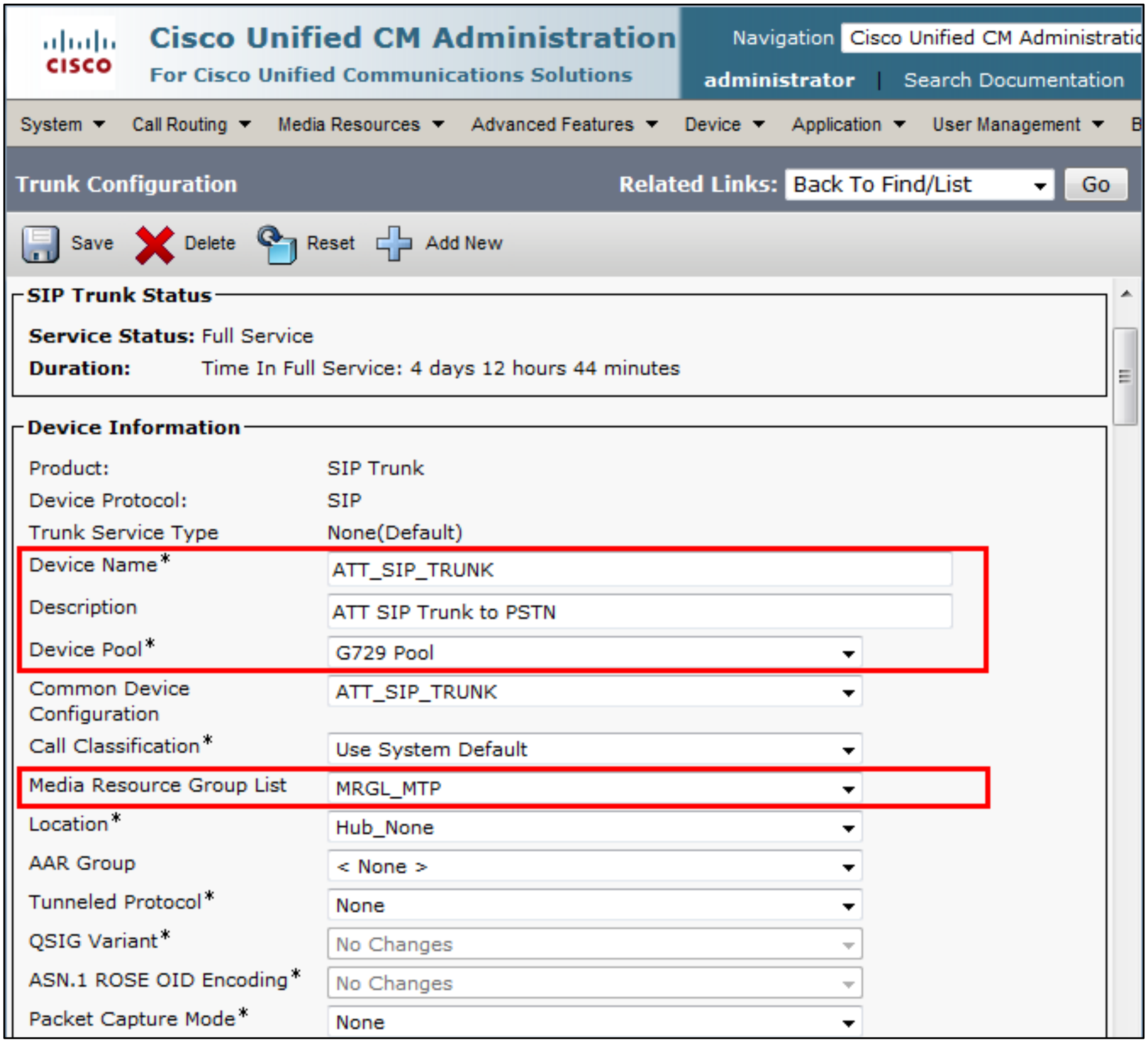
Navigation: Device → Trunk

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

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

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

Set Media Resource Group List = MRGL_MTP.



Trunk Configuration

SIP Trunk Status

Service Status: Full Service

Duration: Time In Full Service: 4 days 12 hours 44 minutes

Device Information

Product: SIP Trunk

Device Protocol: SIP

Trunk Service Type None(Default)

Device Name* ATT_SIP_TRUNK

Description ATT SIP Trunk to PSTN

Device Pool* G729 Pool

Common Device Configuration ATT_SIP_TRUNK

Call Classification* Use System Default

Media Resource Group List MRGL_MTP

Location* Hub_None

AAR Group < None >

Tunneled Protocol* None

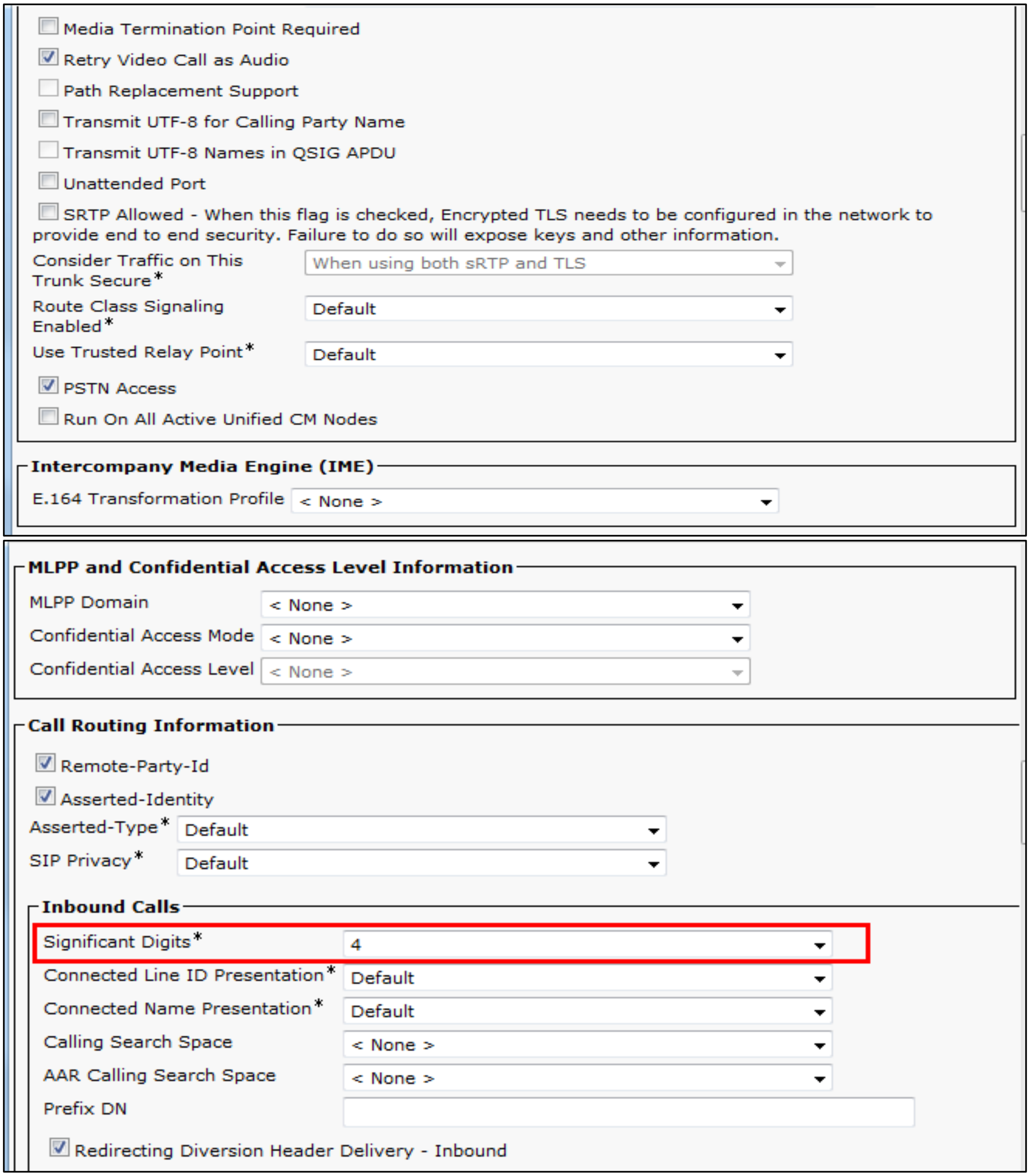
QSIG Variant* No Changes

ASN.1 ROSE OID Encoding* No Changes

Packet Capture Mode* None

SIP Trunk to Cisco UBE Configuration (Continued…)

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



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

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

☑ PSTN Access
☐ Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile	< None >

MLPP and Confidential Access Level Information

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

Call Routing Information

☑ Remote-Party-Id
☑ Asserted-Identity

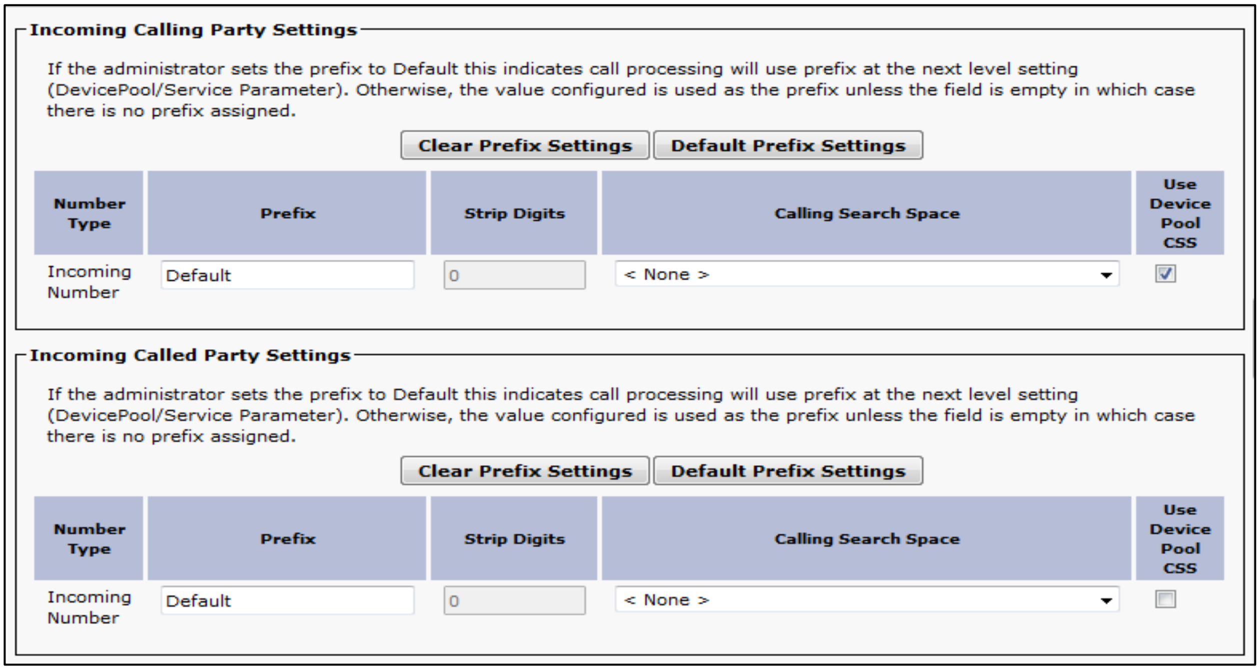
Asserted-Type*	Default
SIP Privacy*	Default

Inbound Calls

Significant Digits*	4
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	

☑ Redirecting Diversion Header Delivery - Inbound

SIP Trunk to Cisco UBE Configuration (Continued…)



Incoming Calling Party Settings

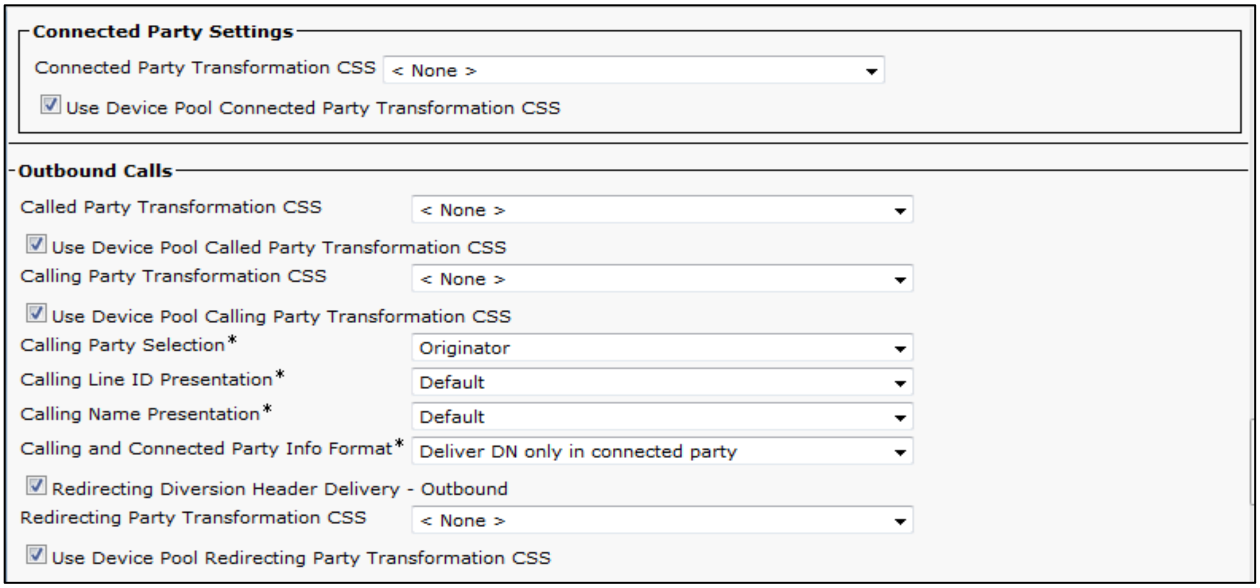
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

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.

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



Connected Party Settings

Connected Party Transformation CSS < None >

☑ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS < None >

☑ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS < None >

☑ Use Device Pool Calling Party Transformation CSS

Calling Party Selection* Originator

Calling Line ID Presentation* Default

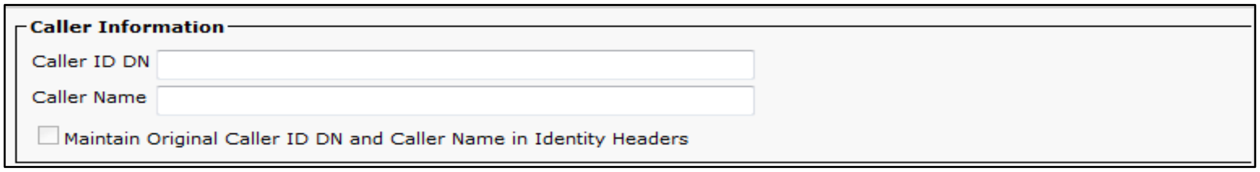
Calling Name Presentation* Default

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

☑ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

☑ Use Device Pool Redirecting Party Transformation CSS



Caller Information

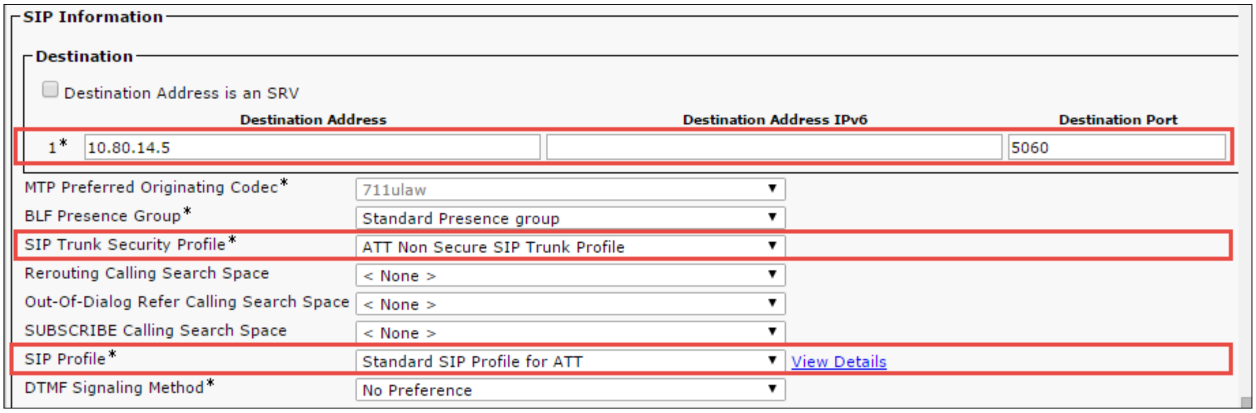
Caller ID DN

Caller Name

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

SIP Trunk to Cisco UBE Configuration (Continued…)

Set Destination Address = Set IP address of ASR-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.



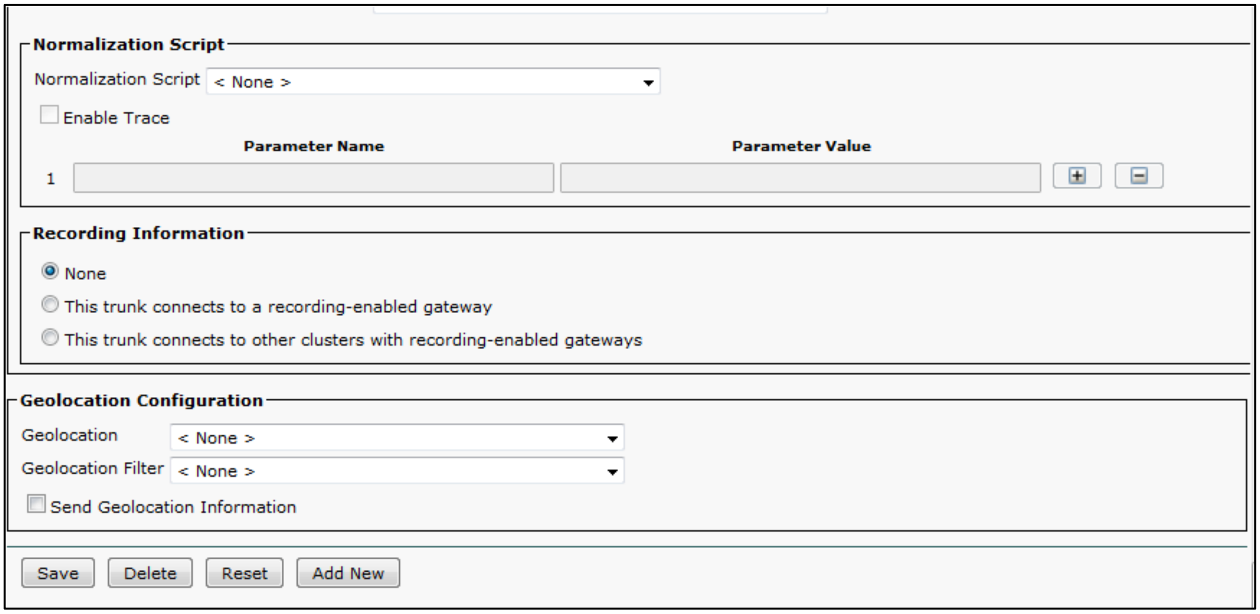
SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	10.80.14.5		5060

MTP Preferred Originating Codec* : 711ulaw
BLF Presence Group* : Standard Presence group
SIP Trunk Security Profile* : ATT Non Secure SIP Trunk Profile
Rerouting Calling Search Space : < None >
Out-Of-Dialog Refer Calling Search Space : < None >
SUBSCRIBE Calling Search Space : < None >
SIP Profile* : Standard SIP Profile for ATT
DTMF Signaling Method* : No Preference



Normalization Script

Normalization Script : < None >

☐ Enable Trace

	Parameter Name	Parameter Value
1		

Recording Information

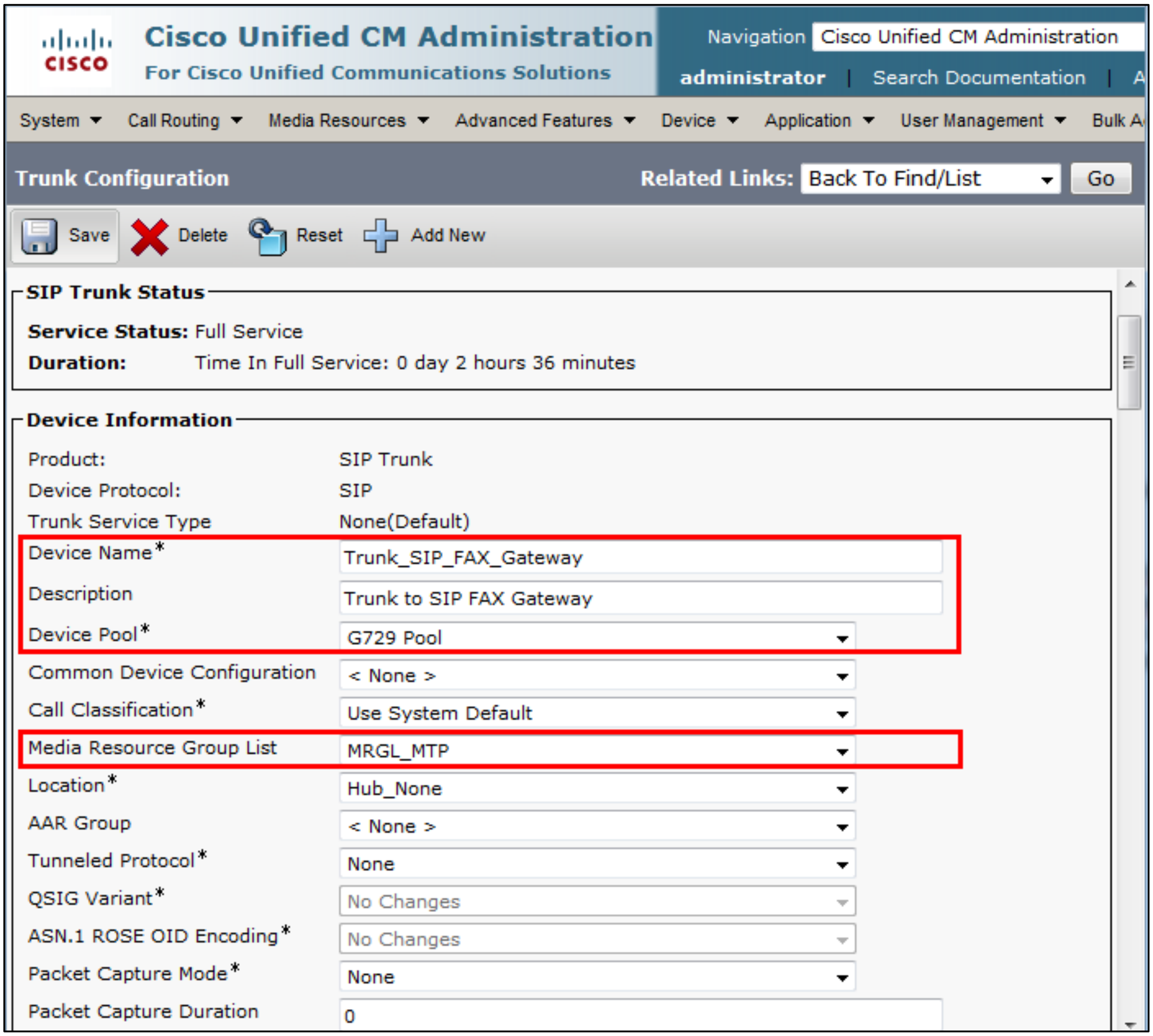
● None
○ This trunk connects to a recording-enabled gateway
○ This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation : < None >
Geolocation Filter : < None >
☐ Send Geolocation Information

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.



Trunk Configuration

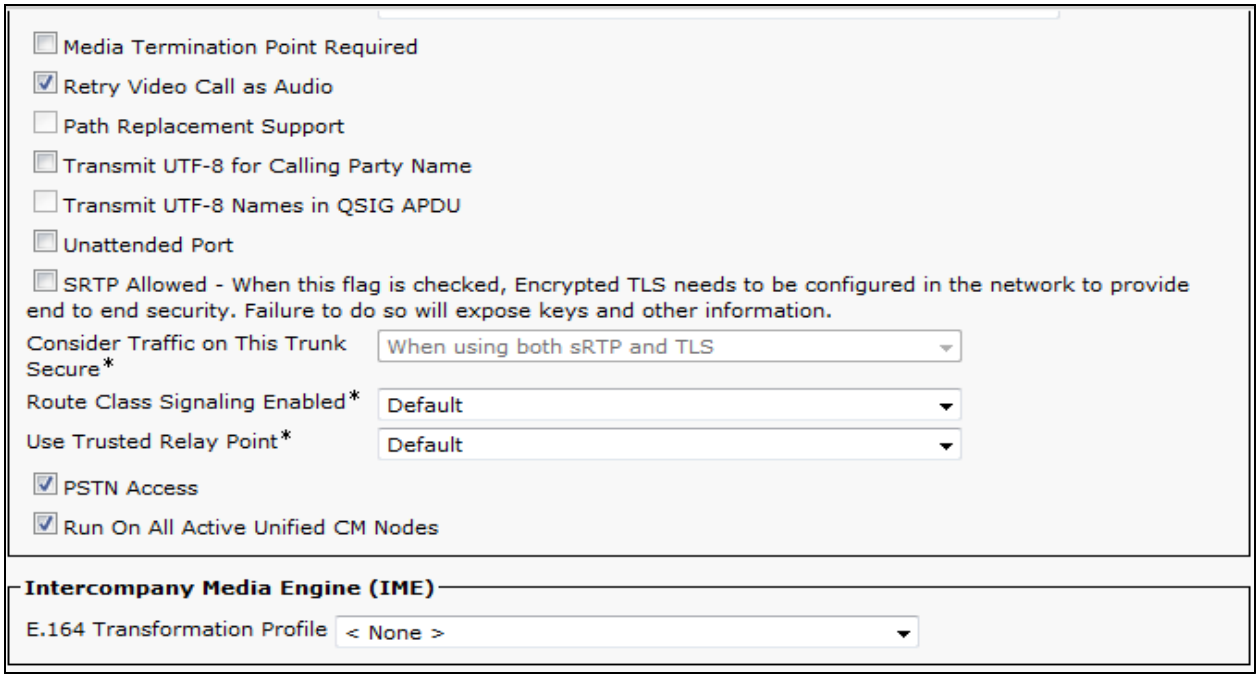
SIP Trunk Status

Service Status: Full Service
Duration: Time In Full Service: 0 day 2 hours 36 minutes

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Trunk_SIP_FAX_Gateway
Description	Trunk to SIP FAX Gateway
Device Pool*	G729 Pool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_MTP
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

SIP Trunk to Fax Gateway Configuration (Continued…)



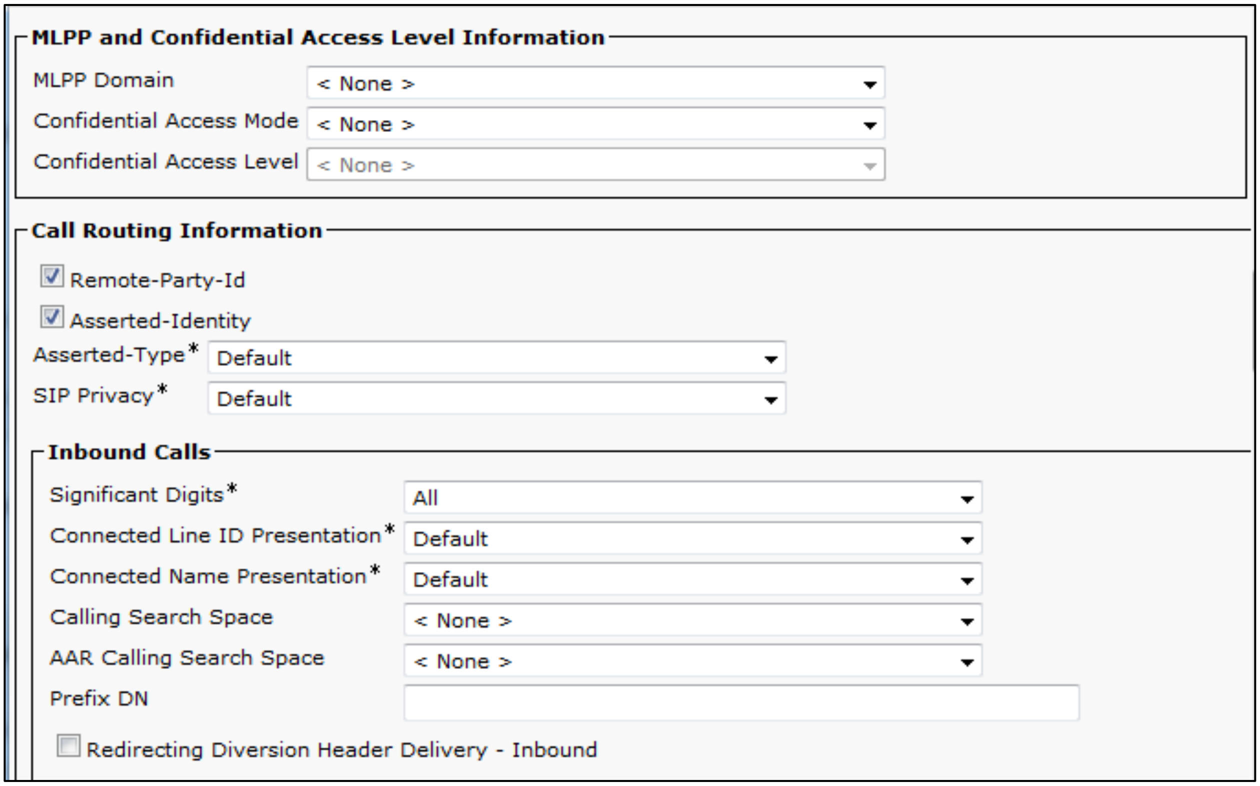
☐ Media Termination Point Required
☑ Retry Video Call as Audio
☐ Path Replacement Support
☐ Transmit UTF-8 for Calling Party Name
☐ Transmit UTF-8 Names in QSIG APDU
☐ Unattended Port
☐ SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end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

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

☑ PSTN Access
☑ Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile: < None >



MLPP and Confidential Access Level Information

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

Call Routing Information

☑ Remote-Party-Id
☑ Asserted-Identity

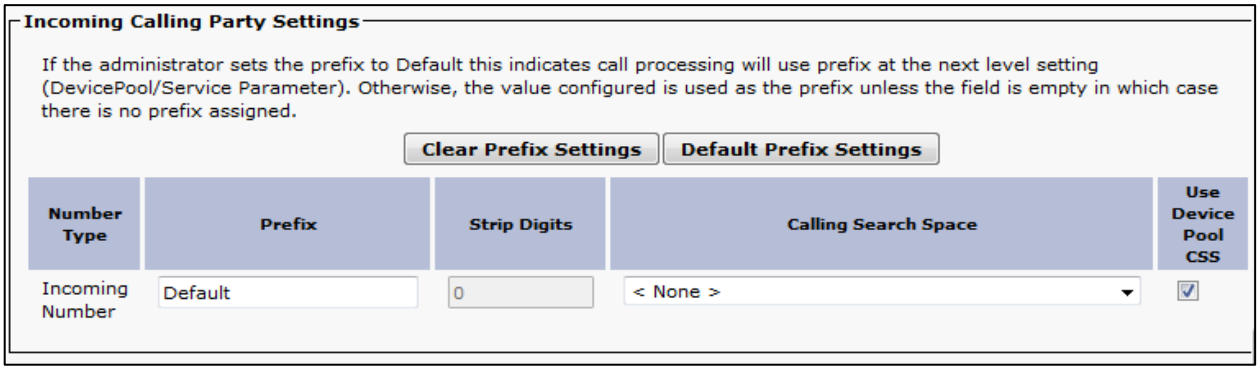
Asserted-Type*: Default
SIP Privacy*: Default

Inbound Calls

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

☐ Redirecting Diversion Header Delivery - Inbound

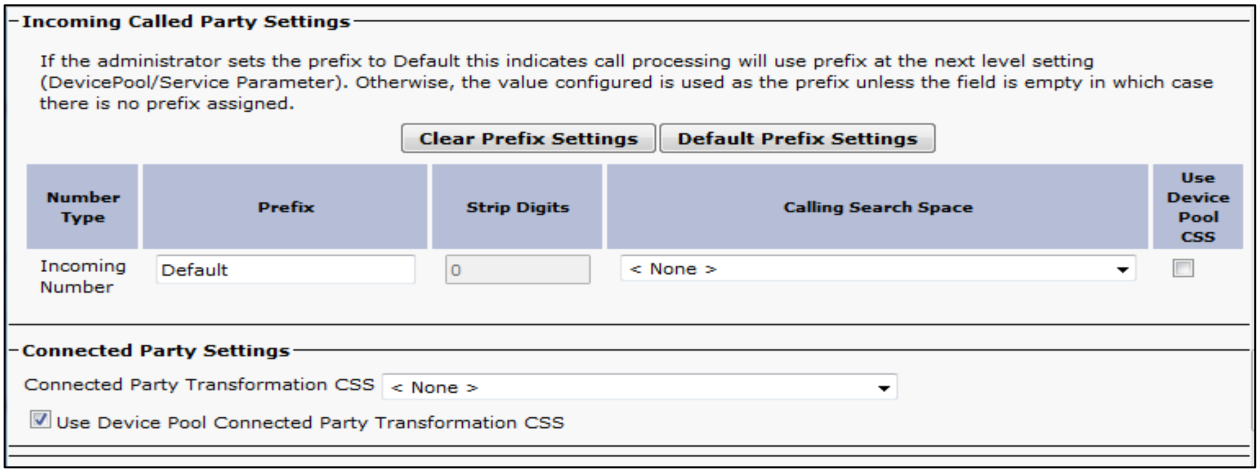
SIP Trunk to Fax Gateway Configuration (Continued…)



Incoming Calling Party Settings

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

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



Incoming Called Party Settings

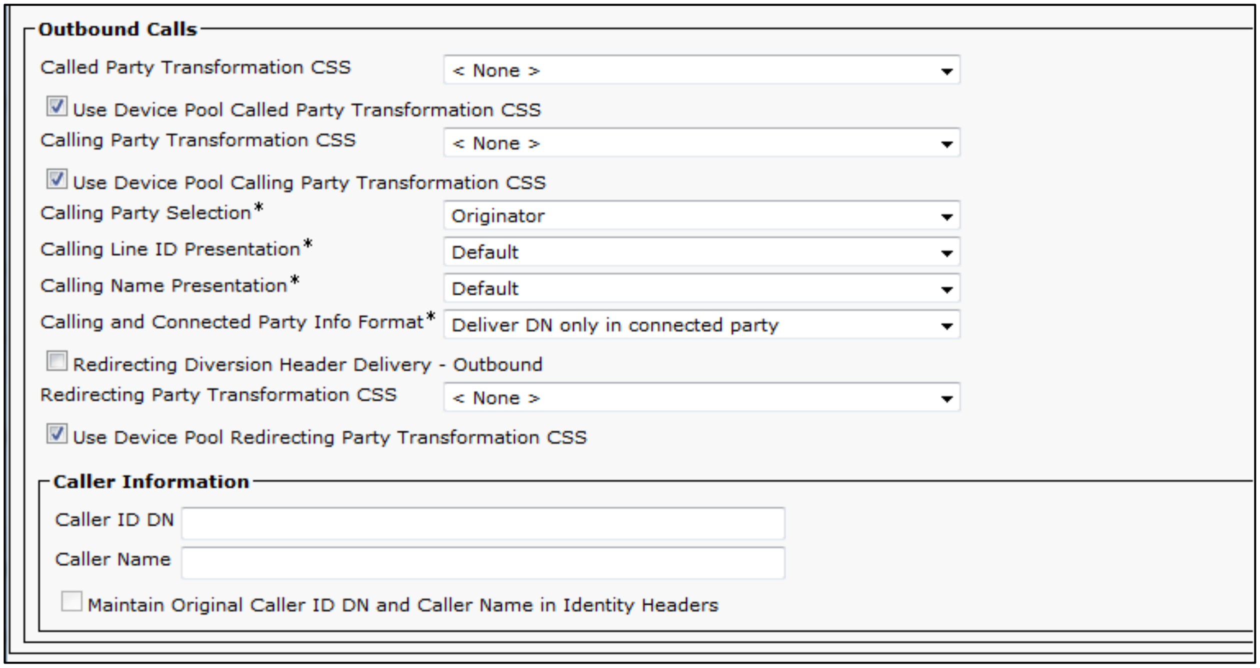
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

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

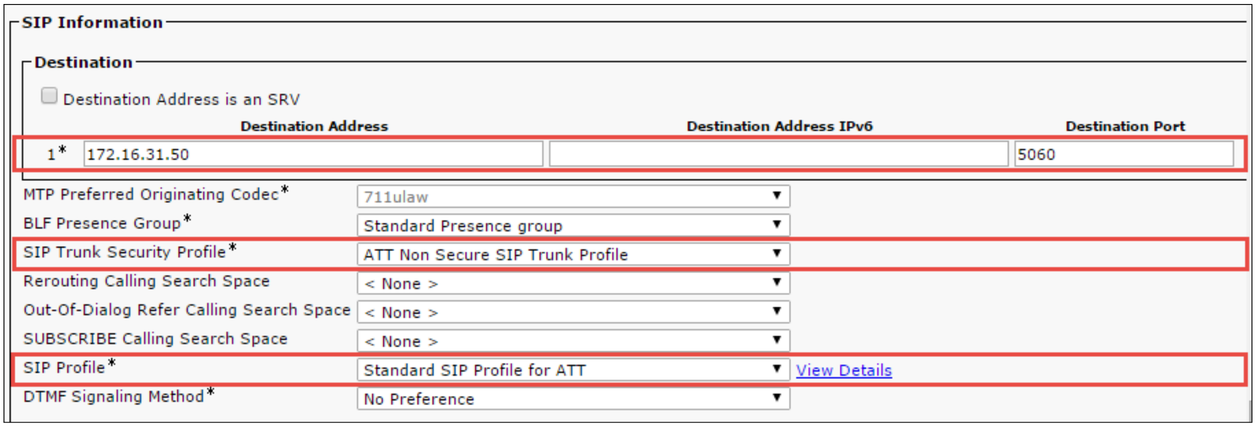
Caller Information

Caller ID DN:

Caller Name:

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

SIP Trunk to Fax Gateway Configuration (Continued…)



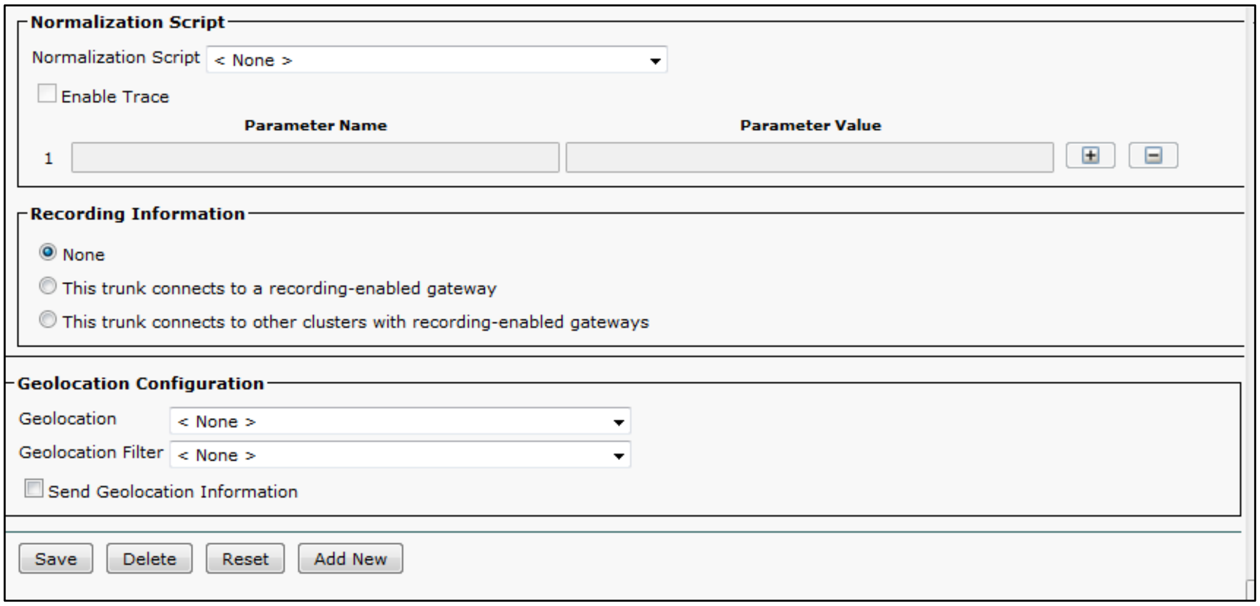
SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	172.16.31.50		5060

MTP Preferred Originating Codec*: 711ulaw
BLF Presence Group*: Standard Presence group
SIP Trunk Security Profile*: ATT Non Secure SIP Trunk Profile
Rerouting Calling Search Space: < None >
Out-Of-Dialog Refer Calling Search Space: < None >
SUBSCRIBE Calling Search Space: < None >
SIP Profile*: Standard SIP Profile for ATT
DTMF Signaling Method*: No Preference



Normalization Script

Normalization Script: < None >

☐ Enable Trace

	Parameter Name	Parameter Value
1		

Recording Information

● None
○ This trunk connects to a recording-enabled gateway
○ This trunk connects to other clusters with recording-enabled gateways

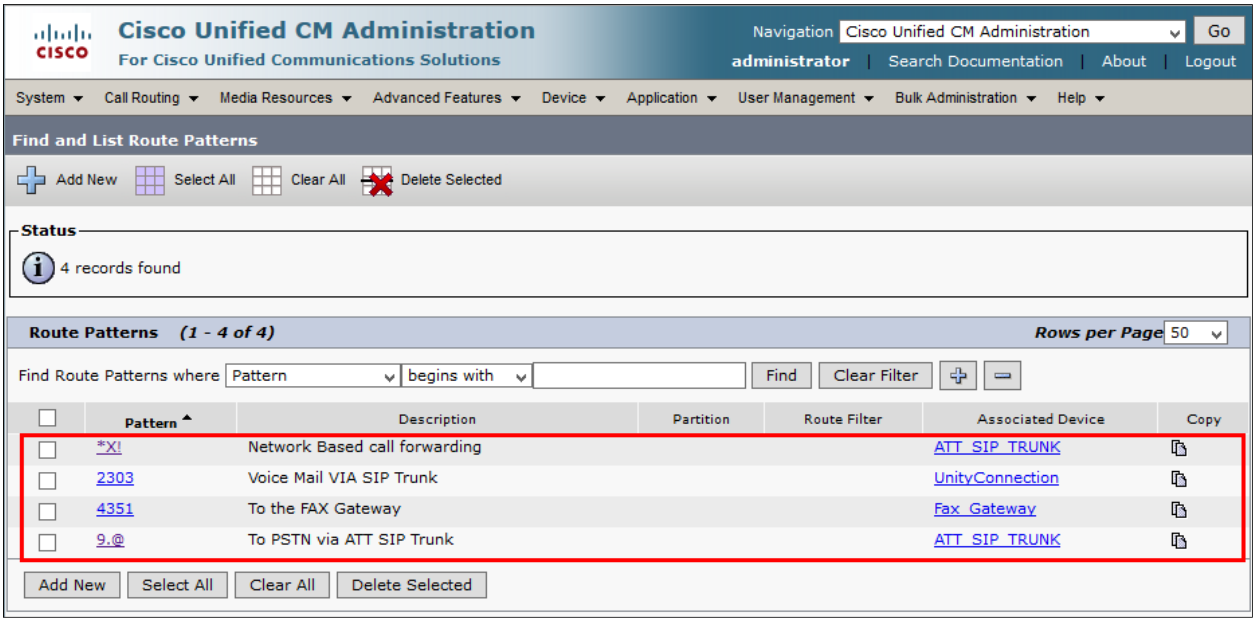
Geolocation Configuration

Geolocation: < None >
Geolocation Filter: < None >
☐ Send Geolocation Information

Route Pattern Configuration

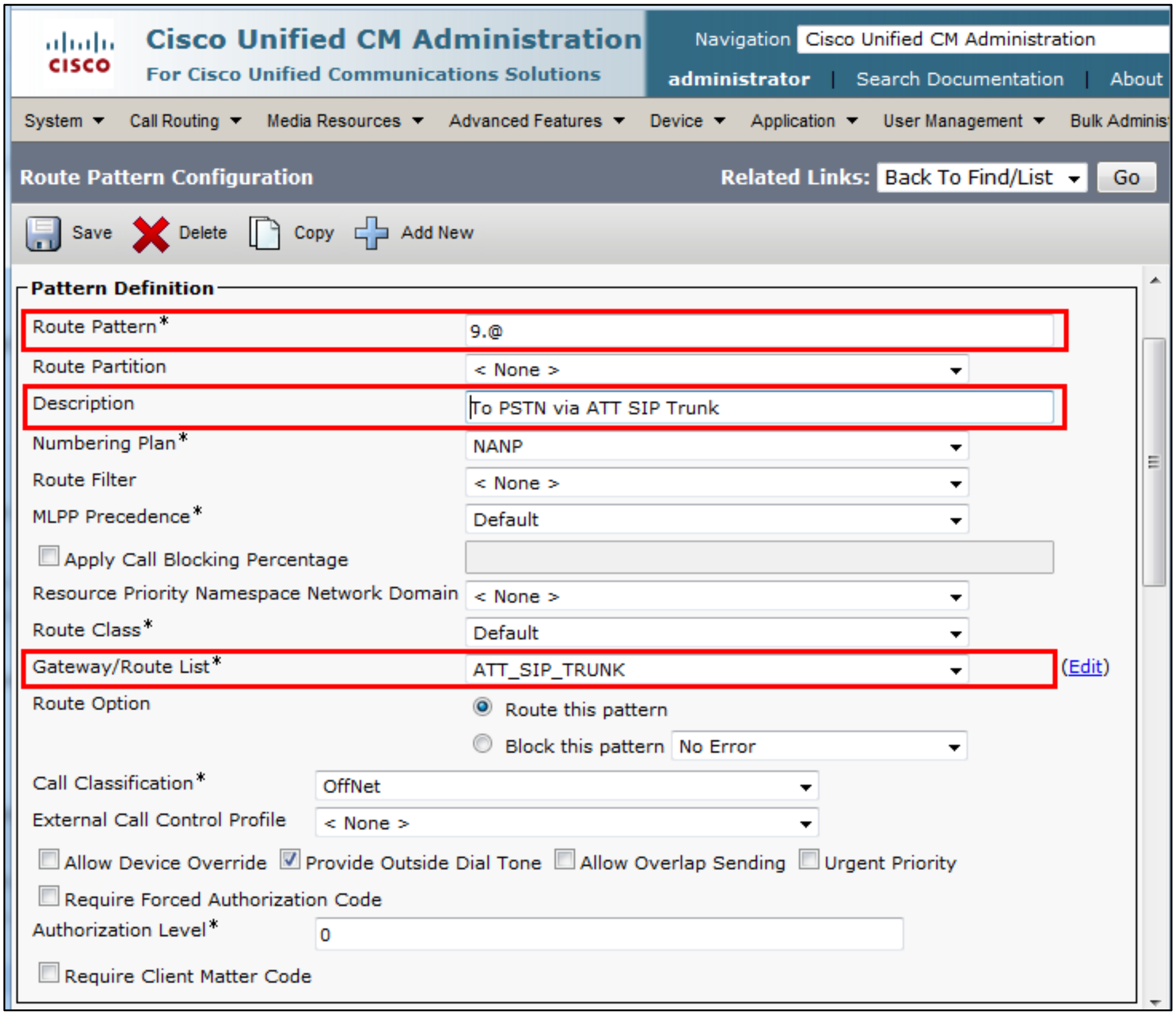
Navigation: Call Routing → Route/Hunt → Route Pattern

Set Route Pattern* = 9. @   This is used to route to AT&T via ASR 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



☐	Pattern	Description	Partition	Route Filter	Associated Device	Copy
☐	*X!	Network Based call forwarding			ATT_SIP_TRUNK	
☐	2303	Voice Mail VIA SIP Trunk			UnityConnection	
☐	4351	To the FAX Gateway			Fax_Gateway	
☐	9.@	To PSTN via ATT SIP Trunk			ATT_SIP_TRUNK	

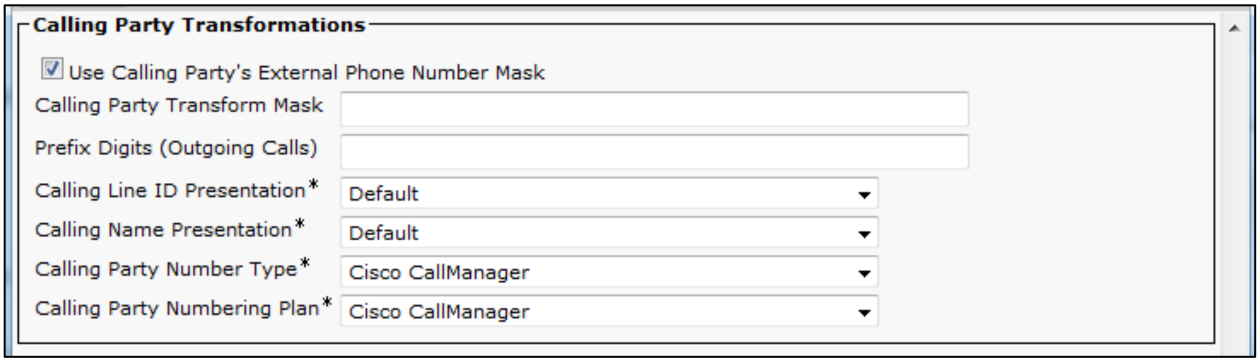
Route Pattern Configuration (Continued…)



Route Pattern Configuration

Pattern Definition

Route Pattern* 9.@
Route Partition < None >
Description To PSTN via ATT SIP Trunk
Numbering Plan* NANP
Route Filter < None >
MLPP Precedence* Default
☐ Apply Call Blocking Percentage
Resource Priority Namespace Network Domain < None >
Route Class* Default
Gateway/Route List* ATT_SIP_TRUNK (Edit)
Route Option ● Route this pattern
○ Block this pattern No Error
Call Classification* OffNet
External Call Control Profile < None >
☐ Allow Device Override ☑ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level* 0
☐ Require Client Matter Code

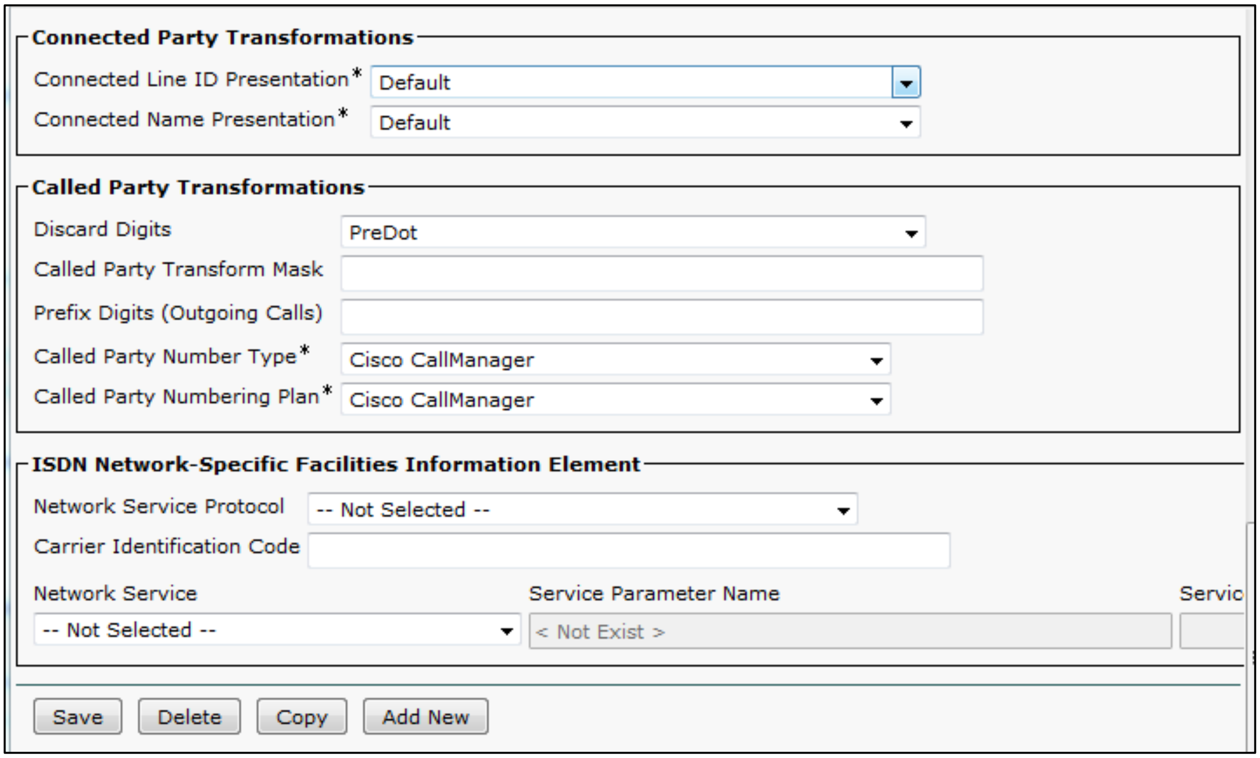


Calling Party Transformations

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

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

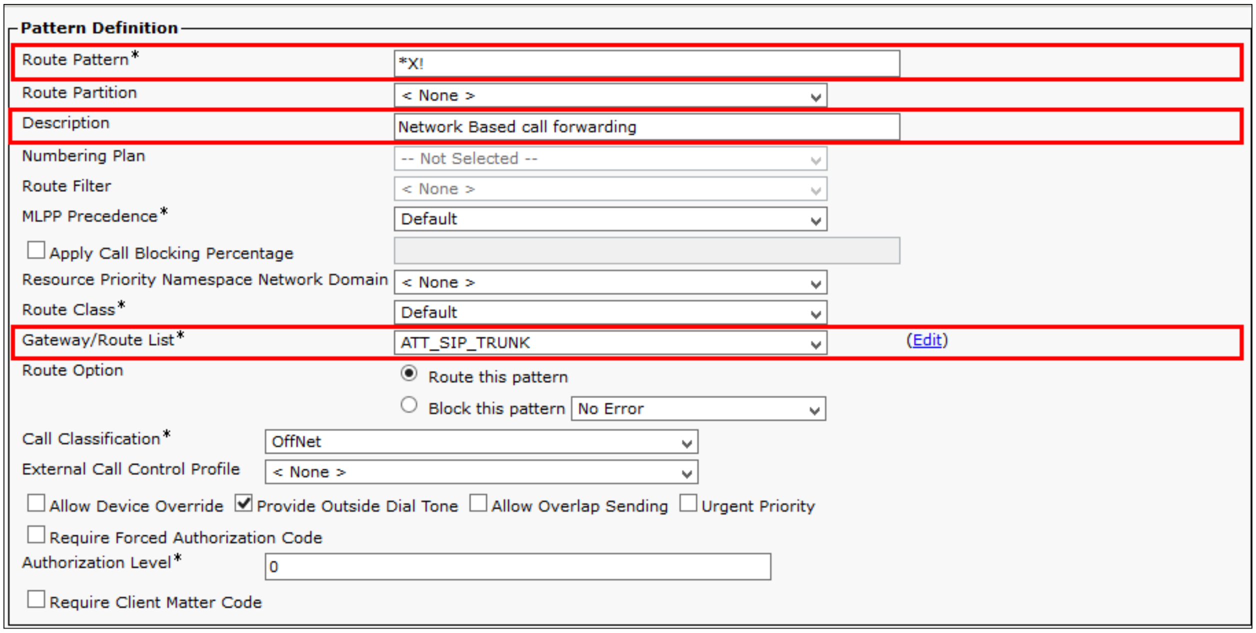
Network Service -- Not Selected --

Service Parameter Name < Not Exist >

Route Pattern Configuration (Continued…)

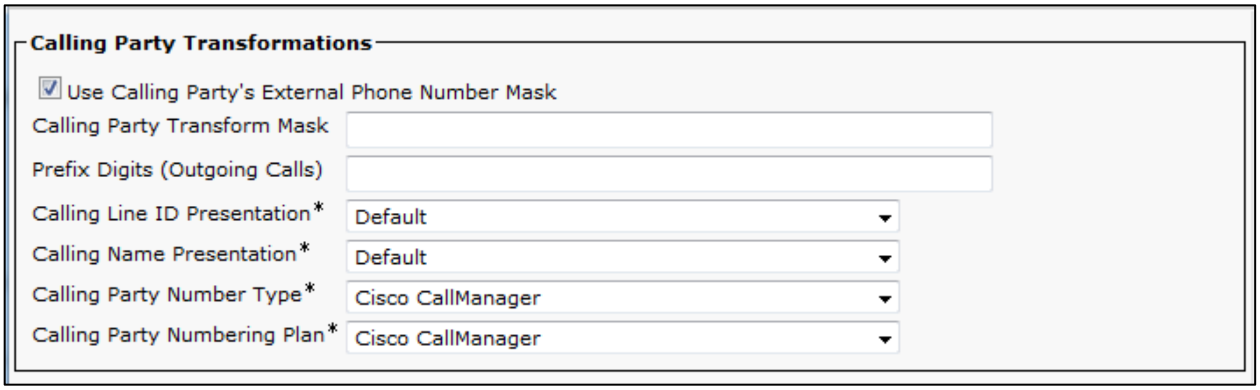
Set Route Pattern* = *X!   This is used to route to AT&T via ASR Cisco UBE.
Set Description = Network-Based Call Forwarding. This text is used to identify this Route Pattern.
Set Gateway/Route List* = ATT_SIP_TRUNK. This is used for this example.
All other values are default

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



Pattern Definition

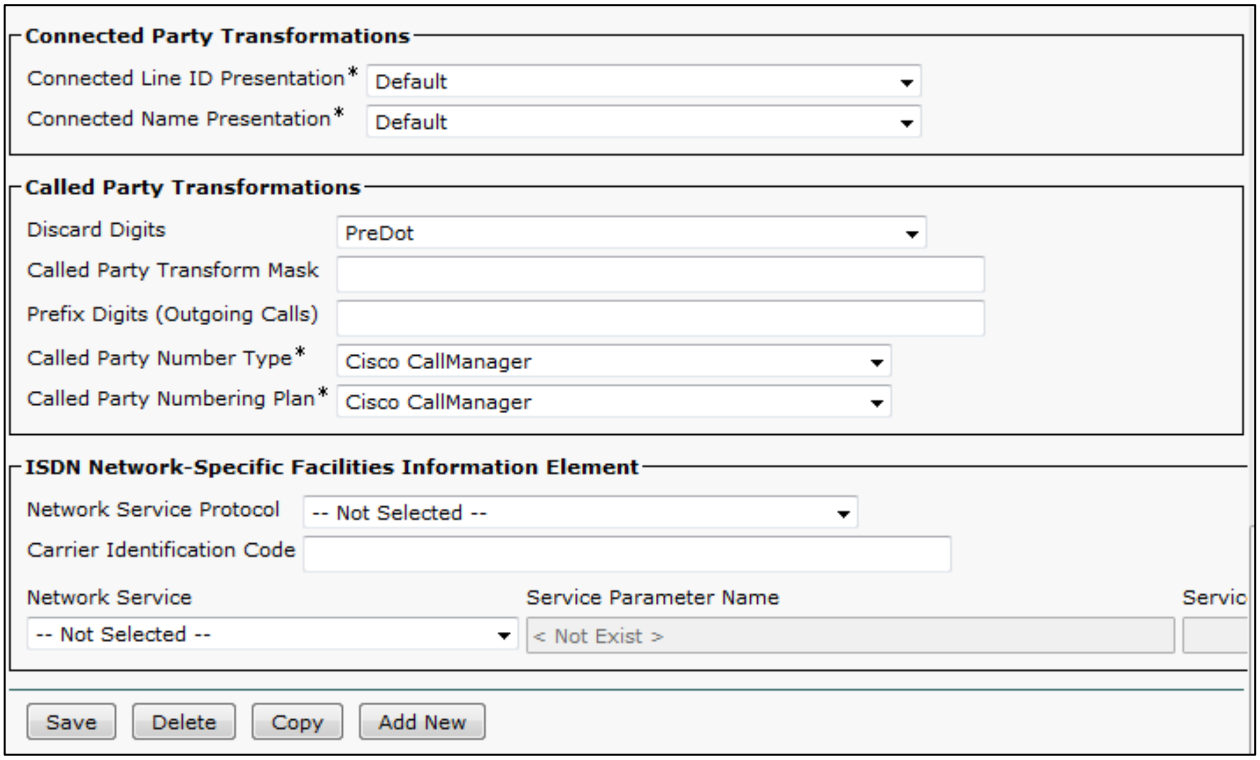
Route Pattern* *X!
Route Partition < None >
Description Network Based call forwarding
Numbering Plan -- Not Selected --
Route Filter < None >
MLPP Precedence* Default
☐ Apply Call Blocking Percentage
Resource Priority Namespace Network Domain < None >
Route Class* Default
Gateway/Route List* ATT_SIP_TRUNK (Edit)
Route Option
● Route this pattern
○ Block this pattern No Error
Call Classification* OffNet
External Call Control Profile < None >
☐ Allow Device Override ☑ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level* 0
☐ Require Client Matter Code



Calling Party Transformations

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

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 --

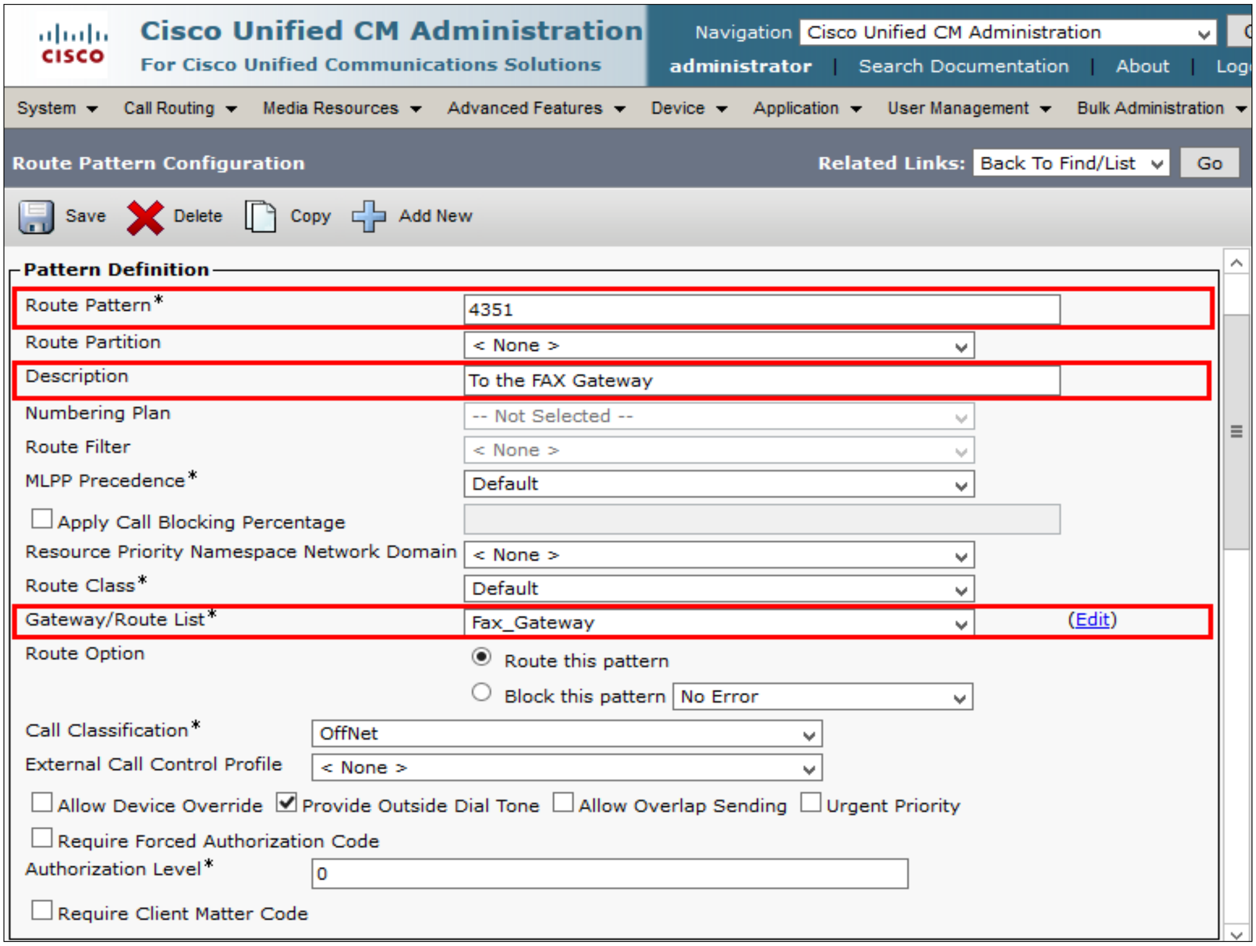
Carrier Identification Code

Network Service -- Not Selected --

Service Parameter Name < Not Exist >

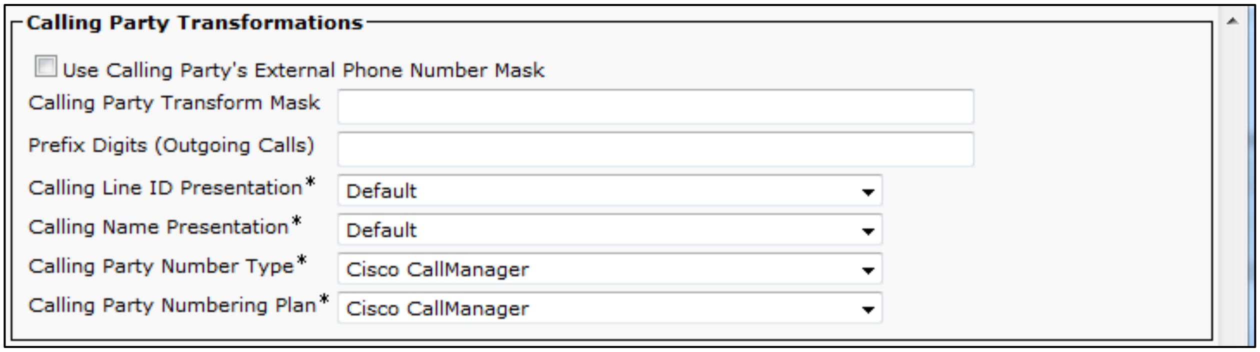
Route Pattern Configuration (Continued…)

Set Route Pattern* = 4351 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



Pattern Definition

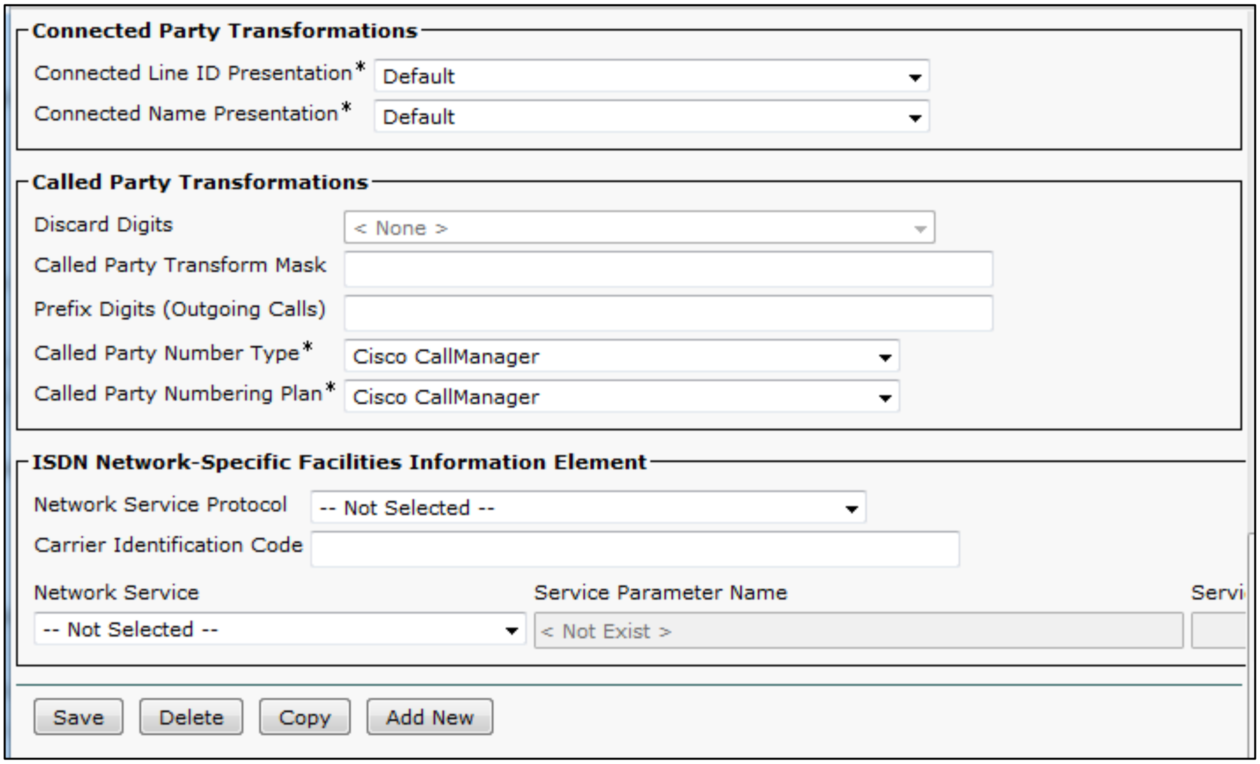
Route Pattern* 4351
Route Partition < None >
Description To the FAX Gateway
Numbering Plan -- Not Selected --
Route Filter < None >
MLPP Precedence* Default
☐ Apply Call Blocking Percentage
Resource Priority Namespace Network Domain < None >
Route Class* Default
Gateway/Route List* Fax_Gateway (Edit)
Route Option ● Route this pattern
○ Block this pattern No Error
Call Classification* OffNet
External Call Control Profile < None >
☐ Allow Device Override ☑ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level* 0
☐ Require Client Matter Code



Calling Party Transformations

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

Route Pattern Configuration (Continued…)



Connected Party Transformations

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Called Party Transformations

Discard Digits < None >

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type* Cisco CallManager

Called Party Numbering Plan* Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected --

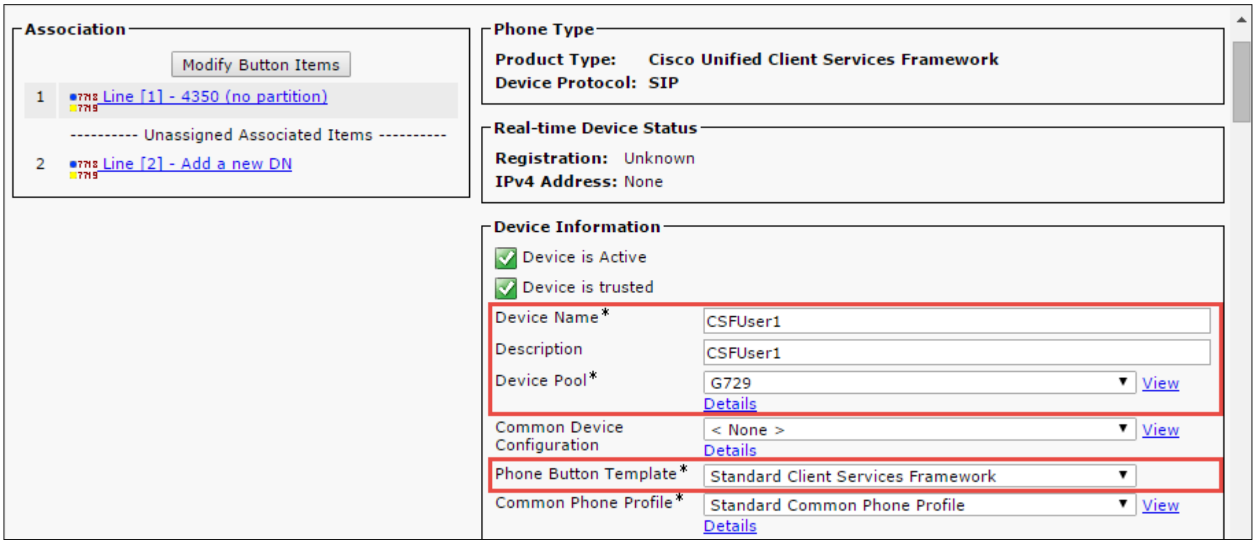
Carrier Identification Code

Network Service	Service Parameter Name	Servi
-- Not Selected --	< Not Exist >	

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.



Phone Type

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

Real-time Device Status

Registration: Unknown
IPv4 Address: None

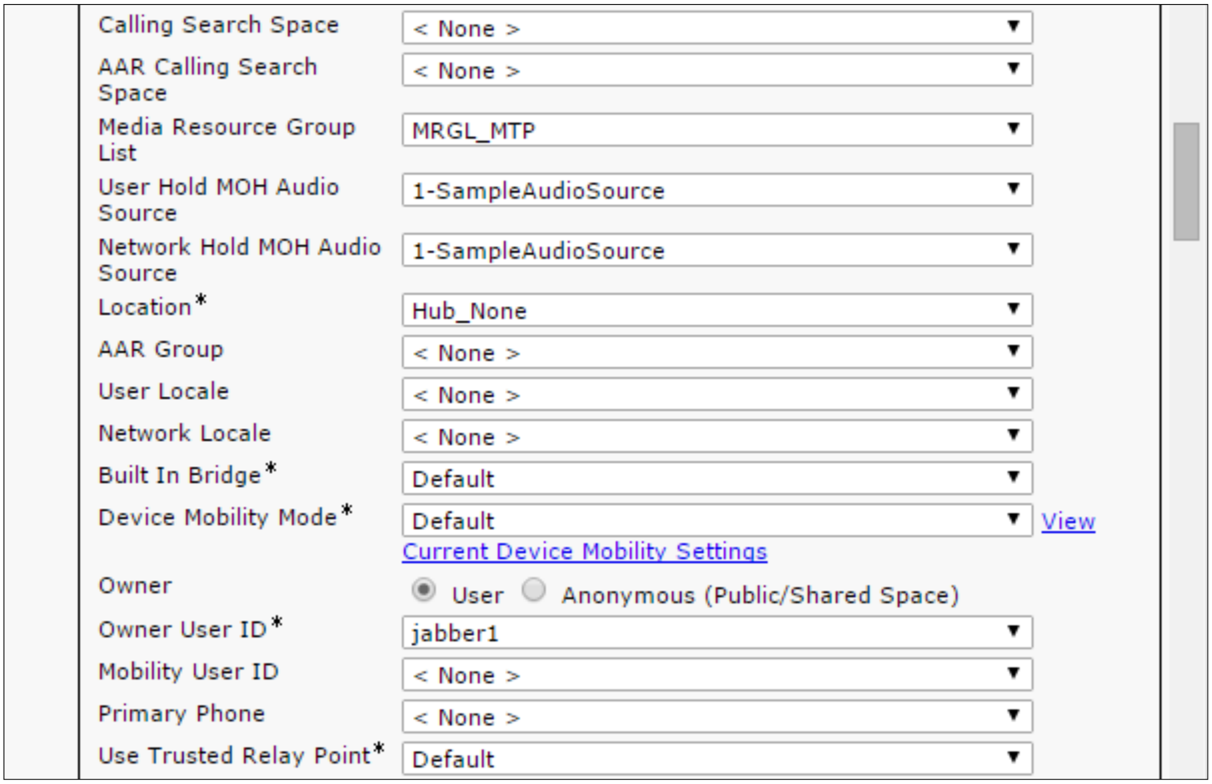
Device Information

☑ Device is Active
☑ Device is trusted

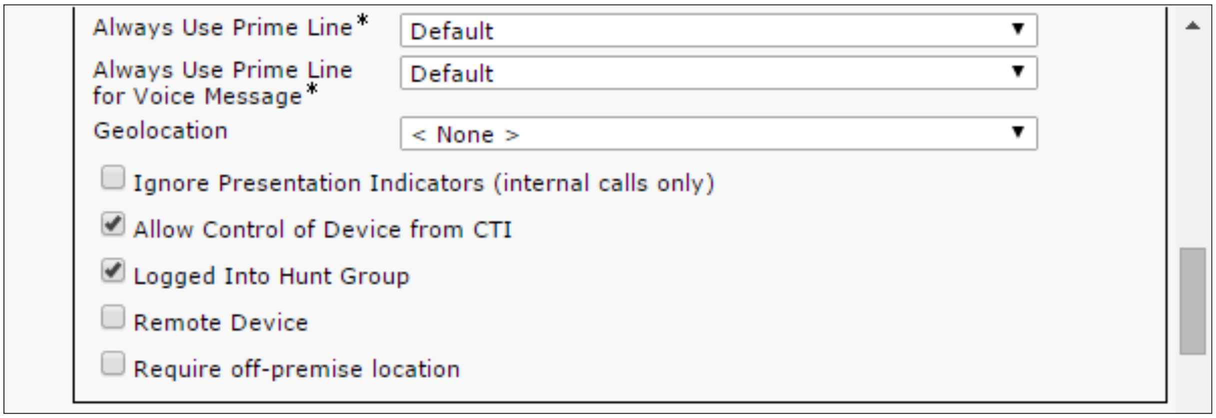
Device Name* CSFUser1
Description CSFUser1
Device Pool* G729 View Details
Common Device Configuration < None > View Details
Phone Button Template* Standard Client Services Framework
Common Phone Profile* Standard Common Phone Profile View Details

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



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



Always Use Prime Line*	Default
Always Use Prime Line for Voice Message*	Default
Geolocation	< None >

☐ Ignore Presentation Indicators (internal calls only)
☑ Allow Control of Device from CTI
☑ Logged Into Hunt Group
☐ Remote Device
☐ Require off-premise location

Jabber Client Configuration (Contd…)

Number Presentation Transformation

Caller ID For Calls From This Phone

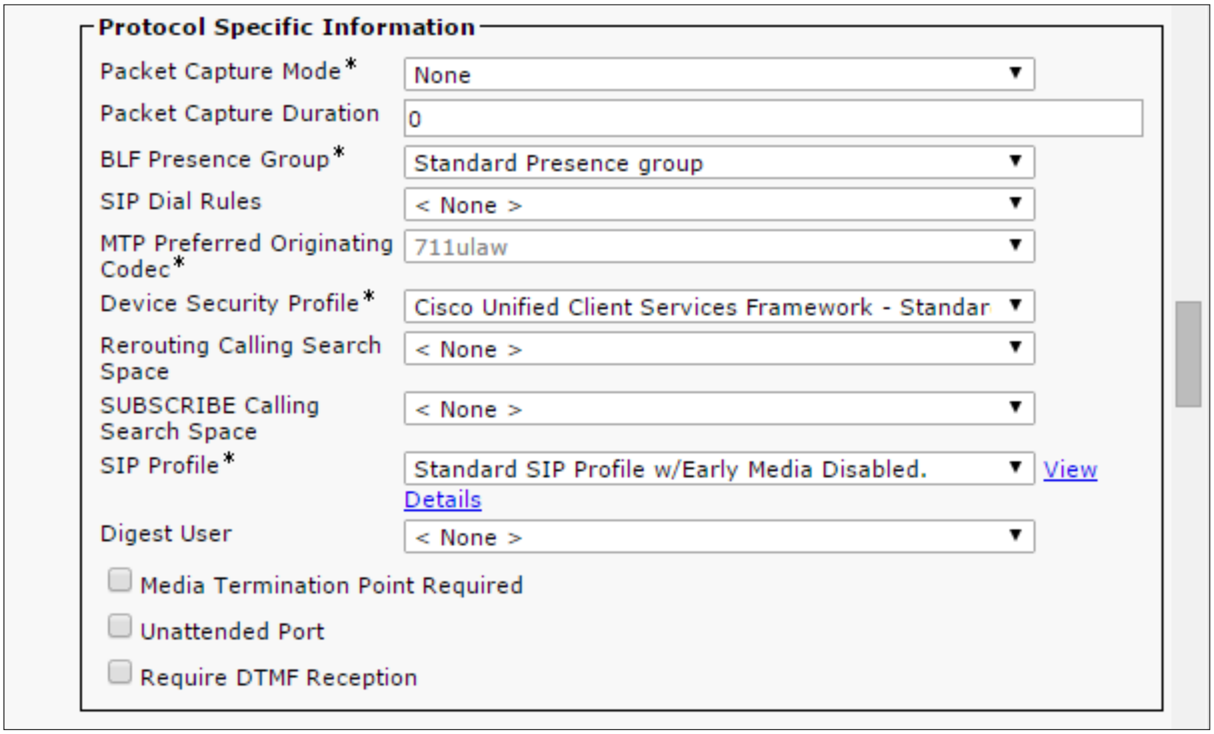
Calling Party Transformation CSS: < None >

☑ 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 Unified Client Services Framework - Standar

Rerouting Calling Search Space: < None >

SUBSCRIBE Calling Search Space: < None >

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

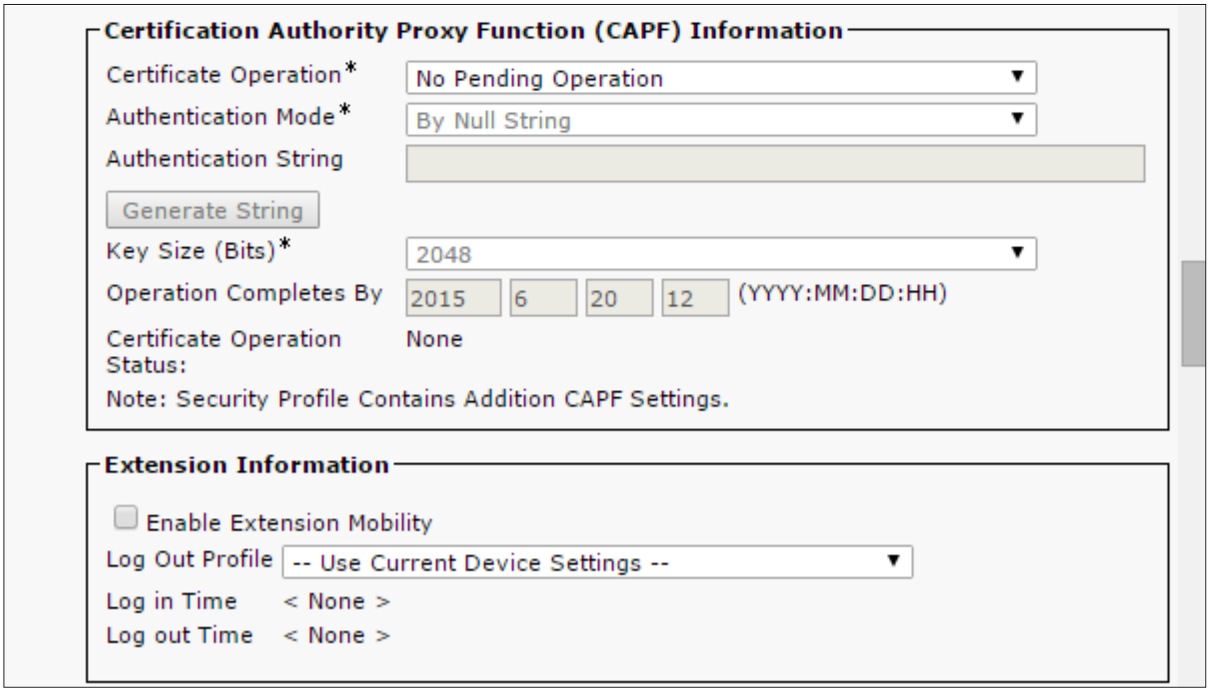
Digest User: < None >

☐ Media Termination Point Required

☐ Unattended Port

☐ Require DTMF Reception

Jabber Client Configuration (Contd…)



Certification Authority Proxy Function (CAPF) Information

Certificate Operation*: No Pending Operation

Authentication Mode*: By Null String

Authentication String:

Key Size (Bits)*: 2048

Operation Completes By: 2015 6 20 12 (YYYY:MM:DD:HH)

Certificate Operation Status: None

Note: Security Profile Contains Addition CAPF Settings.

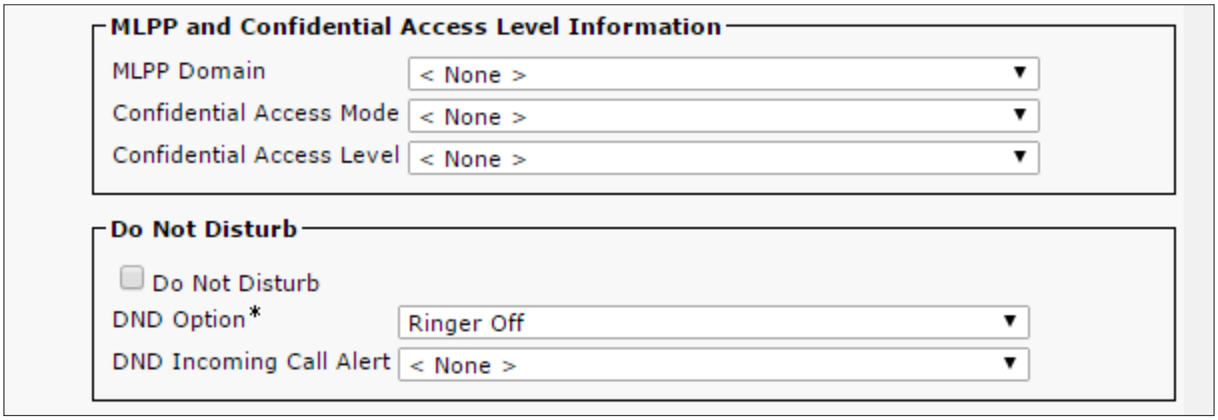
Extension Information

☐ Enable Extension Mobility

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

Log in Time: < None >

Log out Time: < None >



MLPP and Confidential Access Level Information

MLPP Domain: < None >

Confidential Access Mode: < None >

Confidential Access Level: < None >

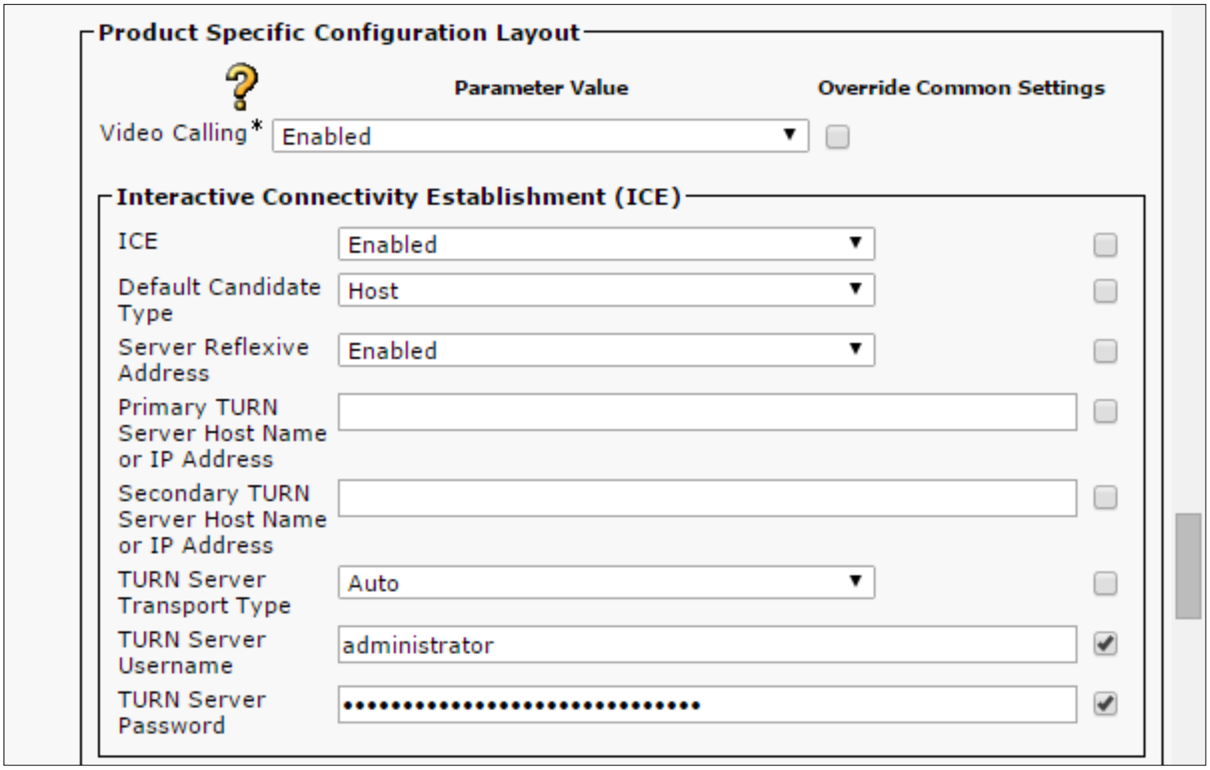
Do Not Disturb

☐ Do Not Disturb

DND Option*: Ringer Off

DND Incoming Call Alert: < None >

Jabber Client Configuration (Contd…)

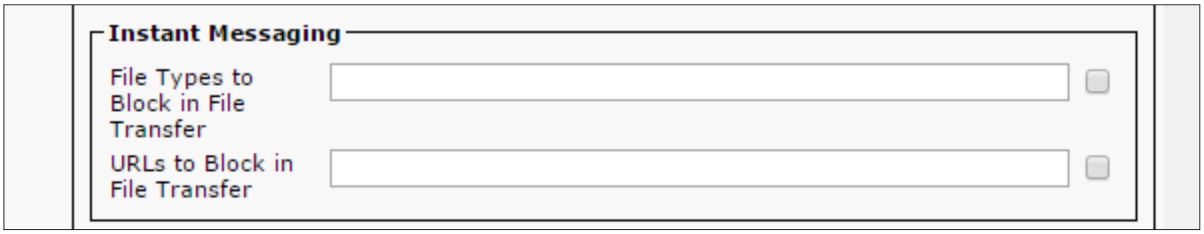


Product Specific Configuration Layout

	Parameter Value	Override Common Settings
Video Calling*	Enabled	☐

Interactive Connectivity Establishment (ICE)

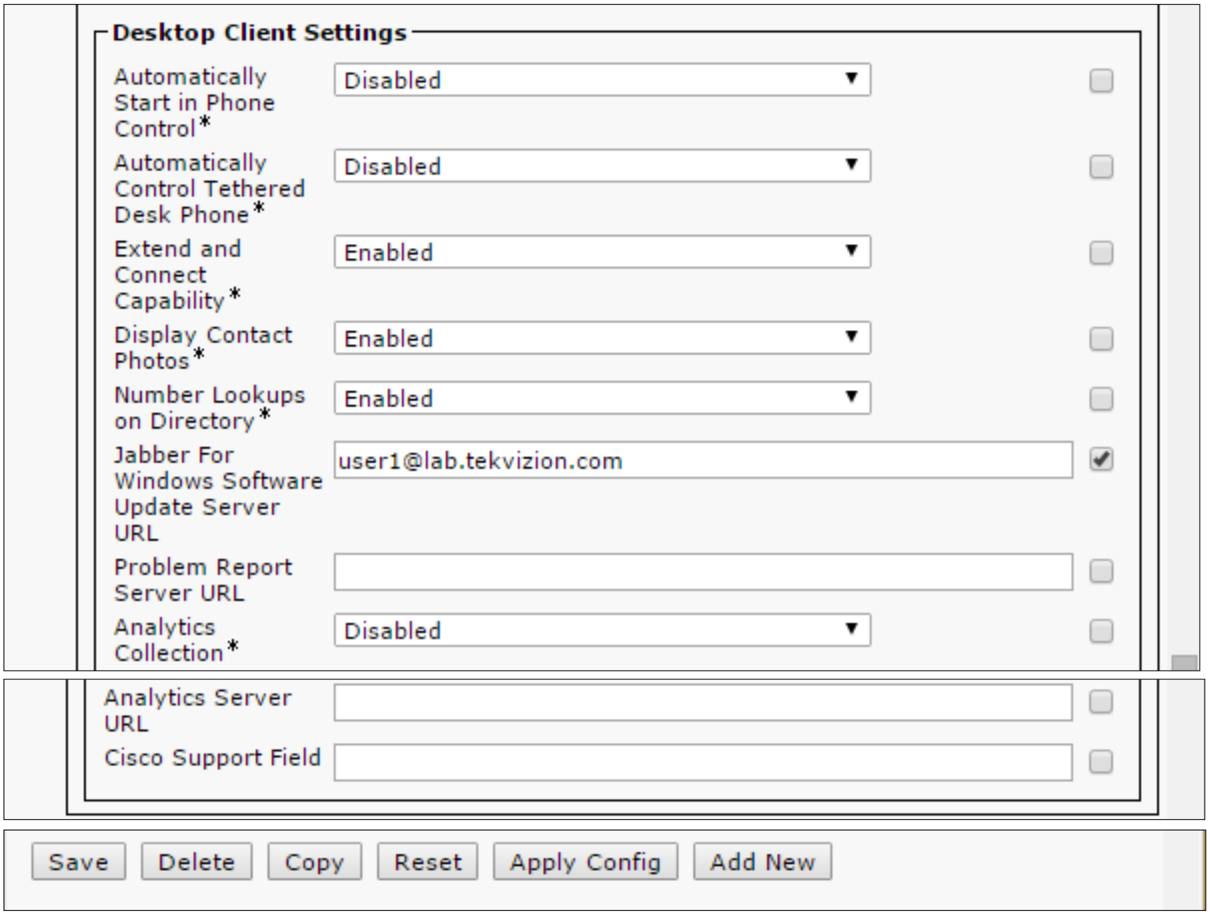
	Parameter Value	Override Common Settings
ICE	Enabled	☐
Default Candidate Type	Host	☐
Server Reflexive Address	Enabled	☐
Primary TURN Server Host Name or IP Address		☐
Secondary TURN Server Host Name or IP Address		☐
TURN Server Transport Type	Auto	☐
TURN Server Username	administrator	☑
TURN Server Password	••••••••••••••••••••••••••••••	☑



Instant Messaging

File Types to Block in File Transfer		☐
URLs to Block in File Transfer		☐

Jabber Client Configuration (Contd…)

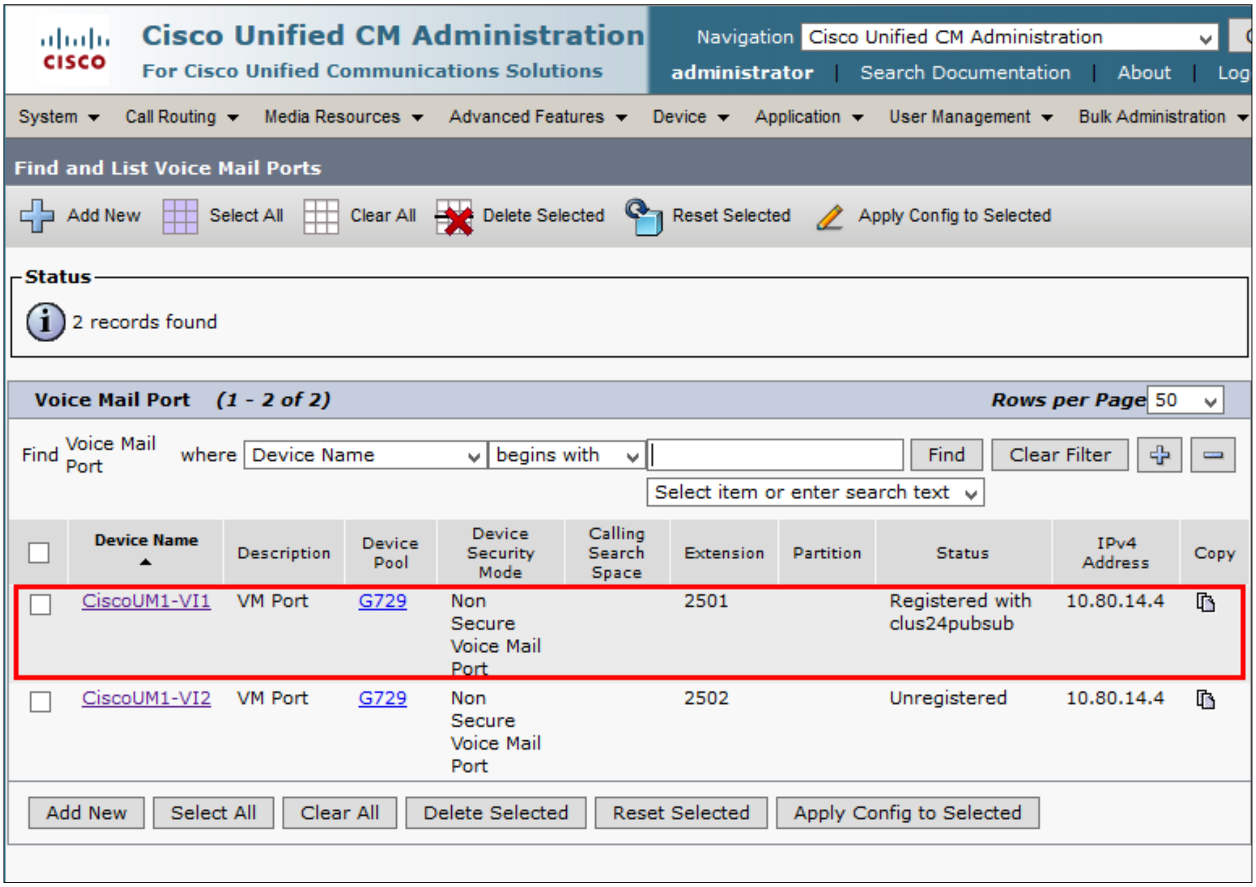


Desktop Client Settings

Automatically Start in Phone Control*	Disabled ▼	☐
Automatically Control Tethered Desk Phone*	Disabled ▼	☐
Extend and Connect Capability*	Enabled ▼	☐
Display Contact Photos*	Enabled ▼	☐
Number Lookups on Directory*	Enabled ▼	☐
Jabber For Windows Software Update Server URL	user1@lab.tekvizion.com	☑
Problem Report Server URL		☐
Analytics Collection*	Disabled ▼	☐
Analytics Server URL		☐
Cisco Support Field		☐

Voicemail Port Configuration

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



Status

2 records found

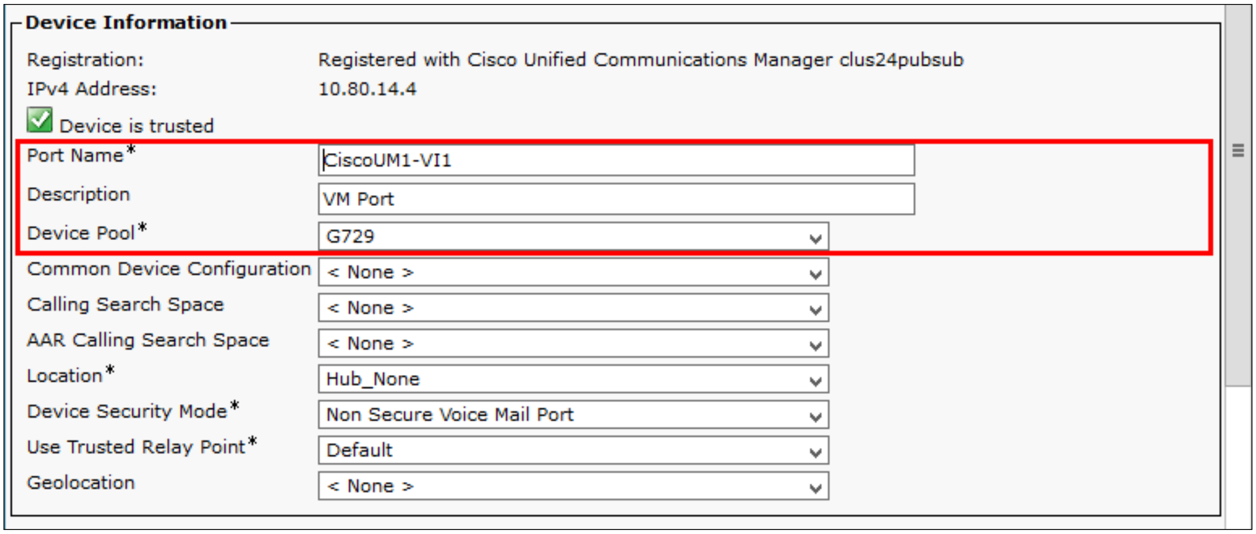
Voice Mail Port *(1 - 2 of 2)* *Rows per Page* 50

Find Voice Mail Port where Device Name begins with

☐	Device Name	Description	Device Pool	Device Security Mode	Calling Search Space	Extension	Partition	Status	IPv4 Address	Copy
☐	CiscoUM1-VI1	VM Port	G729	Non Secure Voice Mail Port		2501		Registered with clus24pubsub	10.80.14.4	
☐	CiscoUM1-VI2	VM Port	G729	Non Secure Voice Mail Port		2502		Unregistered	10.80.14.4	

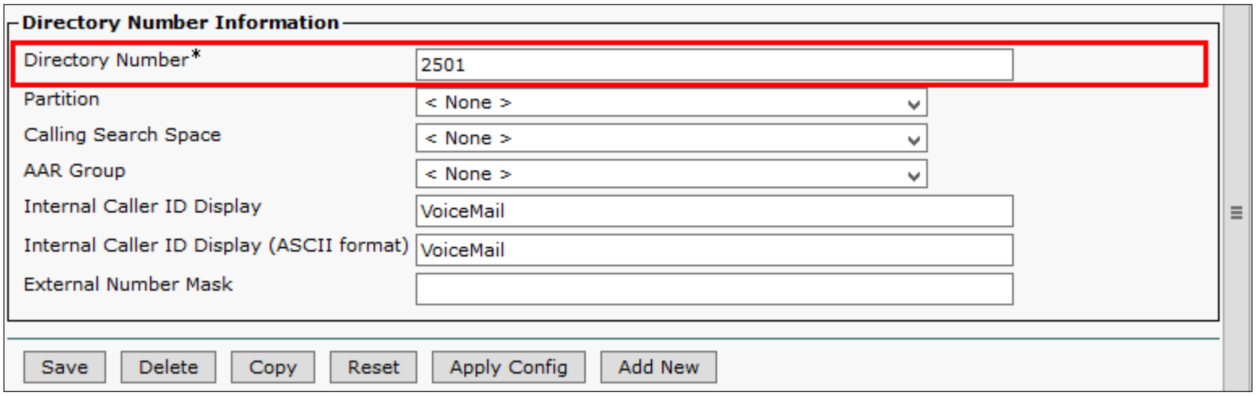
Voicemail Port Configuration (Continued…)

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* = 2501. This is used in this example.



Device Information

Registration: Registered with Cisco Unified Communications Manager clus24pubsub
IPv4 Address: 10.80.14.4
☑ Device is trusted
Port Name*: CiscoUM1-VI1
Description: VM Port
Device Pool*: G729
Common Device Configuration: < None >
Calling Search Space: < None >
AAR Calling Search Space: < None >
Location*: Hub_None
Device Security Mode*: Non Secure Voice Mail Port
Use Trusted Relay Point*: Default
Geolocation: < None >



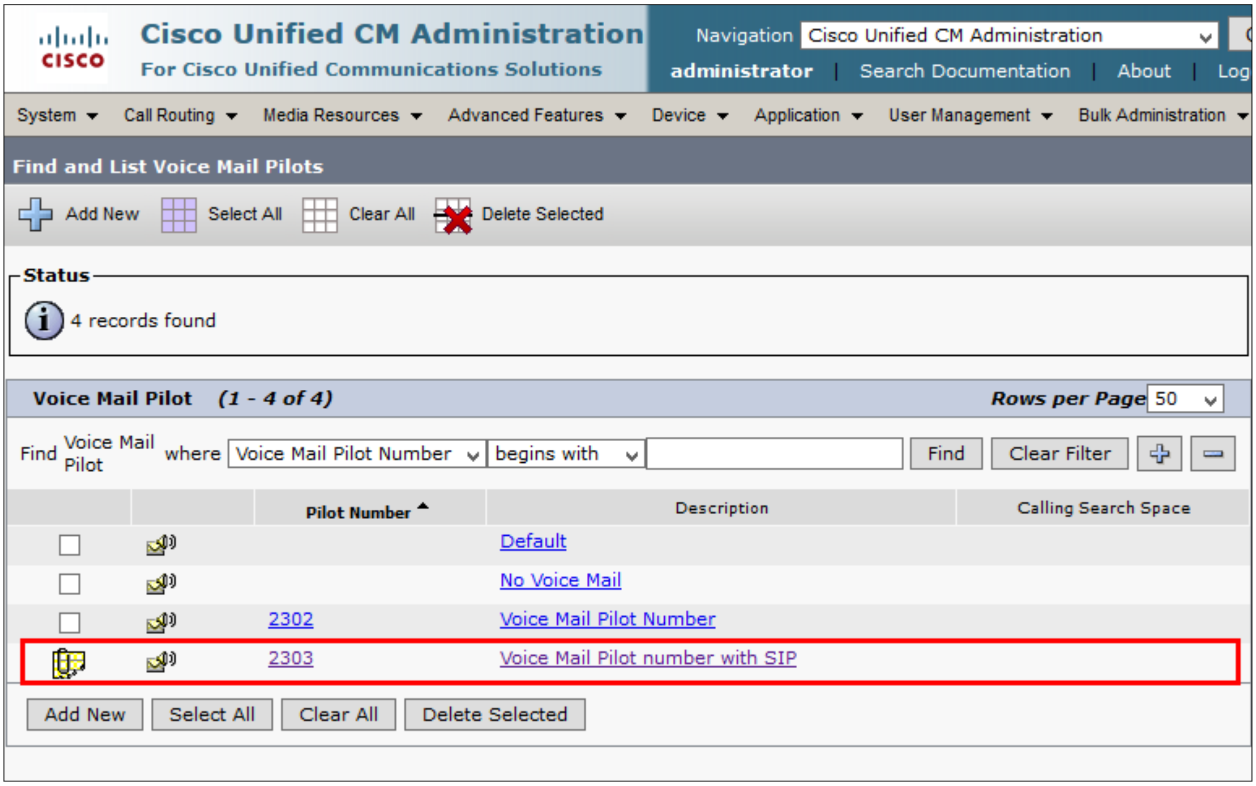
Directory Number Information

Directory Number*: 2501
Partition: < None >
Calling Search Space: < None >
AAR Group: < None >
Internal Caller ID Display: VoiceMail
Internal Caller ID Display (ASCII format): VoiceMail
External Number Mask:

Voicemail Pilot Configuration

Navigation: Advanced Features → Voice Mail → Voice Mail Pilot

Set Voice mail Pilot Number = 2303. This is used for this example
Set Description = VoiceMail Pilot number with SIP

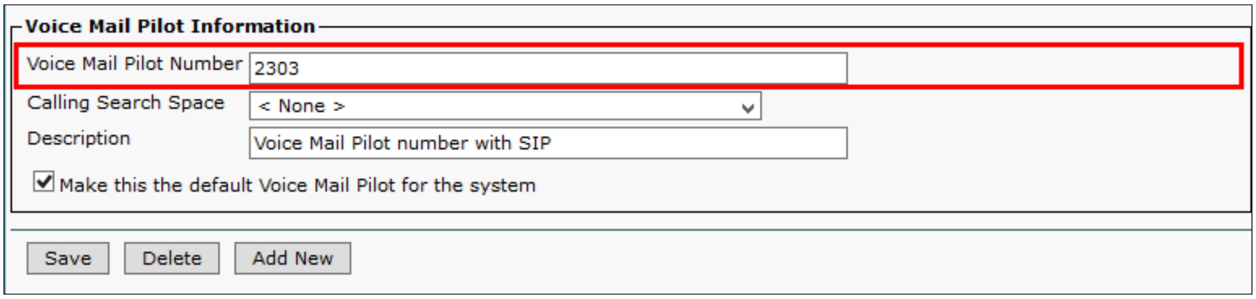


Status

4 records found

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

		Pilot Number ▲	Description	Calling Search Space
☐			Default	
☐			No Voice Mail	
☐		2302	Voice Mail Pilot Number	
		2303	Voice Mail Pilot number with SIP	



Voice Mail Pilot Information

Voice Mail Pilot Number: 2303
Calling Search Space: < None >
Description: Voice Mail Pilot number with SIP
☑ Make this the default Voice Mail Pilot for the system

FAX Gateway Configuration

```
voice service voip
 no ip address trusted authenticate
 allow-connections sip to sip
 redirect ip2ip
 fax protocol pass-through g711ulaw
 no fax-relay sg3-to-g3
 sip
  midcall-signaling passthru
  g729 annexb-all

voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8



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

```
voice-port 0/0/1
 ring frequency 50
 no echo-cancel enable
 no vad
 cptone IN
 description **telephone analog/fax**
 station-id name fax test
 station-id number 4351
 caller-id enable

dial-peer voice 101 pots
 huntstop
 service session
 destination-pattern 4351
 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 4351
 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

 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
```

CUC Version

Version 10.5.2.10000-5

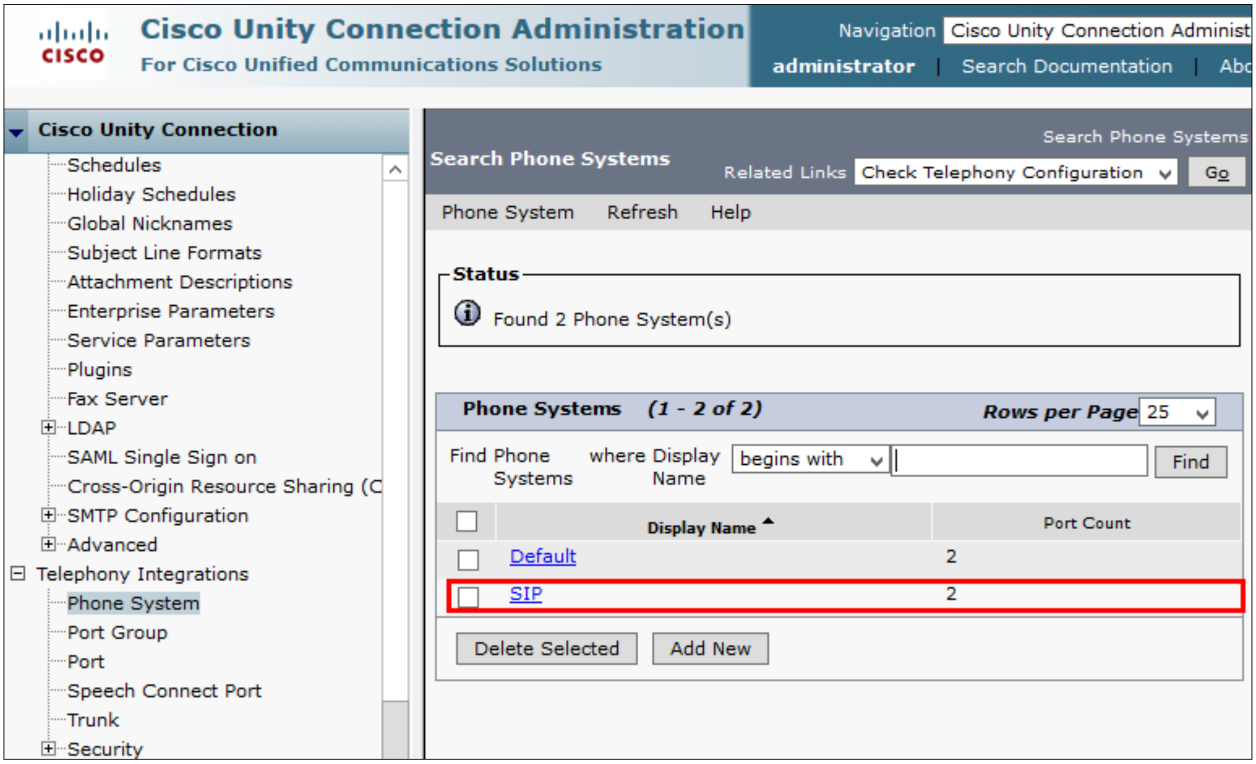
Copyright © 1999 - 2015 Cisco Systems, Inc.
All rights reserved.

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.

CUC Telephony Integration with Cisco UCM

Navigation: Telephony Integrations → Phone system

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



Status

Found 2 Phone System(s)

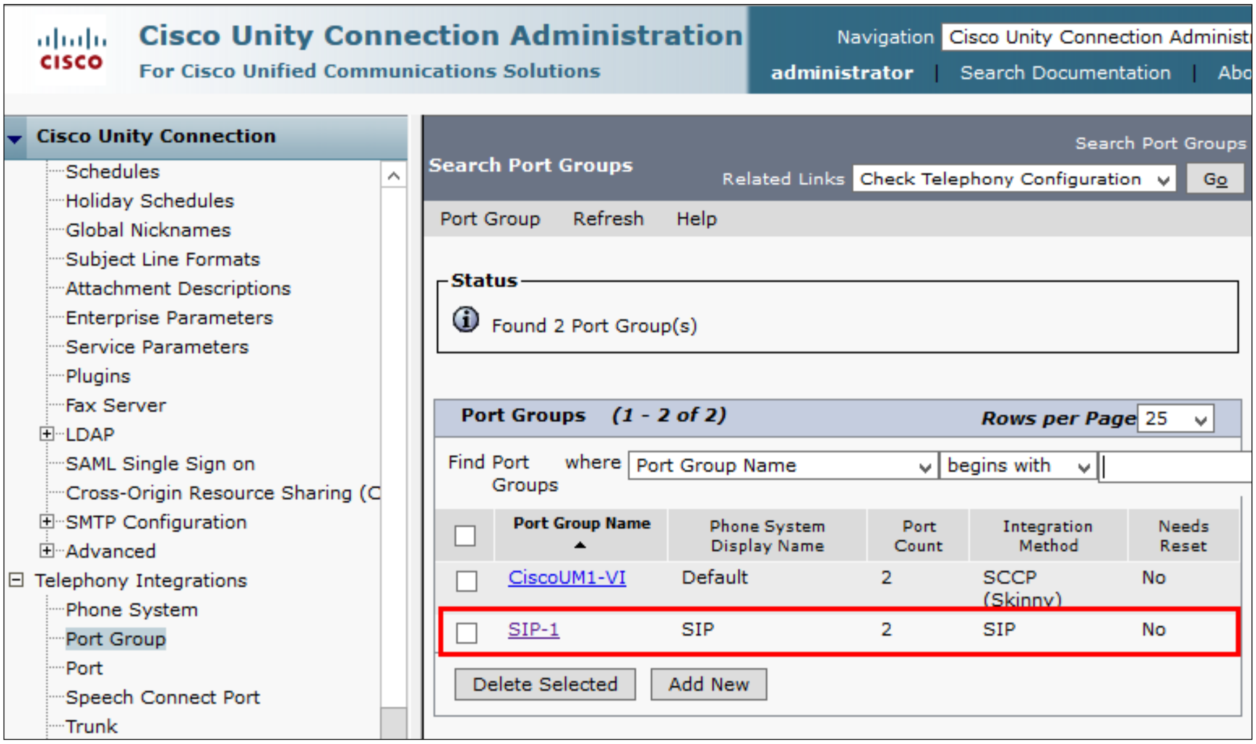
Phone Systems (1 - 2 of 2) Rows per Page 25

Find Phone Systems where Display Name begins with [ ] Find

☐	Display Name	Port Count
☐	Default	2
☐	SIP	2

CUC Port Group

Navigation: Telephony Integration → Port Group



Status

Found 2 Port Group(s)

Port Groups *(1 - 2 of 2)* **Rows per Page** 25

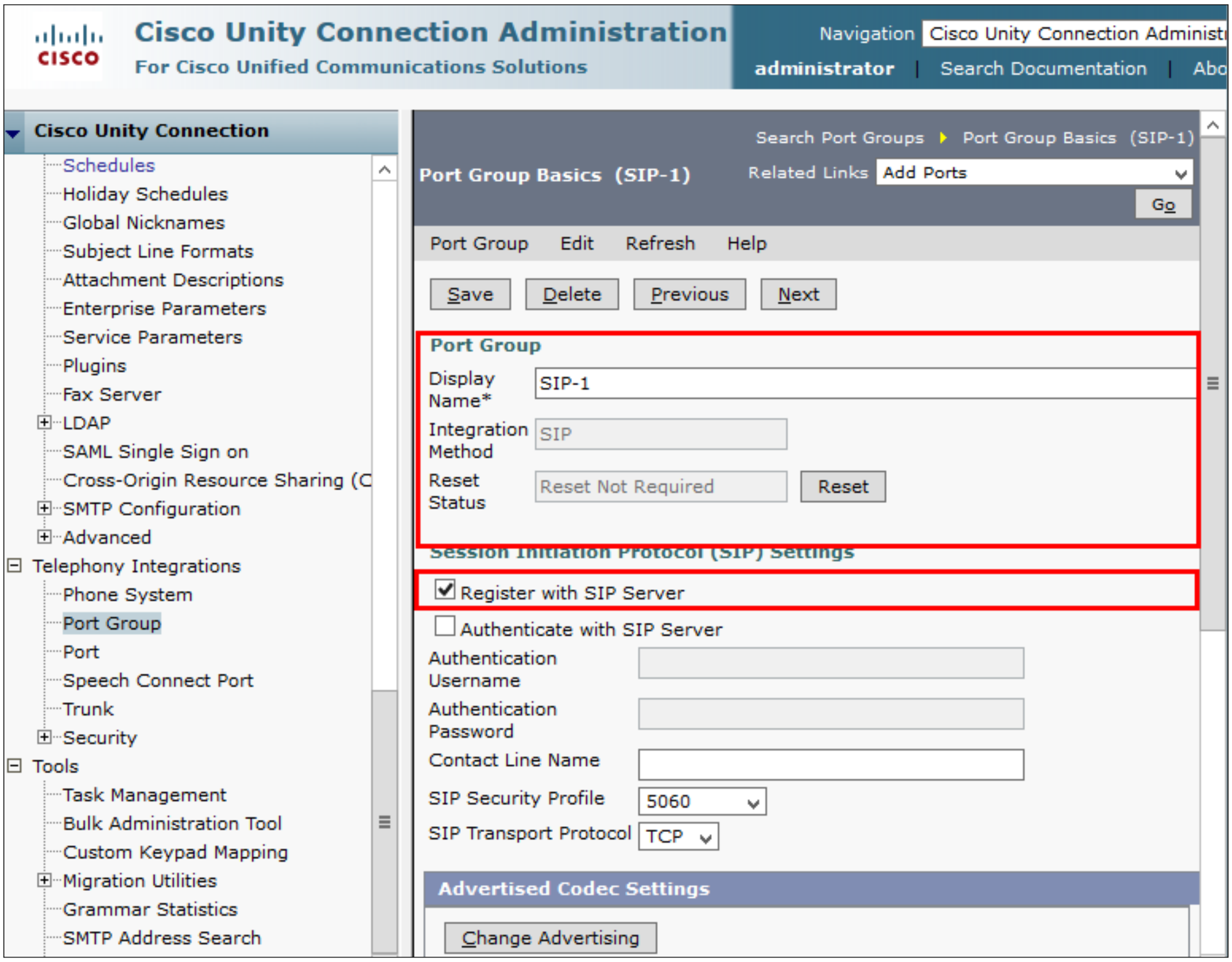
Find Port Groups where Port Group Name begins with

☐	Port Group Name	Phone System Display Name	Port Count	Integration Method	Needs Reset
☐	CiscoUM1-VI	Default	2	SCCP (Skinny)	No
☐	SIP-1	SIP	2	SIP	No

CUC Port Group(continued…)

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

Check Register with SIP Server.



Port Group Basics (SIP-1)

Port Group

Display Name*: SIP-1

Integration Method: SIP

Reset Status: Reset Not Required

Session Initiation Protocol (SIP) Settings

☑ Register with SIP Server

☐ Authenticate with SIP Server

Authentication Username:

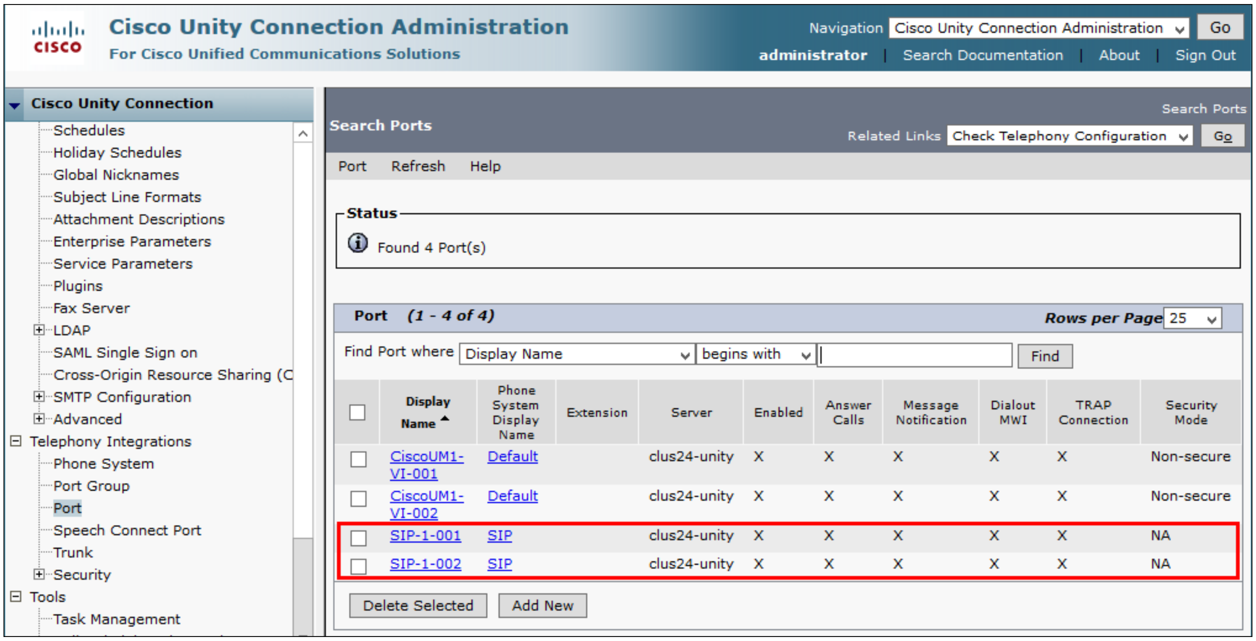
Authentication Password:

Contact Line Name:

SIP Security Profile: 5060

SIP Transport Protocol: TCP

CUC Port Settings



Search Ports

Status

Found 4 Port(s)

Port (1 - 4 of 4) — **Rows per Page** 25

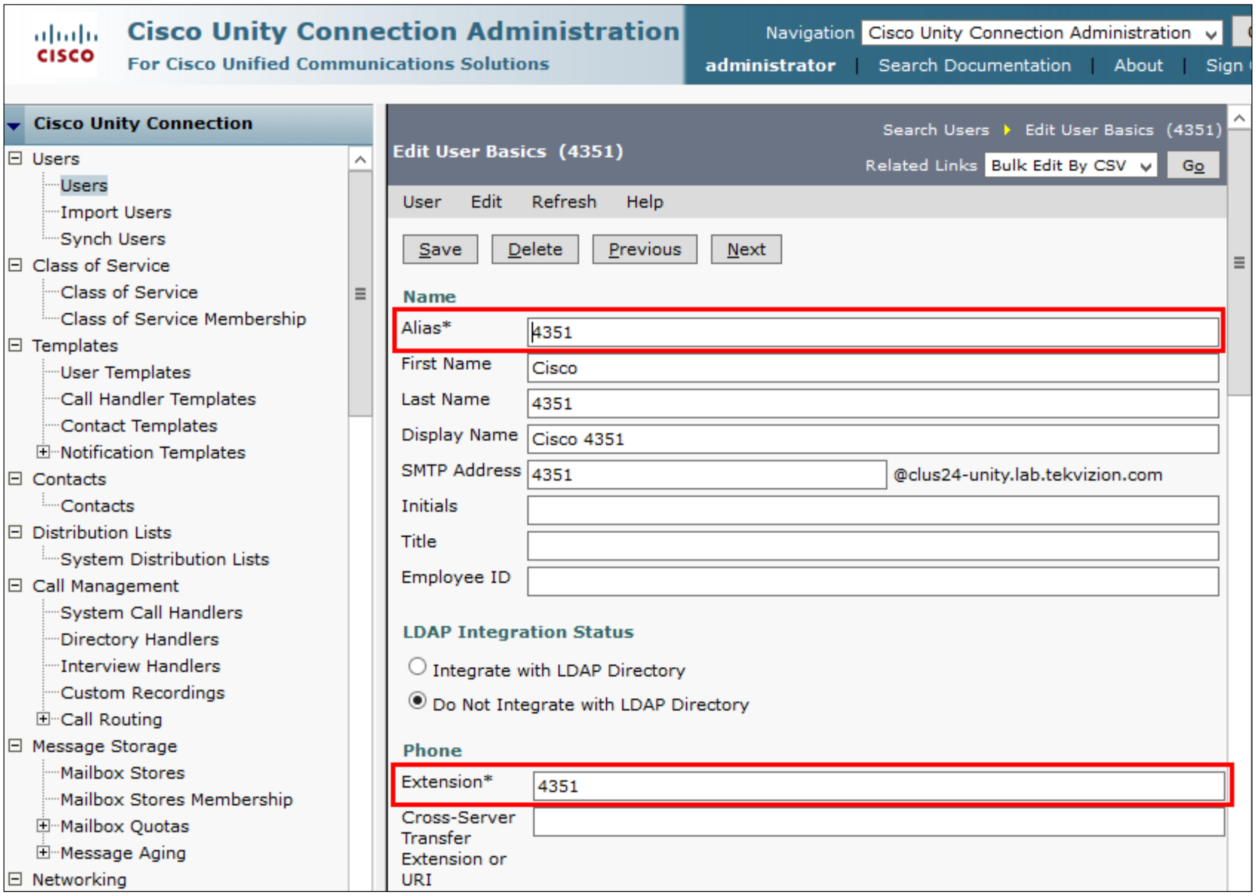
Find Port where Display Name begins with [ ] Find

☐	Display Name	Phone System Display Name	Extension	Server	Enabled	Answer Calls	Message Notification	Dialout MWI	TRAP Connection	Security Mode
☐	CiscoUM1-VI-001	Default		clus24-unity	X	X	X	X	X	Non-secure
☐	CiscoUM1-VI-002	Default		clus24-unity	X	X	X	X	X	Non-secure
☐	SIP-1-001	SIP		clus24-unity	X	X	X	X	X	NA
☐	SIP-1-002	SIP		clus24-unity	X	X	X	X	X	NA

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.



Edit User Basics (4351)

Name

Alias* 4351
First Name Cisco
Last Name 4351
Display Name Cisco 4351
SMTP Address 4351 @clus24-unity.lab.tekvizion.com
Initials
Title
Employee ID

LDAP Integration Status

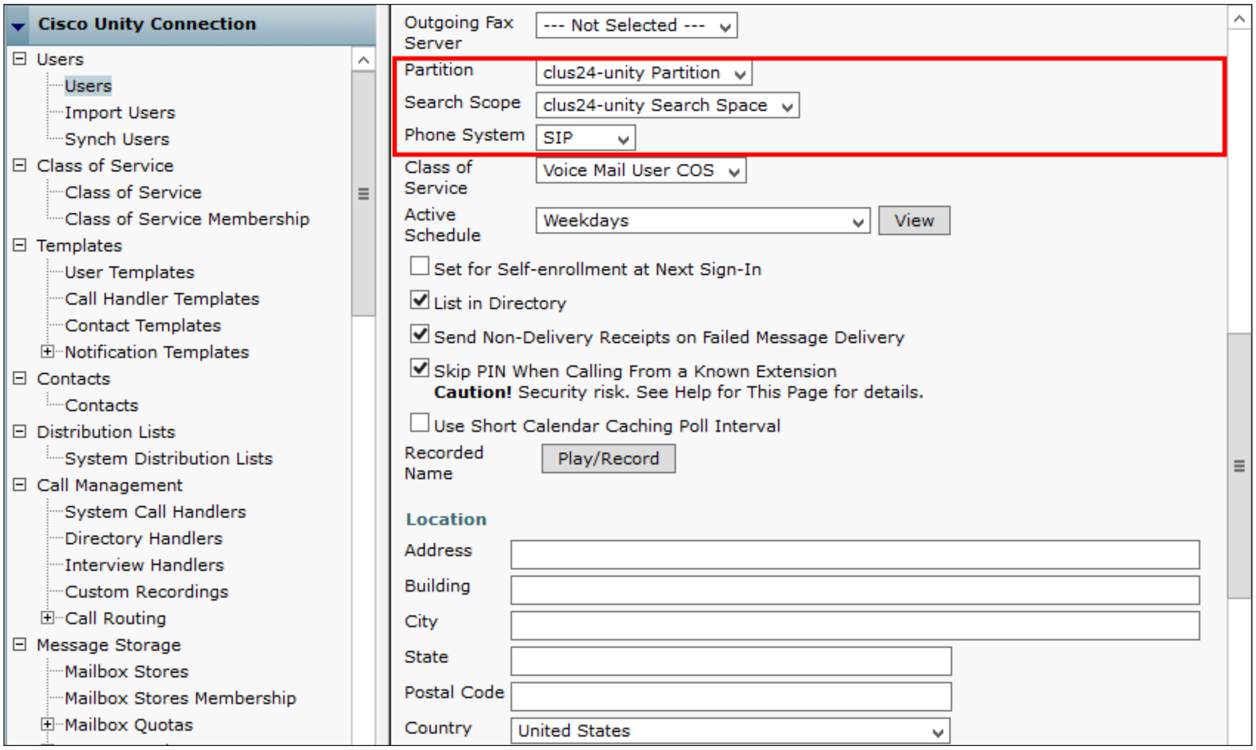
○ Integrate with LDAP Directory
● Do Not Integrate with LDAP Directory

Phone

Extension* 4351
Cross-Server Transfer Extension or URI

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.



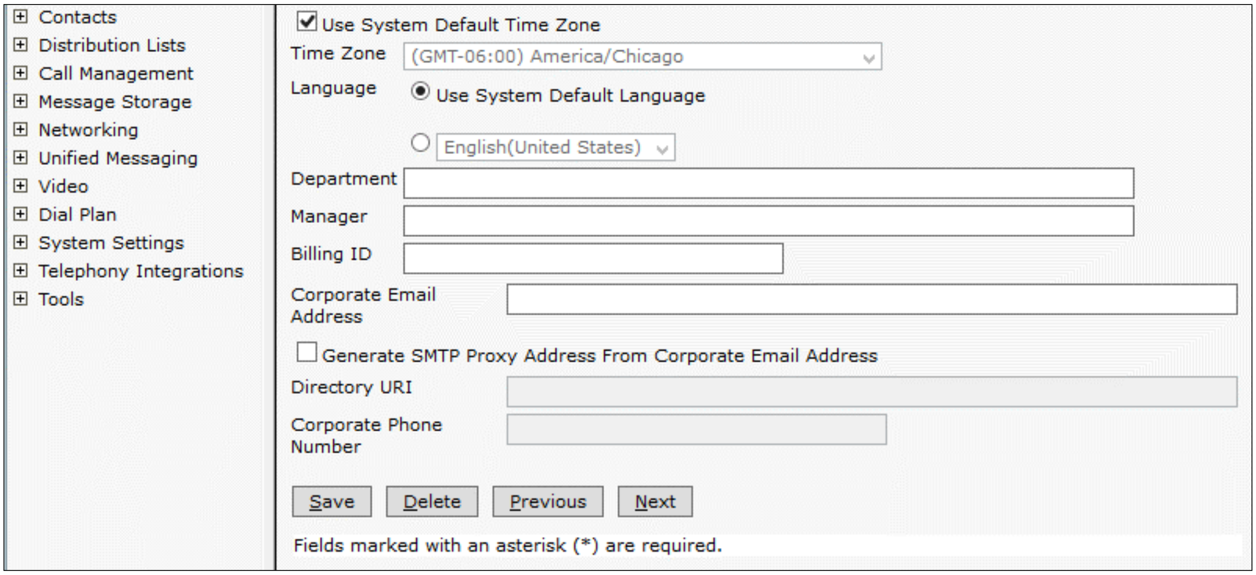
Outgoing Fax Server: --- Not Selected ---
Partition: clus24-unity Partition
Search Scope: clus24-unity Search Space
Phone System: SIP
Class of Service: Voice Mail User COS
Active Schedule: Weekdays

☐ Set for Self-enrollment at Next Sign-In
☑ List in Directory
☑ Send Non-Delivery Receipts on Failed Message Delivery
☑ Skip PIN When Calling From a Known Extension
Caution! Security risk. See Help for This Page for details.
☐ Use Short Calendar Caching Poll Interval

Recorded Name

Location

Address
Building
City
State
Postal Code
Country: United States



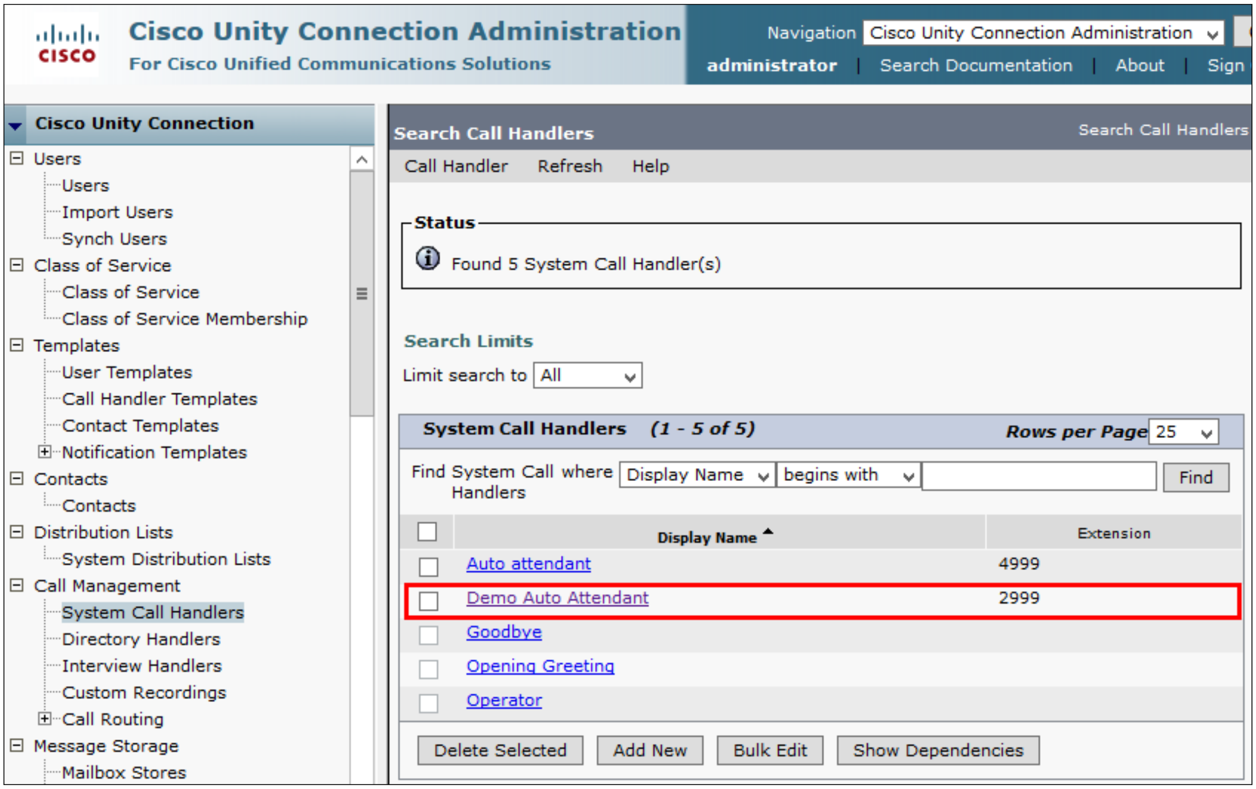
☑ Use System Default Time Zone
Time Zone: (GMT-06:00) America/Chicago
Language: ● Use System Default Language
○ English(United States)
Department
Manager
Billing ID
Corporate Email Address
☐ Generate SMTP Proxy Address From Corporate Email Address
Directory URI
Corporate Phone Number

Fields marked with an asterisk (*) are required.

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.



Found 5 System Call Handler(s)

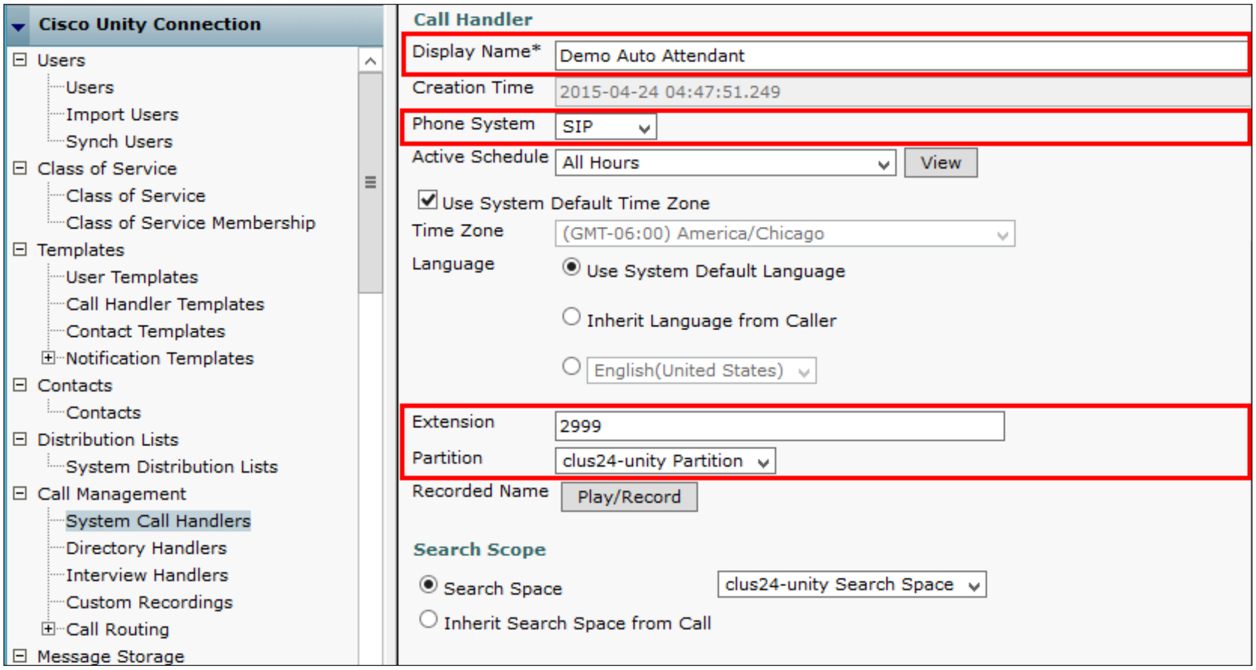
Limit search to: All

Rows per Page: 25

Find System Call Handlers where: Display Name / begins with

☐	Display Name	Extension
☐	Auto attendant	4999
☐	Demo Auto Attendant	2999
☐	Goodbye	
☐	Opening Greeting	
☐	Operator	

Auto Attendant (Continued…)



Call Handler

Display Name* Demo Auto Attendant

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

Phone System SIP

Active Schedule All Hours

☑ Use System Default Time Zone

Time Zone (GMT-06:00) America/Chicago

Language
● Use System Default Language
○ Inherit Language from Caller
○ English(United States)

Extension 2999

Partition clus24-unity Partition

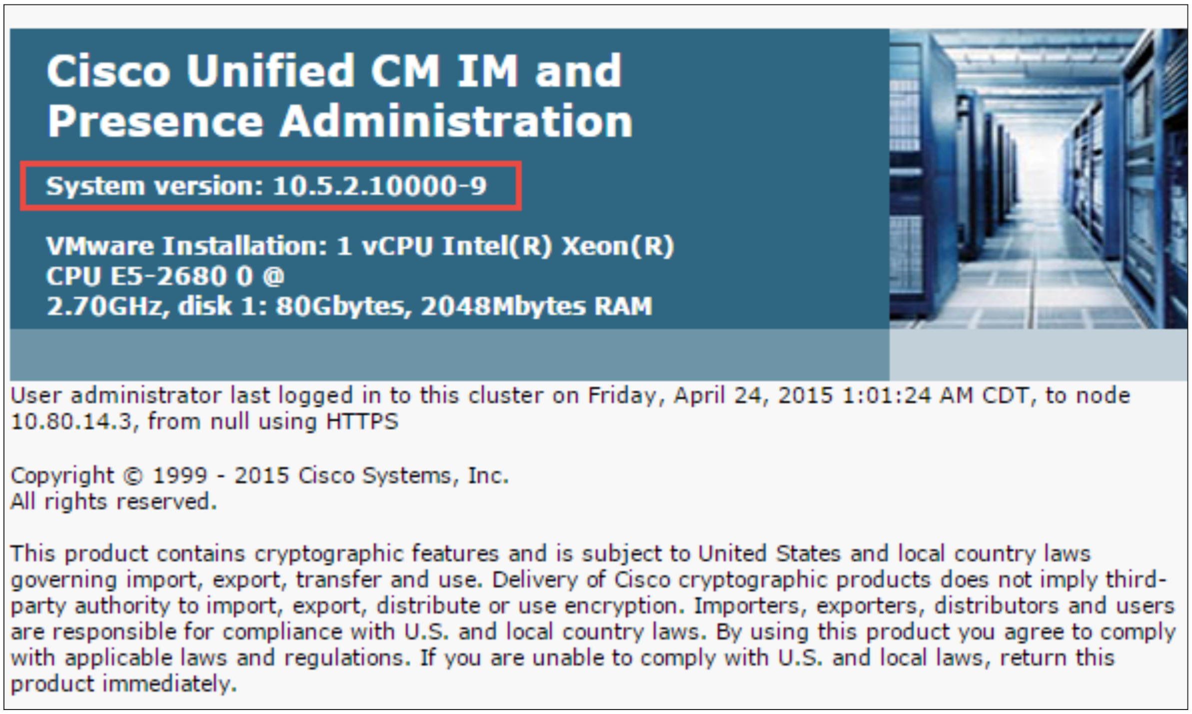
Recorded Name

Search Scope

● Search Space clus24-unity Search Space
○ Inherit Search Space from Call

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

CUP/IMP Version



Cisco Unified CM IM and Presence Administration

System version: 10.5.2.10000-9

VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz, disk 1: 80Gbytes, 2048Mbytes RAM

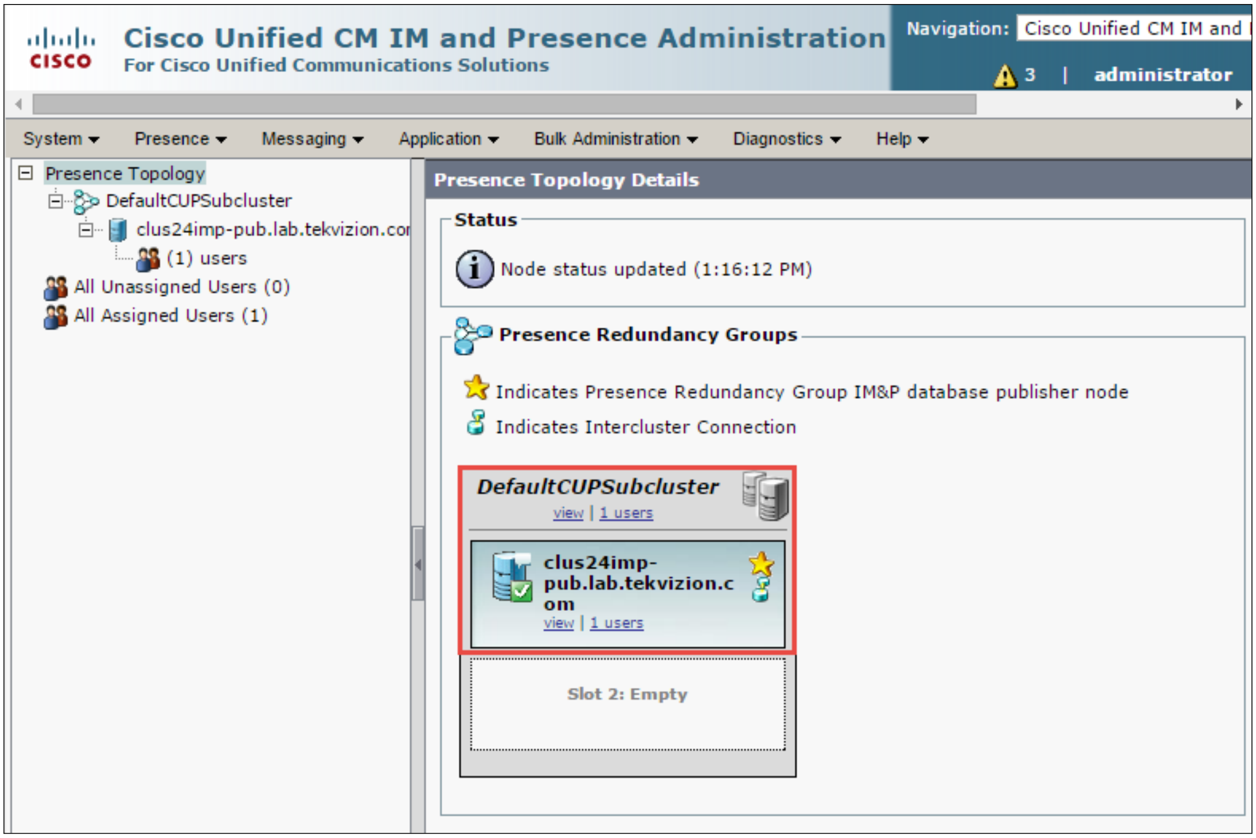
User administrator last logged in to this cluster on Friday, April 24, 2015 1:01:24 AM CDT, to node 10.80.14.3, from null using HTTPS

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.

Presence Topology

Navigation: System → Presence Topology



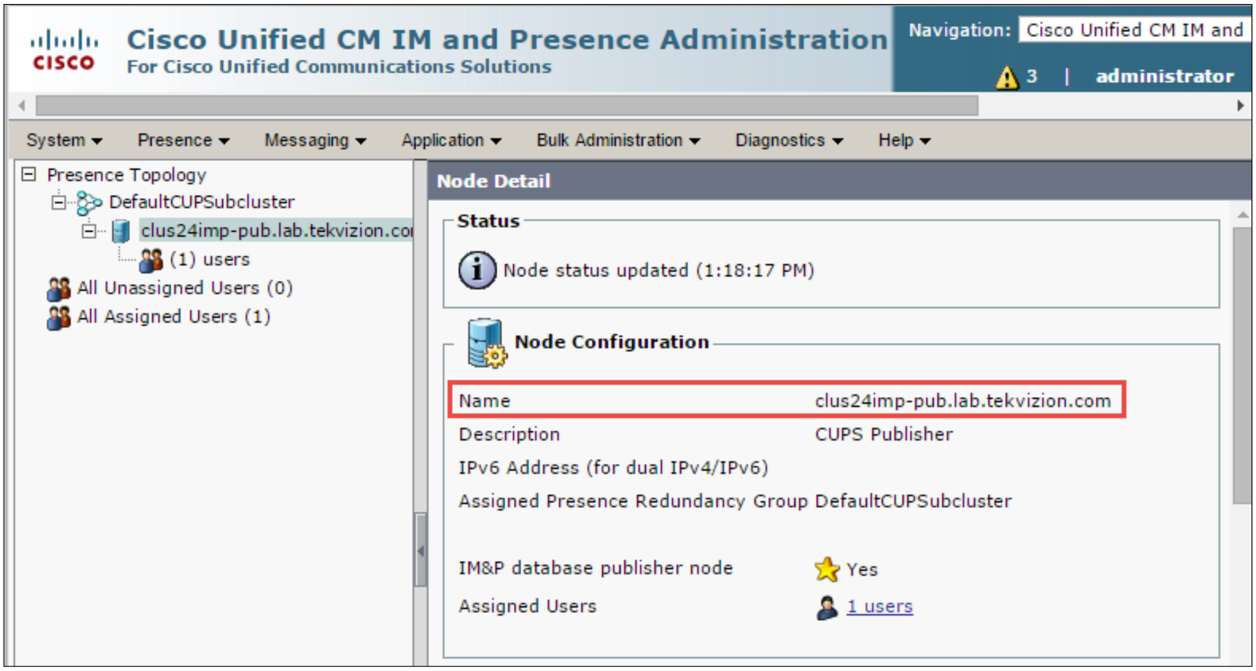
Node status updated (1:16:12 PM)

Indicates Presence Redundancy Group IM&P database publisher node

Indicates Intercluster Connection

Node Configuration

Navigation: System → Presence Topology → Fully Qualified Domain Name



Status

Node status updated (1:18:17 PM)

Node Configuration

Name clus24imp-pub.lab.tekvizion.com

Description CUPS Publisher

IPv6 Address (for dual IPv4/IPv6)

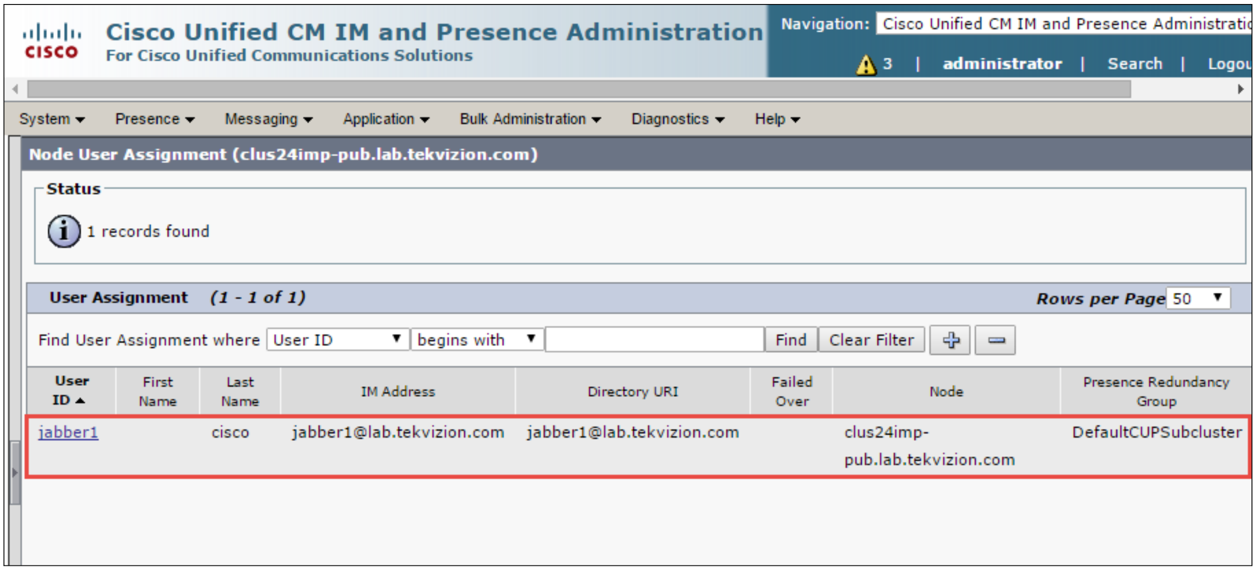
Assigned Presence Redundancy Group DefaultCUPSubcluster

IM&P database publisher node Yes

Assigned Users 1 users

Users

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



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

Status

1 records found

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

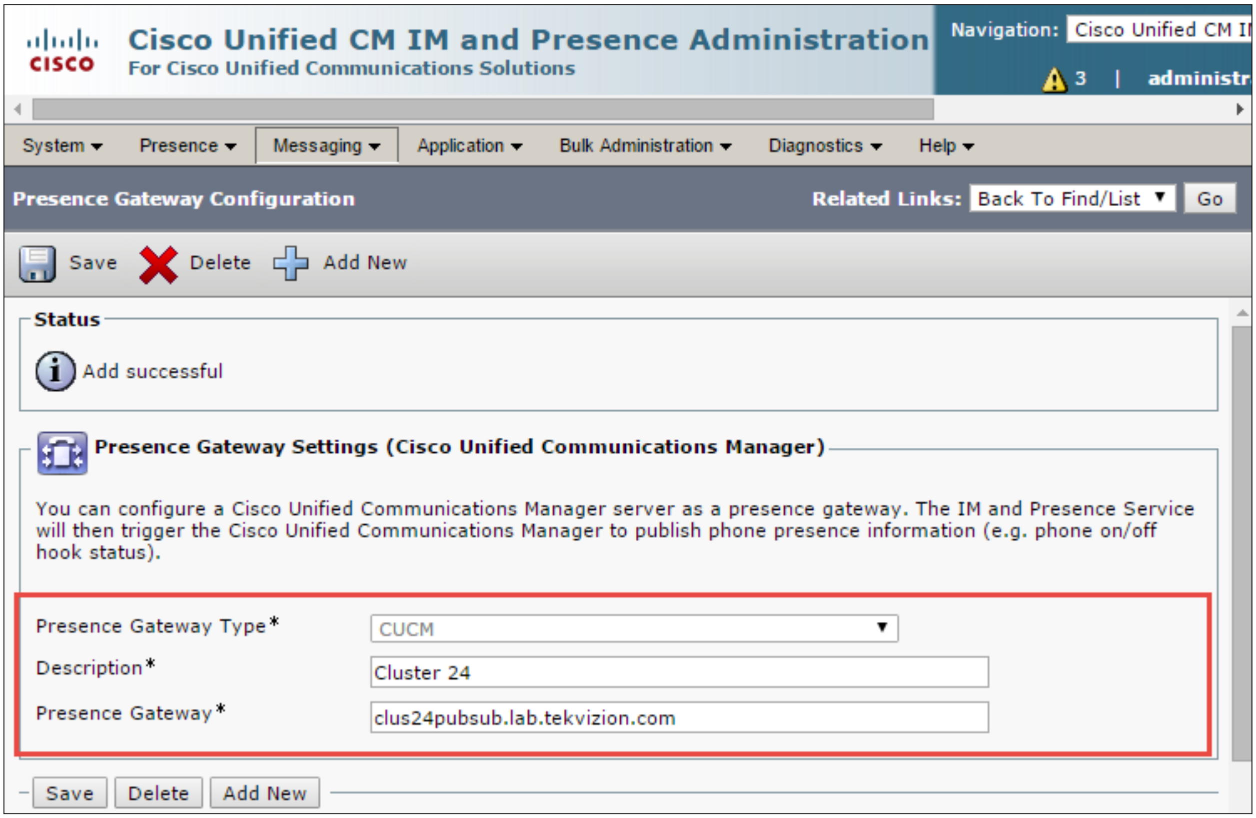
Find User Assignment where User ID begins with

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

Presence gateway configuration

Navigation: Presence → Gateways

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



Status

Add successful

Presence Gateway Settings (Cisco Unified Communications Manager)

You can configure a Cisco Unified Communications Manager server as a presence gateway. The IM and Presence Service will then trigger the Cisco Unified Communications Manager to publish phone presence information (e.g. phone on/off hook status).

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

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
ASR	Aggregation Services Router
MGCP	Media Gateway Control Protocol
MIS	Managed Internet Services
PNT	Private Network Transport
PSTN	Public switched telephone network
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol
SP	Service Provider
TDM	Time-division multiplexing

Important Information

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

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

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

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

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

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