Application Note

Cox SIP Trunking:

Cisco Unified Communications Manager 11.0.1 with Cisco Unified Border Element (CUBE 11.5.0) on ISR4321 [IOS-XE 3.17] using SIP

April 2017

Table of Contents

Table of Figures

Introduction

Service Providers today, such as Cox, 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 centralized IP to TDM POP gateways to provide on-net and off-net services.

Cox is a service provider that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity. A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and Cox Session Border Controller (EdgeMarc), Cisco Unified Border Element (CUBE 11.5.0) on ISR 4321 with IOS-XE 3.17 can be used. The Cisco Unified Border Element provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.0.1 connected to Cox IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of CUCM (Cisco Unified Communications Manager). Only configuration settings specifically required for Cox interoperability are presented. Feature configuration, and most importantly the dial plan, are customer specific and need an individual approach.

- This application note describes how to configure a Cisco Unified Communications Manager (Cisco UCM) 11.0.1 and Cisco Unified Border Element (CUBE 11.5.0) on ISR 4321 for connectivity to Cox SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 11.0.1) to PSTN (Cox).
- Testing was performed in accordance to Cox generic SIP Trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold, Semi-attendant and attendant transfers, call forward, conferences, and interoperability with Cisco Unity Connection
- The CUCM configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Cox SIP network and Cisco Unified Communications. The configuration described in this document details the important configuration settings to have enabled for interoperability to be successful and care must be taken by the network administrator deploying Cisco UCM to interoperate to Cox SIP trunking network.

This application note does not cover the use of Calling Search Spaces (CSS) or partitions on Cisco Unified Communications Manager. To understand and learn how to apply CSS and partitions refer to the cisco.com link below:
http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/collab09/dialplan.html#wpmkr1044275

Network Topology

Basic Call Setup

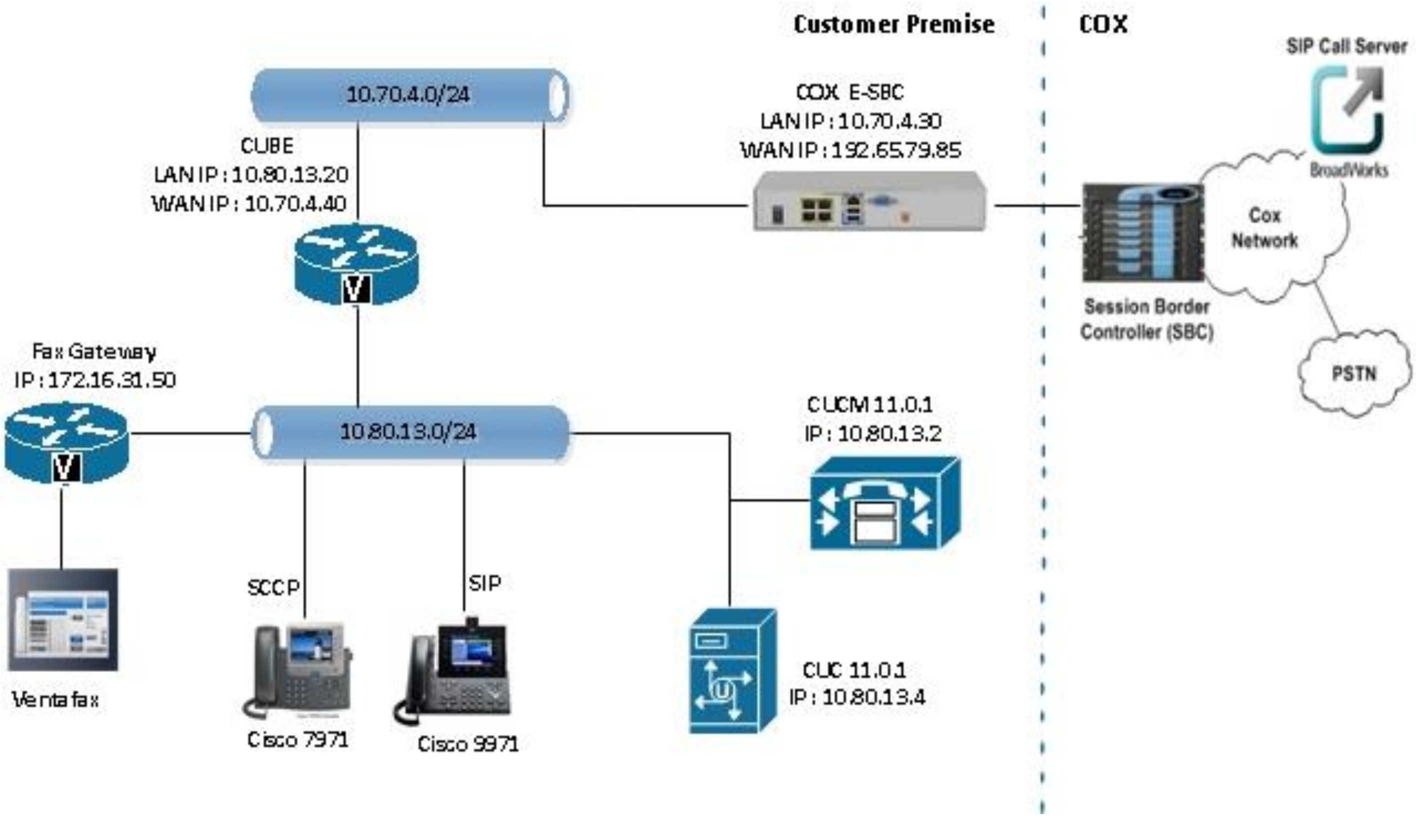


Figure 1: Network Topology

System Components

Hardware Components

- Cisco UCM and Unity Connection run on VMware
- ISR 2851 router as Fax Gateway
- ISR 4321 router as CUBE
- IP phones 9971(SIP) and 7971(SCCP)

Software Requirements

- Cisco Unified Communications Manager 11.0.1
- IOS-15.6(1)S for Cisco Unified Border Element on ISR4321
- IOS 15.1(4)M5 for Fax Gateways on ISR2851
- Cisco Unity Connection 11.0.1

Features Supported

- Incoming and outgoing off-net calls using G711Ulaw (Cox only offer G711Ulaw) with 20ms packetization
- Call hold
- Call transfer (Semi-Attendant and Attendant)
- Call conference
- Call forward (all, busy, no answer)
- Calling line (number) identification presentation (CLIP)
- Calling line (number) identification restriction (CLIR)
- DTMF (RFC2833)
- Media flow-through on CUBE
- Fax G.711 pass-through

Features Not Supported

- Outbound SIP REFER with Replaces. Cisco UCM does not currently support generation of an outbound SIP REFER with Replaces
- Cisco IP phones used in this test do not support Blind Transfer, only Semi-attendant and Attendant transfers were tested

Caveats

- The caller ID of the DUT is being seen instead of the originator of the call that is transferred or forwarded
- Defect ID: PAI/PPI support for INVITE/UPDATE Request/Response in CUBE – CSCuv04539

Configuration

Configuring the Cisco Unified Border Element

Network Interface

Configure Ethernet IP address and sub interface. The IP address and VLAN encapsulation used are for illustration purposes only and the actual IP address can vary. For SIP trunks, two IP addresses must be configured—LAN and WAN.

```
interface GigabitEthernet0/0/0
 description COX CUBE LAN
 ip address 10.80.13.20 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0/0/1
 description COX CUBE WAN
 ip address 10.70.4.40 255.255.255.0
 negotiation auto
```

Global CUBE Settings

In order to enable CUBE IP2IP gateway functionality, following command has to be entered:

```
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element
 allow-connections sip to sip
 no supplementary-service sip handle-replaces
 fax protocol pass-through g711ulaw
 sip
  session refresh
  asserted-id pai
  early-offer forced
```

midcall-signaling passthru

privacy-policy passthru

privacy-policy send-always

g729 annexb-all

Explanation

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
fax protocol	Specifies the fax protocol
asserted-id	Specifies the type of privacy header in the outgoing SIP requests and response messages
early-offer forced	Enables SIP Delayed-Offer to Early-Offer globally
midcall-signaling passthru	Passes SIP messages from one IP leg to another IP leg

Media Passing Through CUBE (media flow-through vs. media flow-around)

Default CUBE configuration enables CUBE to work in flow-through mode (this test use flow-through mode). In order to enable flow-around mode, please perform the following actions:

voice service voip

media flow-around

Codecs

Cox offer only G.711ulaw codec for voice call, it allows codecs other than G.711ulaw but will only accept G.711ulaw.

For customers using **G.711 ulaw** codec:

voice class codec 1

codec preference 1 g711ulaw

Dial Peer

CUCM uses dial-peer to route the call based on the digit to route the call accordingly.

```
dial-peer voice 20 voip
 description " Incoming to IP-PBX - IP-PBX facing side "
 destination-pattern 402614....
 no modem passthrough
 session protocol sipv2
 session target ipv4:10.80.13.2:5060
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 21 voip
 description " Incoming IP-PBX - EdgeMarc facing side "
 no modem passthrough
 session protocol sipv2
 incoming called-number 402614....
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
```

```
 voice-class sip privacy-policy passthru
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 15 voip
 description "Outgoing To EdgeMarc" -EdgeMarc facing side
 destination-pattern [0-9]T
 no modem passthrough
 session protocol sipv2
 session target ipv4:10.70.4.30:5060
 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
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
```

```
dial-peer voice 16 voip
 description "Outgoing To EdgeMarc" -IP PBX facing side"
 no modem passthrough
 session protocol sipv2
 incoming called-number [0-9]T
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
```

ISR Configuration

The following configuration snippet contains a sample configuration of Cisco Unified Border Element with all parameters mentioned previously.

```
COX_CUBE#sh running-config

Building configuration...

!

version 15.6

service timestamps debug datetime msec

service timestamps log datetime msec

no platform punt-keepalive disable-kernel-core

!

hostname COX_CUBE

!

boot-start-marker

boot system flash:isr4300-universalk9.03.17.00.S.156-1.S-std.SPA.bin

boot system bootflash:isr4300-universalk9.03.17.00.S.156-1.S-std.SPA.bin

boot-end-marker

!

!

vrf definition Mgmt-intf

 !

 address-family ipv4

 exit-address-family

 !

 address-family ipv6

 exit-address-family

!

no aaa new-model

no ip domain lookup

ip domain name tekvizion.com
```

```
!
subscriber templating
multilink bundle-name authenticated
!
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 50
 allow-connections sip to sip
 no supplementary-service sip handle-replaces
 fax protocol pass-through g711ulaw
 sip
  session refresh
  asserted-id pai
  early-offer forced
  midcall-signaling passthru
  privacy-policy passthru
  privacy-policy send-always
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g711ulaw
!
voice class sip-profiles 1
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:402614\1@\2>"
!
license udi pid ISR4321/K9 sn FDO17860MW3
license boot level appxk9
license boot level uck9
```

```
!
spanning-tree extend system-id
!
redundancy
 mode none
!
vlan internal allocation policy ascending
!
interface GigabitEthernet0/0/0
 description Cox CUBE LAN
 ip address 10.80.13.20 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0/0/1
 description COX CUBE WAN
 ip address 10.70.4.40 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
interface Vlan1
 no ip address
 shutdown
!
ip forward-protocol nd
no ip http server
```

```
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 10.70.52.1
ip route 10.64.0.0 255.255.0.0 10.80.13.1
ip route 172.16.0.0 255.255.0.0 10.80.13.1
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 20 voip
 description " Incoming to IP-PBX - IP-PBX facing side "
 destination-pattern 402614....
 session protocol sipv2
 session target ipv4:10.80.13.2:5060
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
```

```
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 21 voip
 description " Incoming IP-PBX - EdgeMarc facing side "
 session protocol sipv2
 incoming called-number 402614....
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 15 voip
 description "Outgoing To EdgeMarc" -EdgeMarc facing side
 destination-pattern [0-9]T
 session protocol sipv2
 session target ipv4:10.70.4.30:5060
 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
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 16 voip
 description "Outgoing To EdgeMarc" -IP PBX facing side"
session protocol sipv2
 incoming called-number [0-9]T
 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
sip-ua
 credentials username 4023159990 password 7 15465B5E577B7E7D716A65 realm 10.70.4.30
```

```
 authentication username 4023159990 password 7 01475656085A535678151E
 registrar ipv4:10.70.4.30 expires 3600
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 password tekV1z10n
 login
 transport input telnet ssh
!
!
End
```

Configuring the Cisco Unified Communications Manager

Cisco UCM Version

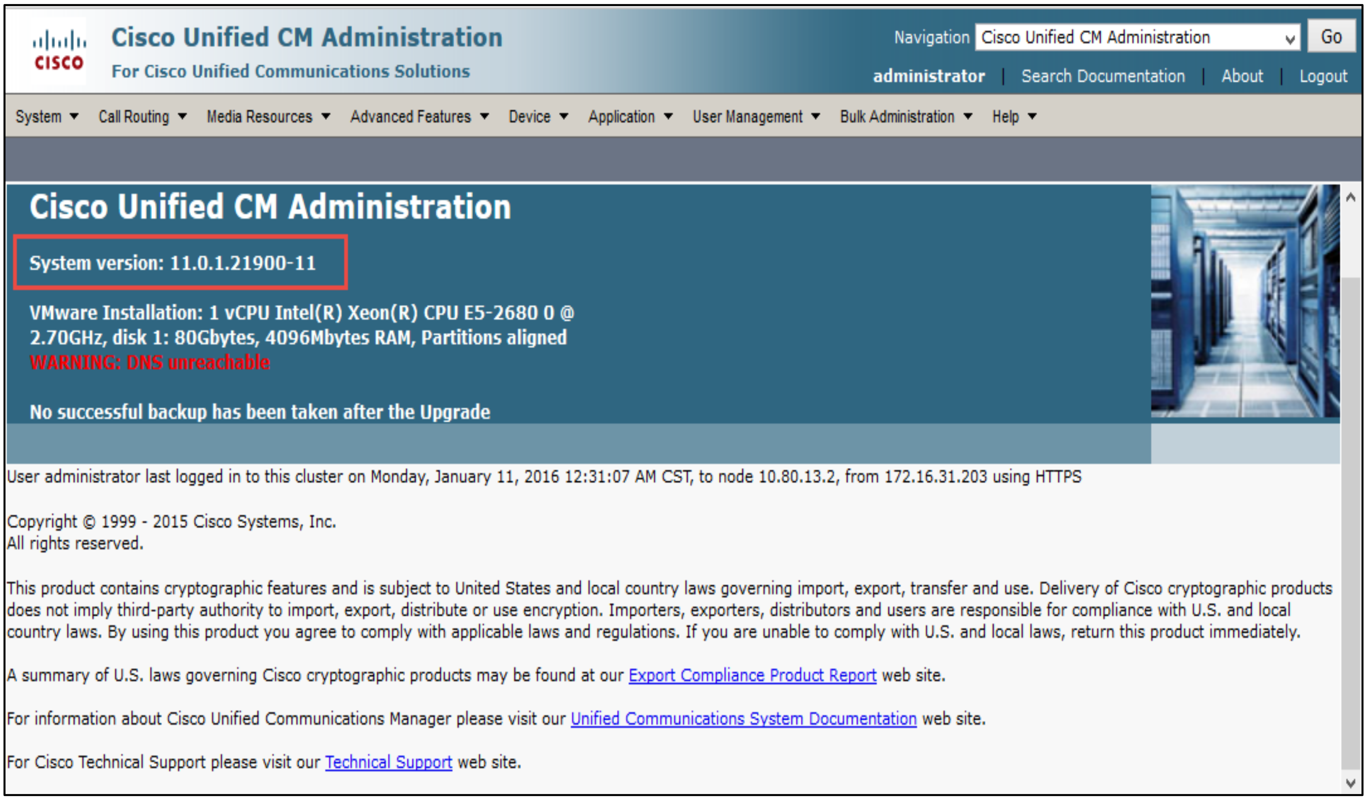


Figure 2: Cisco UCM Version

Cisco UCM Audio Codec Preference List

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

Cisco UCM 11.0.1 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)

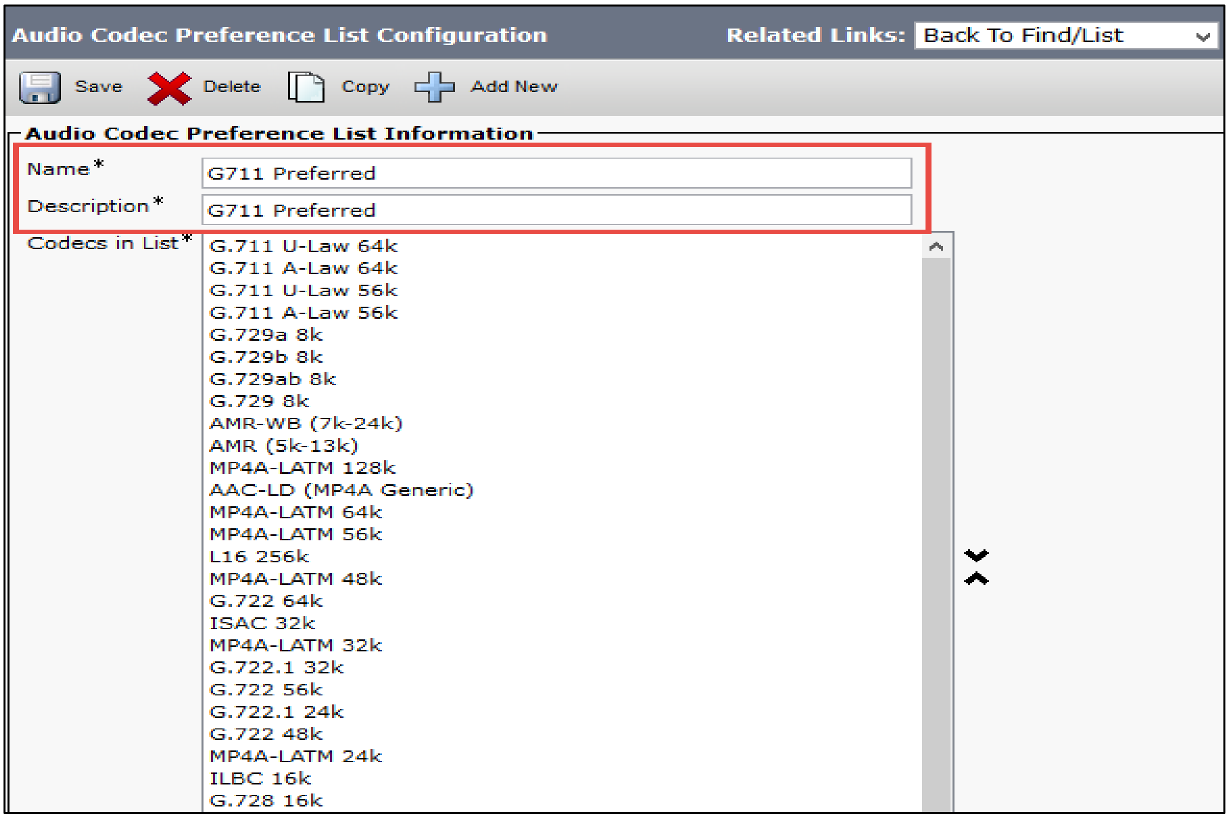


Figure 3: Audio Codec Preference List

Cisco UCM Region Configuration

Navigation Path: System → Region Information → Region

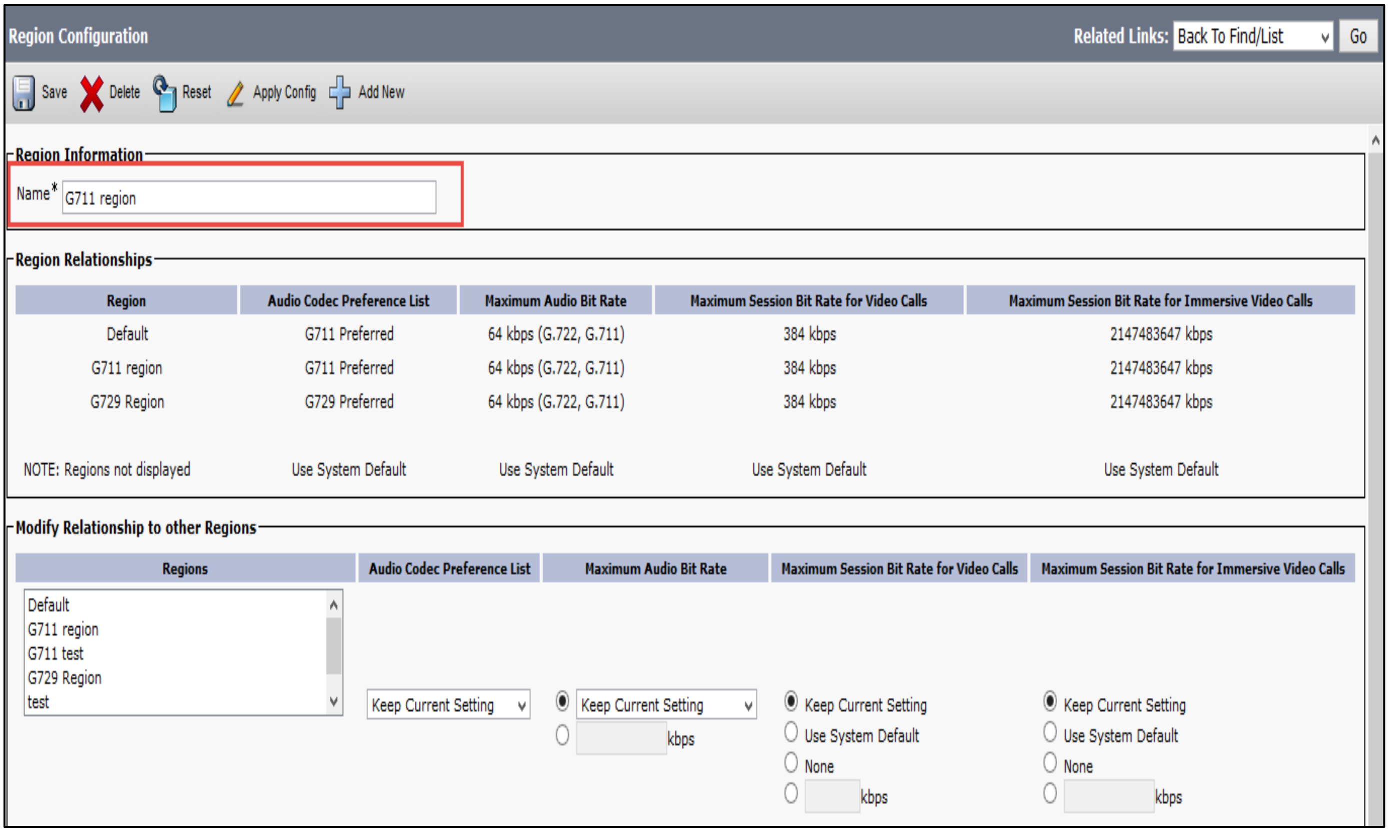


Figure 4: Region Configuration

Device Pool Configuration

Navigation Path: System → Device Pool

“G711 pool” Device Pool is configured for testing the interoperability. No special consideration needs to be taken when configuring the Device Pools. Optionally, a Media Resource Group List can be added to the Device Pools, if needed, to assign selected Media Resources (Conference Bridges, Transcoders, MoH servers, Annunciators) to devices.

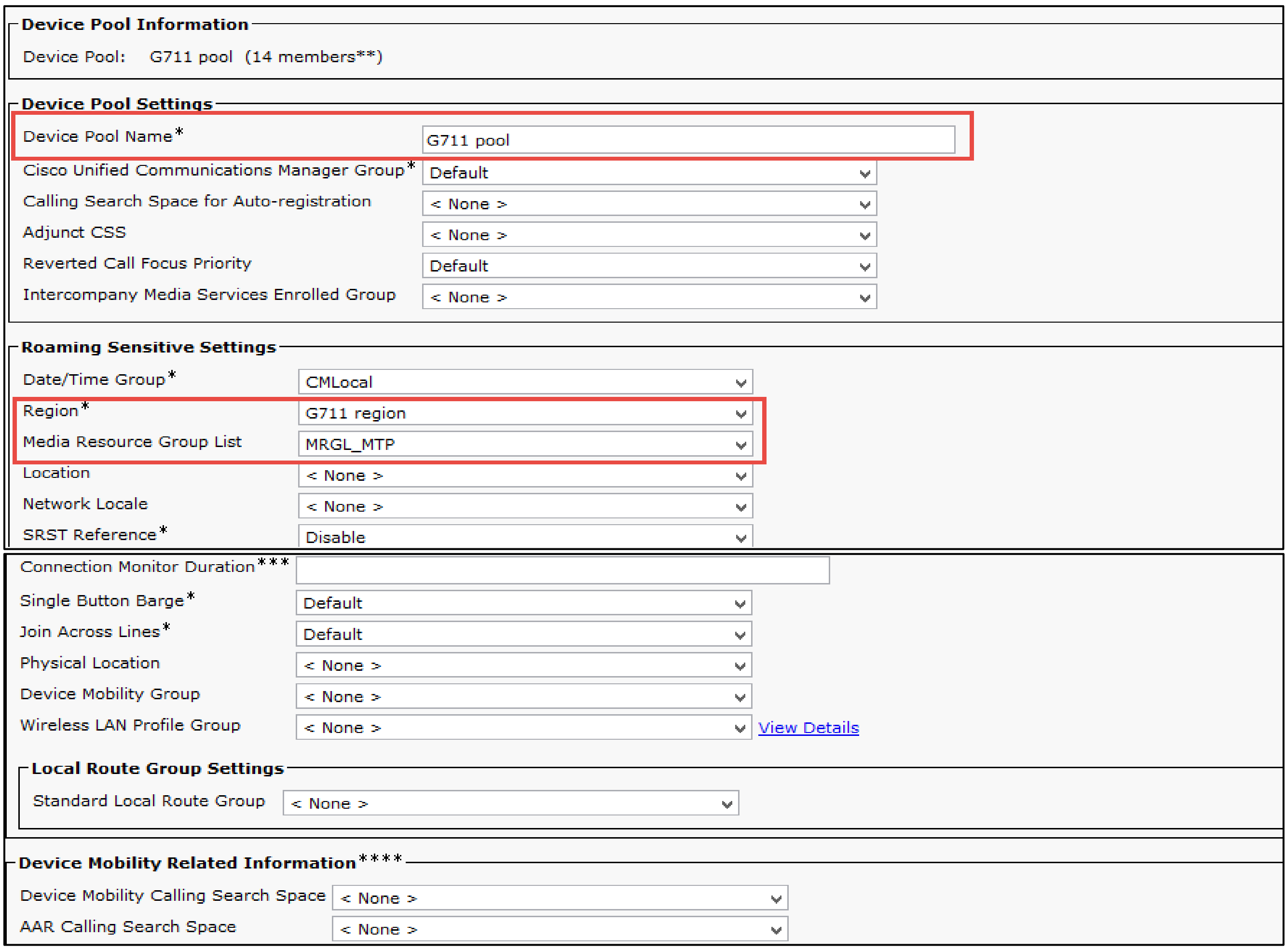


Figure 5: Device Pool Configuration

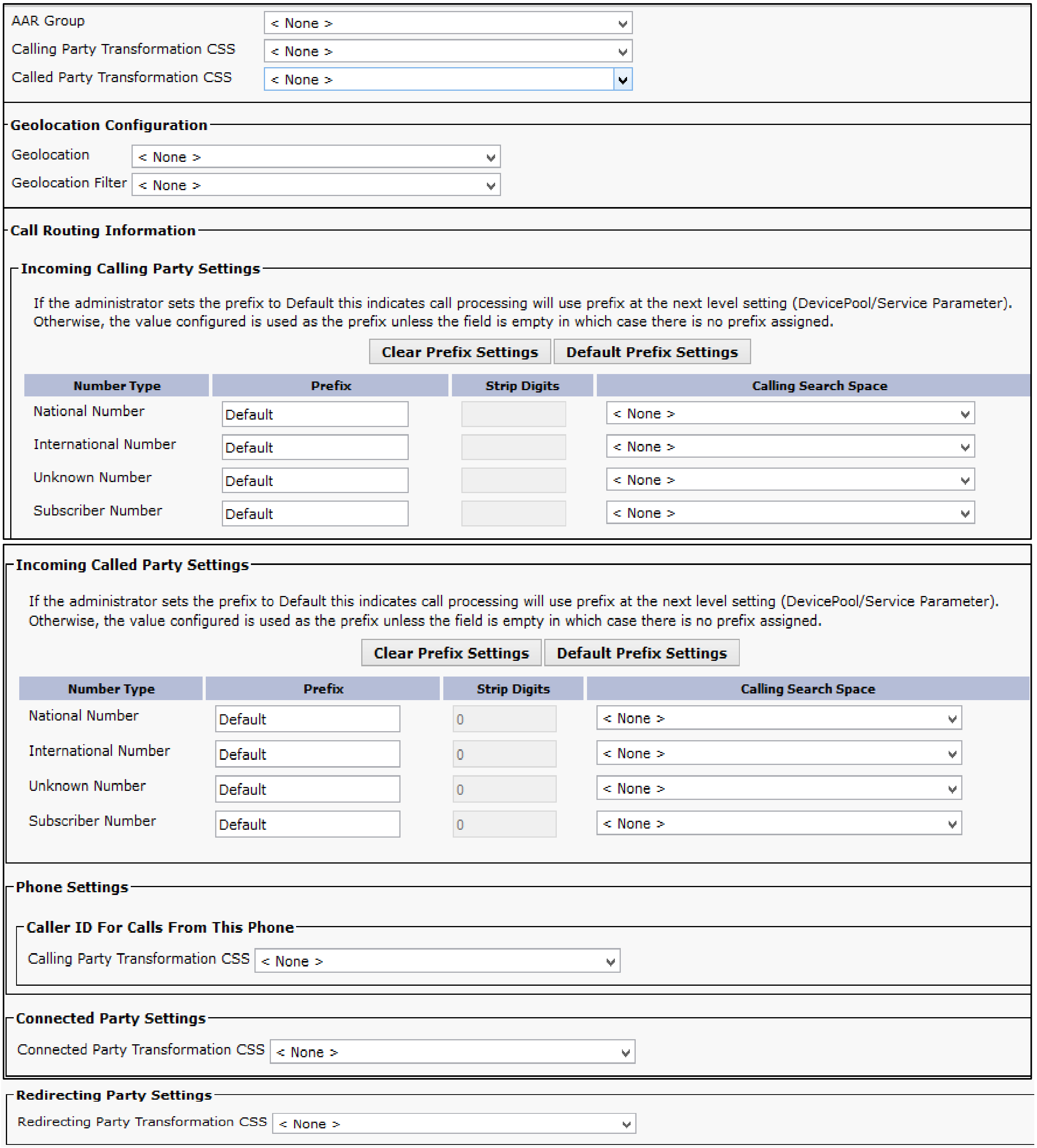


Figure 6: Device Pool Configuration – Cont.

Annunciator Configuration

Navigation: Media Resource → Annunciator

Set Name* = ANN_2
Set Description = ANN_clus23pubsub. This is used for this example.
Set Device Pool* = G711 pool

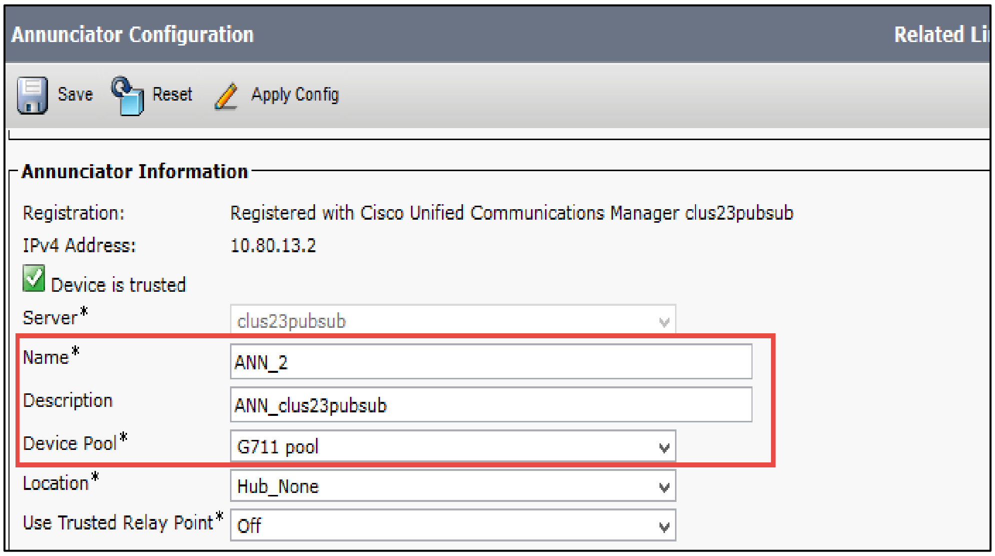


Figure 7: Annunciator Configuration

Conference Bridge Configuration

Navigation: Media Resources → Conference Bridge

Set Conference Bridge Type* = Cisco Conference Bridge Software
Set Host Server = clus23pubsub. This is used for this example
Set Conference Bridge Name* = CFB_2
Set Description = CFB_clus23pubsub. This is used in this example.
Set Device Pool* = G711 pool

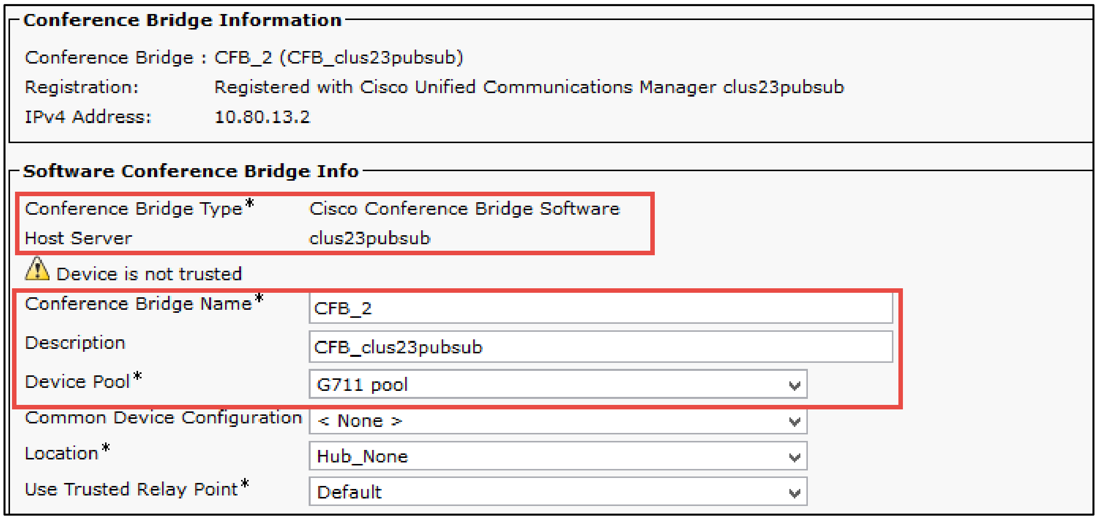


Figure 8: Conference Bridge Configuration

Media Termination Point Configuration

NOTE*** ***Cox certification is a SIP trunk certification between Cox Business eSBC, Cisco CUBE and Cisco CUCM only. In the event the customer requires DTMF between dissimilar endpoints an external MTP is required"***

Navigation: Media Resource → Media Termination Point

Set Media Termination Point Name* = MTP_2
Set Description = MTP_clus23pubsub is used for this example
Set Device pool* = G711 pool

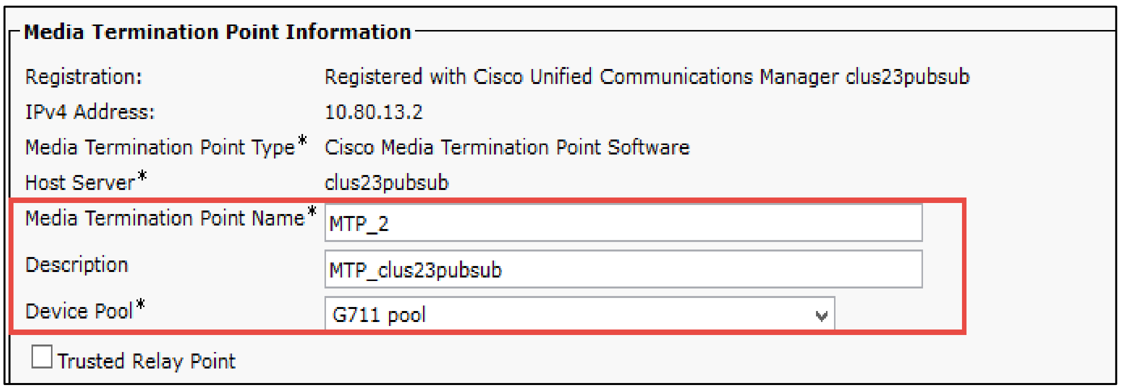


Figure 9: Media Termination Point Configuration

Music on Hold Server Configuration

Navigation: Media Resources → Music on Hold Server

Set Music on Hold Server Name* = MOH_2
Set Description = MOH_clus23pubsub. This is used for this example.
Set Device Pool* = G711 pool

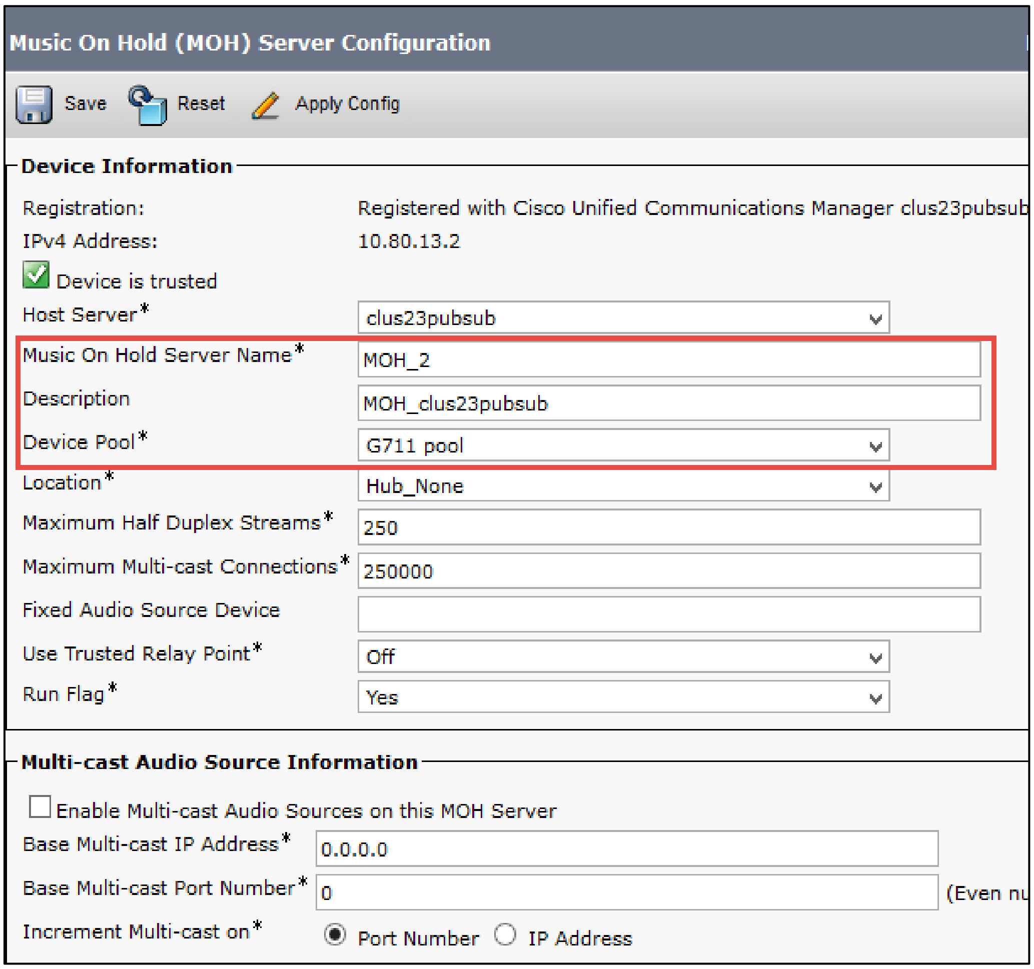


Figure 10: MOH Configuration

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 with MTP. This text is used to define this Media Resource Group List.
Set all Resources in the selected Media Resources Box

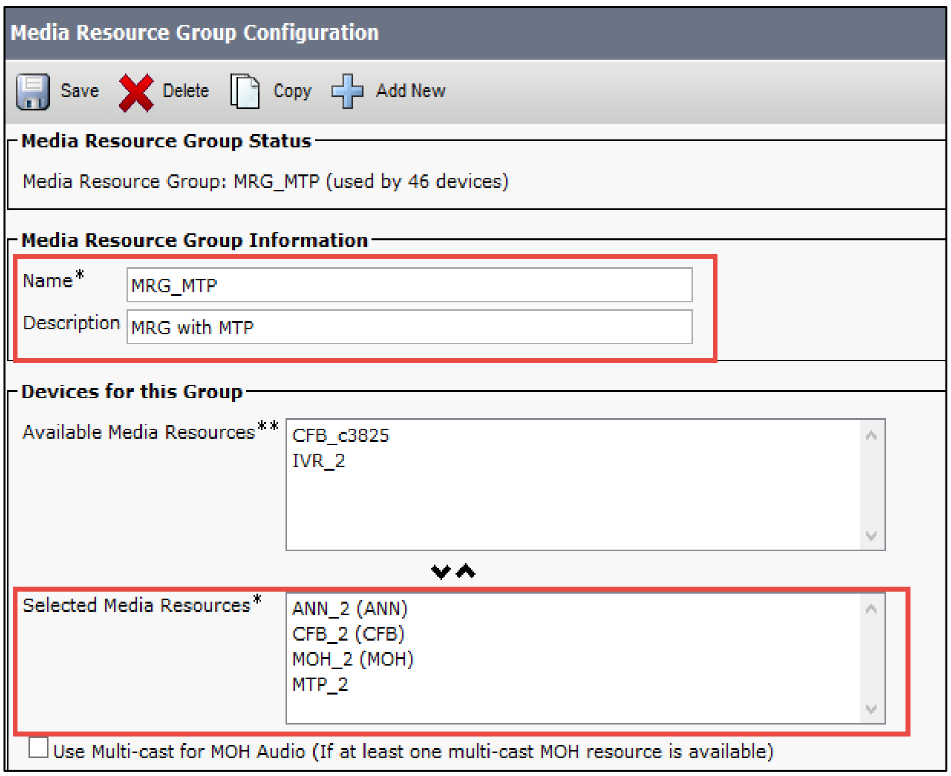


Figure 11: Media Resource Group Configuration

Media Resource Group List Configuration

Navigation Path: Media Resources → Media Resource Group List

Set Name = MRGL_MTP
Set selected Media Resource Groups = MRG_MTP

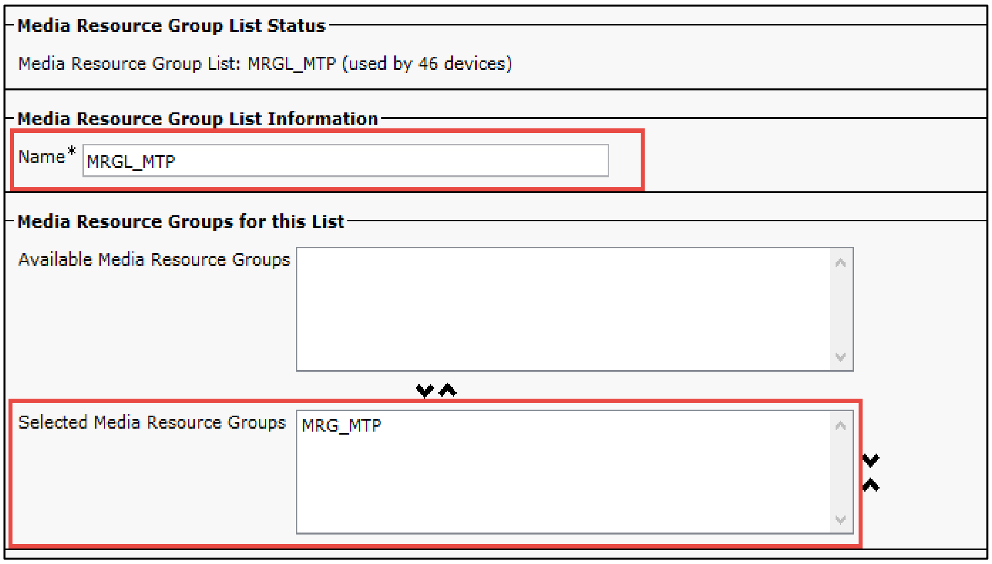


Figure 12: Media Resource Group List Configuration

Cisco IP Phone 9971 SIP Configuration

Set MAC Address* = the below mac is used in this example
Set Description = EXT-COX-1399. This text is used to identify this Phone
Set Device Pool*= G711 pool. This is used in this example
Set Phone Button Template*= Standard 9971 SIP. This is used in this example
Set Soft key Template = Standard User. This is used in this example
Set Media Resource Group List = MRGL_MTP. This is used in this example
Set User Hold MOH Audio Source = 1-SampleAudioSource
Set Network Hold MOH Audio Source = 1-SampleAudioSource

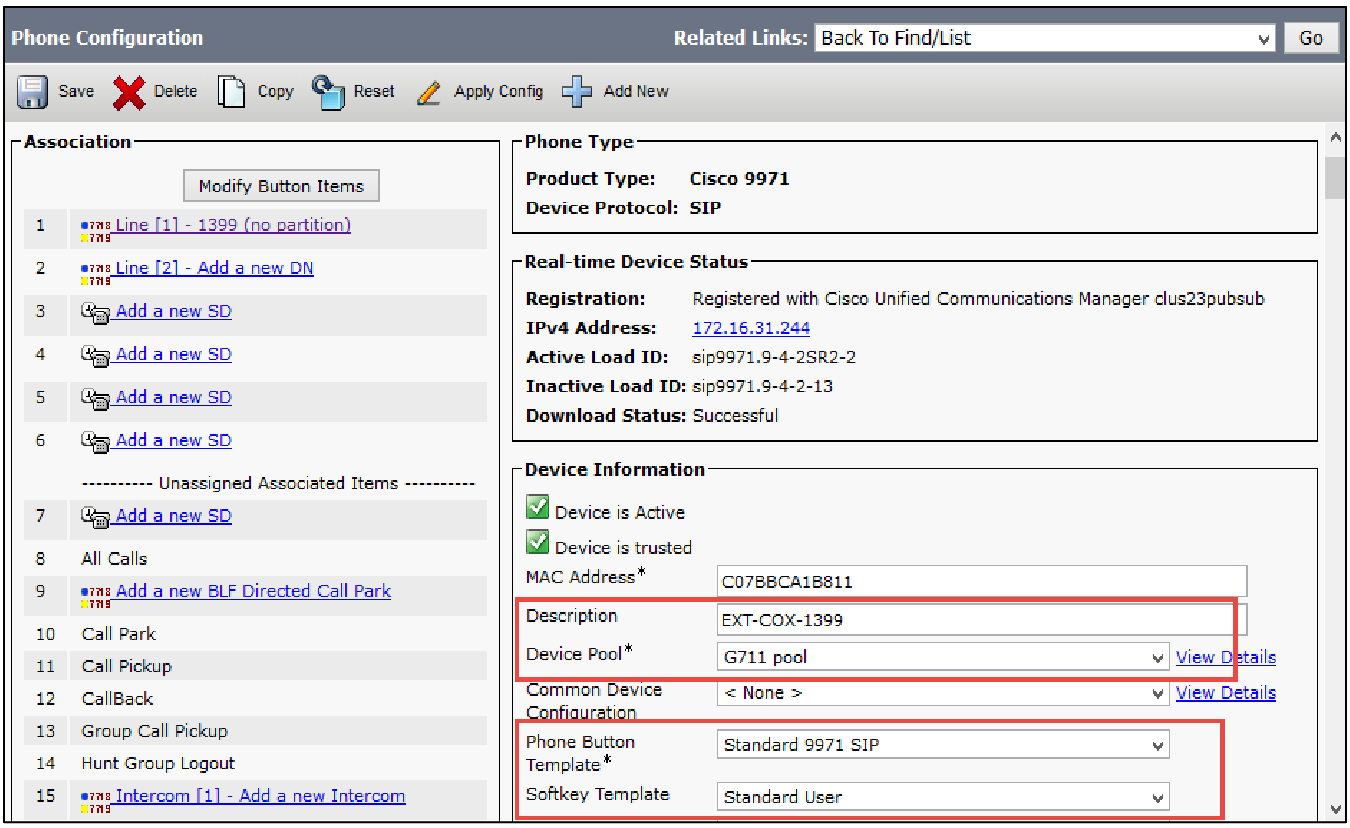


Figure 13: Cisco IP Phone 9971 SIP Configuration

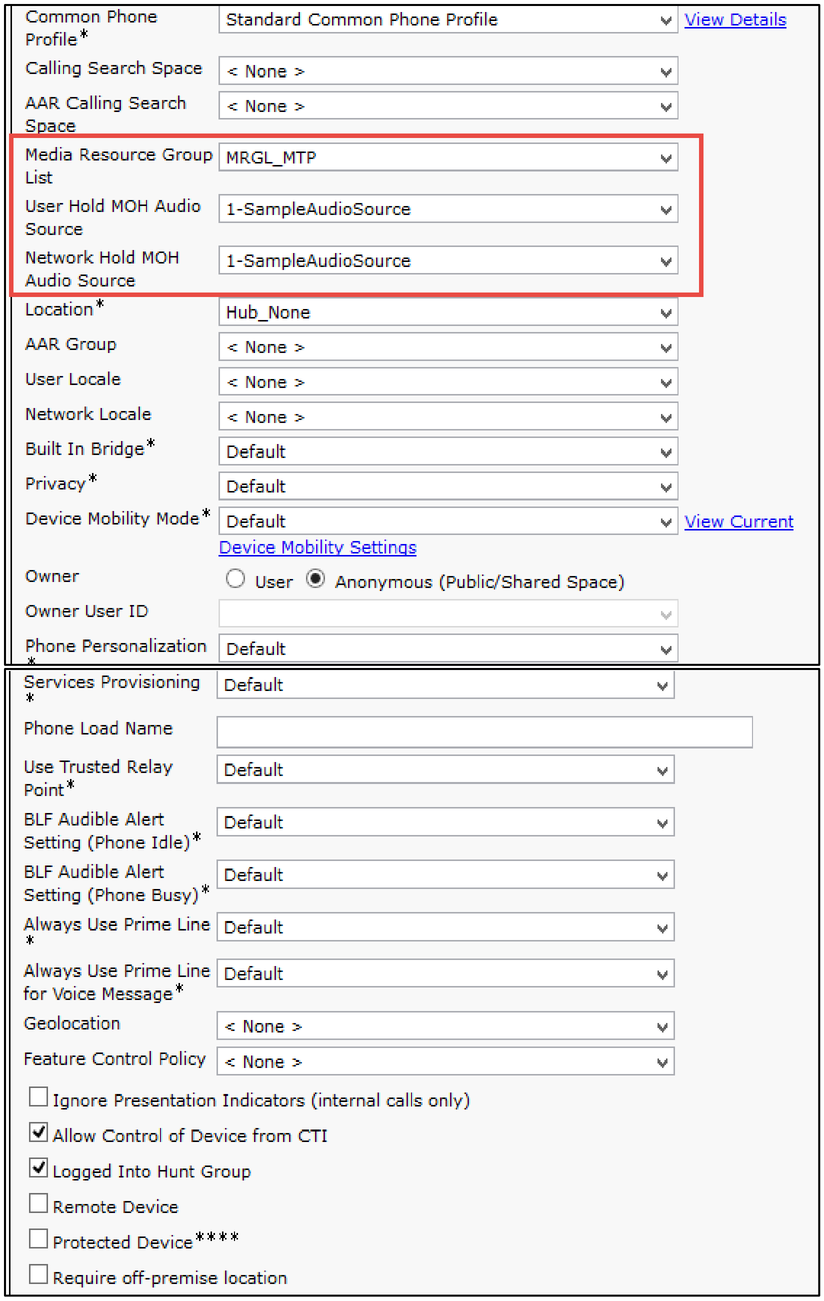


Figure 14: Cisco IP Phone 9971 SIP Configuration – Cont.

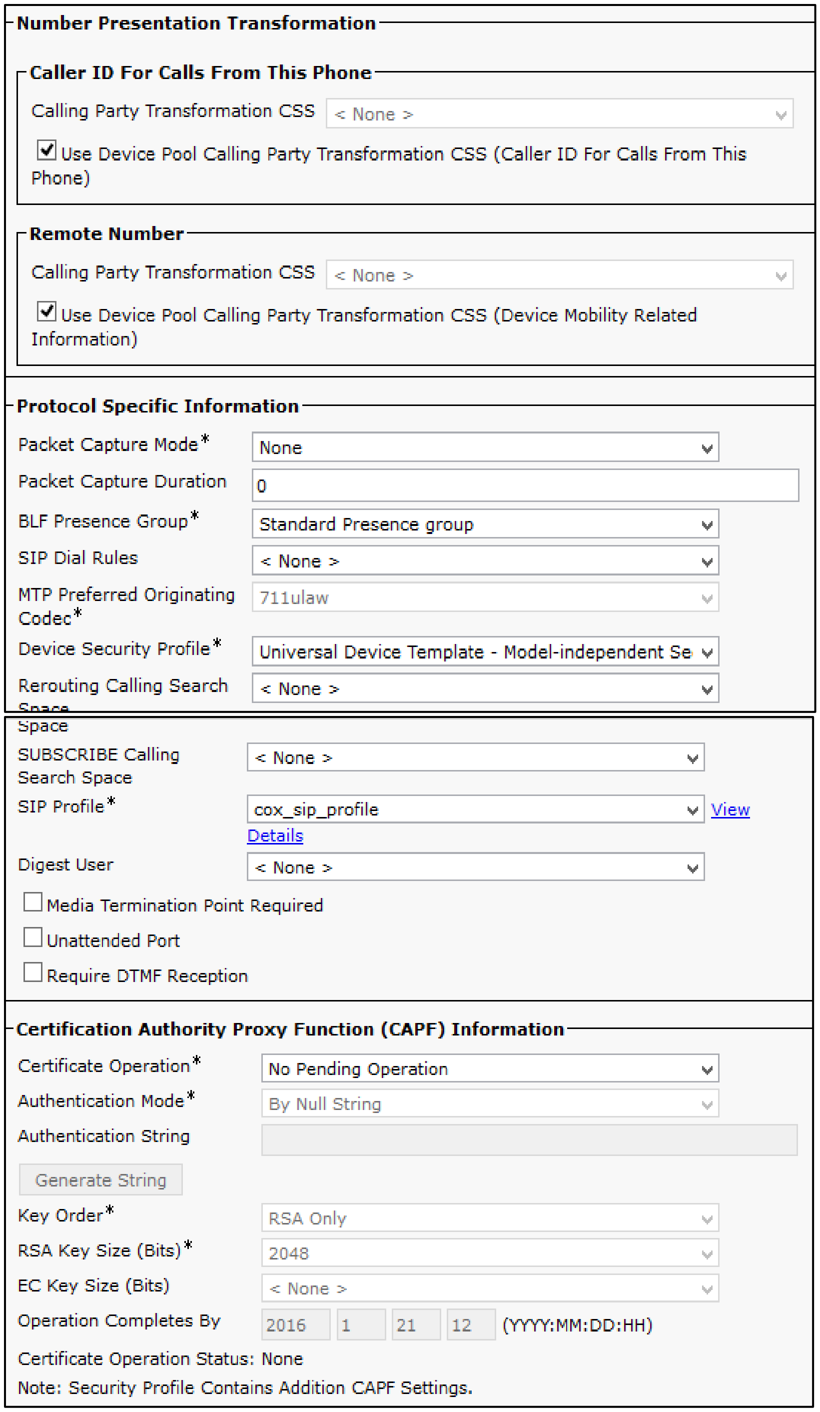


Figure 15: Cisco IP Phone 9971 SIP Configuration – Cont.

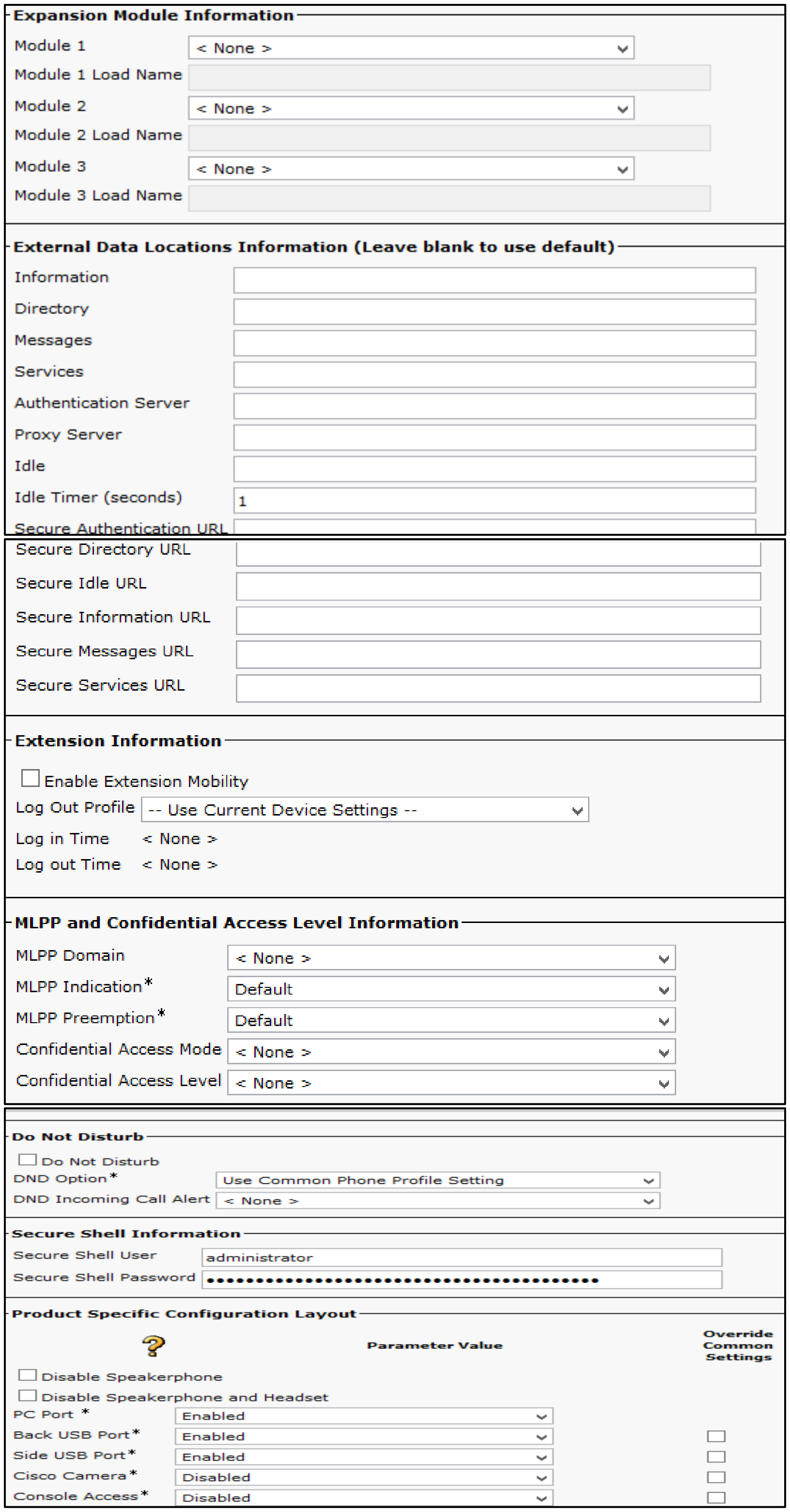


Figure 16: Cisco IP Phone 9971 SIP Configuration – Cont.

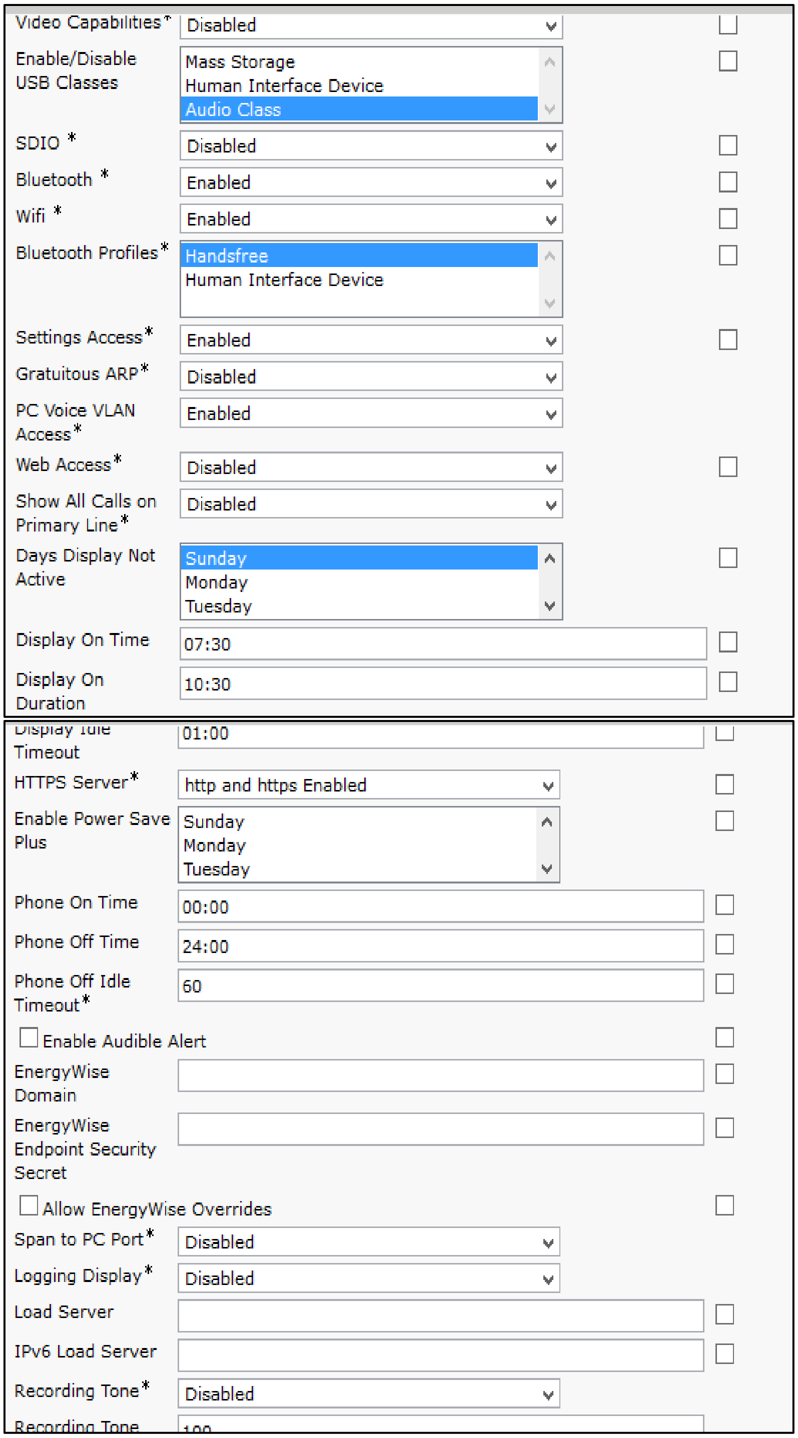


Figure 17: Cisco IP Phone 9971 SIP Configuration – Cont.

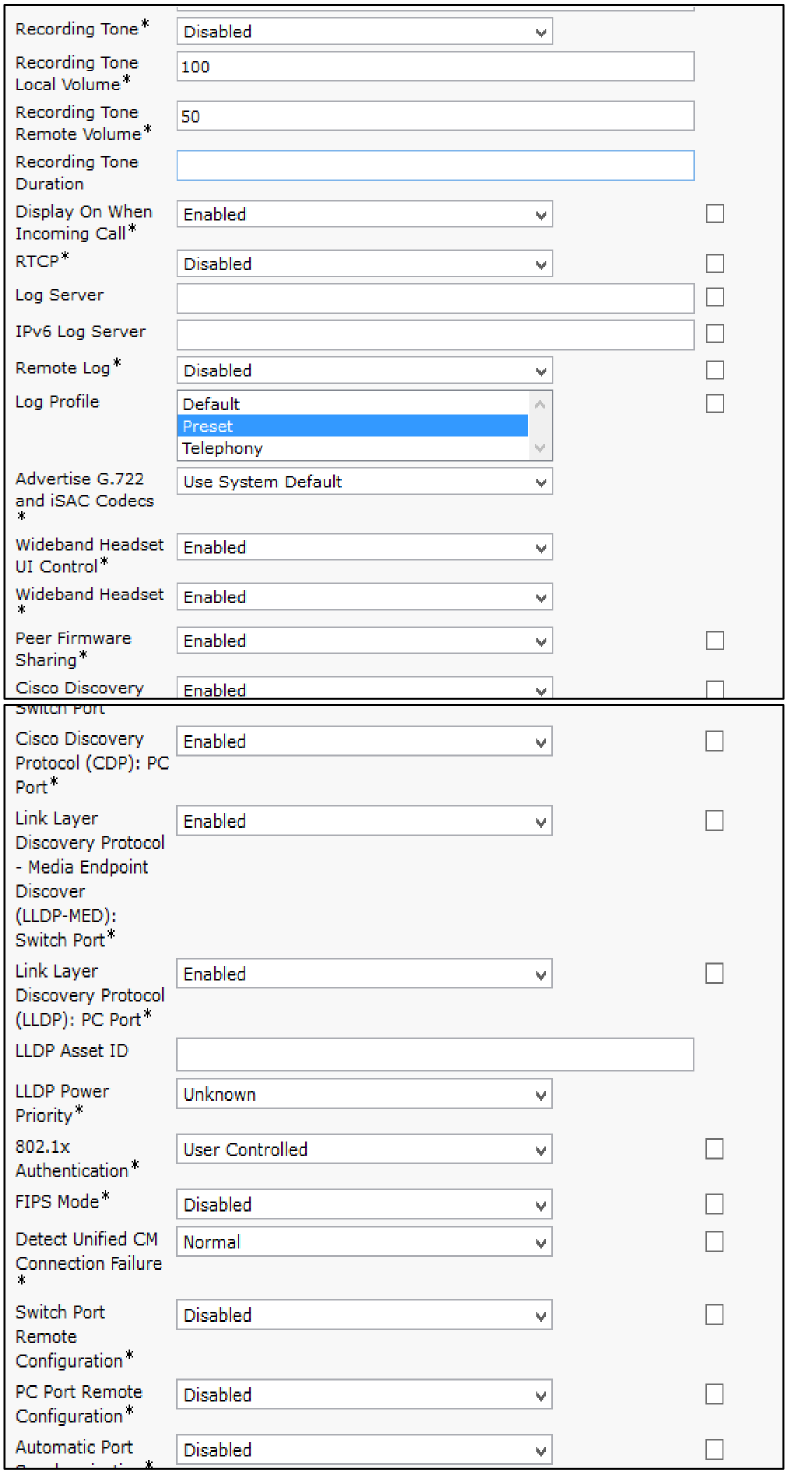
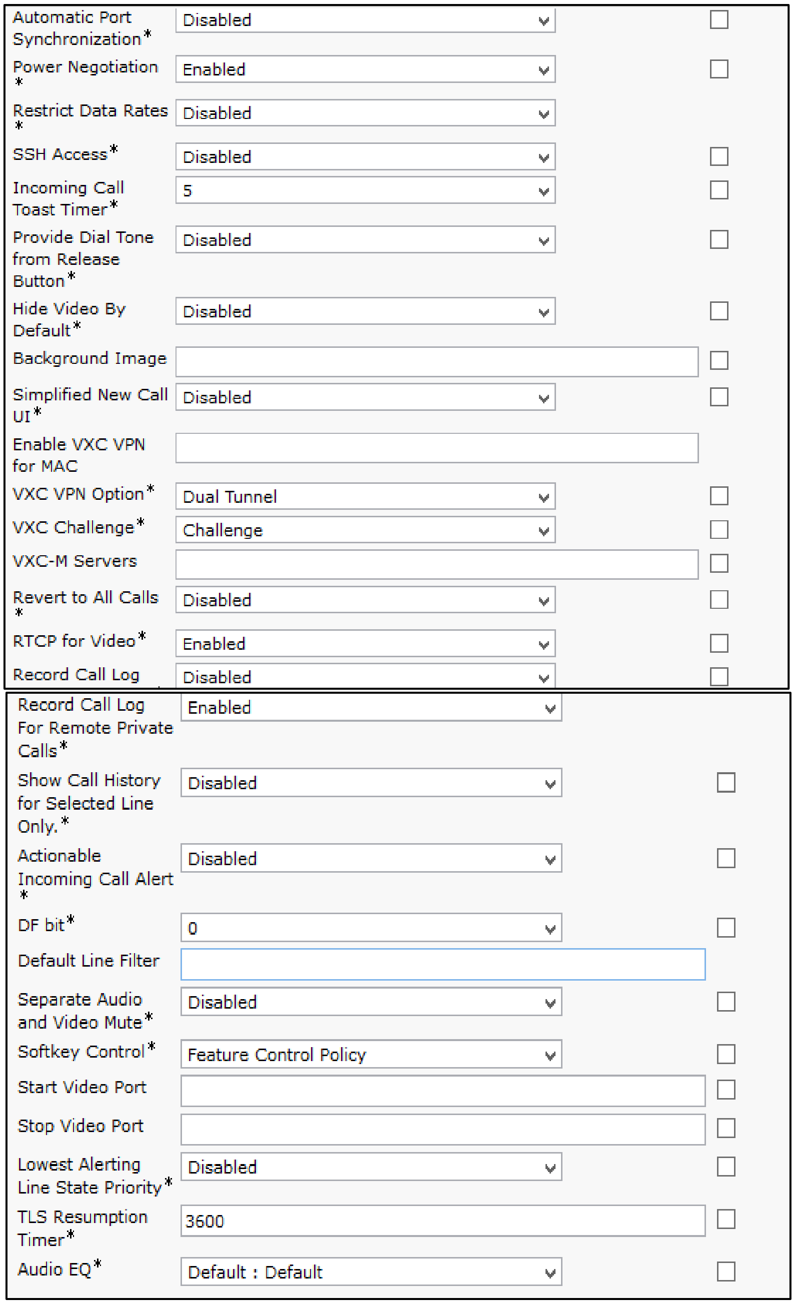


Figure 18: Cisco IP Phone 9971 SIP Configuration – Cont.



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
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	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

Figure 19: Cisco IP Phone 9971 SIP Configuration – Cont.

Set Directory Number* = 1399. This is used in this example.
Set Description = SIP-1399. This is used in this example.
Set Alerting Name = SIP-1399. This is used in this example.
Set ASCII Alerting Name = SIP-1399. This is used in this example.

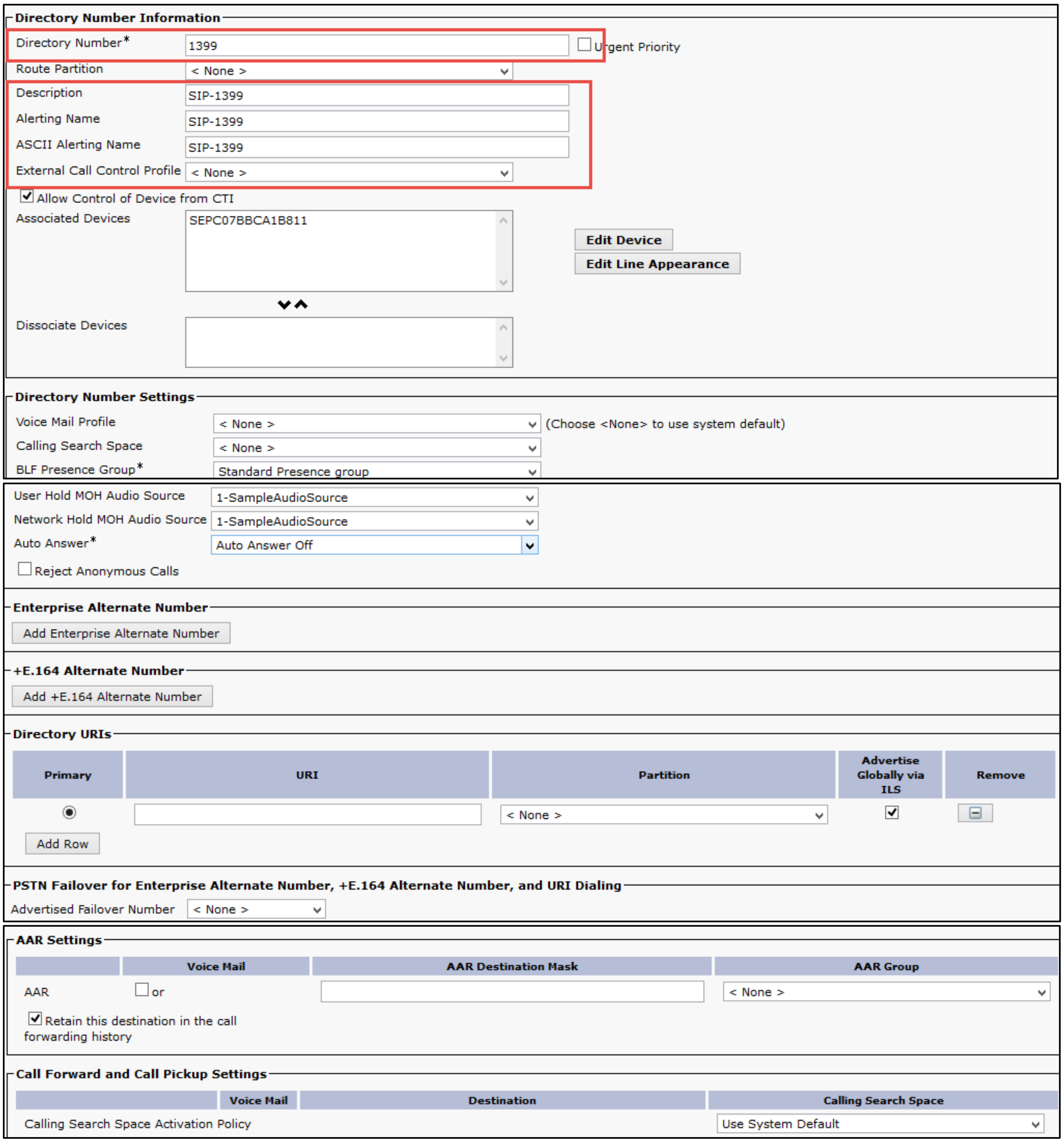


Figure 20: Cisco IP Phone 9971 SIP Configuration – Cont.

Set Display (Caller ID) = SIP-1399. This is used in this example
Set ASCII Display (Caller ID) = SIP-1399. This is used in this example
Set Line Text Label = SIP-1399. This is used in this example
Set External Phone Number Mask = 4026141399. This is used in this example

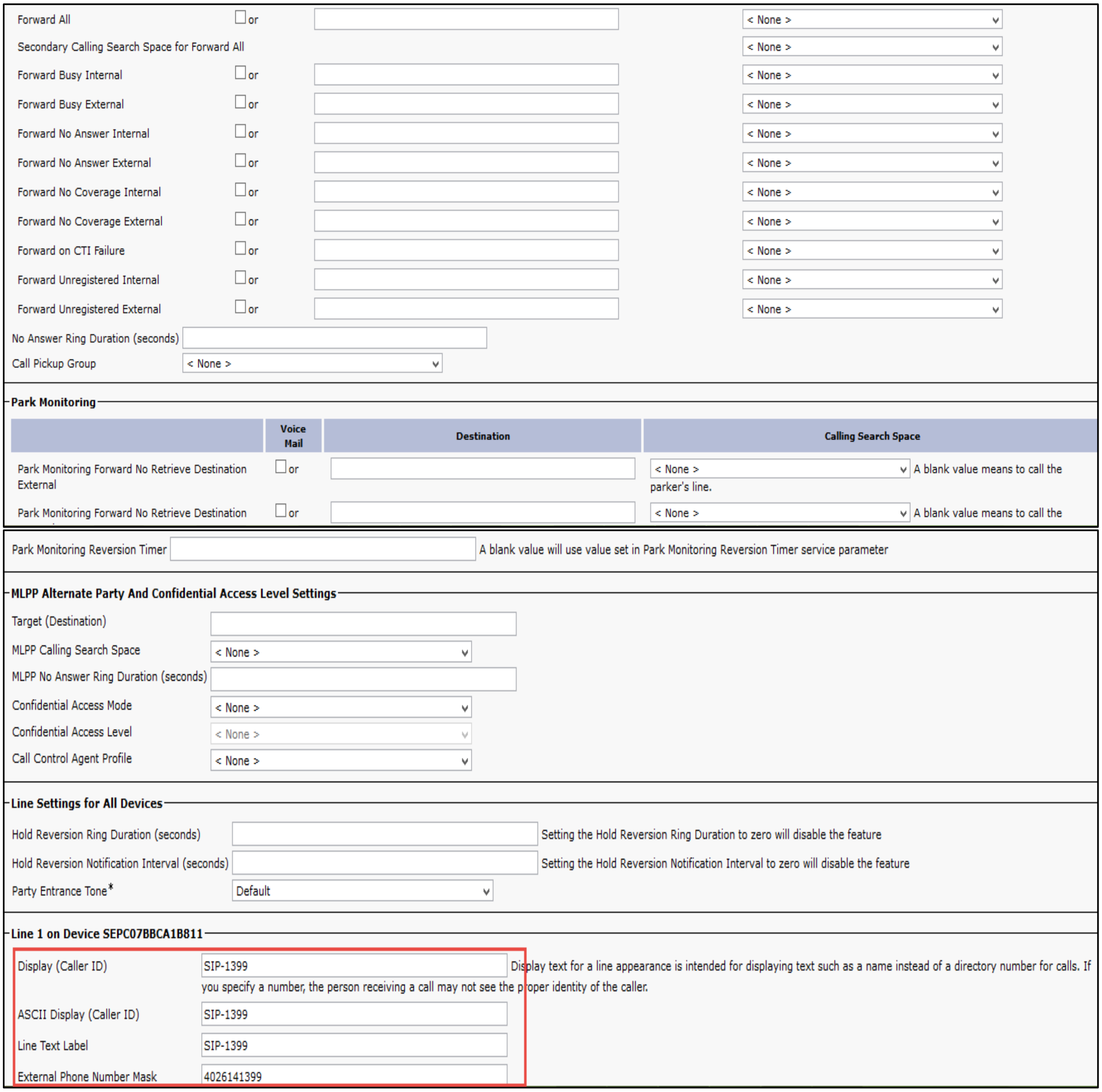


Figure 21: Cisco IP Phone 9971 SIP Configuration – Cont.

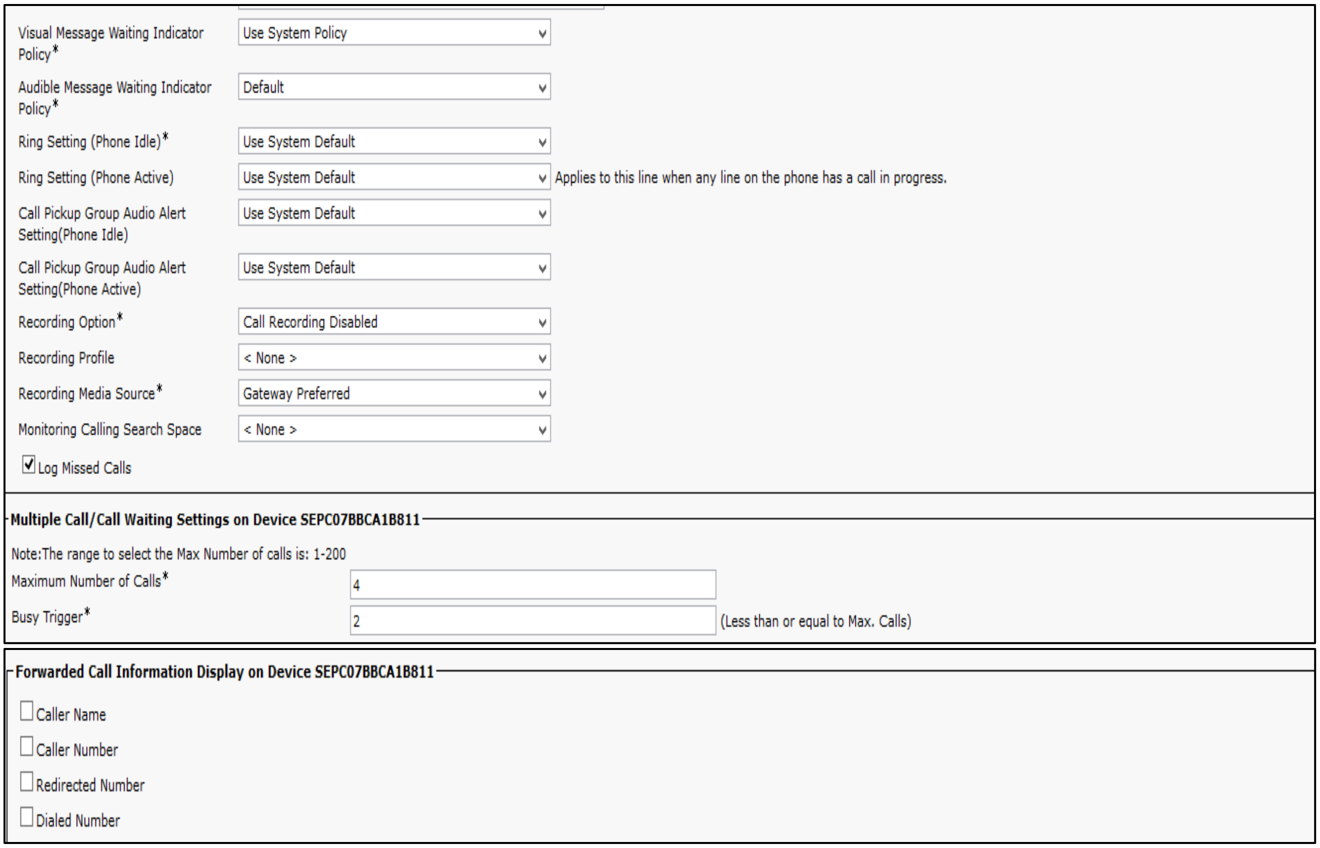


Figure 22: Cisco IP Phone 9971 SIP Configuration – Cont.

Cisco IP Phone 7971 SCCP Configuration

Set MAC Address* = the below mac is used in this example
Set Description = EXT-COX-1958. This text is used to identify this Phone
Set Device Pool*= G711 pool. This is used in this example
Set Phone Button Template*= Standard 7971 SCCP. This is used in this example
Set Soft key Template = Standard User. This is used in this example
Set Media Resource Group List = MRGL_MTP. This is used in this example
Set User Hold MOH Audio Source = 1-SampleAudioSource
Set Network Hold MOH Audio Source = 1-SampleAudioSource

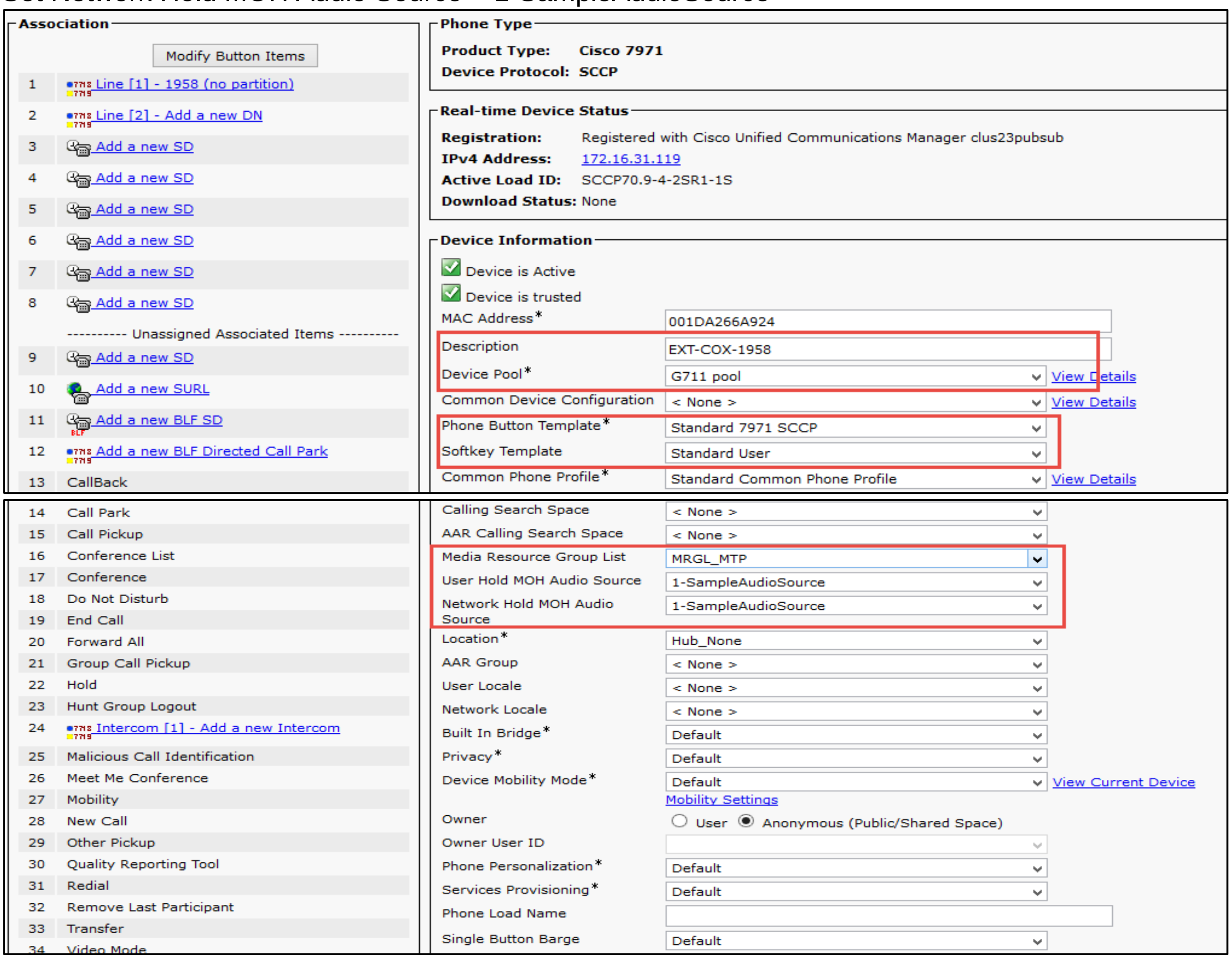


Figure 23: Cisco IP Phone 7971 SCCP Configuration

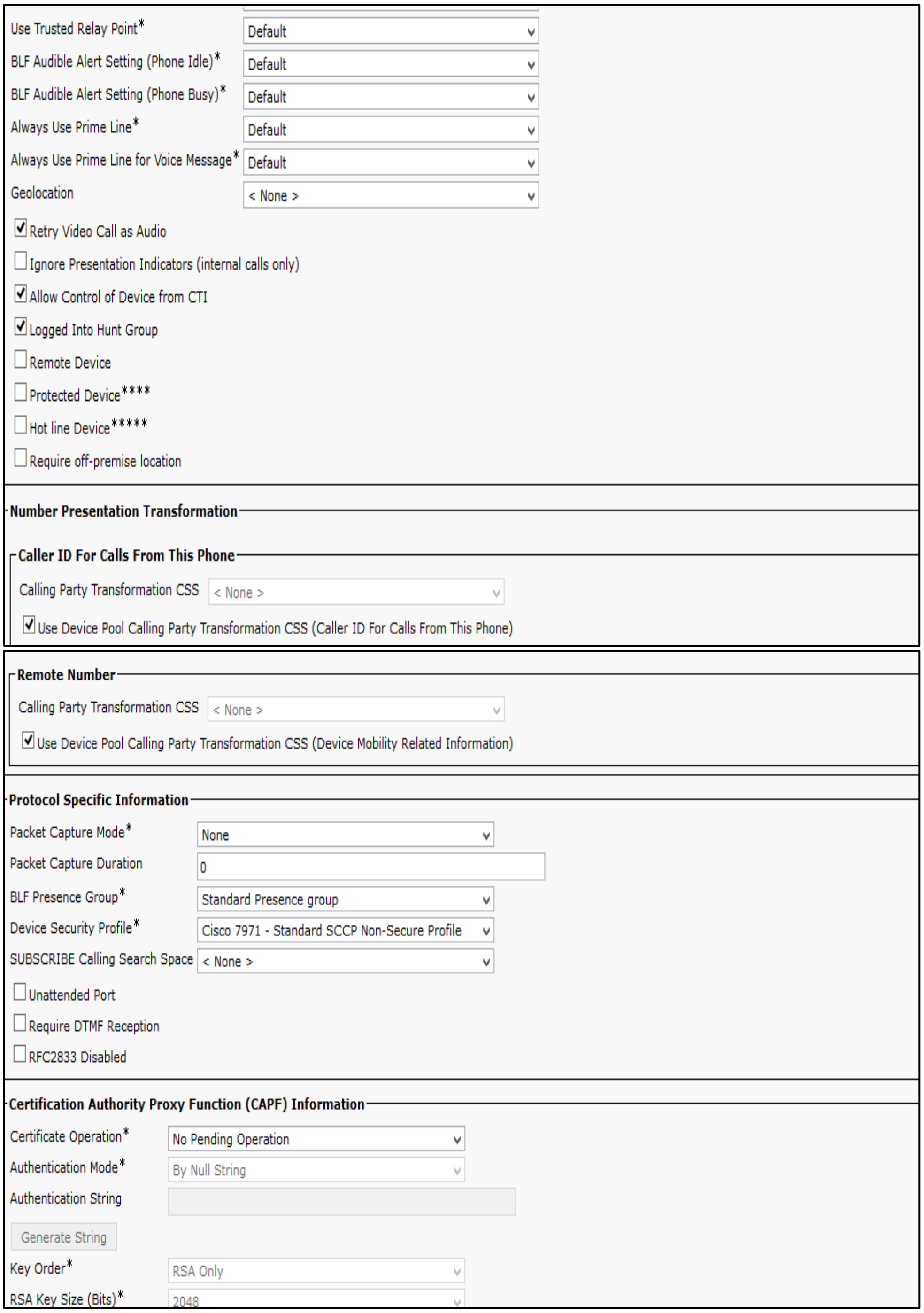


Figure 24: Cisco IP Phone 7971 SCCP Configuration – Cont.

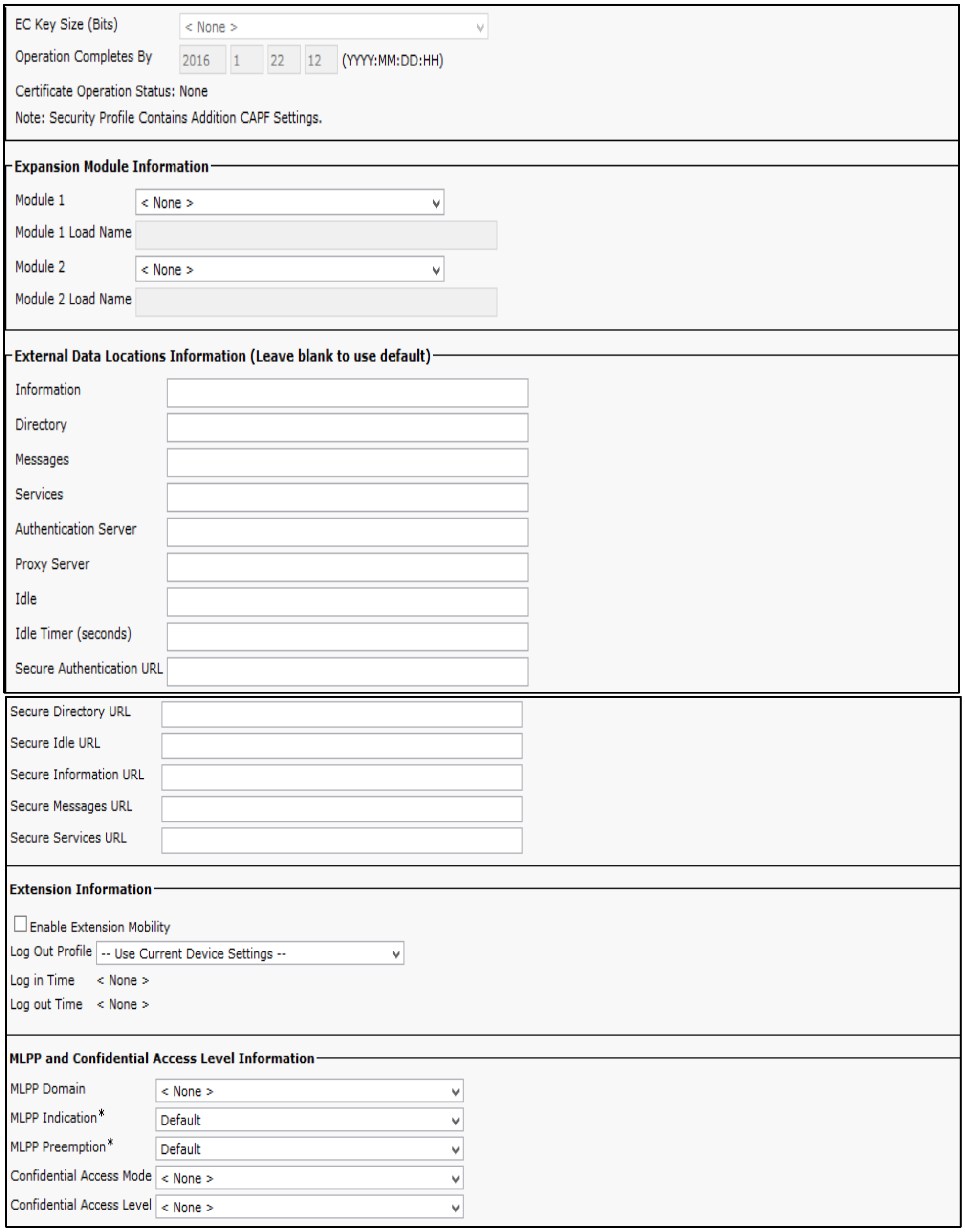


Figure 25: Cisco IP Phone 7971 SCCP Configuration – Cont.

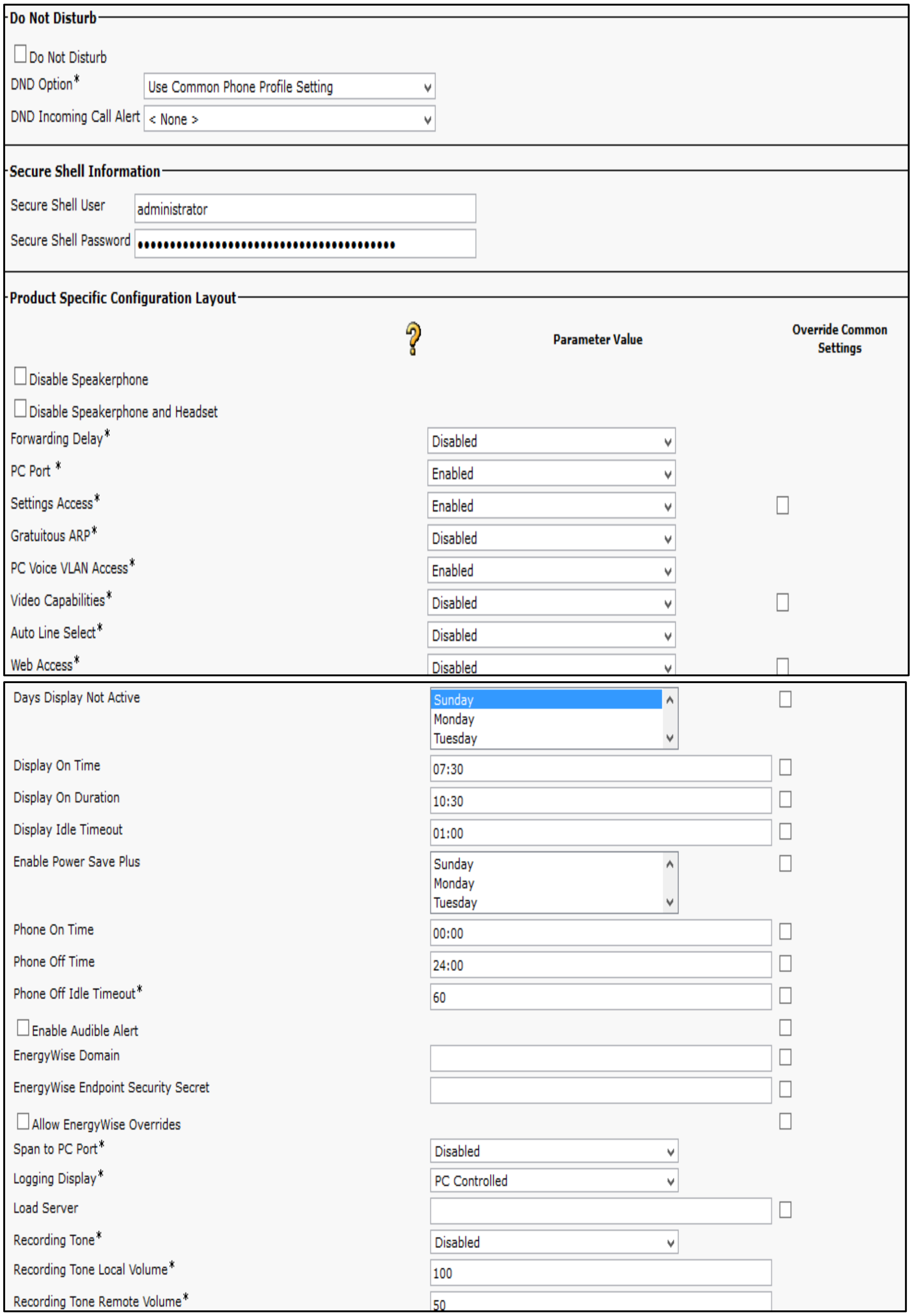


Figure 26: Cisco IP Phone 7971 SCCP Configuration – Cont.

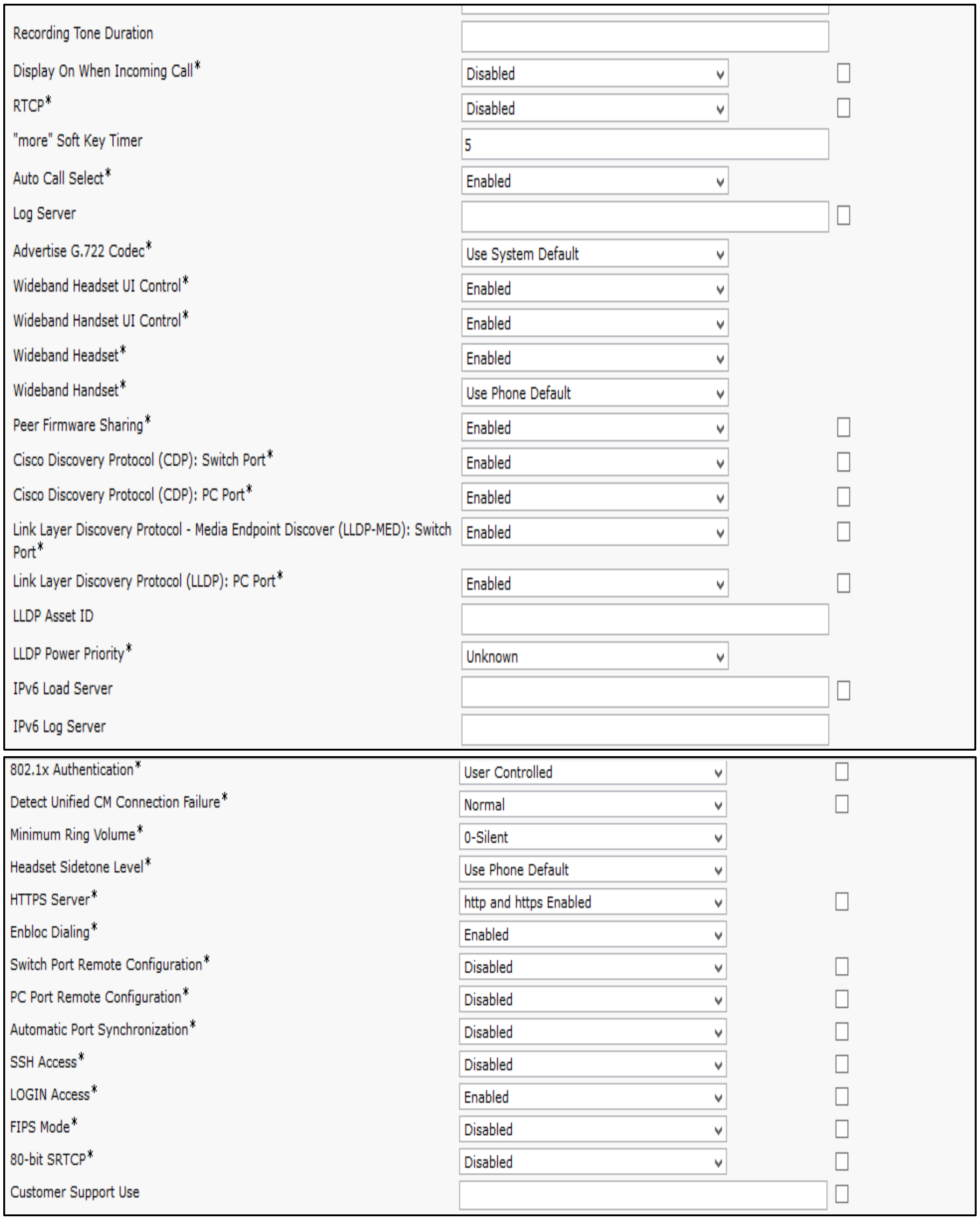


Figure 27: Cisco IP Phone 7971 SCCP Configuration – Cont.

Set Directory Number* = 1958 is used in this example
Set Description = SCCP-1958 is used in this example
Set Alerting Name = SCCP-1958 is used in this example
Set ASCII Alerting Name = SCCP-1958 is used in this example

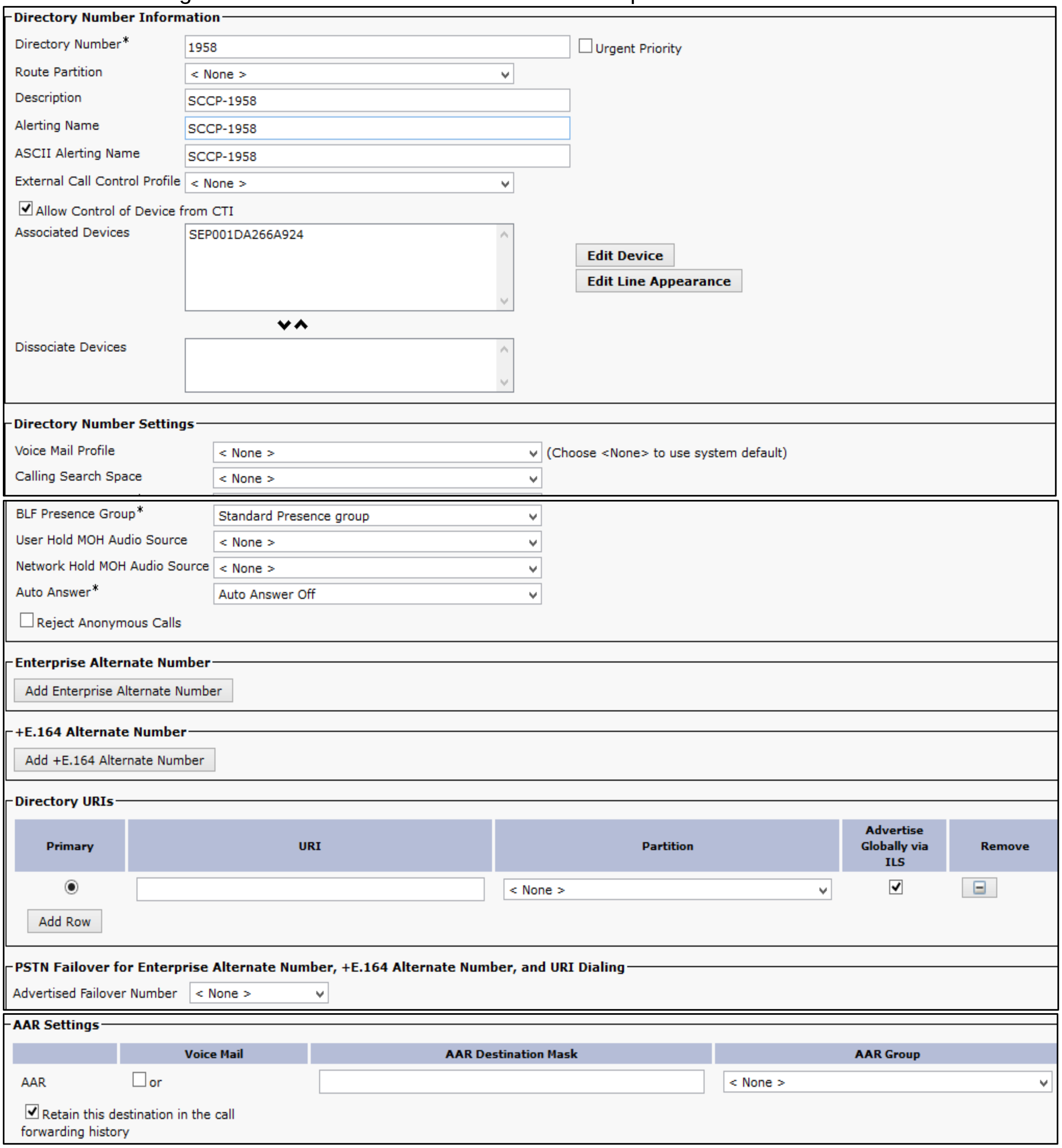
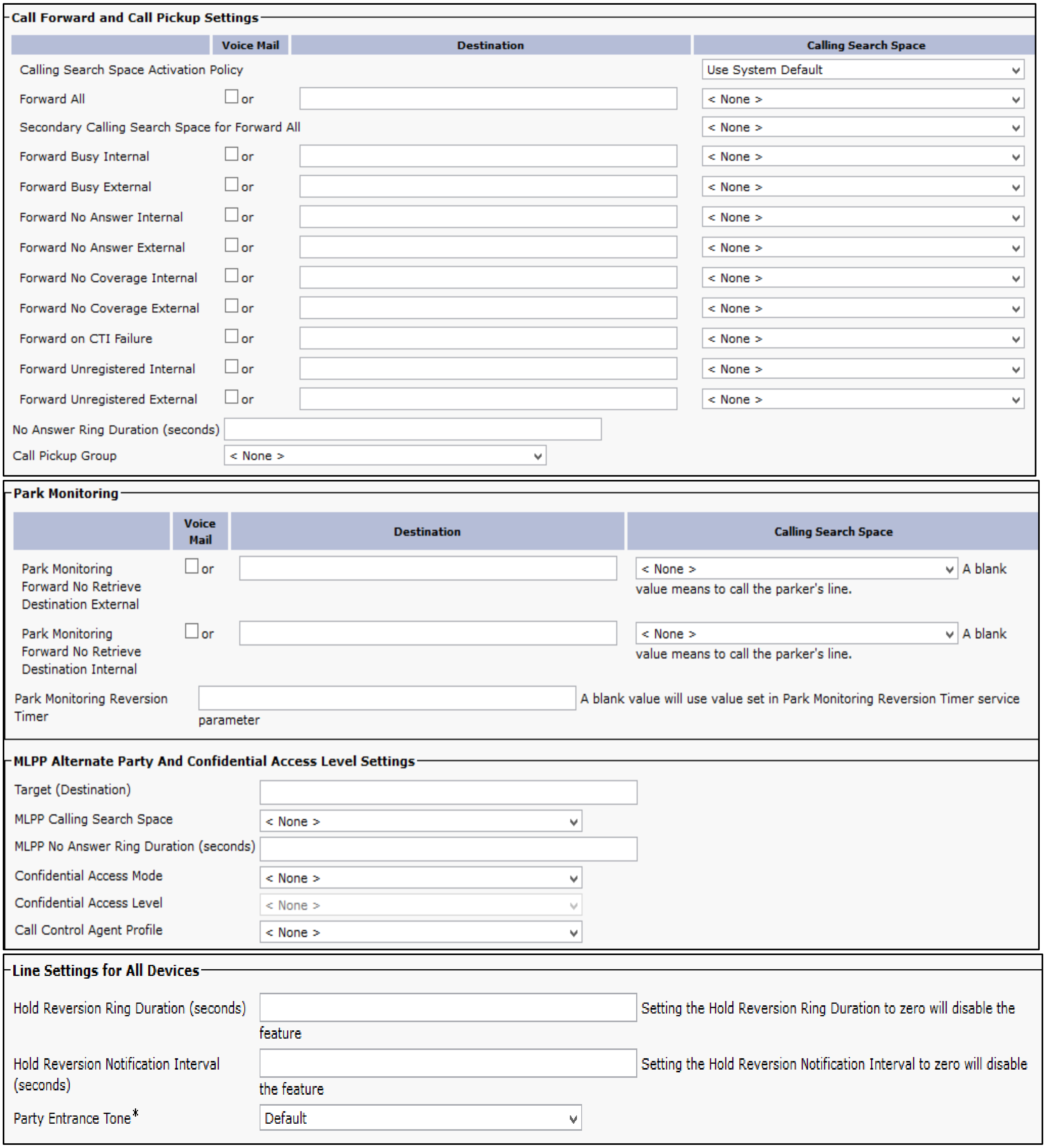


Figure 28: Cisco IP Phone 7971 SCCP Configuration – Cont.



Call Forward and Call Pickup Settings

	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			Use System Default
Forward All	☐ or		< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	☐ or		< None >
Forward Busy External	☐ or		< None >
Forward No Answer Internal	☐ or		< None >
Forward No Answer External	☐ or		< None >
Forward No Coverage Internal	☐ or		< None >
Forward No Coverage External	☐ or		< None >
Forward on CTI Failure	☐ or		< None >
Forward Unregistered Internal	☐ or		< None >
Forward Unregistered External	☐ or		< None >
No Answer Ring Duration (seconds)			
Call Pickup Group	< None >		

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	

Figure 29: Cisco IP Phone 7971 SCCP Configuration – Cont.

Set Display (Caller ID) = SCCP-1958 is used in this example
Set ASCII Display (Caller ID) = SCCP-1958 is used in this example
Set Line Text Label = SCCP-1958 is used in this example
Set External Phone Number Mask = 4026141958 is used in this example

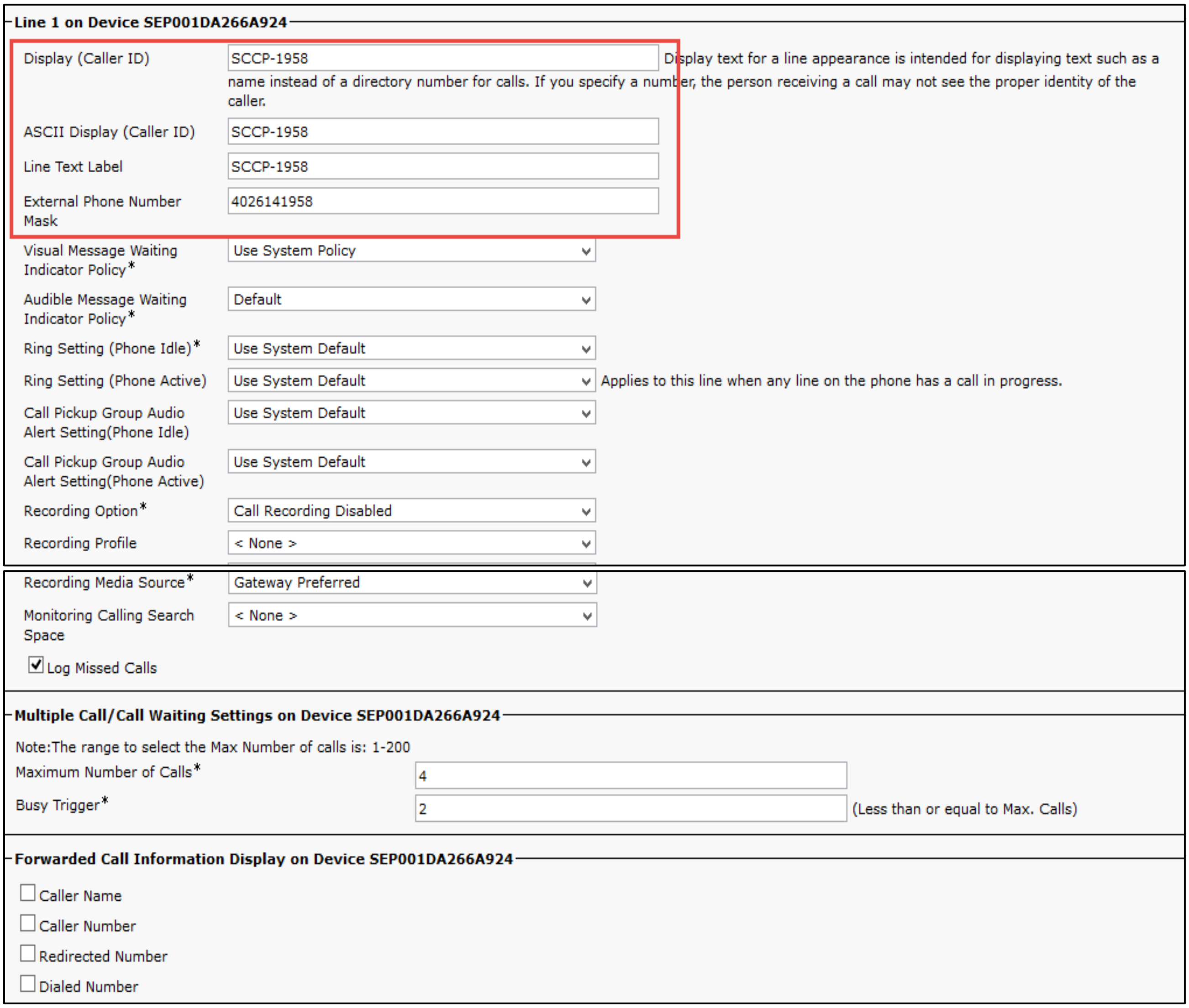


Figure 30: Cisco IP Phone 7971 SCCP Configuration – Cont.

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

Set Name* =cox non secure sip trunk profile is used in this example
Set Description = Non Secure SIP Trunk Profile authenticated by null String is used in this example
Set Incoming Transport Type* = TCP+UDP
Set Outgoing Transport Type = UDP. SIP trunks to Cox E-SBC should use UDP as a transport protocol for SIP. This is configured using the SIP Trunk Security profile, which is later assigned to the SIP trunk itself.

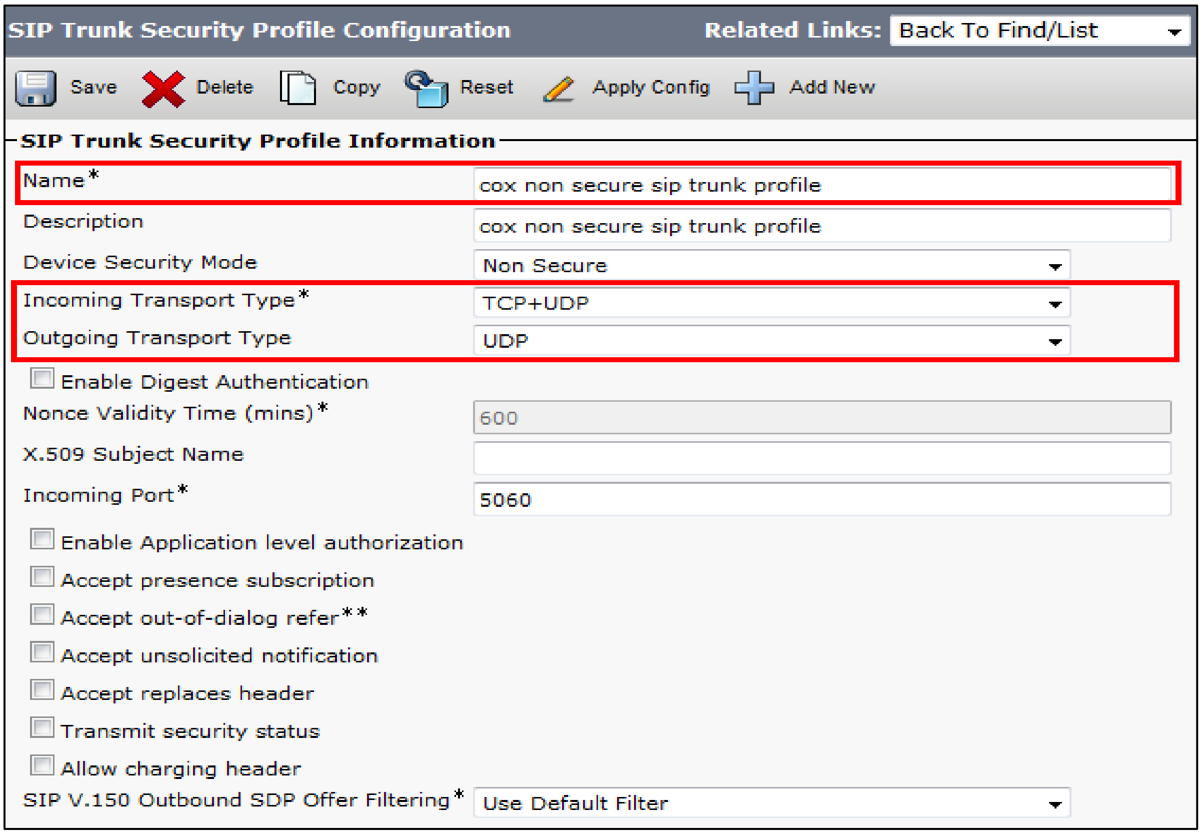


Figure 31: SIP Trunk Security Profile

SIP Profile

Navigation: Device → Device Settings → SIP Profile

Set SIP profile Name * = cox_sip_profile. This is used for this example
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".

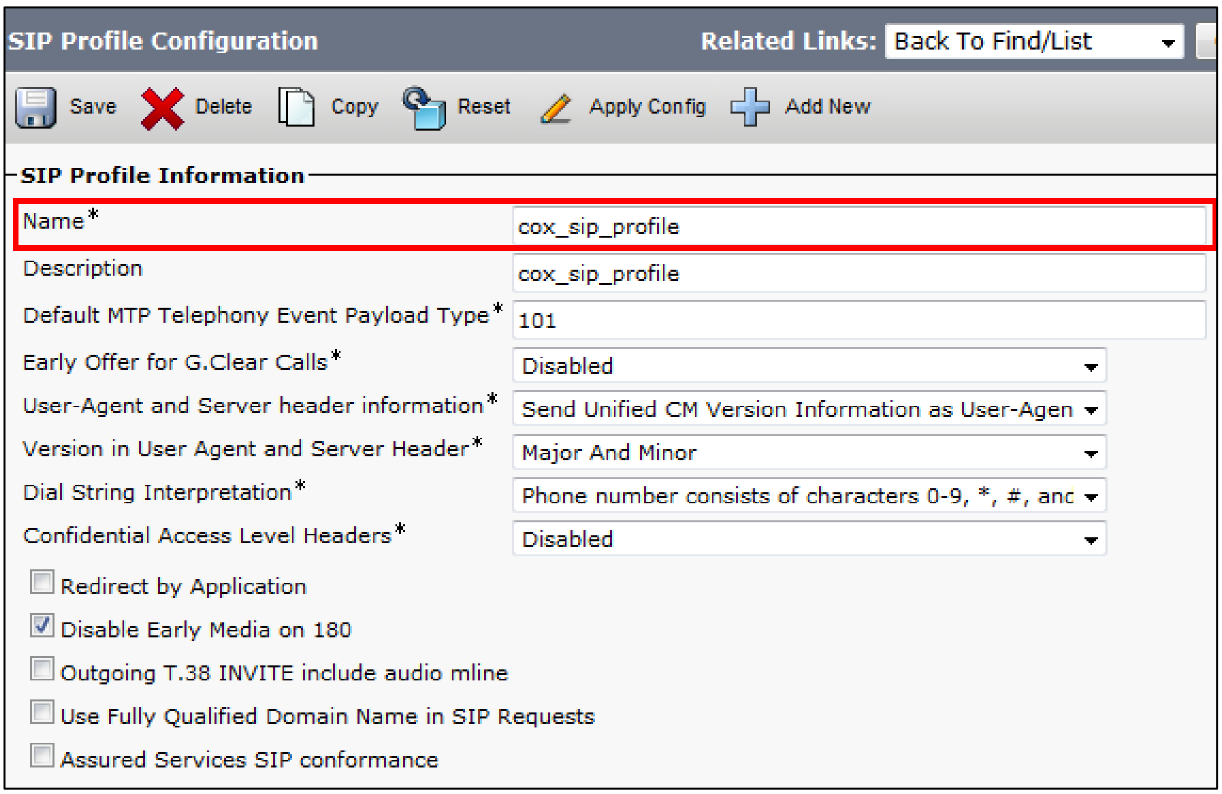
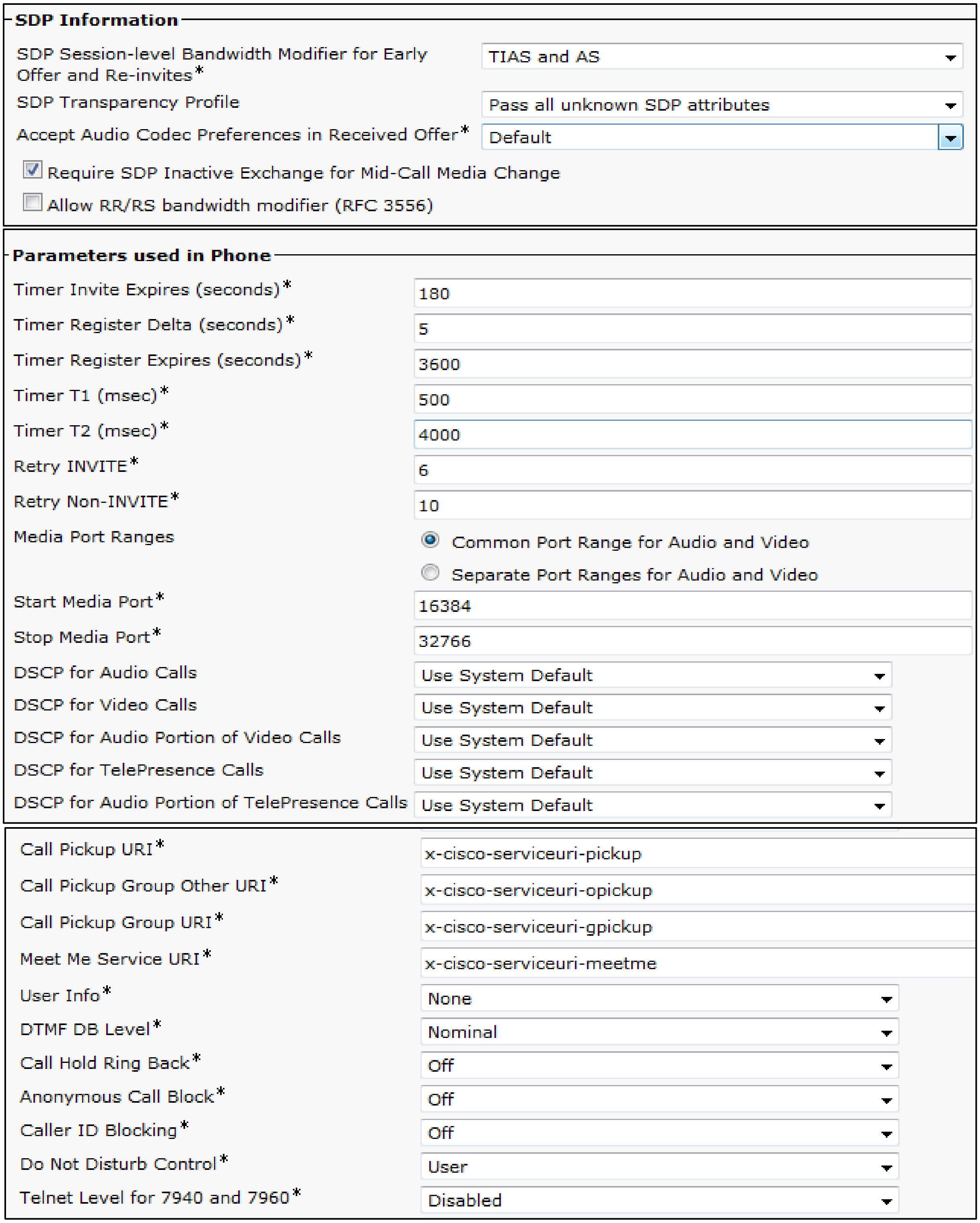


Figure 32: SIP Profile Configuration



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

- [x] 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
Media Port Ranges	(●) Common Port Range for Audio and Video ( ) Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766
DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default
DSCP for Audio Portion of Video Calls	Use System Default
DSCP for TelePresence Calls	Use System Default
DSCP for Audio Portion of TelePresence Calls	Use System Default

Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled

Figure 33: SIP Profile Configuration – Cont.

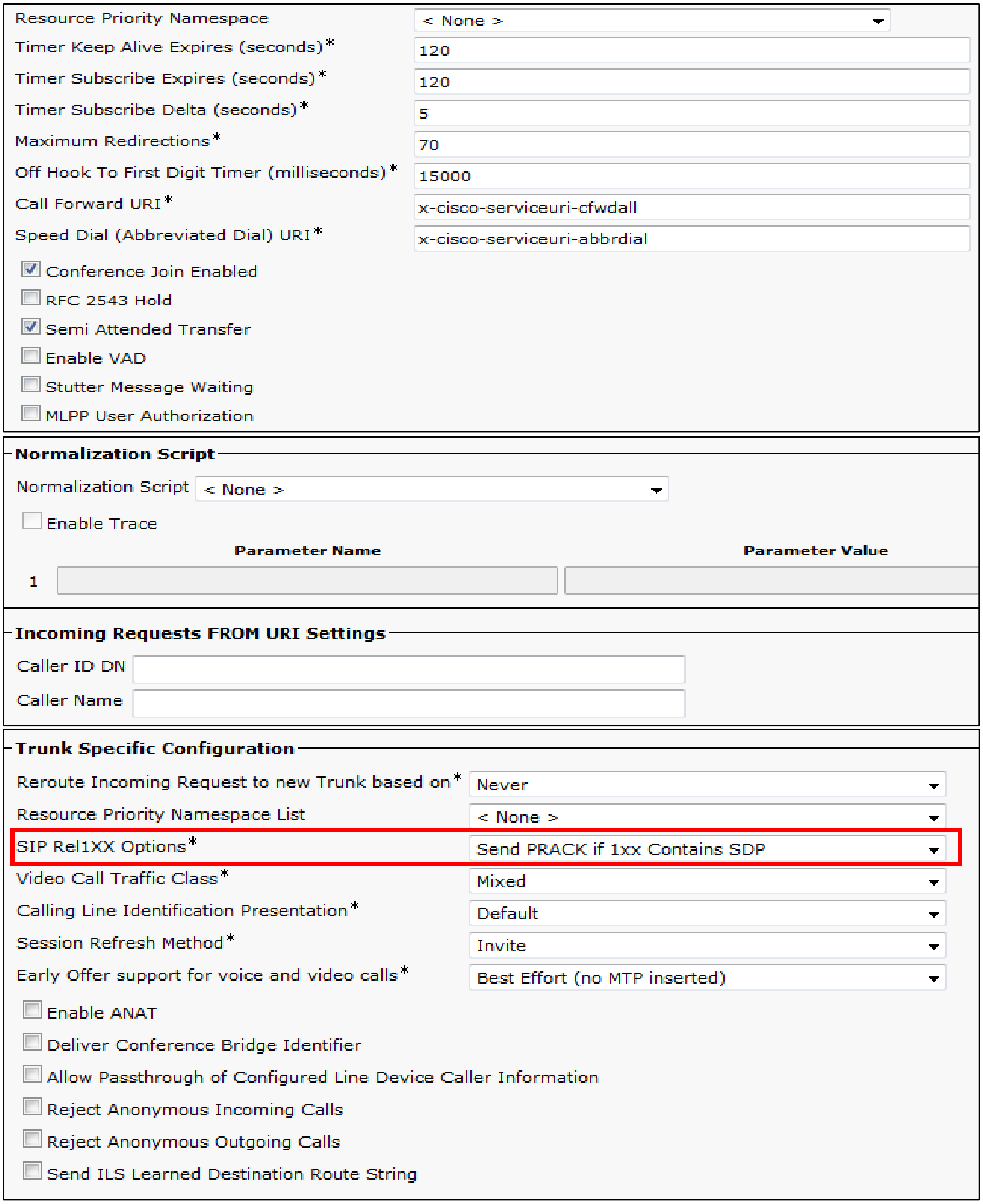


Figure 34: SIP Profile Configuration – Cont.

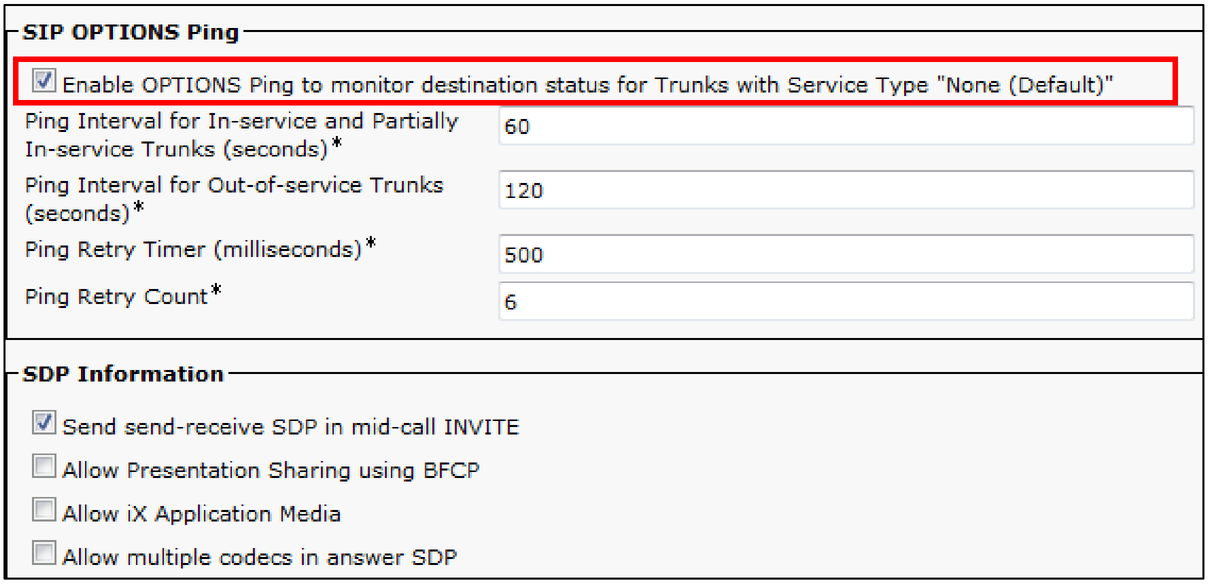


Figure 35: SIP Profile Configuration – Cont.

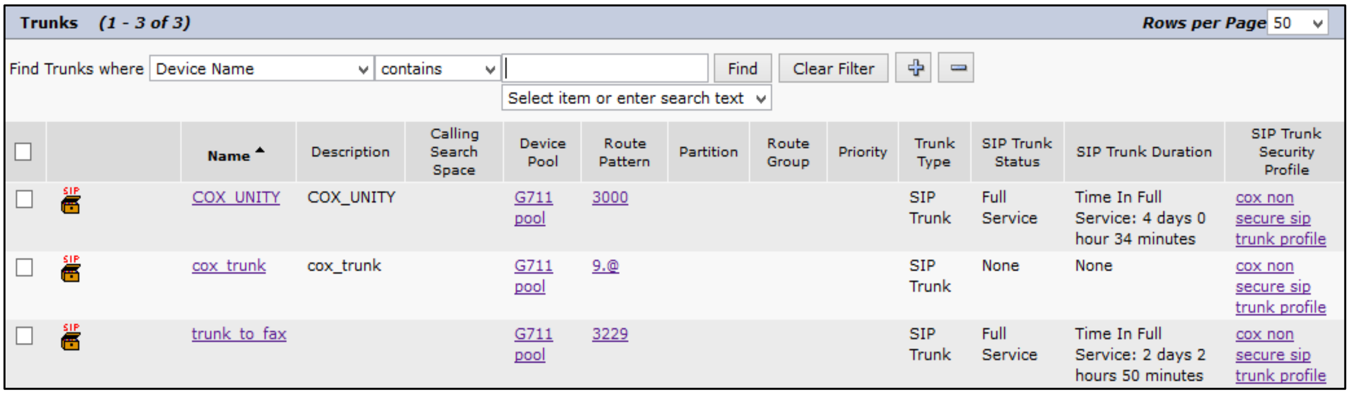
SIP Trunk to Cisco UBE Configuration

Navigation: Device → Trunk

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

Set Description = cox_trunk. This is used for this example

Set Media Resource Group List = MRGL_MTP



Trunks (1 - 3 of 3) Rows per Page 50

Find Trunks where Device Name contains Find Clear Filter

Select item or enter search text

		Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile
	SIP	COX_UNITY	COX_UNITY		G711 pool	3000				SIP Trunk	Full Service	Time In Full Service: 4 days 0 hour 34 minutes	cox non secure sip trunk profile
	SIP	cox_trunk	cox_trunk		G711 pool	9.@				SIP Trunk	None	None	cox non secure sip trunk profile
	SIP	trunk_to_fax			G711 pool	3229				SIP Trunk	Full Service	Time In Full Service: 2 days 2 hours 50 minutes	cox non secure sip trunk profile

Figure 36: SIP Trunk List

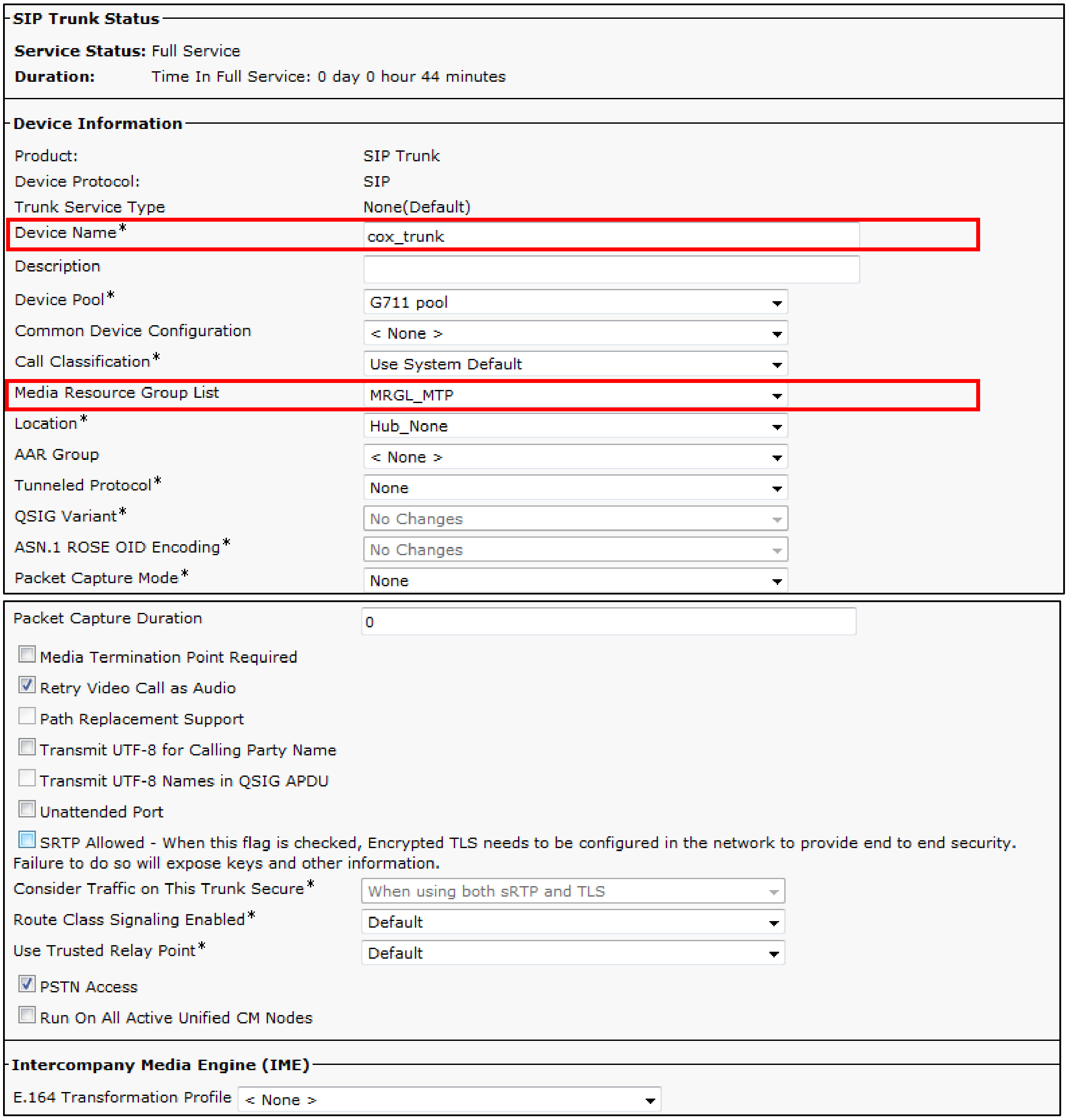


Figure 37: SIP Trunk to CUBE

Set Significant Digits* = 4. 4 digits Extension for all CPE phones

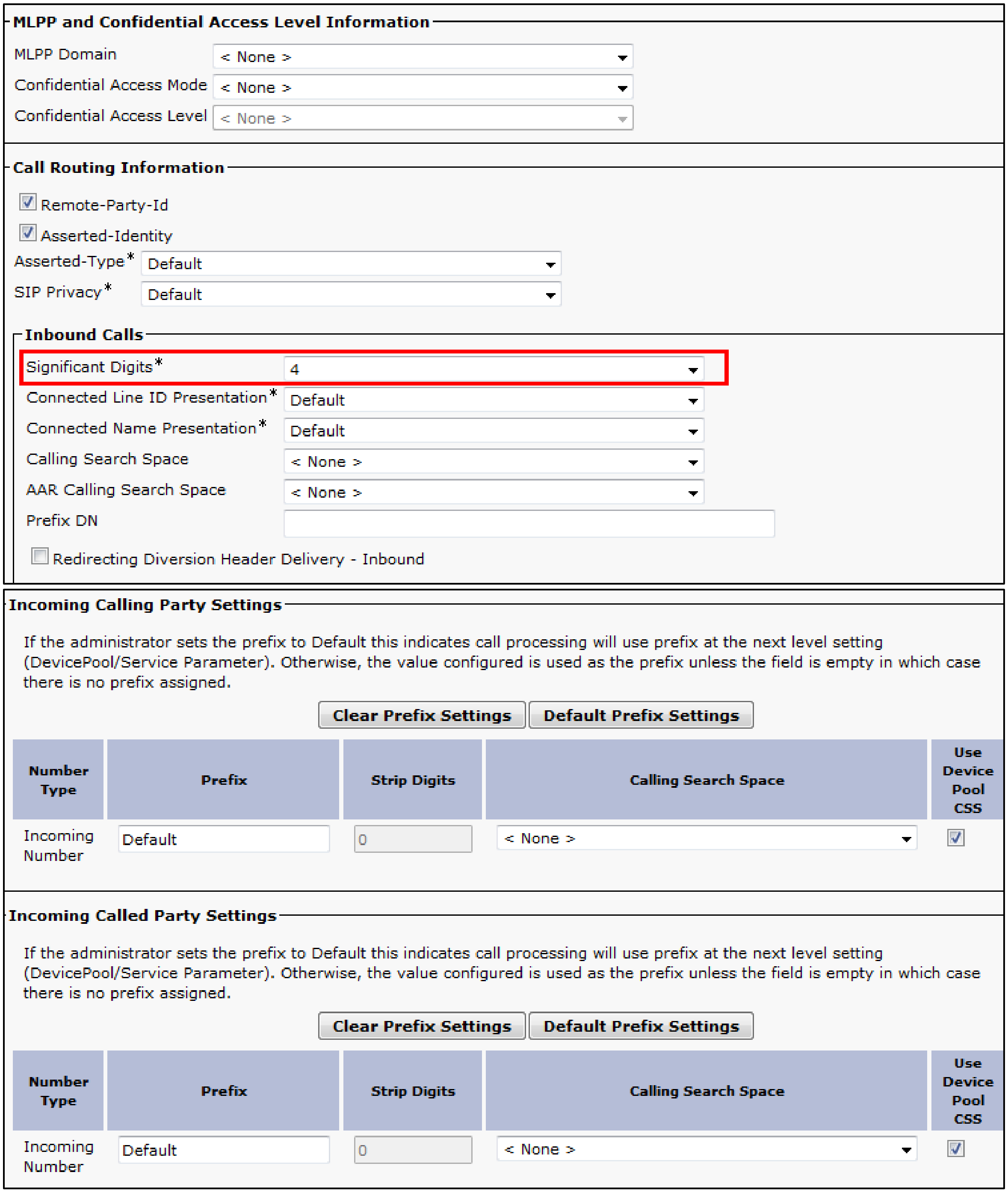


Figure 38: SIP Trunk to CUBE Configuration – Cont.

Set Destination Address = 10.80.13.20 .IP address of the CUBE
Set SIP Trunk Security Profile = Cox Non Secure SIP Trunk Profile. SIP Trunk Security Profile configured earlier
Set SIP Profile = Cox SIP Profile. SIP Profile configured earlier
DTMF Signaling Method = RFC 2833. RFC 2833 is supported for DTMF transport to/from Cox

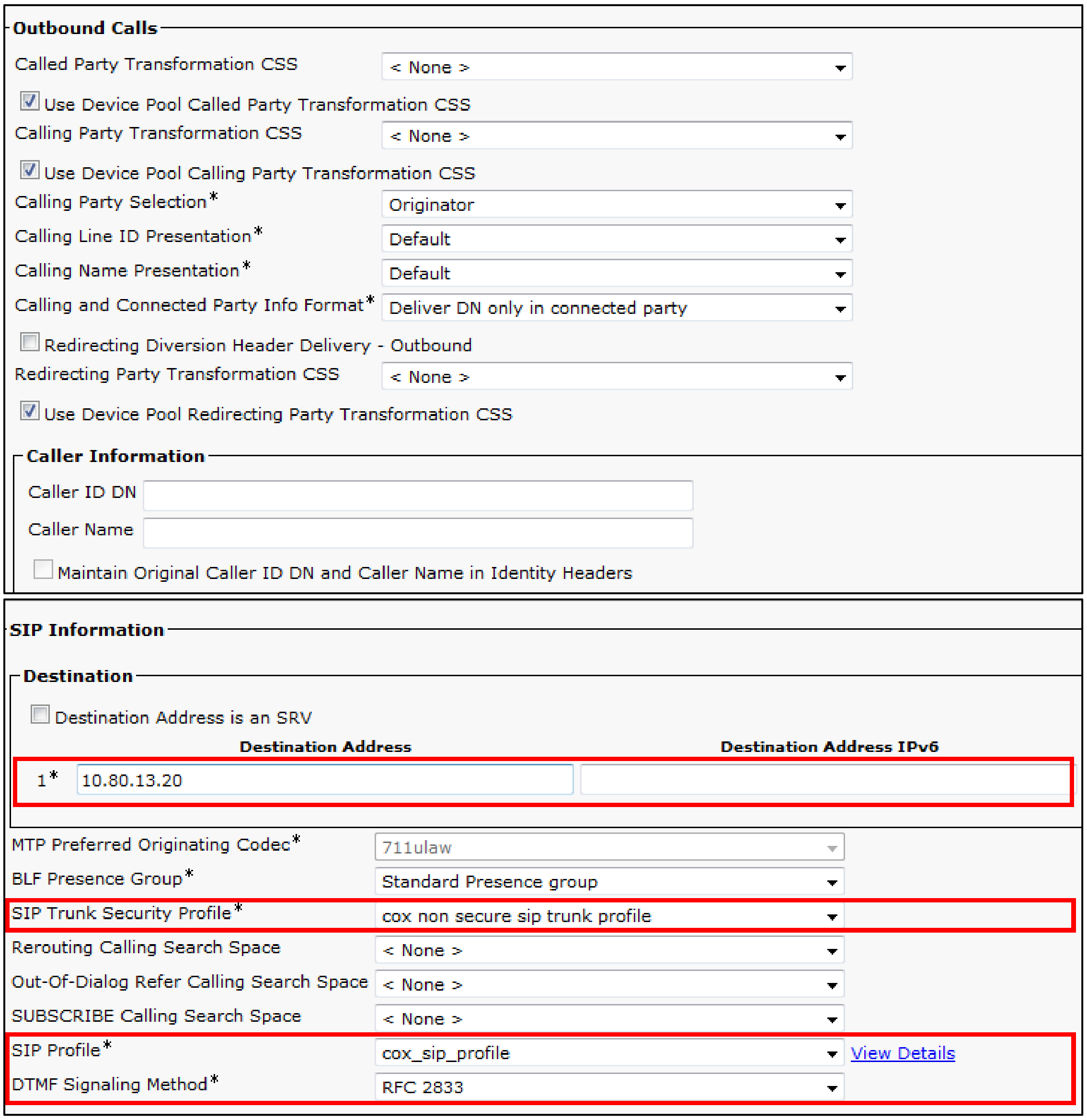


Figure 39: SIP Trunk to CUBE Configuration – Cont.

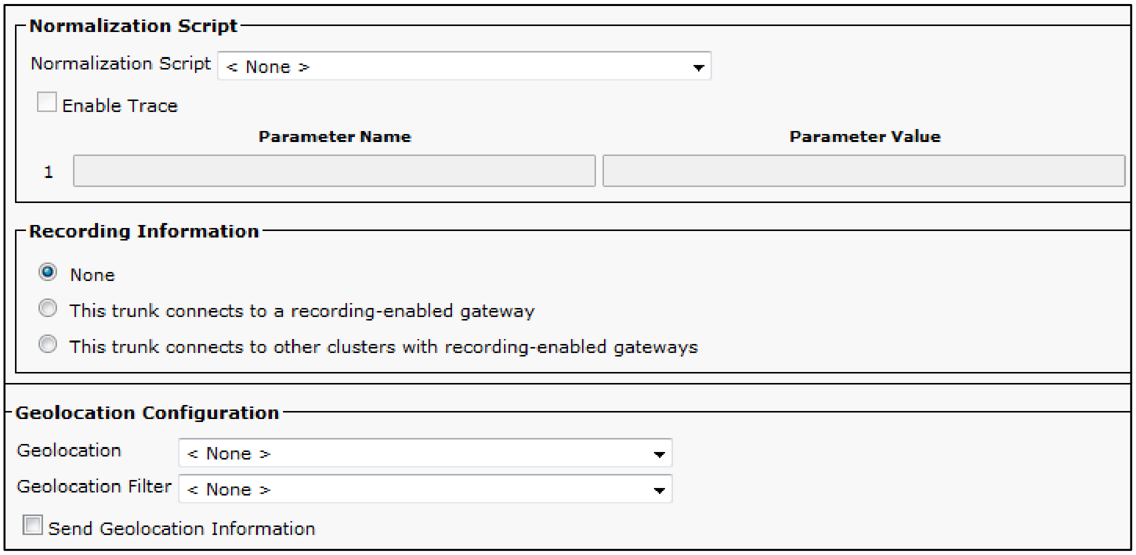


Figure 40: SIP Trunk to CUBE Configuration – Cont.

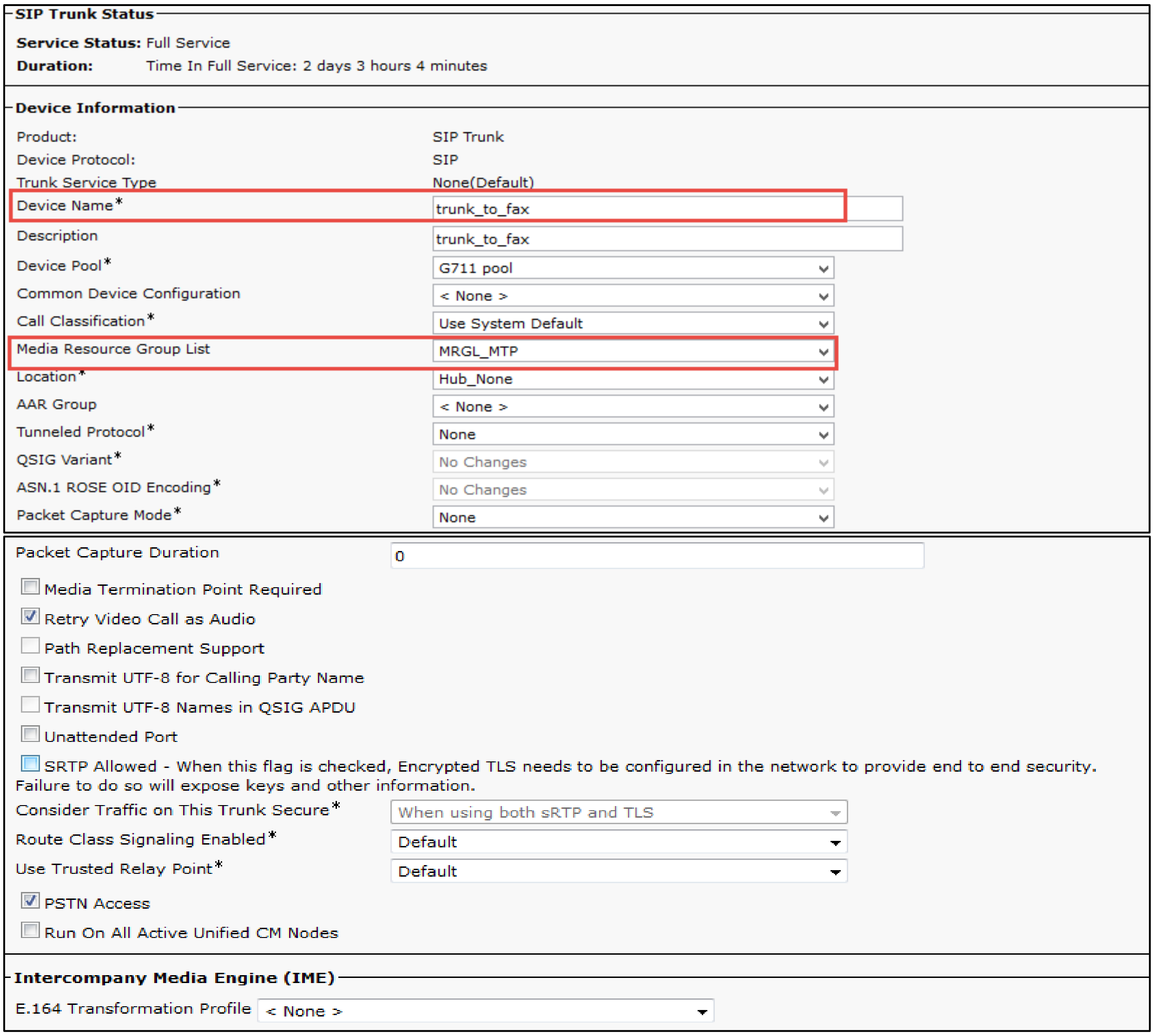
*Reset the trunk after the configuration is completed. Apply same procedure to create SIP trunks to Cisco Unity Connection.

SIP Trunk to Fax Gateway Configuration

Navigation: Device → Trunk

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

Set Media Resource Group List = MRGL_MTP



SIP Trunk Status

Service Status: Full Service

Duration: Time In Full Service: 2 days 3 hours 4 minutes

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	trunk_to_fax
Description	trunk_to_fax
Device Pool*	G711 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

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

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

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

Intercompany Media Engine (IME)

E.164 Transformation Profile: < None >

Figure 41: SIP trunk to FAX Gateway Configuration

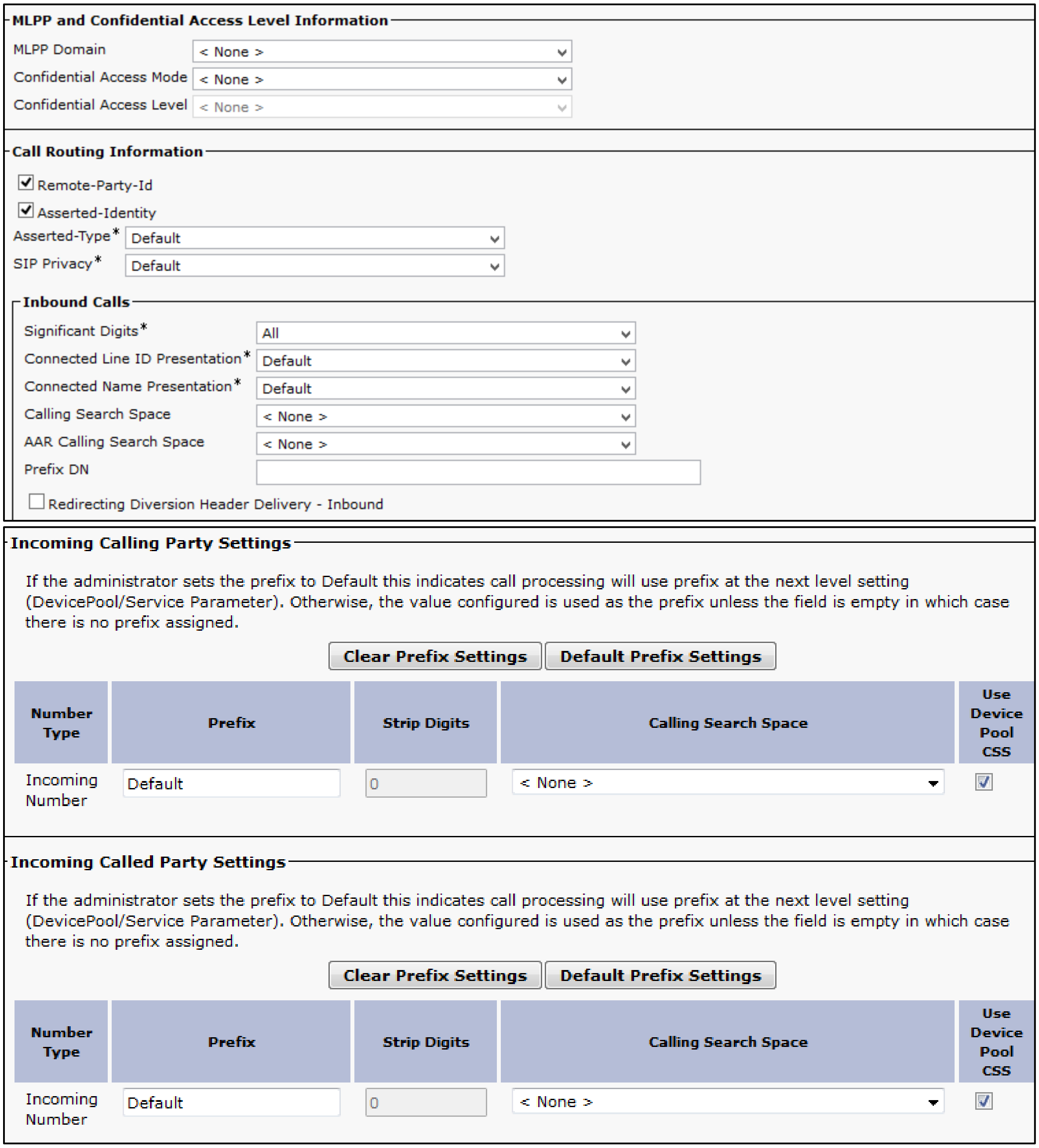


Figure 42: SIP Trunk to FAX Gateway Configuration – Cont.

Set Destination Address = 172.16.31.50 .IP address of the FAX Gateway
Set SIP Trunk Security Profile = Cox Non Secure SIP Trunk Profile. SIP Trunk Security Profile configured earlier
Set SIP Profile = Cox SIP Profile. SIP Profile configured earlier

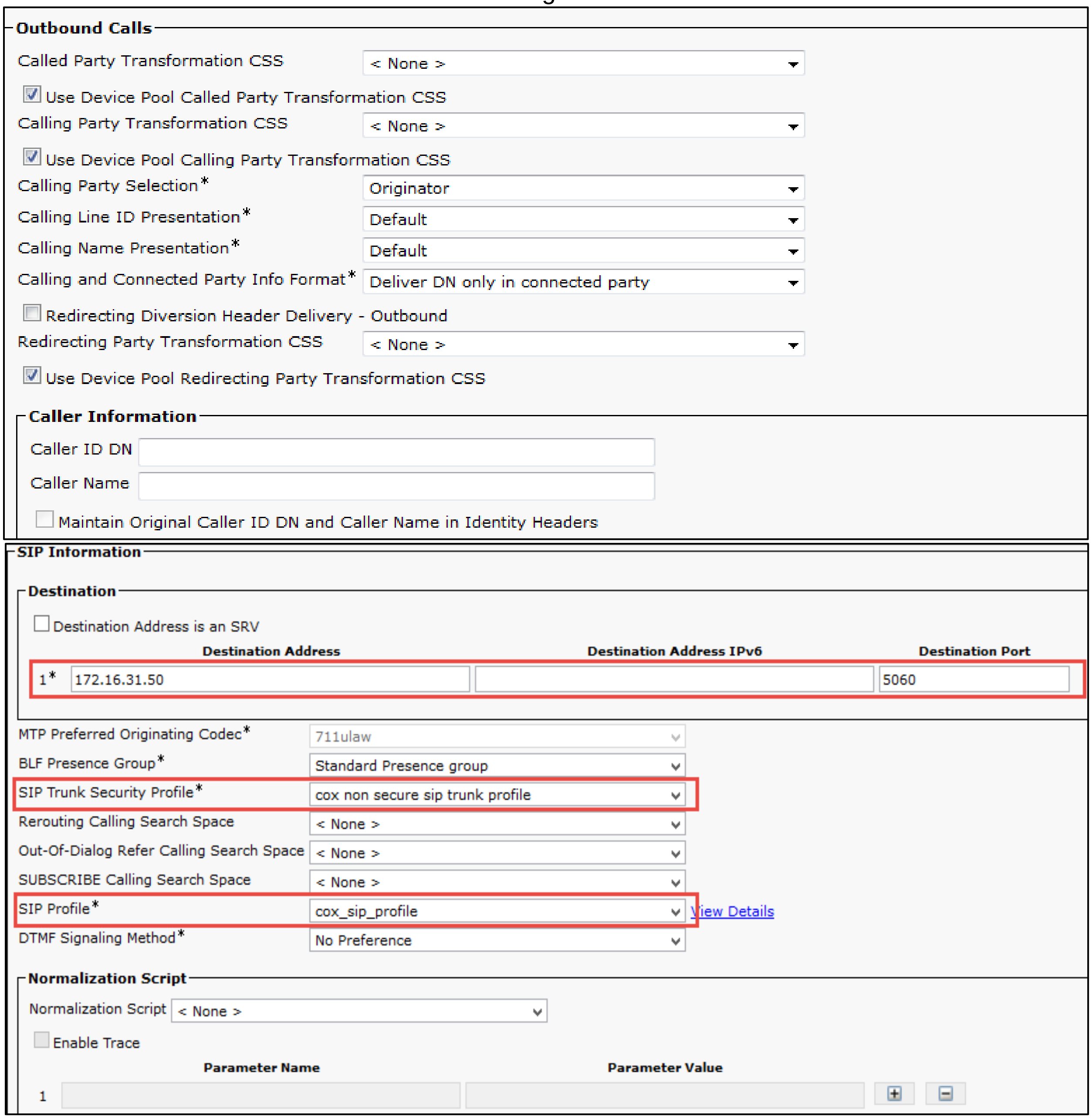


Figure 43: SIP Trunk to FAX Gateway Configuration – Cont.

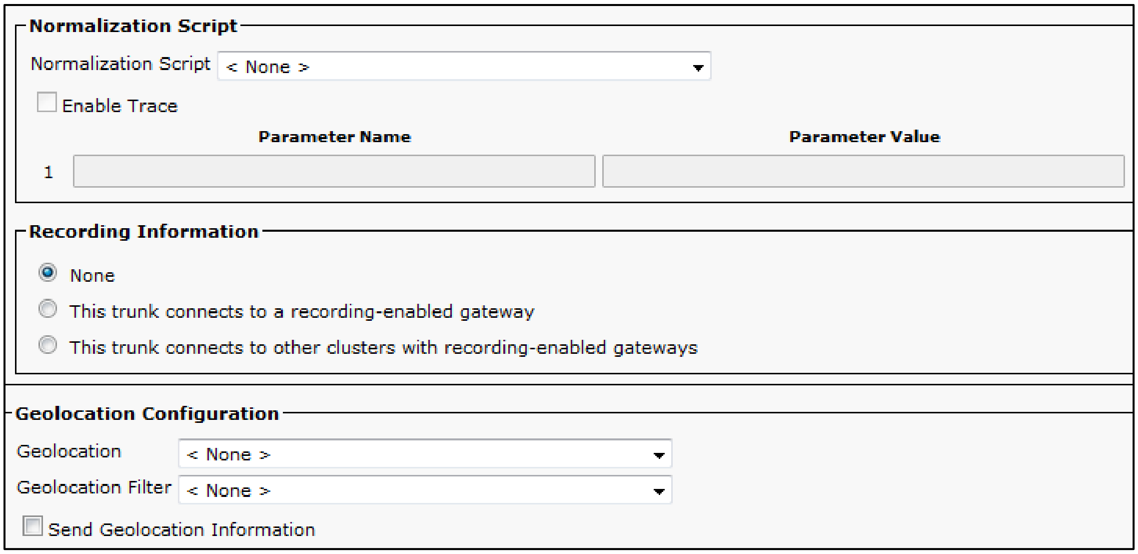


Figure 44: SIP Trunk to Fax Gateway Configuration – Cont.

*Reset the trunk after the configuration is completed. Apply same procedure to create SIP trunks to Cisco Unity Connection.

SIP Trunk to CUC

Navigation: Device → Trunk

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

Set Device Pool* = G711 pool. This is used for this example

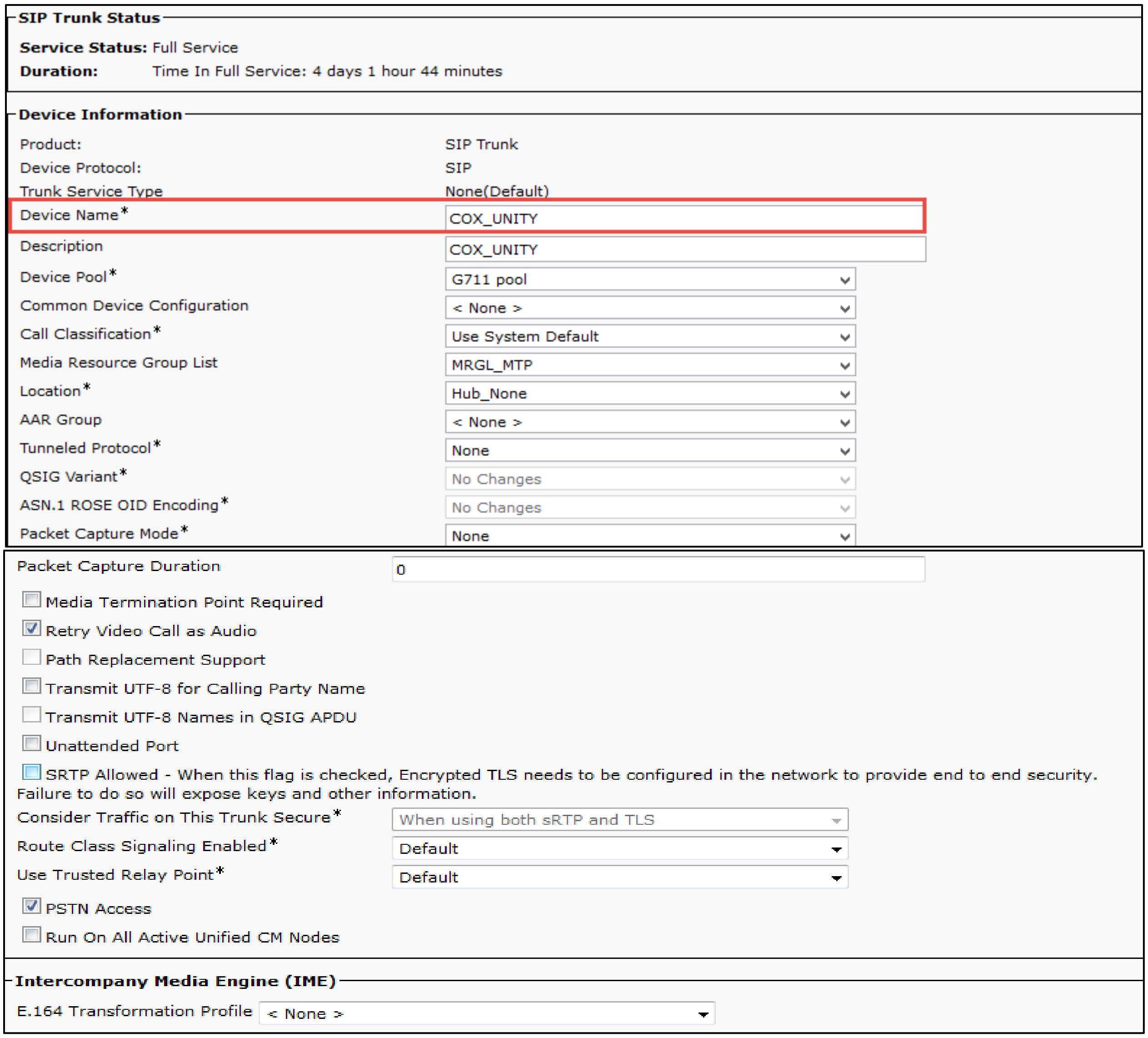


Figure 45: SIP Trunk to Unity

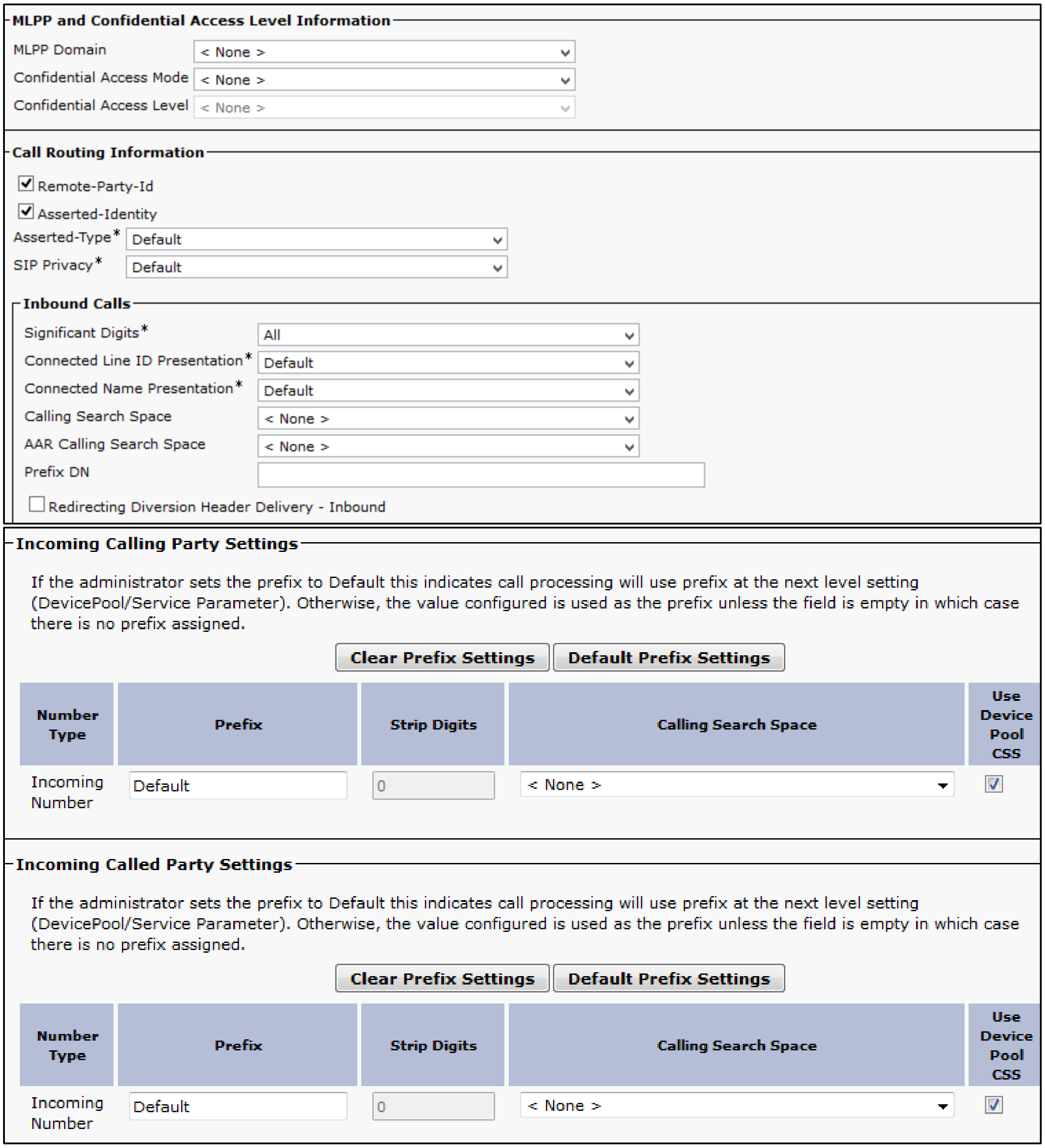


Figure 46: SIP Trunk to Unity – Cont.

Set Destination Address = 10.80.13.4 .IP address of the Unity
Set SIP Trunk Security Profile = Cox Non Secure SIP Trunk Profile. SIP Trunk Security Profile configured earlier
Set SIP Profile = Cox SIP Profile. SIP Profile configured earlier

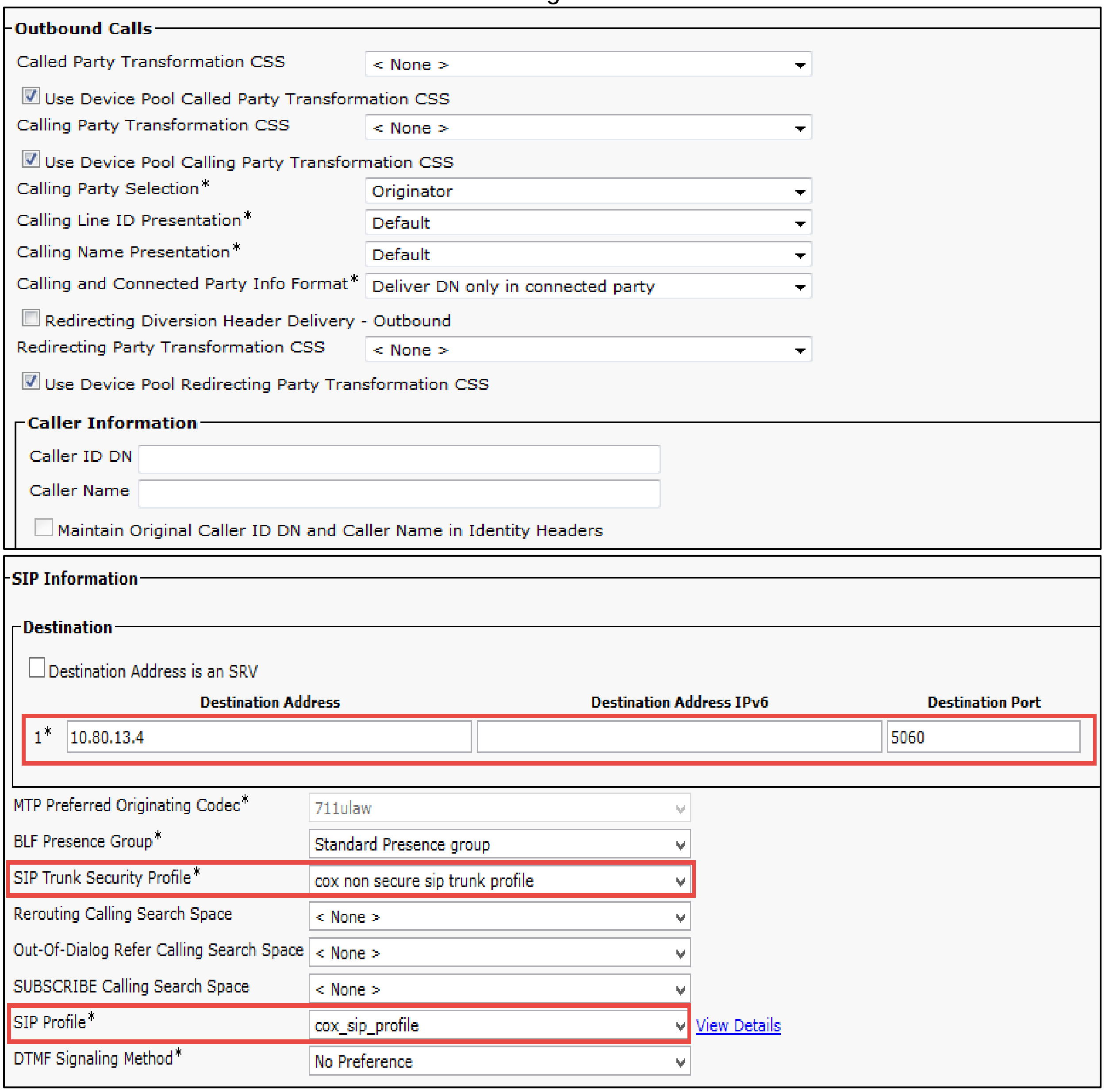


Figure 47: SIP Trunk to Unity – Cont.

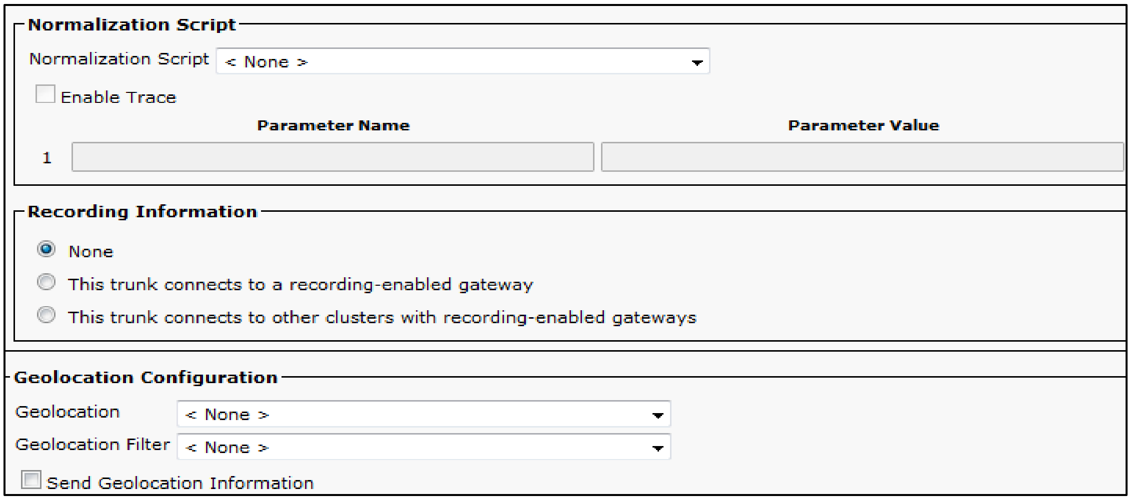


Figure 48: SIP Trunk to Unity – Cont.

*Reset the trunk after the configuration is completed. Apply same procedure to create SIP trunks to Cisco Unity Connection.

Voicemail Pilot Configuration

Navigation: Advanced Features → Voice Mail → Voice Mail Pilot

Set Voice Mail Pilot Number =3000. This is used for this example
Set Description = Pilot to Unity. This is used for this example.

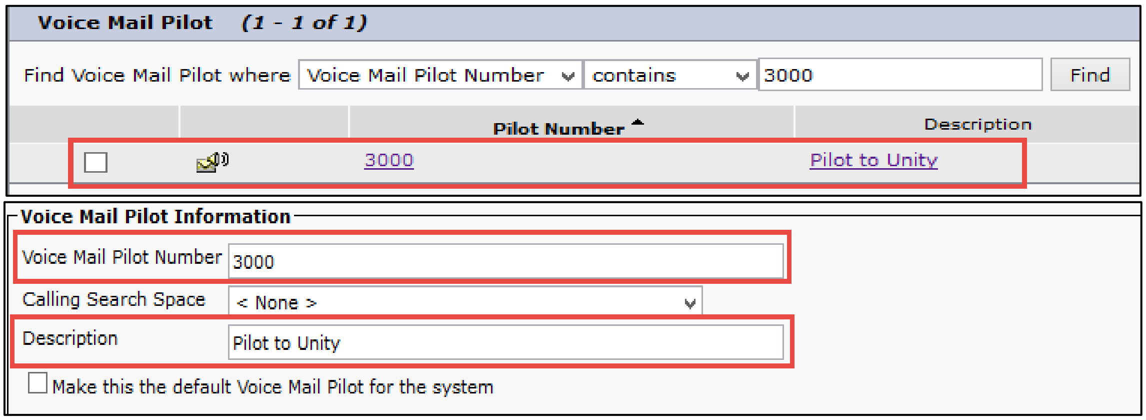


Figure 49: Voicemail Pilot Configuration

Voicemail Profile Configuration

Navigation: Advanced Features → Voice Mail → Voice Mail Profiles

Set Voice Mail Profile Name* =SIP. This is used for this example
Set Description = SIP TO UNITY. This is used for this example
Set Voice Mail Pilot** = 3000. This is used for this example

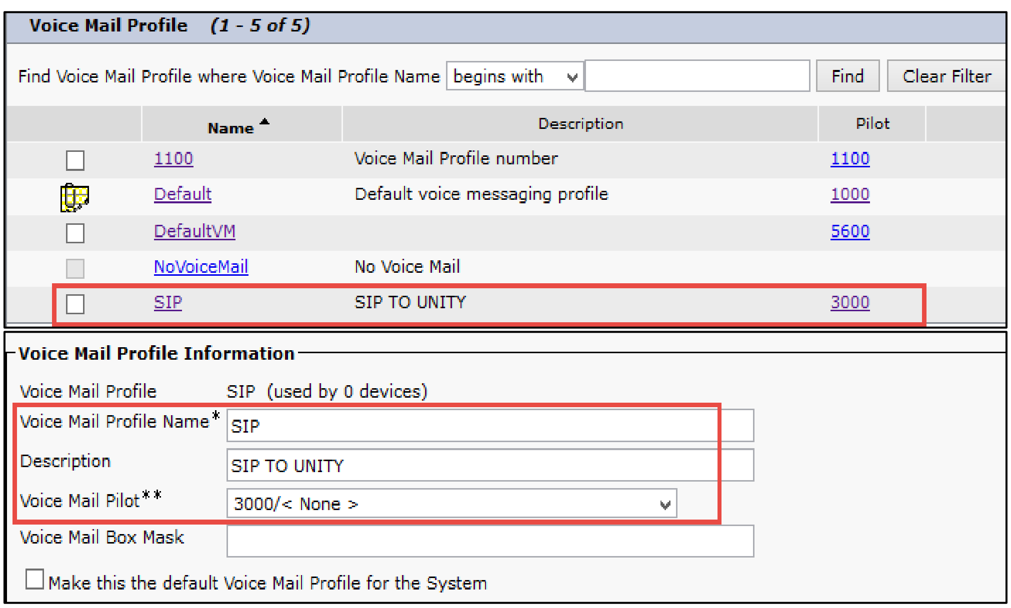


Figure 50: Voicemail Profile Configuration

Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

- Cisco IP phones dial 9+10 digits number to access PSTN via CUBE
- “9” is removed before send to CUBE
- For FAX call, Access Code 9 is used at fax gateway
- “9” is removed at UCM and 10 digits number is send to CUBE to Cox network
- Incoming fax call to 3229 will send to fax gateway
- 3000 is the Pilot Number for voicemail to Unity Connection

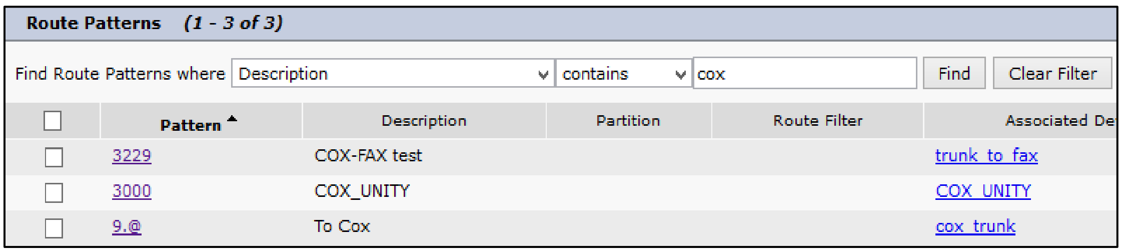


Figure 51: Route Patterns

Set Route Pattern* =9.@ for Voice call ,3000 for Unity and 3229 for the Fax call. Specify appropriate Route Pattern

Set Gateway/Route List* = cox_trunk. SIP Trunk name configured earlier

Discard Digits = PreDot. Specifies how to modify digit before they are sending to Cox ESBC

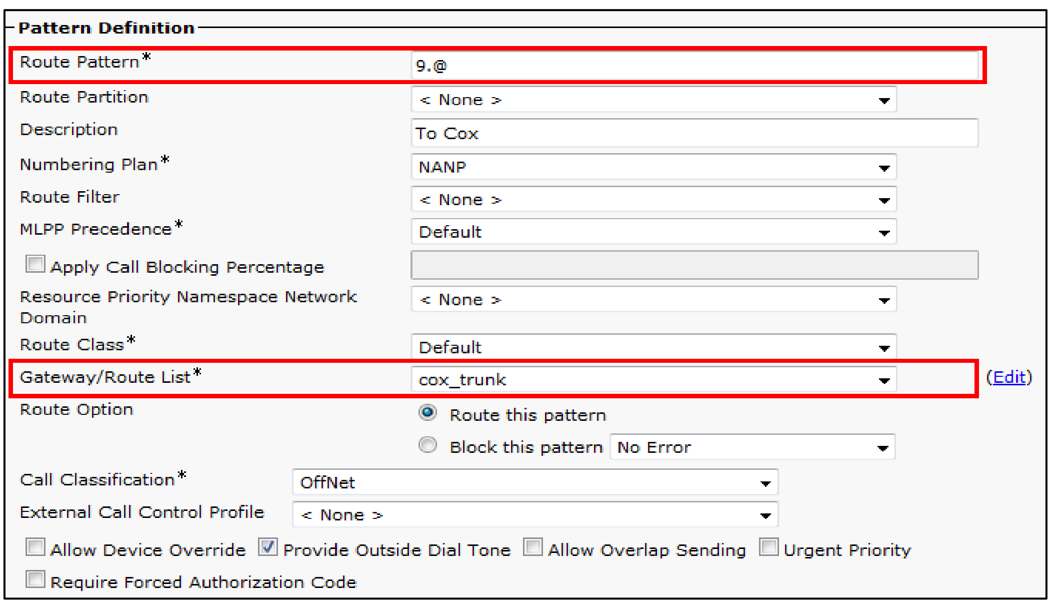


Figure 52: Route Patterns for Voice

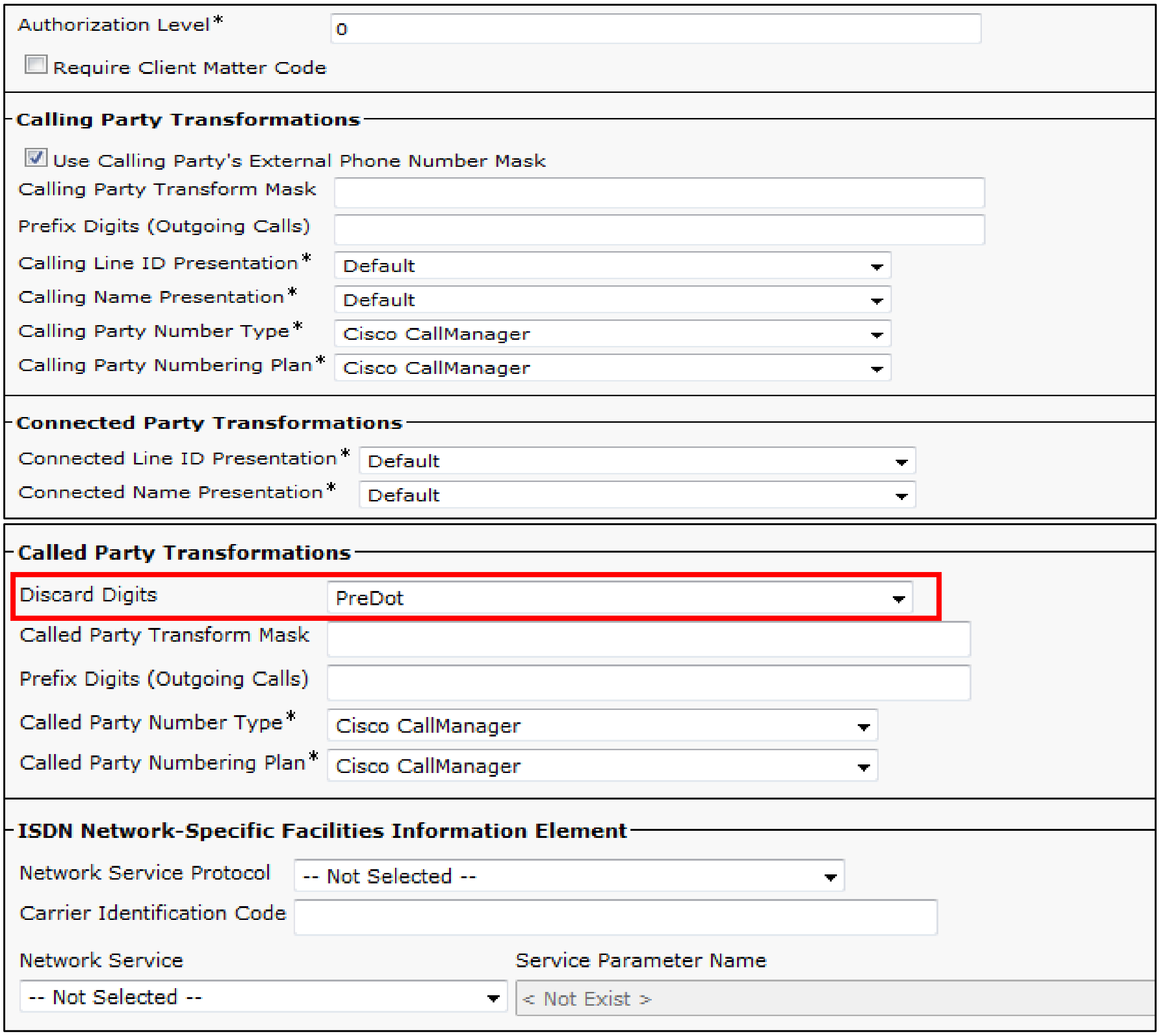


Figure 53: Route Patterns for Voice – Cont.

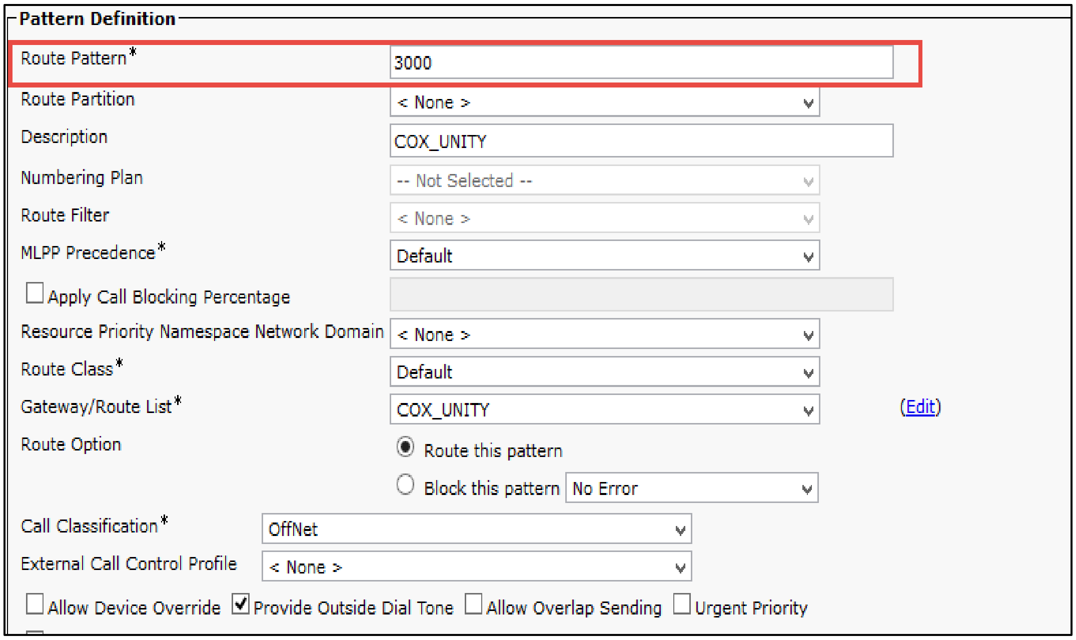


Figure 54: Route Patterns for Unity

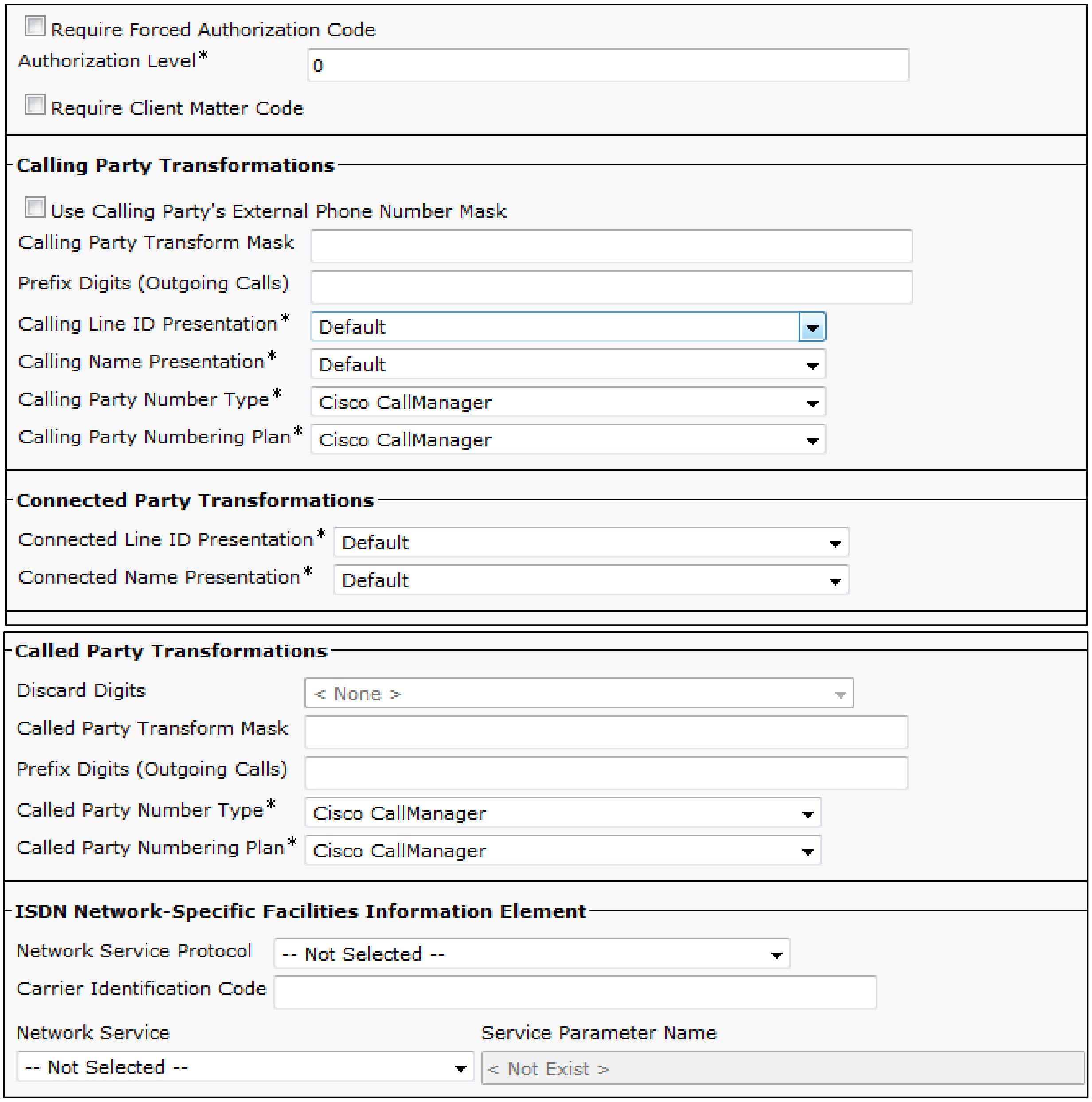


Figure 55: Route Patterns for Unity – Cont.

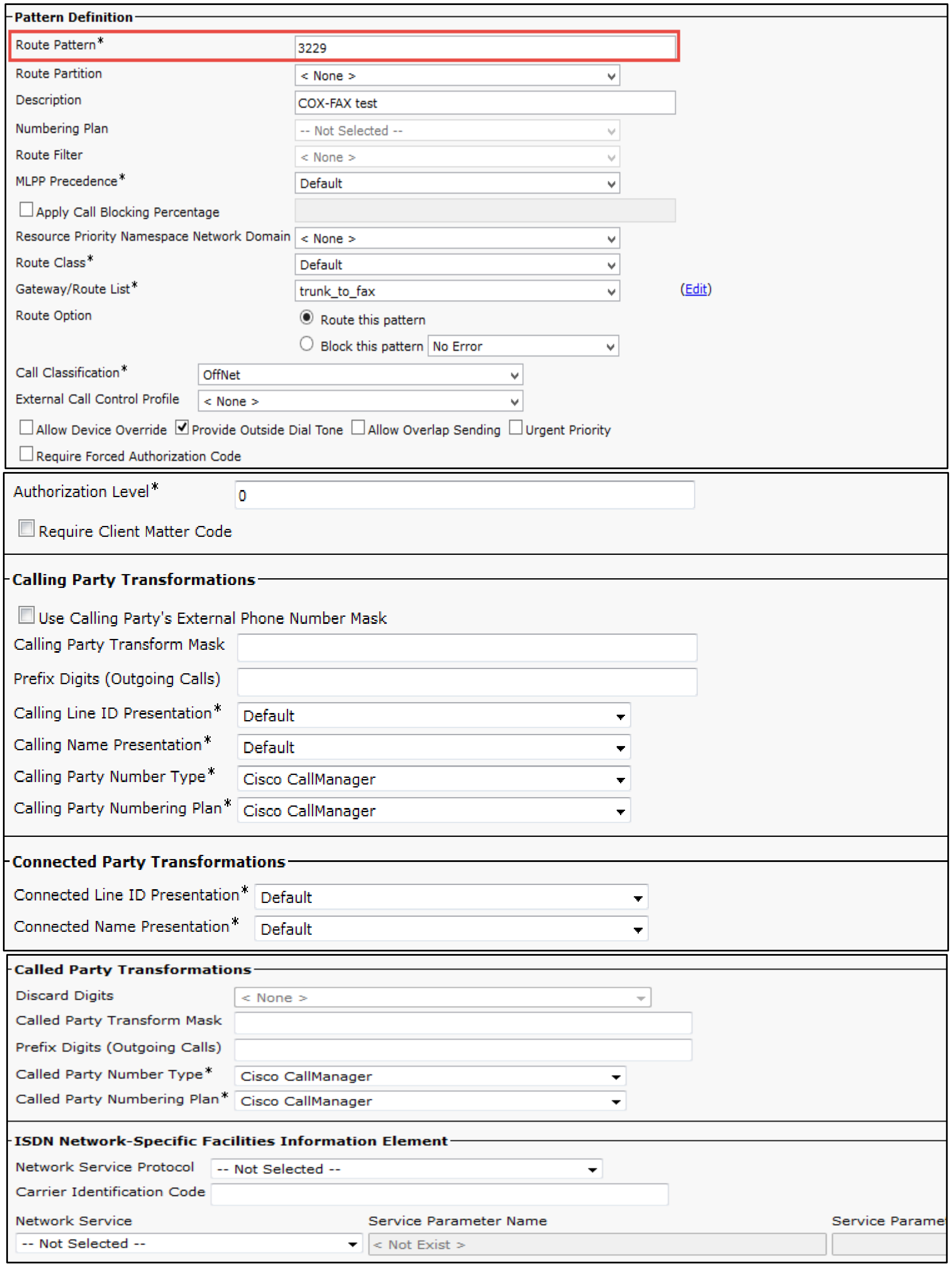


Figure 56: Route Patterns for Fax

FAX Gateway Configuration

```
cme.in.tekvizion.com#sh running config
!
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname cme.in.tekvizion.com
!
boot-start-marker
boot-end-marker
!
aaa new-model
!
!
aaa authentication login local_auth local
!
aaa session-id common
clock timezone IST 5 30
network-clock-participate wic 2
network-clock-participate wic 3
!
dot11 syslog
ip source-route
!
!
ip cef
```

```
!
ip host Clus1-862-Pub 172.16.26.2
no ipv6 cef
multilink bundle-name authenticated
!
isdn switch-type primary-qsig
voice rtp send-recv
!
voice service pots
!
voice service voip
 no ip address trusted authenticate
 allow-connections sip to sip
 no supplementary-service sip handle-replaces
 redirect ip2ip
 fax protocol pass-through g711ulaw
 sip
  g729 annexb-all
!
voice class codec 3
 codec preference 1 g711ulaw
!
voice-card 0
!
crypto pki token default removal timeout 0
!
!
!
!
```

```
license udi pid CISCO2851 sn FHK7867F4LY
username cisco password 0
!
interface GigabitEthernet0/0
 ip address 172.16.31.50 255.255.255.0
 duplex auto
 speed auto
!
interface GigabitEthernet0/1
 no ip address
 ip nat outside
 ip virtual-reassembly in
 shutdown
 duplex auto
 speed auto

ip forward-protocol nd
!
ip http server
no ip http secure-server
!
ip route 0.0.0.0 0.0.0.0 172.16.31.1
!
access-list 1 permit 172.16.31.0 0.0.0.255
!
snmp-server community public RO
snmp-server location Chennai
!
control-plane
```

```
!
voice-port 0/1/1
 cptone IN
 station-id number
 caller-id enable
!
dial-peer voice 777 pots
 huntstop
 destination-pattern 3229
 port 0/1/1
 forward-digits all
!
dial-peer voice 9224 voip
 description CUCM to Gateway
session protocol sipv2
 session transport udp
 incoming called-number 3229
 voice-class codec 3
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 92240 voip
 description Gateway to CUCM
 destination-pattern 9T
 session protocol sipv2
 session target ipv4:10.80.13.2
```

```
 session transport udp
 voice-class codec 3
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
!
gateway
 timer receive-rtp 1200
!
sip-ua
 retry register 5
 timers connection aging 30
 timers update 1000
 no timers hold
 timers register 1000
!
!
line con 0
line aux 0
line 66
 no activation-character
 no exec
 transport preferred none
 transport input all
 transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
line 194
```

```
 no activation-character
 no exec
 transport preferred none
 transport input all
 transport output all
line vty 0 4
 session-timeout 180
 exec-timeout 0 0
 password
 login authentication local_auth
 transport input all
!
scheduler allocate 20000 1000
ntp server
end
```

Cisco UCM SIP Integration with Cisco Unity Connection (CUC)

CUC Version

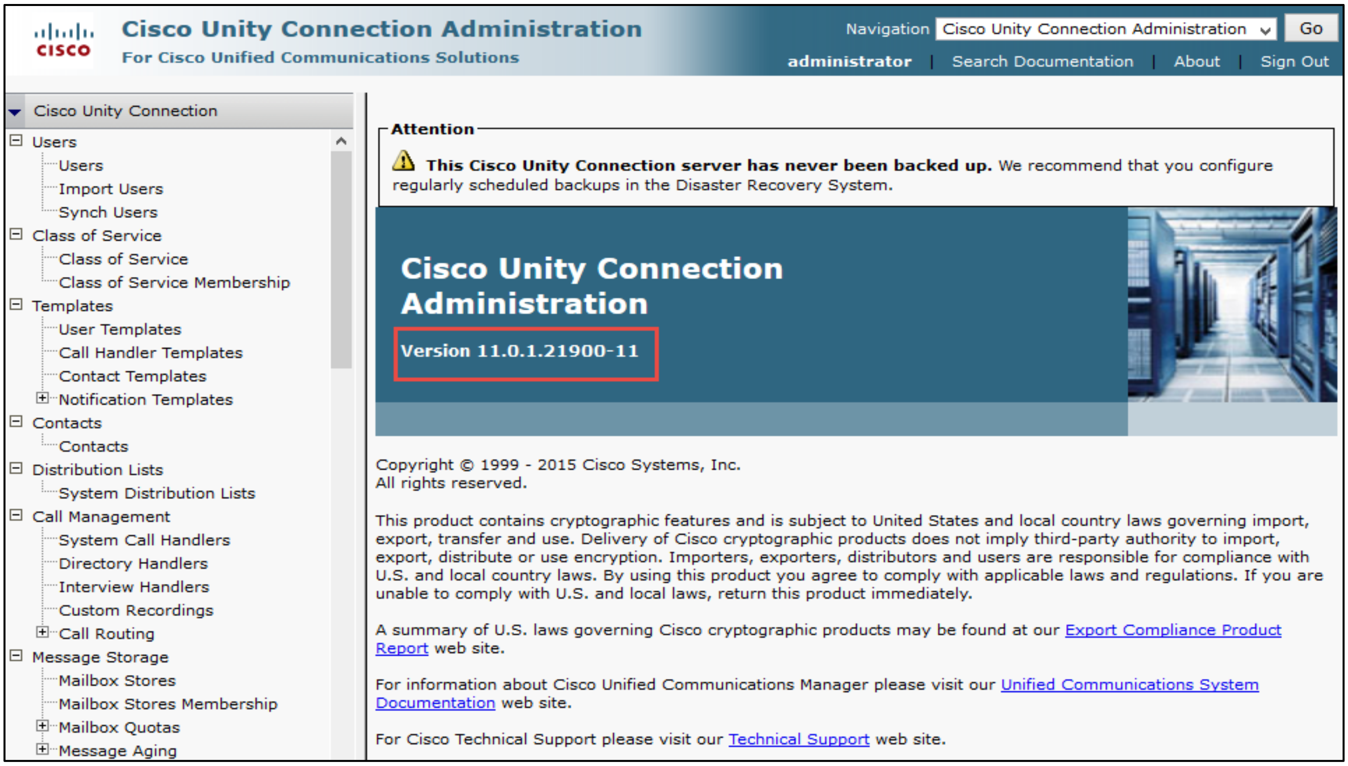


Figure 57: CUC Version

CUC Telephony Integration with Cisco UCM

Navigation: Telephony Integrations → Phone system

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

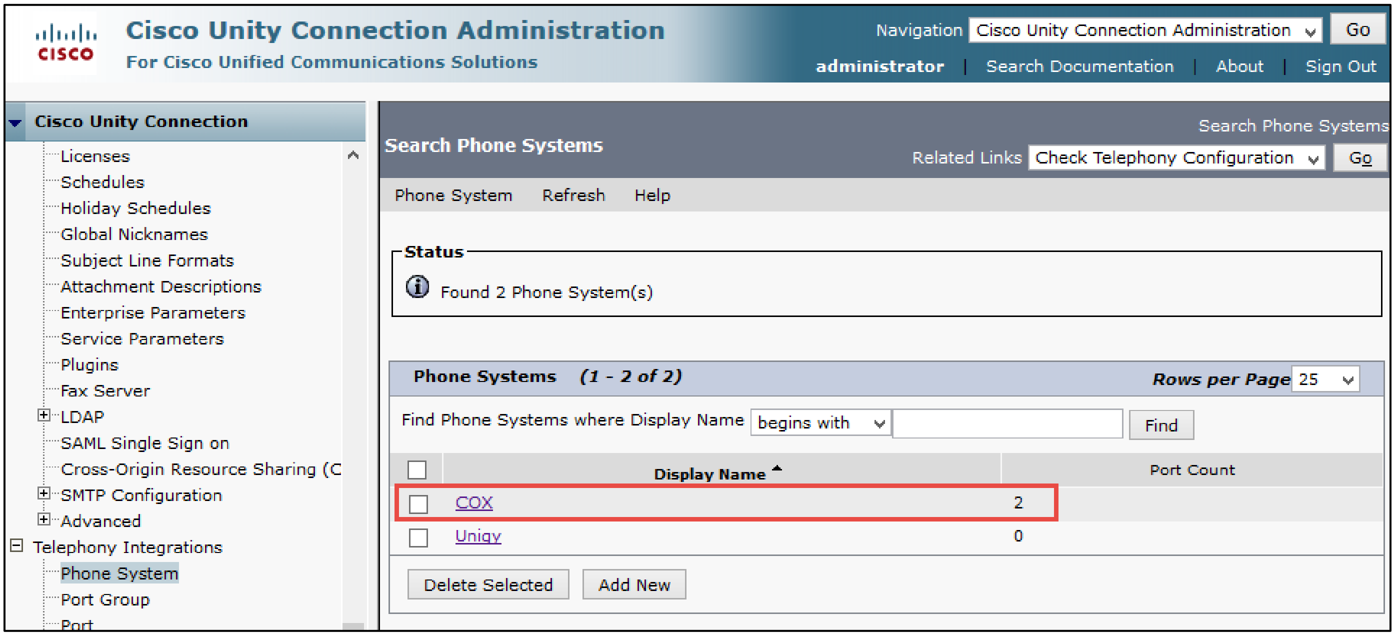


Figure 58: CUC Phone System

CUC Port Group

Navigation: Telephony Integration → Port Group

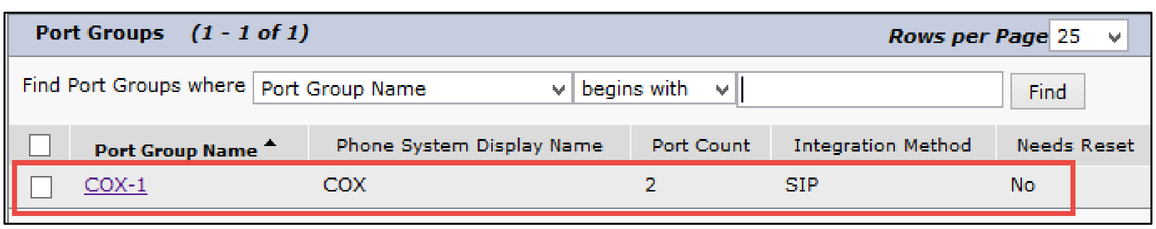


Figure 59: CUC Port Group

Set Display Name* = COX-1. This is used in this example
Set Integration Method = SIP
Check “Enable Message Waiting Indicator Settings “box

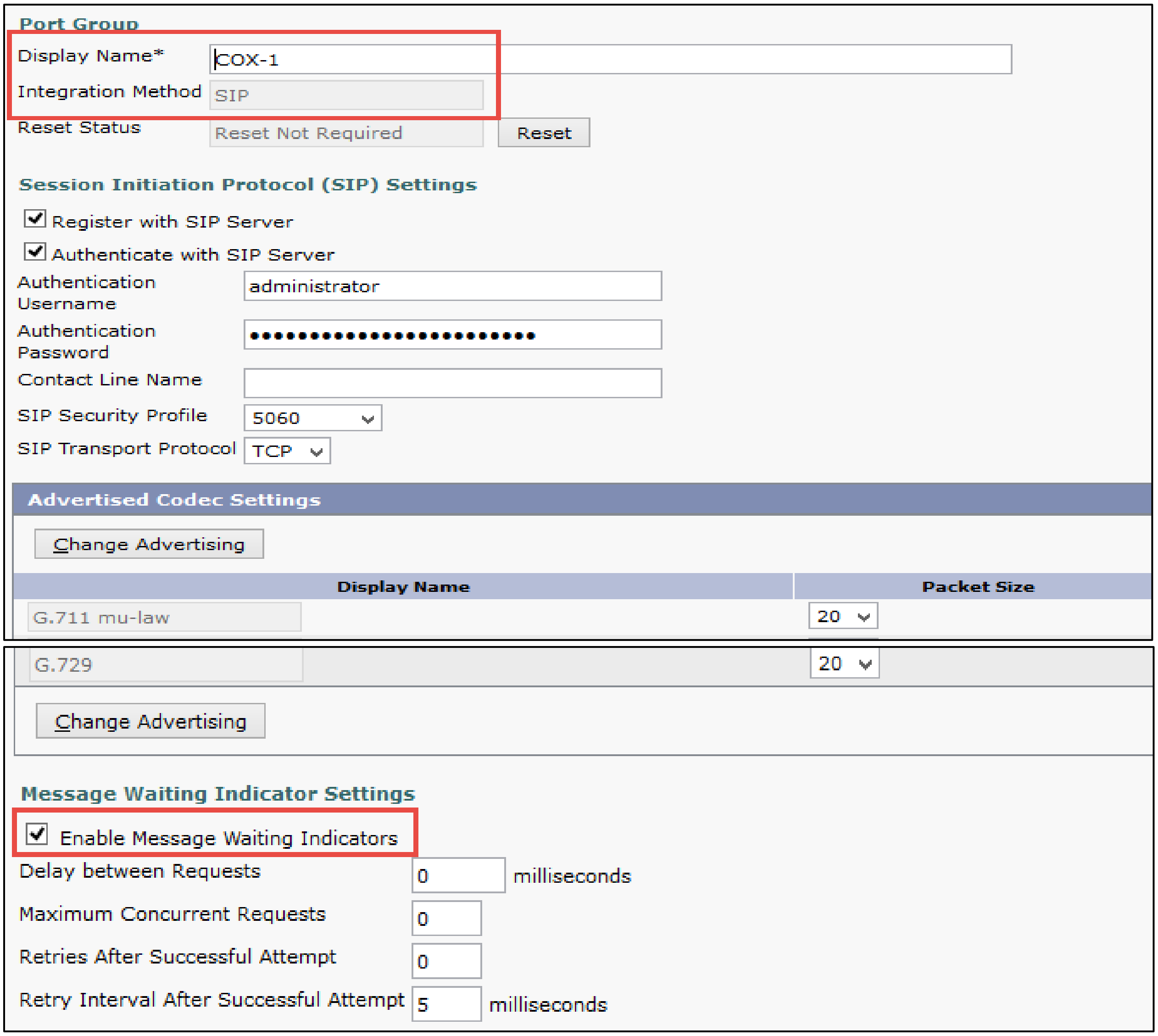


Figure 60: CUC Port Group – Cont.

CUC Port Settings

Navigation: Telephony Integration → Port

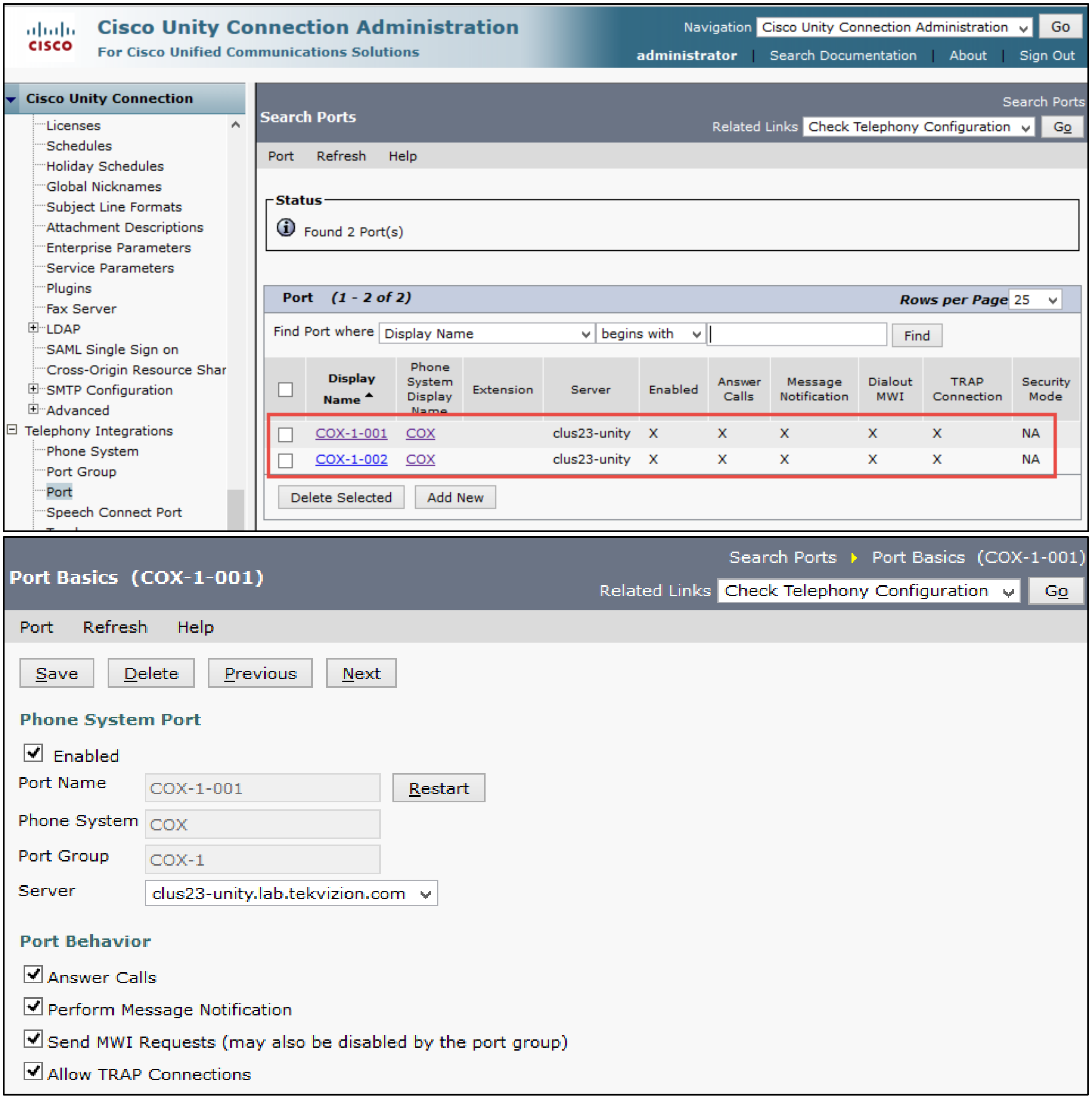


Figure 61: CUC Port Settings

CUC Sample User Basic Settings

Navigation: Cisco Unity Connection → Users → Users

Set Alias = 1399 is one of the extensions used for this testing
Set Extension = 1399 is used for this example
Set Partition = clus23-unity Partition is used for this example
Select Search Space = clus23-unity Search Space
Set Phone System = COX

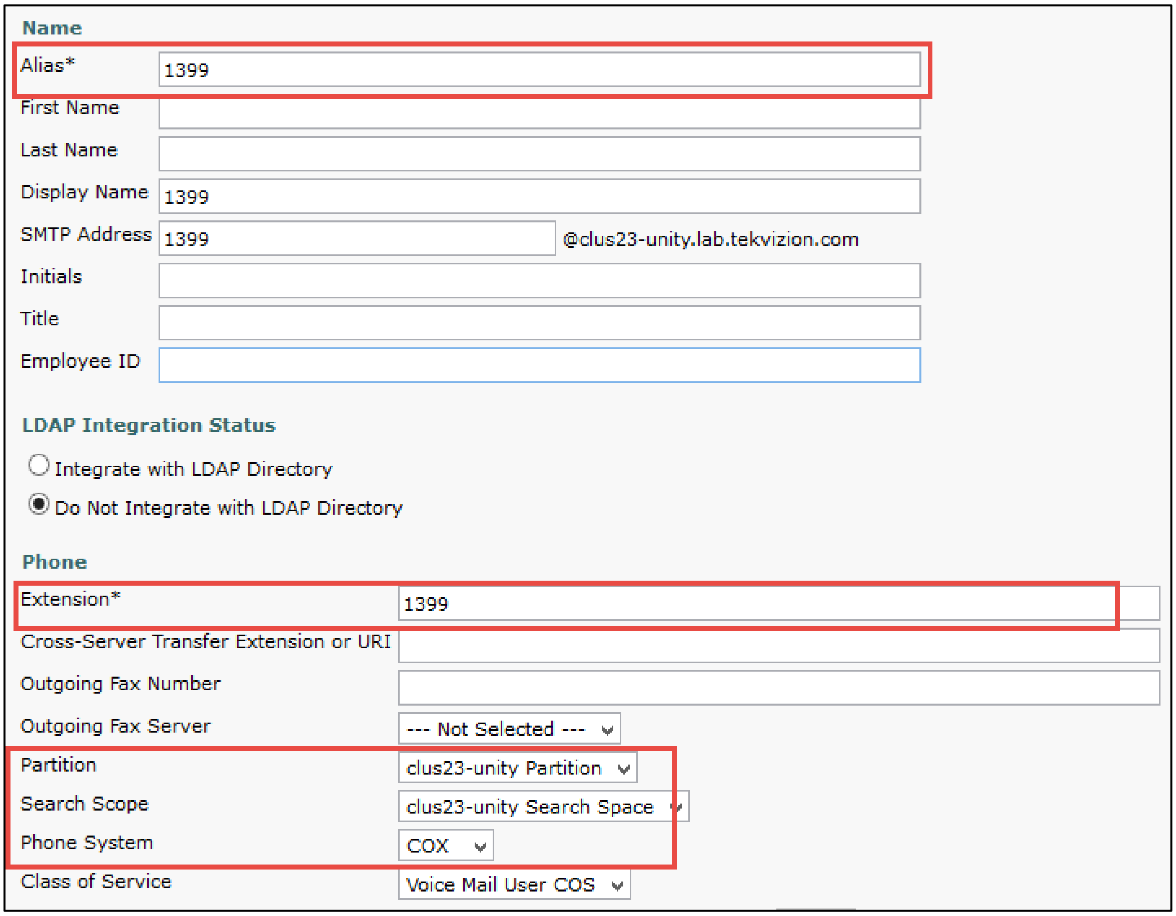


Figure 62: CUC Sample User Basic Settings

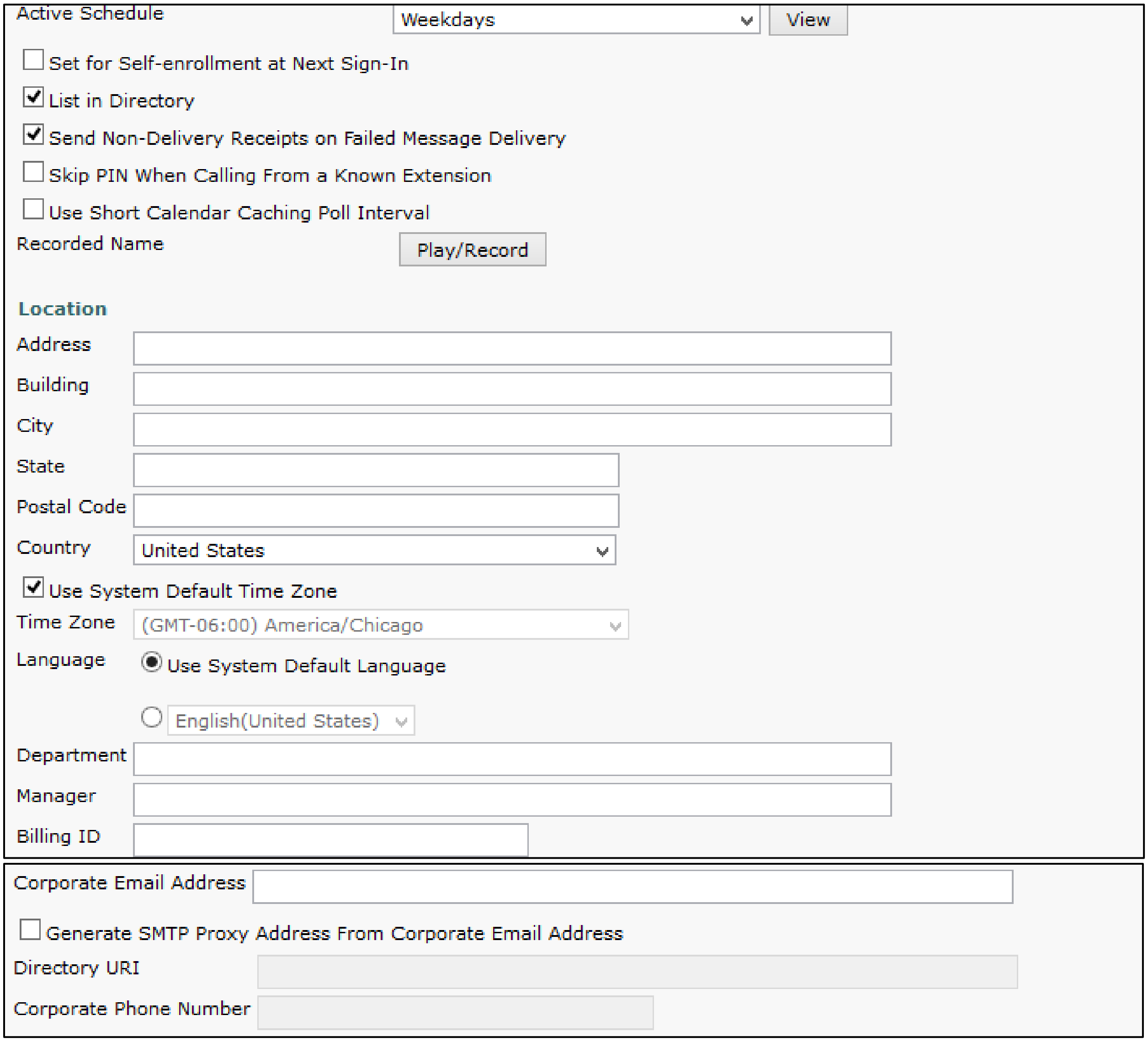


Figure 63: CUC Sample User Basic Settings – Cont.

Acronyms

Acronym	Definitions

CPE	Customer Premise Equipment
CUBE	Cisco Unified Border Element
CUCM	Cisco Unified Communications Manager
MTP	Media Termination Point
POP	Point of Presence
PSTN	Public Switched Telephone Network
ESBC	Enterprise Session Border Controller
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol

Important Information