Application Note

Cox SIP Trunking:

Cisco Unified Communications Manager 11.0.1 with Cisco Unified Border Element (CUBE 11.1.0) on ISR 4K [IOS-XE 3.16] using SIP

November 11, 2015

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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) ISR 15.5(3)S can be used. The Cisco Unified Border Element ISR 15.5(3)S 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 11.1.0) on ISR 4K [IOS-XE3.16] 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

This application note does not cover the configuration of the EdgeMarc E-SBC. The Cox E-SBC is the Edgewater Networks EdgeMarc appliance. The EdgeMarc is the service demarcation point between customer's LAN network and Cox's WAN network and provides firewall/NAT traversal, B2BUA and SIP Application-level gateway. The EdgeMarc has diverse routes to a primary and secondary Acme SBC. For more info regarding the EdgeMarc E-SBC visit www.edgewaternetworks.com.

Network Topology

Basic Call Setup

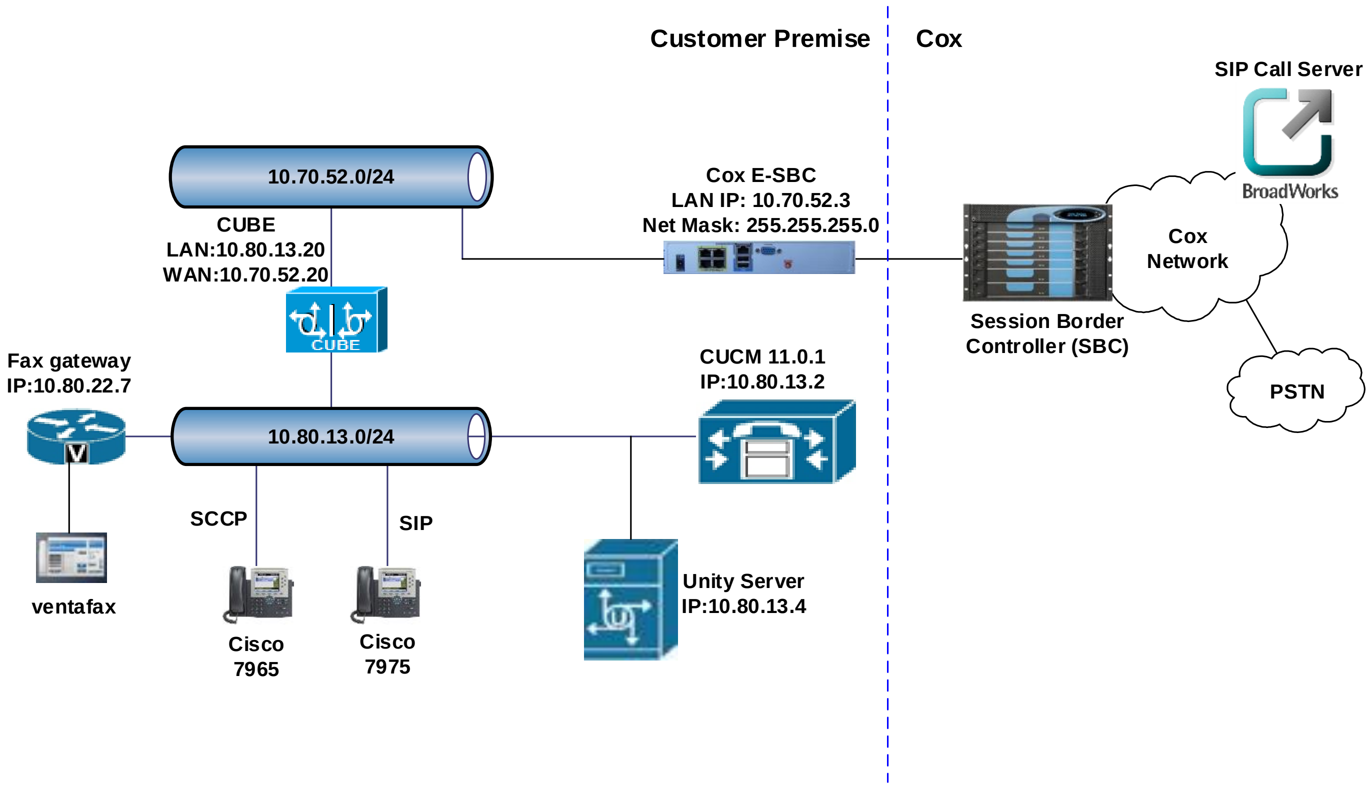

Figure 1: Network Topology

System Components

Hardware Components

- Cisco UCM and Unity Connection run on VMware
- ISR G2 2901/K9 router as Fax Gateway
- ISR 4321/K9 router as CUBE
- IP phones 7975(SIP) and 7965(SCCP)(please consult "Features not supported" for restrictions)

Software Requirements

- Cisco Unified Communications Manager 11.0.1
- IOS-XE 3.16 for Cisco Unified Border Element on ISR4321
- IOS 15.4(3)M1 for Fax Gateways on ISR2901
- 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

- CUBE High Availability (HA) was not tested in this setup due to lack of hardware
- 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 - CSCua03687

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 CUBE1 LAN
 ip address 10.80.13.20 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0/0/1
 description COX CUBE1 WAN
 ip address 10.70.52.20 255.255.255.0
 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 10 voip

description "Outgoing To edgemark"-edgemark facing side

```
 huntstop
 destination-pattern [0-9]T
 no modem passthrough
 session protocol sipv2
 session target ipv4:10.70.52.3:5060
 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 GigabitEthernet0/0/1
 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 11 voip
 description "Outgoing To edgemark .IP PBX facing side"
 no modem passthrough
 session protocol sipv2
 incoming called-number [0-9]
 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 GigabitEthernet0/0/0
```

```
 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 20 voip
 description " Incoming edgemark to IP-PBX - IP-PBX facing side "
 huntstop
 destination-pattern 402502....
 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-relay sg3-to-g3
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
```

```
!
dial-peer voice 21 voip
 description " Incoming edgemark to IP-PBX . AT&T facing side "
 huntstop
 no modem passthrough
 session protocol sipv2
 incoming called-number 40250.....
 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
```

Call Flow

In the sample configuration presented here, CUCM is provisioned with four-digit directory numbers corresponding to the last four DID digits. No digit manipulation is performed on the CUBE.

For incoming PSTN calls, the CUBE presents the full ten-digit DID number to CUCM. The CUCM Translation Pattern strips all but the last four digits and routes the call based on those digits. Voice calls are routed to IP phones; fax calls are routed via a 4-digit route pattern to a SIP trunk that terminates on the fax gateway

CPE callers make outbound PSTN calls by dialing a “9” prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, fax gateway sends to Cisco UCM the DID with leading access code “9”. A “9.@” route pattern strips the prefix and routes the call with the remaining digits via a SIP trunk terminating on the CUBE for voice call or outbound fax.

Configuration Example

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

```
COX_CUBE1#sh running-config

Building configuration...


Current configuration : 7526 bytes

!

version 15.5

service timestamps debug datetime msec

service timestamps log datetime msec

no platform punt-keepalive disable-kernel-core

!

hostname COX_CUBE1

!

boot-start-marker

boot system flash:isr4300-universalk9.03.16.00.S.155-3.S-ext.SPA.bin

boot system bootflash:isr4300-universalk9.03.16.00.S.155-3.S-ext.SPA.bin

boot-end-marker

!

!

vrf definition Mgmt-intf

 !

 address-family ipv4

 exit-address-family

 !

 address-family ipv6

 exit-address-family
```

```
!
enable secret 5 $1$4Gfa$O/1WQEcuut.YXLcn3acUP1
enable password tekV1z10n
!
no aaa new-model
!
!
subscriber templating
!
multilink bundle-name authenticated
!
!
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
!
```

```
voice class codec 1
 codec preference 1 g711ulaw
!
voice class codec 2
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
!
license udi pid ISR4321/K9 sn FDO19220MQ8
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 CUBE1 LAN
 ip address 10.80.13.20 255.255.255.0
 negotiation auto
 !
interface GigabitEthernet0/0/1
 description COX CUBE1 WAN
 ip address 10.70.52.20 255.255.255.0
```

```
 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 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 10 voip
 description "Outgoing To edgemark"-edgemark facing side
 huntstop
 destination-pattern [0-9]T
 no modem passthrough
 session protocol sipv2
 session target ipv4:10.70.52.3:5060
 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 GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
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 fax rate disable
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 no vad
!
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 session protocol sipv2
 incoming called-number [0-9]
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```

```
 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 GigabitEthernet0/0/0
 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 20 voip
 description " Incoming edgemark to IP-PBX - IP-PBX facing side "
 huntstop
 destination-pattern 402502....
 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-relay sg3-to-g3
```

```
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 21 voip
 description " Incoming edgemark to IP-PBX . AT&T facing side "
 huntstop
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 voice-class codec 1
 voice-class sip asymmetric payload full
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip 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
!
!
sip-ua
 credentials username 4023159990 password 7 15465B5E577B7E7D716A65 realm 10.70.52.3
 authentication username 4023159990 password 7 01475656085A535678151E
 registrar ipv4:10.70.52.3 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 Call Manager Service Parameters

1. Navigate to **System > Service Parameters**
2. Set **Duplex Streaming:** enabled = true

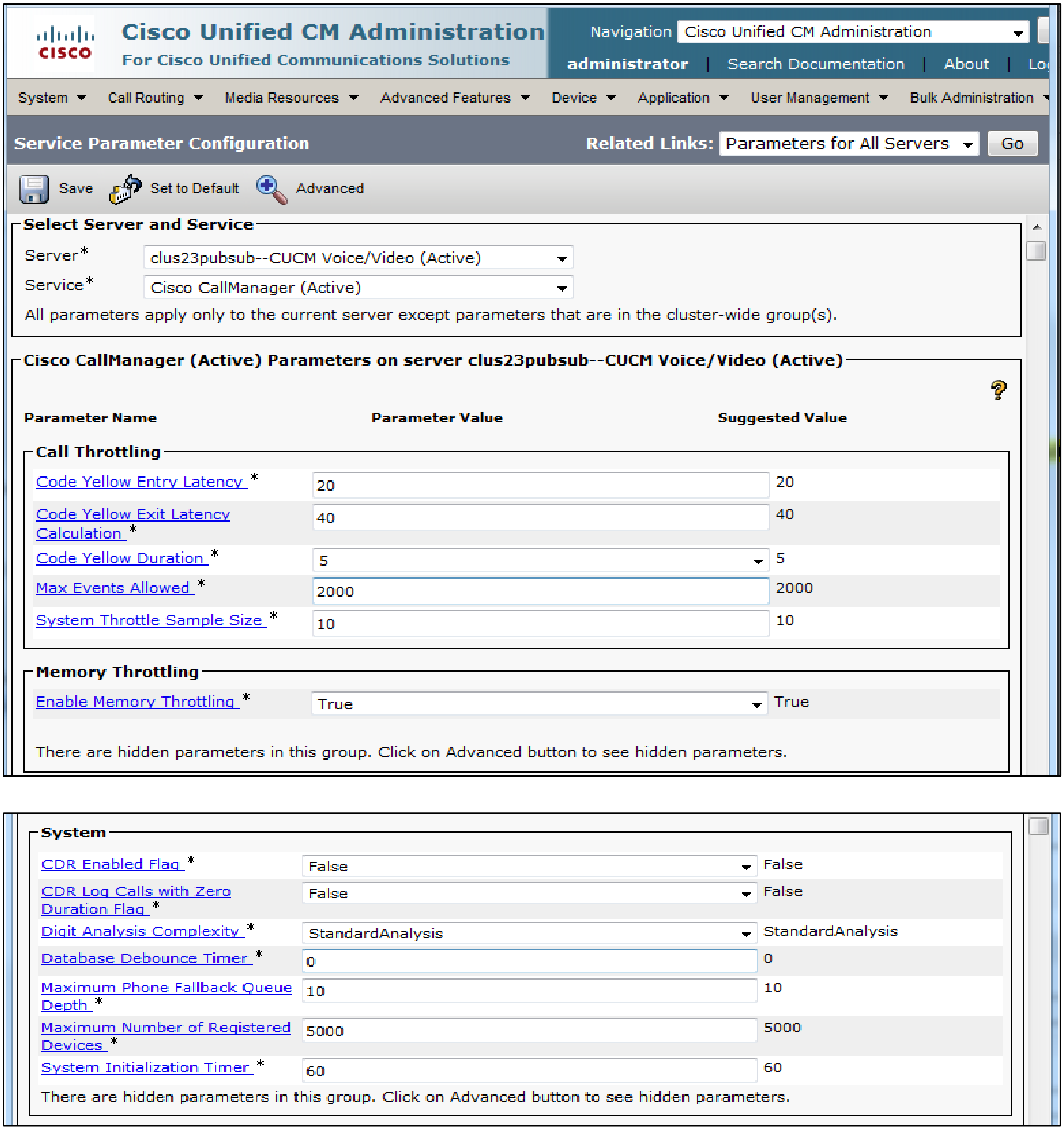


Figure 2: Service Parameters

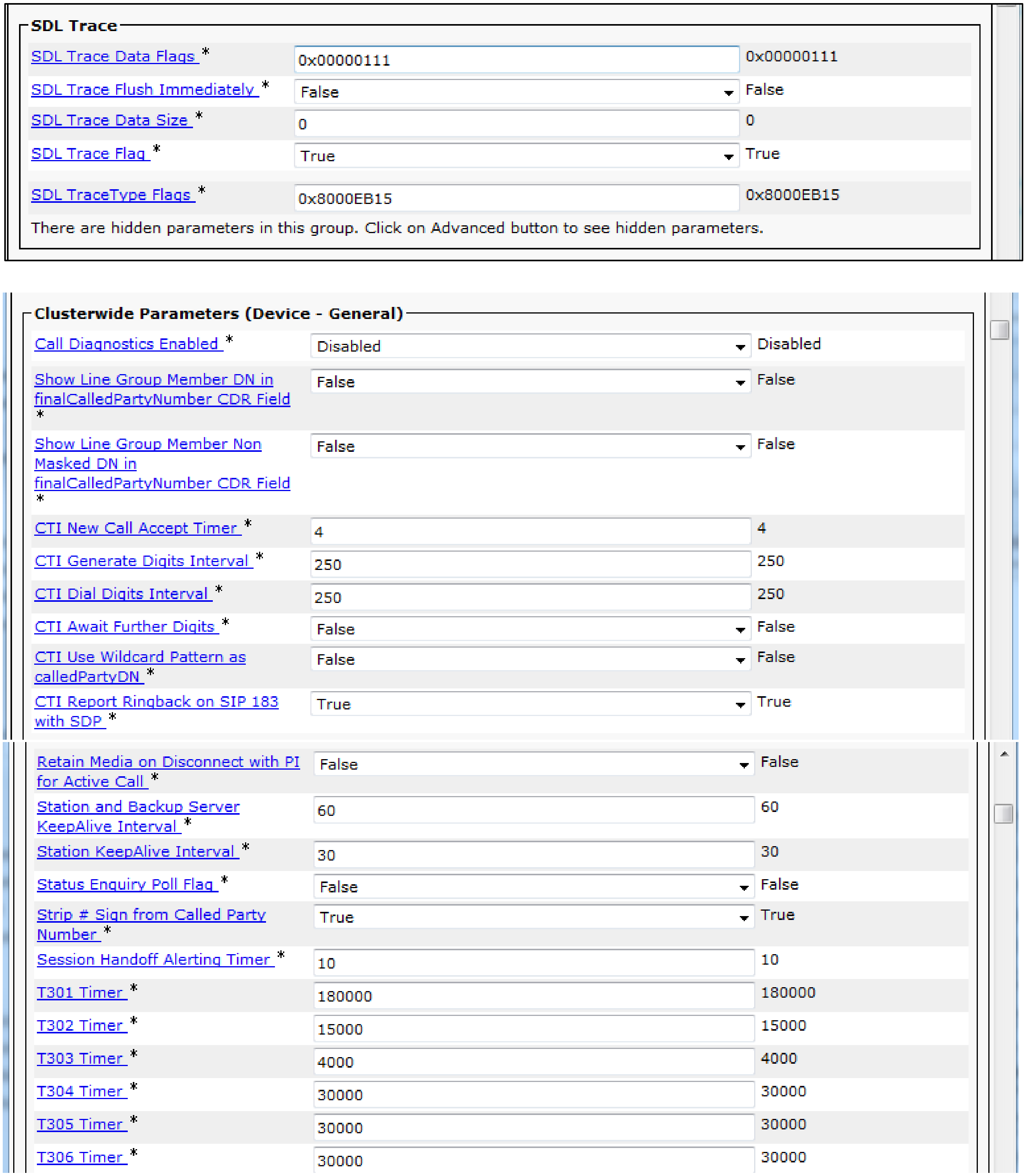


Figure 3: Service Parameters - Cont.

T308 Timer *	4000	4000
T309 Timer *	90000	90000
T310 Timer *	60000	60000
T313 Timer *	4000	4000
T316 Timer *	120000	120000
T317 Timer *	100000	100000
T321 Timer *	30000	30000
T322 Timer *	4000	4000
Tone on Hold Timer *	10	10
Unknown Caller ID Flag *	True	True
Call Classification *	OffNet	OffNet
Always Display Original Dialed Number *	False	False
Name Display for Original Dialed Number When Translated *	Show the Display Name for Original Dialed Number ev	Show the Display Name for Original Dialed Number even if Translated
Always Use PIs With Original Dialed Number *	False	False
Fail Call If Trusted Relay Point Allocation Fails *	True	True
Display Calling/Called ID When PI is Not Available *	False	False
Enable Transit Counter Processing on QSIG Trunks *	False	False
Egress FacilityIE Count *	6	6

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

Clusterwide Parameters (Device - Phone)

Always Use Prime Line *	False	False
Always Use Prime Line for Voice Message *	False	False
Builtin Bridge Enable *	Off	Off
Device Mobility Mode *	Off	Off
Display Device Mobility Location During Phone Registration *	True	True
Auto Answer Timer *	1	1
Extension Display on Cisco IP Phone Model 7910 *	False	False
Alternate Idle Phone Auto-Answer Behavior Enabled *	False	False
Hold Type *	False	False
Line State Update Enabled *	True	True
Off-hook to First Digit Timer *	15000	15000

Figure 4: Service Parameters - Cont.

Override Auto Answer If Speaker Is Disabled *	True	True
Out-of-Bandwidth Text *	Not Enough Bandwidth	Not Enough Bandwidth
Forced Authorization Code Prompt Text *	Enter Authorization Code	Enter Authorization Code
Client Matter Code Prompt Text *	Enter Client Matter Code	Enter Client Matter Code
AAR Network Congestion Rerouting Text *	Network Congestion. Rerouting.	Network Congestion. Rerouting.
Ring Setting of Busy Station Policy *	Only Apply Ring Setting of Busy Station When Incomin	Only Apply Ring Setting of Busy Station When Incoming Call Arrives
Transfer On-hook Enabled *	False	False
Ring Setting of Busy Station *	Beep Only	Beep Only
Ring Setting of Idle Station *	Ring	Ring
Call Pickup Group Audio Alert Setting of Idle Station *	Ring Once	Ring Once
Call Pickup Group Audio Alert Setting of Busy Station *	Beep Only	Beep Only
BLF Pickup Audio Alert Setting of Idle Station *	Disable	Disable
BLF Pickup Audio Alert Setting of Busy Station *	Disable	Disable
Privacy Setting *	True	True
Enforce Privacy Setting on Held Calls *	False	False
SIP Station KeepAlive Interval *	120	120
SIP Station Realm *	ccmsipline	ccmsipline
Hunt Group Logoff Notification *	None	None
Speed Dial Await Further Digits *	False	False
Display CTI Route Point Name or DN *	False	False
Display Original Calling Number on Transfer from Cisco Unity *	False	False
URI Dialing Display Preference *	DN	DN
Insert Hyphens in 12-Digit Numbers *	False	False
Allow Call Waiting During an In-Progress Outbound Analog Call *	True	True

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

Figure 5: Service Parameters - Cont.

Clusterwide Parameters (Device - PRI and MGCP Gateway)

Calling Party Number Screening Indicator *	CallManager sets the screening indicator value - Defau	CallManager sets the screening indicator value - Default setting
Enable Outbound NetworkTrunk CallingParty Restriction *	False	False
Clear Calls Flag When Datalink Is Down *	True	True
Device Status Poll Interval *	3000	3000
Disable Alerting Progress Indicator *	False	False
Discard Non Inband Progress in Overlap Sending *	False	False
Disable Resume from Shared-line MGCP FXS Port *	True	True
DTMF Silence Tone Flag *	False	False
Enable Display IE in Codeset 6 *	False	False
Enable Sending PRI NI2 Service Message *	False	False
Flash Hook Duration *	500	500
Gateway Poll Timer *	10	10
Location In PRI Progress Indicator IE (User Side Only) *	Use the Network Side PRI progress indicator IE	Use the Network Side PRI progress indicator IE
Matching Calling Party with Attendant Flag *	False	False
MGCP Database Query Delay Timer *	1000	1000
MGCP FXS On-Hook Pending Timer *	3	3
MGCP Response Timer *	30	30
MGCP Timer *	3	3
Numbering Plan Info *	1	1
Overlap Receiving Flag for PRI *	True	True
Outgoing Media Connect Time for PRI *	Connect ASAP	Connect ASAP
Port Release Timer *	0	0
SMDI Call Delay Timer *	0	0
Stable in State 4 Flag *	False	False

Figure 6: Service Parameters - Cont.

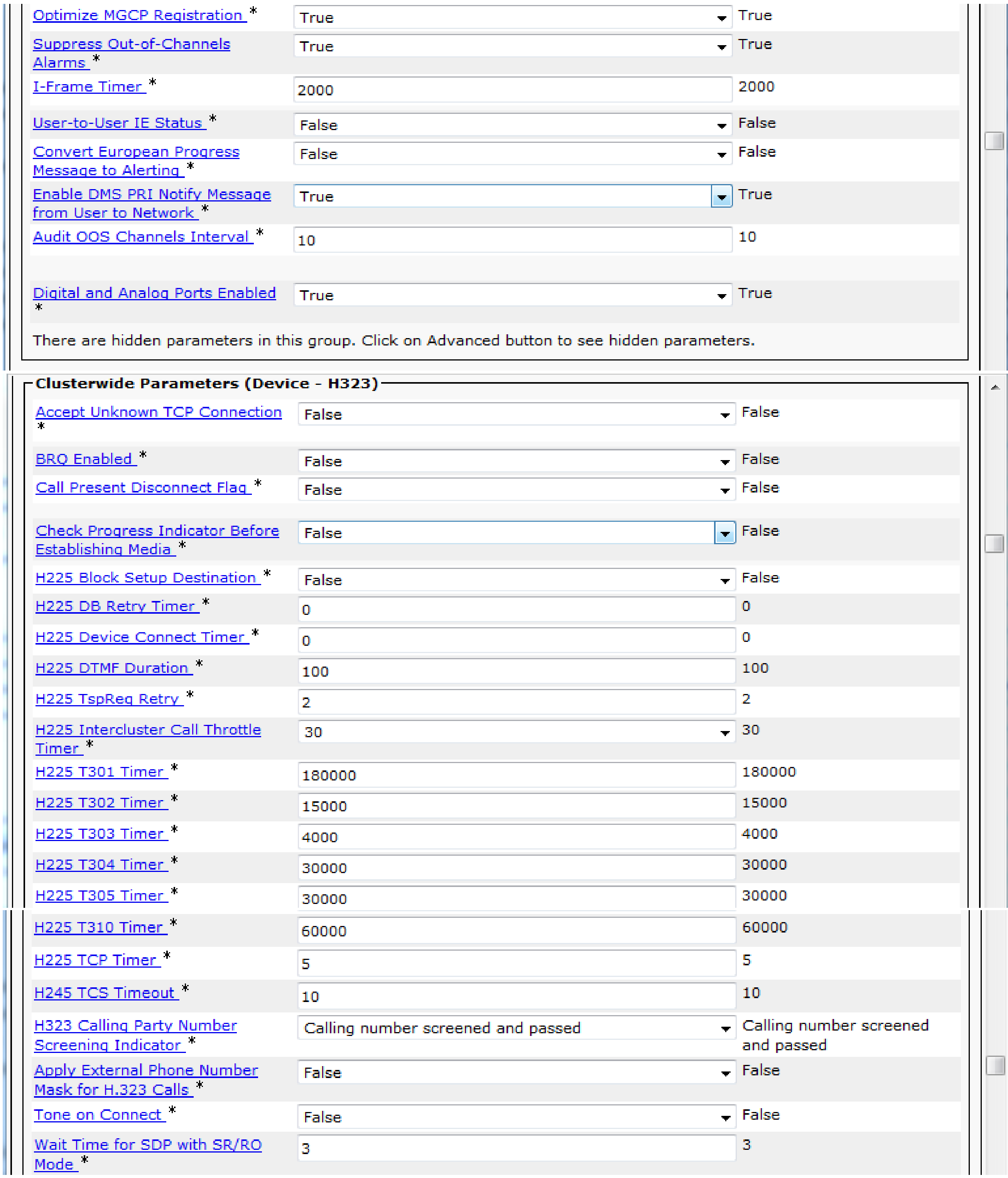


Figure 7: Service Parameters - Cont.

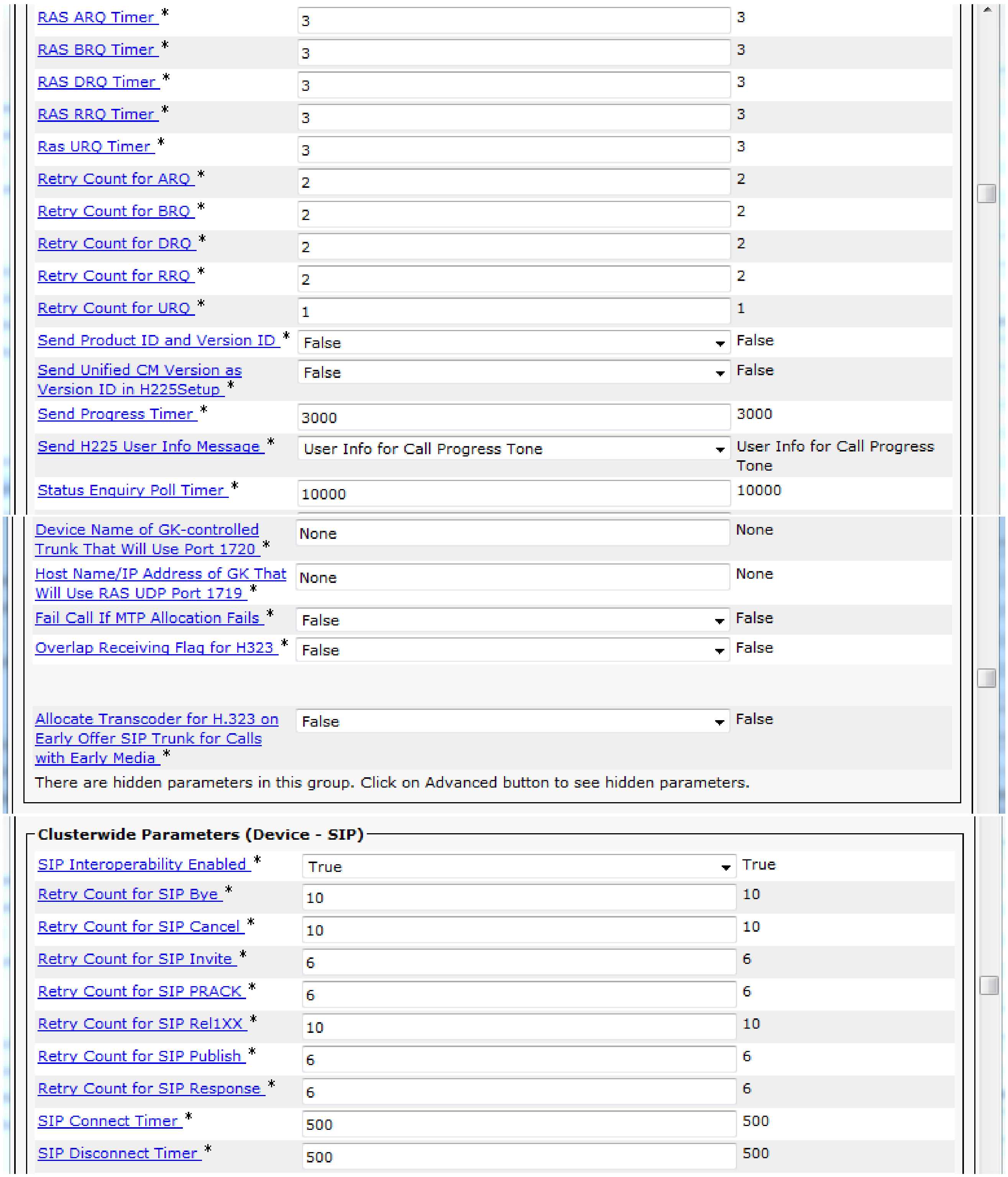


Figure 8: Service Parameters - Cont.

SIP Expires Timer *	180000	180000
SIP PRACK Timer *	500	500
SIP Rel1XX Timer *	500	500
SIP Trying Timer *	500	500
SIP Publish Timer *	500	500
SIP Min-SE Value *	1800	1800
SIPS URI Handling *	Reject	Reject
SIP statistics Periodic update Timer *	2	2
SIP Session Expires Timer *	1800	1800
SIP Trunk TspReq Retry *	2	2
SIP TCP Unused Connection Timer *	14	14
SIP TCP Timer *	5	5
SIP Station TCP Port Throttle Threshold *	100	100
SIP Trunk TCP Port Throttle Threshold *	500	500
SIP V.150 Outbound SDP Offer Filtering *	No Filtering	No Filtering
Send SIP Multicast TTL in SDP *	False	False
Default PUBLISH Expiration Timer *	3600	3600
Minimum PUBLISH Expiration Timer *	60	60
IM and Presence Publish Trunk	IMPTrunk	
Send 181 Call Is Being Forwarded *	False	False
Delay Sending 181 until 180/183 message is received *	True	True
Fail Call Over SIP Trunk if MTP Allocation Fails *	False	False
Log Call-Related REFER/NOTIFY /SUBSCRIBE SIP Messages for Session Trace *	True	True
Port Received Timer for Outbound Call Setup *	2	2

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

Figure 9: Service Parameters - Cont.

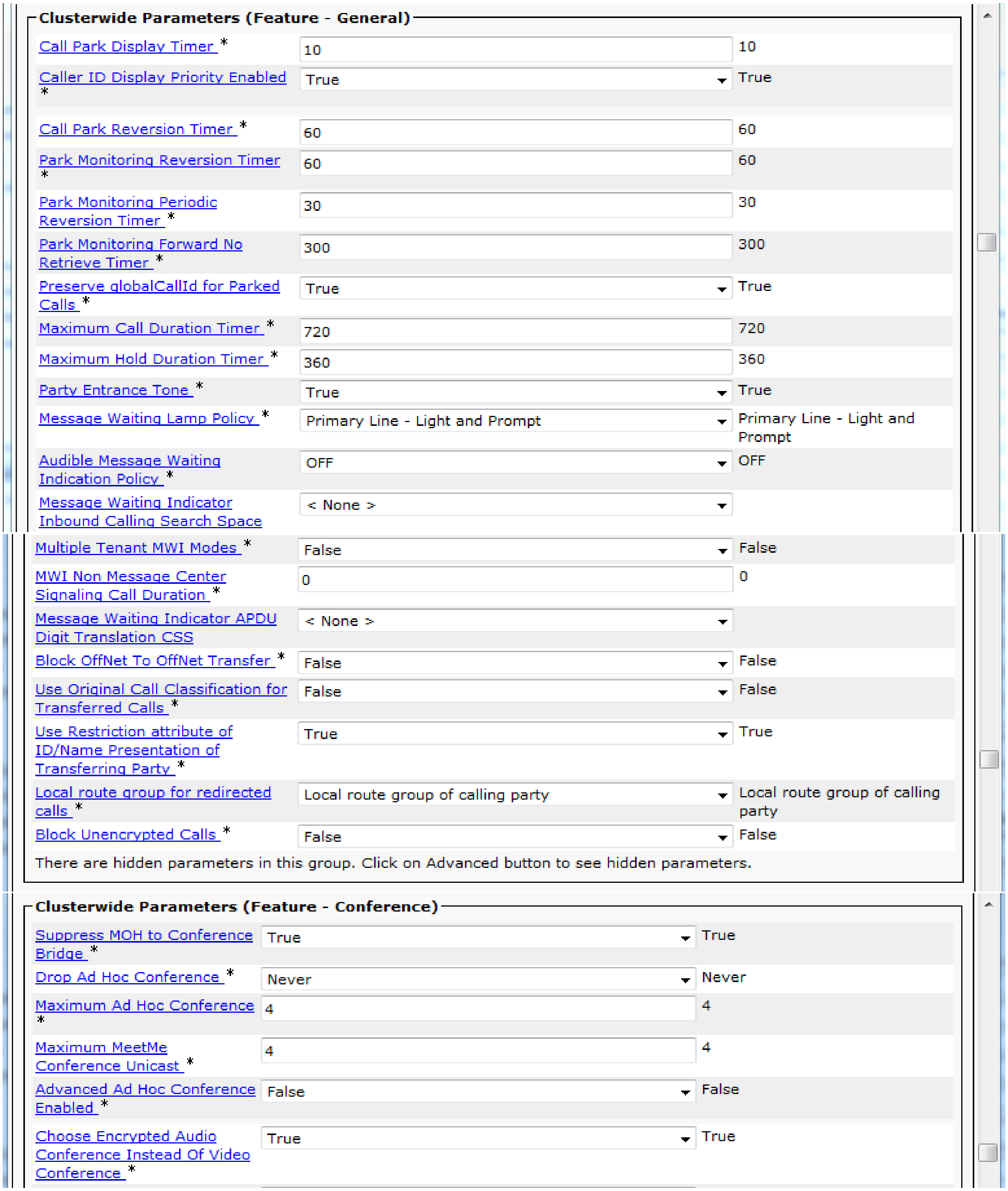


Figure 10: Service Parameters - Cont.

Minimum Video Capable Participants To Allocate Video Conference *	2	2
Enable Click-to-Conference for Third-Party Applications *	False	False
IMS Conference Factory URI *	cucm-conference-factory@cucm1.company.com	cucm-conference-factory@cucm1.company.com
Cluster Conferencing Prefix Identifier		

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

Clusterwide Parameters (Feature - Call Secure Status Policy)

Secure Call Icon Display Policy *	All media except BFCP and iX transports must be encry	All media except BFCP and iX transports must be encrypted

Clusterwide Parameters (Feature - Forward)

Forward Maximum Hop Count *	12	12
Forward No Answer Timer *	12	12
Max Forward Hops to DN *	12	12
Retain Forward Information *	False	False
Forward By Reroute Enabled *	False	False
Transform Forward by Reroute Destination *	True	True
Always Forward Switch Voice Mail Calls *	True	True
Forward By Reroute T1 Timer *	10	10
Include Original Called Info for Q.SIG Call Diversions *	Only after the first diversion	Only after the first diversion
Set Private Numbering Plan for Call Forward *	False	False
Set Type of Number for Call Forward *	Level1RegionalNumber	Level1RegionalNumber
Max Forward UnRegistered Hops to DN *	0	0
CFA CSS Activation Policy *	With Configured CSS	With Configured CSS
Cause Code When Maximum Forward Hop Count is Triggered *	Normal Unspecified	Normal Unspecified

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

Clusterwide Parameters (Feature - Hold Reversion)

Hold Reversion Duration *	0	0
Hold Reversion Notification Interval *	30	30
CFA Destination Override *	False	False

Figure 11: Service Parameters - Cont.

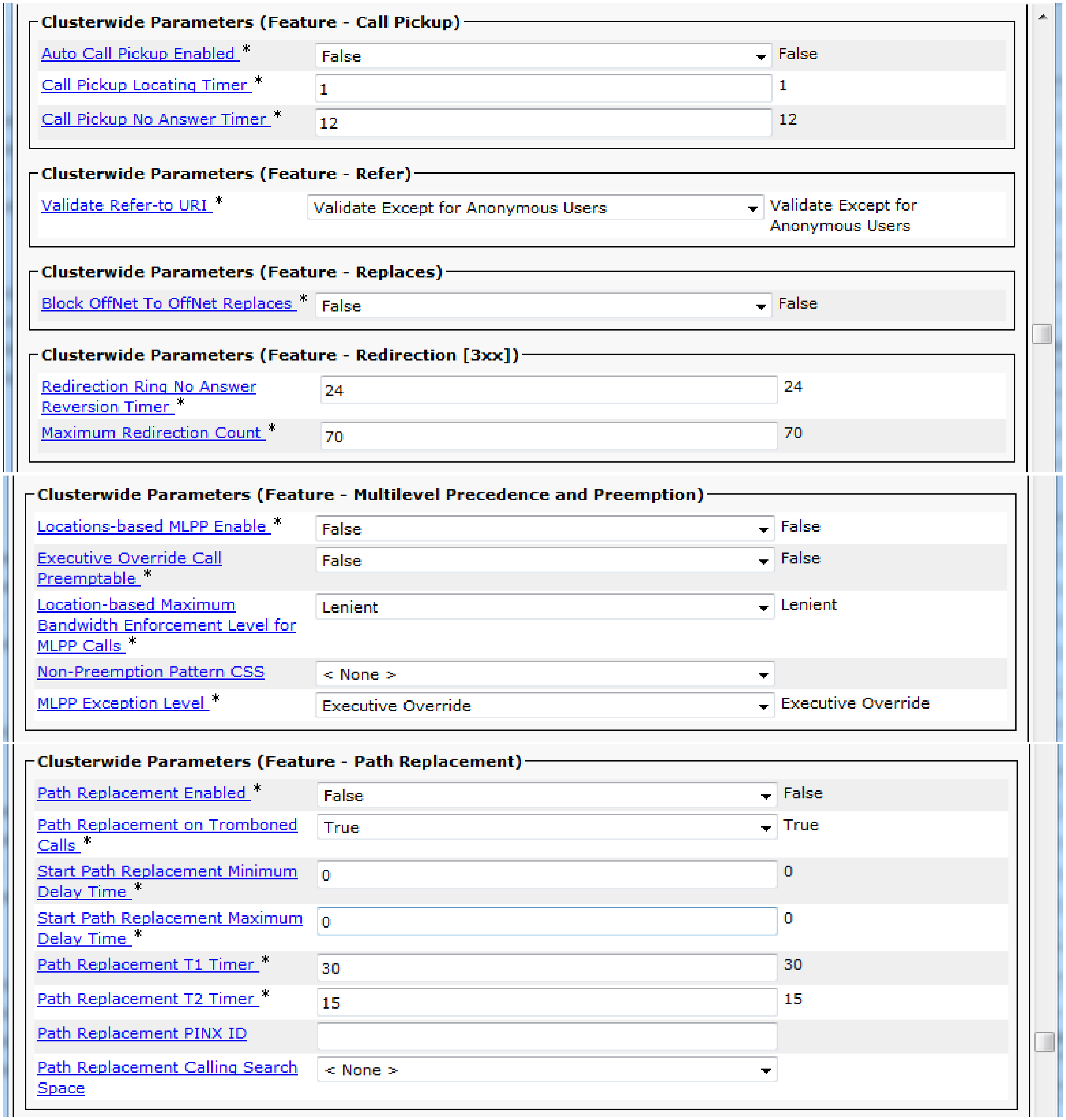


Figure 12: Service Parameters - Cont.

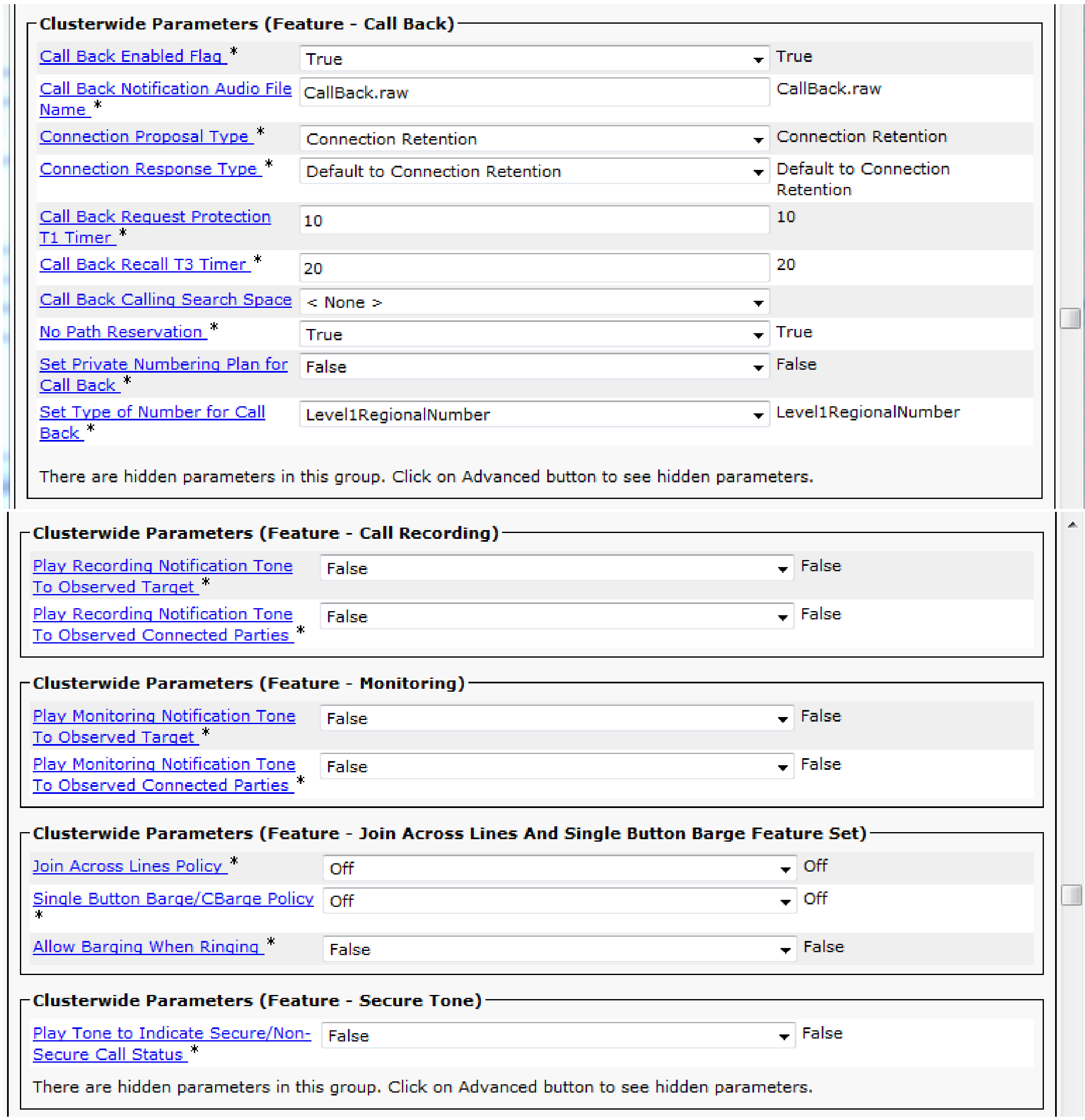


Figure 13: Service Parameters - Cont.

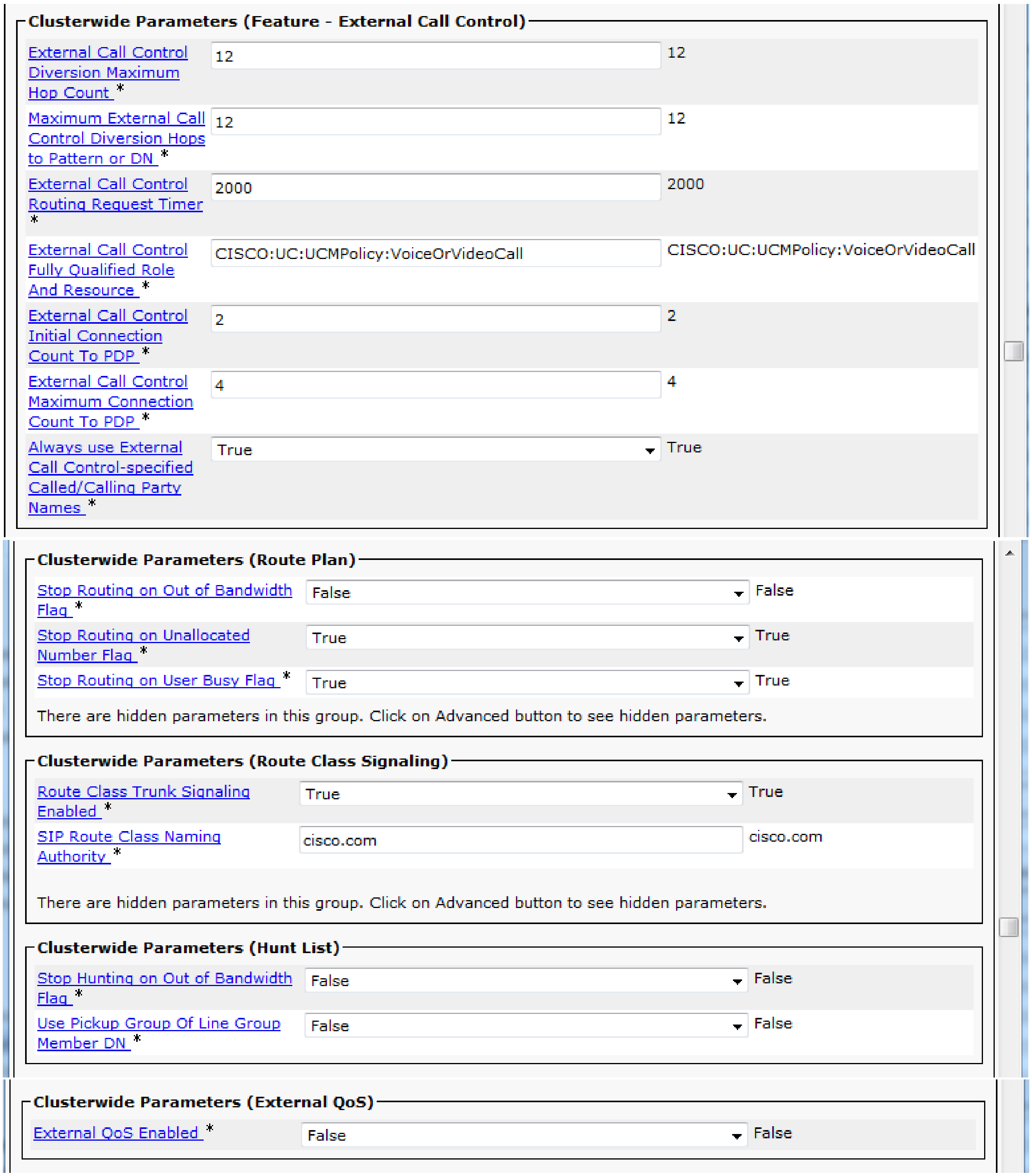


Figure 14: Service Parameters - Cont.

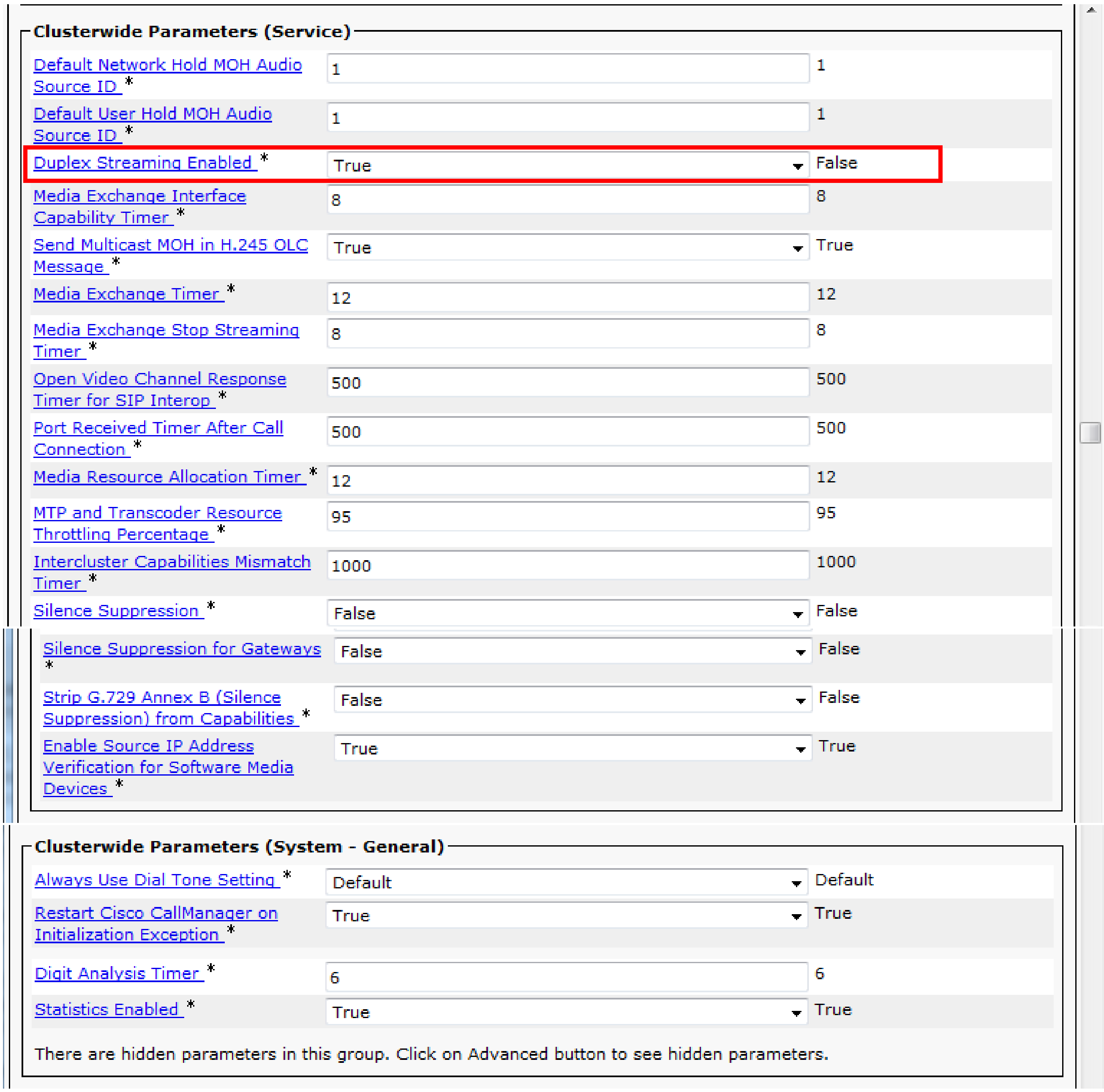


Figure 15: Service Parameters - Cont.

Clusterwide Parameters (System - QOS)

Priority Class *	Normal Priority	Normal Priority
DSCP for Audio Calls *	46 (101110)	46 (101110)
DSCP for Video Calls *	34 (100010)	34 (100010)
DSCP for Audio Portion of Video Calls *	34 (100010)	34 (100010)
DSCP for TelePresence Calls *	32 (100000)	32 (100000)
DSCP for Audio Portion of TelePresence Calls *	32 (100000)	32 (100000)
DSCP for Priority Audio Calls *	45 (101101)	45 (101101)
DSCP for Immediate Audio Calls *	44 (101100)	44 (101100)
DSCP for Flash Audio Calls *	41 (101001)	41 (101001)
DSCP for Flash Override Audio Calls *	42 (101010)	42 (101010)
DSCP for Executive Override Audio Calls *	42 (101010)	42 (101010)
DSCP for Priority Video Calls *	39 (100111)	39 (100111)
DSCP for Immediate Video Calls *	37 (100101)	37 (100101)
DSCP for Flash Video Calls *	35 (100011)	35 (100011)
DSCP for Flash Override Video Calls *	33 (100001)	33 (100001)
DSCP for Executive Override Video Calls *	33 (100001)	33 (100001)
DSCP for G.Clear Calls *	46 (101110)	46 (101110)
DSCP for Priority G.Clear Calls *	45 (101101)	45 (101101)
DSCP for Immediate G.Clear Calls *	44 (101100)	44 (101100)
DSCP for Flash G.Clear Calls *	41 (101001)	41 (101001)
DSCP for Flash Override G.Clear Calls *	42 (101010)	42 (101010)
DSCP for Executive Override G.Clear Calls *	42 (101010)	42 (101010)
DSCP for Audio Calls when RSVP Fails *	0 (000000)	0 (000000)
DSCP for Video Calls when RSVP Fails *	0 (000000)	0 (000000)
DSCP for ICCP Protocol Links *	24 (011000)	24 (011000)

Clusterwide Parameters (System - SDL)

SDL Listening Port Number *	8002	8002
SDL Max Router Latency *	20	20
Suppress Debug Info for Router Death *	0	0
Asynchronous SDL Logging Enabled *	False	False

Figure 16: Service Parameters - Cont.

Clusterwide Parameters (System - Location and Region)

Enforce Millisecond Packet Size *	True	True
Locations Trace Details Enabled *	False	False
Preferred G.711 Millisecond Packet Size *	20	20
Preferred G.722 Millisecond Packet Size *	20	20
Preferred G.723.1 Millisecond Packet Size *	30	30
Preferred G.729 Millisecond Packet Size *	20	20
Always Use Preferred G.729 Packet Size For SIP Trunk Answers *	False	False
Preferred GSM EFR Bytes Packet Size *	31	31
G.711 A-law Codec Enabled *	Enabled for All Devices	Enabled for All Devices
G.711 mu-law Codec Enabled *	Enabled for All Devices	Enabled for All Devices
G.722 Codec Enabled *	Enabled for All Devices	Enabled for All Devices
iLBC Codec Enabled *	Enabled for All Devices	Enabled for All Devices
iSAC Codec Enabled *	Enabled for All Devices	Enabled for All Devices
Opus Codec Enabled *	Enabled for All Devices	Enabled for All Devices
Default Intraregion Max Audio Bit Rate *	64 kbps (G.722, G.711)	64 kbps (G.722, G.711)
Default Interregion Max Audio Bit Rate *	8 kbps (G.729)	8 kbps (G.729)
Default Intraregion Max Video Call Bit Rate (Includes Audio) *	384	384
Default Interregion Max Video Call Bit Rate (Includes Audio) *	384	384
Default Intraregion Max Immersive Video Call Bit Rate (Includes Audio) *	2000000000	2000000000
Default Interregion Max Immersive Video Call Bit Rate (Includes Audio) *	2000000000	2000000000
Use Video BandwidthPool for Immersive Video Calls *	True	True
Default Intraregion and Interregion Link Loss Type *	Low Loss	Low Loss
Default Audio Codec List between Regions *	Factory Default low loss	Factory Default low loss
Default Audio Codec List within Region *	Factory Default low loss	Factory Default low loss
Accept Audio Codec Preferences in Received Offer *	Off	Off
G.Clear Bandwidth Override *	False	False

Figure 17: Service Parameters - Cont.

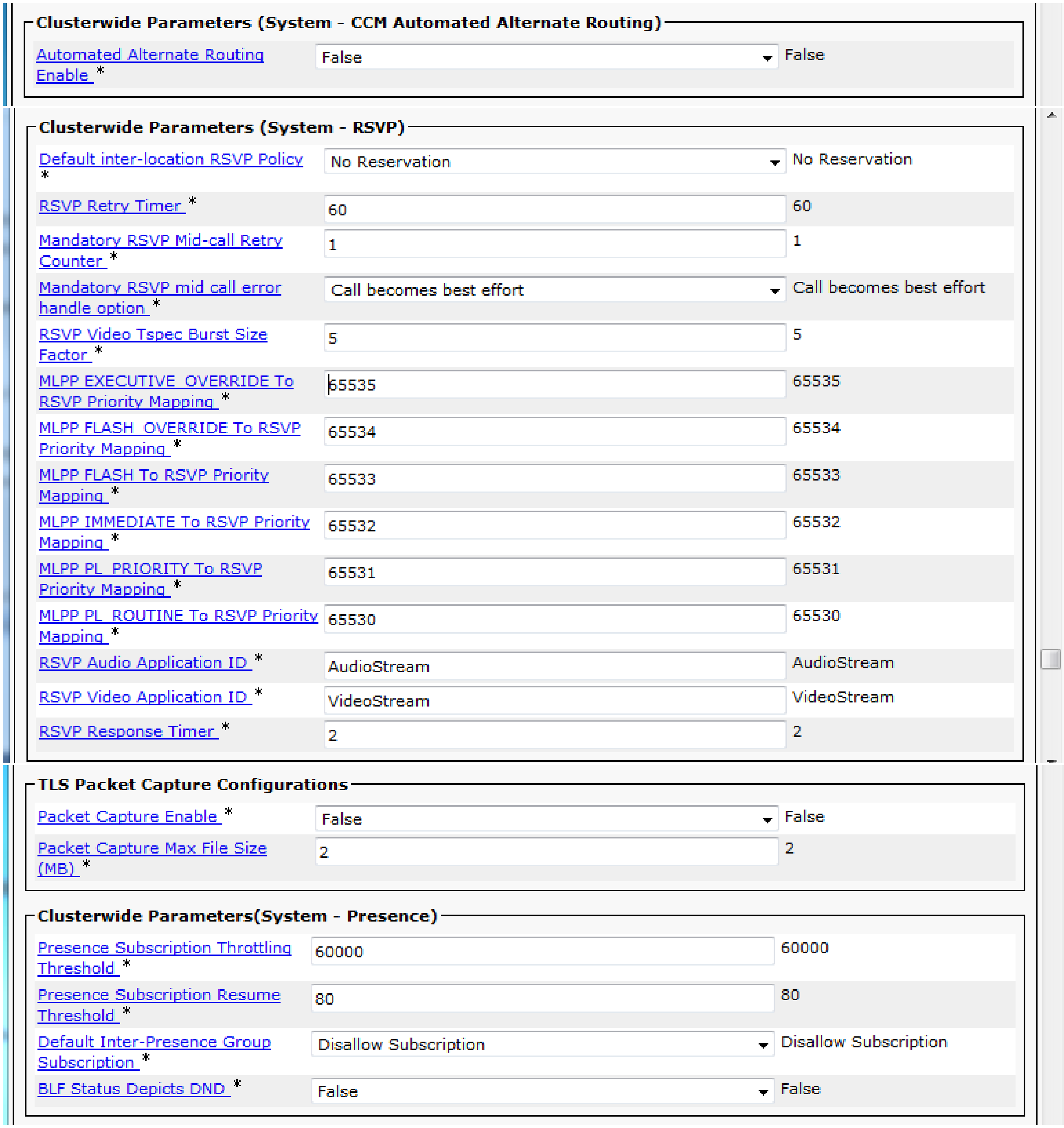


Figure 18: Service Parameters - Cont.

Clusterwide Parameters (System - Mobility)

Enterprise Feature Access Code for Hold *	*81	*81
Enterprise Feature Access Code for Exclusive Hold *	*82	*82
Enterprise Feature Access Code for Resume *	*83	*83
Enterprise Feature Access Code for Transfer *	*84	*84
Enterprise Feature Access Code for Conference *	*85	*85
Enterprise Feature Access Code for Session Handoff *	*74	*74
Enterprise Feature Access Code for Starting Selective Recording *	*86	*86
Enterprise Feature Access Code for Stopping Selective Recording *	*87	*87
Smart Mobile Phone Interdigit Timer *	500	500
Non-Smart Mobile Phone Interdigit Timer *	2000	2000
Send Call to Mobile Menu Timer *	60	60
SIP Dual Mode Alert Timer *	1500	1500
Call Screening Timer *	4000	4000
Session Resumption Await Timer *	180	180
Inbound Calling Search Space for Remote Destination *	Trunk or Gateway Inbound Calling Search Space	Trunk or Gateway Inbound Calling Search Space
Enable Enterprise Feature Access *	False	False
Dial-via-Office Forward Service Access Number		
Enable Mobile Voice Access *	False	False
Mobile Voice Access Number		
Matching Caller ID with Remote Destination *	Complete Match	Complete Match
Number of Digits for Caller ID Partial Match *	10	10
System Remote Access Blocked Numbers		
Enable Use of Called Party Transformed Number for Mobile-terminated Calls *	False	False
Honor Gateway or Trunk Outbound Calling Party Selection for Mobile Connect Calls *	False	False

Figure 19: Service Parameters - Cont.

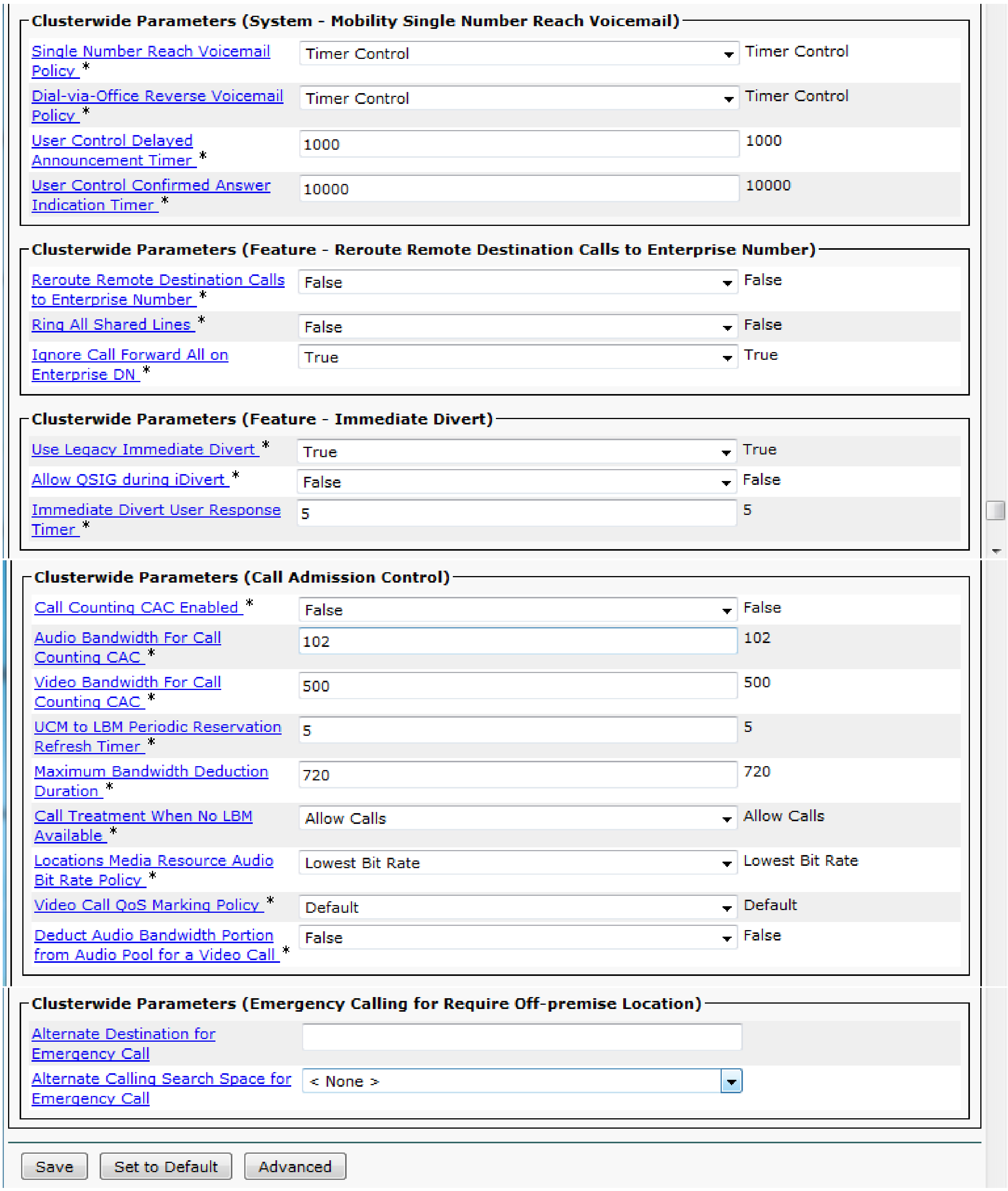


Figure 20: Service Parameters - Cont.

Offnet Calls via Cox SIP Trunk

Off-net calls are served by SIP trunks configured between CUCM and the Cox EdgeMarc E-SBC. Calls are routed via the CUBE.

SIP Trunk Security Profile

1. Navigate to **System > Security > SIP Trunk Security Profile**
2. Click **Add New**

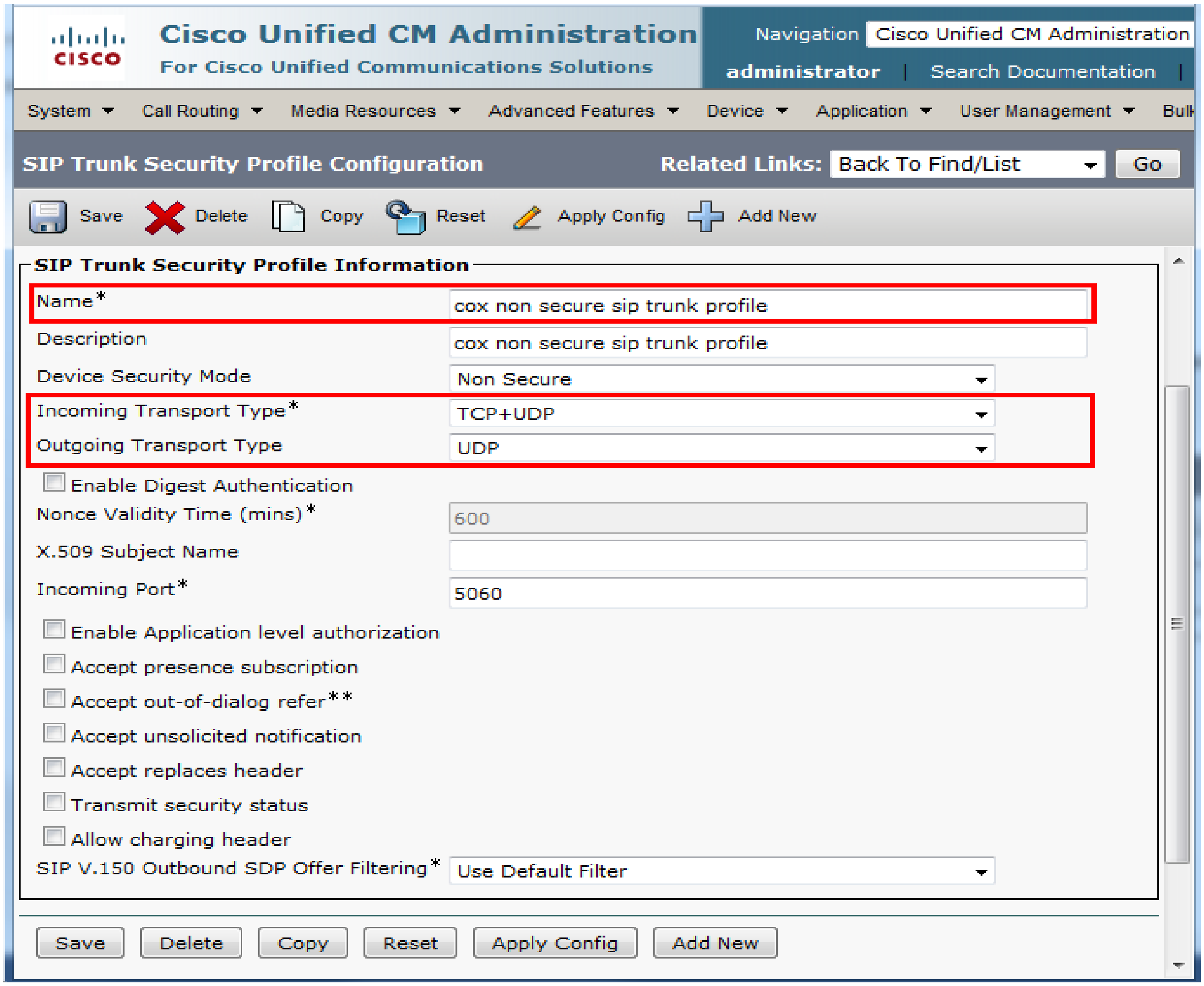


Figure 21: SIP Trunk Security Profile

Parameter	Value	Description

Incoming Transport Type	TCP + UDP	
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.

SIP Profile

The SIP Profile will be associated later with the SIP trunk

1. Navigate to **Device** > **Device Settings** > **SIP Profile**
2. Click **Copy** to modify the default SIP Profile

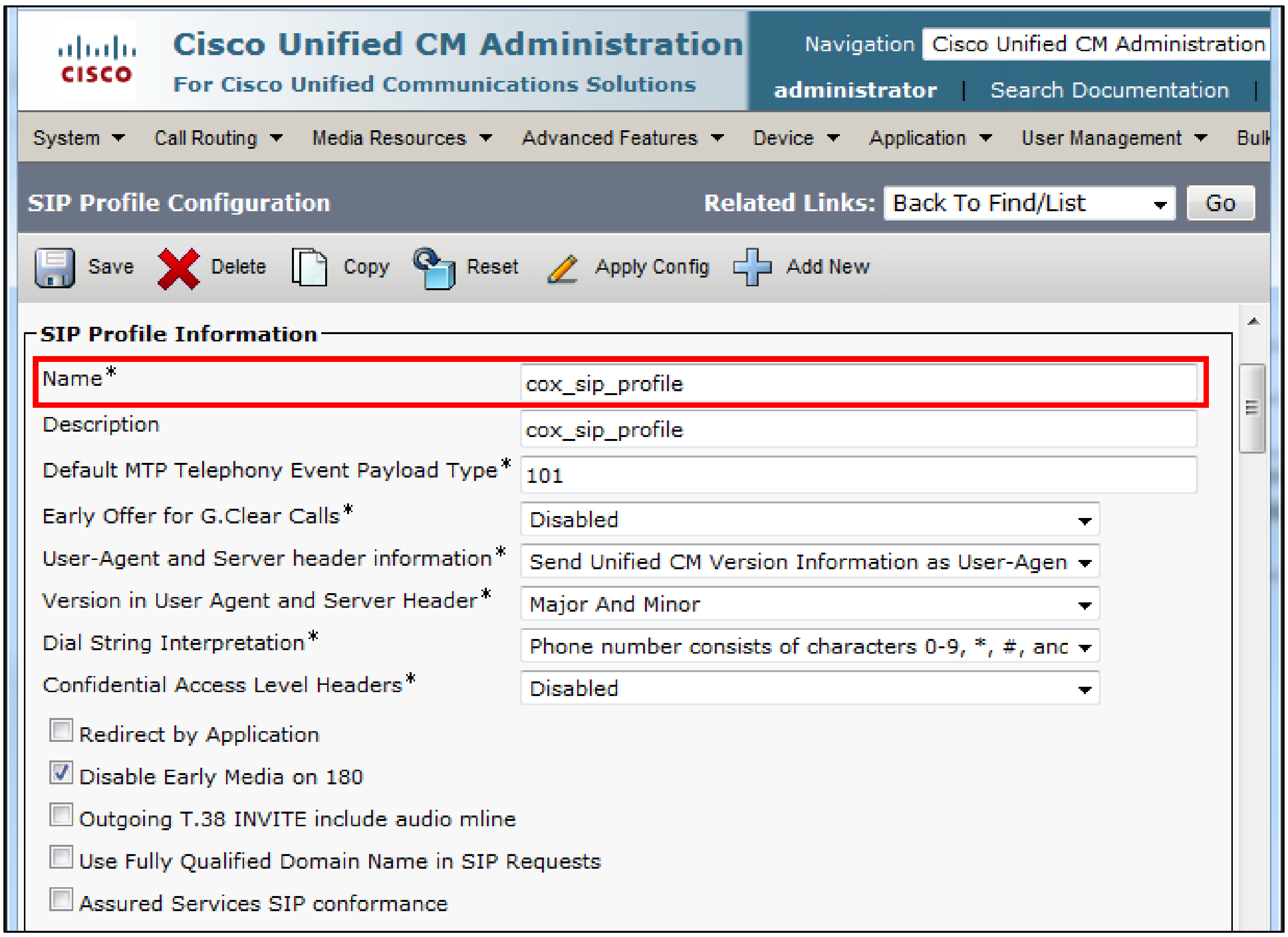


Figure 22: SIP Profile

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 23: SIP Profile - Cont.

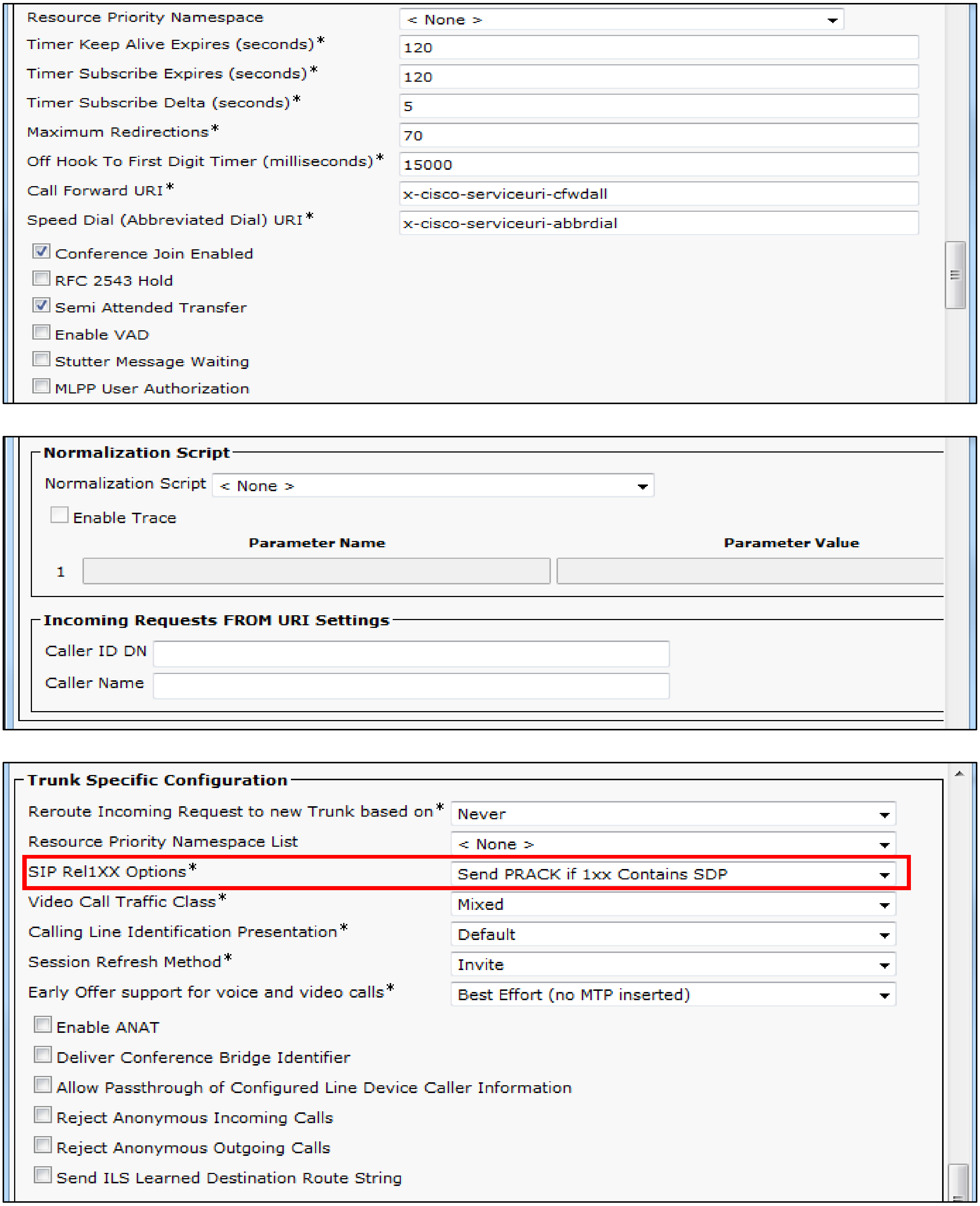


Figure 24: SIP Profile - Cont.

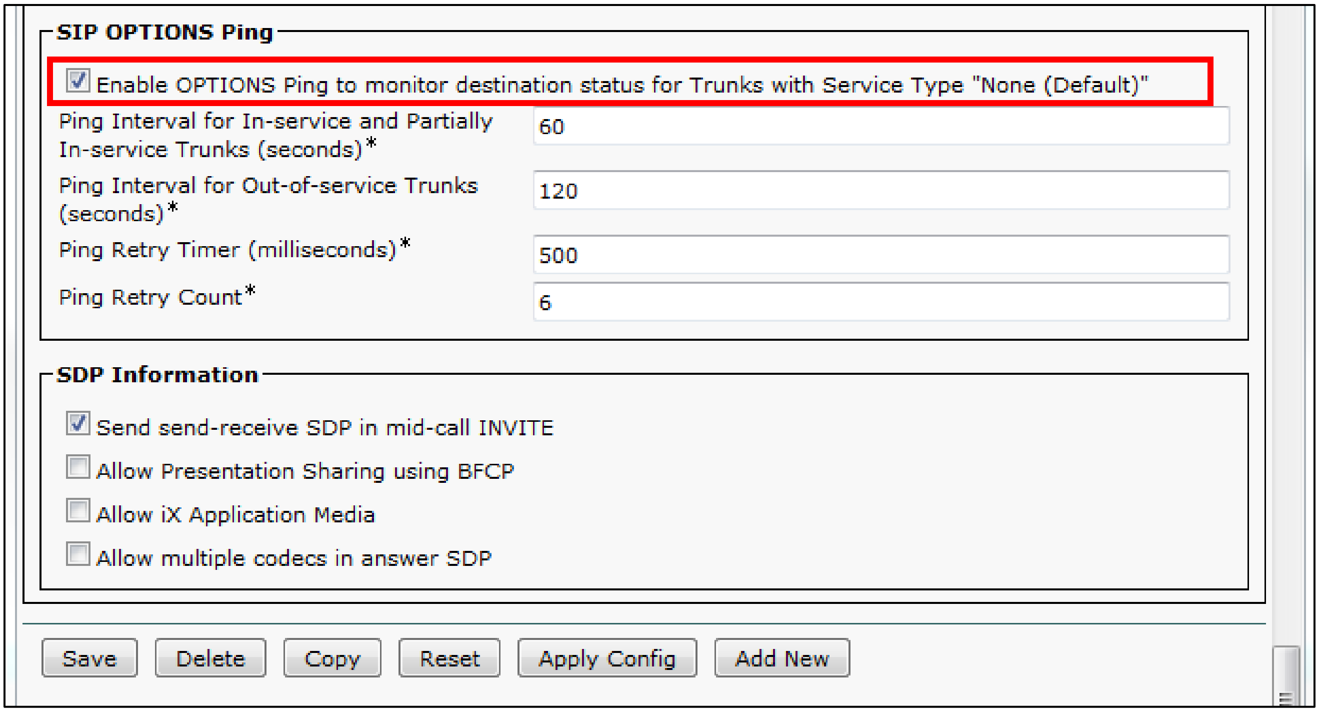


Figure 25: SIP Profile – Cont.

Parameter	Value	Description
Default MTP Telephony Event Payload Type	101	RFC2833 DTMF payload type
Require SDP Inactive Exchange for Mid-Call Media Change	Checked	Send SDP with Inactive when call on hold
SIP Rel1XX Options	Send PRACK for 1xx Messages	Enable Provisional Acknowledgements (Reliable 100 messages)
Ping Interval for In-service and Partially In-service Trunks (seconds)	300	OPTIONS message parameters- interval time
Ping Interval for Out-of-service Trunks (seconds)	5	OPTIONS message parameters- interval time

SIP Trunk Configuration

To create SIP trunks to Cox

1. Navigate to **Device** > **Trunk**
2. Click **Add New**
3. Apply to create SIP trunks to Cisco Unity Connection and Fax Gateway

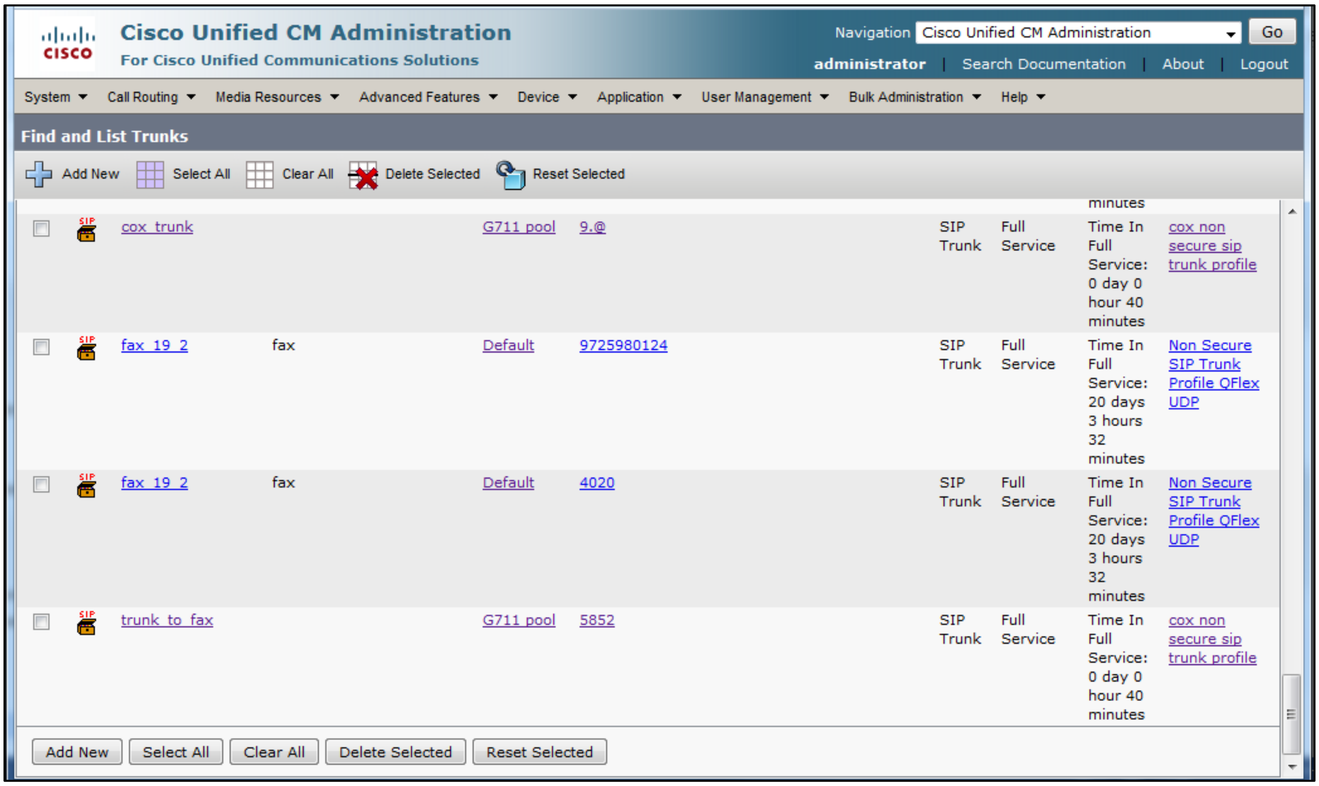


Figure 26: SIP Trunk List

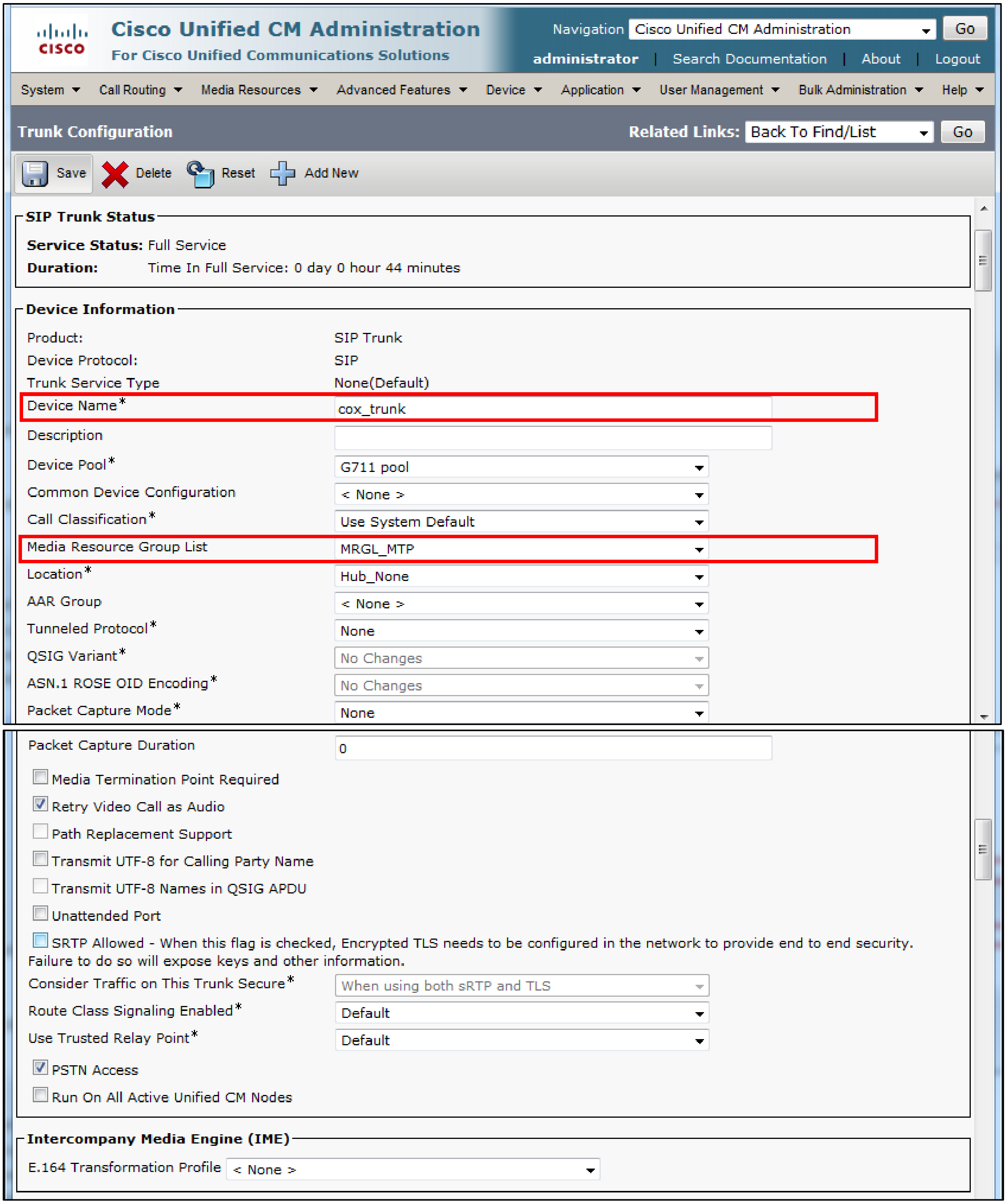


Figure 27: SIP Trunk to CUBE

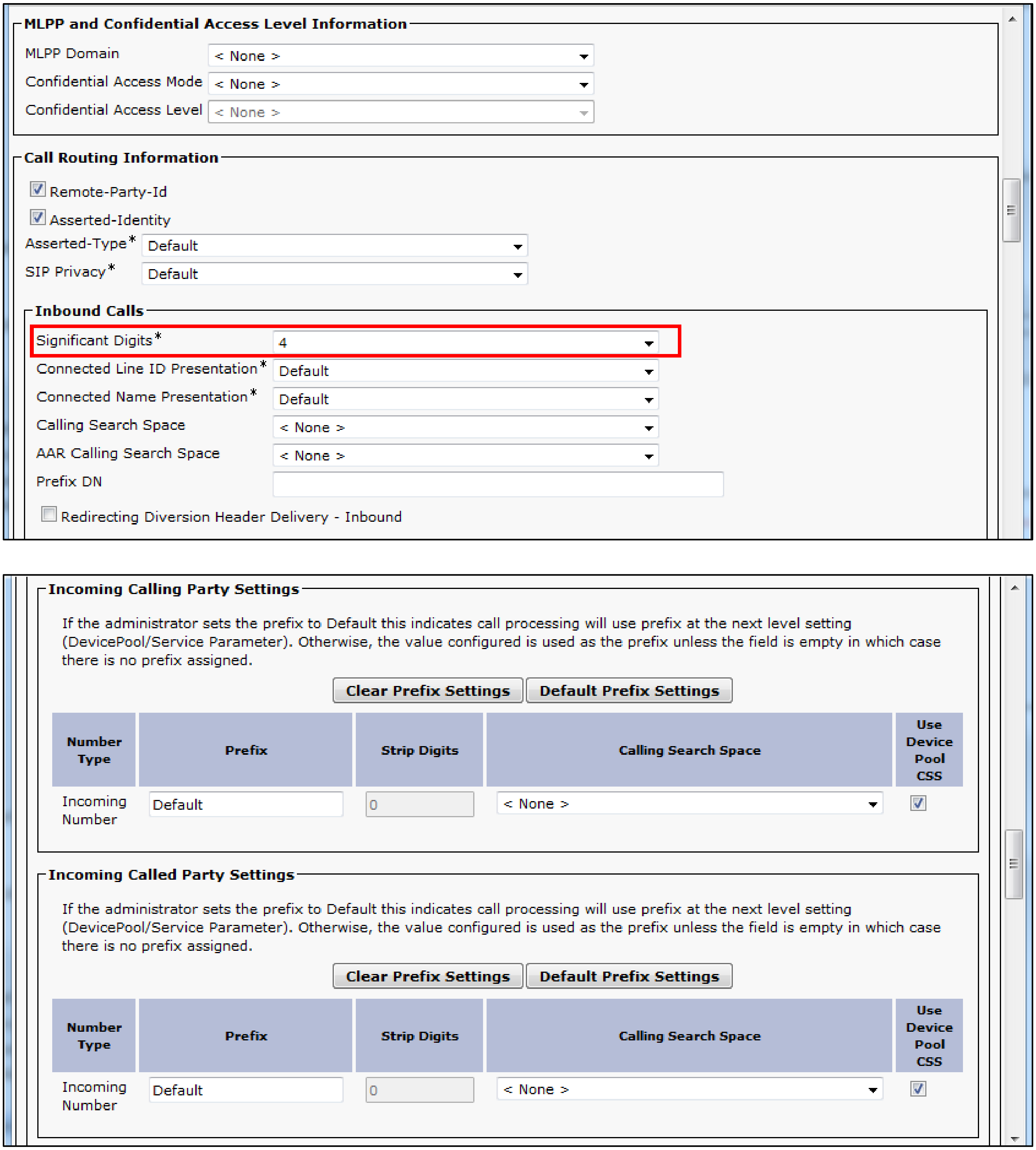
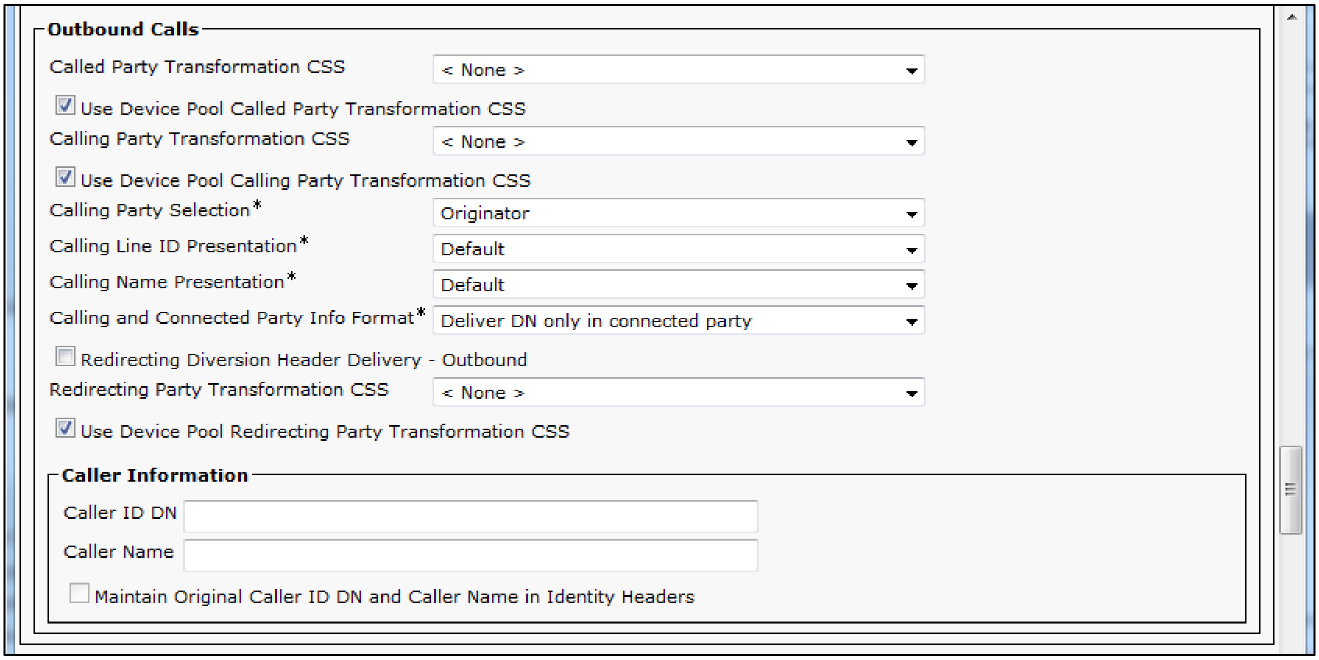


Figure 28: SIP Trunk to CUBE - Cont.



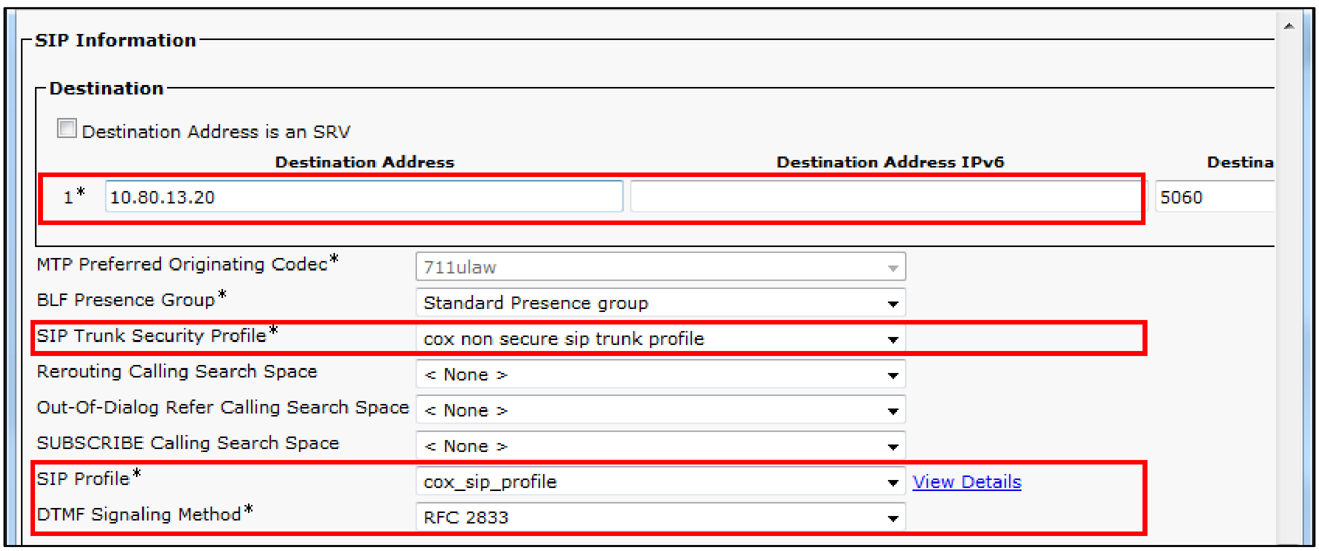


Figure 29: SIP Trunk to CUBE - Cont.

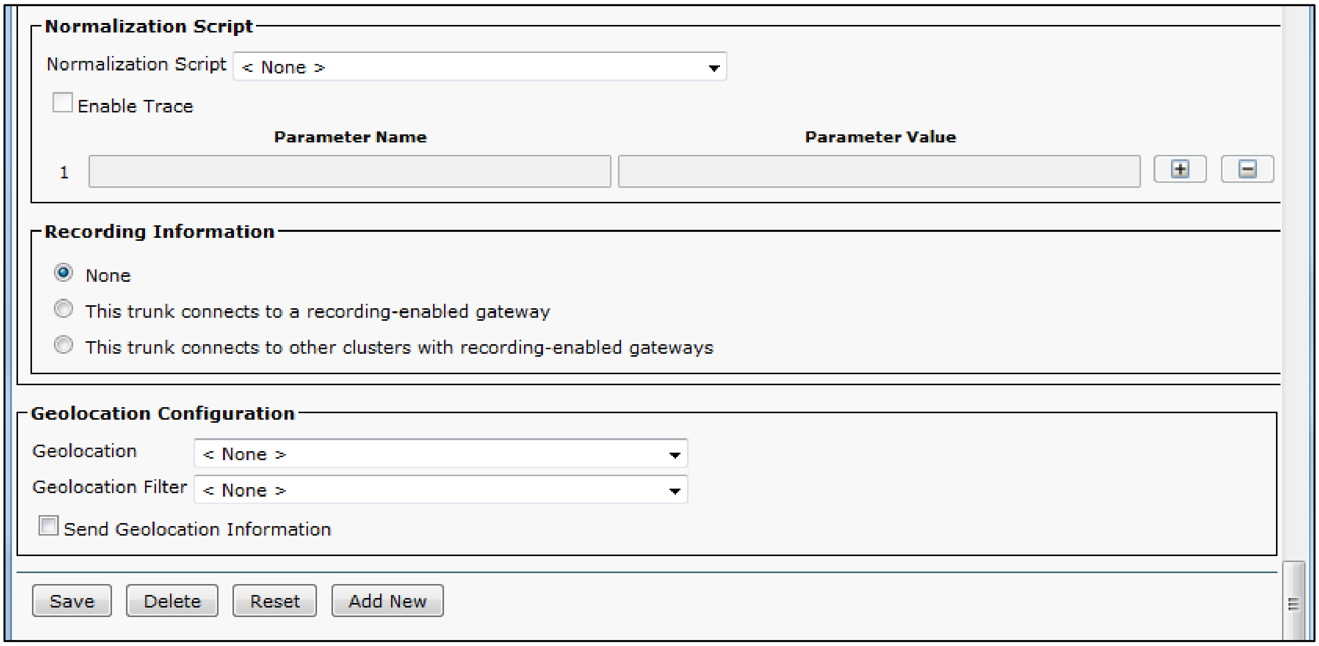


Figure 30: SIP Trunk to CUBE - Cont.

Parameter	Value	Description
Device Name	Cox_Trunk	Name for the trunk
Device Pool	G711 Pool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_MTP	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.13.20	Virtual IP address of the CUBE
SIP Trunk Security Profile	Cox Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	Cox SIP Profile	SIP Profile configured earlier
DTMF Signaling Method	RFC 2833	RFC 2833 is supported for DTMF transport to/from Cox

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

Dial Plan

Route Pattern Configuration

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 5852 will send to fax gateway
- 1000 is the Pilot Number for voicemail to Unity Connection

To Create Route Patterns

1. Navigate to **Call Routing** > **Route/Hunt** > **Route Pattern**
2. Click **Add New**

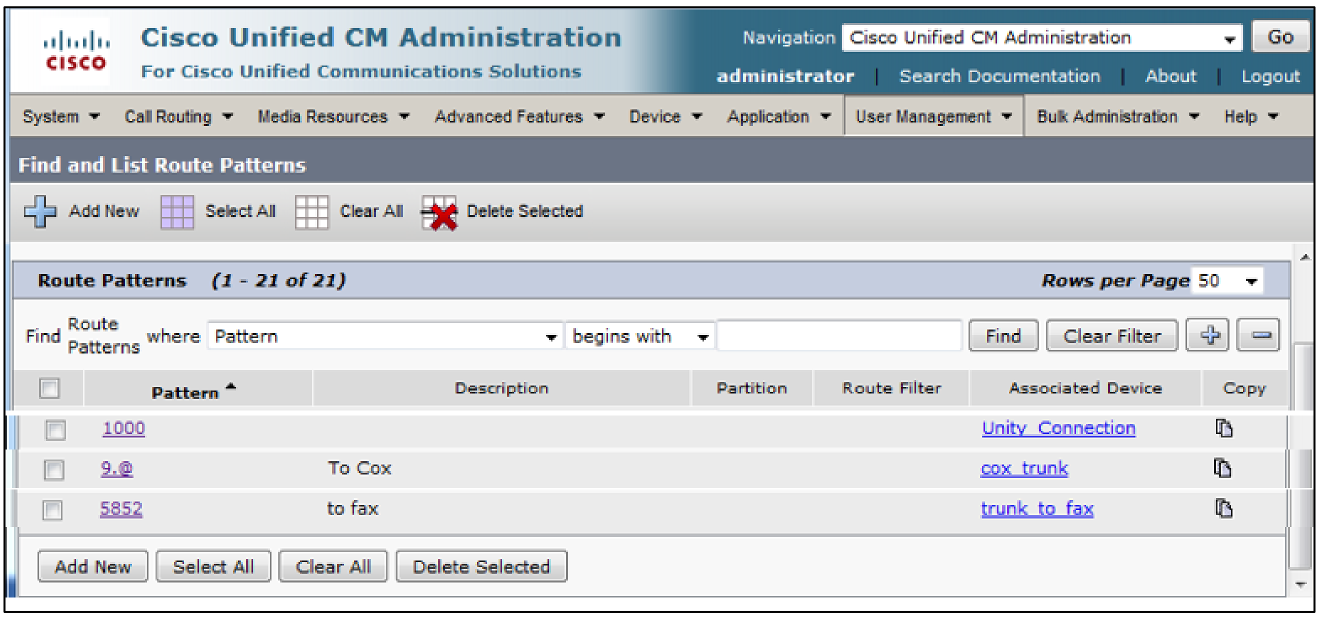


Figure 31: Route Patterns

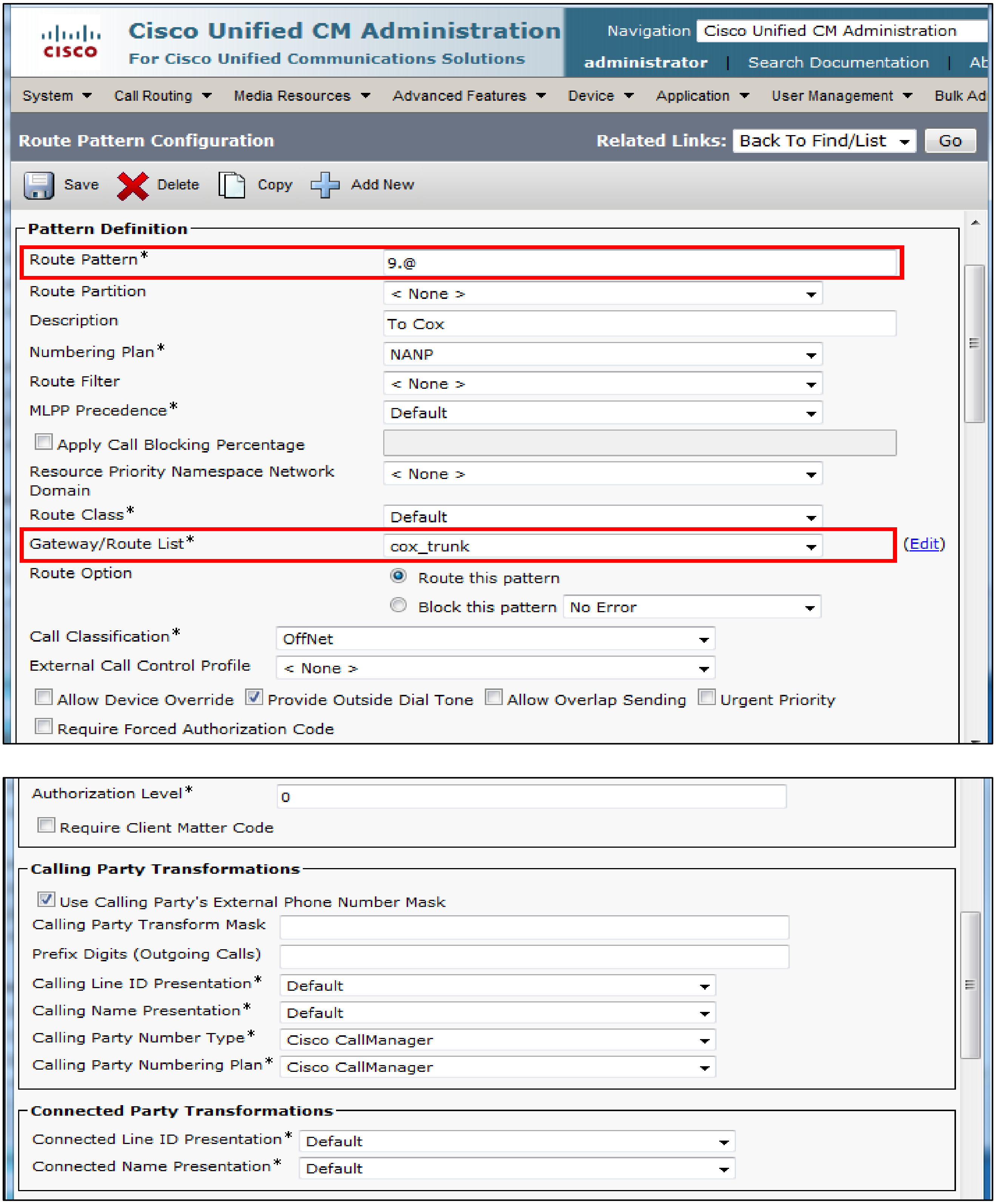


Figure 32: Route Patterns for Voice

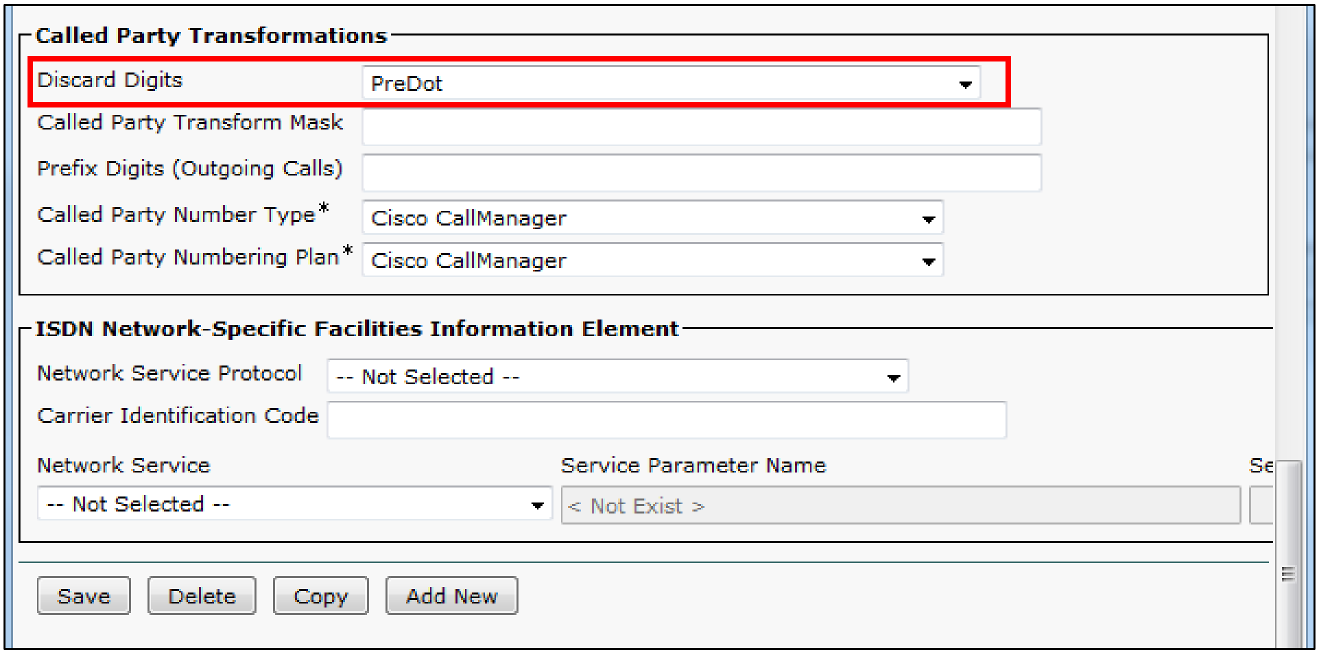


Figure 33: Route Patterns for Voice - Cont.

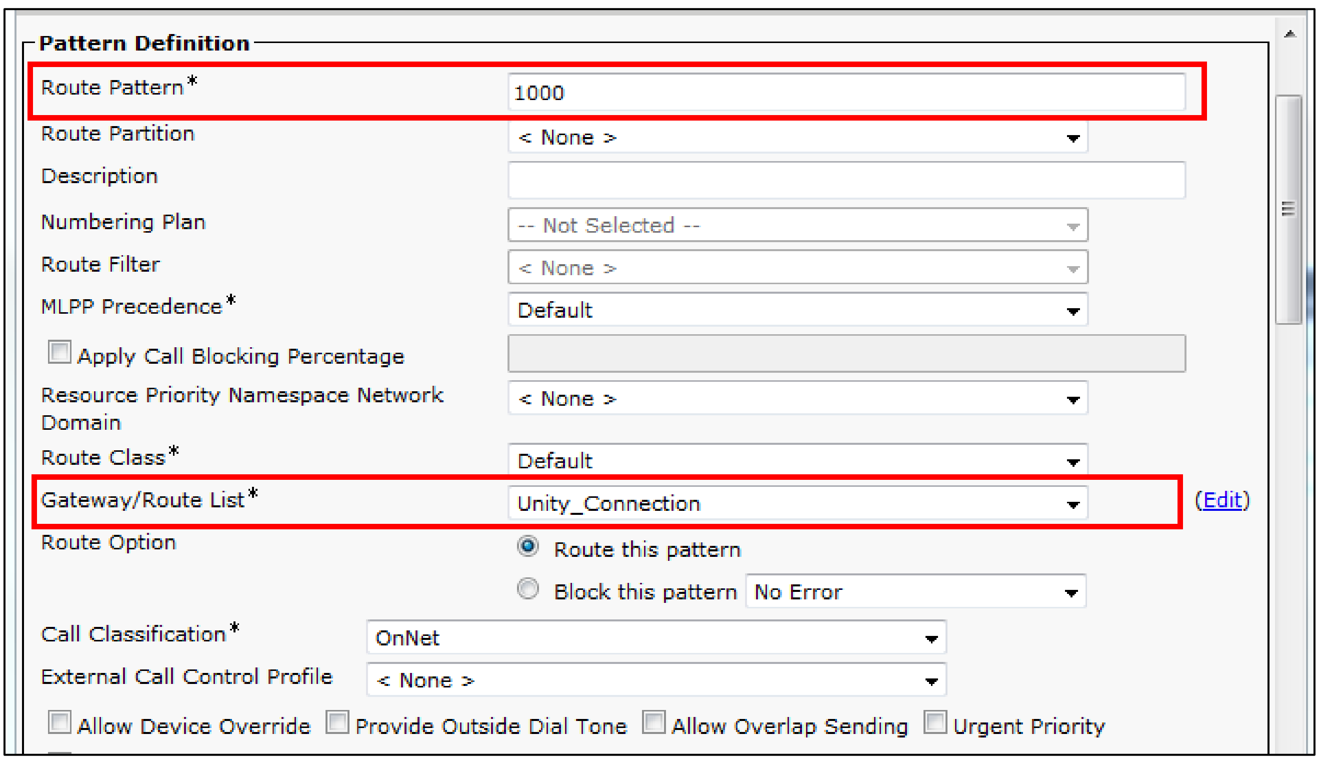
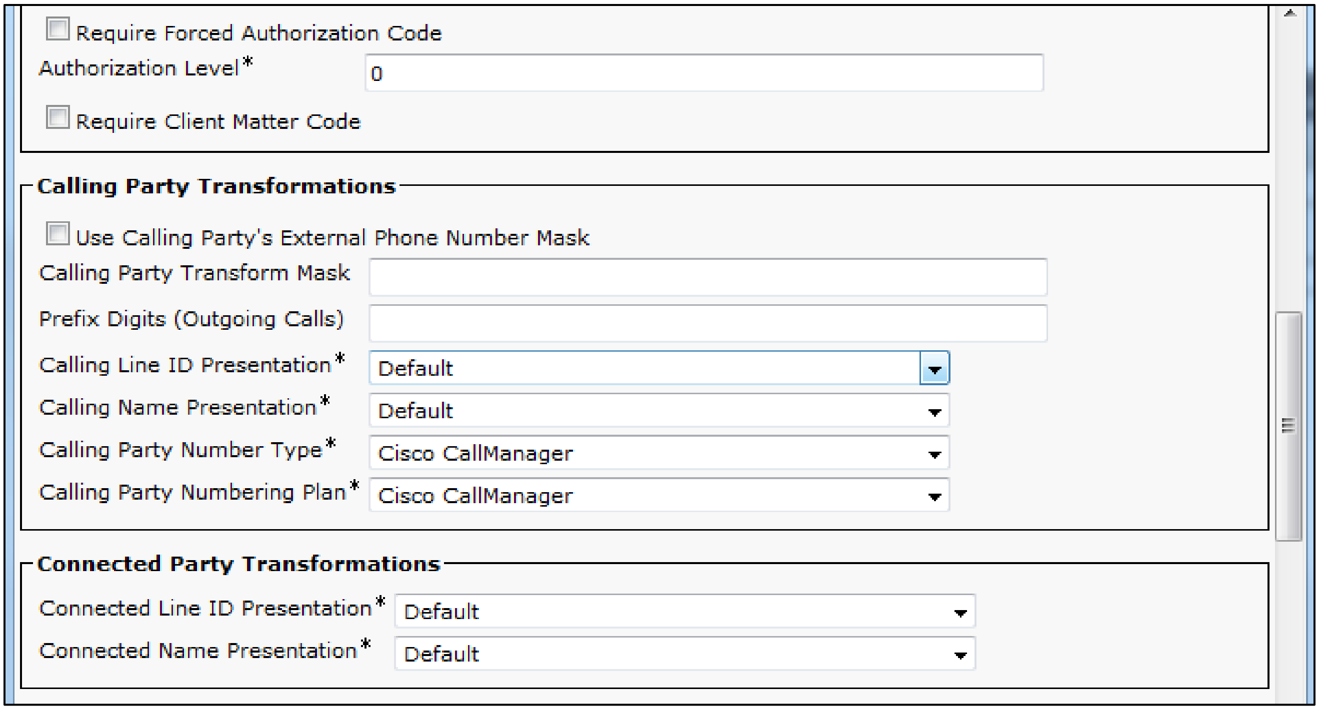


Figure 34: Route Patterns for Unity



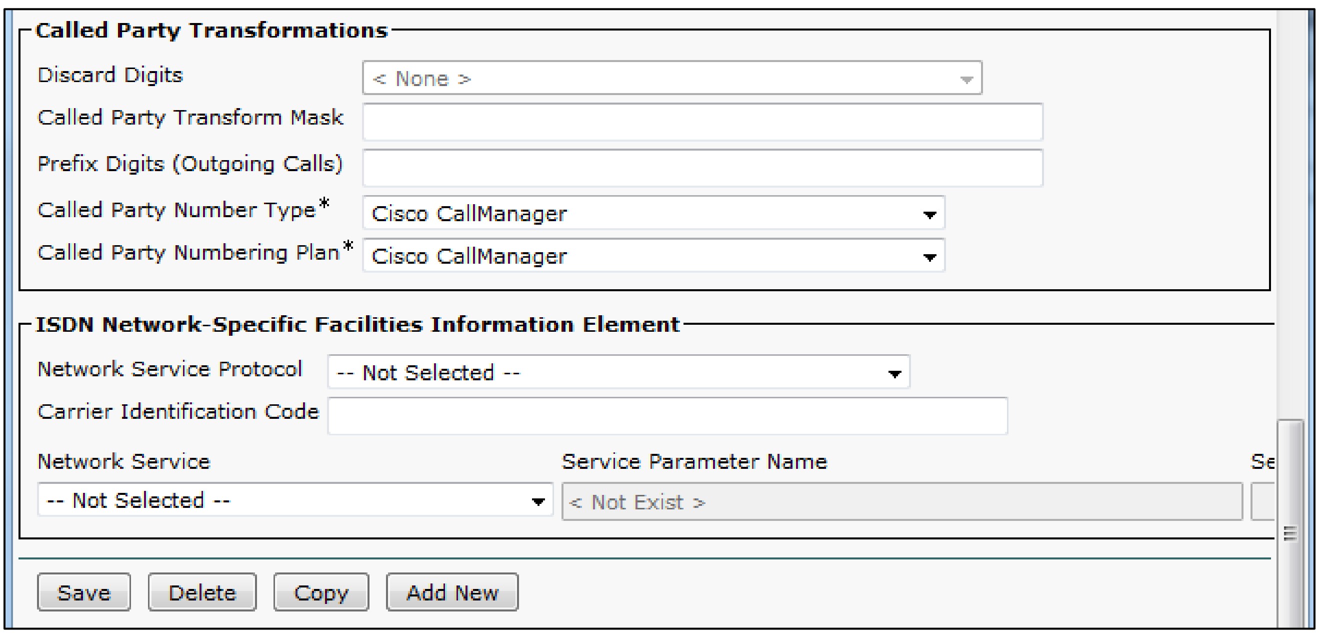


Figure 35: Route Patterns for Unity - Cont.

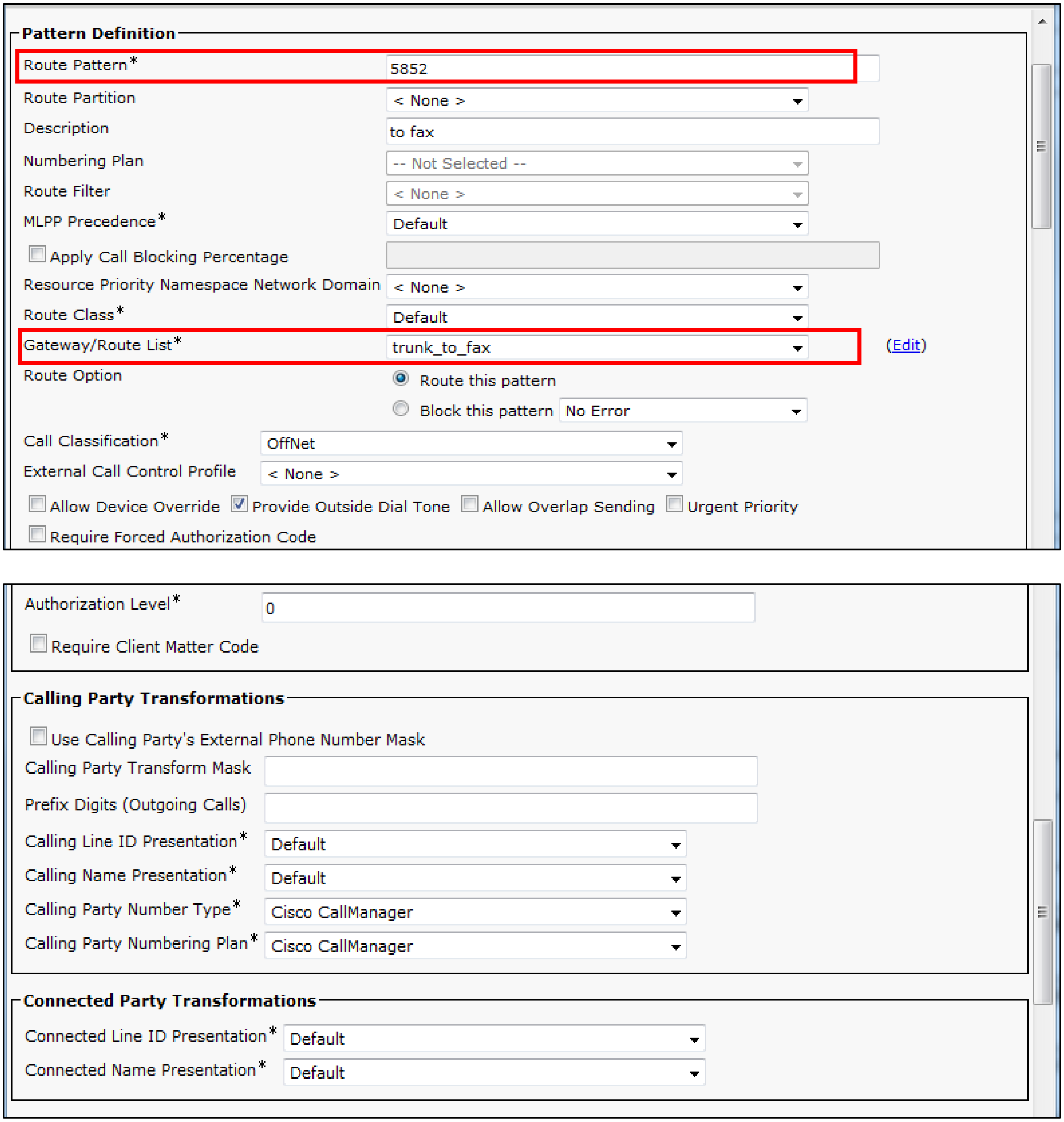


Figure 36: Route Patterns for Fax

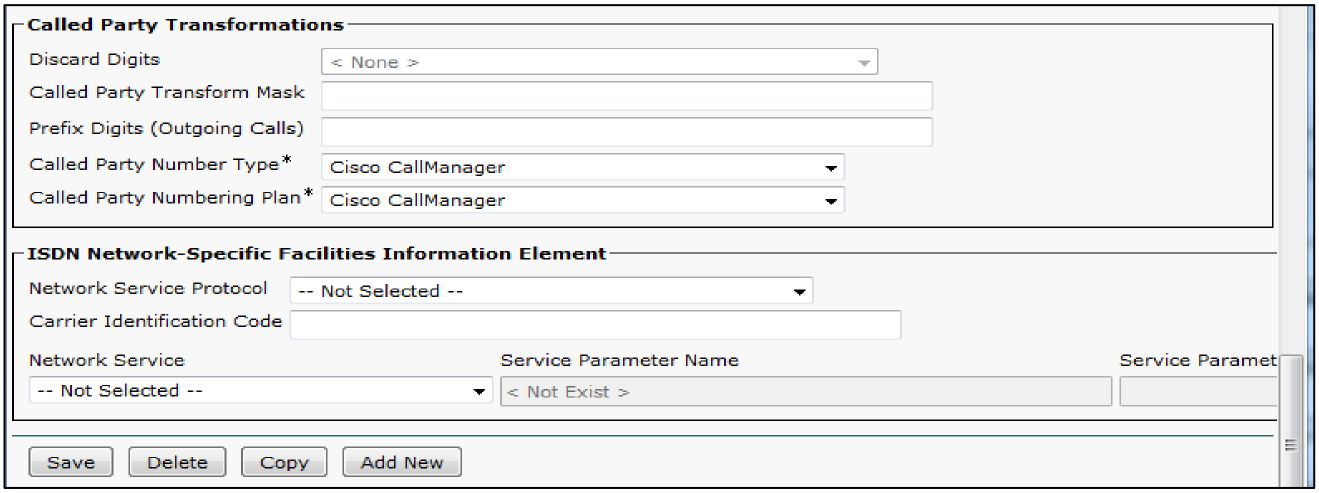


Figure 37: Route Patterns for Fax – Cont.

Setting	Value	Description
Route Pattern	9.@ for Voice call and 8.@ for fax call	Specify appropriate Route Pattern
Gateway/Route List	COX	SIP Trunk name configured earlier
Require Forced Authorization Code	Checked when doing Authorization Code test	Specify if Authorization Code required when make call through this Route Pattern
Require Client Matter Code	Check when doing Account Code test	Specify if Account Code required when make call through this Route Pattern
Calling Party Transform mask	678238XXXX	Specify the Calling Line ID for outgoing call through this Route Pattern
Discard Digits	PreDot for RP 9.@	Specifies how to modify digit before they are sending to Cox ESBC

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

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Appendix A: Test Results