Application Note

Bell Canada SIP Trunking:

Cisco Unified Communications Manager 11.0.1 with Cisco Unified Border Element (CUBE 11.5.0) on ISR 4K [IOS-XE 3.17] using SIP

May 10, 2016

Table of Contents

Table of Figures

Introduction

Service Providers today, such as Bell Canada, 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.

Bell Canada 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 Bell Canada network, Cisco Unified Border Element (CUBE) ISR 15.6(1) S1 can be used. The Cisco Unified Border Element ISR 15.6(1) S1 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.0.1 connected to Bell Canada 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 Bell Canada 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 4K [IOS-XE3.17] for connectivity to Bell Canada SIP trunking service. The deployment model covered in this application note is CPE (Cisco UCM 11.0.1) to PSTN (Bell Canada).
- Testing was performed in accordance to Bell Canada 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 Bell Canada 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 Bell Canada 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/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/dialplan.html

Network Topology

Basic Call Setup

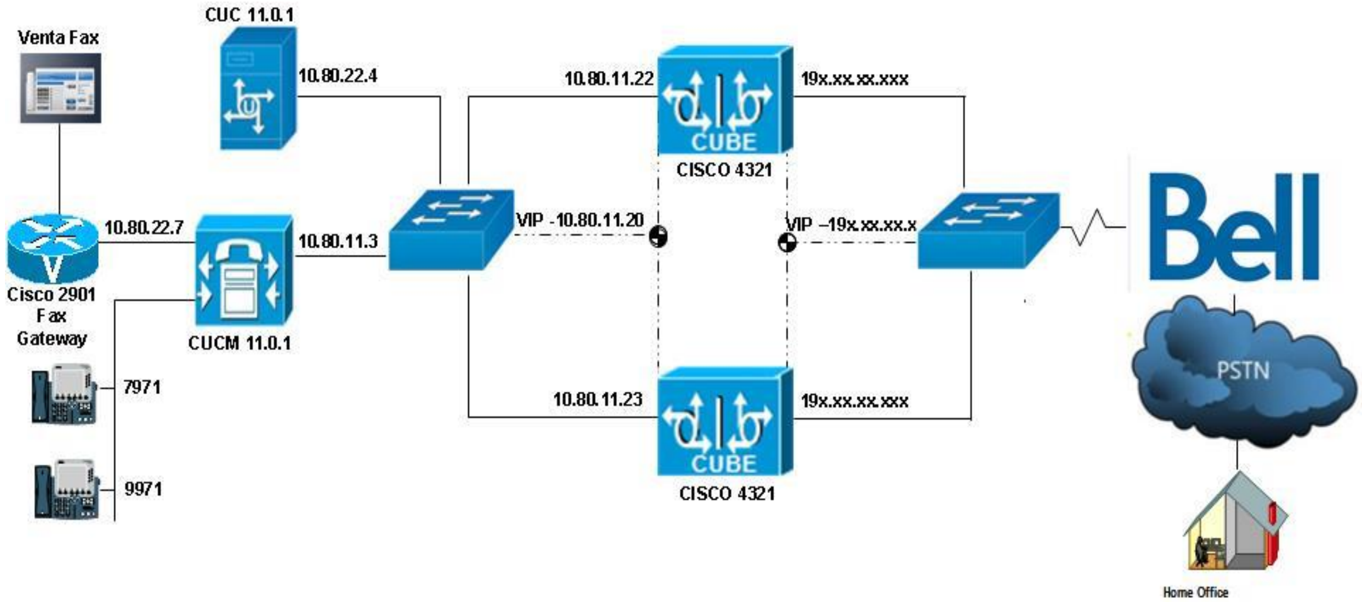


Figure 1: Network Topology

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 7971(SCCP) and 9971(SIP)
  - Please consult “Features not supported” Section for restrictions

Software Requirements

- Cisco Unified Communications Manager 11.0.1
- IOS-XE 3.17 for Cisco Unified Border Element (CUBE 11.5.0) on ISR4321
- IOS 15.4(3)M1 for Fax Gateways on ISR2901
- Cisco Unity Connection 11.0.1

Features

Features Supported

- Incoming and outgoing off-net calls using G729 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
- Fax T.38 is not supported by the Service Provider

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 - 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 BellCanada LAN MS4 1/0/7
 ip address 10.80.13.22 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 5
 redundancy group 2 ip 10.80.13.20 exclusive
!
interface GigabitEthernet0/0/1
 description BellCanada WAN MS4 1/0/8
 ip address 192.65.XXX.XXX 255.255.255.128
 negotiation auto
 redundancy rii 6
 redundancy group 2 ip 192.65.XXX.XXX exclusive
```

Global CUBE Settings

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

```
voice service voip
 ip address trusted list
  ipv4 0.0.0.0 0.0.0.0
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 2
 redirect ip2ip
 fax protocol pass-through g711ulaw
 sip
  asserted-id pai
  localhost dns:venxxxx.xxx.xxxxxnetvoice.ca
  early-offer forced
  history-info
  midcall-signaling passthru
  privacy-policy passthru
  g729 annexb-all
  pass-thru subscribe-notify-events all

 sip-ua
 keepalive target ipv4:69.158.XXX.XXX:5060
 authentication username VEND3_6132606011_01A password 7
 retry invite 2
 sip-server ipv4:69.158.XXX.XXX:5060
```

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
keepalive target	identify SIP server that will receive keepalive packets from SIP gateway
authentication	Enables the SIP digest authentication

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

G729 is used as the preferred codec for this testing

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

SIP Profiles

Static ONND

The following SIP profiles are used in the Dial Peers for Static ONND:

```
voice class sip-profiles 101
 request INVITE sip-header To modify "@.*>" "@siptrunking.bell.ca>"
 request INVITE sip-header SIP-Req-URI modify "69.158.XXX.XXX" "siptrunking.bell.ca"
```

```
 request INVITE sip-header Contact modify "@" ";tgrp=VEND3_613260XXXX_01A;trunk-context=siptrunking.bell.ca@"
 request INVITE sip-header From modify "(@.*)>" "@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request INVITE sip-header P-Asserted-Identity modify "(@.*)>" "@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request INVITE sip-header Diversion modify "reason=unconditional" ""
 request INVITE sip-header Diversion modify "(@.*)>" "@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header From modify "(@.*)>" "@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header P-Asserted-Identity modify "(@.*)>" "@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header Diversion modify "(@.*)>" "@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request INVITE sip-header Diversion modify "reason=user-busy" ""
 request INVITE sip-header Diversion modify "reason=no-answer" ""
!
voice class sip-profiles 103
 request REINVITE sip-header From modify "(@.*)>" "@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header P-Asserted-Identity modify "(@.*)>" "@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header Diversion modify "(@.*)>" "@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>" "<sip:613260\1@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 response 181 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>" "<sip:613260\1@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 response 180 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>" "<sip:613260\1@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
```

Dynamic ONND

The following SIP profiles are used in the Dial Peers for Dynamic ONND

```
voice class sip-profiles 101
 request INVITE sip-header To modify "@.*>" "@siptrunking.bell.ca>"
 request INVITE sip-header SIP-Req-URI modify "69.158.XXX.XXX" "siptrunking.bell.ca"
 request INVITE sip-header Contact modify "@" ";tgrp=VEND3_6132606011_01B;trunk-context=siptrunking.bell.ca@"
 request INVITE sip-header From modify "(@.*)>" "@lxx.xxxxxnetvoice.ca>"
 request INVITE sip-header P-Asserted-Identity modify "(@.*)>" "@lxx.xxxxxnetvoice.ca>"
 request INVITE sip-header Diversion modify "reason=unconditional" ""
 request INVITE sip-header Diversion modify "(@.*)>" "@lxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header From modify "(@.*)>" "@lxx.xxxxxnetvoice.ca>"
 request REINVITE sip-header P-Asserted-Identity modify "(@.*)>" "@lxx.xxxxxnetvoice.ca>"
 request REINVITE sip-header Diversion modify "(@.*)>" "@.lxx.xxxxxnetvoice.ca;user=phone>"
 request INVITE sip-header Diversion modify "reason=user-busy" ""
 request INVITE sip-header Diversion modify "reason=no-answer" ""
!
voice class sip-profiles 103
 request REINVITE sip-header From modify "(@.*)>" "@lxx.xxxxxnetvoice.ca>"
 request REINVITE sip-header P-Asserted-Identity modify "(@.*)>" "@lxx.xxxxxnetvoice.ca>"
 request REINVITE sip-header Diversion modify "(@.*)>" "@lxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>" "<sip:613260\1@lxx.xxxxxnetvoice.ca>"
 response 181 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>" "<sip:613260\1@lxx.xxxxxnetvoice.ca>"
 response 180 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>" "<sip:613260\1@lxx.xxxxxnetvoice.ca>"
```

Dial Peers

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

```
dial-peer voice 100 voip
 description incoming from Bell Canada
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number 613260....
 voice-class codec 1
 voice-class sip profiles 103
 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 vad
!
dial-peer voice 101 voip
 description "incoming call from Bell facing SME"
 destination-pattern 613260....
 session protocol sipv2
 session target ipv4:10.80.11.3
 session transport udp
 voice-class codec 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 vad
!
dial-peer voice 201 voip
 description incoming from SME
```

```
 session protocol sipv2
 session target sip-server
 incoming called-number .T
 voice-class codec 1
 voice-class sip profiles 103
  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 vad
!
dial-peer voice 200 voip
 description outgoing call to BellCanada facing Bell
 translation-profile outgoing AddPlus
 preference 1
 destination-pattern .T
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip profiles 101
 voice-class sip options-keepalive down-interval 20
 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 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 "8" 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 "8". A "8.@" 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

Active Cisco UBE

```
Bell_CUBE1#show running-config
Building configuration...
Current configuration : 8782 bytes
!
! Last configuration change at 11:13:26 UTC Wed May 4 2016 by cisco
!
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
service sequence-numbers
no platform punt-keepalive disable-kernel-core
!
hostname Bell_CUBE1
!
boot-start-marker
boot system bootflash:isr4300-universalk9.03.17.01.S.156-1.S1-std.SPA.bin
boot-end-marker
!
```

```
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
logging queue-limit 10000
logging buffered 999999999
no logging rate-limit
no logging console
no logging monitor
!
aaa new-model
!
aaa session-id common
!
no ip domain lookup
ip domain name tekvizion.com
!
subscriber templating
!
multilink bundle-name authenticated
!
```

```
voice service voip
 ip address trusted list
  ipv4 0.0.0.0 0.0.0.0
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 2
 redirect ip2ip
 fax protocol pass-through g711ulaw
 sip
  asserted-id pai
  localhost dns:venxxxx.xxx.xxxxxnetvoice.ca
  early-offer forced
  history-info
  midcall-signaling passthru
  privacy-policy passthru
  g729 annexb-all
  pass-thru subscribe-notify-events all
!
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
!
voice class sip-profiles 101
 request INVITE sip-header To modify "@.*>" "@siptrunking.bell.ca>"
 request INVITE sip-header SIP-Req-URI modify "69.158.XXX.XXX" "siptrunking.bell.ca"
 request INVITE sip-header Contact modify "@" ";tgrp=VEND3_6132606011_01A;trunk-
context=siptrunking.bell.ca@"
```

```
 request INVITE sip-header From modify "(@.*)>" "@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request INVITE sip-header P-Asserted-Identity modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request INVITE sip-header Diversion modify "reason=unconditional" ""
 request INVITE sip-header Diversion modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header From modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header P-Asserted-Identity modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header Diversion modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request INVITE sip-header Diversion modify "reason=user-busy" ""
 request INVITE sip-header Diversion modify "reason=no-answer" ""
!
voice class sip-profiles 103
 request REINVITE sip-header From modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header P-Asserted-Identity modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header Diversion modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>"
"<sip:613260\1@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 response 181 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>"
"<sip:613260\1@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 response 180 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>"
"<sip:613260\1@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
!
voice translation-rule 108
 rule 1 /\(^..........$\)/ /+\1/
!
voice translation-profile AddPlus
translate called 108
```

```
license udi pid ISR4321/K9 sn FDO19220MW3
license boot level appxk9
license boot level uck9
!
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 2
   name BellCanada
   priority 100 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/1/0 protocol 1
   data GigabitEthernet0/1/0
   track 1 shutdown
   track 2 shutdown
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
 description BellCanada LAN MS4 1/0/7
 ip address 10.80.13.22 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 5
 redundancy group 2 ip 10.80.13.20 exclusive
```

```
!
interface GigabitEthernet0/0/1
 description BellCanada WAN MS4 1/0/8
 ip address 192.65.XXX.XXX 255.255.255.128
 negotiation auto
 redundancy rii 6
 redundancy group 2 ip 192.65.XXX.XXX exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA MS5 3/0/36
 ip address 10.89.20.8 255.255.255.0
 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 rtcp report interval 3000
ip route 0.0.0.0 0.0.0.0 192.65.XXX.XXX
ip route 10.64.0.0 255.255.0.0 10.80.13.1
ip route 10.80.11.0 255.255.255.0 10.80.13.1
```

```
ip route 10.80.13.0 255.255.255.0 10.80.13.1
ip route 10.89.9.0 255.255.255.0 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 100 voip
 description incoming from Bell Canada
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number 613260....
 voice-class codec 1
 voice-class sip profiles 103
 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 vad
!
dial-peer voice 101 voip
 description "incoming call from Bell facing SME"
 destination-pattern 613260....
 session protocol sipv2
```

```
 session target ipv4:10.80.11.3
 session transport udp
 voice-class codec 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 vad
!
dial-peer voice 201 voip
 description incoming from SME
 session protocol sipv2
 session target sip-server
 incoming called-number .T
 voice-class codec 1
 voice-class sip profiles 103
 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 vad
!
dial-peer voice 200 voip
 description outgoing call to BellCanada facing Bell
 translation-profile outgoing AddPlus
 preference 1
 destination-pattern .T
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip profiles 101
 voice-class sip options-keepalive down-interval 20
```

```
 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 vad
!
dial-peer voice 300 voip
 description PBX to PBX call via Bell - WAN facing
 destination-pattern 06T
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip profiles 101
 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 vad
!
dial-peer voice 301 voip
 description PBX to PBX call via Bell - LAN facing
 session protocol sipv2
 session target sip-server
 incoming called-number 06T
 voice-class codec 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 vad
!
dial-peer voice 400 voip
 description International outgoing call to BellCanada facing Bell
```

```
 preference 1
 destination-pattern 011T
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip profiles 101
 voice-class sip options-keepalive down-interval 20
 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 vad
!
dial-peer voice 401 voip
 description International incoming from SME
 session protocol sipv2
 session target sip-server
 incoming called-number 011T
 voice-class codec 1
 voice-class sip profiles 103
 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 vad
!
!
sip-ua
 keepalive target ipv4:69.158.XXX.XXX:5060
 authentication username VEND3_6132606011_01A password 7
 retry invite 2
 sip-server ipv4:69.158.XXX.XXX:5060
```

```
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password xxxxxxxxxx
 transport input telnet ssh
!
!
end
```

Standby Cisco UBE

```
Bell_CUBE2#show running-config
Building configuration...

Current configuration : 8792 bytes
!
! Last configuration change at 11:10:43 UTC Wed May 4 2016 by cisco
!
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname Bell_CUBE2
!
boot-start-marker
boot system bootflash:isr4300-universalk9.03.17.01.S.156-1.S1-std.SPA.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
```

```
logging queue-limit 10000
logging buffered 100000000
logging rate-limit 10000
no logging console
!
no aaa new-model
!
no ip domain lookup
ip domain name tekvizion.com
!
subscriber templating
!
multilink bundle-name authenticated
!
```

```
voice service voip
 ip address trusted list
  ipv4 0.0.0.0 0.0.0.0
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 2
 redirect ip2ip
 fax protocol pass-through g711ulaw
 sip
  asserted-id pai
  localhost dns:venxxxx.xxx.xxxxxnetvoice.ca
  early-offer forced
  history-info
  midcall-signaling passthru
  privacy-policy passthru
  g729 annexb-all
  pass-thru subscribe-notify-events all
!
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
!
```

```
voice class sip-profiles 101
 request INVITE sip-header To modify "@.*>" "@siptrunking.bell.ca>"
 request INVITE sip-header SIP-Req-URI modify "69.158.XXX.XXX" "siptrunking.bell.ca"
 request INVITE sip-header Contact modify "@" ";tgrp=VEND3_6132606011_01A;trunk-
context=siptrunking.bell.ca@"
 request INVITE sip-header From modify "(@.*)>" "@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request INVITE sip-header P-Asserted-Identity modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request INVITE sip-header Diversion modify "reason=unconditional" ""
 request INVITE sip-header Diversion modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header From modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header P-Asserted-Identity modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header Diversion modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request INVITE sip-header Diversion modify "reason=user-busy" ""
 request INVITE sip-header Diversion modify "reason=no-answer" ""
!
voice class sip-profiles 103
 request REINVITE sip-header From modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header P-Asserted-Identity modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header Diversion modify "(@.*)>"
"@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 request REINVITE sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>"
"<sip:613260\1@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 response 181 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>"
"<sip:613260\1@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
 response 180 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>"
"<sip:613260\1@venxxxx.xxx.xxxxxnetvoice.ca;user=phone>"
```

```
voice translation-rule 108
 rule 1 /\(^...........$\)/ /+\1/
!
voice translation-profile AddPlus
translate called 108
!
license udi pid ISR4321/K9 sn FDO19220MQ9
!
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 2
   name BellCanada
   priority 100 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/1/0 protocol 1
   data GigabitEthernet0/1/0
   track 1 shutdown
   track 2 shutdown
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
!
```

```
interface GigabitEthernet0/0/0
 description BellCanada CUBE2 LAN MS4 1/0/11
 ip address 10.80.13.23 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 5
 redundancy group 2 ip 10.80.13.20 exclusive
!
interface GigabitEthernet0/0/1
 description BellCanada CUBE2 WAN MS4 1/0/12
 ip address 192.65.XXX.XXX 255.255.255.128
 negotiation auto
 redundancy rii 6
 redundancy group 2 ip 192.65.XXX.XXX exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA MS5 3/0/38
 ip address 10.89.20.10 255.255.255.0
 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 rtcp report interval 300
ip route 0.0.0.0 0.0.0.0 192.65.XXX.XXX
ip route 10.64.0.0 255.255.0.0 10.80.13.1
ip route 10.80.11.0 255.255.255.0 10.80.13.1
ip route 10.80.13.0 255.255.255.0 10.80.13.1
ip route 10.89.9.0 255.255.255.0 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 100 voip
 description incoming from Bell Canada
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number 613260....
 voice-class codec 1
```

```
 voice-class sip profiles 103
 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 vad
!
dial-peer voice 101 voip
 description "incoming call from Bell facing SME"
 destination-pattern 613260....
 session protocol sipv2
 session target ipv4:10.80.11.3
 session transport udp
 voice-class codec 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 vad
!
dial-peer voice 201 voip
 description incoming from SME
 session protocol sipv2
 session target sip-server
 incoming called-number .T
 voice-class codec 1
 voice-class sip profiles 103
 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 vad
```

```
!
dial-peer voice 200 voip
 description outgoing call to BellCanada facing Bell
 translation-profile outgoing AddPlus
 preference 1
 destination-pattern .T
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip profiles 101
 voice-class sip options-keepalive down-interval 20
 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 vad
!
dial-peer voice 300 voip
 description PBX to PBX call via Bell - WAN facing
 destination-pattern 06T
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip profiles 101
 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 vad
!
dial-peer voice 301 voip
```

```
 description PBX to PBX call via Bell - LAN facing
 session protocol sipv2
 session target sip-server
 incoming called-number 06T
 voice-class codec 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 vad
!
dial-peer voice 400 voip
 description International outgoing call to BellCanada facing Bell
 preference 1
 destination-pattern 011T
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip profiles 101
 voice-class sip options-keepalive down-interval 20
 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 vad
!

dial-peer voice 401 voip
 description International incoming from SME
 session protocol sipv2
 session target sip-server
```

```
 incoming called-number 011T
 voice-class codec 1
 voice-class sip profiles 103
 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 vad
!
!
sip-ua
 keepalive target ipv4:69.158.XXX.XXX:5060
 authentication username VEND3_6132606011_01A password 7
 retry invite 2
 sip-server ipv4:69.158.XXX.XXX:5060
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password xxxxxxxxxx
 login local
!
!
end
```

Configuring the Cisco Unified Communications Manager

Cisco Call Manager Service Parameters

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

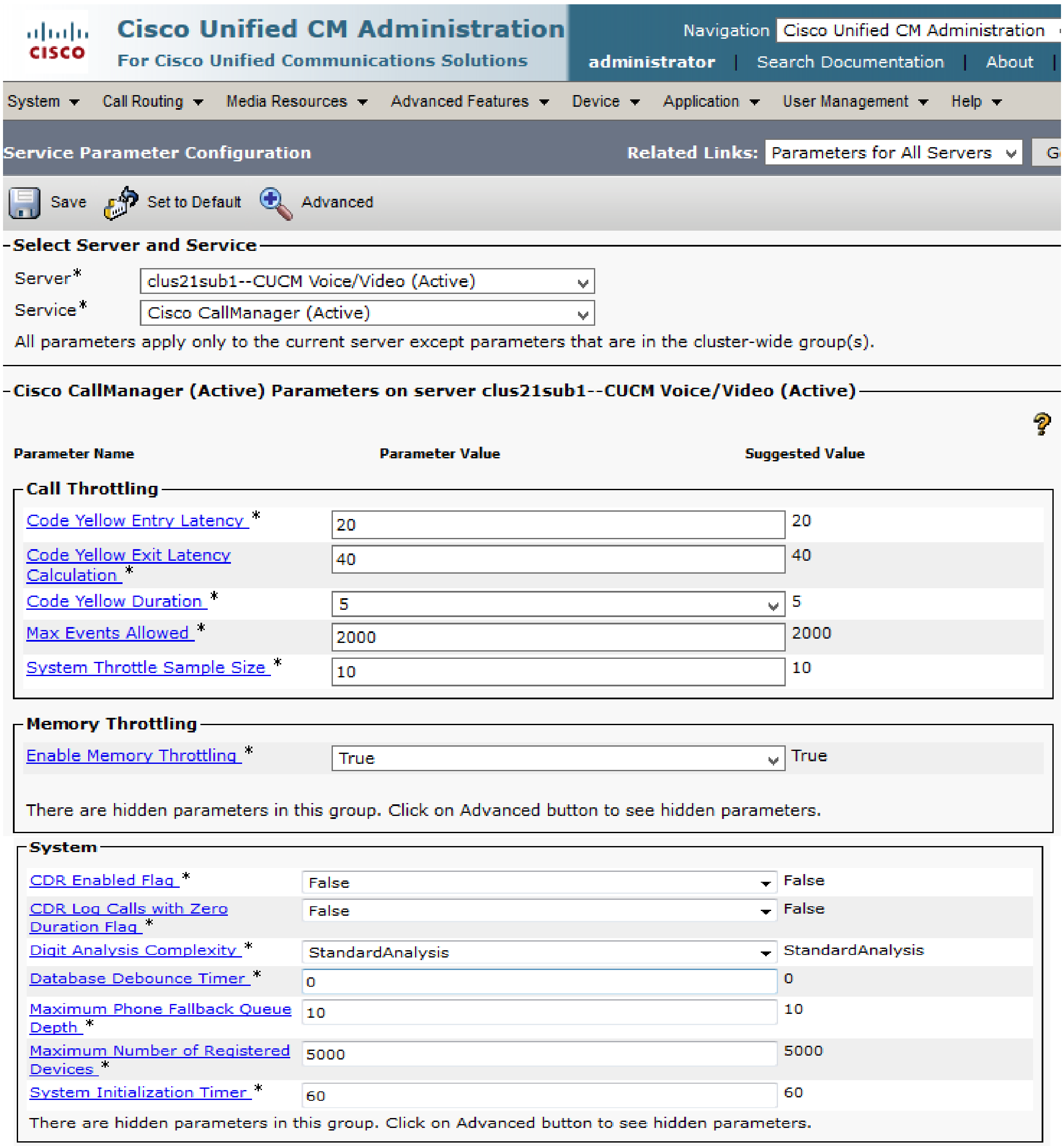


Figure 2: Service Parameters

SDL Trace

SDL Trace Data Flags *	0x00000111	0x00000111
SDL Trace Flush Immediately *	False	False
SDL Trace Data Size *	0	0
SDL Trace Flag *	True	True
SDL TraceType Flags *	0x8000EB15	0x8000EB15

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

Clusterwide Parameters (Device - General)

Call Diagnostics Enabled *	Disabled	Disabled
Show Line Group Member DN in finalCalledPartyNumber CDR Field *	False	False
Show Line Group Member Non Masked DN in finalCalledPartyNumber CDR Field *	False	False
CTI New Call Accept Timer *	4	4
CTI Generate Digits Interval *	250	250
CTI Dial Digits Interval *	250	250
CTI Await Further Digits *	False	False
CTI Use Wildcard Pattern as calledPartyDN *	False	False
CTI Report Ringback on SIP 183 with SDP *	True	True
Retain Media on Disconnect with PI for Active Call *	False	False
Station and Backup Server KeepAlive Interval *	60	60
Station KeepAlive Interval *	30	30
Status Enquiry Poll Flag *	False	False
Strip # Sign from Called Party Number *	True	True
Session Handoff Alerting Timer *	10	10
T301 Timer *	180000	180000
T302 Timer *	15000	15000
T303 Timer *	4000	4000
T304 Timer *	30000	30000
T305 Timer *	30000	30000
T306 Timer *	30000	30000

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

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

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.

Optimize MGCP Registration *	True	True
Suppress Out-of-Channels Alarms *	True	True
I-Frame Timer *	2000	2000
User-to-User IE Status *	False	False
Convert European Progress Message to Alerting *	False	False
Enable DMS PRI Notify Message from User to Network *	True	True
Audit OOS Channels Interval *	10	10
Digital and Analog Ports Enabled *	True	True

Clusterwide Parameters (Device - H323)

Accept Unknown TCP Connection *	False	False
BRQ Enabled *	False	False
Call Present Disconnect Flag *	False	False
Check Progress Indicator Before Establishing Media *	False	False
H225 Block Setup Destination *	False	False
H225 DB Retry Timer *	0	0
H225 Device Connect Timer *	0	0
H225 DTMF Duration *	100	100
H225 TspReq Retry *	2	2
H225 Intercluster Call Throttle Timer *	30	30
H225 T301 Timer *	180000	180000
H225 T302 Timer *	15000	15000
H225 T303 Timer *	4000	4000
H225 T304 Timer *	30000	30000
H225 T305 Timer *	30000	30000
H225 T310 Timer *	60000	60000
H225 TCP Timer *	5	5
H245 TCS Timeout *	10	10
H323 Calling Party Number Screening Indicator *	Calling number screened and passed	Calling number screened and passed
Apply External Phone Number Mask for H.323 Calls *	False	False
Tone on Connect *	False	False
Wait Time for SDP with SR/RO Mode *	3	3

Figure 7: Service Parameters – Cont.

RAS ARQ Timer *	3	3
RAS BRQ Timer *	3	3
RAS DRQ Timer *	3	3
RAS RRQ Timer *	3	3
Ras URQ Timer *	3	3
Retry Count for ARQ *	2	2
Retry Count for BRQ *	2	2
Retry Count for DRQ *	2	2
Retry Count for RRQ *	2	2
Retry Count for URQ *	1	1
Send Product ID and Version ID *	False	False
Send Unified CM Version as Version ID in H225Setup *	False	False
Send Progress Timer *	3000	3000
Send H225 User Info Message *	User Info for Call Progress Tone	User Info for Call Progress Tone
Status Enquiry Poll Timer *	10000	10000
Device Name of GK-controlled Trunk That Will Use Port 1720 *	None	None
Host Name/IP Address of GK That Will Use RAS UDP Port 1719 *	None	None
Fail Call If MTP Allocation Fails *	False	False
Overlap Receiving Flag for H323 *	False	False
Allocate Transcoder for H.323 on Early Offer SIP Trunk for Calls with Early Media *	False	False

Clusterwide Parameters (Device - SIP)

SIP Interoperability Enabled *	True	True
Retry Count for SIP Bye *	10	10
Retry Count for SIP Cancel *	10	10
Retry Count for SIP Invite *	6	6
Retry Count for SIP PRACK *	6	6
Retry Count for SIP Rel1XX *	10	10
Retry Count for SIP Publish *	6	6
Retry Count for SIP Response *	6	6
SIP Connect Timer *	500	500
SIP Disconnect Timer *	500	500

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

Figure 9: Service Parameters – Cont.

Clusterwide Parameters (Feature - General)

Call Park Display Timer *	10	10
Caller ID Display Priority Enabled *	True	True
Call Park Reversion Timer *	60	60
Park Monitoring Reversion Timer *	60	60
Park Monitoring Periodic Reversion Timer *	30	30
Park Monitoring Forward No Retrieve Timer *	300	300
Preserve globalCallId for Parked Calls *	True	True
Maximum Call Duration Timer *	720	720
Maximum Hold Duration Timer *	360	360
Party Entrance Tone *	True	True
Message Waiting Lamp Policy *	Primary Line - Light and Prompt	Primary Line - Light and Prompt
Audible Message Waiting Indication Policy *	OFF	OFF
Message Waiting Indicator Inbound Calling Search Space	< None >	
Multiple Tenant MWI Modes *	False	False
MWI Non Message Center Signaling Call Duration *	0	0
Message Waiting Indicator APDU Digit Translation CSS	< None >	
Block OffNet To OffNet Transfer *	False	False
Use Original Call Classification for Transferred Calls *	False	False
Use Restriction attribute of ID/Name Presentation of Transferring Party *	True	True
Local route group for redirected calls *	Local route group of calling party	Local route group of calling party
Block Unencrypted Calls *	False	False

Clusterwide Parameters (Feature - Conference)

Suppress MOH to Conference Bridge *	True	True
Drop Ad Hoc Conference *	Never	Never
Maximum Ad Hoc Conference *	4	4
Maximum MeetMe Conference Unicast *	4	4
Advanced Ad Hoc Conference Enabled *	False	False
Choose Encrypted Audio Conference Instead Of Video Conference *	True	True

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 encr	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.

Clusterwide Parameters (Feature - Call Pickup)

Auto Call Pickup Enabled *	False	False
Call Pickup Locating Timer *	1	1
Call Pickup No Answer Timer *	12	12

Clusterwide Parameters (Feature - Refer)

Validate Refer-to URI *	Validate Except for Anonymous Users	Validate Except for Anonymous Users

Clusterwide Parameters (Feature - Replaces)

Block OffNet To OffNet Replaces *	False	False

Clusterwide Parameters (Feature - Redirection [3xx])

Redirection Ring No Answer Reversion Timer *	24	24
Maximum Redirection Count *	70	70

Clusterwide Parameters (Feature - Multilevel Precedence and Preemption)

Locations-based MLPP Enable *	False	False
Executive Override Call Preemptable *	False	False
Location-based Maximum Bandwidth Enforcement Level for MLPP Calls *	Lenient	Lenient
Non-Preemption Pattern CSS	< None >	
MLPP Exception Level *	Executive Override	Executive Override

Clusterwide Parameters (Feature - Path Replacement)

Path Replacement Enabled *	False	False
Path Replacement on Tromboned Calls *	True	True
Start Path Replacement Minimum Delay Time *	0	0
Start Path Replacement Maximum Delay Time *	0	0
Path Replacement T1 Timer *	30	30
Path Replacement T2 Timer *	15	15
Path Replacement PINX ID		
Path Replacement Calling Search Space	< None >	

Figure 12: Service Parameters – Cont.

Clusterwide Parameters (Feature - Call Back)

Call Back Enabled Flag *	True	True
Call Back Notification Audio File Name *	CallBack.raw	CallBack.raw
Connection Proposal Type *	Connection Retention	Connection Retention
Connection Response Type *	Default to Connection Retention	Default to Connection Retention
Call Back Request Protection T1 Timer *	10	10
Call Back Recall T3 Timer *	20	20
Call Back Calling Search Space	< None >	
No Path Reservation *	True	True
Set Private Numbering Plan for Call Back *	False	False
Set Type of Number for Call Back *	Level1RegionalNumber	Level1RegionalNumber

Clusterwide Parameters (Feature - Call Recording)

Play Recording Notification Tone To Observed Target *	False	False
Play Recording Notification Tone To Observed Connected Parties *	False	False

Clusterwide Parameters (Feature - Monitoring)

Play Monitoring Notification Tone To Observed Target *	False	False
Play Monitoring Notification Tone To Observed Connected Parties *	False	False

Clusterwide Parameters (Feature - Join Across Lines And Single Button Barge Feature Set)

Join Across Lines Policy *	Off	Off
Single Button Barge/CBarge Policy *	Off	Off
Allow Barging When Ringing *	False	False

Clusterwide Parameters (Feature - Secure Tone)

Play Tone to Indicate Secure/Non-Secure Call Status *	False	False

Figure 13: Service Parameters – Cont.

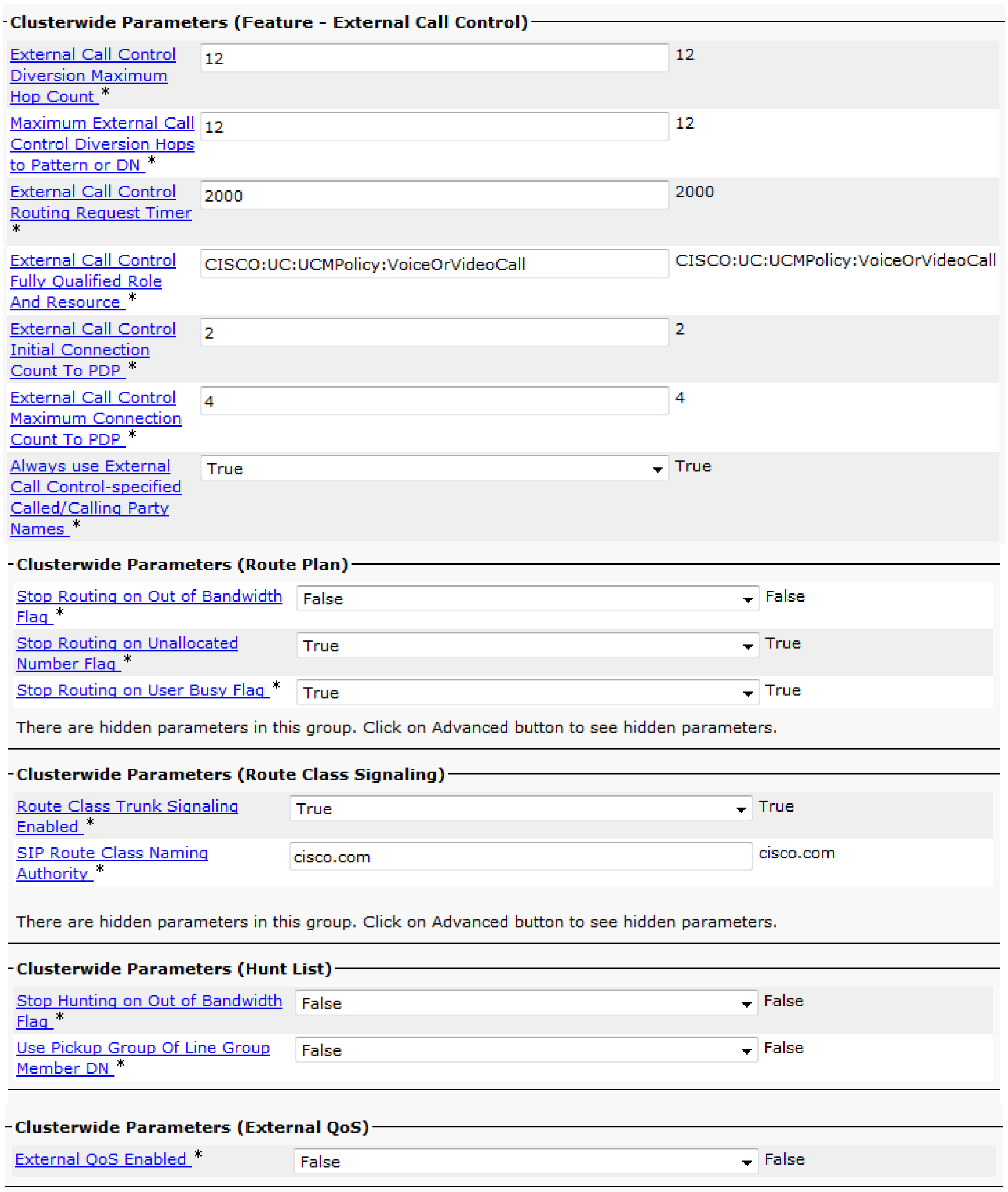


Figure 14: Service Parameters – Cont.

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
Intercluster Capabilities Mismatch Timer *	1000	1000
Silence Suppression *	False	False
Silence Suppression for Gateways *	False	False
Strip G.729 Annex B (Silence Suppression) from Capabilities *	False	False
Enable Source IP Address Verification for Software Media Devices *	True	True

Clusterwide Parameters (System - General)

Always Use Dial Tone Setting *	Default	Default
Restart Cisco CallManager on Initialization Exception *	True	True
Digit Analysis Timer *	6	6
Statistics Enabled *	True	True

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.

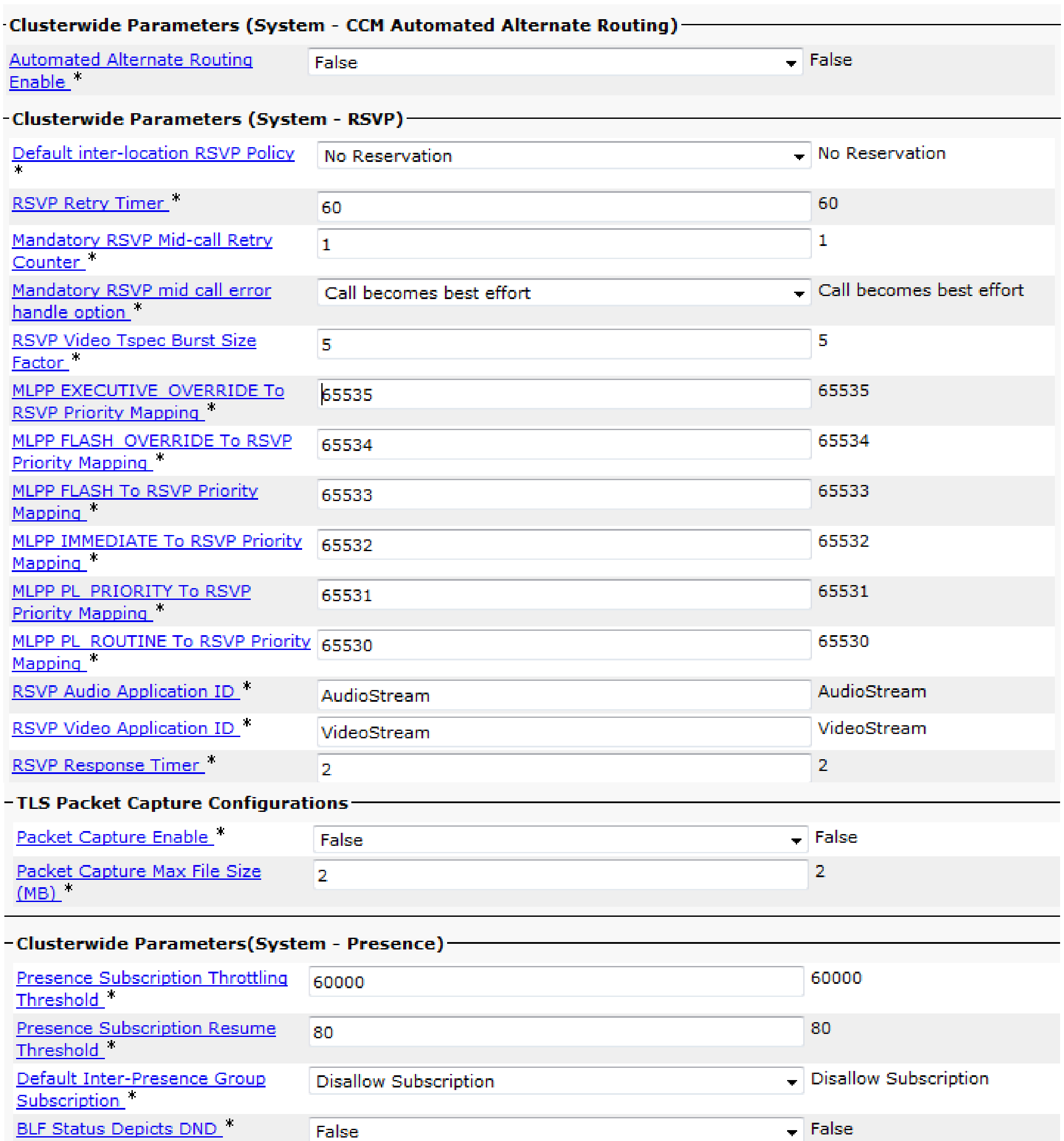


Figure 18: Service Parameters – Cont.

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.

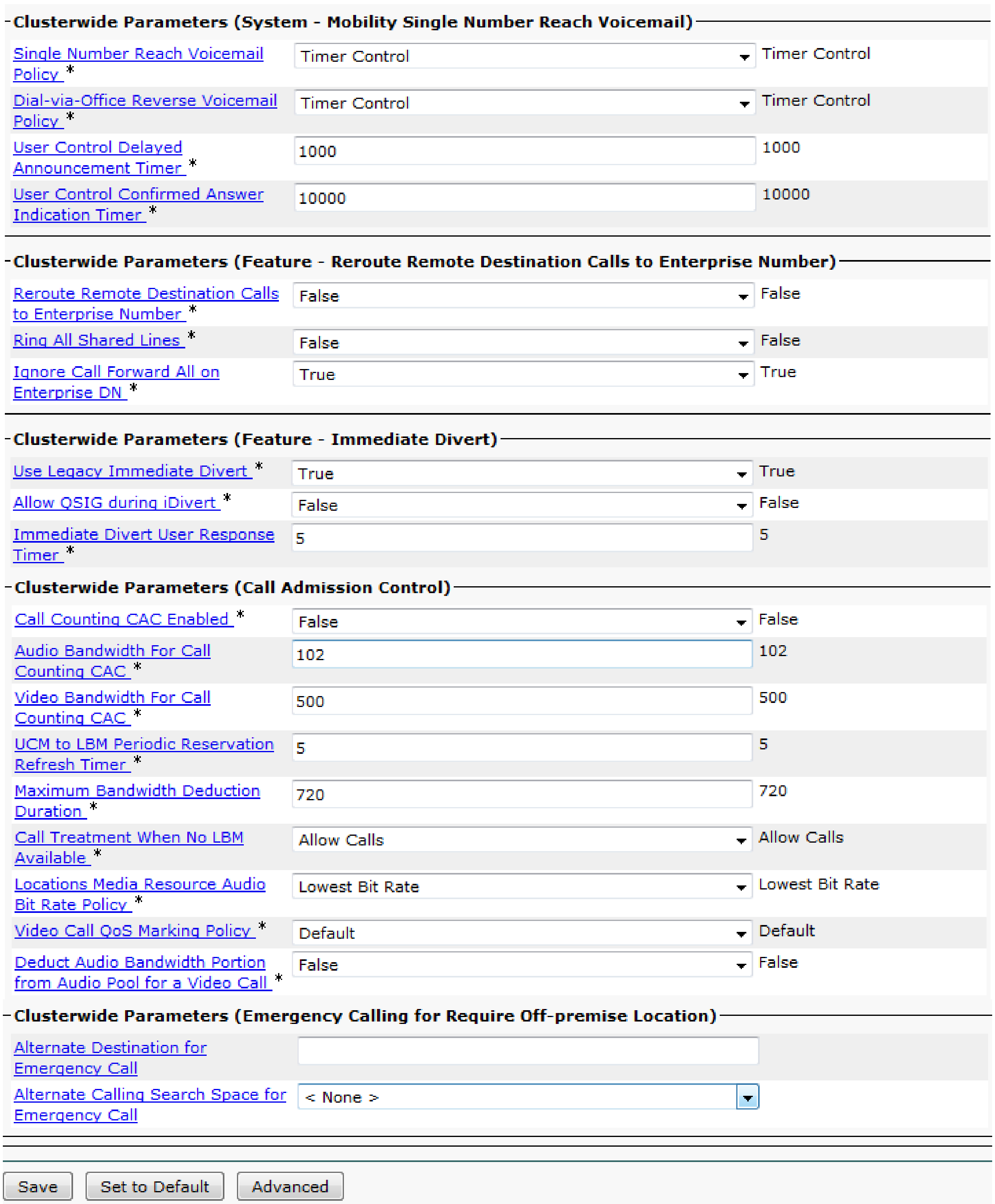


Figure 20: Service Parameters – Cont.

Off-net Calls via Bell Canada SIP Trunk

Off-net calls are served by SIP trunks configured between CUCM and the Bell Canada Network. Calls are routed via the CUBE.

SIP Trunk Security Profile

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

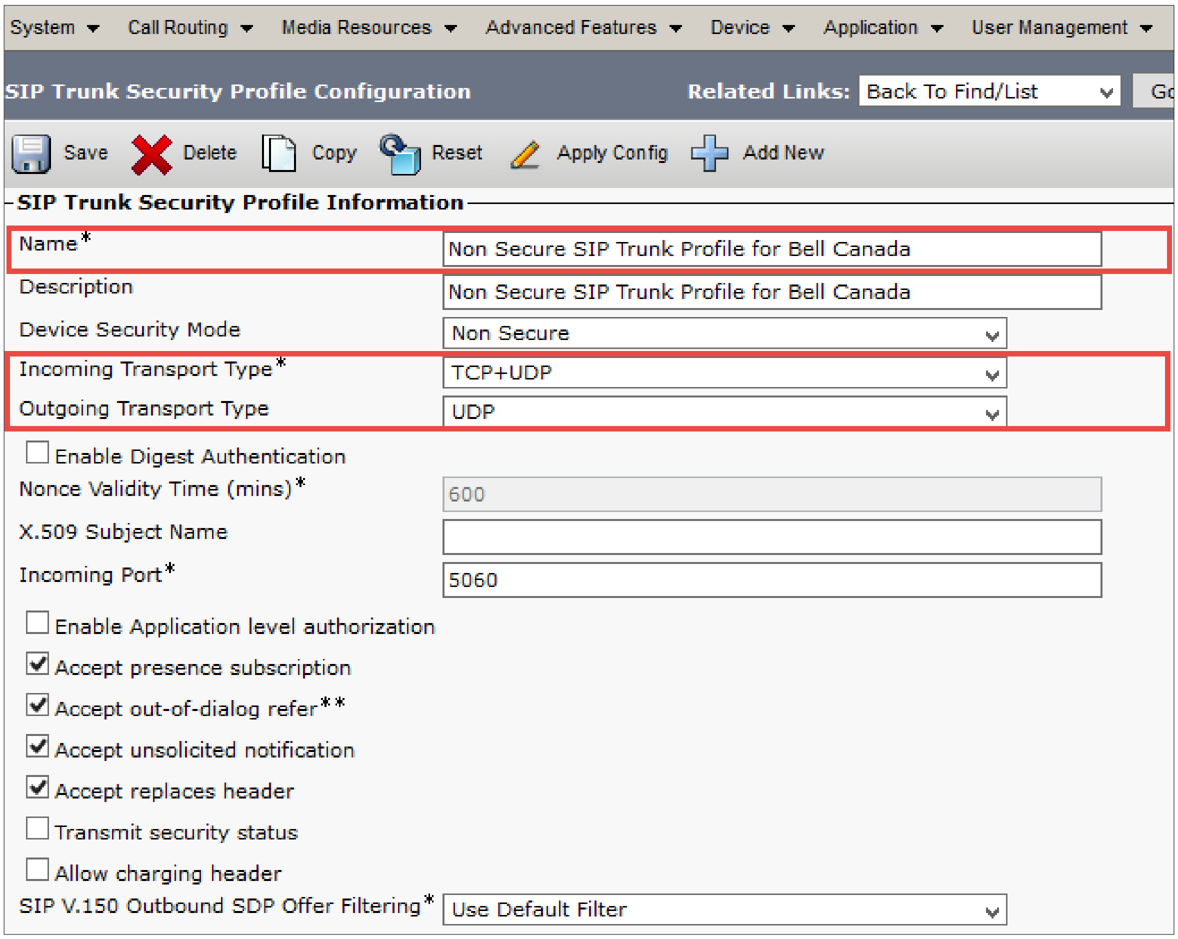


Figure 21: SIP Trunk Security Profile

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to Bell Canada 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

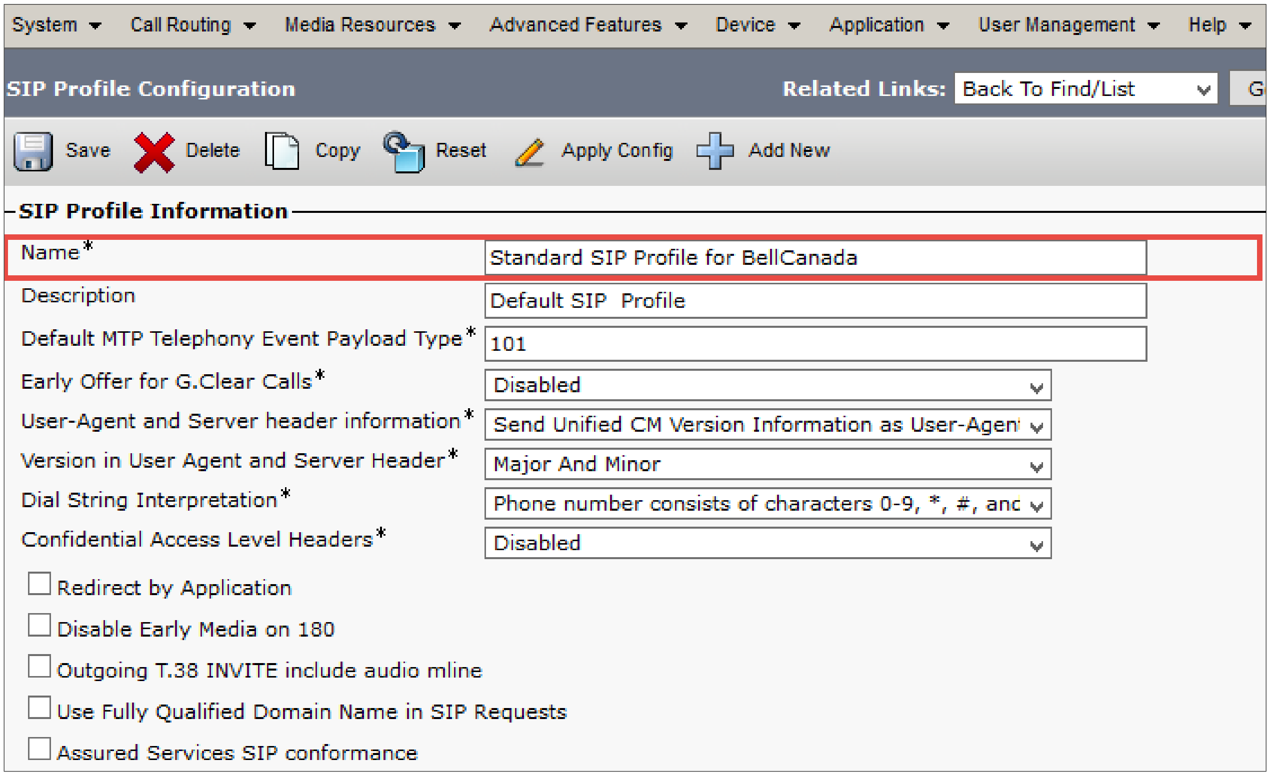


Figure 22: SIP Profile

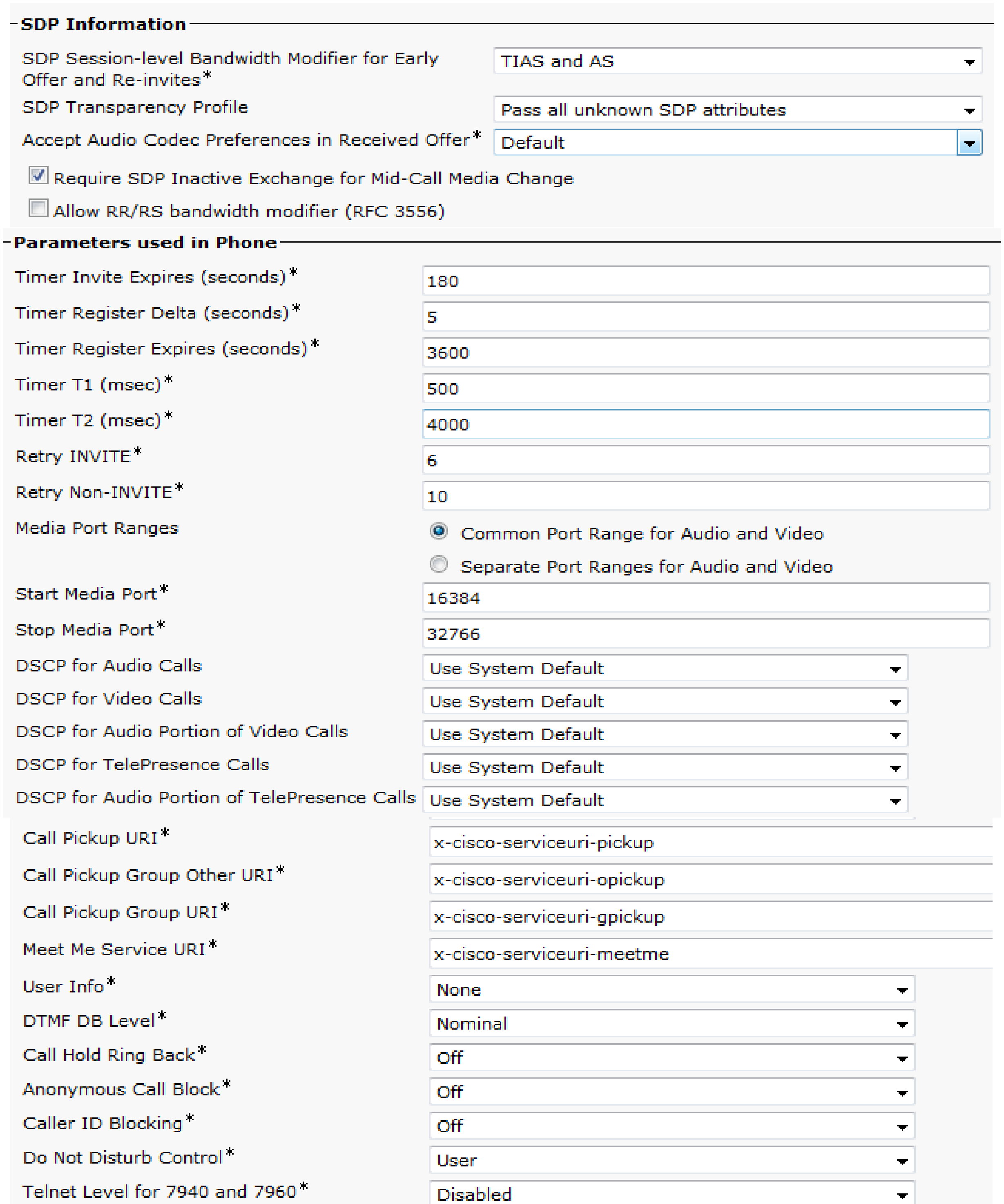


Figure 23: SIP Profile – Cont.

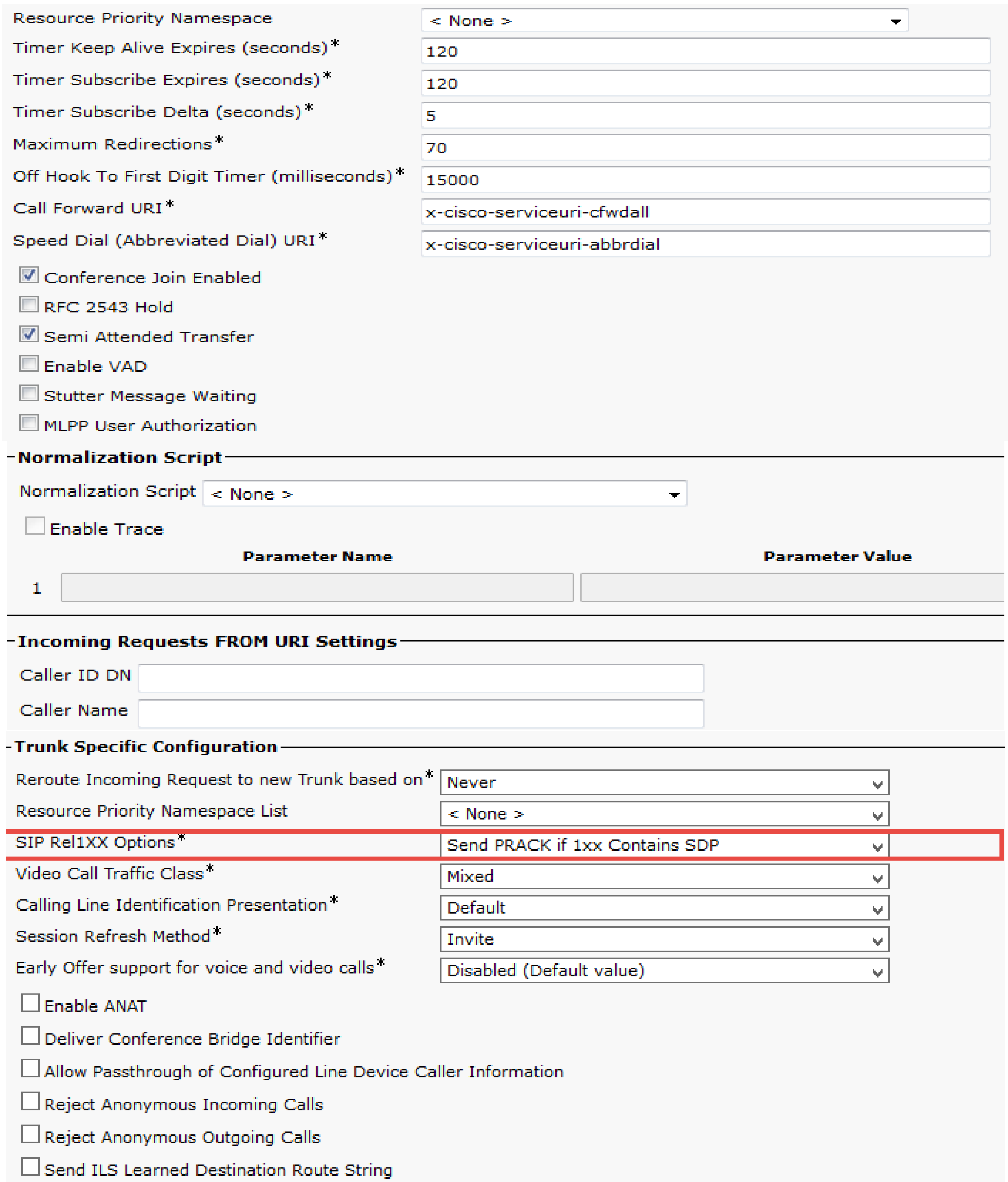


Figure 24: SIP Profile – Cont.

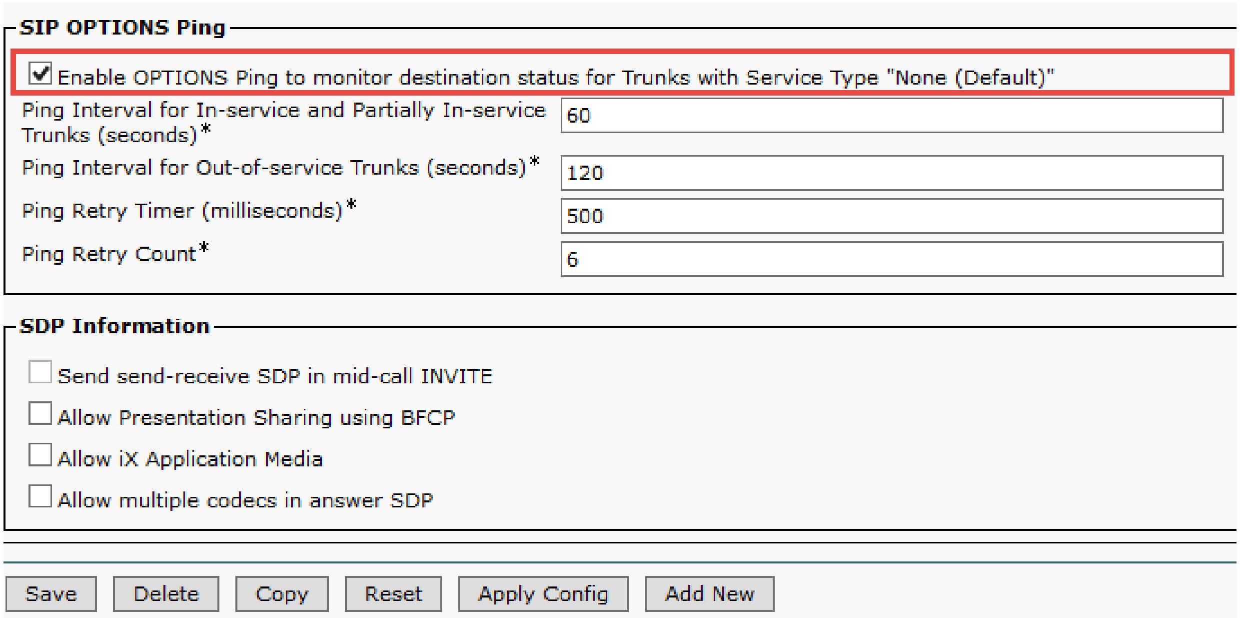


Figure 25: SIP Profile – Cont.

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 Bell Canada

1. Navigate to **Device** > **Trunk**
2. Click **Add New**
3. Follow the same procedure to create SIP trunks to Fax Gateway

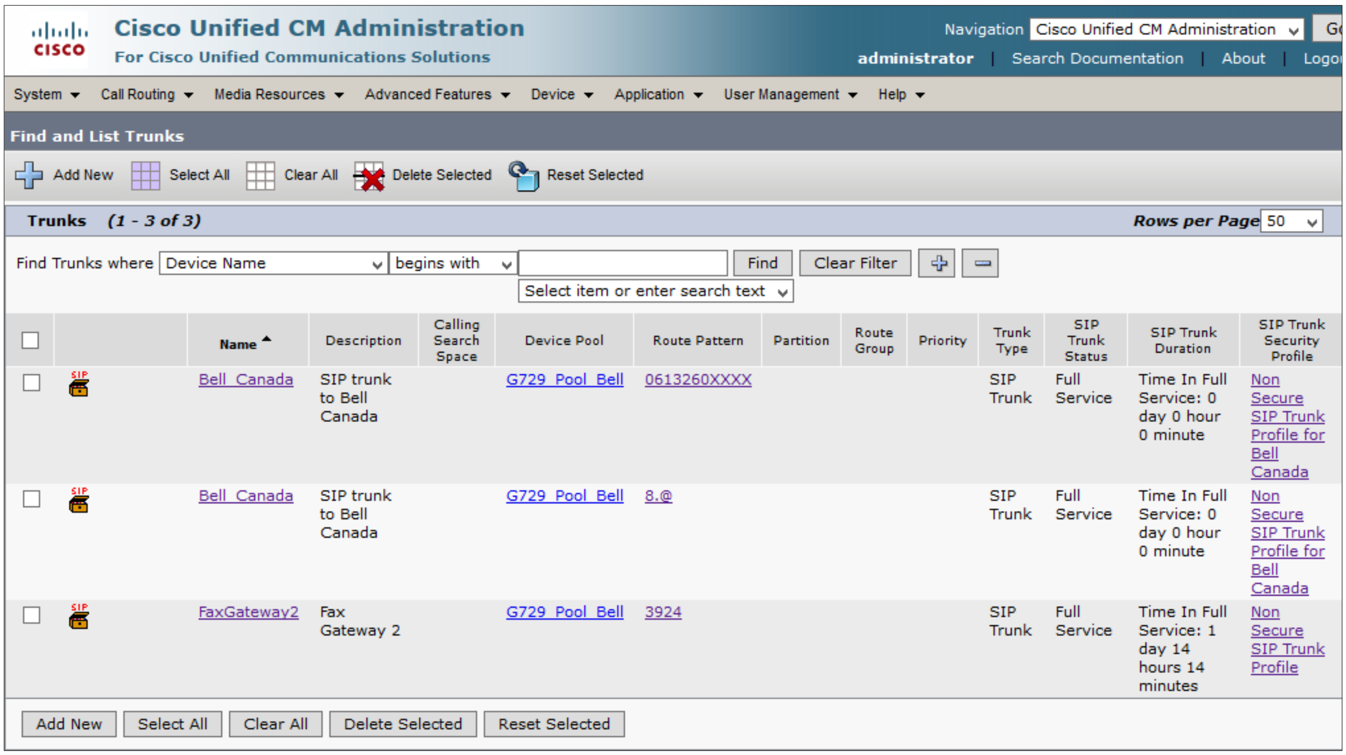
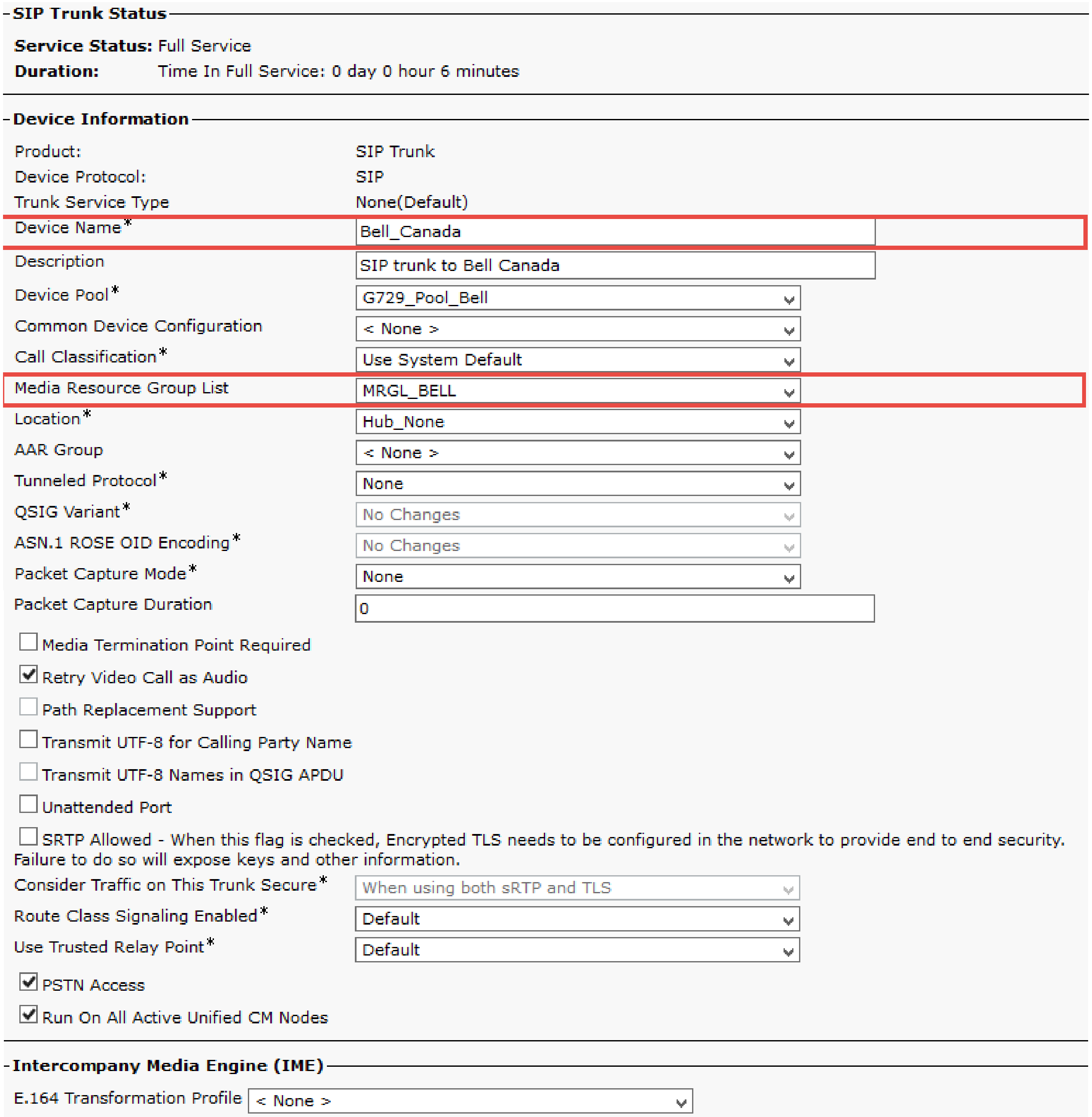


Figure 26: SIP Trunk List



SIP Trunk Status

Service Status: Full Service
Duration: Time In Full Service: 0 day 0 hour 6 minutes

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Bell_Canada
Description	SIP trunk to Bell Canada
Device Pool*	G729_Pool_Bell
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_BELL
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
- [x] Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile	< None >

Figure 27: SIP Trunk to CUBE

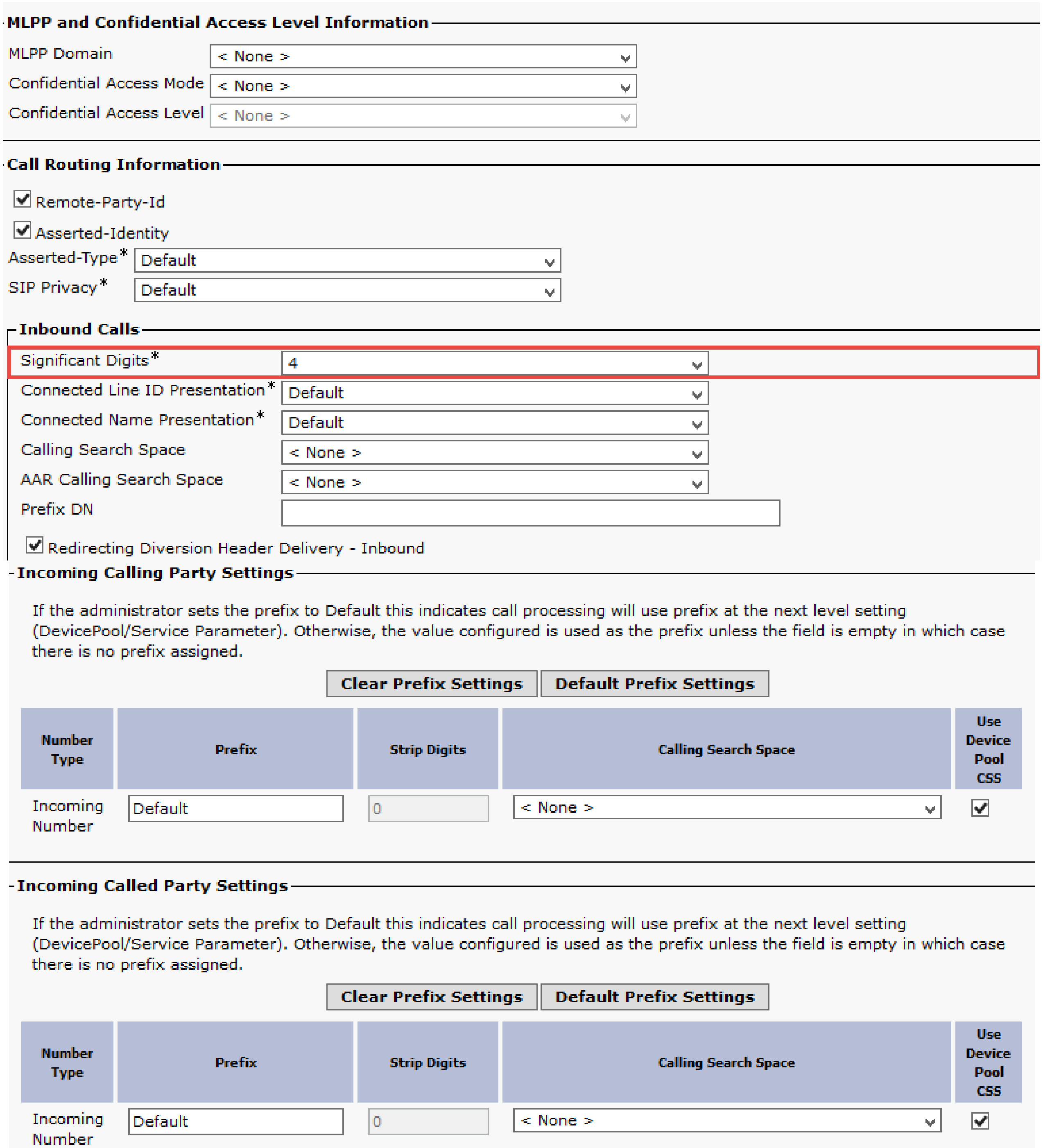


Figure 28: SIP Trunk to CUBE – Cont.

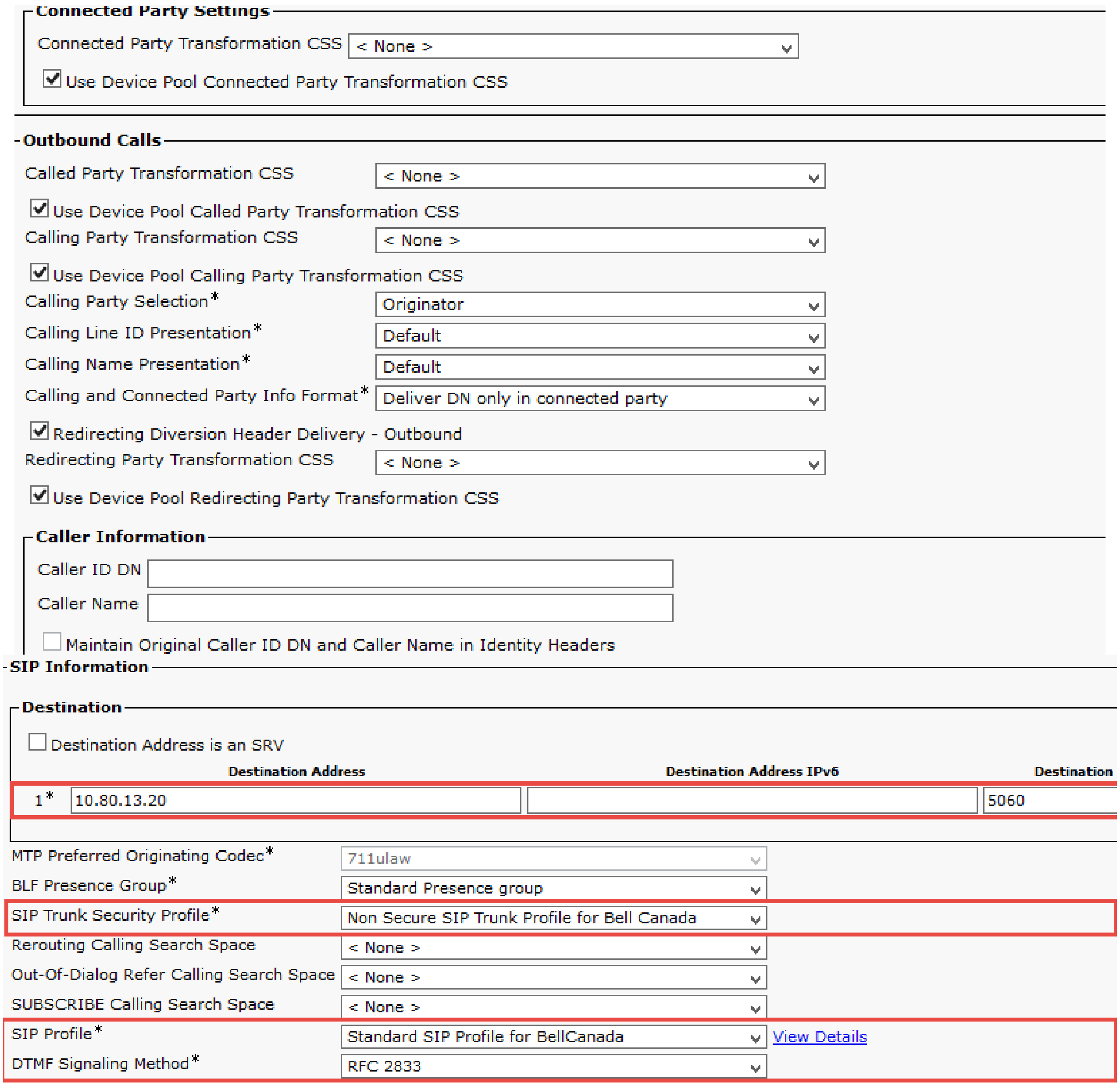


Figure 29: SIP Trunk to CUBE – Cont.

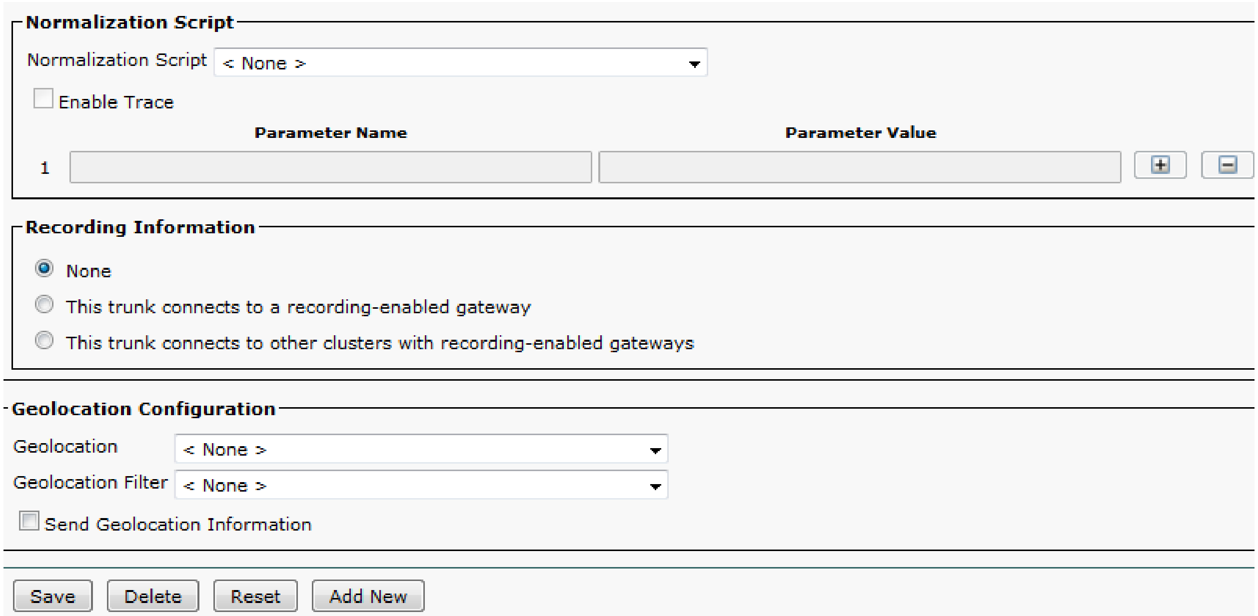


Figure 30: SIP Trunk to CUBE – Cont.

Parameter	Value	Description
Device Name	Bell_Canada	Name for the trunk
Device Pool	G729_Pool_Bell	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_Bell	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	Non Secure SIP Trunk Profile for Bell Canada	SIP Trunk Security Profile configured earlier
SIP Profile	Standard SIP Profile for BellCanada	SIP Profile configured earlier
DTMF Signaling Method	RFC 2833	RFC 2833 is supported for DTMF transport to/from Bell Canada

Reset the trunk after the configuration is completed.

Dial Plan

Route Pattern Configuration

Route patterns are configured as below:

- Cisco IP phones dial 8+10 digits number to access PSTN via CUBE
- “8” is removed before send to CUBE
- 0613260XXXX is used for the PBX to PBX calls that routes via the Bell Canada Network.
- Incoming fax call to 3924 will be sent to Cisco Fax gateway

To Create Route Patterns

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

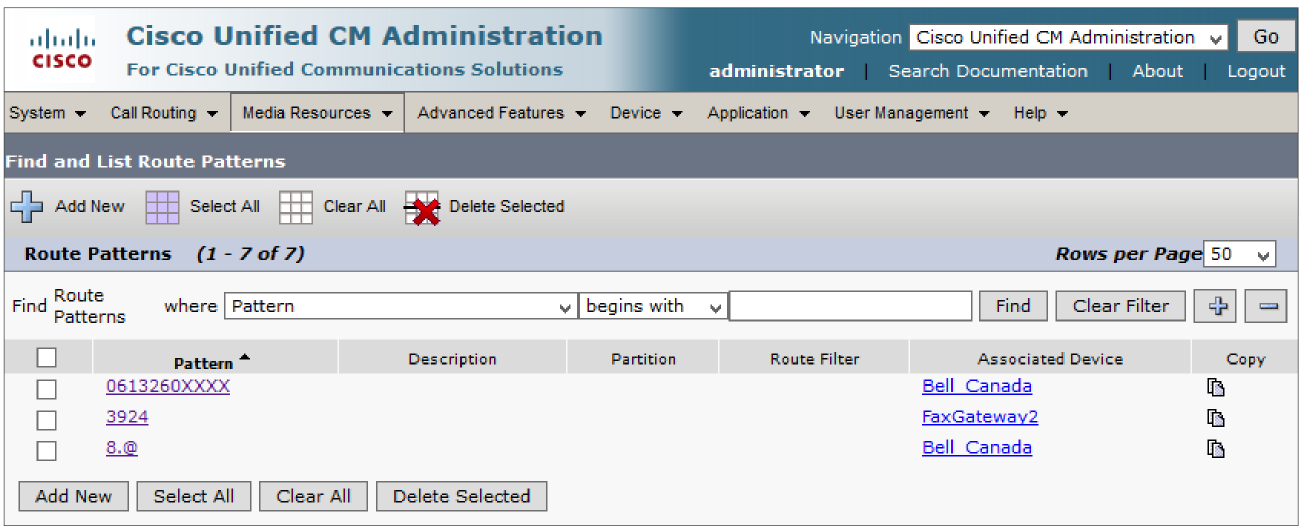


Figure 31: Route Patterns

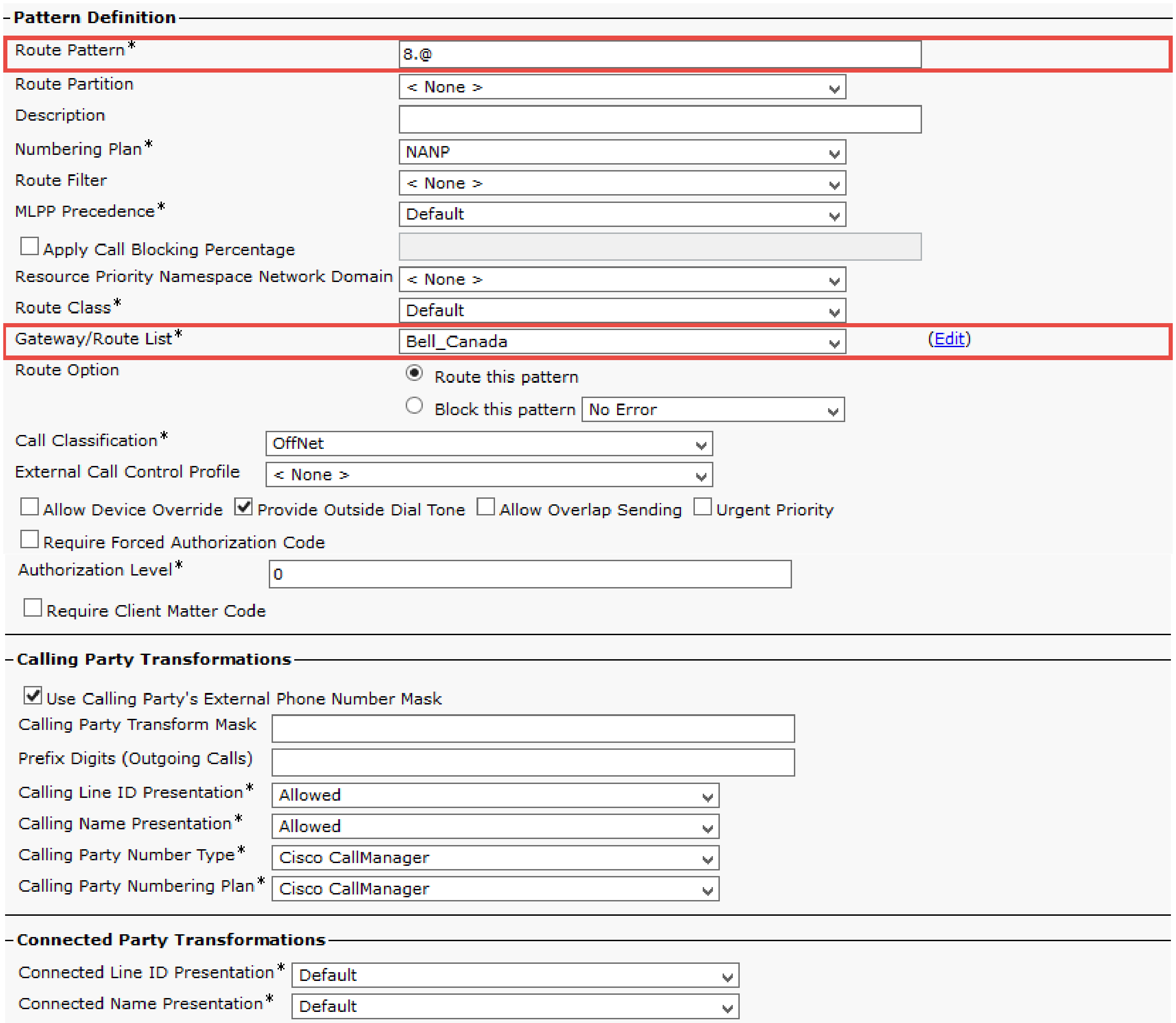


Figure 32: Route Patterns for PSTN

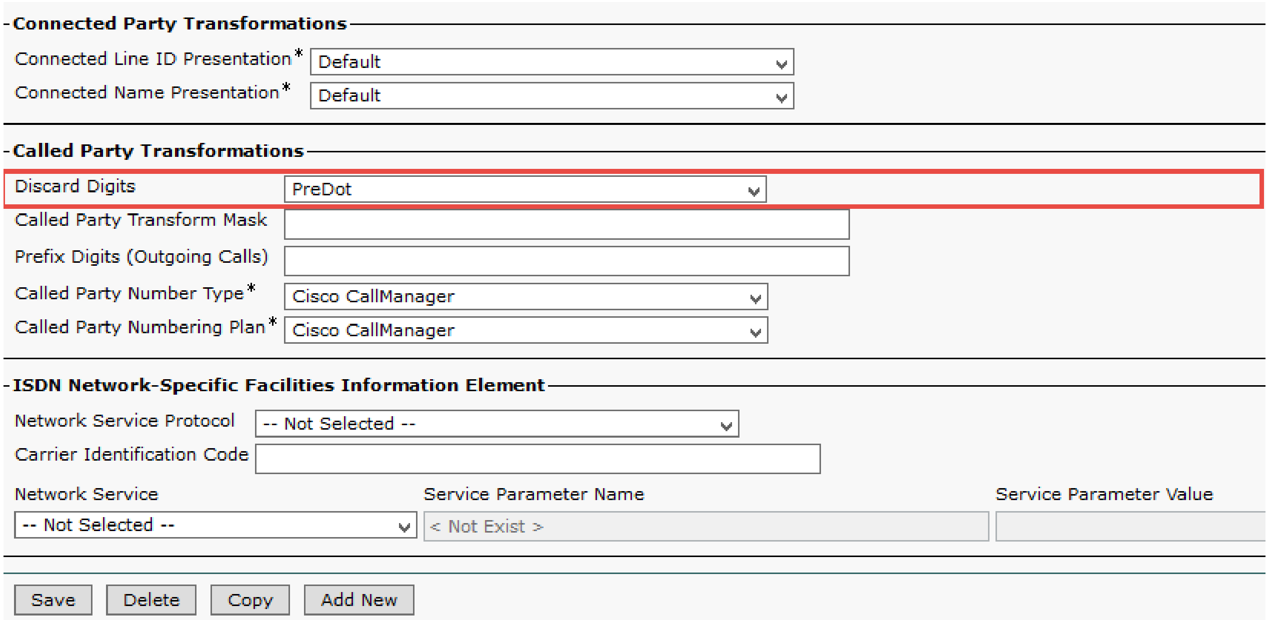


Figure 33: Route Patterns for PSTN – Cont.

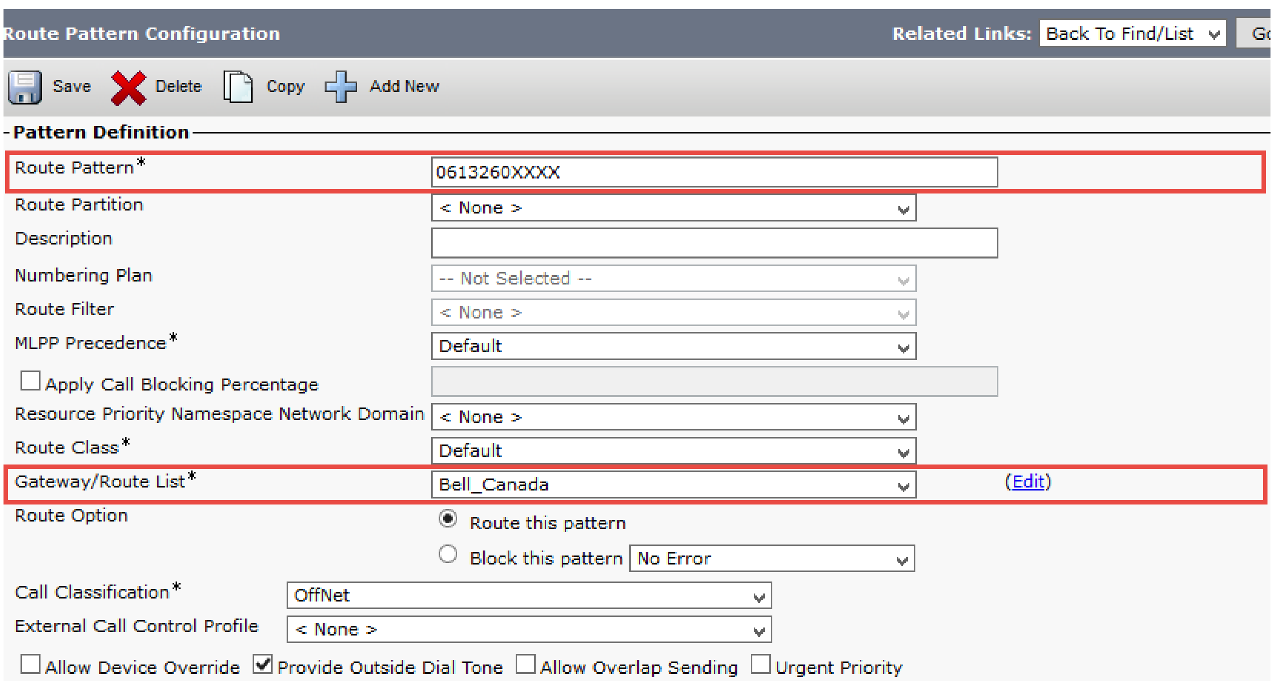


Figure 34: Route Patterns for PBX to PBX Via Bell Canada

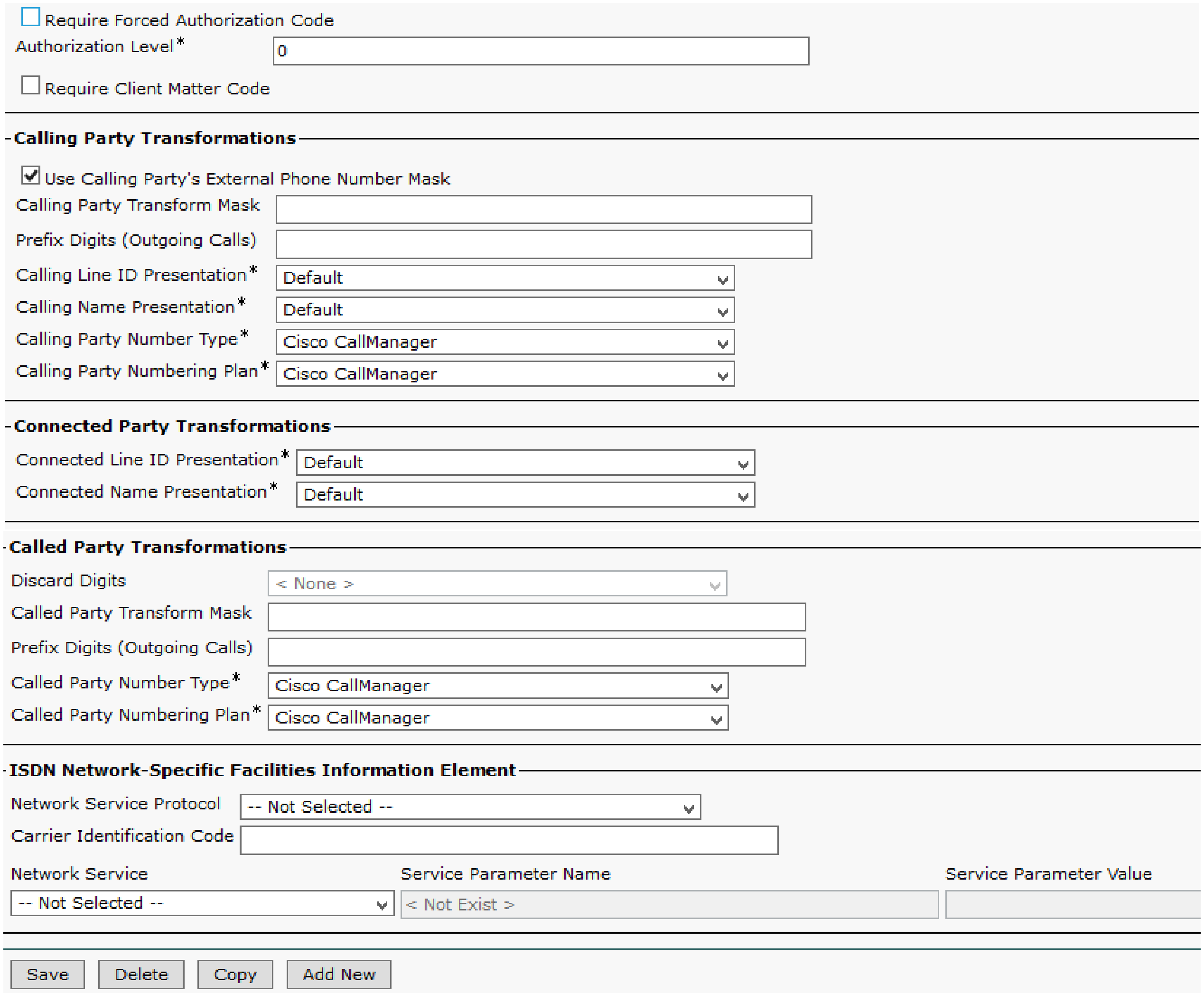


Figure 35: Route Patterns for PBX to PBX Via Bell Canada – Cont.

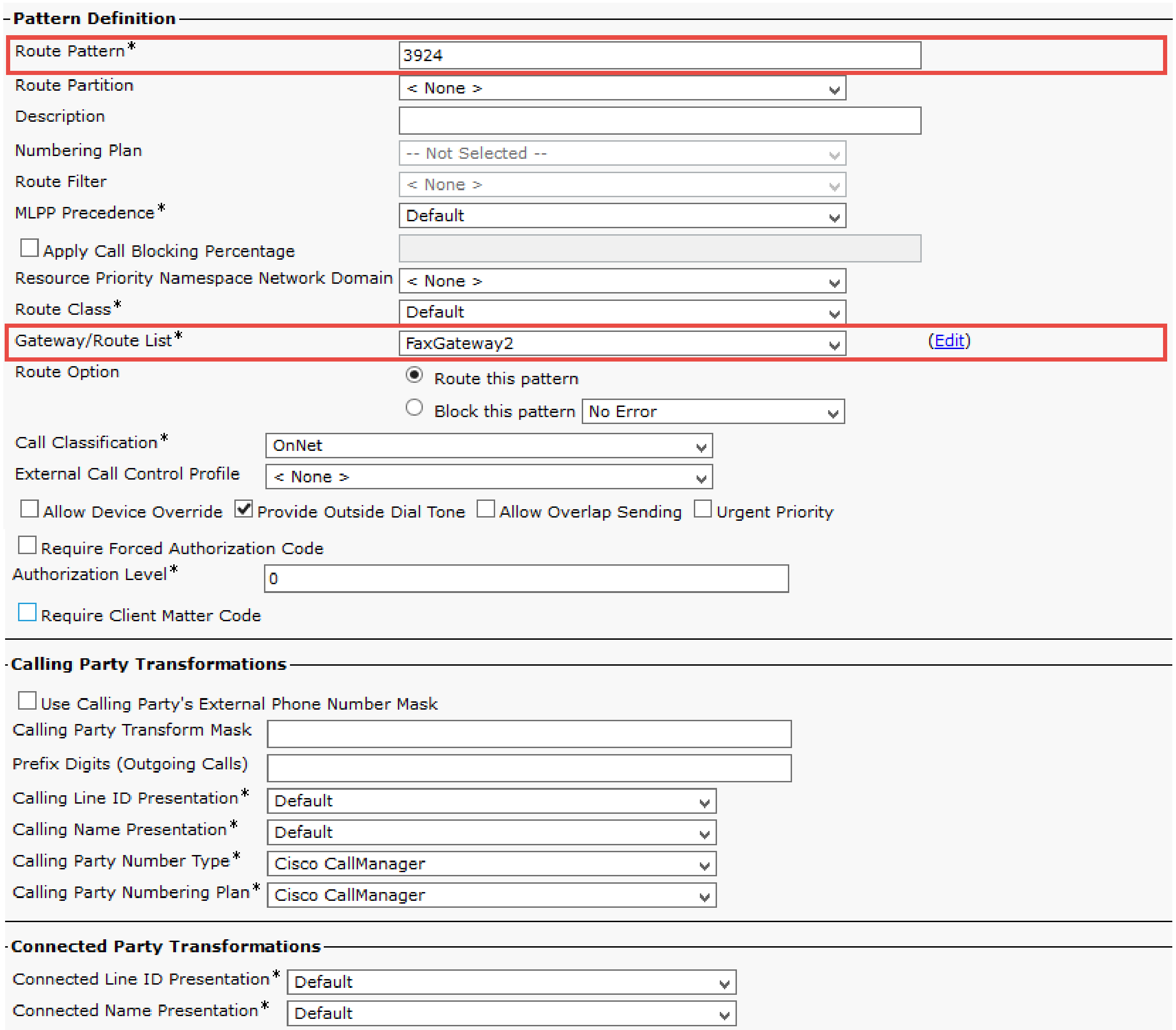


Figure 36: Route Patterns for Fax

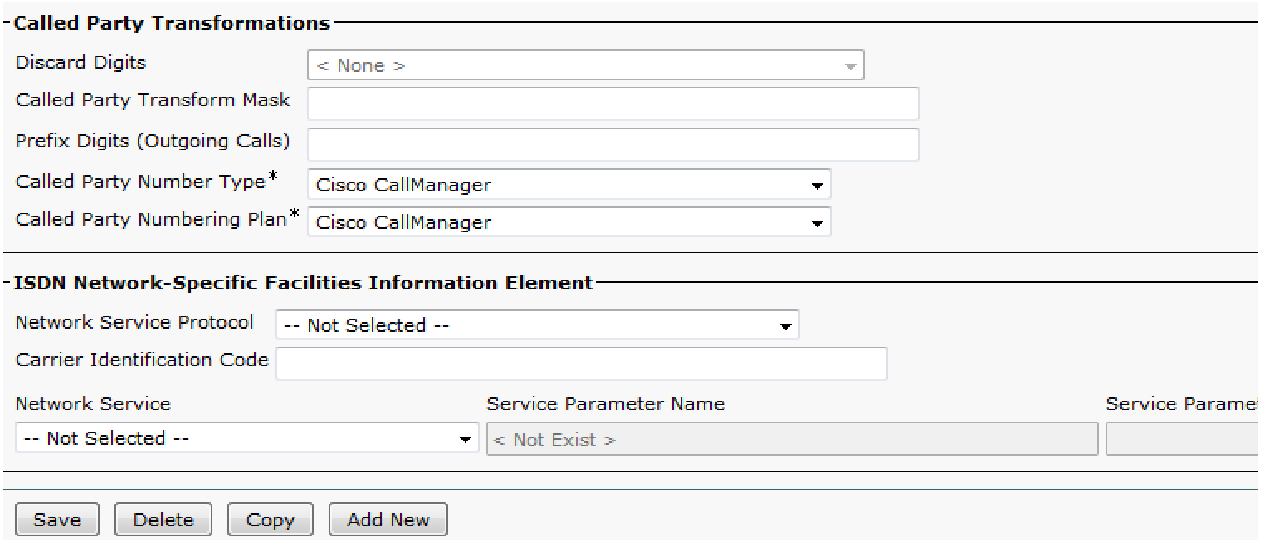


Figure 37: Route Patterns for Fax – Cont.

Setting	Value	Description
Route Pattern	8.@ for Voice call and 3924 for fax call	Specify appropriate Route Pattern
Gateway/Route List	Bell Canada	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	613260XXXX	Specify the Calling Line ID for outgoing call through this Route Pattern
Discard Digits	PreDot for RP 8.@	Specifies how to modify digit before they are sending to Bell Canada

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
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol

Important Information

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESSED OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS. IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.