

## 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

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## 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.

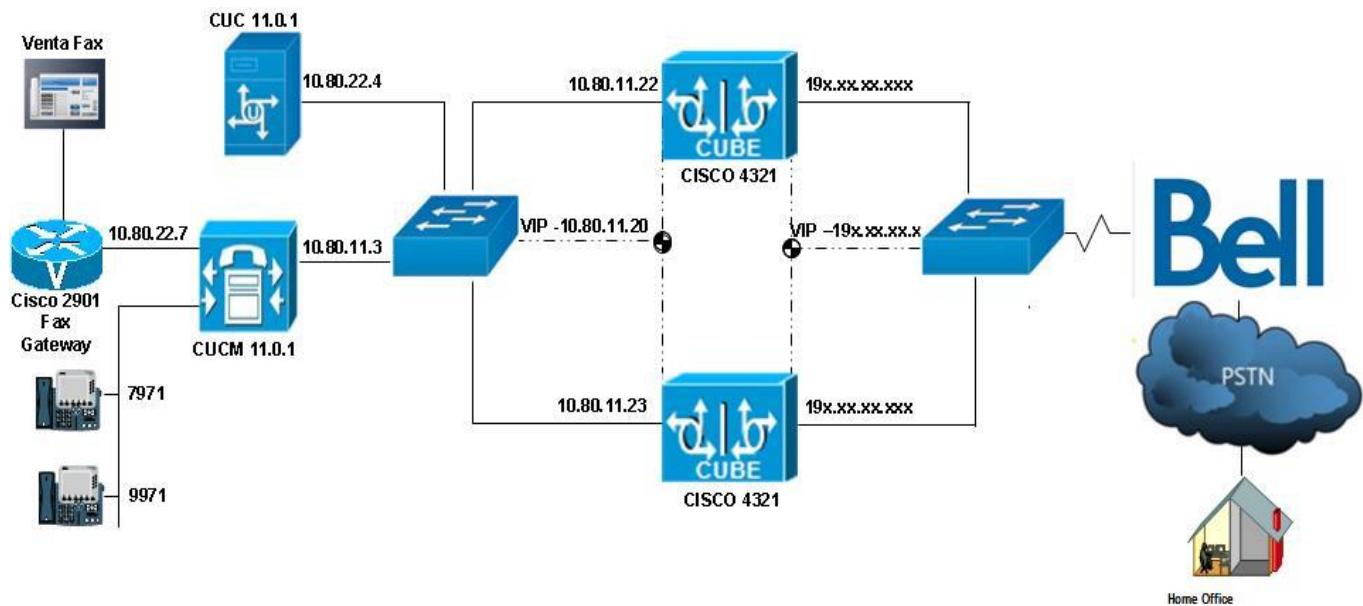
- This application note describes how to configure a Cisco Unified Communications Manager (Cisco UCM) 11.0.1 and Cisco Unified Border Element (CUBE 11.5.0) on ISR 4K [IOS-XE3.17] for connectivity to Bell Canada SIP trunking service. The deployment model covered in this application note is CPE (Cisco 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](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.xxxxxxnetvoice.ca;user=phone>"

request INVITE sip-header P-Asserted-Identity modify "(@.*>" 
"@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"

request INVITE sip-header Diversion modify "reason=unconditional" ""

request INVITE sip-header Diversion modify "(@.*>" 
"@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"

request REINVITE sip-header From modify "(@.*>" 
"@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"

request REINVITE sip-header P-Asserted-Identity modify "(@.*>" 
"@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"

request REINVITE sip-header Diversion modify "(@.*>" 
"@venxxxx.xxx.xxxxxxnetvoice.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.xxxxxxnetvoice.ca;user=phone>"

request REINVITE sip-header P-Asserted-Identity modify "(@.*>" 
"@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"

request REINVITE sip-header Diversion modify "(@.*>" 
"@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"

request REINVITE sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>" 
"<sip:613260\1@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"

response 181 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>" 
"<sip:613260\1@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"

response 180 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>" 
"<sip:613260\1@venxxxx.xxx.xxxxxxnetvoice.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.xxxxxxnetvoice.ca>"  
request INVITE sip-header P-Asserted-Identity modify "(@.*>" "@lxx.xxxxxxnetvoice.ca>"  
request INVITE sip-header Diversion modify "reason=unconditional" ""  
request INVITE sip-header Diversion modify "(@.*>" "@lxx.xxxxxxnetvoice.ca;user=phone>"  
request REINVITE sip-header From modify "(@.*>" "@lxx.xxxxxxnetvoice.ca>"  
request REINVITE sip-header P-Asserted-Identity modify "(@.*>" "@lxx.xxxxxxnetvoice.ca>"  
request REINVITE sip-header Diversion modify "(@.*>" "@.lxx.xxxxxxnetvoice.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.xxxxxxnetvoice.ca>"  
request REINVITE sip-header P-Asserted-Identity modify "(@.*>" "@lxx.xxxxxxnetvoice.ca>"  
request REINVITE sip-header Diversion modify "(@.*>" "@lxx.xxxxxxnetvoice.ca;user=phone>"  
request REINVITE sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>"  
<sip:613260\1@lxx.xxxxxxnetvoice.ca>"  
response 181 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>"  
<sip:613260\1@lxx.xxxxxxnetvoice.ca>"  
response 180 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>"  
<sip:613260\1@lxx.xxxxxxnetvoice.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.xxxxnetvoice.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.xxxxxxnetvoice.ca;user=phone>"  
request INVITE sip-header P-Asserted-Identity modify "(@.*>"  
"@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"  
request INVITE sip-header Diversion modify "reason=unconditional" ""  
request INVITE sip-header Diversion modify "(@.*>"  
"@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"  
request REINVITE sip-header From modify "(@.*>"  
"@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"  
request REINVITE sip-header P-Asserted-Identity modify "(@.*>"  
"@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"  
request REINVITE sip-header Diversion modify "(@.*>"  
"@venxxxx.xxx.xxxxxxnetvoice.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.xxxxxxnetvoice.ca;user=phone>"  
request REINVITE sip-header P-Asserted-Identity modify "(@.*>"  
"@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"  
request REINVITE sip-header Diversion modify "(@.*>"  
"@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"  
request REINVITE sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>"  
"<sip:613260\1@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"  
response 181 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>"  
"<sip:613260\1@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"  
response 180 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>"  
"<sip:613260\1@venxxxx.xxx.xxxxxxnetvoice.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 xxxxxxxxxxxx
  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.xxxxxxnetvoice.ca;user=phone"
request INVITE sip-header P-Asserted-Identity modify "(@.*)" "@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone"
request INVITE sip-header Diversion modify "reason=unconditional" ""
request INVITE sip-header Diversion modify "(@.*)" "@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone"
request REINVITE sip-header From modify "(@.*)" "@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone"
request REINVITE sip-header P-Asserted-Identity modify "(@.*)" "@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone"
request REINVITE sip-header Diversion modify "(@.*)" "@venxxxx.xxx.xxxxxxnetvoice.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.xxxxxxnetvoice.ca;user=phone"
request REINVITE sip-header P-Asserted-Identity modify "(@.*)" "@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone"
request REINVITE sip-header Diversion modify "(@.*)" "@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone"
request REINVITE sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>" "<sip:613260\1@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"
response 181 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>" "<sip:613260\1@venxxxx.xxx.xxxxxxnetvoice.ca;user=phone>"
response 180 sip-header P-Asserted-Identity modify "<sip:(....)@(.*)>" "<sip:613260\1@venxxxx.xxx.xxxxxxnetvoice.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 xxxxxxxxxxxx
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

**Cisco Unified CM Administration** For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration administrator | Search Documentation | About |

System Call Routing Media Resources Advanced Features Device Application User Management Help

Service Parameter Configuration Related Links: Parameters for All Servers G

Save Set to Default Advanced

-Select Server and Service-

Server\* clus21sub1--CUCM Voice/Video (Active)  
Service\* Cisco CallManager (Active)

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

-Cisco CallManager (Active) Parameters on server clus21sub1--CUCM Voice/Video (Active)-

Parameter Name Parameter Value Suggested Value

**Call Throttling**

<u>Code Yellow Entry Latency</u> *	20	20
<u>Code Yellow Exit Latency Calculation</u> *	40	40
<u>Code Yellow Duration</u> *	5	5
<u>Max Events Allowed</u> *	2000	2000
<u>System Throttle Sample Size</u> *	10	10

**Memory Throttling**

<u>Enable Memory Throttling</u> *	True	True
-----------------------------------	------	------

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

**System**

<u>CDR Enabled Flag</u> *	False	False
<u>CDR Log Calls with Zero Duration Flag</u> *	False	False
<u>Digit Analysis Complexity</u> *	StandardAnalysis	StandardAnalysis
<u>Database Debounce Timer</u> *	0	0
<u>Maximum PhoneFallback Queue Depth</u> *	10	10
<u>Maximum Number of Registered Devices</u> *	5000	5000
<u>System Initialization Timer</u> *	60	60

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

Figure 2: Service Parameters

<b>-SDL Trace-</b>		
<u>SDL Trace Data Flags</u> *	0x00000111	0x00000111
<u>SDL Trace Flush Immediately</u> *	False	▼ False
<u>SDL Trace Data Size</u> *	0	0
<u>SDL Trace Flag</u> *	True	▼ True
<u>SDL TraceType Flags</u> *	0x8000EB15	0x8000EB15
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		
<b>-Clusterwide Parameters (Device - General)-</b>		
<u>Call Diagnostics Enabled</u> *	Disabled	▼ Disabled
<u>Show Line Group Member DN in finalCalledPartyNumber CDR Field</u> *	False	▼ False
<u>Show Line Group Member Non Masked DN in finalCalledPartyNumber CDR Field</u> *	False	▼ False
<u>CTI New Call Accept Timer</u> *	4	4
<u>CTI Generate Digits Interval</u> *	250	250
<u>CTI Dial Digits Interval</u> *	250	250
<u>CTI Await Further Digits</u> *	False	▼ False
<u>CTI Use Wildcard Pattern as calledPartyDN</u> *	False	▼ False
<u>CTI Report Ringback on SIP 183 with SDP</u> *	True	▼ True
<u>Retain Media on Disconnect with PI for Active Call</u> *	False	▼ False
<u>Station and Backup Server KeepAlive Interval</u> *	60	60
<u>Station KeepAlive Interval</u> *	30	30
<u>Status Enquiry Poll Flag</u> *	False	▼ False
<u>Strip # Sign from Called Party Number</u> *	True	▼ True
<u>Session Handoff Alerting Timer</u> *	10	10
<u>T301 Timer</u> *	180000	180000
<u>T302 Timer</u> *	15000	15000
<u>T303 Timer</u> *	4000	4000
<u>T304 Timer</u> *	30000	30000
<u>T305 Timer</u> *	30000	30000
<u>T306 Timer</u> *	30000	30000

**Figure 3: Service Parameters – Cont.**



<u>T308 Timer</u> *	4000	4000
<u>T309 Timer</u> *	90000	90000
<u>T310 Timer</u> *	60000	60000
<u>T313 Timer</u> *	4000	4000
<u>T316 Timer</u> *	120000	120000
<u>T317 Timer</u> *	100000	100000
<u>T321 Timer</u> *	30000	30000
<u>T322 Timer</u> *	4000	4000
<u>Tone on Hold Timer</u> *	10	10
<u>Unknown Caller ID Flag</u> *	True	True
<u>Call Classification</u> *	OffNet	OffNet
<u>Always Display Original Dialed Number</u> *	False	False
<u>Name Display for Original Dialed Number When Translated</u> *	Show the Display Name for Original Dialed Number even if Translated	Show the Display Name for Original Dialed Number even if Translated
<u>Always Use PIs With Original Dialed Number</u> *	False	False
<u>Fail Call If Trusted Relay Point Allocation Fails</u> *	True	True
<u>Display Calling/Called ID When PI is Not Available</u> *	False	False
<u>Enable Transit Counter Processing on QSIG Trunks</u> *	False	False
<u>Egress FacilityIE Count</u> *	6	6
<b>- Clusterwide Parameters (Device - Phone)</b>		
<u>Always Use Prime Line</u> *	False	False
<u>Always Use Prime Line for Voice Message</u> *	False	False
<u>Builtin Bridge Enable</u> *	Off	Off
<u>Device Mobility Mode</u> *	Off	Off
<u>Display Device Mobility Location During Phone Registration</u> *	True	True
<u>Auto Answer Timer</u> *	1	1
<u>Extension Display on Cisco IP Phone Model 7910</u> *	False	False
<u>Alternate Idle Phone Auto-Answer Behavior Enabled</u> *	False	False
<u>Hold Type</u> *	False	False
<u>Line State Update Enabled</u> *	True	True
<u>Off-hook to First Digit Timer</u> *	15000	15000

**Figure 4: Service Parameters – Cont.**



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

**Figure 5: Service Parameters – Cont.**

**- Clusterwide Parameters (Device - PRI and MGCP Gateway)**

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

**Figure 6: Service Parameters – Cont.**



<u>Optimize MGCP Registration</u> *	True	True
<u>Suppress Out-of-Channels Alarms</u> *	True	True
<u>I-Frame Timer</u> *	2000	2000
<u>User-to-User IE Status</u> *	False	False
<u>Convert European Progress Message to Alerting</u> *	False	False
<u>Enable DMS PRI Notify Message from User to Network</u> *	True	True
<u>Audit OOS Channels Interval</u> *	10	10
 <u>Digital and Analog Ports Enabled</u> *	True	True
<b>- Clusterwide Parameters (Device - H323)</b>		
<u>Accept Unknown TCP Connection</u> *	False	False
<u>BRQ Enabled</u> *	False	False
<u>Call Present Disconnect Flag</u> *	False	False
<u>Check Progress Indicator Before Establishing Media</u> *	False	False
<u>H225 Block Setup Destination</u> *	False	False
<u>H225 DB Retry Timer</u> *	0	0
<u>H225 Device Connect Timer</u> *	0	0
<u>H225 DTMF Duration</u> *	100	100
<u>H225 TspReq Retry</u> *	2	2
<u>H225 Intercluster Call Throttle Timer</u> *	30	30
<u>H225 T301 Timer</u> *	180000	180000
<u>H225 T302 Timer</u> *	15000	15000
<u>H225 T303 Timer</u> *	4000	4000
<u>H225 T304 Timer</u> *	30000	30000
<u>H225 T305 Timer</u> *	30000	30000
<u>H225 T310 Timer</u> *	60000	60000
<u>H225 TCP Timer</u> *	5	5
<u>H245 TCS Timeout</u> *	10	10
<u>H323 Calling Party Number Screening Indicator</u> *	Calling number screened and passed	Calling number screened and passed
<u>Apply External Phone Number Mask for H.323 Calls</u> *	False	False
<u>Tone on Connect</u> *	False	False
<u>Wait Time for SDP with SR/RO Mode</u> *	3	3

**Figure 7: Service Parameters – Cont.**



<u>RAS ARQ Timer</u> *	3	3
<u>RAS BRQ Timer</u> *	3	3
<u>RAS DRQ Timer</u> *	3	3
<u>RAS RRQ Timer</u> *	3	3
<u>Ras URO Timer</u> *	3	3
<u>Retry Count for ARQ</u> *	2	2
<u>Retry Count for BRQ</u> *	2	2
<u>Retry Count for DRQ</u> *	2	2
<u>Retry Count for RRQ</u> *	2	2
<u>Retry Count for URO</u> *	1	1
<u>Send Product ID and Version ID</u> *	False	▼ False
<u>Send Unified CM Version as Version ID in H225Setup</u> *	False	▼ False
<u>Send Progress Timer</u> *	3000	3000
<u>Send H225 User Info Message</u> *	User Info for Call Progress Tone	▼ User Info for Call Progress Tone
<u>Status Enquiry Poll Timer</u> *	10000	10000
<u>Device Name of GK-controlled Trunk That Will Use Port 1720</u> *	None	None
<u>Host Name/IP Address of GK That Will Use RAS UDP Port 1719</u> *	None	None
<u>Fail Call If MTP Allocation Fails</u> *	False	▼ False
<u>Overlap Receiving Flag for H323</u> *	False	▼ False
 <u>Allocate Transcoder for H.323 on Early Offer SIP Trunk for Calls with Early Media</u> *	False	▼ False
<hr/> <b>- Clusterwide Parameters (Device - SIP)</b>		
<u>SIP Interoperability Enabled</u> *	True	▼ True
<u>Retry Count for SIP Bye</u> *	10	10
<u>Retry Count for SIP Cancel</u> *	10	10
<u>Retry Count for SIP Invite</u> *	6	6
<u>Retry Count for SIP PRACK</u> *	6	6
<u>Retry Count for SIP Rel1XX</u> *	10	10
<u>Retry Count for SIP Publish</u> *	6	6
<u>Retry Count for SIP Response</u> *	6	6
<u>SIP Connect Timer</u> *	500	500
<u>SIP Disconnect Timer</u> *	500	500

**Figure 8: Service Parameters – Cont.**

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

**Figure 9: Service Parameters – Cont.**

<b>- Clusterwide Parameters (Feature - General)</b>	
<u>Call Park Display Timer</u> *	10
<u>Caller ID Display Priority Enabled</u> *	True
<u>Call Park Reversion Timer</u> *	60
<u>Park Monitoring Reversion Timer</u> *	60
<u>Park Monitoring Periodic Reversion Timer</u> *	30
<u>Park Monitoring Forward No Retrieve Timer</u> *	300
<u>Preserve globalCallId for Parked Calls</u> *	True
<u>Maximum Call Duration Timer</u> *	720
<u>Maximum Hold Duration Timer</u> *	360
<u>Party Entrance Tone</u> *	True
<u>Message Waiting Lamp Policy</u> *	Primary Line - Light and Prompt
<u>Audible Message Waiting Indication Policy</u> *	OFF
<u>Message Waiting Indicator Inbound Calling Search Space</u>	< None >
<u>Multiple Tenant MWI Modes</u> *	False
<u>MWI Non Message Center Signaling Call Duration</u> *	0
<u>Message Waiting Indicator APDU Digit Translation CSS</u>	< None >
<u>Block OffNet To OffNet Transfer</u> *	False
<u>Use Original Call Classification for Transferred Calls</u> *	False
<u>Use Restriction attribute of ID/Name Presentation of Transferring Party</u> *	True
<u>Local route group for redirected calls</u> *	Local route group of calling party
<u>Block Unencrypted Calls</u> *	False
<b>- Clusterwide Parameters (Feature - Conference)</b>	
<u>Suppress MOH to Conference Bridge</u> *	True
<u>Drop Ad Hoc Conference</u> *	Never
<u>Maximum Ad Hoc Conference</u> *	4
<u>Maximum MeetMe Conference Unicast</u> *	4
<u>Advanced Ad Hoc Conference Enabled</u> *	False
<u>Choose Encrypted Audio Conference Instead Of Video Conference</u> *	True

**Figure 10: Service Parameters – Cont.**



Minimum Video Capable Participants To Allocate Video Conference \*

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.**

<b>-Clusterwide Parameters (Feature - Call Pickup)-</b>	
<u>Auto Call Pickup Enabled</u> *	False
<u>Call Pickup Locating Timer</u> *	1
<u>Call Pickup No Answer Timer</u> *	12
<b>-Clusterwide Parameters (Feature - Refer)-</b>	
<u>Validate Refer-to URI</u> *	Validate Except for Anonymous Users
	▼ Validate Except for Anonymous Users
<b>-Clusterwide Parameters (Feature - Replaces)-</b>	
<u>Block OffNet To OffNet Replaces</u> *	False
<b>-Clusterwide Parameters (Feature - Redirection [3xx])-</b>	
<u>Redirection_Ring_No_Answer</u>	24
<u>Reversion Timer</u> *	
<u>Maximum Redirection Count</u> *	70
<b>-Clusterwide Parameters (Feature - Multilevel Precedence and Preemption)-</b>	
<u>Locations-based MLPP Enable</u> *	False
<u>Executive Override Call Preemptable</u> *	False
<u>Location-based Maximum Bandwidth Enforcement Level for MLPP Calls</u> *	Lenient
<u>Non-Preemption Pattern CSS</u>	< None >
<u>MLPP Exception Level</u> *	Executive Override
<b>-Clusterwide Parameters (Feature - Path Replacement)-</b>	
<u>Path Replacement Enabled</u> *	False
<u>Path Replacement on Tromboned Calls</u> *	True
<u>Start Path Replacement Minimum Delay Time</u> *	0
<u>Start Path Replacement Maximum Delay Time</u> *	0
<u>Path Replacement T1 Timer</u> *	30
<u>Path Replacement T2 Timer</u> *	15
<u>Path Replacement PINX ID</u>	
<u>Path Replacement Calling Search Space</u>	< None >

**Figure 12: Service Parameters – Cont.**

<b>- Clusterwide Parameters (Feature - Call Back)</b>	
<u>Call Back Enabled Flag</u> *	True
<u>Call Back Notification Audio File Name</u> *	CallBack.raw
<u>Connection Proposal Type</u> *	Connection Retention
<u>Connection Response Type</u> *	Default to Connection Retention
<u>Call Back Request Protection T1 Timer</u> *	10
<u>Call Back Recall T3 Timer</u> *	20
<u>Call Back Calling Search Space</u>	< None >
<u>No Path Reservation</u> *	True
<u>Set Private Numbering Plan for Call Back</u> *	False
<u>Set Type of Number for Call Back</u> *	Level1RegionalNumber
<b>- Clusterwide Parameters (Feature - Call Recording)</b>	
<u>Play Recording Notification Tone To Observed Target</u> *	False
<u>Play Recording Notification Tone To Observed Connected Parties</u> *	False
<b>- Clusterwide Parameters (Feature - Monitoring)</b>	
<u>Play Monitoring Notification Tone To Observed Target</u> *	False
<u>Play Monitoring Notification Tone To Observed Connected Parties</u> *	False
<b>- Clusterwide Parameters (Feature - Join Across Lines And Single Button Barge Feature Set)</b>	
<u>Join Across Lines Policy</u> *	Off
<u>Single Button Barge/CBarge Policy</u> *	Off
<u>Allow Barging When Ringing</u> *	False
<b>- Clusterwide Parameters (Feature - Secure Tone)</b>	
<u>Play Tone to Indicate Secure/Non-Secure Call Status</u> *	False

**Figure 13: Service Parameters – Cont.**



- Clusterwide Parameters (Feature - External Call Control)

External Call Control	12	12
Diversion Maximum Hop Count *		
Maximum External Call Control Diversion Hops to Pattern or DN *	12	12
External Call Control Routing Request Timer *	2000	2000
External Call Control Fully Qualified Role And Resource *	CISCO:UC:UCMPolicy:VoiceOrVideoCall	CISCO:UC:UCMPolicy:VoiceOrVideoCall
External Call Control Initial Connection Count To PDP *	2	2
External Call Control Maximum Connection Count To PDP *	4	4
Always use External Call Control-specified Called/Calling Party Names *	True	True

- Clusterwide Parameters (Route Plan)

Stop Routing on Out of Bandwidth Flag *	False	False
Stop Routing on Unallocated Number Flag *	True	True
Stop Routing on User Busy Flag *	True	True

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

- Clusterwide Parameters (Route Class Signaling)

Route Class Trunk Signaling Enabled *	True	True
SIP Route Class Naming Authority *	cisco.com	cisco.com

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

- Clusterwide Parameters (Hunt List)

Stop Hunting on Out of Bandwidth Flag *	False	False
Use Pickup Group Of Line Group Member DN *	False	False

- Clusterwide Parameters (External QoS)

External QoS Enabled *	False	False
------------------------	-------	-------

Figure 14: Service Parameters – Cont.

**- Clusterwide Parameters (Service)**

<a href="#">Default Network Hold MOH Audio</a>	1	1
<a href="#">Source ID</a> *		
<a href="#">Default User Hold MOH Audio</a>	1	1
<a href="#">Source ID</a> *		
<a href="#">Duplex Streaming Enabled</a> *	True	▼ False
<a href="#">Media Exchange Interface Capability Timer</a> *	8	8
<a href="#">Send Multicast MOH in H.245 OLC Message</a> *	True	▼ True
<a href="#">Media Exchange Timer</a> *	12	12
<a href="#">Media Exchange Stop Streaming Timer</a> *	8	8
<a href="#">Open Video Channel Response Timer for SIP Interop</a> *	500	500
<a href="#">Port Received Timer After Call Connection</a> *	500	500
<a href="#">Media Resource Allocation Timer</a> *	12	12
<a href="#">MTP and Transcoder Resource Throttling Percentage</a> *	95	95
<a href="#">Intercluster Capabilities Mismatch Timer</a> *	1000	1000
<a href="#">Silence Suppression</a> *	False	▼ False
<a href="#">Silence Suppression for Gateways</a> *	False	▼ False
<a href="#">Strip G.729 Annex B (Silence Suppression) from Capabilities</a> *	False	▼ False
<a href="#">Enable Source IP Address Verification for Software Media Devices</a> *	True	▼ True
<b>- Clusterwide Parameters (System - General)</b>		
<a href="#">Always Use Dial Tone Setting</a> *	Default	▼ Default
<a href="#">Restart Cisco CallManager on Initialization Exception</a> *	True	▼ True
<a href="#">Digit Analysis Timer</a> *	6	6
<a href="#">Statistics Enabled</a> *	True	▼ True

**Figure 15: Service Parameters – Cont.**



<b>- Clusterwide Parameters (System - QoS) —</b>		
<u>Priority Class</u> *	Normal Priority	▼ Normal Priority
<u>DSCP for Audio Calls</u> *	46 (101110)	▼ 46 (101110)
<u>DSCP for Video Calls</u> *	34 (100010)	▼ 34 (100010)
<u>DSCP for Audio Portion of Video Calls</u> *	34 (100010)	▼ 34 (100010)
<u>DSCP for TelePresence Calls</u> *	32 (100000)	▼ 32 (100000)
<u>DSCP for Audio Portion of TelePresence Calls</u> *	32 (100000)	▼ 32 (100000)
<u>DSCP for Priority Audio Calls</u> *	45 (101101)	▼ 45 (101101)
<u>DSCP for Immediate Audio Calls</u> *	44 (101100)	▼ 44 (101100)
<u>DSCP for Flash Audio Calls</u> *	41 (101001)	▼ 41 (101001)
<u>DSCP for Flash Override Audio Calls</u> *	42 (101010)	▼ 42 (101010)
<u>DSCP for Executive Override Audio Calls</u> *	42 (101010)	▼ 42 (101010)
<u>DSCP for Priority Video Calls</u> *	39 (100111)	▼ 39 (100111)
<u>DSCP for Immediate Video Calls</u> *	37 (100101)	▼ 37 (100101)
<u>DSCP for Flash Video Calls</u> *	35 (100011)	▼ 35 (100011)
<u>DSCP for Flash Override Video Calls</u> *	33 (100001)	▼ 33 (100001)
<u>DSCP for Executive Override Video Calls</u> *	33 (100001)	▼ 33 (100001)
<u>DSCP for G.Clear Calls</u> *	46 (101110)	▼ 46 (101110)
<u>DSCP for Priority G.Clear Calls</u> *	45 (101101)	▼ 45 (101101)
<u>DSCP for Immediate G.Clear Calls</u> *	44 (101100)	▼ 44 (101100)
<u>DSCP for Flash G.Clear Calls</u> *	41 (101001)	▼ 41 (101001)
<u>DSCP for Flash Override G.Clear Calls</u> *	42 (101010)	▼ 42 (101010)
<u>DSCP for Executive Override G.Clear Calls</u> *	42 (101010)	▼ 42 (101010)
<u>DSCP for Audio Calls when RSVP Fails</u> *	0 (000000)	▼ 0 (000000)
<u>DSCP for Video Calls when RSVP Fails</u> *	0 (000000)	▼ 0 (000000)
<u>DSCP for ICCP Protocol Links</u> *	24 (011000)	▼ 24 (011000)
<b>- Clusterwide Parameters (System - SDL) —</b>		
<u>SDL Listening Port Number</u> *	8002	8002
<u>SDL Max Router Latency</u> *	20	20
<u>Suppress Debug Info for Router Death</u> *	0	0
<u>Asynchronous SDL Logging Enabled</u> *	False	▼ False

Figure 16: Service Parameters – Cont.

**Clusterwide Parameters (System - Location and Region)**

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

**Figure 17: Service Parameters – Cont.**



<b>- Clusterwide Parameters (System - CCM Automated Alternate Routing)</b>	
<a href="#">Automated Alternate Routing Enable</a> *	<input type="checkbox"/> False
<b>- Clusterwide Parameters (System - RSVP)</b>	
<a href="#">Default inter-location RSVP Policy</a> *	No Reservation
<a href="#">RSVP Retry Timer</a> *	60
<a href="#">Mandatory RSVP Mid-call Retry Counter</a> *	1
<a href="#">Mandatory RSVP mid call error handle option</a> *	Call becomes best effort
<a href="#">RSVP Video Tspec Burst Size Factor</a> *	5
<a href="#">MLPP EXECUTIVE OVERRIDE To RSVP Priority Mapping</a> *	65535
<a href="#">MLPP FLASH OVERRIDE To RSVP Priority Mapping</a> *	65534
<a href="#">MLPP FLASH To RSVP Priority Mapping</a> *	65533
<a href="#">MLPP IMMEDIATE To RSVP Priority Mapping</a> *	65532
<a href="#">MLPP PL_PRIORITY To RSVP Priority Mapping</a> *	65531
<a href="#">MLPP PL_ROUTINE To RSVP Priority Mapping</a> *	65530
<a href="#">RSVP Audio Application ID</a> *	AudioStream
<a href="#">RSVP Video Application ID</a> *	VideoStream
<a href="#">RSVP Response Timer</a> *	2
<b>- TLS Packet Capture Configurations</b>	
<a href="#">Packet Capture Enable</a> *	<input type="checkbox"/> False
<a href="#">Packet Capture Max File Size (MB)</a> *	2
<b>- Clusterwide Parameters (System - Presence)</b>	
<a href="#">Presence Subscription Throttling Threshold</a> *	60000
<a href="#">Presence Subscription Resume Threshold</a> *	80
<a href="#">Default Inter-Presence Group Subscription</a> *	Disallow Subscription
<a href="#">BLF Status Depicts DND</a> *	<input type="checkbox"/> False

Figure 18: Service Parameters – Cont.

**- Clusterwide Parameters (System - Mobility)**

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

**Figure 19: Service Parameters – Cont.**

<b>- Clusterwide Parameters (System - Mobility Single Number Reach Voicemail)</b>		
<a href="#">Single Number Reach Voicemail Policy</a> *	<input type="text" value="Timer Control"/>	<input type="text" value="Timer Control"/>
<a href="#">Dial-via-Office Reverse Voicemail Policy</a> *	<input type="text" value="Timer Control"/>	<input type="text" value="Timer Control"/>
<a href="#">User Control Delayed Announcement Timer</a> *	<input type="text" value="1000"/>	<input type="text" value="1000"/>
<a href="#">User Control Confirmed Answer Indication Timer</a> *	<input type="text" value="10000"/>	<input type="text" value="10000"/>
<b>- Clusterwide Parameters (Feature - Reroute Remote Destination Calls to Enterprise Number)</b>		
<a href="#">Reroute Remote Destination Calls to Enterprise Number</a> *	<input type="text" value="False"/>	<input type="text" value="False"/>
<a href="#">Ring All Shared Lines</a> *	<input type="text" value="False"/>	<input type="text" value="False"/>
<a href="#">Ignore Call Forward All on Enterprise DN</a> *	<input type="text" value="True"/>	<input type="text" value="True"/>
<b>- Clusterwide Parameters (Feature - Immediate Divert)</b>		
<a href="#">Use Legacy Immediate Divert</a> *	<input type="text" value="True"/>	<input type="text" value="True"/>
<a href="#">Allow QSIG during iDivert</a> *	<input type="text" value="False"/>	<input type="text" value="False"/>
<a href="#">Immediate Divert User Response Timer</a> *	<input type="text" value="5"/>	<input type="text" value="5"/>
<b>- Clusterwide Parameters (Call Admission Control)</b>		
<a href="#">Call Counting CAC Enabled</a> *	<input type="text" value="False"/>	<input type="text" value="False"/>
<a href="#">Audio Bandwidth For Call Counting CAC</a> *	<input type="text" value="102"/>	<input type="text" value="102"/>
<a href="#">Video Bandwidth For Call Counting CAC</a> *	<input type="text" value="500"/>	<input type="text" value="500"/>
<a href="#">UCM to LBM Periodic Reservation Refresh Timer</a> *	<input type="text" value="5"/>	<input type="text" value="5"/>
<a href="#">Maximum Bandwidth Deduction Duration</a> *	<input type="text" value="720"/>	<input type="text" value="720"/>
<a href="#">Call Treatment When No LBM Available</a> *	<input type="text" value="Allow Calls"/>	<input type="text" value="Allow Calls"/>
<a href="#">Locations Media Resource Audio Bit Rate Policy</a> *	<input type="text" value="Lowest Bit Rate"/>	<input type="text" value="Lowest Bit Rate"/>
<a href="#">Video Call QoS Marking Policy</a> *	<input type="text" value="Default"/>	<input type="text" value="Default"/>
<a href="#">Deduct Audio Bandwidth Portion from Audio Pool for a Video Call</a> *	<input type="text" value="False"/>	<input type="text" value="False"/>
<b>- Clusterwide Parameters (Emergency Calling for Require Off-premise Location)</b>		
<a href="#">Alternate Destination for Emergency Call</a>	<input type="text"/>	
<a href="#">Alternate Calling Search Space for Emergency Call</a>	<input type="text" value="&lt; None &gt;"/>	
<input type="button" value="Save"/> <input type="button" value="Set to Default"/> <input type="button" value="Advanced"/>		

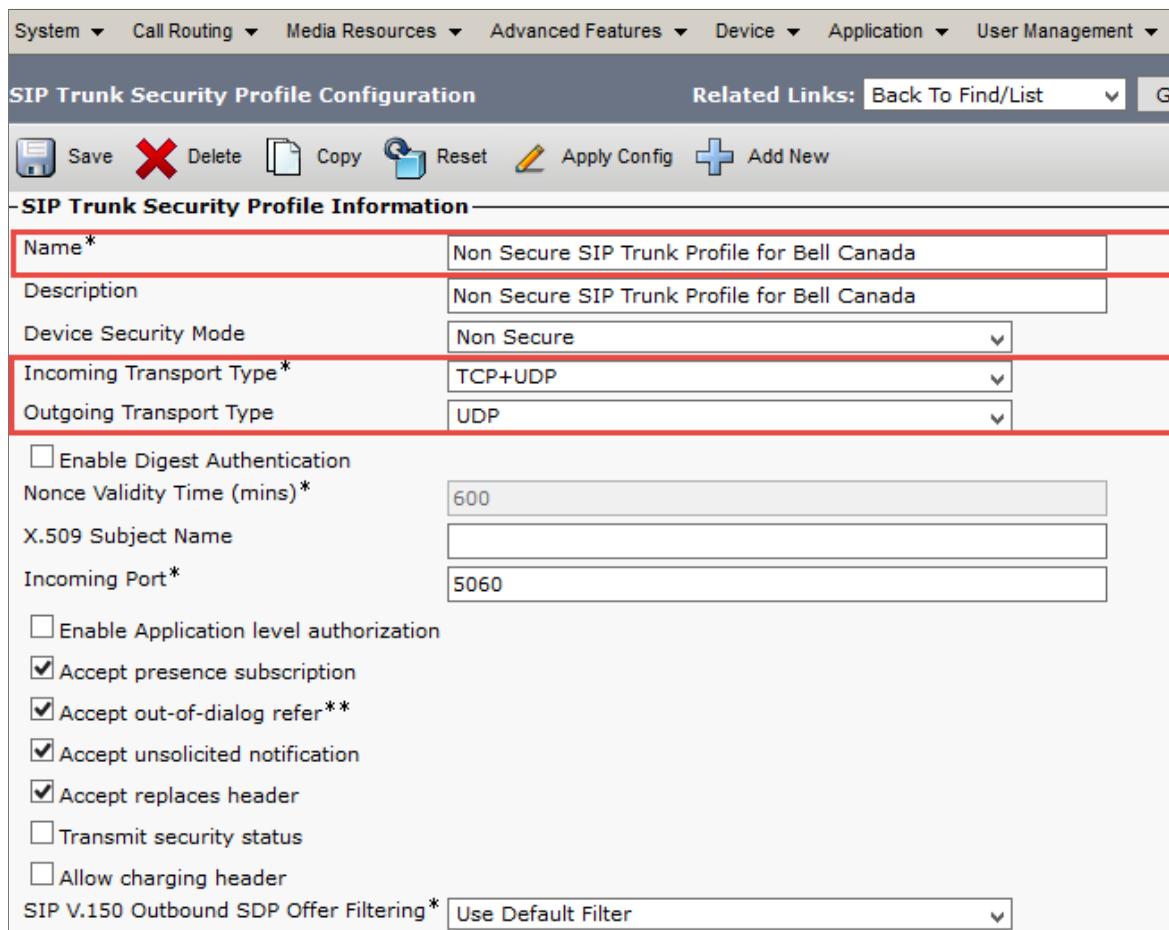
**Figure 20: Service Parameters – Cont.**

## 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**



The screenshot shows the 'SIP Trunk Security Profile Configuration' page. The 'Name\*' field is set to 'Non Secure SIP Trunk Profile for Bell Canada'. The 'Description' field is also 'Non Secure SIP Trunk Profile for Bell Canada'. The 'Device Security Mode' is set to 'Non Secure'. The 'Incoming Transport Type\*' is 'TCP+UDP' and the 'Outgoing Transport Type' is 'UDP'. There are several checkboxes for security features: 'Enable Digest Authentication' (unchecked), 'Nonce Validity Time (mins)\*' (set to 600), 'X.509 Subject Name' (empty), 'Incoming Port\*' (set to 5060), 'Enable Application level authorization' (unchecked), 'Accept presence subscription' (checked), 'Accept out-of-dialog refer\*\*' (checked), 'Accept unsolicited notification' (checked), 'Accept replaces header' (checked), 'Transmit security status' (unchecked), 'Allow charging header' (unchecked), and 'SIP V.150 Outbound SDP Offer Filtering\*' (set to 'Use Default Filter').

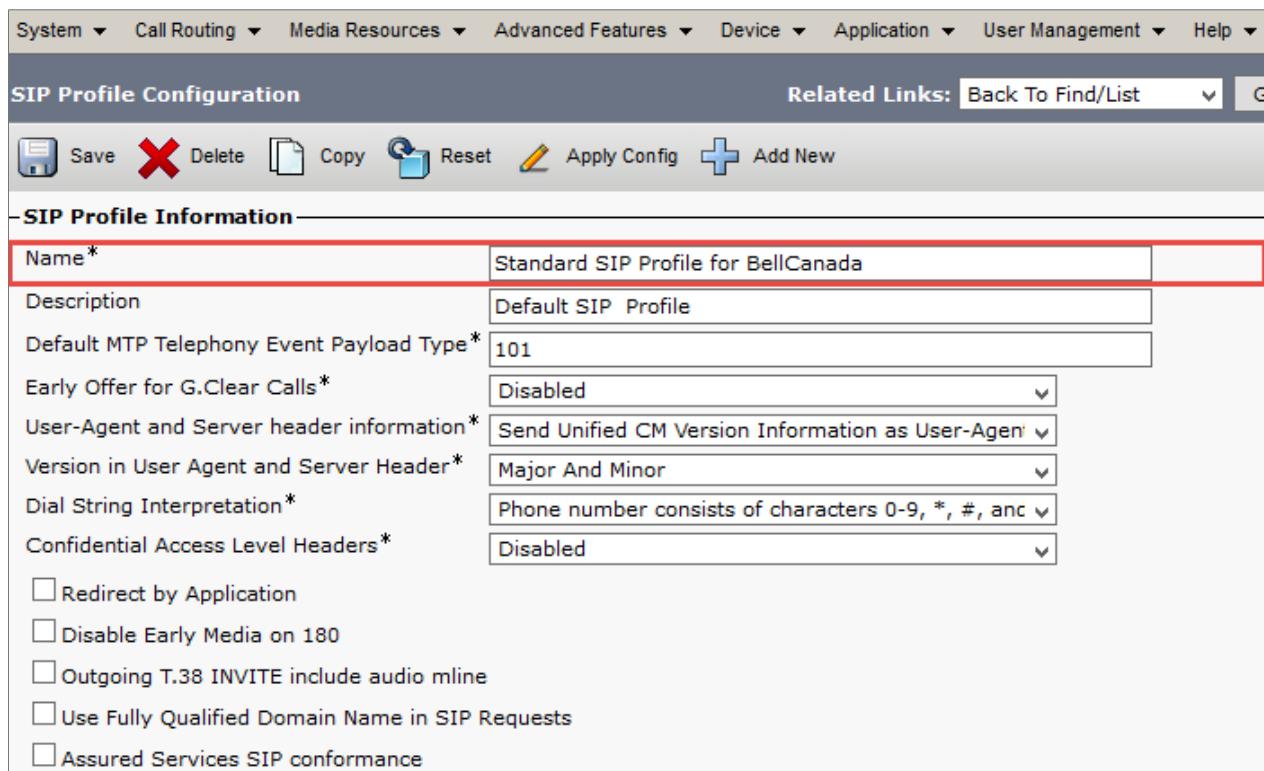
**Figure 21: SIP Trunk Security Profile**

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



**SIP Profile Configuration**

**SIP Profile Information**

Name*	Standard SIP Profile for BellCanada
Description	Default SIP Profile
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
User-Agent and Server header information*	Send Unified CM Version Information as User-Agent
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, and .
Confidential Access Level Headers*	Disabled
<input type="checkbox"/> Redirect by Application <input type="checkbox"/> Disable Early Media on 180 <input type="checkbox"/> Outgoing T.38 INVITE include audio mline <input type="checkbox"/> Use Fully Qualified Domain Name in SIP Requests <input type="checkbox"/> Assured Services SIP conformance	

**Figure 22: SIP Profile**

**- SDP Information**

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	<input type="text" value="TIAS and AS"/>
SDP Transparency Profile	<input type="text" value="Pass all unknown SDP attributes"/>
Accept Audio Codec Preferences in Received Offer*	<input type="text" value="Default"/>
<input checked="" type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change	
<input type="checkbox"/> Allow RR/RS bandwidth modifier (RFC 3556)	

---

**- Parameters used in Phone**

Timer Invite Expires (seconds)*	<input type="text" value="180"/>
Timer Register Delta (seconds)*	<input type="text" value="5"/>
Timer Register Expires (seconds)*	<input type="text" value="3600"/>
Timer T1 (msec)*	<input type="text" value="500"/>
Timer T2 (msec)*	<input type="text" value="4000"/>
Retry INVITE*	<input type="text" value="6"/>
Retry Non-INVITE*	<input type="text" value="10"/>
Media Port Ranges	<input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> Separate Port Ranges for Audio and Video
Start Media Port*	<input type="text" value="16384"/>
Stop Media Port*	<input type="text" value="32766"/>
DSCP for Audio Calls	<input type="text" value="Use System Default"/>
DSCP for Video Calls	<input type="text" value="Use System Default"/>
DSCP for Audio Portion of Video Calls	<input type="text" value="Use System Default"/>
DSCP for TelePresence Calls	<input type="text" value="Use System Default"/>
DSCP for Audio Portion of TelePresence Calls	<input type="text" value="Use System Default"/>
Call Pickup URI*	<input type="text" value="x-cisco-serviceuri-pickup"/>
Call Pickup Group Other URI*	<input type="text" value="x-cisco-serviceuri-opickup"/>
Call Pickup Group URI*	<input type="text" value="x-cisco-serviceuri-gpickup"/>
Meet Me Service URI*	<input type="text" value="x-cisco-serviceuri-meetme"/>
User Info*	<input type="text" value="None"/>
DTMF DB Level*	<input type="text" value="Nominal"/>
Call Hold Ring Back*	<input type="text" value="Off"/>
Anonymous Call Block*	<input type="text" value="Off"/>
Caller ID Blocking*	<input type="text" value="Off"/>
Do Not Disturb Control*	<input type="text" value="User"/>
Telnet Level for 7940 and 7960*	<input type="text" value="Disabled"/>

**Figure 23: SIP Profile – Cont.**

Resource Priority Namespace	<input type="button" value="&lt; None &gt;"/>				
Timer Keep Alive Expires (seconds)*	120				
Timer Subscribe Expires (seconds)*	120				
Timer Subscribe Delta (seconds)*	5				
Maximum Redirections*	70				
Off Hook To First Digit Timer (milliseconds)*	15000				
Call Forward URI*	x-cisco-serviceuri-cfwdall				
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial				
<input checked="" type="checkbox"/> Conference Join Enabled					
<input type="checkbox"/> RFC 2543 Hold					
<input checked="" type="checkbox"/> Semi Attended Transfer					
<input type="checkbox"/> Enable VAD					
<input type="checkbox"/> Stutter Message Waiting					
<input type="checkbox"/> MLPP User Authorization					
<b>-Normalization Script-</b>					
Normalization Script	<input type="button" value="&lt; None &gt;"/>				
<input type="checkbox"/> Enable Trace					
<table border="1"> <thead> <tr> <th>Parameter Name</th> <th>Parameter Value</th> </tr> </thead> <tbody> <tr> <td>1</td> <td></td> </tr> </tbody> </table>		Parameter Name	Parameter Value	1	
Parameter Name	Parameter Value				
1					
<b>-Incoming Requests FROM URI Settings-</b>					
Caller ID DN					
Caller Name					
<b>-Trunk Specific Configuration-</b>					
Reroute Incoming Request to new Trunk based on*	Never				
Resource Priority Namespace List	<input type="button" value="&lt; None &gt;"/>				
SIP Rel1XX Options*	Send PRACK if 1xx Contains SDP				
Video Call Traffic Class*	Mixed				
Calling Line Identification Presentation*	Default				
Session Refresh Method*	Invite				
Early Offer support for voice and video calls*	Disabled (Default value)				
<input type="checkbox"/> Enable ANAT					
<input type="checkbox"/> Deliver Conference Bridge Identifier					
<input type="checkbox"/> Allow Passthrough of Configured Line Device Caller Information					
<input type="checkbox"/> Reject Anonymous Incoming Calls					
<input type="checkbox"/> Reject Anonymous Outgoing Calls					
<input type="checkbox"/> Send ILS Learned Destination Route String					

**Figure 24: SIP Profile – Cont.**

**SIP OPTIONS Ping**

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

Ping Interval for In-service and Partially In-service Trunks (seconds)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6

**SDP Information**

Send send-receive SDP in mid-call INVITE  
 Allow Presentation Sharing using BFCP  
 Allow iX Application Media  
 Allow multiple codecs in answer SDP

**Action Buttons:** Save, Delete, Copy, Reset, Apply Config, Add New

**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

Cisco Unified CM Administration													
For Cisco Unified Communications Solutions													
Find and List Trunks													
<input type="button" value="Add New"/>	<input type="button" value="Select All"/>	<input type="button" value="Clear All"/>	<input type="button" value="Delete Selected"/>	<input type="button" value="Reset Selected"/>									
<b>Trunks (1 - 3 of 3)</b>													
Find Trunks where <input type="text" value="Device Name"/> begins with <input type="text"/> <input type="button" value="Find"/> <input type="button" value="Clear Filter"/> <input type="button" value="+"/> <input type="button" value="="/> <input type="text" value="Select item or enter search text"/>													
		Name 	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile
<input type="checkbox"/>		<a href="#">Bell_Canada</a>	SIP trunk to Bell Canada		<a href="#">G729_Pool_Bell</a>	<a href="#">0613260XXXX</a>				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 0 minute	<a href="#">Non Secure SIP Trunk Profile for Bell Canada</a>
<input type="checkbox"/>		<a href="#">Bell_Canada</a>	SIP trunk to Bell Canada		<a href="#">G729_Pool_Bell</a>	<a href="#">8.@"</a>				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 0 minute	<a href="#">Non Secure SIP Trunk Profile for Bell Canada</a>
<input type="checkbox"/>		<a href="#">FaxGateway2</a>	Fax Gateway 2		<a href="#">G729_Pool_Bell</a>	<a href="#">3924</a>				SIP Trunk	Full Service	Time In Full Service: 1 day 14 hours 14 minutes	<a href="#">Non Secure SIP Trunk Profile</a>
<input type="button" value="Add New"/> <input type="button" value="Select All"/> <input type="button" value="Clear All"/> <input type="button" value="Delete Selected"/> <input type="button" value="Reset Selected"/>													

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  
 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\*

Route Class Signaling Enabled\*

Use Trusted Relay Point\*

PSTN Access  
 Run On All Active Unified CM Nodes

**-Intercompany Media Engine (IME)**

E.164 Transformation Profile

Figure 27: SIP Trunk to CUBE

**-MLPP and Confidential Access Level Information**

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

---

**-Call Routing Information**

<input checked="" type="checkbox"/> Remote-Party-Id	
<input checked="" type="checkbox"/> Asserted-Identity	
Asserted-Type*	Default
SIP Privacy*	Default

---

**Inbound Calls**

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

Redirecting Diversion Header Delivery - Inbound

---

**-Incoming Calling Party Settings**

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

<b>Clear Prefix Settings</b>		<b>Default Prefix Settings</b>		
Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

---

**-Incoming Called Party Settings**

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

<b>Clear Prefix Settings</b>		<b>Default Prefix Settings</b>		
Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**Figure 28: SIP Trunk to CUBE – Cont.**



**Connected Party Settings**

Connected Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Connected Party Transformation CSS	

**Outbound Calls**

Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling and Connected Party Info Format*	Deliver DN only in connected party
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	
Redirecting Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Redirecting Party Transformation CSS	

**Caller Information**

Caller ID DN	
Caller Name	
<input type="checkbox"/> Maintain Original Caller ID DN and Caller Name in Identity Headers	

**SIP Information**

**Destination**

<input type="checkbox"/> Destination Address is an SRV		
Destination Address	Destination Address IPv6	Destination
1 * 10.80.13.20		5060

MTP Preferred Originating Codec\* 711ulaw

BLF Presence Group\* Standard Presence group

SIP Trunk Security Profile\* Non Secure SIP Trunk Profile for Bell Canada

Rerouting Calling Search Space < None >

Out-Of-Dialog Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile\* Standard SIP Profile for BellCanada [View Details](#)

DTMF Signaling Method\* RFC 2833

Figure 29: SIP Trunk to CUBE – Cont.

**Normalization Script**

Normalization Script < None >

Enable Trace

Parameter Name	Parameter Value
1	

**Recording Information**

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

**Geolocation Configuration**

Geolocation < None >  
Geolocation Filter < None >  
 Send Geolocation Information

**Action Buttons**

Save | Delete | Reset | Add New

**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**

The screenshot shows the Cisco Unified CM Administration interface under the 'Route Patterns' section. The page title is 'Cisco Unified CM Administration' with the subtitle 'For Cisco Unified Communications Solutions'. The navigation bar includes links for 'Cisco Unified CM Administration', 'administrator', 'Search Documentation', 'About', and 'Logout'. The main menu has dropdowns for 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', and 'Help'. Below the menu, there's a toolbar with buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'. A search bar allows filtering by 'Pattern' with an 'Advanced' dropdown and a 'Find' button. The table lists three route patterns:

	Pattern	Description	Partition	Route Filter	Associated Device	Copy
<input type="checkbox"/>	<a href="#">0613260XXXX</a>				<a href="#">Bell_Canada</a>	
<input type="checkbox"/>	<a href="#">3924</a>				<a href="#">FaxGateway2</a>	
<input type="checkbox"/>	<a href="#">8.@</a>				<a href="#">Bell_Canada</a>	

At the bottom of the table are buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'.

Figure 31: Route Patterns

**-Pattern Definition-**

Route Pattern*	8.@	
Route Partition	< None >	
Description		
Numbering Plan*	NANP	
Route Filter	< None >	
MLPP Precedence*	Default	
<input type="checkbox"/> Apply Call Blocking Percentage		
Resource Priority Namespace Network Domain	< None >	
Route Class*	Default	
Gateway/Route List*	Bell_Canada	(Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern	
Call Classification*	OffNet	
External Call Control Profile	< None >	
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority		
<input type="checkbox"/> Require Forced Authorization Code		
Authorization Level*	0	
<input type="checkbox"/> Require Client Matter Code		

---

**-Calling Party Transformations-**

<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	Allowed
Calling Name Presentation*	Allowed
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

---

**-Connected Party Transformations-**

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

**Figure 32: Route Patterns for PSTN**

**-Connected Party Transformations**

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

---

**-Called Party Transformations**

Discard Digits	PreDot
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager

---

**-ISDN Network-Specific Facilities Information Element**

Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

---

**Action Buttons:** Save | Delete | Copy | Add New

**Figure 33: Route Patterns for PSTN – Cont.**

**Route Pattern Configuration**

**Related Links:** Back To Find/List | Go

**Pattern Definition**

Route Pattern*	0613260XXXX	
Route Partition	< None >	
Description		
Numbering Plan	-- Not Selected --	
Route Filter	< None >	
MLPP Precedence*	Default	
<input type="checkbox"/> Apply Call Blocking Percentage		
Resource Priority Namespace Network Domain	< None >	
Route Class*	Default	
Gateway/Route List*	Bell_Canada	(Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern	
Call Classification*	OffNet	
External Call Control Profile	< None >	
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority		

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



Require Forced Authorization Code  
Authorization Level\*

Require Client Matter Code

**-Calling Party Transformations**

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

**-Connected Party Transformations**

Connected Line ID Presentation\*   
Connected Name Presentation\*

**-Called Party Transformations**

Discard Digits   
Called Party Transform Mask   
Prefix Digits (Outgoing Calls)   
Called Party Number Type\*   
Called Party Numbering Plan\*

**-ISDN Network-Specific Facilities Information Element**

Network Service Protocol   
Carrier Identification Code   
Network Service  Service Parameter Name  Service Parameter Value

**Buttons:** Save | Delete | Copy | Add New

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

**- Pattern Definition -**

Route Pattern *	3924
Route Partition	< None >
Description	
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence *	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class *	Default
Gateway/Route List *	FaxGateway2
<a href="#">(Edit)</a>	
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern
Call Classification *	OnNet
External Call Control Profile	< None >
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level *	0
<input type="checkbox"/> Require Client Matter Code	

---

**- Calling Party Transformations -**

<input type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation *	Default
Calling Name Presentation *	Default
Calling Party Number Type *	Cisco CallManager
Calling Party Numbering Plan *	Cisco CallManager

---

**- Connected Party Transformations -**

Connected Line ID Presentation *	Default
Connected Name Presentation *	Default

**Figure 36: Route Patterns for Fax**



-Called Party Transformations-

Discard Digits < None >

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type\* Cisco CallManager

Called Party Numbering Plan\* Cisco CallManager

-ISDN Network-Specific Facilities Information Element-

Network Service Protocol -- Not Selected --

Carrier Identification Code

Network Service Service Parameter Name Service Parameter

-- Not Selected -- < Not Exist >

Save Delete Copy Add New

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



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