Application Note

BroadCloud SIP Trunking:

Cisco Unified Communications Manager 11.0.1 with Cisco Unified Border Element (CUBE 11.5.0) on ISR4321/K9 [IOS-XE 3.17.1 – 15.6(1)S1] using SIP

August 29, 2016

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Introduction

Service Providers today, such as BroadCloud, 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.

BroadCloud is a service provider offering 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 BroadCloud network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS-XE 3.17.1 – 15.6(1)S1 can be used. The Cisco Unified Border Element provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.0.1 connected to BroadCloud IP network.

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

- This application note describes how to configure Cisco Unified Communications Manager (Cisco UCM) 11.0.1, and Cisco Unified Border Element (Cisco UBE) on ISR 4321/K9 [IOS-XE 3.17.1 – 15.6(1)S1] for connectivity to BroadCloud SIP Trunking service. The deployment model covered in this application note is CPE (Cisco Unified Communications Manager 11.0.1) to PSTN (BroadCloud).
- Testing was performed in accordance to BroadCloud generic SIP Trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold (MOH), unattended and attended transfers, call forward, conferences and interoperability with Cisco Unity Connection (CUC).
- The Cisco UCM configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between BroadCloud 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 BroadCloud SIP Trunking network.

This application note does not cover the use of Calling Search Spaces (CSS) or partitions on Cisco UCM. 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

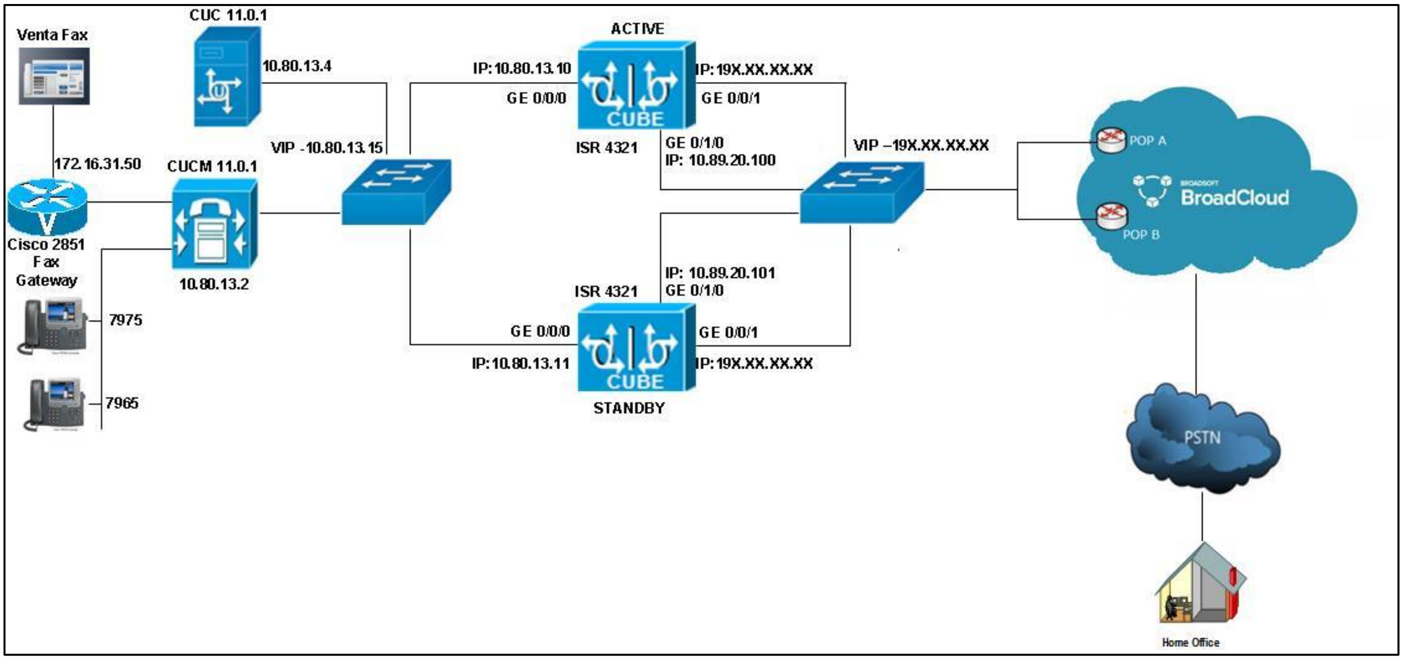


Figure 1: Network Topology

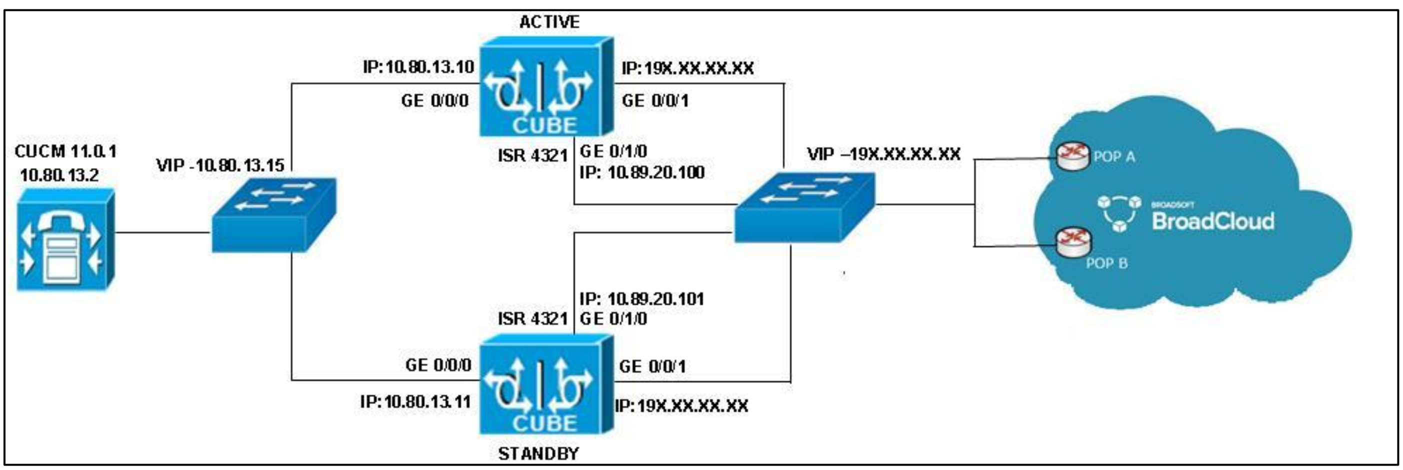


Figure 2: Cisco UBE High Availability

System Components

Hardware Requirements

- Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4321/K9 router as CUBE
- Cisco ISR4321/K9 (1RU) processor with 1647061K/6147K bytes of memory with 3 Gigabit Ethernet interfaces
- Processor board ID FLM1925W0WZ
- Cisco 2851 Fax Gateway
- IP phones 7965 (SIP), 7975 (SCCP)

Software Requirements

- Cisco Unified Communications Manager 11.0.1.21900-11
- Cisco Unity Connection 11.0.1.21900-11
- IOS-XE 15.6(1)S1 for ISR 4321/K9 Cisco Unified Border Element
- Cisco IOS Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.6(1)S1, RELEASE SOFTWARE (fc3)
- Cisco IOS XE Software, Version 03.17.01.S
- IOS 15.0(1)XA for Cisco 2851 Fax Gateway

Features

Features Supported

- Incoming and outgoing off-net calls using G711ULAW & G729 voice codecs

- Call hold
- Call transfer (unattended and attended)
- Call Conference
- Call forward (all, busy, no answer)
- Calling Line (number) Identification Presentation (CLIP)
- Calling Line (number) Identification Restriction (CLIR)
- DTMF relay (both directions) (RFC2833)
- Media flow-through on Cisco UBE
- Fax (T38 & G711 pass-through)

Features Not Supported

- In HA Redundancy mode the Primary cube will not take over the Primary/Active role after a reboot/network outage

Caveats

- Caller ID is not updated after attended or semi-attended transfers to off-net phones. This is due to a limitation on Cisco UBE and will be resolved in the next release. The issue does not impact the calls.
- BroadCloud does not support faxes using SuperG3. However, fax test cases are executed using G3.
- 911 Calls were not tested since BroadCloud lab environment does not have the ability to route to a 911 PSAP calls
- Early media call that requires PRACK with SDP Failed with enabling “require100rel” since BroadCloud does not support “require100rel”. However, the call was successful without PRACK.

Configuration

Configuring the Cisco Unified Border Element for Registration SIP Trunk Testing

Network interface

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

```
interface GigabitEthernet0/0/0
 description bcloud LAN
 ip address 10.80.13.10 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 6
 redundancy group 2 ip 10.80.13.15 exclusive
!
interface GigabitEthernet0/0/1
 description bcloud WAN
 ip address 192.XX.XX.XX 255.255.255.128
 negotiation auto
 redundancy rii 7
 redundancy group 2 ip 192.XX.XX.XX exclusive
!
```

Global Cisco UBE settings

In order to enable Cisco UBE IP2IP gateway functionality, enter the following:

```
!
voice service voip
 ip address trusted list
  ipv4 199.XX.XX.XX
  ipv4 199.XX.XX.XX
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 2
 sip
  bind control source-interface GigabitEthernet0/0/1
  rel1xx supported "rel100"
  session refresh
  asserted-id pai
  privacy pstn
  outbound-proxy dns:XXX@domain.com
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
!
```

Explanation

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
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

Codecs

G711ulaw and G729 voice codecs are used for this testing. Codec preferences used to change according to the test plan description

```
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 codec preference 3 g722-56
```

Dial peer

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

```
dial-peer voice 500 voip
 description Outgoing call to Broadcloud - LAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number XXXXXX....
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 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 510 voip
 description Outgoing call to Broadcloud - WAN facing
 huntstop
 destination-pattern XXXXXX....
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip outbound-proxy dns:XXX@domain.com
 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 520 voip
 description Incoming call to PBX - WAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number XXXXXX….
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 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 530 voip
 description Incoming call to PBX - LAN facing
 huntstop
 destination-pattern XXXXXX….
 session protocol sipv2
 session target ipv4:10.80.13.2:5060
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 voice-class sip options-keepalive
 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
!
```

Configuration example

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

Active Cisco UBE

```
bcloud1#sh run


version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname bcloud1
!
boot-start-marker
boot system flash 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
!
no aaa new-model
!
ip name-server 8.26.56.xx
```

```
!
subscriber templating
!
multilink bundle-name authenticated
!
voice service voip
 ip address trusted list
  ipv4 199.XX.XX.XX
  ipv4 199.XX.XX.XX
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 2
 sip
  bind control source-interface GigabitEthernet0/0/1
  rel1xx supported "rel100"
  session refresh
  asserted-id pai
  outbound-proxy dns:XXX@domain.com
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 codec preference 3 g722-56
!
voice class codec 2
 codec preference 1 g729r8
```

```
 codec preference 2 g711ulaw
 codec preference 3 g722-56
!
voice class sip-profiles 101
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:XXXXXX\1@\2"
!
voice translation-rule 1
 rule 1 /^.*\(..........\)/ /\1/
!
voice translation-profile BroadCloud
translate called 1
!
license udi pid ISR4321/K9 sn FDO19220MQ8
license boot level appxk9
license boot level uck9
!
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 2
   name b2bhaBcloud
   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 bcloud LAN
 ip address 10.80.13.10 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 6
 redundancy group 2 ip 10.80.13.15 exclusive
!
interface GigabitEthernet0/0/1
 description bcloud WAN
 ip address 192.XX.XX.XX 255.255.255.128
 negotiation auto
 redundancy rii 7
 redundancy group 2 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA
 ip address 10.89.20.100 255.255.255.0
 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 GigabitEthernet0/0/1
ip route 0.0.0.0 0.0.0.0 192.XX.XX.XX
ip route 10.80.0.0 255.255.0.0 10.80.13.1
ip route 172.16.0.0 255.255.0.0 10.80.13.1
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 500 voip
 description Outgoing call to Broadcloud - LAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number XXXXXX....
 voice-class codec 1
 voice-class sip asserted-id pai
```

```
 no voice-class sip outbound-proxy
 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 510 voip
 description Outgoing call to Broadcloud - WAN facing
 huntstop
 destination-pattern XXXXXX....
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip outbound-proxy dns:XXX@domain.com
 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 520 voip
 description Incoming call to PBX - WAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number XXXXXX….
 voice-class codec 1
 voice-class sip asserted-id pai
```

```
 voice-class sip options-keepalive
 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 530 voip
 description Incoming call to PBX - LAN facing
 huntstop
 destination-pattern XXXXXX….
 session protocol sipv2
 session target ipv4:10.80.13.2:5060
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 voice-class sip options-keepalive
 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 600 voip
 description PBX to PBX Outgoing Call - LAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number 1XXXXXX….
 voice-class codec 1
 voice-class sip asserted-id pai
```

```
 no voice-class sip outbound-proxy
 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 610 voip
 description PBX to PBX Outgoing Call - WAN facing
 translation-profile outgoing BroadCloud
 huntstop
 destination-pattern 1XXXXXX….
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip outbound-proxy dns:XXX@domain.com
 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
!
sip-ua
 credentials number XXXXXXXXXX username XXXXXXXXXX password 7 realm XXX@customerdomain.com
 authentication username XXXXXXXXXX password 7
 no remote-party-id
 retry invite 2
 timers keepalive active 10
 registrar dns:XXX@customerdomain.com:8933 expires 60
```

```
 sip-server dns:XXX@customerdomain.com:8933
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password XXXXXX
 login
!
end
```

Standby Cisco UBE

```
bcloud2#sh run


version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname bcloud2
!
boot-start-marker
boot system flash 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
!
no aaa new-model
!
ip name-server 8.26.56.xx
!
subscriber templating
multilink bundle-name authenticated
```

```
!
voice service voip
 ip address trusted list
  ipv4 199.XX.XX.XX
  ipv4 199.XX.XX.XX
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 2
 sip
  bind control source-interface GigabitEthernet0/0/1
  rel1xx supported "rel100"
  session refresh
  asserted-id pai
  outbound-proxy dns:XXX@domain.com
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 codec preference 3 g722-56
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711ulaw
 codec preference 3 g722-56
!
!
```

```
voice class sip-profiles 101
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:XXXXXX\1@\2"
!
voice translation-rule 1
 rule 1 /^.*\(..........\)/ /\1/!
!
voice translation-profile BroadCloud
translate called 1
!
license udi pid ISR4321/K9 sn FDO19220MQ9
!
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 2
   name b2bhaBcloud
   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 bcloud LAN
 ip address 10.80.13.11 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 6
 redundancy group 2 ip 10.80.13.15 exclusive
!
interface GigabitEthernet0/0/1
 description bcloud WAN
 ip address 192.XX.XX.XX 255.255.255.128
 negotiation auto
 redundancy rii 7
 redundancy group 2 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA
 ip address 10.89.20.101 255.255.255.0
 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 route 0.0.0.0 0.0.0.0 192.XX.XX.XX
ip route 10.80.0.0 255.255.0.0 10.80.13.1
ip route 172.16.0.0 255.255.0.0 10.80.13.1
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 500 voip
 description Outgoing call to Broadcloud - LAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number XXXXXX....
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 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 510 voip
 description Outgoing call to Broadcloud - WAN facing
 huntstop
 destination-pattern XXXXXX....
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip outbound-proxy dns:XXX@domain.com
 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 520 voip
 description Incoming call to PBX - WAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number XXXXXX….
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 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 530 voip
 description Incoming call to PBX - LAN facing
 huntstop
 destination-pattern XXXXXX….
 session protocol sipv2
 session target ipv4:10.80.13.2:5060
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 voice-class sip options-keepalive
 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 600 voip
 description PBX to PBX Outgoing Call - LAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number 1XXXXXX….
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 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 610 voip
 description PBX to PBX Outgoing Call - WAN facing
 translation-profile outgoing BroadCloud
 huntstop
 destination-pattern 1XXXXXX….
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip outbound-proxy dns:XXX@domain.com
 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
!
!
sip-ua
 credentials number XXXXXXXXXX username XXXXXXXXXX password 7 realm XXX@customerdomain.com
 authentication username XXXXXXXXXX password 7
 no remote-party-id
 retry invite 2
 timers keepalive active 180
 registrar dns:XXX@customerdomain.com:8933 expires 60
 sip-server dns:XXX@customerdomain.com:8933
!
line con 0
 exec-timeout 0 0
 stopbits 1
```

```
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password XXXXXXX
 login
!
!
end
```

Call flow

In the sample configuration presented here, Cisco UCM is provisioned with four-digit directory numbers corresponding to the last four DID digits. No digit manipulation is performed on the Cisco UBE.
For incoming PSTN calls, the Cisco UBE presents the full ten-digit DID number to Cisco UCM. The Cisco UCM picks up the last 4 significant Digits configured under SIP Trunk and routes the call based on those 4 digits. Voice calls are routed to IP phones; Fax calls are routed via a 4-digit route pattern over a SIP trunk that terminates on the Fax Gateway and in turn to the VentaFax client connected to the Fax Gateway.

CPE callers make outbound PSTN calls by dialing a “7” prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, Cisco fax Gateway sends to Cisco UCM the DID with leading access code “7”. A “7.@” route pattern strips the prefix and routes the call with the remaining digits via a SIP trunk terminating on the Cisco UBE for Voice call or Fax. For PBX to PBX via BroadCloud, Caller dial 7 prefix followed by the target 10-digits number, 7 was stripped and the remaining digits were send to Cisco UBE, Cisco UBE pass the DID under Dial Peer 510 and send to BroadCloud network which will direct back to Cisco UBE and handled same as normal incoming PSTN call.

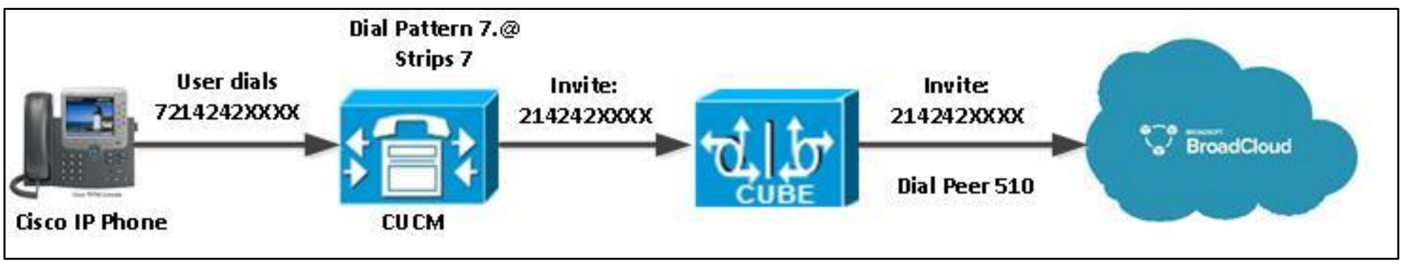


Figure 3: Outbound Voice Call

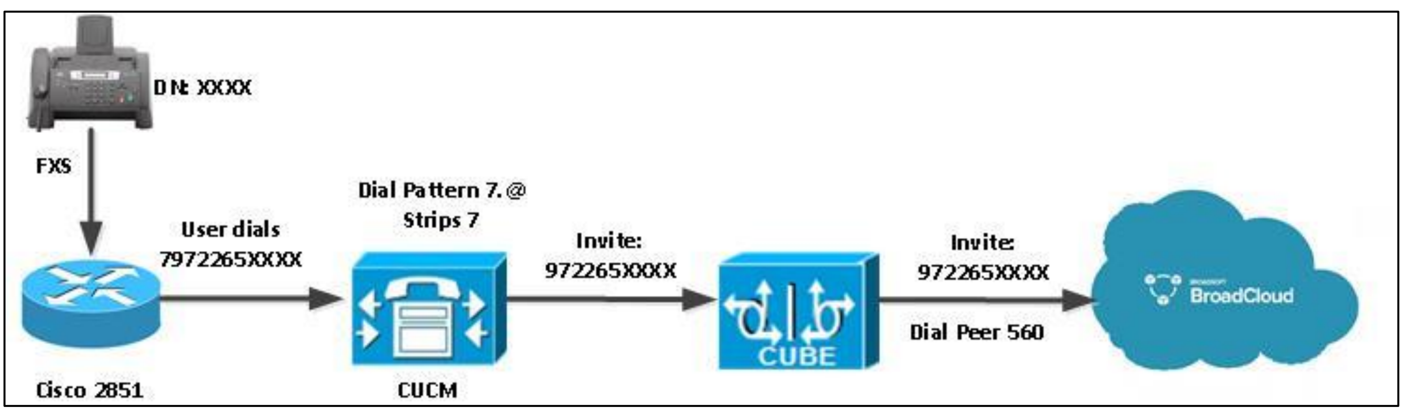


Figure 4: Outbound Fax Call

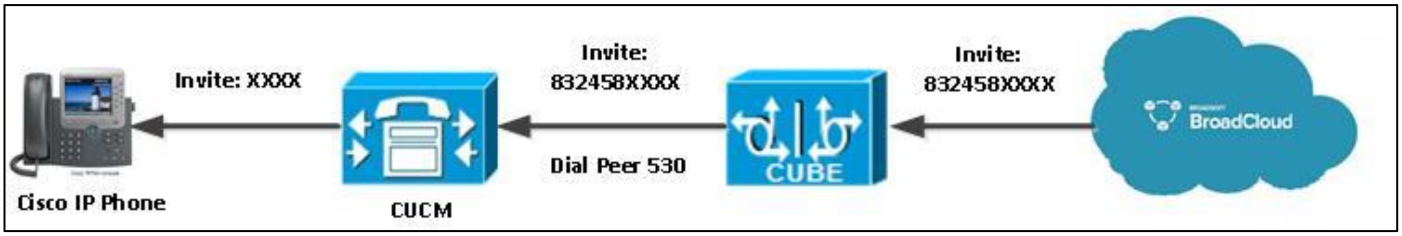


Figure 5: Inbound Voice Call

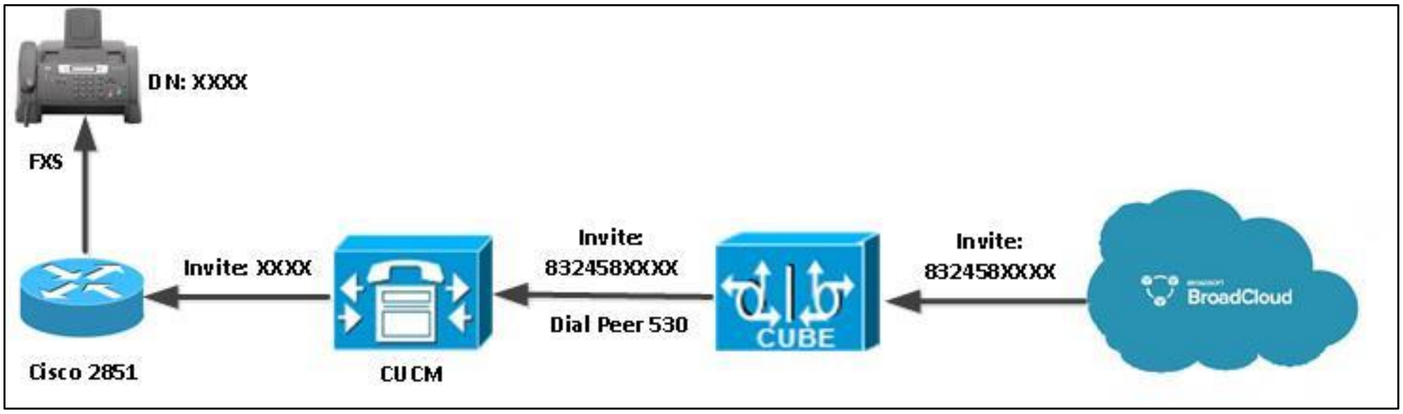


Figure 6 : Inbound Fax Call

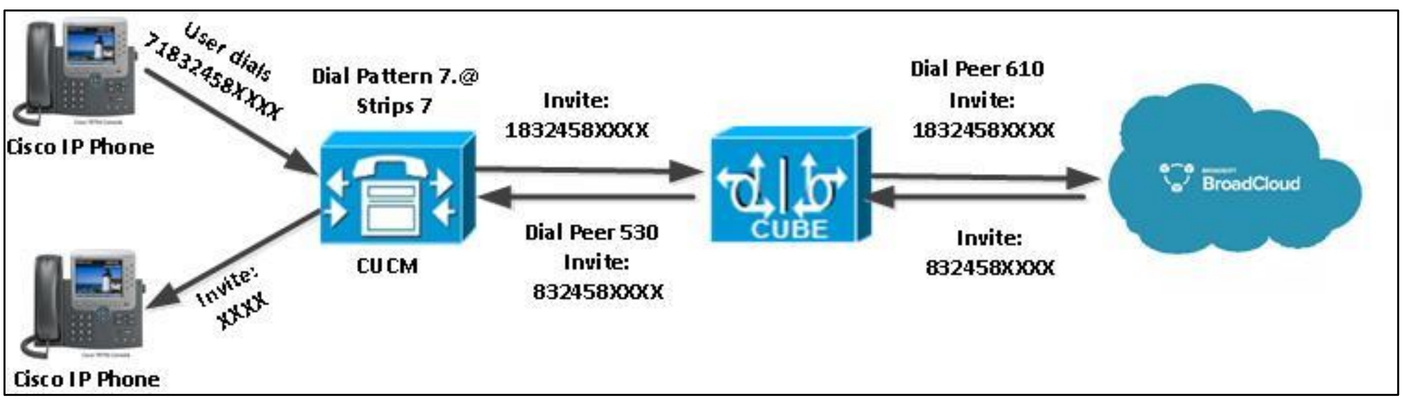


Figure 7 : PBX to PBX via BroadCloud Call

Configuring Cisco Unified Communications Manager

Cisco UCM Version

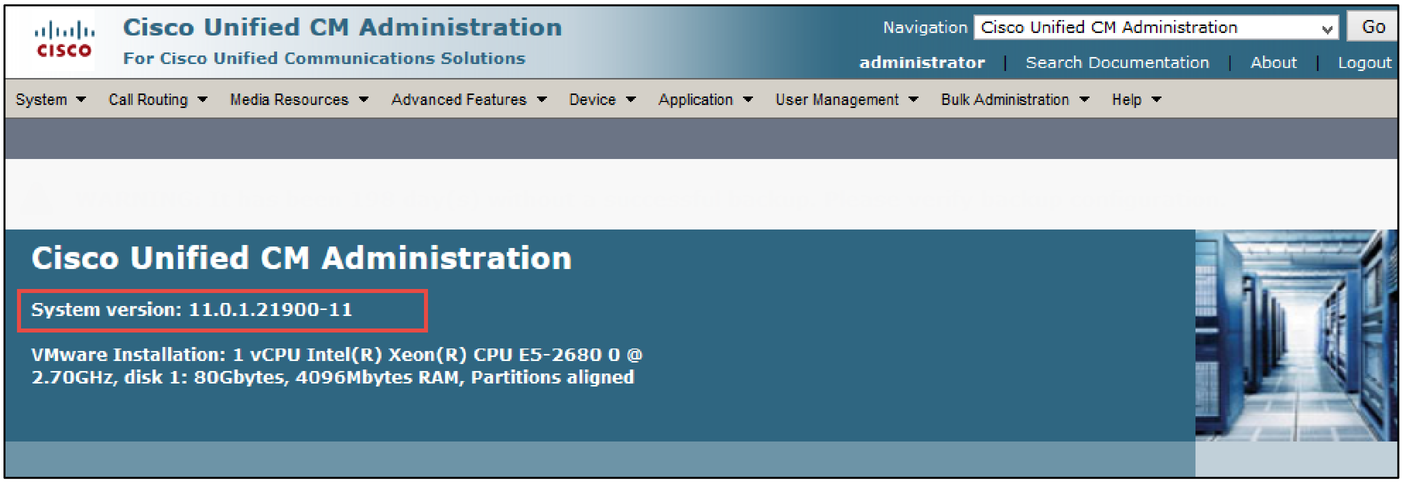


Figure 8: Cisco UCM Version

Cisco Call manager Service Parameters

Navigation: System → Service Parameters

1. Select Server* = clus23pubsub--CUCM Voice/Video (Active)
2. Select Service*= Cisco CallManager (Active)
3. All other fields are set to default values

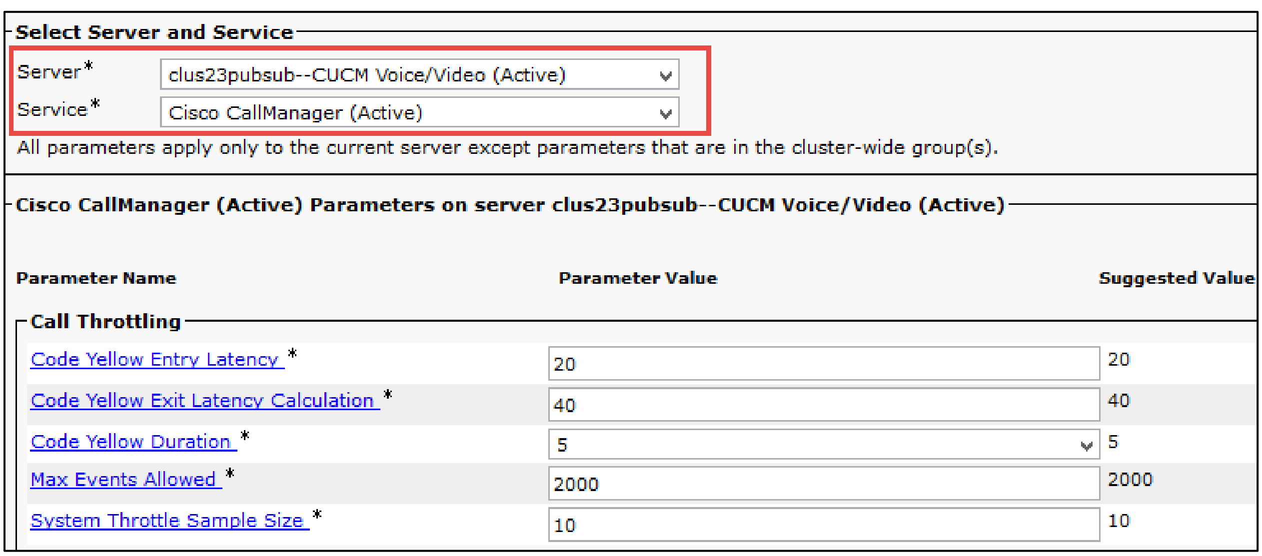


Figure 9: Service Parameters

Offnet Calls via BroadCloud SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and BroadCloud Network and calls are routed via Cisco UBE

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

1. **Name=** *BroadCloud Non Secure SIP Trunk Profile* is used as an example
2. **Description** = *Non Secure non secure trunk profile* is used as an example

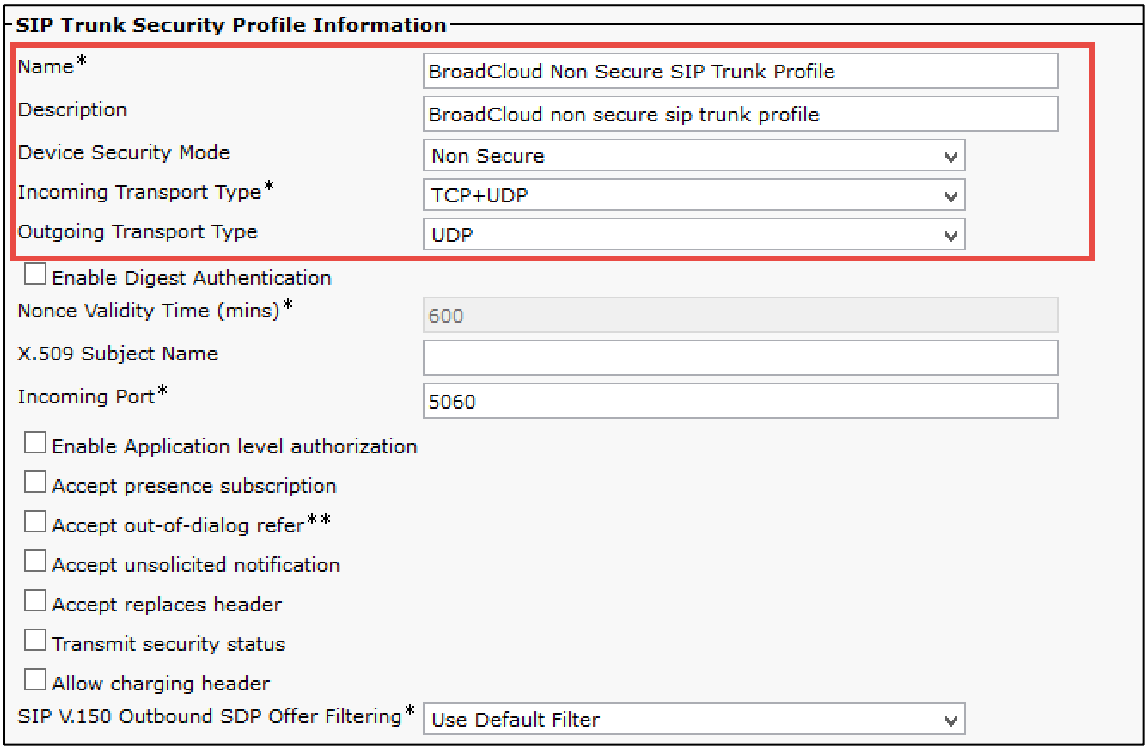


Figure 10: SIP Trunk Security Profile

Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to BroadCloud SBC should use UDP as a transport protocol for SIP. This is configured using SIP Trunk Security profile, which is later assigned to the SIP trunk itself.

SIP Profile Configuration

NOTE: SIP Profile will be later associated with the SIP trunk

Navigation: Device > Device Settings > SIP Profile

1. **Name**= *BroadCloud SIP Profile* is used as an example
2. **Description** = *BroadCloud SIP Profile* is used as an example

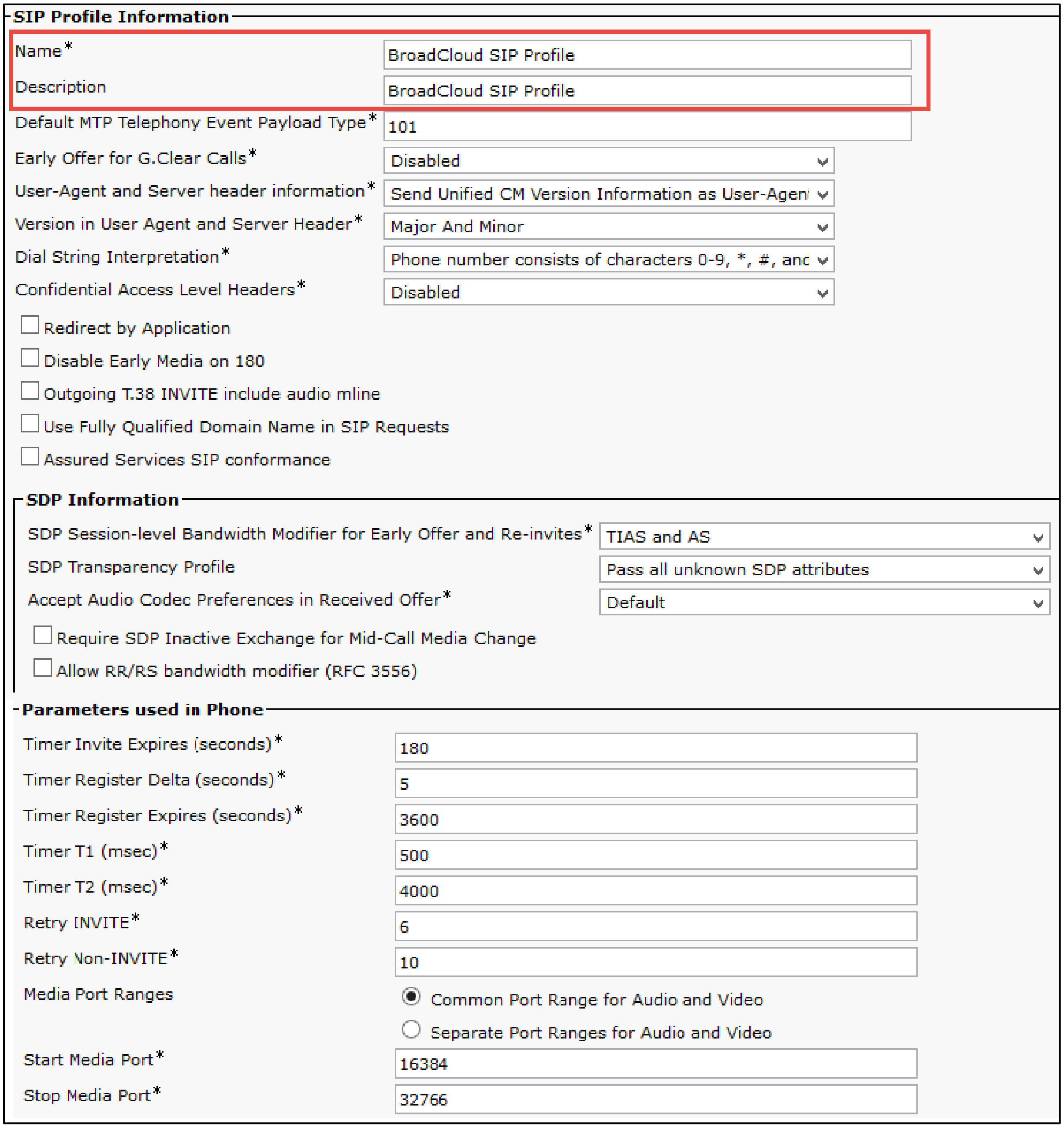


Figure 11: SIP Profile

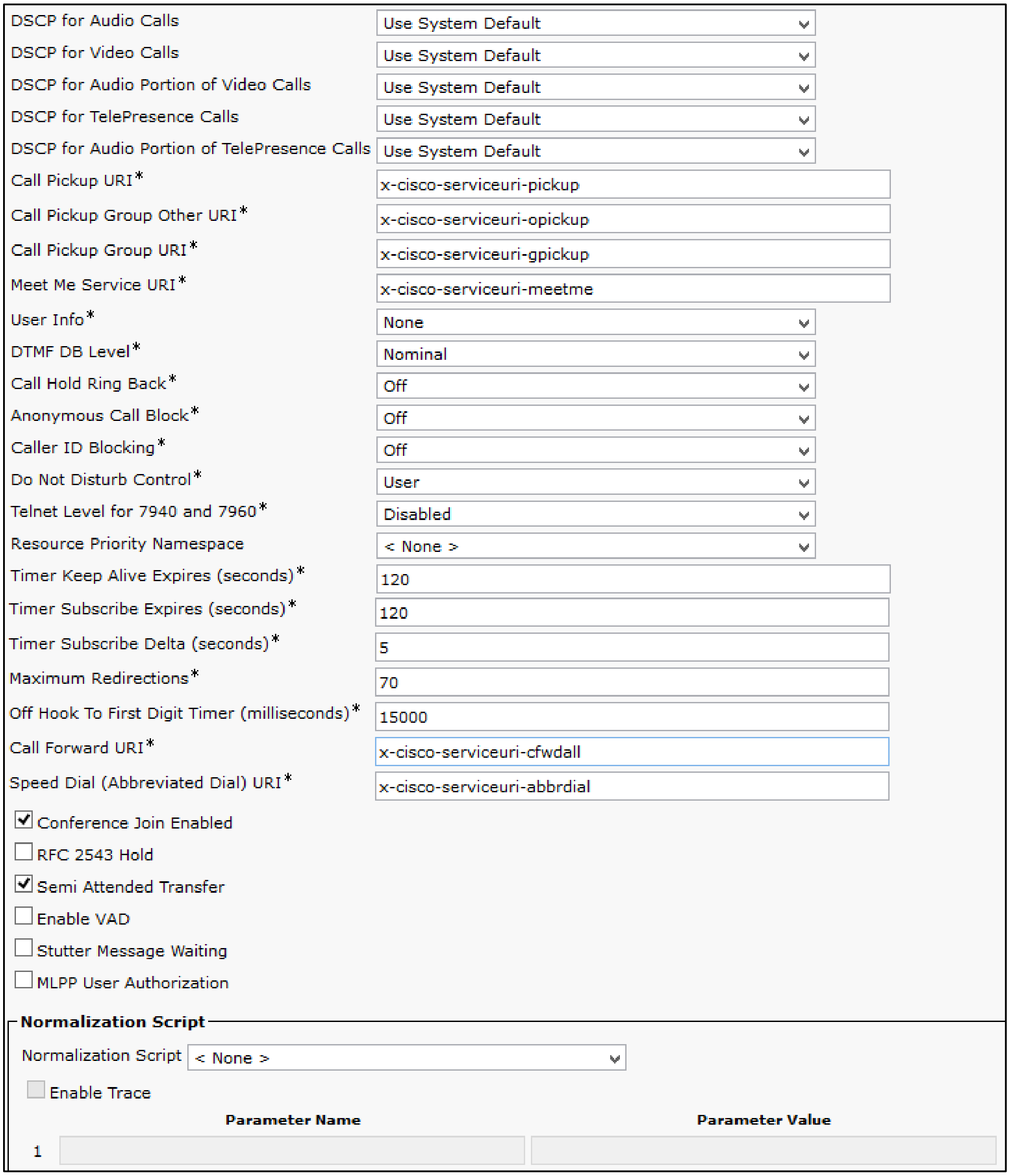


Figure 12: SIP Profile (Cont.)

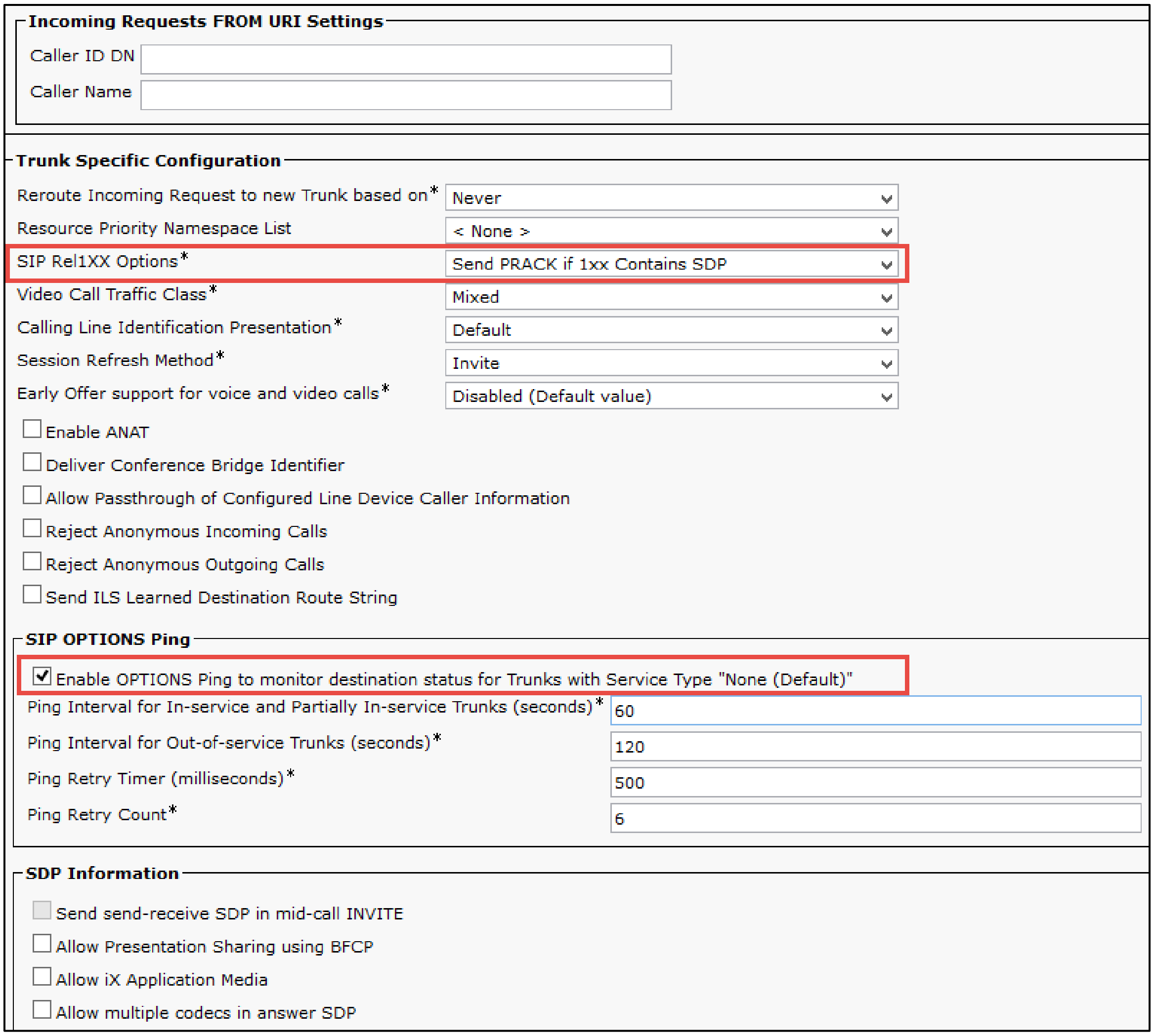


Figure 13: SIP Profile (Cont.)

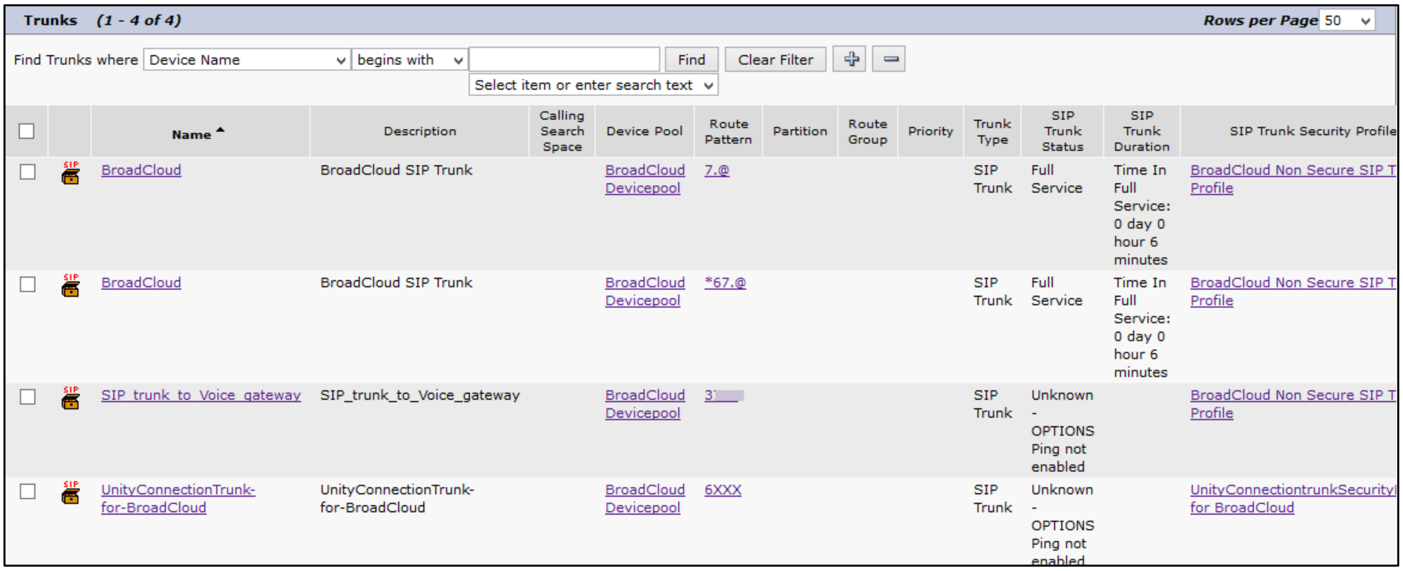
Explanation

Parameter	Value	Description
Default MTP Telephony Event Payload Type	101	RFC2833 DTMF payload type
SIP Rel1XX Options	Send PRACK for 1xx Contains SDP	Enable Provisional Acknowledgements (Reliable 100 messages)
Ping Interval for In-service and Partially In-service Trunks (seconds)	60	OPTIONS message parameters- interval time
Ping Interval for Out-of-service Trunks (seconds)	120	OPTIONS message parameters- interval time

SIP Trunk Configuration

Create SIP trunks to Cisco UBE

Navigation: Device > Trunk



Trunks (1 - 4 of 4) — *Rows per Page* 50

Find Trunks where Device Name | begins with | | Find | Clear Filter

Select item or enter search text

	Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile
	BroadCloud	BroadCloud SIP Trunk		BroadCloud Devicepool	7.@				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 6 minutes	BroadCloud Non Secure SIP T Profile
	BroadCloud	BroadCloud SIP Trunk		BroadCloud Devicepool	*67.@				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 6 minutes	BroadCloud Non Secure SIP T Profile
	SIP_trunk_to_Voice_gateway	SIP_trunk_to_Voice_gateway		BroadCloud Devicepool	3				SIP Trunk	Unknown - OPTIONS Ping not enabled		BroadCloud Non Secure SIP T Profile
	UnityConnectionTrunk-for-BroadCloud	UnityConnectionTrunk-for-BroadCloud		BroadCloud Devicepool	6XXX				SIP Trunk	Unknown - OPTIONS Ping not enabled		UnityConnectiontrunkSecurity for BroadCloud

Figure 14: SIP Trunks List

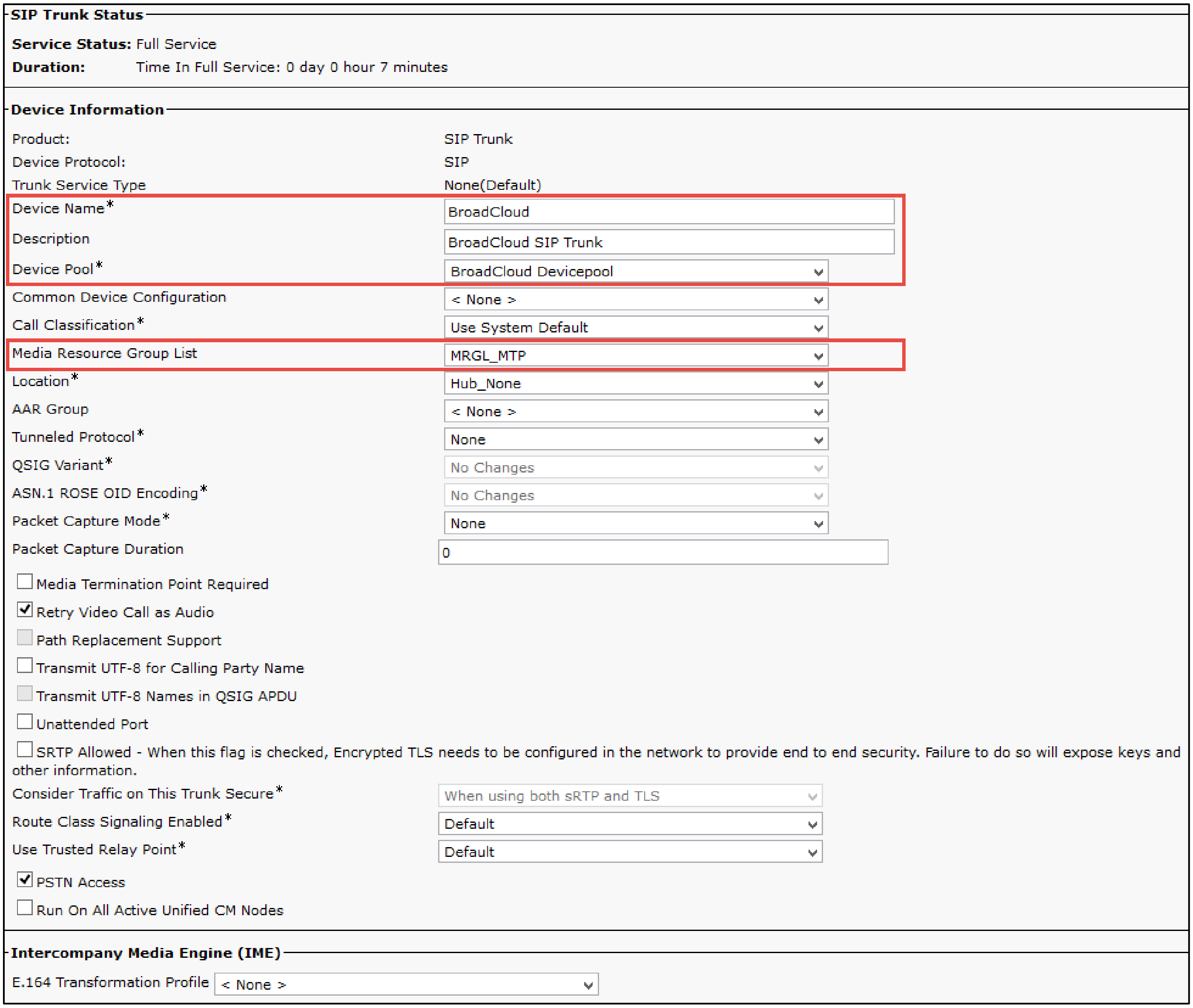


Figure 15: SIP Trunk to Cisco UBE

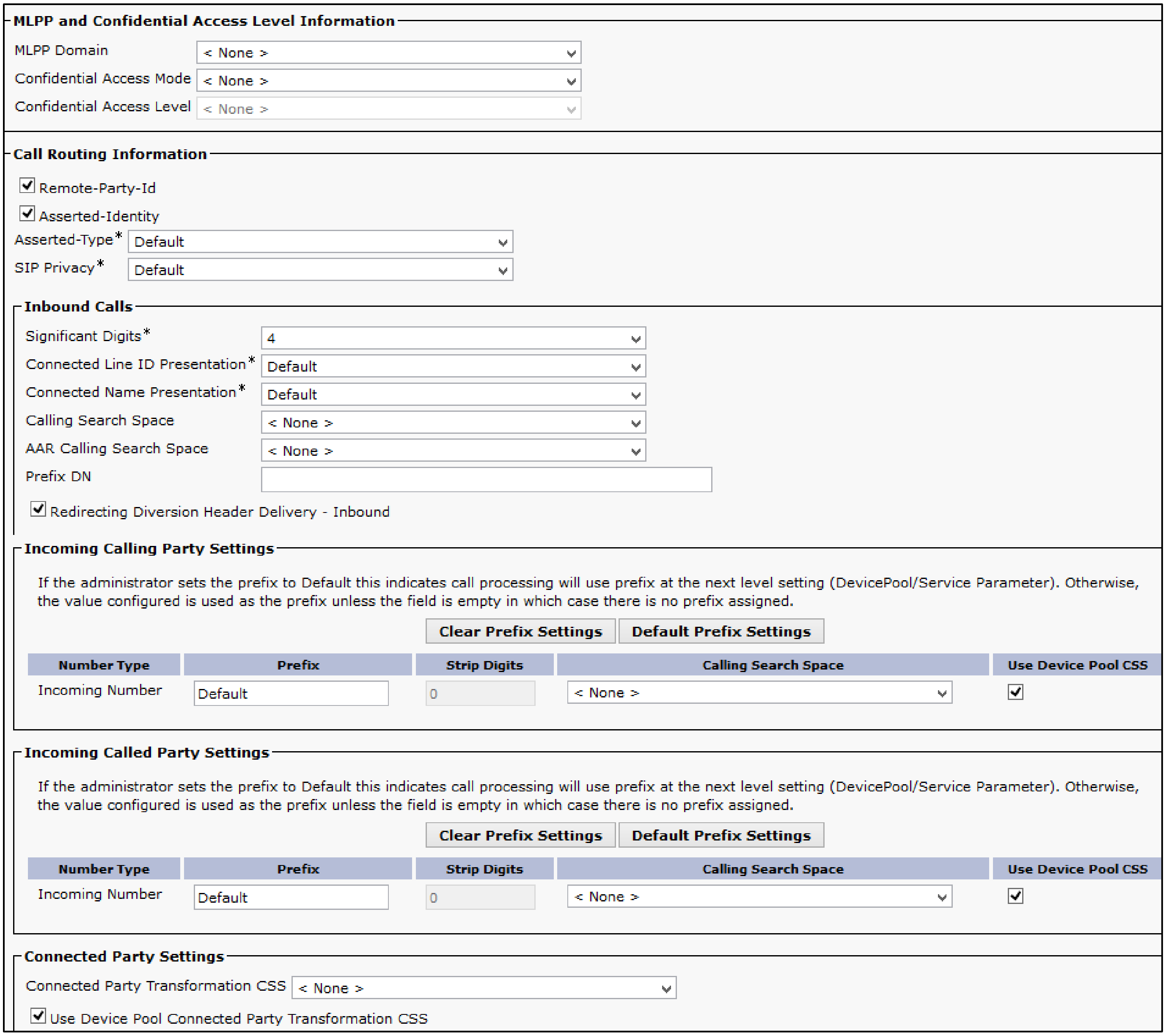
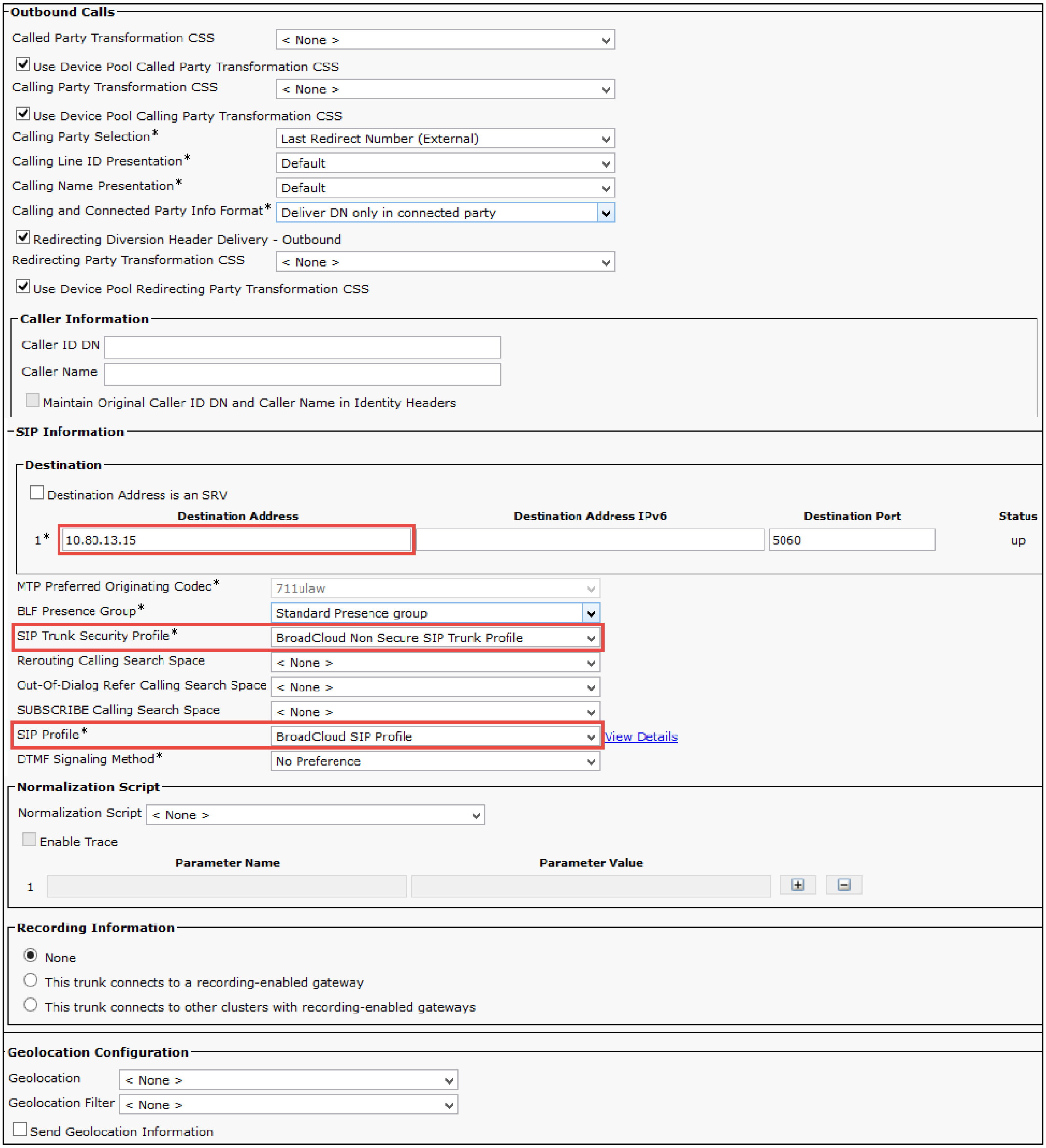


Figure 16: SIP Trunk to Cisco UBE (Cont.)



Outbound Calls

Called Party Transformation CSS: < None >

☑ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS: < None >

☑ Use Device Pool Calling Party Transformation CSS

Calling Party Selection*: Last Redirect Number (External)

Calling Line ID Presentation*: Default

Calling Name Presentation*: Default

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

☑ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS: < None >

☑ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

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

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status
1*	10.80.13.15		5060	up

MTP Preferred Originating Codec*: 711ulaw

BLF Presence Group*: Standard Presence group

SIP Trunk Security Profile*: BroadCloud Non Secure SIP Trunk Profile

Rerouting Calling Search Space: < None >

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

SUBSCRIBE Calling Search Space: < None >

SIP Profile*: BroadCloud SIP Profile View Details

DTMF Signaling Method*: No Preference

Normalization Script

Normalization Script: < None >

☐ Enable Trace

	Parameter Name	Parameter Value
1		

Recording Information

◉ None

○ This trunk connects to a recording-enabled gateway

○ This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation: < None >

Geolocation Filter: < None >

☐ Send Geolocation Information

Figure 17: SIP Trunk to Cisco UBE (Cont.)

Explanation

Parameter	Value	Description
Device Name	BroadCloud	Name for the trunk
Device Pool	BroadCloud Devicepool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_MTP_Group	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.13.15	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	BroadCloud Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	BroadCloud SIP Profile	SIP Profile configured earlier

Dial Plan

Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

1. Cisco IP phone dial "7"+10 digits number to access PSTN via Cisco UBE
   - "7" is removed before sending to Cisco UBE
2. For FAX call, Access Code "7"+10 digits number is used at Cisco Fax gateway
   - "7" is removed at Cisco UCM
   - The rest of the number is sent to Cisco UBE to BroadCloud network
3. Incoming fax call to 3XXX will be sent to Cisco Fax gateway
4. For Anonymous call, access code "*67"+10 digits number is used.
   - "*67" is removed at Cisco UCM
   - The rest of the number is sent to Cisco UBE to BroadCloud network

Route Patterns *(1 - 4 of 4)* **Rows per Page** 50

Find Route Patterns where Pattern begins with [Find] [Clear Filter]

	Pattern	Description	Partition	Route Filter	Associated Device	Copy
	*67.@	BroadCloud RP for Anonymous calls			BroadCloud	
	3	BroadCloud RP for FAX			SIP_trunk_to_Voice_gateway	
	7.@	BroadCloud Route pattern			BroadCloud	

Figure 18: Route Patterns List

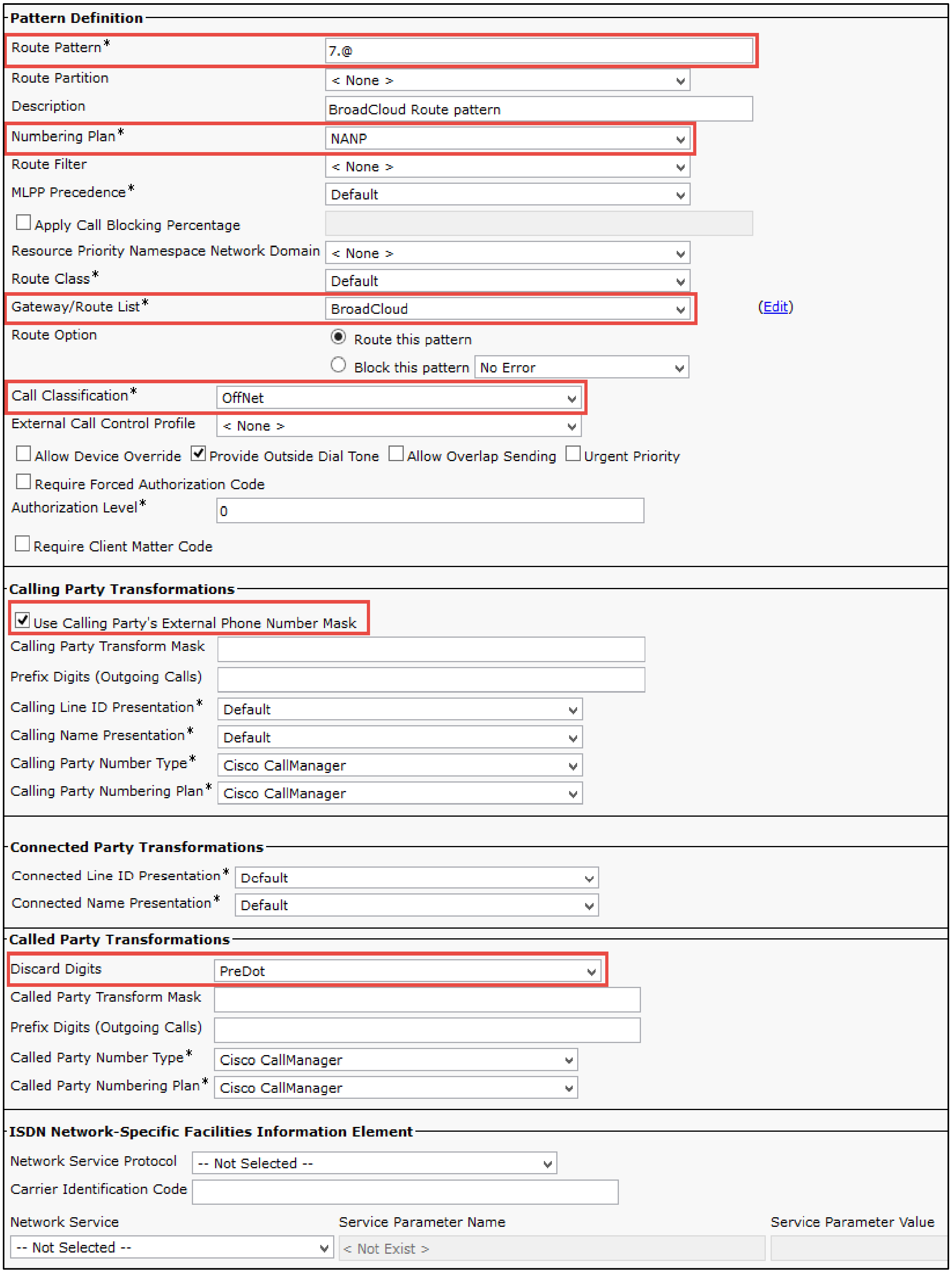


Figure 19: Route Pattern for Voice

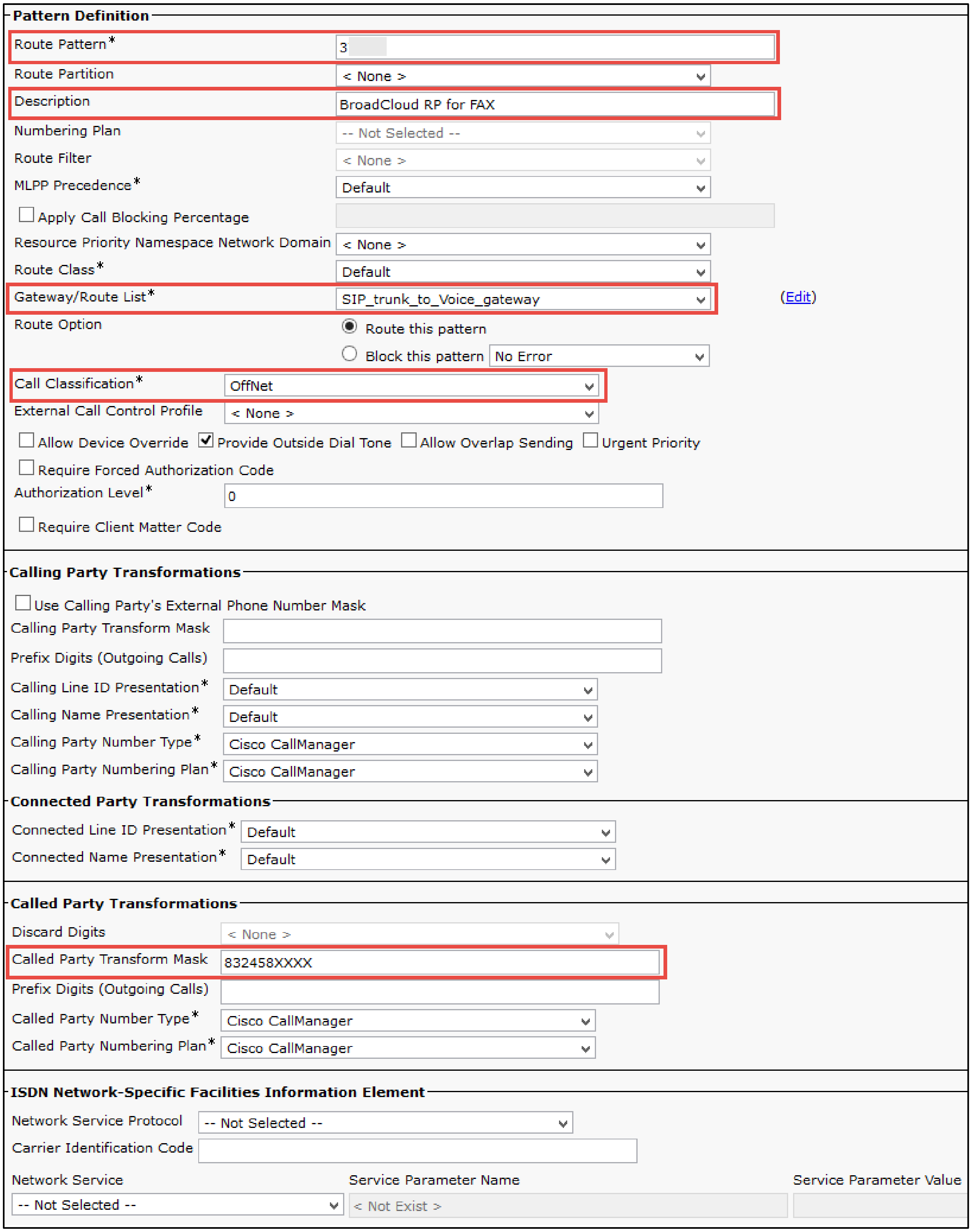


Figure 20: Route Pattern for Voice (Cont.)

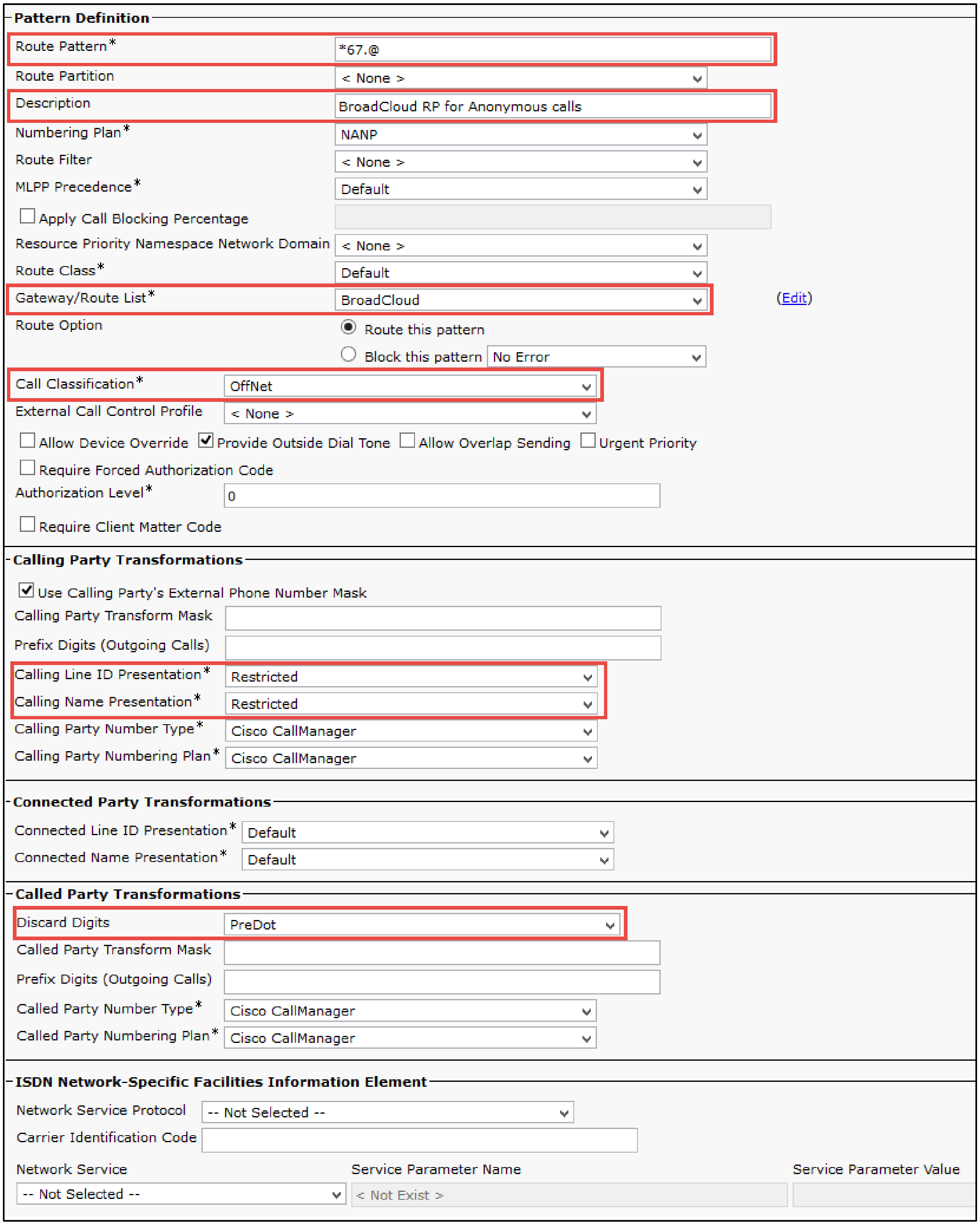


Figure 21: Route Pattern for Fax

Explanation

Setting	Value	Description
Route Pattern	7.@ for Voice & International Calls, 3XXX for Fax Call and *67.@ for Anonymous Call.	Specify appropriate Route Pattern
Gateway/Route List	BroadCloud for Route Pattern 7.@, *67.@ and SIP_Trunk_To_Voice_Gateway for Route Pattern 3XXX	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 7.@ and *67.@	North American Numbering Plan
Call Classification	OffNet for Route Pattern 7.@, 3XXX and *67.@	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 7.@ and *67.@	Specifies how to modify digit before they are sending to BroadCloud network
Calling Line ID Presentation & Calling Name Presentation	Restricted for Route Pattern *67.@	Restrict the Caller ID Display of Calling party's at external user endpoint.

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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