Application Note

Comcast SIP Trunking:

Cisco Unified Communications Manager 11.0.1 with Cisco Unified Border Element (CUBE 11.5.0) on ISR 4321 [IOS-XE 3.17- 15.6(1)S] using SIP

May 04, 2016

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Introduction

Service Providers today, such as Comcast, 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.

Comcast 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 Comcast network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS 15.6(1)S can be used. The Cisco Unified Border Element 11.5.0 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.0.1 connected to Comcast IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco UCM (Cisco Unified Communications Manager). Only configuration settings specifically required for Comcast 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 a Cisco Unified Communications Manager (Cisco UCM) 11.0.1 and Cisco Unified Border Element (Cisco UBE) on ISR 4321/K9 [IOS-XE 3.17 - 15.6(1)S] for connectivity to Comcast SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 11.0.1) to PSTN (Comcast).
- Testing was performed in accordance to Comcast 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 Comcast 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 Comcast 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

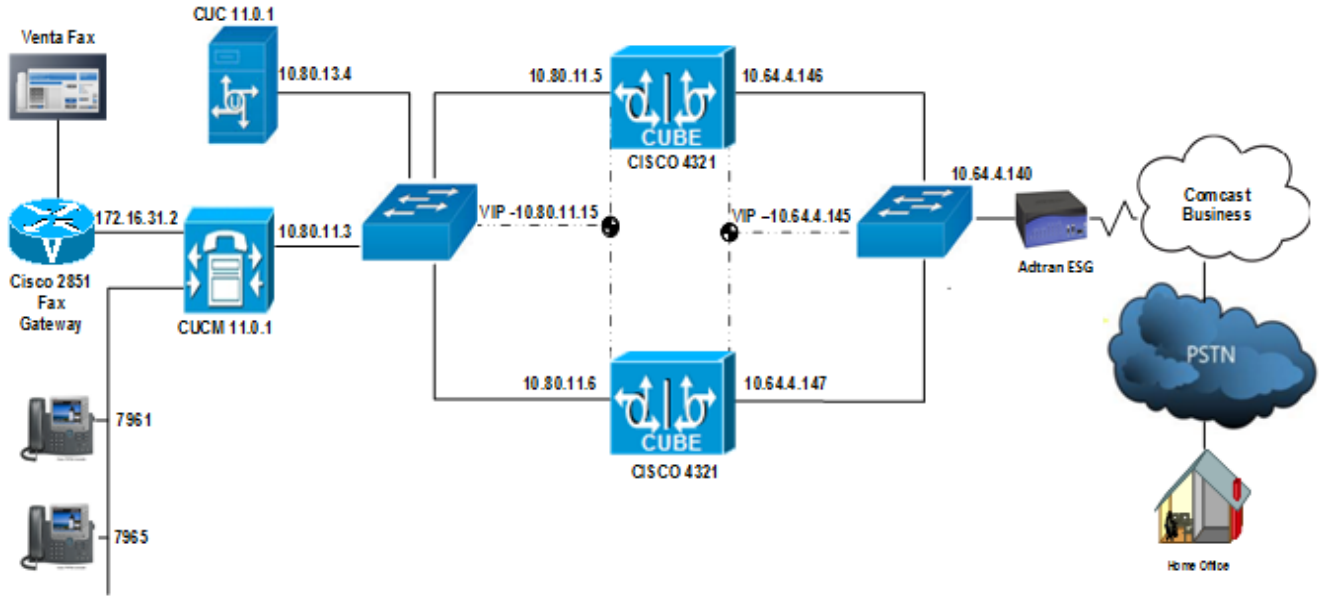

Figure 1: Network Topology

- Cisco IP Phones 7961 and 7965 phones are the devices primarily used throughout the testing to place or receive calls
- VentaFax Soft Client is used to perform all fax related scenarios. The fax client is connected to SIP Gateway via FXS port which in turn communicates with Cisco UCM over SIP.

System Components

Hardware Requirements

- Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4321/K9 router as CUBE
- Cisco 2851 Fax Gateway
- IP phones 7961 (SIP) and 7965 (SCCP)
- Comcast(Adtran) ESG – Provided and managed by Comcast

Software Requirements

- Cisco Unified Communications Manager 11.0.1
- Cisco Unity Connection 11.0.1
- IOS-XE 3.17.0 – 15.6(1)S for ISR 4321/K9 Cisco Unified Border Element
- IOS 15.0(1)XA for Cisco 2851 Fax Gateway
- Comcast(Adtran) ESG R11.4.3.V – Provided and managed by Comcast

Features

Features Supported

- Incoming and outgoing off-net calls using G711ULaw
- Call hold
- Call transfer (unattended and attended)
- Call forward (all, busy and no answer)
- Calling Line (number) Identification Presentation (CLIP)
- Calling Line (number) Identification Restriction (CLIR)
- DTMF (RFC2833)
- Media flow-through on Cisco UBE
- Fax (G.711 pass-through)

Features Not Supported

- Cisco IP phones used in this test do not support blind transfer
- Fax (T.38) is not supported by Service Provider

Caveats

- CLID is not updated on PSTN phones for transfer (attended and unattended) OffNet PSTN scenarios. Caller ID is not updated at PSTN once transfer is completed by PBX. Cisco UBE modify PAI/PPI header and forward to network in the tested release. CISCO BUG ID: CSCuv04539.
- For testing, 911 calls were routed internally in the Comcast lab

Configuration

Configuring Cisco Unified Border Element

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 - for LAN and WAN.

```
interface GigabitEthernet0/0/0
 description Comcast CUBE1 LAN MS4 1/0/5
 ip address 10.80.11.5 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 3
 redundancy group 1 ip 10.80.11.15 exclusive
!
interface GigabitEthernet0/0/1
 description Comcast CUBE1 WAN MS4 1/0/6
 ip address 10.64.4.146 255.255.0.0
 negotiation auto
 redundancy rii 4
 redundancy group 1 ip 10.64.4.145 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 0.0.0.0 0.0.0.0
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 no supplementary-service sip handle-replaces
 fax protocol pass-through g711ulaw
 sip
  rel1xx supported "rel100"
  session refresh
  asserted-id pai
  privacy pstn
  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
fax protocol	Specifies the fax protocol
asserted-id	Specifies the type of privacy header in the outgoing SIP requests and response messages
early-offer forced	Enables SIP Delayed-Offer to Early-Offer globally
midcall-signaling passthru	Passes SIP messages from one IP leg to another IP leg

Media Passing through Cisco UBE (media flow-through vs. media flow-around)

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

```
voice service voip
    media flow-around
```

Codecs

G711ulaw is used as the preferred codec for this testing

```
    codec g711ulaw
```

Dial Peer

Cisco UBE uses dial-peers to route the call accordingly based on the digits

```
!
dial-peer voice 100 voip
 description "Outbound from PBX to Adtran"
 destination-pattern .T
 session protocol sipv2
 session target ipv4:10.64.4.140
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 codec g711ulaw
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
```

```
dial-peer voice 101 voip
 description ""inbound from PBX to Adtran"
 session protocol sipv2
 incoming called-number .T
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 codec g711ulaw
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 200 voip
 description "Inbound call from Adrtan to CUCM"
 destination-pattern 856333....
 session protocol sipv2
 session target ipv4:10.80.11.3
 voice-class sip asserted-id pai
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 codec g711ulaw
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
```

```
!
dial-peer voice 201 voip
 session protocol sipv2
 incoming called-number 856333....
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 codec g711ulaw
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
```

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 “9” 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 “9”. A “9.@” 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 Comcast, Caller dial 9 prefix followed by the target 1+10-digits number, 9 was stripped and the remaining digits were send to Cisco UBE, Cisco UBE pass the DID under Dial Peer 100 and send to Comcast network which will direct back to Cisco UBE and handled same as normal incoming PSTN call.

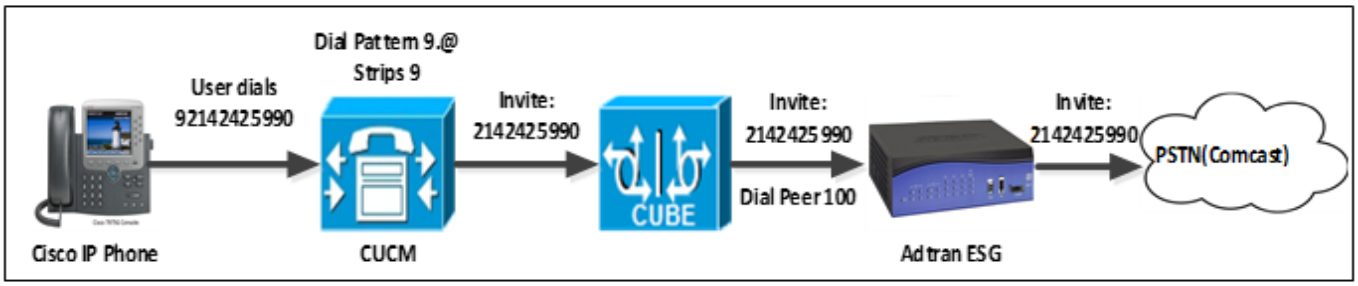

Figure 2: Outbound Voice Call

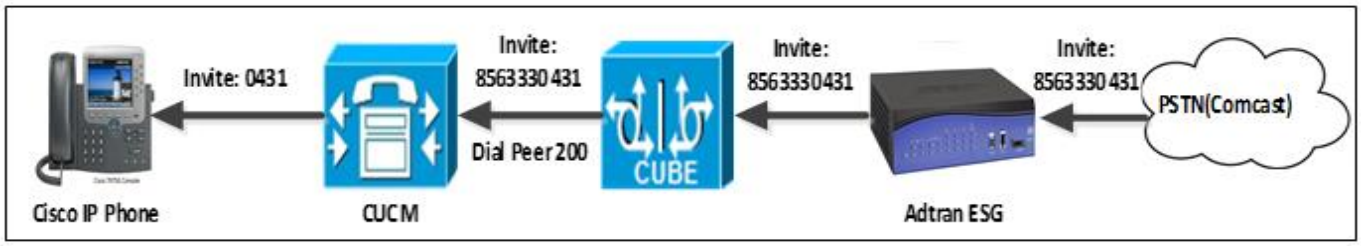

Figure 3: Inbound Voice Call

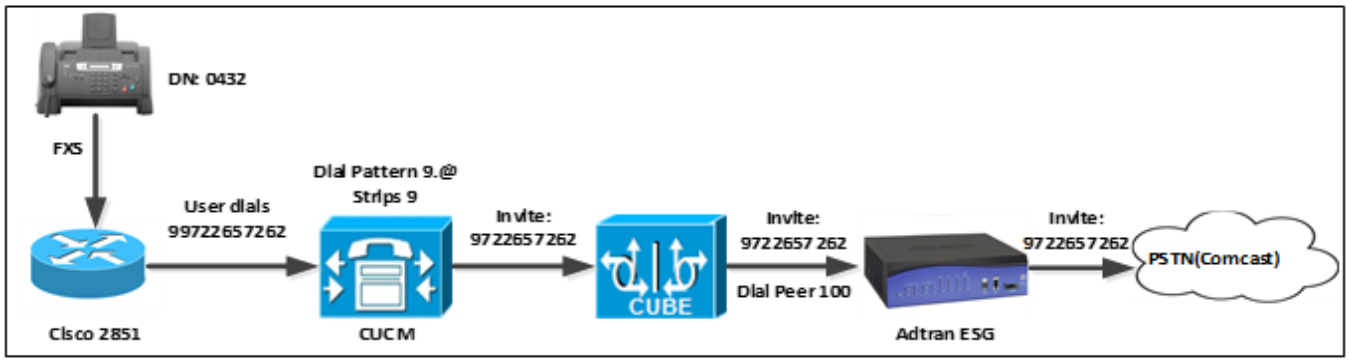

Figure 4: Outbound Fax Call

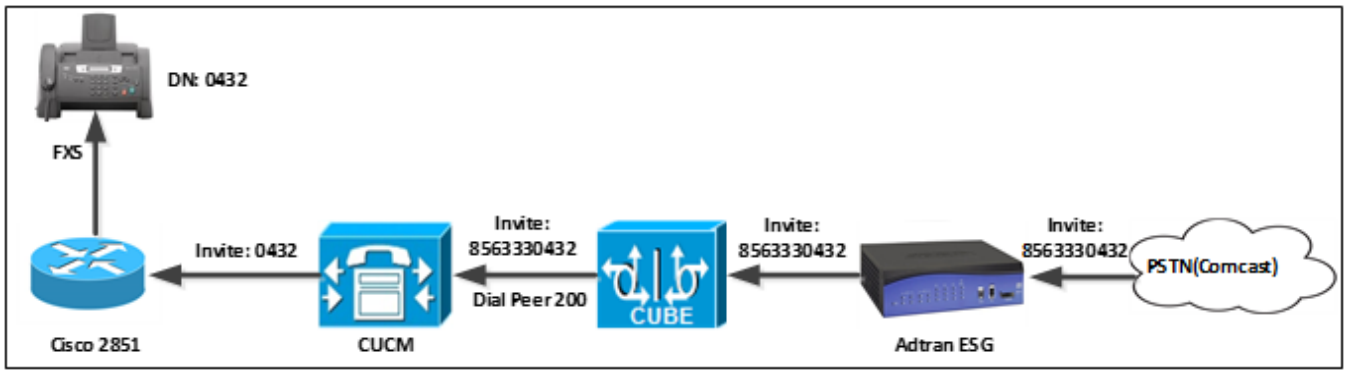

Figure 5: Inbound Fax Call

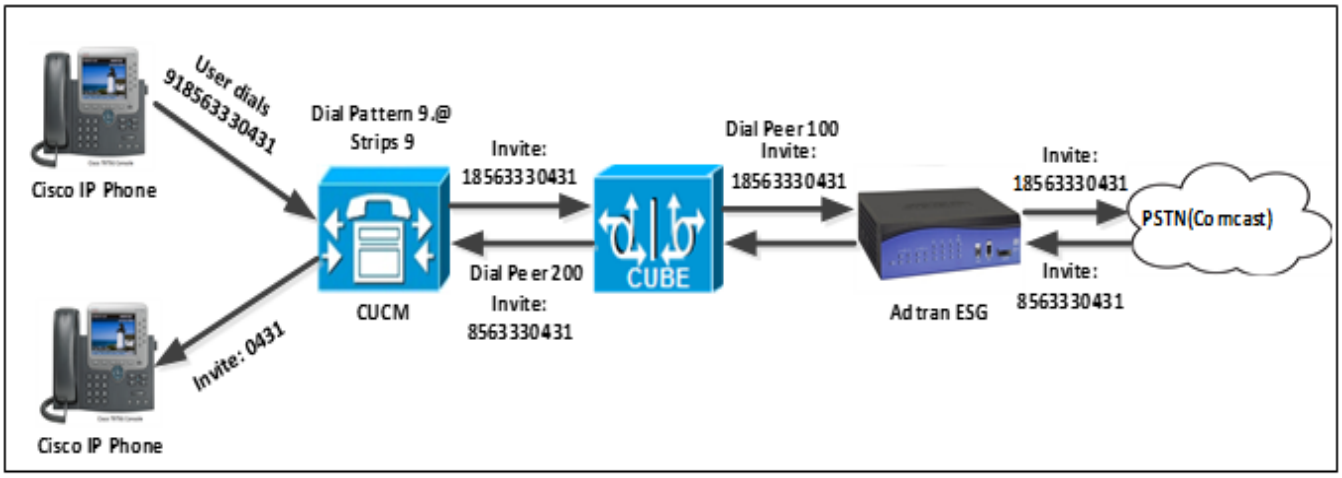

Figure 6: PBX to PBX via Comcast Call

Configuration Example

The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously

Active Cisco UBE

```
User Access Verification


Comcast_VUBE1>en
Comcast_VUBE1#sh running-config
!
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname Comcast_VUBE1
!
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
!
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
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 no supplementary-service sip handle-replaces
 fax protocol pass-through g711ulaw
 sip
  rel1xx supported "rel100"
  session refresh
  asserted-id pai
  privacy pstn
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
!
!
```

```
voice class sip-profiles 1
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:856333\1@\2>"
!
application
 group-params 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 1
   name Comcast
   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 Comcast CUBE1 LAN MS4 1/0/5
 ip address 10.80.11.5 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 3
 redundancy group 1 ip 10.80.11.15 exclusive
!
interface GigabitEthernet0/0/1
 description Comcast CUBE1 WAN MS4 1/0/6
 ip address 10.64.4.146 255.255.0.0
 negotiation auto
 redundancy rii 4
 redundancy group 1 ip 10.64.4.145 exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA MS5 3/0/35
 ip address 10.89.20.7 255.255.255.0
 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 rtcp report interval 300
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.80.11.0 255.255.255.0 10.80.11.1
ip route 172.16.24.0 255.255.248.0 10.80.11.1
ip route 172.16.31.0 255.255.255.0 10.80.11.1
!
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 "Outbound from PBX to Adtran"
 destination-pattern .T
 session protocol sipv2
 session target ipv4:10.64.4.140
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 codec g711ulaw
 fax-relay ecm disable
 fax rate disable
```

```
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 101 voip
 description ""inbound from PBX to Adtran"
 session protocol sipv2
 incoming called-number .T
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 codec g711ulaw
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
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dial-peer voice 200 voip
 description "Inbound call from Adrtan to CUCM"
 destination-pattern 856333....
 session protocol sipv2
 session target ipv4:10.80.11.3
 voice-class sip asserted-id pai
 voice-class sip bind control source-interface GigabitEthernet0/0/0
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 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 codec g711ulaw
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
!
gateway
 timer receive-rtp 1200
!
sip-ua
 protocol mode ipv4
!
!
line con 0
 stopbits 1
line aux 0
```

```
 stopbits 1
line vty 0
 password
 transport input telnet ssh
line vty 1
 exec-timeout 0 0
 password
 transport input telnet ssh
line vty 2 4
 password
 transport input telnet ssh
!
!
end
```

Standby Cisco UBE

```
Comcast_CUBE2#sh running-config
!
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname Comcast_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
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 no supplementary-service sip handle-replaces
 fax protocol pass-through g711ulaw
 sip
  rel1xx supported "rel100"
  session refresh
  asserted-id pai
  privacy pstn
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
!
!
voice class sip-profiles 1
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:856333\1@\2>"
!
license udi pid ISR4321/K9 sn FDO19220MQ9
!
spanning-tree extend system-id
!
```

```
redundancy
 mode none
 application redundancy
  group 2
   name Comcast
   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 Comcast CUBE2 LAN MS4 1/0/11
 ip address 10.80.11.6 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 5
 redundancy group 2 ip 10.80.11.15 exclusive
!
interface GigabitEthernet0/0/1
 description Comcast CUBE2 WAN MS4 1/0/12
 ip address 10.64.4.147 255.255.255.128
 negotiation auto
 redundancy rii 6
 redundancy group 2 ip 10.64.4.145 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 300
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.80.11.0 255.255.255.0 10.80.11.1
ip route 172.16.24.0 255.255.248.0 10.80.11.1
ip route 172.16.31.0 255.255.255.0 10.80.11.1
!
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 "Outbound from PBX to Adtran"
 destination-pattern .T
 session protocol sipv2
 session target ipv4:10.64.4.140
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 codec g711ulaw
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 101 voip
 description "inbound from PBX to Adtran"
 session protocol sipv2
 incoming called-number .T
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 codec g711ulaw
 fax-relay ecm disable
```

```
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 200 voip
 description "Inbound call from Adrtan to CUCM"
 destination-pattern 856333....
 session protocol sipv2
 session target ipv4:10.80.11.3
 voice-class sip asserted-id pai
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 codec g711ulaw
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 201 voip
 session protocol sipv2
 incoming called-number 856333....
 voice-class sip asserted-id pai
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 codec g711ulaw
 fax-relay ecm disable
```

```
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad

!
!
sip-ua
protocol mode ipv4
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0
 password
 transport input telnet ssh
line vty 1
 exec-timeout 0 0
 password
 transport input telnet ssh
line vty 2 4
 password
 transport input telnet ssh
!
!
end
```

Configuring Cisco Unified Communications Manager

Cisco UCM Version

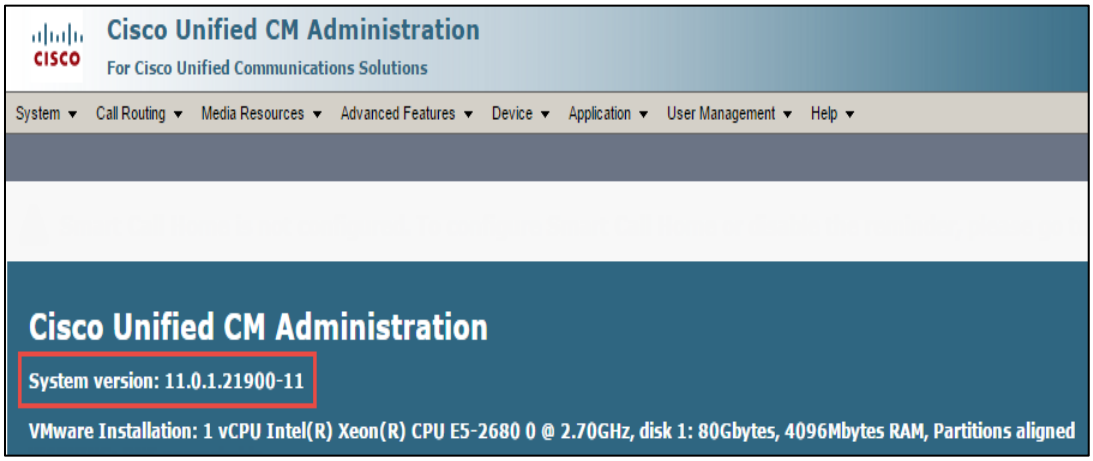

Figure 7: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation Path: System > Service Parameters

Select Server* = Clus21sub1--CUCM Voice/Video (Active)

Select Service*= Cisco CallManager (Active)

All other fields are set to default values

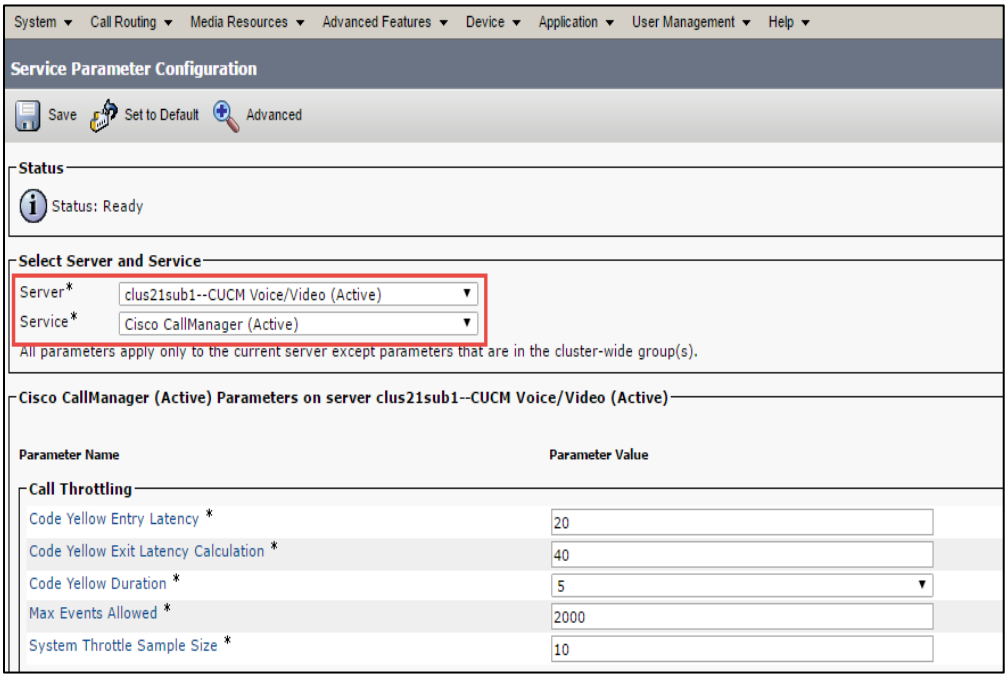

Figure 8: Service Parameters

Offnet Calls via Comcast SIP Trunk

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

SIP Trunk Security Profile

Navigation Path: System > Security > SIP Trunk Security Profile

Name*= Comcast Non Secure SIP Trunk Profile

Description = Non Secure SIP Trunk Profile

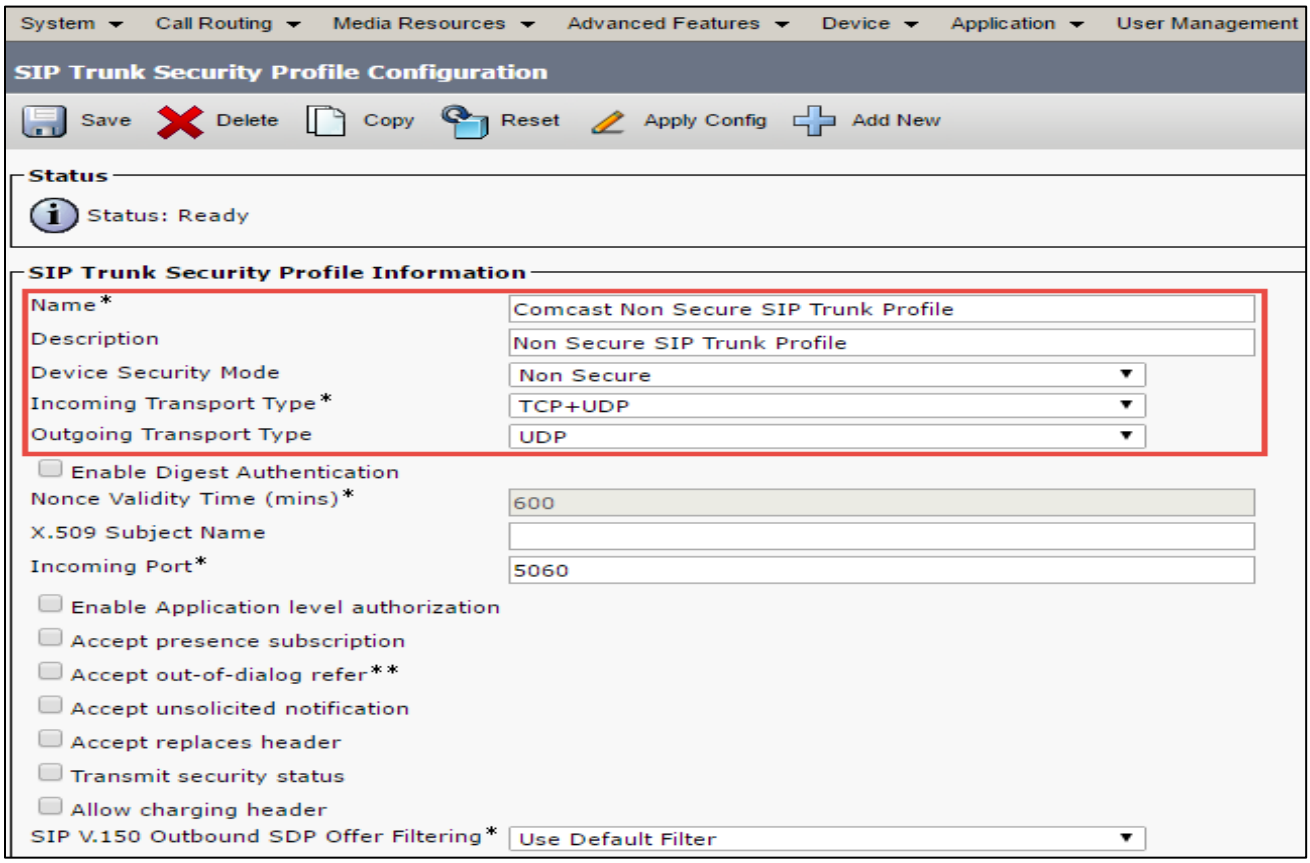

Figure 9: SIP Trunk Security Profile

Explanation

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

SIP Profile will be later associated with the SIP trunk

Navigation Path: Device > Device Settings > SIP Profile

Name*= Standard SIP Profile_Comcast

Description = Default SIP Profile

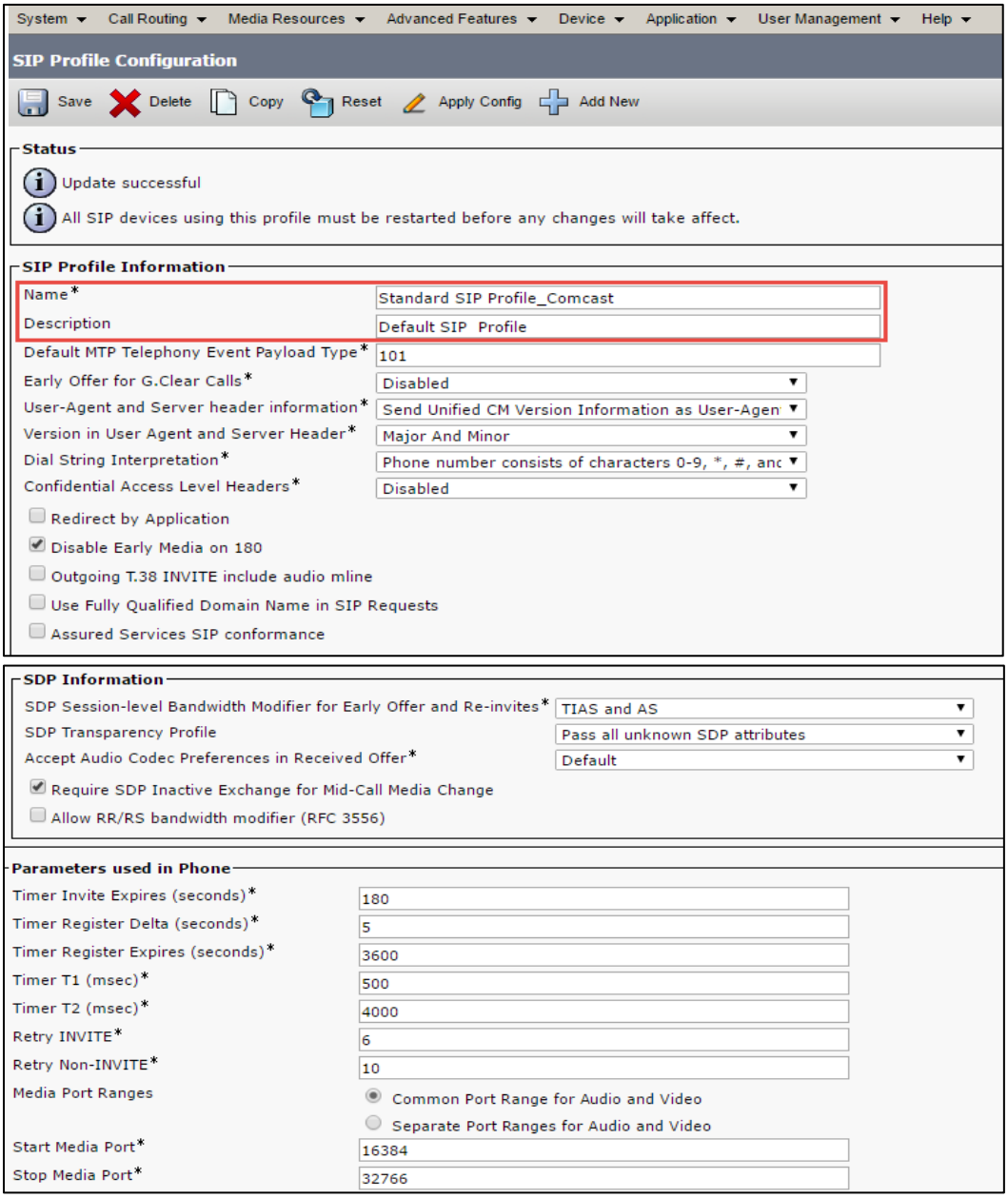

Figure 10: SIP Profile

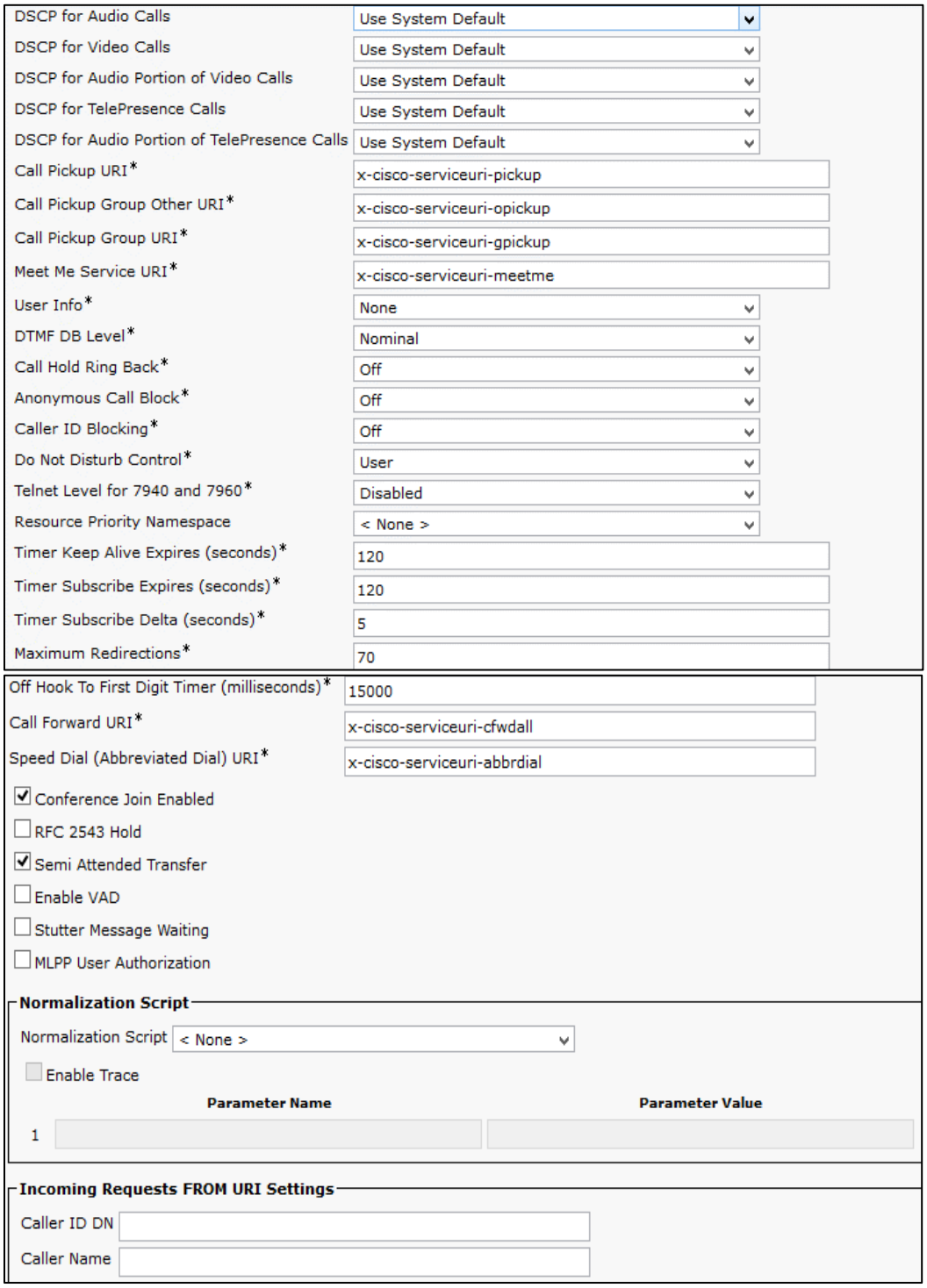

Figure 11: SIP Profile (Cont.)

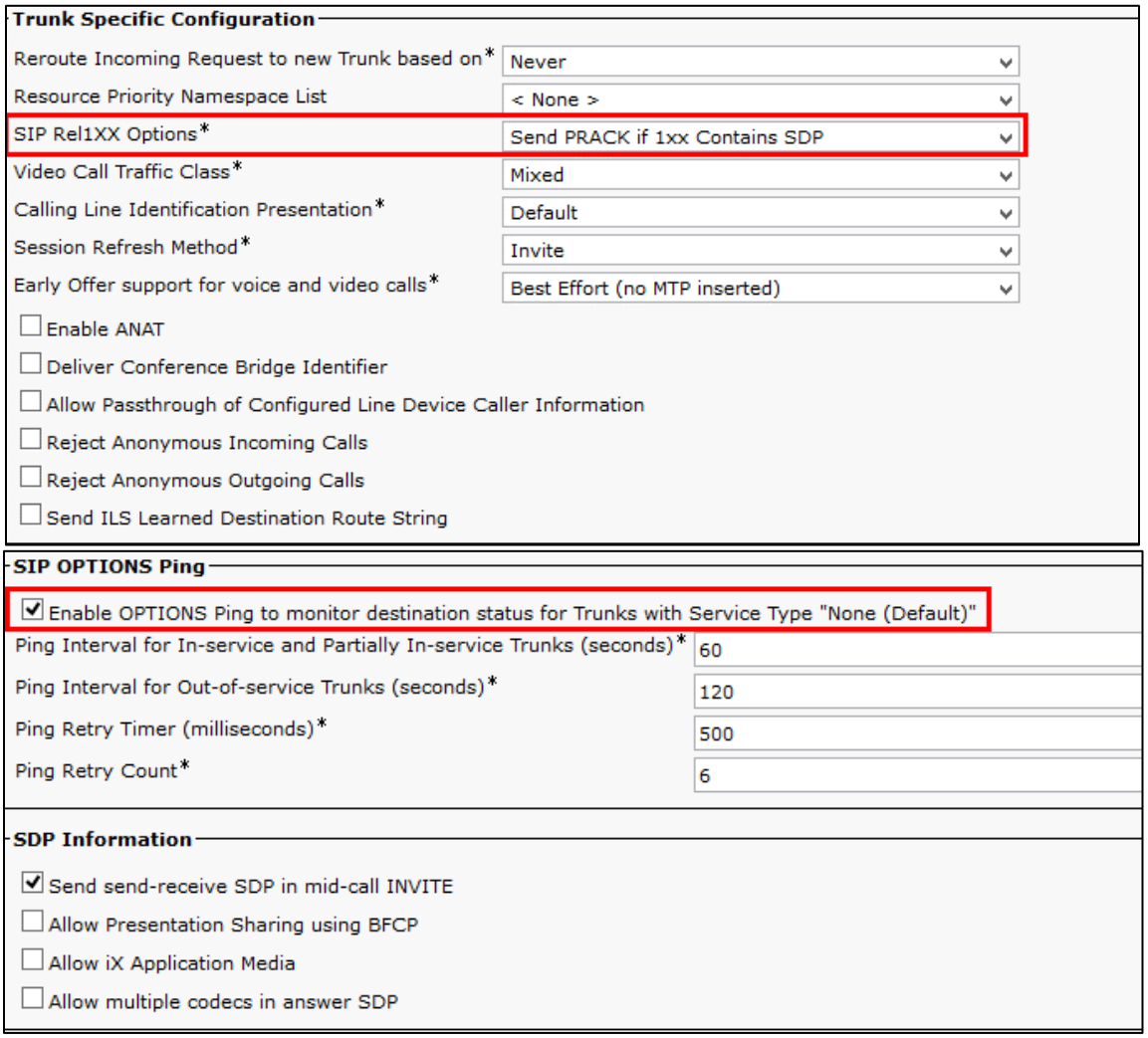

Figure 12: SIP Profile (Cont.)

Explanation

Parameter	Value	Description
Default MTP Telephony Event Payload Type	101	RFC2833 DTMF payload type
SIP Rel1XX Options	Send PRACK for 1xx Messages	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 Path: Device > Trunk

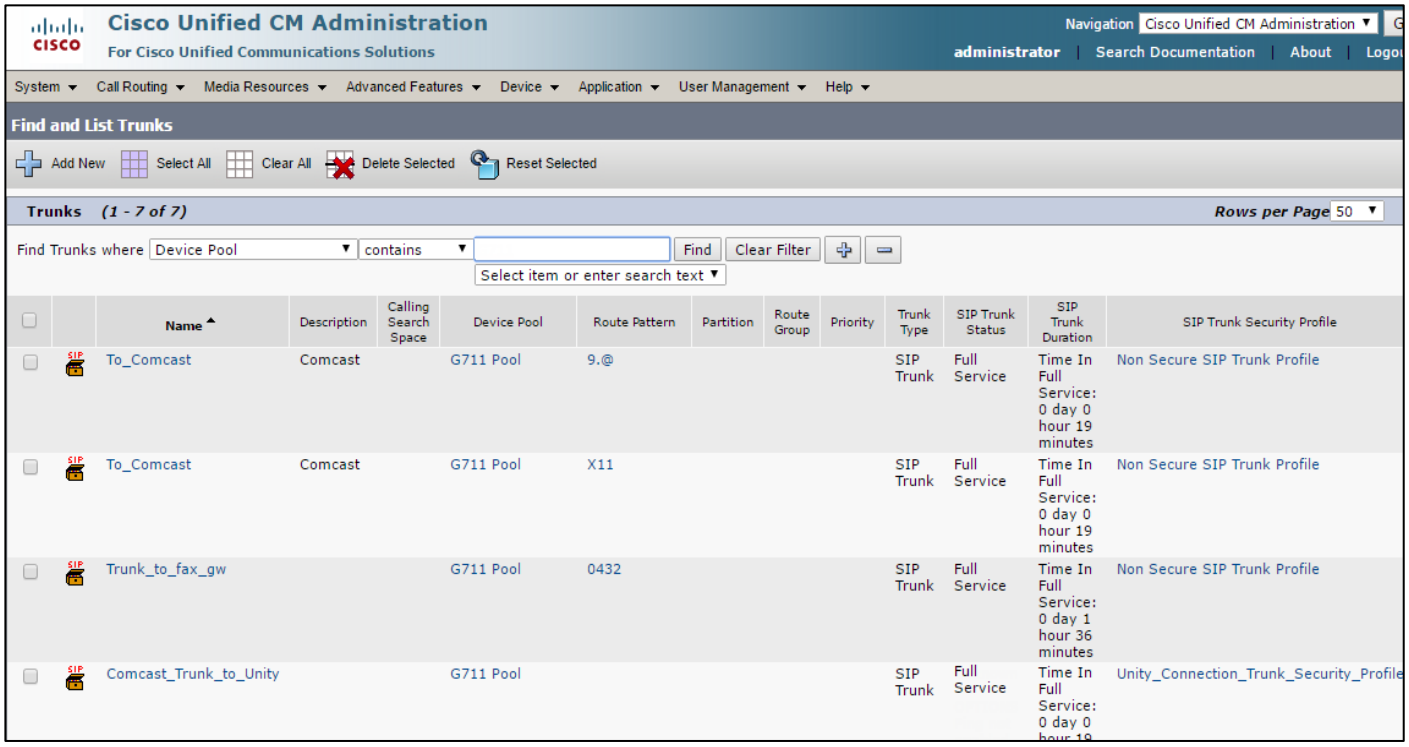

Figure 13: SIP Trunks List

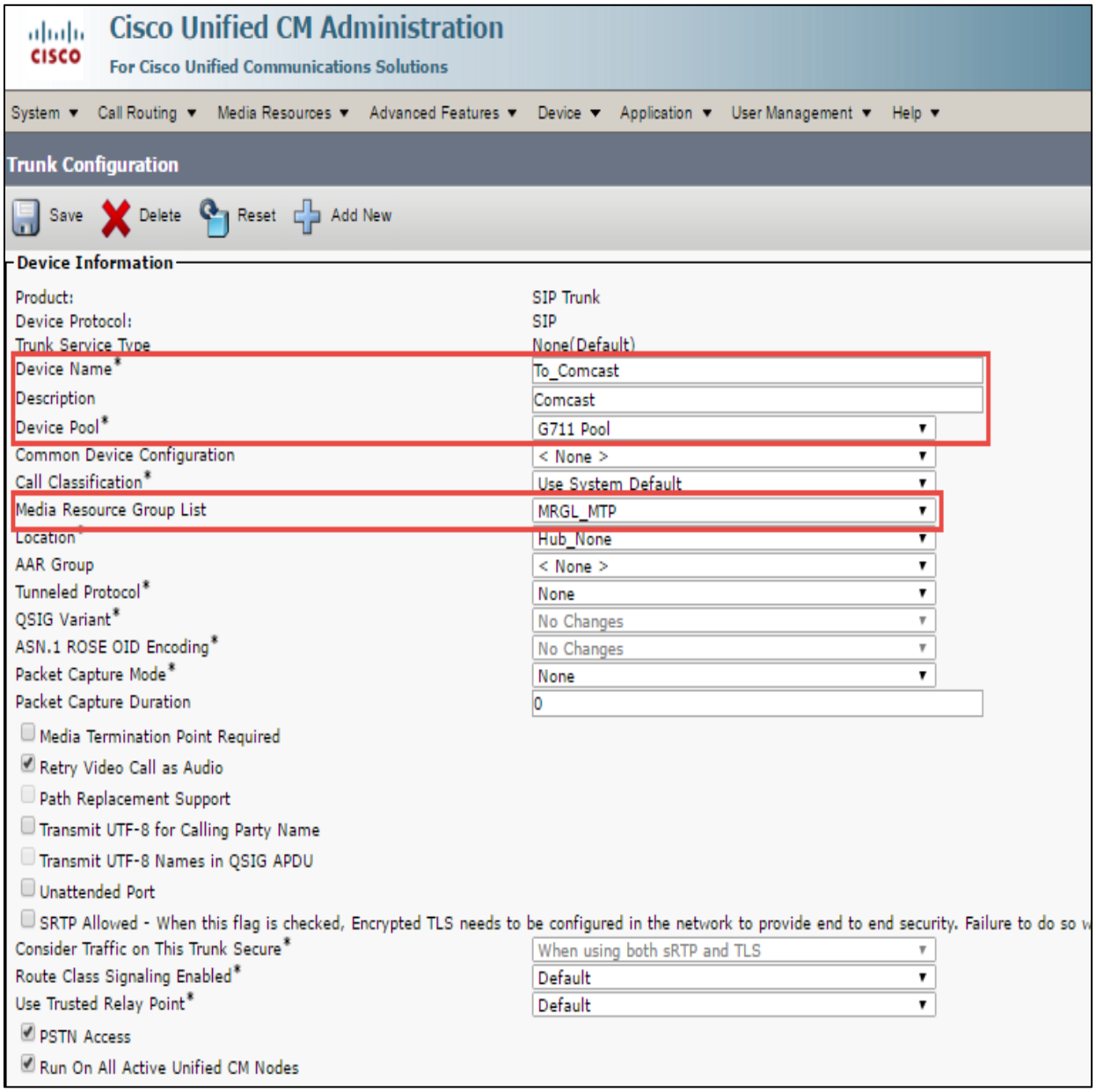

Figure 14: SIP Trunk to Cisco UBE

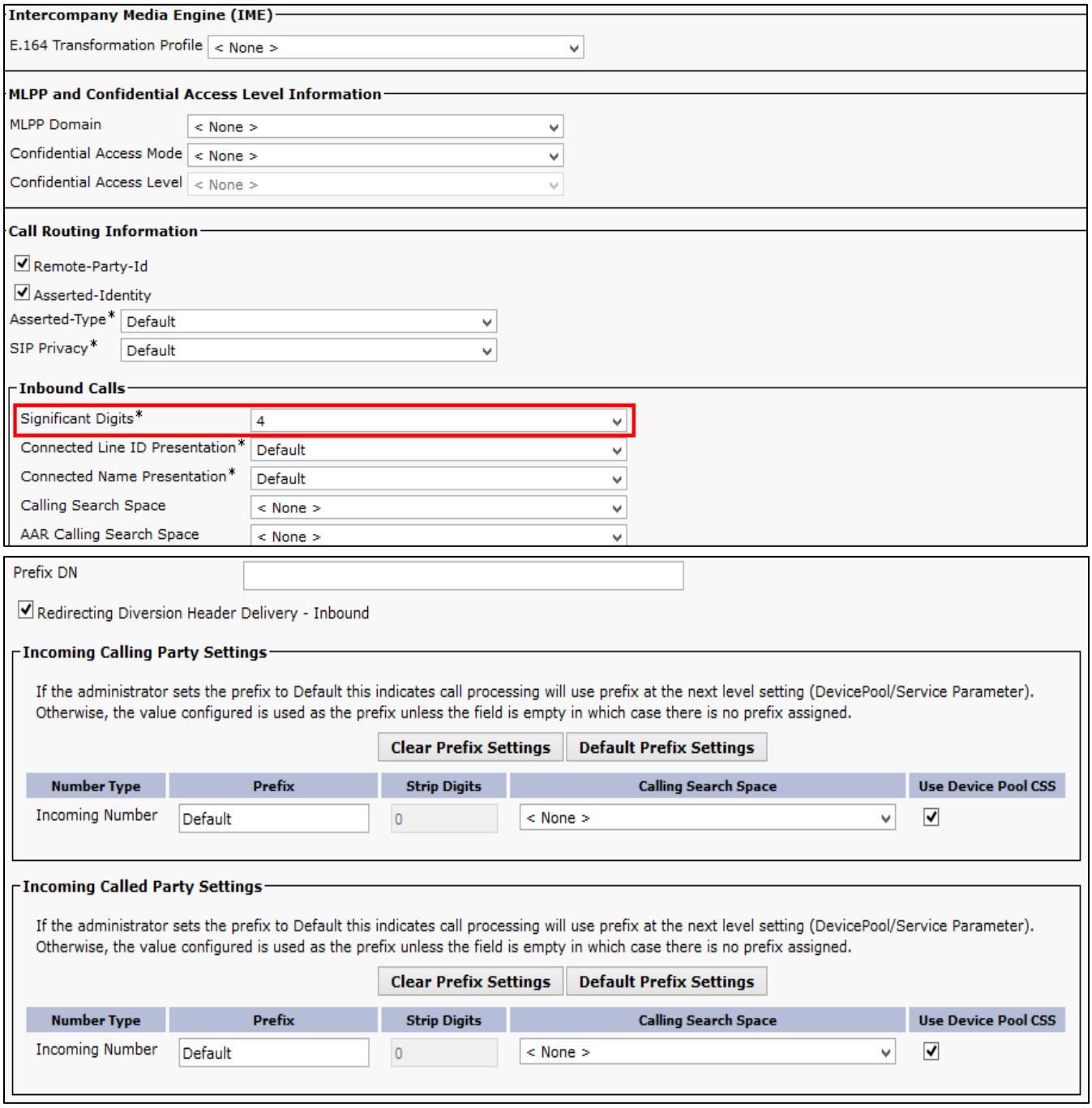

Figure 15: SIP Trunk to Cisco UBE (Cont.)

Connected Party Settings

Connected Party Transformation CSS: < None >

☑ Use Device Pool Connected Party Transformation CSS

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*: Originator

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*	10.80.11.15		5060	up

MTP Preferred Originating Codec*: 711ulaw

BLF Presence Group*: Standard Presence group

SIP Trunk Security Profile*: Comcast 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*: Standard SIP Profile_Comcast 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 16: SIP Trunk to Cisco UBE (Cont.)

Explanation

Parameter	Value	Description
Device Name	To_Comcast	Name for the trunk
Device Pool	G711_pool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_Default	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.11.15	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	Comcast Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	Comcast Standard SIP Profile	SIP Profile configured earlier

Dial Plan

Route Pattern Configuration

Navigation Path: Call Routing > Route/Hunt > Route Pattern

Route patterns are configured as below:

- Cisco IP phones dial “9”+10 digits number to access PSTN via Cisco UBE
  - “9” is removed before sending to Cisco UBE
- For FAX calls, Access Code “9”+10 digits number is used at Cisco Fax gateway
  - “9” is removed at Cisco UCM
  - The rest of the number is sent to Cisco UBE to Comcast network
- Incoming fax calls to 0432 will be sent to Cisco Fax gateway
- Cisco IP phones dial 4XX and 9XX for emergency call and will send all digits from Cisco UBE to Comcast network

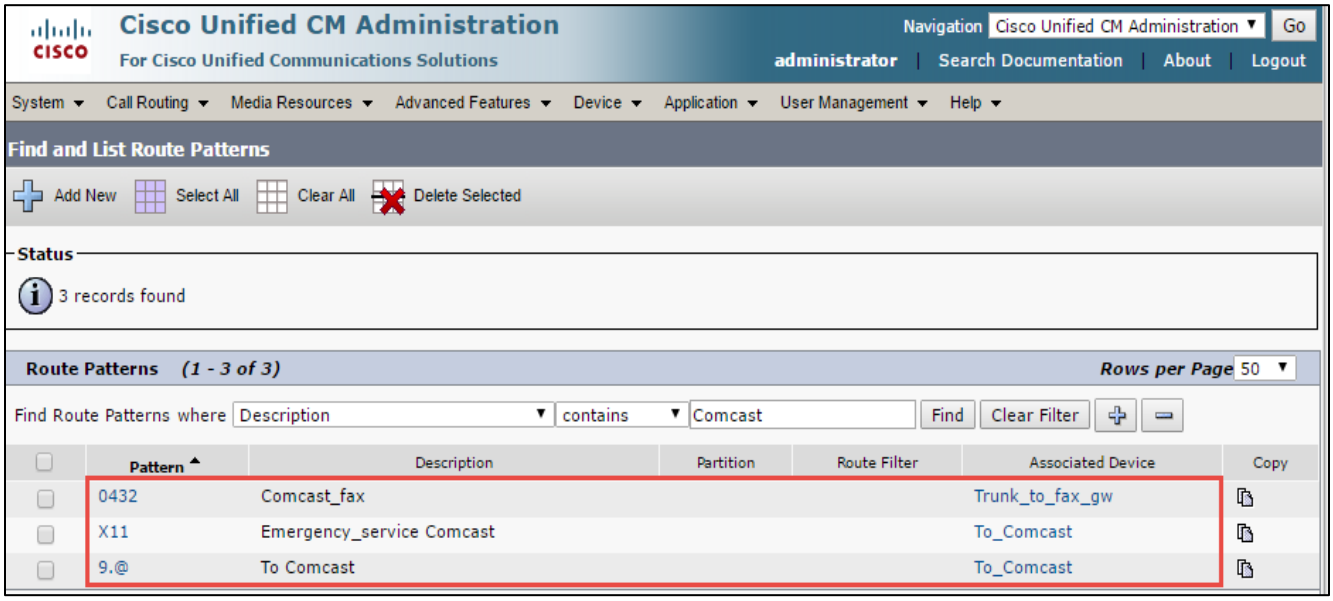

Figure 17: Route Patterns List

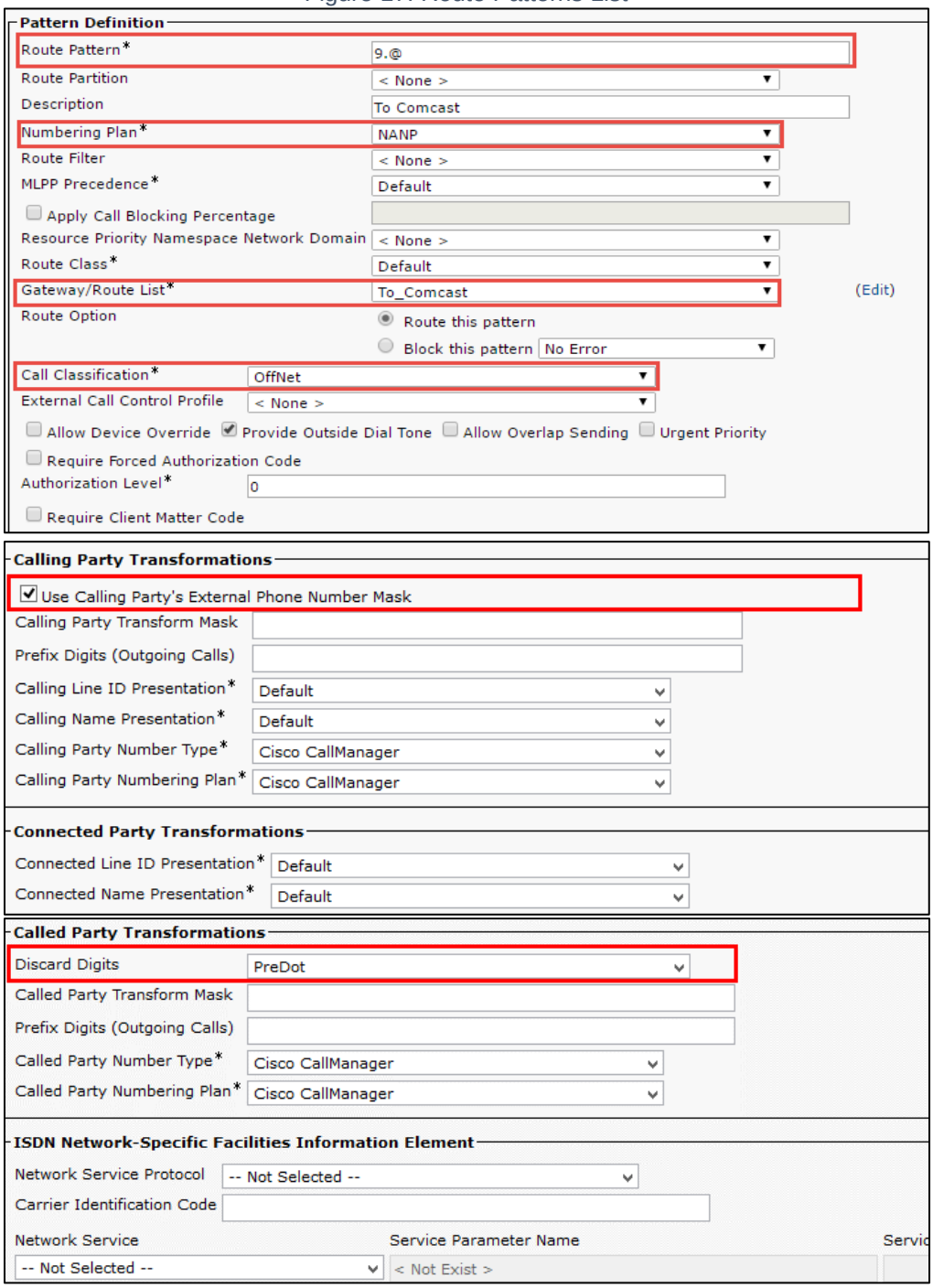

Figure 18: Route Pattern for Voice

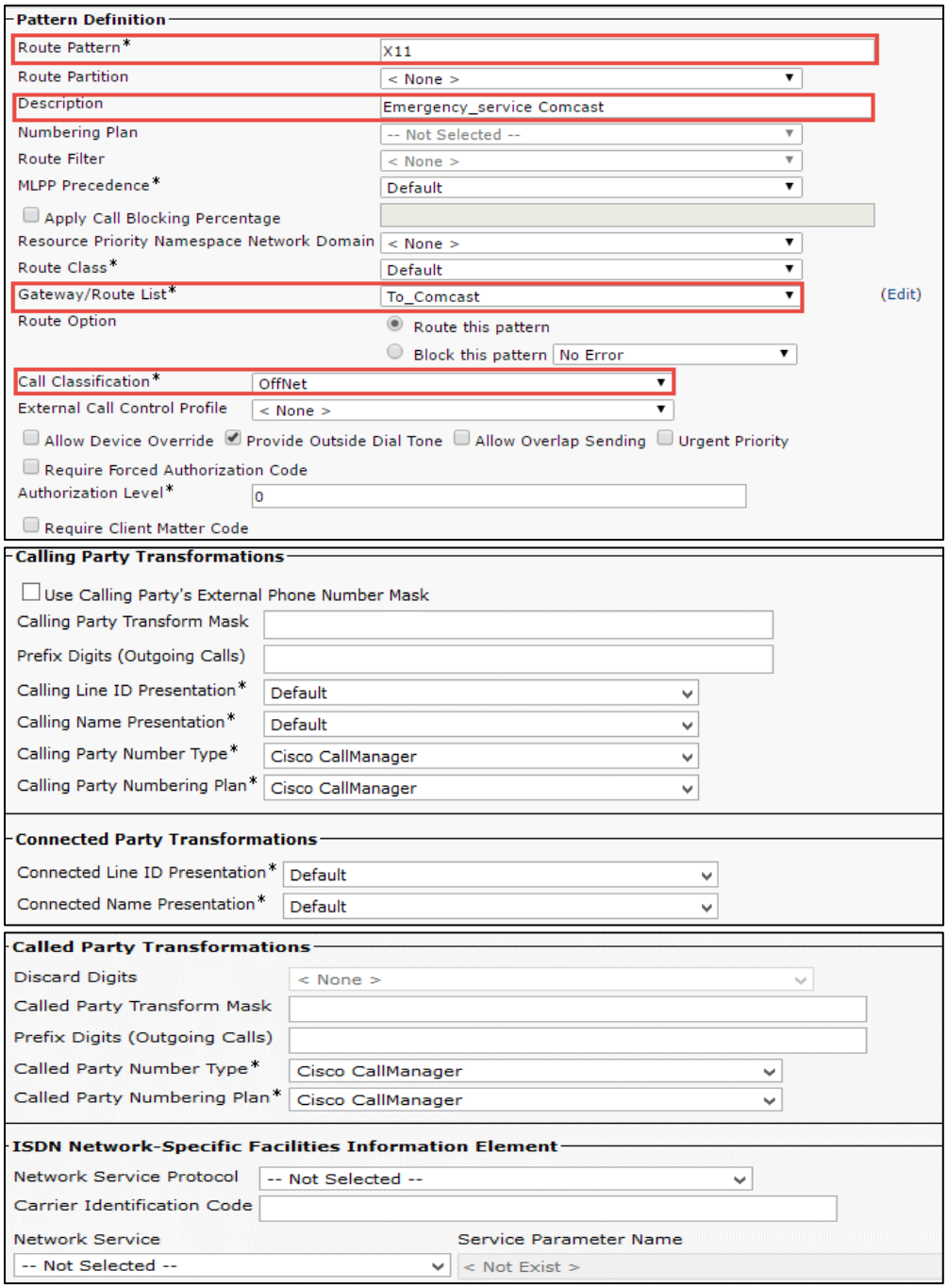

Figure 19: Route Pattern for Voice (Cont.)

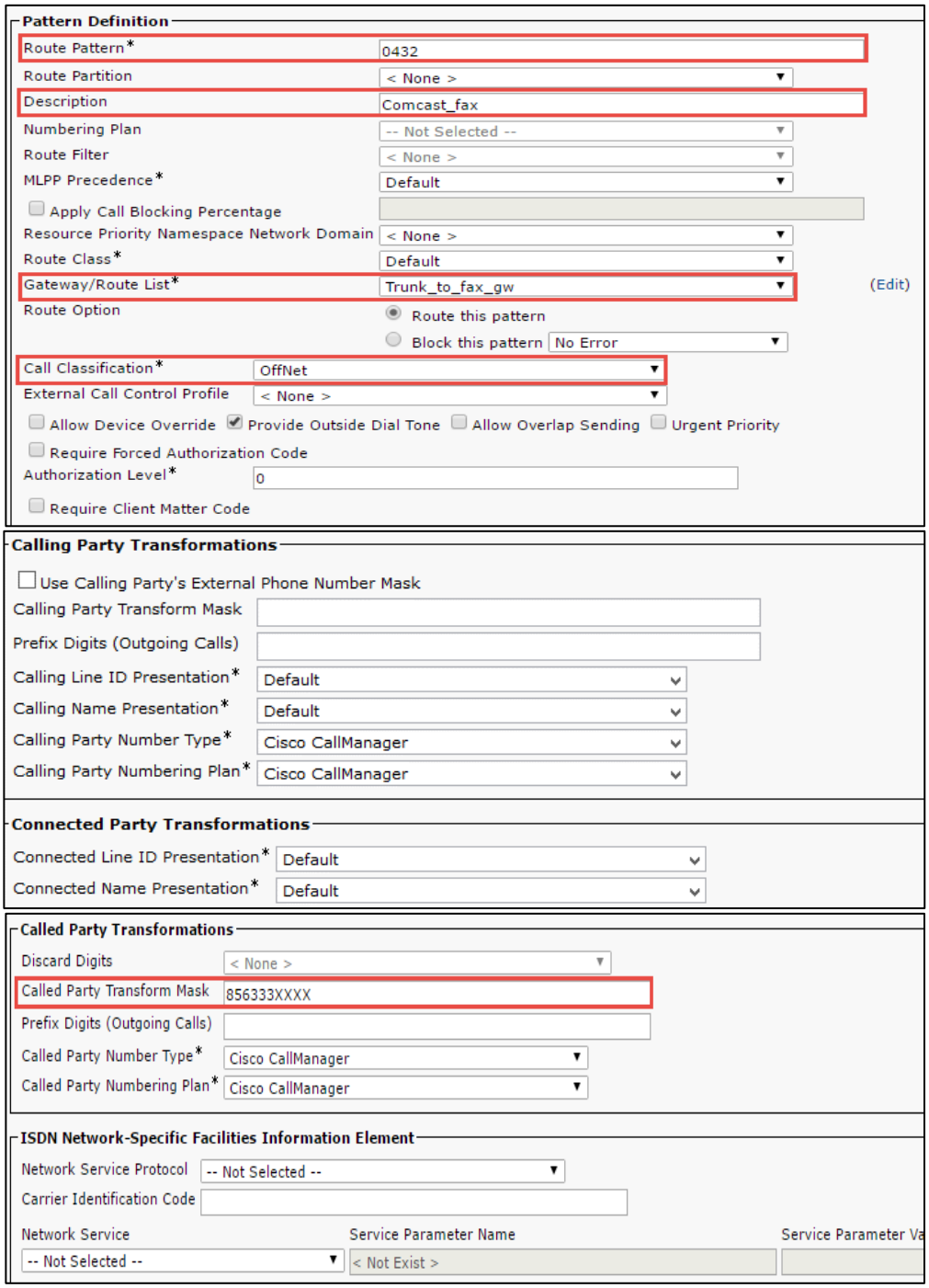

Figure 20: Route Pattern for Fax

Explanation

Setting	Value	Description
Route Pattern	9.@ for Voice & International Calls, 0432 for fax call and X11 for operator call and emergency services	Specify appropriate Route Pattern
Gateway/Route List	To_Comcast for Route Pattern 9.@, X11and Trunk_to_fax_gw for Route Pattern 0432	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 9.@	North American Numbering Plan
Call Classification	Offnet for Route Pattern 9.@, 0432 and X11	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 9.@	Specifies how to modify digit before they are sending to Comcast network

Acronyms

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

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