Application Note

Comcast SIP Trunking:

Cisco Unified Communications Manager 12.0.1 with Cisco Unified Border Element (CUBE 12.0) on ISR 4321 [IOS-XE – 16.06.02] using SIP

July 26, 2018

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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-XE 16.06.02 can be used. The Cisco Unified Border Element 12.0 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 12.0.1 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) 12.0.1 and Cisco Unified Border Element (Cisco UBE) on ISR 4321/K9 [IOS-XE – 16.06.02] for connectivity to Comcast SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 12.0.1) 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

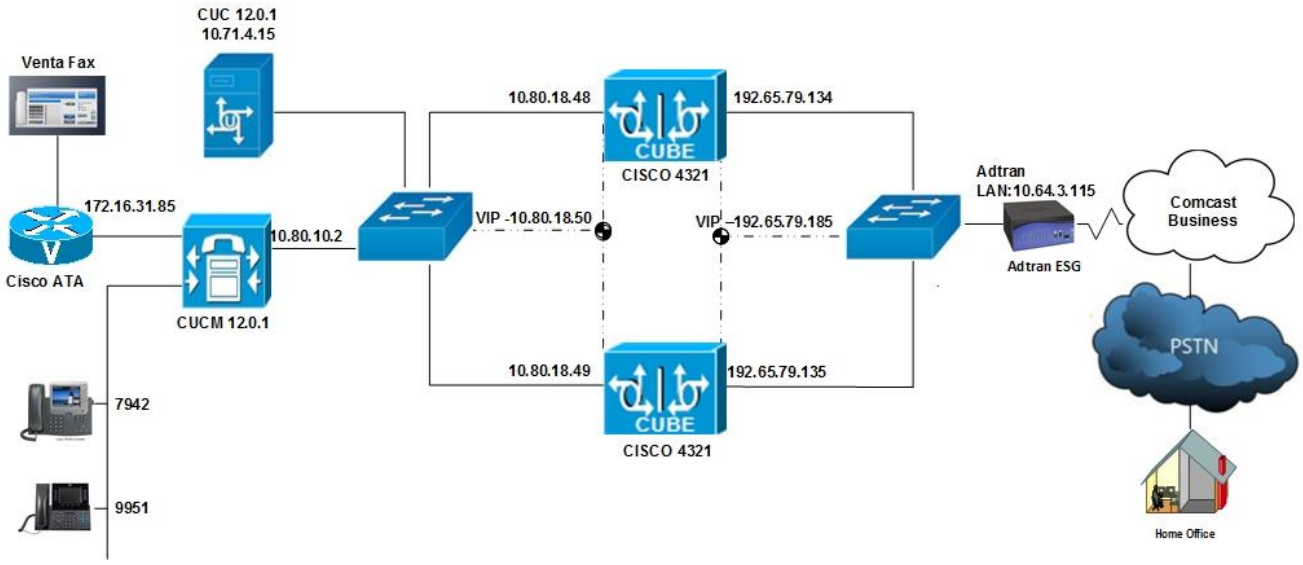

Figure 1: Network Topology

- Cisco IP Phones 7942, 9951 and 9971 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 (Cisco ATA) 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 ATA SPA112
- IP phones 9951 (SIP), 9971 (SIP) and 7942 (SCCP)
- Comcast(Adtran) ESG – Provided and managed by Comcast

Software Requirements

- Cisco Unified Communications Manager 12.0.1
- Cisco Unity Connection 12.0.1
- IOS-XE 16.06.02 for ISR 4321/K9 Cisco Unified Border Element
- Firmware Version 1.3.5 (004p) for Cisco ATA SPA112
- 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
- Comcast does not support Fax with SIP service

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 ComcastCube WAN
 ip address 192.65.79.134 255.255.255.128
 shutdown
 media-type rj45
 negotiation auto
 redundancy rii 15
 redundancy group 1 ip 192.65.79.185 exclusive
!
interface GigabitEthernet0/0/1
 description ComcastCube LAN
 ip address 10.80.18.48 255.255.255.0
 negotiation auto
 redundancy rii 16
 redundancy group 1 ip 10.80.18.50 exclusive
!
!
```

Global Cisco UBE Settings

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

```
!
!
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 no supplementary-service sip refer
 supplementary-service media-renegotiate
 sip
  session refresh
  asserted-id pai
  privacy pstn
  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
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.

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

Dial Peer

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

```
!
dial-peer voice 100 voip
 description Outbound from CUCM to CUBE LAN
 session protocol sipv2
 session transport udp
 incoming called-number .T
 voice-class codec 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
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 101 voip
 description Outbound from CUBE WAN to Adtran
 destination-pattern .T
 session protocol sipv2
 session target ipv4:10.64.3.115
```

```
 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
 fax protocol pass-through g711ulaw
 no vad
 !
dial-peer voice 102 voip
 description Incoming from Adtran to CUBE WAN
 session protocol sipv2
 incoming called-number 856333....
 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
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 103 voip
 description Incoming from CUBE LAN to CUCM
 destination-pattern 856333....
 session protocol sipv2
 session target ipv4:10.80.10.2
 voice-class codec 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
 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 101 and send to Comcast network which will direct back to Cisco UBE and handled same as normal incoming PSTN call.

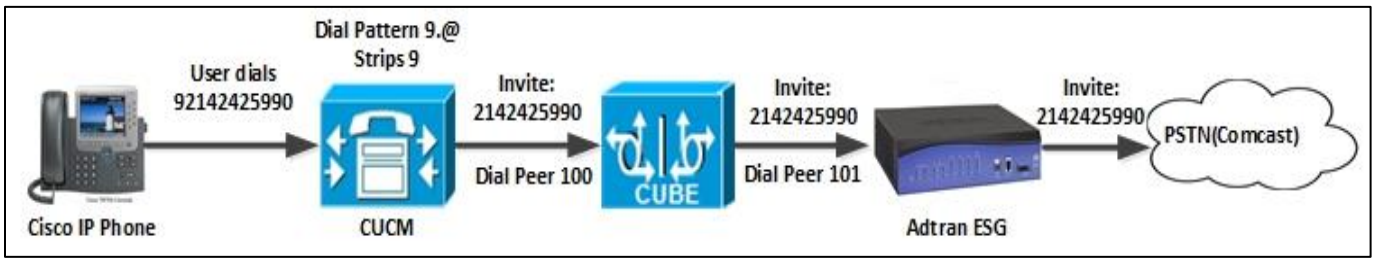

Figure 2: Outbound Voice Call

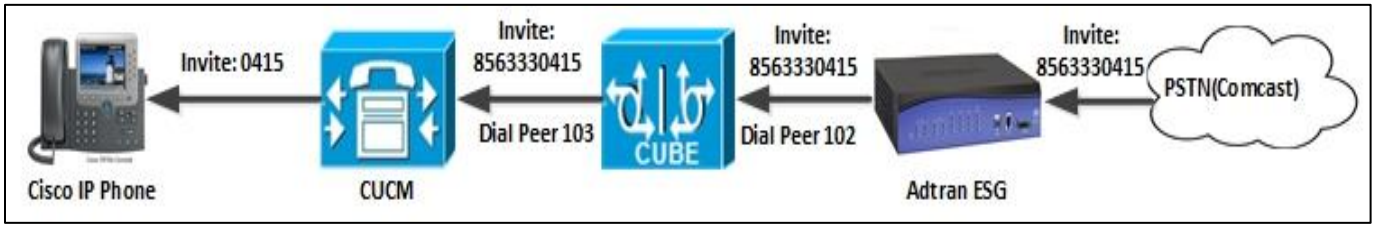

Figure 3: Inbound Voice Call

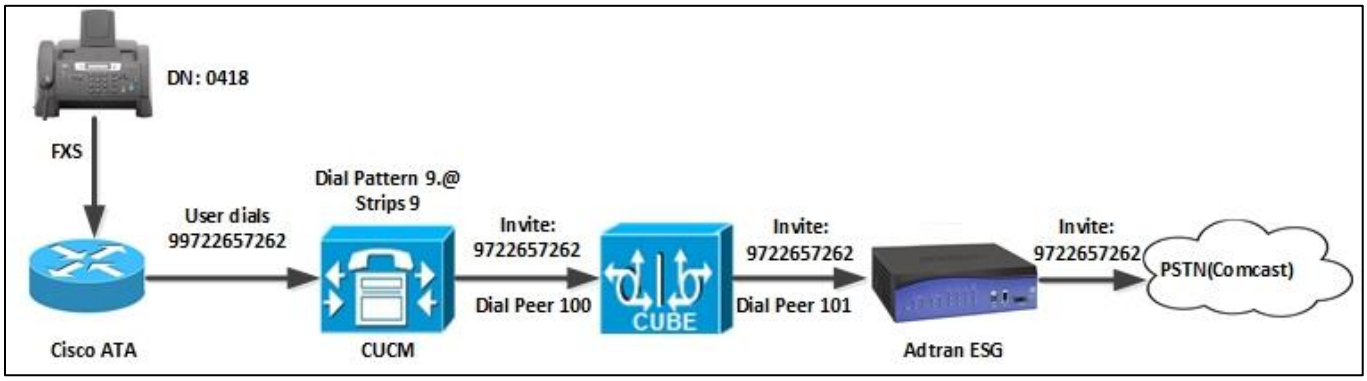

Figure 4: Outbound Fax Call

Figure 5: Inbound Fax Call

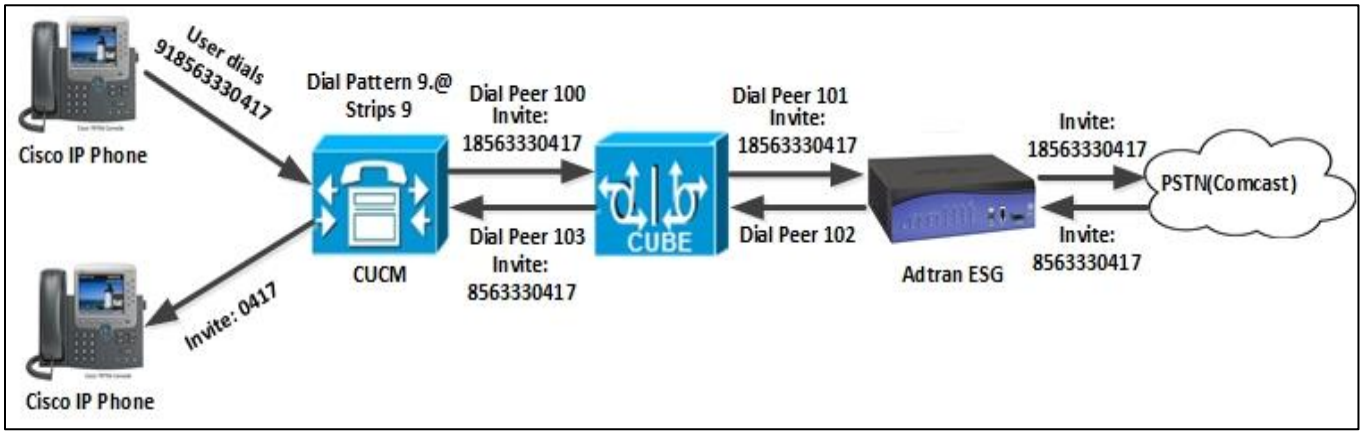

Figure 6: PBX to PBX via Comcast Call

Configuration Example

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

Active Cisco UBE

```
ComcastCube1>en
Password:
ComcastCube1#show running-config
!
version 16.6
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname ComcastCube1
!
boot-start-marker
boot system flash isr4300-universalk9.16.06.02.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
no logging queue-limit
logging buffered 10000000
```

```
no logging rate-limit
no logging monitor
enable secret 5 $1
!
aaa new-model
!
aaa session-id common
!
multilink bundle-name authenticated
!
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 no supplementary-service sip refer
 supplementary-service media-renegotiate
 sip
  session refresh
  asserted-id pai
  privacy pstn
  midcall-signaling passthru
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
license udi pid ISR4321/K9 sn FDO19244MQ8
diagnostic bootup level minimal
```

```
spanning-tree extend system-id
!
!
redundancy
 mode none
 application redundancy
  group 1
   name b2bHAcomcast
   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
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
!
interface GigabitEthernet0/0/0
 description ComcastCube WAN
 ip address 192.65.79.134 255.255.255.128
 shutdown
 media-type rj45
 negotiation auto
 redundancy rii 15
 redundancy group 1 ip 192.65.79.185 exclusive
!
interface GigabitEthernet0/0/1
 description ComcastCube LAN
```

```
 ip address 10.80.18.48 255.255.255.0
 negotiation auto
 redundancy rii 16
 redundancy group 1 ip 10.80.18.50 exclusive
!
interface GigabitEthernet0/1/0
 description ComcastCube HA Interface
 ip address 10.70.50.100 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 shutdown
 negotiation auto
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.70.50.0 255.255.255.0 10.80.18.1
ip route 10.80.10.0 255.255.255.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 10.80.18.1
!
!
!
!
!
!
!
```

```
!
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 CUCM to CUBE LAN
 session protocol sipv2
 session transport udp
 incoming called-number .T
 voice-class codec 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
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 101 voip
 description Outbound from CUBE WAN to Adtran
 destination-pattern .T
 session protocol sipv2
 session target ipv4:10.64.3.115
```

```
 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
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 102 voip
 description Incoming from Adtran to CUBE WAN
 session protocol sipv2
 incoming called-number 856333....
 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
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 103 voip
 description Incoming from CUBE LAN to CUCM
 destination-pattern 856333....
 session protocol sipv2
 session target ipv4:10.80.10.2
 voice-class codec 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
 fax protocol pass-through g711ulaw
 no vad
!
```

```
!
gateway
 timer receive-rtp 1200
!
sip-ua
!
!
line con 0
 password password
 transport input none
 stopbits 1
line aux 0
 password password
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password password
 transport preferred telnet
!
wsma agent exec
!
wsma agent config
!
wsma agent filesys
!
wsma agent notify
!
end

ComcastCube1#
```

Standby Cisco UBE

```
ComcastCube2#show running-config
!
version 16.6
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
service password-encryption
service internal
service sequence-numbers
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname ComcastCube2
!
boot-start-marker
boot system flash isr4300-universalk9.16.06.02.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
no logging queue-limit
```

```
logging buffered 30000000
no logging rate-limit
no logging console
logging monitor notifications
enable secret 5 $1$aEKr$TYnj4bdYn.
!
no aaa new-model
subscriber templating
!
multilink bundle-name authenticated
!
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 no supplementary-service sip refer
 supplementary-service media-renegotiate
 sip
  session refresh
  privacy pstn
  conn-reuse
  midcall-signaling passthru
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
```

```
!
license udi pid ISR4321/K9 sn FDO19440MW3
license boot level appxk9
license boot level uck9
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 7 13111A
!
redundancy
 mode none
 application redundancy
  group 1
   name b2bHAcomcast
   priority 150 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/1/0 protocol 1
   data GigabitEthernet0/1/0
   track 1 shutdown
   track 2 shutdown
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
!
!
!
!
```

```
interface GigabitEthernet0/0/0
 description ComcastCube WAN
 ip address 192.65.79.135 255.255.255.128
 media-type rj45
 negotiation auto
 redundancy rii 15
 redundancy group 1 ip 192.65.79.185 exclusive
!
interface GigabitEthernet0/0/1
 description ComcastCube LAN
 ip address 10.80.18.49 255.255.255.0
 negotiation auto
 redundancy rii 16
 redundancy group 1 ip 10.80.18.50 exclusive
!
interface GigabitEthernet0/1/0
 description ComcastCube HA Interface
 ip address 10.70.50.110 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 shutdown
 negotiation auto
!
ip forward-protocol nd
no ip http server
```

```
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.70.50.0 255.255.255.0 10.80.18.1
ip route 10.80.10.0 255.255.255.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 10.80.18.1
!
!
!
ip access-list standard my_access_list
 permit any
!
!
!
!
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
!
!
!
```

```
!
telephony-service
 max-conferences 8 gain -6
 transfer-system full-consult
!
!
dial-peer voice 100 voip
 description Outbound from CUCM to CUBE LAN
 session protocol sipv2
 session transport udp
 incoming called-number .T
 voice-class codec 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
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 101 voip
 description Outbound from CUBE WAN to Adtran
 destination-pattern .T
 session protocol sipv2
 session target ipv4:10.64.3.115
 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
```

```
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 102 voip
 description Incoming from Adtran to CUBE WAN
 session protocol sipv2
 incoming called-number 856333....
 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
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 103 voip
 description Incoming from CUBE LAN to CUCM
 destination-pattern 856333....
 session protocol sipv2
 session target ipv4:10.80.10.2
 voice-class codec 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
 fax protocol pass-through g711ulaw
 no vad
!
!
gateway
```

```
 timer receive-rtp 1200
!
!
line con 0
 exec-timeout 0 0
 login local
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 login local
 transport input telnet ssh
!
wsma agent exec
!
wsma agent config
!
wsma agent filesys
!
wsma agent notify
!
!
end

ComcastCube2#
```

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* = Clus20pub1--CUCM Voice/Video (Active)

Select Service*= Cisco CallManager (Active)

All other fields are set to default values

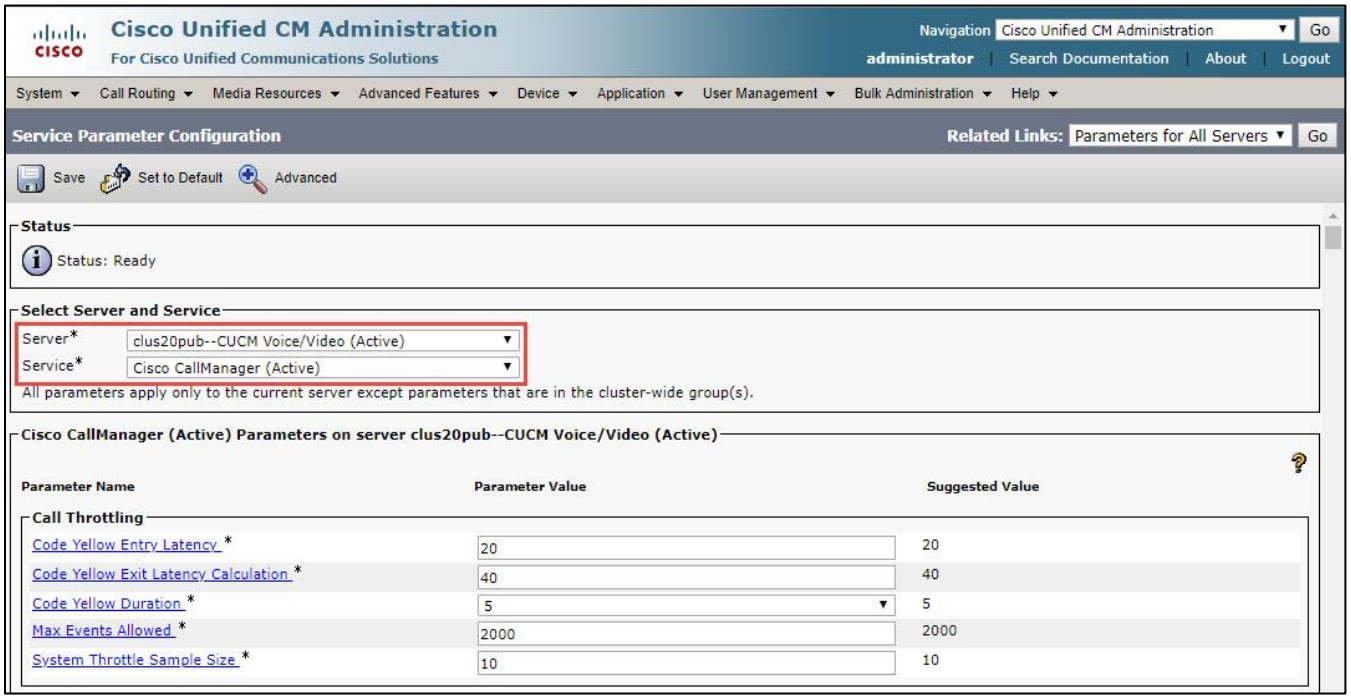

Figure 8: Service Parameters

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

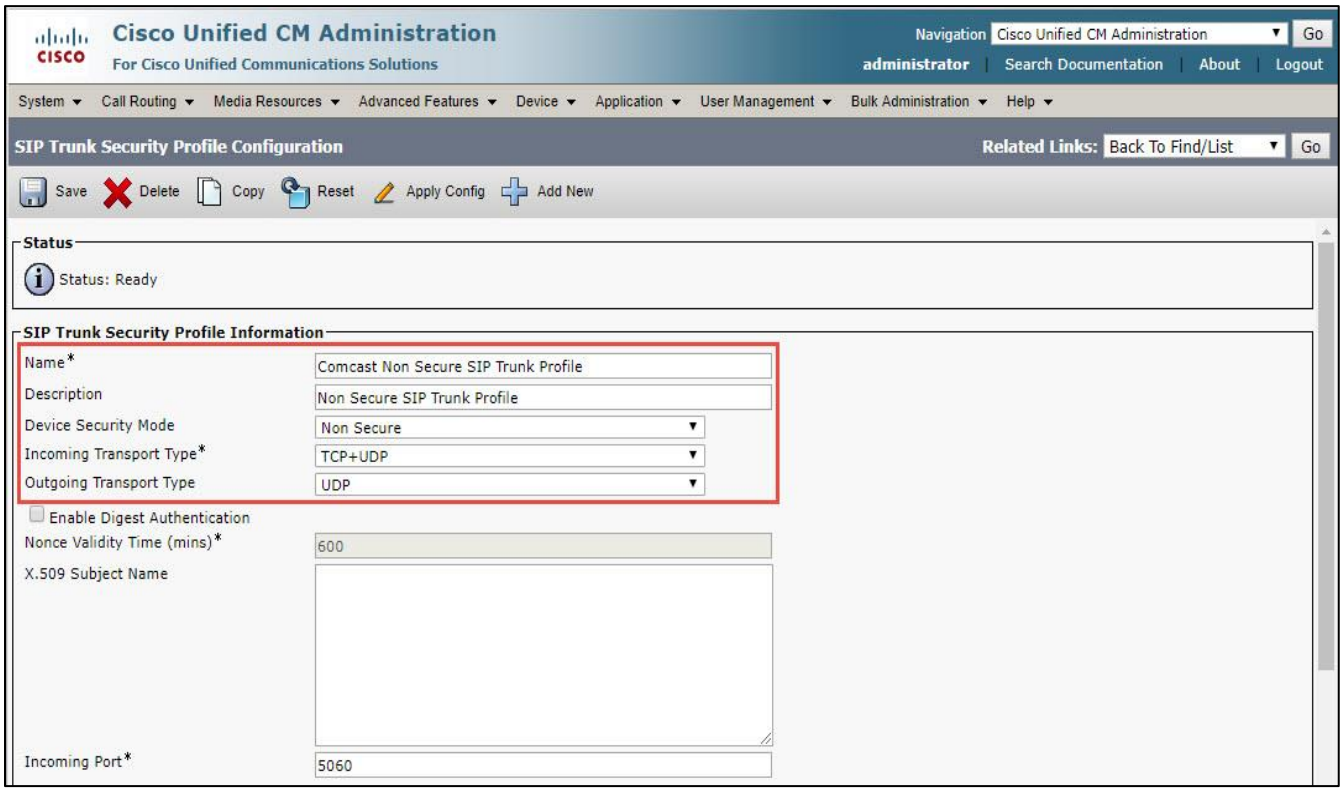

Figure 9: SIP Trunk Security Profile

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

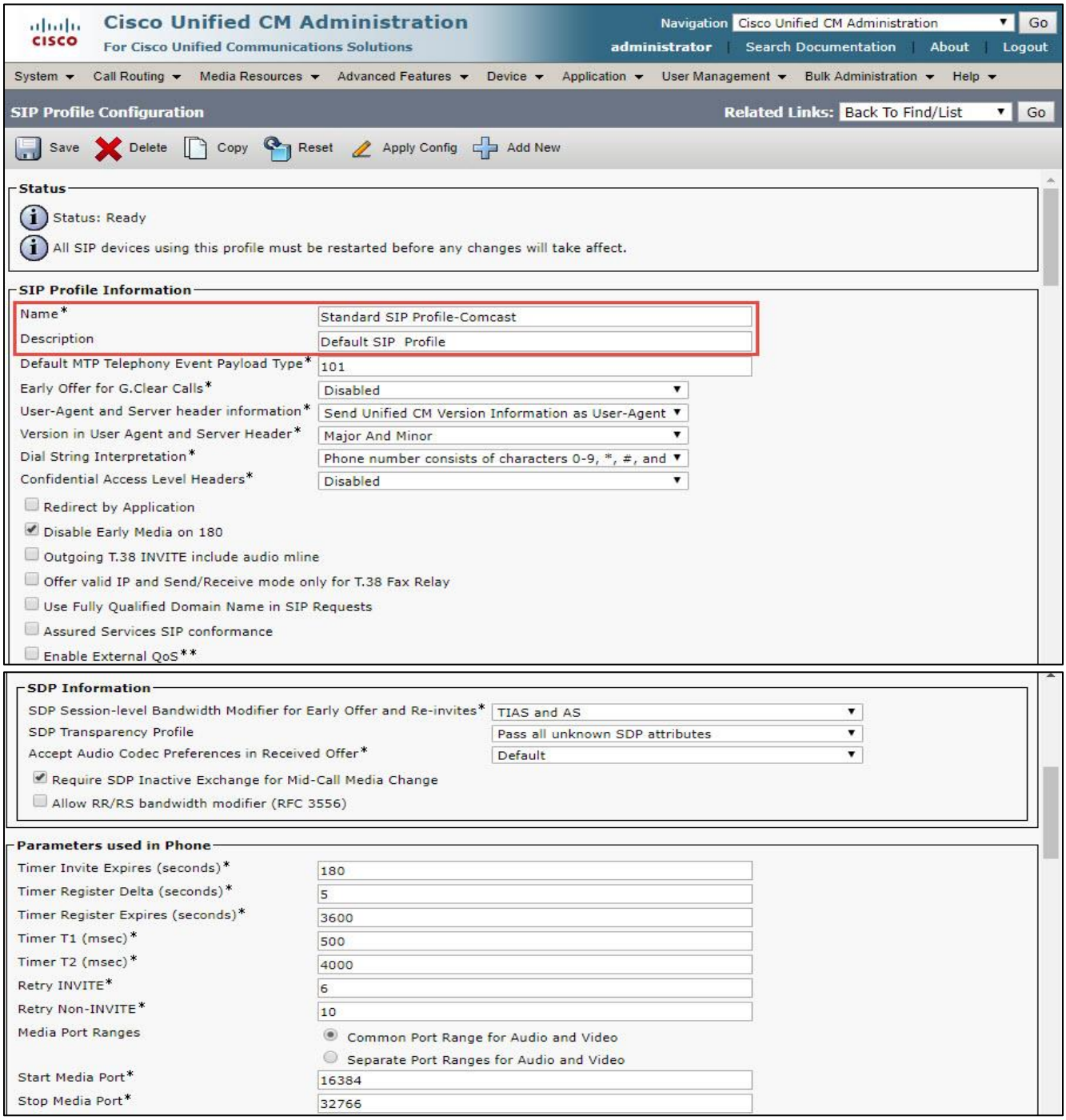

Figure 10: SIP Profile

DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default
DSCP for Audio Portion of Video Calls	Use System Default
DSCP for TelePresence Calls	Use System Default
DSCP for Audio Portion of TelePresence Calls	Use System Default
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
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

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] Semi Attended Transfer
- [ ] Enable VAD
- [ ] Stutter Message Waiting
- [ ] MLPP User Authorization

Normalization Script

Normalization Script < None >

Enable Trace

	Parameter Name	Parameter Value
1		

Incoming Requests FROM URI Settings

Caller ID DN	
Caller Name	

Figure 11: SIP Profile (Cont.)

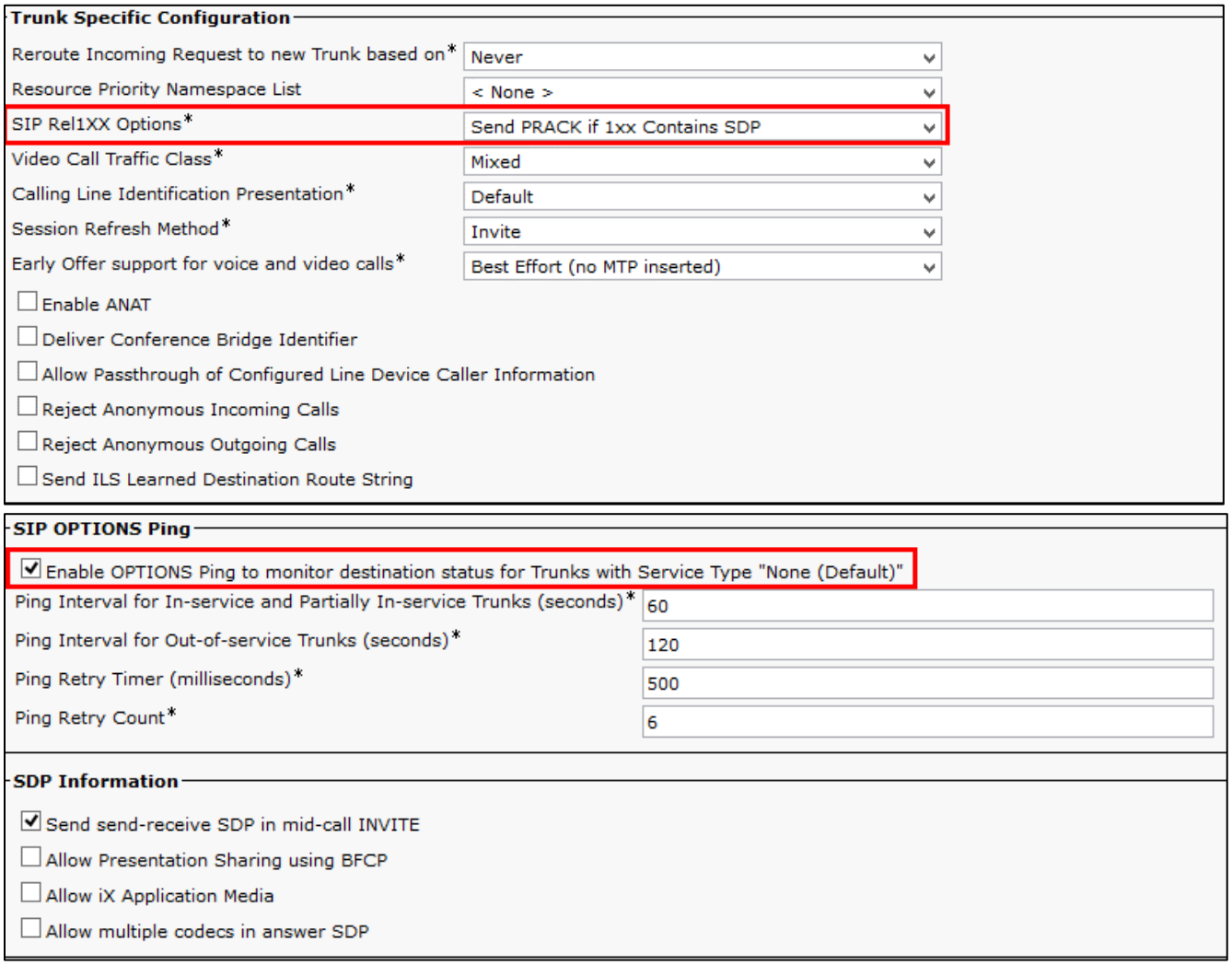

Figure 12: SIP Profile (Cont.)

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

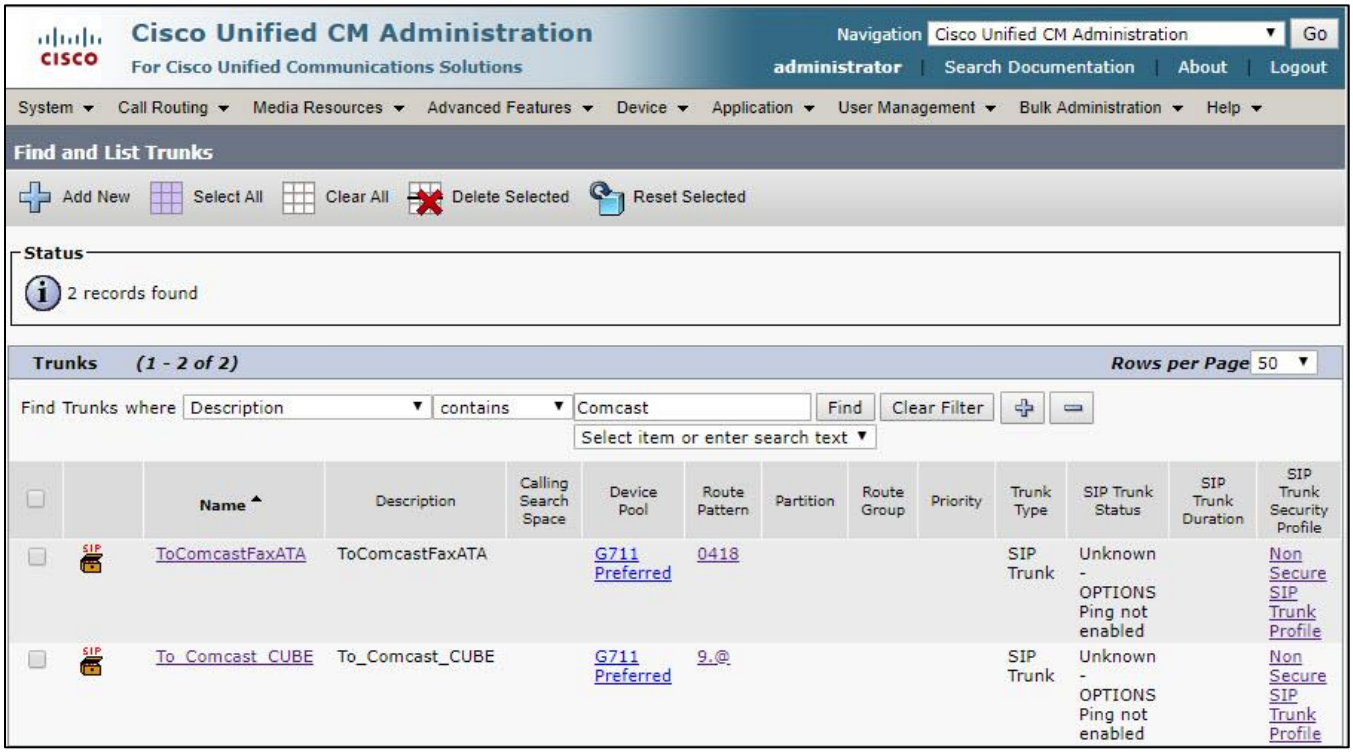

Figure 13: SIP Trunks List

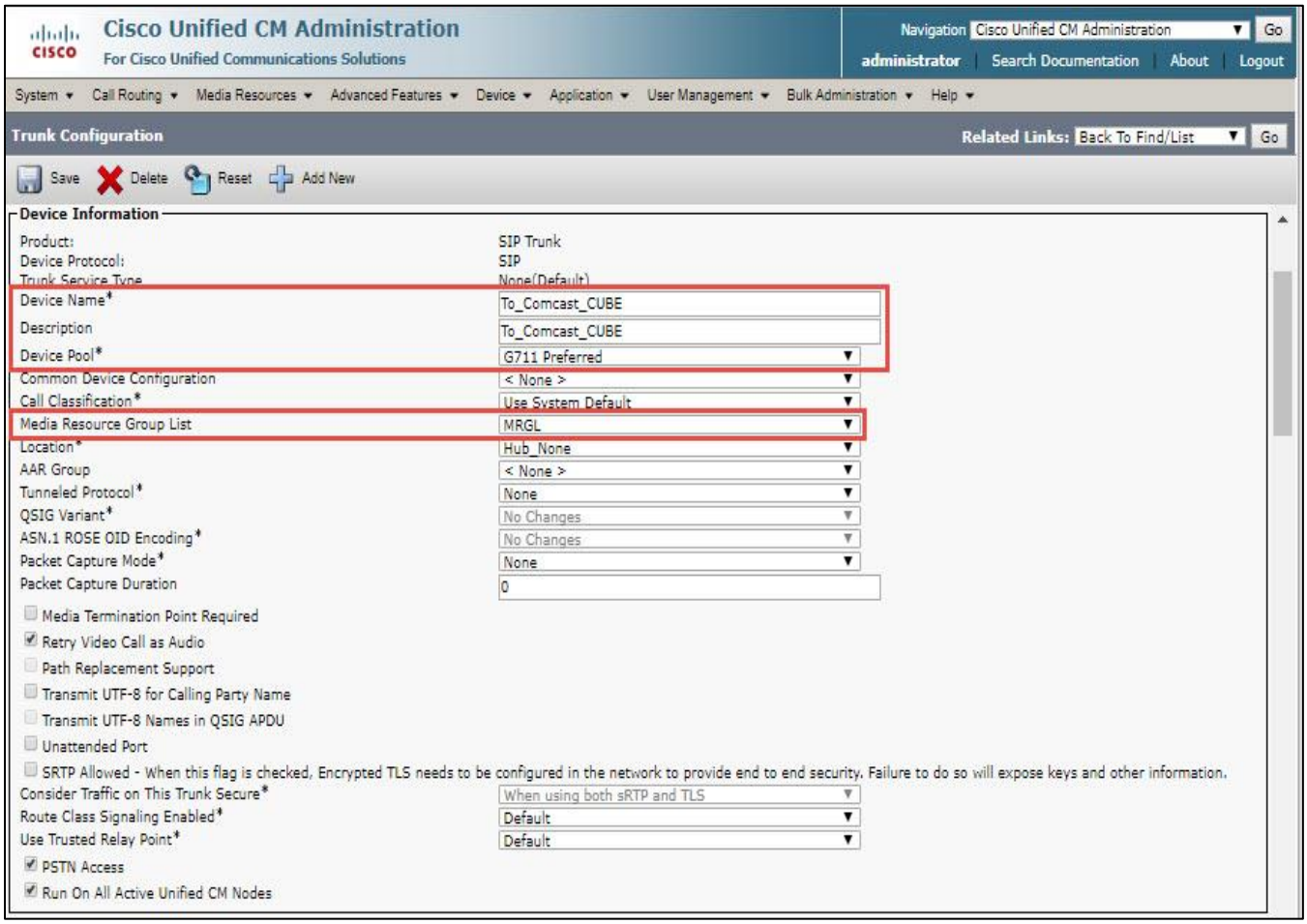

Figure 14: SIP Trunk to Cisco UBE

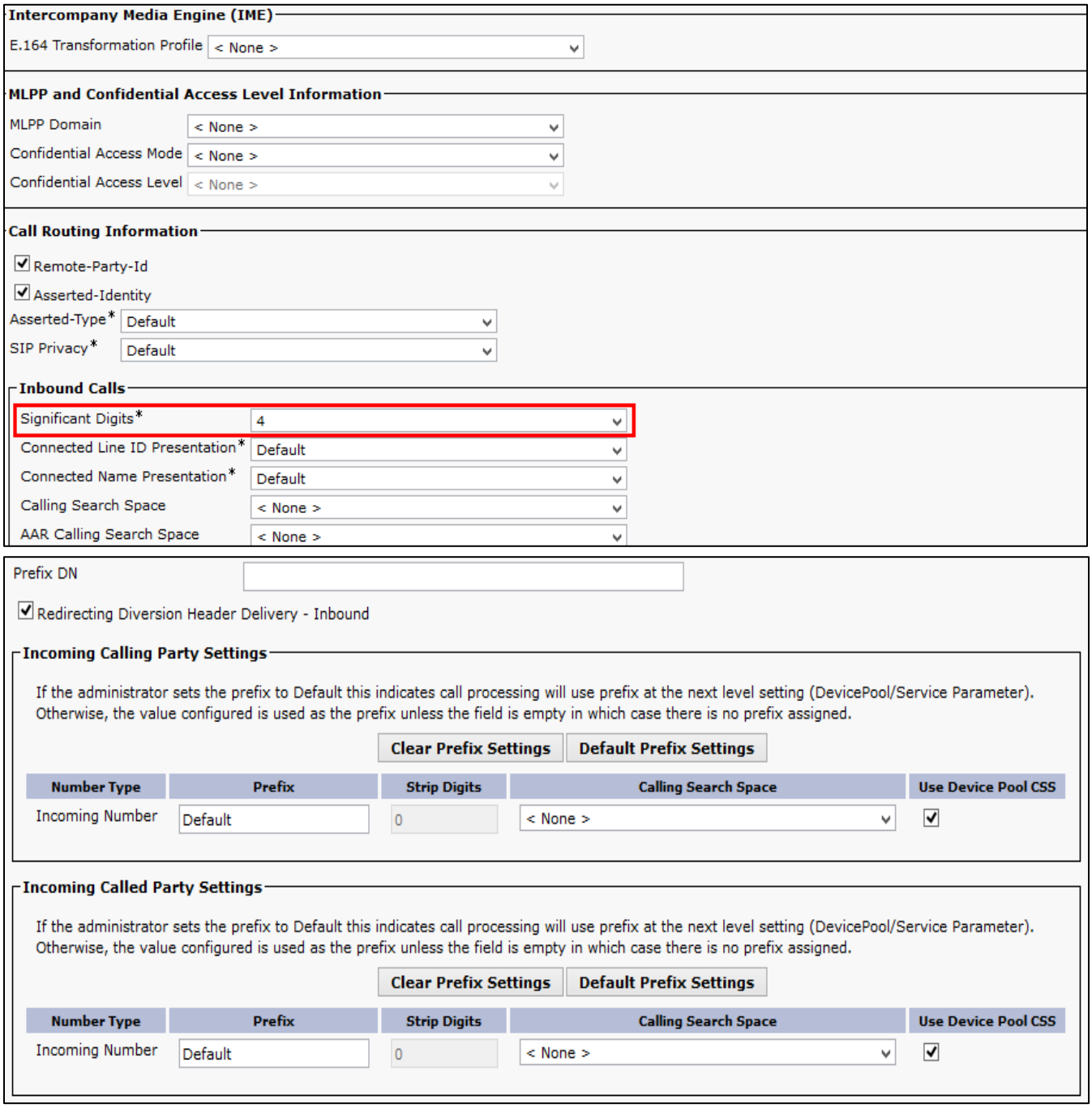

Figure 15: SIP Trunk to Cisco UBE (cont.)

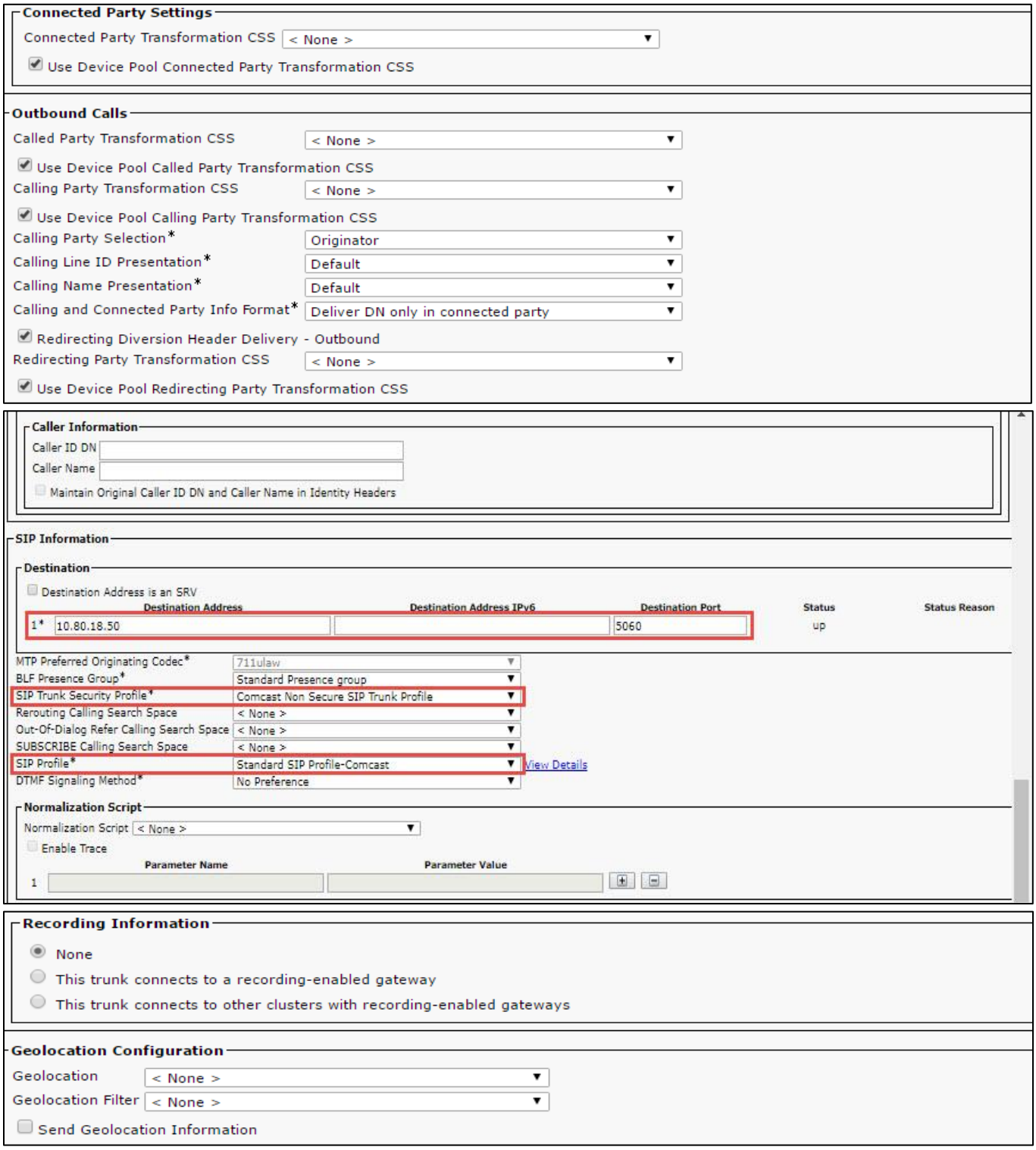

Figure 16: SIP Trunk to Cisco UBE (Cont.)

Explanation

Parameter	Value	Description
Device Name	To_Comcast_CUBE	Name for the trunk
Device Pool	G711 Preferred	Default Device Pool is used for this trunk
Media Resource Group List	MRGL	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.18.50	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	Standard SIP Profile-Comcast	SIP Profile configured earlier

Dial Plan

Route Pattern Configuration

Navigation: Call Routing > Route/Hunt > Route Pattern

Route patterns are configured as below:

- Cisco IP phone dial “9”+10 digits number to access PSTN via Cisco UBE
  - “9” is removed before sending to Cisco UBE
- For FAX call, 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 call to 0418 will be sent to Cisco ATA
- Cisco IP phones dial X11 for emergency call and will send all digits to Cisco UBE to Comcast network

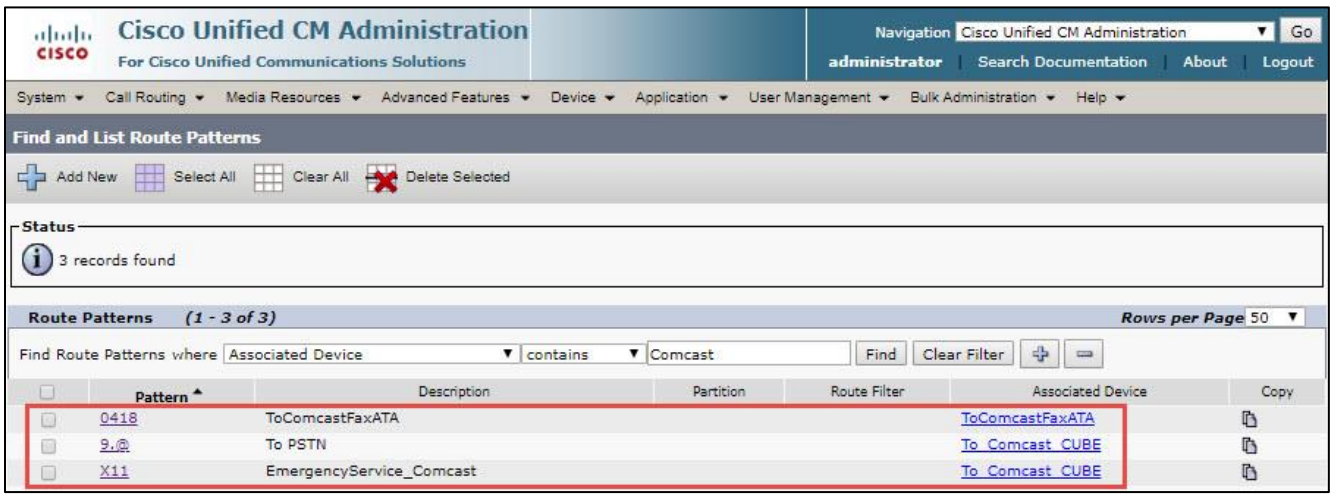

Figure 17: Route Patterns List

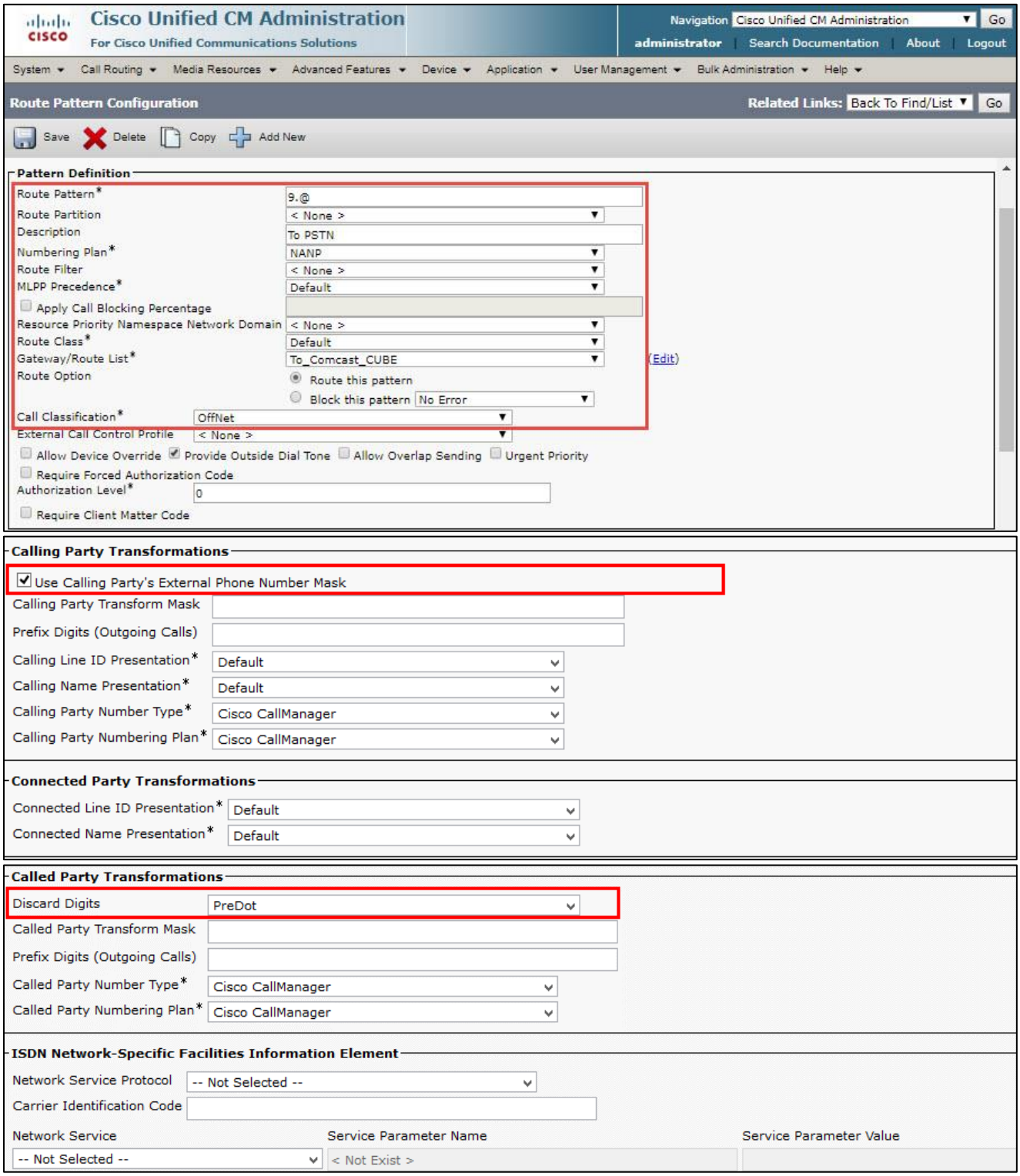

Figure 18: Route Pattern for Voice

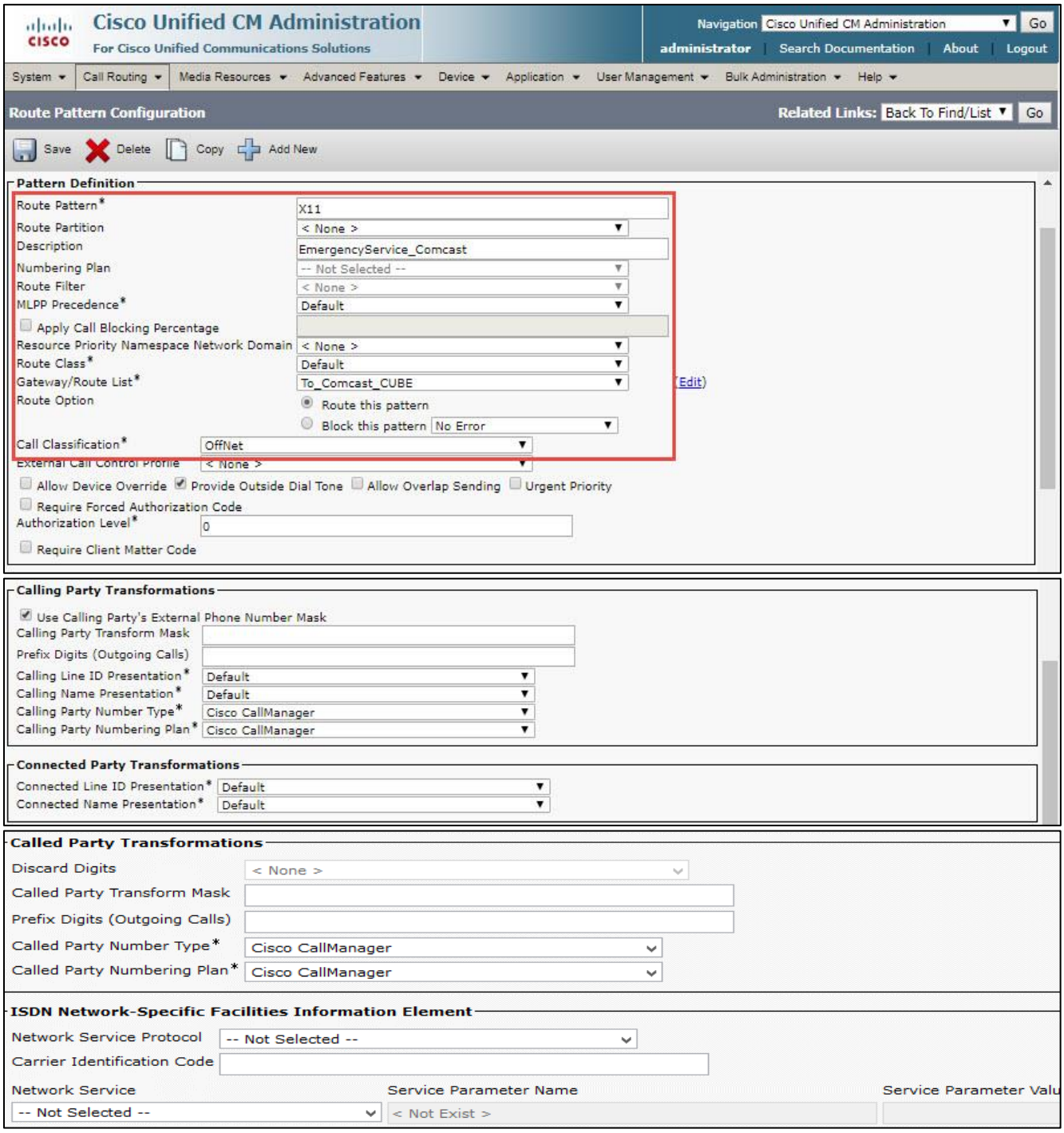

Figure 19: Route Pattern for Voice (Cont.)

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For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | Go
administrator | Search Documentation | About | Logout

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

Route Pattern Configuration — Related Links: Back To Find/List | Go

Save | Delete | Copy | Add New

Status

Update successful

Pattern Definition

Route Pattern*	0418
Route Partition	< None >
Description	ToComcastFaxATA
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	ToComcastFaxATA (Edit)
Route Option	Route this pattern / Block this pattern: No Error
Call Classification*	OffNet
External Call Control Profile	< None >

Allow Device Override | Provide Outside Dial Tone | Allow Overlap Sending | Urgent Priority
Require Forced Authorization Code
Authorization Level* 0
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*	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

Called Party Transformations

Discard Digits	< None >
Called Party Transform Mask	856333XXXX
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 >	

Figure 20: Route Pattern for Fax

Explanation

Setting	Value	Description
Route Pattern	9.@ for Voice & International Calls, 0418 for fax call and X11 for operator call and emergency services	Specify appropriate Route Pattern
Gateway/Route List	To_Comcast_CUBE for Route Pattern 9.@, X11and T0ComcastFaxATA for Route Pattern 0418	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 9.@	North American Numbering Plan
Call Classification	Offnet for Route Pattern 9.@, 0418 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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Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters
CiscoSystems
International BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

Americas Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

AsiaPacific Headquarters
Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
www.cisco.com
Tel: +65 317 7777
Fax: +65 317 7799

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