Application Note

Cisco Unified Communications Manager 12.5 to COX Business SIP Trunk via Cisco Unified Border Element v12.0.0 on ISR4331 [IOS-XE 16.06.05]

July 16, 2019

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Introduction

Service Providers today, such as COX Business, are offering alternative methods to connect to the PSTN via their IP networks. 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.

A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and COX Business network, Cisco Unified Border Element v12.0.0 (ISR4331) running IOS-XE 16.06.05 can be used. The Cisco Unified Border Element v12.0.0 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 12.5 connected to COX Business 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 COX Business 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) 12.5 and Cisco Unified Border Element (ISR4331) running IOS-XE 16.06.05 for connectivity to COX Business SIP Trunking service. The deployment model covered in this application note is (Cisco UCM 12.5) to PSTN (COX Business) via Cisco unified border element 12.0.0
- Testing was performed in accordance to COX Business generic SIP Trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold (MOH), attended and semi-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 COX Business 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 COX Business SIP Trunking network.

Network Topology

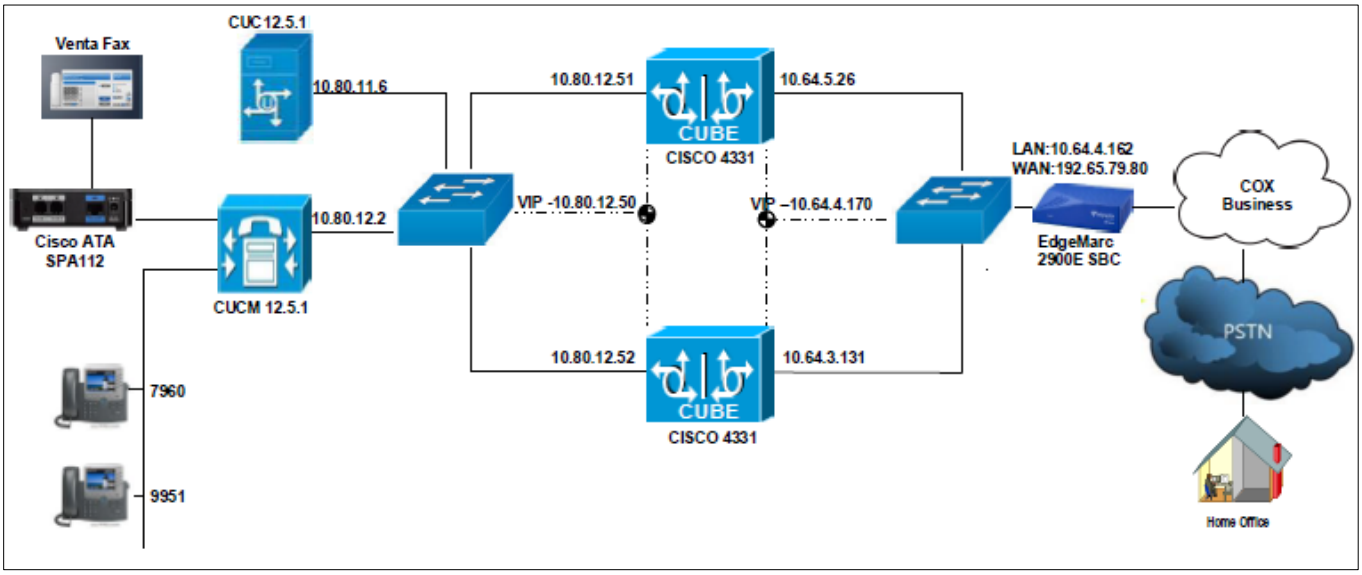

Figure 1: Network Topology

System Components

Hardware Requirements

- Cisco UCSC-C240-M3S VMWare host running ESXi 5.5 or later
- Cisco Unified Border Element (Cisco UBE)
- Cisco ISR4331/K9 (1RU) processor with 3843979K/6147K bytes of memory with 3 Gigabit Ethernet interfaces.
- Processor board ID FLM2141V252 & FLM2141V250
- EdgeMarc 2900E SBC
- Cisco ATA SPA112
- IP phones 7960 (SIP) and 9951 (SIP)

Software Requirements

- Cisco Unified Communications Manager 12.5.1.10000-22
- Cisco Unity Connection 12.5.1.10000-23
- IOS-XE 16.06.05 for Cisco Unified Border Element version 12.0.0
- Cisco IOS Software [Everest], ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 16.6.5, RELEASE SOFTWARE (fc3)
- Cisco IOS XE Software, Version 16.06.05
- EdgeMarc 2900E SBC 14.9.2

Features

Features Supported

- Incoming and outgoing off-net calls using G711ULaw voice codec
- Call hold
- Call transfer (semi-attended and attended)
- Call conference
- Call forward (all, busy and no answer)
- DTMF (RFC2833)
- Fax- G.711U Pass-Through

Features Not Supported

- Cisco IP phones used in this test do not support blind transfer

The following are not supported by COX Business

- G.729 voice codec
- T.38 fax
- SG3 fax transmission speed

Features Not Supported

- None

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.

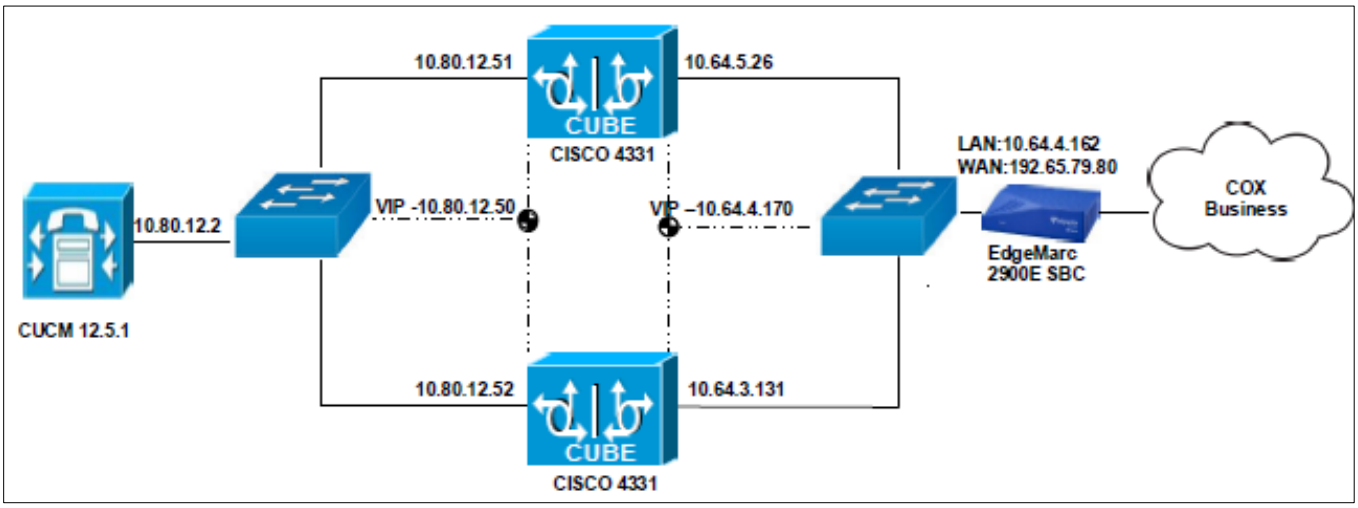

Figure 2 High Availability topology

CUBE 1:

```
interface GigabitEthernet0/0/0
 description WAN
 ip address 10.64.5.26 255.255.0.0
 negotiation auto
 redundancy rii 16
 redundancy group 2 ip 10.64.4.170 exclusive
!
interface GigabitEthernet0/0/1
 description LAN
 ip address 10.80.12.51 255.255.255.0
 negotiation auto
 redundancy rii 17
 redundancy group 2 ip 10.80.12.50 exclusive
!
interface GigabitEthernet0/0/2
 description HA_Interface
 ip address 10.70.50.101 255.255.255.0
 negotiation auto
```

CUBE 2:

```
interface GigabitEthernet0/0/0
 description WAN
 ip address 10.64.3.131 255.255.0.0
 negotiation auto
 redundancy rii 16
 redundancy group 2 ip 10.64.4.170 exclusive
!
interface GigabitEthernet0/0/1
 description LAN
 ip address 10.80.12.52 255.255.255.0
 negotiation auto
 redundancy rii 17

 redundancy group 2 ip 10.80.12.50 exclusive
!
interface GigabitEthernet0/0/2
 description HA_interface
 ip address 10.70.50.102 255.255.255.0
 negotiation auto
```

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
 allow-connections sip to sip
 redundancy-group 2
 no supplementary-service sip handle-replaces
 fax protocol pass-through g711ulaw
 sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  early-offer forced
  midcall-signaling passthru
```

Explanation

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
fax protocol	Specifies the fax protocol
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 voice codec is used for this testing. Codec preferences used to change according to the test plan description

```
voice class codec 1
 codec preference 1 g711ulaw
```

Dial Peer

Active Cisco UBE

```
dial-peer voice 1 voip
 description WAN-Facing Outgoing
 huntstop
 destination-pattern .T
 translate-outgoing called 110
 session protocol sipv2
 session target ipv4:10.64.4.162:5060
 session transport udp
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 2 voip
 description LAN facing Outgoing
 huntstop
 session protocol sipv2
 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
 no vad
!
```

```
dial-peer voice 3 voip
 description Incoming CUCM Facing
 huntstop
 destination-pattern 402.......
 session protocol sipv2
 session target ipv4:10.80.12.2:5060
 session transport udp
 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
 no vad
!
dial-peer voice 4 voip
 description Incoming WAN Facing
 huntstop
 session protocol sipv2
 incoming called-number 402.......
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
```

Configuration Example

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

Active Cisco UBE

```
Current configuration : 4583 bytes
!
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 COX_CUBE1
!
boot-start-marker
boot system flash:isr4300-universalk9.16.06.05.SPA.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
logging buffered 99999
!
no aaa new-model
subscriber templating
multilink bundle-name authenticated
```

```
!
crypto pki trustpoint TP-self-signed-3616943619
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-3616943619
 revocation-check none
 rsakeypair TP-self-signed-3616943619
!
crypto pki certificate chain TP-self-signed-3616943619
!
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element
 allow-connections sip to sip
 redundancy-group 2
 no supplementary-service sip handle-replaces
 fax protocol pass-through g711ulaw
 sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  early-offer forced
  midcall-signaling passthru
!
voice class codec 1
 codec preference 1 g711ulaw
!
voice-card 0/4
 no watchdog
!
license udi pid ISR4331/K9 sn FDO24881FEY
diagnostic bootup level minimal
spanning-tree extend system-id
!
```

```
!
redundancy
 mode none
 application redundancy
  group 2
   name COX
   priority 150 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/0/2 protocol 1
   data GigabitEthernet0/0/2
   track 1 shutdown
   track 2 shutdown
!
track 1 interface GigabitEthernet0/0/1 line-protocol
track 2 interface GigabitEthernet0/0/0 line-protocol
!
translation-rule 110
 Rule 1 14025059697 4025059697
 Rule 2 14029165692 4029165692
 Rule 3 14025056016 4025056016
!
interface GigabitEthernet0/0/0
 description WAN
 ip address 10.64.5.26 255.255.0.0
 negotiation auto
 redundancy rii 16
 redundancy group 2 ip 10.64.4.170 exclusive
!
interface GigabitEthernet0/0/1
 description LAN
 ip address 10.80.12.51 255.255.255.0
 negotiation auto
 redundancy rii 17
 redundancy group 2 ip 10.80.12.50 exclusive
```

```
!
interface GigabitEthernet0/0/2
 description HA_Interface
 ip address 10.70.50.101 255.255.255.0
 negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.80.12.0 255.255.255.0 10.80.12.1
!
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 1 voip
 description WAN-Facing Outgoing
 huntstop
 destination-pattern .T
 translate-outgoing called 110
```

```
 session protocol sipv2
 session target ipv4:10.64.4.162:5060
 session transport udp
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 2 voip
 description LAN facing Outgoing
 huntstop
 session protocol sipv2
 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
 no vad
!
dial-peer voice 3 voip
 description Incoming CUCM Facing
 huntstop
 destination-pattern 402.......
 session protocol sipv2
 session target ipv4:10.80.12.2:5060
 session transport udp
 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
 no vad
!
```

```
dial-peer voice 4 voip
 description Incoming WAN Facing
 huntstop
 session protocol sipv2
 incoming called-number 402.......
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
alias exec tr show run | begin voice translation-rule
alias exec dp show run | begin dial-peer voice
!
line con 0
 exec-timeout 0 0
 password tekV1z10n
 login
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password tekV1z10n
 login
```

Standby Cisco UBE

```
Current configuration : 4111 bytes

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 COX_CUBE2
!
boot-start-marker
boot-end-marker
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
no aaa new-model
subscriber templating
multilink bundle-name authenticated
!
voice service voip
 address-hiding
 mode border-element
 allow-connections sip to sip
 redundancy-group 2
 no supplementary-service sip handle-replaces
 fax protocol pass-through g711ulaw
 sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  early-offer forced
  midcall-signaling passthru
!
voice class codec 1
 codec preference 1 g711ulaw
!
voice-card 0/1
 no watchdog
!
```

```
voice-card 0/4
 no watchdog
!
license udi pid ISR4331/K9 sn FDO24881GMV
diagnostic bootup level minimal
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 2
   name COX
   priority 100 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/0/2 protocol 1
   data GigabitEthernet0/0/2
   track 1 shutdown
   track 2 shutdown
!
track 1 interface GigabitEthernet0/0/1 line-protocol
track 2 interface GigabitEthernet0/0/0 line-protocol
!
translation-rule 110
 Rule 1 14025059697 4025059697
 Rule 2 14029165692 4029165692
 Rule 3 14025056016 4025056016
!
interface GigabitEthernet0/0/0
 description WAN
 ip address 10.64.3.131 255.255.0.0
 negotiation auto
 redundancy rii 16
 redundancy group 2 ip 10.64.4.170 exclusive
!
interface GigabitEthernet0/0/1
 description LAN
 ip address 10.80.12.52 255.255.255.0
 negotiation auto
 redundancy rii 17
 redundancy group 2 ip 10.80.12.50 exclusive
!
interface GigabitEthernet0/0/2
 description HA_interface
 ip address 10.70.50.102 255.255.255.0
 negotiation auto
!
```

```
interface Service-Engine0/1/0
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 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 10.64.1.1
ip route 10.80.12.0 255.255.255.0 10.80.12.1
!
control-plane
!
voice-port 0/1/0
voice-port 0/1/1
!
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 1 voip
 description WAN-Facing Outgoing
 huntstop
 destination-pattern .T
 translate-outgoing called 110
 session protocol sipv2
 session target ipv4:10.64.4.162:5060
 session transport udp
 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 2 voip
 description LAN facing Outgoing
 huntstop
 session protocol sipv2
 incoming called-number .T
 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 3 voip
 description Incoming CUCM Facing
 huntstop
 destination-pattern 402.......
 session protocol sipv2
 session target ipv4:10.80.12.2:5060
 session transport udp
 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 4 voip
 description Incoming WAN Facing
 huntstop
 session protocol sipv2
 incoming called-number 402.......
 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
!
line con 0
 exec-timeout 0 0
 password tekV1z10n
 login
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password tekV1z10n
 login
!
end
```

Configuring Cisco Unified Communications Manager

Cisco UCM Version

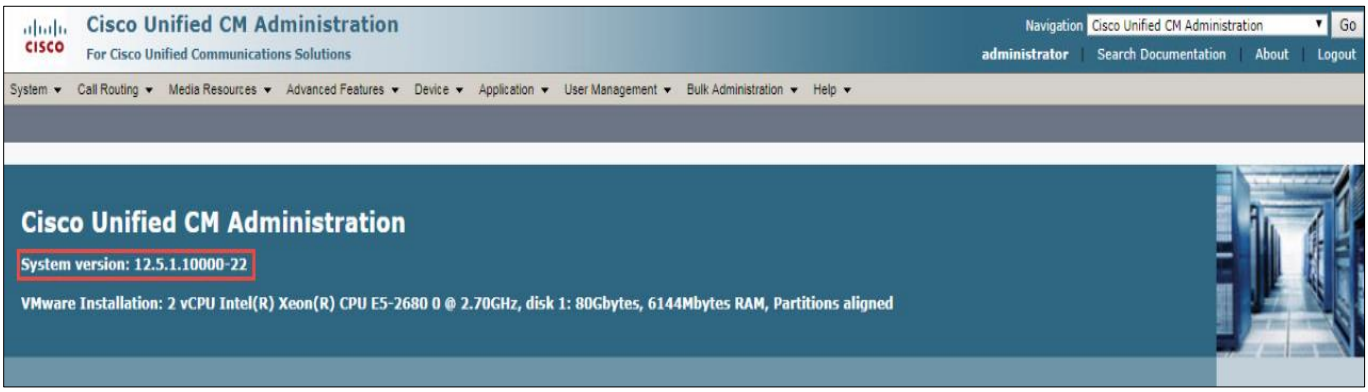

Figure 3 cisco UCM Version

Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

1. Select Server*: Clus22pub--CUCM Voice/Video (Active)
2. Select Service*: Cisco CallManager (Active)
3. All other fields are set to default values

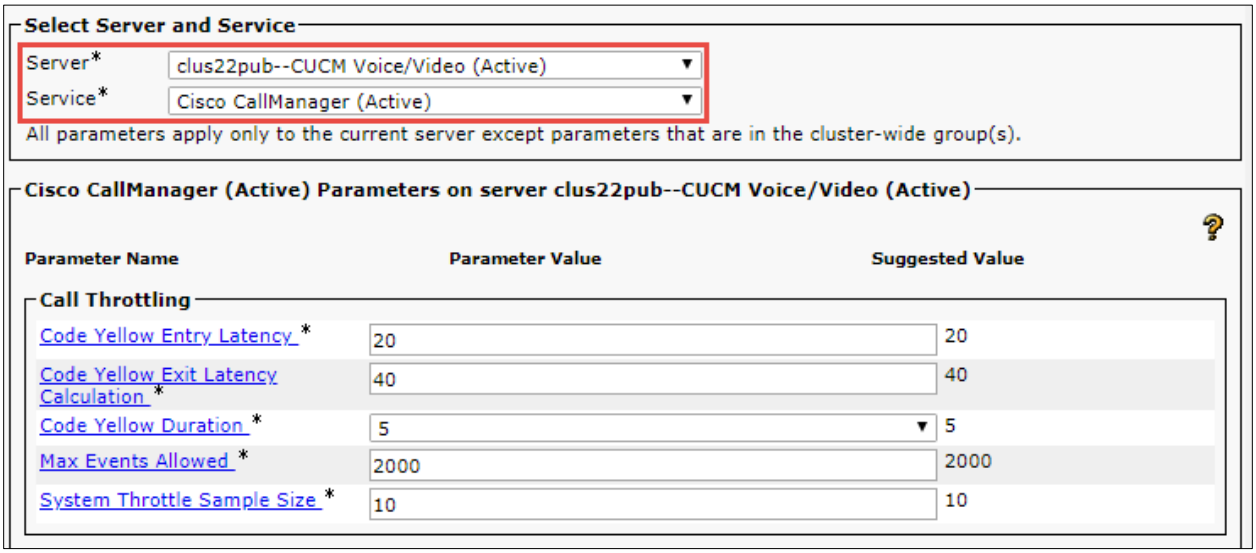

Figure 4: Service Parameters

Offnet Calls via COX Business SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and COX Business network via Cisco UBE and EdgeMarc.

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

1. Name*: Non Secure SIP Trunk Profile
2. Description: Non Secure SIP Trunk Profile authenticated by null String

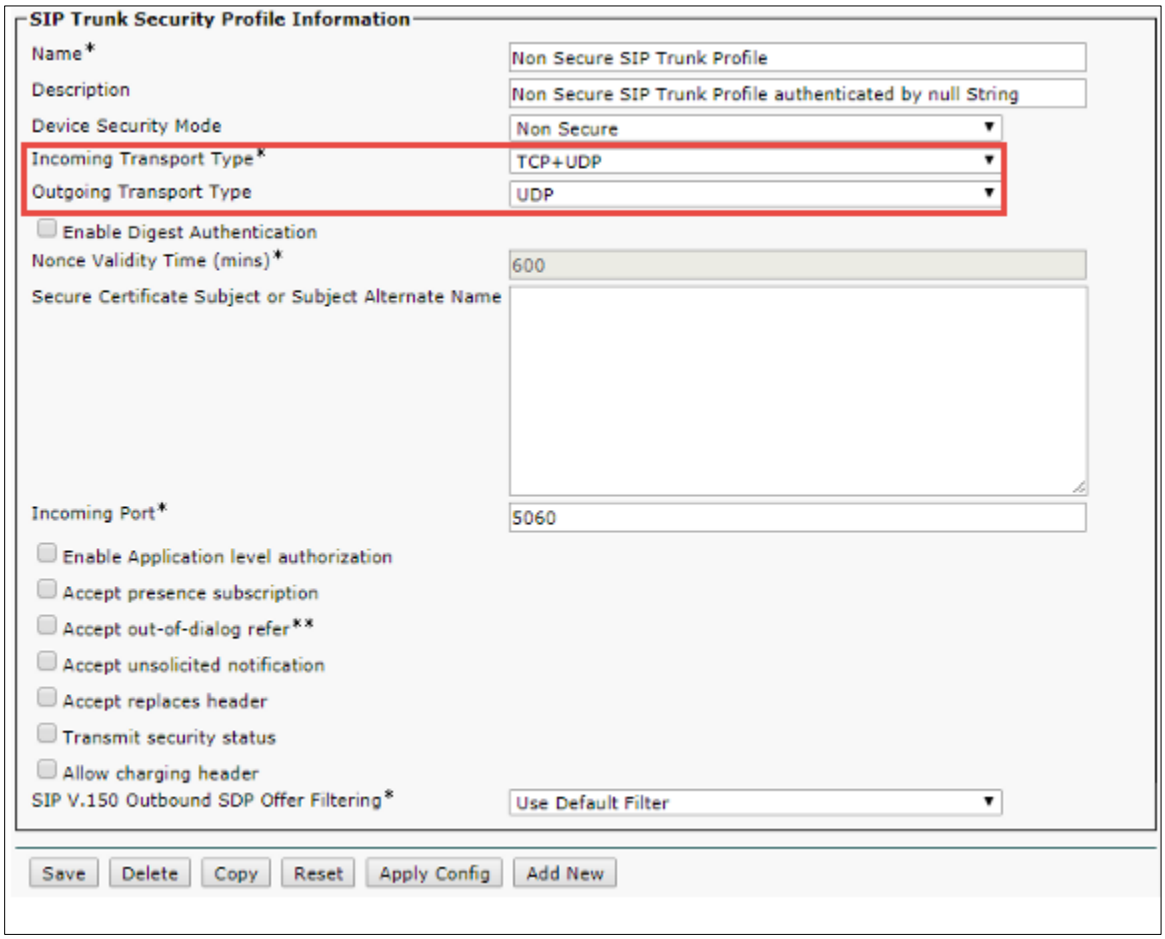

Figure 5: SIP Trunk Security Profile

Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to Cisco UBE 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: Device → Device Settings → SIP Profile

1. Name*: COX SIP Profile, for example

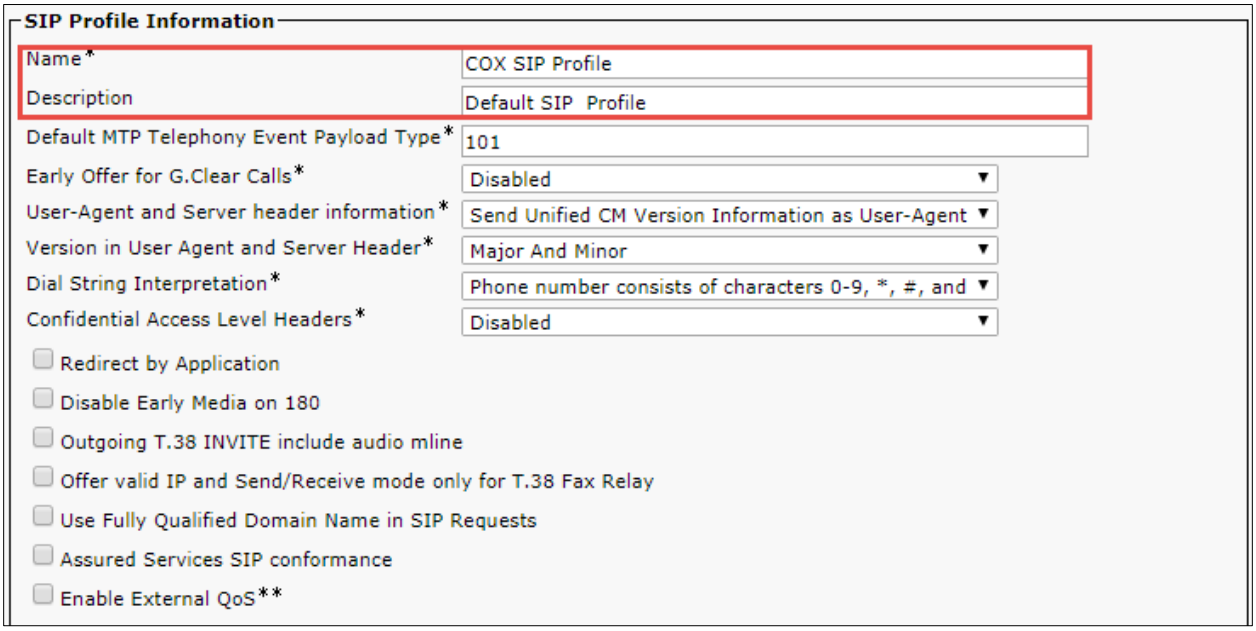

Figure 6: SIP Profile

SDP Information

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer*	Default

- [ ] Require SDP Inactive Exchange for Mid-Call Media Change
- [ ] Allow RR/RS bandwidth modifier (RFC 3556)

Parameters used in Phone

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Media Port Ranges	(•) Common Port Range for Audio and Video ( ) Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766
DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default
DSCP for Audio Portion of Video Calls	Use System Default

Figure 6: SIP Profile (Cont.)

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

Figure 6: SIP Profile (Cont.)

Normalization Script

Normalization Script < None >

Enable Trace

	Parameter Name	Parameter Value
1		

External Presentation Information

Anonymous External Presentation

External Presentation Number

External Presentation Name

Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*	Never
Resource Priority Namespace List	< None >
SIP Rel1XX Options*	Disabled
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default
Session Refresh Method*	Invite
Early Offer support for voice and video calls*	Disabled (Default value)

Enable ANAT

Deliver Conference Bridge Identifier

Enable External Presentation Name and Number

Reject Anonymous Incoming Calls

Figure 6: SIP Profile (Cont.)

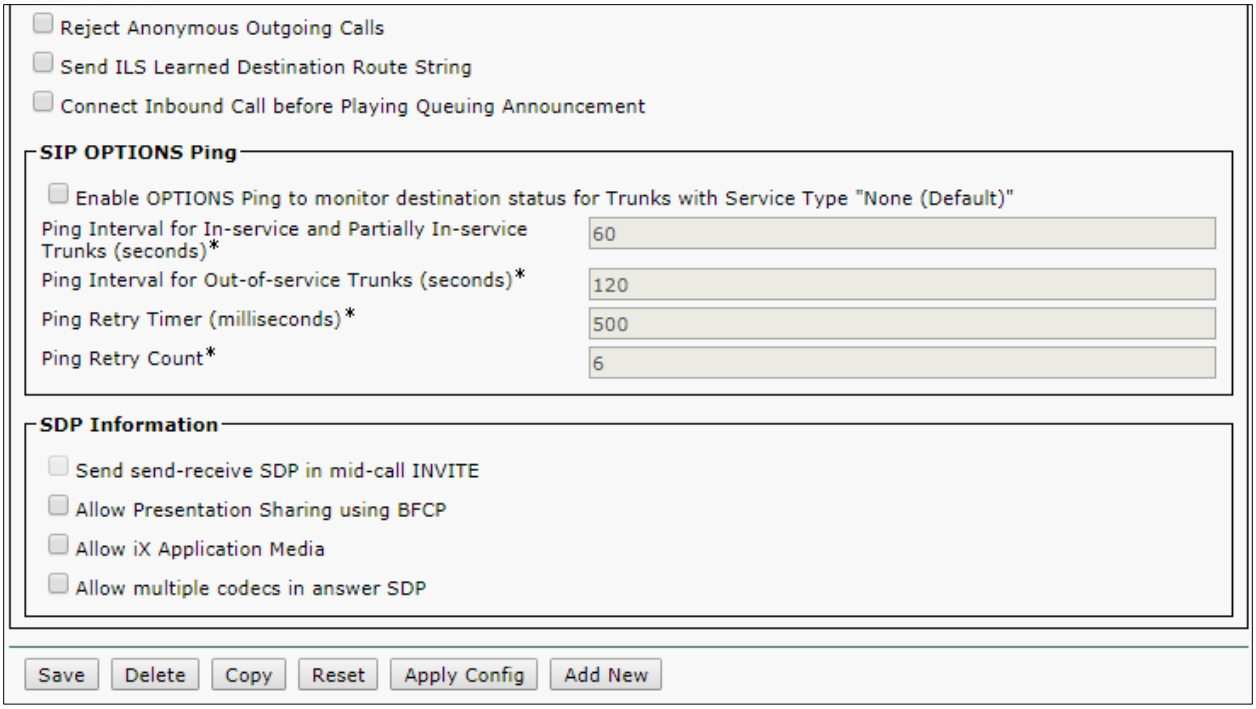

Figure 6: SIP Profile (Cont.)

Explanation

Parameter	Value	Description
Name	COX SIP Profile	Name given to refer the SIP profile created for COX Business
Description	Default SIP Profile	Customized description of the SIP profile used for COX Business
Default MTP Telephony Event Payload Type	101	RFC2833 DTMF payload type

SIP Trunk Configuration

Create SIP trunks to Cisco UBE

Navigation: Device → Trunk

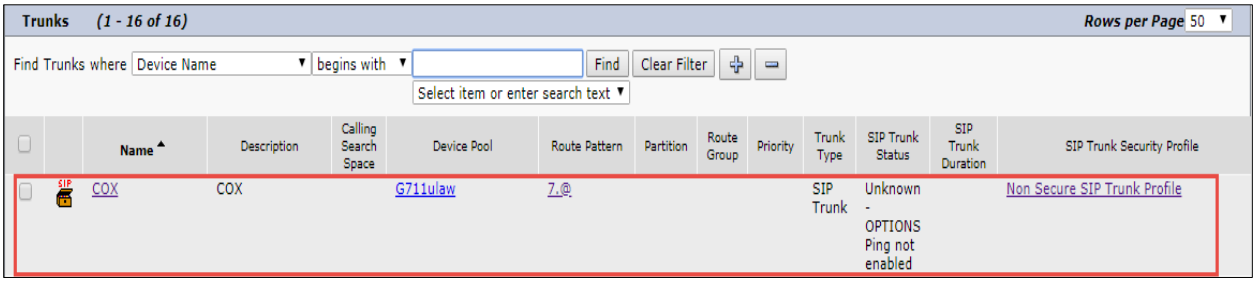

Figure 7:SIP Trunks List

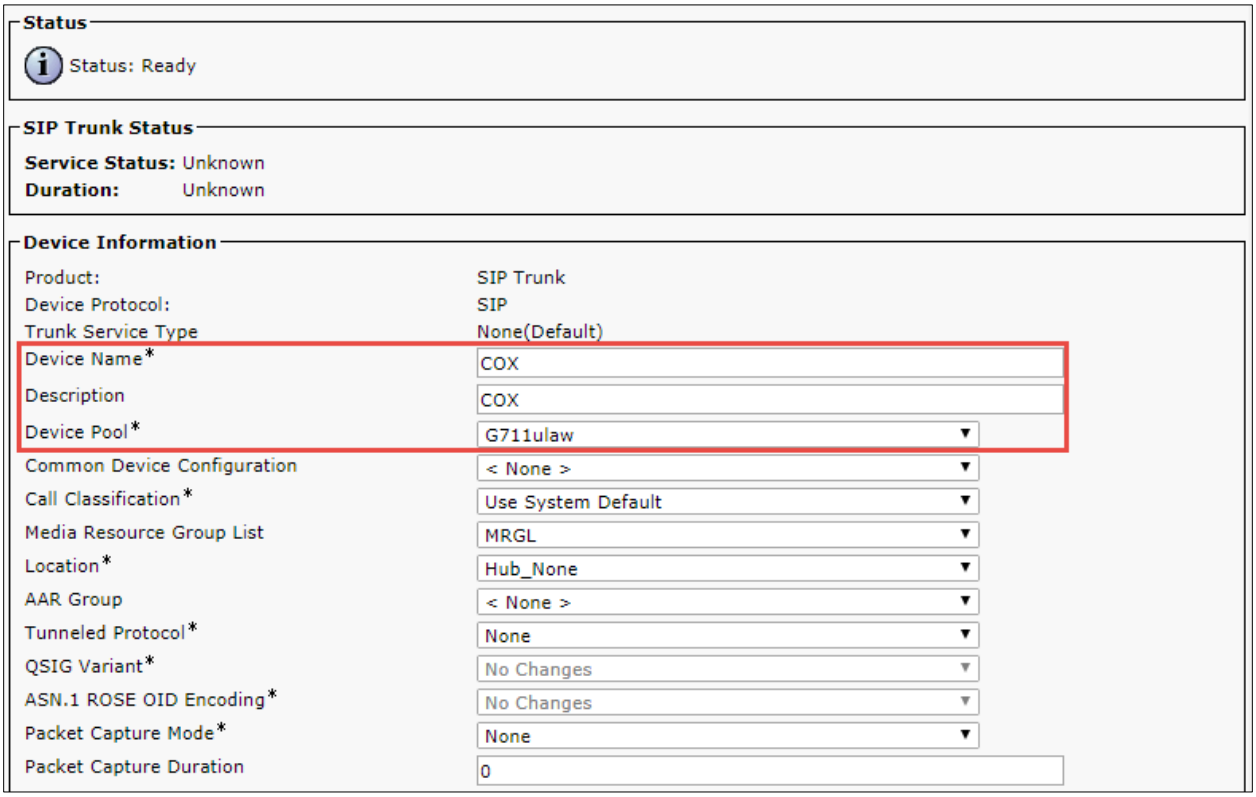

Figure 8: SIP Trunk to Cisco UBE

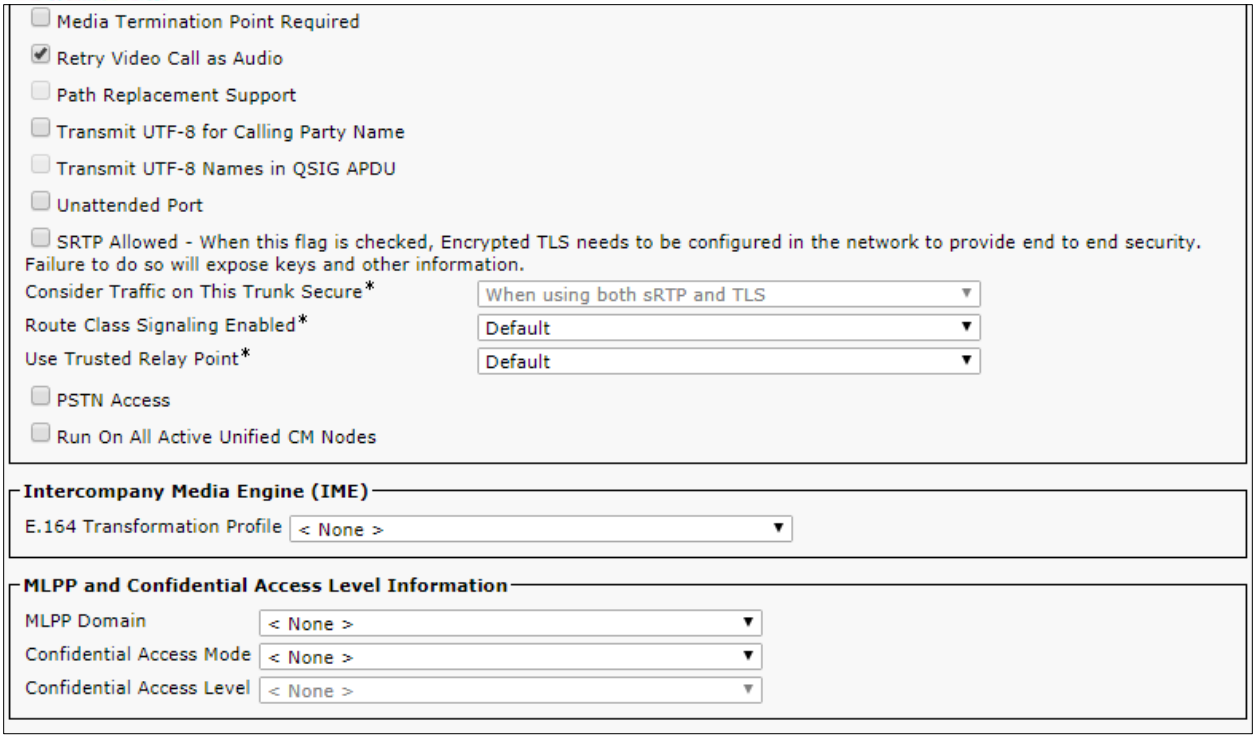

Figure 8: SIP Trunk to Cisco UBE (Cont.)

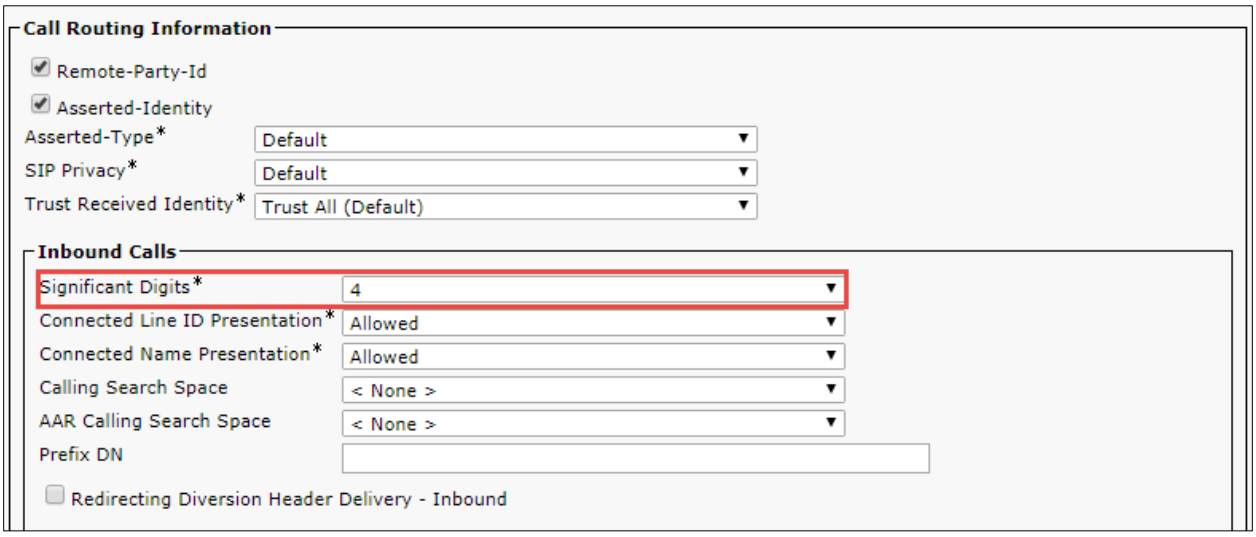

Figure 8: SIP Trunk to Cisco UBE (Cont.)

Incoming Calling Party Settings

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

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	☑

Incoming Called Party Settings

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

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	☑

Figure 8: 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* Allowed

Calling Name Presentation* Allowed

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

Presentation Information

☐ Anonymous Presentation

Presentation Number

Presentation Name

☐ Send Presentation Name and Number only in the FROM header and not in the other identity headers

Figure 8: SIP Trunk to Cisco UBE (Cont.)

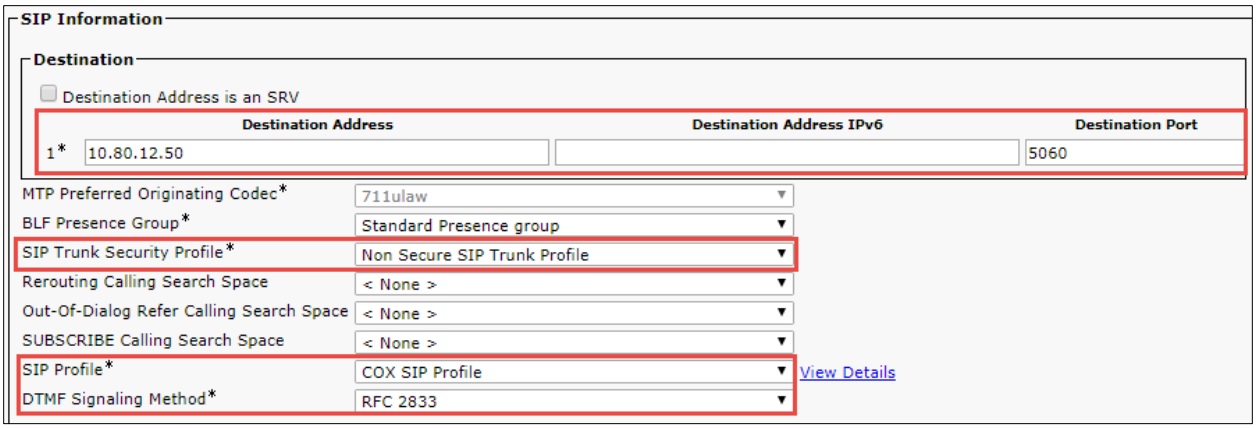

Figure 8: SIP Trunk to Cisco UBE (Cont.)

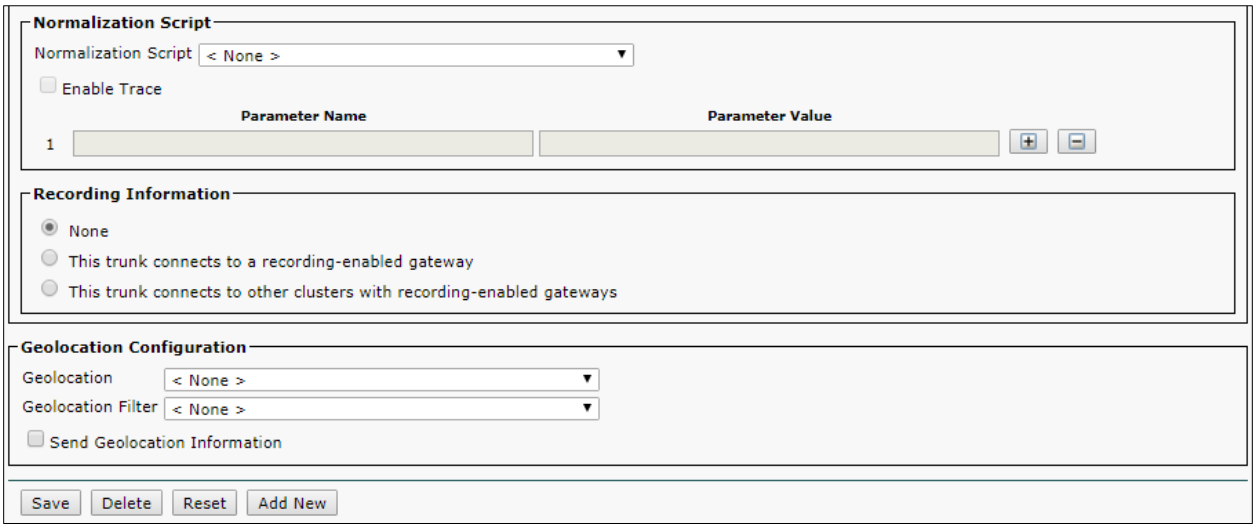

Figure 8: SIP Trunk to Cisco UBE (Cont.)

Explanation

Parameter	Value	Description
Device Name	COX	Name for the trunk
Device Pool	G711ulaw	Device Pool used for SIP trunk to COX
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	10.80.12.50	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	COX SIP Profile	SIP Profile configured earlier

Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

- Cisco IP phone dial "7"+10 digits number to access PSTN via Cisco UBE
  - "7" is removed before sending to Cisco UBE

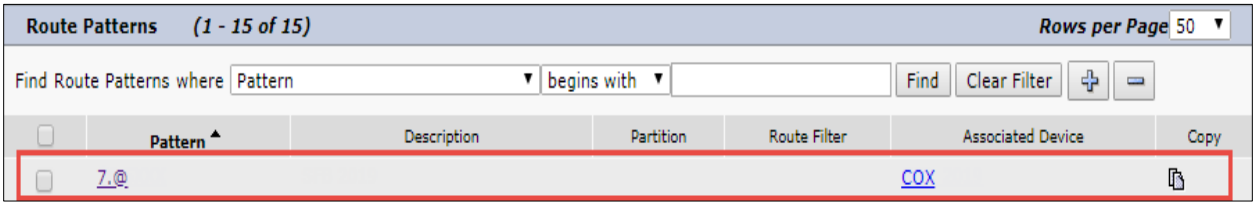

Figure 9: Route Patterns List

Pattern Definition

Route Pattern*	7.@
Route Partition	< None >
Description	
Numbering Plan*	NANP
Route Filter	< None >
MLPP Precedence*	Default
Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	COX (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

Figure 10: Route Pattern for Voice

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

Figure 10: Route Pattern for Voice(cont..)

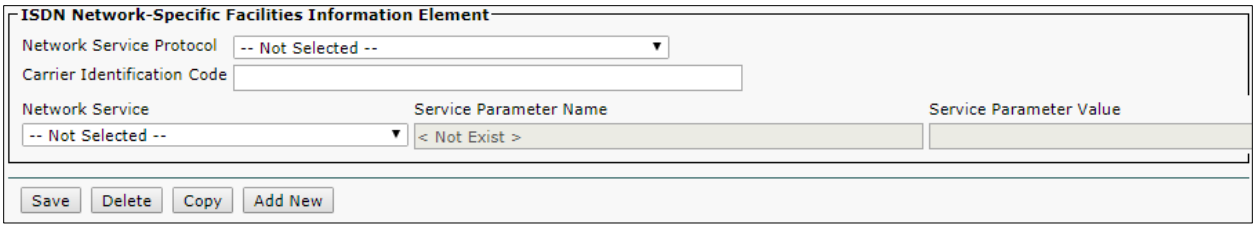

Figure 10: Route Pattern for Voice(cont..)

Explanation

Setting	Value	Description
Route Pattern	7.@ for Voice & International Calls	Specify appropriate Route Pattern
Gateway/Route List	COX for Route Pattern 7.@	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 7.@	North American Numbering Plan
Call Classification	OffNet for Route Pattern 7.@	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 7.@	Specifies how to modify digit before they are sent to COX Business network

Acronyms

Acronym	Definition
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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