



Comcast SIP Trunking:

Cisco Unified Communications Manager 10.5.2 with Cisco Unified Border Element (CUBE 11.5.0) on ISR4321/K9 [IOS-XE 3.17.1 – 15.6(1)S1] using SIP

July 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-XE 3.17 – 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 10.5.2 connected to Comcast 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 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 Cisco Unified Communications Manager (Cisco UCM) 10.5.2, and Cisco Unified Border Element (Cisco UBE) on ISR 4321/K9 [IOS-XE 3.17 15.6(1)S1] for connectivity to Comcast SIP Trunking service. The deployment model covered in this application note is CPE (Cisco Unified Communications Manager 10.5.2) 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

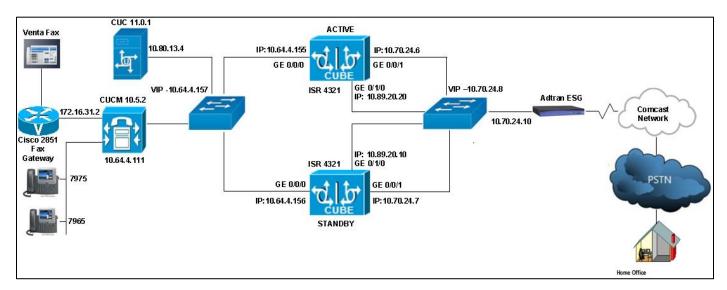


Figure 1 Network Topology

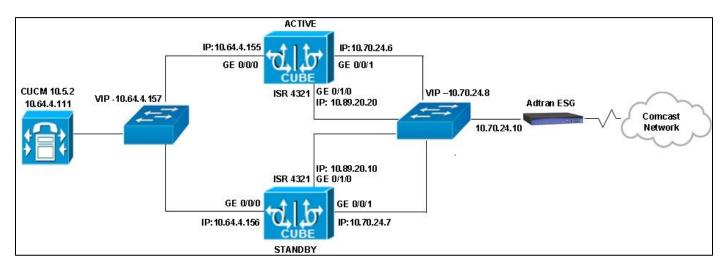


Figure 2: Cisco UBE High Availability



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 FLM1925W0X2
- Cisco 2851 Fax Gateway
- IP phones 7965 (SIP) and 7975 (SCCP)
- Comcast(Adtran) ESG Provided and managed by Comcast

Software Requirements

- Cisco Unified Communications Manager 10.5.2.13900-12
- Cisco Unity Connection 11.0.1.21900-11
- IOS-XE 3.17.1 15.6(1)S 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
- 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 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 (G711 passthrough)

Features Not Supported

- Cisco IP phones used in this test do not support Blind Transfer
- Fax (T.38) is not supported by Service Provider
- 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.
- For testing, 911 calls were routed internally in the Comcast lab



Configuration

Configuring the Cisco Unified Border Element

Network interface

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

```
interface GigabitEthernet0/0/0
description MS4 1/0/9
ip address 10.64.4.155 255.255.0.0
media-type rj45
negotiation auto
redundancy rii 11
redundancy group 2 ip 10.64.4.157 exclusive
!
interface GigabitEthernet0/0/1
description MS4 1/0/10
ip address 10.70.24.6 255.255.255.0
negotiation auto
redundancy rii 12
redundancy group 2 ip 10.70.24.8 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 2
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



Codecs

G711ulaw is used as the preferred codec for this testing and changed the preferences according to the test plan description

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

Dial peer

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

```
dial-peer voice 100 voip
description Outgoing call from PBX to PSTN - WAN side
huntstop
destination-pattern .T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 101
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
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
```



```
dial-peer voice 200 voip
   description Outgoing call to Comcast - LAN side
   huntstop
   session protocol sipv2
   session transport udp
   incoming called-number .T
   voice-class codec 1
   voice-class sip asserted-id pai
   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
   fax-relay ecm disable
   fax rate disable
   fax nsf 000000
   fax protocol pass-through g711ulaw
   no vad
  dial-peer voice 300 voip
   description Incoming call From PSTN - WAN side
   huntstop
   session protocol sipv2
   session transport udp
   incoming called-number 856......
   voice-class codec 1
   voice-class sip asserted-id pai
   voice-class sip profiles 101
   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
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
dial-peer voice 400 voip
description Inbound from PSTN to PBX - LAN side
huntstop
destination-pattern 856......
session protocol sipv2
session target ipv4:10.64.4.111:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
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
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.

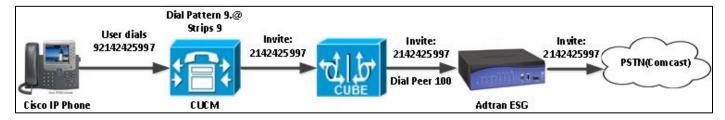


Figure 3: Outbound Voice Call

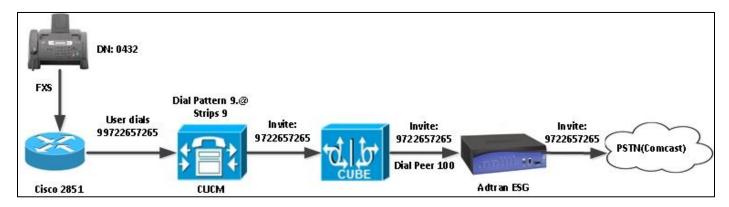


Figure 4: Outbound Fax Call



Figure 5: Inbound Voice Call

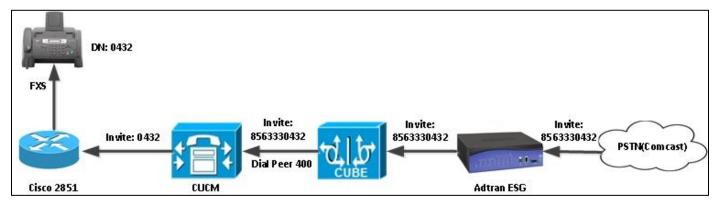


Figure 6: Inbound Fax Call

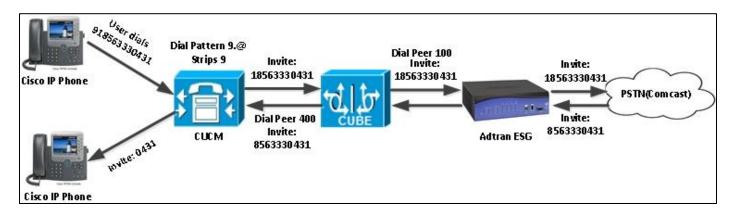


Figure 7: PBX to PBX via Comcast Call



Configuration example

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

Active Cisco UBE

```
ComcastCube1#sh running-config
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
hostname ComcastCube1
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
Ţ
address-family ipv4
exit-address-family
ļ
address-family ipv6
exit-address-family
enable secret 5 $1$Fla5$WzRjkSNL8NDpOOc7tuSBw/
no aaa new-model
```



```
no ip domain lookup
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 2
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 codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
voice class sip-profiles 101
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:856333\1@\2>
license udi pid ISR4321/K9 sn FDO19220MSQ
```



```
!
spanning-tree extend system-id
!
redundancy
mode none
application redundancy
 group 2
 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
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 MS4 1/0/9
ip address 10.64.4.155 255.255.0.0
media-type rj45
negotiation auto
redundancy rii 11
redundancy group 2 ip 10.64.4.157 exclusive
```



```
interface GigabitEthernet0/0/1
description MS4 1/0/10
ip address 10.70.24.6 255.255.255.0
negotiation auto
redundancy rii 12
redundancy group 2 ip 10.70.24.8 exclusive
interface GigabitEthernet0/1/0
description CUBE HA MS5 3/0/36
ip address 10.89.20.20 255.255.255.0
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 route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.64.0.0 255.255.0.0 10.64.1.1
ip route 10.70.0.0 255.255.0.0 10.70.24.1
ip route 172.16.0.0 255.255.0.0 10.64.1.1
```



```
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 Outgoing call from PBX to PSTN - WAN side
huntstop
destination-pattern .T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 101
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
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
```



dial-peer voice 200 voip description Outgoing call to Comcast - LAN side huntstop session protocol sipv2 session transport udp incoming called-number .T voice-class codec 1 voice-class sip asserted-id pai 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 fax-relay ecm disable fax rate disable fax nsf 000000 fax protocol pass-through g711ulaw no vad dial-peer voice 300 voip description Incoming call From PSTN - WAN side huntstop session protocol sipv2 session transport udp incoming called-number 856...... voice-class codec 1 voice-class sip asserted-id pai voice-class sip profiles 101 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
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
dial-peer voice 400 voip
description Inbound from PSTN to PBX - LAN side
huntstop
destination-pattern 856......
session protocol sipv2
session target ipv4:10.64.4.111:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
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
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
sip-ua
```



keepalive target ipv4:10.70.24.10:5060 timers keepalive active 180

sip-server ipv4:10.70.24.10:5060

!

End



Standby Cisco UBE

```
ComcastCube2#sh running-config
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
hostname ComcastCube2
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
ļ
address-family ipv6
exit-address-family
no aaa new-model
no ip domain lookup
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 2
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 codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
voice class sip-profiles 101
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 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
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 LAN MS4 1/0/11
ip address 10.64.4.156 255.255.0.0
media-type rj45
negotiation auto
redundancy rii 11
redundancy group 2 ip 10.64.4.157 exclusive
```



```
interface GigabitEthernet0/0/1
description Comcast WAN MS4 1/0/12
ip address 10.70.24.7 255.255.255.0
negotiation auto
redundancy rii 12
redundancy group 2 ip 10.70.24.8 exclusive
interface GigabitEthernet0/1/0
ip address 10.89.20.10 255.255.255.0
negotiation auto
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
negotiation auto
interface Vlan1
no ip address
shutdown
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.64.0.0 255.255.0.0 10.64.1.1
ip route 10.70.0.0 255.255.0.0 10.70.24.1
ip route 172.16.0.0 255.255.0.0 10.64.1.1
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 Outgoing call from PBX to PSTN - WAN side
huntstop
destination-pattern .T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 101
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
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
dial-peer voice 200 voip
description Outgoing call to Comcast - LAN side
```



fax-relay ecm disable

huntstop session protocol sipv2 session transport udp incoming called-number .T voice-class codec 1 voice-class sip asserted-id pai 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 fax-relay ecm disable fax rate disable fax nsf 000000 fax protocol pass-through g711ulaw no vad dial-peer voice 300 voip description Incoming call From PSTN - WAN side huntstop session protocol sipv2 session transport udp incoming called-number 856...... voice-class codec 1 voice-class sip asserted-id pai voice-class sip profiles 101 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



```
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
dial-peer voice 400 voip
description Inbound from PSTN to PBX - LAN side
huntstop
destination-pattern 856......
session protocol sipv2
session target ipv4:10.64.4.111:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
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
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
sip-ua
keepalive target ipv4:10.70.24.10:5060
timers keepalive active 180
sip-server ipv4:10.70.24.10:5060
```



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



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* = ipcsme1sub1--CUCM Voice/Video (Active)
- 2. Select Service*= Cisco CallManager (Active)
- 3. All other fields are set to default values

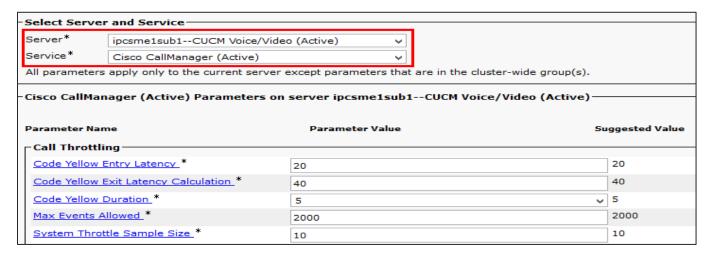


Figure 9: Service Parameters

Offnet Calls via 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: System > Security > SIP Trunk Security Profile

- 1. Name*= Comcast Non Secure SIP Trunk Profile
- 2. Description = Non Secure SIP Trunk Profile authenticated by null String

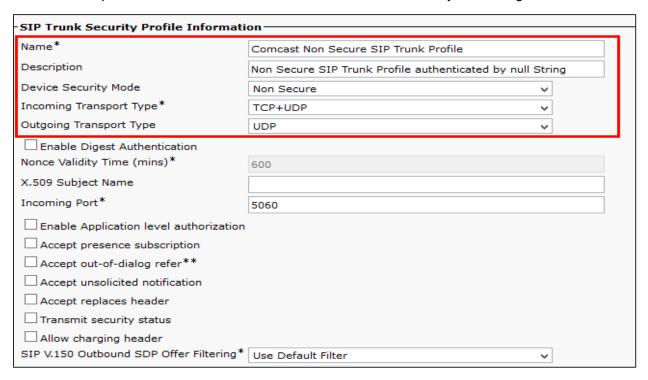


Figure 10: SIP Trunk Security Profile

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: Device > Device Settings > SIP Profile

- 1. Name*= Comcast SIP Profile
- 2. Description = Default SIP Profile

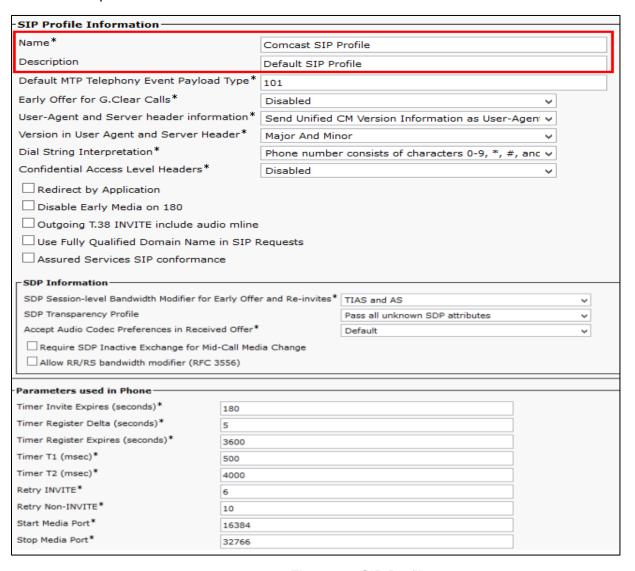


Figure 11: SIP Profile



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	V	
DTMF DB Level*	Nominal	~	
Call Hold Ring Back*	Off	<u> </u>	
Anonymous Call Block*	Off	<u> </u>	
Caller ID Blocking*	Off	~	
Do Not Disturb Control*	User	~	
Telnet Level for 7940 and 7960*	Disabled	~	
Resource Priority Namespace	< None >	v	
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		
✓ Conference Join Enabled			
RFC 2543 Hold			
✓ Semi Attended Transfer			
☐ Enable VAD			
Stutter Message Waiting			
MLPP User Authorization			
Normalization Script			
Normalization Script < None >	V		
Enable Trace			
Parameter Name		Parameter Value	
1			± =
- Incoming Dequests EDOM LIDI Settings			
Caller ID DN		1	
Caller ID DN			
Caller Name			

Figure 12: SIP Profile (Cont.)



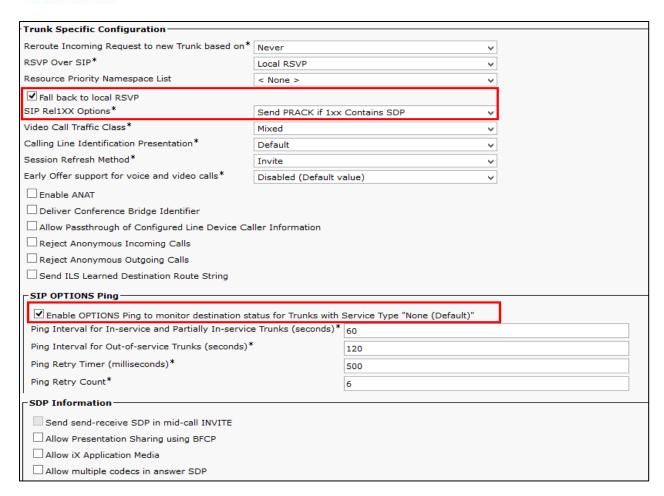


Figure 13: SIP Profile (Cont.)

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 Path:** Device > Trunk

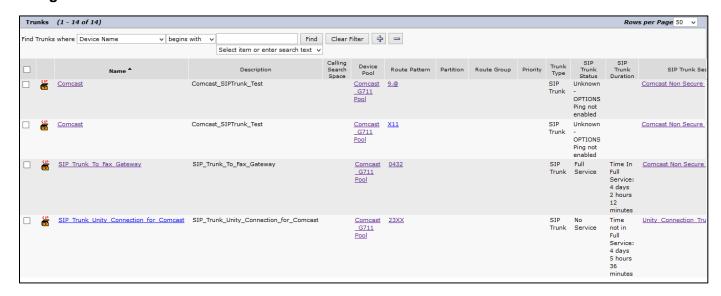


Figure 14: SIP Trunks List



SIP Trunk Status		
Service Status: Full Service		
Duration: Time In Full Service: 0 day 0 hour 13 minutes		
Device Information		
Product:	SIP Trunk	
Device Protocol:	SIP	
Trunk Service Type	None(Default)	
Device Name*	Comcast	
Description	Comcast_SIPTrunk_Test	
Device Pool*	Comcast _G711 Pool	▽
Common Device Configuration	< None >	▽
Call Classification*	Use System Default	<u> </u>
Media Resource Group List	MRG_MTP_Group	▽
Location*	Hub_None	<u> </u>
AAR Group	< None >	▼
Tunneled Protocol*	None	▽
QSIG Variant*	No Changes	▼
ASN.1 ROSE OID Encoding*	No Changes	<u> </u>
Packet Capture Mode*	None	~
Packet Capture Duration	0	
Media Termination Point Required		
▼ Retry Video Call as Audio		
Path Replacement Support		
☐ Transmit UTF-8 for Calling Party Name		
Transmit UTF-8 Names in QSIG APDU		
Unattended Port		
SRTP Allowed - When this flag is checked, Encrypted TLS needs	to be configured in the network to provide er	d to end security. Failure to do so will expose keys
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS	V
Route Class Signaling Enabled*	Default	▼
Use Trusted Relay Point*	Default	~
✓ PSTN Access		
Run On All Active Unified CM Nodes		
-Intercompany Media Engine (IME)		
E.164 Transformation Profile < None >	<u> </u>	
MLPP and Confidential Access Level Information		
Thore P	<u> </u>	
Confidential Access Mode < None >	<u> </u>	
Confidential Access Level < None >	V	

Figure 15: SIP Trunk to Cisco UBE



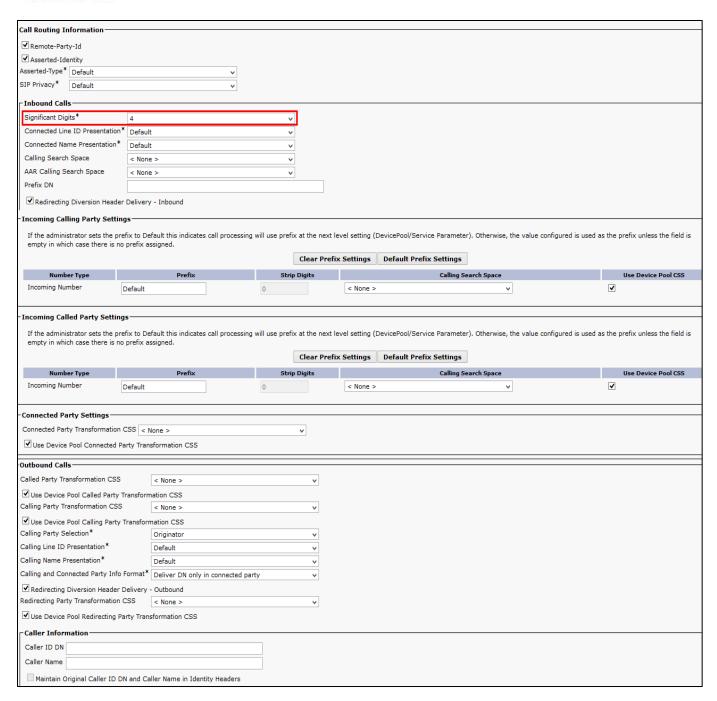


Figure 16: SIP Trunk to Cisco UBE (Cont.)



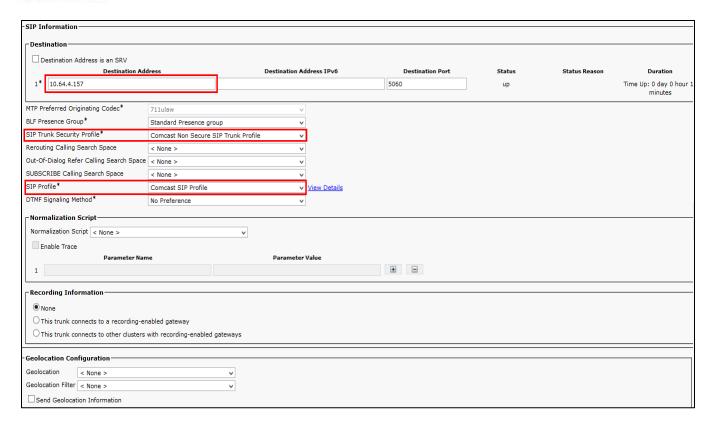


Figure 17: SIP Trunk to Cisco UBE (Cont.)

Explanation

Parameter	Value	Description
Device Name	Comcast	Name for the trunk
Device Pool	Comcast_G711 Pool	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	10.64.4.157	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security	Comcast Non Secure SIP	SIP Trunk Security Profile configured
Profile	Trunk Profile	earlier
SIP Profile	Comcast 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 "9"+10 digits number to access PSTN via Cisco UBE
 - o "9" is removed before sending to Cisco UBE
- 2. For FAX call, Access Code "9"+10 digits number is used at Cisco Fax gateway
 - o "9" is removed at Cisco UCM
 - o The rest of the number is sent to Cisco UBE to Comcast network
- 3. Incoming fax call to 0432 will be sent to Cisco Fax gateway
- Cisco IP phones dial 4XX and 9XX for emergency call and will send all digits to Cisco UBE to Comcast Network



Figure 18: Route Patterns List



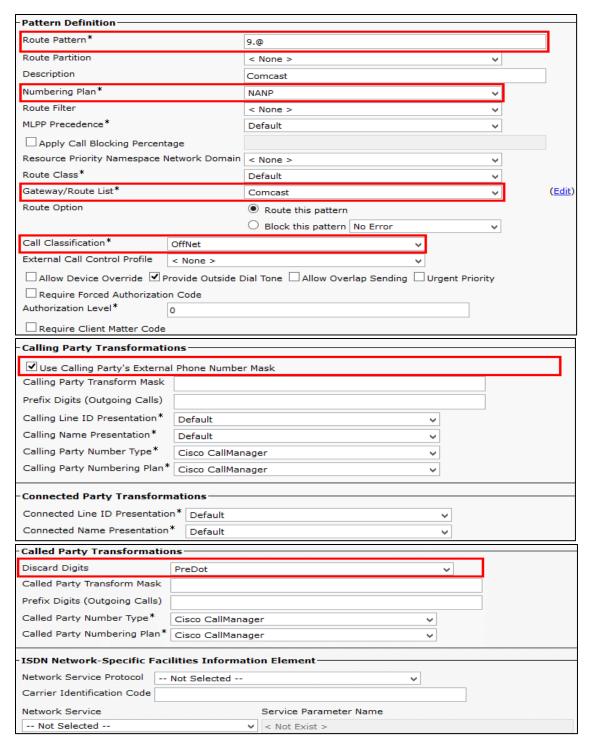


Figure 19: Route Pattern for Voice



Pattern Definition						
Route Pattern*		X11				
Route Partition		< No	one >	~		
Description		Com	cast_Emergency_calls			
Numbering Plan		N	ot Selected	~		
Route Filter		< No	one >	~		
MLPP Precedence*		Defa	ault	~		
Apply Call Blocking Percent	age				,	
Resource Priority Namespace N	_	< No	one >	~		
Route Class*		Defa	ault	~		
Gateway/Route List*		Con	ncast	~	(E	Edit)
Route Option		● I	Route this pattern			
		O 1	Block this pattern No Error	~		
Call Classification*	OffNet		>			
External Call Control Profile	< None >		>			
☐ Allow Device Override ☑ P	rovide Outside [Dial To	one Allow Overlap Sending U	rgent Priority		
Require Forced Authorization	n Code					
Authorization Level*	0					
Require Client Matter Code						
-Calling Party Transformation	s					
Use Calling Party's External P	hone Number Ma	sk				
Calling Party Transform Mask						
Prefix Digits (Outgoing Calls)						
Calling Line ID Presentation*	Default		V			
Calling Name Presentation*	Default		<u> </u>			
Calling Party Number Type*	Cisco CallManage	r	~			
Calling Party Numbering Plan*						
Connected Party Transformations						
Connected Line ID Presentation*						
Connected Name Presentation*	Default		~			
• Called Party Transformati Discard Digits						
Called Party Transform Mask	< None >				7	
]	
Prefix Digits (Outgoing Calls)]	
Called Party Number Type*	Cisco CallMa			~		
Called Party Numbering Plan	Cisco CallMa	nage	er	~		
ISDN Network-Specific Facilities Information Element						
Network Service Protocol Not Selected						
Carrier Identification Code						
Network Service			Service Parameter Name			
Not Selected		v	< Not Exist >			
		-				

Figure 20: Route Pattern for Voice (Cont.)



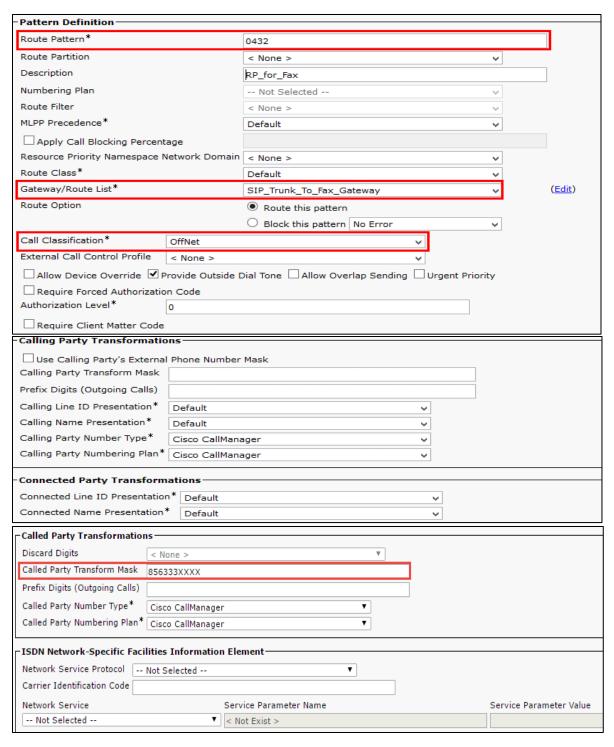


Figure 21: Route Pattern for Fax





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	Comcast for Route Pattern 9.@, X11 and SIP_Trunk_To_Fax_Gateway 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
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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