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:

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.

```plaintext
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
```

<table>
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<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>fax protocol</td>
<td>Specifies the fax protocol</td>
</tr>
<tr>
<td>asserted-id</td>
<td>Specifies the type of privacy header in the outgoing SIP requests and response messages</td>
</tr>
<tr>
<td>early-offer forced</td>
<td>Enables SIP Delayed-Offer to Early-Offer globally</td>
</tr>
<tr>
<td>midcall-signaling passthru</td>
<td>Passes SIP messages from one IP leg to another IP leg</td>
</tr>
</tbody>
</table>
**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-nce
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711-ulaw
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-nce
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711-ulaw
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
!
address-family ipv4
exit-address-family
!
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-n te
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 asserte- 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-n te
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
!
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
huntstop
session protocol sipv2
session transport udp
incoming called-number .T
voice-class codec 1
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voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
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dial-peer voice 300 voip
description Incoming call From PSTN - WAN side
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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
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voice-class sip options-keepalive
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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

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

![Cisco UCM Version](image)

**Figure 8:** Cisco UCM Version

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

![Service Parameters](image)

**Figure 9:** Service Parameters
**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

![SIP Trunk Security Profile Information](image)

**Explanation**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Transport Type</td>
<td>TCP + UDP</td>
<td>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.</td>
</tr>
<tr>
<td>Outgoing Transport Type</td>
<td>UDP</td>
<td></td>
</tr>
</tbody>
</table>

Figure 10: SIP Trunk Security Profile
**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

![SIP Profile Information Table]

- **Name:** Comcast SIP Profile
- **Description:** Default SIP Profile
- **Default MTP Telephony Event Payload Type:** 101 (Disabled)
- **Early Offer for G.729 Calls:** Enabled
- **User-Agent and Server header information:** Send Unified CM Version Information as User-Agent
- **Version in User Agent and Server Header:** Major And Minor
- **Dial String Interpretation:** Phone number consists of characters 0-9, *, #, and v
- **Redirect by Application:** Disabled
- **Disable Early Media on 180:** Disabled
- **Outgoing T.38 INVITE include audio min:** Disabled
- **Use Fully Qualified Domain Name in SIP Requests:** Disabled
- **Assured Services SIP conformance:** Disabled

![SDP Information Table]

- **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:** Disabled
- **Allow RR/RS bandwidth modifier (RFC 3556):** Disabled

![Parameters used in Phone Table]

- **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
- **Start Media Port:** 16384
- **Stop Media Port:** 32766

**Figure 11: SIP Profile**
<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Pickup URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>x-cisco-serviceuri-meetme</td>
</tr>
<tr>
<td>User Info</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level</td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control</td>
<td>User</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7960</td>
<td>Disabled</td>
</tr>
<tr>
<td>Resource Priority Namespace</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirections</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>x-cisco-serviceuri-cfwdall</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>x-cisco-serviceuri-abbrevdial</td>
</tr>
</tbody>
</table>

**Normalization Script**

| Normalization Script | < None > |

**Incoming Requests FROM URI Settings**

| Caller ID DN | Caller Name |

*Figure 12: SIP Profile (Cont.)*
### Figure 13: SIP Profile (Cont.)

#### Explanation

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

<table>
<thead>
<tr>
<th>Device Information</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Product:</strong></td>
<td>SIP Trunk</td>
<td></td>
</tr>
<tr>
<td><strong>Device Protocol:</strong></td>
<td>SIP</td>
<td></td>
</tr>
<tr>
<td><strong>Trunk Service Type:</strong></td>
<td>None (Default)</td>
<td></td>
</tr>
<tr>
<td><strong>Device Name:</strong></td>
<td>Comcast</td>
<td></td>
</tr>
<tr>
<td><strong>Description:</strong></td>
<td>Comcast_SIPTrunk_Test</td>
<td></td>
</tr>
<tr>
<td><strong>Device Pool:</strong></td>
<td>Comcast_0711_Pool</td>
<td></td>
</tr>
<tr>
<td><strong>Common Device Configuration:</strong></td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td><strong>Call Classification:</strong></td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td><strong>Media Resource Group List:</strong></td>
<td>MRG_MTP_Group</td>
<td></td>
</tr>
<tr>
<td><strong>Location:</strong></td>
<td>Hub_None</td>
<td></td>
</tr>
<tr>
<td><strong>AAR Group:</strong></td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td><strong>Tunnel Protocol:</strong></td>
<td>None</td>
<td></td>
</tr>
<tr>
<td><strong>QSIG Variant:</strong></td>
<td>No Changes</td>
<td></td>
</tr>
<tr>
<td><strong>ASN.1 ROSE CID Encoding:</strong></td>
<td>No Changes</td>
<td></td>
</tr>
<tr>
<td><strong>Packet Capture Mode:</strong></td>
<td>None</td>
<td></td>
</tr>
<tr>
<td><strong>Packet Capture Duration:</strong></td>
<td>0</td>
<td></td>
</tr>
</tbody>
</table>

- Media Termination Point Required
- Retry Video Call on Audio
- Path Replacement Support
- Transmit UDT-8 for Calling Party Name
- Transmit UDT-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 end to end security. Failure to do so will expose keys and traffic.
- Consider Traffic on This Trunk Secure
- Route Class Signaling Enabled
- Use Trusted Relay Point
- PSTN Access
- Run On All Active Unified CH Nodes

<table>
<thead>
<tr>
<th>Intercompany Media Engine (IME)</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>E.164 Transformation Profile:</strong></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>MLPP and Confidential Access Level Information</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>MLPP Domain:</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Confidential Access Mode:</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Confidential Access Level:</strong></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

---

**Figure 15: SIP Trunk to Cisco UBE**
**Call Routing Information**

- **Remote-Party-ID**
- **Asserted Identity**
- **Asserted-Type** *(Default)*
- **SIP Privacy** *(Default)*

**Inbound Calls**

- **Signaling Digits** *(4)*
- **Connected Line ID Presentation** *(Default)*
- **Connected Name Presentation** *(Default)*
- **Calling Search Space** *(< None >)*
- **AAR Calling Search Space** *(< None >)*
- **Prefix DN**

- **Redirecting Diversion Header Delivery - Inbound**

**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.

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td>✓</td>
</tr>
</tbody>
</table>

**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.

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td>✓</td>
</tr>
</tbody>
</table>

**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**
- **Called Party Transformation CSS** *(< None >)*
- **Use Device Pool Called Party Transformation CSS**
- **Called Party Selection** *(Originator)*
- **Calling Line ID Presentation** *(Default)*
- **Calling Name Presentation** *(Default)*
- **Calling and Connected Party Info Format** *(Deliver DN only in connected party)*
- **Redirecting Diversion Header Delivery - Outbound**
- **Redirecting Party Transformation CSS** *(< None >)*
- **Use Device Pool Redirecting Party Transformation CSS**

**Caller Information**

- **Caller ID ON**
- **Caller Name**

- **Maintain Original Caller ID ON and Caller Name in Identity Headers**

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

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Name</td>
<td>Comcast</td>
<td>Name for the trunk</td>
</tr>
<tr>
<td>Device Pool</td>
<td>Comcast_G711 Pool</td>
<td>Default Device Pool is used for this trunk</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL_MTP_Group</td>
<td>MRG with resources: ANN, CFB, MOH and MTP</td>
</tr>
<tr>
<td>Significant Digits</td>
<td>4</td>
<td>4 digits Extension for all CPE phones</td>
</tr>
<tr>
<td>Destination Address</td>
<td>10.64.4.157</td>
<td>IP address of the Cisco UBE Virtual LAN</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
<td>Comcast Non Secure SIP Trunk Profile</td>
<td>SIP Trunk Security Profile configured earlier</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>Comcast SIP Profile</td>
<td>SIP Profile configured earlier</td>
</tr>
</tbody>
</table>
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
     - “9” is removed before sending to Cisco UBE
  2. 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
  3. Incoming fax call to 0432 will be sent to Cisco Fax gateway
  4. Cisco IP phones dial 4XX and 9XX for emergency call and will send all digits to Cisco UBE to Comcast Network

<table>
<thead>
<tr>
<th>Pattern</th>
<th>Description</th>
<th>Partition</th>
<th>Route Filter</th>
<th>Associated Device</th>
</tr>
</thead>
<tbody>
<tr>
<td>0432</td>
<td>RP_for_Fax</td>
<td></td>
<td></td>
<td>SIP_Trunk_To_Fax_Gateway</td>
</tr>
<tr>
<td>9-0</td>
<td>Comcast</td>
<td></td>
<td></td>
<td>Comcast</td>
</tr>
<tr>
<td>X11</td>
<td>Comcast_Emergency_calls</td>
<td></td>
<td></td>
<td>Comcast</td>
</tr>
</tbody>
</table>

Figure 18: Route Patterns List
Figure 19: Route Pattern for Voice
Figure 20: Route Pattern for Voice (Cont.)
### Figure 21: Route Pattern for Fax

<table>
<thead>
<tr>
<th>Pattern Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Route Pattern</strong></td>
</tr>
<tr>
<td><strong>Route Partition</strong></td>
</tr>
<tr>
<td><strong>Description</strong></td>
</tr>
<tr>
<td><strong>Numbering Plan</strong></td>
</tr>
<tr>
<td><strong>Route Filter</strong></td>
</tr>
<tr>
<td><strong>MLPP Precedence</strong></td>
</tr>
<tr>
<td><strong>Apply Call Blocking Percentage</strong></td>
</tr>
<tr>
<td><strong>Resource Priority Namespace Network Domain</strong></td>
</tr>
<tr>
<td><strong>Route Class</strong></td>
</tr>
<tr>
<td><strong>Gateway/Route List</strong></td>
</tr>
<tr>
<td><strong>Route Option</strong></td>
</tr>
<tr>
<td><strong>Call Classification</strong></td>
</tr>
<tr>
<td><strong>External Call Control Profile</strong></td>
</tr>
<tr>
<td><strong>Allow Device Override</strong></td>
</tr>
<tr>
<td><strong>Provide Outside Dial Tone</strong></td>
</tr>
<tr>
<td><strong>Allow Overlap Sending</strong></td>
</tr>
<tr>
<td><strong>Urgent Priority</strong></td>
</tr>
<tr>
<td><strong>Require Forced Authorization Code</strong></td>
</tr>
<tr>
<td><strong>Authorization Level</strong></td>
</tr>
<tr>
<td><strong>Require Client Meter Code</strong></td>
</tr>
</tbody>
</table>

### Calling Party Transformations
- **Use Calling Party's External Phone Number Mask**
- **Calling Party Transform Mask**
- **Prefix Digits (Outgoing Calls)**
- **Calling Line ID Presentation**
- **Calling Name Presentation**
- **Calling Party Number Type**
- **Calling Party Numbering Plan**

### Connected Party Transformations
- **Connected Line ID Presentation**
- **Connected Name Presentation**

### Called Party Transformations
- **Discard Digits**
- **Called Party Transform Mask**
- **Prefix Digits (Outgoing Calls)**
- **Called Party Number Type**
- **Called Party Numbering Plan**

### ISDN Network-Specific Facilities Information Element
- **Network Service Protocol**
- **Network Service**
- **Service Parameter Name**
- **Service Parameter Value**
### Explanation

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>9.@ for Voice &amp; International Calls, 0432 for Fax Call and X11 for Operator Call and Emergency Services</td>
<td>Specify appropriate Route Pattern</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>Comcast for Route Pattern 9.@, X11 and SIP_Trunk_To_Fax_Gateway for Route Pattern 0432</td>
<td>SIP Trunk name configured earlier</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>NANP for Route Pattern 9.@</td>
<td>North American Numbering Plan</td>
</tr>
<tr>
<td>Call Classification</td>
<td>OffNet for Route Pattern 9.@, 0432 and X11</td>
<td>Restrict the transferring of an external call to an external device</td>
</tr>
<tr>
<td>Discard Digits</td>
<td>PreDot for Route Pattern 9.@</td>
<td>Specifies how to modify digit before they are sending to Comcast network</td>
</tr>
</tbody>
</table>

### Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>CPE</td>
<td>Customer Premise Equipment</td>
</tr>
<tr>
<td>CUBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>CUCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>MTP</td>
<td>Media Termination Point</td>
</tr>
<tr>
<td>POP</td>
<td>Point of Presence</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
</tbody>
</table>
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<table>
<thead>
<tr>
<th>Corporate Headquarters</th>
<th>European Headquarters</th>
<th>Americas Headquarters</th>
<th>Asia Pacific Headquarters</th>
</tr>
</thead>
<tbody>
<tr>
<td>170 West Tasman Drive</td>
<td>Haarlerbergpark</td>
<td>170 West Tasman Drive</td>
<td>Capital Tower</td>
</tr>
<tr>
<td>San Jose, CA 95134-1706</td>
<td></td>
<td>San Jose, CA 95134-1706</td>
<td>168 Robinson Road</td>
</tr>
</tbody>
</table>