Comcast SIP Trunking:

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

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Introduction

Service Providers today, such as Comcast, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and centralized IP to TDM POP gateways to provide on-net and off-net services.

Comcast is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity. A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and Comcast network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS-XE 16.06.02 can be used. The Cisco Unified Border Element 12.0 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 12.0.1 connected to Comcast IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco UCM (Cisco Unified Communications Manager). Only configuration settings specifically required for Comcast interoperability are presented. Feature configuration and most importantly the dial plan are customer specific and need individual approach.

- This application note describes how to configure a Cisco Unified Communications Manager (Cisco UCM) 12.0.1 and Cisco Unified Border Element (Cisco UBE) on ISR 4321/K9 [IOS-XE – 16.06.02] for connectivity to Comcast SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 12.0.1) to PSTN (Comcast).

- Testing was performed in accordance to Comcast generic SIP Trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold (MOH), unattended and attended transfers, call forward, conferences and interoperability with Cisco Unity Connection (CUC)

- The Cisco UCM configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Comcast SIP network and Cisco Unified Communications. The configuration described in this document details the important configuration settings to have enabled for interoperability to be successful and care must be taken by the network administrator deploying Cisco UCM to interoperate to Comcast SIP Trunking network.

This application note does not cover the use of Calling Search Spaces (CSS) or partitions on Cisco UCM. To understand and learn how to apply CSS and partitions refer to the cisco.com link below:

Cisco IP Phones 7942, 9951 and 9971 phones are the devices primarily used throughout the testing to place or receive calls.

VentaFax Soft Client is used to perform all fax related scenarios. The fax client is connected to SIP Gateway (Cisco ATA) via FXS port which in turn communicates with Cisco UCM over SIP.
System Components

Hardware Requirements
- Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4321/K9 router as CUBE
- Cisco ATA SPA112
- IP phones 9951 (SIP), 9971 (SIP) and 7942 (SCCP)
- Comcast(Adtran) ESG – Provided and managed by Comcast

Software Requirements
- Cisco Unified Communications Manager 12.0.1
- Cisco Unity Connection 12.0.1
- IOS-XE 16.06.02 for ISR 4321/K9 Cisco Unified Border Element
- Firmware Version 1.3.5 (004p) for Cisco ATA SPA112
- Comcast(Adtran) ESG R11.4.3.V – Provided and managed by Comcast

Features

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

Features Not Supported
- Cisco IP phones used in this test do not support blind transfer
- Comcast does not support Fax with SIP service

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

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

interface GigabitEthernet0/0/0
description ComcastCube WAN
ip address 192.65.79.134 255.255.255.128
shutdown
media-type rj45
negotiation auto
redundancy rii 15
redundancy group 1 ip 192.65.79.185 exclusive
!
interface GigabitEthernet0/0/1
description ComcastCube LAN
ip address 10.80.18.48 255.255.255.0
negotiation auto
redundancy rii 16
redundancy group 1 ip 10.80.18.50 exclusive
!
!
Global Cisco UBE Settings
In order to enable Cisco UBE IP2IP gateway functionality, enter the following:

!  !
voice service voip
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip refer
supplementary-service media-renegotiate
sip
session refresh
asserted-id pai
privacy pstn
midcall-signaling passthru
g729 annexb-all
!
!

Explanation

<table>
<thead>
<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>midcall-signaling passthru</td>
<td>Passes SIP messages from one IP leg to another IP leg</td>
</tr>
</tbody>
</table>
Media Passing through Cisco UBE (media flow-through vs. media flow-around)

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

voice service voip

media flow-around

Codecs

G711ulaw is used as the preferred codec for this testing.

voice class codec 1

codec preference 1 g711ulaw

codec preference 2 g729r8

Dial Peer

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

!

dial-peer voice 100 voip

description Outbound from CUCM to CUBE LAN

session protocol sipv2

session transport udp

incoming called-number .T

voice-class codec 1

voice-class sip bind control source-interface GigabitEthernet0/0/1

voice-class sip bind media source-interface GigabitEthernet0/0/1

dtmf-relay rtp-nte

fax protocol pass-through g711ulaw

no vad

!

dial-peer voice 101 voip

description Outbound from CUBE WAN to Adtran

destination-pattern .T

session protocol sipv2

session target ipv4:10.64.3.115
session transport udp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n-nte
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 102 voip
description Incoming from Adtran to CUBE WAN
session protocol sipv2
incoming called-number 856333....
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n-nte
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 103 voip
description Incoming from CUBE LAN to CUCM
destination-pattern 856333....
session protocol sipv2
session target ipv4:10.80.10.2
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-n-nte
fax protocol pass-through g711ulaw
no vad
!
Call Flow

In the sample configuration presented here, Cisco UCM is provisioned with four-digit directory numbers corresponding to the last four DID digits. No digit manipulation is performed on the Cisco UBE.

For incoming PSTN calls, the Cisco UBE presents the full ten-digit DID number to Cisco UCM. The Cisco UCM picks up the last 4 significant Digits configured under SIP Trunk and routes the call based on those 4 digits. Voice calls are routed to IP phones; Fax calls are routed via a 4-digit route pattern over a SIP trunk that terminates on the Fax Gateway and in turn to the VentaFax client connected to the Fax Gateway.

CPE callers make outbound PSTN calls by dialing a “9” prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, Cisco fax Gateway sends to Cisco UCM the DID with leading access code “9”. A “9.@” route pattern strips the prefix and routes the call with the remaining digits via a SIP trunk terminating on the Cisco UBE for Voice call or Fax. For PBX to PBX via Comcast, Caller dial 9 prefix followed by the target 1+10-digits number, 9 was stripped and the remaining digits were send to Cisco UBE. Cisco UBE pass the DID under Dial Peer 101 and send to Comcast network which will direct back to Cisco UBE and handled same as normal incoming PSTN call.

Figure 2: Outbound Voice Call

Figure 3: Inbound Voice Call
Figure 4: Outbound Fax Call

Figure 5: Inbound Fax Call

Figure 6: PBX to PBX via Comcast Call
Configuration Example
The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously

Active Cisco UBE

ComcastCube1>en
Password:
ComcastCube1#show running-config
!
version 16.6
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname ComcastCube1
!
boot-start-marker
boot system flash isr4300-universalk9.16.06.02.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no logging queue-limit
logging buffered 1000000
no logging rate-limit
no logging monitor
enable secret 5 $1
!
aaa new-model
!
aaa session-id common
!
multilink bundle-name authenticated
!
voice service voip
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip refer
supplementary-service media-renegotiate
sip
  session refresh
  asserted-id pai
privacy pstn
  midcall-signaling passthru
  g729 annexb-all
!
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g729r8
!
license udi pid ISR4321/K9 sn FDO19244MQ8
diagnostic bootup level minimal
spanning-tree extend system-id
!
!
redundancy
mode none
application redundancy
group 1
name b2bHAcomcast
priority 100 failover threshold 75
timers delay 30 reload 60
control GigabitEthernet0/1/0 protocol 1
data GigabitEthernet0/1/0
track 1 shutdown
track 2 shutdown
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
!
interface GigabitEthernet0/0/0
description ComcastCube WAN
ip address 192.65.79.134 255.255.255.128
shutdown
media-type rj45
negotiation auto
redundancy rii 15
redundancy group 1 ip 192.65.79.185 exclusive
!
interface GigabitEthernet0/0/1
description ComcastCube LAN
ip address 10.80.18.48 255.255.255.0
negotiation auto
redundancy rii 16
redundancy group 1 ip 10.80.18.50 exclusive

interface GigabitEthernet0/1/0
description ComcastCube HA Interface
ip address 10.70.50.100 255.255.255.0
negotiation auto

interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto

ip forward-protocol nd
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.70.50.0 255.255.255.0 10.80.18.1
ip route 10.80.10.0 255.255.255.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 10.80.18.1

!
control-plane

mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable

mgcp profile default

dial-peer voice 100 voip
  description Outbound from CUCM to CUBE LAN
  session protocol sipv2
  session transport udp
  incoming called-number .T
  voice-class codec 1
  voice-class sip bind control source-interface Gigabitethernet0/0/1
  voice-class sip bind media source-interface Gigabitethernet0/0/1
dtmf-relay rtp-nte
  fax protocol pass-through g711ulaw
  no vad

dial-peer voice 101 voip
  description Outbound from CUBE WAN to Adtran
  destination-pattern .T
  session protocol sipv2
  session target ipv4:10.64.3.115
session transport udp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n-te
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 102 voip
description Incoming from Adtran to CUBE WAN
session protocol sipv2
incoming called-number 856333....
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n-te
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 103 voip
description Incoming from CUBE LAN to CUCM
destination-pattern 856333....
session protocol sipv2
session target ipv4:10.80.10.2
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-n-te
fax protocol pass-through g711ulaw
no vad
!
! gateway
  timer receive-rtp 1200
!
sip-ua
!
!
line con 0
  password password
  transport input none
  stopbits 1
line aux 0
  password password
  stopbits 1
line vty 0 4
  exec-timeout 0 0
  password password
  transport preferred telnet
!
wsma agent exec
!
wsma agent config
!
wsma agent filesys
!
wsma agent notify
!
end

ComcastCube1#
Standby Cisco UBE

ComcastCube2#show running-config

!

version 16.6

service timestamps debug datetime msec localtime

service timestamps log datetime msec localtime

service password-encryption

service internal

service sequence-numbers

platform qfp utilization monitor load 80

no platform punt-keepalive disable-kernel-core

!

hostname ComcastCube2

!

boot-start-marker

boot system flash isr4300-universalk9.16.06.02.SPA.bin

boot-end-marker

!

vrf definition Mgmt-intf

!

address-family ipv4

exit-address-family

!

address-family ipv6

exit-address-family

!

no logging queue-limit
logging buffered 30000000
no logging rate-limit
no logging console
logging monitor notifications
enable secret 5 $1$aEKr$TYnj4bdYn.
!
no aaa new-model
subscriber templating
!
multilink bundle-name authenticated
!
voice service voip
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip refer
supplementary-service media-renegotiate
sip
session refresh
privacy pstn
conn-reuse
midcall-signaling passthru
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
license udi pid ISR4321/K9 sn FDO19440MW3
license boot level appxk9
license boot level uck9
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 7 13111A
!
redundancy
mode none
application redundancy
group 1
   name b2bHAComcast
   priority 150 failover threshold 75
timers delay 30 reload 60
control GigabitEthernet0/1/0 protocol 1
data GigabitEthernet0/1/0
track 1 shutdown
track 2 shutdown
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
!
!
!
interface GigabitEthernet0/0/0
description ComcastCube WAN
ip address 192.65.79.135 255.255.255.128
media-type rj45
negotiation auto
redundancy rii 15
redundancy group 1 ip 192.65.79.185 exclusive
!
interface GigabitEthernet0/0/1
description ComcastCube LAN
ip address 10.80.18.49 255.255.255.0
negotiation auto
redundancy rii 16
redundancy group 1 ip 10.80.18.50 exclusive
!
interface GigabitEthernet0/1/0
description ComcastCube HA Interface
ip address 10.70.50.110 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
ip forward-protocol nd
no ip http server
no ip http secure-server

ip tftp source-interface GigabitEthernet0

ip route 0.0.0.0 0.0.0.0 192.65.79.129

ip route 10.64.0.0 255.255.0.0 10.80.18.1

ip route 10.70.50.0 255.255.255.0 10.80.18.1

ip route 10.80.10.0 255.255.255.0 10.80.18.1

ip route 172.16.24.0 255.255.248.0 10.80.18.1

!

!

!

ip access-list standard my_access_list

   permit any

!

!

!

!

control-plane

!

!

!

!

!

mgcp behavior rsip-range tgcp-only

mgcp behavior comedia-role none

mgcp behavior comedia-check-media-src disable

mgcp behavior comedia-sdp-force disable

!

mgcp profile default

!

!
!telephony-service
max-conferences 8 gain -6
transfer-system full-consult
!
!
dial-peer voice 100 voip
description Outbound from CUCM to CUBE LAN
session protocol sipv2
session transport udp
incoming called-number .T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 101 voip
description Outbound from CUBE WAN to Adtran
destination-pattern .T
session protocol sipv2
session target ipv4:10.64.3.115
session transport udp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 102 voip
description Incoming from Adtran to CUBE WAN
session protocol sipv2
incoming called-number 856333....
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 103 voip
description Incoming from CUBE LAN to CUCM
destination-pattern 856333....
session protocol sipv2
session target ipv4:10.80.10.2
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
no vad
!
!
gateway
timer receive-rtp 1200
!
!
line con 0
exec-timeout 0 0
login local
transport input none
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 0 0
login local
transport input telnet ssh
!
wsma agent exec
!
wsma agent config
!
wsma agent filesys
!
wsma agent notify
!
!
end

ComcastCube2#
Configuring Cisco Unified Communications Manager

Cisco UCM Version

![Cisco UCM Version](image)

**Figure 7: Cisco UCM Version**

Cisco Call Manager Service Parameters

**Navigation path:** System > Service Parameters

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

Select Service* = Cisco CallManager (Active)

All other fields are set to default values

![Service Parameters](image)

**Figure 8: Service Parameters**
Off-net Calls via Comcast SIP Trunk

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

SIP Trunk Security Profile

Navigation Path: System > Security > SIP Trunk Security Profile
Name* = Comcast Non Secure SIP Trunk Profile
Description = Non Secure SIP Trunk Profile

![SIP Trunk Security Profile Configuration](image)

Figure 9: SIP Trunk Security Profile

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>

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SIP Profile Configuration
SIP Profile will be later associated with the SIP trunk

**Navigation Path:** Device > Device Settings > SIP Profile

Name* = Standard SIP Profile-Comcast
Description = Default SIP Profile

![SIP Profile Configuration](image)

**Figure 10:** SIP Profile
<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>DSCP for Audio Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup 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>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>5000</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>x-cisco-serviceuri-sfwdall</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>x-cisco-serviceuri-sbdial</td>
</tr>
</tbody>
</table>

 normalization script

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td></td>
</tr>
</tbody>
</table>

- **Incoming Requests FROM URI Settings**

<table>
<thead>
<tr>
<th>Caller ID DN</th>
<th>Caller Name</th>
</tr>
</thead>
</table>

**Figure 11: SIP Profile (Cont.)**
## Trunk Specific Configuration

<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 if 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>

### 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 if 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 13: SIP Trunks List**

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Device Pool</th>
<th>Route Pattern</th>
<th>Partition</th>
<th>Route Group</th>
<th>Priority</th>
<th>Trunk Type</th>
<th>SIP Trunk Status</th>
<th>SIP Trunk Duration</th>
<th>SIP Trunk Security Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>ToComcastATAP</td>
<td>ToComcastATAP</td>
<td>6711</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Unknown</td>
<td></td>
<td>Non Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>To_Comcast_CUBE</td>
<td>To_Comcast_CUBE</td>
<td>6711</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Unknown</td>
<td></td>
<td>Non Secure SIP Trunk Profile</td>
</tr>
</tbody>
</table>
Figure 14: SIP Trunk to Cisco UBE
### Intercompany Media Engine (IME)

- **E.164 Transformation Profile**: <None>

### MLPP and Confidential Access Level Information

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>MLPP Domain</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Confidential Access Mode</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Confidential Access Level</td>
<td>&lt;None&gt;</td>
</tr>
</tbody>
</table>

### Call Routing Information

- Remote-Party-Id
- Asserted-Identity
- **Asserted-Type**: Default
- **SIP Privacy**: Default

#### Inbound Calls

<table>
<thead>
<tr>
<th><strong>Significant Digits</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Connected Line ID Presentation</strong>: Default</td>
<td></td>
</tr>
<tr>
<td><strong>Connected Name Presentation</strong>: Default</td>
<td></td>
</tr>
<tr>
<td><strong>Calling Search Space</strong>: &lt;None&gt;</td>
<td></td>
</tr>
<tr>
<td><strong>AAR Calling Search Space</strong>: &lt;None&gt;</td>
<td></td>
</tr>
</tbody>
</table>

- 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><strong>Incoming Number</strong></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><strong>Incoming Number</strong></td>
<td>Default</td>
<td>0</td>
<td>&lt;None&gt;</td>
<td>✓</td>
</tr>
</tbody>
</table>

---

Figure 15: SIP Trunk to Cisco UBE (cont.)
Figure 16: SIP Trunk to Cisco UBE (Cont.)
Explanations

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Name</td>
<td>To_Comcast_CUBE</td>
<td>Name for the trunk</td>
</tr>
<tr>
<td>Device Pool</td>
<td>G711 Preferred</td>
<td>Default Device Pool is used for this trunk</td>
</tr>
<tr>
<td>Media Resource Group</td>
<td>MRGL</td>
<td>MRG with resources: ANN, CFB, MOH and MTP</td>
</tr>
<tr>
<td>List</td>
<td></td>
<td></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>10.80.18.50</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>Standard SIP Profile-Comcast</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:

- Cisco IP phone dial “9”+10 digits number to access PSTN via Cisco UBE
  - “9” is removed before sending to Cisco UBE
- For FAX call, Access Code “9”+10 digits number is used at Cisco Fax gateway
  - “9” is removed at Cisco UCM
  - The rest of the number is sent to Cisco UBE to Comcast network
- Incoming fax call to 0418 will be sent to Cisco ATA
- Cisco IP phones dial X11 for emergency call and will send all digits to Cisco UBE to Comcast network

![Figure 17: Route Patterns List](image-url)
Figure 18: Route Pattern for Voice
Figure 19: Route Pattern for Voice (Cont.)
Figure 20: Route Pattern for Fax
## 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, 0418 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>To_Comcast_CUBE for Route Pattern 9.@, X11 and T0ComcastFaxATA for Route Pattern 0418</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.@, 0418 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>Cisco UBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>Cisco UCM</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>ESBC</td>
<td>Enterprise Session Border Controller</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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<th>European Headquarters</th>
<th>Americas Headquarters</th>
<th>AsiaPacific Headquarters</th>
</tr>
</thead>
<tbody>
<tr>
<td>170 West Tasman Drive</td>
<td>International BV</td>
<td>170 West Tasman Drive</td>
<td>Capital Tower</td>
</tr>
<tr>
<td>San Jose, CA 95134-1706 USA</td>
<td>Haarlerbergpark</td>
<td>San Jose, CA 95134-1706 USA</td>
<td>168 Robinson Road</td>
</tr>
<tr>
<td>USA</td>
<td>Haarlerbergweg 13-19</td>
<td>USA</td>
<td>#22-01 to #29-01</td>
</tr>
<tr>
<td><a href="http://www.cisco.com">www.cisco.com</a></td>
<td>1101 CH Amsterdam</td>
<td><a href="http://www.cisco.com">www.cisco.com</a></td>
<td>Singapore 068912</td>
</tr>
<tr>
<td>Tel: 408 526-4000</td>
<td>The Netherlands</td>
<td>Tel: 408 526-7660</td>
<td><a href="http://www.cisco.com">www.cisco.com</a></td>
</tr>
<tr>
<td>800 553-NETS (6387)</td>
<td>www-europe.cisco.com</td>
<td>Fax: 408 527-0883</td>
<td>Tel: +65 317 7777</td>
</tr>
<tr>
<td>Fax: 408 526-4100</td>
<td></td>
<td>Fax: 31 0 20 357 1100</td>
<td>Fax: +65 317 7799</td>
</tr>
</tbody>
</table>

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