Comcast SIP Trunking:

Cisco Unified Communications Manager 11.0.1 with Cisco Unified Border Element (CUBE 11.5.0) on ISR 4321 [IOS-XE 3.17- 15.6(1)S] using SIP

May 04, 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 15.6(1)S can be used. The Cisco Unified Border Element 11.5.0 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.0.1 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) 11.0.1 and Cisco Unified Border Element (Cisco UBE) on ISR 4321/K9 [IOS-XE 3.17 - 15.6(1)S] for connectivity to Comcast SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 11.0.1) 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 7961 and 7965 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 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 2851 Fax Gateway
- IP phones 7961 (SIP) and 7965 (SCCP)
- Comcast(Adtran) ESG – Provided and managed by Comcast

Software Requirements
- Cisco Unified Communications Manager 11.0.1
- Cisco Unity Connection 11.0.1
- IOS-XE 3.17.0 – 15.6(1)S for ISR 4321/K9 Cisco Unified Border Element
- 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 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
- Fax (T.38) is not supported by Service Provider

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: CSCuvo4539.
- 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 Comcast CUBE1 LAN MS4 1/0/5
    ip address 10.80.11.5 255.255.255.0
    media-type rj45
    negotiation auto
    redundancy ri 3
    redundancy group 1 ip 10.80.11.15 exclusive
    !
interface GigabitEthernet0/0/1
    description Comcast CUBE1 WAN MS4 1/0/6
    ip address 10.64.4.146 255.255.0.0
    negotiation auto
    redundancy ri 4
    redundancy group 1 ip 10.64.4.145 exclusive
    !
Global Cisco UBE Settings
In order to enable Cisco UBE IP2IP gateway functionality, enter the following:

! voice service voip
    ip address trusted list
    ipv4 0.0.0.0 0.0.0.0
    address-hiding
    mode border-element license capacity 20
    allow-connections sip to sip
    redundancy-group 1
    no supplementary-service sip handle-replaces
    fax protocol pass-through g711ulaw
    sip
    relxx supported "rel100"
    session refresh
    asserted-id pai
    privacy pstn
    early-offer forced
    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>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>
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

```
codec g711ulaw
```

**Dial Peer**

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

```
!  
dial-peer voice 100 voip
description "Outbound from PBX to Adtran"
destination-pattern .T
  session protocol sipv2
  session target ipv4:10.64.4.140
  voice-class sip asserted-id pai
  voice-class sip profiles 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
  codec g711ulaw
  fax-relay ecm disable
  fax rate disable
  fax nsf 000000
  fax protocol pass-through g711ulaw
  no vad
!```
dial-peer voice 101 voip
description ""inbound from PBX to Adtran"
session protocol sipv2
incoming called-number .T
voice-class sip asserted-id pai
voice-class sip profiles 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
codec g711ulaw
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 200 voip
description "Inbound call from Adrtan to CUCM"
destination-pattern 856333....
session protocol sipv2
session target ipv4:10.80.11.3
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
codec g711ulaw
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 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

User Access Verification

Comcast_VUBE1>en
Comcast_VUBE1#sh running-config
!
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname Comcast_VUBE1
!
boot-start-marker
boot system 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
!
logging queue-limit 10000
logging buffered 10000000
logging rate-limit 10000
no logging console
!

aaa new-model
aaa session-id common
!

no ip domain lookup
ip domain name tekvizion.com
!

subscriber templating
multilink bundle-name authenticated
!

voice service voip
ip address trusted list
ipv4 0.0.0.0 0.0.0.0
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip handle-replaces
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 sip-profiles 1
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:856333\1\2>"

application
group-params 1

license udi pid ISR4321/K9 sn FDO19220MQ8
license boot level appxk9
license boot level uck9

spanning-tree extend system-id

redundancy
mode none
application redundancy
group 1
name Comcast
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 CUBE1 LAN MS4 1/0/5
  ip address 10.80.11.5 255.255.255.0
  media-type rj45
  negotiation auto
  redundancy rii 3
  redundancy group 1 ip 10.80.11.15 exclusive

interface GigabitEthernet0/0/1
  description Comcast CUBE1 WAN MS4 1/0/6
  ip address 10.64.4.146 255.255.0.0
  negotiation auto
  redundancy rii 4
  redundancy group 1 ip 10.64.4.145 exclusive

interface GigabitEthernet0/1/0
  description CUBE HA MS5 3/0/35
  ip address 10.89.20.7 255.255.255.0
  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 rtcp report interval 300
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.80.11.0 255.255.255.0 10.80.11.1
ip route 172.16.24.0 255.255.255.0 10.80.11.1
ip route 172.16.31.0 255.255.255.0 10.80.11.1
!
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 PBX to Adtran"
destination-pattern .T
session protocol sipv2
session target ipv4:10.64.4.140
voice-class sip asserted-id pai
voice-class sip profiles 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
codec g711ulaw
tax-relay ecm disable
tax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad!

dial-peer voice 101 voip
description ""inbound from PBX to Adtran"
session protocol sipv2
incoming called-number .T
voice-class sip asserted-id pai
voice-class sip profiles 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
codec g711ulaw
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad!

dial-peer voice 200 voip
description "Inbound call from Adtran to CUCM"
destination-pattern 856333....
session protocol sipv2
session target ipv4:10.80.11.3
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
codec g711ulaw
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 201 voip
  session protocol sipv2
  incoming called-number 856333....
  voice-class sip asserted-id pai
  voice-class sip profiles 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
codec g711ulaw
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
!
gateway
  timer receive-rtp 1200
!
sip-ua
  protocol mode ipv4
!
!
line con 0
  stopbits 1
line aux 0
stopbits 1
line vty 0
  password
  transport input telnet ssh
line vty 1
  exec-timeout 0 0
  password
  transport input telnet ssh
line vty 2 4
  password
  transport input telnet ssh

!  
!  
end
Standby Cisco UBE

Comcast_CUBE2#sh running-config
!
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname Comcast_CUBE2
!
boot-start-marker
boot system 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
!
logging queue-limit 10000
logging buffered 100000000
logging rate-limit 10000
no logging console
!
no aaa new-model
no ip domain lookup
ip domain name tekvizion.com
subscriber templating
!
multilink bundle-name authenticated
!
voice service voip
ip address trusted list
ipv4 0.0.0.0 0.0.0.0
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip handle-replaces
fax protocol pass-through g711ulaw
sip
rel1xx supported "rel100"
session refresh
asserted-id pai
privacy pstn
early-offer forced
midcall-signaling passthru
g729 annexe-b-all
!
!
voice class sip-profiles 1
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 Comcast
  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 CUBE2 LAN MS4 1/0/11
ip address 10.80.11.6 255.255.255.0
media-type rj45
negotiation auto
redundancy rii 5
redundancy group 2 ip 10.80.11.15 exclusive
!
interface GigabitEthernet0/0/1
description Comcast CUBE2 WAN MS4 1/0/12
ip address 10.64.4.147 255.255.255.128
negotiation auto
redundancy rii 6
redundancy group 2 ip 10.64.4.145 exclusive
interface GigabitEthernet0/1/0
description CUBE HA MS5 3/0/36
ip address 10.89.20.8 255.255.255.0
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 rtcp report interval 300
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.80.11.0 255.255.255.0 10.80.11.1
ip route 172.16.24.0 255.255.248.0 10.80.11.1
ip route 172.16.31.0 255.255.255.0 10.80.11.1
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comediasrc disable
mgcp behavior comediasdp-force disable
! mgcp profile default

! dial-peer voice 100 voip
description "Outbound from PBX to Adtran"
destination-pattern .T
session protocol sipv2
session target ipv4:10.64.4.140
voice-class sip asserted-id pai
voice-class sip profiles 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
codec g711ulaw
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 101 voip
description "inbound from PBX to Adtran"
session protocol sipv2
incoming called-number .T
voice-class sip asserted-id pai
voice-class sip profiles 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
codec g711ulaw
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 200 voip
description "Inbound call from Adrtan to CUCM"
destination-pattern 856333....
session protocol sipv2
session target ipv4:10.80.11.3
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
codec g711ulaw
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 201 voip
session protocol sipv2
incoming called-number 856333....
voice-class sip asserted-id pai
voice-class sip profiles 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
codec g711ulaw
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad

!
!
sip-ua
protocol mode ipv4
!
!
line con 0
stopbits 1
line aux 0
stopbits 1
line vty 0
password
transport input telnet ssh
line vty 1
exec-timeout 0 0
password
transport input telnet ssh
line vty 2 4
password
transport input telnet ssh
!
!
end
Configuring Cisco Unified Communications Manager

Cisco UCM Version

Figure 7: Cisco UCM Version

Cisco Call Manager Service Parameters

**Navigation Path:** System > Service Parameters

Select Server* = Clus21sub1--CUCM Voice/Video (Active)

Select Service* = Cisco CallManager (Active)

All other fields are set to default values

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

---

**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>Parameter</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 URL</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URL</td>
<td>x-cisco-serviceuri-opickup</td>
</tr>
<tr>
<td>Call Pickup Group URL</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>= 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 Redirects</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URL</td>
<td>x-cisco-serviceuri-cfwdall</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URL</td>
<td>x-cisco-serviceuri-abbrevdial</td>
</tr>
</tbody>
</table>

**Conference Join Enabled**
- [ ] RFC 2843 Hold
- [x] Semi-Attended Transfer
- [ ] Enable VAD
- [ ] Silence Message Waiting
- [ ] MLPP User Authorization

**Normalization Script**
- Normalization Script: <None>
- [ ] Enable Trace

**Parameter Name** | **Parameter Value**
--- | ---
1 | 1

**Incoming Requests FROM URI Settings**
- Caller ID DN
- Caller Name

Figure 11: SIP Profile (Cont.)
**Figure 12: 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 Messages</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
**Figure 14: SIP Trunk to Cisco UBE**
Intercompany Media Engine (IME)
E.164 Transformation Profile < None >

MLPP and Confidential Access Level Information
MLPP Domain < None >
Confidential Access Mode < None >
Confidential Access Level < None >

Call Routing Information
✓ Remote-Party-Id
✓ Asserted-Identity
Asserted-Type Default
SIP Privacy Default

- Inbound Calls

<table>
<thead>
<tr>
<th>Significant Digits</th>
<th>4</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Line ID Presentation</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&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>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>

Figure 15: SIP Trunk to Cisco UBE (Cont.)
### Connected Party Settings

- **Connected Party Transformation CSS**: < None >
- **Use Device Pool Connected Party Transformation CSS**

### Outbound Calls

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

### Caller Information

- **Caller ID DN**: 
- **Caller Name**: 
- **Maintain Original Caller ID DN and Caller Name in Identity Headers**

### SIP Information

#### Destination

<table>
<thead>
<tr>
<th>Destination Address</th>
<th>Destination Address IPv6</th>
<th>Destination Port</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>10.80.11.15</td>
<td></td>
<td>5060</td>
<td>up</td>
</tr>
</tbody>
</table>

- **MTP Preferred Originating Codec**: 711/sip
- **BLS Presence Group**: Standard Presence group
- **SIP Trunk Security Profile**: Comcast Non Secure SIP Trunk Profile
- **Redirecting Calling Search Space**: < None >
- **Out-Of-Dialog Refer Calling Search Space**: < None >
- **SUBSCRIBE Calling Search Space**: < None >
- **SIP Profile**: Standard SIP Profile, Comcast
- **DTMF Signaling Method**: No Preference

#### Normalization Script

- **Normalization Script**: < None >

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

#### Recording Information

- **None**
- **This trunk connects to a recording-enabled gateway**
- **This trunk connects to other clusters with recording-enabled gateways**

#### Geolocation Configuration

- **Geolocation**: < None >
- **Geolocation Filter**: < None >
- **Send Geolocation Information**
**Explanation**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Name</td>
<td>To_Comcast</td>
<td>Name for the trunk</td>
</tr>
<tr>
<td>Device Pool</td>
<td>G711_pool</td>
<td>Default Device Pool is used for this trunk</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRG_Default</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>10.80.11.15</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 Standard SIP Profile</td>
<td>SIP Profile configured earlier</td>
</tr>
</tbody>
</table>

**Dial Plan**

**Route Pattern Configuration**

**Navigation Path:** Call Routing > Route/Hunt > Route Pattern

Route patterns are configured as below:

- Cisco IP phones dial “9”+10 digits number to access PSTN via Cisco UBE
  - “9” is removed before sending to Cisco UBE
- For FAX calls, 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 calls to 0432 will be sent to Cisco Fax gateway
- Cisco IP phones dial 4XX and 9XX for emergency call and will send all digits from Cisco UBE to Comcast network
Figure 17: Route Patterns List

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, 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>To_Comcast for Route Pattern 9.@, X11and Trunk_to_fax_gw 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>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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