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

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Introduction

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

A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and COX Business network, Cisco Unified Border Element v12.0.0 (ISR4331) running IOS-XE 16.06.05 can be used. The Cisco Unified Border Element v12.0.0 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 12.5 connected to COX Business network.

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

- This application note describes how to configure a Cisco Unified Communications Manager (Cisco UCM) 12.5 and Cisco Unified Border Element (ISR4331) running IOS-XE 16.06.05 for connectivity to COX Business SIP Trunking service. The deployment model covered in this application note is (Cisco UCM 12.5) to PSTN (COX Business) via Cisco unified border element 12.0.0

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

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

Figure 1: Network Topology
System Components

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

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

Features

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

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

The following are not supported by COX Business
- G.729 voice codec
- T.38 fax
- SG3 fax transmission speed

Features Not Supported
- None
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.

![High Availability topology](image)

Figure 2 High Availability topology
CUBE 1:

interface GigabitEthernet0/0/0
  description WAN
  ip address 10.64.5.26 255.255.0.0
  negotiation auto
  redundancy rii 16
  redundancy group 2 ip 10.64.4.170 exclusive

interface GigabitEthernet0/0/1
  description LAN
  ip address 10.80.12.51 255.255.255.0
  negotiation auto
  redundancy rii 17
  redundancy group 2 ip 10.80.12.50 exclusive

interface GigabitEthernet0/0/2
  description HA_Interface
  ip address 10.70.50.101 255.255.255.0
  negotiation auto
interface GigabitEthernet0/0/0
   description WAN
   ip address 10.64.3.131 255.255.0.0
   negotiation auto
   redundancy rii 16
   redundancy group 2 ip 10.64.4.170 exclusive

interface GigabitEthernet0/0/1
   description LAN
   ip address 10.80.12.52 255.255.255.0
   negotiation auto
   redundancy rii 17
   redundancy group 2 ip 10.80.12.50 exclusive

interface GigabitEthernet0/0/2
   description HA_interface
   ip address 10.70.50.102 255.255.255.0
   negotiation auto
Global Cisco UBE Settings

In order to enable Cisco UBE IP2IP gateway functionality, enter the following:

```
voice service voip
no ip address trusted authenticate
address-hiding
mode border-element
allow-connections sip to sip
redundancy-group 2
no supplementary-service sip handle-replaces
fax protocol pass-through g711ulaw
sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  early-offer forced
  midcall-signaling passthru
```

**Explanation**

<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>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 voice codec is used for this testing. Codec preferences used to change according to the test plan description.

```
voice class codec 1
  codec preference 1 g711ulaw
```
Dial Peer

Active Cisco UBE

dial-peer voice 1 voip
description WAN-Facing Outgoing
huntstop
destination-pattern .T
translate-outgoing called 110
session protocol sipv2
session target ipv4:10.64.4.162:5060
session transport udp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2 voip
description LAN facing Outgoing
huntstop
session protocol sipv2
incoming called-number .T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 3 voip

description Incoming CUCM Facing

huntstop
destination-pattern 402........
session protocol sipv2
session target ipv4:10.80.12.2:5060
session transport udp

voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-n-te
no vad

!

dial-peer voice 4 voip

description Incoming WAN Facing

huntstop

session protocol sipv2
incoming called-number 402........
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n-te
no vad
Configuration Example
The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously

**Active Cisco UBE**

Current configuration : 4583 bytes

```
! version 16.6
service timestamps debug datetime msec
service timestamps log datetime msec

platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname COX_CUBE1
!
boot-start-marker
boot system flash:isr4300-universalk9.16.06.05.SPA.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
!
  address-family ipv4
  exit-address-family
  !
  address-family ipv6
  exit-address-family
  !
logging buffered 99999
!
no aaa new-model
subscriber templating
multilink bundle-name authenticated
```

crypto pki trustpoint TP-self-signed-3616943619
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-3616943619
  revocation-check none
  rsakeypair TP-self-signed-3616943619
!
crypto pki certificate chain TP-self-signed-3616943619
!
voice service voip
  no ip address trusted authenticate
  address-hiding
  mode border-element
  allow-connections sip to sip
  redundancy-group 2
  no supplementary-service sip handle-replaces
  fax protocol pass-through g711ulaw
  sip
    bind control source-interface GigabitEthernet0/0/0
    bind media source-interface GigabitEthernet0/0/0
      early-offer forced
      midcall-signaling passthru
!
voice class codec 1
  codec preference 1 g711ulaw
!
voice-card 0/4
  no watchdog
!
license udi pid ISR4331/K9 sn FDO24881FEY
diagnostic bootup level minimal
spanning-tree extend system-id
!
! redundancy
  mode none
  application redundancy
  group 2
    name COX
    priority 150 failover threshold 75
    timers delay 30 reload 60
    control GigabitEthernet0/0/2 protocol 1
    data GigabitEthernet0/0/2
    track 1 shutdown
    track 2 shutdown
  
  track 1 interface GigabitEthernet0/0/1 line-protocol
  track 2 interface GigabitEthernet0/0/0 line-protocol

  ! translation-rule 110
    Rule 1 14025059697 4025059697
    Rule 2 14029165692 4029165692
    Rule 3 14025056016 4025056016

  ! interface GigabitEthernet0/0/0
    description WAN
    ip address 10.64.5.26 255.255.0.0
    negotiation auto
    redundancy rii 16
    redundancy group 2 ip 10.64.4.170 exclusive

  ! interface GigabitEthernet0/0/1
    description LAN
    ip address 10.80.12.51 255.255.255.0
    negotiation auto
    redundancy rii 17
    redundancy group 2 ip 10.80.12.50 exclusive
interface GigabitEthernet0/0/2
    description HA_Interface
    ip address 10.70.50.101 255.255.255.0
    negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
    vrf forwarding Mgmt-intf
    no ip address
    negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.80.12.0 255.255.255.0 10.80.12.1
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedия-role none
mgcp behavior comedía-check-media-src disable
mgcp behavior comedía-sdp-force disable
!
mgcp profile default
!
dial-peer voice 1 voip
    description WAN-Facing Outgoing
    huntstop
    destination-pattern .T
    translate-outgoing called 110
session protocol sipv2
session target ipv4:10.64.4.162:5060
session transport udp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nre
no vad
!
dial-peer voice 2 voip
description LAN facing Outgoing
huntstop
session protocol sipv2
incoming called-number .T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nre
no vad
!
dial-peer voice 3 voip
description Incoming CUCM Facing
huntstop
destination-pattern 402.......
dial-peer voice 4 voip
description Incoming WAN Facing
huntstop
session protocol sipv2
incoming called-number 402......
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nce
no vad
!
alias exec tr show run | begin voice translation-rule
alias exec dp show run | begin dial-peer voice
!
line con 0
  exec-timeout 0 0
  password tekV1z10n
  login
  transport input none
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 0 0
  password tekV1z10n
  login
Current configuration : 4111 bytes

version 16.6
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname COX_CUBE2
!
boot-start-marker
boot-end-marker
!
vrf definition Mgmt-intf
!
  address-family ipv4
  exit-address-family
!
  address-family ipv6
  exit-address-family
!
no aaa new-model
subscriber templating
multilink bundle-name authenticated
!
voice service voip
  address-hiding
  mode border-element
  allow-connections sip to sip
  redundancy-group 2
  no supplementary-service sip handle-replaces
  fax protocol pass-through g711ulaw
  sip
    bind control source-interface GigabitEthernet0/0/0
    bind media source-interface GigabitEthernet0/0/0
    early-offer forced
    midcall-signaling passthru
!
voice class codec 1
  codec preference 1 g711ulaw
!
voice-card 0/1
  no watchdog
!
voice-card 0/4
  no watchdog
!
license udi pid ISR4331/K9 sn FDO24881GMV
diagnostic bootup level minimal
spanning-tree extend system-id
!
redundancy
  mode none
application redundancy
  group 2
    name COX
    priority 100 failover threshold 75
timers delay 30 reload 60
  control GigabitEthernet0/0/2 protocol 1
data GigabitEthernet0/0/2
  track 1 shutdown
  track 2 shutdown
!
  track 1 interface GigabitEthernet0/0/1 line-protocol
  track 2 interface GigabitEthernet0/0/0 line-protocol
!
translation-rule 110
  Rule 1 14025059697 4025059697
  Rule 2 14029165692 4029165692
  Rule 3 14025056016 4025056016
!
interface GigabitEthernet0/0/0
  description WAN
  ip address 10.64.3.131 255.255.0.0
  negotiation auto
  redundancy rii 16
  redundancy group 2 ip 10.64.4.170 exclusive
!
interface GigabitEthernet0/0/1
  description LAN
  ip address 10.80.12.52 255.255.255.0
  negotiation auto
  redundancy rii 17
  redundancy group 2 ip 10.80.12.50 exclusive
!
interface GigabitEthernet0/0/2
  description HA_interface
  ip address 10.70.50.102 255.255.255.0
  negotiation auto
!
interface Service-Engine0/1/0
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  negotiation auto
!
ip forward-protocol nd
no ip http server
no ip http secure
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.80.12.0 255.255.255.0 10.80.12.1
!
control-plane
!
voice-port 0/1/0
voice-port 0/1/1
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 1 voip
  description WAN-Facing Outgoing
  huntstop
  destination-pattern .T
  translate-outgoing called 110
  session protocol sipv2
  session target ipv4:10.64.4.162:5060
  session transport udp
  voice-class sip bind control source-interface GigabitEthernet0/0/0
  voice-class sip bind media source-interface GigabitEthernet0/0/0
  dtmf-relay rtp-nte
  no vad
!

dial-peer voice 2 voip
  description LAN facing Outgoing
  huntstop
  session protocol sipv2
  incoming called-number .T
  voice-class sip bind control source-interface GigabitEthernet0/0/1
  voice-class sip bind media source-interface GigabitEthernet0/0/1
  dtmf-relay rtp-n-te
  no vad
!
dial-peer voice 3 voip
  description Incoming CUCM Facing
  huntstop
  destination-pattern 402.......  
  session protocol sipv2
  session target ipv4:10.80.12.2:5060
  session transport udp
  voice-class sip bind control source-interface GigabitEthernet0/0/1
  voice-class sip bind media source-interface GigabitEthernet0/0/1
  dtmf-relay rtp-n-te
  no vad
!
dial-peer voice 4 voip
  description Incoming WAN Facing
  huntstop
  session protocol sipv2
  incoming called-number 402.......  
  voice-class sip bind control source-interface GigabitEthernet0/0/0
  voice-class sip bind media source-interface GigabitEthernet0/0/0
  dtmf-relay rtp-n-te
  no vad
!
line con 0
  exec-timeout 0 0
  password tekV1z10n
  login
  transport input none
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 0 0
  password tekV1z10n
  login
!
end
Configuring Cisco Unified Communications Manager

Cisco UCM Version

![Cisco Unified CM Administration Interface](image)

**Figure 3: Cisco UCM Version**

Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

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

![Select Server and Service Interface](image)

![Call Throttling Table](image)

**Figure 4: Service Parameters**
Offnet Calls via COX Business SIP Trunk

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

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

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

![Figure 5: SIP Trunk Security Profile](image)
<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></td>
</tr>
<tr>
<td>Outgoing Transport Type</td>
<td>UDP</td>
<td>SIP trunks to Cisco UBE should use UDP as a transport protocol for SIP. This is configured using SIP Trunk Security profile, which is later assigned to the SIP trunk itself.</td>
</tr>
</tbody>
</table>

**SIP Profile Configuration**

SIP Profile will be later associated with the SIP trunk

Navigation: Device → Device Settings → SIP Profile

1. Name*: COX SIP Profile, for example

![SIP Profile Information](image)

Figure 6: SIP Profile
### SDP Information

- **SDP Session-level Bandwidth Modifier for Early Offer and Re-invites**: TIAS and AS
- **SDP Transparency Profile**: Pass all unknown SDP attributes
- **Accept Audio Coded Preferences in Received Offer**: Default
- **Require SDP Inactive Exchange for Mid-Call Media Change**: Default
- **Allow RR/RS bandwidth modifier (RFC 3556)**: Default

### Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)</td>
<td>3600</td>
</tr>
<tr>
<td>Timer T1 (msec)</td>
<td>500</td>
</tr>
<tr>
<td>Timer T2 (msec)</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE</td>
<td>10</td>
</tr>
<tr>
<td>Media Port Ranges</td>
<td></td>
</tr>
<tr>
<td>- Common Port Range for Audio and Video</td>
<td></td>
</tr>
<tr>
<td>- Separate Port Ranges for Audio and Video</td>
<td></td>
</tr>
<tr>
<td>Start Media Port</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port</td>
<td>32766</td>
</tr>
<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 Port of Video Calls</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

*Figure 6: SIP Profile (Cont.)*
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<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 Other URI *</td>
<td>x-cisco-serviceuri-opickup</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-abbrdial</td>
</tr>
<tr>
<td>Conference Join Enabled</td>
<td></td>
</tr>
<tr>
<td>RFC 2541 Hold</td>
<td></td>
</tr>
<tr>
<td>Semi Attended Transfer</td>
<td></td>
</tr>
<tr>
<td>Enable VAD</td>
<td></td>
</tr>
<tr>
<td>Stutter Message Waiting</td>
<td></td>
</tr>
<tr>
<td>MLPP User Authorization</td>
<td></td>
</tr>
</tbody>
</table>

*Figure 6: SIP Profile (Cont.)*
### Normalization Script

**Normalization Script**

- **Normalization Script**: <None>▼
- **Enable Trace**

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

### External Presentation Information

- **Anonymous External Presentation**
- **External Presentation Number**
- **External Presentation Name**

### Trunk Specific Configuration

- **Reroute Incoming Request to new Trunk based on**
- **Resource Priority Namespace List**
- **SIP Relay Options**
- **Video Call Traffic Class**
- **Calling Line Identification Presentation**
- **Session Refresh Method**
- **Early Offer support for voice and video calls**
- **Enable ANAT**
- **Deliver Conference Bridge Identifier**
- **Enable External Presentation Name and Number**
- **Reject Anonymous Incoming Calls**

---

*Figure 6: SIP Profile (Cont.)*
Figure 6: SIP Profile (Cont.)

### Explanation

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name</td>
<td>COX SIP Profile</td>
<td>Name given to refer the SIP profile created for COX Business</td>
</tr>
<tr>
<td>Description</td>
<td>Default SIP Profile</td>
<td>Customized description of the SIP profile used for COX Business</td>
</tr>
<tr>
<td>Default MTP Telephony Event Payload Type</td>
<td>101</td>
<td>RFC2833 DTMF payload type</td>
</tr>
</tbody>
</table>
Create SIP trunks to Cisco UBE

Navigation: Device → Trunk

Figure 7: SIP Trunks List

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

### Intercompany Media Engine (IME)

<table>
<thead>
<tr>
<th>Option</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>E.164 Transformation Profile</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### MLPP and Confidential Access Level Information

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<thead>
<tr>
<th>Option</th>
<th>Setting</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

<table>
<thead>
<tr>
<th>Option</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Remote-Party-Id</td>
<td>Default</td>
</tr>
<tr>
<td>Asserted-Identity</td>
<td>Default</td>
</tr>
<tr>
<td>Asserted-Type</td>
<td>Default</td>
</tr>
<tr>
<td>STIP Privacy</td>
<td>Default</td>
</tr>
<tr>
<td>Trust Received Identity</td>
<td>Trust All (Default)</td>
</tr>
</tbody>
</table>

### Inbound Calls

<table>
<thead>
<tr>
<th>Option</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Significant Digits</td>
<td>4</td>
</tr>
<tr>
<td>Connected Line ID Presentation</td>
<td>Allowed</td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td>Allowed</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Prefix DN</td>
<td></td>
</tr>
<tr>
<td>Redirecting Diversion Header Delivery - Inbound</td>
<td></td>
</tr>
</tbody>
</table>

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

SIP Information

- Destination
  - Destination Address is an SRV
  - Destination Address: 10.80.12.50
  - Destination Address IPv6: None
  - Destination Port: 5060

- SIP Trunk Security Profile
  - Non Secure SIP Trunk Profile

- DTMF Signaling Method
  - RFC 2833

- SIP Profile
  - COX SIP Profile

- Out-Of-Dialog Refer Calling Search Space
  - < None >

- Subscribe Calling Search Space
  - < None >

- Redirect Calling Search Space
  - < None >

- BLF Presence Group
  - Standard Presence Group

- MTP Preferred Originating Codec
  - 711ulaw

- Normalization Script
  - < None >

- Geolocation Configuration
  - Geolocation: < None >
  - Geolocation Filter: < None >

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

Figure 8: 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>COX</td>
<td>Name for the trunk</td>
</tr>
<tr>
<td>Device Pool</td>
<td>G711ulaw</td>
<td>Device Pool used for SIP trunk to COX</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL</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>10.80.12.50</td>
<td>IP address of the Cisco UBE Virtual LAN</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
<td>Non Secure SIP Trunk Profile</td>
<td>SIP Trunk Security Profile configured earlier</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>COX SIP Profile</td>
<td>SIP Profile configured earlier</td>
</tr>
</tbody>
</table>

### Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

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

![Figure 9: Route Patterns List](image-url)
### Figure 10: Route Pattern for Voice

**Pattern Definition**

| Route Pattern | 7.0  |
| Description   |      |
| Numbering Plan | NANP |
| Route Filter  | < None > |
| MLPP Precedence | Default |
| Apply Call Blocking Percentage |   |
| Resource Priority Namespace Network Domain | < None > |
| Route Class   | Default |
| Gateway/Route List | COX |
| Route Option  | 
| Call Classification | ONNet |
| External Call Control Profile | < None > |
| Allow Device Override | ✓ Provide Outside Dial Tone | ☐ Allow Overlap Sending | ☐ Urgent Priority |
| Require Forced Authorization Code |   |
| Authorization Level | 0 |
| Require Client Matter Code |   |

**Calling Party Transformations**

- Use Calling Party’s External Phone Number Mask
- Calling Party Transform Mask: 
- Prefix Digits (Outgoing Calls): 
- Calling Line ID Presentation: Default
- Calling Name Presentation: Default
- Calling Party Number Type: Cisco CallManager
- Calling Party Numbering Plan: Cisco CallManager

**Connected Party Transformations**

- Connected Line ID Presentation: Default
- Connected Name Presentation: Default

**Called Party Transformations**

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

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

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>7.@ for Voice &amp; International Calls</td>
<td>Specify appropriate Route Pattern</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>COX for Route Pattern 7.@</td>
<td>SIP Trunk name configured earlier</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>NANP for Route Pattern 7.@</td>
<td>North American Numbering Plan</td>
</tr>
<tr>
<td>Call Classification</td>
<td>OffNet for Route Pattern 7.@</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 7.@</td>
<td>Specifies how to modify digit before they are sent to COX Business network</td>
</tr>
</tbody>
</table>

### Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definition</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>Asia Pacific Headquarters</th>
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<td>Haarlerbergpark</td>
<td>170 West Tasman Drive</td>
<td>Capital Tower</td>
</tr>
<tr>
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<td>Haarlerbergweg 13-19</td>
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<td>168 Robinson Road</td>
</tr>
<tr>
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<td>1101 CH Amsterdam</td>
<td>USA</td>
<td>#22-01 to #29-01</td>
</tr>
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<td>www-europe.cisco.com</td>
<td>Tel: 408 526-7660</td>
<td>Tel: +65 317 7777</td>
</tr>
<tr>
<td>800 553-NETS (6387)</td>
<td></td>
<td>Fax: 408 527-0883</td>
<td>Fax: +65 317 7799</td>
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<td></td>
</tr>
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