Connecting Cisco Unified Communication Manager [v11.5.1] to CableOne SIP Trunks via Cisco Unified Border Element v12.0.0 [IOS-XE 16.07.01]

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# Table of Contents

Introduction .................................................................................................................. 4

Network Topology ......................................................................................................... 5

System Components ....................................................................................................... 6
  Hardware Requirements ............................................................................................... 6
  Software Requirements .............................................................................................. 6

Features .......................................................................................................................... 6
  Features Supported ...................................................................................................... 6
  Features Not Supported .............................................................................................. 6

Caveats ............................................................................................................................ 7

Configuration .................................................................................................................. 8
  Configuring Cisco Unified Border Element ................................................................. 8
    Network Interface ....................................................................................................... 8
    Global Cisco UBE settings ......................................................................................... 11
    Codecs ....................................................................................................................... 11
    Dial peer .................................................................................................................... 12
    Configuration example ................................................................................................. 14

Configuring Cisco UCM 11.5 Cluster ............................................................................. 28
  Cisco UCM Version ...................................................................................................... 28
  Cisco Call Manager Service Parameters ..................................................................... 29
  SIP Trunk Security Profile ........................................................................................... 30
  SIP Profile .................................................................................................................... 32
  Trunk configuration ...................................................................................................... 36
  Routing configuration .................................................................................................. 42

Acronyms ......................................................................................................................... 47

Important Information .................................................................................................... 47
**Table of Figures**

<table>
<thead>
<tr>
<th>Figure</th>
<th>Description</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>Figure 1</td>
<td>Network Topology</td>
<td>5</td>
</tr>
<tr>
<td>Figure 2</td>
<td>High Availability topology</td>
<td>8</td>
</tr>
<tr>
<td>Figure 3</td>
<td>Cisco UCM Version</td>
<td>28</td>
</tr>
<tr>
<td>Figure 4</td>
<td>Service Parameters</td>
<td>29</td>
</tr>
<tr>
<td>Figure 5</td>
<td>Service Parameters (Cont.)</td>
<td>30</td>
</tr>
<tr>
<td>Figure 6</td>
<td>SIP Trunk Security Profile</td>
<td>31</td>
</tr>
<tr>
<td>Figure 7</td>
<td>SIP Profile</td>
<td>32</td>
</tr>
<tr>
<td>Figure 8</td>
<td>SIP Profile (Cont.)</td>
<td>33</td>
</tr>
<tr>
<td>Figure 9</td>
<td>SIP Profile (Cont.)</td>
<td>34</td>
</tr>
<tr>
<td>Figure 10</td>
<td>SIP Profile (Cont.)</td>
<td>35</td>
</tr>
<tr>
<td>Figure 11</td>
<td>Add New Trunk to Cisco UBE</td>
<td>36</td>
</tr>
<tr>
<td>Figure 12</td>
<td>Add SIP Trunk Type</td>
<td>37</td>
</tr>
<tr>
<td>Figure 13</td>
<td>SIP Trunk to Cisco UBE</td>
<td>38</td>
</tr>
<tr>
<td>Figure 14</td>
<td>SIP Trunk to Cisco UBE (Cont.)</td>
<td>39</td>
</tr>
<tr>
<td>Figure 15</td>
<td>SIP Trunk to Cisco UBE (Cont.)</td>
<td>40</td>
</tr>
<tr>
<td>Figure 16</td>
<td>SIP Trunk to Cisco UBE (Cont.)</td>
<td>41</td>
</tr>
<tr>
<td>Figure 17</td>
<td>Add New Route Pattern for Cisco UBE</td>
<td>42</td>
</tr>
<tr>
<td>Figure 18</td>
<td>Route Pattern Configuration for Cisco UBE-PSTN Access</td>
<td>43</td>
</tr>
<tr>
<td>Figure 19</td>
<td>Route Pattern Configuration for Cisco UBE-PSTN Access (Cont.)</td>
<td>44</td>
</tr>
<tr>
<td>Figure 20</td>
<td>Route Pattern Configuration for Cisco UBE-PSTN Access - Anonymous Call</td>
<td>45</td>
</tr>
<tr>
<td>Figure 21</td>
<td>Route Pattern Configuration for Cisco UBE-PSTN Access - Anonymous Call (Cont.)</td>
<td>46</td>
</tr>
</tbody>
</table>
Introduction

Service Providers today, such as CableOne, 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.

CableOne 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 (Cisco UCM) and CableOne network, Cisco Unified Border Element (Cisco UBE) ISR 4331/K9 running IOS 16.07.01 can be used. The Cisco Unified Border Element 16.07.01 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.1 connected to CableOne IP network.

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

- This application note describes how to configure Cisco UCM 11.5.1, and Cisco UBE on ISR 4331/K9 [IOS – 16.07.01] for connectivity to CableOne SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM) to PSTN (CableOne) via Cisco UBE v12.0 [IOS-XE] 16.07.01.

- Testing was performed in accordance to CableOne 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 High Availability.

- The Cisco UBE configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between CableOne 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 CableOne SIP Trunking network.
The network topology includes the Cisco UCM Cluster, Unity Voicemail system, Cisco Fax gateway and 2 Cisco Endpoints. Cisco UCM has a trunk configured to Cisco UBE’s Virtual IP Address. CableOne was used as the service provider with SIP trunk to the Cisco UBE using the WAN Virtual IP Address.

- 2 Cisco Unified Border Elements are used here for High Availability.
- SIP Trunk transport type used between Cisco UBE and Cisco UCM is TCP and to CableOne is UDP.

### Cisco UCM and Cisco UBE Settings:

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transport from Cisco UBE to Cisco UCM</td>
<td>TCP with RTP</td>
</tr>
<tr>
<td>Transport from Cisco UBE to CableOne</td>
<td>UDP with RTP</td>
</tr>
<tr>
<td>Voice Mail Support</td>
<td>YES</td>
</tr>
<tr>
<td>Session Refresh</td>
<td>YES</td>
</tr>
</tbody>
</table>
System Components

Hardware Requirements
- Cisco UBE on Cisco ISR 4331/K9 router
- CUCM cluster on UCS C240, 1 Publisher node and 2 Subscriber nodes
- Cisco 2851 with FXS ports and Analog Fax machine
- Generic Cisco IP-Phones
- Adtran NetVanta 3140

Software Requirements
- CUBE-Version: 12.0.0 running IOS-XE 16.07.01
- CUCM UCOS 11.5.1.12900-21 for 1 Publisher and 2 Subscriber
- Cisco IOS v15.1(4)M5 for the fax gateway
- ADTRAN, Inc. OS version R12.3.3.E

Features

Features Supported
- Incoming and outgoing calls using G711ulaw codec
- International Calls and digit manipulations
- Call Conference
- CPE Voice Mail support
- Call hold & Resume with and without MoH
- Unattended and Attended Call transfer
- Call forward (all, busy and no answer)
- Inband DTMF
- Fax (G711 Pass-through)
- IP-PBX Calling number privacy
- High Availability

Features Not Supported
- Cisco UCM does not support Blind Call transfer
- In HA Redundancy mode, the Primary Cisco UBE will not take over the Primary/Active role after a reboot/network outage
- CableOne does not support T.38 Fax
- G.729 Codec for voice calls is not supported by CableOne
Caveats

- Caller ID is not updated on attended and unattended transfer scenarios.
- Only one IP PBX used for the testing.
- The Cisco UBE HA tested is layer 2 box to box Cisco UBE redundancy.
- Transcoder has been enabled on Cisco UBE to support Inband DTMF.
Configuration

Configuring Cisco Unified Border Element

Network Interface
The IP address used are for illustration only, the actual IP address can vary. The Active/Standby pair share the same virtual IP address and continually exchange status messages.

Figure 2 High Availability topology
Cisco UBE 1:

interface GigabitEthernet0/0/0
    description WAN
    ip address 10.64.4.131 255.255.0.0
    negotiation auto
    redundancy rii 1
    redundancy group 1 ip 10.64.5.12 exclusive
!
interface GigabitEthernet0/0/1
    description LAN
    ip address 10.80.18.11 255.255.255.0
    negotiation auto
    redundancy rii 2
    redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/0/2
    description CUBE_HA
    ip address 10.89.20.12 255.255.255.0
    negotiation auto
!
interface Service-Engine0/4/0

Cisco UBE 2:

interface GigabitEthernet0/0/0
    description WAN
    ip address 10.64.4.170 255.255.0.0
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.64.5.12 exclusive
!
interface GigabitEthernet0/0/1
description LAN
ip address 10.80.18.12 255.255.255.0
negotiation auto
redundancy rii 2
redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/0/2
description CUBE_HA
ip address 10.89.20.11 255.255.255.0
negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
Global Cisco UBE settings

In order to enable Cisco UBE IP2IP SBC functionality, following command has to be entered:

```
voice service voip
  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
  asserted-id pai
  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>redundancy-group 1</td>
<td>Enable High Availability for the VoIP service</td>
</tr>
<tr>
<td>fax protocol</td>
<td>Specifies the fax protocol</td>
</tr>
<tr>
<td>asserted-id</td>
<td>Specifies the privacy header in the outgoing SIP requests and response messages</td>
</tr>
<tr>
<td>midcall-signaling passthru</td>
<td>Pass-through media change method optimizes or consumes mid-call, media-related signaling within the call</td>
</tr>
</tbody>
</table>

Codecs

G711ulaw codec is used for this testing.

```
voice class codec 1
  codec preference 1 g711ulaw!
```
Dial peer

Outbound Dial-peer to CableOne:

description Incoming Call from CUCM International
session protocol sipv2
session transport tcp
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 2 voip
description Outgoing Call to CableOne
destination-pattern .T
session protocol sipv2
session target ipv4:10.64.5.7:5060
voice-class codec 1
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
no dtmf-interworking
no vad

Inbound Dial-peer from CableOne:

dial-peer voice 3 voip
description Incoming call from CableOne
session protocol sipv2
incoming called-number 480.....
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
no vad
!
dial-peer voice 4 voip
  description Outgoing to CUCM
destination-pattern 480......
  session protocol sipv2
  session target ipv4:10.80.18.2:5060
  session transport tcp
  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-nce
no vad
Configuration example
The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously.

**Active Cisco UBE:**

```plaintext
version 16.7
service config
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname CUBE10Cableone
!
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
!
crypto pki trustpoint TP-self-signed-2930804041
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-2930804041
  revocation-check none
  rsakeypair TP-self-signed-2930804041
!
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
!

voice service voip
  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
    asserted-id pai
    midcall-signaling passthru
! voice class codec 1  
   codec preference 1 g711ulaw
!
voice class sip-profiles 1  
   request ANY sip-header Diversion modify "sip:(.*)@" "sip:480562\1@"
!
voice-card 0/4  
   dsp services dspfarm
   no watchdog
!
license udi pid ISR4331/K9 sn FDO21381F17  
diagnostic bootup level minimal  
spanning-tree extend system-id
!
redundancy  
   mode none  
   application redundancy
      group 1  
         name voice-HA
         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 GigabitEthernet0/0/0 line-protocol  
track 2 interface GigabitEthernet0/0/1 line-protocol
interface GigabitEthernet0/0/0
description WAN
ip address 10.64.4.131 255.255.0.0
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.64.5.12 exclusive

interface GigabitEthernet0/0/1
description LAN
ip address 10.80.18.11 255.255.255.0
negotiation auto
redundancy rii 2
redundancy group 1 ip 10.80.18.10 exclusive

interface GigabitEthernet0/0/2
description CUBE_HA
ip address 10.89.20.12 255.255.255.0
negotiation auto

interface Service-Engine0/4/0

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

ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet0/0/0
ip tftp source-interface GigabitEthernet0/0/0
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.80.18.0 255.255.255.0 10.80.18.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
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
!
dspfarm profile 50 transcode
  codec g711ulaw
  maximum sessions 12
  associate application CUBE
!
dial-peer voice 1 voip
  description Incoming Call from CUCM International
  session protocol sipv2
  session transport tcp
  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 2 voip
description Outgoing Call to CableOne
destination-pattern .T
session protocol sipv2
session target ipv4:10.64.5.7:5060
voice-class codec 1
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
no dtmf-interworking
no vad
!
dial-peer voice 3 voip
description Incoming call from CableOne
session protocol sipv2
incoming called-number 480......
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
no vad
!
dial-peer voice 4 voip
description Outgoing to CUCM
destination-pattern 480......
session protocol sipv2
session target ipv4:10.80.18.2:5060
session transport tcp
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-nce
no vad
!
!
sip-ua
    connection-reuse
!
!
line con 0
    exec-timeout 0 0
    password xxxxxxxxxx
    login
    transport input none
    stopbits 1
line aux 0
    stopbits 1
line vty 0 4
    exec-timeout 0 0
    password xxxxxxxxxx
    login
    transport input telnet
!
end
Standby Cisco UBE:

version 16.7
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname CUBE8Cableone
!
boot-start-marker
boot-end-marker
!
!
vrft 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
!
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 license capacity 20
  allow-connections sip to sip
  redundancy-group 1
  no supplementary-service sip handle-replaces
  fax protocol pass-through g711ulaw
  sip
    midcall-signaling passthru
  
voice class codec 1
  codec preference 1 g711ulaw
!

voice class sip-profiles 1
  request ANY sip-header Diversion modify "sip:(.*)@" "sip:480562\1@"
!
voice-card 0/4
  dsp services dspfarm
  no watchdog
!
license udi pid ISR4331/K9 sn FDO21381FEY
no license smart enable
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
redundancy
mode none
application redundancy
group 1
  name voice-HA
  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 GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
  description WAN
  ip address 10.64.4.170 255.255.0.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.64.5.12 exclusive
!
interface GigabitEthernet0/0/1
  description LAN
  ip address 10.80.18.12 255.255.255.0
negotiation auto
redundancy rii 2
redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/0/2
description CUBE_HA
ip address 10.89.20.11 255.255.255.0
negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet0/0/0
ip tftp source-interface GigabitEthernet0/0/0
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.80.18.0 255.255.255.0 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
!
dspfarm profile 50 transcode
   codec g711ulaw
   maximum sessions 12
   associate application CUBE
!
dial-peer voice 1 voip
   description Incoming Call from CUCM International
   session protocol sipv2
   session transport tcp
   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 2 voip
   description Outgoing Call to CableOne
   destination-pattern .T
   session protocol sipv2
   session target ipv4:10.64.5.7:5060
   voice-class codec 1
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
no dtmf-interworking
no vad
!
dial-peer voice 3 voip
  description Incoming call from CableOne
  session protocol sipv2
  incoming called-number 480......
  voice-class codec 1
  voice-class sip options-keepalive
  voice-class sip bind control source-interface GigabitEthernet0/0/0
  voice-class sip bind media source-interface GigabitEthernet0/0/0
  no vad
!
dial-peer voice 4 voip
  description Outgoing to CUCM
  destination-pattern 480......
  session protocol sipv2
  session target ipv4:10.80.18.2:5060
  session transport tcp
  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
!
!
sip-ua
  connection-reuse

!
!
line con 0
  exec-timeout 0 0
  password xxxxxxxxxx
  login
  transport input none
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 0 0
  password xxxxxxxxxx
  login
!
wsma agent exec
!
wsma agent config
!
wsma agent filesys
!
wsma agent notify
!
!
end

CUBE8Cableone#
Configuring Cisco UCM 11.5 Cluster

Cisco UCM Version

![Cisco UCM Version](image)

Figure 3: Cisco UCM Version
Cisco Call Manager Service Parameters

**Navigation:** System → Service Parameters

- Select Server* = Clus28pub--CUCM Voice/Video (Active)
- Select Service* = Cisco CallManager (Active)
- Duplex Streaming Enabled* = True
- All other fields are set to default values

![Cisco Unified CM Administration](Image)

**Figure 4: Service Parameters**
### SIP Trunk Security Profile

**Navigation:** System → Security → SIP Trunk Security Profile

- **Name** = **Non Secure SIP Trunk Profile** is used as an example
- **Description** = **Non Secure SIP Trunk Profile authenticated by null String** is used as an example
- **Device Security Mode** = **Non Secure**
- **Incoming Transport Type** = **TCP + UDP**
- **Outgoing Transport Type** = **TCP**
Figure 6: SIP Trunk Security Profile
SIP Profile

**Navigation:** Device → Device Settings → SIP Profile

- Name* = **CableOne SIP Profile** is used as an example
- **SIP Rel1XX Options:** Send PRACK if 1XX contains SDP is selected for this example

![SIP Profile Configuration](image)

- **Status:** Status: Ready
- All SIP devices using this profile must be restarted before any changes will take affect.

![SIP Profile Information](image)

- **Name:** CableOne SIP Profile
- **Default MTU Telephony Event Payload Type:** 101
- **Early Offer for G.332 Calls:** Disabled
- **User-Agent and Server header information:** Send Unified CM Version Information as User-Agent
- **Version in User Agent and Server Header:** Major And Minor
- **Dial String Interpretation:** Phone number consists of characters 0-9, +, *, and
- **Confidential Access Level Headers:** Disabled

![SDP Information](image)

- **SDP Session-level Bandwidth Modifier for Early Offer and Re-invites:** TIAS and AS
- **SDP Transparency Profile:** Pass all unknown SDP attributes
- **Accept Audio Codec Preferences in Received Offer:** Default
- **Require SDP Inactive Exchange for Mid-Call Media Change:**
- **Allow RR/RS bandwidth modifier (RFC 3558):**

**Figure 7: SIP Profile**
<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>3500</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>15384</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>
<tr>
<td>DSCP for TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Port 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-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></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>
</tbody>
</table>

Figure 8: SIP Profile (Cont.)
Figure 9: SIP Profile (Cont.)
Figure 10: SIP Profile (Cont.)

**SIP OPTIONS Ping**

<table>
<thead>
<tr>
<th>Option</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable OPTIONS Ping to monitor destination status for Trunks with Service Type &quot;None (Default)&quot;</td>
<td></td>
</tr>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)</td>
<td>60</td>
</tr>
<tr>
<td>Ping Interval for Out-of-service Trunks (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Ping Retry Timer (milliseconds)</td>
<td>500</td>
</tr>
<tr>
<td>Ping Retry Count</td>
<td>6</td>
</tr>
</tbody>
</table>

**SDP Information**

<table>
<thead>
<tr>
<th>Option</th>
</tr>
</thead>
<tbody>
<tr>
<td>Send send-receive SDP in mid-call INVITE</td>
</tr>
<tr>
<td>Allow Presentation Sharing using BFCP</td>
</tr>
<tr>
<td>Allow iX Application Media</td>
</tr>
<tr>
<td>Allow multiple codecs in answer SDP</td>
</tr>
</tbody>
</table>
Trunk configuration

Trunk configuration from Cisco UCM to the LAN side of the Cisco UBE:

**Navigation:** Device → Trunk → Add New

- **Select ‘Trunk Type’ as SIP Trunk and ‘Device Protocol’ as SIP and select ‘Next’ as shown below.**
- **Set Device Name:** CableOne is given for this example
- **Set Description:** CableOne SIP Trunk Certification is given for this example
- **Select Device Pool:** “Default” is selected for this example.
- **Set Significant Digits:** 4
- **Under Destination:**
  - **Set Destination Address:** 10.80.18.12 is given for this example. This is the CUBE LAN side VIP IP address.
  - **Set Destination Port:** 5060
  - **Set BLF Presence Group:** Standard presence group is selected from dropdown menu
  - **Set SIP Trunk security profile:** Non-Security SIP trunk Profile is selected from drop-down menu
  - **Set SIP Profile:** CableOne SIP Profile is selected from drop-down menu
Figure 12: Add SIP Trunk Type
**Figure 13: SIP Trunk to Cisco UBE**

<table>
<thead>
<tr>
<th>Device Information</th>
<th>SIP Trunk</th>
</tr>
</thead>
<tbody>
<tr>
<td>Product:</td>
<td>SIP Trunk</td>
</tr>
<tr>
<td>Device Protocol:</td>
<td>SIP</td>
</tr>
<tr>
<td>Trunk Service Type:</td>
<td>None (Default)</td>
</tr>
<tr>
<td><strong>Device Name</strong></td>
<td>CableOne</td>
</tr>
<tr>
<td><strong>Description</strong></td>
<td>CableOne SIP Trunk Certification</td>
</tr>
<tr>
<td><strong>Device Pool</strong></td>
<td>Default</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td></td>
</tr>
<tr>
<td>Call Classification</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL_Default</td>
</tr>
<tr>
<td>Location</td>
<td>Hub_None</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Tunneled Protocol</td>
<td>None</td>
</tr>
<tr>
<td>QSIG Variant</td>
<td>No Changes</td>
</tr>
<tr>
<td>ASN.1 ROSE OID Encoding</td>
<td>No Changes</td>
</tr>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
<tr>
<td>Media Termination Point Required</td>
<td></td>
</tr>
<tr>
<td>Retry Video Call as Audio</td>
<td></td>
</tr>
<tr>
<td>Path Replacement Support</td>
<td></td>
</tr>
<tr>
<td>Transmit UTF-8 for Calling Party Name</td>
<td></td>
</tr>
<tr>
<td>Transmit UTF-8 Names in QSIG APDU</td>
<td></td>
</tr>
<tr>
<td>Unattended Port</td>
<td></td>
</tr>
<tr>
<td>SRTP Allowed</td>
<td>When using both SRTP and TLS</td>
</tr>
<tr>
<td>Consider Traffic on This Trunk Secure</td>
<td>Default</td>
</tr>
<tr>
<td>Route Class Signaling Enabled</td>
<td>Default</td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>Default</td>
</tr>
</tbody>
</table>

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Configure the Virtual LAN IP address of the Cisco UBE and the Destination Port
Configure the SIP Trunk Security Profile and SIP Profile as shown below
The rest of the configuration is all default
• Click Save and Reset after completion

**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 there is no prefix assigned.

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

**Connected Party Settings**

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

**Outbound Calls**

- Called Party Transformation CSS: < None >
  - Use Device Pool Called Party Transformation CSS
  - Calling Party Transformation CSS: < None >
  - 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:
  - 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

---

Figure 15: SIP Trunk to Cisco UBE (Cont.)
Figure 16: SIP Trunk to Cisco UBE (Cont.)
Routing configuration

Route Pattern for Cisco UBE:

Navigation: Call Routing → Route/Hunt → Route Pattern → Add New

Route patterns are configured as below:
Cisco IP phone dial “11” 10 digits number to access PSTN via Cisco UBE
  o “1” is removed before sending to Cisco UBE
  o The rest of the number is sent to Cisco UBE and to Cable One Network
• *6 is used to dial the outbound calls with caller ID blocking

Figure 17: Add New Route Pattern for Cisco UBE
Figure 18: Route Pattern Configuration for Cisco UBE-PSTN Access
Figure 19: Route Pattern Configuration for Cisco UBE-PSTN Access (Cont.)
Figure 20: Route Pattern Configuration for Cisco UBE-PSTN Access - Anonymous Call
Figure 21: Route Pattern Configuration for Cisco UBE-PSTN Access - Anonymous Call (Cont.)
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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