Intelepeer SIP Trunking:

Connecting Cisco Unified Communications Manager [v12.0.1] to Intelepeer SIP Trunks via Cisco Unified Border Element v12.2.0 [IOS-XE 16.09.01]

January 24, 2019
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Introduction

Service Providers today, such as Intelepeer, 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.

Intelepeer 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 Intelepeer network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS-XE 16.09.01 can be used. The Cisco Unified Border Element v12.2 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 12.0.1 connected to Intelepeer 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 Intelepeer 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 12.0.1, and Cisco UBE on ISR 4321/K9 [IOS-XE – 16.09.01] for connectivity to Intelepeer SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM) to PSTN (Intelepeer) via Cisco UBE v12.2.0 [IOS-XE] 16.09.01.

- Testing was performed in accordance to Intelepeer 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 Intelepeer 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 Intelepeer SIP Trunking network.
**Network Topology**

- 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. Intelepeer 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 UDP and to Intelepeer 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>UDP with RTP</td>
</tr>
<tr>
<td>Transport from Cisco UBE to Intelepeer</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 4321/K9 router
- CUCM cluster on UCS C240, 1 Publisher node, 2 Subscriber nodes and Cisco Unity Connection
- Cisco 2851 with FXS ports and Analog Fax machine
- Generic Cisco IP-Phones: 7841 and 7941

Software Requirements
- CUBE-Version: 12.2.0 running IOS-XE 16.09.01
- CUCM UCOS 12.0.1.22900-11 for 1 Publisher and 2 Subscriber
- Cisco Unity Connection 12.0.1.22900-14
- Cisco IOS v15.0(1) for the fax gateway

Features

Features Supported
- Incoming and outgoing calls using G711ulaw and G729 codecs
- 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)
- DTMF (RFC2833)
- Fax (T38 and 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

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.
- Intelepeer has negotiated the fax rate only on G3 speed. However on T.38 Loop back scenarios the fax rate was on SG3 speed.
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 Intelepeer CUBE LAN
  ip address 10.64.5.8 255.255.0.0
  negotiation auto
  redundancy rii 4
  redundancy group 2 ip 10.64.5.28 exclusive

interface GigabitEthernet0/0/1
  description Intelepeer CUBE WAN
  ip address 192.65.xx.xx 255.255.255.224
  negotiation auto
  redundancy rii 3
  redundancy group 2 ip 192.65.xx.xx exclusive

interface GigabitEthernet0/1/0
  description Intelepeer CUBE HA
  ip address 10.80.10.48 255.255.255.0
  negotiation auto

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

Cisco UBE 2:

interface GigabitEthernet0/0/0
  description Intelepeer CUBE LAN
  ip address 10.64.5.9 255.255.0.0
  negotiation auto
  redundancy rii 4
  redundancy group 2 ip 10.64.5.28 exclusive
interface GigabitEthernet0/0/1
  description Intelepeer CUBE WAN
  ip address 192.65.xx.xx 255.255.255.224
  negotiation auto
  redundancy rii 3
  redundancy group 2 ip 192.65.xx.xx exclusive

interface GigabitEthernet0/1/0
  description Intelepeer CUBE HA
  ip address 10.80.10.49 255.255.255.0
  negotiation auto

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
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 2
sip
rel1xx 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>redundancy-group 1</td>
<td>Enable High Availability for the VoIP service</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 and G729 voice codecs are used for this testing. Codec preferences used to change according to the test plan description.

voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711ulaw
  codec preference 3 g711alaw

voice class codec 2
  codec preference 1 g711ulaw
  codec preference 2 g711alaw
  codec preference 3 g729r8

Dial Peer

Outbound Dial-peer to Intelepeer:

dial-peer voice 1 voip
description Ingress CUCM to PSTN - LAN Interface
huntstop
session protocol sipv2
session transport udp
incoming called-number .T
incoming uri via cucm
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
! 
dial-peer voice 2 voip
description Egress CUCM to PSTN - WAN Interface
huntstop
destination-pattern .T
session protocol sipv2
session target ipv4:68.68.xx.xx: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-npe
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad

Inbound Dial-peer from Intelepeer:

dial-peer voice 3 voip
description Ingress PSTN to CUCM - WAN Interface
huntstop
session protocol sipv2
session transport udp
incoming called-number 719270....
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-npe
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 4 voip
description Egress PSTN to CUCM - LAN Interface
huntstop
destination-pattern 719270....
session protocol sipv2
session target ipv4:10.80.10.3:5060
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-n-te
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
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 16.9
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname IntelepeerCUBE1
!
boot-start-marker
boot system flash isr4300-universalk9.16.09.01.SPA.bin
boot-end-marker
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
address-family ipv6
exit-address-family
!
enable secret 5 $1$nU
!
no aaa new-model
ip admission watch-list expiry-time 0
!
subscriber templating
multilink bundle-name authenticated
```
voice service voip
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 2
sip
  rel1xx supported "rel100"
  session refresh
  asserted-id pai
  privacy pstn
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
!
voice class uri cucm sip
  host 10.80.10.3
voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711ulaw
  codec preference 3 g711alaw
!
voice class codec 2
  codec preference 1 g711ulaw
  codec preference 2 g711alaw
  codec preference 3 g729r8
!
license udi pid ISR4321/K9 sn FDO19220MSQ
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level securityk9
no license smart enable
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
username **** privilege 15 password 0 *****
!
redundancy
  mode none
  application redundancy
    group 2
      name IntelepeerB2BHA
      priority 100 failover threshold 75
      timers delay 30 reload 60
      control GigabitEthernet0/1/0 protocol 1
      data GigabitEthernet0/1/0
      track 1 shutdown
      track 2 shutdown
    !
    track 1 interface GigabitEthernet0/0/0 line-protocol
    !
    track 2 interface GigabitEthernet0/0/1 line-protocol
    !
interface GigabitEthernet0/0/0
  description Intelepeer CUBE LAN
ip address 10.64.5.8 255.255.0.0
negotiation auto
redundancy rii 4
redundancy group 2 ip 10.64.5.28 exclusive
!
interface GigabitEthernet0/0/1
description Intelepeer CUBE WAN
ip address 192.65.xx.xx 255.255.255.224
negotiation auto
redundancy rii 3
redundancy group 2 ip 192.65.xx.xx exclusive
!
interface GigabitEthernet0/1/0
description Intelepeer CUBE HA
ip address 10.80.10.48 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 192.65.xx.xx
ip route 10.80.10.0 255.255.255.0 10.64.1.1
ip route 172.16.24.0 255.255.248.0 10.64.1.1
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 1 voip
description Ingress CUCM to PSTN - LAN Interface
huntstop
session protocol sipv2
session transport udp
incoming called-number .T
incoming uri via cucm
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 2 voip
description Egress CUCM to PSTN - WAN Interface
huntstop
destination-pattern .T
session protocol sipv2
session target ipv4:68.68.xx.xx: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-npe
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 3 voip
description Ingress PSTN to CUCM - WAN Interface
huntstop
session protocol sipv2
session transport udp
incoming called-number 719270....
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-npe
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 4 voip
description Egress PSTN to CUCM - LAN Interface
huntstop
destination-pattern 719270....
session protocol sipv2
session target ipv4:10.80.10.3:5060
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-n te
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
line con 0
  exec-timeout 0 0
  password ******
  login local
  transport input none
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 0 0
  password ******
  login local
end
Standby Cisco UBE:

version 16.9
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname IntelepeerCUBE2
!
boot-start-marker
boot system flash:isr4300-universalk9.16.09.01.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
  !
  address-family ipv4
  exit-address-family
  address-family ipv6
  exit-address-family
!
enable secret 5 $1$u
!
no aaa new-model
ip admission watch-list expiry-time 0
!
subscriber templating
!
multilink bundle-name authenticated
!
voice service voip
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 2
sip
rel1xx supported "rel100"
session refresh
asserted-id pai
privacy pstn
early-offer forced
midcall-signaling passthru
g729 annexb-all!

voice class uri cucm sip
host 10.80.10.3
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g711alaw!

voice class codec 2
codec preference 1 g711ulaw
codec preference 2 g711alaw
codec preference 3 g729r8!

license udi pid ISR4321/K9 sn FDO19220MQ9
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level uck9
license boot level securityk9
no license smart enable
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
username **** privilege 15 password 0 ******
!
redundancy
  mode none
  application redundancy
    group 2
      name IntelepeerB2BHA
      priority 100 failover threshold 75
      timers delay 30 reload 60
      control GigabitEthernet0/1/0 protocol 1
      data GigabitEthernet0/1/0
      track 1 shutdown
      track 2 shutdown
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
  description Intelepeer CUBE LAN
ip address 10.64.5.9 255.255.0.0
negotiation auto
redundancy rii 4
redundancy group 2 ip 10.64.5.28 exclusive
!
interface GigabitEthernet0/0/1
description Intelepeer CUBE WAN
ip address 192.65.xx.xx 255.255.255.224
negotiation auto
redundancy rii 3
redundancy group 2 ip 192.65.xx.xx exclusive
!
interface GigabitEthernet0/1/0
description Intelepeer CUBE HA
ip address 10.80.10.49 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
ip forward-protocol nd
ip http server
ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 192.65.xx.xx
ip route 10.80.10.0 255.255.255.0 10.64.1.1
ip route 172.16.24.0 255.255.248.0 10.64.1.1
!
control-plane
!
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 1 voip
   description Ingress CUCM to PSTN - LAN Interface
   huntstop
   session protocol sipv2
   session transport udp
   incoming called-number .T
   incoming uri via cucm
   voice-class codec 1
   voice-class sip bind control source-interface GigabitEthernet0/0/0
   voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 2 voip
   description Egress CUCM to PSTN - WAN Interface
   huntstop
destination-pattern .T
session protocol sipv2
session target ipv4:68.68.xx.xx: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-nle
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 3 voip
description Ingress PSTN to CUCM - WAN Interface
huntstop
session protocol sipv2
session transport udp
incoming called-number 719270....
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-nle
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 4 voip
description Egress PSTN to CUCM - LAN Interface
huntstop
destination-pattern 719270....
session protocol sipv2
session target ipv4:10.80.10.3:5060
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
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
gateway
  timer receive-rtp 1200
!
line con 0
  exec-timeout 0 0
  transport input none
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  login local
  transport input telnet
end
Configuring Cisco UCM 12.0.1 Cluster

Cisco UCM Version

Figure 3: Cisco UCM Version
Cisco Call Manager Service Parameters

**Navigation:** System → Service Parameters

- Select Server* = Clus20pub--CUCM Voice/Video (Active)
- Select Service* = Cisco CallManager (Active)
- 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 for Intelepeer is used as an example
- Description = Non Secure SIP Trunk Profile Intelepeer is used as an example
- Device Security Mode = Non Secure
- Incoming Transport Type* = TCP + UDP
- Outgoing Transport Type = UDP
Figure 5: SIP Trunk Security Profile
SIP Profile

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

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

![SIP Profile Configuration](image)

**Figure 6:** SIP Profile
### 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>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 Portion of Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup URI*</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI*</td>
<td>x-cisco-serviceuri-opickup</td>
</tr>
<tr>
<td>Call Pickup Group URI*</td>
<td>x-cisco-serviceuri-gpickup</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>
</tbody>
</table>

**Figure 7: SIP Profile (Cont.)**
<table>
<thead>
<tr>
<th>Call Forward URI*</th>
<th>x-cisco-serviceuri-cfwdail</th>
</tr>
</thead>
<tbody>
<tr>
<td>Speed Dial (Abbreviated Dial) URI*</td>
<td>x-cisco-serviceuri-abbrdial</td>
</tr>
</tbody>
</table>

- **Conference Join Enabled**
- **RFC 2543 Hold**
- **Semi Attended Transfer**
- **Enable VAD**
- **Stutter Message Waiting**
- **MLPP User Authorization**

**Normalization Script**

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

**Incoming Requests FROM URI Settings**

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

**Trunk Specific Configuration**

<table>
<thead>
<tr>
<th>Reroute Incoming Request to new Trunk based on*</th>
<th>Never</th>
</tr>
</thead>
<tbody>
<tr>
<td>Resource Priority Namespace List</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SIP Rel1XX Options*</td>
<td>Send FRACK if 1xx Contains SDP</td>
</tr>
<tr>
<td>Video Call Traffic Class*</td>
<td>Mixed</td>
</tr>
<tr>
<td>Calling Line Identification Presentation*</td>
<td>Default</td>
</tr>
<tr>
<td>Session Refresh Method*</td>
<td>Invite</td>
</tr>
<tr>
<td>Early Offer support for voice and video calls*</td>
<td>Disabled (Default value)</td>
</tr>
</tbody>
</table>

- **Enable ANAT**
- **Deliver Conference Bridge Identifier**
- **Allow Passthrough of Configured Line Device Caller Information**
- **Reject Anonymous Incoming Calls**
- **Reject Anonymous Outgoing Calls**
- **Send ILS Learned Destination Route String**
- **Connect Inbound Call before Playing Queuing Announcement**

Figure 8: SIP Profile (Cont.)
### SIP OPTIONS Ping

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable OPTIONS Ping to monitor destination status for Trunks with Service Type “None (Default)”</td>
<td><strong>on</strong></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

- [ ] Send send-receive SDP in mid-call INVITE
- [ ] Allow Presentation Sharing using BFCP
- [ ] Allow iX Application Media
- [ ] Allow multiple codecs in answer SDP

---

Figure 9: SIP Profile (Cont.)
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:** Intelepeer_SIP_trunk_to_CUBE is given for this example
- Set **Description:** Intelepeer SIP trunk to CUBE is given for this example
- Select **Device Pool:** “Intelepeer G711Ulawn” is selected for this example.
- Set **Significant Digits:** 4
- Under **Destination:**
  - Set **Destination Address:** 10.64.5.28 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 Secured SIP Trunk Profile for Intelepeer is selected from drop down menu
  - Set **SIP Profile:** Standard SIP Profile for Intelepeer is selected from drop down menu
Figure 11: Add SIP Trunk Type

<table>
<thead>
<tr>
<th>Trunk Type*</th>
<th>SIP Trunk</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Protocol*</td>
<td>SIP</td>
</tr>
<tr>
<td>Trunk Service Type*</td>
<td>None(Default)</td>
</tr>
</tbody>
</table>

* indicates required item.
Figure 12: SIP Trunk to Cisco UBE
Figure 13: SIP Trunk to Cisco UBE (Cont.)

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

**Connected Party Settings**

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

**Outbound Calls**

- Called Party Transformation CSS: < None >
- Use Device Pool Called Party Transformation CSS
- Calling Party Transformation CSS: < None >
- Use Device Pool Calling Party Transformation CSS
- Calling Party Selection*:
- Calling Line ID Presentation*:
- Calling Name Presentation*:
- Calling and Connected Party Info Format*:
- 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 14: SIP Trunk to Cisco UBE (Cont.)
Figure 15: SIP Trunk to Cisco UBE (Cont.)
Trunk configuration from Cisco UCM to Fax Gateway:

Navigation: Device → Trunk → Add New

Figure 16: Add New Trunk to Fax Gateway

- Select ‘Trunk Type’ as SIP Trunk and ‘Device Protocol’ as SIP and select ‘Next’ as shown below.
- Set Device Name: FaxGatewayIntelepeer is given for this example
- Set Description: faxGateway is given for this example
- Select Device Pool: “Intelepeer G711Ulaw” is selected for this example.
- Set Significant Digits: All
- Under Destination:
  - Set Destination Address: 172.16.31.50 is given for this example. This is the Fax gateway IP address.
  - Set Destination Port: 5060
- Set BLF Presence Group: Standard presence group is selected from dropdown menu
- Set SIP Trunk security profile: Non Secured SIP Trunk Profile for Intelepeer is selected from drop down menu
- Set SIP Profile: Standard SIP Profile for Intelepeer is selected from drop down menu
Figure 17: Add SIP Trunk Type

Figure 18: SIP Trunk to Fax Gateway
Figure 19: SIP Trunk to Fax Gateway (Cont.)

- Configure the IP address of the Fax gateway 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 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>
</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 >

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

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 20: SIP Trunk to Fax Gateway (Cont.)
Figure 21: SIP Trunk to Fax Gateway (Cont.)
Routing configuration

Route Pattern for Cisco UBE:

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

![Add New Route Pattern for Cisco UBE](image)

Figure 22: Add New Route Pattern for Cisco UBE
Figure 23: Route Pattern Configuration for Cisco UBE-PSTN Access
Figure 24: Route Pattern Configuration for Cisco UBE-PSTN Access (Cont.)

<table>
<thead>
<tr>
<th>Connected Party Transformations</th>
<th>Called Party Transformations</th>
<th>ISDN Network-Specific Facilities Information Element</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Line ID Presentation</td>
<td>Discard Digits</td>
<td>Service Protocol</td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td>Called Party Transform Mask</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td></td>
<td>Prefix Digits (Outgoing Call)</td>
<td>Carrier Identification Code</td>
</tr>
<tr>
<td></td>
<td>Called Party Number Type</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Called Party Numbering Plan</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Network Service Protocol</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>-- Not Selected --</td>
<td>&lt; Not Exist &gt;</td>
<td></td>
</tr>
</tbody>
</table>

Save  Delete  Copy  Add New

Figure 25 Route Pattern Configuration for Cisco UBE-PSTN Access National- Anonymous Call
Figure 26: Route Pattern Configuration for Cisco UBE-PSTN Access National-Anonymous Call (Cont.)
Route Pattern for Fax Gateway:

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

![Cisco Unified CM Administration](image)

Figure 27: Add New Route pattern to Fax Gateway

![Route Pattern Configuration](image)

Figure 28: Route Pattern Configuration for Fax Gateway
Figure 29: Route Pattern Configuration for Fax Gateway (Cont.)
Configuring Cisco Voice Gateway for Fax

Global Settings
voice service voip
  allow-connections sip to sip
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol t38 version 0 hs-redundancy 0 ls-redundancy 0 fallback none
no fax-relay sg3-to-g3
sip
  midcall-signaling passthru
  g729 annexb-all

Codecs

G711ulaw and G729 voice codecs are used for this testing. Codec preferences used to change according to the test plan description.

voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711ulaw
  codec preference 3 g711alaw
!
voice class codec 2
  codec preference 1 g711ulaw
  codec preference 2 g711alaw
  codec preference 3 g729r8
!

Dial Peer

Inbound Dial-peer from Cisco UCM:
dial-peer voice 4500 voip
  description Inbound DP for CUCM-FAX
  session protocol sipv2
  session target sip-server
  session transport udp
  incoming called-number 71927045x.
  voice-class codec 1
  dtmf-relay rtp-nre
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  no vad

Outbound Dial-peer to Cisco UCM:

dial-peer voice 4501 voip
  description Outbound DP for CUCM-FAX
  destination-pattern 8.
  session protocol sipv2
  session target ipv4:10.80.10.3:5060
  session transport udp
  voice-class codec 1
  dtmf-relay rtp-nre
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  no vad

POTS and Port Configuration

Based on configured POTS destination pattern, gateway forwards the call to designated voice port.

dial-peer voice 4502 pots
  service session
  destination-pattern 71927045xx
  no digit-strip
  port 0/0/0
  forward-digits all
voice-port 0/0/0
no echo-cancel enable
no vad
cptone IN
description Fax Test-Intelepeer
station-id number 71927045xx
caller-id enable
!

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>

Important Information

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