IntelePeer SIP Trunking:

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

May 20, 2016
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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 and IntelePeer network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS-XE 3.17 – 15.6(1)S can be used. The Cisco Unified Border Element 11.5.0 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.0.1 connected to IntelePeer IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco UCM (Cisco Unified Communications Manager). Only configuration settings specifically required for 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 a Cisco Unified Communications Manager (Cisco UCM) 11.0.1 and Cisco Unified Border Element (Cisco UBE) on ISR 4321/K9 [IOS-XE 3.17 – 15.6(1)S] for connectivity to IntelePeer SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 11.0.1) to PSTN (IntelePeer).

- 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 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 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.

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

Network Topology

- Cisco IP Phones 7975, 7965 and 9971 phones are the devices primarily used throughout the testing to place or receive calls
- VentaFax Soft Client is used to perform all fax related scenarios. The fax client is connected to SIP Gateway via FXS port which in turn communicates with Cisco UCM over SIP.

System Components

Hardware Requirements
- Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4321/K9 Router as CUBE
- Cisco 2851 Fax Gateway
- IP Phones 9971 (SIP), 7965 (SCCP) and 7975 (SIP)

Software Requirements
- Cisco Unified Communications Manager 11.0.1
- Cisco Unity Connection 11.0.1
- IOS-XE 3.17 – 15.6(1)S for ISR 4321/K9 Cisco Unified Border Element (CUBE 11.5.0)
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway
Features

Features Supported
- Incoming and outgoing off-net calls using G711ULaw and G729
- Call Hold
- Call Transfer (unattended and attended)
- Call Forward (all, busy and no answer)
- Calling Line (number) Identification Presentation (CLIP)
- Calling Line (number) Identification Restriction (CLIR)
- DTMF (RFC2833)
- Media flow-through on Cisco UBE
- Fax (G.711 pass-through and T.38)

Features Not Supported
- Cisco IP phones used in this test do not support blind transfer
- Fax re-invite is not supported by Service Provider

Caveats
- CLID is not updated on PSTN phones for transfer (attended and unattended) OffNet PSTN scenarios. Caller ID is not updated at PSTN once transfer is completed by PBX. Cisco UBE modify PAI/PPI header and forward to network in the tested release. CISCO BUG ID: CSCuv04539
- Privacy is not supported on Loopback to PBX. Anonymous calling out to network and back into same PBX.
  o Inbound to PBX Privacy works without issue
  o Outbound from PBX Privacy works without issue
- Intermittent no way speech path observed for an inbound calls. This issue is not indicative of CUBE/CUCM and Taqua standard behavior but is a Layer 3 issue. The intermittent audio issue is due to the nature of the lab used for testing by IntelePeer. IntelePeer has CUBE/CUCM customers running this deployment load on Taqua in production without audio issues
Configuration
Configuring Cisco Unified Border Element

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

interface GigabitEthernet0/0/0
ip address 10.80.18.21 255.255.255.0
media-type rj45
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.80.18.20 exclusive
!
interface GigabitEthernet0/0/1
ip address 192.65.x.x 255.255.255.128
negotiation auto
redundancy rii 2
redundancy group 1 ip 192.65.x.x exclusive
!

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

voice service voip
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
fax protocol pass-through G711ULaw
sip
rel1xx supported "rel100"
session refresh
asserted-id pai
privacy pstn
early-offer forced
midcall-signaling passthru
privacy-policy passthru
g729 annexb-all

Explanation

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>fax protocol</td>
<td>Specifies the fax protocol</td>
</tr>
<tr>
<td>asserted-id</td>
<td>Specifies the type of privacy header in the outgoing SIP requests and response messages</td>
</tr>
<tr>
<td>early-offer forced</td>
<td>Enables SIP Delayed-Offer to Early-Offer globally</td>
</tr>
<tr>
<td>midcall-signaling passthru</td>
<td>Passes SIP messages from one IP leg to another IP leg</td>
</tr>
</tbody>
</table>

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

```
voice service voip
    media flow-around
```

Codecs
G729 is used as the preferred codec for this testing and changed the codecs according to the test plan description

voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g729br8
codec preference 3 g711ulaw
Dial Peer

Cisco UBE uses dial-peers to route the call accordingly based on the digits
dial-peer voice 200 voip
description Outbound-from IP PBX to PSTN - WAN facing
translation-profile outgoing e16four
huntstop
destination-pattern .T
session protocol sipv2
session target sip-server
session transport tcp
voice-class codec 2
voice-class sip asserted-id pai
voice-class sip profiles 100
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 210 voip
description outgoing call to intelepeer - LAN facing
huntstop
session protocol sipv2
session target sip-server
incoming called-number .T
voice-class codec 2
voice-class sip asserted-id pai
voice-class sip profiles 100
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nre
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 500 voip
description cube-dp incoming call from PSTN
huntstop
session protocol sipv2
session transport tcp
incoming called-number +1949204....
voice-class codec 2
voice-class sip asserted-id pai
voice-class sip profiles 100
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nre
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 100 voip
description Inbound-from PSTN to IP PBX - LAN facing
translation-profile outgoing noe16four
huntstop
destination-pattern +1949204....
session protocol sipv2
session target ipv4:10.80.18.3:5060
voice-class codec 2
voice-class sip asserted-id pai
voice-class sip profiles 100
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nre
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!

Call Flow

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

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

CPE callers make outbound PSTN calls by dialing a “9” prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, Cisco fax Gateway sends to Cisco UCM the DID with leading access code “9”. A route pattern strips the prefix and routes the call with the remaining digits via a SIP trunk terminating on the Cisco UBE for Voice call or Fax. For PBX to PBX via IntelePeer, Caller dial 9 prefix followed by the 10 digit DID provided by IntelePeer, 9 was stripped and the ten-digit number was send to Cisco UBE, Cisco UBE translate the 10 digits extension number to its full ten-digits DID with E.164 format under Dial Peer 200 and send to IntelePeer network which will direct back to Cisco UBE and handled same as normal incoming PSTN call.

Figure 2: Outbound Voice Call

Figure 3: Inbound Voice Call

Figure 4: Outbound Fax Call
Figure 5: Inbound Fax Call

Figure 6: PBX to PBX via IntelePeer Call

Configuration Example
The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously
Active Cisco UBE

IN_CUBE1#sh running
!
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname IN_CUBE1
!
boot-start-marker
boot system bootflash:isr4300-universalk9.03.17.01.S.156-1.S1-std.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no aaa new-model
!
no ip domain lookup
ip domain name tekvizion.com
!
subscriber templating
!
multilink bundle-name authenticated

voice service voip
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
fax protocol pass-through g711ulaw
sip
rel1xx supported "rel100"
session refresh
asserted-id pai
privacy pstn
early-offer forced
midcall-signaling passthru
g729 annexb-all

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

voice class codec 2
codec preference 1 g729r8
codec preference 2 g729br8
codec preference 3 g711ulaw
voice class sip-profiles 100

request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:+1949204\1@\2"

voice translation-rule 1
rule 1 /\(...........\)$>/ /+1\1/

voice translation-rule 2
rule 1 /\(...........\)$>/ /+1\1/
rule 2 /\(...........\)$>/ /+1\1/
rule 3 /\(442037\...\)\.\)/ /+\1/

voice translation-rule 3
rule 1 /\+1\1\(...........\)\)/ /\1/

voice translation-rule 4
rule 1 /\+1\1\(...........\)\)/ /\1/

voice translation-profile e16four
translate calling 1
translate called 2

voice translation-profile noe16four
translate calling 3
translate called 4

license udi pid ISR4321/K9 sn FDO19280MSQ

spanning-tree extend system-id
! redundancy
mode none
application redundancy
group 1
  name voice-b2bhaIntelePeer
  priority 100 failover threshold 75
timers delay 30 reload 60
control GigabitEthernet0/1/0 protocol 1
data GigabitEthernet0/1/0
  track 1 shutdown
  track 2 shutdown

! vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
translation-rule 1
!
interface GigabitEthernet0/0/0
  ip address 10.80.18.21 255.255.255.0
  media-type rj45
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.18.20 exclusive
!
interface GigabitEthernet0/0/1
ip address 192.65.XX.XXX 255.255.255.128
negotiation auto
redundancy rii 2
redundancy group 1 ip 192.65.x.x exclusive
!
interface GigabitEthernet0/1/0
description CUBE HA MS5 3/0/37
ip address 10.89.20.9 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
negotiation auto
!
interface Vlan1
no ip address
shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 192.65.XX.XXX
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.80.0.0 255.255.0.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 10.80.18.1
ip route 172.16.31.0 255.255.255.0 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

dial-peer voice 200 voip

description Outbound-from IP PBX to PSTN - WAN facing
translation-profile outgoing e16four
huntstop
destination-pattern .T
session protocol sipv2
session target sip-server
session transport tcp
voice-class codec 2
voice-class sip asserted-id pai
voice-class sip profiles 100
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
! 

dial-peer voice 210 voip

description outgoing call to intelepeer - LAN facing

huntstop

session protocol sipv2

session target sip-server

incoming called-number .T

voice-class codec 2

voice-class sip asserted-id pai

voice-class sip profiles 100

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

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

dt mf-relay rtp-n te

fax-relay ecm disable

fax rate 14400

fax nsf 000000

fax protocol pass-through g711ulaw

no vad

! 

dial-peer voice 500 voip

description cube-dp incoming call from PSTN

huntstop

session protocol sipv2

session transport tcp

incoming called-number +19492......

voice-class codec 2

voice-class sip asserted-id pai

voice-class sip profiles 100

voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-n-te
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 100 voip
description Inbound-from PSTN to IP PBX - LAN facing
translation-profile outgoing noe16four
huntstop
destination-pattern +19492......
session protocol sipv2
session target ipv4:10.80.18.3:5060
voice-class codec 2
voice-class sip asserted-id pai
voice-class sip profiles 100
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-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
!
sip-ua
keepalive target ipv4:68.68.XXX.xx:5060
timers keepalive active 180
sip-server ipv4:68.68.XX.xx:5060
!
!
line con 0
   stopbits 1
line aux 0
   stopbits 1
line vty 0 4
   exec-timeout 0 0
   password xxxx
   login local
!
!
End
Standby Cisco UBE

IN_CUBE2#sh run

! version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core

! hostname IN_CUBE2

! boot-start-marker
boot system bootflash:isr4300-universalk9.03.17.01.S.156-1.S1-std.SPA.bin
boot-end-marker

! vrf definition Mgmt-intf

! address-family ipv4
exit-address-family

! address-family ipv6
exit-address-family

! no aaa new-model

! no ip domain lookup
ip domain name tekvizion.com

! subscriber templating
multilink bundle-name authenticated
!
voice service voip
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
fax protocol pass-through g711ulaw
sip
rel1xx supported "rel100"
session refresh
asserted-id pai
privacy pstn
early-offer forced
midcall-signaling passthru
g729 annexb-all
!
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g729br8
codec preference 3 g711ulaw
!
!
voice class sip-profiles 100
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:+1949204\1@\2"

! voice translation-rule 1
rule 1 \(.........\$/ /+1\1/
!

voice translation-rule 2
rule 1 \(.........\$/ /+1\1/
rule 2 \(.........\$/ /+1\1/
rule 3 \(442037\$/ /+1\1/
!

voice translation-rule 3
rule 1 \+1\1\1\$/ /1/
!

voice translation-rule 4
rule 1 \+1\1\1\$/ /1/
!

voice translation-profile e16four
translate calling 1
translate called 2
!

voice translation-profile noe16four
translate calling 3
translate called 4
!

license udi pid ISR4321/K9 sn FDO19290MQ9
!

spanning-tree extend system-id
!

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redundancy
mode none
application redundancy

group 1
 name voice-b2bhaIntelePeer
 priority 100 failover threshold 75
 timers delay 30 reload 60
 control GigabitEthernet0/1/0 protocol 1
 data GigabitEthernet0/1/0
 track 1 shutdown
 track 2 shutdown!

! vlan internal allocation policy ascending
!
 track 1 interface GigabitEthernet0/0/0 line-protocol
 track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
 ip address 10.80.18.22 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.18.20 exclusive
!
interface GigabitEthernet0/0/1
 ip address 192.65.XX.XXX 255.255.255.128
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.65.XX.XXX exclusive
interface GigabitEthernet0/1/0
description CUBE HA MS5 3/0/38
ip address 10.89.20.10 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
interface Vlan1
no ip address
shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 192.65.XX.XXX
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.80.0.0 255.255.0.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 10.80.18.1
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 200 voip
description Outbound-from IP PBX to PSTN - WAN facing
translation-profile outgoing e16four
huntstop
destination-pattern .T
session protocol sipv2
session target sip-server
session transport tcp
voice-class codec 2
voice-class sip asserted-id pai
voice-class sip profiles 100
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 210 voip
description outgoing call to intelepeer - LAN facing
huntstop

session protocol sipv2
session target sip-server
incoming called-number .T
voice-class codec 2
voice-class sip asserted-id pai
voice-class sip profiles 100
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-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 500 voip
description cube-dp incomming call from PSTN
huntstop
session protocol sipv2
session transport tcp
incoming called-number +19492……
voice-class codec 2
voice-class sip asserted-id pai
voice-class sip profiles 100
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-n-te
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 100 voip
description Inbound-from PSTN to IP PBX - LAN facing
translation-profile outgoing noe16four
huntstop
destination-pattern +19492......
session protocol sipv2
session target ipv4:10.80.18.3:5060
voice-class codec 2
voice-class sip asserted-id pai
voice-class sip profiles 100
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-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
sip-ua
keepalive target ipv4:68.68.123.174:5060
timers keepalive active 180
sip-server ipv4:68.68.123.174:5060
!
!
line con 0
stopbits 1
line aux 0
stopbits 1
line vty 0 4
password xxxx
login local
!
!
end

IN_CUBE2#
Cisco Call Manager Service Parameters

**Navigation Path:** System > Service Parameters

Select Server* = Clus28Sub1--CUCM Voice/Video (Active)

Select Service* = Cisco CallManager (Active)

All other fields are set to default values

---

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Figure 8: Service Parameters

Offnet Calls via IntelePeer SIP Trunk

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

SIP Trunk Security Profile

Navigation Path: System > Security > SIP Trunk Security Profile

Name* = Intelepeer Non Secure SIP Trunk Profile

Description = non Secure SIP Trunk Profile authenticated by null String

![SIP Trunk Security Profile Configuration](image)

Figure 9: SIP Trunk Security Profile

Explanation

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Transport Type</td>
<td>TCP + UDP</td>
<td>SIP trunks to IntelePeer SBC should use UDP as a transport protocol for SIP. This is configured using SIP Trunk Security profile, which is later assigned to the SIP trunk itself.</td>
</tr>
<tr>
<td>Outgoing Transport Type</td>
<td>UDP</td>
<td></td>
</tr>
</tbody>
</table>
SIP Profile Configuration

SIP Profile will be later associated with the SIP trunk

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

- **Name:** Intelepeer SIP Profile
- **Description:** Default SIP Profile

![SIP Profile Configuration](image)

**Figure 10: SIP Profile**
Figure 11: SIP Profile (Cont.)
## Explanation

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default MTP Telephony Event Payload Type</td>
<td>101</td>
<td>RFC2833 DTMF payload type</td>
</tr>
<tr>
<td>SIP Rel1XX Options</td>
<td>Send PRACK for 1xx Messages</td>
<td>Enable Provisional Acknowledgements (Reliable 100 messages)</td>
</tr>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)</td>
<td>30</td>
<td>OPTIONS message parameters- interval time- This is used in this example</td>
</tr>
<tr>
<td>Ping Interval for Out-of-service Trunks (seconds)</td>
<td>120</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
</tbody>
</table>
SIP Trunk Configuration
Create SIP trunks to Cisco UBE

Navigation Path: Device > Trunk

Figure 13: SIP Trunks List
### Figure 14: SIP Trunk to Cisco UBE (Cont.)
### Call Routing Information

- **Remote-Party-ID**
- **Asserted-Identity**
- **Asserted-Type**
- **SIP Privacy**

#### Inbound Calls

- **Significant Digits**
- **Connected Line ID Presentation**
- **Connected Name Presentation**
- **Calling Search Space**
- **AAR Calling Search Space**
- **Prefix DN**

- **Redirecting Diversion Header Delivery - Inbound**

#### Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

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

#### Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

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

#### Connected Party Settings

- **Connected Party Transformation CSS**
- **Use Device Pool Connected Party Transformation CSS**

#### Outbound Calls

- **Called Party Transformation CSS**
- **Use Device Pool Called Party Transformation CSS**
- **Calling Party Transformation CSS**
- **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**
- **Use Device Pool Redirecting Party Transformation CSS**

#### Caller Information

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

### SIP Information

#### Destinations

<table>
<thead>
<tr>
<th>Destination Address is an SRV</th>
<th>Destination Address</th>
<th>Destination Address (IPv6)</th>
<th>Destination Port</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>✗</td>
<td>10.80.10.20</td>
<td>10.80.10.20</td>
<td>5060</td>
<td>down</td>
</tr>
</tbody>
</table>

---

Figure 15: SIP Trunk to Cisco UBE (Cont.)
Figure 16: 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>Intelepeer</td>
<td>Name for the trunk</td>
</tr>
<tr>
<td>Device Pool</td>
<td>G711_pool</td>
<td>Default Device Pool is used for this trunk</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL_Default</td>
<td>MRG with resources: ANN, CFB, MOH and MTP</td>
</tr>
<tr>
<td>Significant Digits</td>
<td>4</td>
<td>4 digits Extension for all CPE phones</td>
</tr>
<tr>
<td>Destination Address</td>
<td>10.80.18.20</td>
<td>IP address of the Cisco UBE Virtual LAN</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
<td>Intelepeer Non Secure SIP Trunk Profile</td>
<td>SIP Trunk Security Profile configured earlier</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>Intelepeer Standard SIP Profile</td>
<td>SIP Profile configured earlier</td>
</tr>
</tbody>
</table>
Dial Plan

Route Pattern Configuration

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

Route patterns are configured as below:

- Cisco IP phone dial "9". 10 digits number to access PSTN via Cisco UBE
  - "9" is removed before sending to Cisco UBE
- For FAX call, Access Code "9"+ 10 digits number is used at Cisco Fax gateway
  - "9" is removed at Cisco UCM
  - The rest of the number is sent to Cisco UBE to IntelePeer network
- Incoming fax call to xxxx will be sent to the Cisco Fax gateway

![Cisco Unified CM Administration](image)

Figure 17: Route Patterns List
Figure 16: Route Pattern for Voice
Figure 19: Route Pattern for Voice
Figure 21: Route Pattern for Voice – Cont.
**Pattern Definition**

<table>
<thead>
<tr>
<th>Route Pattern</th>
<th>2900</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Partition</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Description</td>
<td>VM pilot Number</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>MUPP Precedence*</td>
<td>Default</td>
</tr>
</tbody>
</table>

- **Apply Call Blocking Percentage**
- **Resource Priority Namespace Network Domain**
- **Route Class***
- **Gateway/Route List**
- **Route Option**
  - Route this pattern
  - Block this pattern
  - No Error

- **Call Classification**
- **External Call Control Profile**
- **Allow Device Override**
- **Provide Outside Dial Tone**
- **Allow Overlap Sending**
- **Urgent Priority**
- **Require Forced Authorization Code**
- **Authorization Level**
- **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**
- **Calling Name Presentation**
- **Calling Party Number Type**
- **Calling Party Numbering Plan**

**Connected Party Transformations**

- **Connected Line ID Presentation**
- **Connected Name Presentation**

**Called Party Transformations**

- **Discard Digits**
- **Called Party Transform Mask**
- **Prefix Digits (Outgoing Calls)**
- **Called Party Number Type**
- **Called Party Numbering Plan**

**ISDN Network-Specific Facilities Information Element**

<table>
<thead>
<tr>
<th>Network Service Protocol</th>
<th>-- Not Selected --</th>
</tr>
</thead>
<tbody>
<tr>
<td>Carrier Identification Code</td>
<td>-- Not Selected --</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Network Service</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>-- Not Selected --</td>
<td>-- Not Exist --</td>
<td></td>
</tr>
</tbody>
</table>

Figure 22: Route Pattern for Voice
**Explanation**

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>9.! For Voice call, 5421 for fax call,*6.! For restricted calls,2900 for Voice mail pilot number</td>
<td>Specify appropriate Route Pattern</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>Routed the calls to the appropriate trunk and gateways</td>
<td>SIP Trunk name configured earlier</td>
</tr>
<tr>
<td>Discard Digits</td>
<td>PreDot for Route Pattern 9.!,&quot;6.!.</td>
<td>Specifies how to modify digit before they are sending to IntelePeer network</td>
</tr>
</tbody>
</table>

**Acronyms**

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>CPE</td>
<td>Customer Premise Equipment</td>
</tr>
<tr>
<td>Cisco UBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>Cisco UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>MTP</td>
<td>Media Termination Point</td>
</tr>
<tr>
<td>POP</td>
<td>Point of Presence</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>ESBC</td>
<td>Enterprise Session Border Controller</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
</tbody>
</table>
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