IntelePeer SIP Trunking:

Cisco Unified Communications Manager 11.5.1 with Cisco Unified Border Element (CUBE 11.6.0) on ISR 4321/K9 [IOS-XE - 16.5.1b] using SIP

July 26, 2017
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Service Providers today, such as IntelePeer, are offering alternative methods to connect to the PSTN via their IP networks. Most of these services utilize SIP as the primary signaling method and centralized IP to TDM POP gateways to provide on-net and off-net services.

A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and Spectrum network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS 16.5.1b can be used. The Cisco Unified Border Element 16.5.1b provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.1 connected to IntelePeer 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) 11.5.1 and Cisco Unified Border Element (Cisco UBE) on ISR 4321/K9 [IOS - 16.5.1b] for connectivity to IntelePeer SIP Trunking service available in the former IntelePeer Business service area. The deployment model covered in this application note is CPE (Cisco UCM 11.5.1) 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:

Figure 1: Network Topology

Figure 2: Cisco UBE High Availability
System Components

Hardware Requirements
- Cisco UCSC-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR4431/K9 router as CUBE
- Cisco ISR4431/K9 (1RU) processor with 1684579K/6147K bytes of memory with 4 Gigabit Ethernet interfaces
- Processor board ID FTX1845AJ9S
- Cisco 2851 Fax Gateway
- IP phones 9971 (SIP) and 8945 (SIP)

Software Requirements
- Cisco Unified Communications Manager 11.5.1
- Cisco Unity Connection 11.5.1
- IOS-XE 16.05.01b for ISR 4431/K9 Cisco Unified Border Element
- Cisco IOS XE Software, Version 03.17.01.S
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway
Features

**Features Supported**
- Incoming and outgoing off-net calls using G711ULaw
- Call hold
- Call transfer (unattended and attended)
- Call conference
- Call forward (all, busy and no answer)
- Calling Line (number) Identification Presentation (CLIP)
- Calling Line (number) Identification Restriction (CLIR)
- DTMF relay (both directions) (RFC2833)
- Media flow-through on Cisco UBE
- Fax (G.711 pass-through and T38)

**Features Not Supported**
- Cisco IP phones used in this test do not support blind transfer
- In HA redundancy mode, the primary cube will not take over the primary/active role after a reboot/network outage

**Caveats**
- Caller ID is not updated after attended or unattended transfers to off-net phones. This is due to a limitation on Cisco UBE and will be resolved in the next release. The issue does not impact the calls.
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 192.65.79.135 255.255.255.128
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 192.65.79.250 exclusive
!
interface GigabitEthernet0/0/2
  ip address 10.80.11.17 255.255.255.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.11.20 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
mode border-element
allow-connections sip to sip
redundancy-group 1
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
rel1xx supported "rel100"
header-passing
  asserted-id pai
  early-offer forced
midcall-signaling passthru
privacy-policy passthru
pass-thru content sdp

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>
Codecs
G711Ulaw is used as the preferred codec for this testing and changed the preferences according to the test plan description

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

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

- dial-peer voice 10 voip
description Incoming from CUCM
huntstop
session protocol sipv2
incoming called-number [0-9]T
voice-class codec 2
voice-class sip profiles 100
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!

- dial-peer voice 20 voip
description Outgoing to Intelepeer
huntstop
destination-pattern [0-9]T
session protocol sipv2
session target dns:tekv.vsiptest.Intelepeer .com
voice-class codec 2
voice-class sip options-ping 60
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-nte
fax-relay sg3-to-g3
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 30 voip
description Incoming from Intelepeer
huntstop
session protocol sipv2
incoming called-number +1980
voice-class codec 2
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-nte
fax-relay sg3-to-g3
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 40 voip
description Outgoing to CUCM
huntstop
destination-pattern +1980
session protocol sipv2
session target ipv4:10.80.11.2
voice-class codec 2
voice-class sip profiles 100
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
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 “7” 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 “7”. A “7.@” 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 7 prefix followed by the target 1+10Digit DID no for that extension number, 7 was stripped and the 1+10 digits number was send to Cisco UBE, Cisco UBE sends the full 1+ 10 digits DID under Dial Peer 20 and send to Spectrum network which will direct back to Cisco UBE and handled same as normal incoming PSTN call. For International calls same pattern 7 followed by 011, country code and calling no is used.

Figure 3: Outbound Voice Call

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

Figure 6: Inbound Fax Call

Figure 7: PBX to PBX via Spectrum Call
Configuration Example

The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously.

**Active Cisco UBE**

*User Access Verification*
*Username: cisco*
*Password:*

```bash
isr4K2Intelepeer #sh run
isr4K2Intelepeer #sh running-config
```

```text
version 16.5
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
platform qfp utilization monitor load 80
no platform punt-keepalive disable-core
!
hostname isr4K2Intelepeer
!
boot-start-marker
boot-end-marker
!
vrf definition Mgmt-intf
  
  address-family ipv4
  exit-address-family
  
  address-family ipv6
  exit-address-family
  
logging console emergencies
enable secret 5 *****
!
no aaa new-model
!
ip name-server 8.8.8.8
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-988930787
enrollment selfsigned
subject-name cn=IOS-Self-Signed-Certificate-988930787
```
revocation-check none
rsakeypair TP-self-signed-988930787

!  
crypto pki certificate chain TP-self-signed-988930787

!  
voice service voip
no ip address trusted authenticate
mode border-element
allow-connections sip to sip
redundancy-group 1
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
rel1xx supported "rel100"
header-passing
asserted-id pai
early-offer forced
midcall-signaling passthru
privacy-policy passthru
pass-thru content sdp

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

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

!  
voice class sip-profiles 100
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:980233\1@\2"

!  
license udi pid ISR4431/K9 sn FOC18232988
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level uck9
license boot level securityk9

!  
diagnostic bootup level minimal
spanning-tree extend system-id

!  
username cisco privilege 15 password 7 *****

!
redundancy
mode none
application redundancy
group 1
  name voice-b2bha
  timers delay 30 reload 60
  control GigabitEthernet0/0/3 protocol 1
data GigabitEthernet0/0/3
  track 1 shutdown
  track 2 shutdown
!
  track 1 interface GigabitEthernet0/0/2 line-protocol
  !
  track 2 interface GigabitEthernet0/0/0 line-protocol
  !
  interface GigabitEthernet0/0/0
    ip address 192.65.79.135 255.255.255.128
    negotiation auto
    redundancy rii 2
    redundancy group 1 ip 192.65.79.250 exclusive
  !
  interface GigabitEthernet0/0/1
    no ip address
    negotiation auto
  !
  interface GigabitEthernet0/0/2
    ip address 10.80.11.17 255.255.255.0
    negotiation auto
    redundancy rii 1
    redundancy group 1 ip 10.80.11.20 exclusive
  !
  interface GigabitEthernet0/0/3
    ip address 10.20.20.1 255.255.255.0
    negotiation auto
  !
  interface GigabitEthernet0
    vrf forwarding Mgmt-intf
    no ip address
    negotiation auto
  !
  threat-visibility
    ip forward-protocol nd
    ip http server
    ip http authentication local
    ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.255.0 10.80.11.1
ip route 172.16.24.0 255.255.248.0 10.80.11.1
!
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
!
dial-peer voice 10 voip
description Incoming from CUCM
huntstop
session protocol sipv2
incoming called-number [0-9]T
voice-class codec 2
voice-class sip profiles 100
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 20 voip
description Outgoing to IntelePeer
huntstop
destination-pattern [0-9]T
session protocol sipv2
session target dns:tekv.vsiptest.Intelepeer.com
voice-class codec 2
voice-class sip options-ping 60
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-nte
fax-relay sg3-to-g3
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 30 voip
description Incoming from IntelePeer
huntstop
session protocol sipv2
incoming called-number +1980
voice-class codec 2
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-npe
fax-relay sg3-to-g3
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 40 voip
description Outgoing to CUCM
huntstop
destination-pattern +1980
session protocol sipv2
session target ipv4:10.80.11.2
voice-class codec 2
voice-class sip profiles 100
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-npe
fax-relay sg3-to-g3
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
sip-ua
!
line con 0
exec-timeout 0 0
transport input none
stopbits 1
line aux 0
stopbits 1
line vty 0 4
  logging synchronous
  login local
!
no network-clock synchronization automatic
ntp server pool.ntp.org
ntp server time-pnp.cisco.com

**Standby Cisco UBE**

isr4k1Intelepeer #sh running-config

version 16.5
service timestamps debug datatime msec
service timestamps log datatime msec
service password-encryption
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname isr4k1Intelepeer
!
boot-start-marker
boot-end-marker
!
vrp definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
logging console emergencies
enable secret 5
!
no aaa new-model
!
ip name-server 8.8.8.8
!
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-1179880555
enrollment selfsigned
subject-name cn=IOS-Self-Signed-Certificate-1179880555
revocation-check none
rsakeypair TP-self-signed-1179880555

voice service voip
no ip address trusted authenticate
mode border-element
allow-connections sip to sip
redundancy-group 1
tax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
bind control source-interface GigabitEthernet0/0/0
bind media source-interface GigabitEthernet0/0/0
rel1xx supported "rel100"
header-passing
asserted-id pai
early-offer forced
midcall-signaling passthru
privacy-policy passthru
pass-thru content sdp

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

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

voice class sip-profiles 100
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:980233\1@\2"

voice-card 0/1
no watchdog

license udi pid ISR4431/K9 sn FOC18261KJL
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level uck9
license boot level securityk9
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 7 *****
!
redundancy
mode none
application redundancy
group 1
  name voice-b2bha
  timers delay 30 reload 60
  control GigabitEthernet0/0/3 protocol 1
data GigabitEthernet0/0/3
  track 1 shutdown
  track 2 shutdown
!
track 1 interface GigabitEthernet0/0/2 line-protocol
!
track 2 interface GigabitEthernet0/0/0 line-protocol
!
interface GigabitEthernet0/0/0
  ip address 192.65.79.134 255.255.255.128
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 192.65.79.250 exclusive
!
interface GigabitEthernet0/0/1
  no ip address
  negotiation auto
!
interface GigabitEthernet0/0/2
  ip address 10.80.11.18 255.255.255.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.11.20 exclusive
!
interface GigabitEthernet0/0/3
  ip address 10.20.20.2 255.255.255.0
  negotiation auto
!
interface Service-Engine0/1/0
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
no ip address
negotiation auto
!
threat-visibility
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.2.100 255.255.255.255 10.80.11.1
ip route 172.16.24.0 255.255.248.0 10.80.11.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
control-plane
!
voice-port 0/1/0
!
voice-port 0/1/1
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 10 voip
description Incoming from CUCM
huntstop
session protocol sipv2
incoming called-number [0-9]T
voice-class codec 2
voice-class sip profiles 100
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 20 voip
description Outgoing to Intelepeer
translation-profile outgoing 10
huntstop
destination-pattern [0-9]T
session protocol sipv2
session target dns:tekv.vsiptest.Intelepeer.com
voice-class codec 2
voice-class sip options-ping 60
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-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 Is-redundancy 0 hs-redundancy 0 fallback none
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!
dial-peer voice 30 voip
description Incoming from Intelepeer
huntstop
session protocol sipv2
incoming called-number +1980
voice-class codec 2
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-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 Is-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 40 voip
description Outgoing to CUCM
huntstop
destination-pattern +1980
session protocol sipv2
session target ipv4:10.80.11.2
voice-class codec 2
voice-class sip profiles 100
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
sip-ua
!
line con 0
  exec-timeout 0 0
  logging synchronous
  transport input none
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  logging synchronous
  login local
!
no network-clock synchronization automatic
ntp server pool.ntp.org
ntp server time-pnp.cisco.com
Configuring Cisco Unified Communications Manager

Cisco UCM Version

Navigation: System → Service Parameters

1. Select **Server**: Clus21Sub1--CUCM Voice/Video (Active)
2. Select **Service**: Cisco CallManager (Active)
3. All other fields are set to default values

Offset Calls via IntelePeer SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and the IntelePeer network and calls are routed via Cisco UBE.
SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

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

---

**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>

---

**Figure 10: SIP Trunk Security Profile**
SIP Profile Configuration

SIP Profile will be later associated with the SIP trunk

Navigation: Device → Device Settings → SIP Profile

1. **Name**: Intelepeer SIP Profile
2. **Description**: Intelepeer SIP Profile

![SIP Profile Information](Figure 11: SIP Profile)
<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>DSCP for Audio Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup URI&lt;sup&gt;*&lt;/sup&gt;</td>
<td>x-cisco-serviceun-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI&lt;sup&gt;*&lt;/sup&gt;</td>
<td>x-cisco-serviceun-pickup</td>
</tr>
<tr>
<td>Call Pickup Group URI&lt;sup&gt;*&lt;/sup&gt;</td>
<td>x-cisco-serviceun-pickup</td>
</tr>
<tr>
<td>Meet Me Service URI&lt;sup&gt;*&lt;/sup&gt;</td>
<td>x-cisco-serviceun-meetme</td>
</tr>
<tr>
<td>User Info&lt;sup&gt;*&lt;/sup&gt;</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level&lt;sup&gt;*&lt;/sup&gt;</td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back&lt;sup&gt;*&lt;/sup&gt;</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block&lt;sup&gt;*&lt;/sup&gt;</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking&lt;sup&gt;*&lt;/sup&gt;</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control&lt;sup&gt;*&lt;/sup&gt;</td>
<td>User</td>
</tr>
<tr>
<td>Taint Level for 7940 and 7960&lt;sup&gt;*&lt;/sup&gt;</td>
<td>Disabled</td>
</tr>
<tr>
<td>Resource Priority Namespace&lt;sup&gt;*&lt;/sup&gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)&lt;sup&gt;*&lt;/sup&gt;</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)&lt;sup&gt;*&lt;/sup&gt;</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)&lt;sup&gt;*&lt;/sup&gt;</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirects&lt;sup&gt;*&lt;/sup&gt;</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)&lt;sup&gt;*&lt;/sup&gt;</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI&lt;sup&gt;*&lt;/sup&gt;</td>
<td>x-cisco-serviceun-cfwall</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI&lt;sup&gt;*&lt;/sup&gt;</td>
<td>x-cisco-serviceun-abbrevdial</td>
</tr>
</tbody>
</table>

- **Conference Join Enabled**
- **RFC 2545 Hold**
- **Semi Attended Transfer**
- **Enable VAD**
- **Stutter Message Waiting**
- **MLPP User Authorization**

### Normalization Script

**Normalization Script**: < None >

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

### Incoming Requests FROM URI Settings

**Caller ID DN**

**Caller Name**

---

*Figure 12: SIP Profile (Cont.)*
**Table:** SIP Profile (Cont.)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default MTP Telephony Event Payload Type</td>
<td>101</td>
<td>RFC2833 DTMF payload type</td>
</tr>
<tr>
<td>SIP Rel1XX Options</td>
<td>Send PRACK for 1xx Messages</td>
<td>Enable Provisional Acknowledgements (Reliable 100 messages)</td>
</tr>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)</td>
<td>60</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
<tr>
<td>Ping Interval for Out-of-service Trunks (seconds)</td>
<td>120</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
</tbody>
</table>
SIP Trunk Configuration
Create SIP trunks to Cisco UBE

Navigation: Device → Trunk

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Calling Search Space</th>
<th>Device Pool</th>
<th>Route Pattern</th>
<th>Portion</th>
<th>Route Group</th>
<th>Priority</th>
<th>Trunk Type</th>
<th>SIP Trunk Status</th>
<th>SIP Trunk Duration</th>
<th>SIP Trunk Security Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>guam-fagateway</td>
<td>guam-fagateway</td>
<td>G711 Pool</td>
<td>BG911</td>
<td>SIP</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Non Secure SIP Trunk Profile</td>
</tr>
</tbody>
</table>

Figure 14: SIP Trunks List

<table>
<thead>
<tr>
<th>Device Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Product: SIP Trunk</td>
</tr>
<tr>
<td>Device Protocol: SIP</td>
</tr>
<tr>
<td>Trunk Service Type: None (Default)</td>
</tr>
<tr>
<td>Device Name: Intelepeer_Trunk</td>
</tr>
<tr>
<td>Description: Intelepeer_Trunk</td>
</tr>
<tr>
<td>Device Pool: G711 Pool</td>
</tr>
<tr>
<td>Common Device Configuration: &lt; None &gt;</td>
</tr>
<tr>
<td>Call Classification: Use System Default</td>
</tr>
<tr>
<td>Media Resource Group List: MRCL_MTP</td>
</tr>
<tr>
<td>Location: Hub_None</td>
</tr>
<tr>
<td>AAR Group: &lt; None &gt;</td>
</tr>
<tr>
<td>Tunneled Protocol: None</td>
</tr>
<tr>
<td>QSIG Variant: No Changes</td>
</tr>
<tr>
<td>ASN.1 ROSE OID Encoding: No Changes</td>
</tr>
<tr>
<td>Packet Capture Mode: None</td>
</tr>
<tr>
<td>Packet Capture Duration: 0</td>
</tr>
<tr>
<td>Media Termination Point Required:</td>
</tr>
<tr>
<td>Retry Video Call as Audio:</td>
</tr>
<tr>
<td>Path Replacement Support:</td>
</tr>
<tr>
<td>Transmit UTF-8 for Calling Party Name:</td>
</tr>
<tr>
<td>Transmit UTF-8 Names in QSIG APDU:</td>
</tr>
<tr>
<td>Unattended Port:</td>
</tr>
<tr>
<td>SRTP Allowed: When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will result in issues.</td>
</tr>
<tr>
<td>Consider Traffic On This Trunk Secure:</td>
</tr>
<tr>
<td>Route Class Signaling Enabled: Default</td>
</tr>
<tr>
<td>Use Trusted Relay Point: Default</td>
</tr>
<tr>
<td>PSTN Access:</td>
</tr>
<tr>
<td>Run On All Active Unified CM Nodes:</td>
</tr>
</tbody>
</table>

Figure 15: SIP Trunk to Cisco UBE
### Intercompany Media Engine (IME)

| E.164 Transformation Profile | < None > |

### MLPP and Confidential Access Level Information

| MLPP Domain | < None > |
| Confidential Access Mode | < None > |
| Confidential Access Level | < None > |

### Call Routing Information

- Remote-Party-Id
- Asserted-Identity
- Asserted-Type: Default
- SIP Privacy: Default

#### Inbound Calls

| Significant Digits | 4 |
| Connected Line ID Presentation | Default |
| Connected Name Presentation | Default |
| Calling Search Space | < None > |
| AAR Calling Search Space | < None > |

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>Yes</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>Yes</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
Figure 17: 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 _Trunk</td>
<td>Name for the trunk</td>
</tr>
<tr>
<td>Device Pool</td>
<td>G711pool</td>
<td>Default Device Pool is used for this trunk</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL_MTP</td>
<td>MRG with resources: ANN, CFB, MOH and MTP</td>
</tr>
<tr>
<td>Significant Digits</td>
<td>4</td>
<td>4 digits Extension for all CPE phones</td>
</tr>
<tr>
<td>Destination Address</td>
<td>10.80.11.20</td>
<td>IP address of the Cisco UBE Virtual LAN</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
<td>Non Secure SIP Trunk Profile</td>
<td>SIP Trunk Security Profile configured earlier</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>Intelepeer SIP Profile</td>
<td>SIP Profile configured earlier</td>
</tr>
</tbody>
</table>
Route patterns are configured as below:

- Cisco IP phone dial “9”.1+10 digits number to access PSTN via Cisco UBE
  - “9” is removed before sending to Cisco UBE
- For FAX call, Access Code “9”+ 1+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 2063 will be sent to Cisco Fax gateway

![Route Patterns List](image-url)

Figure 18: Route Patterns List
### Calling Party Transformations

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use Calling Party’s External Phone Number Mask</td>
<td>Yes</td>
</tr>
<tr>
<td>Calling Party Transform Mask</td>
<td></td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td></td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Name Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Party Number Type</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Calling Party Numbering Plan</td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

### Connected Party Transformations

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td>Default</td>
</tr>
</tbody>
</table>

### Called Party Transformations

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Discard Digits</td>
<td>PreDot</td>
</tr>
<tr>
<td>Called Party Transform Mask</td>
<td></td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td></td>
</tr>
<tr>
<td>Called Party Number Type</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Called Party Numbering Plan</td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

### ISDN Network-Specific Facilities Information Element

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

Figure 19: Route Pattern for Voice
<table>
<thead>
<tr>
<th>Pattern Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Route Pattern</strong></td>
</tr>
<tr>
<td><strong>Route Partition</strong></td>
</tr>
<tr>
<td><strong>Description</strong></td>
</tr>
<tr>
<td><strong>Numbering Plan</strong></td>
</tr>
<tr>
<td><strong>Route Filter</strong></td>
</tr>
<tr>
<td><strong>MLPP Precedence</strong></td>
</tr>
<tr>
<td><strong>Apply Call Blocking Percentage</strong></td>
</tr>
<tr>
<td><strong>Resource Priority Namespace Network Domain</strong></td>
</tr>
<tr>
<td><strong>Route Class</strong></td>
</tr>
<tr>
<td><strong>Gateway/Route List</strong></td>
</tr>
<tr>
<td><strong>Route Option</strong></td>
</tr>
<tr>
<td><strong>Call Classification</strong></td>
</tr>
<tr>
<td><strong>External Call Control Profile</strong></td>
</tr>
<tr>
<td><strong>Allow Device Override</strong></td>
</tr>
<tr>
<td><strong>Provide Outside Dial Tone</strong></td>
</tr>
<tr>
<td><strong>Allow Overlap Sending</strong></td>
</tr>
<tr>
<td><strong>Urgent Priority</strong></td>
</tr>
<tr>
<td><strong>Require Forced Authorization Code</strong></td>
</tr>
<tr>
<td><strong>Authorization Level</strong></td>
</tr>
<tr>
<td><strong>Require Client Caller ID</strong></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Calling Party Transformations</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Use Calling Party’s External Phone Number Mask</strong></td>
</tr>
<tr>
<td><strong>Calling Party Transform Mask</strong></td>
</tr>
<tr>
<td><strong>Prefix Digits (Outgoing Calls)</strong></td>
</tr>
<tr>
<td><strong>Calling Line ID Presentation</strong></td>
</tr>
<tr>
<td><strong>Calling Name Presentation</strong></td>
</tr>
<tr>
<td><strong>Calling Party Number Type</strong></td>
</tr>
<tr>
<td><strong>Calling Party Numbering Plan</strong></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Connected Party Transformations</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Connected Line ID Presentation</strong></td>
</tr>
<tr>
<td><strong>Connected Name Presentation</strong></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Called Party Transformations</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Discard Digits</strong></td>
</tr>
<tr>
<td><strong>Called Party Transform Mask</strong></td>
</tr>
<tr>
<td><strong>Prefix Digits (Outgoing Calls)</strong></td>
</tr>
<tr>
<td><strong>Called Party Number Type</strong></td>
</tr>
<tr>
<td><strong>Called Party Numbering Plan</strong></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>ISDN Network-Specific Facilities Information Element</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Network Service Protocol</strong></td>
</tr>
<tr>
<td><strong>Carrier Identification Code</strong></td>
</tr>
</tbody>
</table>

Figure 20: Route Pattern for Fax

Explanation
<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>7.@ for Voice &amp; International Calls and 8021 for Fax Call</td>
<td>Specify appropriate Route Pattern</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>Intelepeer for Route Pattern 7.@ and 8021 for SIP Trunk To Fax Gateway.</td>
<td>SIP Trunk name configured earlier</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>NANP for Route Pattern 7.@</td>
<td>North American Numbering Plan</td>
</tr>
<tr>
<td>Call Classification</td>
<td>OffNet for Route Pattern 7.@ and 8021</td>
<td>Restrict the transferring of an external call to an external device</td>
</tr>
<tr>
<td>Discard Digits</td>
<td>PreDot for Route Pattern 7.@</td>
<td>Specifies how to modify digit before they are sent to Spectrum network</td>
</tr>
</tbody>
</table>

**Acronyms**

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