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

November 15, 2017
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Introduction

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 IntelePeer network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS 16.5.1b can be used. The Cisco Unified Border Element 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-XE 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:

Network Topology

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 16.05.01b for ISR 4431/K9 Cisco Unified Border Element
- Cisco IOS-XE Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 16.05.01b, RELEASE SOFTWARE (fc1)
- 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.
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 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
ip address 10.80.11.17 255.255.255.0
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.80.11.30 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 license capacity 20
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"
session refresh
header-passing
asserted-id pai
early-offer forced
midcall-signaling passthru
privacy-policy passthru

Explanation

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>fax protocol</td>
<td>Specifies the fax protocol</td>
</tr>
<tr>
<td>asserted-id</td>
<td>Specifies the type of privacy header in the outgoing SIP</td>
</tr>
<tr>
<td></td>
<td>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 1
- 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-nte
- fax-relay sg3-to-g3
- fax rate 14400
- 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 ipv4:65.158.193.102
- voice-class codec 1
- voice-class sip conn-reuse
- 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 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 1
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 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 1
voice-class sip options-ping 60
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-nte
fax-relay sg3-to-g3
fax rate 14400
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 10Digit DID no for that extension number, 7 was stripped and the 10 digits number was send to Cisco UBE, Cisco UBE sends the full 10 digits DID under Dial Peer 20 and send to IntelePeer 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 IntelePeer Call
Configuration Example
The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously.

Active Cisco UBE

```
intelPeerCube2#sh run
Building configuration...
Current configuration : 5590 bytes
!
! Last configuration change at 08:13:43 UTC Thu Oct 5 2017 by cisco
!
version 16.5
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname intelPeerCube2
!
boot-start-marker
boot-end-marker
!
vrf definition Mgmt-intf
!
  address-family ipv4
  exit-address-family
!
  address-family ipv6
  exit-address-family
!
enable secret 5
!
no aaa new-model
!
subscriber templating
!
multilink bundle-name authenticated
!
voice service voip
  no ip address trusted authenticate
  mode border-element license capacity 20
  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"
session refresh
header-passing
asserted-id pai
early-offer forced
midcall-signaling passthru
privacy-policy passthru
!
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 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
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 0 **********
!
redundancy
mode none
application redundancy
group 1
name voice-b2bha
timers delay 30 reload 60
control GigabitEthernet0/1/0 protocol 1
data GigabitEthernet0/1/0
  track 1 shutdown
  track 2 shutdown
!
track 1 interface GigabitEthernet0/0/1 line-protocol
!
track 2 interface GigabitEthernet0/0/0 line-protocol
!
interface GigabitEthernet0/0/0
  ip address xxx.xxx.xxx.xxx 255.255.255.128
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip xxx.xxx.xxx.xxx exclusive
!
interface GigabitEthernet0/0/1
  ip address 10.80.11.17 255.255.255.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.11.30 exclusive
!
interface GigabitEthernet0/1/0
  ip address 20.0.0.2 255.255.255.252
  negotiation auto
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  shutdown
  negotiation auto
!
threat-visibility
  ip forward-protocol nd
  ip http server
  ip http authentication local
  ip http secure-server
  ip tftp source-interface GigabitEthernet0
  ip route 0.0.0.0 0.0.0.0 192.65.79.129
  ip route 10.64.0.0 255.255.0.0 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 1
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-nte
fax-relay sg3-to-g3
fax rate 14400
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 ipv4:xxx.xxx.xxx.xxx
voice-class codec 1
voice-class sip conn-reuse
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 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 1
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 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 1
voice-class sip options-ping 60
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-nte
fax-relay sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
sip-ua
!
line con 0
transport input none
stopbits 1
line aux 0
stopbits 1
line vty 0 4
logging synchronous
login local
!
no network-clock synchronization automatic
Standby Cisco UBE

intelPeerCube1#sh run
Building configuration...
Current configuration : 5617 bytes
!
! Last configuration change at 09:12:23 UTC Thu Oct 5 2017 by cisco
!
version 16.5
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname intelPeerCube1
!
boot-start-marker
boot system flash isr4300-universalk9.16.05.01b.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
enable secret 5
!
no aaa new-model
!
subscriber templating
!
multilink bundle-name authenticated
!
voice service voip
no ip address trusted authenticate
mode border-element license capacity 20
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"
session refresh
header-passing
asserted-id pai
early-offer forced
midcall-signaling passthru
privacy-policy passthru
!
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 ISR4321/K9 sn FDO19220MSQ
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level securityk9
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 0
!
redundancy
mode none
application redundancy
group 1
  name voice-b2bha
timers delay 30 reload 60
control GigabitEthernet0/1/0 protocol 1
data GigabitEthernet0/1/0
track 1 shutdown
  track 2 shutdown

track 1 interface GigabitEthernet0/0/1 line-protocol
  track 2 interface GigabitEthernet0/0/0 line-protocol

interface GigabitEthernet0/0/0
  ip address xxx.xxx.xxx.xxx 255.255.255.128
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip xxx.xxx.xxx.xxx exclusive

interface GigabitEthernet0/0/1
  ip address 10.80.11.16 255.255.255.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.11.30 exclusive

interface GigabitEthernet0/1/0
  ip address 20.0.0.1 255.255.255.0
  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 1
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-nte
fax-relay sg3-to-g3
fax rate 14400
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 ipv4:xxx.xxx.xxx.xxx
voice-class codec 1
voice-class sip conn-reuse
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 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 1
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 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 1
voice-class sip options-ping 60
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-nte
fax-relay sg3-to-g3
fax rate 14400
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 time-pnp.cisco.com
!
end
Configuring Cisco Unified Communications Manager

Cisco UCM Version

![Cisco Unified CM Administration (System Version: 11.5.1.12900-21)](image)

VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz, disk 1: 80Gbytes, 4096Mbytes RAM, Partitions aligned

Figure 8: Cisco UCM Version

Cisco Call Manager Service Parameters

**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

![Select Server and Service](image)

All parameters apply only to the current server except parameters that are in the cluster-wide group(s).

### Cisco CallManager (Active) Parameters on server Clus21Sub1--CUCM Voice/Video (Active)

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Throttling</td>
<td></td>
</tr>
<tr>
<td>Code Yellow Entry Latency</td>
<td></td>
</tr>
<tr>
<td>Code Yellow Exit Latency Calculation</td>
<td></td>
</tr>
<tr>
<td>Code Yellow Duration</td>
<td></td>
</tr>
<tr>
<td>Max Events Allowed</td>
<td></td>
</tr>
<tr>
<td>System Throttle Sample Size</td>
<td></td>
</tr>
</tbody>
</table>

Figure 9: Service Parameters
Offnet 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

![SIP Trunk Security Profile](image)

**Figure 10: 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: Device → Device Settings → SIP Profile

1. **Name**: Intelepeer SIP Profile, for example
2. **Description**: Intelepeer SIP Profile, for example

### SIP Profile Information

<table>
<thead>
<tr>
<th>Name</th>
<th>Intelepeer sip profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>Description</td>
<td>Intelepeer sip profile</td>
</tr>
</tbody>
</table>

- Default NTTP Telephony Event Payload Type: 101
- Early Offer for G.729 Calls: Disabled
- User-Agent and Server header information: Send Unified CM Version Information as User-Agent
- Version in User Agent and Server Header: Major And Minor
- Dial String Interpretation: Phone number consists of characters 0-9, *, #, and v
- Confidential Access Level Headers: Disabled
- Redirect by Application
- Disable Early Media on 180
- Outgoing T38 INVITE include audio mline
- Offer valid IP and Send/Receive mode only for T.38 Fax Relay
- Use Fully Qualified Domain Name in SIP Requests
- Assured Services SIP conformance
- Enable External QoS

### SDP Information

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

### 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>5</td>
</tr>
<tr>
<td>Retry Non-INVITE</td>
<td>10</td>
</tr>
<tr>
<td>Media Port Ranges</td>
<td>Common Port Range for Audio and Video</td>
</tr>
<tr>
<td>Start Media Port</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port</td>
<td>32766</td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Feature</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>Call Pickup URI†</td>
<td>x-cisco-serviceum-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI†</td>
<td>x-cisco-serviceum-opickup</td>
</tr>
<tr>
<td>Call Pickup Group URI†</td>
<td>x-cisco-serviceum-opickup</td>
</tr>
<tr>
<td>Meet Me Service URI†</td>
<td>x-cisco-serviceum-meetme</td>
</tr>
<tr>
<td>User Info†</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level†</td>
<td>Default</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>Off</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7960†</td>
<td>Disabled</td>
</tr>
<tr>
<td>Resource Priority Namespace</td>
<td>Default</td>
</tr>
<tr>
<td>Timer: Keep Alive Expires (seconds)†</td>
<td>120</td>
</tr>
<tr>
<td>Timer: Subscribe Expires (seconds)†</td>
<td>120</td>
</tr>
<tr>
<td>Timer: Subscribe Delta (seconds)†</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirections†</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook to First Digit Timer (milliseconds)†</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI†</td>
<td>x-cisco-serviceum-cfwdial</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI†</td>
<td>x-cisco-serviceum-abbrevdial</td>
</tr>
<tr>
<td>Conference Join Enabled</td>
<td>Default</td>
</tr>
<tr>
<td>RFC 2543 Hold</td>
<td>Default</td>
</tr>
<tr>
<td>Semi Attended Transfer</td>
<td>Default</td>
</tr>
<tr>
<td>Enable VAD</td>
<td>Default</td>
</tr>
<tr>
<td>Stutter Message Waiting</td>
<td>Default</td>
</tr>
<tr>
<td>MLPP User Authorization</td>
<td>Default</td>
</tr>
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</table>

#### Normalization Script

<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

<table>
<thead>
<tr>
<th>Caller ID DN</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller Name</td>
<td></td>
</tr>
</tbody>
</table>
Figure 13: 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>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

![SIP Trunks List](image)

**Figure 14: SIP Trunks List**

![SIP Trunk to Cisco UBE](image)

**Figure 15: SIP Trunk to Cisco UBE**
Figure 16: SIP Trunk to Cisco UBE (Cont.)
**Figure 17: SIP Trunk to Cisco UBE (Cont.)**

- **Connected Party Settings**
  - Connected Party Transformation CSS: [Select]
  - Use Device Pool Connected Party Transformation CSS

- **Outbound Calls**
  - Called Party Transformation CSS: [Select]
  - Use Device Pool Called Party Transformation CSS
  - Calling Party Transformation CSS: [Select]
  - Use Device Pool Calling Party Transformation CSS
  - **Calling Party Selection**: [Select]
    - Originate
  - Calling Line ID Presentation: [Select]
    - Default
  - Calling Name Presentation: [Select]
    - Default
  - Calling and Connected Party Info Format: [Select]
    - Deliver DN only in connected party
  - Redirecting Diversion Header Delivery - Outbound
  - Redirecting Party Transformation CSS: [Select]
    - Use Device Pool Redirecting Party Transformation CSS

- **Caller Information**
  - Caller ID DN
  - Caller Name
  - [Select] Maintain Original Caller ID DN and Caller Name in Identity Headers

- **SIP Information**
  - **Destination**
    - Destination Address: [Select]
      - 10.80.11.30
    - Destination Address IPv6
    - Destination Port: [Select]
      - 5060
  - MTP Preferred Originating Codes: [Select]
    - 711ulaw
  - BLF Presence Group: [Select]
    - Standard Presence group
  - **SIP Trunk Security Profile**: [Select]
    - Non Secure SIP Trunk Profile
  - Resolving Calling Search Space: [Select]
    - < None >
  - Out-Of-Dialog Refer Calling Search Space: [Select]
    - < None >
  - SUBSCRIBE Calling Search Space: [Select]
    - < None >
  - **SIP Profile**: [Select]
    - Interop3 sip profile
    - [View Details]
  - DTMF Signaling Method: [Select]
    - [Select]
    - No Preference

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

- **Geolocation Configuration**
  - Geolocation: [Select]
    - < None >
  - Geolocation Filter: [Select]
    - < None >
  - [Select] Send Geolocation Information
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>10.80.11.30</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>

Dial Plan

Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

- Cisco IP phone dial "7"+10 digits number to access PSTN via Cisco UBE
  - "7" is removed before sending to Cisco UBE
- For FAX call, Access Code “7”+10 digits number is used at Cisco Fax gateway
  - "7" is removed at Cisco UCM
  - The rest of the number is sent to Cisco UBE to IntelePeer network
- Incoming fax call to 8021 will be sent to Cisco Fax gateway

Find Route Patterns where Pattern begin with v

<table>
<thead>
<tr>
<th>Pattern</th>
<th>Description</th>
<th>Partition</th>
<th>Route Filter</th>
<th>Associated Device</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>7*</td>
<td>Intelepeer_Trunk</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>8021</td>
<td>ccm-faxgateway</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Figure 18: Route Patterns List
### Pattern Definition

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>7.0</td>
</tr>
<tr>
<td>Route Partition</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>NANC</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>MLPP Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Apply Call Blocking Percentage</td>
<td></td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Route Class</td>
<td>Default</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>Intelepeer Trunk</td>
</tr>
<tr>
<td>Route Option</td>
<td>Route this pattern</td>
</tr>
<tr>
<td>Call Classification</td>
<td>OffSet</td>
</tr>
<tr>
<td>External Call Control Profile</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Allow Device Override</td>
<td></td>
</tr>
<tr>
<td>Provide Outside Dial Tone</td>
<td></td>
</tr>
<tr>
<td>Allow Overlap Sending</td>
<td></td>
</tr>
<tr>
<td>Urgent Priority</td>
<td></td>
</tr>
<tr>
<td>Require Forced Authorization Code</td>
<td></td>
</tr>
<tr>
<td>Authorization Level</td>
<td>0</td>
</tr>
<tr>
<td>Require Client Matter Code</td>
<td></td>
</tr>
</tbody>
</table>

### Calling Party Transformations

- Use Calling Party's External Phone Number Mask

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<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>Field</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>Field</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>Field</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>&lt; Not Exist &gt;</td>
</tr>
</tbody>
</table>

---

Figure 19: Route Pattern for Voice
**Figure 20: Route Pattern for Fax**

<table>
<thead>
<tr>
<th>Pattern Definition</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>8021</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td>Route Filter</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td>MLLP Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Apply Call Blocking Percentage</td>
<td></td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td>Route Class</td>
<td>Default</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>gcm-faxgateway</td>
</tr>
<tr>
<td>Route Option</td>
<td></td>
</tr>
<tr>
<td>Call Classification</td>
<td>Offnet</td>
</tr>
<tr>
<td>External Call Control Profile</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td>Allow Device Override</td>
<td></td>
</tr>
<tr>
<td>Provide Outside Dial Tone</td>
<td></td>
</tr>
<tr>
<td>Allow Overlap Sending</td>
<td></td>
</tr>
<tr>
<td>Urgent Priority</td>
<td></td>
</tr>
<tr>
<td>Require Forced Authorization Code</td>
<td></td>
</tr>
<tr>
<td>Authorization Level</td>
<td>0</td>
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<tr>
<td>Require Client Matter Code</td>
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</table>

<table>
<thead>
<tr>
<th>Calling Party Transformations</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Use Calling Party's External Phone Number Mask</td>
<td></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>

<table>
<thead>
<tr>
<th>Connected Party Transformations</th>
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</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>

<table>
<thead>
<tr>
<th>Connected Party Transformations</th>
<th></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>

<table>
<thead>
<tr>
<th>Called Party Transformations</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Discard Digits</td>
<td>-- Not Selected --</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>

<table>
<thead>
<tr>
<th>ISDN Network-Specific Facilities Information Element</th>
<th></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>&lt; Not Exist &gt;</td>
</tr>
<tr>
<td>Setting</td>
<td>Value</td>
</tr>
<tr>
<td>----------------------</td>
<td>----------------------------------------------------------------------</td>
</tr>
<tr>
<td>Route Pattern</td>
<td>7.@ for Voice &amp; International Calls and 8021 for Fax Call</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>IntelePeer for Route Pattern 7.@ and 8021 for SIP Trunk To Fax Gateway.</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>NANP for Route Pattern 7.@</td>
</tr>
<tr>
<td>Call Classification</td>
<td>OffNet for Route Pattern 7.@ and 8021</td>
</tr>
<tr>
<td>Discard Digits</td>
<td>PreDot for Route Pattern 7.@</td>
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</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>

### Important Information

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS. IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.
<table>
<thead>
<tr>
<th>Corporate Headquarters</th>
<th>European Headquarters</th>
<th>Americas Headquarters</th>
<th>Asia Pacific Headquarters</th>
</tr>
</thead>
<tbody>
<tr>
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<td>Haarlerbergpark</td>
<td>170 West Tasman Drive</td>
<td>Capital Tower</td>
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<td>Haarlerbergweg 13-19</td>
<td>San Jose, CA 95134-1706</td>
<td>168 Robinson Road</td>
</tr>
<tr>
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<td>USA</td>
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<tr>
<td>Fax: 408 526-4100</td>
<td>Fax: 31 0 20 357 1100</td>
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