Spectrum Enterprise 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 15, 2017
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<td>37</td>
</tr>
</tbody>
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Introduction

Service Providers today, such as Spectrum Enterprise, 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-XE 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 Spectrum 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 Spectrum Enterprise 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 Spectrum Enterprise SIP Trunking service available in the former Charter Spectrum Business service area1 (hereafter referred to as Spectrum Enterprise (L-Charter)). The deployment model covered in this application note is CPE (Cisco UCM 11.5.1) to PSTN (Spectrum Enterprise).

- Testing was performed in accordance to Spectrum Enterprise 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 Spectrum Enterprise (L-Charter) 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 Spectrum Enterprise (L-Charter) 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:


1 Refers to the former Charter Spectrum Business service area for SIP trunking, prior to the acquisition of Time Warner Cable and BrightHouse Networks in 2016.
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)
- Adtran Total_Access_908e_2nd_Gen – Provided and managed by Spectrum Enterprise

Software Requirements

- Cisco Unified Communications Manager 11.5.1
- Cisco Unity Connection 11.5.1
- IOS-XE 16.5.1b for ISR 4321/K9 Cisco Unified Border Element
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway
- Adtran Total_Access_908e_2nd_Gen /R11.4.6.E - Provided and managed by Spectrum Enterprise
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)

Features Not Supported
- Cisco IP phones used in this test do not support blind transfer
- Fax (T.38) and G729 is not supported by Spectrum Enterprise (L-Charter)
- 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.
- For testing, 911 calls were terminated by Spectrum Enterprise
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 10.64.4.19 255.255.0.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.64.4.30 exclusive
!
interface GigabitEthernet0/0/2
  ip address 10.80.11.17 255.255.255.0
  negotiation auto
  redundancy rii 2
  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 pass-through g711ulaw
sip
rel1xx supported "rel100"
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 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 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 rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 20 voip
description Outgoing to Adtran
huntstop
destination-pattern [0-9]T
session protocol sipv2
session target ipv4:10.64.4.18:5060
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 ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 30 voip
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 “9.@” 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 Spectrum Enterprise, Caller dial 9 prefix followed by the target 1+10Digit DID no for that extension number, 9 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 200 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 9 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:
  Password:
isr4k1spectrum#sh running-config
Building configuration...
Current configuration : 5286 bytes
!
! Last configuration change at 06:17:09 UTC Tue May 30 2017 by cisco
!
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-kernel-core
!
hostname isr4k1spectrum
!
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 $1$JbZh$5I3Lr7oSNCAEckNMyGP0
!
no aaa new-model
ipv6 unicast-routing

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


crypto pki certificate chain TP-self-signed-1179880555

voice service voip
  no ip address trusted authenticate
  mode border-element
  allow-connections sip to sip
  redundancy-group 1
  fax protocol pass-through g711ulaw
  sip
    rel1xx supported "rel100"
    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 sip-profiles 100
  request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:303835\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 09584B022F540D435B02
!
redundancy
mode none
application redundancy
group 1
   name voice-b2bha
   timers delay 30
   control GigabitEthernet0/0/3 protocol 1
data GigabitEthernet0/0/3
track 1 shutdown
track 2 shutdown
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/2 line-protocol
!
interface GigabitEthernet0/0/0
ip address 10.64.4.19 255.255.0.0
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.64.4.30 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 2
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 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 10.64.1.1
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 1
description Incoming from CUCM
huntstop
session protocol sipv2
incoming called-number [0-9]T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-npe
tax-relay ecm disable
no tax-relay sg3-to-g3
tax rate disable
tax protocol pass-through g711ulaw
no vad
!
dial-peer voice 20 voip
description Outgoing to Adtran
huntstop
destination-pattern [0-9]T
session protocol sipv2
session target ipv4:10.64.4.18:5060
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-npe
tax-relay ecm disable
no tax-relay sg3-to-g3
tax rate disable
tax protocol pass-through g711ulaw
no vad
!
dial-peer voice 30 voip
description Incoming from Adtran
huntstop
session protocol sipv2
incoming called-number [37][02][30]T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 40 voip
description Outgoing to CUCM
huntstop
destination-pattern [37][02][30]T
session protocol sipv2
session target ipv4:10.80.11.2
voice-class codec 1
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
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711ulaw
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
login local
Standby Cisco UBE
User Access Verification

Username: isr4k2spectrum
Password: spectrum

Building configuration...

Current configuration: 5218 bytes

! Last configuration change at 06:20:11 UTC Tue May 30 2017 by cisco

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-kernel-core

hostname isr4k2spectrum

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 $1$YMA$OqK8fiN9WnsjC82D73OUY/

no aaa new-model

ipv6 unicast-routing

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
rel1xx supported "rel100"
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 sip-profiles 100
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:303835\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 051F0304171D5458490B

redundancy
  mode none
  application redundancy
    group 1
      name voice-b2bha
      timers delay 30
      control GigabitEthernet0/0/3 protocol 1
      data GigabitEthernet0/0/3
      track 1 shutdown
      track 2 shutdown

track 1 interface GigabitEthernet0/0/0 line-protocol

track 2 interface GigabitEthernet0/0/2 line-protocol

interface GigabitEthernet0/0/0
  ip address 10.64.4.20 255.255.0.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.64.4.30 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 2
  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 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 10.64.1.1
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 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 rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 20 voip
description Outgoing to Adtran
huntstop
destination-pattern [0-9]T
session protocol sipv2
session target ipv4:10.64.4.18:5060
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 ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 30 voip
description Incoming from Adtran
huntstop
session protocol sipv2
incoming called-number [37][02][30]T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 40 voip
description Outgoing to CUCM
huntstop
destination-pattern [37][02][30]T
session protocol sipv2
session target ipv4:10.80.11.2
voice-class codec 1
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 rate disable
fax protocol pass-through g711ulaw
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
login local
Configuring Cisco Unified Communications Manager

Cisco UCM Version

![Cisco Unified CM Administration](image)

**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

![Service Parameters](image)

**Figure 9: Service Parameters**
Offnet Calls via Spectrum Enterprise SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and the Spectrum Enterprise network and calls are routed via Cisco UBE

**SIP Trunk Security Profile**

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

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

<table>
<thead>
<tr>
<th><strong>Parameter</strong></th>
<th><strong>Value</strong></th>
<th><strong>Description</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Transport Type</td>
<td>TCP + UDP</td>
<td>SIP trunks to Spectrum Enterprise 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** = Spectrum Enterprise SIP Profile
2. **Description** = Spectrum Enterprise SIP Profile

---

### SIP Profile Information

<table>
<thead>
<tr>
<th>Name*</th>
<th>Spectrum Enterprise SIP Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>Description</td>
<td>Spectrum Enterprise SIP Profile</td>
</tr>
<tr>
<td>Default MTP Telephony Event Payload Type*</td>
<td>101</td>
</tr>
<tr>
<td>Early Offer for G.Clear Cells*</td>
<td>Disabled</td>
</tr>
<tr>
<td>User-Agent and Server header information*</td>
<td>Send Unified CM Version Information as User-Agent</td>
</tr>
<tr>
<td>Version in User Agent and Server Header*</td>
<td>Major And Minor</td>
</tr>
<tr>
<td>Dial String Interpretation*</td>
<td>Phone number consists of characters 0-9, *, #, and v</td>
</tr>
<tr>
<td>Confidential Access Level Headers*</td>
<td>Disabled</td>
</tr>
</tbody>
</table>

- Redirect by Application
- Disable Early Media on 180
- Outgoing T.38 INVITE include audio line
- 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>6</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*
Figure 12: SIP Profile (Cont.)
### 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></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

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Calling Search</th>
<th>Device Pool</th>
<th>Route Pattern</th>
<th>Partition</th>
<th>Route Group</th>
<th>Priority</th>
<th>SIP Trunk Type</th>
<th>SIP Trunk Status</th>
<th>SIP Trunk Duration</th>
<th>SIP Trunk Security Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco to FXGateways</td>
<td>SIP Trunk to FXGateways</td>
<td>GT11 Pool</td>
<td>263</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Unknown -OPTIONS</td>
<td>Ring not enabled</td>
<td>Non-Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>Spectrum_Enterprise_Trunk</td>
<td>Charter Trunk</td>
<td>GT11 Pool</td>
<td></td>
<td>263</td>
<td></td>
<td></td>
<td></td>
<td>SIP Trunk</td>
<td>Unknown -OPTIONS</td>
<td>Ring not enabled</td>
<td>Spectrum Non-Secure SIP Trunk Profile</td>
</tr>
</tbody>
</table>

**Figure 14: SIP Trunks List**

## SIP Trunk to Cisco UBE

<table>
<thead>
<tr>
<th>Device Information</th>
<th>SIP Trunk Status: Unknown</th>
<th>Duration: Unknown</th>
</tr>
</thead>
<tbody>
<tr>
<td>Product:</td>
<td>SIP Trunk Protocol: SIP</td>
<td>Trunk Service Type: None (Default)</td>
</tr>
<tr>
<td>Device Name*</td>
<td>Spectrum_Enterprise_Trunk</td>
<td>Description: Spectrum_Enterprise_Trunk</td>
</tr>
<tr>
<td>Device Pool*</td>
<td>GT11 Pool</td>
<td>Call Classification: Use System Default</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
<td>Media Resource Group List: MRCL_MTP</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Use System Default</td>
<td>Location: Hub_None</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
<td>Tunnel Protocol*</td>
</tr>
<tr>
<td>Trimmed Protocol*</td>
<td>None</td>
<td>QSIG Variant*</td>
</tr>
<tr>
<td>ASK1 ROSE OID Encoding*</td>
<td>No Changes</td>
<td>Packet Capture Mode*</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
<td>Packet Capture Duration</td>
</tr>
<tr>
<td>Media Termination Point Required</td>
<td>✓</td>
<td>Retrying Video Call as Audio</td>
</tr>
<tr>
<td>Transmit UTP-8 for Calling Party Name</td>
<td>✓</td>
<td>Path Replacement Support</td>
</tr>
<tr>
<td>Transmit UTP-5 Names in QSIG AFDU</td>
<td>✓</td>
<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 expose keys and other information.</td>
<td>✓</td>
<td>Consider Traffic on This Trunk Secure*</td>
</tr>
<tr>
<td>Route Class Signaling Enabled*</td>
<td>Default</td>
<td>Use Trusted Relay Point*</td>
</tr>
<tr>
<td>Use PSTN Access</td>
<td>✓</td>
<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

<table>
<thead>
<tr>
<th>Domain</th>
<th>&lt;None&gt;</th>
<th>&lt;None&gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Confidential Access Mode</td>
<td>&lt;None&gt;</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Confidential Access Level</td>
<td>&lt;None&gt;</td>
<td>&lt;None&gt;</td>
</tr>
</tbody>
</table>

### Call Routing Information

- `[ ]` Remote-Party-Id
- `[ ]` Asserted-Identity
- Assumed Type: `<Default>`
- SIP Privacy: `<Default>`

### Inbound Calls

<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>Inbound Call</td>
<td>Significant Digits: 4</td>
<td>Connected Line ID Presentation: <code>&lt;Default&gt;</code></td>
<td>Calling Search Space: <code>&lt;None&gt;</code></td>
<td>Use Device Pool CSS: <code>&lt;Default&gt;</code></td>
</tr>
<tr>
<td>Connected Name Presentation: <code>&lt;Default&gt;</code></td>
<td>Calling Search Space: <code>&lt;None&gt;</code></td>
<td>AAR Calling Search Space: <code>&lt;None&gt;</code></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Prefix ON: `<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.

### 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.

---

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

### 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**: Default
- **Redirecting Party Transformation CSS**: <None>
- Use Device Pool Redirecting Party Transformation CSS

### Caller Information
- **Caller ID DN**: [Input Field]
- **Caller Name**: [Input Field]
- **Maintain Original Caller ID DN and Caller Name in Identity Headers**: [Checkbox]

### SIP Information

<table>
<thead>
<tr>
<th>Destination Address</th>
<th>Destination Address IPv6</th>
<th>Destination Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>10.80.11.20</td>
<td></td>
<td>5060</td>
</tr>
</tbody>
</table>

- **MTP Preferred Originating Code**: 711 (Silent)
- **BLF Presence Group**: Standard Presence Group
- **SIP Trunk Security Profile**: Spectrum Enterprise Non Secure SIP Trunk Profile
- **Redirecting Calling Search Space**: <None>
- **Out-Of-Dialog Refer Calling Search Space**: Default
- **SUBSCRIBE Calling Search Space**: <None>
- **SIP Profile**: Spectrum Enterprise SIP Profile
- **DTMF Signaling Method**: No Preference

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

### Geolocation Configuration
- **Geolocation**: <None>
- **Geolocation Filter**: <None>
- **Send Geolocation Information**: [Checkbox]
Explanation

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Name</td>
<td>Spectrum Enterprise</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>Spectrum Enterprise Non Secure SIP Trunk Profile</td>
<td>SIP Trunk Security Profile configured earlier</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>Spectrum Enterprise 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 “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 Spectrum Enterprise network
- Incoming fax call to 2063 will be sent to Cisco Fax gateway

![Figure 18: Route Patterns List](image)
Figure 19: Route Pattern for Voice
### Pattern Definition

<table>
<thead>
<tr>
<th>Route Pattern</th>
<th>2063</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Partition</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Description</td>
<td>Route pattern to FAX Gateway</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>MPLS Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>CUCH_to_FAXgateway</td>
</tr>
<tr>
<td>Route Option</td>
<td>Route this pattern</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Offset</td>
</tr>
</tbody>
</table>

### Calling Party Transformations

<table>
<thead>
<tr>
<th>Calling Party Transform Mask</th>
<th></th>
</tr>
</thead>
<tbody>
<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>Connected Line ID Presentation</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Name Presentation</td>
<td>Default</td>
</tr>
</tbody>
</table>

### Called Party Transformations

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

---

**Figure 21: 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>9.@ for Voice &amp; International Calls, 2063 for Fax Call and 2302 for Unity Connection</td>
<td>Specify appropriate Route Pattern</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>Spectrum Enterprise for Route Pattern 9.@, 2063 for SIP Trunk To Fax Gateway and 2032 for Unity Connection</td>
<td>SIP Trunk name configured earlier</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>NANP for Route Pattern 9.@</td>
<td>North American Numbering Plan</td>
</tr>
<tr>
<td>Call Classification</td>
<td>OffNet for Route Pattern 9.@, 2063 and 2032</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 9.@</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>
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<th>Americas Headquarters</th>
<th>Asia Pacific Headquarters</th>
</tr>
</thead>
<tbody>
<tr>
<td>170 West Tasman Drive</td>
<td>Haarlerbergpark</td>
<td>170 West Tasman Drive</td>
<td>Capital Tower</td>
</tr>
<tr>
<td>San Jose, CA 95134-1706</td>
<td>Haarlerbergweg 13-19</td>
<td>San Jose, CA 95134-1706</td>
<td>168 Robinson Road</td>
</tr>
<tr>
<td>USA</td>
<td>1101 CH Amsterdam</td>
<td>USA</td>
<td>#22-01 to #29-01</td>
</tr>
<tr>
<td><a href="http://www.cisco.com">www.cisco.com</a></td>
<td>The Netherlands</td>
<td><a href="http://www.cisco.com">www.cisco.com</a></td>
<td>Singapore 068912</td>
</tr>
<tr>
<td>Tel: 408 526-4000</td>
<td>www-europe.cisco.com</td>
<td>Tel: 408 526-7660</td>
<td><a href="http://www.cisco.com">www.cisco.com</a></td>
</tr>
<tr>
<td>800 553-NETS (6387)</td>
<td>31 0 20 357 1000</td>
<td>Fax: 408 527-0883</td>
<td>Tel: +65 317 7777</td>
</tr>
<tr>
<td>Fax: 408 526-4100</td>
<td>31 0 20 357 1100</td>
<td>Fax: +65 317 7799</td>
<td></td>
</tr>
</tbody>
</table>

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