Spectrum Enterprise SIP Trunking:

Cisco Unified Communications Manager 12.0.x with Cisco Unified Border Element (CUBE 12.0.0) on ISR 4321/K9 [IOS-XE – 16.6.1] using SIP

January 22, 2018
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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 (CUBE) ISR 4321/K9 running IOS 16.x can be used. The Cisco Unified Border Element 12.0.0 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 12.0.x 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 (CUCM) 12.0.x and Cisco Unified Border Element (CUBE) on ISR 4321/K9 [IOS-XE – 16.6.1] for connectivity to Spectrum Enterprise SIP Trunking services. The deployment model covered in this application note is CPE (CUCM 12.0.x) 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 SIP Trunking Service(s) 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 SIP Trunking services.

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:


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1 Spectrum Enterprise is a division of Charter Communications following a merger with Time Warner Cable (TWC) and acquisition of Bright House Networks.

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Network Topology

Figure 1: Network Topology

Figure 2: Cisco UBE High Availability
System Components

Hardware Requirements
- Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4321/K9 router as Cisco UBE
- Cisco ISR4321/K9 (1RU) processor with 1651964K/6147K bytes of memory with 3 Gigabit Ethernet interfaces
- Processor board ID FLM1925W0WZ
- Cisco 2921 Fax Gateway
- IP phones 99X1 (SIP), 8945 (SIP) and 7965 (SCCP)
- Spectrum eSBC – Provided and Managed by Spectrum Enterprise

Software Requirements
- Cisco Unified Communications Manager 12.0.1.10000-10
- Cisco Unity Connection 12.0.1.10000-10
- IOS-XE 16.6.1 for ISR 4321/K9 Cisco Unified Border Element
- Cisco IOS Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 16.6.1, RELEASE SOFTWARE (fc4)
- Cisco IOS XE Software, Version 16.06.01
- IOS 15.1(4)M5 for Cisco 2921 Fax Gateway
- Spectrum eSBC – Provided and Managed by Spectrum Enterprise
Features

Features Supported

- Incoming and outgoing off-net calls using G711ULAW voice codecs
- Call Hold
- Call Transfer (unattended and attended)
- Call Conference
- Call Forward (all, busy, 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 (G711 Pass-through)
- T38 fax. Note T38 Fax was not available for testing with the Spectrum Enterprise L-Charter SIP trunk variant.

Features Not Supported

- Cisco IP phones used in this test do not support Blind Transfer
- G729 is not supported by Service Provider
- Cisco does not support Blind Call Transfer

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.
- T38 fax loopback call via Spectrum Enterprise (L-TWC variant) does not work,
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 10.64.5.8 255.255.0.0
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 10.64.5.10 exclusive
!
interface GigabitEthernet0/0/1
ip address 10.80.10.48 255.255.255.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.10.50 exclusive
```
Global CUBE Settings
In order to enable CUBE 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
  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 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 g711alaw

Dial Peer

The CUBE uses dial-peer to route the call based on the digit to route the call accordingly.
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 GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 20 voip
description Outgoing to Spectrum SIP trunk
destination-pattern [0-9]T
session protocol sipv2
session target ipv4:10.64.5.1 ---(Note: 10.64.5.11 for L-Charter variant)
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 sg3-to-g3
fax protocol pass-through g711ulaw
no vad
! dial-peer voice 30 voip
description Outgoing from Spectrum SIP trunk
huntstop
session protocol sipv2
incoming called-number 469573....
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 sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 40 voip
description Outgoing to CUCM from Spectrum SIP trunk
huntstop
destination-pattern 469573....
session protocol sipv2
session target ipv4:10.80.10.2
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 50 voip
description Incoming from Spectrum SIP trunk
huntstop
session protocol sipv2
incoming called-number 303547....
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 sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 60 voip
description Outgoing to CUCM from Spectrum SIP trunk
huntstop
destination-pattern 303547....
session protocol sipv2
session target ipv4:10.80.10.2
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
Call Flow

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

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

CPE callers make outbound PSTN calls by dialing a “5” 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 “5”. A “5.@” 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 5 prefix followed by the target 1+10Digit DID for that extension number, 5 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 to Spectrum network which will direct back to Cisco UBE and handled same as normal incoming PSTN call. For International calls same pattern 5 followed by 011, country code and calling number 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 the CUBE with all parameters mentioned previously

**Active Cisco UBE**
Cube3SpecCube1#show run

```
version 16.6
service config
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 Cube3SpecCube1
!
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 $1$GDSM$JeD6RWIHx4hcko/eAmvcEXc.
```
no aaa new-model

subscriber templating

multilink bundle-name authenticated

crypto pki trustpoint TP-self-signed-1582728230
    enrollment selfsigned
    subject-name cn=IOS Self-Signed Certificate-1582728230
    revocation-check none
    rsakeypair TP-self-signed-1582728230

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
        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 g711alaw

license udi pid ISR4321/K9 sn FDO19220XSQ
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 7 15060E0732FB3175783D

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 10.64.5.8 255.255.0.0
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 10.64.5.10 exclusive

interface GigabitEthernet0/0/1
  ip address 10.80.10.48 255.255.255.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.10.50 exclusive

interface GigabitEthernet0/1/0
  ip address 1.1.1.1 255.255.255.0
  negotiation auto

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

ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet0/0/1
ip route 0.0.0.0 0.0.0.0 10.64.5.1
ip route 10.80.19.0 255.255.255.0 10.80.10.1
ip route 172.16.24.0 255.255.248.0 10.80.10.1

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 GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 20 voip
description Outgoing to Spectrum SIP trunk
destination-pattern [0-9]T
session protocol sipv2
session target ipv4:10.64.5.1 (Change to 10.64.5.11 for SIP trunk L-Charter variant)
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-npe
fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 30 voip
description Outgoing from Spectrum
huntstop
session protocol sipv2
incoming called-number 469573....
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-npe
fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 40 voip
description Outgoing to CUCM
huntstop
destination-pattern 469573....
session protocol sipv2
session target ipv4:10.80.10.2
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 50 voip
description Incoming from Spectrum SIP trunk
huntstop
session protocol sipv2
incoming called-number 303547....
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 sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 60 voip
description Outgoing to CUCM from Spectrum SIP trunk
huntstop
destination-pattern 303547....
session protocol sipv2
session target ipv4:10.80.10.2
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
line con 0
   exec-timeout 0 0
   transport input none
   stopbits 1
line aux 0
   stopbits 1
line vty 0 4
   login local
   transport input telnet
!
ntp server 34.202.215.187
ntp server pool.ntp.org
!
end
Standby Cisco UBE

Cube4SpecCube2#show run

version 16.6
service config
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 Cube4SpecCube2
!
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 $1$JlS4$FlJ1pipGXrkwKV3i/1c5.
!
no aaa new-model
subscriber templating
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-1017057749
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-1017057749
  revocation-check none
  rsakeypair TP-self-signed-1017057749
!
crypto pki certificate chain TP-self-signed-1017057749
!
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
    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 g711alaw
!
!
license udi pid ISR4321/K9 sn FDO19220XQ9
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 071B244778X80354471C
!
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 10.64.5.9 255.255.0.0
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 10.64.5.10 exclusive

interface GigabitEthernet0/0/1
  ip address 10.80.10.49 255.255.255.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.10.50 exclusive

interface GigabitEthernet0/1/0
  ip address 1.1.1.2 255.255.255.0
  negotiation auto

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

ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet0/0/1
ip route 0.0.0.0 0.0.0.0 10.64.5.1
ip route 10.80.19.0 255.255.255.0 10.80.10.1
ip route 172.16.24.0 255.255.248.0 10.80.10.1
!
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 GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 20 voip
description Outgoing to Spectrum SIP trunk
destination-pattern [0-9]T
session protocol sipv2
session target ipv4:10.64.5.1 (Change to 10.64.5.11 for SIP trunk L-Charter variant)
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 sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 30 voip
description Outgoing from Spectrum
huntstop
session protocol sipv2
incoming called-number 469573....
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 sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 40 voip
description Outgoing to CUCM
huntstop
destination-pattern 469573....
session protocol sipv2
session target ipv4:10.80.10.2
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 50 voip
description Incoming from Spectrum SIP trunk
huntstop
session protocol sipv2
incoming called-number 303547....
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 sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 60 voip
description Outgoing to CUCM from Spectrum SIP trunk
huntstop
destination-pattern 303547....
session protocol sipv2
session target ipv4:10.80.10.2
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad!
line con 0
  exec-timeout 0 0
  transport input none
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  login local
  transport input telnet
!
ntp server 34.208.249.133
!
end
Configuring Cisco Unified Communications Manager

CUCM Version

![CUCM Version](image)

Figure 8: CUCM Version

Cisco Call Manager Service Parameters

**Navigation:** System → Service Parameters

1. Select **Server**: clus20pubSub1--CUCM Voice/Video (Active)
2. Select **Service**: Cisco Call Manager (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 CUCM and the Spectrum Enterprise network and calls are routed via the CUBE

**SIP Trunk Security Profile**

Navigation: System → Security → SIP Trunk Security Profile

1. Name*: Spectrum SIP Trunk Security Profile
2. Description: Non Secure Profile for Spectrum

![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 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>

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SIP Profile Configuration
SIP Profile will be later associated with the SIP trunk

Navigation: Device → Device Settings → SIP Profile

1. Name*: Spectrum Standard SIP Profile
2. Description: Default SIP Profile

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 RelXX Options</td>
<td>Send PRACK if 1xx Contains SDP</td>
<td>Enable Provisional Acknowledgements(Reliable 100 messages)</td>
</tr>
<tr>
<td>Early Offer support for voice and video calls</td>
<td>Best Effort (no MTP inserted)</td>
<td>Enable early media call</td>
</tr>
<tr>
<td>Enable OPTIONS Ping to monitor destination status</td>
<td>Checked</td>
<td>SIP OPTIONS Ping enabled between CUCM and CUBE</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 CUBE

Navigation: Device → Trunk

**Figure 14: SIP Trunks List**

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

The image shows a SIP Information window with the following details:

- **Destination Address**: 10.80.10.50
- **Destination Address IPv6**: Not specified
- **SIP Trunk Security Profile**: Spectrum SIP Trunk Security Profile
- **SIP Profile**: Spectrum Standard SIP Profile
- **Normalization Script**: Spectrum
- **Recording Information**: None
- **Geolocation Configuration**: None

The screen also includes options for normalization parameters and trace settings.
Figure 18: SIP Trunk to Fax Gateway
Figure 19: SIP Trunk to Fax gateway Cont.
Figure 20: SIP trunk to Cisco Unity Connection
Figure 21: SIP Trunk to Unity Connection Cont.
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Name</td>
<td>Spectrum_SIP_Trunk for SIP Trunk to CUBE; SpectrumFax for SIP trunk to Fax Gateway; Unity for SIP trunk to Cisco Unity Connection</td>
<td>Name for the trunk</td>
</tr>
<tr>
<td>Device Pool</td>
<td>G711 Preferred</td>
<td>G711 Device Pool is used for this trunk</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL</td>
<td>MRG with resources: ANN, CFB, MOH and MTP</td>
</tr>
<tr>
<td>Significant Digits</td>
<td>4 for SIP Trunk to CUBE; All for SIP trunk to Fax gateway and Cisco Unity Connection</td>
<td>4 digits Extension for all CPE phones/Fax/Unity Connection Pilot number</td>
</tr>
<tr>
<td>Destination Address</td>
<td>10.80.10.50/10.80.19.7/10.80.10.5 for SIP trunk to CUBE/Fax Gateway/Cisco Unity Connection</td>
<td>IP address of the Cisco UBE Virtual LAN/Fax Gateway/Cisco Unity Connection</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>
<tr>
<td>Normalization Script</td>
<td>Spectrum for SIP trunk to CUBE only</td>
<td>Convert 4 digits EXT to 11 digits DID for Diversion header</td>
</tr>
<tr>
<td>Diversion-Mask</td>
<td>1469573XXXX/1303547XXXX</td>
<td>Used in the Manipulation Script for Timer Warner Cable/L-Charter</td>
</tr>
</tbody>
</table>
Normalization Script

A SIP Normalization Script is used to convert SIP Diversion Headers from 4-digit EXT to the full 11-digit telephone number, this is required for call redirecting over Spectrum SIP network. Navigate to Device>Device Settings>SIP Normalization Script

![SIP Normalization Script Configuration](image)

**Figure 22: SIP Normalization Script**

SIP Normalization Script (Text):

\[
M = {}
\]

\[
local \ mask = \text{scriptParameters.getValue("Diversion-Mask")}
\]

\[
-- \text{handle the mask of the diversion header for non-911 calls}
\]

\[
\text{function M.outbound_INVITE(message)}
\]

\[
\text{if mask then}
\]

\[
\text{message:applyNumberMask("Diversion", mask)}
\]

\[
\text{end}
\]

\[
\text{return } M
\]
Dial Plan

Route Pattern Configuration

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

Route patterns are configured as below:

- Cisco IP phone dial “5”+1+10 digits number to access PSTN via Cisco UBE
  - “5” is removed before sending to Cisco UBE
- For FAX call, Access Code “5”+ 1+10 digits number is used at Cisco Fax gateway
  - “5” is removed at Cisco UCM
  - The rest of the number is sent to Cisco UBE to Spectrum Enterprise network
- Incoming fax call to 898X (for L-TWC) or 126X (for L-Charter) will be sent to Cisco Fax gateway
- Call to Voice Mail 2302 will be send to Cisco Unity Connection

![Route Patterns List](image)

*Figure 23: Route Patterns List*
**Figure 24: Route Pattern for Voice**
Figure 25: Route Pattern for Fax (L-TWC)
Figure 26: Route Pattern for Fax (L-Charter)
Figure 27: Route Pattern to Cisco Unity Connection
**Explanation**

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>5.@ for Voice &amp; International Calls, 898X for Fax Call and 2302 for Unity Connection</td>
<td>Specify appropriate Route Pattern</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>Spectrum _SIP_Trunk for Route Pattern 5.@, SpectrumFax for Pattern 898X and 126X; and Cisco Unity Connection for Pattern 2302</td>
<td>SIP Trunk name configured earlier</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>NANP for Route Pattern 5.@</td>
<td>North American Numbering Plan</td>
</tr>
<tr>
<td>Discard Digits</td>
<td>PreDot for Route Pattern 5.@</td>
<td>Specifies how to modify digit before they are sent to Spectrum network</td>
</tr>
</tbody>
</table>

**Fax Gateway Example Configuration**

```
FAX-GATEWAY#show run

version 15.4

service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service sequence-numbers
!
hostname FAX-GATEWAY
!
boot-start-marker
boot-end-marker
!
aqm-register-fnf
!
logging queue-limit 1000000000
logging buffered 30000000
logging rate-limit 10000
no logging console
```
enable secret 4 iR3uUX3Bo6oYbT6ajhFwJx39FR4g.1QCmm7yYduKGZI
!
no aaa new-model
clock timezone CST -6 0
clock summer-time CDT recurring
!
ip domain name lab.tekvizion.com
ip name-server 10.64.1.3
ip cef
no ipv6 cef
multilink bundle-name authenticated
!
stcapp register capability 0/0/0 both
stcapp ccm-group 15
!
stcapp feature access-code
!
stcapp feature speed-dial
!
stcapp supplementary-services
  port 0/0/0
    fallback-dn 4037
!
cts logging verbose
!
crypto pki trustpoint TP-self-signed-1120430079
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-1120430079
  revocation-check none
rsakeypair TP-self-signed-1120430079
!
!
crypto pki certificate chain TP-self-signed-1120430079
certificate self-signed 01
3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030 31312F30 2D060355 04031326 494F532D 53656C66 2D5369676E65642D 436574696669636174652D 31313203 230343305A17 0D3230303130303030305A3031312F302D06035504031326494F532D53656C662D5369676E65642D 436574696669636174652D 31323034 33303037 3930819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281 8100D201 333D5645 360F1451 B7BC6B7F F8C6715 F1658731 5ABE4743 6B6EA3B0 1C7D1D76 17B5BD08 BA6894F9 80F790EA 8C46036E 7FDE5827 3BF9B45F 7B71D959 B6955C33 A47811B4 7DA425F8 81EC15A8 1AFC1E03 3690A9E4 DF1C9AF1 974151B1 15742950 E0ABEBBB DDE57C06 97D3C3A4 4205ADE2 36CB165E 47967DC4 8F41F25B 17430203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603 551D2304 18301680 14107F60 ACB1F013 843D556C F6C614CF 491E69DA E5301D06 03551D0E 04160414 107F60AC B1F01384 3D556CF6 C614CF49 1E69DAE5 300D0609 2A864886 F70D0101 05050003 81810029 2333B82C 9C91341D 6921E674 E3657F3A 840DB5A3 54C20F8B E004BE5E 741B5B53 3978D629 DBE53B11 51C2F563 9A82DCC4 37D269C6 B8A5F80D 969C90E1 DA963C31 D49B3F9C B7A1E484 C792A0E7 1BEA5D90 0A501640 BD4045A8 649E269C B4891C0D FF317730 DD446398 2E0130DB 6DCD39EA 8C68EC80 308E03E1 80C6C503 CD4A4 quit
voice-card 0
!
voice service voip
no ip address trusted authenticate
allow-connections sip to sip
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol pass-through g711ulaw
sip
rel1xx disable
header-passing
error-passthru
midcall-signaling passthru
!
license udi pid CISCO2901/K9 sn FTX174081TS
hw-module pvdm 0/0
!
username cisco privilege 15 secret 5 $1$vLg0$o8vBEuXW0zO4WrX6xkgOY/
!
redundancy
!
translation-rule 898
  Rule 1 8985 4695738985
  Rule 2 8986 4695738986
!
translation-rule 126
  Rule 1 1267 3035471267
  Rule 2 1268 3035471268
!
interface Embedded-Service-Engine0/0
  no ip address
  shutdown
!
interface GigabitEthernet0/0
description FAX Gateway for IP Toll Free
ip address 10.80.19.7 255.255.255.0
duplex auto
speed auto
!
interface GigabitEthernet0/1
no ip address
shutdown
duplex auto
speed auto
!
ip forward-protocol nd
!
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
ip route 0.0.0.0 0.0.0.0 10.80.19.1
!
!
control-plane
!
!
voice-port 0/0/0
 no echo-cancel enable
 no vad
cptone IN
caller-id enable

voice-port 0/0/1
no echo-cancel enable
no vad

cptone IN
caller-id enable

no mgcp timer receive-rtcp
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

do ccm-manager fax protocol cisco


dial-peer voice 203 voip
description Gateway to CUCM
service session
destination-pattern 5T
translate-outgoing calling 898 (898 for L-TWC and 126 for L-Charter)
session protocol sipv2
session target ipv4:10.80.10.2
session transport udp
codec g711ulaw
fax rate 14400
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 204 voip
description CUCM to Gateway
service session
session protocol sipv2
session transport udp
incoming called-number 898. (898. for L-TWC and 126. for L-Charter)
codec g711ulaw
fax rate 14400
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 8985 pots
service session
destination-pattern 8985
no digit-strip
port 0/0/0
forward-digits 0
!
dial-peer voice 1268 pots
service session
destination-pattern 1268
no digit-strip
port 0/0/1
forward-digits 0
!
! gateway
timer receive-rtp 1200
!
sip-ua
no timers hold
!
!
!
gatekeeper
shutdown
!
!
!
telephony-service
max-conferences 8 gain -6
transfer-system full-consult
!
!
line con 0
exec-timeout 0 0
login local
line aux 0
line vty 0 4
session-timeout 180
exec-timeout 0 0
login local
transport input telnet ssh
line vty 5 15
privilege level 15
login local

transport input telnet ssh
!
scheduler allocate 20000 1000
!
End

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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<td>Haarlerbergpark</td>
<td>170 West Tasman Drive</td>
<td>Capital Tower</td>
</tr>
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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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<td>Fax: 31 0 20 357 1100</td>
<td></td>
<td></td>
</tr>
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