Spectrum Enterprise SIP Trunking:

Cisco Unified Communications Manager 11.5.x with Cisco Unified Border Element (CUBE 11.5.2) on ISR 4321/K9 [IOS-XE 16.3.2] using SIP

October 24, 2017
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Introduction

Service Providers today, such as Spectrum Enterprise\(^1\), 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 16.x provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.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) 11.5.x and Cisco Unified Border Element (CUBE) on ISR 4321/K9 [IOS - 16.x] for connectivity to Spectrum Enterprise SIP Trunking services. The deployment model covered in this application note is CPE (CUCM 11.5.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 and acquisition of Bright House Networks.
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 2851 Fax Gateway
- IP phones 9971 (SIP) and 7975 (SCCP)
- Spectrum eSBC – Provided and Managed by Spectrum Enterprise

Software Requirements
- Cisco Unified Communications Manager 11.5.x
- Cisco Unity Connection 11.5.x
- IOS 16.3.2 for ISR 4321/K9 Cisco Unified Border Element
- Cisco IOS Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 16.3.2, RELEASE SOFTWARE (fc4)
- Cisco IOS XE Software, Version 16.03.02
- IOS 15.1(4)M5 for Cisco 2851 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)

Features Not Supported
- Cisco IP phones used in this test do not support Blind Transfer
- G729 is not supported by Service Provider
- T38 Fax was not available for testing during the test cycle
- 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.
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 10.249.248.110 255.255.0.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.249.248.98 exclusive
!
interface GigabitEthernet0/0/1
  description LAN
  ip address 10.80.18.12 255.255.255.0
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 10.80.18.10 exclusive
Global CUBE Settings

In order to enable CUBE IP2IP gateway functionality, enter the following:

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

Explanation

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>fax protocol</td>
<td>Specifies the fax protocol</td>
</tr>
<tr>
<td>asserted-id</td>
<td>Specifies the type of privacy header in the outgoing SIP requests and response messages</td>
</tr>
<tr>
<td>early-offer forced</td>
<td>Enables SIP Delayed-Offer to Early-Offer globally</td>
</tr>
<tr>
<td>midcall-signaling passthru</td>
<td>Passes SIP messages from one IP leg to another IP leg</td>
</tr>
</tbody>
</table>
**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**
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 early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 20 voip
description Outgoing to Spectrum
huntstop
destination-pattern [0-9]T
session protocol sipv2
session target ipv4:10.249.248.100
dial-peer voice 30 voip
description Incoming from Spectrum
huntstop
session protocol sipv2
incoming called-number 469T
voice-class codec 1
voice-class sip bind control 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

dtmf-relay rtp-nte
fax-relay ecm disable
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
description Outgoing to CUCM
huntstop
destination-pattern 469T
session protocol sipv2
session target ipv4:10.80.18.3
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 ecm disable
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711ulaw
no vad
Call Flow

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

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

CPE callers make outbound PSTN calls by dialing a “9” prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, Cisco fax Gateway sends to Cisco UCM the DID with leading access code “9”. A “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 20 and send to Spectrum network which will direct back to Cisco UBE and handled same as normal incoming PSTN call. For International calls same pattern 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 the CUBE with all parameters mentioned previously

**Active Cisco UBE**
User Access Verification
Username: cisco
Password:
CUBE2#sh run
Building configuration...

version 16.3
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname CUBE2
!
boot-start-marker
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no aaa new-model
no ip domain lookup

subscriber templating

multilink bundle-name authenticated

crypto pki trustpoint TP-self-signed-2548443246
   enrollment selfsigned
   subject-name cn=IOS-Self-Signed-Certificate-2548443246
   revocation-check none
   rsakeypair TP-self-signed-2548443246

---

crypto pki certificate chain TP-self-signed-2548443246
   certificate self-signed 01
   30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
   31312F30 2D060355 04031326 494F532D 536566C66 2D536967 6E65642D 43657274
   69666666 6176452D 32335432 34343333 3436301E 170D3137 30363237 32303334
   34345A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
   4F532D53 656C662D 5369676E 65642D43 65727469 66696666 74652D32 35343834
   34333234 36308201 22300D06 092A8648 86F70D01 01005500 0382010F 00308201
   0A028201 0100C689 F505FDD45 ED38A673 62CE87C5 63533BBB 961BA455 815FF15F
   B176F1FC CFE7B1CB B7339913 CE6C099B 81BFC015 E07BBF68 28447C7F 1110A35D
   9CB584B5 ABF264EC 2E3B759F D011A2C4 F15AE9D0 4B0639B7 F10F9BC5 6FC8DCE3
   30A95FC9 4D2428E5 D87B4A36 02F264C9 9EEC37A2 8234BE96 64AC5269 AB37F8A4
   CDFCF43B 9DAA4608 45A68382 210032EA ECE3DD81 CDAC3CAC 33F04215 7F9AF16
   DA46BAAD 4AE6D7A2 50647AB6 243B37F1 3ED7660D D9B83510 9E68B935 AA9FC6B0
   9C4FD9FA F77A3B037 9802FE33 16F7CC59 A0FA9835 3755D690 693A1156 156641B3
   F6271738 3AC8A0E9 389A49E3 FA0F04CE 3C690B25 5908ED3B 4AD33C83 218704D6
quit

voice service voip

no ip address trusted authenticate

address-hiding

mode border-element license capacity 20

allow-connections sip to sip

redundancy-group 1

fax protocol pass-through g711ulaw

sip

rel1xx supported "rel100"

session refresh

header-passing

asserted-id pai

privacy pstn

conn-reuse

early-offer forced
midcall-signaling passthru
privacy-policy passthru
g729 annexb-all
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
!
voice class sip-profiles 1
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:469573\1\2>"
!
license udi pid ISR4321/K9 sn FDO19220MW3
license boot level appxk9
license boot level uck9
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 0 cisco
!
redundancy
mode none
application redundancy
group 1
name voice-b2bha
priority 100 failover threshold 75
timers delay 30 reload 60
control GigabitEthernet0/1/0 protocol 1
data GigabitEthernet0/1/0
track 1 shutdown
track 2 shutdown
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
  ip address 10.249.248.110 255.255.0.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.249.248.98 exclusive
!
interface GigabitEthernet0/0/1
  description LAN
  ip address 10.80.18.12 255.255.255.0
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/1/0
  ip address 10.89.20.10 255.255.255.0
  negotiation auto
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
interface Vlan1
no ip address
shutdown
!
ip forward-protocol nd
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 10.249.248.97
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.80.0.0 255.255.0.0 10.80.18.1
ip route 172.16.31.0 255.255.255.0 10.80.18.1
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 10 voip
description Incoming from CUCM
huntstop

session protocol sipv2

incoming called-number [0-9]T

to voice-class codec 1

voice-class sip early-offer forced

voice-class sip bind control source-interface GigabitEthernet0/0/1

voice-class sip bind media source-interface GigabitEthernet0/0/1

dtmf-relay rtp-nce

fax-relay ecm disable

no fax-relay sg3-to-g3

fax rate disable

fax protocol pass-through g711ulaw

no vad

!

dial-peer voice 20 voip

description Outgoing to Spectrum

huntstop

destination-pattern [0-9]T

session protocol sipv2

session target ipv4:10.249.248.100

voice-class codec 1

voice-class sip options-ping 60

voice-class sip early-offer forced

voice-class sip profiles 1

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

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

dtmf-relay rtp-nce

fax-relay ecm disable

no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 30 voip
description Incoming from Spectrum
huntstop
session protocol sipv2
incoming called-number 469T
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 469T
session protocol sipv2
session target ipv4:10.80.18.3
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 ecm disable
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
line con 0
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 0 0
  privilege level 15
  password cisco
  logging synchronous
  login local
  transport input telnet ssh
!
end
CUBE2#

**Standby Cisco UBE**

User Access Verification
Username: cisco
Password:

CUBE1#sh run
Building configuration...

```
version 16.3
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname CUBE1
!
boot-start-marker
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no aaa new-model
!
```
no ip domain lookup
!
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-1270583006
    enrollment selfsigned
    subject-name cn=IOS-Self-Signed-Certificate-1270583006
    revocation-check none
    rsakeypair TP-self-signed-1270583006
!
crypto pki certificate chain TP-self-signed-1270583006
    certificate self-signed 01
    30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
    31312F30 2D060355 04031326 49F4F532D 53656C66 2D536967 6E65642D 43657274
    69666966 61746562 31323730 35383330 30363031E 170D3137 30363237 32303539
    31385A17 0D323030 31303130 305A3031 312F302D 06035504 03132649 1270583006
    4F532D53 656C662D 53656C66 2D536967 6E65642D 43657274 69666966 61746562 3132373035
    38333030 36308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
    0A028201 01009357 89CDDFF2 740408D8 9D185E93 E6426DED 9FDA526D E01BE0B9
    2589A47F 06FEDD14 6349C1F3 55C8EA44 4D57D07E 20632E1D C43930BD A0F3EF9F
    A65D105B 9CEB70FC 99865700 6E282D45 41D865DB D57057A4 D5A8B2D4 FFBEB680
    9A579600 F421DB9A C644B2B7 AF57547B C704EE31 3EB85920 C0BDD17C DD438273
    194EE1F4 868B3722 F6879823 96CAE06A 2F3F52E3 E4EC16F2 180ED78B C85F472C
    9C9850DD 0051069A 89FE6636 42EEA22 BF39D1D3 039973CC 4F1D7B41 4924767C
    EFBD18A0 5B379A2D 19A7F46E A50A5C36 7B7A1796 F4E50235 92BABE2D 29CDC9E7
    96498FCD 84505728 2E2371AD 2AF67552 77347558 1AE693D0 0990E608 A486C23
    AA1217B0 D5470203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
301F0603 551D2304 18301680 14871077 5EF40EA7 CDF094B7 98F70346 512C8849
73301D06 03551D0E 04160414 8710775E F40EA7CD F094B798 F7034651 2C884973
300D0609 2A864886 F70D0101 05050003 82010100 207D6A92 9E325E56 827134D3
3B537BD5 F4E10F9B 290B3836 77AA1E5A 9B7F945E 025603F8 D8BF4CF0 A85B5219
4CB96446 76FCC5BD 31744F18 BA94CFC2 86DDD92 EBD72FE3 8470D1A4 05A2BF95
46DCA505 E46C0D4C FFAA5B35 A2B142AF 658DD7D7 98D9E168 771E3AAD 2CA4BCF9
C5F835F3 92FC4F87 DB4E4ACA 33038758 1AA0F13F 317C2A98 375D7E0B 5C5F85C7
D296FA05 7750D490 7B75FED0 136DEE87 5818B934 470058BF 16341380 AD22A121
71A7B85B 4AE79B16 70215C44 D9137A13 101F5E3D B949745B 2FE99F42 671D5AC7
E9BF5D6B EB46D767 8ABEB24D 1649FDB0 7C3866E2 24B2ADB6 11B35200 4EEC2D92
E7845239 534D484B DF4D44C6 7C05BFAA D459BD31

`quit`

voice service voip

no ip address trusted authenticate

address-hiding

mode border-element license capacity 20

allow-connections sip to sip

redundancy-group 1

fax protocol pass-through g711ulaw

sip

rel1xx supported "rel100"

session refresh

header-passing

asserted-id pai

privacy pstn

conn-reuse

early-offer forced

midcall-signaling passthru
privacy-policy passthru
g729 annexb-all

! 
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g729r8
!
voice class sip-profiles 1
  request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:469573\1@\2>"

!
license udi pid ISR4321/K9 sn FDO19220MQ8
license boot level appxk9
license boot level uck9
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 0 cisco
!
redundancy
  mode none
  application redundancy
    group 1
      name voice-b2bha
      priority 100 failover threshold 75
      timers delay 30 reload 60
      control GigabitEthernet0/1/0 protocol 1
      data GigabitEthernet0/1/0
track 1 shutdown
track 2 shutdown

! vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
  ip address 10.249.248.109 255.255.0.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.249.248.98 exclusive
!
interface GigabitEthernet0/0/1
  description LAN
  ip address 10.80.18.11 255.255.255.0
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/1/0
  ip address 10.89.20.11 255.255.255.0
  negotiation auto
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
shutdown
negotiation auto
!
interface Vlan1
no ip address
shutdown
!
ip forward-protocol nd
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 10.249.248.97
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.80.0.0 255.255.0.0 10.80.18.1
ip route 172.16.31.0 255.255.255.0 10.80.18.1
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 10 voip
description Incoming from CUCM
huntstop
session protocol sipv2
incoming called-number [0-9]T
voice-class codec 1
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nre
fax-relay ecm disable
no fax-relay sg3-to-g3
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 20 voip
description Outgoing to Spectrum
huntstop
destination-pattern [0-9]T
session protocol sipv2
session target ipv4:10.249.248.100
voice-class codec 1
voice-class sip options-ping 60
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nre
fax-relay ecm disable
no fax-relay sg3-to-g3
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 30 voip
description Incoming from Spectrum
huntstop
session protocol sipv2
incoming called-number 469T
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 469T
session protocol sipv2
session target ipv4:10.80.18.3
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 ecm disable
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 889 voip
!
line con 0
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 0 0
  privilege level 15
  password cisco
  logging synchronous
  login local
  transport input telnet ssh
!
end
Configuring Cisco Unified Communications Manager

CUCM Version

![Cisco Unified CM Administration](image)

**Figure 8: CUCM Version**

**Cisco Call Manager Service Parameters**

**Navigation:** System → Service Parameters

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

![Select Server and Service](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 Enterprise 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 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 SIP Profile
2. **Description**: Spectrum SIP Profile

![SIP Profile Configuration](image)

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>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 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.)
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 19: Route Pattern for Voice
### Pattern Definition

**Route Pattern**: 2302
**Route Partition**: < None >
**Description**: Route pattern to unity

**Numbering Plan**
**Route Filter**
**MLPP Precedence**: Default
**Apply Call Blocking Percentage**
**Resource Priority Namespace Network Domain**
**Route Class**: Default

**Gateway/Route List**: UnityConnection
**Route Option**: Route this pattern

**Call Classification**: OffNet
**External Call Control Profile**: < None >

**Require Forced Authorization Code**
**Authorization Level**: 0
**Require Client Matter Code**

### Calling Party Transformations

- **Use Calling Party's External Phone Number Mask**
- **Calling Party Transform Mask**: 3961
- **Prefix Digits (Outgoing Calls)**
- **Calling Line ID Presentation**: Default
- **Calling Name Presentation**: Default
- **Calling Party Number Type**: Cisco CallManager
- **Calling Party Numbering Plan**: Cisco CallManager

### Connected Party Transformations

- **Connected Line ID Presentation**: Default
- **Connected Name Presentation**: Default

### Called Party Transformations

- **Discard Digits**: < None >
- **Called Party Transform Mask**
- **Prefix Digits (Outgoing Calls)**
- **Called Party Number Type**: Cisco CallManager
- **Called Party Numbering Plan**: Cisco CallManager

### ISDN Network-Specific Facilities Information Element

<table>
<thead>
<tr>
<th>Network Service Protocol</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>-- Not Selected --</td>
<td>&lt; Not Exist &gt;</td>
<td></td>
</tr>
</tbody>
</table>

---

Figure 20: Route Pattern for Voice (Cont.)
### Pattern Definition

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>6984</td>
</tr>
<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>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>custom-faxgateway</td>
</tr>
<tr>
<td>Route Option</td>
<td>Route this pattern</td>
</tr>
<tr>
<td>Call Classification</td>
<td>OffNet</td>
</tr>
<tr>
<td>External Call Control Profile</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Allow Device Override</td>
<td>Provide Outside Dial Tone</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

<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 Number 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>&lt; None &gt;</td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td>Default</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></td>
</tr>
<tr>
<td>Network Service</td>
<td>Service Parameter Name</td>
</tr>
</tbody>
</table>
### 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, 8984 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.@, 8984 for CUCM- Fax Gateway and 2302 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.@, 8984 and 2302</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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<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>1101 CH Amsterdam</td>
<td>USA</td>
<td>#22-01 to #29-01</td>
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<td><a href="http://www.cisco.com">www.cisco.com</a></td>
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<td><a href="http://www.cisco.com">www.cisco.com</a></td>
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<tr>
<td>800 553-NETS (6387)</td>
<td>Tel: 31 0 20 357 1000</td>
<td>Fax: 408 527-0883</td>
<td>Tel: +65 317 7777</td>
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<tr>
<td>Fax: 408 526-4100</td>
<td>Fax: 31 0 20 357 1100</td>
<td></td>
<td>Fax: +65 317 7799</td>
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</tbody>
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