



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**

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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:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/dialplan.html

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

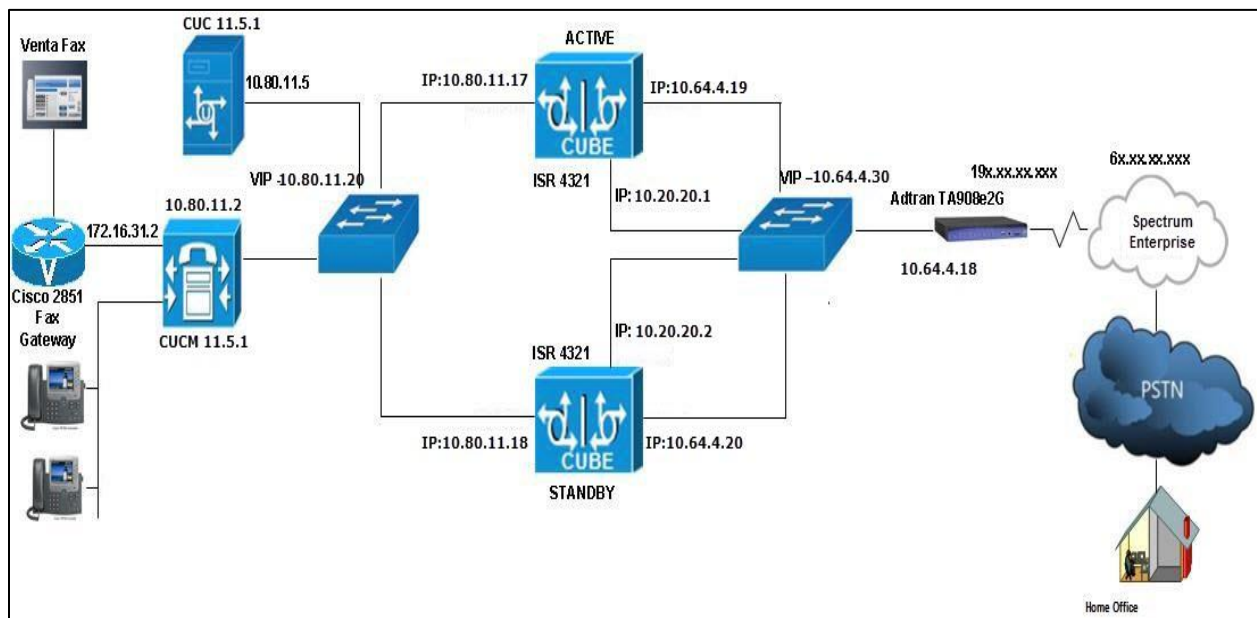


Figure 1: Network Topology

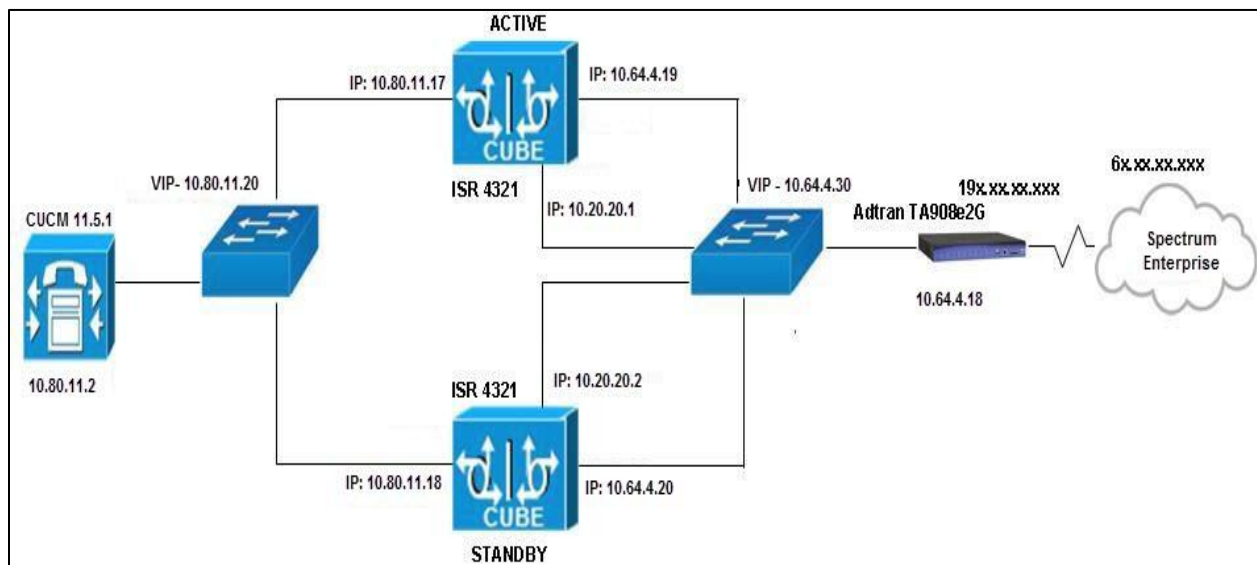


Figure 2: Cisco UBE High Availability



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

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
fax protocol	Specifies the fax protocol
asserted-id	Specifies the type of privacy header in the outgoing SIP requests and response messages
early-offer forced	Enables SIP Delayed-Offer to Early-Offer globally
midcall-signaling passthru	Passes SIP messages from one IP leg to another IP leg

Codecs

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

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```
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
```



```
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
!
```

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.

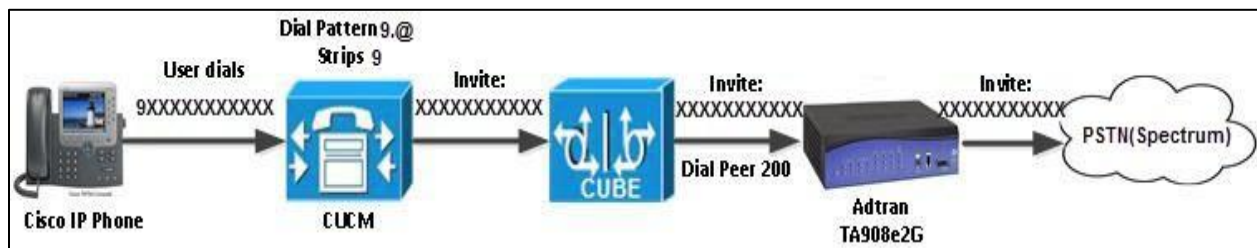


Figure 3: Outbound Voice Call

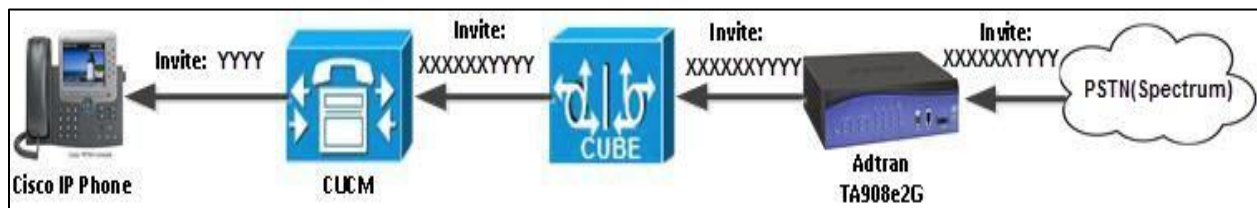


Figure 4: Inbound Voice Call

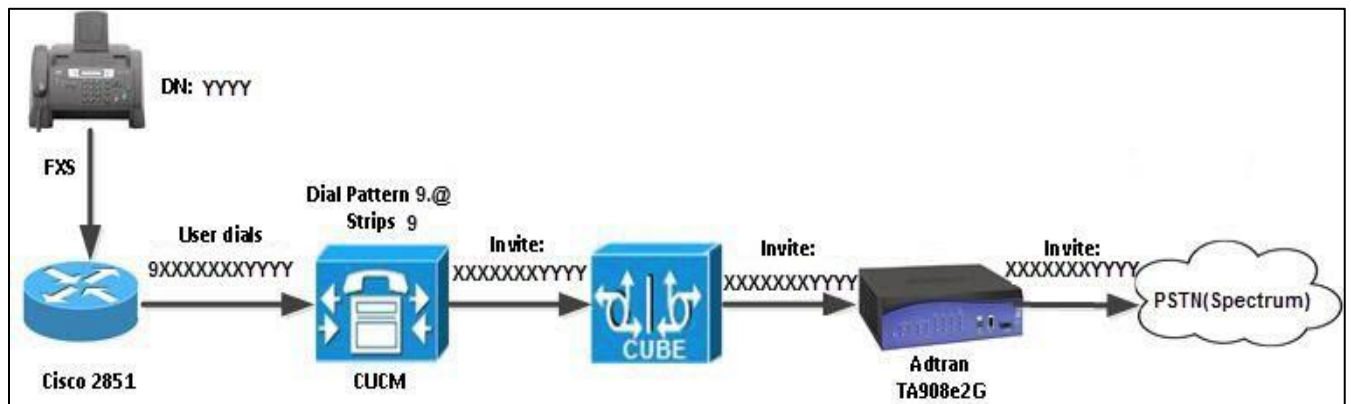


Figure 5: Outbound Fax Call

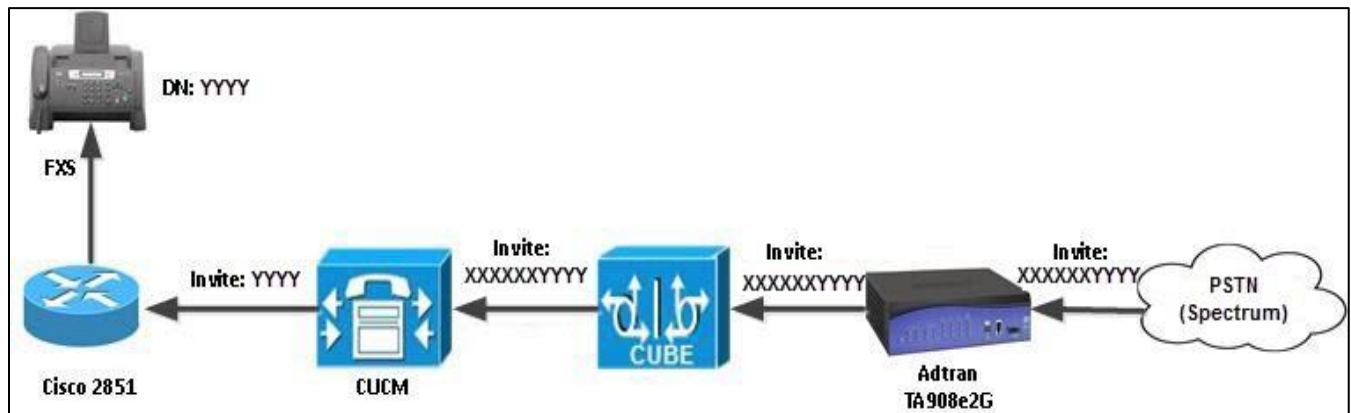


Figure 6: Inbound Fax Call

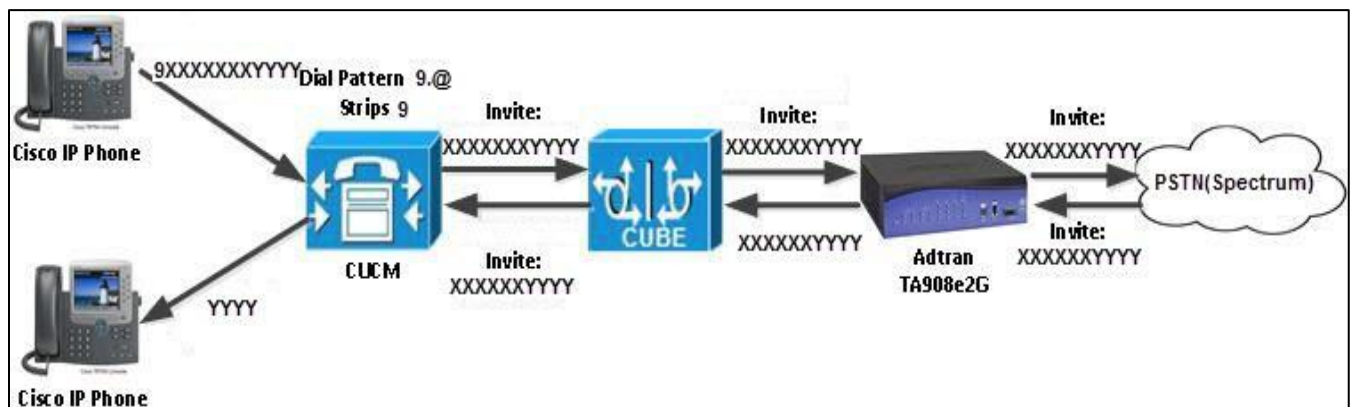


Figure 7: PBX to PBX via Spectrum Call



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$5l3Lr7oSNCAEcKgkNMyGP0
```

```
!
```

```
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  
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  
!  
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  
no fax-relay sg3-to-g3  
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
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:

Password:

```
isr4k2spectrum#sh running-config
```

```
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$7YMA$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:30383511@12"
!
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

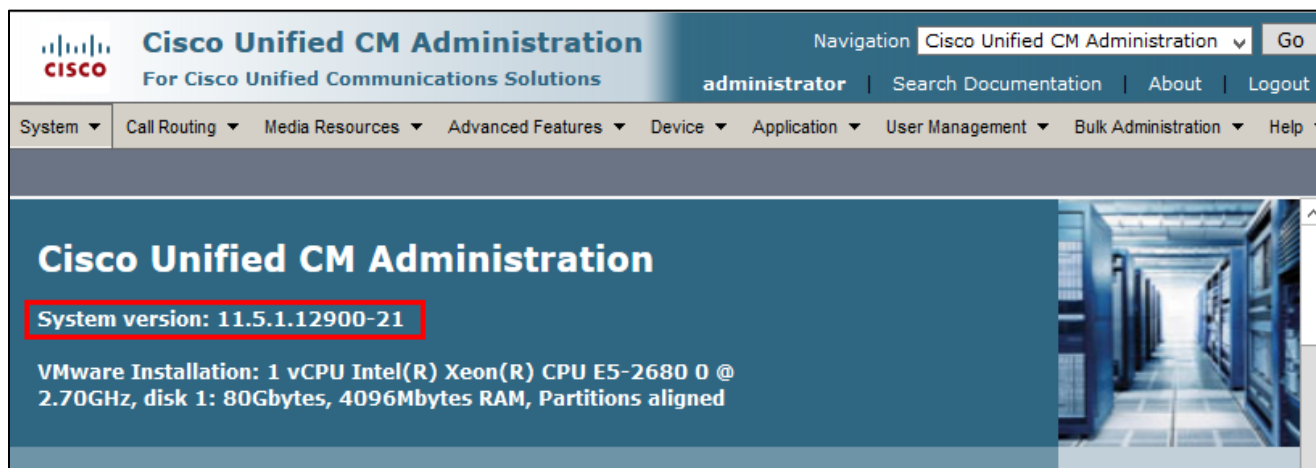


Figure 8: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation: System > Service Parameters

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

Select Server and Service

Server*

Service*

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

Cisco CallManager (Active) Parameters on server clus21sub1--CUCM Voice/Video (Active)

Parameter Name	Parameter Value	Suggested Value
Call Throttling		
Code Yellow Entry Latency *	<input type="text" value="20"/>	20
Code Yellow Exit Latency Calculation *	<input type="text" value="40"/>	40
Code Yellow Duration *	<input type="text" value="5"/>	5
Max Events Allowed *	<input type="text" value="2000"/>	2000
System Throttle Sample Size *	<input type="text" value="10"/>	10

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

SIP Trunk Security Profile Information

Name*	Spectrum Enterprise Non Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null Strin
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP
Outgoing Transport Type	UDP
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
X.509 Subject Name	
Incoming Port*	5060
<input type="checkbox"/> Enable Application level authorization	
<input checked="" type="checkbox"/> Accept presence subscription	
<input checked="" type="checkbox"/> Accept out-of-dialog refer**	
<input checked="" type="checkbox"/> Accept unsolicited notification	
<input checked="" type="checkbox"/> Accept replaces header	
<input type="checkbox"/> Transmit security status	
<input type="checkbox"/> Allow charging header	
SIP V.150 Outbound SDP Offer Filtering*	Use Default Filter

Figure 10: SIP Trunk Security Profile

Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	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.



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	
Name*	Spectrum Enterprise SIP Profile
Description	Spectrum Enterprise SIP Profile
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
User-Agent and Server header information*	Send Unified CM Version Information as User-Agent
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, and
Confidential Access Level Headers*	Disabled
<input type="checkbox"/> Redirect by Application	
<input type="checkbox"/> Disable Early Media on 180	
<input type="checkbox"/> Outgoing T.38 INVITE include audio mline	
<input type="checkbox"/> Offer valid IP and Send/Receive mode only for T.38 Fax Relay	
<input type="checkbox"/> Use Fully Qualified Domain Name in SIP Requests	
<input type="checkbox"/> Assured Services SIP conformance	
<input type="checkbox"/> 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
<input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change	
<input type="checkbox"/> Allow RR/RS bandwidth modifier (RFC 3556)	

Parameters used in Phone	
Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Media Port Ranges	<input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766

Figure 11: SIP Profile



DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default
DSCP for Audio Portion of Video Calls	Use System Default
DSCP for TelePresence Calls	Use System Default
DSCP for Audio Portion of TelePresence Calls	Use System Default
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70

Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial
<input checked="" type="checkbox"/> Conference Join Enabled <input type="checkbox"/> RFC 2543 Hold <input checked="" type="checkbox"/> Semi Attended Transfer <input type="checkbox"/> Enable VAD <input type="checkbox"/> Stutter Message Waiting <input type="checkbox"/> MLPP User Authorization	

Normalization Script							
Normalization Script	< None >						
<input type="checkbox"/> Enable Trace							
	<table border="1"> <thead> <tr> <th></th> <th>Parameter Name</th> <th>Parameter Value</th> </tr> </thead> <tbody> <tr> <td>1</td> <td></td> <td></td> </tr> </tbody> </table>		Parameter Name	Parameter Value	1		
	Parameter Name	Parameter Value					
1							

Incoming Requests FROM URI Settings	
Caller ID DN	
Caller Name	

Figure 12: SIP Profile (Cont.)



Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*

Resource Priority Namespace List

SIP Rel1XX Options*

Video Call Traffic Class*

Calling Line Identification Presentation*

Session Refresh Method*

Early Offer support for voice and video calls*

☐ Enable ANAT

☐ Deliver Conference Bridge Identifier

☐ Allow Passthrough of Configured Line Device Caller Information

☐ Reject Anonymous Incoming Calls

☐ Reject Anonymous Outgoing Calls

☐ Send ILS Learned Destination Route String

SIP OPTIONS Ping

☒ Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)*

Ping Interval for Out-of-service Trunks (seconds)*

Ping Retry Timer (milliseconds)*

Ping Retry Count*

SDP Information

☒ Send send-receive SDP in mid-call INVITE

☐ Allow Presentation Sharing using BFCP

☐ Allow iX Application Media

☐ Allow multiple codecs in answer SDP

Figure 13: SIP Profile (Cont.)

Explanation

Parameter	Value	Description
Default MTP Telephony Event Payload Type	101	RFC2833 DTMF payload type
SIP Rel1XX Options	Send PRACK for 1xx Messages	Enable Provisional Acknowledgements (Reliable 100 messages)
Ping Interval for In-service and Partially In-service Trunks (seconds)	60	OPTIONS message parameters- interval time
Ping Interval for Out-of-service Trunks (seconds)	120	OPTIONS message parameters- interval time



SIP Trunk Configuration

Create SIP trunks to Cisco UBE

Navigation: Device → Trunk

<input type="checkbox"/>	Name ^	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile
<input type="checkbox"/>	CUCM to FAXgateway	SIP Trunk to FAX Gateway		G711 Pool	2063				SIP Trunk	Unknown - OPTIONS Ping not enabled		Non Secure SIP Trunk Profile
<input type="checkbox"/>	Spectrum_Enterprise_Trunk	Charter Trunk		G711 Pool	9.@				SIP Trunk	Unknown - OPTIONS Ping not enabled		Spectrum Non Secure SIP Trunk Profile

Figure 14: SIP Trunks List

SIP Trunk Status
Service Status: Unknown
Duration: Unknown

Device Information
Product: SIP Trunk
Device Protocol: SIP
Trunk Service Type: None(Default)
Device Name*:
Description:
Device Pool*:
Common Device Configuration:
Call Classification*:
Media Resource Group List:
Location*:
AAR Group:
Tunneled Protocol*:

QSIG Variant*:
ASN.1 ROSE OID Encoding*:
Packet Capture Mode*:
Packet Capture Duration:
☐ Media Termination Point Required
☒ Retry Video Call as Audio
☐ Path Replacement Support
☐ Transmit UTF-8 for Calling Party Name
☐ Transmit UTF-8 Names in QSIG APDU
☐ Unattended Port
☐ 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.
Consider Traffic on This Trunk Secure*:
Route Class Signaling Enabled*:
Use Trusted Relay Point*:
☒ PSTN Access
☒ Run On All Active Unified CM Nodes

Figure 15: SIP Trunk to Cisco UBE



Intercompany Media Engine (IME)
E.164 Transformation Profile < None >

MLPP and Confidential Access Level Information
MLPP Domain < None >
Confidential Access Mode < None >
Confidential Access Level < None >

Call Routing Information
☒ Remote-Party-Id
☒ Asserted-Identity
Asserted-Type* Default
SIP Privacy* Default

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

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.
Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

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.
Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

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



Connected Party Settings		
Connected Party Transformation CSS < None >		
<input checked="" type="checkbox"/> Use Device Pool Connected Party Transformation CSS		
Outbound Calls		
Called Party Transformation CSS < None >		
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS		
Calling Party Transformation CSS < None >		
<input checked="" type="checkbox"/> 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	
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound		
Redirecting Party Transformation CSS	< None >	
<input checked="" type="checkbox"/> Use Device Pool Redirecting Party Transformation CSS		
Caller Information		
Caller ID DN	<input type="text"/>	
Caller Name	<input type="text"/>	
<input type="checkbox"/> Maintain Original Caller ID DN and Caller Name in Identity Headers		
SIP Information		
Destination		
<input type="checkbox"/> Destination Address is an SRV		
1*	10.80.11.20	5060
MTP Preferred Originating Codec*	711ulaw	
BLF Presence Group*	Standard Presence group	
SIP Trunk Security Profile*	Spectrum Enterprise Non Secure SIP Trunk Profile	
Rerouting Calling Search Space	< None >	
Out-Of-Dialog Refer Calling Search Space	< None >	
SUBSCRIBE Calling Search Space	< None >	
SIP Profile*	Spectrum Enterprise SIP Profile View Details	
DTMF Signaling Method*	No Preference	
Recording Information		
<input checked="" type="radio"/> None		
<input type="radio"/> This trunk connects to a recording-enabled gateway		
<input type="radio"/> This trunk connects to other clusters with recording-enabled gateways		
Geolocation Configuration		
Geolocation	< None >	
Geolocation Filter	< None >	
<input type="checkbox"/> Send Geolocation Information		

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



Pattern Definition		
Route Pattern*	9.@"	
Route Partition	< None >	
Description	To dialogic	
Numbering Plan*	NANP	
Route Filter	< None >	
MLPP Precedence*	Default	
<input type="checkbox"/> Apply Call Blocking Percentage		
Resource Priority Namespace Network Domain	< None >	
Route Class*	Default	
Gateway/Route List*	Spectrum_Enterprise_Trunk	(Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error	
Call Classification*	OffNet	
External Call Control Profile	< None >	
<input type="checkbox"/> Allow Device Override	<input checked="" type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Overlap Sending
<input type="checkbox"/> Require Forced Authorization Code		
Authorization Level*	0	
<input type="checkbox"/> Require Client Matter Code		

Calling Party Transformations	
<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	
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	PreDot
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		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Figure 19: Route Pattern for Voice



- Pattern Definition -	
Route Pattern*	2302
Route Partition	< None >
Description	Route pattern to unity
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	Unity_Connection (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error
Call Classification*	OffNet
External Call Control Profile	< None >
<input type="checkbox"/> Allow Device Override	<input checked="" type="checkbox"/> Provide Outside Dial Tone
<input type="checkbox"/> Require Forced Authorization Code	<input type="checkbox"/> Allow Overlap Sending
Authorization Level*	0
<input type="checkbox"/> Require Client Matter Code	<input type="checkbox"/> Urgent Priority

- Calling Party Transformations -	
<input type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	
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 -		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Figure 20: Route Pattern for Voice (Cont.)



- Pattern Definition -		
Route Pattern*	2063	
Route Partition	< None >	
Description	Route pattern to FAX Gateway	
Numbering Plan	-- Not Selected --	
Route Filter	< None >	
MLPP Precedence*	Default	
<input type="checkbox"/> Apply Call Blocking Percentage		
Resource Priority Namespace Network Domain	< None >	
Route Class*	Default	
Gateway/Route List*	CUCM_to_FAXgateway	(Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error	
Call Classification*	OffNet	
External Call Control Profile	< None >	
<input type="checkbox"/> Allow Device Override	<input checked="" type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Overlap Sending
<input type="checkbox"/> Require Forced Authorization Code	<input type="checkbox"/> Urgent Priority	
Authorization Level*	0	
<input type="checkbox"/> Require Client Matter Code		

- Calling Party Transformations -	
<input type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	
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

- 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 -		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Figure 21: Route Pattern for Fax



Explanation

Setting	Value	Description
Route Pattern	9.@ for Voice & International Calls, 2063 for Fax Call and 2302 for Unity Connection	Specify appropriate Route Pattern
Gateway/Route List	Spectrum Enterprise for Route Pattern 9.@, 2063 for SIP Trunk To Fax Gateway and 2032 for Unity Connection	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 9.@	North American Numbering Plan
Call Classification	OffNet for Route Pattern 9.@, 2063 and 2032	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 9.@	Specifies how to modify digit before they are sent to Spectrum network

Acronyms

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



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