



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 Enterprise¹, 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:

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

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



Network Topology

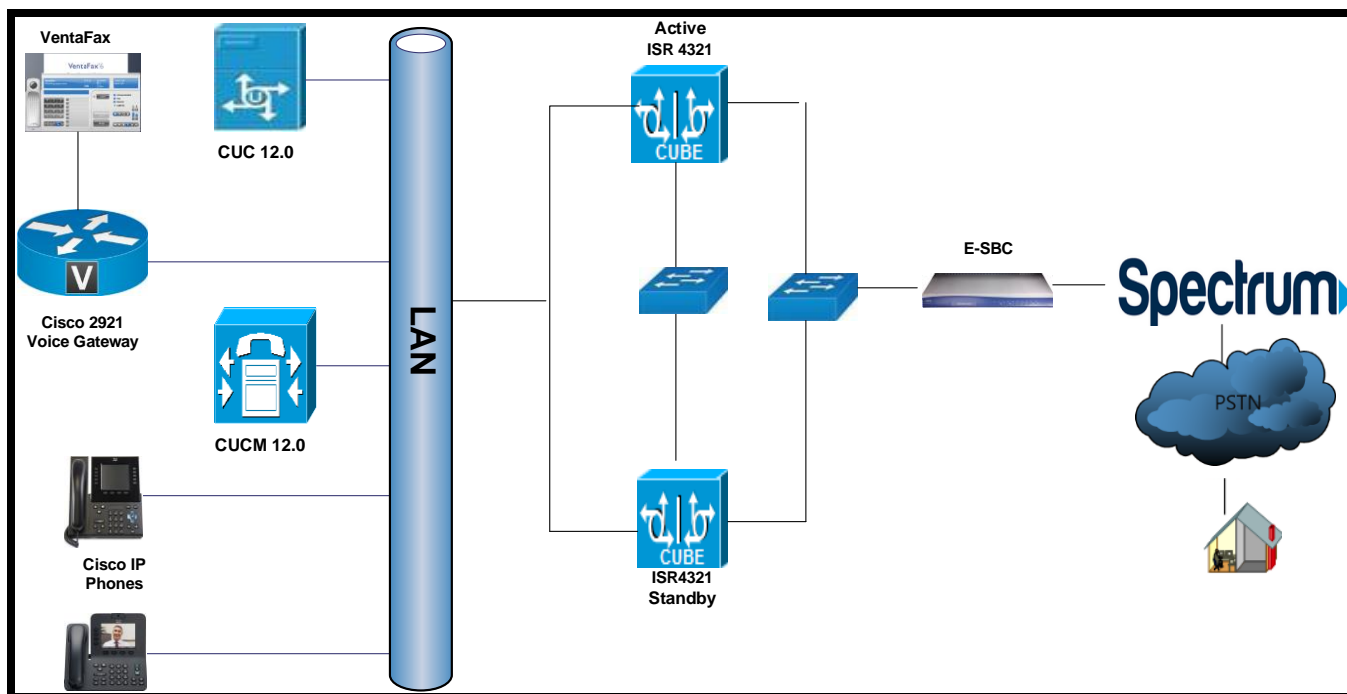


Figure 1: Network Topology

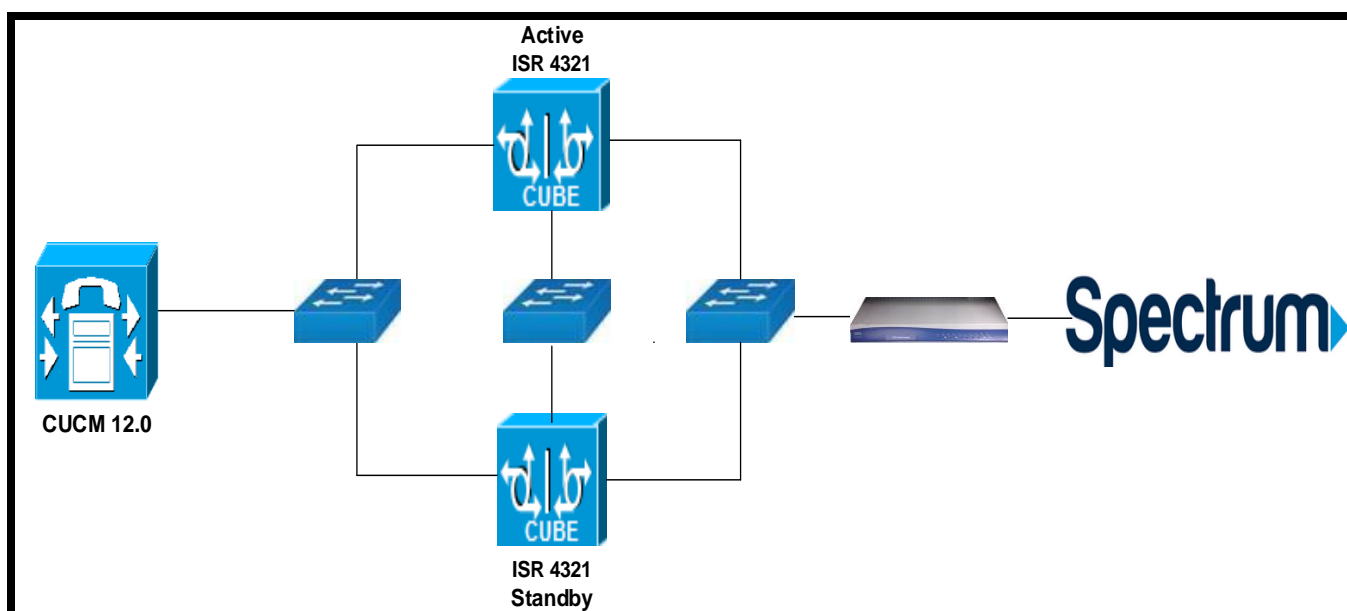


Figure 2: Cisco UBE High Availability



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 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

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

```
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.

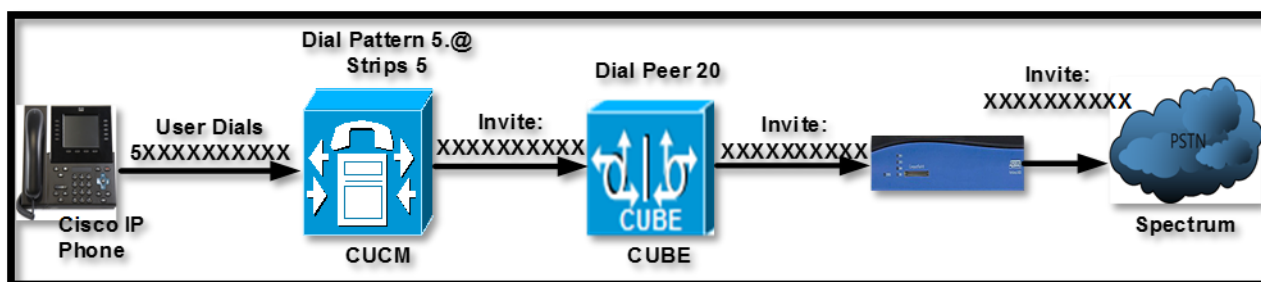


Figure 3: Outbound Voice Call

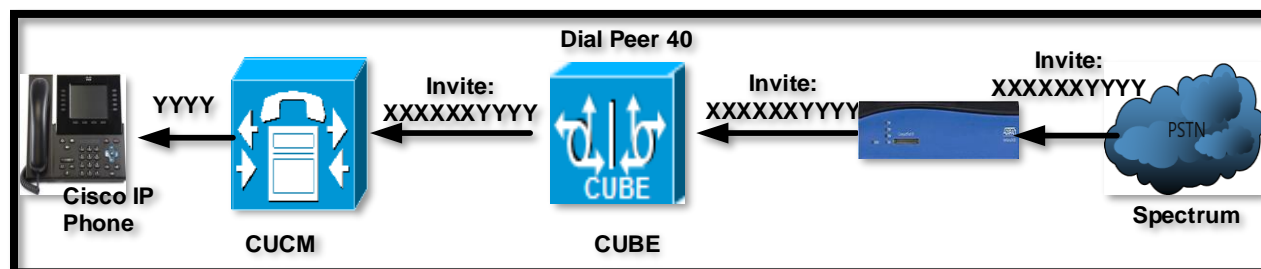


Figure 4: Inbound Voice Call

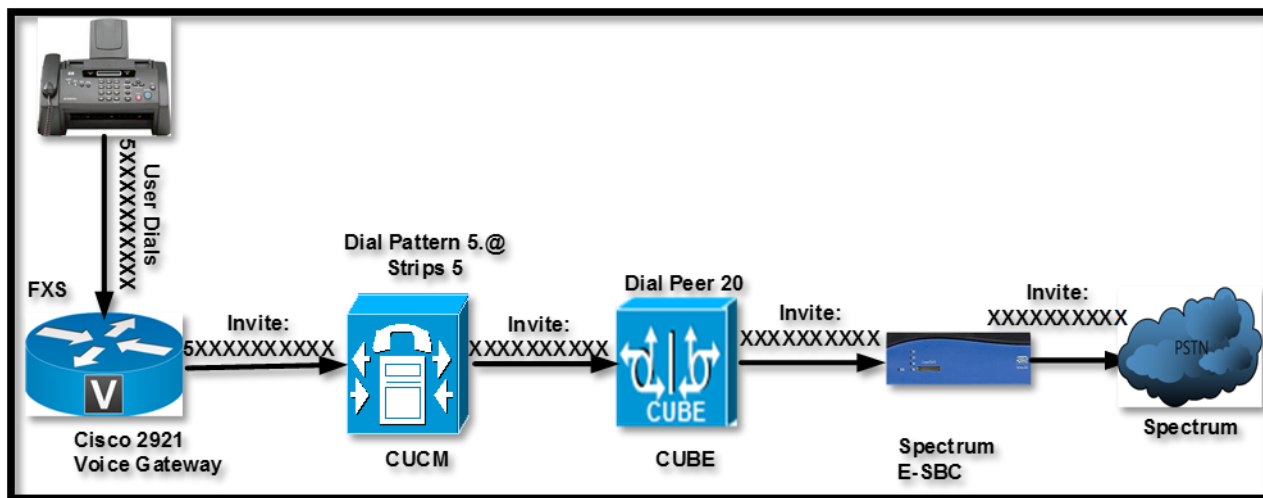


Figure 5: Outbound Fax Call

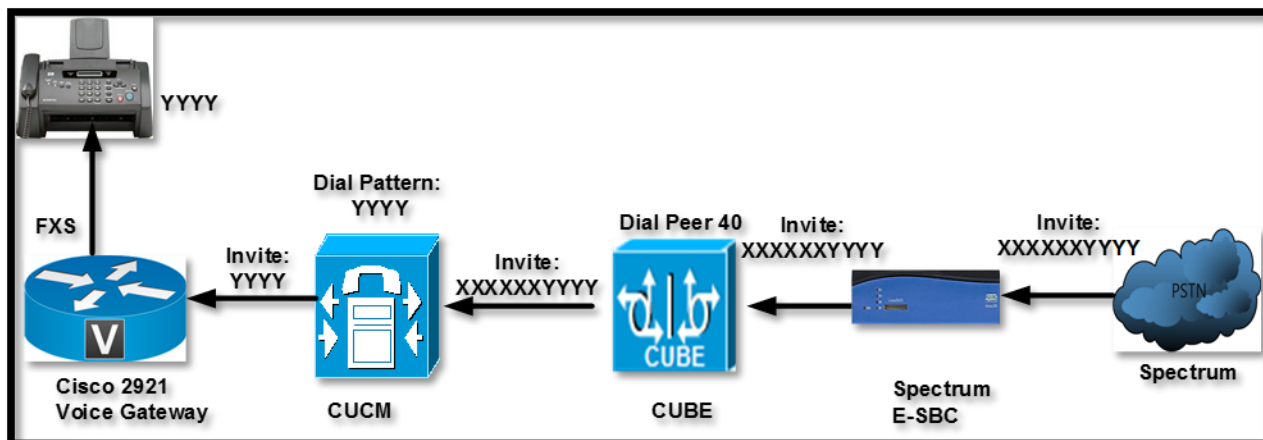


Figure 6: Inbound Fax Call

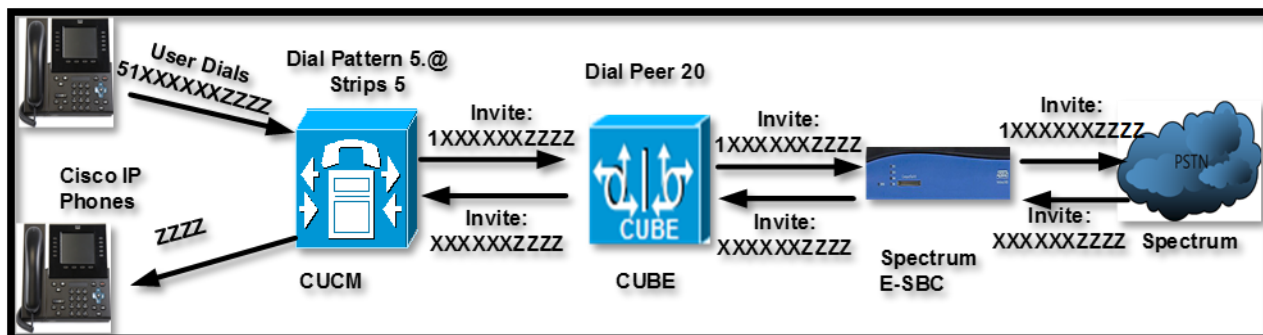


Figure 7: PBX to PBX via Spectrum Call



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  
!  
crypto pki certificate chain 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-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 34.202.215.187
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 $1$JIS4$FtJ1pipGXrkwyKV3i/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

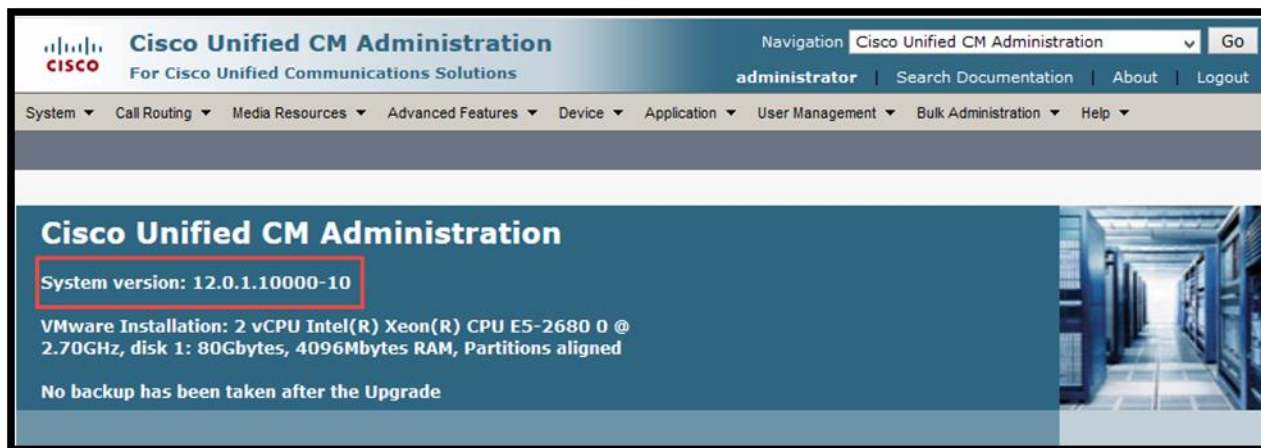


Figure 8: CUCM Version

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

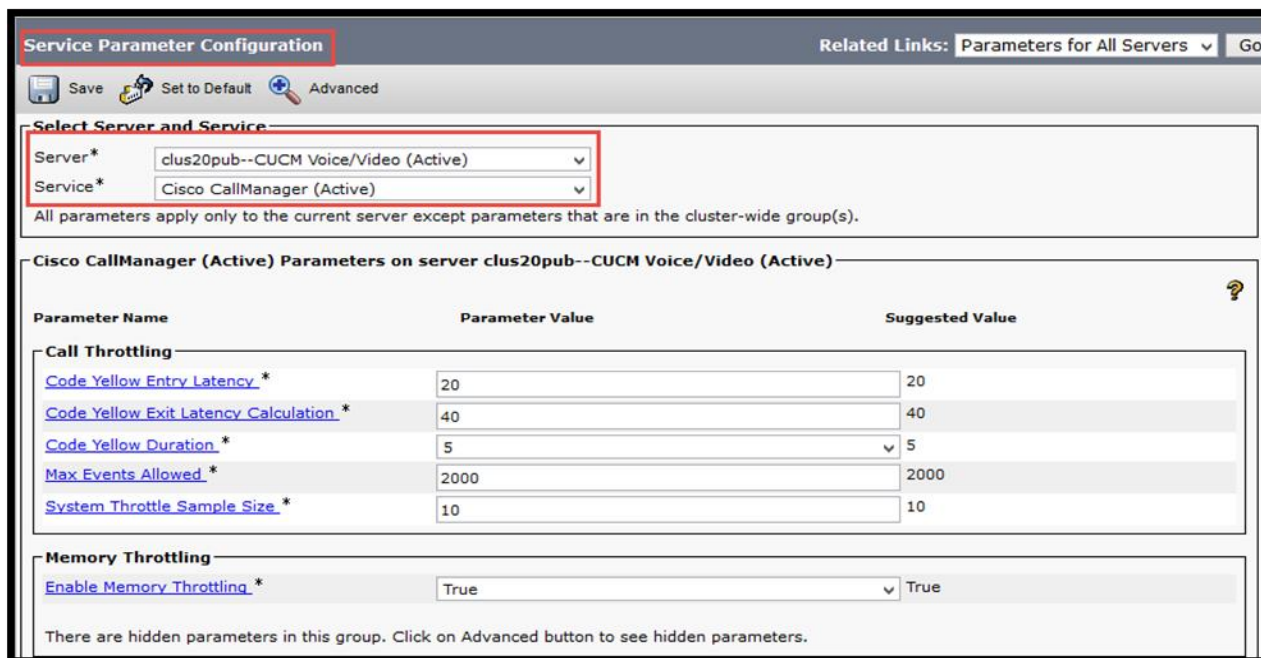


Figure 9: Service Parameters



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 Configuration

Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Reset Apply Config Add New

SIP Trunk Security Profile Information

Name* Spectrum SIP Trunk Security Profile

Description None Secure Profile for Spectrum

Device Security Mode Non Secure

Incoming Transport Type* TCP+UDP

Outgoing Transport Type UDP

☐ Enable Digest Authentication

Nonce Validity Time (mins)* 600

X.509 Subject Name

Incoming Port* 5060

☐ Enable Application level authorization

☐ Accept presence subscription

☐ Accept out-of-dialog refer**

☐ Accept unsolicited notification

☐ Accept replaces header

☐ Transmit security status

☐ 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 Standard SIP Profile
2. **Description**: Default SIP Profile

SIP Profile Configuration Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Reset Apply Config Add New

SIP Profile Information

Name* Spectrum Standard SIP Profile

Description Default 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-Agen

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

☐ Redirect by Application

☐ Disable Early Media on 180

☐ Outgoing T.38 INVITE include audio mline

☐ Offer valid IP and Send/Receive mode only for T.38 Fax Relay

☐ Use Fully Qualified Domain Name in SIP Requests

☐ Assured Services SIP conformance

☐ Enable External QoS**

SDP Information

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites* TIAS and AS

SDP Transparency Profile Pass all unknown SDP attributes

Accept Audio Codec Preferences in Received Offer* Default

☐ Require SDP Inactive Exchange for Mid-Call Media Change

☐ Allow RR/RS bandwidth modifier (RFC 3556)

Parameters used in Phone

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

☒ Common Port Range for Audio and Video

☐ Separate Port Ranges for Audio and Video

Figure 11: SIP Profile



Start Media Port*	16384								
Stop Media Port*	32766								
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><thead><tr><th></th><th>Parameter Name</th><th>Parameter Value</th><th></th></tr></thead><tbody><tr><td>1</td><td></td><td></td><td><input type="button" value="+"/> <input type="button" value="-"/></td></tr></tbody></table>			Parameter Name	Parameter Value		1			<input type="button" value="+"/> <input type="button" value="-"/>
	Parameter Name	Parameter Value							
1			<input type="button" value="+"/> <input type="button" value="-"/>						
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* Never

Resource Priority Namespace List < None >

SIP RelXX Options* Send PRACK if 1xx Contains SDP

Video Call Traffic Class* Mixed

Calling Line Identification Presentation* Default

Session Refresh Method* Invite

Early Offer support for voice and video calls* Best Effort (no MTP inserted)

☐ 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

☐ Connect Inbound Call before Playing Queuing Announcement

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)* 60

Ping Interval for Out-of-service Trunks (seconds)* 120

Ping Retry Timer (milliseconds)* 500

Ping Retry Count* 6

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

Navigation: Device → Trunk





Find and List Trunks									
<div> + Add New ☐ Select All ☐ Clear All ✖ Delete Selected ↺ Reset Selected </div>									
<input type="checkbox"/>		SpectrumFax	SpectrumFax	G711 Preferred	898X	SIP Trunk	Full Service	Time In Full Service: 1 day 3 hours 7 minutes	Spectrum SIP Trunk Security Profile
<input type="checkbox"/>		SpectrumFax	SpectrumFax	G711 Preferred	126X	SIP Trunk	Full Service	Time In Full Service: 1 day 3 hours 7 minutes	Spectrum SIP Trunk Security Profile
<input type="checkbox"/>		Spectrum SIP Trunk	Spectrum SIP Trunk	G711 Preferred	5.0	SIP Trunk	Full Service	Time In Full Service: 0 day 21 hours 41 minutes	Spectrum SIP Trunk Security Profile
<input type="checkbox"/>		Unity	Unity	Default	2302	SIP Trunk	Unknown - OPTIONS Ping not enabled		SIP Trunk Profile for Unity Connection

Figure 14: SIP Trunks List

Trunk Configuration

Related Links:
Back To Find/List
Go

Save
Delete
Reset
Add New

Device Information

Product:
SIP Trunk

Device Protocol:
SIP

Trunk Service Type
None(Default)

Device Name*
Spectrum_SIP_Trunk

Description
Spectrum SIP Trunk

Device Pool*
G711 Preferred

Common Device Configuration
< None >

Call Classification*
Use System Default

Media Resource Group List
MRGL

Location*
Hub_None

AAR Group
< None >

Tunneled Protocol*
None

QSIG Variant*
No Changes

ASN.1 ROSE OID Encoding*
No Changes

Packet Capture Mode*
None

Packet Capture Duration
0

☐ 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*
When using both sRTP and TLS

Route Class Signaling Enabled*
Default

Use Trusted Relay Point*
Default

☐ PSTN Access

☐ Run On All Active Unified CM Nodes

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 >

Figure 15: SIP Trunk to CUBE



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

Inbound Calls
Significant Digits*
Connected Line ID Presentation*
Connected Name Presentation*
Calling Search Space
AAR Calling Search Space
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	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" < 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	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" < None >"/>	<input checked="" type="checkbox"/>

Connected Party Settings
Connected Party Transformation CSS
☒ Use Device Pool Connected Party Transformation CSS

Outbound Calls
Called Party Transformation CSS
☒ Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS
☒ Use Device Pool Calling Party Transformation CSS
Calling Party Selection*
Calling Line ID Presentation*
Calling Name Presentation*
Calling and Connected Party Info Format*
☒ Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS
☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information
Caller ID DN
Caller Name
☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

Figure 16: SIP Trunk to CUBE (Cont.)



SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status
1 *	10.80.10.50		5060	up

MTP Preferred Originating Codec* 711ulaw

BLF Presence Group* Standard Presence group

SIP Trunk Security Profile* Spectrum SIP Trunk Security Profile

Rerouting Calling Search Space < None >

Out-Of-Dialog Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile* Spectrum Standard SIP Profile [View Details](#)

DTMF Signaling Method* No Preference

Normalization Script

Normalization Script Spectrum

☐ Enable Trace

	Parameter Name	Parameter Value
1	Diversion-Mask	1469573XXXX

Recording Information

☒ None

☐ This trunk connects to a recording-enabled gateway

☐ This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation < None >

Geolocation Filter < None >

☐ Send Geolocation Information

Figure 17: SIP Trunk to CUBE (Cont.)



Trunk Configuration

Related Links: [Back To Find/List](#)

Device Information

Product:

SIP Trunk

Device Protocol:

SIP

Trunk Service Type

None(Default)

Device Name*

SpectrumFax

Description

SpectrumFax

Device Pool*

G711 Preferred

Common Device Configuration

< None >

Call Classification*

Use System Default

Media Resource Group List

MRGL

Location*

Hub_None

AAR Group

< None >

Tunneled Protocol*

None

QSIG Variant*

No Changes

ASN.1 ROSE OID Encoding*

No Changes

Packet Capture Mode*

None

Packet Capture Duration

0

☐ 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*

When using both sRTP and TLS

Route Class Signaling Enabled*

Default

Use Trusted Relay Point*

Default

☐ PSTN Access

☐ Run On All Active Unified CM Nodes

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

Trust Received Identity*

Trust All (Default)

Inbound Calls

Significant Digits*

All

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"/>

Figure 18: SIP Trunk to Fax Gateway

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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 SettingsDefault Prefix Settings

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

Connected Party Settings

Connected Party Transformation CSS < None >

☒ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS < None >

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS < None >

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection* Originator

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Calling and Connected Party Info Format* Deliver DN only in connected party

☐ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status
1*	10.80.19.7		5060	up

MTP Preferred Originating Codec* 711ulaw

BLF Presence Group* Standard Presence group

SIP Trunk Security Profile* Spectrum SIP Trunk Security Profile

Rerouting Calling Search Space < None >

Out-Of-Dialog Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile* Spectrum Standard SIP Profile [View Details](#)

DTMF Signaling Method* No Preference

Normalization Script

Normalization Script < None >

☐ Enable Trace

	Parameter Name	Parameter Value
1		

Recording Information

☒ None

☐ This trunk connects to a recording-enabled gateway

☐ This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation < None >

Geolocation Filter < None >

☐ Send Geolocation Information

Figure 19: SIP Trunk to Fax gateway Cont.



Trunk Configuration

Related Links: [Back To Find/List](#)

Device Information

Product:

SIP Trunk

Device Protocol:

SIP

Trunk Service Type

None(Default)

Device Name*

Unity

Description

Unity

Device Pool*

G711 Preferred

Common Device Configuration

< None >

Call Classification*

Use System Default

Media Resource Group List

MRGL

Location*

Hub_None

AAR Group

< None >

Tunneled Protocol*

None

QSIG Variant*

No Changes

ASN.1 ROSE OID Encoding*

No Changes

Packet Capture Mode*

None

Packet Capture Duration

0

☐ 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*

When using both sRTP and TLS

Route Class Signaling Enabled*

Default

Use Trusted Relay Point*

Default

☒ PSTN Access

☐ Run On All Active Unified CM Nodes

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

Trust Received Identity*

Trust All (Default)

Inbound Calls

Significant Digits*

All

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"/>

Figure 20: SIP trunk to Cisco Unity Connection

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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 SettingsDefault Prefix Settings

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

Connected Party Settings

Connected Party Transformation CSS < None >

☒ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS < None >

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS < None >

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection* Originator

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Calling and Connected Party Info Format* Deliver DN only in connected party

☒ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status
1*	10.80.10.5		5060	N/A

MTP Preferred Originating Codec* 711ulaw

BLF Presence Group* Standard Presence group

SIP Trunk Security Profile* SIP Trunk Profile for Unity Connection

Rerouting Calling Search Space < None >

Out-Of-Dialog Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile* Standard SIP Profile [View Details](#)

DTMF Signaling Method* No Preference

Normalization Script

Normalization Script < None >

☐ Enable Trace

	Parameter Name	Parameter Value
1		

Recording Information

☒ None

☐ This trunk connects to a recording-enabled gateway

☐ This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation < None >

Geolocation Filter < None >

☐ Send Geolocation Information

Figure 21: SIP Trunk to Unity Connection Cont.

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Explanation

Parameter	Value	Description
Device Name	Spectrum_SIP_Trunk for SIP Trunk to CUBE; SpectrumFax for SIP trunk to Fax Gateway; Unity for SIP trunk to Cisco Unity Connection	Name for the trunk
Device Pool	G711 Preferred	G711 Device Pool is used for this trunk
Media Resource Group List	MRGL	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4 for SIP Trunk to CUBE; All for SIP trunk to Fax gateway and Cisco Unity Connection	4 digits Extension for all CPE phones/Fax/Unity Connection Pilot number
Destination Address	10.80.10.50/10.80.19.7/10.80.10.5 for SIP trunk to CUBE/Fax Gateway/Cisco Unity Connection	IP address of the Cisco UBE Virtual LAN/Fax Gateway/Cisco Unity Connection
SIP Trunk Security Profile	Spectrum Enterprise Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	Spectrum Enterprise SIP Profile	SIP Profile configured earlier
Normalization Script	Spectrum for SIP trunk to CUBE only	Convert 4 digits EXT to 11 digits DID for Diversion header
Diversion-Mask	1469573XXXX/1303547XXXX	Used in the Manipulation Script for Timer Warner Cable/L-Charter



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

Related Links: [Back To Find/List](#) [Go](#)

Save Delete Reset Add New Import File

SIP Normalization Script Info

Name* Spectrum

Description convert 4 digits Diversion to 11 digits

Content*

```
M = {}
local mask = scriptParameters.getValue("Diversion-Mask")

-- handle the mask of the diversion header for non-911 calls
function M.outbound_INVITE(message)
    if mask
    then
        message:applyNumberMask("Diversion", mask)
    end
end
return M
```

Script Execution Error Recovery Action* Message Rollback Only

System Resource Error Recovery Action* Disable Script

Memory Threshold* 50 kilobytes

Lua Instruction Threshold* 1000 instructions

Figure 22: SIP Normalization Script

SIP Normalization Script (Text):

```
M = {}
local mask = scriptParameters.getValue("Diversion-Mask")

-- handle the mask of the diversion header for non-911 calls
function M.outbound_INVITE(message)
    if mask
    then
        message:applyNumberMask("Diversion", mask)
    end
end
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









Find and List Route Patterns			
	Add New		Select All
	Clear All		Delete Selected
<input type="checkbox"/>	126X	SpectrumFax	
<input type="checkbox"/>	2302	Unity	
<input type="checkbox"/>	5.@	Spectrum SIP Trunk	
<input type="checkbox"/>	898X	SpectrumFax	

Figure 23: Route Patterns List

Route Pattern Configuration
Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Add New

Pattern Definition

Route Pattern*	5.@	
Route Partition	< None >	
Description		
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_SIP_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"/> Urgent Priority <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 24: Route Pattern for Voice



Route Pattern Configuration

Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Add New

Pattern Definition

Route Pattern*

898X

Route Partition

< None >

Description

Numbering Plan

-- Not Selected --

Route Filter

< None >

MLPP Precedence*

Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

< None >

Route Class*

Default

Gateway/Route List*

SpectrumFax

[\(Edit\)](#)

Route Option

☒ Route this pattern

☐ Block this pattern

No Error

Call Classification*

OffNet

External Call Control Profile

< None >

☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority

☐ 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

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

-- Not Selected --

Service Parameter Name

< Not Exist >

Service Parameter Value

Figure 25: Route Pattern for Fax (L-TWC)



Route Pattern ConfigurationRelated Links: [Back To Find/List](#) [Go](#)

Save

Delete

Copy

Add New

Pattern Definition

Route Pattern*

126X

Route Partition

< None >

Description

Numbering Plan

-- Not Selected --

Route Filter

< None >

MLPP Precedence*

Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

< None >

Route Class*

Default

Gateway/Route List*

SpectrumFax

(Edit)

Route Option

☒ Route this pattern

☐ Block this pattern

No Error

Call Classification*

OffNet

External Call Control Profile

< None >

☐ Allow Device Override

☒ Provide Outside Dial Tone

☐ Allow Overlap Sending

☐ Urgent Priority

☐ 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

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

-- Not Selected --

Service Parameter Name

< Not Exist >

Service Parameter Value

Figure 26: Route Pattern for Fax (L-Charter)



Route Pattern Configuration

Related Links: [Back To Find/List](#) [Go](#)

Save

Delete

Copy

Add New

Pattern Definition

Route Pattern*

2302

Route Partition

< None >

Description

Numbering Plan

-- Not Selected --

Route Filter

< None >

MLPP Precedence*

Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

< None >

Route Class*

Default

Gateway/Route List*

Unity

[\(Edit\)](#)

Route Option

☒ Route this pattern

☐ Block this pattern

No Error

Call Classification*

OffNet

External Call Control Profile

< None >

☐ Allow Device Override

☒ Provide Outside Dial Tone

☐ Allow Overlap Sending

☐ Urgent Priority

☐ 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

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

-- Not Selected --

Service Parameter Name

< Not Exist >

Service Parameter Value

Figure 27: Route Pattern to Cisco Unity Connection



Explanation

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

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



rsa keypair 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 2D536967 6E65642D 43657274
69666963 6174652D 31313230 34333030 3739301E 170D3133 31303031 32313434
33305A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 31323034
33303037 3930819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
8100D201 333D5645 360F1451 B7BC6EBB 7FFC6715 F1658731 5ABE4743 6B6EA3B0
1C7D1D76 17B5BD08 BA68F94B 80F790EA 8C46036E 7FDE5827 3BF9B45F 7B71D959
B6943C33 A4781184 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 E365F73A
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 CD45A4**

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 pvdn 0/0

!

username cisco privilege 15 secret 5 \$1\$vLg0\$08vBEuXW0zO4WrX6xkgOY/

!

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

!

no 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

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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