



Sprint SIP Trunking:

Cisco Unified Communications Manager 11.5.1 with Cisco Unified Border Element (CUBE 12.0.0) on ISR 4321 [IOS-XE 16.06.01] using SIP

October 25, 2017



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Introduction

Service Providers today, such as Sprint, 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 Sprint network, Cisco Unified Border Element (CUBE) ISR 4321/K9 running IOS 16.5.1b can be used. The Cisco Unified Border Element 16.5.1b provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.1 connected to Sprint network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco Unified Communications Manager (CUCM). Only configuration settings specifically required for Sprint interoperability are presented. Feature configuration and most importantly the dial plan are customer specific and need individual approach.

- This application note describes how to configure a Cisco Unified Communications Manager (CUCM) 11.5.1 and Cisco Unified Border Element (CUBE) on ISR 4321/K9 [IOS - 16.5.1b] for connectivity to Sprint SIP Trunking service available in the former Sprint Business service area¹. The deployment model covered in this application note is CPE (CUCM 11.5.1) to PSTN (Sprint).
- Testing was performed in accordance to Sprint 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 CUCM configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Sprint 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 CUCM to interoperate to Sprint SIP Trunking network.

This application note does not cover the use of Calling Search Spaces (CSS) or partitions on CUCM. 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



Network Topology

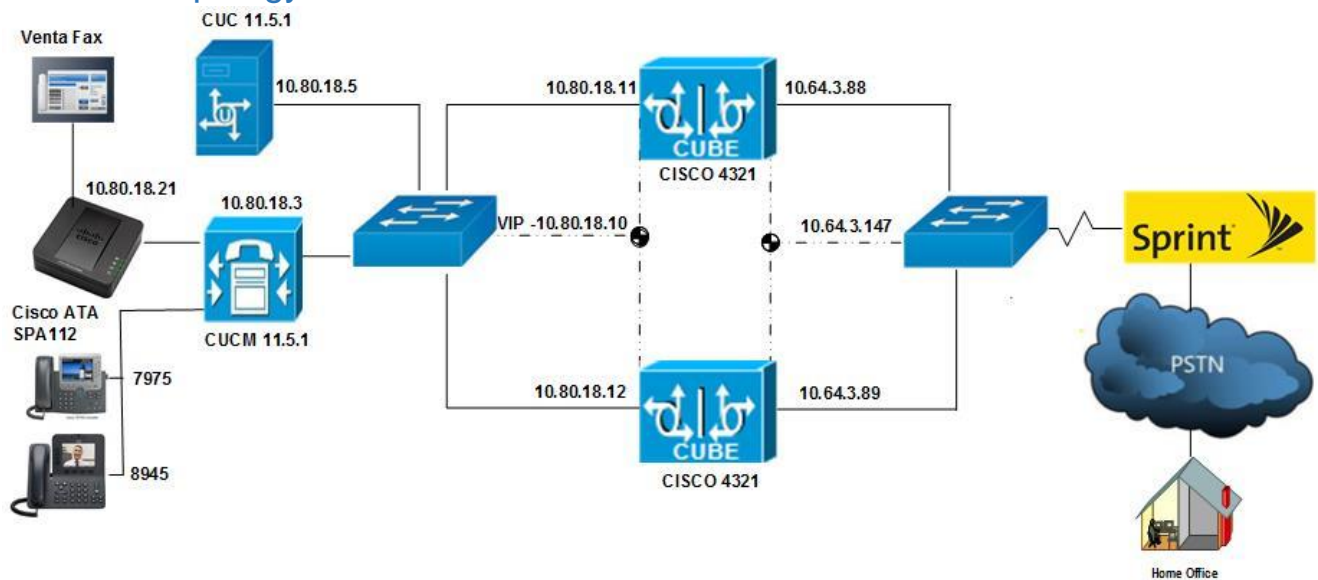


Figure 1: Network Topology

System Components

Hardware Requirements

- Cisco UCSC-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR4321/K9 router as CUBE
- Cisco ISR4321/K9 (1RU) processor with 1797107K/6147K bytes of memory with 3 Gigabit Ethernet interfaces
- Processor board ID FTX1845AJ9S
- Cisco 2851 Fax Gateway
- IP phones 9971 (SIP), 7960 (SIP) and 8945 (SIP)

Software Requirements

- Cisco Unified Communications Manager 11.5.1
- Cisco Unity Connection 11.5.1
- IOS 16.06.01 for ISR 4321/K9 Cisco Unified Border Element
- Cisco IOS Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 16.06.01, RELEASE SOFTWARE (fc1)
- Cisco IOS XE Software, Version 03.17.01.S
- Cisco SPA112 for FAX



Features

Features Supported

- Incoming and outgoing off-net calls using G729
- 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 and T38)

Features Not Supported

- Cisco IP phones used in this test do not support blind transfer
- 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 CUBE and will be resolved in the next release. The issue does not impact the calls.



Configuration

Configuring Cisco Unified Border Element (CUBE)

Network Interface

The IP address used are for illustration only, the actual IP address can vary. The Active/Standby pair share the same virtual IP address and continually exchange status messages.

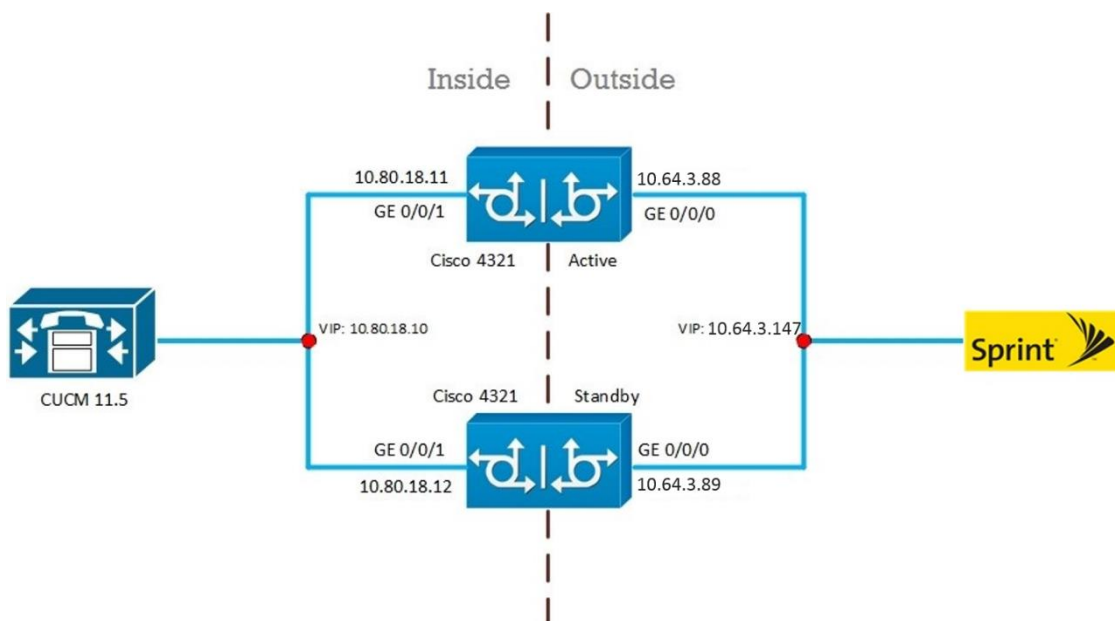


Figure 2: High Availability Topology



CUBE 1:

```
interface GigabitEthernet0/0/0
description WAN
ip address 10.64.3.88 255.255.0.0
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.64.3.147 exclusive
!
interface GigabitEthernet0/0/1
description LAN
ip address 10.80.18.11 255.255.255.0
negotiation auto
redundancy rii 2
redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/1/0
description CUBE HA
ip address 10.89.20.10 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
negotiation auto
!
```




Cisco UBE 2:

```
interface GigabitEthernet0/0/0
description WAN
ip address 10.64.3.89 255.255.0.0
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.64.3.147 exclusive
!
interface GigabitEthernet0/0/1
description LAN
ip address 10.80.18.12 255.255.255.0
negotiation auto
redundancy rii 2
redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/1/0
description CUBE HA
ip address 10.89.20.11 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
```



Global CUBE Settings

In order to enable CUBE IP2IP SBC functionality, following command has to be entered:

```
voice service voip
no ip address trusted authenticate
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback cisco
sip
midcall-signaling passthru
pass-thru content unsupp
pass-thru content sdp
!
```

Explanation

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
redundancy-group 1	Enable High Availability for the VoIP service
fax protocol	Specifies the fax protocol

Codecs

G729 is used primarily towards Sprint until specified otherwise.

```
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711ulaw
!
```



Dial peer

Outbound Dial-peer to Sprint:

```
dial-peer voice 1 voip
description Incoming call from CUCM
session protocol sipv2
incoming called-number .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 protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback cisco
no vad
!
dial-peer voice 2 voip
description Outgoing to Sprint
destination-pattern .T
session protocol sipv2
session target ipv4:199.11.XXX.XX:5060
voice-class codec 1
voice-class sip options-ping 60
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 protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback cisco
no vad
```



Inbound Dial-Peer from Sprint:

```
dial-peer voice 3 voip
description Incoming call from Sprint
session protocol sipv2
incoming called-number 913827....
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 protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback cisco
no vad
!
dial-peer voice 4 voip
description Outgoing to CUCM
destination-pattern 913827....
session protocol sipv2
session target ipv4:10.80.18.2:5060
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 protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback cisco
no vad
```



Configuration example

The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously.

Active Cisco UBE:

Current configuration : 5016 bytes

!

! Last configuration change at 19:32:03 UTC Tue Aug 29 2017

! NVRAM config last updated at 19:32:07 UTC Tue Aug 29 2017

!

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 Sprint_CUBE1

!

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  
!  
no aaa new-model  
!  
subscriber templating  
!  
multilink bundle-name authenticated  
!  
crypto pki trustpoint TP-self-signed-1270583006  
    enrollment selfsigned  
    subject-name cn=IOS-Self-Signed-Certificate-1270583006  
    revocation-check none  
    rsakeypair TP-self-signed-1270583006  
!  
crypto pki certificate chain TP-self-signed-1270583006  
!  
voice service voip  
    no ip address trusted authenticate  
    mode border-element license capacity 20  
    allow-connections sip to sip  
    redundancy-group 1  
    no supplementary-service sip handle-replaces  
    redirect ip2ip  
    fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback cisco  
    sip  
    midcall-signaling passthru  
    pass-thru content unsupp  
    pass-thru content sdp
```



```
!  
voice class codec 1  
  codec preference 1 g729r8  
  codec preference 2 g711ulaw  
!  
license udi pid ISR4321/K9 sn FDO19220MQ8  
license boot level appxk9  
license boot level uck9  
diagnostic bootup level minimal  
spanning-tree extend system-id  
!  
redundancy  
  mode none  
  application redundancy  
  group 1  
    name voice-HA  
    priority 150 failover threshold 75  
    timers delay 30 reload 60  
    control GigabitEthernet0/1/0 protocol 1  
    data GigabitEthernet0/1/0  
    track 1 shutdown  
    track 2 shutdown  
!  
track 1 interface GigabitEthernet0/0/0 line-protocol  
!  
track 2 interface GigabitEthernet0/0/1 line-protocol  
!  
interface GigabitEthernet0/0/0  
  description WAN
```



```
ip address 10.64.3.88 255.255.0.0
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.64.3.147 exclusive
!
interface GigabitEthernet0/0/1
description LAN
ip address 10.80.18.11 255.255.255.0
negotiation auto
redundancy rii 2
redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/1/0
description CUBE HA
ip address 10.89.20.10 255.255.255.0
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 GigabitEthernet0/0/0
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.64.205.0 255.255.255.0 10.80.18.1
```




```
ip route 10.80.0.0 255.255.0.0 10.80.18.1
!
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 1 voip
description Incoming call from CUCM
session protocol sipv2
incoming called-number .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 protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback cisco
no vad
!
dial-peer voice 2 voip
description Outgoing to Sprint
destination-pattern .T
session protocol sipv2
```



```
session target ipv4:199.11.104.70:5060
voice-class codec 1
voice-class sip options-ping 60
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 protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback cisco
no vad
!
dial-peer voice 3 voip
description Incoming call from Sprint
session protocol sipv2
incoming called-number 913827....
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 protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback cisco
no vad
!
dial-peer voice 4 voip
description Outgoing to CUCM
destination-pattern 913827....
session protocol sipv2
session target ipv4:10.80.18.2:5060
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 protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback cisco
no vad
!
line con 0
exec-timeout 0 0
password 7
login
transport input none
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 0 0
password 7
login
!
ntp server 34.202.XXX.XXX
ntp server pool.ntp.org
!
End
```



Configuring Cisco UCM 11.5 Cluster

Cisco Unified CM Version

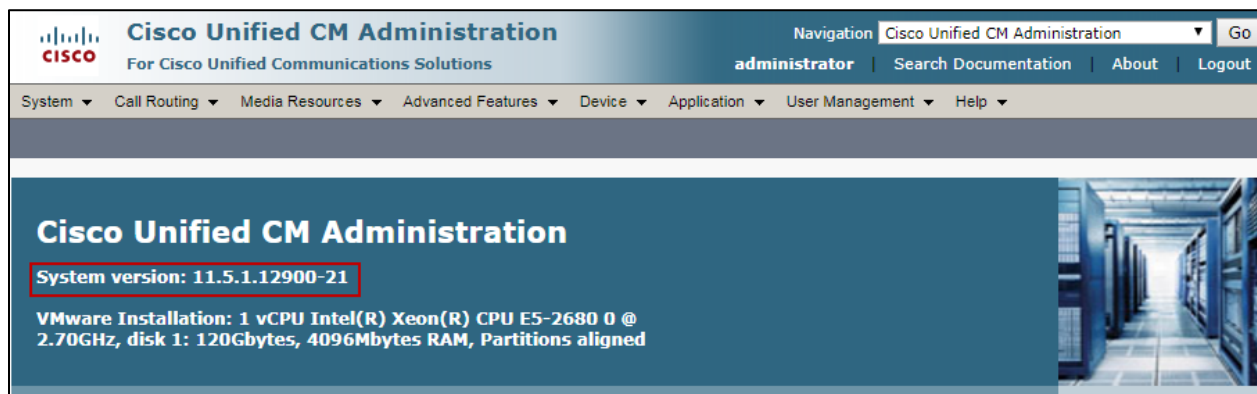


Figure 3: Cisco UCM Version



Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

1. Select **Server***: Clus28Sub1--CUCM Voice/Video (Active)
2. Select **Service***: Cisco CallManager (Active)
3. All other fields are set to default values

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration
administrator | Search Documentation | About | Log

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Help ▾

Service Parameter Configuration Related Links: Parameters for All Servers ▾ Go

Save Set to Default Advanced

Status
Status: Ready

Select Server and Service
Server* Clus28Sub1--CUCM Voice/Video (Active) ▾
Service* Cisco CallManager (Active) ▾
All parameters apply only to the current server except parameters that are in the cluster-wide group(s).

Cisco CallManager (Active) Parameters on server Clus28Sub1--CUCM Voice/Video (Active)

Parameter Name	Parameter Value	Suggested Value
Call Throttling		
Code Yellow Entry Latency *	20	20
Code Yellow Exit Latency Calculation *	40	40
Code Yellow Duration *	5	5
Max Events Allowed *	2000	2000
System Throttle Sample Size *	10	10

Figure 4: Service Parameters



Clusterwide Parameters (Service)		
Default Network Hold MOH Audio Source ID *	1	1
Default User Hold MOH Audio Source ID *	1	1
Duplex Streaming Enabled *	True	False
Media Exchange Interface Capability Timer *	8	8
Send Multicast MOH in H.245 OLC Message *	True	True
Media Exchange Timer *	12	12
Media Exchange Stop Streaming Timer *	8	8
Open Video Channel Response Timer for SIP Interop *	500	500
Port Received Timer After Call Connection *	500	500
Media Resource Allocation Timer *	12	12
MTP and Transcoder Resource Throttling Percentage *	95	95
Intercluster Capabilities Mismatch Timer *	1000	1000
Silence Suppression *	True	False
Silence Suppression for Gateways *	True	False
Strip G.729 Annex B (Silence Suppression) from Capabilities *	True	False
Enable Source IP Address Verification for Software Media Devices *	True	True

Figure 5: Service Parameters (Cont.)



Off-Net Calls via Sprint SIP Trunk

Off-net calls are served by SIP trunks configured between CUCM and the Sprint network and calls are routed via CUBE

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

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

SIP Trunk Security Profile Information

Name* Non Secure SIP Trunk Profile

Description Non Secure SIP Trunk Profile authenticated by null String

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 6: SIP Trunk Security Profile

Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to Sprint 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***: Standard SIP Profile
2. **Description**: Default SIP Profile

SIP Profile Information	
Name*	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-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	< None >
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 7: 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							

Figure 8: SIP Profile (Cont.)



Incoming Requests FROM URI Settings	
Caller ID DN	<input type="text"/>
Caller Name	<input type="text"/>

Trunk Specific Configuration	
Reroute Incoming Request to new Trunk based on*	<input type="text" value="Never"/>
Resource Priority Namespace List	<input type="text" value="< None >"/>
SIP Rel1XX Options*	<input type="text" value="Disabled"/>
Video Call Traffic Class*	<input type="text" value="Mixed"/>
Calling Line Identification Presentation*	<input type="text" value="Default"/>
Session Refresh Method*	<input type="text" value="Invite"/>
Early Offer support for voice and video calls*	<input type="text" value="Disabled (Default value)"/>
<input type="checkbox"/> Enable ANAT	
<input type="checkbox"/> Deliver Conference Bridge Identifier	
<input type="checkbox"/> Allow Passthrough of Configured Line Device Caller Information	
<input type="checkbox"/> Reject Anonymous Incoming Calls	
<input type="checkbox"/> Reject Anonymous Outgoing Calls	
<input type="checkbox"/> Send ILS Learned Destination Route String	
<input type="checkbox"/> Connect Inbound Call before Playing Queuing Announcement	

SIP OPTIONS Ping	
<input type="checkbox"/> 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)*	<input type="text" value="60"/>
Ping Interval for Out-of-service Trunks (seconds)*	<input type="text" value="120"/>
Ping Retry Timer (milliseconds)*	<input type="text" value="500"/>
Ping Retry Count*	<input type="text" value="6"/>

SDP Information	
<input type="checkbox"/> Send send-receive SDP in mid-call INVITE	
<input type="checkbox"/> Allow Presentation Sharing using BFCP	
<input type="checkbox"/> Allow iX Application Media	
<input type="checkbox"/> Allow multiple codecs in answer SDP	

Figure 9: SIP Profile (Cont.)



SIP Trunk Configuration

Create SIP trunks to CUBE

Navigation: Device → Trunk

Device Information	
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Sprint
Description	Sprint SIP Trunk certification
Device Pool*	G729_Sprint_Pool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_Default
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
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end protection. 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
<input type="checkbox"/> PSTN Access	
<input type="checkbox"/> 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 >

Figure 10: SIP Trunk to CUBE



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
1*	10.80.18.10		5060

MTP Preferred Originating Codec* 711ulaw

BLF Presence Group* Standard Presence group

SIP Trunk Security Profile* Sprint

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* RFC 2833

Figure 11: SIP Trunk to CUBE (Cont.)



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 12: SIP Trunk to CUBE (Cont.)

Explanation

Parameter	Value	Description
Device Name	Sprint	Name for the trunk
Device Pool	G729_Sprint_pool	Default Device Pool is used for this trunk
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.11.10	IP address of the CUBE Virtual LAN
SIP Trunk Security Profile	Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	Standard SIP Profile	SIP Profile configured earlier



Dial Plan

Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

- Cisco IP phone dial “71” 10 digits number to access PSTN via CUBE
 - “71” is removed before sending to CUBE
 - “+1” is added to the calling number for outbound calls
 - The rest of the number is sent to CUBE to Sprint network
- Incoming fax call to 1491 will be sent to Cisco Fax ATA

Pattern Definition	
Route Pattern*	71.@
Route Partition	< None >
Description	to Sprint
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*	Sprint
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"/> Urgent Priority	<input type="checkbox"/> Require Client Matter Code

Figure 13: Route Pattern for Voice



Calling Party Transformations	
<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	<input type="text"/>
Prefix Digits (Outgoing Calls)	<input type="text" value="+1"/>
Calling Line ID Presentation*	<input type="text" value="Default"/>
Calling Name Presentation*	<input type="text" value="Default"/>
Calling Party Number Type*	<input type="text" value="Cisco CallManager"/>
Calling Party Numbering Plan*	<input type="text" value="Cisco CallManager"/>
Connected Party Transformations	
Connected Line ID Presentation*	<input type="text" value="Default"/>
Connected Name Presentation*	<input type="text" value="Default"/>
Called Party Transformations	
Discard Digits	<input type="text" value="PreDot"/>
Called Party Transform Mask	<input type="text"/>
Prefix Digits (Outgoing Calls)	<input type="text"/>
Called Party Number Type*	<input type="text" value="Cisco CallManager"/>
Called Party Numbering Plan*	<input type="text" value="Cisco CallManager"/>
ISDN Network-Specific Facilities Information Element	
Network Service Protocol	<input type="text" value="-- Not Selected --"/>
Carrier Identification Code	<input type="text"/>
Network Service	Service Parameter Name
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>

Figure 14: Route Pattern for Voice (Cont.)



Pattern Definition	
Route Pattern*	1491
Route Partition	< None >
Description	
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*	CiscoATA
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

Figure 15: Route Pattern for Voice (Cont.)



-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
-- Not Selected -- ▼	< Not Exist >

Figure 16: Route Pattern for Voice (Cont.)

Explanation

Setting	Value	Description
Route Pattern	71.@ for Voice & International Calls and 1491 for Fax Call	Specify appropriate Route Pattern
Gateway/Route List	Sprint for Route Pattern 71.@ and 8021 for SIP Trunk To Fax ATA	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 71.@	North American Numbering Plan
Call Classification	OffNet for Route Pattern 71.@ and 1491	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 71.@	Specifies how to modify digit before they are sent to Sprint network
Prefix Digits (Outgoing Call)	" +1 " is added as prefix from calling party number to convert to E.164 format.	



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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