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
# Table of Contents

- **Introduction** ................................................................. 4
- **Network Topology** .......................................................... 5
- **System Components** ......................................................... 5
  - **Hardware Requirements** .................................................. 5
  - **Software Requirements** .................................................. 5
- **Features** ............................................................................. 6
  - **Features Supported** .......................................................... 6
  - **Features Not Supported** .................................................... 6
  - **Caveats** ............................................................................ 6
- **Configuration** ..................................................................... 7
  - **Configuring Cisco Unified Border Element (CUBE)** ............. 7
    - **Network Interface** ......................................................... 7
    - **Global CUBE Settings** ................................................... 10
    - **Codecs** ......................................................................... 10
    - **Dial peer** ....................................................................... 11
    - **Configuration example** .................................................. 13
  - **Configuring Cisco UCM 11.5 Cluster** ............................... 20
    - **Cisco Unified CM Version** .............................................. 20
    - **Cisco Call Manager Service Parameters** ............................ 21
    - **Off-Net Calls via Sprint SIP Trunk** .................................. 23
    - **Dial Plan** ....................................................................... 30
- **Acronyms** ........................................................................... 34
- **Important Information** ....................................................... 35
### Table of Figures

- **Figure 1:** Network Topology ........................................................................................................ 5
- **Figure 2:** High Availability Topology ........................................................................................... 7
- **Figure 3:** Cisco UCM Version ........................................................................................................ 20
- **Figure 4:** Service Parameters ....................................................................................................... 21
- **Figure 5:** Service Parameters (Cont.) ........................................................................................... 22
- **Figure 6:** SIP Trunk Security Profile ............................................................................................... 23
- **Figure 7:** SIP Profile .................................................................................................................... 24
- **Figure 8:** SIP Profile (Cont.) ........................................................................................................ 25
- **Figure 9:** SIP Profile (Cont.) ........................................................................................................ 26
- **Figure 10:** SIP Trunk to CUBE ...................................................................................................... 27
- **Figure 11:** SIP Trunk to CUBE (Cont.) ........................................................................................... 28
- **Figure 12:** SIP Trunk to CUBE (Cont.) ........................................................................................... 29
- **Figure 13:** Route Pattern for Voice ................................................................................................. 30
- **Figure 14:** Route Pattern for Voice (Cont.) .................................................................................... 31
- **Figure 15:** Route Pattern for Voice (Cont.) .................................................................................... 32
- **Figure 16:** Route Pattern for Voice (Cont.) .................................................................................... 33
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:

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.
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

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>redundancy-group 1</td>
<td>Enable High Availability for the VoIP service</td>
</tr>
<tr>
<td>fax protocol</td>
<td>Specifies the fax protocol</td>
</tr>
</tbody>
</table>

Codecs
G729 is used primarily towards Sprint until specified otherwise.

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

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Page 10 of 36
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
!
vrfs 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
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-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

![Cisco Unified CM Version](image)

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](image)

**Figure 4: Service Parameters**
### Clusterwide Parameters (Service)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default Network Hold MOH Audio Source ID</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Default User Hold MOH Audio Source ID</td>
<td>1</td>
<td></td>
</tr>
<tr>
<td><strong>Duplex Streaming Enabled</strong></td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Media Exchange Interface Capability Timer</td>
<td>8</td>
<td>8</td>
</tr>
<tr>
<td>Send Multicast MOH in H.245 DLC Message</td>
<td>True</td>
<td></td>
</tr>
<tr>
<td>Media Exchange Timer</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>Media Exchange Stop Streaming Timer</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>Open Video Channel Response Timer for SIP Interop</td>
<td>900</td>
<td>900</td>
</tr>
<tr>
<td>Port Received Timer After Call Connection</td>
<td>500</td>
<td></td>
</tr>
<tr>
<td>Media Resource Allocation Timer</td>
<td>500</td>
<td></td>
</tr>
<tr>
<td>MTP and Transcoder Resource Throttling Percentage</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>Intercluster Capabilities Mismatch Timer</td>
<td>1000</td>
<td></td>
</tr>
<tr>
<td>Silence Suppression</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Silence Suppression for Gateways</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Strip G.729 Annex B (Silence Suppression) from Capabilities</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Enable Source IP Address Verification for Software Media Devices</td>
<td>True</td>
<td>True</td>
</tr>
</tbody>
</table>

**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

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Transport Type</td>
<td>TCP + UDP</td>
<td>SIP trunks to 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.</td>
</tr>
<tr>
<td>Outgoing Transport Type</td>
<td>UDP</td>
<td></td>
</tr>
</tbody>
</table>
**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](image)

**Figure 7: SIP Profile**
Figure 8: SIP Profile (Cont.)

<table>
<thead>
<tr>
<th>Configuration Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>DSCP for Audio Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>x-cisco-serviceun-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI</td>
<td>x-cisco-serviceuri-opickup</td>
</tr>
<tr>
<td>Call Pickup Group URI*</td>
<td>x-cisco-serviceuri-gpickup</td>
</tr>
<tr>
<td>Meet Me Service URI*</td>
<td>x-cisco-serviceuri-meetme</td>
</tr>
<tr>
<td>User Info*</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level</td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control</td>
<td>User</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7950*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Resource Priority Namespace</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirections</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI*</td>
<td>x-cisco-serviceun-cfwdail</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI*</td>
<td>x-cisco-serviceun-abbrdial</td>
</tr>
<tr>
<td>Conference Join Enabled</td>
<td>True</td>
</tr>
<tr>
<td>RFC 2543 Hold</td>
<td>True</td>
</tr>
<tr>
<td>Semi Attended Transfer</td>
<td>True</td>
</tr>
<tr>
<td>Enable VAD</td>
<td>True</td>
</tr>
<tr>
<td>Stutter Message Waiting</td>
<td>True</td>
</tr>
<tr>
<td>MLPP User Authorization</td>
<td>True</td>
</tr>
</tbody>
</table>

Normalization Script

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td></td>
</tr>
</tbody>
</table>
## Incoming Requests FROM URI Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller ID DN</td>
<td></td>
</tr>
<tr>
<td>Caller Name</td>
<td></td>
</tr>
</tbody>
</table>

## Trunk Specific Configuration

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Reroute Incoming Request to new Trunk based on</td>
<td>Never</td>
</tr>
<tr>
<td>Resource Priority Namespace List</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SIP Rel1XX Options</td>
<td>Disabled</td>
</tr>
<tr>
<td>Video Call Traffic Class</td>
<td>Mixed</td>
</tr>
<tr>
<td>Calling Line Identification Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Session Refresh Method</td>
<td>Invite</td>
</tr>
<tr>
<td>Early Offer support for voice and video calls</td>
<td>Disabled (Default value)</td>
</tr>
<tr>
<td>Enable ANAT</td>
<td></td>
</tr>
<tr>
<td>Deliver Conference Bridge Identifier</td>
<td></td>
</tr>
<tr>
<td>Allow Passthrough of Configured Line Device Caller Information</td>
<td></td>
</tr>
<tr>
<td>Reject Anonymous Incoming Calls</td>
<td></td>
</tr>
<tr>
<td>Reject Anonymous Outgoing Calls</td>
<td></td>
</tr>
<tr>
<td>Send ILS Learned Destination Route String</td>
<td></td>
</tr>
<tr>
<td>Connect Inbound Call before Playing Queuing Announcement</td>
<td></td>
</tr>
</tbody>
</table>

## SIP OPTIONS Ping

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable OPTIONS Ping to monitor destination status for Trunks with Service Type &quot;None (Default)&quot;</td>
<td></td>
</tr>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)</td>
<td>60</td>
</tr>
<tr>
<td>Ping Interval for Out-of-service Trunks (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Ping Retry Timer (milliseconds)</td>
<td>500</td>
</tr>
<tr>
<td>Ping Retry Count</td>
<td>6</td>
</tr>
</tbody>
</table>

## SDP Information

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Send send-receive SDP in mid-call INVITE</td>
<td></td>
</tr>
<tr>
<td>Allow Presentation Sharing using BFCP</td>
<td></td>
</tr>
<tr>
<td>Allow iX Application Media</td>
<td></td>
</tr>
<tr>
<td>Allow multiple codecs in answer SDP</td>
<td></td>
</tr>
</tbody>
</table>

---

Figure 9: SIP Profile (Cont.)
### SIP Trunk Configuration
Create SIP trunks to CUBE

**Navigation:** Device → Trunk

<table>
<thead>
<tr>
<th>Device Information</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Product</td>
<td>SIP Trunk</td>
</tr>
<tr>
<td>Device Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Trunk Service Type</td>
<td>None (Default)</td>
</tr>
<tr>
<td><strong>Device Name</strong></td>
<td>Sprint</td>
</tr>
<tr>
<td><strong>Description</strong></td>
<td>Sprint SIP Trunk certification</td>
</tr>
<tr>
<td><strong>Device Pool</strong></td>
<td>G729_Sprint_Pool</td>
</tr>
</tbody>
</table>

**Common Device Configuration**
- 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
- 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>

---

Figure 10: SIP Trunk to CUBE

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Page 27 of 36
### Outbound Calls

- **Called Party Transformation CSS**: < None >
- **Use Device Pool Called Party Transformation CSS**: < None >
- **Calling Party Transformation CSS**: < None >
- **Use Device Pool Calling Party Transformation CSS**: < None >
- **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**: < None >
- **Redirecting Party Transformation CSS**: < None >

### Caller Information

- **Caller ID DN**
- **Caller Name**

- **Maintain Original Caller ID DN and Caller Name in Identity Headers**

---

### SIP Information

**Destination**

- **Destination Address**: 10.80.18.10
- **Destination Address IPv6**: 
- **Destination Port**: 5060

- **HTTP Preferred Originating Codec**: 711ulew
- **SLF Presence Group**: Standard Presence Group
- **SIP Trunk Security Profile**: Sprint
- **Redirecting Calling Search Space**: < None >
- **Out-Of-Dialing Refer Calling Search Space**: < None >
- **SUBSCRIBE Calling Search Space**: < None >

### SIP Profile

- **SIP Profile**: Standard SIP Profile

### DTMF Signaling Method

- **RFC 2833**

---

**Figure 11**: SIP Trunk to CUBE (Cont.)
Figure 12: SIP Trunk to CUBE (Cont.)

Explanation

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Name</td>
<td>Sprint</td>
<td>Name for the trunk</td>
</tr>
<tr>
<td>Device Pool</td>
<td>G729_Sprint_pool</td>
<td>Default Device Pool is used for this trunk</td>
</tr>
<tr>
<td>Significant Digits</td>
<td>4</td>
<td>4 digits Extension for all CPE phones</td>
</tr>
<tr>
<td>Destination Address</td>
<td>10.80.11.10</td>
<td>IP address of the CUBE Virtual LAN</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
<td>Non Secure SIP Trunk Profile</td>
<td>SIP Trunk Security Profile configured earlier</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>Standard SIP Profile</td>
<td>SIP Profile configured earlier</td>
</tr>
</tbody>
</table>
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 Table](image)

Figure 13: Route Pattern for Voice
### Calling Party Transformations

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use Calling Party’s External Phone Number Mask</td>
<td>Yes</td>
</tr>
<tr>
<td>Calling Party Transform Mask</td>
<td></td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td>+1</td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Name Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Party Number Type</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Calling Party Numbering Plan</td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

### Connected Party Transformations

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td>Default</td>
</tr>
</tbody>
</table>

### Called Party Transformations

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Discard Digits</td>
<td>PreDot</td>
</tr>
<tr>
<td>Called Party Transform Mask</td>
<td></td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td></td>
</tr>
<tr>
<td>Called Party Number Type</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Called Party Numbering Plan</td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

### ISDN Network Specific Facilities Information Element

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Network Service Protocol</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td>Carrier Identification Code</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Network Service</th>
<th>Service Parameter Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>-- Not Selected --</td>
<td>&lt; Not Exist &gt;</td>
</tr>
</tbody>
</table>
**Figure 15: Route Pattern for Voice (Cont.)**

<table>
<thead>
<tr>
<th>Pattern Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Route Pattern</strong></td>
</tr>
<tr>
<td><strong>Route Partition</strong></td>
</tr>
<tr>
<td><strong>Description</strong></td>
</tr>
<tr>
<td><strong>Numbering Plan</strong></td>
</tr>
<tr>
<td><strong>Route Filter</strong></td>
</tr>
<tr>
<td><strong>MLPP Precedence</strong></td>
</tr>
<tr>
<td><strong>Apply Call Blocking Percentage</strong></td>
</tr>
<tr>
<td><strong>Resource Priority Namespace Network Domain</strong></td>
</tr>
<tr>
<td><strong>Route Class</strong></td>
</tr>
<tr>
<td><strong>Gateway/Route List</strong></td>
</tr>
<tr>
<td><strong>Route Option</strong></td>
</tr>
<tr>
<td><strong>Call Classification</strong></td>
</tr>
<tr>
<td><strong>External Call Control Profile</strong></td>
</tr>
<tr>
<td><strong>Allow Device Override</strong></td>
</tr>
<tr>
<td><strong>Provide Outside Dial Tone</strong></td>
</tr>
<tr>
<td><strong>Allow Overlap Sending</strong></td>
</tr>
<tr>
<td><strong>Urgent Priority</strong></td>
</tr>
<tr>
<td><strong>Require Forced Authorization Code</strong></td>
</tr>
<tr>
<td><strong>Authorization Level</strong></td>
</tr>
</tbody>
</table>

**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 |
Figure 16: Route Pattern for Voice (Cont.)

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

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