BroadCloud SIP Trunking:

Cisco Unified Communications Manager 11.0.1 with Cisco Unified Border Element (CUBE 11.5.0) on ISR4321/K9 [IOS-XE 3.17.1 – 15.6(1)S1] using SIP

August 29, 2016
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Introduction
Service Providers today, such as BroadCloud, are offering alternative methods to connect to the PSTN via their IP network. 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.

BroadCloud is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity. A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and BroadCloud network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS-XE 3.17.1 – 15.6(1)S1 can be used. The Cisco Unified Border Element provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.0.1 connected to BroadCloud IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco Unified Communications Manager. Only configuration settings specifically required for BroadCloud 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 Cisco Unified Communications Manager (Cisco UCM) 11.0.1, and Cisco Unified Border Element (Cisco UBE) on ISR 4321/K9 [IOS-XE 3.17.1 – 15.6(1)S1] for connectivity to BroadCloud SIP Trunking service. The deployment model covered in this application note is CPE (Cisco Unified Communications Manager 11.0.1) to PSTN (BroadCloud).

- Testing was performed in accordance to BroadCloud 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 BroadCloud SIP network and Cisco Unified Communications. The configuration described in this document details the important configuration settings to have enabled for interoperability to be successful and care must be taken by the network administrator deploying Cisco UCM to interoperate to BroadCloud SIP Trunking network.

This application note does not cover the use of Calling Search Spaces (CSS) or partitions on Cisco UCM. To understand and learn how to apply CSS and partitions refer to the cisco.com link below:


Network Topology
System Components

Hardware Requirements
Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
Cisco ISR 4321/K9 router as CUBE
Cisco ISR4321/K9 (1RU) processor with 1647061K/6147K bytes of memory with 3 Gigabit Ethernet interfaces
Processor board ID FLM1925W0WZ
Cisco 2851 Fax Gateway
IP phones 7965 (SIP), 7975 (SCCP)

Software Requirements

- Cisco Unified Communications Manager 11.0.1.21900-11
- Cisco Unity Connection 11.0.1.21900-11
- IOS-XE 15.6(1)S1 for ISR 4321/K9 Cisco Unified Border Element
- Cisco IOS Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.6(1)S1, RELEASE SOFTWARE (fc3)
- Cisco IOS XE Software, Version 03.17.01.S
- IOS 15.0(1)XA for Cisco 2851 Fax Gateway

Features

Features Supported

- Incoming and outgoing off-net calls using G711ULAW & G729 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 (T38 & G711 pass-through)

### Features Not Supported
- 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 semi-attended 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.
- BroadCloud does not support faxes using SuperG3. However, fax test cases are executed using G3.
- 911 Calls were not tested since BroadCloud lab environment does not have the ability to route to a 911 PSAP calls
- Early media call that requires PRACK with SDP Failed with enabling "require100rel" since BroadCloud does not support "require100rel". However, the call was successful without PRACK.

### Configuration

Configuring the Cisco Unified Border Element for Registration SIP Trunk Testing
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—LAN and WAN.

```plaintext
interface GigabitEthernet0/0/0
  description bcloud LAN
  ip address 10.80.13.10 255.255.255.0
  media-type rj45
  negotiation auto
  redundancy rii 6
  redundancy group 2 ip 10.80.13.15 exclusive
!
interface GigabitEthernet0/0/1
  description bcloud WAN
  ip address 192.XX.XX.XX 255.255.255.128
  negotiation auto
  redundancy rii 7
  redundancy group 2 ip 192.XX.XX.XX exclusive
!
```
Global Cisco UBE settings

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

```
! 
voice service voip 
ip address trusted list 
ipv4 199.XX.XX.XX 
ipv4 199.XX.XX.XX 
address-hiding 
mode border-element license capacity 20 
allow-connections sip to sip 
redundancy-group 2 
sip 
bind control source-interface GigabitEthernet0/0/1 
rel1xx supported "rel100" 
session refresh 
asserted-id pai 
privacy pstn 
outbound-proxy dns:XXX@domain.com 
early-offer forced 
midcall-signaling passthru 
g729 annexb-all 
!
```

### Explanation

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>asserted-id</td>
<td>Specifies the type of privacy header in the outgoing SIP requests and response messages</td>
</tr>
<tr>
<td>early-offer forced</td>
<td>Enables SIP Delayed-Offer to Early-Offer globally</td>
</tr>
<tr>
<td>midcall-signaling passthru</td>
<td>Passes SIP messages from one IP leg to another IP leg</td>
</tr>
</tbody>
</table>
Codecs

G711ulaw and G729 voice codecs are used for this testing. Codec preferences used to change according to the test plan description

voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  codec preference 3 g722-56

Dial peer
Cisco UBE uses dial-peer to route the call based on the digit to route the call accordingly.

dial-peer voice 500 voip
description Outgoing call to Broadcloud - LAN facing
  huntstop
  session protocol sipv2
  session transport udp
  incoming called-number XXXXXX...
  voice-class codec 1
  voice-class sip asserted-id pai
  no voice-class sip outbound-proxy
  voice-class sip bind control source-interface GigabitEthernet0/0/0
  voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
  no vad
!
dial-peer voice 510 voip
description Outgoing call to Broadcloud - WAN facing
huntstop
destination-pattern XXXXXX....
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip outbound-proxy dns:XXX@domain.com
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 520 voip
description Incoming call to PBX - WAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number XXXXXX....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 530 voip
description Incoming call to PBX - LAN facing
huntstop
destination-pattern Xxxxxx....
session protocol sipv2
session target ipv4:10.80.13.2:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
no voice-class sip outbound-proxy
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
Configuration example

The following configuration snippet contains a sample configuration of Cisco Unified Border Element with all parameters mentioned previously.

Active Cisco UBE

bcloud1#sh run

version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname bcloud1
!
boot-start-marker
boot system flash bootflash:isr4300-universalk9.03.17.01.S.156-1.S1-std.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no aaa new-model
!
ip name-server 8.26.56.xx
subscriber templating

multilink bundle-name authenticated

voice service voip

ip address trusted list
ipv4 199.xx.xx.xx
ipv4 199.xx.xx.xx

address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 2

sip
bind control source-interface GigabitEthernet0/0/1
rel1xx supported "rel100"

session refresh
asserted-id pai
outbound-proxy dns:XXX@domain.com

early-offer forced
midcall-signaling passthru
g729 annexb-all

voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
codec preference 3 g722-56

voice class codec 2
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g722-56

voice class sip-profiles 101
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:XXXXXX\1@2"

voice translation-rule 1
  rule 1 /^.(.*)\(.*)$/ /1/

voice translation-profile BroadCloud
translate called 1

license udi pid ISR4321/K9 sn FDO19220MQ8
license boot level appxk9
license boot level uck9

spanning-tree extend system-id

redundancy
  mode none
  application redundancy
    group 2
      name b2bhaBcloud
      priority 100 failover threshold 75
      timers delay 30 reload 60
      control GigabitEthernet0/1/0 protocol 1
      data GigabitEthernet0/1/0
      track 1 shutdown
      track 2 shutdown
vlan internal allocation policy ascending

track 1 interface GigabitEthernet0/0/0 line-protocol

track 2 interface GigabitEthernet0/0/1 line-protocol

interface GigabitEthernet0/0/0
    description bcloud LAN
    ip address 10.80.13.10 255.255.255.0
    media-type rj45
    negotiation auto
    redundancy rii 6
    redundancy group 2 ip 10.80.13.15 exclusive

interface GigabitEthernet0/0/1
    description bcloud WAN
    ip address 192.XX.XX.XX 255.255.255.128
    negotiation auto
    redundancy rii 7
    redundancy group 2 ip 192.XX.XX.XX exclusive

interface GigabitEthernet0/1/0
    description CUBE HA
    ip address 10.89.20.100 255.255.255.0
    negotiation auto

interface GigabitEthernet0
    vrf forwarding Mgmt-intf
    no ip address
    negotiation auto
! 
interface Vlan1
no ip address
shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0/0/1
ip route 0.0.0.0 0.0.0.0 192.XX.XX.XX
ip route 10.80.0.0 255.255.0.0 10.80.13.1
ip route 172.16.0.0 255.255.0.0 10.80.13.1
!
control-plane
!
mgcp behavior rsip-range tgcponly
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 500 voip
description Outgoing call to Broadcloud - LAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number XXXXXX....
voice-class codec 1
voice-class sip asserted-id pai
no voice-class sip outbound-proxy
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 510 voip
description Outgoing call to Broadcloud - WAN facing
huntstop
destination-pattern XXXXXX....
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip outbound-proxy dns:XXX@domain.com
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 520 voip
description Incoming call to PBX - WAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number XXXXXX....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 530 voip
description Incoming call to PBX - LAN facing
huntstop
destination-pattern XXXXXX....
session protocol sipv2
session target ipv4:10.80.13.2:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
no voice-class sip outbound-proxy
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 600 voip
description PBX to PBX Outgoing Call - LAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number 1XXXXXXX....
voice-class codec 1
voice-class sip asserted-id pai
no voice-class sip outbound-proxy
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n-te
no vad
!
dial-peer voice 610 voip
description PBX to PBX Outgoing Call - WAN facing
translation-profile outgoing BroadCloud
huntstop
destination-pattern 1XXXXXX....
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip outbound-proxy dns:XXX@domain.com
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-n-te
no vad
!
sip-ua
credentials number XXXXXXXXXX username XXXXXXXXXX password 7 realm XXX@customerdomain.com
authentication username XXXXXXXXXX password 7
no remote-party-id
retry invite 2
timers keepalive active 10
registrar dns:XXX@customerdomain.com:8933 expires 60
sip-server dns:XXX@customerdomain.com:8933
!
!
line con 0
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 0 0
  password XXXXXX
  login
!
end
Standby Cisco UBE

bcloud2#sh run

version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname bcloud2
!
boot-start-marker
boot system flash bootflash:isr4300-universalk9.03.17.01.S.156-1.S1-std.SPA.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no aaa new-model
!
ip name-server 8.26.56.xx
!
subscriber templating
multilink bundle-name authenticated
! voice service voip
ip address trusted list
ipv4 199.XX.XX.XX
ipv4 199.XX.XX.XX
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 2
sip
bind control source-interface GigabitEthernet0/0/1
rel1x supported "rel100"
session refresh
asserted-id pai
outbound-proxy dns:XXX@domain.com
early-offer forced
midcall-signaling passthru
g729 annexb-all
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
codec preference 3 g722-56
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g722-56
!
!
voice class sip-profiles 101
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:XXXXXXX\1@\2"

! 
voice translation-rule 1
rule 1 /^\.*\((\........\))/ /1/

! 
voice translation-profile BroadCloud
translate called 1

! 
license udi pid ISR4321/K9 sn FDO19220MQ9

! 
spanning-tree extend system-id

! 
redundancy
mode none
application redundancy
group 2
name b2bhaBcloud
priority 100 failover threshold 75
timers delay 30 reload 60
control GigabitEthernet0/1/0 protocol 1
data GigabitEthernet0/1/0
track 1 shutdown
track 2 shutdown

! 
vlan internal allocation policy ascending

! 
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol

!
interface GigabitEthernet0/0/0
  description bcloud LAN
  ip address 10.80.13.11 255.255.255.0
  media-type rj45
  negotiation auto
  redundancy rii 6
  redundancy group 2 ip 10.80.13.15 exclusive
!
interface GigabitEthernet0/0/1
  description bcloud WAN
  ip address 192.XX.XX.XX 255.255.255.128
  negotiation auto
  redundancy rii 7
  redundancy group 2 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/1/0
  description CUBE HA
  ip address 10.89.20.101 255.255.255.0
  negotiation auto
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  negotiation auto
!
interface Vlan1
  no ip address
  shutdown
!
  ip forward-protocol nd
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.XX.XX.XX
ip route 10.80.0.0 255.255.0.0 10.80.13.1
ip route 172.16.0.0 255.255.0.0 10.80.13.1
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 500 voip
  description Outgoing call to Broadcloud - LAN facing
  huntstop
  session protocol sipv2
  session transport udp
  incoming called-number XXXXXX....
  voice-class codec 1
  voice-class sip asserted-id pai
  no voice-class sip outbound-proxy
  voice-class sip bind control source-interface GigabitEthernet0/0/0
  voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n-te
  no vad
!

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dial-peer voice 510 voip
  description Outgoing call to Broadcloud - WAN facing
  huntstop
  destination-pattern XXXXXX....
  session protocol sipv2
  session target sip-server
  session transport udp
  voice-class codec 1
  voice-class sip asserted-id pai
  voice-class sip outbound-proxy dns:XXX@domain.com
  voice-class sip profiles 101
  voice-class sip bind control source-interface GigabitEthernet0/0/1
  voice-class sip bind media source-interface GigabitEthernet0/0/1
  dtmf-relay rtp-nte
  no vad
!
dial-peer voice 520 voip
  description Incoming call to PBX - WAN facing
  huntstop
  session protocol sipv2
  session transport udp
  incoming called-number XXXXXX....
  voice-class codec 1
  voice-class sip asserted-id pai
  voice-class sip options-keepalive
  voice-class sip bind control source-interface GigabitEthernet0/0/1
  voice-class sip bind media source-interface GigabitEthernet0/0/1
  dtmf-relay rtp-nte
  no vad
!
dial-peer voice 530 voip
description Incoming call to PBX - LAN facing
huntstop
destination-pattern XXXXXX....
session protocol sipv2
session target ipv4:10.80.13.2:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
no voice-class sip outbound-proxy
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 600 voip
description PBX to PBX Outgoing Call - LAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number 1XXXXXX....
voice-class codec 1
voice-class sip asserted-id pai
no voice-class sip outbound-proxy
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 610 voip
description PBX to PBX Outgoing Call - WAN facing
translation-profile outgoing BroadCloud
huntstop
destination-pattern 1XXXXX....
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip outbound-proxy dns:XXX@domain.com
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
no vad
!
!
sip-ua
credentials number XXXXXXXXX username XXXXXXXXXX password 7 realm XXX@customerdomain.com
authentication username XXXXXXXXXX password 7
no remote-party-id
retry invite 2
timers keepalive active 180
registrar dns:XXX@customerdomain.com:8933 expires 60
sip-server dns:XXX@customerdomain.com:8933
!
line con 0
exec-timeout 0 0
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 0 0
password XXXXXXX
login
!
!
end
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 “7” 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 “7”. A “7.@” 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 BroadCloud, Caller dial 7 prefix followed by the target 10-digits number, 7 was stripped and the remaining digits were send to Cisco UBE, Cisco UBE pass the DID under Dial Peer 510 and send to BroadCloud network which will direct back to Cisco UBE and handled same as normal incoming PSTN call.

![Figure 3: Outbound Voice Call](image)

![Figure 4: Outbound Fax Call](image)
Figure 5: Inbound Voice Call

Figure 6: Inbound Fax Call

Figure 7: PBX to PBX via BroadCloud Call
Configuring Cisco Unified Communications Manager

Cisco UCM Version

Figure 8: Cisco UCM Version

Cisco Call manager Service Parameters

Navigation: System → Service Parameters

1. Select Server* = clus23pubsub--CUCM Voice/Video (Active)
2. Select Service *= Cisco CallManager (Active)
3. All other fields are set to default values

Figure 9: Service Parameters
Offnet Calls via BroadCloud SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and BroadCloud Network and calls are routed via Cisco UBE

**SIP Trunk Security Profile**

**Navigation:** System → Security → SIP Trunk Security Profile

1. **Name** = *BroadCloud Non Secure SIP Trunk Profile* is used as an example
2. **Description** = *Non Secure non secure trunk profile* is used as an example

![SIP Trunk Security Profile Information](image)

**Figure 10: SIP Trunk Security Profile**

**Explanation**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Transport Type</td>
<td>TCP + UDP</td>
<td>SIP trunks to BroadCloud 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

NOTE: SIP Profile will be later associated with the SIP trunk

Navigation: Device > Device Settings > SIP Profile

1. Name = BroadCloud SIP Profile is used as an example
2. Description = BroadCloud SIP Profile is used as an example

![SIP Profile Configuration Table]

Figure 11: SIP Profile
### SIP Profile (Cont.)

<table>
<thead>
<tr>
<th>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 URL</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URL</td>
<td>x-cisco-serviceuri-opickup</td>
</tr>
<tr>
<td>Call Pickup Group URL</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 7960</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 URL</td>
<td>x-cisco-serviceuri-cfwcall</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>x-cisco-serviceuri-abrdial</td>
</tr>
</tbody>
</table>

**Figure 12:** SIP Profile (Cont.)
![Figure 13: SIP Profile (Cont.)](image)

**Explanation**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default MTP Telephony Event Payload Type</td>
<td>101</td>
<td>RFC2833 DTMF payload type</td>
</tr>
<tr>
<td>SIP Rel1XX Options</td>
<td>Send PRACK if 1xx Contains SDP</td>
<td>Enable Provisional Acknowledgements (Reliable 100 messages)</td>
</tr>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)</td>
<td>60</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
<tr>
<td>Ping Interval for Out-of-service Trunks (seconds)</td>
<td>120</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
</tbody>
</table>
### SIP Trunk Configuration

Create SIP trunks to Cisco UBE

**Navigation:** Device > Trunk

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Ceiling Search Space</th>
<th>Device Pool</th>
<th>Route Pattern</th>
<th>Partition</th>
<th>Route Group</th>
<th>Priority</th>
<th>Trunk Type</th>
<th>SIP Trunk Status</th>
<th>SIP Trunk Duration</th>
<th>SIP Trunk Security Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>BroadCloud</td>
<td>BroadCloud SIP Trunk</td>
<td></td>
<td>BroadCloud</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Full Service</td>
<td>0 day, 0 hour, 0 minutes</td>
<td>BroadCloud Non-Secure SIP T Profile</td>
</tr>
<tr>
<td>BroadCloud</td>
<td>BroadCloud SIP Trunk</td>
<td></td>
<td>BroadCloud</td>
<td><em>SIP</em></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Full Service</td>
<td>0 day, 0 hour, 0 minutes</td>
<td>BroadCloud Non-Secure SIP T Profile</td>
</tr>
<tr>
<td>SIP_trunk_to_Voice_gateway</td>
<td>SIP_trunk_to_Voice_gateway</td>
<td></td>
<td>BroadCloud</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Unknown - OPTIONS Ping not enabled</td>
<td>0 day, 0 hour, 0 minutes</td>
<td>BroadCloud Non-Secure SIP T Profile</td>
</tr>
<tr>
<td>UnityConnectionTrunk-for-BroadCloud</td>
<td>UnityConnectionTrunk-for-BroadCloud</td>
<td></td>
<td>BroadCloud</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Unknown - OPTIONS Ping not enabled</td>
<td>0 day, 0 hour, 0 minutes</td>
<td>UnityConnectionTrunk Security for BroadCloud</td>
</tr>
</tbody>
</table>

Figure 14: SIP Trunks List
### SIP Trunk Status

**Service Status:** Full Service  
**Duration:** Time In Full Service: 0 day 0 hour 7 minutes

<table>
<thead>
<tr>
<th>Device Information</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Product:</strong> SIP Trunk</td>
</tr>
<tr>
<td><strong>Device Protocol:</strong> SIP</td>
</tr>
<tr>
<td><strong>Trunk Service Type:</strong> None (Default)</td>
</tr>
<tr>
<td><strong>Device Name:</strong> BroadCloud</td>
</tr>
<tr>
<td><strong>Description:</strong> BroadCloud SIP Trunk</td>
</tr>
<tr>
<td><strong>Device Pool:</strong> BroadCloud DevicesPool</td>
</tr>
<tr>
<td><strong>Common Device Configuration:</strong> &lt; None &gt;</td>
</tr>
<tr>
<td><strong>Call Classification:</strong> Use System Default</td>
</tr>
<tr>
<td><strong>Media Resource Group List:</strong> MRQL_MTP</td>
</tr>
<tr>
<td><strong>Location:</strong> Hub_None</td>
</tr>
<tr>
<td><strong>AAR Group:</strong> &lt; None &gt;</td>
</tr>
<tr>
<td><strong>Tunneled Protocol:</strong> None</td>
</tr>
<tr>
<td><strong>QSIG Variant:</strong> No Changes</td>
</tr>
<tr>
<td><strong>ASN.1 ROSE OID Encoding:</strong> No Changes</td>
</tr>
<tr>
<td><strong>Packet Capture Mode:</strong> None</td>
</tr>
<tr>
<td><strong>Packet Capture Duration:</strong> 0</td>
</tr>
</tbody>
</table>

- 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 Fort
- 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**

<table>
<thead>
<tr>
<th>Route Class Signaling Enabled</th>
<th>Use Trusted Relay Point</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Default</td>
</tr>
</tbody>
</table>

**Intercompany Media Engine (IME)**

| E.164 Transformation Profile |  
|------------------------------|---

---

**Figure 15:** SIP Trunk to Cisco UBE
### MLPP and Confidential Access Level Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>MLPP Domain</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Confidential Access Mode</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Confidential Access Level</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Call Routing Information

- **Remote-Party-Id**
- **Asserted-Identity**
- **Asserted-Type** | Default
- **SIP Privacy**   | Default

### Inbound Calls

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Significant Digits</td>
<td>4</td>
</tr>
<tr>
<td>Connected Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Prefix DN</td>
<td></td>
</tr>
</tbody>
</table>

- **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.

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
</tbody>
</table>

### 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.

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
</tbody>
</table>

### Connected Party Settings

- **Connected Party Transformation CSS** | < None > |
- **Use Device Pool Connected Party Transformation CSS**

---

Figure 16: SIP Trunk to Cisco UBE (Cont.)
Figure 17: SIP Trunk to Cisco UBE (Cont.)
Explanation

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Name</td>
<td>BroadCloud</td>
<td>Name for the trunk</td>
</tr>
<tr>
<td>Device Pool</td>
<td>BroadCloud Devicepool</td>
<td>Default Device Pool is used for this trunk</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL_MTP_Group</td>
<td>MRG with resources: ANN, CFB, MOH and MTP</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>10.80.13.15</td>
<td>IP address of the Cisco UBE Virtual LAN</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
<td>BroadCloud Non Secure SIP Trunk Profile</td>
<td>SIP Trunk Security Profile configured earlier</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>BroadCloud 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:

1. Cisco IP phone dial “7”+10 digits number to access PSTN via Cisco UBE
   - “7” is removed before sending to Cisco UBE
2. For FAX call, Access Code “7”+10 digits number is used at Cisco Fax gateway
   - “7” is removed at Cisco UCM
   - The rest of the number is sent to Cisco UBE to BroadCloud network
3. Incoming fax call to 3XXX will be sent to Cisco Fax gateway
4. For Anonymous call, access code “*67”+10 digits number is used.
   - “*67” is removed at Cisco UCM
   - The rest of the number is sent to Cisco UBE to BroadCloud network

<table>
<thead>
<tr>
<th>Pattern</th>
<th>Description</th>
<th>Partition</th>
<th>Route Filter</th>
<th>Associated Device</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>*67.@</td>
<td>BroadCloud RP for Anonymous calls</td>
<td></td>
<td></td>
<td>BroadCloud</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>BroadCloud RP for FAX</td>
<td></td>
<td></td>
<td>SIP trunk to Voice gateway</td>
<td></td>
</tr>
<tr>
<td>7.@</td>
<td>BroadCloud Route pattern</td>
<td></td>
<td></td>
<td>BroadCloud</td>
<td></td>
</tr>
</tbody>
</table>

*Figure 18: Route Patterns List*
### Figure 19: Route Pattern for Voice

<table>
<thead>
<tr>
<th><strong>Pattern Definition</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Route Pattern</strong></td>
<td>7.00</td>
</tr>
<tr>
<td><strong>Route Partition</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Description</strong></td>
<td>BroadCloud Route pattern</td>
</tr>
<tr>
<td><strong>Numbering Plan</strong></td>
<td>TAPI</td>
</tr>
<tr>
<td><strong>Route Filter</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>MLPP Precedence</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Apply Call Blocking Percentage</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Resource Priority Namespace Network Domain</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Route Class</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Gateway/Route List</strong></td>
<td>BroadCloud</td>
</tr>
<tr>
<td><strong>Route Option</strong></td>
<td></td>
</tr>
</tbody>
</table>

- **Call Classification**: OIRNet
- **External Call Control Profile**: < None >
- **Provider Outside Dialed Tone**: Provide Outside Dialed Tone
- **Overload Sending**: Yes
- **Urgent Priority**: Yes

**Use Calling Party's External Phone Number 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** |  |  
| **Carrier Identification Code** |  |  

- **Network Service**: Not Selected
- **Service Parameter Name**: Not Selected
- **Service Parameter Value**: Not Selected
Figure 20: Route Pattern for Voice (Cont.)
### Pattern Definition

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td><em>67.</em></td>
</tr>
<tr>
<td>Route Partition</td>
<td>None</td>
</tr>
<tr>
<td>Description</td>
<td>BroadCloud RP for Anonymous calls</td>
</tr>
</tbody>
</table>

### Gateway/Route List
- BroadCloud

### Call Classification
- Offset: 

### External Call Control Profile
- Allow Device Override: False
- Provide Outside Dial Tone: True
- Allow Overlap Sending: False
- Urgent Priority: False

### Authorization Level
- 0

### 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

<table>
<thead>
<tr>
<th>Network Service Protocol</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>-- Not Selected --</td>
<td>&lt; Not Exist &gt;</td>
<td></td>
</tr>
<tr>
<td>Setting</td>
<td>Value</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------------------</td>
<td>----------------------------------------------------------------------</td>
<td>----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Route Pattern</td>
<td>7.@ for Voice &amp; International Calls, 3XXX for Fax Call and *67.@ for Anonymous Call.</td>
<td>Specify appropriate Route Pattern</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>BroadCloud for Route Pattern 7.@, *67.@ and SIP_Trunk_To_Voice_Gateway for Route Pattern 3XXX</td>
<td>SIP Trunk name configured earlier</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>NANP for Route Pattern 7.@ and *67.@</td>
<td>North American Numbering Plan</td>
</tr>
<tr>
<td>Call Classification</td>
<td>OffNet for Route Pattern 7.@, 3XXX and *67.@</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 7.@ and *67.@</td>
<td>Specifies how to modify digit before they are sending to BroadCloud network</td>
</tr>
<tr>
<td>Calling Line ID Presentation &amp; Calling Name Presentation</td>
<td>Restricted for Route Pattern *67.@</td>
<td>Restrict the Caller ID Display of Calling party’s at external user endpoint.</td>
</tr>
</tbody>
</table>
## Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>CPE</td>
<td>Customer Premise Equipment</td>
</tr>
<tr>
<td>CUBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>CUCM</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>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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<th>Americas Headquarters</th>
<th>Asia Pacific Headquarters</th>
</tr>
</thead>
<tbody>
<tr>
<td>170 West Tasman Drive</td>
<td>Haarlerbergpark</td>
<td>170 West Tasman Drive</td>
<td>Capital Tower</td>
</tr>
<tr>
<td>San Jose, CA 95134-1706</td>
<td>Haarlerbergweg 13-19</td>
<td>San Jose, CA 95134-1706</td>
<td>168 Robinson Road</td>
</tr>
<tr>
<td>USA</td>
<td></td>
<td>USA</td>
<td>#22-01 to #29-01</td>
</tr>
<tr>
<td><a href="http://www.cisco.com">www.cisco.com</a></td>
<td>1101 CH Amsterdam</td>
<td><a href="http://www.cisco.com">www.cisco.com</a></td>
<td>Singapore 068912</td>
</tr>
<tr>
<td>Tel: 408 526-4000</td>
<td>The Netherlands</td>
<td>Tel: 408 526-7660</td>
<td><a href="http://www.cisco.com">www.cisco.com</a></td>
</tr>
<tr>
<td>800 553-NETS (6387)</td>
<td></td>
<td>Fax: 408 527-0883</td>
<td>Tel: +65 317 7777</td>
</tr>
<tr>
<td>Fax: 408 526-4100</td>
<td></td>
<td>Fax: 31 0 20 357 1100</td>
<td>Fax: +65 317 7799</td>
</tr>
</tbody>
</table>

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