



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

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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:

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

Network Topology

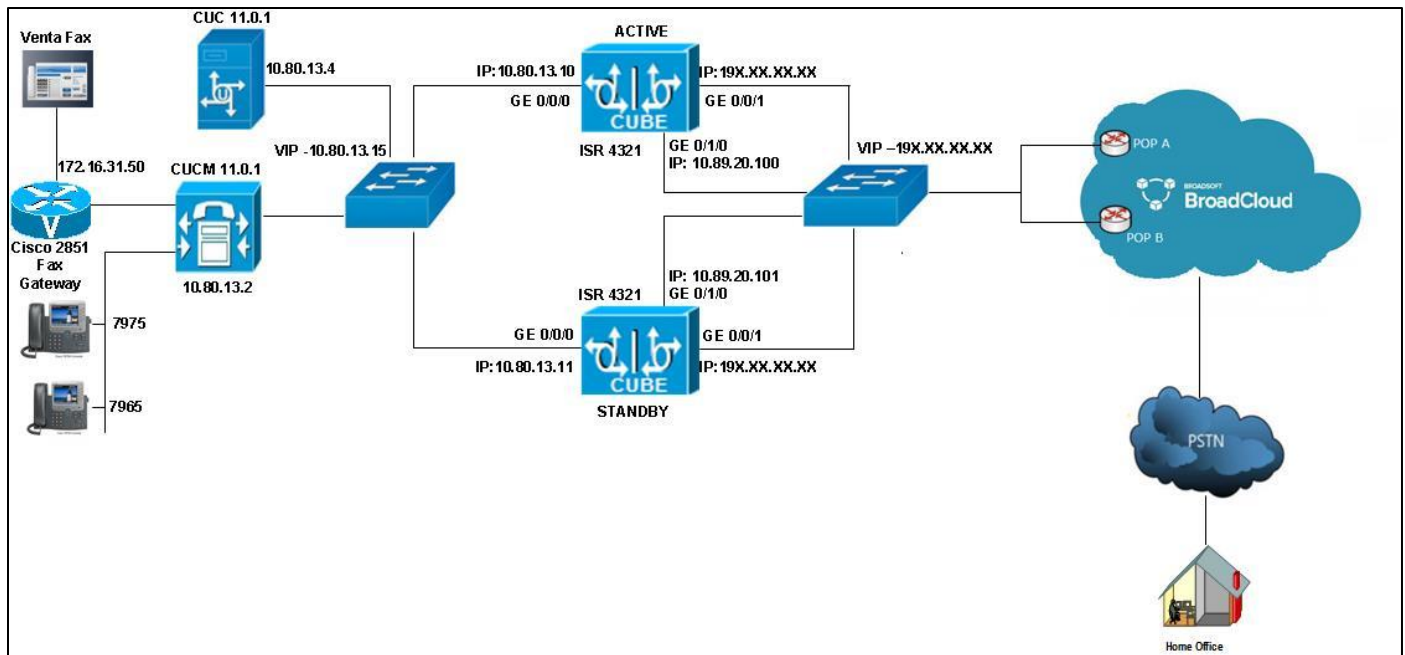


Figure 1: Network Topology

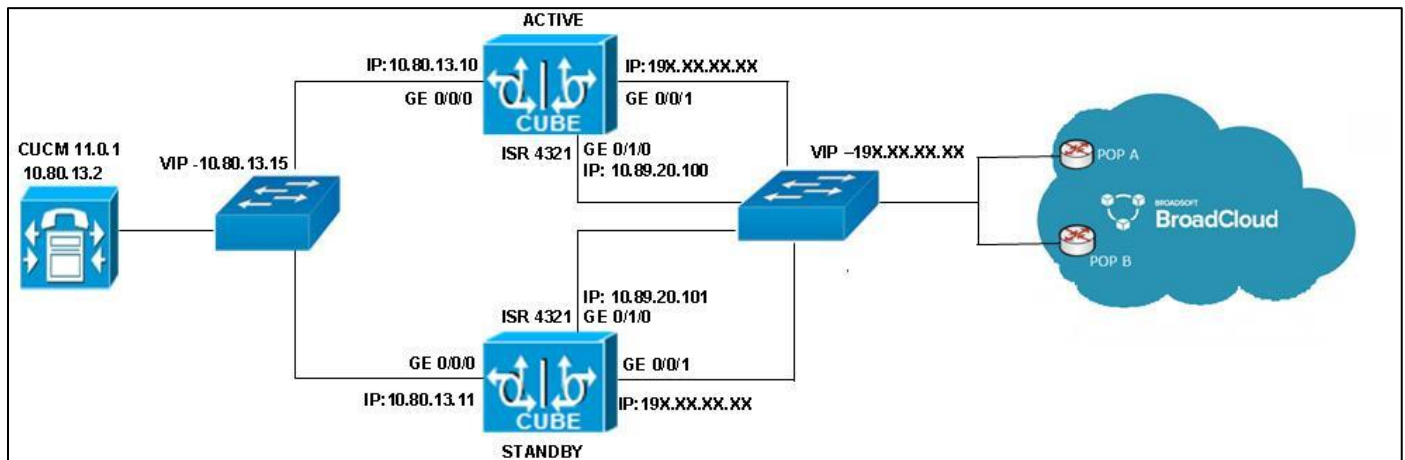


Figure 2: Cisco UBE High Availability

System Components

Hardware Requirements



- Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4321/K9 router as 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



Network interface

Configure Ethernet IP address and sub interface. The IP address and VLAN encapsulation used are for illustration only, the actual IP address can vary. For SIP trunks two IP addresses must be configured—LAN and WAN.

```
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

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



Codecs

G711ulaw 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 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-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 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 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-nte
no vad
!
sip-ua
credentials number XXXXXXXXXXXX username XXXXXXXXXXXX password 7 realm XXX@customerdomain.com
authentication username XXXXXXXXXXXX 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  
    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 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-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 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 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-nte
no vad
!
!
sip-ua
credentials number XXXXXXXXXXXX username XXXXXXXXXXXX password 7 realm XXX@customerdomain.com
authentication username XXXXXXXXXXXX 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 XXXXXXXX
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.

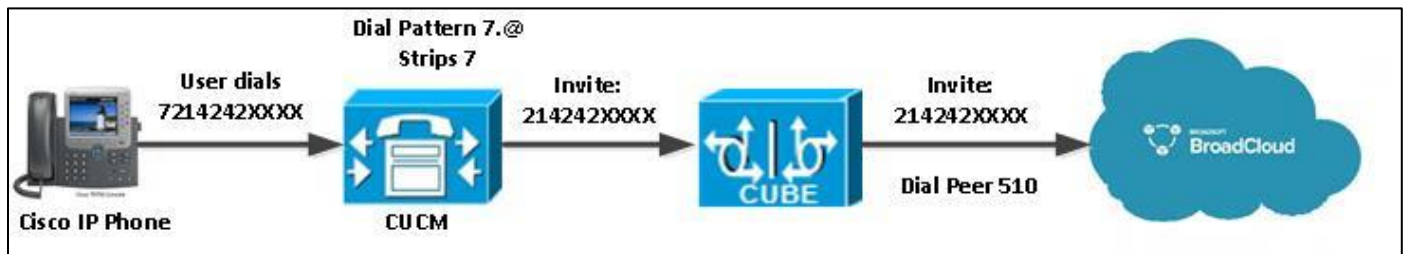


Figure 3: Outbound Voice Call

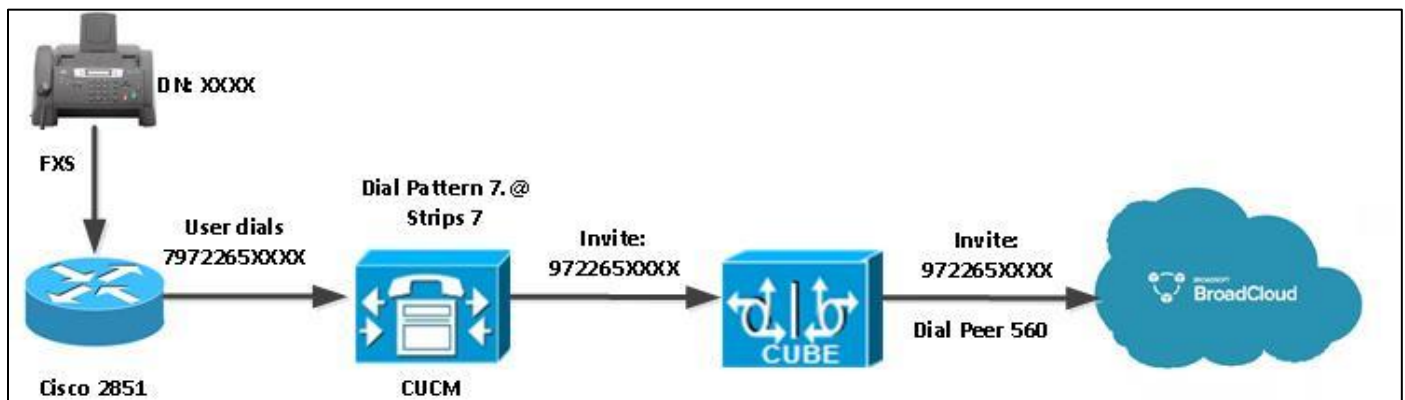


Figure 4: Outbound Fax Call

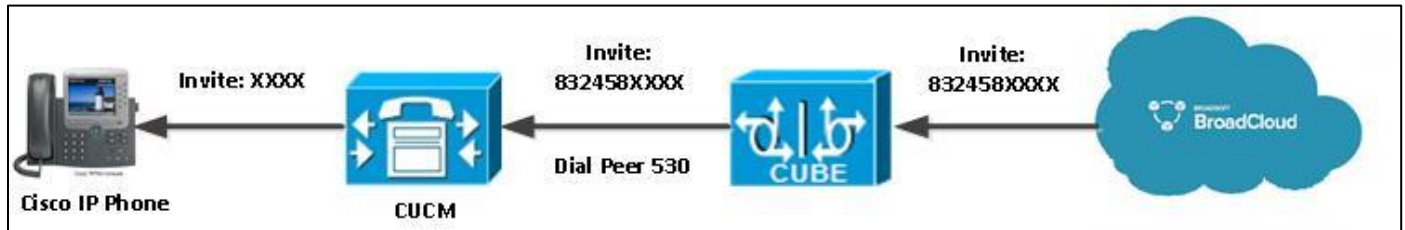


Figure 5: Inbound Voice Call

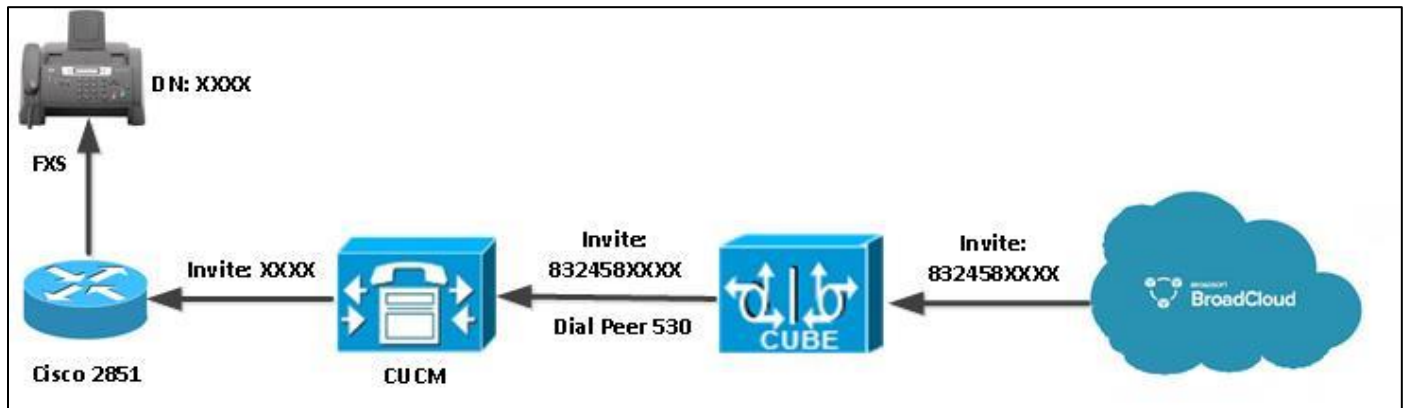


Figure 6 : Inbound Fax Call

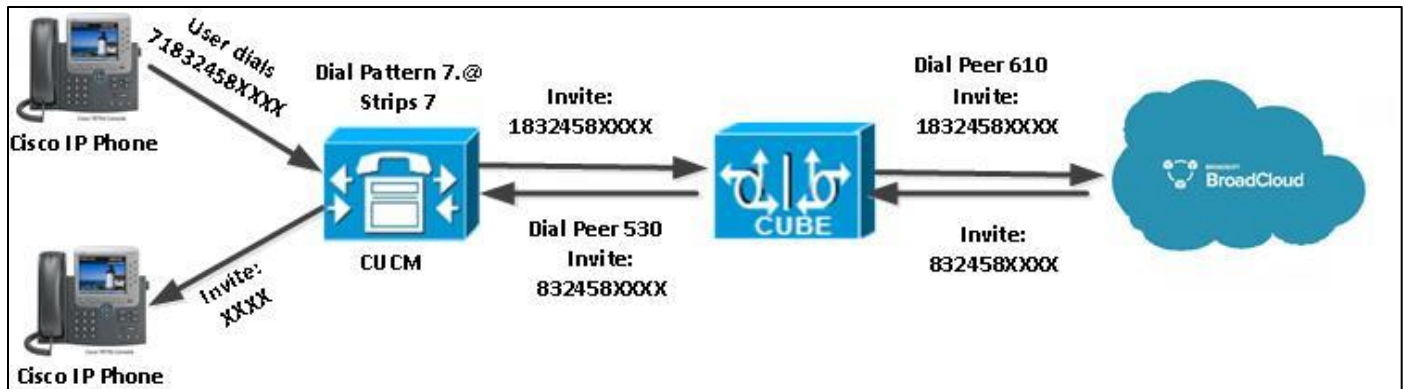


Figure 7 : PBX to PBX via BroadCloud Call



Configuring Cisco Unified Communications Manager

Cisco UCM Version

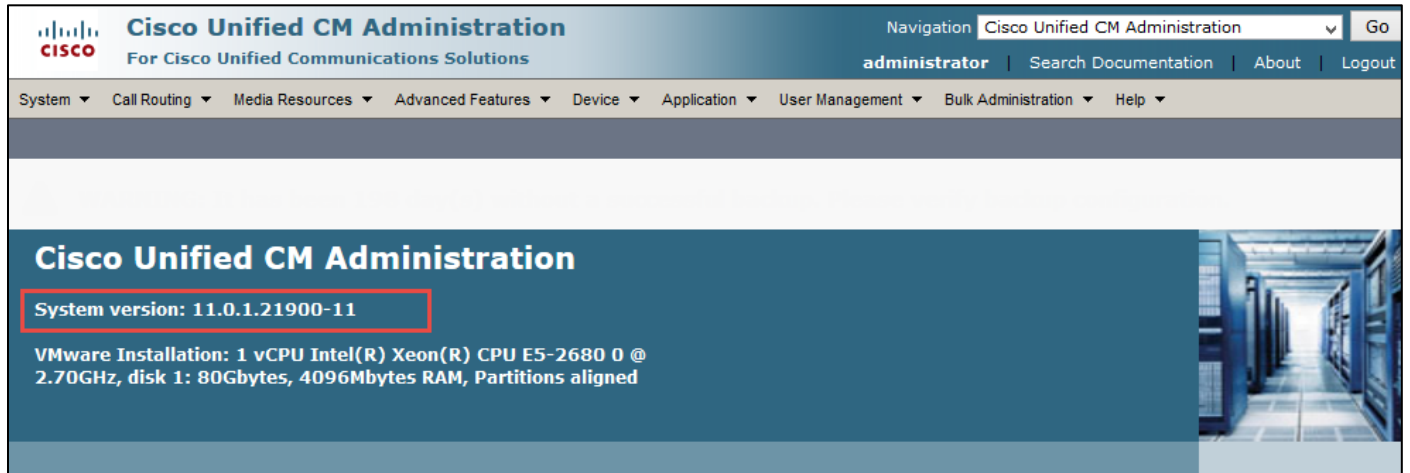


Figure 8: Cisco UCM Version

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

Select Server and Service

Server*

Service*

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

Cisco CallManager (Active) Parameters on server clus23pubsub--CUCM Voice/Video (Active)

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

Figure 9: Service Parameters



Offnet Calls via 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

Figure 10: SIP Trunk Security Profile

Explanation

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



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 Information	
Name*	BroadCloud SIP Profile
Description	BroadCloud 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"/> Use Fully Qualified Domain Name in SIP Requests	
<input type="checkbox"/> Assured Services SIP conformance	
SDP Information	
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer*	Default
<input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change	
<input type="checkbox"/> Allow RR/RS bandwidth modifier (RFC 3556)	
Parameters used in Phone	
Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Media Port Ranges	<input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766

Figure 11: SIP Profile



DSCP for Audio Calls	Use System Default						
DSCP for Video Calls	Use System Default						
DSCP for Audio Portion of Video Calls	Use System Default						
DSCP for TelePresence Calls	Use System Default						
DSCP for Audio Portion of TelePresence Calls	Use System Default						
Call Pickup URI*	x-cisco-serviceuri-pickup						
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup						
Call Pickup Group URI*	x-cisco-serviceuri-gpickup						
Meet Me Service URI*	x-cisco-serviceuri-meetme						
User Info*	None						
DTMF DB Level*	Nominal						
Call Hold Ring Back*	Off						
Anonymous Call Block*	Off						
Caller ID Blocking*	Off						
Do Not Disturb Control*	User						
Telnet Level for 7940 and 7960*	Disabled						
Resource Priority Namespace	< None >						
Timer Keep Alive Expires (seconds)*	120						
Timer Subscribe Expires (seconds)*	120						
Timer Subscribe Delta (seconds)*	5						
Maximum Redirections*	70						
Off Hook To First Digit Timer (milliseconds)*	15000						
Call Forward URI*	x-cisco-serviceuri-cfwdall						
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial						
<input checked="" type="checkbox"/> Conference Join Enabled <input type="checkbox"/> RFC 2543 Hold <input checked="" type="checkbox"/> Semi Attended Transfer <input type="checkbox"/> Enable VAD <input type="checkbox"/> Stutter Message Waiting <input type="checkbox"/> MLPP User Authorization							
Normalization Script							
Normalization Script < None >							
<input type="checkbox"/> Enable Trace							
	<table border="1"> <thead> <tr> <th></th> <th>Parameter Name</th> <th>Parameter Value</th> </tr> </thead> <tbody> <tr> <td>1</td> <td></td> <td></td> </tr> </tbody> </table>		Parameter Name	Parameter Value	1		
	Parameter Name	Parameter Value					
1							

Figure 12: 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*	Never
Resource Priority Namespace List	< None >
SIP Rel1XX Options*	Send PRACK if 1xx Contains SDP
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default
Session Refresh Method*	Invite
Early Offer support for voice and video calls*	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	
SIP OPTIONS Ping	
<input checked="" 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)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	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 13: SIP Profile (Cont.)

Explanation

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



SIP Trunk Configuration

Create SIP trunks to Cisco UBE

Navigation: Device > Trunk





Trunks (1 - 4 of 4)													Rows per Page 50
Find Trunks where Device Name begins with Find Clear Filter													
Select item or enter search text													
		Name ^	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile
<input type="checkbox"/>		BroadCloud	BroadCloud SIP Trunk		BroadCloud Devicepool	7.0				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 6 minutes	BroadCloud Non Secure SIP T Profile
<input type="checkbox"/>		BroadCloud	BroadCloud SIP Trunk		BroadCloud Devicepool	*67.0				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 6 minutes	BroadCloud Non Secure SIP T Profile
<input type="checkbox"/>		SIP_trunk_to_Voice_gateway	SIP_trunk_to_Voice_gateway		BroadCloud Devicepool	3				SIP Trunk	Unknown - OPTIONS Ping not enabled		BroadCloud Non Secure SIP T Profile
<input type="checkbox"/>		UnityConnectionTrunk-for-BroadCloud	UnityConnectionTrunk-for-BroadCloud		BroadCloud Devicepool	6XXX				SIP Trunk	Unknown - OPTIONS Ping not enabled		UnityConnectiontrunkSecurity for BroadCloud

Figure 14: SIP Trunks List



SIP Trunk Status	
Service Status: Full Service	
Duration: Time In Full Service: 0 day 0 hour 7 minutes	
Device Information	
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	BroadCloud
Description	BroadCloud SIP Trunk
Device Pool*	BroadCloud Devicepool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_MTP
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 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
<input checked="" type="checkbox"/> PSTN Access	
<input type="checkbox"/> Run On All Active Unified CM Nodes	
Intercompany Media Engine (IME)	
E.164 Transformation Profile	< None >

Figure 15: SIP Trunk to Cisco UBE



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

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

Inbound Calls
Significant Digits* 4
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Calling Search Space < None >
AAR Calling Search Space < None >
Prefix DN
☒ Redirecting Diversion Header Delivery - Inbound

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

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

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

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

Connected Party Settings
Connected Party Transformation CSS < None >
☒ Use Device Pool Connected Party Transformation CSS

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



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*

Last Redirect Number (External)

Calling Line ID Presentation*

Default

Calling Name Presentation*

Default

Calling and Connected Party Info Format*

Deliver DN only in connected party

☒ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

< None >

☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

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

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status
1 *	10.80.13.15		5060	up

MTP Preferred Originating Codec*

711ulaw

BLF Presence Group*

Standard Presence group

SIP Trunk Security Profile*

BroadCloud Non Secure SIP Trunk Profile

Rerouting Calling Search Space

< None >

Cut-Of-Dialog Refer Calling Search Space

< None >

SUBSCRIBE Calling Search Space

< None >

SIP Profile*

BroadCloud SIP Profile

[View Details](#)

DTMF Signaling Method*

No Preference

Normalization Script

Normalization Script

< None >

☐ Enable Trace

	Parameter Name	Parameter Value	
1			<div><div>+</div><div>-</div></div>

Recording Information

☒ None

☐ This trunk connects to a recording-enabled gateway

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

Geolocation Configuration

Geolocation

< None >

Geolocation Filter

< None >

☐ Send Geolocation Information

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



Explanation

Parameter	Value	Description
Device Name	BroadCloud	Name for the trunk
Device Pool	BroadCloud Devicepool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_MTP_Group	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.13.15	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	BroadCloud Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	BroadCloud SIP Profile	SIP Profile configured earlier



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

Route Patterns (1 - 4 of 4)						Rows per Page 50
Find Route Patterns where <input type="text" value="Pattern"/> <input type="text" value="begins with"/> <input type="button" value="Find"/> <input type="button" value="Clear Filter"/> <input type="button" value="+"/> <input type="button" value="-"/>						
<input type="checkbox"/>	Pattern ^	Description	Partition	Route Filter	Associated Device	Copy
<input type="checkbox"/>	*67.@	BroadCloud RP for Anonymous calls			BroadCloud	
<input type="checkbox"/>	3'	BroadCloud RP for FAX			SIP trunk to Voice gateway	
<input type="checkbox"/>	7.@	BroadCloud Route pattern			BroadCloud	

Figure 18: Route Patterns List

Pattern Definition		
Route Pattern*	7.@	
Route Partition	< None >	
Description	BroadCloud Route pattern	
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*	BroadCloud	
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error	
Call Classification*	OffNet	
External Call Control Profile	< None >	
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority		
<input type="checkbox"/> Require Forced Authorization Code		
Authorization Level*	0	
<input type="checkbox"/> Require Client Matter Code		
(Edit)		
Calling Party Transformations		
<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask		
Calling Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Calling Line ID Presentation*	Default	
Calling Name Presentation*	Default	
Calling Party Number Type*	Cisco CallManager	
Calling Party Numbering Plan*	Cisco CallManager	
Connected Party Transformations		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
Called Party Transformations		
Discard Digits	PreDot	
Called Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Called Party Number Type*	Cisco CallManager	
Called Party Numbering Plan*	Cisco CallManager	
ISDN Network-Specific Facilities Information Element		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Figure 19: Route Pattern for Voice

Pattern Definition		
Route Pattern*	3	
Route Partition	< None >	
Description	BroadCloud RP for FAX	
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*	SIP_trunk_to_Voice_gateway (Edit)	
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error	
Call Classification*	OffNet	
External Call Control Profile	< None >	
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority		
<input type="checkbox"/> Require Forced Authorization Code		
Authorization Level*	0	
<input type="checkbox"/> Require Client Matter Code		
Calling Party Transformations		
<input type="checkbox"/> Use Calling Party's External Phone Number Mask		
Calling Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Calling Line ID Presentation*	Default	
Calling Name Presentation*	Default	
Calling Party Number Type*	Cisco CallManager	
Calling Party Numbering Plan*	Cisco CallManager	
Connected Party Transformations		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
Called Party Transformations		
Discard Digits	< None >	
Called Party Transform Mask	832458XXX	
Prefix Digits (Outgoing Calls)		
Called Party Number Type*	Cisco CallManager	
Called Party Numbering Plan*	Cisco CallManager	
ISDN Network-Specific Facilities Information Element		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Figure 20: Route Pattern for Voice (Cont.)

Pattern Definition		
Route Pattern*	*67. @	
Route Partition	< None >	
Description	BroadCloud RP for Anonymous calls	
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*	BroadCloud	
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error	
Call Classification*	OffNet	
External Call Control Profile	< None >	
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority <input type="checkbox"/> Require Forced Authorization Code Authorization Level* 0 <input type="checkbox"/> Require Client Matter Code		
Calling Party Transformations		
<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask		
Calling Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Calling Line ID Presentation*	Restricted	
Calling Name Presentation*	Restricted	
Calling Party Number Type*	Cisco CallManager	
Calling Party Numbering Plan*	Cisco CallManager	
Connected Party Transformations		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
Called Party Transformations		
Discard Digits	PreDot	
Called Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Called Party Number Type*	Cisco CallManager	
Called Party Numbering Plan*	Cisco CallManager	
ISDN Network-Specific Facilities Information Element		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Figure 21: Route Pattern for Fax



Explanation

Setting	Value	Description
Route Pattern	7.@ for Voice & International Calls, 3XXX for Fax Call and *67.@ for Anonymous Call.	Specify appropriate Route Pattern
Gateway/Route List	BroadCloud for Route Pattern 7.@, *67.@ and SIP_Trunk_To_Voice_Gateway for Route Pattern 3XXX	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 7.@ and *67.@	North American Numbering Plan
Call Classification	OffNet for Route Pattern 7.@, 3XXX and *67.@	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 7.@ and *67.@	Specifies how to modify digit before they are sending to BroadCloud network
Calling Line ID Presentation & Calling Name Presentation	Restricted for Route Pattern *67.@	Restrict the Caller ID Display of Calling party's at external user endpoint.



Acronyms

Acronym	Definitions
CPE	Customer Premise Equipment
CUBE	Cisco Unified Border Element
CUCM	Cisco Unified Communications Manager
MTP	Media Termination Point
POP	Point of Presence
PSTN	Public Switched Telephone Network
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol



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