Application Note

Orange Business Services Business Talk IP (France and International): Connecting Cisco Unified Communications Manager 8.0 via the Cisco Unified Border Element 1.4 using SIP

April 28, 2011

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Note: Testing was conducted by Orange Business Services.

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Introduction

Service Providers today, such as Orange Business Services, 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. Orange Business Services offers voice trunking services under following names:

- Business Talk IP – in France
- Business Talk – internationally (outside of France)

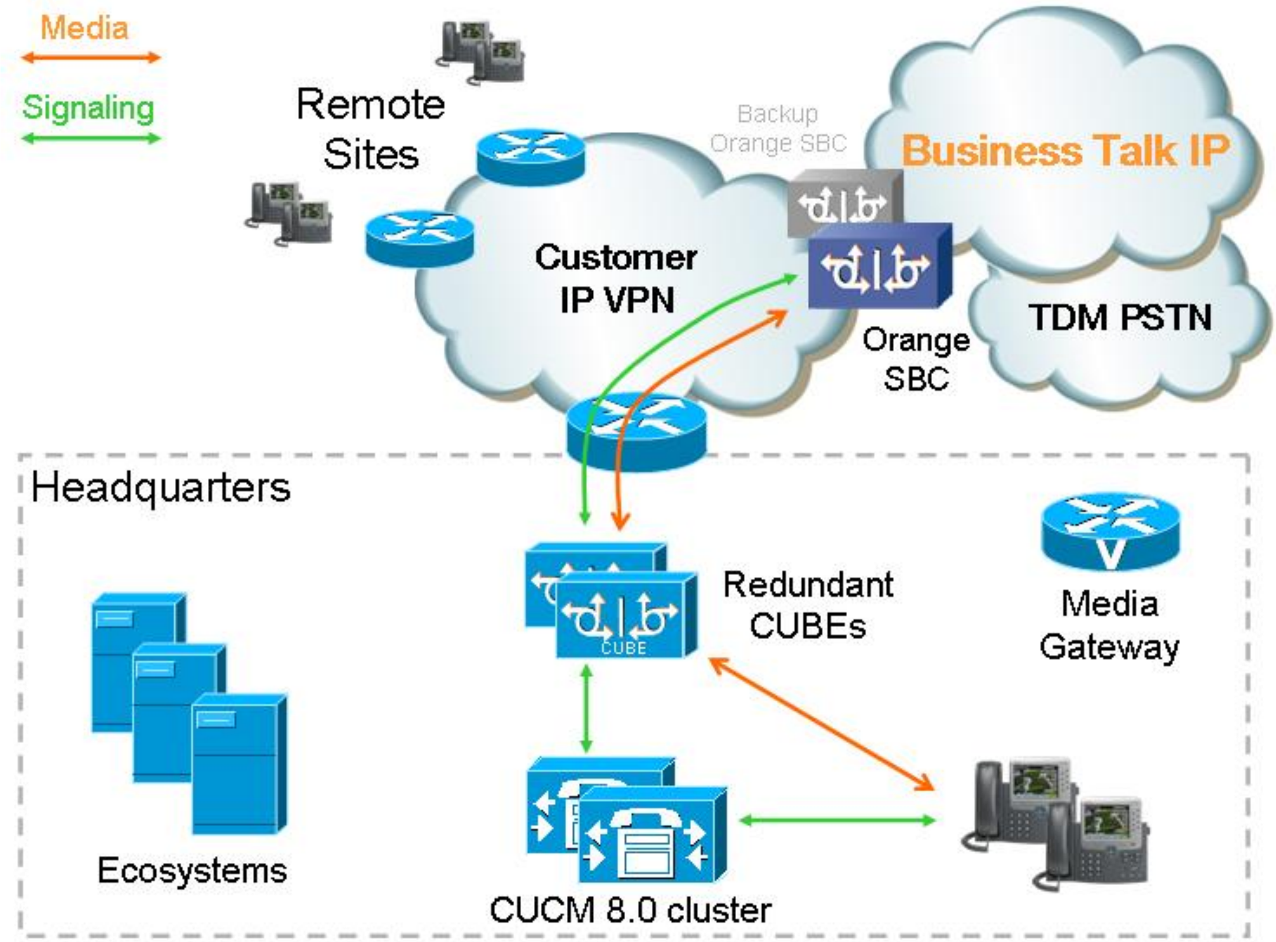
These services are hereafter referred collectively as **Business Talk IP**. Business Talk IP is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity. A demarcation device between these services and customer owned services is recommended. The Cisco Unified Border Element provides demarcation, security, interworking and session management services for Cisco Unified Communications Manager 8.0 connected to Business Talk IP.

- This application note describes how to configure a Cisco Unified Communications Manager (Cisco UCM) 8.0 with a Cisco Unified Border Element (Cisco UBE) for connectivity to Business Talk IP SIP trunking service. The deployment model covered in this application note is CPE (Cisco UCM 8.0 with Cisco UBE) to PSTN (Business Talk IP). This document does not address 911 emergency outbound calls. For 911 feature service details contact Orange Business Services, directly.
- Testing was performed in accordance to Orange Business Services generic SIP trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold, blind and supervised transfers, call forward, conferences, hunt groups, call pickup, call park, Mobile Connect and interoperability with various Cisco ecosystems (Unity Connection, Unified Contact Center Express, MeetingPlace, etc).
- The Cisco Unified Border Element configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Business Talk IP network and Cisco Unified Communications. The configuration described in this document details the important commands to be configured for interoperability to be successful and care must be taken, by the network administrator deploying Cisco UBE, to ensure these commands are set per each dial-peer requiring to interoperate to Orange Business Services SIP network.
- This application note does not cover the use of calling search spaces (CSS) or partitions on Cisco Unified Communications Manager. To understand and learn how to apply CSS and partitions refer to the cisco.com link below:
  http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/8x/dialplan.html#wp1044299

Note: Testing was conducted by Orange Business Services.

Network Topology

Figure 1. Basic Call Setup



System Components

Hardware Components

- UCS-B or UCS-C or MCS 7800 servers for CUCM and ecosystems
- ISR G1 2800/3800 series routers with PVDM2 modules for media resources and CUBE functionality
- ISR G2 2900/3900 series routers with PVDM3 modules for media resources and CUBE functionality
- IP phones 7900/8900/9900 series (different models, both SIP and SCCP where supported)
- IP phones 6900 series supported only with SCCP protocol
- Cisco Voice Gateways 200 series (VG202/204/224)

Note: Testing was conducted by Orange Business Services.

Software Requirements

For exact versions please contact your Orange Business Services account team:

- Cisco Unified Communications Manager 8.0 and later 8.x releases
- Cisco Unified Border Element Release 1.4 IOS version 15.1.1T
- IOS 15.1.1T for IOS gateways (required for DSP media resources and local PSTN failover)
- Cisco Unity Connection 8.x
- Unified Contact Center Express 8.x
- Cisco Unified Attendant Consoled (CUDAC/CUBAC/CUEAC) 8.x
- Cisco Unified Presence Server 8.x with Cisco Unified Personal Communicator
- Cisco Unified MeetingPlace 8.x

Note: Testing was conducted by Orange Business Services.

Features

Features Supported

- Incoming and outgoing offnet calls using G711 a-law or G.729 (only 1 of them must be used for all calls), both with 20ms packetization
- Call hold
- Call transfer (blind and supervised)
- Call conference
- Call forward (all, busy, no answer, unregistered)
- Call park
- Hunt groups
- Calling line (number) identification presentation (CLIP)
- Calling line (number) identification restriction (CLIR)
- Calling ID restriction
- DTMF (RFC2833)
- Unified Mobility (Single Number Reach feature)
- Mobile connect

Features Not Supported

- Fax over IP (T.38 and G.711 pass-through). Customers are recommended to use local fax routing through their own centralized or distributed dedicated PSTN gateways.
- Software MTPs (not supported MTP based on CUCM and on other ISR router). Customers are recommended to use hardware MTPs based on PVDM modules.
- Media flow-around on CUBE (only flow-through mode is supported)

Note: Testing was conducted by Orange Business Services.

Caveats

- For incoming offnet calls to ecosystems (IVR, voicemail impacted) up to 1 second of media clipping might occur for the initial automated prompt. Some amount of silence can be added to the initial prompt as a workaround
- Media hairpinning for offnet calls to/from IP phones located in Remote Sites and off-net to off-net transfers and call forwarding
- Transcoders must only be configured for devices requiring them and excluded from the configuration of all other devices (most important: they should be not available on SIP trunks)
- UCCX does not support RFC2833 DTMF transport method used by Business Talk IP. Transcoders are mandatory for DTMF interoperability to out of band form. Dedicated SIP trunks have to be configured between CUCM <-> CUBE to limit transcoder usage only to calls to UCCX.
- Cisco IP phones 6900 and 7940/60 series supported only with SCCP firmware

Note: Testing was conducted by Orange Business Services.

- Configuration

Configuring Cisco Unified Border Element

Network interface

Configure Ethernet IP address and sub interface. The IP address and VLAN encapsulation used are for illustration only, the actual IP address can vary.

```
interface FastEthernet0/0

description **CUBE voice**

no ip address

duplex auto

speed auto

!

interface FastEthernet0/0.123

description **CUBE voice**

encapsulation dot1Q 123

ip address 10.108.105.201 255.255.255.0

!
```

SNMP server

A snmp-server community named public is created with Read Only access in order to allow equipment supervision.

Snmp server manager is activated for Embedded Event Manager purpose (see Embedded Event Manager applet configuration).

```
snmp-server community public RO

snmp-server manager

!
```

Global SIP settings

```
voice service voip

mode border-element
```


Note: Testing was conducted by Orange Business Services.

```
allow-connections sip to sip

sip

       header-passing

        error-passthru

       no update-callerid

       early-offer forced

       midcall-signaling passthru

       sip-profiles 1

              ip address trusted list

                     ipv4 A.B.C.D   ! primary Business Talk IP SBC IP address

                     ipv4 E.F.G.H   ! backup Business Talk IP SBC IP address
```

Explanation

Command	**Description**
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
header-passing, error-passthru	Error messages are passed through CUBE (SIP error transparency)
no update-callerid	Transparency regarding Caller ID
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
sip-profiles 1	Apply sip profile at global level

Codecs

Business Talk IP requires monocodec configuration. That means, that only a single codec should be offered by CUBE. This is configured using codec class which is then applied to specific dial-peer. Business Talk IP supports either G.711 A-law or G.729 in WAN. G.711 u-law is not supported.

For customers using **G.711 alaw** codec:

```
voice class codec 1

   codec preference 1 g711alaw
```


Note: Testing was conducted by Orange Business Services.

For customers using **G.729** codec use following configuration:

```
voice class codec 1

   codec preference 1 g729r8
```

SIP Profiles

voice class sip-profiles command allows to fine-tune SIP signaling on CUBE. This feature is used to solve some interoperability issues with Business Talk IP service and to provide current software versions used by customer. Sip-profile is later assigned to dial-peers pointing to Business Talk IP.

Sip-profile configuration is removed and created automatically after each router reboot by Embedded Event Manager (see the chapter on EEM configuraiton later on). See following example of configuration generated by EEM:

```
voice class sip-profiles 1

request INVITE sip-header User-Agent modify ".*" "User-Agent: CUBE 15.1.1T + CUCM 8.0 "

response 183 sip-header Call-Info add "P-EARLY-MEDIA: sendrecv"

request INVITE sip-header Supported modify "timer," ""

response 180 sip-header Server modify ".*" "Server: CUBE 15.1.1T + CUCM 8.0 "
```

Explanation

Command	**Description**
request INVITE sip-header User-Agent modify ".*" "User-Agent: CUBE 15.1.1T + CUCM 8.0 "	Insert CUBE and CUCM versioning in SIP Invite Message header (command created by EEM script). Refer to Embedded Event Manager Applet configuration chapter.
response 183 sip-header Call-Info add "P-EARLY-MEDIA: sendrecv"	Insert P-early-media attribute in SIP 183 messages to force remote RBT generation
request INVITE sip-header Supported modify "timer," ""	Modify timer information in SIP header in sip profile
response 180 sip-header Server modify “.*” “Server: CUBE 15.1.1T + CUCM 8.0"	Insert CUBE and CUCM versioning in SIP 180 Ringing Message header (command created by EEM script). Cf. Embedded Event Manager Applet configuration

SIP user agent

SIP signaling parameters are configured in the sip user agent section.

```
sip-ua

   retry invite 2
```


Note: Testing was conducted by Orange Business Services.

```
retry response 2

retry bye 2

retry cancel 2

reason-header override

connection-reuse

g729-annexb override
```

Explanation

Command	**Description**
retry …	Specifies number of retries for different SIP message types
reason-header override	Enable cause code passing from one SIP leg to another
connection-reuse	Always use the same port for both source and destination (UDP 5060)
g729-annexb override	Required for interoperability with Business Talk IP infrastructure, when G.729 codec is used

Dial peers

Basically, four dial peers are created and each of them can be customized according to customer dial plan. For instance, dial peers 1 and 2 are used for inbound and outbound calls on CUCM side, one for the CUCM Subscriber, the other for the CUCM Publisher. Dial peer 101 and 102 are used for incoming and outgoing calls on SBC side (please note, that the actual IP addresses will probably differ).

```
dial-peer voice 1 voip

 description ** DP to/from HQ devices - CUCM2 Subscriber **

 preference 1

 answer-address 180....

 destination-pattern 180....

 voice-class codec 1

 voice-class sip profiles 1

 voice-class sip options-keepalive up-interval 300 down-interval 300 retry 5
```


Note: Testing was conducted by Orange Business Services.

```
 session protocol sipv2
 session target ipv4:10.108.101.2
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 2 voip
 description ** DP to/from HQ devices - CUCM1 Publisher **
 preference 2
 answer-address 180....
 destination-pattern 180....
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip options-keepalive up-interval 300 down-interval 300 retry 5
 session protocol sipv2
 session target ipv4:10.108.101.1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 101 voip
 description ** Incoming calls - Inbound dial peer - SBC side **
 answer-address +.T
 voice-class codec 1
```


Note: Testing was conducted by Orange Business Services.

```
 voice-class sip profiles 1
 session protocol sipv2
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 102 voip
 description ** Outgoing calls - Outbound dial peer - SBC1 side **
 preference 1
 destination-pattern 0.T
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip options-keepalive up-interval 300 down-interval 300 retry 5
 session protocol sipv2
 session target ipv4:172.22.244.217
 dtmf-relay rtp-nte
 no vad
!
```

Note: Keepalives: IPBX SIP profile recommendations are to send OPTIONS every 300s minimum not to overload the aSBCs: 300s for in-service trunks is the min interval required.

Explanation

Command	**Description**
answer-address XXX….	Matches calls from CUCM by their calling number (ie site code)
destination-pattern XXX….	Matches calls from CUCM by their called number (ie site code)

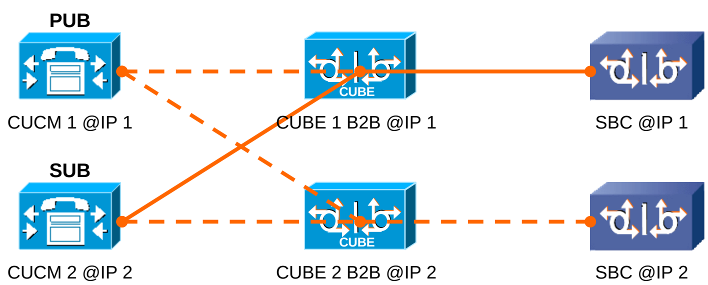
Note: Testing was conducted by Orange Business Services.

voice-class codec 1	Apply codecs of codec class 1 configured earlier
voice-class sip profiles 1	Apply SIP profile 1 configured earlier
voice-class sip options-keepalive	Enable SIP OPTION PING mechanisms between CUCM and CUBE
voice-class sip options-keepalive up-interval 300	Enable SIP OPTION PING mechanisms between CUBE and SBC (fixed to 5 minutes)
session target ipv4:IP_ADDRESS	Destination IP address for SIP messages sent through this dial-peer
dtmf-relay rtp-nte	Allows DTMF relay using NTE RTP packets. DTMF tones are encoded in the NTE format and transported in the same RTP channel as the voice.
no vad	Disables voice activity detection (not supported by Business Talk IP)

Redundancy

Outgoing calls

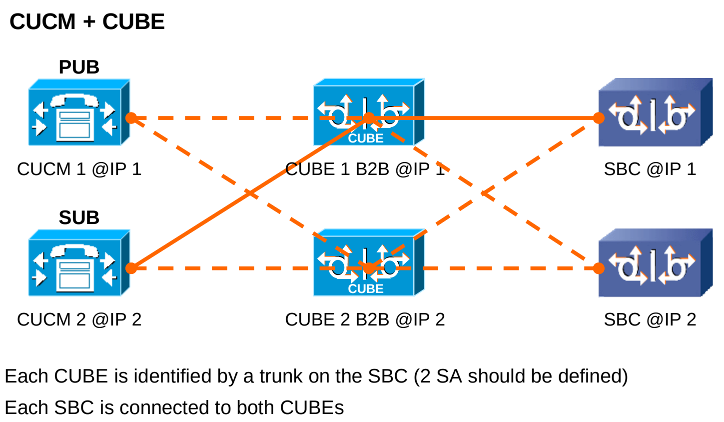
CUCM + CUBE



Each CUCM is connected to both CUBEs
Each CUBE is connected to only one SBC
For redundancy purpose, 2 CUBEs are mandatory on the architecture

Note: Testing was conducted by Orange Business Services.

Iincoming calls



Support for Privacy and P-Asserted Identity

The Cisco Unified Border Element supports the use of P-Asserted Identity (PAID) and Privacy collectively known as P-headers, in INVITE messages**.**

If the user is subscribed to a privacy service, the Cisco Unified Border Element can support privacy using the Privacy header method. If the Privacy header is set to None, the calling number is delivered to the called party. If the Privacy header is set to a Privacy:id value, the calling number is not delivered to the called party.

To enable the privacy settings for the header on a specific dial peer, use the **voice-class sip privacy id** command in dial peer voice configuration mode.

```
dial-peer voice tag  voip

   voice-class sip privacy id
```

To enable the translation to PAID privacy headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id pai** command in dial peer voice configuration mode.

```
dial-peer voice tag voip

   voice-class sip asserted-id pai
```

Embedded Event Manager

For release management process and Business Talk IP management systems, it's required to include in the SIP signaling messages sent from CUBE the current version of the software running on the customer's endpoints (i.e. CUCM and CUBE). This has to be done automatically in order to provide up-to-date information. Unfortunately, IOS and CUCM don't offer such functionality by default, a workaround needs to be used.

Note: Testing was conducted by Orange Business Services.

Embedded Event Manager (EEM) is a flexible subsystem in Cisco IOS that provides real-time network event detection and onboard automation. EEM 3.0 released offers the possibility to poll remote equipment's MIB in order to get information such as version id for instance. Using an EEM applet we can add current IOS version for CUBE and CUCM release in the INVITE SIP-Header message, so this information can be tracked in management systems for later use.

The IOS version applet is activated after each router reboot (e.g. after IOS upgrade). It then checks the IOS and CUCM versions using SNMP queries. Using this information it automatically invokes IOS commands to generate SIP profile including

```
event manager environment _sip_header_1 request INVITE sip-header User-Agent modify    ".*"
"User-Agent: CUBE

event manager environment _sip_header_2 "

event manager environment _sip_header_3 response 183 sip-header Call-Info add "P-EARLY-MEDIA:
sendrecv"

event manager environment _sip_header_4 request INVITE sip-header Supported modify "timer," ""

event manager environment _sip_header_5 response 180 sip-header Server modify ".*" "Server: CUBE



event manager applet IOSversion

   event timer countdown name IOSversion time 50

   action 1.0 info type snmp oid 1.3.6.1.2.1.47.1.1.1.1.10.3 get-type exact

   action 1.1 set _info_snmp_iosversion "$_info_snmp_value"

   action 1.2 info type snmp oid 1.3.6.1.4.1.9.9.156.1.1.2.1.4.1 get-type exact community public
ipaddr 10.108.101.1

   action 2.0 cli command "enable"

   action 2.1 cli command "config t"

   action 3.0 cli command "no voice class sip-profiles 1"

   action 4.0 cli command "voice class sip-profiles 1"
```


Note: Testing was conducted by Orange Business Services.

```
   action 4.1 cli command "$_sip_header_1 $_info_snmp_iosversion + CUCM $_info_snmp_value
$_sip_header_2"
   action 4.2 cli command "$_sip_header_3"
   action 4.3 cli command "$_sip_header_4"
   action 4.4 cli command "$_sip_header_5 $_info_snmp_iosversion + CUCM $_info_snmp_value
$_sip_header_2"
   action 5.0 cli command "voice service voip"
   action 5.1 cli command "sip"
   action 5.2 cli command "sip-profiles 1"
   action 6.0 cli command "end"
```

Explanation

Each time, 50 seconds after the router reboots, the EEM applet will be executed and will build a new voice classe sip-profiles with correct ID versions of CUBE and CUCM in the INVITE SIP-Header.

This way, CUBE IOS version and CUCM release will be added in all SIP INVITE and SIP 180 Ringing header messages sent from the CUBE.

Parameter	**Description**
1.3.6.1.2.1.47.1.1.1.1.10.3	SNMP OID which contains CUBE IOS version id. Local SNMP polling.
_info_snmp_iosversion	the IOS version will be stored in this variable
1.3.6.1.4.1.9.9.156.1.1.2.1.4.1	SNMP OID which contains CUCM version id. Remote SNMP polling available since EEM 3.0
10.108.101.1	CUCM Publisher IP address. This MIB is only available on Publisher Server

Verification

EEM applet configuration can be verified with following commands:

- show event manager policy registered
- show run | section sip-profile
- debug voip ccsip messages

Note: Testing was conducted by Orange Business Services.

```
HQ8-CUBE1#sh event manager policy registered

No.  Class     Type    Event Type          Trap  Time Registered           Name

1    applet    user    timer countdown     Off   Fri Jun 26 16:33:07 2009  IOSversion

 name {IOSversion} time 50.000

 maxrun 20.000

 action 1.0 info type snmp oid 1.3.6.1.2.1.47.1.1.1.1.10.3 get-type exact

 action 1.1 set _info_snmp_iosversion "$_info_snmp_value"

 action 1.2 info type snmp oid 1.3.6.1.4.1.9.9.156.1.1.2.1.4.1 get-type exact community public
ipaddr 10.108.101.1

 action 2.0 cli command "enable"

 action 2.1 cli command "config t"

 action 3.0 cli command "no voice class sip-profiles 1"

 action 4.0 cli command "voice class sip-profiles 1"

 action 4.1 cli command "$_sip_header_1 $_info_snmp_iosversion + CUCM $_info_snmp_value
$_sip_header_2"

 action 4.2 cli command "$_sip_header_3"

 action 4.3 cli command "$_sip_header_4"

 action 4.4 cli command "$_sip_header_5 $_info_snmp_iosversion + CUCM $_info_snmp_value
$_sip_header_2"

 action 5.0 cli command "voice service voip"

 action 5.1 cli command "sip"

 action 5.2 cli command "sip-profiles 1"

 action 6.0 cli command "end"
```


Note: Testing was conducted by Orange Business Services.

```
HQ8-CUBE1#show run | section sip-profiles

voice class sip-profiles 1

request INVITE sip-header User-Agent modify ".*" "User-Agent: CUBE 15.1.1T + CUCM 8.0"

response 183 sip-header Call-Info add "P-EARLY-MEDIA: sendrecv"

request INVITE sip-header Supported modify "timer," ""

response 180 sip-header Server modify ".*" "Server: CUBE 15.1.1T + CUCM 8.0 "
```

This can be verified using **debug voip ccsip messages** command or using packet sniffer:

```
HQ300CUBE1#debug ccsip messages

HQ300CUBE1#term mon



151172: Mar 11 13:46:10.527: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:

Sent:

INVITE sip:00480048602331630@172.22.246.49:5060 SIP/2.0

Date: Thu, 11 Mar 2010 12:46:10 GMT

Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO,
REGISTER

From: sip:0003001100@6.3.0.2;tag=C1FAA1E4-1030

Allow-Events: telephone-event

Supported: 100rel,resource-priority,replaces,sdp-anat

Min-SE:  1800

Remote-Party-ID: <sip:0003001100@6.3.0.71>;party=calling;screen=yes;privacy=off

Cisco-Guid: 3881251155-742527455-2564919463-3436266594
```


Note: Testing was conducted by Orange Business Services.

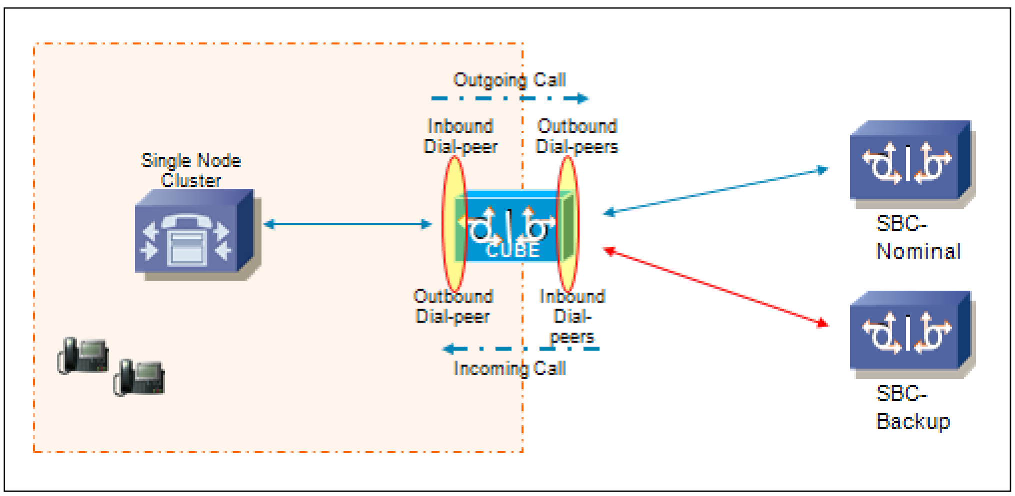
```
Timestamp: 1268311570

Content-Length: 235

User-Agent: CUBE 15.1.1T + CUCM 8.0

To: <sip:0048004860233l630@172.22.246.49>

Contact: <sip:000300l100@6.3.0.71:5060>

Expires: 180

Content-Disposition: session;handling=required

Content-Type: application/sdp

Call-ID: E75905CA-2C4211DF-98E790A7-CCD14462@6.3.0.71

Via: SIP/2.0/UDP 6.3.0.71:5060;branch=z9hG4bK3DCEB1870

CSeq: 101 INVITE

Session-Expires:  1800

Max-Forwards: 69



v=0

o=CiscoSystemsSIP-GW-UserAgent 3823 1607 IN IP4 6.3.0.71

s=SIP Call

c=IN IP4 6.3.0.71

t=0 0

m=audio 18626 RTP/AVP 8 101

c=IN IP4 6.3.0.71

a=rtpmap:8 PCMA/8000
```

```
a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

a=ptime:20

[...]
```

Note: Warning! In oder to provide such information, the SNMP server must be activated on CUBE (0), and a public community must be created on CUCM.

Specific design for a single CUCM server

In case of a specific design for a single CUCM server which can be a CUCM Business Edition (CUCMBE), the configuration and integration should take care of the rerouting decision based on SIP Error and Release Cause received from infrastructure.



Call rerouting should be done at the CUCM level in such scenario. This way, on reception of an error from the infrastructure, the CUBE will forward the error to the CUCM node. On reception of an error from the CUCM, the CUBE will forward the error to the infrastructure.

CUCM needs to be configured with the list of Release Cause value that needs overflow: **1, 16, 17, 21, 22, 28, 34, 63, 127** (see CUCM configuration guide). CUBE needs be configured with the command **huntstop** under each dial-peer toward the SBCs, to disable local rerouting.

CUBE can be configured with OPTIONS ping. If an SBC is not available, the dial-peer will be in a busy out state and the error will be reported back to CUCM that will overflow the call accordingly.

Configuration

CUBE needs to be configured with physical interface will be configured with a secondary IP address.

```
interface FastEthernet 0/0.167

   ip address 10.227.101.201 255.255.255.0
```


Note: Testing was conducted by Orange Business Services.

```
   ip address 10.227.101.203 255.255.255.0 secondary
```

CUCM will be configured with 2 different SIP trunks :

- 1st SIP trunk pointing to the primary address of the CUBE
- 2nd SIP trunk pointing to the secondary address of the CUBE

CUCM will be configured with a Route List composed of (at least) 2 Route Groups. Each route group will include one of the SIP trunk configured. On each route group parameters, a specific **prefix** should be defined. This way the CUBE will be able to route the outgoing calls to the right SBC, depending on this prefix value:

```
dial-peer voice 11 voip

   answer-address 227....

   destination-pattern 11T

   session-target <SBC1_IP>

   […]

dial-peer voice 12 voip

   answer-address 227....

   destination-pattern 12T

   session-target <SBC2_IP>

   […]
```

The prefix should be stripped using voice translation rules before sending the call to the infrastructure.

Configuration example

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

```
!

! Last configuration change at 12:07:33 CEST Wed Jul 1 2009

!

version 15.1
```


Note: Testing was conducted by Orange Business Services.

```
service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

!

hostname BO27-CUBE1

!

boot-start-marker

boot system flash:c2900-universalk9_npe-mz.SPA.151-1.T.bin

boot-end-marker

!

logging message-counter syslog

logging buffered 2000000

no logging console

enable password cisco

!

no aaa new-model

clock timezone CET 1

clock summer-time CEST recurring last Sun Mar 2:00 last Sun Oct 3:00

!

dot11 syslog

ip source-route

!
```


Note: Testing was conducted by Orange Business Services.

```
!
ip cef
!
no ipv6 cef
multilink bundle-name authenticated
!
voice service voip
 allow-connections sip to sip
 sip
  header-passing
  error-passthru
  no update-callerid
  early-offer forced
  midcall-signaling passthru
  sip-profiles 1
!
voice class codec 1
 codec preference 1 g711alaw
!
voice class sip-profiles 1
 request INVITE sip-header User-Agent modify ".*" "User-Agent: CUBE 15.1(1)T + CUCM 8.0.3.10000-
8 "
```


Note: Testing was conducted by Orange Business Services.

```
 response 183 sip-header Call-Info add "P-EARLY-MEDIA: sendrecv"
 request INVITE sip-header Supported modify "timer," ""
 response 180 sip-header Server modify ".*" "Server: CUBE 15.1(1)T + CUCM 8.0.3.10000-8 "
!
voice translation-rule 1
 rule 1 /^80\(.*\)/ /\1/
!
voice translation-profile Strip80Cld
 translate called 1
!
voice-card 0
!
archive
 log config
  hidekeys
!
interface FastEthernet0/0
 description **CUBE voice**
 no ip address
 duplex auto
 speed auto
!
```


Note: Testing was conducted by Orange Business Services.

```
interface FastEthernet0/0.167
 description **CUBE voice**
 encapsulation dot1Q 167
 ip address 10.227.101.201 255.255.255.0
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 10.227.101.254
no ip http server
no ip http secure-server
!
access-list 1 deny   10.227.101.1
access-list 1 permit any
snmp-server community public RO
snmp-server manager
!
control-plane
```

```
!
ccm-manager fax protocol cisco
!
mgcp fax t38 ecm
!
dial-peer voice 1 voip
 description ** DP to/from BO devices - CUCM SUB **
 preference 1
 answer-address 227....
 destination-pattern 227....
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip options-keepalive
 session protocol sipv2
 session target ipv4:10.227.101.2
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 2 voip
 description ** DP to/from BO devices - CUCM PUB **
 preference 2
 answer-address 227....
```


Note: Testing was conducted by Orange Business Services.

```
 destination-pattern 227....
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip options-keepalive
 session protocol sipv2
 session target ipv4:10.227.101.1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 101 voip
 description ** Incoming calls - Inbound dial peer - SBC side **
 answer-address +.T
 voice-class codec 1
 voice-class sip profiles 1
 session protocol sipv2
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 102 voip
 description ** Outgoing calls - Outbound dial peer - SBC1 side **
 preference 1
 destination-pattern 0.T
```


Note: Testing was conducted by Orange Business Services.

```
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip options-keepalive up-interval 300
 session protocol sipv2
 session target ipv4:172.22.244.217
 dtmf-relay rtp-nte
 no vad
!
gateway
 timer receive-rtp 1200
!
sip-ua
 retry invite 2
 retry response 2
 retry bye 2
 retry cancel 2
 reason-header override
 connection-reuse
 g729-annexb override
!
banner login ^CC
```


Note: Testing was conducted by Orange Business Services.

```
***********************************************
cube1
***********************************************
^C
!
line con 0
line aux 0
line vty 0 4
 password cisco
 login
 no editing
 international
!
scheduler allocate 20000 1000
ntp server 10.20.0.254
event manager environment _sip_header_1 request INVITE sip-header User-Agent modify ".*" "User-
Agent: CUBE
event manager environment _sip_header_2 "
event manager environment _sip_header_3 response 183 sip-header Call-Info add "P-EARLY-MEDIA:
sendrecv"
event manager environment _sip_header_4 request INVITE sip-header Supported modify "timer," ""
event manager environment _sip_header_5 response 180 sip-header Server modify ".*" "Server: CUBE
event manager applet IOSversion
```


Note: Testing was conducted by Orange Business Services.

```
 event timer countdown name IOSversion time 50
 action 1.0 info type snmp oid 1.3.6.1.2.1.47.1.1.1.1.10.3 get-type exact
 action 1.1 set _info_snmp_iosversion "$_info_snmp_value"
 action 1.2 info type snmp oid 1.3.6.1.4.1.9.9.156.1.1.2.1.4.1 get-type exact community public
ipaddr 10.108.101.1
 action 2.0 cli command "enable"
 action 2.1 cli command "config t"
 action 3.0 cli command "no voice class sip-profiles 1"
 action 4.0 cli command "voice class sip-profiles 1"
 action 4.1 cli command "$_sip_header_1 $_info_snmp_iosversion + CUCM $_info_snmp_value
$_sip_header_2"
 action 4.2 cli command "$_sip_header_3"
 action 4.3 cli command "$_sip_header_4"
 action 4.4 cli command "$_sip_header_5 $_info_snmp_iosversion + CUCM $_info_snmp_value
$_sip_header_2"
 action 5.0 cli command "voice service voip"
 action 5.1 cli command "sip"
 action 5.2 cli command "sip-profiles 1"
 action 6.0 cli command "end"
!
End
```


Note: Testing was conducted by Orange Business Services.

CUBE configuration checklist

Step	Item	Configuration commands
1.	Enable SNMP server	`snmp-server manager`

2.	Enable IP2IP gateway	`voice service voip`
		`mode border-element`
		`allow-connections sip to sip`

3.	Configure global SIP settings	`sip`
		`header-passing`
		`error-passthru`
		`no update-callerid`
		`early-offer forced`
		`midcall-signaling passthru`
		`sip-profiles 1`
		`ip address trusted list`
		`ipv4 A.B.C.D`
		`ipv4 E.F.G.H`

4.	Configure codec	`voice class codec 1` `codec preference 1 g711alaw OR g729r8`

5.	Configure SIP User Agent	`sip-ua`
		`retry invite 2`
		`retry response 2`

Note: Testing was conducted by Orange Business Services.

		retry bye 2 retry cancel 2 reason-header override connection-reuse g729-annexb override
6.	Configure dial-peers to CUCM cluster	dial-peer voice 1 voip description ** DP to/from HQ devices - CUCM2 Subscriber ** preference 1 answer-address 180.... destination-pattern 180.... voice-class codec 1 voice-class sip profiles 1 voice-class sip options-keepalive up-interval 300 down-interval 300 retry 5 session protocol sipv2 session target ipv4:10.108.101.2 dtmf-relay rtp-nte no vad ! dial-peer voice 2 voip description ** DP to/from HQ devices - CUCM1

Note: Testing was conducted by Orange Business Services.

		Publisher ** preference 2 answer-address 180.... destination-pattern 180.... voice-class codec 1 voice-class sip profiles 1 voice-class sip options-keepalive up-interval 300 down-interval 300 retry 5 session protocol sipv2 session target ipv4:10.108.101.1 dtmf-relay rtp-nte no vad
7.	Configure dial-peers to Business Talk IP core SBC	dial-peer voice 101 voip description ** Incoming calls - Inbound dial peer - SBC side ** answer-address +.T voice-class codec 1 voice-class sip profiles 1 session protocol sipv2 dtmf-relay rtp-nte no vad !

Note: Testing was conducted by Orange Business Services.

		dial-peer voice 102 voip description ** Outgoing calls - Outbound dial peer - SBC1 side ** preference 1 destination-pattern 0.T voice-class codec 1 voice-class sip profiles 1 voice-class sip options-keepalive up-interval 300 down-interval 300 retry 5 session protocol sipv2 session target ipv4:V.X.Y.Z dtmf-relay rtp-nte no vad
8.	Configure Embedded Event Manager	event manager environment _sip_header_1 request INVITE sip-header User-Agent modify ".*" "User-Agent: CUBE event manager environment _sip_header_2 " event manager environment _sip_header_3 response 183 sip-header Call-Info add "P-EARLY-MEDIA: sendrecv" event manager environment _sip_header_4 request INVITE sip-header Supported modify "timer," ""

Note: Testing was conducted by Orange Business Services.

		event manager environment _sip_header_5 response 180 sip-header Server modify ".*" "Server: CUBE event manager applet IOSversion event timer countdown name IOSversion time 50 action 1.0 info type snmp oid **1.3.6.1.2.1.47.1.1.1.1.10.3** get-type exact action 1.1 set **_info_snmp_iosversion** "$_info_snmp_value" action 1.2 info type snmp oid **1.3.6.1.4.1.9.9.156.1.1.2.1.4.1** get-type exact community public ipaddr **10.108.101.1** action 2.0 cli command "enable" action 2.1 cli command "config t" action 3.0 cli command "no voice class sip-profiles 1" action 4.0 cli command "voice class sip-profiles 1" action 4.1 cli command "$_sip_header_1 $_info_snmp_iosversion + CUCM $_info_snmp_value $_sip_header_2" action 4.2 cli command "$_sip_header_3" action 4.3 cli command "$_sip_header_4" action 4.4 cli command "$_sip_header_5

Note: Testing was conducted by Orange Business Services.

		$_info_snmp_iosversion + CUCM $_info_snmp_value $_sip_header_2" action 5.0 cli command "voice service voip" action 5.1 cli command "sip" action 5.2 cli command "sip-profiles 1" action 6.0 cli command "end"

Note: Testing was conducted by Orange Business Services.

Configuring the Cisco Unified Communications Manager

Note: Please note that all screenshots in this chapter are presented as an example and actual naming of configuration elements in CUCM will probably differ in your deployment. E.g. HQ8 and BO27 are site names specific to Orange Business Services Engineering lab.

General

These configuration guidelines for Cisco Unified Communications Manager permit correct interoperability with CUBE and Orange Business Services Business Talk IP infrastructure.

These recommendations include the following components:

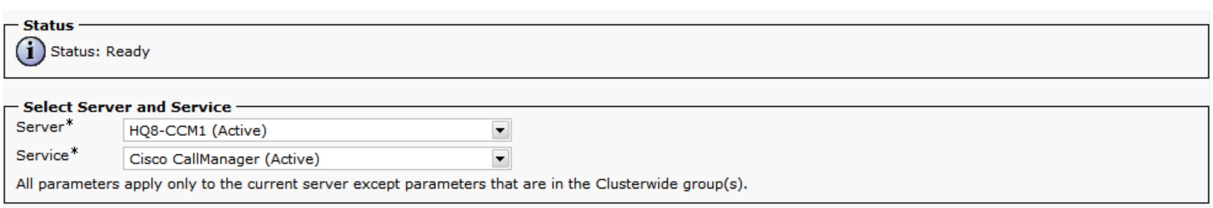
Voice packetization and payload configuration

- Region configuration allow to define the codec used on the LAN/WAN,
- Call Routing to Business Talk IP infrastructure: There are different configuration components to route the calls to our OBS Infrastructure.

Service parameters

The configuration of service parameters on the Cisco Unified Communications Manager is detailed below:

- Select the menu **System > Service Parameters**
- Then select the server of the cluster you want to configure
- Then select the service



Cisco CallManager service

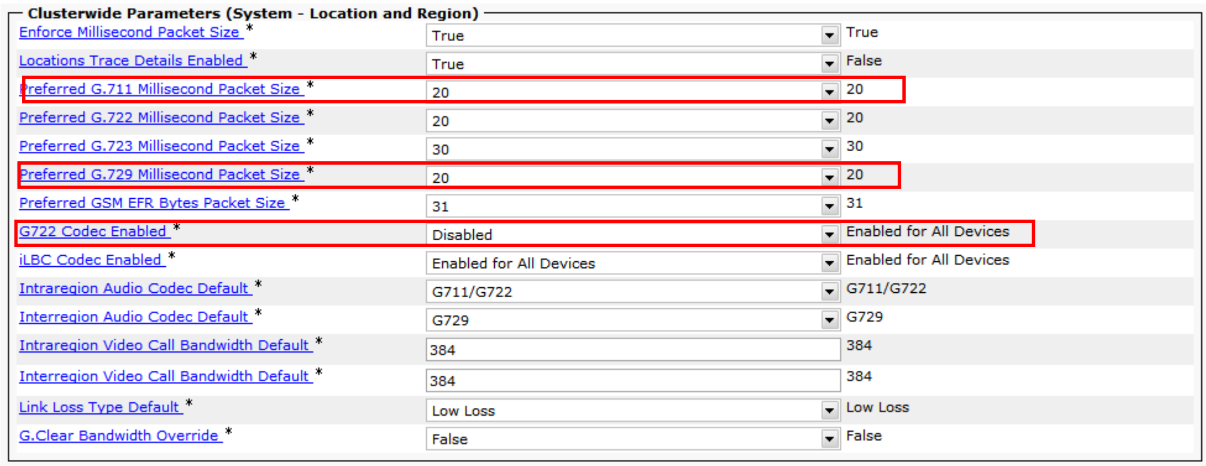
Codec and payload configuration

Navigate to Cisco CallManager service parameters page and go to the **Clusterwide Parameters (System – Location and Region)** section. In this section you will find parameters relative to codec media payload configuration. Please make sure that the following parameters are configured:

Parameter	Value	Description
Preferred G.711 Millisecond Packet Size	20 ms (default value)	-

Note: Testing was conducted by Orange Business Services.

Preferred G.729 Millisecond Packet Size	20 ms (default value)	-
G.722 Codec Enabled	Disabled	G.722 is not supported in Business Talk IP

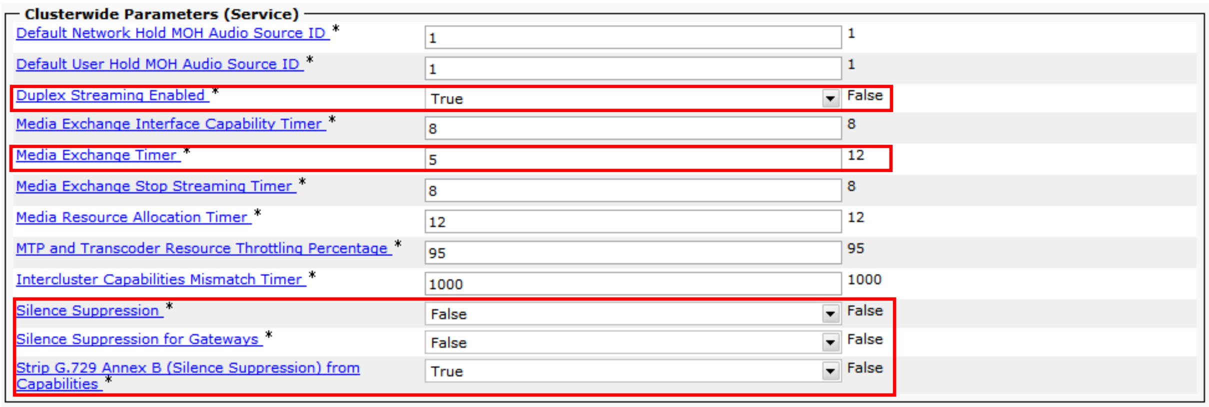


Media configuration

Still on the Cisco CallManager service parameters page, navigate to the **Clusterwide Parameters (Service)** section. In this section you will find other parameters related to media payload configuration. Please configure the following parameters as below:

Parameter	**Value**	**Description**
Duplex Streaming Enabled	True	Determines whether music on hold and annunciator use duplex streaming
Media Exchange Timer	5	This value is recommended by Cisco Systems Engineering
Silence Suppression	False (default value)	-
Silence Suppression for Gateways	False (default value)	-
Strip G.729 Annex B (Silence Suppression) from Capabilities	True	VAD is not used

Note: Testing was conducted by Orange Business Services.

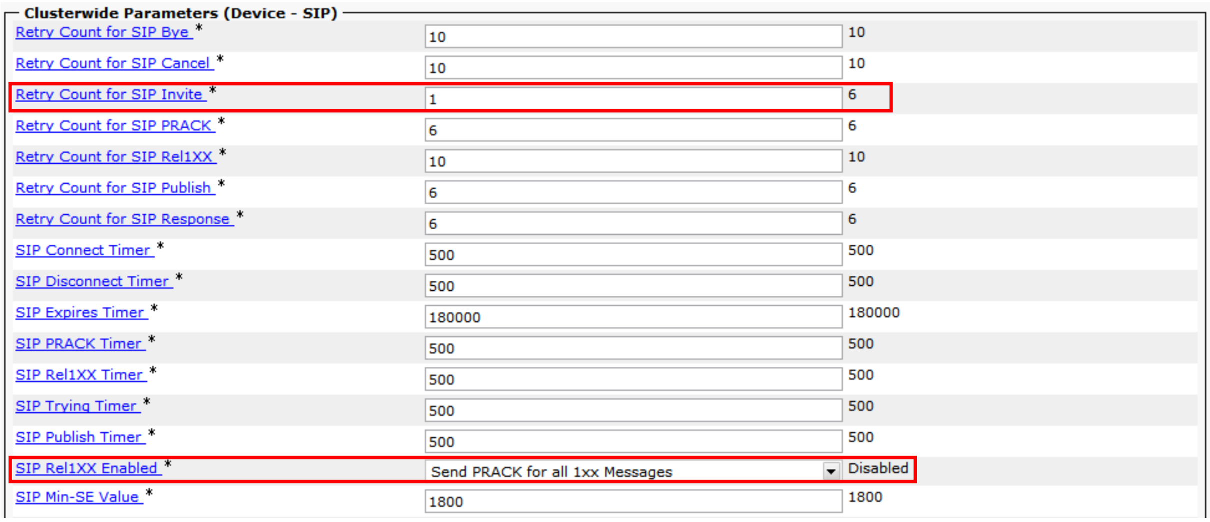


SIP Parameters

Still on the Cisco CallManager service parameters page, navigate to the **Clusterwide Parameters (Device - SIP).** In this section you will find parameters specific to SIP protocol. Please configure the following parameters as below:

Parameter	**Value**	**Description**
Retry Count for SIP Invite	1	This parameter determines number of retry that must be sent to destination when no answer is received. It is set to 1 to reduce at the maximum the delay of rerouting on different CUBEs of the design. This parameter is only manageable at SIP global level and can not be specified for a specific trunk.
SIP Rel1XX Enabled	Send PRACK for all 1xx Messages	This parameter determines whether SIP provisional responses (other than 100 trying message) are sent or received reliably by using the SIP PRACK message. PRACK is negotiated between the Cisco Unified Communications Manager and the remote SIP device by providing the "100Rel" SIP extension in the INVITE and 1xx messages.

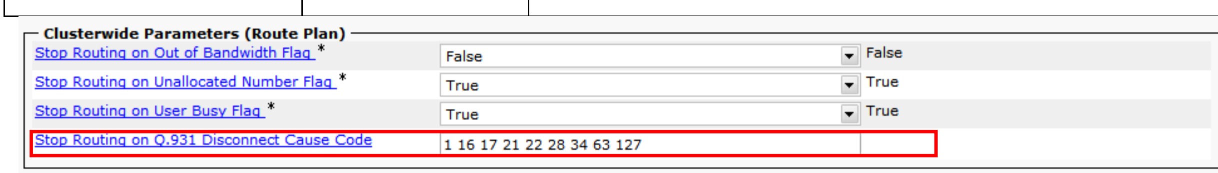
Note: Testing was conducted by Orange Business Services.



Route Plan Parameters

Still on the Cisco CallManager service parameters page, navigate to the **Clusterwide Parameters (Route Plan).** In this section you will found parameters relative to routing rules configuration. Please configure the following parameters as below:

Parameter	Value	Description
Stop Routing on Out of Bandwidth Flag	False (default value)	This parameter determines routing behavior for calls through intercluster trunks when not enough bandwidth exists.
Stop Routing on Unallocated Number Flag	True (default value)	This parameter determines routing behavior for intercluster trunk calls to an unallocated number. An unallocated number represents a dialed directory number that does not exist in a Cisco cluster.
Stop Routing on User Busy Flag	True (default value)	This parameter determines routing behavior for intercluster trunk calls to a busy phone at a remote Cisco cluster.
Stop Routing on Q.931Disconnect Cause Code	1 16 17 21 22 28 34 63 127	This parameter determines routing behavior when a call that is being routed to a remote site through a route list is released and a Q.931 cause code is sent to Cisco CallManager. If the cause code encountered in the message matches a cause code specified in this parameter, a local Cisco CallManager will stop routing the call (it will not be sent to the next device in the route list). Valid values specify any standard valid Q.931 cause codes in the integer range 1 to 127. Use a space to separate multiple cause codes.

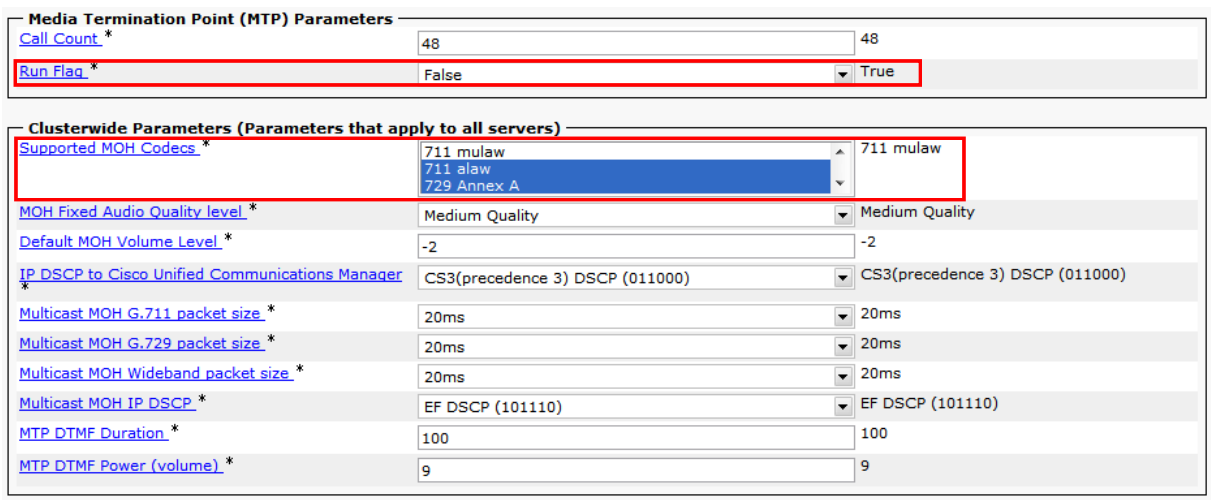


Note: Testing was conducted by Orange Business Services.

Cisco IP Voice Media Streaming Application service

Navigate to the Cisco IP Voice Media Streaming Application service parameters page and configure the following parameters **on all servers in the cluster**:

Parameter	**Value**	**Description**
MTP Run Flag	False	Software MTP is not used with Business Talk IP services
Supported MOH Codec	G711alaw, G729 Annex A	This parameter specifies the codec (compression/decompression) types that the Music on Hold system should support. G711ulaw should not be selected



Note: Testing was conducted by Orange Business Services.

Region and Device Pool Configuration

Region configuration

Region configuration (**System** > **Region** menu in CUCM administration) allows specifying codec interactions between different devices in the cluster. To do this, all devices configured in the CUCM (IP phone, SIP Trunks, GW...) are associated with a region. This section describes how to change configuration parameters, but do not provide the exact design to integrate and codec to use between all regions within you cluster, as each cluster will be different.

The minimal possible number of regions needed is equal to the number of codecs used in the deployment (e.g Region_G.711 and region_G729). However, **configuring each site/zone with a dedicated region provides the highest level of flexibility** (e.g. Region_HQ, Region_BO, Region_RS, Region_WAN, Region_MOH). If specific devices require different codec interaction with other region they must be configured with a dedicated region002E

Note: Regions configuration allows only to specific the upper limit for bandwidth used between 2 regions. When you select G.711 codec between 2 regions it means in fact, that any codec with bandwidth equal or lower then G.711 can be used between them (e.g. G.729)

Codecs authorized for use with Business Talk IP SIP trunking services:

- G.729
- G.711 ALaw - CUCM doesn't allow to specify G.711 companding type (A-law or u-law), so simply choose G.711

Region configuration example

Please consider the following customer design:

- central site (HQ) with CUCM cluster
- a single remote site (RS) with call processing on HQ

In such scenario, following 3 regions could be used:

Region	**Purpose**
HQ	Assigned to devices (IP phones and gateways) in the HQ site`
RS1	Assigned to devices in the Remote Site 1
WAN	Assigned to SIP trunk to Business Talk IP
MoH	Assigned to Music On Hold server, to allow specifying dedicated codec for MoH

Example of the codec selection for for customer using G.729 (G.711 for intrasite calls and low-rate G.729 for calls over the WAN)

Note: Testing was conducted by Orange Business Services.

From To	HQ	RS	MOH	WAN
Default	G711	G729	G711	G729
RS	G729	G711	G729	G729
MOH	G711	G729	G711	G729
WAN	G729	G729	G729	G729

Device Pool configuration

Depending on the number of Cisco Unified Communications Manager Group and Region, the number of Device pool may vary (**System > Device Pool** menu in CUCM administration).

An example with a site with 2 server (1 Publisher and 1 subscriber): the Cisco recommendation indicates only 1 Call Manager group for this type of configuration with the Primary (SQL primary database) server acting as the subscriber (backup call manager for call routing) and the Secondary (SQL replicated database) backup server acting as the publisher (main call manager for call routing).

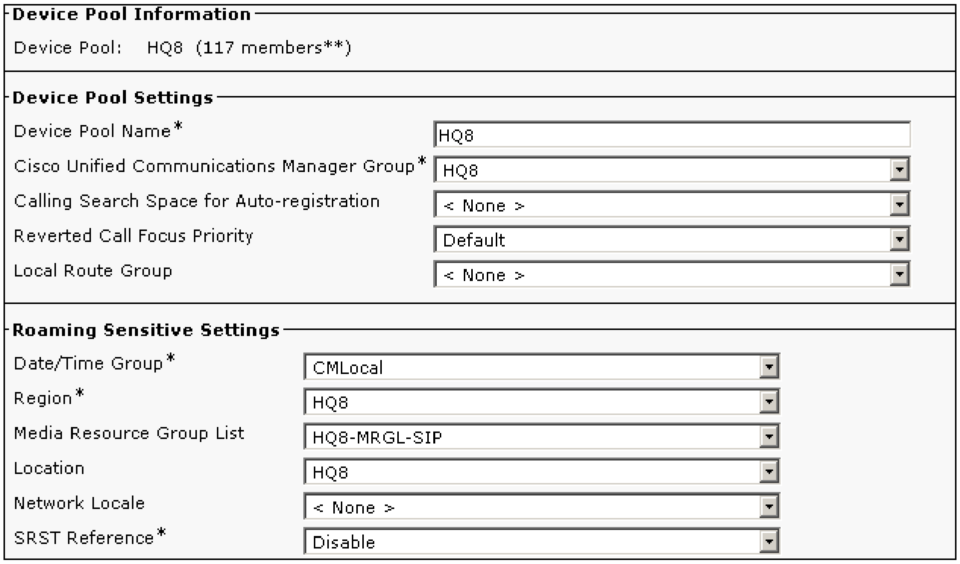
As this CUCM deployment model is limited to only one CCM group, the number of Devices pool is limited to the number of region. Indeed, we will have at lease 1 Device pool per site. All these Device Pools will have the following characteristics:

- The same CallManager Group
- Date/time Group : the same for all device pool if all the sites are in the same time zone and use the same date and time format
- Region : each Device Pool should probably include
- Media Ressource

Configuration Example:

Device Pool	Region	CallManager Group
DP_CentralSite	default	default
DP_Remote_Site_A	Remote_Site_A	default
DP_Remote_Site_B	Remote_Site_B	default
DP_MOH	MOH	default
DP_WAN	WAN	default

Note: Testing was conducted by Orange Business Services.



Call Admission Control (Locations)

Locations (**System > Device Pool** menu in CUCM administration) provide means necessary to provide Call Admission Control (CAC) to calls from/to CUCM cluster. This feature allows specifying maximum bandwidth for calls between any two locations (each device is configured with one). CAC protects voice from voice, i.e. it won't allow more voice calls then there is provisioned for by QoS (e.g. Priority Queueing).

Note: Warning! RSVP-based locations are not supported!

For customers using IP VPN to connect all their locations, Static Locations CAC feature is well-suited. In such case, **the default Hub_None location with unlimited bandwidth should be used to represent the IP VPN cloud (no devices should be associated with it).** Each site should have a dedicated location to track bandwidth used on it's WAN link.

For calculation purpose, the CUCM assumes that each call consumes the following amount of bandwidth:

- G.711 call uses 80 kbps
- G.729 call uses 24 kbps

Configuration example for customer using G.729 codec in the IP VPN WAN (both offnet calls via Business Talk IP and calls between different clusters):

- Remote Site 1 Location (2 calls maximum) = 24*2=48
- Remote Site 2 Location (3 calls maximum) = 24*3=72
- Central Site Location - Headquarter (10 calls maximum) = 24*10=240 kbps

Note: Testing was conducted by Orange Business Services.

Media resources

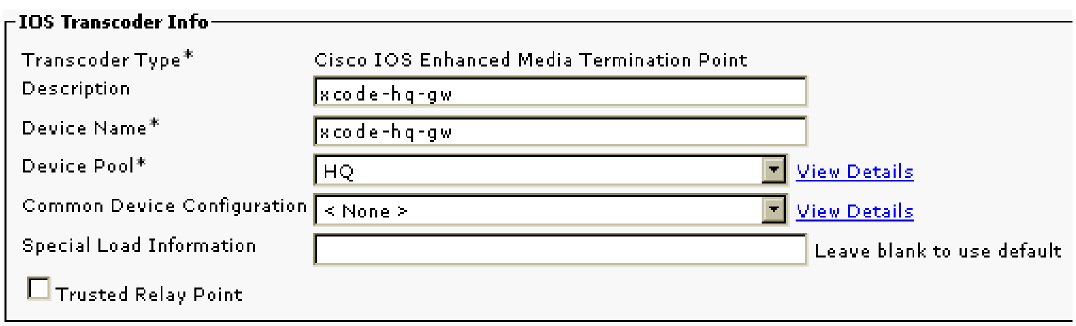
This section addresses the configuration to be done on the Cisco Cisco Unified Communications Manager. Some parts of configuration should be also completed on the Cascaded IOS media gateway Router.

Transcoder configuration

Warning! Software MTP are not supported by Business Talk IP SIP Trunking design.

Transcoder can be added to the cluster after selecting **Media Resources > Transcoder** from the CUCM Administration menu. Use following settings when configuring it:

Setting	**Value**	**Description**
Transcoder Type	Cisco IOS Enhanced Media Termination Point	This is the appropriate transcoder type for PVDM2 modules in ISR 2800/3800 series routers
Device Name	use the name configured in sccp ccm group in the IOS	-
Device Pool	use the HQ device pool	-



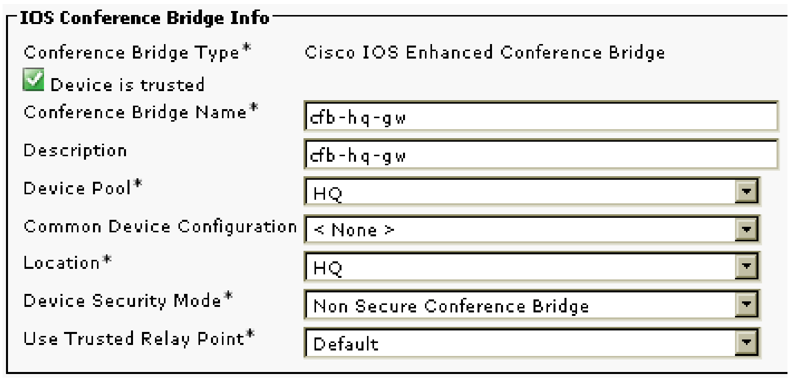
Conference Bridge configuration

Hardware Conference Bridge is necessary for conferences involving G.729 flows or more than 3 participants in G.711. Transcoder can be added to the cluster after selecting **Media Resources > Conference Bridge** from the CUCM Administration menu. Use following settings to configure it:

Setting	**Value**	**Description**
Conference Bridge Type	Cisco IOS Enhanced Media Termination Point	This is the appropriate transcoder type for PVDM2 modules in ISR 2800/3800 series routers
Device Name	use the name configured in sccp ccm group in the IOS	-

Note: Testing was conducted by Orange Business Services.

Device Pool	use the HQ device pool	-



Unicast Music on Hold

Music On Hold servers are configured in **Media Resources** > **Music On Hold Server**. They are software media resources automatically provisioned when IP Media Voice Streaming service is activated.

Media Resource Group Lists configuration

Media Resource Groups **(MRG)** and Media Resource Group Lists **(MRGL)** are used by CUCM to control access to media resources from different device classes. MRGs are included into MRGL to indicate their preferences. MRGLs can be associated with devices and Device Pools.

Each MRG groups similar media resources (e.g. software resources in HQ, hardware resources in HQ, hardware resources in RS, etc.). CUCM uses load balancing to select media resources within a single MRG, but respects the priority of MRGs on the MRGL.

Note: Warning! Media Resources, which are not associated with any MRG are available to every device in the cluster by default.

In order to control media resources usage exactly, a dedicated MRG including all resources must be created (eg. MRG_All). This way each device can use only media resources explicitly configured in its MRGL. This is required to avoid issues discovered during Engineering validation.

Media Resource Groups and Media Resource Group Lists are configured on the Cisco Unified Communications Manager in following places:

- Media Resources > Media Resource Group
- Media Resources > Media Resource Group List

Note: **Warning:** To avoid issues with unexpected transcoders involvement, we recommend **not to** associate transcoder resource on MRGL used on SIP Trunk between CUCM and CUBE and assign all transcoding resources to a dedicated MRG to prevent their default availability.

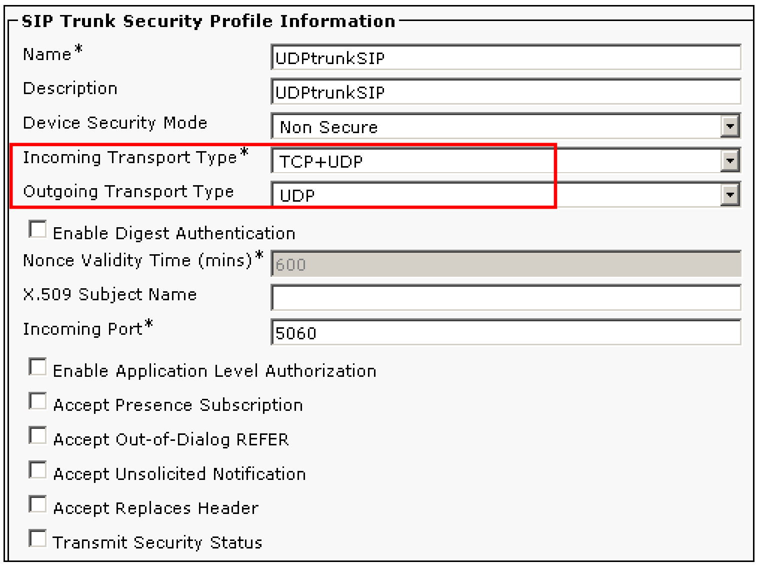
Off-net calling via Business Talk IP

Off-net calls are served by two SIP trunks configured between CUCM and both CUBEs.

SIP Trunk Security profile

SIP trunks to CUBEs 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.

- Add a new SIP Trunk Secuirity profile by going to **System > Security Profile > SIP Trunk Security Profile**
- Select **UDP** for **Outgoing Transport Type**



SIP Trunk to Cisco Unified Border Element

Configure two new SIP trunks to both CUBEs in **Device > Trunk.** Use following configuration settings:

Setting	Value	Description
Trunk Type	SIP	-
Device Protocol	SIP	-
Redirecting Diversion Header Delivery for Outbound Calls	NOT checked	not supported due to existing issue: Cisco DDTS CSCso92803
Destination address	IP address of CUBE	-
SIP Trunk Security Profile	previously created SIP Trunk Security Profile	-
SIP Profile	Standard SIP Profile	-
DTMF Signaling Method	RFC 2833	RFC 2833 is the only method supported by

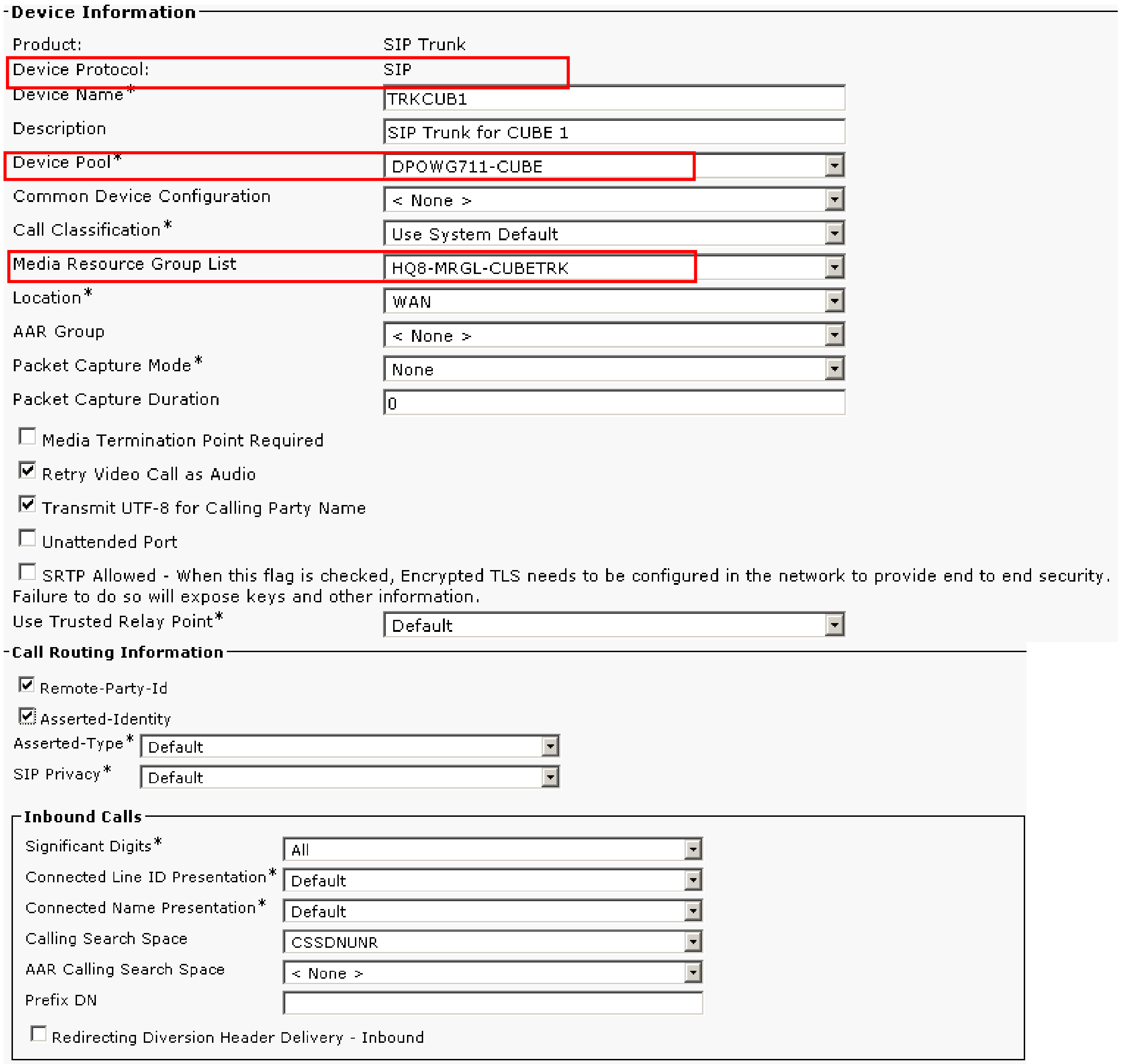
		Business Talk IP service. This setting is required to avoid transcoder/MTP usage

Note: 2 SIP Trunk must be configured for fallback process. Repeat the same process for the SIP trunk to the backup CUBE. These 2 SIP Trunks differ only by a different destination address.

Note: Warning: Because of the issues discovered during Engineering valiadation, the SIP trunks configured in CUCM should not be associated with Media Resource Group List containing transcoders. Transcoders should not be available in the default Media Resource Group. See chapter on MRGL configuration for more information.

Cisco Unified Communications Managers always prefers G711 u-law from A-law, when both are available. OBS infrastructure offers only G711 A-law. This behavior can be problematic in case of transfer and conference scenario for instance. Because of this, it is better to force CUCM to use G711 Alaw by configuring media resource without transcoder.

Note: Testing was conducted by Orange Business Services.



Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	TRKCUB1
Description	SIP Trunk for CUBE 1
Device Pool*	DPOWG711-CUBE
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	HQ8-MRGL-CUBETRK
Location*	WAN
AAR Group	< None >
Packet Capture Mode*	None
Packet Capture Duration	0

- [ ] Media Termination Point Required
- [x] Retry Video Call as Audio
- [x] Transmit UTF-8 for Calling Party Name
- [ ] 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.

Use Trusted Relay Point*	Default

Call Routing Information

- [x] Remote-Party-Id
- [x] Asserted-Identity

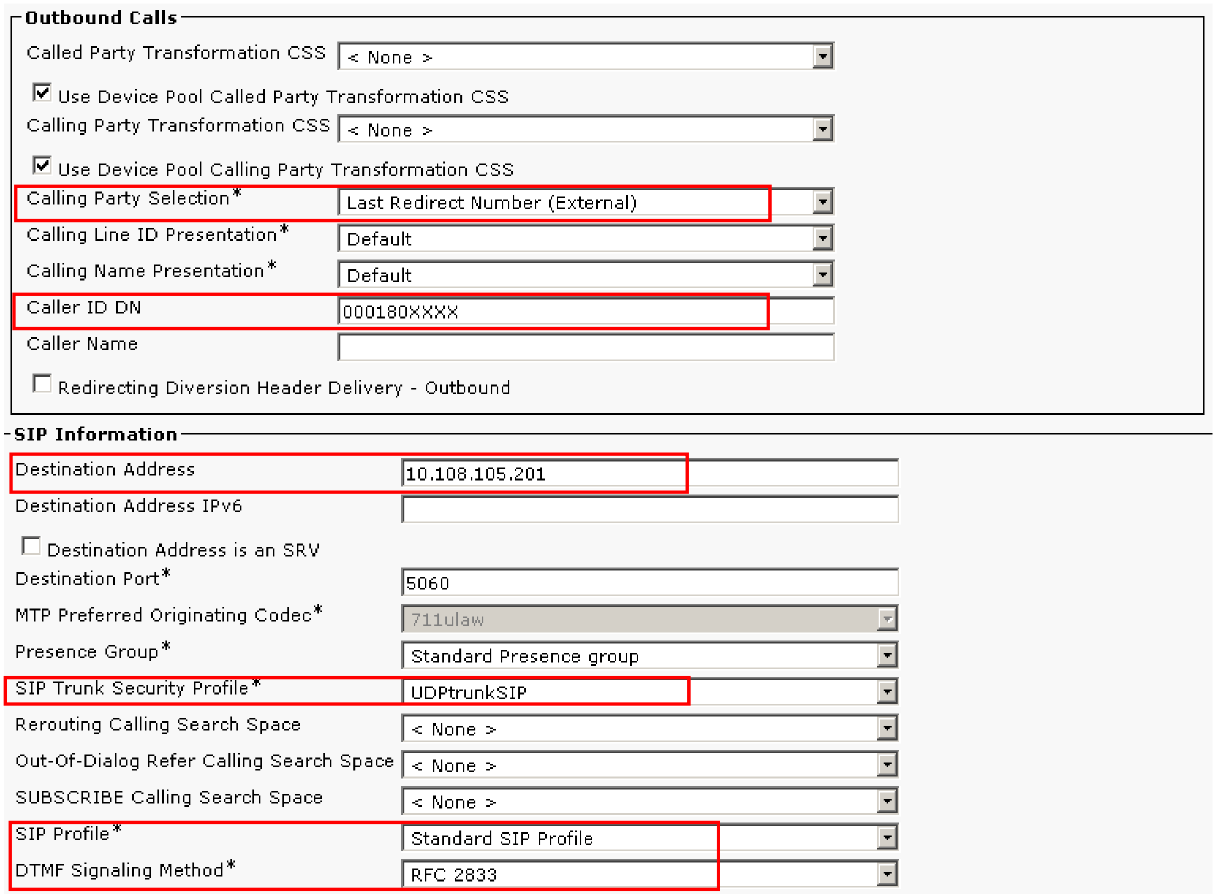
Asserted-Type*	Default
SIP Privacy*	Default

Inbound Calls

Significant Digits*	All
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
Calling Search Space	CSSDNUNR
AAR Calling Search Space	< None >
Prefix DN	

- [ ] Redirecting Diversion Header Delivery - Inbound

Note: Testing was conducted by Orange Business Services.

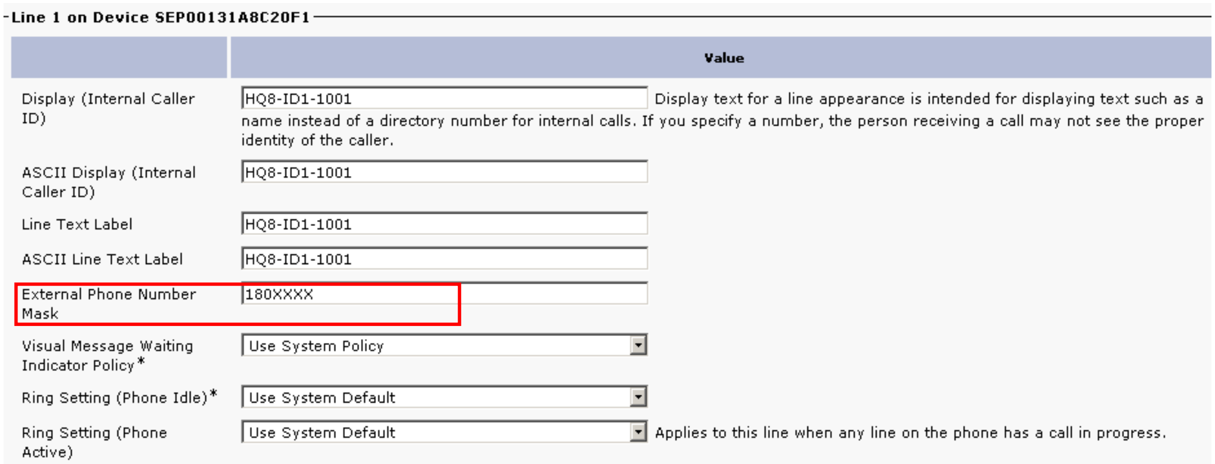


Note: **Warning:** Redirecting Diversion Header Delivery is not supported on SIP Trunk between CUCM and CUBE due to existing issue: Cisco DDTS CSCso92803 (Phone mask not being used in SIP Diversion Header for CFA calls).

Line configuration

In case “Last redirect (External)” is selected in the SIP Trunk configuration, the line configuration has to be updated: **External Phone Number Mask** should be set to **Site code + XXXX (extension).**

Note: Testing was conducted by Orange Business Services.


Line 1 on Device SEP00131A8C20F1

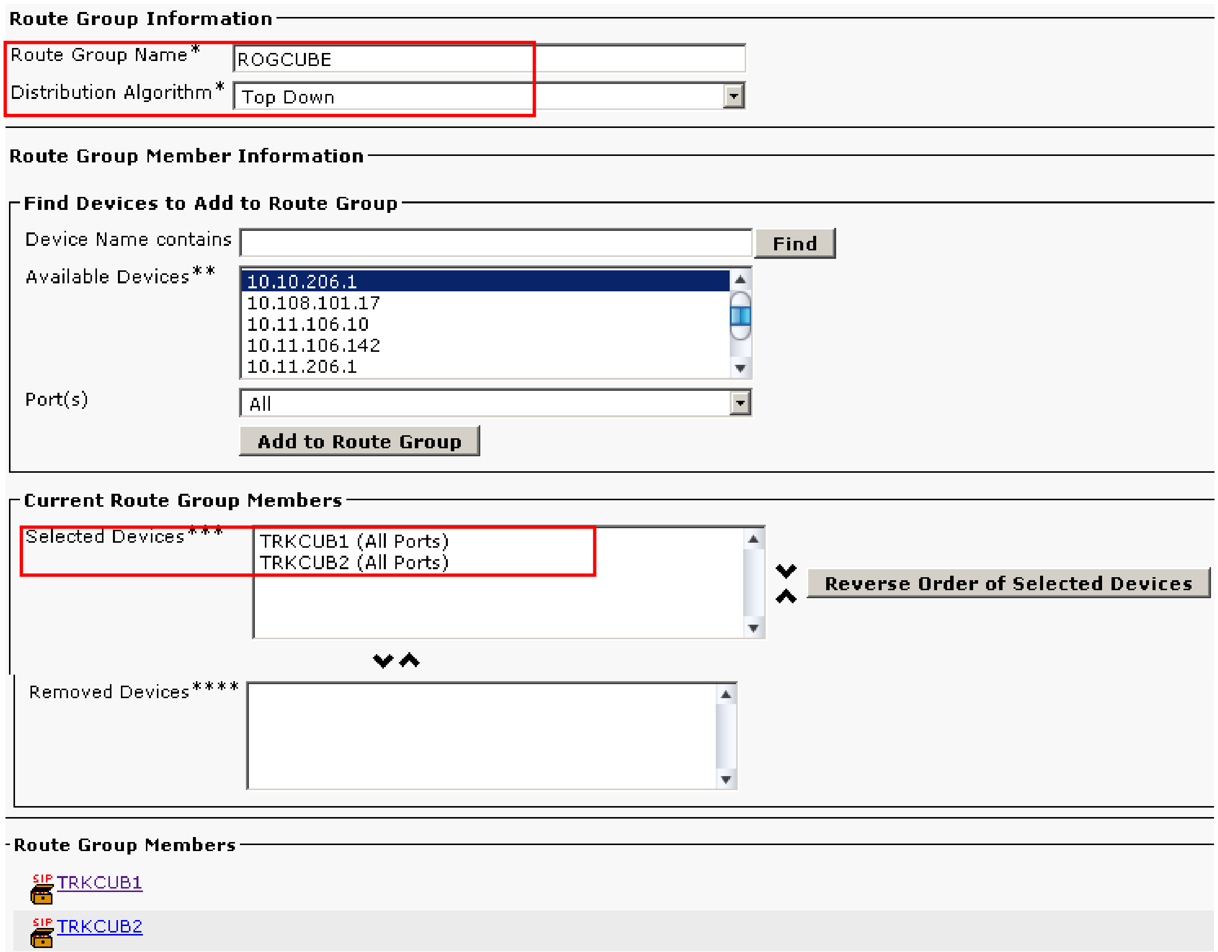
	Value
Display (Internal Caller ID)	HQ8-ID1-1001 Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID)	HQ8-ID1-1001
Line Text Label	HQ8-ID1-1001
ASCII Line Text Label	HQ8-ID1-1001
External Phone Number Mask	180XXXX
Visual Message Waiting Indicator Policy*	Use System Policy
Ring Setting (Phone Idle)*	Use System Default
Ring Setting (Phone Active)	Use System Default Applies to this line when any line on the phone has a call in progress.

Route group

Configure a new Route Group for the newly configured SIP trunks to CUBE by going to **Call Routing > Route/Hunt > Route group**. Use following settings when configuring it:

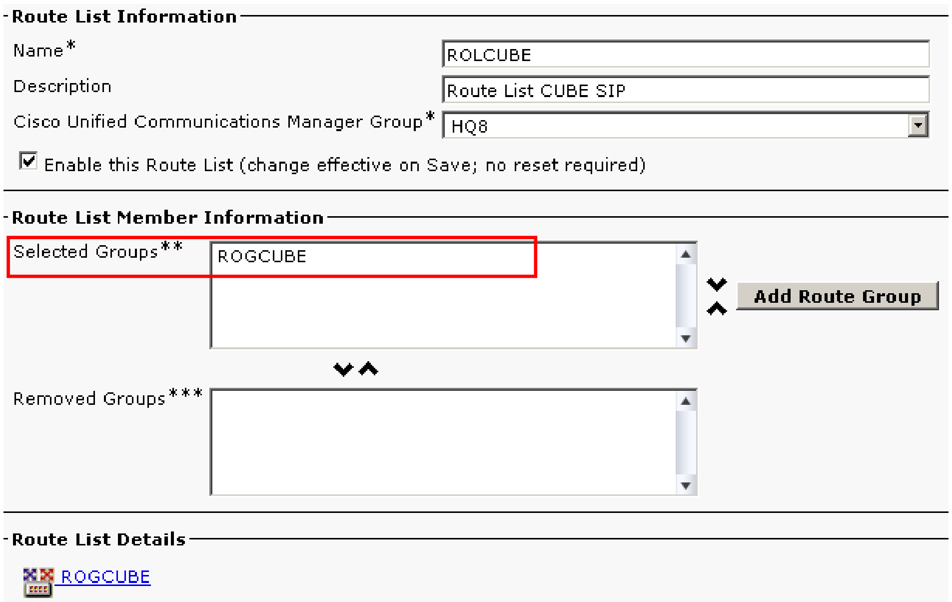
Setting	**Value**	**Description**
Route Group Name	any meaningful name can be used	-
Distribution algorithm	Top Down	Primary CUBE should always be tried first
Selected devices	both SIP trunks to CUBEs	use the trunks configured earlier to primary and backup CUBEs. The order is important.

Note: Testing was conducted by Orange Business Services.



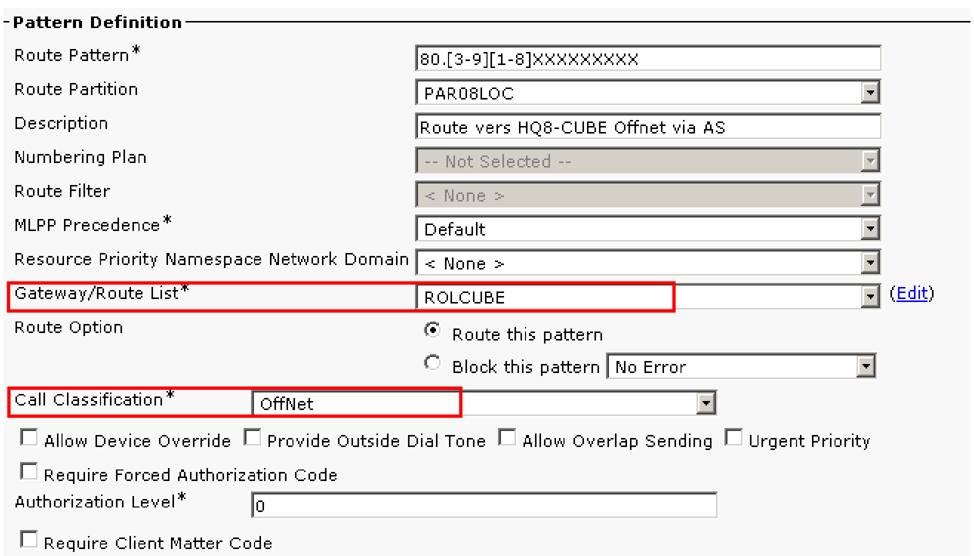
Route list

Create a new Route List by going to **Call Routing > Route/Hunt > Route list**. Add the previously configured Route Group.



Route Patterns

Configure all required Route Patterns (**Route Plan > Route/Hunt > Route Pattern**) for all off-net destinations and associate them with previously configured Route List containing SIP trunks to CUBEs. Example below of Off-net route pattern



Digit manipulation

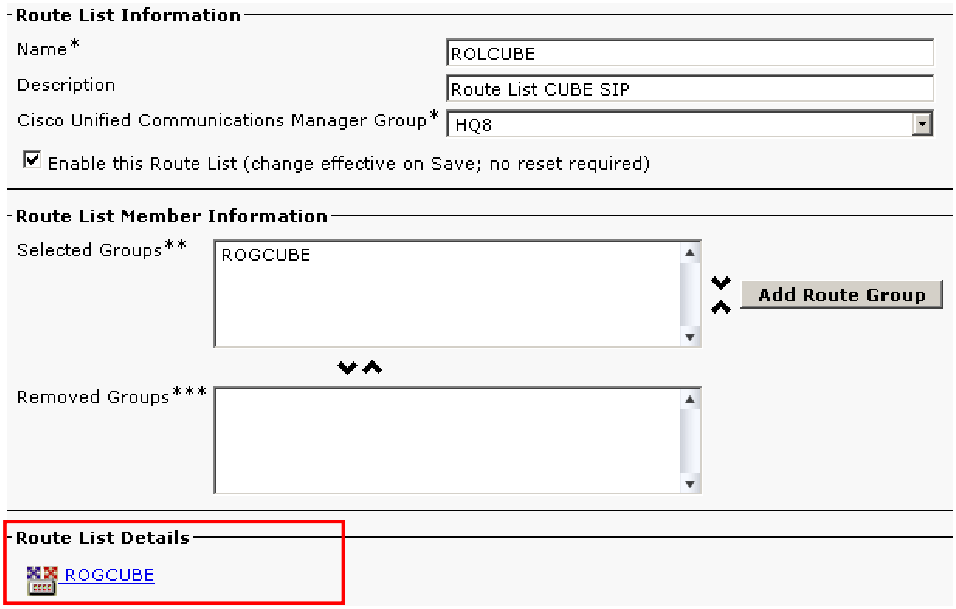
Business Talk IP infrastructure expects the DNIS (called number) to be in specific form: **00 prefix + E.164 destination address**, e.g. 0033172697967 for OBS conference bridge.

Configuration of the DNIS can be made in the **Route List Detail** configuration:

- Go to the Route List configuration: Call Routing > Route/Hunt > Route list

Note: Testing was conducted by Orange Business Services.

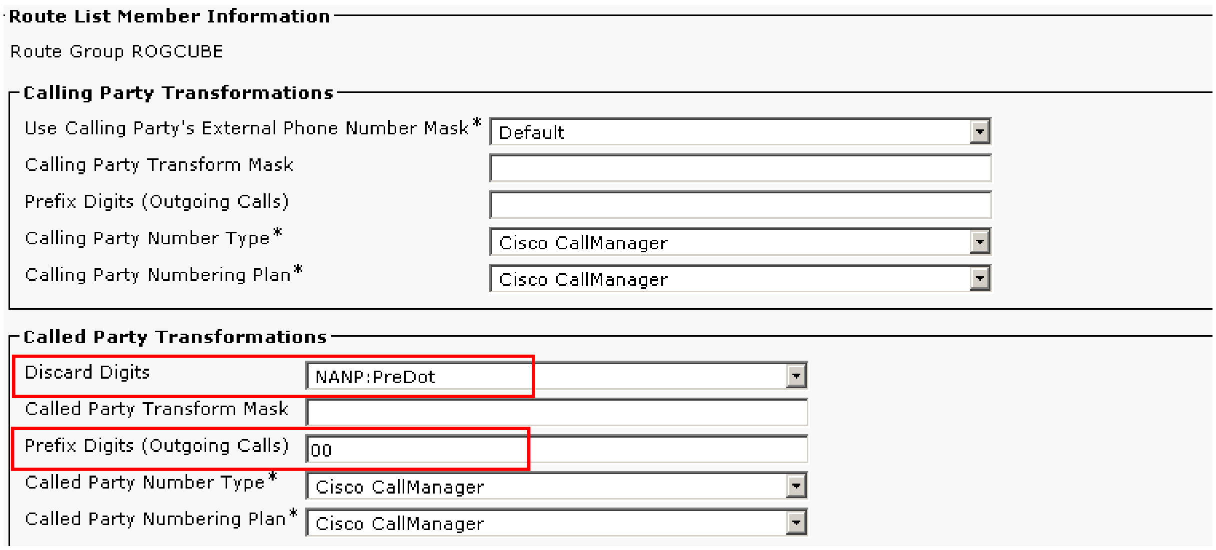
- Select the Route List containing SIP trunks to CUBEs
- Click on the Route Group in "Route List Member Information" area



In the route list detail configure following number manipulation in the **Called Party Transformation** area:

Setting	**Value**	**Description**
Discard digits	NANP: PreDot	Strips the digits in front of dot in the route Pattern (e.g. strips 9 from 9.0033XXXXXXXXX)
Prefix Digits (Outgoing calls)	00	Prefixes the dialed number with 00

Note: Testing was conducted by Orange Business Services.



On-net calling between two sites

In order to interconnect two clusters belonging to the same customer, a **direct SIP Trunk** can be configured between these 2 sites.

The **configuration of such intercluster SIP Trunk is the same as the one described for off-net calls** (SIP Trunks between CUCM and CUBE), the only difference being the Destination Address (which should point to remote cluster).

Note: Testing was conducted by Orange Business Services.

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	TRKBO27
Description	SIP Trunk to BO27
Device Pool*	DPOWG711
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	HQ8-MRGL-SIP
Location*	BO27
AAR Group	< None >
Packet Capture Mode*	None
Packet Capture Duration	0

- [ ] Media Termination Point Required
- [x] Retry Video Call as Audio
- [x] Transmit UTF-8 for Calling Party Name
- [ ] 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.

Use Trusted Relay Point*	Default

SIP Information

Destination Address	10.227.101.1
Destination Address IPv6	

- [ ] Destination Address is an SRV

Destination Port*	5060
MTP Preferred Originating Codec*	711ulaw
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	UDPtrunkSIP
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
DTMF Signaling Method*	RFC 2833

Note: For redundancy, at least two SIP trunks pointing at 2 different servers in the remote cluster should be configured. Repeat the same process for each SIP trunk. These 2 SIP Trunks differ only by a different destination address.

Note: Because of the issues discovered during Engineering validation, the SIP trunks configured in CUCM should not be associated with Media Resource Group List containing transcoders. Transcoders should not be available in the default Media Resource Group. See chapter on MRGL configuration for more information.

Numbering format expected by Business Talk IP infrastructure

Calling/Called number format rquired by Business Talk IP

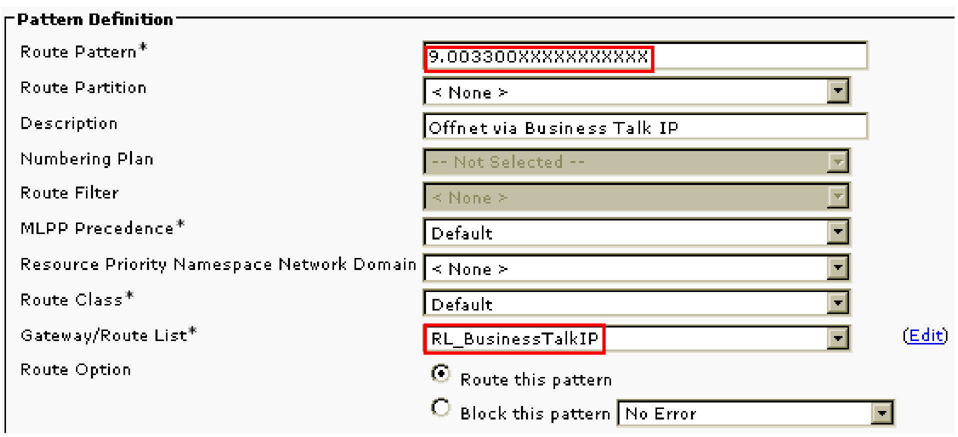
Both calling number (ANI) and called number (DNIS) must follow specific format rules to be correctly handled by the BT infrastructure. This treatment has become mandatory in BT SIP Trunking, because the format of the private numbering plan cannot be identified.

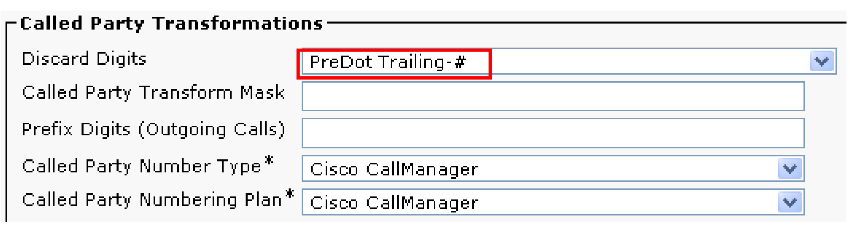
Supported format for the calling and called identity digits are:

- Private Number
- "+CCNSN" (international number)
- "[international prefix]CCNSN" (international number)
- "[national prefix]NSN" (national number)

Route Pattern configuration example

Go to Call Routing > Route/Hunt > Route Pattern and press Add new button.





Setting	Value	Description
Route Pattern	Specific Route Pattern. It's only an example	Specify appropriate Route Pattern
Gateway/Route List	Route List name	Route List name earlier configured
Discard Digits	PreDot Trailing#	It specifies how to inspect digit before they are sent to CUBE

Note: Testing was conducted by Orange Business Services.

Acronyms

Acronym	Definitions
SIP	Session Initiation Protocol
MGCP	Media Gateway Control Protocol
SCCP	Skinny Client Control Protocol
Cisco UCM	Cisco Unified Communications Manager
CUCM	Cisco Unified Communications Manager
Cisco UBE	Cisco Unified Border Element
CUBE	Cisco Unified Border Element

Note: Testing was conducted by Orange Business Services.

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