Application Note

Orange Business Services

Business Talk IP (France and International) connecting:

Cisco Unified Communications Manager 12.0 with Cisco Unified Border Element 12.0 (IOS-XE 16.6.3 Everest) using SIP

April 2, 2019

Testing was conducted in Orange Business Services lab.

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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. This device is by default Orange Business Services' IPVPN router, the Customer Premises Equipment (CPE). As an intermediary device between Customer IP-PBX and Orange Session Border Controller, Cisco Unified Border Element (CUBE) 12.0 can be used. The Cisco Unified Border Element 12.0 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 12.0 connected to Business Talk IP.

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

- This application note describes how to configure a Cisco Unified Communications Manager (Cisco UCM) 12.0 and Cisco Unified Border Element (CUBE) 12.0 for connectivity to Business Talk IP SIP trunking service. The deployment model covered in this application note is CPE (Cisco UCM 12.0) to PSTN (Orange Business Services Business Talk IP). This document does not address 112 emergency outbound calls. For 112 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, etc).
- The CUCM configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Orange Business Services 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 Orange Business Services SIP trunking 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:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12.html

Testing was conducted in Orange Business Services lab.

Network Topology

Basic Call Setup

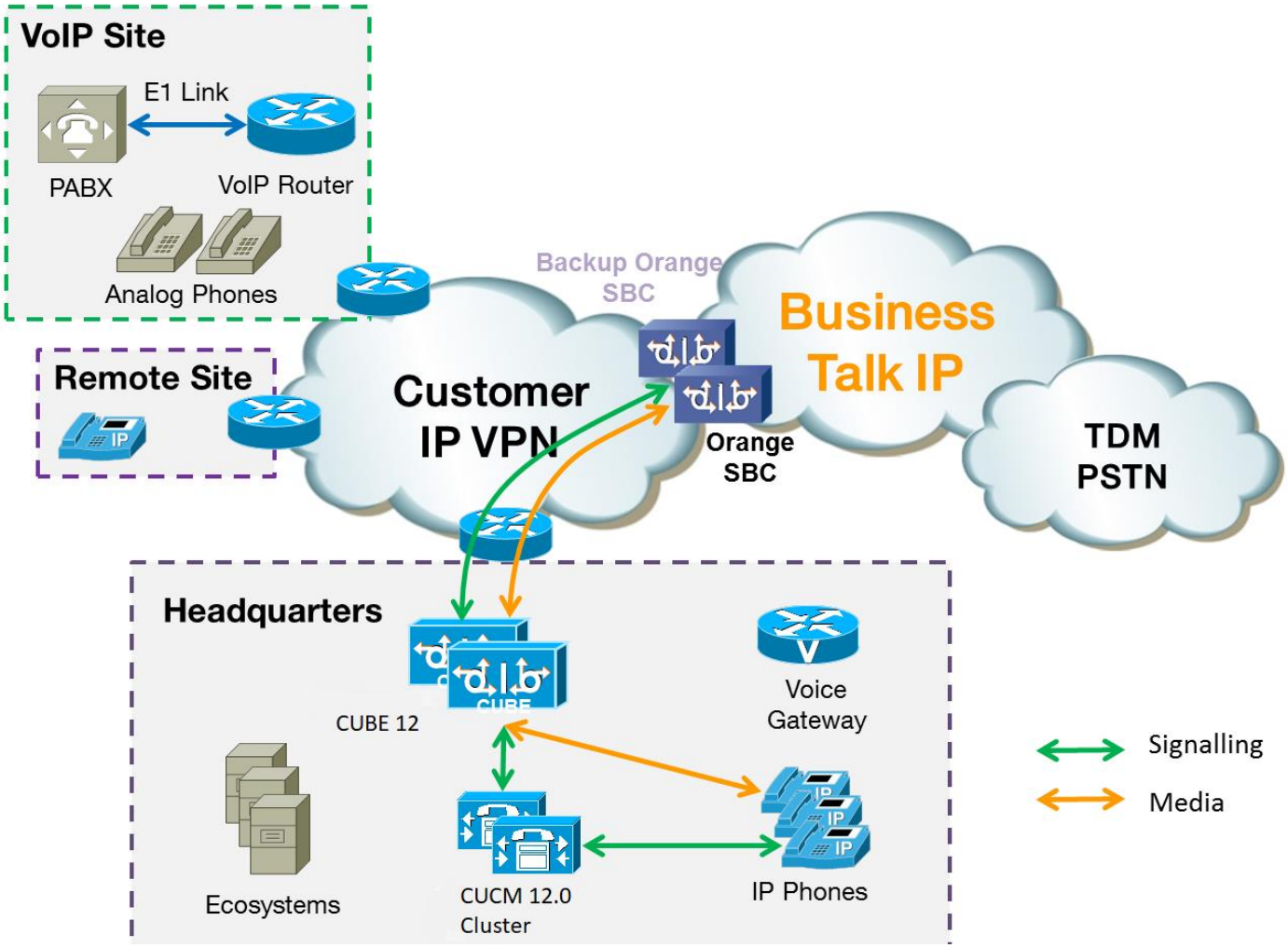


Figure 1 Network Topology

Testing was conducted in Orange Business Services lab.

System Components

Hardware Components

- UCS-B, UCS-C or UCS-E servers for CUCM and ecosystems
- ISR G2 2900/3900 series routers with PVDM3 modules for media resources
- ISR4300/4400 series routers with PVDM4 modules for media resources (excluding video resources)
- IP phones 6900/7800/7900/8800/8900/9900 series and DX650/70/80 (different models, both SIP and SCCP where supported, please consult "Features not supported" for restrictions)
- Cisco ATA190
- Cisco Voice Gateways series (VG20xXM/310/320/350)

Software Requirements

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

- Cisco Unified Communications Manager 12.0 (please note that we recommend to use 12.0.1.21900-7 version)
- IOS 16.06.3 (ISR G3) or IOS 15.7.3M (ISR G2) for Cisco Unified Border Element 12.0
- IOS 15.7.3M for IOS ISR gateways (required for DSP media resources and local PSTN failover)
- IOS 15.7.3M for IOS Voice Gateways
- Cisco Unity Connection 12.0

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
- SIP Early Offer (short media cut-through times)
- 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)
- Media flow-through and flow-around on CUBE

Testing was conducted in Orange Business Services lab.

Features Not Supported

- Offnet calling through SIP trunk to Business Talk IP trunk in SRST mode (e.g. CUCM cluster down, but IP VPN available). Recommended SRST deployments should use alternate TDM offnet for all survivable calls.
- 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.

Testing was conducted in Orange Business Services lab.

Caveats

- Business Talk IP supports G.711 A-law companding. For incoming offnet calls to ecosystems supporting only G.711 u-law, transcoding resources are required. Examples of such ecosystems include Cisco Unity Express and Unity.
- Business Talk IP complies with SIP RFC2833 and uses this method of choice to convey DTMF in-band. However, because of JTAPI limitations, this method is not supported by all Cisco ecosystems. Cisco Unified Contact Center Express, Cisco Unity Express (and possibly other ecosystems integrating with CUCM using JTAPI) do not support inband DTMFs natively and require additional MTP or transcoder resource to translate between inband and out-of-band DTMF transport. Because of media resource selection algorithm used by CUCM, this might lead in some cases to hairpinning of the RTP stream (e.g. transcoder/MTP allocated in headquarters for incoming offnet call to a remote site).
- CUCM requires MTP or transcoding resources for external ad-hoc conference participants using RFC2833 DTMFs. Every offnet participant connected to a conference via Business Talk IP will use additional transcoding/MTP resource apart from the one conferencing resource.
- IP phones running SCCP firmware must use Skinny protocol version higher than v20 in order to support Early Offer for outgoing calls via SIP trunks (Business Talk IP requirement). Following IP phone models are confirmed to support Early Offer when running newest SCCP firmware: 6901, 6911, 6921, 6941, 6945, 6951, 6961, 7906, 7911, 7931, 7941, 7942, 7961, 7962, 7965, 7945, 7970, 7971, 7975. Cisco WiFi phones and legacy devices like 7940/7960 do not support Early Offer when running SCCP firmware. Software MTP resources can enable mentioned legacy device for Early Offer support.
- Only IOS-based Enhanced Software MTP is supported as MTP with BT/BTIP SIP trunking. Hardware IOS-based Software MTP co-located with CUCM server (enabled via IP Voice Media Streaming Application service) is not supported.
- Transfers involving ATA190 analog gateway and BT/BTIP SIP trunk require MTP resources to complete the call successfully.
- Business Talk IP requires CUCM to include and send exact version information in User-Agent/Server SIP header. There is no need to send version information in SIP normalization script (manipulation of User-Agent/Server SIP header) because CUCM 12.0 sends this info natively.
- Other non-compliances to Orange trunking require application of SIP normalization script on CUCM, including SIP Error and Release Cause re-mapping, so the CUCM re-reroutes the call correctly in case of primary Business Talk IP entity failure.
- Semi-attended call transferring via CUxAC requires transcoder resources during the ringing state of call semi-attended transfer.
- Delayed semi-attended transfer initiated from CUxAC desktop application fails. Workaround available through setting external access number to be the same as pattern for off-net calls (i.e 8.!), and setting the maximum internal device digit length to 7 (to distinguish onnet and offnet calls). But it also makes it impossible to perform onnet connections to numbers beginning with 8 (i.e LO BLB) as even though they are seven digits numbers, they are treated as external numbers.
- Small gap in ringing tone and MoH during semi-attended transfer from CUxAC**.** Call itself is successful.
- MS Exchange 2010/2013 does not support G.729. Transcoder resources required for compliance with BT/BTIP when G.729 codec usage is required.
- SIP Normalization Script name to be provided by Cisco. How to apply the script has to be detailed by Cisco
- Supervised transfer between SIP phones with Send send-receive parameter enabled- MTP resources are used in the moment of transfer.
- CUCM sends 500 internal error when performing semi-attended transfer from SIP phone, although call transfer is successful.
- Conference initiated by 8941 – one way audio with VG2xx.
- WebEx Meeting Server sends in communication Comfort Noise (CN) packets. Cisco TAC can deactivate CN on manually using admin level reserved only for Cisco.

Testing was conducted in Orange Business Services lab.

Configuration

In the case of deployment with CUCM only (without CUBE) the configuration in CUBE section shall be omitted. In CUCM SIP Trunk Configuration section, SBC IP address should be configured in destination address field (instead of CUBE address).

Configuring the 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. For two SIP trunks two IP addresses must be configured.

```
interface GigabitEthernet0/0
 description CUBE Voice Interface
 no ip address
 duplex auto
 speed auto
!
interface GigabitEthernet0/0.433
 description *** HQ433-CUBE -> VLAN433 ***
 encapsulation dot1Q 433
 ip address 6.4.33.30 255.255.255.192
```

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
```


Testing was conducted in Orange Business Services lab.

Global CUBE settings

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

```
voice service voip
      mode border-element license capacity [session count]
      allow-connections sip to sip
      sip
          header-passing
          error-passthru
          pass-thru headers unsupp
          no update-callerid
          early-offer forced
          midcall-signaling passthru
          sip-profiles 1
             ip address trusted list
                    ipv4 A.B.C.D   ! primary SBC IP address
                    ipv4 E.F.G.H   ! backup SBC IP address
```

Explanation

Command	**Description**
mode border-element license capacity [session count]	[session count] – indicate the session count based on the license purchased for CUBE
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

Testing was conducted in Orange Business Services lab.

Please note that there is a difference between 12.4T and 15.5(3)M4 trains regarding two commands “header-passing” and “error-passthru”, which should be taken into account while making an update between the two IOS versions. With 12.4T they should be invoked together as “header-passing error-passthru” while in 15.5(3)M4 they should be invoked as 2 separate commands: “header-passing” and “error-passthru”

Media Passing through CUBE (media flow-through vs. media flow-around)

Default CUBE configuration enables CUBE to work in flow-through mode. In order to enable flow-around mode, please perform the following actions:

```
voice service voip
   media flow-around
```

Codecs

BT/BTIP requires currently 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.

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

```
voice class codec 1
   codec preference 1 g711alaw
```

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

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

For BT customers in areas requiring **G.711 uLaw** codec:

```
voice class codec 1
   codec preference 1 g711ulaw
```


Testing was conducted in Orange Business Services lab.

SIP user agent

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

```
sip-ua
   retry invite 1
   retry response 2
   retry bye 2
   retry cancel 2
   reason-header override
   connection-reuse
   g729-annexb override
   timers options 1000
```


Testing was conducted in Orange Business Services lab.

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 BT/BTIP infrastructure, when G.729 codec is used

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
```


Testing was conducted in Orange Business Services lab.

Design for a CUCM cluster and two CUBEs

In case of a design for with a CUCM cluster and two CUBEs, the configuration and integration should take care of the rerouting decision based on SIP Error and Release Cause received from infrastructure.

Trunking redundancy for **outgoing calls** is presented below:

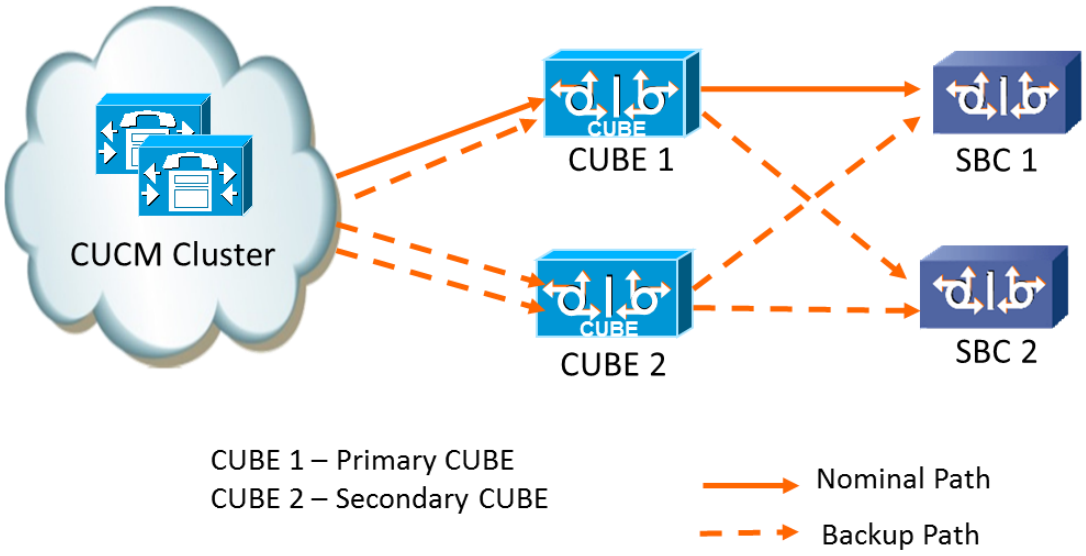


Figure 2 Redundancy for CUCM cluster with two CUBEs – Outgoing call

For **incoming calls**, trunking redundancy is shown below:

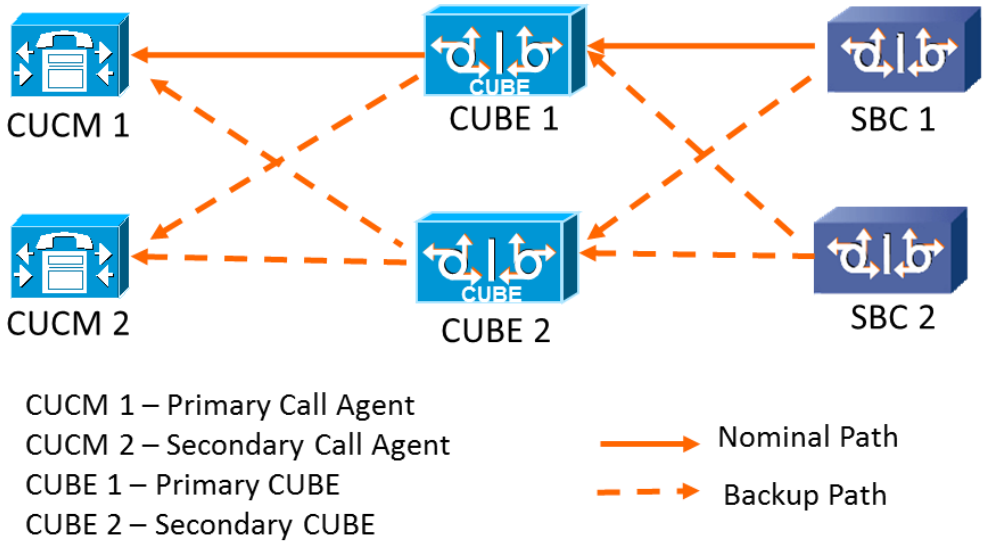


Figure 3 Redundancy for CUCM cluster with two CUBEs – Incoming call

Testing was conducted in Orange Business Services lab.

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.

Each 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.

Redundancy in Active/Standby mode with High Availability feature by using HSRP (Hot Standby Router Protocol) has been tested but it is not a standard feature. It can be done on customer request only. Main limitation of Active/Standby redundancy is the requirement of geographical colocation of redundant routers.

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

   ip address 10.227.101.203 255.255.255.0 secondary
```

CUCM cluster will be configured with 4 different SIP trunks :

- 1st SIP trunk pointing to the primary address of Primary CUBE
- 2nd SIP trunk pointing to the secondary address of Primary CUBE
- 3rd SIP trunk pointing to primary address of Secondary CUBE
- 4th SIP trunk pointing to secondary address of Secondary CUBE

CUCM will be configured with a Route List composed of (at least) 4 Route Groups. Each route group will include SIP trunk to one of CUBE IP Address (Primary or Secondary). On each route group parameters, a specific prefix should be defined (one prefix for each RG). This way the CUBE will be able to route the outgoing calls to the right SBC, depending on this prefix value:

For incoming and outgoing calls for CUCMs side

```
dial-peer voice 1 voip

 description ** to/from site devices - Primary CUCM **

 answer-address 433....

 destination-pattern 433....

 session protocol sipv2
```


Testing was conducted in Orange Business Services lab.

```
 session target ipv4:<PRIMARY_CUCM_IP_ADDR>
 voice-class codec 1
 voice-class sip options-keepalive up-interval 300 down-interval 300 retry 5
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 2 voip
 description ** to/from site devices - Backup CUCM **
 preference 1
 answer-address 433....
 destination-pattern 433....
 session protocol sipv2
 session target ipv4:<SECONDARY_CUCM_IP_ADDR>
 voice-class codec 1
 voice-class sip options-keepalive up-interval 300 down-interval 300 retry 5
 dtmf-relay rtp-nte
 no vad

!For outgoing calls (with a prefix to select the target SBC)
dial-peer voice 102 voip
 description ** Outgoing calls - Outbound dial peer - Primary SBC side **
 translation-profile outgoing 113
 huntstop
 destination-pattern 113T
  session protocol sipv2
 session target ipv4:<PRIMARY_SBC_IP_ADDR>
 voice-class codec 1
 voice-class sip options-keepalive up-interval 300 down-interval 300 retry 5
```


Testing was conducted in Orange Business Services lab.

```
 voice-class sip send 180 sdp
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 103 voip
 description ** Outgoing calls - Outbound dial peer - Backup SBC side **
 translation-profile outgoing 114
 huntstop
 destination-pattern 114T
 session protocol sipv2
 session target ipv4:<SECONDARY_SBC_IP_ADDR>
 voice-class codec 1
 voice-class sip options-keepalive up-interval 300 down-interval 300 retry 5
 voice-class sip send 180 sdp
 dtmf-relay rtp-nte
 no vad

!For incoming calls
dial-peer voice 100 voip
 description ** Incoming calls - Inbound dial peer - SBC side **
 answer-address +.T
 session protocol sipv2
 voice-class codec 1
 voice-class sip send 180 sdp
 dtmf-relay rtp-nte
 no vad
!
```

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

Testing was conducted in Orange Business Services lab.

Design for a single CUCM server and one CUBE

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.

TRUNKING Redundancy – Outgoing calls

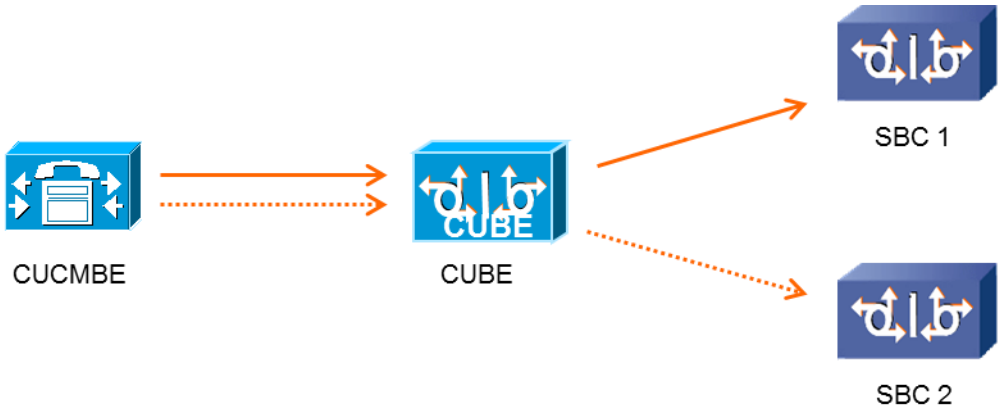


Figure 4 Redundancy for single CUCM with single CUBE – outgoing calls

TRUNKING Redundancy – Incoming calls

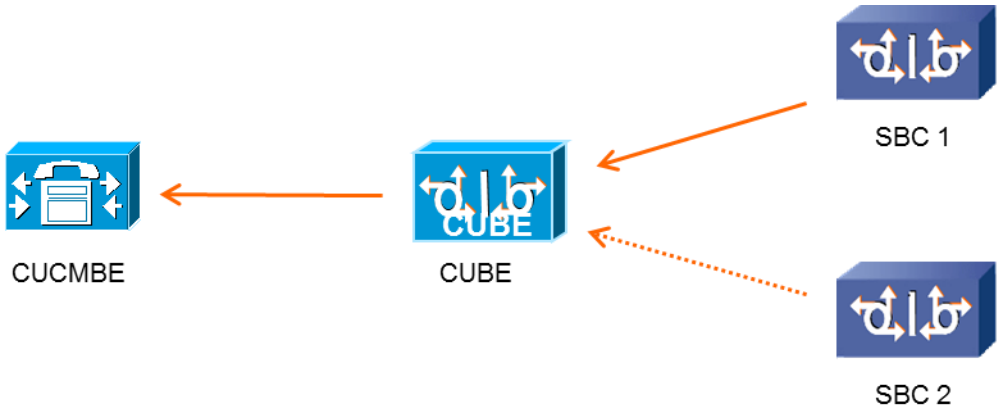


Figure 5 Redundancy for single CUCM with single CUBE – incoming calls

Testing was conducted in Orange Business Services lab.

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 (from 15.4(3)M2). 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

   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 1 voip

   description ** DP to/from BO devices - CUCMBE**

   answer-address 227....

   destination-pattern 227....

   session target ipv4:<CUCMBE_IP>

   […]


!For outgoing calls (with a prefix to select the target SBC)

dial-peer voice 11 voip

   description ** Outgoing calls - Outbound dial peer - SBC1 side **

   answer-address 227....
```


Testing was conducted in Orange Business Services lab.

```
   destination-pattern 11T
   session-target <SBC1_IP>
   […]

dial-peer voice 12 voip
   description ** Outgoing calls - Outbound dial peer - SBC2 side **
   answer-address 227....
   destination-pattern 12T
   session-target <SBC2_IP>
   […]

dial-peer voice 101 voip
 description ** Incoming calls - Inbound dial peer - SBC side **
 answer-address +.T
 voice-class codec 1
 voice-class sip send 180 sdp
 session protocol sipv2
 dtmf-relay rtp-nte
no vad
!
```

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

Design for a CUCM cluster and one CUBE

In case of a specific design for a CUCM cluster and only one CUBE (the customer does not require the redundancy of the CUBE and accept the risks involved with failure of the CUBE), the configuration and integration should take care of the rerouting decision based on SIP Error and Release Cause received from infrastructure like see previously.

Testing was conducted in Orange Business Services lab.

TRUNKING Redundancy – Outgoing calls

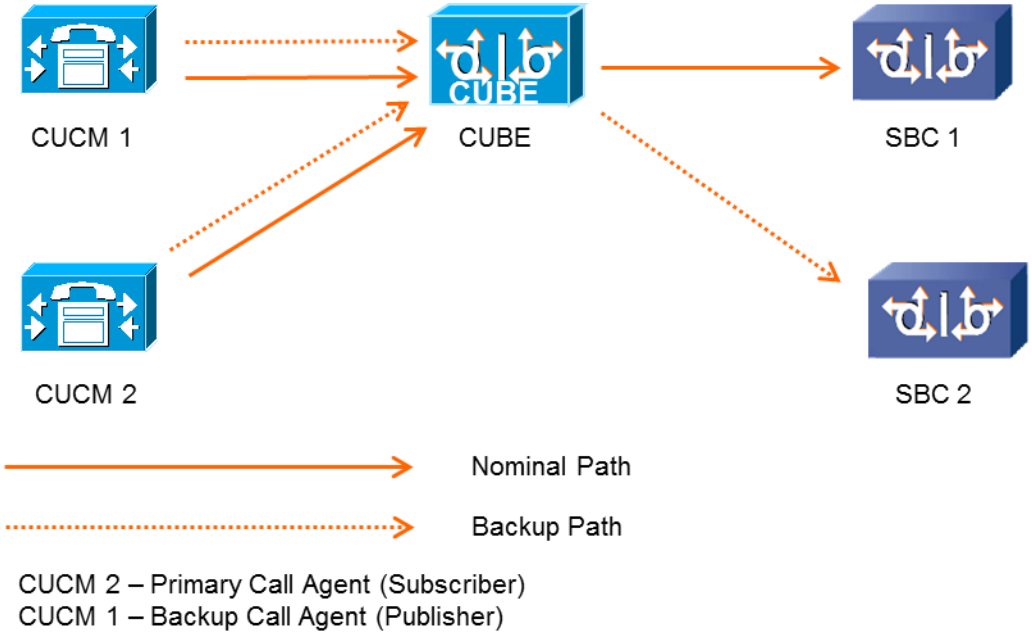


Figure 6 Redundancy for CUCM cluster and single CUBE – outgoing calls

TRUNKING Redundancy – Incoming calls

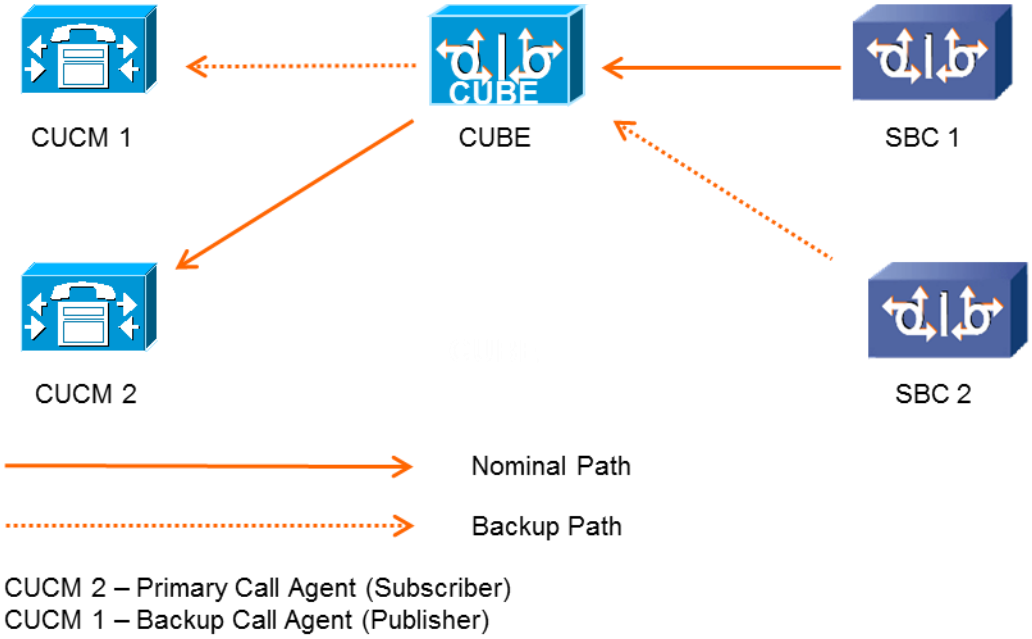


Figure 7 Redundancy for CUCM cluster and single CUBE – outgoing calls

Testing was conducted in Orange Business Services lab.

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.

Each 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 (from 15.4(3)M2). 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

   ip address 10.227.101.203 255.255.255.0 secondary
```

CUCM cluster 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:

For incoming and outgoing calls for CUCMs side

```
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

   session target ipv4:<CUCM2_IP>

   […]
```


Testing was conducted in Orange Business Services lab.

```
dial-peer voice 2 voip
   description ** DP to/from BO devices - CUCM PUB**
   preference 2
   answer-address 227....
   destination-pattern 227....
   voice-class codec 1
   session target ipv4:<CUCM1_IP>
   […]
```

For outgoing calls (with a prefix to select the target SBC)

```
dial-peer voice 11 voip
   preference 1
   answer-address 227....
   destination-pattern 11T
   session-target <SBC1_IP>
   […]
dial-peer voice 12 voip
   preference 2
   answer-address 227....
   destination-pattern 12T
   session-target <SBC2_IP>
   […]
```

For incoming calls

```
dial-peer voice 101 voip
 description ** Incoming calls - Inbound dial peer - SBC side **
 answer-address +.T
```


Testing was conducted in Orange Business Services lab.

```
 voice-class codec 1
 voice-class sip send 180 sdp
 session protocol sipv2
 dtmf-relay rtp-nte
 no vad
!
```

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

Testing was conducted in Orange Business Services lab.

Configuration example

The following configuration snippet contains a sample configuration of Cisco Unified Border Element with all parameters mentioned previously. This is configuration for case with CUCM cluster and 2 CUBES (configuration of one CUBE only is given below)

```
no service pad
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service sequence-numbers
!
hostname RID35_HQ433_CUBE1
!
boot-start-marker
boot system bootflash:/isr4400-universalk9.16.06.03.SPA.bin
boot-end-marker
!
!
no logging buffered
enable password cisco
!
no aaa new-model
!
clock timezone CET 1 0
clock summer-time CEST recurring last Sun Mar 2:00 last Sun Oct 3:00
network-clock-participate wic 1
!
no ip source-route
!
!
```


Testing was conducted in Orange Business Services lab.

```
no ip cef
no ipv6 cef
multilink bundle-name authenticated
!
voice-card 0
!
!
!
voice service voip
 ip address trusted list
  ipv4 172.22.246.33
  ipv4 172.22.246.73
 mode border-element
 allow-connections sip to sip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  header-passing
  error-passthru
  pass-thru headers unsupp
  no update-callerid
  early-offer forced
  midcall-signaling passthru
  sip-profiles 1
!
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g729r8
!
```


Testing was conducted in Orange Business Services lab.

```
voice class sip-profiles 1
 request INVITE sip-header User-Agent modify ".*" "User-Agent: CUBE 16.06.03 + CUCM
12.0.1.21900-7 "
 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 16.06.03 + CUCM 12.0.1.21900-7 "
!
interface GigabitEthernet0/0
 description Gi0/0
 no ip address
 duplex auto
 speed auto
!
interface GigabitEthernet0/0.433
 description CUBE Interface
 encapsulation dot1Q 433
 ip address 6.4.33.30 255.255.255.192
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
!
snmp-server community public RO
snmp-server manager
!
control-plane
!
```


Testing was conducted in Orange Business Services lab.

```
mgcp profile default
!
dial-peer voice 1 voip
 description ** to/from site devices - Primary CUCM **
 answer-address 433....
 destination-pattern 433....
 session protocol sipv2
 session target ipv4:6.4.33.2
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip options-keepalive up-interval 300 down-interval 300 retry 5
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 2 voip
 description ** to/from site devices - Backup CUCM **
 preference 1
 answer-address 433....
 destination-pattern 433....
 session protocol sipv2
 session target ipv4:6.4.33.1
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip options-keepalive up-interval 300 down-interval 300 retry 5
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 100 voip
```


Testing was conducted in Orange Business Services lab.

```
 description ** Incoming calls - Inbound dial peer - SBC side **
 answer-address +.T
 session protocol sipv2
 voice-class codec 1
 voice-class sip profiles 1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 102 voip
 description ** Outgoing calls - Outbound dial peer - Backup SBC side **
 preference 1
 destination-pattern 0.T
 progress_ind alert strip
 session protocol sipv2
 session target ipv4:172.22.246.33
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip options-keepalive up-interval 300 down-interval 300 retry 5
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 103 voip
 description ** Outgoing calls - Outbound dial peer - Backup SBC side **
 preference 2
 destination-pattern 0.T
 progress_ind alert strip
 session protocol sipv2
 session target ipv4:172.22.246.73
```


Testing was conducted in Orange Business Services lab.

```
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip options-keepalive up-interval 300 down-interval 300 retry 5
 dtmf-relay rtp-nte
 no vad
!
!
sip-ua
 retry invite 1
 retry response 2
 retry bye 2
 retry cancel 2
 timers options 1000
 reason-header override
 connection-reuse
 g729-annexb override
!
gatekeeper
 shutdown
!
line con 0
 exec-timeout 0 0
 privilege level 15
 password visit1
 accounting commands 15 OBS_LAB
 logging synchronous
line vty 0 4
 exec-timeout 15 0
```


Testing was conducted in Orange Business Services lab.

```
 privilege level 15
 accounting commands 15 OBS_LAB
 accounting exec OBS_LAB
 logging synchronous
 login authentication OBS_LAB
 transport input telnet ssh
!
scheduler allocate 20000 1000
ntp server 10.25.0.120 prefer
ntp server 10.10.0.5
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 1.3.6.1.2.1.47.1.1.1.1.10.1000 get-type exact
 action 1.1 set _info_snmp_iosversion "$_info_snmp_value"
 action 1.2 info type snmp oid 1.3.6.1.4.1.9.9.156.1.1.2.1.4.1 get-type exact community public ipaddr 6.4.33.1
 action 2.0 cli command "enable"
 action 2.1 cli command "config t"
 action 3.0 cli command "no voice class sip-profiles 1"
 action 4.0 cli command "voice class sip-profiles 1"
 action 4.1 cli command "$_sip_header_1 $_info_snmp_iosversion + CUCM $_info_snmp_value $_sip_header_2"
```


Testing was conducted in Orange Business Services lab.

```
 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"
```

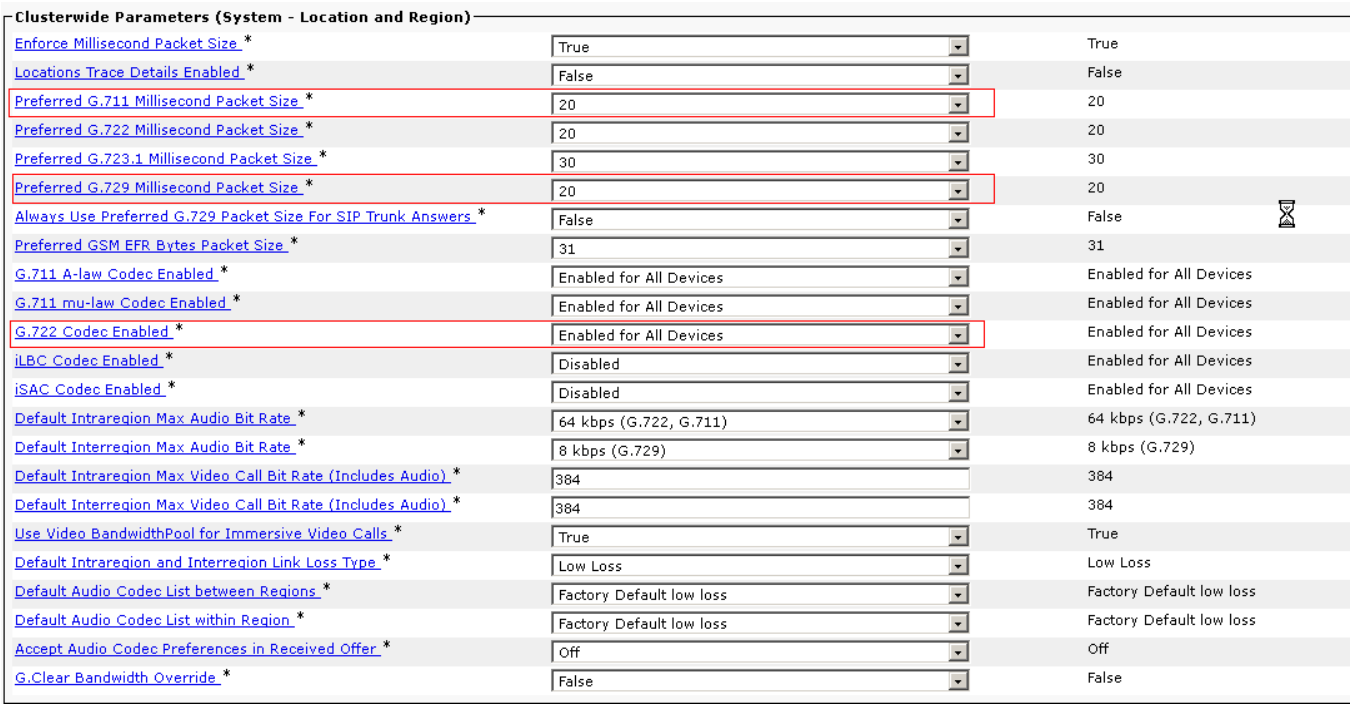

Testing was conducted in Orange Business Services lab.

Configuring the Cisco Unified Communications Manager

Cisco CallManager service

Go to **System > Service Parameters > Appropriate server > Cisco CallManager (Active)** and click **Advanced** button to see all parameters.

Codec payload configuration



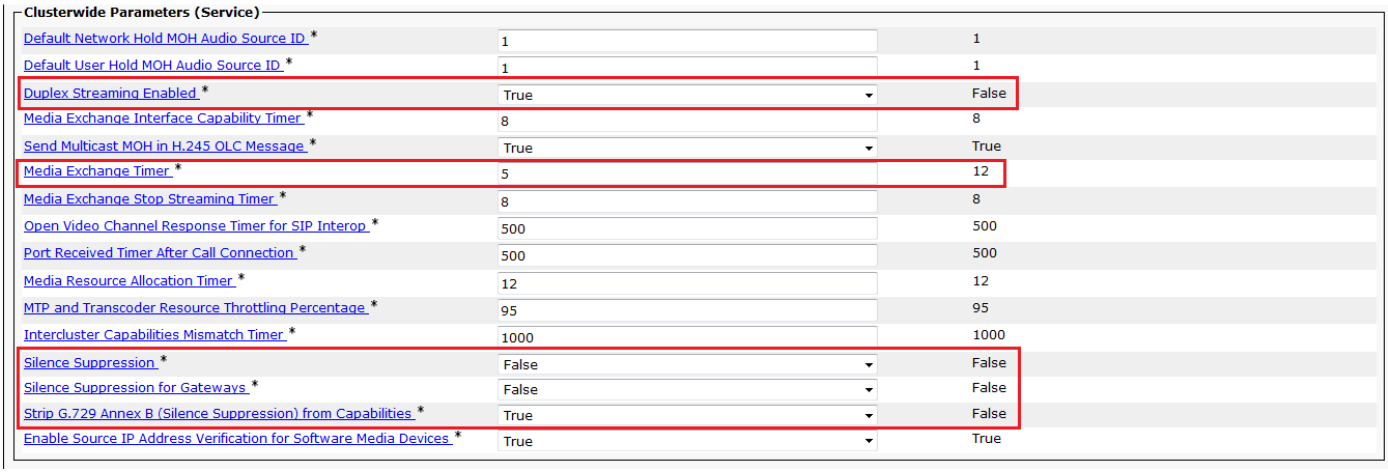
Clusterwide Parameters (System - Location and Region)

Enforce Millisecond Packet Size *	True	True
Locations Trace Details Enabled *	False	False
Preferred G.711 Millisecond Packet Size *	20	20
Preferred G.722 Millisecond Packet Size *	20	20
Preferred G.723.1 Millisecond Packet Size *	30	30
Preferred G.729 Millisecond Packet Size *	20	20
Always Use Preferred G.729 Packet Size For SIP Trunk Answers *	False	False
Preferred GSM EFR Bytes Packet Size *	31	31
G.711 A-law Codec Enabled *	Enabled for All Devices	Enabled for All Devices
G.711 mu-law Codec Enabled *	Enabled for All Devices	Enabled for All Devices
G.722 Codec Enabled *	Enabled for All Devices	Enabled for All Devices
iLBC Codec Enabled *	Disabled	Enabled for All Devices
iSAC Codec Enabled *	Disabled	Enabled for All Devices
Default Intraregion Max Audio Bit Rate *	64 kbps (G.722, G.711)	64 kbps (G.722, G.711)
Default Interregion Max Audio Bit Rate *	8 kbps (G.729)	8 kbps (G.729)
Default Intraregion Max Video Call Bit Rate (Includes Audio) *	384	384
Default Interregion Max Video Call Bit Rate (Includes Audio) *	384	384
Use Video BandwidthPool for Immersive Video Calls *	True	True
Default Intraregion and Interregion Link Loss Type *	Low Loss	Low Loss
Default Audio Codec List between Regions *	Factory Default low loss	Factory Default low loss
Default Audio Codec List within Region *	Factory Default low loss	Factory Default low loss
Accept Audio Codec Preferences in Received Offer *	Off	Off
G.Clear Bandwidth Override *	False	False

Parameter	Value	Description
Preferred G.711 Millisecond Packet Size	20 ms (default value)	Default value
Preferred G.729 Millisecond Packet Size	20 ms (default value)	Default value
G.722 Codec Enabled	Enabled	Default value

Testing was conducted in Orange Business Services lab.

Media configuration

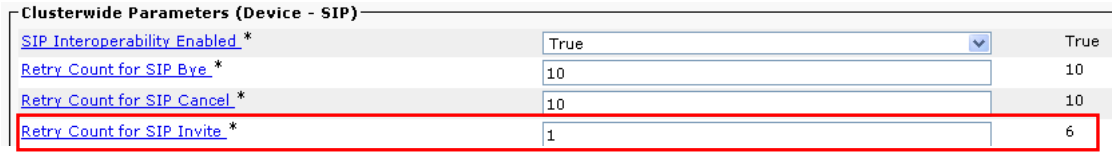


Clusterwide Parameters (Service)

Default Network Hold MOH Audio Source ID *	1	1
Default User Hold MOH Audio Source ID *	1	1
Duplex Streaming Enabled *	True	False
Media Exchange Interface Capability Timer *	8	8
Send Multicast MOH in H.245 OLC Message *	True	True
Media Exchange Timer *	5	12
Media Exchange Stop Streaming Timer *	8	8
Open Video Channel Response Timer for SIP Interop *	500	500
Port Received Timer After Call Connection *	500	500
Media Resource Allocation Timer *	12	12
MTP and Transcoder Resource Throttling Percentage *	95	95
Intercluster Capabilities Mismatch Timer *	1000	1000
Silence Suppression *	False	False
Silence Suppression for Gateways *	False	False
Strip G.729 Annex B (Silence Suppression) from Capabilities *	True	False
Enable Source IP Address Verification for Software Media Devices *	True	True

Parameter	Value	Description
Duplex Streaming Enabled	True	Determines whether music on hold and annunciator use duplex (two-way) 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 with Business Talk IP

SIP parameters



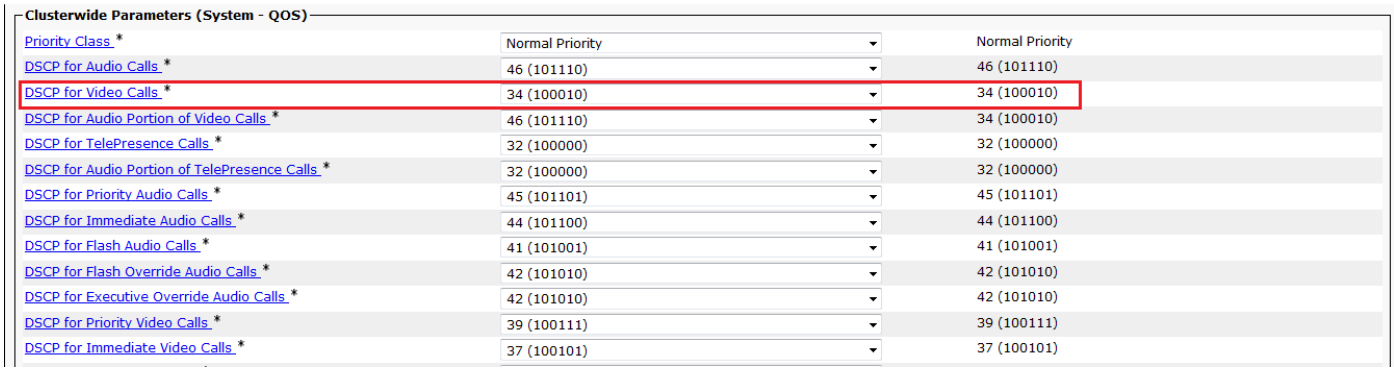
Clusterwide Parameters (Device - SIP)

SIP Interoperability Enabled *	True	True
Retry Count for SIP Bye *	10	10
Retry Count for SIP Cancel *	10	10
Retry Count for SIP Invite *	1	6

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.

Testing was conducted in Orange Business Services lab.

System – QoS parameters



Clusterwide Parameters (System - QOS)

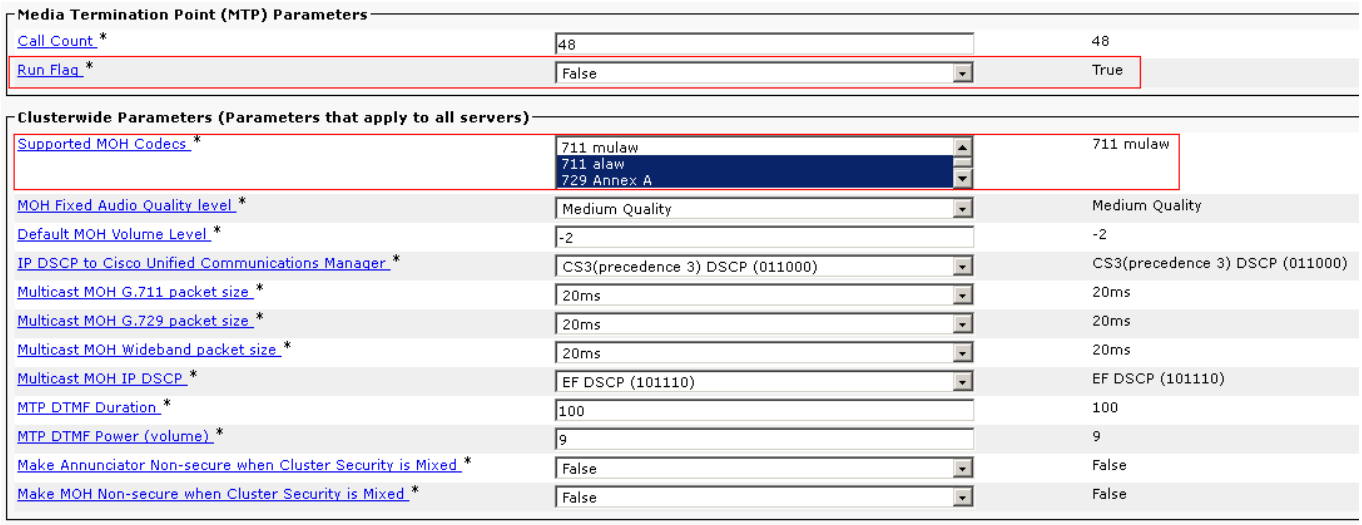
Priority Class *	Normal Priority	Normal Priority
DSCP for Audio Calls *	46 (101110)	46 (101110)
DSCP for Video Calls *	34 (100010)	34 (100010)
DSCP for Audio Portion of Video Calls *	46 (101110)	34 (100010)
DSCP for TelePresence Calls *	32 (100000)	32 (100000)
DSCP for Audio Portion of TelePresence Calls *	32 (100000)	32 (100000)
DSCP for Priority Audio Calls *	45 (101101)	45 (101101)
DSCP for Immediate Audio Calls *	44 (101100)	44 (101100)
DSCP for Flash Audio Calls *	41 (101001)	41 (101001)
DSCP for Flash Override Audio Calls *	42 (101010)	42 (101010)
DSCP for Executive Override Audio Calls *	42 (101010)	42 (101010)
DSCP for Priority Video Calls *	39 (100111)	39 (100111)
DSCP for Immediate Video Calls *	37 (100101)	37 (100101)

Parameter	Value	Description
DSCP for Video Calls	34 (100010) (default value)	This parameter specifies the Differentiated Service Code Point (DSCP) value for video calls.

Testing was conducted in Orange Business Services lab.

Cisco IP Voice Media Streaming Application service

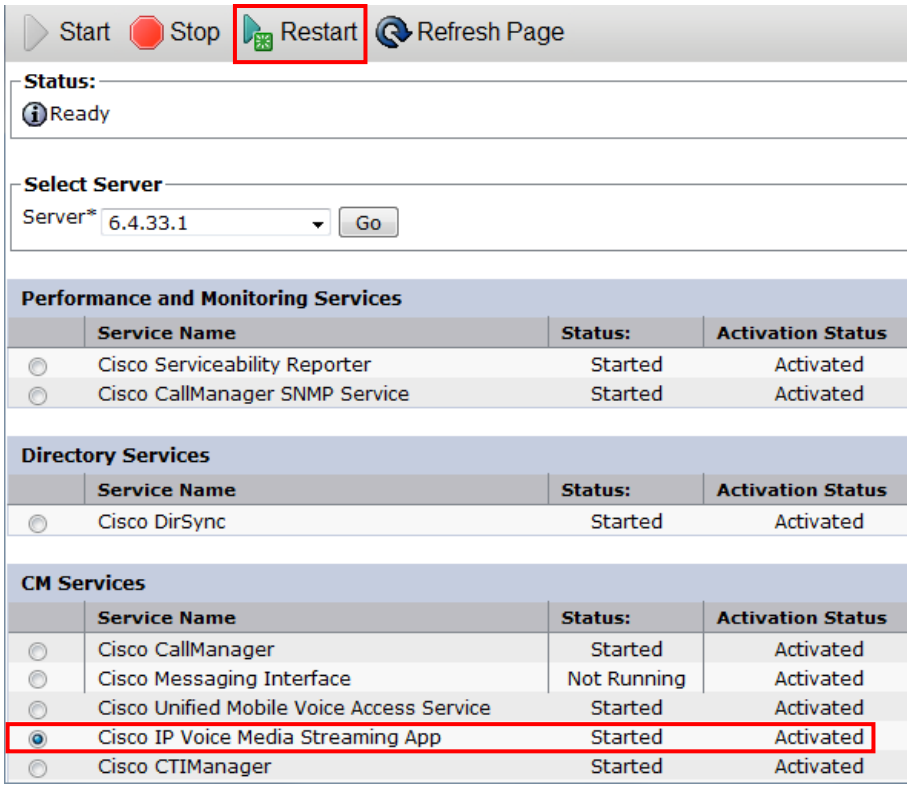
Go to **System** > **Service Parameters** > **Appropriate CUCM server** > **Cisco IP Voice Media Streaming App (Active)**. Configure the following parameters on all IP Voice Media Streaming Applications services running in the cluster (on all servers)



Parameter	Value	Description
MTP Run Flag	False	Software MTP is not used with Business Talk IP
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, as it's not supported by Business Talk IP.

Please reset the Cisco IP Voice Media Streaming App from **Cisco Unified Serviceability** -> **Tools** -> **Feature Services**, where you should select CUCM Server and find within CM Services this service. By clicking the Restart button, this service will be restarted on this service.

Testing was conducted in Orange Business Services lab.



Testing was conducted in Orange Business Services lab.

Region configuration

Regions are configured at **System > Region Information > Region**. They need to be associated with proper device pools later. Business Talk IP service currently supports only monocodec configuration, i.e. for any given customer only single codec must be used for all calls over the WAN. Mulitcodec support is considered as a non-standard solution and can be applied only on customer request raised to Orange Business Services. Only one of the 2 following codecs is supported:

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

Consider the following customer design:

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

There should be at least one region per site and additional one dedicated for SIP trunk to BTIP (called WAN).

Region	**Purpose**
HQ	Assigned to devices in the HQ site
RS	Assigned to devices in the Remote Site 1
WAN	Assigned to SIP trunk to Business Talk IP

Region configuration example for customer using G.729 in WAN

From **To**	**HQ**	**RS**	**WAN**
HQ	64 kbps (G.722, G.711)	8 kbps (G.729)	8 kbps (G.729)
RS	8 kbps (G.729)	64 kbps (G.722, G.711)	8 kbps (G.729)
WAN	8 kbps (G.729)	8 kbps (G.729)	8 kbps (G.729)

Region configuration example for customer using G.711 in WAN

From **To**	**HQ**	**RS**	**WAN**
HQ	64 kbps (G.722, G.711)	64 kbps (G.722, G.711)	64 kbps (G.722, G.711)
RS	64 kbps (G.722, G.711)	64 kbps (G.722, G.711)	64 kbps (G.722, G.711)
WAN	64 kbps (G.722, G.711)	64 kbps (G.722, G.711)	64 kbps (G.722, G.711)

Testing was conducted in Orange Business Services lab.

Location configuration

To configure locations go to **System > Location Info > Location** than click on **Add New**. Only static locations feature is supported, **RSVP is not supported** with Business Talk IP. For customers using IP VPN to connect all their locations, Static Locations CAC feature in CUCM 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.

Device Pool configuration

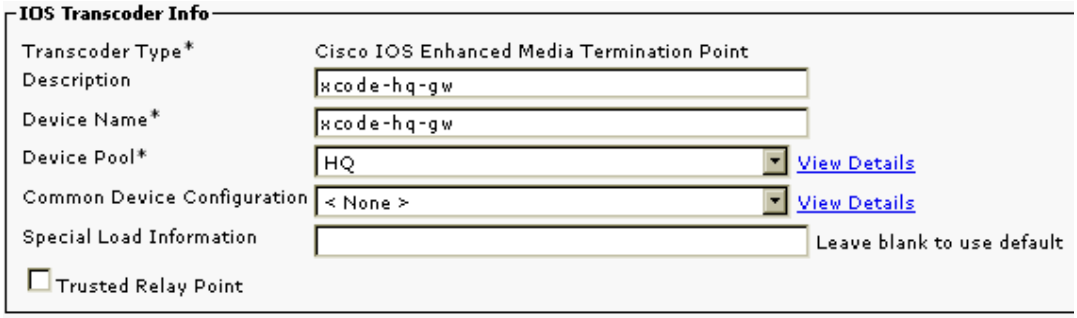
Go to **System > Device Pool** and click on **Add New.** When configuring Device Pools take into account following:

- The number of Device Pools should be at least the same as the number of site or greater, according to customer specific requirements
- Every Device Pool should have appropriate Region (used for codec selection) and Location (used for CAC / bandwidth tracking) value configured
- Every device in the cluster (IP phones, SIP trunks, media resources, etc.) should have correct Device Pool selected
- A dedicated Device Pool for SIP trunk should be created with proper region (with G711/G722 or G729 selected to/from all other regions)

Testing was conducted in Orange Business Services lab.

Transcoder

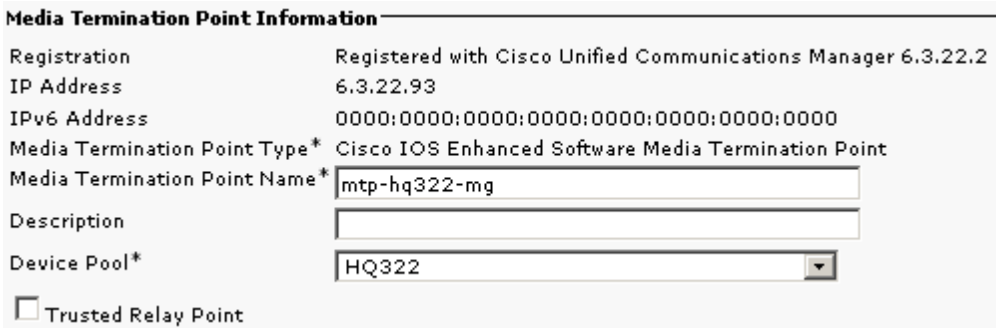
To configure hardware IOS-based transcoder and software IOS-based MTP go to **Media Resources > Transcoder** and press **Add new** button. An example of transcoder configuration follows:



Setting	Value	Description
Transcoder Type	Cisco IOS Enhanced Media Termination Point	This is the appropriate transcoder type for PVDM3/4 modules in ISR 2900/3900/4000 series routers
Device Name	Use the name configured in sccp ccm group in the IOS	-
Device Pool	Use the appropriate Device Pool	The Device Pool defines the Region (codec selection) and Location (CAC) used for media resource.

Media Termination Point

To configure software IOS-based MTP go to **Media Resources > Media Termination Point** and press **Add New** button. An example of MTP configuration follows



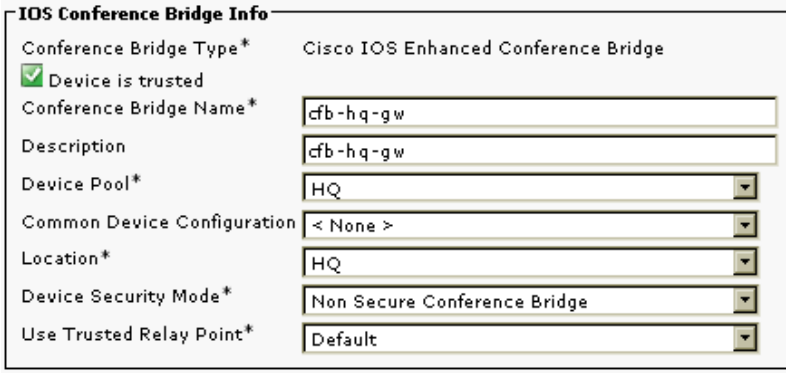
Setting	Value	Description

Testing was conducted in Orange Business Services lab.

Media Termination Point Type	Cisco IOS Enhanced Media Termination Point	This is the appropriate type MTP running on IOS in software.
Device Name	Use the name configured in sccp ccm group in the IOS	-
Device Pool	Use the appropriate Device Pool	The Device Pool defines the Region (codec selection) and Location (CAC) used for media resource.

Conference Bridge configuration

Go to **Media Resources > Conference Bridge** and click on **Add New**.



Setting	Value	Description
Conference Bridge Type	Cisco IOS Enhanced Media Conference Bridge	This is the appropriate transcoder type for PVDM3/4 modules in ISR 2900/3900/4000 series routers
Device Name	Use the name configured in sccp ccm group in the IOS	-
Device Pool	Use the appropriate Device Pool	The Device Pool defines the Region (codec selection) and Location (CAC) used for media resource.

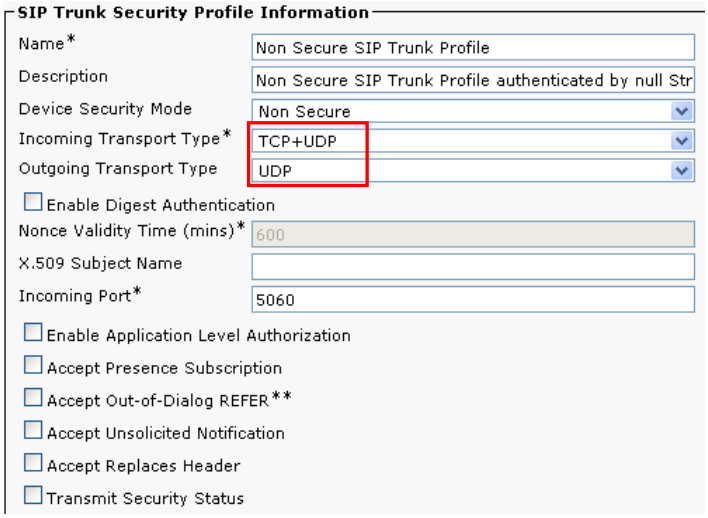
Testing was conducted in Orange Business Services lab.

Offnet calls via Business Talk IP

Off-net calls are served by SIP trunks configured between CUCM and Business Talk IP primary and secondary SBCs. Calls are routed via CUBE.

SIP Trunk Security Profile

Go to **System > Security > SIP Trunk Security Profile** and click on **Add New**.



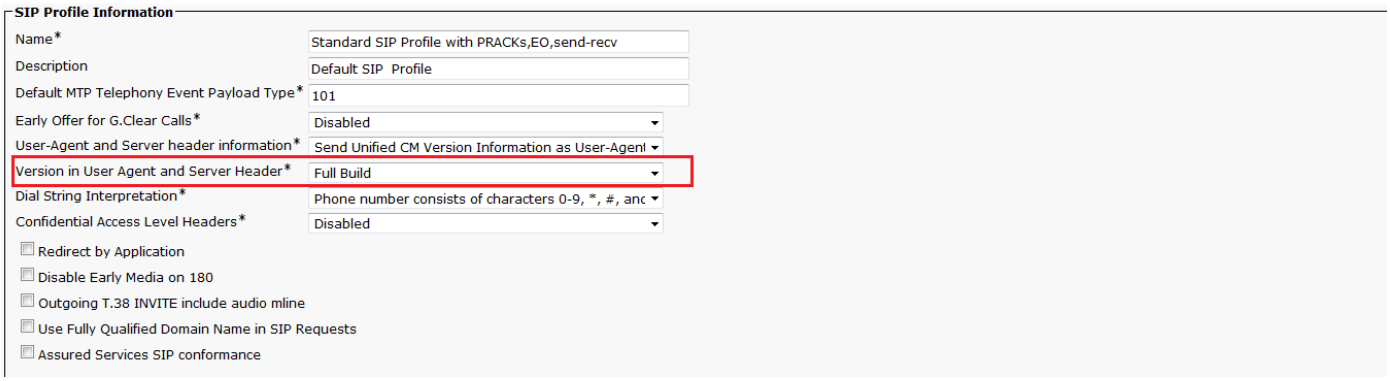
Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to Orange Core SBC should use UDP as a transport protocol for SIP. This is configured using SIP Trunk Security profile, which is later assigned to the SIP trunk itself.

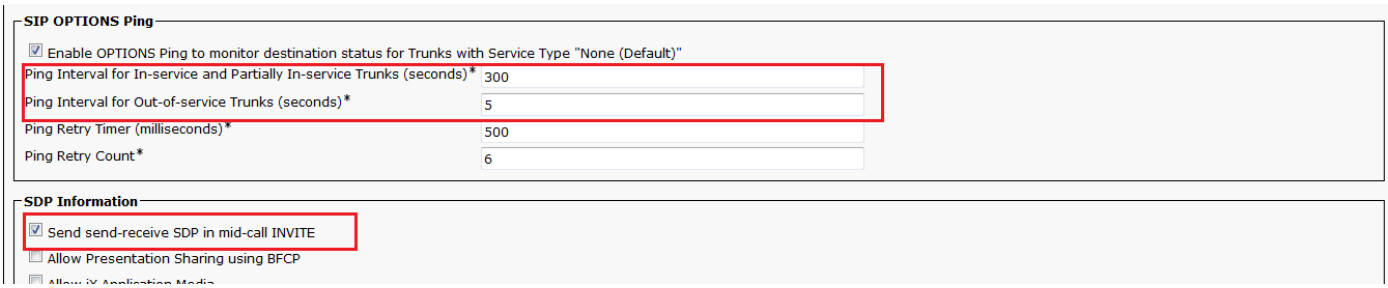
Testing was conducted in Orange Business Services lab.

SIP Profile

SIP Profile will be later associated with the SIP trunk.

Go to **Device** > **Device Settings** > **SIP Profile** and modify default SIP Profile by clicking on a **Copy** button in its row.





Parameter	Value	Description
SIP Rel1XX Options	Send PRACK for 1xx Messages	Enable Provisional Acknowlesgements (Reliable 100 messages)
Early Offer support for voice and video calls (insert MTP if needed)	Checked	Early Offer required for Business Talk

Testing was conducted in Orange Business Services lab.

Send send-receive SDP in mid-call INVITE	Checked	CUCM will not send INVITE a=inactive SDP during call hold or media break during supplementary services. This basically avoids putting the remote SIP device media state to inactive mode
Ping Interval for In-service and Partially In-service Trunks (seconds)	300	OPTIONS message parameters- interval time
Ping Interval for Out-of-service Trunks (seconds)	5	OPTIONS message parameters- interval time
Version in User Agent and Sever Header	Full build	Inject info about full version of CUCM
Session Refresh Method	INVITE or UPDATE	"INVITE" should be used by default.

SIP Normalization Script

SIP Normalization Script is applied to SIP trunk and is required to adapt the SIP signaling to the form expected by Business Talk IP infrastructure. Create the script: **Device > Device Settings > SIP normalization script** and press **Add new** button (you may name it "BT adaptation").

```
-- Orange SIP Normalization Script v11
-- this is normalization script for uc 12.0
M = {}



-- This is called when an INVITE message is sent
function M.outbound_INVITE(msg)
    local sdp = msg:getSdp()
    if sdp
    then
       -- remove b=TIAS:
       sdp = sdp:gsub("b=TIAS:%d*\r\n", "")
       -- store the updated sdp in the message object
```


Testing was conducted in Orange Business Services lab.

```
        msg:setSdp(sdp)
    end
end


--modifying of Server header in 183 messages
function M.outbound_183_INVITE(msg)
-- change 183 to 180 if sdp
 local sdp = msg:getSdp()
 if sdp
 then
  msg:setResponseCode(180, "Ringing")
 end
end

--modifying of Server header in 488 messages
function M.outbound_488_INVITE(msg)
 -- change 488 to 503 if sdp
  msg:setResponseCode(503, "Service Unavailable")
end

--handling of 400 errors
function M.inbound_400_INVITE(msg)
 local reason = msg:getHeader("Reason")
 if reason
 then
  msg:modifyHeader("Reason", "Q.850; cause=27")
 else
  msg:addHeader("Reason", "Q.850; cause=27")
 end
```


Testing was conducted in Orange Business Services lab.

```
end

--handling of 403 errors
function M.inbound_403_INVITE(msg)
 local reason = msg:getHeader("Reason")
 if reason
 then
  msg:modifyHeader("Reason", "Q.850; cause=2")
 end
end

--handling of 408 errors
function M.inbound_408_INVITE(msg)
 local reason = msg:getHeader("Reason")
 if reason
 then
  msg:removeHeader("Reason")
 end
end

-- handling of 480 errors
function M.inbound_480_INVITE(msg)
 local reason = msg:getHeader("Reason")
 if not reason
 then
  msg:addHeader("Reason", "Q.850; cause=20")
 end
end

--handling of 481 errors
```


Testing was conducted in Orange Business Services lab.

```
function M.inbound_481_INVITE(msg)
 local reason = msg:getHeader("Reason")
 if reason
 then
  msg:modifyHeader("Reason", "Q.850; cause=27")
 else
  msg:addHeader("Reason", "Q.850; cause=27")
 end
end

--handling of 487 errors
function M.inbound_487_INVITE(msg)
 local reason = msg:getHeader("Reason")
 if not reason
 then
  msg:addHeader("Reason", "Q.850; cause=16")
 end
end

--handling of 488 errors
function M.inbound_488_INVITE(msg)
 local reason = msg:getHeader("Reason")
 if not reason
 then
  msg:addHeader("Reason", "Q.850; cause=127")
 end
end

--handling of 500 errors
function M.inbound_500_INVITE(msg)
```

```
 local reason = msg:getHeader("Reason")
 if reason
 then
  msg:modifyHeader("Reason", "Q.850; cause=2")
 else
  msg:addHeader("Reason", "Q.850; cause=2")
 end
end

--handling of 501 errors
function M.inbound_501_INVITE(msg)
 local reason = msg:getHeader("Reason")
 if reason
 then
  msg:modifyHeader("Reason", "Q.850; cause=2")
 else
  msg:addHeader("Reason", "Q.850; cause=2")
 end
end

--handling of 502 errors
function M.inbound_502_INVITE(msg)
 local reason = msg:getHeader("Reason")
 if reason
 then
  msg:removeHeader("Reason")
 end
end

-- handling of 503 errors
```

```
function M.inbound_503_INVITE(msg)
 local reason = msg:getHeader("Reason")
 if reason
 then
  msg:modifyHeader("Reason", "Q.850; cause=38")
 else
  msg:addHeader("Reason", "Q.850; cause=38")
 end
end

-- handling of 505 errors
function M.inbound_505_INVITE(msg)
 local reason = msg:getHeader("Reason")
 if reason
 then
  msg:modifyHeader("Reason", "Q.850; cause=38")
 else
  msg:addHeader("Reason", "Q.850; cause=38")
 end
end

-- handling of 513 errors
function M.inbound_513_INVITE(msg)
 local reason = msg:getHeader("Reason")
 if reason
 then
  msg:modifyHeader("Reason", "Q.850; cause=38")
 else
  msg:addHeader("Reason", "Q.850; cause=38")
 end
```

```
end

-- addition of PAI header if incoming INVITE includes Privacy header
function M.inbound_INVITE(msg)
 -- get Privacy header
 local privacy = msg:getHeader("Privacy")
 if privacy
 then
  -- get From and Pai
  from = msg:getHeader("From")
  pai = msg:getHeader("P-Asserted-Identity")
  --check if Pai header is not present
  if pai==nil
  then
   -- add Pai header filled with From URI value
   local uri = string.match(from, "(<.+>)")
   msg:addHeader("P-Asserted-Identity", uri)
  end
 end
end

return M
```

SIP Trunk Configuration

Create 2 SIP trunks to BT/BTIP by going to **Device** > **Trunk** and clicking **Add New** button.

For a connection via CUBE, please configure separate SIP Trunk to each of the IP Addresses configured on CUBE (both Primary or Secondary) at **Device** > **Trunk**. For a design with two CUBEs, overall 4 SIP Trunks should be configured. Please refer to the previous section, describing CUBE configuration to get more detailed information on the design with CUBE.

Testing was conducted in Orange Business Services lab.

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	ACME1
Description	Primary BT/BTIP SBC
Device Pool*	WAN
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_MTP_XCODE
Location*	WAN
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

Inbound Calls

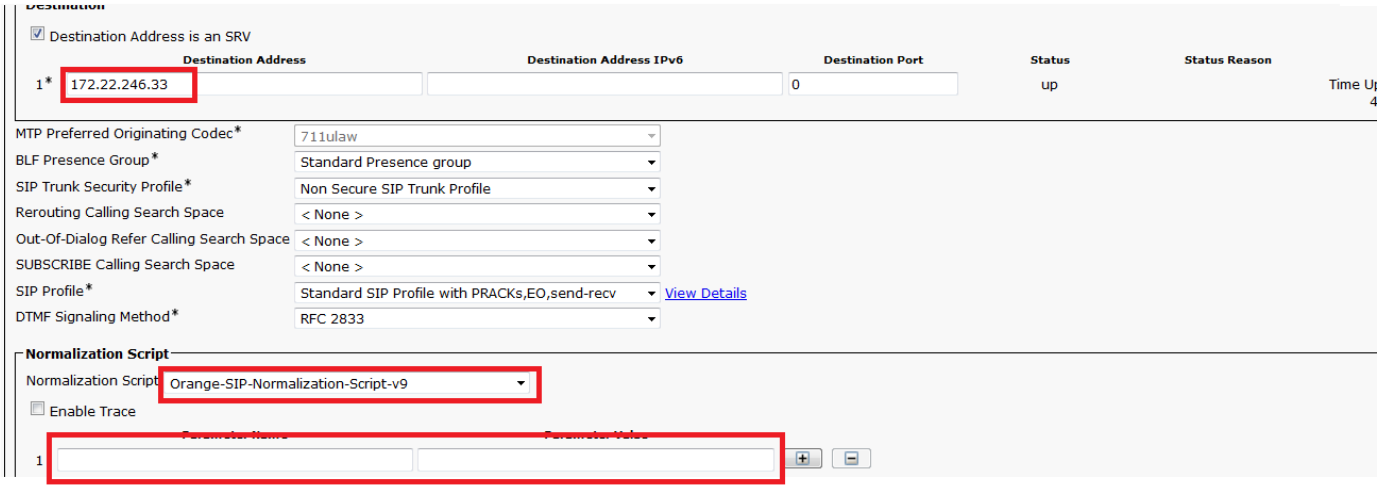
Significant Digits*	All
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	

☑ Redirecting Diversion Header Delivery - Inbound

Outbound Calls

Called Party Transformation CSS	< None >
☑ Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
☑ Use Device Pool Calling Party Transformation CSS	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling and Connected Party Info Format*	Deliver DN only in connected party
☑ Redirecting Diversion Header Delivery - Outbound	
Redirecting Party Transformation CSS	DIV-HEADER-CSS
☑ Use Device Pool Redirecting Party Transformation CSS	

Testing was conducted in Orange Business Services lab.



Parameter	Value	Description
Device Pool	Choose Device Pool which include Region and Location value	Region used for this DP should use G729 OR G711 to all other regions.
Media Resource Group List	MRGL	MRG with resources: conference, transcoder, annuciator (Subscribers), MOH Server (Subscribers)
Redirecting Diversion Header Delivery - Inbound	Checked	Adding Diversion Header for calls initiated from other site or offnet
Redirecting Diversion Header Delivery - outbound	Checked	Adding Diversion Header for calls outband from site
Redirecting Party Transformation CSS	DIV-HEADER-CSS	Adding Diversion Header for calls outband from site
Destination Address	IP address of the CUBE	The IP address is selected by customer based on IP VPN addressing scheme used and communicated to Orange Business Services.
SIP Trunk Security Profile	SIP Trunk Secure Profile name	SIP Trunk Security Profile configured earlier
SIP Profile	Standard SIP Profile name	SIP Profile configured earlier

Testing was conducted in Orange Business Services lab.

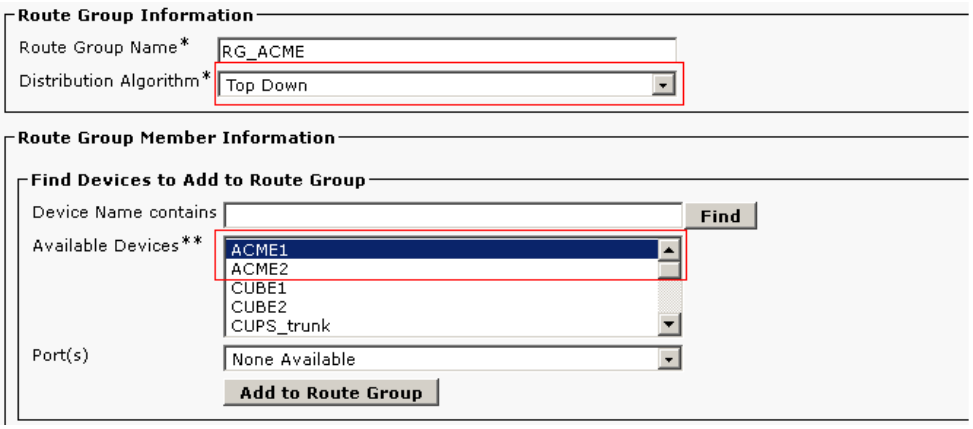
DTMF Signaling Method	RFC 2833	Only RFC 2833 is supported for DTMF transport to/from Business Talk IP
Normalization Script	SIP Normalization Script name	SIP Normalization Script configured earlier. There is no need to send version information in SIP normalization script (manipulation of User-Agent/Server SIP header) because CUCM 12.0 sends this info natively (option "Version in User Agent and Server Header" – value "full build" in SIP Profile)
Enable Trace	Unchecked	

Reset the trunk after the configuration is completed.

Route group

For a connection via CUBE - please create separate Route Group to each of CUBE IP Addresses (primary and secondary). For a design with two CUBEs, four Route Groups have to be configured.

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



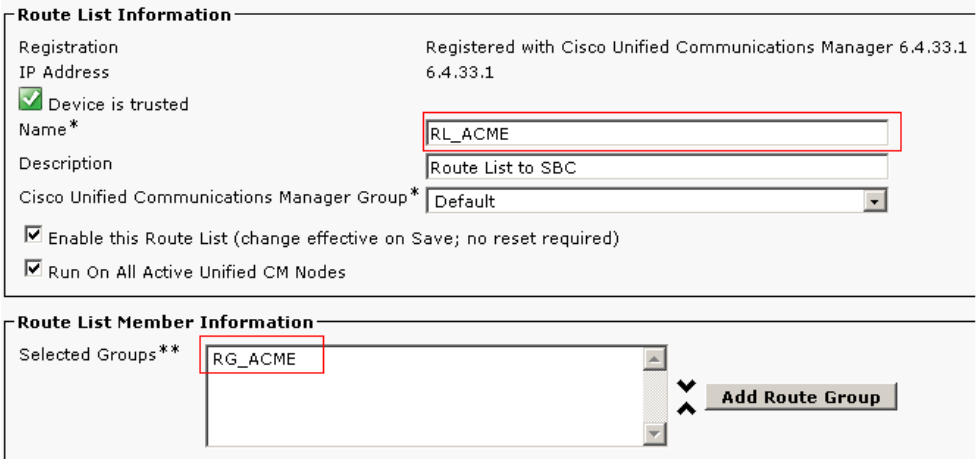
Setting	Value	Description
Distribution algorithm	Top Down	Primary Orange Core SBC should always be tried first
Selected devices	SIP Trunk to CUBE IP Address	use the trunks configured earlier to primary or secondary IP Address of Primary or Backup CUBEs. Only one SIP Trunk per one Route Group.

Testing was conducted in Orange Business Services lab.

Route list

Create Route List which will be later associated with the Route Pattern.

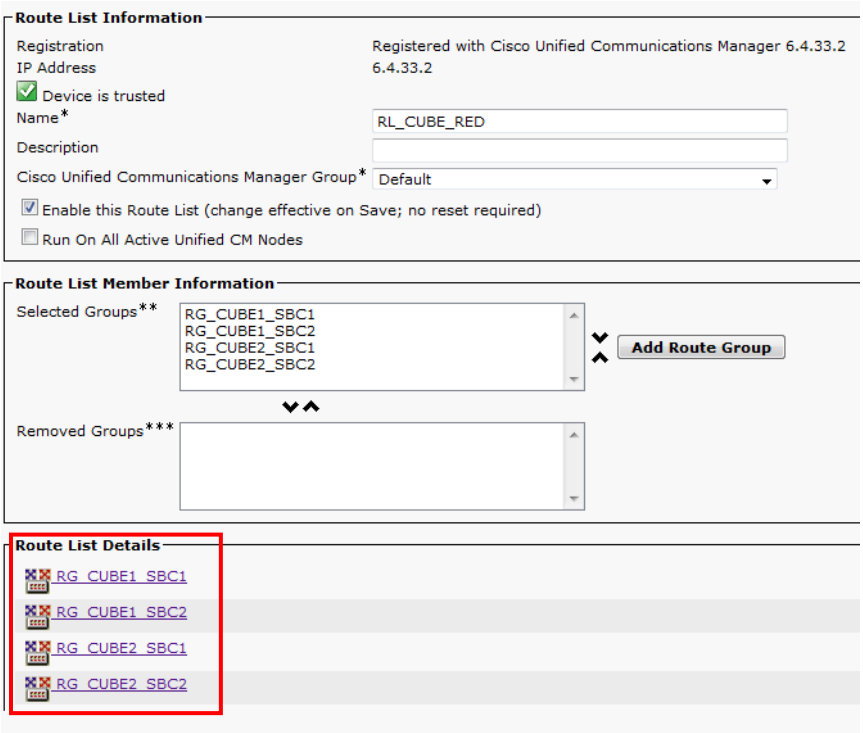
Go to **Call Routing** > **Route/Hunt** > **Route list** and click on Add New. Add the previously configured Route Groups.



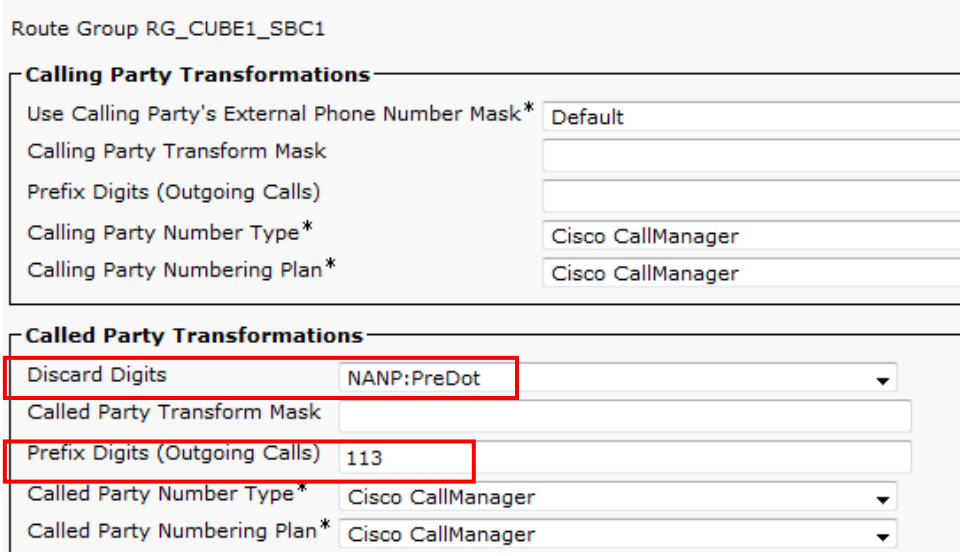
Setting	Value	Description
Selected Groups	Route Group with SIP trunks to CUBE	Select all the Route Groups configured in previous steps. Please apply the required order.

For a design with CUBE, we have to add the prefixes specific for each of the Route Groups. This could be done after the Route List is configured with Route Groups. At the bottom of the Route List configuration window, you will find the Link to Route Group configuration:

Testing was conducted in Orange Business Services lab.



Click each of the Route Groups and configure the following options:Discard Digits (set to NANP:PreDot) and Prefix Digits (Outgoing Calls) (set according to customer preference, as prefix digits will be removed on CUBE). This actions may also be done during Route Group addition:



Dialplan

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.

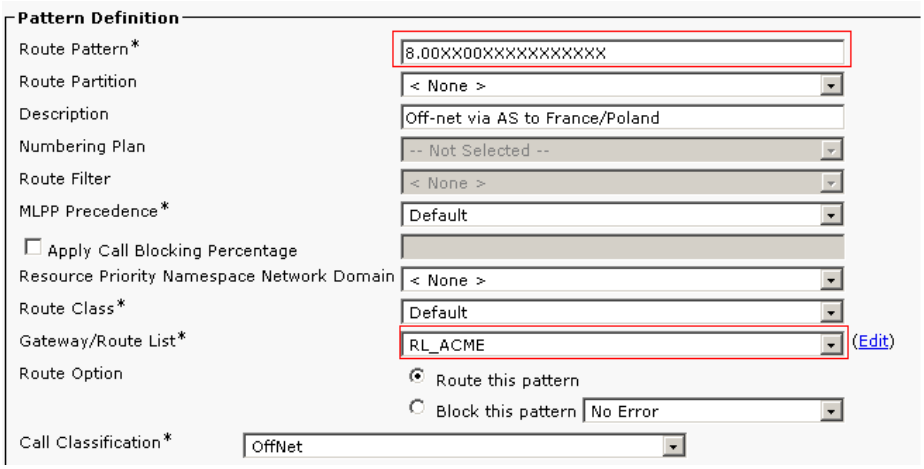
Testing was conducted in Orange Business Services lab.

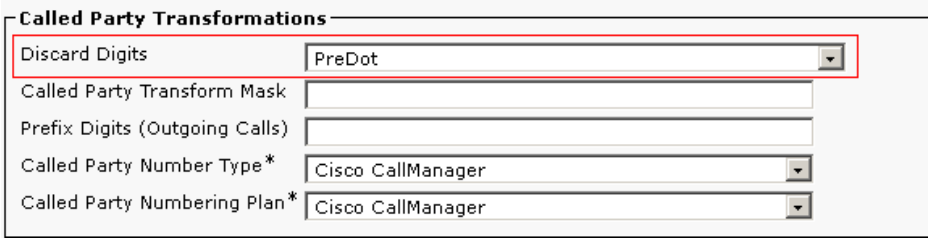
Supported format for the calling and called identity digits are:

- 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	It specifies how to inspect digit before they are sending to Orange Core SBC

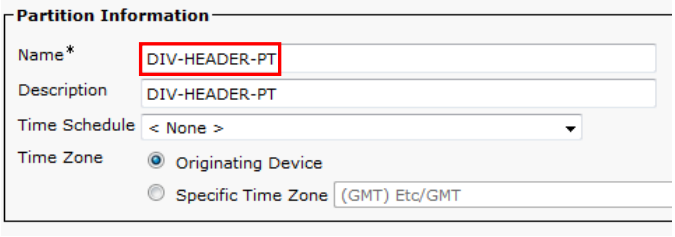
Testing was conducted in Orange Business Services lab.

Diversion Header manipulation

When call is forwarded CUCM sends extension number in Diversion Header. BT/BTIP requires site and extension number to be sent. Manipulation is allowed since release **8.6.2**.

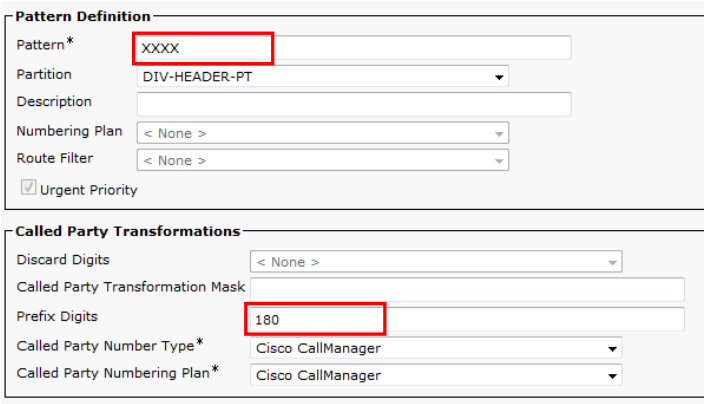
Configuration steps:

Go to **Call Routing** -> **Class of Control** -> **Partition** and create new partition by clicking on **Add New** button.



Setting	Value	Description
Name	Name for partition that will be assigned to CSS	Specify name for partition.

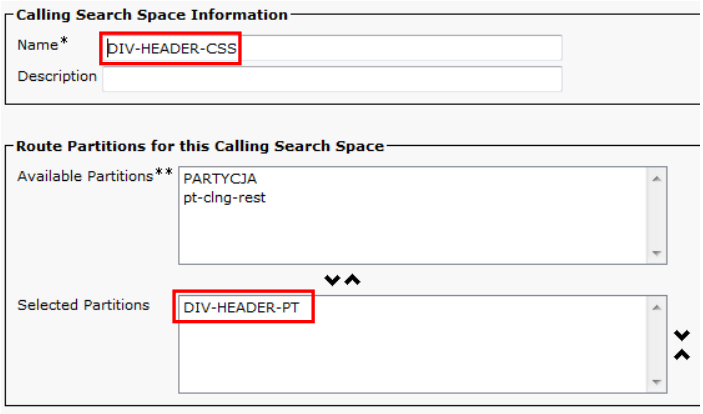
Go to **Call Routing** -> **Transformation** -> **Transformation Pattern** -> **Called PartyTransformation Pattern** and add New transformation pattern by clicking on **Add New** button.



Setting	Value	Description
Pattern	XXXX	Specify pattern that match extensions numbers
Prefix Digits	Site prefix	Prefix proper for configured site.

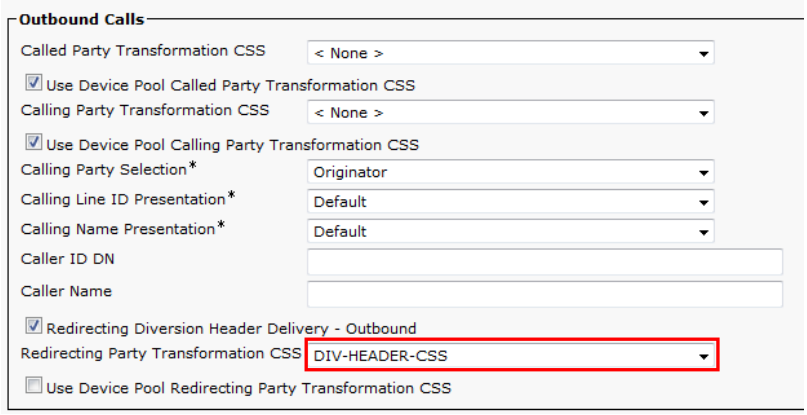
Go to **Call Routing** -> **Class of Control** -> **Calling Search Space** and add New CSS by clicking on **Add New** button:

Testing was conducted in Orange Business Services lab.



Setting	Value	Description
Name	CSS name	Specify name for Calling Search Space
Selected Partitions	Previously created partition	Assign previously created Partition to new Calling Search Space.

Go to **Device** -> **Trunk**, click on **Find** button and select trunk towards BT/BTIP and choose newly created CSS.



Setting	Value	Description
Redirecting Party Transformation CSS	Previously created CSS	Select Calling Search Space created for Diversion Header modification

This step must be repeated for all trunks towards SBCs.

Testing was conducted in Orange Business Services lab.

Acronyms

Acronym	Definitions
BTIP	Business Talk IP
CUBE	Cisco Unified Border Element
CUCM	Cisco Unified Communications Manager
CUACx	Cisco Unified Attendant Console (Standard/Advanced)
MTP	Media Termination Point
OBS	Orange Business Services
POP	Point of Presence
SBC	Session Border Controller
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol
SRST	Survivable Remote Service Telephony

Testing was conducted in Orange Business Services lab.

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