Vodafone: Connecting Cisco Unified Communications Manager 10.5 via the Cisco Unified Border Element [IOS 15.4(1)T] Using SIP

November 20, 2014
Media Resource Group Configuration ............................................................................................................. 71
Media Resource Group List Configuration ..................................................................................................... 72
Off-net calls via Vodafone’s network .............................................................................................................. 73
Dialplan .......................................................................................................................................................... 84
Cisco Analog Gateway VG224 configuration .................................................................................................. 87
Acronyms ....................................................................................................................................................... 93
Test Results.................................................................................................................................................... 95
Introduction

Service Providers today, such as Vodafone, 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 a centralized IP to TDM gateway to provide on-net and off-net services. Vodafone is a SP offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity via either Analog or T1 lines. 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.

This application note describes how to configure a Cisco Unified Communications Manager (CUCM) 10.5 with a Cisco Unified Border Element (CUBE) for connectivity to Vodafone SIP trunk service. The deployment model covered in this application note is CPE (CUCM 10.5/CUBE) to PSTN (Vodafone). This document does not address emergency outbound calls. For emergency feature service details contact Vodafone directly.

- Testing was performed in accordance to Vodafone and all features were verified. Key features verified are:
  - Basic Calls
  - Basic Calls with Calling Name and Number as allowed or restricted
  - DTMF Relay
  - Call Conference
  - Call Transfer (Blind, Attended, Early Attended)
  - Hold and Resume
  - Voice Mail
  - Simultaneous Calls
  - Auto Attendant
  - International calls
  - G3 Fax with T.38 and G.711 pass-through
  - Call Forwarding - Find Me (Unconditional, Busy, No Reply)
  - Dial Plans
  - PRACK with SDP early-media cut-through

- 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 Vodafone SIP network and Cisco Unified Communications. The configuration described in this document details the important commands to have enabled for interoperability to be successful and care must be taken, by the network administrator deploying CUBE, to ensure these commands are set per each dial-peer requiring to interoperate to Vodafone SIP network.

- This application note does not cover the use of Calling Search Spaces (CSS) or Partitions on Cisco Unified Communications Manager.
Network Topology

System Components

Hardware Components
- Cisco UCS-C240-M3S VMware host running ESXi5.5 standard
- ISR G2 3900 series routers (2 routers were used for HA setup)
- IP phones 9900/7900 (different models, both SIP and SCCP where supported)
- Cisco Voice Gateways VG224

Software Requirements
- Cisco Unified Communications Manager Release 10.5.1.10000-7
- Cisco Unity Connection release 10.5.1.10000-7
- Cisco Unified Border Element IOS Version 15.4(1)T, CUBE software release 10.0
- Cisco Voice Gateway VG224 IOS Version 15.1(3)T3.
Features

Features Supported

- Incoming and outgoing off-net calls using G711 a-law and G729 with 20ms packetization
- SIP Early Offer
- Call hold/Resume
- Call transfer (Blind, semi-attended and attended)
- Call conference
- Call forward (all, busy, no answer)
- Calling ID restriction
- DTMF (RFC2833)
- Simultaneous Calls
- Auto Attendant
- International calls
- G3 fax with T.38 and G711 pass-through
- CUBE High Availability

Features Not Supported

- SG3 Fax
- Calling Name presentation
Caveats

- On outbound calls, calling name presentation is not being presented at the terminating phone
- No ring back for the call originator on semi attended call transfers
- To enable conference on Vodafone and Cisco UCM SIP trunk, it is required to configure a conference bridge (CFB) resource to initiate a three-way conference between end-points. See configuration section for details.
Configuration

Configuring Cisco Unified Border Element (CUBE)

Network Interface and CUBE HA

Configure CUBE High Availability (HA) using HSRP (Hot Standby Router Protocol). Two identical ISR G2s equipped with UC Technology Package License installed, 1G DRAM memory and Cisco IOS software release 15.4.1.T is required. Both routers must be physically located on the same Ethernet LAN. The CUBE configuration of both routers need to be identical except slight difference in HSRP configuration between the Active and standby routers. In our lab test, Dual-Attached deployment is used as shown in chapter Network Topology.

<table>
<thead>
<tr>
<th>Active CUBE</th>
<th>Standby CUBE</th>
</tr>
</thead>
<tbody>
<tr>
<td>ipc zone default</td>
<td>ipc zone default</td>
</tr>
<tr>
<td>association 1</td>
<td>association 1</td>
</tr>
<tr>
<td>no shutdown</td>
<td>no shutdown</td>
</tr>
<tr>
<td>protocol scpt</td>
<td>protocol scpt</td>
</tr>
<tr>
<td>local-port 5000</td>
<td>local-port 5000</td>
</tr>
<tr>
<td>local-ip 10.80.10.10</td>
<td>local-ip 10.80.10.16</td>
</tr>
<tr>
<td>remote-port 5000</td>
<td>remote-port 5000</td>
</tr>
<tr>
<td>remote-ip 10.80.10.16</td>
<td>remote-ip 10.80.10.16</td>
</tr>
</tbody>
</table>

Explanation

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ipc zone default</td>
<td>Configures the Inter-Device Communication Protocol(IPC) and enters IPC zone configuration mode</td>
</tr>
<tr>
<td>Association 1</td>
<td>Configures an association between the 2 routers</td>
</tr>
<tr>
<td>No shutdown</td>
<td>Restarts a disabled association and its associated transport protocol</td>
</tr>
<tr>
<td>Protocol scpt</td>
<td>Configure Stream Control Transmission Protocol(SCTP) as the transport protocol</td>
</tr>
<tr>
<td>Local-port port_num</td>
<td>Configures the local SCTP port number to communicate with redundant peer, 5000 must be used.</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Local-ip ip_addr</td>
<td>Defines the local router’s IP address to use to communicate with redundant peer</td>
</tr>
<tr>
<td>Remote-port port_num</td>
<td>Configures the remote SCTP port number, 5000 must be used</td>
</tr>
<tr>
<td>Remote-ip ip_addr</td>
<td>Defines remote router’s IP address to use to communicate with redundant peer</td>
</tr>
</tbody>
</table>

### Active CUBE

```
voice service voip
  ip address trusted list
    ipv4 0.0.0.0 0.0.0.0
    ipv4 62.140.159.241
  address-hiding
  mode border-element
  allow-connections sip to sip
  redundancy
  !
  ....
  redundancy inter-device
  scheme standby SB
  !
  track 1 interface GigabitEthernet/1 line-protocol
  !
  track 2 interface GigabitEthernet/0 line-protocol
```

### Standby CUBE

```
voice service voip
  ip address trusted list
    ipv4 0.0.0.0 0.0.0.0
    ipv4 62.140.159.241
  address-hiding
  mode border-element
  allow-connections sip to sip
  redundancy
  !
  ....
  redundancy inter-device
  scheme standby SB
  !
  track 1 interface GigabitEthernet/1 line-protocol
  !
  track 2 interface GigabitEthernet/0 line-protocol
```
### Explanation

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mode border-element</td>
<td>Enable CUBE on both routers</td>
</tr>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>redundancy</td>
<td>Enable CUBE redundancy and call checkpointing on both routers</td>
</tr>
<tr>
<td>Redundancy inter-device</td>
<td>Enable HSRP</td>
</tr>
<tr>
<td>Scheme standby SB</td>
<td>Enable standby(HSRP) as redundancy state tracking scheme with group name---SB</td>
</tr>
<tr>
<td>Tracking <code>obj_num</code> interface <code>int_id</code> <code>line-protocol</code></td>
<td>Create a tracking list to track the line-protocol state of an interface</td>
</tr>
<tr>
<td></td>
<td>• The <code>obj_num</code> identify the tracked object with range from 1 to 500.</td>
</tr>
<tr>
<td></td>
<td>• The <code>int_id</code> is the interface being tracked.</td>
</tr>
</tbody>
</table>
Active CUBE

interface GigabitEthernet0/0
description WAN Interface to Vodafone
ip address 10.64.3.186 255.255.0.0
standby delay minimum 30 reload 60
standby version 2
standby 0 ip 10.64.3.185
standby 0 priority 60
standby 0 preempt delay minimum 10
standby 0 name SB
standby 0 track 1 decrement 20
standby 0 track 2 decrement 20
duplex auto
speed auto
!

interface GigabitEthernet0/1
description LAN Interface to CUCM
ip address 10.80.10.10 255.255.255.0
standby delay minimum 30 reload 60
standby version 2
standby 6 ip 10.80.10.11
standby 6 priority 60
standby 6 preempt delay minimum 10
standby 6 track 1 decrement 20
standby 6 track 2 decrement 20
duplex auto
speed auto
!

Standby CUBE

interface GigabitEthernet0/0
description WAN interface to Vodafone
ip address 10.64.3.187 255.255.0.0
standby delay minimum 30 reload 60
standby version 2
standby 0 ip 10.64.3.185
standby 0 priority 50
standby 0 preempt delay minimum 10
standby 0 name SB
standby 0 track 1 decrement 20
standby 0 track 2 decrement 20
duplex auto
speed auto
!

interface GigabitEthernet0/1
description LAN interface to CUCM
ip address 10.80.10.16 255.255.255.0
standby delay minimum 30 reload 60
standby version 2
standby 6 ip 10.80.10.11
standby 6 priority 50
standby 6 preempt delay minimum 10
standby 6 track 1 decrement 20
standby 6 track 2 decrement 20
duplex auto
speed auto
!
Explanation

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Interface type number</td>
<td>Configures an interface type and enters interface configuration mode</td>
</tr>
<tr>
<td>Ip address ip-addr mask</td>
<td>Specifies the ip address and mask for the interface</td>
</tr>
<tr>
<td>Standby delay minimum min-sec</td>
<td>Configures the delay period before the initialization of HSRP groups</td>
</tr>
<tr>
<td>reload reload-sec</td>
<td></td>
</tr>
<tr>
<td>Standby version ver</td>
<td>Specify the version of HSRP groups, ver1 or ver2</td>
</tr>
<tr>
<td>Standby grp ip ip-addr</td>
<td>Configures the HSRP group and associated virtual IP address</td>
</tr>
<tr>
<td>Standby grp priority pri</td>
<td>Configures HSRP group grp priority</td>
</tr>
<tr>
<td>Standby grp preempt delay</td>
<td>Configures HSRP preemption and preemption delay</td>
</tr>
<tr>
<td>minimum sec</td>
<td></td>
</tr>
<tr>
<td>Standby grp name name</td>
<td>Configures HSRP group name</td>
</tr>
<tr>
<td>Standby grp track obj_num</td>
<td>Configures HSRP to track an object and change the Hot Standby priority on the</td>
</tr>
<tr>
<td>decrement pri</td>
<td>basis of the state of the object</td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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

Default CUBE configuration enables CUBE to work in media flow-through mode (this mode was used in this certification test). In order to enable flow-around mode, please perform the following actions:

**voice service voip**

- media flow-around

**Codecs**

Vodafone’s network negotiates G729r8 codec for voice

**voice class codec 1**

codec preference 1 g729r8

codec preference 2 g711alaw
Translation rule

In this test, we configured translation rules to modify the number we wanted to send, include calling and called number. You can create the rules using command translation rule, then assign it for either calling or called in each dial-peer.

translation-rule 1

Rule 1 2090 0387002090
Rule 2 2091 0387002091
Rule 3 2095 0387002095
Rule 4 2092 0387002092
!

translation-rule 2

Rule 1 0387002090 2090
Rule 2 0387002091 2091
Rule 3 0387002092 2092
Rule 4 0387002095 2095

dial-peer voice 15 voip

description *** CUCM PBX 1 -> VDF VOF International PSTN ***
huntstop
preference 1
destination-pattern 001T
signaling forward unconditional
translate-outgoing calling 1
session protocol sipv2
session target ipv4:62.140.159.241
session transport udp
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
## Explanation

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>translation-rule $n$</td>
<td>Enters translation rule configuration mode</td>
</tr>
<tr>
<td>Rule $x$</td>
<td>Detail match and replace rule $x$</td>
</tr>
<tr>
<td>translate-outgoing calling $n$</td>
<td>Specifies the translation rule set to apply to the calling number</td>
</tr>
<tr>
<td>translate-outgoing called $n$</td>
<td>Specifies the translation rule set to apply to the called number</td>
</tr>
</tbody>
</table>

## Dial Peer

CUCM uses dial-peer to route the call based on the digits. The following is an example for this test only in our lab environment.

!-----incoming call from Vodafone and route to the end users at CUCM -----!

dial-peer voice 990 voip

description ***VOV GW->PBX incoming dial peer ****
translation-profile incoming vov_prefix_99
preference 1
session protocol sipv2
session transport udp
incoming uri from 99
voice-class codec 1
voice-class sip dtmf-relay force rtp-nte
voice-class sip options-keepalive
dtmf-relay rtp-nte
fax-relay ecm disable
no vad

! dial-peer voice 990 voip

description *** VOV GW->PBX
translation-profile outgoing vov_prefix_del_99
preference 1
destination-pattern 03T
session protocol sipv2
session target ipv4:10.80.10.3
session transport udp
voice-class codec 1
voice-class sip dtmf-relay force rtp-nte
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!

----- outgoing call to Vodafone network with 11 digits dialing -----!

dial-peer voice 17 voip
description *** CUCM PBX 1 -> VDF VOF International PSTN *** Incoming Dial Peer ***
huntstop
preference 1
session protocol sipv2
session transport udp
incoming called-number 001T
voice-class codec 1
voice-class sip options-keepalive
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!

dial-peer voice 15 voip
description *** CUCM PBX 1 -> VDF VOF International PSTN *** Outgoing Dial Peer ***
huntstop
preference 1
destination-pattern 001T
signaling forward unconditional
translate-outgoing calling 1
session protocol sipv2
session target ipv4:62.140.159.241
session transport udp
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!

**Explanation**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dial-peer voice <em>n</em> voip</td>
<td>Enters dial peer configuration mode</td>
</tr>
<tr>
<td>destination-pattern <em>string</em></td>
<td>Configures the dial peer’s destination pattern so that system can reconcile dialed digits with a telephone number</td>
</tr>
<tr>
<td>session target <em>tar</em></td>
<td>Defines target device IP address or host name/FQDN</td>
</tr>
<tr>
<td>session transport <em>udp/tcp</em></td>
<td>Specifies the protocol for the call</td>
</tr>
<tr>
<td>no voice-class sip outbound-proxy</td>
<td>This will remove the global outbound proxy for this dial-peer</td>
</tr>
<tr>
<td>voice-class sip bind control source-interface <em>int</em></td>
<td>Sets a source interface for signaling packets</td>
</tr>
<tr>
<td>voice-class sip bind media source-interface <em>int</em></td>
<td>Sets a source interface for media packets</td>
</tr>
</tbody>
</table>
Conference Bridge

The following is an example of conference bridge configuration, for this test only in our lab environment.

voice-card 0
dspfarm
dsp services dspfarm
!
sccp local GigabitEthernet0/1
sccp ccm 10.80.10.2 identifier 2 version 7.0
sccp ccm 10.80.10.3 identifier 1 priority 1 version 7.0
sccp
!
sccp ccm group 1
  bind interface GigabitEthernet0/1
  associate ccm 1 priority 1
  associate ccm 2 priority 2
  associate profile 1 register my-Conf
!
---The name should match the conference bridge configured in Conference Bridge ---!
!
dspfarm profile 1 conference
  codec g729r8
  codec g711alaw
  codec g711ulaw
  maximum sessions 8
  associate application SCCP
Configuration example

The following contains the sample configuration of Primary CUBE with all parameters mentioned previously. This is the configuration for CUCM and 2 CUBEs in an HA setup.

Primary CUBE Configuration
CUCM_Vodafone#show run
Building configuration...
Current configuration : 13952 bytes

! Last configuration change at 23:32:34 UTC Wed Nov 19 2014 by cisco
!
version 15.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname CUCM_Vodafone
!
boot-start-marker
boot-end-marker
!
aqm-register-fnf
!
! card type command needed for slot/vwic-slot 0/0
logging buffered 51200 warnings
enable secret 5 $1$qxj7$lrgJ7sS7zFYWF3bUbPpWy0
!
!
ipc zone default
association 1
no shutdown
protocol sctp
local-port 5000
local-ip 10.80.10.10
remote-port 5000
remote-ip 10.80.10.16
no aaa new-model

ip domain name yourdomain.com
ip cef
no ipv6 cef

multilink bundle-name authenticated

crypto pki trustpoint TP-self-signed-61138991
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-61138991
  revocation-check none
  rsakeypair TP-self-signed-61138991

crypto pki certificate chain TP-self-signed-61138991
  certificate self-signed 01
    30820227 30820190 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
    2F312D30 2B060355 04031324 494F532D 53656C66 2D536967 6E65642D 436574696669636174652D3631333839393130019F300D06092A864886 F70D0101 05050030
  6A65666963642D3130
    5A170D32 30303030 30303030 30303030 302F312D 302B0603 5504031324494F532D 53656C66 2D536967 6E65642D 436574696669636174652D3631333839393130019F300D06092A864886 F70D0101 05050030
    6A65666963642D3130
      5A170D32 30303030 30303030 30303030 302F312D 302B0603 5504031324494F532D 53656C66 2D536967 6E65642D 436574696669636174652D3631333839393130019F300D06092A864886 F70D0101 05050030
    6A65666963642D3130
      5A170D32 30303030 30303030 30303030 302F312D 302B0603 5504031324494F532D 53656C66 2D536967 6E65642D 436574696669636174652D3631333839393130019F300D06092A864886 F70D0101 05050030
quit
voice-card 0
dspfarm
dsp services dspfarm
!
!
voice call send-alert
voice rtp send-receive
!
voice service voip
ip address trusted list
ipv4 0.0.0.0 0.0.0.0
ipv4 62.140.159.241
address-hiding
mode border-element
allow-connections sip to sip
redundancy
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711alaw
sip
header-passing
error-pass-through
privacy pstn
early-offer forced
midcall-signaling pass-through
privacy-policy pass-through
g729 annexb-all
pass-thru headers unsupplied
sip-profiles 102
!
!
voice class uri 99 sip
  host ipv4:62.140.159.241
  host ipv4:62.140.159.242
voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711alaw

voice class sip-profiles 102
  request INVITE sip-header Diversion modify "<sip:20(..)@" "<sip:03870020:1@"
  response ANY sip-header Allow-Header modify "UPDATE," ""

voice translation-rule 2

voice translation-rule 99
  rule 1 /\+\+\+/ /+99/
  rule 2 /\+/ /99/

voice translation-rule 991
  rule 1 /\+99/ /+
  rule 2 /\99/ //

voice translation-profile vov_prefix_99
  translate called 99

voice translation-profile vov_prefix_del_99
  translate called 991

license udi pid C3900-SPE200/K9 sn FOC174142WS

hw-module pvdm 0/0

hw-module pvdm 0/1
username cisco privilege 15 password 0 tekV1z10n

redundancy inter-device
  scheme standby SB

redundancy

track 1 interface GigabitEthernet0/1 line-protocol

track 2 interface GigabitEthernet0/0 line-protocol

translation-rule 1
  Rule 1 2090 0387002090
  Rule 2 2091 0387002091
  Rule 3 2095 0387002095
  Rule 4 2092 0387002092

translation-rule 2
  Rule 1 0387002090 2090
  Rule 2 0387002091 2091
  Rule 3 0387002092 2092
  Rule 4 0387002095 2095

interface GigabitEthernet0/0
  description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
  ip address 10.64.3.186 255.255.0.0
  standby delay minimum 30 reload 60
  standby version 2
  standby 0 ip 10.64.3.185
  standby 0 priority 60
standby 0 preempt delay minimum 10
standby 0 name SB
standby 0 track 1 decrement 20
standby 0 track 2 decrement 20
duplex auto
speed auto
!
interface GigabitEthernet0/1
  ip address 10.80.10.10 255.255.255.0
  standby delay minimum 30 reload 60
  standby version 2
  standby 6 ip 10.80.10.11
  standby 6 priority 60
  standby 6 preempt delay minimum 10
  standby 6 track 1 decrement 20
  standby 6 track 2 decrement 20
duplex auto
speed auto
!
interface GigabitEthernet0/2
  no ip address
  shutdown
duplex auto
speed auto
!
interface GigabitEthernet0/3
  no ip address
  shutdown
duplex auto
speed auto
!
ip forward-protocol nd
!
no ip http server
ip http access-class 23
ip http authentication local
no ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 172.16.0.0 255.255.0.0 10.80.10.1
!
!
nls resp-timeout 1
cpd cr-id 1
!
access-list 23 permit 10.10.10.0 0.0.0.7
!
control-plane
!
!
voice-port 0/1/0
!
voice-port 0/1/1
!
voice-port 0/1/2
!
voice-port 0/1/3
!
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
scerp local GigabitEthernet0/1
scerp ccm 10.80.10.2 identifier 2 version 7.0
scerp ccm 10.80.10.3 identifier 1 priority 1 version 7.0
scerp
!
sccp ccm group 1
bind interface GigabitEthernet0/1
associate ccm 1 priority 1
associate ccm 2 priority 2
associate profile 1 register my-Conf
!
!
dspfarm profile 1 conference
  codec g729r8
  codec g711alaw
  codec g711ulaw
  maximum sessions 8
  associate application SCCP

!  dial-peer voice 10 voip
description *** CUCM PBX 1 -> VDF CNolIP Mobile***
preference 1
destination-pattern 06T
  signaling forward unconditional
  translate-outgoing calling 1
  session protocol sipv2
  session target ipv4:62.140.159.242
  session transport udp
  voice-class codec 1
  voice-class sip options-keepalive
  voice-class sip bind control source-interface GigabitEthernet0/0
  voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
  fax-relay ecm disable
no vad
!
dial-peer voice 12 voip
description *** CUCM PBX 1 -> VDF VOF PSTN ***
preference 1
destination-pattern 0T
  signaling forward unconditional
translate-outgoing calling 1
session protocol sipv2
session target ipv4:62.140.159.241
session transport udp
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 11 voip
description *** CUCM PBX 1 -> VDF CNoIP Short numbers***
huntstop
preference 1
destination-pattern .T
signaling forward unconditional
translate-outgoing calling 1
session protocol sipv2
session target ipv4:62.140.159.242
session transport udp
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 13 voip
description *** CUCM PBX 1 -> VDF VOV Emergency ***
preference 1
destination-pattern 112
signaling forward unconditional
translate-outgoing calling 1
session protocol sipv2
session target ipv4:62.140.159.241
session transport udp
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 14 voip
description *** CUCM PBX 1 -> VDF VOV Service Numbers ***
  preference 1
destination-pattern 140T
signaling forward unconditional
translate-outgoing calling 1
session protocol sipv2
session target ipv4:62.140.159.241
session transport udp
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 99 voip
description *** VOV GW->PBX
translation-profile outgoing vov_prefix_del_99
  preference 1
destination-pattern 03T
session protocol sipv2
session target ipv4:10.80.10.3
session transport udp
voice-class codec 1
voice-class sip dtmf-relay force rtp-nte
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 991 voip
description *** VoV +CNoIP -> CUCM PBX 1 ***
translation-profile outgoing vov_prefix_del_99
huntstop
preference 1
destination-pattern 99T
translate-outgoing called 2
session protocol sipv2
session target ipv4:10.80.10.3
session transport udp
voice-class codec 1
voice-class sip dtmf-relay force rtp-nte
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 992 voip
description *** Vodafone VoV + CNoIP -> CUCM PBX 1 ***
translation-profile outgoing vov_prefix_del_99
huntstop
preference 1
destination-pattern +99T
translate-outgoing called 2
session protocol sipv2
session target ipv4:10.80.10.3
session transport udp
voice-class codec 1
voice-class sip dtmf-relay force rtp-nte
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 15 voip
description *** CUCM PBX 1 -> VDF VOF International PSTN ***
huntstop
preference 1
destination-pattern 001T
signaling forward unconditional
translate-outgoing calling 1
session protocol sipv2
session target ipv4:62.140.159.241
session transport udp
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 16 voip
description CUCM PBX --> Service Numbers
preference 1
destination-pattern 12T
signaling forward unconditional
translate-outgoing calling 1
session protocol sipv2
session target ipv4:62.140.159.241
session transport udp
voice-class codec 1
voice-class sip options-keepalive
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 990 voip
description ***VOV GW->PBX incoming dial peer ****
translation-profile incoming vov_prefix_99
preference 1
session protocol sipv2
session transport udp
incoming uri from 99
voice-class codec 1
voice-class sip dtmf-relay force rtp-nte
voice-class sip options-keepalive
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 17 voip
description *** CUCM PBX 1 -> VDF VOF International PSTN *** Incoming Dial P
huntstop
preference 1
session protocol sipv2
session transport udp
incoming called-number 001T
voice-class codec 1
voice-class sip options-keepalive
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
!
gateway
  media-inactivity-criteria all
timer receive-rtcp 5
timer receive-rtp 1200
!
sip-ua
  keepalive target ipv4:62.140.159.241:5060
  no remote-party-id
disable-early-media 180
timers keepalive active 30
  connection-reuse
  !
  !
  !
gatekeeper
  shutdown
  !
  !
banner exec ^C
% Password expiration warning.

-----------------------------------------------------------------------
Cisco Configuration Professional (Cisco CP) is installed on this device
and it provides the default username "cisco" for one-time use. If you have
already used the username "cisco" to login to the router and your IOS image
supports the "one-time" user option, then this username has already expired.
You will not be able to login to the router with this username after you exit
this session.

It is strongly suggested that you create a new username with a privilege level
of 15 using the following command.

username <myuser> privilege 15 secret 0 <mypassword>

Replace <myuser> and <mypassword> with the username and password you want to
use.
Cisco Configuration Professional (Cisco CP) is installed on this device. This feature requires the one-time use of the username "cisco" with the password "cisco". These default credentials have a privilege level of 15.

YOU MUST USE CISCO CP or the CISCO IOS CLI TO CHANGE THESE PUBLICLY-KNOWN CREDENTIALS

Here are the Cisco IOS commands.

```
username <myuser> privilege 15 secret 0 <mypassword>
no username cisco
```

Replace <myuser> and <mypassword> with the username and password you want to use.

IF YOU DO NOT CHANGE THE PUBLICLY-KNOWN CREDENTIALS, YOU WILL NOT BE ABLE TO LOG INTO THE DEVICE AGAIN AFTER YOU HAVE LOGGED OFF.

For more information about Cisco CP please follow the instructions in the QUICK START GUIDE for your router or go to http://www.cisco.com/go/ciscocp
line vty 5 15
exec-timeout 0 0
privilege level 15
login local
transport input telnet ssh
!
scheduler allocate 20000 1000
!
end
Standby CUBE Configuration
Current configuration : 10868 bytes

! Last configuration change at 18:36:21 UTC Sun Nov 23 2014 by cisco
!
version 15.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname TWC_CUBE2
!
boot-start-marker
boot system flash0:c3900e-universalk9-mz.SPA.154-1.T.bin
boot-end-marker
!
aqm-register-fnf
!
logging buffered 51200 warnings
enable secret 5 $1$q/Bk$BuOl4yptT4JPxDeWSCcBd.
!
!
ipc zone default
association 1
no shutdown
protocol scvp
    local-port 5000
    local-ip 10.80.10.16
    remote-port 5000
    remote-ip 10.80.10.10
!
no aaa new-model
!
!
ip name-server 10.64.1.3
ip cef
no ipv6 cef
!
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-3709846528

  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-3709846528
  revocation-check none
  rsakeypair TP-self-signed-3709846528
!
!
crypto pki certificate chain TP-self-signed-3709846528

certificate self-signed 01
  0382022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
  31312F30 2D060355 04031326 49F532D 53656C66 2D536967 6E65642D 43657274
  69666963 6174652D 33373039 38343635 3238301E 170D3134 30383236 32313335
  35325A17 0D323030 31303310 30303030 305A320F 300D0609 2A864886 F70D0101 01050003
  31303030 300D0609 2A864886 F70D0101 01050003 31303030 300D0609 2A864886 F70D0101 01050003
  31303030 300D0609 2A864886 F70D0101 01050003 31303030 300D0609 2A864886 F70D0101 01050003

1810CE51 F561CD41 24990148 0E798600 71068690 36683A6B A7E16F02 A66F8471
71E35FA6 C13EBD9D C6887395 683BB37A 27B11487 97EEDF44 0E881127 EC99BC0F
488D3C31 B36459DC FAA585B5 DD209151 8AEDCEA7 847D8ACB 9DEB0523 3818EF93
B21AD7EB B41CEC57 39FB6D65 F4BD27E6 6B548ECC 7C85320F 00436C79 F5978280
44250203 010001A3 53305130 0F603035 1D310101 FF040530 030101FF 30180609
551D2304 18301680 14841E1D 28893357 F087CC1E BBD38D76 C91253B9 4E301D06
03551D00 04160414 841E1D28 893357F0 87CC1EBB D3BD76C9 1253B94E 300D0609
2A864886 F70D0101 05050003 81810013 876F5E4D 896D48AB B4E92489 B1C42EE6
60EAC45D BD88C5A7 39EA149E F2576DD3 95177726 7C63256F B1746B16 2A22BEBE
06DCC8B3 0B8A373E 5F2813D B70E577D 54926FA5 6B17CFB3 97575471 9587DC43
7428A023 11E71071 9E6EFD10 473A4DA6 FBD2209C 1DE25F6D 4CDF4AF5 A0EF1B13
8994EB81 B772150C 6A0416ED E295DA
quit
voice-card 0
dspfarm
dsp services dspfarm
voice call send-alert
voice rtp send-recv

voice service voip
ip address trusted list
ipv4 0.0.0.0 0.0.0.0
ipv4 62.140.159.241
no ip address trusted authenticate
address-hiding
mode border-element
allow-connections sip to sip
redundancy
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711alaw
sip
header-passing
error-passthru
privacy pstn
early-offer forced
midcall-signaling passthru
privacy-policy passthru
g729 annexb-all
pass-thru headers unsupp
sip-profiles 102
!
!
voice class uri 99 sip
host ipv4:62.140.159.241
host ipv4:62.140.159.242
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711alaw
!
!voice class sip-profiles 102
request INVITE sip-header Diversion modify "<sip:20(\..)@" "<sip:03870020\1@"
response ANY sip-header Allow-Header modify "UPDATE," ""

!

voice translation-rule 99
rule 1 /^\+/ /+99/
rule 2 /^/ /99/


voice translation-rule 991
rule 1 /^\+99/ /+
rule 2 /^99/ //


voice translation-profile vov_prefix_99
translate called 99


voice translation-profile vov_prefix_del_99
translate called 991


license udi pid C3900-SPE250/K9 sn FOC15391VLH
license accept end user agreement
license boot module c3900e technology-package securityk9


hw-module pvdm 0/0


username cisco privilege 15 password 0 tekV1z10n


redundancy inter-device
scheme standby SB


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

translation-rule 1
Rule 1 2090 0387002090
Rule 2 2091 0387002091
Rule 3 2095 0387002095
Rule 4 2092 0387002092
!

translation-rule 2
Rule 1 0387002090 2090
Rule 2 0387002091 2091
Rule 3 0387002092 2092
Rule 4 0387002095 2095
!

interface GigabitEthernet0/0
(description CUBE WAN)
ip address 10.64.3.187 255.255.0.0
standby delay minimum 30 reload 60
standby version 2
standby 0 ip 10.64.3.185
standby 0 priority 50
standby 0 preempt delay minimum 10
standby 0 name SB
standby 0 track 1 decrement 20
standby 0 track 2 decrement 20
duplex auto
speed auto
!

interface GigabitEthernet0/1
(description CUBE LAN)
ip address 10.80.10.16 255.255.255.0
standby delay minimum 30 reload 60
standby version 2
standby 6 ip 10.80.10.11
standby 6 priority 50
standby 6 preempt delay minimum 10
standby 6 track 1 decrement 20
standby 6 track 2 decrement 20
duplex auto
speed auto
!
interface GigabitEthernet0/2
  no ip address
  shutdown
duplex auto
  speed auto
!
interface GigabitEthernet0/3
  no ip address
  shutdown
duplex auto
  speed auto
!
ip forward-protocol nd
!
no ip http server
ip http access-class 23
ip http authentication local
no ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
ip route 0.0.0.0 0.0.0.0 10.64.1.1
!
!
nls resp-timeout 1
cpd cr-id 1
access-list 23 permit 10.10.10.0 0.0.0.7

control-plane

mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable

mgcp profile default

sccp local GigabitEthernet0/1
sccp ccm 10.80.10.2 identifier 2 version 7.0
sccp ccm 10.80.10.3 identifier 1 priority 1 version 7.0

sccp ccm group 1
bind interface GigabitEthernet0/1
associate ccm 1 priority 1
associate ccm 2 priority 2
associate profile 1 register my-Conf

dspfarm profile 1 conference
codec g729r8
codec g711alaw
codec g711ulaw
maximum sessions 8
associate application SCCP
!
dial-peer voice 10 voip
description *** CUCM PBX 1 -> VDF CNoIP Mobile***
preference 1
destination-pattern 06T
signaling forward unconditional
translate-outgoing calling 1
session protocol sipv2
session target ipv4:62.140.159.242
session transport udp
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 12 voip
description *** CUCM PBX 1 -> VDF VOF PSTN ***
preference 1
destination-pattern 0T
signaling forward unconditional
translate-outgoing calling 1
session protocol sipv2
session target ipv4:62.140.159.241
session transport udp
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
! 

dial-peer voice 11 voip  
  description *** CUCM PBX 1 -> VDF CNoIP Short numbers*** 
  huntstop 
  preference 1 
  destination-pattern .T 
  signaling forward unconditional 
  translate-outgoing calling 1 
  session protocol sipv2 
  session target ipv4:62.140.159.242 
  session transport udp 
  voice-class codec 1 
  voice-class sip options-keepalive 
  voice-class sip bind control source-interface GigabitEthernet0/0 
  voice-class sip bind media source-interface GigabitEthernet0/0 
  dtmf-relay rtp-nte 
  fax-relay ecm disable 
  no vad 
! 

dial-peer voice 13 voip  
  description *** CUCM PBX 1 -> VDF VOV Emergency *** 
  preference 1 
  destination-pattern 112 
  signaling forward unconditional 
  translate-outgoing calling 1 
  session protocol sipv2 
  session target ipv4:62.140.159.241 
  session transport udp 
  voice-class codec 1 
  voice-class sip options-keepalive 
  voice-class sip bind control source-interface GigabitEthernet0/0 
  voice-class sip bind media source-interface GigabitEthernet0/0 
  dtmf-relay rtp-nte 
  fax-relay ecm disable 
  no vad 
!
dial-peer voice 14 voip
description *** CUCM PBX 1 -> VDF VOV Service Numbers ***
preference 1
destination-pattern 140T
signaling forward unconditional
translate-outgoing calling 1
session protocol sipv2
session target ipv4:62.140.159.241
session transport udp
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 99 voip
description *** VOV GW->PBX
translation-profile outgoing vov_prefix_del_99
preference 1
destination-pattern 03T
session protocol sipv2
session target ipv4:10.80.10.3
session transport udp
voice-class codec 1
voice-class sip dtmf-relay force rtp-nte
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 991 voip
description *** VoV +CNoIP -> CUCM PBX 1 ***
translation-profile outgoing vov_prefix_del_99
huntstop
preference 1
destination-pattern 99T
translate-outgoing called 2
session protocol sipv2
session target ipv4:10.80.10.3
session transport udp
voice-class codec 1
voice-class sip dtmf-relay force rtp-nte
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 992 voip
description *** Vodafone VoV + CNoIP -> CUCM PBX 1 ***
translation-profile outgoing vov_prefix_del_99
huntstop
preference 1
destination-pattern +99T
translate-outgoing called 2
session protocol sipv2
session target ipv4:10.80.10.3
session transport udp
voice-class codec 1
voice-class sip dtmf-relay force rtp-nte
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 15 voip
description *** CUCM PBX 1 -> VDF VOF International PSTN ***
huntstop
preference 1
destination-pattern 001T
signaling forward unconditional
translate-outgoing calling 1
session protocol sipv2
session target ipv4:62.140.159.241
session transport udp
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 16 voip
description CUCM PBX --> Service Numbers
preference 1
destination-pattern 12T
signaling forward unconditional
translate-outgoing calling 1
session protocol sipv2
session target ipv4:62.140.159.241
session transport udp
voice-class codec 1
voice-class sip options-keepalive
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 990 voip
description ***VOV GW->PBX incoming dial peer ****
translation-profile incoming vov_prefix_99
preference 1
session protocol sipv2
session transport udp
incoming uri from 99
voice-class codec 1
voice-class sip dtmf-relay force rtp-nte
dtmf-relay rtp-nte
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
dial-peer voice 17 voip
description *** CUCM PBX 1 -> VDF VOF International PSTN *** Incoming Dial P
huntstop
preference 1
session protocol sipv2
session transport udp
incoming called-number 001T
voice-class codec 1
voice-class sip options-keepalive
dtmf-relay rtp-nte
dtmf-relay rtp-nte
fax-relay ecm disable
no vad
!
!
gateway
mediainactivitycriteria all
timer receive-rtcp 5
timer receive-rtp 1200
!
sip-ua
no remote-party-id
disable-early-media 180
timers keepalive active 30
connection-reuse
!
gatekeeper
  shutdown
!
line con 0
  login local
line aux 0
line vty 0 4
  exec-timeout 0 0
  privilege level 15
  logging synchronous
  login local
  transport input telnet ssh
line vty 5 15
  exec-timeout 0 0
  privilege level 15
  logging synchronous
  login local
  transport input telnet ssh
!
scheduler allocate 20000 1000
!
end
Figure 2 CUCM Version
Cisco CallManager service parameter

Go to System > Service Parameters, select the appropriate server (clus20pub) and then Cisco CallManager (Active) from the drop-down menu. We leave all fields in the service parameter as default values for this test.

Figure 3: CUCM Service Parameter
### Figure 4: CUCM Service Parameter cont.

#### SDL Trace
- **SDL Trace Data Flags**
  - Value: 0x000000111
- **SDL Trace Flush Immediately**
  - Value: False
- **SDL Trace Data Size**
  - Value: 0
- **SDL Trace Flags**
  - Value: True
- **SDL Trace Type Flags**
  - Value: 0x8000EB15

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

#### Clusterwide Parameters (Device - General)
- **Call Diagnostics Enabled**
  - Value: Disabled
- **Show Line Group Member DN in FinalCalledParty/Number CDR Field**
  - Value: False
- **Show Line Group Member Non Masked DN in FinalCalledParty/Number CDR Field**
  - Value: False
- **CGI New Call Accept Timer**
  - Value: 4
- **CGI Generate Digits Interval**
  - Value: 250
- **CGI Dial Digits Interval**
  - Value: 250
- **CGI Await Further Digits**
  - Value: False
- **CGI Use Wildcard Pattern as calledPartyDN**
  - Value: False
- **Retain Media on Disconnect with PI for Active Call**
  - Value: False
- **Station KeepAlive Interval**
  - Value: 60
- **Status Inquiry Poll Flag**
  - Value: False
- **Strip # Sign from Called Party Number**
  - Value: True
- **Session Handoff Alerting Timer**
  - Value: 10
- **T301 Timer**
  - Value: 180000
- **T302 Timer**
  - Value: 15000
- **T303 Timer**
  - Value: 4000
- **T305 Timer**
  - Value: 30000
- **T306 Timer**
  - Value: 30000
- **T308 Timer**
  - Value: 4000
- **T309 Timer**
  - Value: 90000
- **T310 Timer**
  - Value: 50000

© 2014 Cisco Systems, Inc. All rights reserved.

Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Page 50 of 97
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>T313 Timer</td>
<td>4000</td>
<td>4000</td>
</tr>
<tr>
<td>T316 Timer</td>
<td>120000</td>
<td>120000</td>
</tr>
<tr>
<td>T317 Timer</td>
<td>100000</td>
<td>100000</td>
</tr>
<tr>
<td>T321 Timer</td>
<td>30000</td>
<td>30000</td>
</tr>
<tr>
<td>T322 Timer</td>
<td>4000</td>
<td>4000</td>
</tr>
<tr>
<td>Tone on Hold Timer</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Unknown Caller ID Flag</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Offset</td>
<td>Offset</td>
</tr>
<tr>
<td>Always Display Original Dialed Number</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Name Display for Original Dialed Number When Translated</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Always Use Pls With Original Dialed Number</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Fail Call If Trusted Relay Point Allocation Fails</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Display Calling/Called ID When PI is Not Available</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Enable Transit Counter Processing on G723 Trunks</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Express FacilityIF Count</td>
<td>5</td>
<td>6</td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Figure 5: CUCM Service Parameter cont.
<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ring Setting of Busy Station Policy</td>
<td>Only Apply Ring Setting of Busy Station When Incoming Call Arrives</td>
<td>Ring Setting of Busy Station</td>
<td>Only Apply Ring Setting of Busy Station When Incoming Call Arrives</td>
</tr>
<tr>
<td>Transfer On-hook Enabled</td>
<td>False</td>
<td>Ring Setting of Idle Station</td>
<td>Ring</td>
</tr>
<tr>
<td>Ring Settings of Busy Station</td>
<td>Ring</td>
<td>Call Pickup Group Audio Alert Setting of Idle Station</td>
<td>Ring once</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting of Busy Station</td>
<td>Ring</td>
<td>Call Pickup Group Audio Alert Setting of Busy Station</td>
<td>Ring once</td>
</tr>
<tr>
<td>BIP Pickup Audio Alert Setting of Idle Station</td>
<td>Disable</td>
<td>Privacy Setting</td>
<td>True</td>
</tr>
<tr>
<td>BIP Pickup Audio Alert Setting of Busy Station</td>
<td>Disable</td>
<td>Enforce Privacy Setting on Held Calls</td>
<td>False</td>
</tr>
<tr>
<td>Privacy Setting</td>
<td>True</td>
<td>SIP Station KeepAlive Interval</td>
<td>120</td>
</tr>
<tr>
<td>Enforce Privacy Setting on Held Calls</td>
<td>False</td>
<td>SIP Station Realm</td>
<td>cmsoapline</td>
</tr>
<tr>
<td>Hunt Group Logoff Notification</td>
<td>None</td>
<td>Speed Dial Await Further Dialed</td>
<td>False</td>
</tr>
<tr>
<td>Speed Dial Await Further Dialed</td>
<td>False</td>
<td>Display CTI Route Point Name or DNI</td>
<td>False</td>
</tr>
<tr>
<td>Display Original Calling Number on Transfer from Cisco Unity</td>
<td>False</td>
<td>DDI Dialing Display Preference</td>
<td>False</td>
</tr>
<tr>
<td>DDI Dialing Display Preference</td>
<td>False</td>
<td>Insert Numerals in 12-Digit Numbers</td>
<td>False</td>
</tr>
<tr>
<td>Insert Numerals in 12-Digit Numbers</td>
<td>True</td>
<td>Allow Call Waiting During an In-Progress Outbound Analog Call</td>
<td>True</td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

---

**Figure 6: CUCM Service Parameter cont.**
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disable Resume from Shared-line MGCP FXS Port</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>DTMF Silence Tone Flag</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Enable Display TE in Codec 6</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Enable Sending PRI N12 Service Message</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Flash Hook Duration</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>Gateway Poll Timer</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Location ID PRI Progress Indicator IE (User Side Only)</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Matching Calling Party with Attendant Flag</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>MGCP Database Query Delay Timer</td>
<td>1000</td>
<td>1000</td>
</tr>
<tr>
<td>MGCP FXS On-Hook Pending Timer</td>
<td>3</td>
<td>3</td>
</tr>
<tr>
<td>MGCP Response Timer</td>
<td>30</td>
<td>30</td>
</tr>
<tr>
<td>MGCP Timer</td>
<td>3</td>
<td>3</td>
</tr>
<tr>
<td>Numbering Plan Info</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Overrun Receiving Flag for PRI</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Outgoing Media Connect Time for PRI</td>
<td>Connect ASAP</td>
<td>Connect ASAP</td>
</tr>
<tr>
<td>Port Release Timer</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>SMTP Cell Delay Timer</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>Stable in State 4 Plan</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Optimize MGCP Registration</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Suppress Out-of-Channels Alarms</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>E-Frame Timer</td>
<td>2000</td>
<td>2000</td>
</tr>
<tr>
<td>User-to-User IE Status</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Convert European Progress Message to Alerting</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Enable CMS PRI Notify Message from User to Network</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Audit CSS Channels Interval</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Digital and Analog Ports Enabled</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Clusterwide Parameters (Device - H323)</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Accept Unknown TCP Connection</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>BRI Enabled</td>
<td>False</td>
<td>False</td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

**Figure 7: CUCM Service Parameter cont.**
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Present Disconnect Flag</td>
<td>False</td>
<td></td>
</tr>
<tr>
<td>Check Progress Indicator Before Establishing Media</td>
<td>False</td>
<td></td>
</tr>
<tr>
<td>H225 Block Setup Destination</td>
<td>False</td>
<td></td>
</tr>
<tr>
<td>H225 DR Retry Timer</td>
<td>0</td>
<td></td>
</tr>
<tr>
<td>H225 Device Connect Timer</td>
<td>0</td>
<td></td>
</tr>
<tr>
<td>H225 DTMF Duration</td>
<td>100</td>
<td></td>
</tr>
<tr>
<td>H225 TopRec Retry</td>
<td>2</td>
<td></td>
</tr>
<tr>
<td>H225 Intercluster Call Throttle Timer</td>
<td>30</td>
<td></td>
</tr>
<tr>
<td>H225 T301 Timer</td>
<td>180000</td>
<td></td>
</tr>
<tr>
<td>H225 T302 Timer</td>
<td>15000</td>
<td></td>
</tr>
<tr>
<td>H225 T303 Timer</td>
<td>4000</td>
<td></td>
</tr>
<tr>
<td>H225 T304 Timer</td>
<td>50000</td>
<td></td>
</tr>
<tr>
<td>H225 T305 Timer</td>
<td>30000</td>
<td></td>
</tr>
<tr>
<td>H225 T310 Timer</td>
<td>60000</td>
<td></td>
</tr>
<tr>
<td>H225 TCP Timer</td>
<td>5</td>
<td></td>
</tr>
<tr>
<td>H245 TCS Timeout</td>
<td>10</td>
<td></td>
</tr>
<tr>
<td>H323 Calling Party Number Screening Indicator</td>
<td>Calling number screened and passed</td>
<td></td>
</tr>
<tr>
<td>Apply External Phone Number Mask for H.323 Calls</td>
<td>False</td>
<td></td>
</tr>
<tr>
<td>Tone on Connect</td>
<td>False</td>
<td></td>
</tr>
<tr>
<td>Wait Time for SDP with SR/RO Mode</td>
<td>3</td>
<td></td>
</tr>
<tr>
<td>RAS ARO Timer</td>
<td>3</td>
<td></td>
</tr>
<tr>
<td>RAS BRQ Timer</td>
<td>3</td>
<td></td>
</tr>
<tr>
<td>RAS DRQ Timer</td>
<td>3</td>
<td></td>
</tr>
<tr>
<td>RAS RRO Timer</td>
<td>3</td>
<td></td>
</tr>
<tr>
<td>Ras URO Timer</td>
<td>3</td>
<td></td>
</tr>
<tr>
<td>Retry Count for ARO</td>
<td>2</td>
<td></td>
</tr>
<tr>
<td>Retry Count for BRQ</td>
<td>2</td>
<td></td>
</tr>
<tr>
<td>Retry Count for DRQ</td>
<td>2</td>
<td></td>
</tr>
<tr>
<td>Retry Count for RRO</td>
<td>2</td>
<td></td>
</tr>
<tr>
<td>Retry Count for URO</td>
<td>1</td>
<td></td>
</tr>
<tr>
<td>Send Product ID and Version ID</td>
<td>False</td>
<td></td>
</tr>
</tbody>
</table>

Figure 8: CUCM Service Parameter cont.
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Send Unified CM Version as Version ID in H223 Setup</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>Send Progress Timer</td>
<td>3000</td>
<td>3000</td>
</tr>
<tr>
<td>Send H223 User Info Message</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>Status Inquiry Poll Timer</td>
<td>10000</td>
<td>10000</td>
</tr>
<tr>
<td>Device Name of GK-controlled Trunk That Will Use Port 1720</td>
<td>None</td>
<td>None</td>
</tr>
<tr>
<td>Host Name/IP Address of GK That Will Use RFC 2970 Port 1720</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>Fail Call if HTP Allocation Fails</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>Overlay Receiving Flow for H323</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>Allocate Transcoder for H.323 on Early Offer SIP Trunk for Calls with Early Media</td>
<td>False</td>
<td>True</td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

<table>
<thead>
<tr>
<th>Clustervide Parameters (Device - SIP)</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Interoperability Enabled</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Retry Count for SIP Bye</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Retry Count for SIP Cancel</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Retry Count for SIP Invite</td>
<td>6</td>
<td>6</td>
</tr>
<tr>
<td>Retry Count for SIP PRACK</td>
<td>6</td>
<td>6</td>
</tr>
<tr>
<td>Retry Count for SIP Rel1XX</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Retry Count for SIP Publish</td>
<td>6</td>
<td>6</td>
</tr>
<tr>
<td>Retry Count for SIP Response</td>
<td>6</td>
<td>6</td>
</tr>
<tr>
<td>SIP Connect Timer</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>SIP Disconnect Timer</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>SIP Expires Timer</td>
<td>160000</td>
<td>160000</td>
</tr>
<tr>
<td>SIP PRACK Timer</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>SIP Rel1XX Timer</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>SIP Trying Timer</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>SIP Publish Timer</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>SIP Min-SE Value</td>
<td>90</td>
<td>1800</td>
</tr>
<tr>
<td>SIP URI Handling</td>
<td>Reject</td>
<td>Reject</td>
</tr>
<tr>
<td>SIP statistics Periodic update Timer</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>SIP Session Expires Timer</td>
<td>1800</td>
<td>1800</td>
</tr>
</tbody>
</table>

Figure 9: CUCM Service Parameter cont.
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Trunk T3/1 Messages Retry</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>SIP TCP Unused Connection Timer</td>
<td>14</td>
<td>14</td>
</tr>
<tr>
<td>SIP TCP Timer</td>
<td>5</td>
<td>3</td>
</tr>
<tr>
<td>SIP Station TCP Port Throttle Threshold</td>
<td>100</td>
<td>100</td>
</tr>
<tr>
<td>SIP Trunk TCP Port Throttle Threshold</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>SIP V.150 Outbound SDP Offer Filtering</td>
<td>No Filtering</td>
<td>No Filtering</td>
</tr>
<tr>
<td>Send SIP Multicast TTL in SDP</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>Default PUBLISH Expiration Timer</td>
<td>3600</td>
<td>3600</td>
</tr>
<tr>
<td>Minimum PUBLISH Expiration Timer</td>
<td>60</td>
<td>60</td>
</tr>
<tr>
<td>IM and Presence Publish Trunk</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Send 180 Call Is Being Forwarded</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>Delay Sending 180 until 180/183 message is received</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Fail Call Over SIP Trunk if HTP Allocation Fails</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>Log Call-Related REFER/NOTIFY/SUBSCRIBE SIP Messages for Session Trace</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Port Received Timer for Outbound Call Setup</td>
<td>2</td>
<td>2</td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

### Clusterwide Parameters (Feature - General)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Park Display Timer</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Caller ID Display Priority Enabled</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Call Park Reversion Timer</td>
<td>60</td>
<td>60</td>
</tr>
<tr>
<td>Park Monitoring Reversion Timer</td>
<td>60</td>
<td>60</td>
</tr>
<tr>
<td>Park Monitoring Periodic Reversion Timer</td>
<td>30</td>
<td>30</td>
</tr>
<tr>
<td>Park Monitoring Forward No Retrieve Timer</td>
<td>300</td>
<td>300</td>
</tr>
<tr>
<td>Preserve globalCalled for Parked Calls</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Maximum Call Duration Timer</td>
<td>720</td>
<td>720</td>
</tr>
<tr>
<td>Maximum Hold Duration Timer</td>
<td>360</td>
<td>360</td>
</tr>
<tr>
<td>Party Entrance Tone</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Message Waiting Lamp Policy</td>
<td>Primary Line - Light and Prompt</td>
<td>Primary Line - Light and Prompt</td>
</tr>
<tr>
<td>Audible Message Waiting Indication Policy</td>
<td>OFF</td>
<td>OFF</td>
</tr>
<tr>
<td>Message Waiting Indicator Inbound Calling Search Space</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Multiple Tenant Mru Modes</td>
<td>False</td>
<td>False</td>
</tr>
</tbody>
</table>

Figure 10: CUCM Service Parameter cont.
### Clusterwide Parameters (Feature - Conference)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Suppress MOH to Conference Bridge</td>
<td>True</td>
</tr>
<tr>
<td>Drop Ad Hoc Conference</td>
<td>Never</td>
</tr>
<tr>
<td>Maximum Ad Hoc Conference</td>
<td>4</td>
</tr>
<tr>
<td>Maximum MeetMe Conference Unicast</td>
<td>4</td>
</tr>
<tr>
<td>Advanced Ad Hoc Conference Enabled</td>
<td>False</td>
</tr>
<tr>
<td>Choose Encrypted Audio Conference Instead Of Video Conference</td>
<td>True</td>
</tr>
<tr>
<td>Minimum Video Capable Participants To Allocate Video Conference</td>
<td>2</td>
</tr>
<tr>
<td>Enable Click-to-Conference for Third-Party Applications</td>
<td>False</td>
</tr>
<tr>
<td>IMS Conference Factory URL</td>
<td><a href="mailto:cucm-conference-factory@cucm1.company.com">cucm-conference-factory@cucm1.company.com</a>, <a href="mailto:cucm-conference-factory@cucm2.company.com">cucm-conference-factory@cucm2.company.com</a></td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

### Clusterwide Parameters (Feature - Call Security Status Policy)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Secure Call Icon Display Policy</td>
<td>All media except BFCP and IX transports must be encrypted</td>
</tr>
</tbody>
</table>

### Clusterwide Parameters (Feature - Forward)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward Maximum Hop Count</td>
<td>12</td>
</tr>
<tr>
<td>Forward No Answer Time</td>
<td>12</td>
</tr>
<tr>
<td>Max Forward Hop to PIN</td>
<td>12</td>
</tr>
<tr>
<td>Retain Forward Information</td>
<td>False</td>
</tr>
<tr>
<td>Forward By Recipient Enabled</td>
<td>False</td>
</tr>
<tr>
<td>Transform Forward By Recipient Destination</td>
<td>True</td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Figure 11: CUCM Service Parameter cont.
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Validated</th>
</tr>
</thead>
<tbody>
<tr>
<td>Always Forward Switch Voice Mail Calls</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Forward By Remote T1 Timer</td>
<td>10</td>
<td></td>
</tr>
<tr>
<td>Include Original Called Info for DSSG Call Div</td>
<td>Only after the first diversion</td>
<td></td>
</tr>
<tr>
<td>Set Private Numbering Plan for Call Forward</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>Set Type of Number for Call Forward</td>
<td>Level1RegionalNumber</td>
<td>Level1RegionalNumber</td>
</tr>
<tr>
<td>Max Forward Unregistered Hops to DN</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>CFA CSS Activation Policy</td>
<td>With Configured CSS</td>
<td>With Configured CSS</td>
</tr>
<tr>
<td>Cause Code When Maximum Forward Hop Count is</td>
<td>Normal Unspecified</td>
<td>Normal Unspecified</td>
</tr>
<tr>
<td>Triggered.</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

---

**Figure 12: CUCM Service Parameter cont.**
<table>
<thead>
<tr>
<th>Clusterwide Parameters (Feature - Path Replacement)</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Path Replacement Enabled</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Path Replacement on Terminated Calls</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Start Path Replacement Minimum Delay Time</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>Start Path Replacement Maximum Delay Time</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>Path Replacement T1 Timer</td>
<td>30</td>
<td>30</td>
</tr>
<tr>
<td>Path Replacement T2 Timer</td>
<td>15</td>
<td>15</td>
</tr>
<tr>
<td>Path Replacement PINX ID</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Path Replacement Calling Search Space</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Clusterwide Parameters (Feature - Call Back)</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Back Enabled Flags</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Call Back Notification Audio File Name</td>
<td>CallBack.raw</td>
<td>CallBack.raw</td>
</tr>
<tr>
<td>Connection Proposal Type</td>
<td>Connection Retention</td>
<td>Connection Retention</td>
</tr>
<tr>
<td>Connection Response Type</td>
<td>Default to Connection Retention</td>
<td>Default to Connection Retention</td>
</tr>
<tr>
<td>Call Back Request Protection T1 Timer</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Call Back Recall T1 Timer</td>
<td>20</td>
<td>20</td>
</tr>
<tr>
<td>Call Back Calling Search Space</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>No Path Reservation</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Set Private Numbering Plan for Call Back</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Set Type of Number for Call Back</td>
<td>Level1RegionalNumber</td>
<td>Level1RegionalNumber</td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

<table>
<thead>
<tr>
<th>Clusterwide Parameters (Feature - Call Recording)</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Replay Recording Notification Tone To Observed Target</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Replay Recording Tone To Connected Parties</td>
<td>False</td>
<td>False</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Clusterwide Parameters (Feature - Monitoring)</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Replay Monitoring Notification Tone To Observed Target</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Replay Monitoring Tone To Connected Parties</td>
<td>False</td>
<td>False</td>
</tr>
</tbody>
</table>

Figure 13: CUCM Service Parameter cont.
### Clusterwide Parameters (Feature - Join Across Lines And Single Button Barge Feature Set)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Join Across Lines Policy</td>
<td>Off</td>
</tr>
<tr>
<td>Single Button Barge/CSarging Policy</td>
<td>Off</td>
</tr>
<tr>
<td>Allow Barging When Ringing</td>
<td>False</td>
</tr>
</tbody>
</table>

### Clusterwide Parameters (Feature - Secure Tone)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Play Tone to Indicate Secure/Non-Secure Call Status</td>
<td>False</td>
</tr>
</tbody>
</table>

### Clusterwide Parameters (Feature - External Call Control)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>External Call Control Diversion Maximum Hop Count</td>
<td>12</td>
</tr>
<tr>
<td>Maximum External Call Control Diversion Hops to Pattern or DN</td>
<td>12</td>
</tr>
<tr>
<td>External Call Control Routing Request Timer</td>
<td>2000</td>
</tr>
<tr>
<td>External Call Control Fully Qualified Role And Resource</td>
<td>Cisco:UC:UCMPolicy:VoiceOrVideoCall</td>
</tr>
<tr>
<td>External Call Control Initial Connection Count To PDP</td>
<td>2</td>
</tr>
<tr>
<td>External Call Control Maximum Connection Count To PDP</td>
<td>4</td>
</tr>
<tr>
<td>Always use External Call Control-specified Called/Calling Party Names</td>
<td>True</td>
</tr>
</tbody>
</table>

### Clusterwide Parameters (Route Plan)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Stop Routing on Out Of Bandwidth Flag</td>
<td>False</td>
</tr>
<tr>
<td>Stop Routing on Unallocated Number Flag</td>
<td>True</td>
</tr>
<tr>
<td>Stop Routing on User Busy Flag</td>
<td>True</td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

### Clusterwide Parameters (Route Class Signaling)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Class Trunk Signaling Enabled</td>
<td>True</td>
</tr>
<tr>
<td>SIP Route Class Naming Authority</td>
<td>Cisco.com</td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

### Clusterwide Parameters (Hunt List)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Stop Hunting on Out Of Bandwidth Flag</td>
<td>False</td>
</tr>
<tr>
<td>Use Pickup Group Of Line Group Member DN</td>
<td>False</td>
</tr>
</tbody>
</table>

Figure 14: CUCM Service Parameter cont.
<table>
<thead>
<tr>
<th>Clusterwide Parameters (External QoS)</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>External QoS Enabled</strong></td>
<td>False</td>
<td>False</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Clusterwide Parameters (Service)</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Default Network Hold MOH Audio Source ID</strong></td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td><strong>Default User Hold MOH Audio Source ID</strong></td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td><strong>Duplex Streaming Enabled</strong></td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td><strong>Media Exchange Interface Capability Timer</strong></td>
<td>6</td>
<td>8</td>
</tr>
<tr>
<td><strong>Send Multicast MOH in H.245 CLC Message</strong></td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td><strong>Media Exchange Timer</strong></td>
<td>12</td>
<td>12</td>
</tr>
<tr>
<td><strong>Media Exchange Stop Streaming Timer</strong></td>
<td>8</td>
<td>8</td>
</tr>
<tr>
<td><strong>Open Video Channel Response Timer for SIP Interop</strong></td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td><strong>Port Received Timer After Call Connection</strong></td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td><strong>Media Resource Allocation Timer</strong></td>
<td>12</td>
<td>12</td>
</tr>
<tr>
<td><strong>MTU and Transcoder Resource Throttling Percentage</strong></td>
<td>95</td>
<td>95</td>
</tr>
<tr>
<td><strong>Intercluster Capabilities Mismatch Timer</strong></td>
<td>1000</td>
<td>1000</td>
</tr>
<tr>
<td><strong>Silence Suppression</strong></td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td><strong>Silence Suppression for Gateways</strong></td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td><strong>Strip G.729 Annex B (Silence Suppression) from Capabilities</strong></td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td><strong>Enable Source IP Address Verification for Software Media Devices</strong></td>
<td>True</td>
<td>True</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Clusterwide Parameters (System - General)</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Always Use Dead Time Setting</strong></td>
<td>Default</td>
<td>False</td>
</tr>
<tr>
<td><strong>Restart Cisco CallManager on Initialization Exception</strong></td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td><strong>Digit Analysis Timer</strong></td>
<td>6</td>
<td>5</td>
</tr>
<tr>
<td><strong>Statistics Enabled</strong></td>
<td>True</td>
<td>True</td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

<table>
<thead>
<tr>
<th>Clusterwide Parameters (System - QOS)</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Priority Class</strong></td>
<td>Normal Priority</td>
<td>Normal Priority</td>
</tr>
<tr>
<td><strong>DSCP for Audio Calls</strong></td>
<td>46 (101110)</td>
<td>46 (101110)</td>
</tr>
<tr>
<td><strong>DSCP for Video Calls</strong></td>
<td>34 (100010)</td>
<td>34 (100010)</td>
</tr>
<tr>
<td><strong>DSCP for Audio Portion of Video Calls</strong></td>
<td>34 (100010)</td>
<td>34 (100010)</td>
</tr>
</tbody>
</table>

Figure 15: CUCM Service Parameter cont.
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DSCP for TelePresence Calls †</td>
<td>32 (100000)</td>
<td>DSCP for TelePresence Calls</td>
</tr>
<tr>
<td>DSCP for Audio Portion of TelePresence Calls †</td>
<td>32 (100000)</td>
<td></td>
</tr>
<tr>
<td>DSCP for Priority Audio Calls †</td>
<td>45 (101101)</td>
<td>DSCP for Priority Audio Calls</td>
</tr>
<tr>
<td>DSCP for Immediate Audio Calls †</td>
<td>44 (101100)</td>
<td></td>
</tr>
<tr>
<td>DSCP for Flash Audio Calls †</td>
<td>41 (101001)</td>
<td></td>
</tr>
<tr>
<td>DSCP for Flash Override Audio Calls †</td>
<td>42 (101010)</td>
<td>DSCP for Flash Override Audio Calls</td>
</tr>
<tr>
<td>DSCP for Executive Override Audio Calls †</td>
<td>42 (101010)</td>
<td></td>
</tr>
<tr>
<td>DSCP for Priority Video Calls †</td>
<td>39 (100111)</td>
<td></td>
</tr>
<tr>
<td>DSCP for Immediate Video Calls †</td>
<td>37 (100101)</td>
<td></td>
</tr>
<tr>
<td>DSCP for Flash Video Calls †</td>
<td>35 (100011)</td>
<td></td>
</tr>
<tr>
<td>DSCP for Flash Override Video Calls †</td>
<td>33 (100001)</td>
<td></td>
</tr>
<tr>
<td>DSCP for Executive Override Video Calls †</td>
<td>33 (100001)</td>
<td></td>
</tr>
<tr>
<td>DSCP for G.Clean Calls †</td>
<td>46 (101110)</td>
<td>DSCP for G.Clean Calls</td>
</tr>
<tr>
<td>DSCP for Priority G.Clean Calls †</td>
<td>45 (101101)</td>
<td></td>
</tr>
<tr>
<td>DSCP for Immediate G.Clean Calls †</td>
<td>44 (101100)</td>
<td></td>
</tr>
<tr>
<td>DSCP for Flash G.Clean Calls †</td>
<td>41 (101001)</td>
<td></td>
</tr>
<tr>
<td>DSCP for Flash Override G.Clean Calls †</td>
<td>42 (101010)</td>
<td>DSCP for Flash Override G.Clean Calls</td>
</tr>
<tr>
<td>DSCP for Executive Override G.Clean Calls †</td>
<td>42 (101010)</td>
<td></td>
</tr>
<tr>
<td>DSCP for Audio Calls when RSVP Fails †</td>
<td>0 (000000)</td>
<td>DSCP for Audio Calls when RSVP Fails</td>
</tr>
<tr>
<td>DSCP for Video Calls when RSVP Fails †</td>
<td>0 (000000)</td>
<td></td>
</tr>
<tr>
<td>DSCP for UCCP Protocol Links †</td>
<td>24 (011000)</td>
<td></td>
</tr>
</tbody>
</table>

**Figure 16: CUCM Service Parameter cont.**
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Preferred G.723.1 Millisecond Packet Size</td>
<td>30</td>
</tr>
<tr>
<td>Preferred G.729 Millisecond Packet Size</td>
<td>20</td>
</tr>
<tr>
<td>Always Use Preferred G.729 Packet Size for SIP Trunk Answers</td>
<td>False</td>
</tr>
<tr>
<td>Preferred G.729 EFR Bytes Packet Size</td>
<td>31</td>
</tr>
<tr>
<td>G.711 A-law Codec Enabled</td>
<td>Enabled for All Devices</td>
</tr>
<tr>
<td>G.711 mu-law Codec Enabled</td>
<td>Enabled for All Devices</td>
</tr>
<tr>
<td>G.722 Codec Enabled</td>
<td>Enabled for All Devices</td>
</tr>
<tr>
<td>ILRC Codec Enabled</td>
<td>Enabled for All Devices</td>
</tr>
<tr>
<td>ISAC Codec Enabled</td>
<td>Enabled for All Devices</td>
</tr>
<tr>
<td>Default Interregion Max Audio Bit Rate</td>
<td>64 kbps (G.722, G.711)</td>
</tr>
<tr>
<td>Default Interregion Max Audio Bit Rate (Includes Audio)</td>
<td>64 kbps (G.722, G.711)</td>
</tr>
<tr>
<td>Default Interregion Max Video Call Bit Rate</td>
<td>6 kbps (G.729)</td>
</tr>
<tr>
<td>Default Interregion Max Video Call Bit Rate (Includes Audio)</td>
<td>8 kbps (G.729)</td>
</tr>
<tr>
<td>Use Video Bandwidth Pool for Immersive Video Calls</td>
<td>True</td>
</tr>
<tr>
<td>Default Interregion and Interregion Link Loss Type</td>
<td>Low Loss</td>
</tr>
<tr>
<td>Default Audio Codec List between Regions</td>
<td>Factory Default low loss</td>
</tr>
<tr>
<td>Default Audio Codec List within Region</td>
<td>Factory Default low loss</td>
</tr>
<tr>
<td>Accept Audio Codec Preferences in Received Offer</td>
<td>Off</td>
</tr>
<tr>
<td>G.Cleart Bandwidth Override</td>
<td>False</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Clusterwide Parameters (System - CCM Automated Alternate Routing)</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Automated Alternate Routing Enable</td>
<td>False</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Clusterwide Parameters (System - RSVP)</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default inter-locatization RSVP Policy</td>
<td>No Reservation</td>
</tr>
<tr>
<td>RSVP Retry Timer</td>
<td>60</td>
</tr>
<tr>
<td>Mandatory RSVP Mid-call Retry Counter</td>
<td>1</td>
</tr>
<tr>
<td>Mandatory RSVP mid-call error handle option</td>
<td>Call becomes best effort</td>
</tr>
<tr>
<td>RSVP Video Traffic Burst Size factor</td>
<td>5</td>
</tr>
<tr>
<td>IMPP EXECUTIVE OVERRIDE To RSVP Priority Mapping</td>
<td>55535</td>
</tr>
</tbody>
</table>

Figure 17: CUCM Service Parameter cont.
**Figure 18: CUCM Service Parameter cont.**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>MLPP FLASH OVERRIDE To RSVP Priority Mapping</td>
<td>65534</td>
</tr>
<tr>
<td>MLPP FLASH To RSVP Priority Mapping</td>
<td>65533</td>
</tr>
<tr>
<td>MLPP IMMEDIATE To RSVP Priority Mapping</td>
<td>65532</td>
</tr>
<tr>
<td>MLPP PL_PRIORITY To RSVP Priority Mapping</td>
<td>65531</td>
</tr>
<tr>
<td>MLPP PL_ROUTINE To RSVP Priority Mapping</td>
<td>65530</td>
</tr>
<tr>
<td>RSVP Audio Application ID</td>
<td>AudioStream</td>
</tr>
<tr>
<td>RSVP Video Application ID</td>
<td>VideoStream</td>
</tr>
<tr>
<td>RSVP Response Timer</td>
<td>2</td>
</tr>
</tbody>
</table>

**TLS Packet Capture Configurations**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet Capture Enable</td>
<td>False</td>
</tr>
<tr>
<td>Packet Capture Max File Size (MB)</td>
<td>2</td>
</tr>
</tbody>
</table>

**Clusterwide Parameters (System - Presence)**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Presence Subscription Throttling Threshold</td>
<td>60000</td>
</tr>
<tr>
<td>Presence Subscription Resume Threshold</td>
<td>80</td>
</tr>
<tr>
<td>Default Inter-Presence Group Subscription</td>
<td>Disallow Subscription</td>
</tr>
<tr>
<td>BLD Status Updates Delay</td>
<td>False</td>
</tr>
</tbody>
</table>

**Clusterwide Parameters (System - Mobility)**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enterprise Feature Access Code for Hold</td>
<td>81</td>
</tr>
<tr>
<td>Enterprise Feature Access Code for Exclusive Hold</td>
<td>82</td>
</tr>
<tr>
<td>Enterprise Feature Access Code for Resume</td>
<td>83</td>
</tr>
<tr>
<td>Enterprise Feature Access Code for Transfer</td>
<td>84</td>
</tr>
<tr>
<td>Enterprise Feature Access Code for Conference</td>
<td>85</td>
</tr>
<tr>
<td>Enterprise Feature Access Code for Session Handoff</td>
<td>74</td>
</tr>
<tr>
<td>Enterprise Feature Access Code for Starting Selective Recording</td>
<td>86</td>
</tr>
<tr>
<td>Enterprise Feature Access Code for Stopping Selective Recording</td>
<td>87</td>
</tr>
<tr>
<td>Smart Mobile Phone Interdigit Timer</td>
<td>500</td>
</tr>
<tr>
<td>Non-Smart Mobile Phone Interdigit Timer</td>
<td>2000</td>
</tr>
<tr>
<td>Send Call to Mobile Menu Timer</td>
<td>60</td>
</tr>
<tr>
<td>SIP Dual Mode Alert Timer</td>
<td>1500</td>
</tr>
<tr>
<td>CUCM Service Parameter cont.</td>
<td></td>
</tr>
<tr>
<td>-----------------------------</td>
<td></td>
</tr>
<tr>
<td><strong>Call Screening Timer</strong></td>
<td>4000</td>
</tr>
<tr>
<td><strong>Session Resumption Wait Timer</strong></td>
<td>1800</td>
</tr>
<tr>
<td><strong>Inbound Calling Search Space for Remote Destination</strong></td>
<td>Trunk or Gateway Inbound Calling Search Space</td>
</tr>
<tr>
<td><strong>Enable Enterprise Feature Access</strong></td>
<td>False</td>
</tr>
<tr>
<td><strong>Enable Mobile Voice Access</strong></td>
<td>False</td>
</tr>
<tr>
<td><strong>Mobile Voice Access Number</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Match Caller ID with Remote Destination</strong></td>
<td>Complete Match</td>
</tr>
<tr>
<td><strong>Number of Digits for Caller ID Partial Match</strong></td>
<td>10</td>
</tr>
<tr>
<td><strong>System Remote Access Blocked Numbers</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Enable Use of Called Party Transformed Number for Mobile-terminated Calls</strong></td>
<td>False</td>
</tr>
<tr>
<td><strong>Honor Gateway or Trunk Outbound Calling Party Selection for Mobile Connect Calls</strong></td>
<td>False</td>
</tr>
</tbody>
</table>

**Clusterwide Parameters (System - Mobility Single Number Reach Voicemail)**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Single Number Reach Voicemail Policy</td>
<td>Timer Control</td>
<td></td>
</tr>
<tr>
<td>Dial Via-Office Reverse Voicemail Policy</td>
<td>Timer Control</td>
<td></td>
</tr>
<tr>
<td>User Control Delayed Announcement Timer</td>
<td>10000</td>
<td></td>
</tr>
<tr>
<td>User Control Confirmed Answer Indication Timer</td>
<td>10000</td>
<td></td>
</tr>
</tbody>
</table>

**Clusterwide Parameters (Feature - Reroute Remote Destination Calls to Enterprise Number)**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Reroute Remote Destination Calls to Enterprise Number</td>
<td>False</td>
<td></td>
</tr>
<tr>
<td>Ring All Shared Lines</td>
<td>False</td>
<td></td>
</tr>
<tr>
<td>Ignore Call Forward All on Enterprise DN</td>
<td>True</td>
<td></td>
</tr>
</tbody>
</table>

**Clusterwide Parameters (Feature - Immediate Divert)**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use Legacy Immediate Divert</td>
<td>True</td>
<td></td>
</tr>
<tr>
<td>Allow OFIC during Divert</td>
<td>False</td>
<td></td>
</tr>
<tr>
<td>Immediate Divert User Response Timer</td>
<td>5</td>
<td></td>
</tr>
</tbody>
</table>

**Clusterwide Parameters (Call Admission Control)**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Counting CAC Enabled</td>
<td>False</td>
<td></td>
</tr>
<tr>
<td>Audio Bandwidth For Call Counting CAC</td>
<td>102</td>
<td></td>
</tr>
<tr>
<td>Video Bandwidth For Call Counting CAC</td>
<td>500</td>
<td></td>
</tr>
<tr>
<td>UCM to LBN Periodic Reservation Refresh Timer</td>
<td>5</td>
<td></td>
</tr>
<tr>
<td>Maximum Bandwidth Deduction Duration</td>
<td>720</td>
<td></td>
</tr>
<tr>
<td>Call Treatment When No LBN Available</td>
<td>Allow Calls</td>
<td></td>
</tr>
<tr>
<td>Locations Media Resource Audio Bit Rate Policy</td>
<td>Lowest Bit Rate</td>
<td></td>
</tr>
<tr>
<td>Video Call QoS Marking Policy</td>
<td>Default</td>
<td></td>
</tr>
</tbody>
</table>

**Clusterwide Parameters (Emergency Calling for Require Off-premise Location)**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Alternate Destination for Emergency Call</td>
<td>None</td>
<td></td>
</tr>
<tr>
<td>Alternate Calling Search Space for Emergency Call</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

---

Figure 19: CUCM Service Parameter cont.
Audio Codec Preference List

Navigate to System > Region Information > Audio Codec Preference List to create Codec Preference List, for this test, G729 preferred is configured.

<table>
<thead>
<tr>
<th>Codec in List</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.729a 8k</td>
</tr>
<tr>
<td>G.711 A-Law 64k</td>
</tr>
<tr>
<td>AMR 8k (7.4 k-24k)</td>
</tr>
<tr>
<td>AMR 16k  (1.28 k-4.96 k)</td>
</tr>
<tr>
<td>MP4A-LATM 128k</td>
</tr>
<tr>
<td>MP4A-LATM 56k</td>
</tr>
<tr>
<td>G.726 16k</td>
</tr>
<tr>
<td>G.722 64k</td>
</tr>
<tr>
<td>G.722 1.32k</td>
</tr>
<tr>
<td>G.722 56k</td>
</tr>
<tr>
<td>G.722 1.24k</td>
</tr>
<tr>
<td>G.722 48k</td>
</tr>
<tr>
<td>MP4A-LATM 24k</td>
</tr>
<tr>
<td>G.711 U-Law 64k</td>
</tr>
<tr>
<td>G.711 U-Law 56k</td>
</tr>
<tr>
<td>G.711 A-Law 56k</td>
</tr>
<tr>
<td>ILBC 16k</td>
</tr>
<tr>
<td>G.728 16k</td>
</tr>
<tr>
<td>GSM Enhanced Full Rate 13k</td>
</tr>
<tr>
<td>GSM Full Rate 13k</td>
</tr>
<tr>
<td>G.729a 8k</td>
</tr>
<tr>
<td>G.729a 8k</td>
</tr>
<tr>
<td>G.728 8k</td>
</tr>
<tr>
<td>GSM Half Rate 6.4k</td>
</tr>
<tr>
<td>G.723.1 7k</td>
</tr>
</tbody>
</table>

Figure 20 Audio Codec Preference List
Region

Navigate to System > Region Information > Region to create Region. G729 Region was created in our test.

<table>
<thead>
<tr>
<th>Region Configuration</th>
<th>Related Links: Back To Find/List Go</th>
</tr>
</thead>
<tbody>
<tr>
<td>Save</td>
<td>Delete</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Region Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name: Vodafone_Region</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Region Relationships</th>
</tr>
</thead>
<tbody>
<tr>
<td>Region</td>
</tr>
<tr>
<td>---------</td>
</tr>
<tr>
<td>Vodafone_Region</td>
</tr>
</tbody>
</table>

NOTE: Regions not displayed
Use System Default | Use System Default | Use System Default | Use System Default |

<table>
<thead>
<tr>
<th>Modify Relationship to other Regions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Regions</td>
</tr>
<tr>
<td>---------</td>
</tr>
<tr>
<td>Default</td>
</tr>
</tbody>
</table>

Figure 21 CUCM Region
Device Pool

Navigate to System > Device Pool and click on Add New to create the Device Pool. In our test, we created G729 Device Pool associated with newly added G729 Region based on the preferred codecs in the Audio Codec Preference List. All SIP/SCCP phones, SIP trunks to VG224 and the SIP trunk to CUBE are assigned to G729 device pool.

Figure 22 CUCM Device Pool
### Call Routing Information

**Incoming Calling Party Settings**

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

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Default Prefix Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td></td>
<td></td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Incoming Called Party Settings**

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

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Default Prefix Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td>0</td>
<td></td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td>0</td>
<td></td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>0</td>
<td></td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td>0</td>
<td></td>
</tr>
</tbody>
</table>

### Phone Settings

**Caller ID For Calls From This Phone**

Calling Party Transformation CSS: < None >

**Connected Party Settings**

Connected Party Transformation CSS: < None >

**Redirecting Party Settings**

Redirecting Party Transformation CSS: < None >

**Figure 23: CUCM Device Pool cont.**
Conference Bridge Configuration

Go to Media Resources > Conference Bridge and click on Add New, my-Conf conference bridge is used in this configuration.

Figure 24: Conference Bridge Configuration
Media Resource Group Configuration

Go to Media Resources > Media Resource Group Configuration and click on Add New. Select MOH, MTP media resources and the conference bridge configured in Conference Bridge Configuration.

Figure 25: Media Resource Group Configuraiton
Media Resource Group List Configuration

Go to Media Resources > Media Resource Group Configuration and click on Add New. Select the media resource group configured in Media Resource Group Configuration.

**Figure 26: Media Resource Group List Configuration**

<table>
<thead>
<tr>
<th>Media Resource Group List Configuration</th>
<th>Related Links:</th>
<th>Status: Ready</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Media Resource Group List Status</strong></td>
<td></td>
<td>Media Resource Group List: Vodafone_MRGL (used by 11 devices)</td>
</tr>
<tr>
<td><strong>Media Resource Group List Information</strong></td>
<td>Name: Vodafone_MRGL</td>
<td></td>
</tr>
<tr>
<td><strong>Media Resource Groups for this List</strong></td>
<td>Available Media Resource Groups</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Selected Media Resource Groups</td>
<td>Vodafone_MRGL</td>
</tr>
</tbody>
</table>

**Figure 26: Media Resource Group List Configuration**
Off-net calls via Vodafone's network

Off-net calls are served by SIP trunks configured between CUCM and SBCs in Vodafone's network. Calls are routed via CUBE.

SIP Trunk Security Profile

Go to System > Security > SIP Trunk Security Profile and click on Add New. Vodafone_ Non Secure SIP Trunk Profile is used in this test.

Figure 27 SIP Trunk Security Profile
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Transport Type</td>
<td>TCP + UDP</td>
<td></td>
</tr>
<tr>
<td>Outgoing Transport Type</td>
<td>UDP</td>
<td>SIP trunks to Vodafone SBC should use UDP as a transport protocol for SIP. This is configured using SIP Trunk Security profile, which is later assigned to the SIP trunk itself.</td>
</tr>
<tr>
<td>Incoming Port</td>
<td>5060</td>
<td>5060 is the default fpirt for TCP/UDP</td>
</tr>
</tbody>
</table>
SIP Profile

Navigate to Device > Device Settings > SIP Profile and modify default SIP Profile by clicking on a **Copy** button. Vodafone_SIP_PROFILE was created in this test and will be later associated with the SIP trunk.

![SIP Profile Configuration Table]

---

Figure 28: CUCM SIP Profile

---

© 2014 Cisco Systems, Inc. All rights reserved.

Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Page 75 of 97
Figure 29: CUCM SIP Profile cont.
### Incoming Requests FROM URI Settings

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

### Trunk Specific Configuration

- **Reroute Incoming Request to new Trunk based on**
  - Never

- **RSVP Over SIP**
  - Local RSVP

- **Resource Priority Namespace List**
  - < None >

- **Fall back to local RSVP**

- **SIP Re1XX Options**
  - Send PRACK if 1xx Contains SDP

- **Video Call Traffic Class**
  - Mixed

- **Calling Line Identification Presentation**
  - Default

- **Session Refresh Method**
  - Invite

- **Early Offer support for voice and video calls**
  - Mandatory [insert RTP if needed]

- **Enable AIMAT**

- **Deliver Conference Bridge Identifier**

- **Allow Pass-through of Configured Line Device Caller Information**

- **Reject Anonymous Incoming Calls**

- **Reject Anonymous Outgoing Calls**

- **Send ILS Learned Destination Route String**

### SIP OPTIONS Ping

- **Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"**

- **Ping Interval for In-service and Partially In-service Trunks (seconds)**
  - 60

- **Ping Interval for Out-of-service Trunks (seconds)**
  - 120

- **Ping Retry Timer (milliseconds)**
  - 500

- **Ping Retry Count**
  - 6

### SDP Information

- **Send send-receive SDP in mid-call INVITE**

- **Allow Presentation Sharing using BFCP**

- **Allow bx Application Media**

- **Allow multiple codecs in answer SDP**

---

**Figure 30: CUCM SIP Profile cont.**
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default MTP Telephony Event Payload Type</td>
<td>101</td>
<td>RFC2833 DTMF payload type</td>
</tr>
<tr>
<td>SIP Rel1XX Options</td>
<td>Send PRACK if 1xx Contains SDP</td>
<td>Enable Provisional Acknowledgements (Reliable 100 messages)</td>
</tr>
<tr>
<td>Early Offer support for voice and video calls</td>
<td>Mandatory (insert MTP if needed)</td>
<td>To create a trunk that supports early offer</td>
</tr>
<tr>
<td>Enable OPTIONS Ping to monitor destination status for Trunks with Service Type &quot;None(Default)&quot;</td>
<td>Checked</td>
<td></td>
</tr>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)</td>
<td>60</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
<tr>
<td>Ping Interval for Out-of-service Trunks (seconds)</td>
<td>120</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
</tbody>
</table>
SIP Trunk Configuration

Create SIP trunks to Vodafone by navigating to Device > Trunk and clicking Add New button. Same apply to create SIP trunks to Cisco Analog gateway VG224.

![SIP Trunk Configuration](image)

Figure 31: SIP Trunk to Vodafone via CUBE
Figure 32: SIP Trunk to Vodafone via CUBE cont.
Figure 33: SIP Trunk to Vodafone via CUBE cont.
Figure 34: SIP Trunk to Vodafone via CUBE cont.
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Name</td>
<td>Vodafone_SIP_TRUNK</td>
<td>Name for the trunk</td>
</tr>
<tr>
<td>Device Pool</td>
<td>Vodafone_Devicepool</td>
<td>G729 Device Pool is used for this trunk</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>Vodafone_MRGL</td>
<td>MRG with resources: ANN, CFB, MOH and MTP</td>
</tr>
<tr>
<td>Significant Digits</td>
<td>4</td>
<td>4 digits Extension for all CPE phones</td>
</tr>
<tr>
<td>Redirecting Diversion Header Delivery - outbound</td>
<td>Checked</td>
<td>Adding Diversion Header for redirecting calls outbound from site</td>
</tr>
<tr>
<td>Destination Address</td>
<td>10.80.10.11</td>
<td>Virtual LAN IP address of the CUBE HA</td>
</tr>
<tr>
<td>Destination Port</td>
<td>5060</td>
<td>Port 5060 is default for TCP/UDP</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
<td>Vodafone_Non Secure SIP Trunk Profile</td>
<td>SIP Trunk Security Profile configured earlier</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>Vodafone_SIP_PROFILE</td>
<td>SIP Profile configured earlier</td>
</tr>
</tbody>
</table>

Reset the trunk after the configuration is completed. Apply same procedure to create SIP trunks to Cisco Analog Gateway VG224.
Dialplan

CUBE translates 10 digits dialed number from Vodafone's network to 4 digits extension. Hence there is no need to configure Translation Pattern under Call Routing. For outgoing call from Cisco CUCM a Route Pattern must be configured. Cisco IP phones dial 9+10/11 digits PSTN number to access PSTN via CUBE. “9” is removed before being sent to CUBE; the same applies to outbound fax call from VG224. For incoming fax call to VG224, Route Pattern 2095 is configured.

Navigate to Call Routing > Route/Hunt > Route Pattern and press Add New button to create Route Patterns.
Figure 35 Route Pattern cont.
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>9001XXXXXXXXXXXX</td>
<td>Specify appropriate Route Pattern</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>Vodafone_SIP_TRUNK</td>
<td>SIP Trunk name configured earlier</td>
</tr>
<tr>
<td>Calling Line ID</td>
<td>Default</td>
<td>Set to Restricted for Anonymous Calls</td>
</tr>
<tr>
<td>Calling Line ID</td>
<td>Default</td>
<td>Set to Restricted for Anonymous Calls</td>
</tr>
<tr>
<td>Called Party Transform</td>
<td>001XXXXXXXXXXXX</td>
<td>Strip ‘9’ on the called party number before sending it out to the CUBE</td>
</tr>
</tbody>
</table>
Cisco Analog Gateway VG224 configuration
Router#show run
Building configuration...
Current configuration : 2908 bytes
!
! Last configuration change at 19:49:05 UTC Sun Mar 7 1993 by cisco
!
version 15.1
no service pad
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
!
boot-start-marker
boot-end-marker
!
!
enable secret 5 $1$2vXb$mom3hjaQF.cY7CZ0YP3Oo.
!
no aaa new-model
crypto pki token default removal timeout 0
!
!
ip source-route
ip cef
!
!
no ipv6 cef
!
!
voice service voip
allow-connections sip to sip
redirect ip2ip
fax protocol t38 version 0 ls-redundancy 1 hs-redundancy 1 fallback pass-through g711alaw
sip
asserted-id pai
early-offer forced
midcall-signaling passthru
!
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711alaw
!
voice class codec 2
codec preference 1 g711alaw
codec preference 2 g729r8
!
!
voice-card 0
!
username cisco privilege 15 password 0 tekV1z10n
!
!
interface FastEthernet0/0
ip address 10.80.10.15 255.255.255.0
duplex auto
speed auto
!
interface FastEthernet0/1
no ip address
shutdown
duplex auto
speed auto
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 10.80.10.1
! control-plane
!

voice-port 2/0
!

voice-port 2/1
  ring frequency 50
cptone DE
description **telephone analog/fax**
  station-id number 2095
caller-id enable
!
voice-port 2/2
!
voice-port 2/3
!
voice-port 2/4
!
voice-port 2/5
!
voice-port 2/6
!
voice-port 2/7
!
voice-port 2/8
!
voice-port 2/9
!
voice-port 2/10
!
voice-port 2/11
!
voice-port 2/12
!
voice-port 2/13
| voice-port 2/14 |
| voice-port 2/15 |
| voice-port 2/16 |
| voice-port 2/17 |
| voice-port 2/18 |
| voice-port 2/19 |
| voice-port 2/20 |
| voice-port 2/21 |
| voice-port 2/22 |
| voice-port 2/23 |

no ccm-manager fax protocol cisco

no mgcp package-capability fxr-package
no mgcp timer receive-rtcp

mgcp profile default

dial-peer voice 100 voip
description outbound call
destination-pattern 90T
session protocol sipv2
session target ipv4:10.80.10.3:5060
session transport udp
voice-class codec 2
voice-class sip asserted-id pai
dtmf-relay rtp-nte
fax rate 14400
no vad
!
dial-peer voice 2093 pots
  service stcapp
  shutdown
  no digit-strip
  port 2/1
  forward-digits 0
  !
dial-peer voice 2095 voip
  description **Incoming Call from SIP Trunk**
  shutdown
  session protocol sipv2
  session target ipv4:10.80.10.3:5060
  session transport udp
  !
dial-peer voice 101 pots
  destination-pattern 2...
  incoming called-number [0-9]T
  no digit-strip
  port 2/1
  !
dial-peer voice 103 voip
  description outbound call
  destination-pattern 2091
  session protocol sipv2
  session target ipv4:10.80.10.3:5060
  session transport udp
  voice-class codec 1
  voice-class sip asserted-id pai
dtmf-relay rtp-nte
fax rate 14400
no vad
! 
line con 0
  speed 115200
line aux 0
line vty 0 4
  session-timeout 900
exec-timeout 960 0
login local
  transport input all
!
end
**Acronyms**

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>MGCP</td>
<td>Media Gateway Control Protocol</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>CUCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>CUBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>HSRP</td>
<td>Hot Standby Router Protocol</td>
</tr>
<tr>
<td>IPC</td>
<td>Inter-Device Communication Protocol</td>
</tr>
<tr>
<td>STCP</td>
<td>Stream Control Transmission Protocol</td>
</tr>
<tr>
<td>FQDN</td>
<td>Fully Qualified Domain Name</td>
</tr>
<tr>
<td>SBC</td>
<td>Session Border Controller</td>
</tr>
<tr>
<td>TCP</td>
<td>Transmission Control Protocol</td>
</tr>
<tr>
<td>UDP</td>
<td>User Datagram Protocol</td>
</tr>
<tr>
<td>DTMF</td>
<td>Dual–Tone Multi-Frequency signaling</td>
</tr>
<tr>
<td>HA</td>
<td>High Availability</td>
</tr>
<tr>
<td>ANN</td>
<td>Annunciator</td>
</tr>
<tr>
<td>CFB</td>
<td>Conference Bridge</td>
</tr>
<tr>
<td>MOH</td>
<td>Music on Hold</td>
</tr>
<tr>
<td>MTP</td>
<td>Media Termination Points</td>
</tr>
<tr>
<td>MRG</td>
<td>Media Resource Group</td>
</tr>
</tbody>
</table>
Important Information

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS.

IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.
Test Results

SP_SIP_master_testplan_V1.xlsx