



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

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Note: Testing was conducted in Tekvizion labs



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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

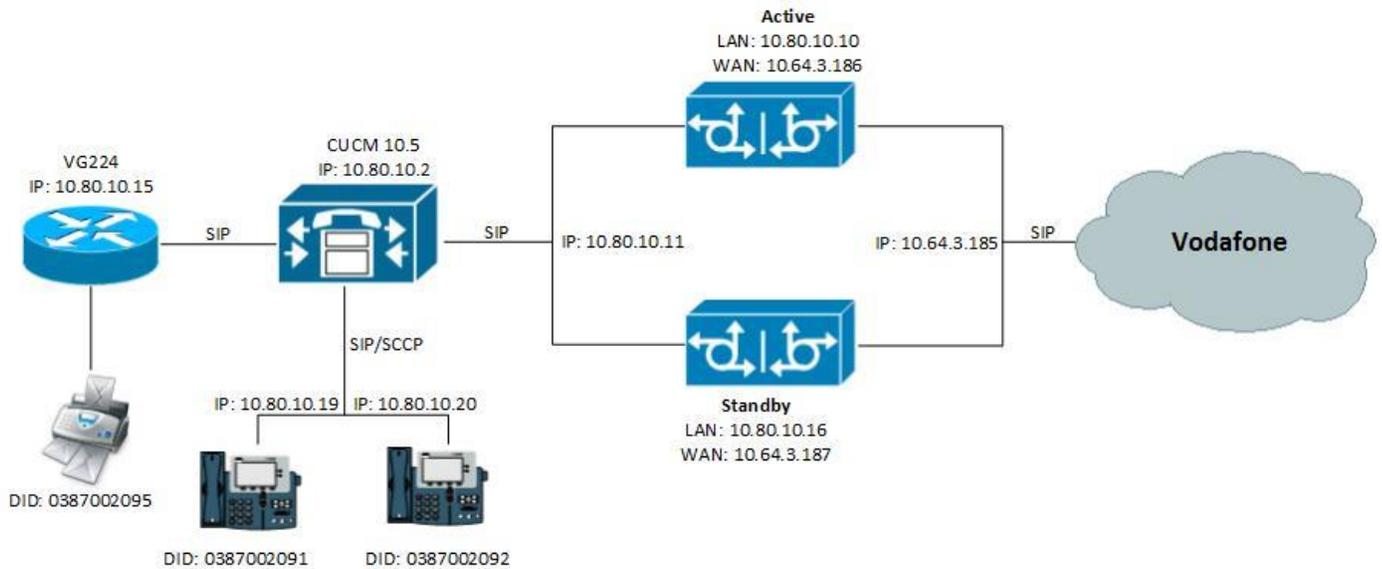


Figure 1 Network Topology

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](#).

Active CUBE

```
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
```

Standby CUBE

```
ipc zone default
association 1
  no shutdown
  protocol sctp
  local-port 5000
  local-ip 10.80.10.16
  remote-port 5000
  remote-ip 10.80.10.10
```

Explanation

Command	Description
ipc zone default	Configures the Inter-Device Communication Protocol(IPC) and enters IPC zone configuration mode
Association 1	Configures an association between the 2 routers
No shutdown	Restarts a disabled association and its associated transport protocol
Protocol sctp	Configure Stream Control Transmission Protocol(SCTP) as the transport protocol
Local-port <i>port_num</i>	Configures the local SCTP port number to communicate with redundant peer, 5000 must be used.



Local-ip <i>ip_addr</i>	Defines the local router's IP address to use to communicate with redundant peer
Remote-port <i>port_num</i>	Configures the remote SCTP port number, 5000 must be used
Remote-ip <i>ip_addr</i>	Defines remote router's IP address to use to communicate with redundant peer

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



Explanation

Command	Description
Mode border-element	Enable CUBE on both routers
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
redundancy	Enable CUBE redundancy and call checkpointing on both routers
Redundancy inter-device	Enable HSRP
Scheme standby SB	Enable standby(HSRP) as redundancy state tracking scheme with group name---SB
Tracking <i>obj_num</i> interface <i>int_id</i> line-protocol	Create a tracking list to track the line-protocol state of an interface <ul style="list-style-type: none">• The <i>obj_num</i> identify the tracked object with range from 1 to 500.• The <i>int_id</i> is the interface being tracked.



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

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

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

Command	Description
translation-rule <i>n</i>	Enters translation rule configuration mode
Rule <i>x</i>	Detail match and replace rule <i>x</i>
translate-outgoing calling <i>n</i>	Specifies the translation rule set to apply to the calling number
translate-outgoing called <i>n</i>	Specifies the translation rule set to apply to the called number

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 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
!
```

!-----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

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



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$IrgJ7sS7zFYWf3bUbPpWy0
```

```
!
```

```
!
```

```
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
```

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```
!  
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 43657274  
    69666963 6174652D 36313133 38393931 301E170D 31343130 33303138 31323336  
    5A170D32 30303130 31303030 3030305A 302F312D 302B0603 55040313 24494F53  
    2D53656C 662D5369 676E6564 2D436572 74696669 63617465 2D363131 33383939  
    3130819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281 8100F1F2  
    7EF0830A 450DE4A0 C5355090 A1A6DDEC 9141BFE4 AF6CD117 008FDE0E E5236D1D  
    0EB78AC6 0EDDFB75 3C796888 75C1C5E0 A45CEC36 6BD5B64C E4366C28 BD4DCE9C  
    32FE38B7 14CFA2B5 C3CC856B 6ECAB1A9 C12DFCE6 70037DFC 72FBC9C0 73987091  
    7207D11F E5DD87B5 3DAF9C73 850EB70D FED2F079 5C65E6D7 AC9F25F0 F9690203  
    010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603 551D2304  
    18301680 14462079 9704B1A7 EB95833B D1989DE9 BB8F876A 2C301D06 03551D0E  
    04160414 46207997 04B1A7EB 95833BD1 989DE9BB 8F876A2C 300D0609 2A864886  
    F70D0101 05050003 818100B8 AD4C2954 2F034EA8 CB1D6132 FC40FC11 1DEEA862
```



```
4F761AE8 30E3A8B5 954CA46B 681B3AEB 849EB483 BD3570EE 7C1C44D9 6430099B
DE03DE91 0C953975 DF46EA7A B3688AFD 4EF0B441 67953D2E 554A0103 CDE31436
E5FEE6D1 80C76BD0 5A6BC79A 25C4586C D61D77C2 F4EBEEE5 298AD702 8085C26D
11E8AD1A 8C64085B 1204EE
```

```
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
```

```
  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 un supp
```

```
    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
!
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
!
!
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
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.



^C

banner login ^C

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>

^C

!

```
line con 0  
  login local  
line aux 0  
line vty 0 4  
  exec-timeout 0 0  
  privilege level 15  
  login local  
  transport input telnet ssh
```



```
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 sctp  
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
    3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
    31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
    69666963 6174652D 33373039 38343635 3238301E 170D3134 30383236 32313335
    35325A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
    4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D33 37303938
    34363532 3830819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
    8100CE51 F561CD41 24990148 0E798600 71068690 366B3A6B A7E16F02 A66F8471
    71E35FA6 C13EBD9D C6887395 683BB37A 27B11487 97EEDF44 0E881127 EC99BC0F
    4B8D3C31 B36459DC FAA585B5 DD209151 8AEDCEA7 847D8ACB 9DEB0523 3818EF93
    B21AD7EB B41CEC57 39FBD6C5 F4BD27E6 6B548ECC 7C85320F 00436C79 F5978280
    44250203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603
    551D2304 18301680 14841E1D 28893357 F087CC1E BBD3BD76 C91253B9 4E301D06
    03551D0E 04160414 841E1D28 893357F0 87CC1EBB D3BD76C9 1253B94E 300D0609
    2A864886 F70D0101 05050003 81810013 876F5E4D 896D48AB B4E92489 B1C42EE6
    60EAC45D BD88C5A7 39EA149E F2576DD3 95177726 7C63256F B1746B16 2A22BEBE
    06DDCB83 0B8A373E 5FE2813D 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
```

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```
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  
!  
!  
voice-port 0/2/0  
!  
voice-port 0/2/1  
!  
voice-port 0/2/2  
!  
voice-port 0/2/3  
!  
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  
!  
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  
!  
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
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
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
```



Configuring the Cisco Unified Communications Manager

Cisco Unified Communications Manager version

The screenshot shows the Cisco Unified CM Administration web interface. At the top left is the Cisco logo. The main header reads "Cisco Unified CM Administration" and "For Cisco Unified Communications Solutions". On the right, there is a navigation menu with "Cisco Unified CM Administration" selected and a "Go" button. Below the header is a navigation bar with "administrator" and links for "Search Documentation", "About", and "Logout". A secondary navigation bar contains dropdown menus for "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The main content area features a large blue banner with the title "Cisco Unified CM Administration" and the text "System version: 10.5.1.10000-7" highlighted with a red box. Below this, it states "VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz, disk 1: 80Gbytes, 4096Mbytes RAM, Partitions aligned". To the right of the banner is a photograph of a server room aisle. At the bottom left of the interface, it says "Last Successful Logon: Tuesday, November 18, 2014 7:26:17 AM CST".

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.

The screenshot shows the Cisco Unified CM Administration interface. The page title is "Service Parameter Configuration" for "Cisco CallManager (Active)" on server "clus20pub--CUCM Voice/Video (Active)". The status is "Ready".

Select Server and Service

Server*: clus20pub--CUCM Voice/Video (Active) (selected)
 Service*: Cisco CallManager (Active) (selected)

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

Cisco CallManager (Active) Parameters on server clus20pub--CUCM Voice/Video (Active)

Parameter Name	Parameter Value	Suggested Value
Call Throttling		
Code Yellow Entry Latency *	20	20
Code Yellow Exit Latency Calculation *	40	40
Code Yellow Duration *	5	5
Max Events Allowed *	2000	2000
System Throttle Sample Size *	10	10
Memory Throttling		
Enable Memory Throttling *	True	True
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		
System		
CDR Enabled Flag *	False	False
CDR Log Calls with Zero Duration Flag *	False	False
Digit Analysis Complexity *	StandardAnalysis	StandardAnalysis
Database Debounce Timer *	0	0
Maximum Phone Fallback Queue Depth *	10	10
Maximum Number of Registered Devices *	5000	5000
System Initialization Timer *	60	60
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		

Figure 3: CUCM Service Parameter

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SDL Trace		
SDL Trace Data Flags *	<input type="text" value="0x00000111"/>	0x00000111
SDL Trace Flush Immediately *	<input type="checkbox" value="False"/>	False
SDL Trace Data Size *	<input type="text" value="0"/>	0
SDL Trace Flag *	<input type="checkbox" value="True"/>	True
SDL TraceType Flags *	<input type="text" value="0x8000EB15"/>	0x8000EB15
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		
Clusterwide Parameters (Device - General)		
Call Diagnostics Enabled *	<input type="checkbox" value="Disabled"/>	Disabled
Show Line Group Member DN in finalCalledPartyNumber CDR Field *	<input type="checkbox" value="False"/>	False
Show Line Group Member Non Masked DN in finalCalledPartyNumber CDR Field *	<input type="checkbox" value="False"/>	False
CTI New Call Accept Timer *	<input type="text" value="4"/>	4
CTI Generate Digits Interval *	<input type="text" value="250"/>	250
CTI Dial Digits Interval *	<input type="text" value="250"/>	250
CTI Await Further Digits *	<input type="checkbox" value="False"/>	False
CTI Use Wildcard Pattern as calledPartyDN *	<input type="checkbox" value="False"/>	False
Retain Media on Disconnect with PI for Active Call *	<input type="checkbox" value="False"/>	False
Station and Backup Server KeepAlive Interval *	<input type="text" value="60"/>	60
Station KeepAlive Interval *	<input type="text" value="30"/>	30
Status Enquiry Poll Flag *	<input type="checkbox" value="False"/>	False
Strip # Sign from Called Party Number *	<input type="checkbox" value="True"/>	True
Session Handoff Alerting Timer *	<input type="text" value="10"/>	10
T301 Timer *	<input type="text" value="180000"/>	180000
T302 Timer *	<input type="text" value="15000"/>	15000
T303 Timer *	<input type="text" value="4000"/>	4000
T304 Timer *	<input type="text" value="30000"/>	30000
T305 Timer *	<input type="text" value="30000"/>	30000
T306 Timer *	<input type="text" value="30000"/>	30000
T308 Timer *	<input type="text" value="4000"/>	4000
T309 Timer *	<input type="text" value="90000"/>	90000
T310 Timer *	<input type="text" value="60000"/>	60000

Figure 4: CUCM Service Parameter cont.



T313 Timer *	4000	4000
T316 Timer *	120000	120000
T317 Timer *	100000	100000
T321 Timer *	30000	30000
T322 Timer *	4000	4000
Tone on Hold Timer *	10	10
Unknown Caller ID Flag *	True	True
Call Classification *	OffNet	OffNet
Always Display Original Dialed Number *	False	False
Name Display for Original Dialed Number When Translated *	Show the Display Name for Original Dialed Number ev	Show the Display Name for Original Dialed Number even if Translated
Always Use PIs With Original Dialed Number *	False	False
Fail Call If Trusted Relay Point Allocation Fails *	True	True
Display Calling/Called ID When PI is Not Available *	False	False
Enable Transit Counter Processing on QSIG Trunks *	False	False
Egress Facility IE Count *	6	6

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

Clusterwide Parameters (Device - Phone)		
Always Use Prime Line *	False	False
Always Use Prime Line for Voice Message *	False	False
Builtin Bridge Enable *	Off	Off
Device Mobility Mode *	Off	Off
Display Device Mobility Location During Phone Registration *	True	True
Auto Answer Timer *	1	1
Extension Display on Cisco IP Phone Model 7910 *	False	False
Alternate Idle Phone Auto-Answer Behavior Enabled *	False	False
Hold Type *	False	False
Line State Update Enabled *	True	True
Off-hook to First Digit Timer *	15000	15000
Override Auto Answer If Speaker Is Disabled *	True	True
Out-of-Bandwidth Text *	Not Enough Bandwidth	Not Enough Bandwidth
Forced Authorization Code Prompt Text *	Enter Authorization Code	Enter Authorization Code

Figure 5: CUCM Service Parameter cont.



Client Matter Code Prompt Text *	Enter Client Matter Code	Enter Client Matter Code
AAR Network Congestion Rerouting Text *	Network Congestion, Rerouting.	Network Congestion, Rerouting.
Ring Setting of Busy Station Policy *	Only Apply Ring Setting of Busy Station When Incomir	Only Apply Ring Setting of Busy Station When Incoming Call Arrives
Transfer On-hook Enabled *	False	False
Ring Setting of Busy Station *	Beep Only	Beep Only
Ring Setting of Idle Station *	Ring	Ring
Call Pickup Group Audio Alert Setting of Idle Station *	Ring Once	Ring Once
Call Pickup Group Audio Alert Setting of Busy Station *	Beep Only	Beep Only
BLF Pickup Audio Alert Setting of Idle Station *	Disable	Disable
BLF Pickup Audio Alert Setting of Busy Station *	Disable	Disable
Privacy Setting *	True	True
Enforce Privacy Setting on Held Calls *	False	False
SIP Station KeepAlive Interval *	120	120
SIP Station Realm *	ccmsipline	ccmsipline
Hunt Group Logoff Notification *	None	None
Speed Dial Await Further Digits *	False	False
Display CTI Route Point Name or DN *	False	False
Display Original Calling Number on Transfer from Cisco Unity *	False	False
URI Dialing Display Preference *	DN	DN
Insert Hyphens in 12-Digit Numbers *	False	False
Allow Call Waiting During an In-Progress Outbound Analog Call *	True	True

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

Clusterwide Parameters (Device - PRI and MGCP Gateway)		
Calling Party Number Screening Indicator *	CallManager sets the screening indicator value - Defat	CallManager sets the screening indicator value - Default setting
Enable Outbound NetworkTrunk CallingParty Restriction *	False	False
Clear Calls Flag When Datalink Is Down *	True	True
Device Status Poll Interval *	3000	3000
Disable Alerting Progress Indicator *	False	False
Discard Non Inband Progress in Overlap Sending *	False	False

Figure 6: CUCM Service Parameter cont.



Disable Resume from Shared-line MGCP FXS Port *	True	True
DTMF Silence Tone Flag *	False	False
Enable Display IE in Codeset 6 *	False	False
Enable Sending PRI NI2 Service Message *	False	False
Flash Hook Duration *	500	500
Gateway Poll Timer *	10	10
Location In PRI Progress Indicator IE (User Side Only) *	Use the Network Side PRI progress indicator IE	Use the Network Side PRI progress indicator IE
Matching Calling Party with Attendant Flag *	False	False
MGCP Database Query Delay Timer *	1000	1000
MGCP FXS On-Hook Pending Timer *	3	3
MGCP Response Timer *	30	30
MGCP Timer *	3	3
Numbering Plan Info *	1	1
Overlap Receiving Flag for PRI *	True	True
Outgoing Media Connect Time for PRI *	Connect ASAP	Connect ASAP
Port Release Timer *	0	0
SMDI Call Delay Timer *	0	0
Stable in State 4 Flag *	False	False
Optimize MGCP Registration *	True	True
Suppress Out-of-Channels Alarms *	True	True
I-Frame Timer *	2000	2000
User-to-User IE Status *	False	False
Convert European Progress Message to Alerting *	False	False
Enable DMS PRI Notify Message from User to Network *	True	True
Audit OOS Channels Interval *	10	10
Digital and Analog Ports Enabled *	True	True
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		
Clusterwide Parameters (Device - H323)		
Accept Unknown TCP Connection *	False	False
BRQ Enabled *	False	False

Figure 7: CUCM Service Parameter cont.



Call Present Disconnect Flag *	False	False
Check Progress Indicator Before Establishing Media *	False	False
H225 Block Setup Destination *	False	False
H225 DB Retry Timer *	0	0
H225 Device Connect Timer *	0	0
H225 DTMF Duration *	100	100
H225 TspReq Retry *	2	2
H225 Intercluster Call Throttle Timer *	30	30
H225 T301 Timer *	180000	180000
H225 T302 Timer *	15000	15000
H225 T303 Timer *	4000	4000
H225 T304 Timer *	30000	30000
H225 T305 Timer *	30000	30000
H225 T310 Timer *	60000	60000
H225 TCP Timer *	5	5
H245 TCS Timeout *	10	10
H323 Calling Party Number Screening Indicator *	Calling number screened and passed	Calling number screened and passed
Apply External Phone Number Mask for H.323 Calls *	False	False
Tone on Connect *	False	False
Wait Time for SDP with SR/RO Mode *	3	3
RAS ARQ Timer *	3	3
RAS BRQ Timer *	3	3
RAS DRQ Timer *	3	3
RAS RRQ Timer *	3	3
Ras URQ Timer *	3	3
Retry Count for ARQ *	2	2
Retry Count for BRQ *	2	2
Retry Count for DRQ *	2	2
Retry Count for RRQ *	2	2
Retry Count for URQ *	1	1
Send Product ID and Version ID *	False	False

Figure 8: CUCM Service Parameter cont.



Send Unified CM Version as Version ID in H225Setup *	False	False
Send Progress Timer *	3000	3000
Send H225 User Info Message *	User Info for Call Progress Tone	User Info for Call Progress Tone
Status Enquiry Poll Timer *	10000	10000
Device Name of GK-controlled Trunk That Will Use Port 1720 *	None	None
Host Name/IP Address of GK That Will Use RAS UDP Port 1719 *	None	None
Fail Call If MTP Allocation Fails *	False	False
Overlap Receiving Flag for H323 *	False	False
Allocate Transcoder for H.323 on Early Offer SIP Trunk for Calls with Early Media *	False	False

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

Clusterwide Parameters (Device - SIP)		
SIP Interoperability Enabled *	True	True
Retry Count for SIP Bye *	10	10
Retry Count for SIP Cancel *	10	10
Retry Count for SIP Invite *	6	6
Retry Count for SIP PRACK *	6	6
Retry Count for SIP Rel1XX *	10	10
Retry Count for SIP Publish *	6	6
Retry Count for SIP Response *	6	6
SIP Connect Timer *	500	500
SIP Disconnect Timer *	500	500
SIP Expires Timer *	180000	180000
SIP PRACK Timer *	500	500
SIP Rel1XX Timer *	500	500
SIP Trying Timer *	500	500
SIP Publish Timer *	500	500
SIP Min-SE Value *	90	1800
SIPS URI Handling *	Reject	Reject
SIP statistics Periodic update Timer *	2	2
SIP Session Expires Timer *	1800	1800

Figure 9: CUCM Service Parameter cont.



SIP Trunk TspReq Retry *	<input type="text" value="2"/>	2
SIP TCP Unused Connection Timer *	<input type="text" value="14"/>	14
SIP TCP Timer *	<input type="text" value="5"/>	5
SIP Station TCP Port Throttle Threshold *	<input type="text" value="100"/>	100
SIP Trunk TCP Port Throttle Threshold *	<input type="text" value="500"/>	500
SIP V.150 Outbound SDP Offer Filtering *	<input type="text" value="No Filtering"/>	▼ No Filtering
Send SIP Multicast TTL in SDP *	<input type="text" value="False"/>	▼ False
Default PUBLISH Expiration Timer *	<input type="text" value="3600"/>	3600
Minimum PUBLISH Expiration Timer *	<input type="text" value="60"/>	60
IM and Presence Publish Trunk	<input type="text" value="< None >"/>	▼
Send 181 Call Is Being Forwarded *	<input type="text" value="False"/>	▼ False
Delay Sending 181 until 180/183 message is received *	<input type="text" value="True"/>	▼ True
Fail Call Over SIP Trunk if MTP Allocation Fails *	<input type="text" value="False"/>	▼ False
Log Call-Related REFER/NOTIFY/SUBSCRIBE SIP Messages for Session Trace *	<input type="text" value="True"/>	▼ True
Port Received Timer for Outbound Call Setup *	<input type="text" value="2"/>	2

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

Clusterwide Parameters (Feature - General)		
Call Park Display Timer *	<input type="text" value="10"/>	10
Caller ID Display Priority Enabled *	<input type="text" value="True"/>	▼ True
Call Park Reversion Timer *	<input type="text" value="60"/>	60
Park Monitoring Reversion Timer *	<input type="text" value="60"/>	60
Park Monitoring Periodic Reversion Timer *	<input type="text" value="30"/>	30
Park Monitoring Forward No Retrieve Timer *	<input type="text" value="300"/>	300
Preserve globalCallId for Parked Calls *	<input type="text" value="True"/>	▼ True
Maximum Call Duration Timer *	<input type="text" value="720"/>	720
Maximum Hold Duration Timer *	<input type="text" value="360"/>	360
Party Entrance Tone *	<input type="text" value="True"/>	▼ True
Message Waiting Lamp Policy *	<input type="text" value="Primary Line - Light and Prompt"/>	▼ Primary Line - Light and Prompt
Audible Message Waiting Indication Policy *	<input type="text" value="OFF"/>	▼ OFF
Message Waiting Indicator Inbound Calling Search Space	<input type="text" value="< None >"/>	▼
Multiple Tenant MWI Modes *	<input type="text" value="False"/>	▼ False

Figure 10: CUCM Service Parameter cont.



MWI Non Message Center Signaling Call Duration *	<input type="text" value="0"/>	0
Message Waiting Indicator APDU Digit Translation CSS	< None >	
Block OffNet To OffNet Transfer *	False	False
Use Original Call Classification for Transferred Calls *	False	False
Use Restriction attribute of ID/Name Presentation of Transferring Party *	True	True
Local route group for redirected calls *	Local route group of calling party	Local route group of calling party
Block Unencrypted Calls *	False	False

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

Clusterwide Parameters (Feature - Conference)

Suppress MOH to Conference Bridge *	True	True
Drop Ad Hoc Conference *	Never	Never
Maximum Ad Hoc Conference *	4	4
Maximum MeetMe Conference Unicast *	4	4
Advanced Ad Hoc Conference Enabled *	False	False
Choose Encrypted Audio Conference Instead Of Video Conference *	True	True
Minimum Video Capable Participants To Allocate Video Conference *	2	2
Enable Click-to-Conference for Third-Party Applications *	False	False
IMS Conference Factory URI *	cucm-conference-factory@cucm1.company.com	cucm-conference-factory@cucm1.company.com
Cluster Conferencing Prefix Identifier	<input type="text"/>	

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

Clusterwide Parameters (Feature - Call Secure Status Policy)

Secure Call Icon Display Policy *	All media except BFCP and iX transports must be encr	All media except BFCP and iX transports must be encrypted
---	--	---

Clusterwide Parameters (Feature - Forward)

Forward Maximum Hop Count *	12	12
Forward No Answer Timer *	12	12
Max Forward Hops to DN *	12	12
Retain Forward Information *	False	False
Forward By Reroute Enabled *	False	False
Transform Forward by Reroute Destination *	True	True

Figure 11: CUCM Service Parameter cont.



Always Forward Switch Voice Mail Calls *	True	True
Forward By Reroute T1 Timer *	10	10
Include Original Called Info for Q.SIG Call Diversions *	Only after the first diversion	Only after the first diversion
Set Private Numbering Plan for Call Forward *	False	False
Set Type of Number for Call Forward *	Level1RegionalNumber	Level1RegionalNumber
Max Forward UnRegistered Hops to DN *	0	0
CFA CSS Activation Policy *	With Configured CSS	With Configured CSS
Cause Code When Maximum Forward Hop Count is Triggered *	Normal Unspecified	Normal Unspecified
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		
Clusterwide Parameters (Feature - Hold Reversion)		
Hold Reversion Duration *	0	0
Hold Reversion Notification Interval *	30	30
CFA Destination Override *	False	False
Clusterwide Parameters (Feature - Call Pickup)		
Auto Call Pickup Enabled *	False	False
Call Pickup Locating Timer *	1	1
Call Pickup No Answer Timer *	12	12
Clusterwide Parameters (Feature - Refer)		
Validate Refer-to URI *	Validate Except for Anonymous Users	Validate Except for Anonymous Users
Clusterwide Parameters (Feature - Replaces)		
Block OffNet To OffNet Replaces *	False	False
Clusterwide Parameters (Feature - Redirection [3xx])		
Redirection Ring No Answer Reversion Timer *	24	24
Maximum Redirection Count *	70	70
Clusterwide Parameters (Feature - Multilevel Precedence and Preemption)		
Locations-based MLPP Enable *	False	False
Executive Override Call Preemptable *	False	False
Location-based Maximum Bandwidth Enforcement Level for MLPP Calls *	Lenient	Lenient

Figure 12: CUCM Service Parameter cont.



Non-Preemption Pattern CSS	< None >	▼	
MLPP Exception Level *	Executive Override	▼	Executive Override

Clusterwide Parameters (Feature - Path Replacement)			
Path Replacement Enabled *	False	▼	False
Path Replacement on Tromboned Calls *	True	▼	True
Start Path Replacement Minimum Delay Time *	0		0
Start Path Replacement Maximum Delay Time *	0		0
Path Replacement T1 Timer *	30		30
Path Replacement T2 Timer *	15		15
Path Replacement PINX ID			
Path Replacement Calling Search Space	< None >	▼	

Clusterwide Parameters (Feature - Call Back)			
Call Back Enabled Flag *	True	▼	True
Call Back Notification Audio File Name *	CallBack.raw		CallBack.raw
Connection Proposal Type *	Connection Retention	▼	Connection Retention
Connection Response Type *	Default to Connection Retention	▼	Default to Connection Retention
Call Back Request Protection T1 Timer *	10		10
Call Back Recall T3 Timer *	20		20
Call Back Calling Search Space	< None >	▼	
No Path Reservation *	True	▼	True
Set Private Numbering Plan for Call Back *	False	▼	False
Set Type of Number for Call Back *	Level1RegionalNumber	▼	Level1RegionalNumber

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

Clusterwide Parameters (Feature - Call Recording)			
Play Recording Notification Tone To Observed Target *	False	▼	False
Play Recording Notification Tone To Observed Connected Parties *	False	▼	False

Clusterwide Parameters (Feature - Monitoring)			
Play Monitoring Notification Tone To Observed Target *	False	▼	False
Play Monitoring Notification Tone To Observed Connected Parties *	False	▼	False

Figure 13: CUCM Service Parameter cont.



Clusterwide Parameters (Feature - Join Across Lines And Single Button Barge Feature Set)		
Join Across Lines Policy *	Off	Off
Single Button Barge/CBarge Policy *	Off	Off
Allow Barging When Ringing *	False	False

Clusterwide Parameters (Feature - Secure Tone)		
Play Tone to Indicate Secure/Non-Secure Call Status *	False	False

Clusterwide Parameters (Feature - External Call Control)		
External Call Control Diversion Maximum Hop Count *	12	12
Maximum External Call Control Diversion Hops to Pattern or DN *	12	12
External Call Control Routing Request Timer *	2000	2000
External Call Control Fully Qualified Role And Resource *	CISCO:UC:UCMPolicy:VoiceOrVideoCall	CISCO:UC:UCMPolicy:VoiceOrVideoCall
External Call Control Initial Connection Count To PDP *	2	2
External Call Control Maximum Connection Count To PDP *	4	4
Always use External Call Control-specified Called/Calling Party Names *	True	True

Clusterwide Parameters (Route Plan)		
Stop Routing on Out of Bandwidth Flag *	False	False
Stop Routing on Unallocated Number Flag *	True	True
Stop Routing on User Busy Flag *	True	True

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

Clusterwide Parameters (Route Class Signaling)		
Route Class Trunk Signaling Enabled *	True	True
SIP Route Class Naming Authority *	cisco.com	cisco.com

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

Clusterwide Parameters (Hunt List)		
Stop Hunting on Out of Bandwidth Flag *	False	False
Use Pickup Group Of Line Group Member DN *	False	False

Figure 14: CUCM Service Parameter cont.



Clusterwide Parameters (External QoS)		
External QoS Enabled *	False	False

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

Clusterwide Parameters (System - General)		
Always Use Dial Tone Setting *	Default	Default
Restart Cisco CallManager on Initialization Exception *	True	True
Digit Analysis Timer *	6	6
Statistics Enabled *	True	True

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

Clusterwide Parameters (System - QoS)		
Priority Class *	Normal Priority	Normal Priority
DSCP for Audio Calls *	46 (101110)	46 (101110)
DSCP for Video Calls *	34 (100010)	34 (100010)
DSCP for Audio Portion of Video Calls *	34 (100010)	34 (100010)

Figure 15: CUCM Service Parameter cont.



DSCP for TelePresence Calls *	32 (100000)	▼	32 (100000)
DSCP for Audio Portion of TelePresence Calls *	32 (100000)	▼	32 (100000)
DSCP for Priority Audio Calls *	45 (101101)	▼	45 (101101)
DSCP for Immediate Audio Calls *	44 (101100)	▼	44 (101100)
DSCP for Flash Audio Calls *	41 (101001)	▼	41 (101001)
DSCP for Flash Override Audio Calls *	42 (101010)	▼	42 (101010)
DSCP for Executive Override Audio Calls *	42 (101010)	▼	42 (101010)
DSCP for Priority Video Calls *	39 (100111)	▼	39 (100111)
DSCP for Immediate Video Calls *	37 (100101)	▼	37 (100101)
DSCP for Flash Video Calls *	35 (100011)	▼	35 (100011)
DSCP for Flash Override Video Calls *	33 (100001)	▼	33 (100001)
DSCP for Executive Override Video Calls *	33 (100001)	▼	33 (100001)
DSCP for G.Clear Calls *	46 (101110)	▼	46 (101110)
DSCP for Priority G.Clear Calls *	45 (101101)	▼	45 (101101)
DSCP for Immediate G.Clear Calls *	44 (101100)	▼	44 (101100)
DSCP for Flash G.Clear Calls *	41 (101001)	▼	41 (101001)
DSCP for Flash Override G.Clear Calls *	42 (101010)	▼	42 (101010)
DSCP for Executive Override G.Clear Calls *	42 (101010)	▼	42 (101010)
DSCP for Audio Calls when RSVP Fails *	0 (000000)	▼	0 (000000)
DSCP for Video Calls when RSVP Fails *	0 (000000)	▼	0 (000000)
DSCP for ICCP Protocol Links *	24 (011000)	▼	24 (011000)

Clusterwide Parameters (System - SDL)

SDL Listening Port Number *	8002		8002
SDL Max Router Latency *	20		20
Suppress Debug Info for Router Death *	0		0
Asynchronous SDL Logging Enabled *	False	▼	False

Clusterwide Parameters (System - Location and Region)

Enforce Millisecond Packet Size *	True	▼	True
Locations Trace Details Enabled *	False	▼	False
Preferred G.711 Millisecond Packet Size *	20	▼	20
Preferred G.722 Millisecond Packet Size *	20	▼	20
Preferred G.723.1 Millisecond Packet Size *	30	▼	30

Figure 16: CUCM Service Parameter cont.



Preferred G.723.1 Millisecond Packet Size *	30	▼	30
Preferred G.729 Millisecond Packet Size *	20	▼	20
Always Use Preferred G.729 Packet Size For SIP Trunk Answers *	False	▼	False
Preferred GSM EFR Bytes Packet Size *	31	▼	31
G.711 A-law Codec Enabled *	Enabled for All Devices	▼	Enabled for All Devices
G.711 mu-law Codec Enabled *	Enabled for All Devices	▼	Enabled for All Devices
G.722 Codec Enabled *	Enabled for All Devices	▼	Enabled for All Devices
iLBC Codec Enabled *	Enabled for All Devices	▼	Enabled for All Devices
iSAC Codec Enabled *	Enabled for All Devices	▼	Enabled for All Devices
Default Intra-region Max Audio Bit Rate *	64 kbps (G.722, G.711)	▼	64 kbps (G.722, G.711)
Default Inter-region Max Audio Bit Rate *	8 kbps (G.729)	▼	8 kbps (G.729)
Default Intra-region Max Video Call Bit Rate (Includes Audio) *	384		384
Default Inter-region Max Video Call Bit Rate (Includes Audio) *	384		384
Default Intra-region Max Immersive Video Call Bit Rate (Includes Audio) *	2000000000		2000000000
Default Inter-region Max Immersive Video Call Bit Rate (Includes Audio) *	2000000000		2000000000
Use Video BandwidthPool for Immersive Video Calls *	True	▼	True
Default Intra-region and Inter-region Link Loss Type *	Low Loss	▼	Low Loss
Default Audio Codec List between Regions *	Factory Default low loss	▼	Factory Default low loss
Default Audio Codec List within Region *	Factory Default low loss	▼	Factory Default low loss
Accept Audio Codec Preferences in Received Offer *	Off	▼	Off
G.Clear Bandwidth Override *	False	▼	False
Clusterwide Parameters (System - CCM Automated Alternate Routing)			
Automated Alternate Routing Enable *	False	▼	False
Clusterwide Parameters (System - RSVP)			
Default inter-location RSVP Policy *	No Reservation	▼	No Reservation
RSVP Retry Timer *	60		60
Mandatory RSVP Mid-call Retry Counter *	1		1
Mandatory RSVP mid call error handle option *	Call becomes best effort	▼	Call becomes best effort
RSVP Video Tspec Burst Size Factor *	5		5
MLPP EXECUTIVE_OVERRIDE To RSVP Priority Mapping *	65535		65535

Figure 17: CUCM Service Parameter cont.



MLPP FLASH OVERRIDE To RSVP Priority Mapping *	65534	65534
MLPP FLASH To RSVP Priority Mapping *	65533	65533
MLPP IMMEDIATE To RSVP Priority Mapping *	65532	65532
MLPP PL PRIORITY To RSVP Priority Mapping *	65531	65531
MLPP PL ROUTINE To RSVP Priority Mapping *	65530	65530
RSVP Audio Application ID *	AudioStream	AudioStream
RSVP Video Application ID *	VideoStream	VideoStream
RSVP Response Timer *	2	2

TLS Packet Capture Configurations		
Packet Capture Enable *	False	False
Packet Capture Max File Size (MB) *	2	2

Clusterwide Parameters(System - Presence)		
Presence Subscription Throttling Threshold *	60000	60000
Presence Subscription Resume Threshold *	80	80
Default Inter-Presence Group Subscription *	Disallow Subscription	Disallow Subscription
BLF Status Depicts DND *	False	False

Clusterwide Parameters (System - Mobility)		
Enterprise Feature Access Code for Hold *	*81	*81
Enterprise Feature Access Code for Exclusive Hold *	*82	*82
Enterprise Feature Access Code for Resume *	*83	*83
Enterprise Feature Access Code for Transfer *	*84	*84
Enterprise Feature Access Code for Conference *	*85	*85
Enterprise Feature Access Code for Session Handoff *	*74	*74
Enterprise Feature Access Code for Starting Selective Recording *	*86	*86
Enterprise Feature Access Code for Stopping Selective Recording *	*87	*87
Smart Mobile Phone Interdigit Timer *	500	500
Non-Smart Mobile Phone Interdigit Timer *	2000	2000
Send Call to Mobile Menu Timer *	60	60
SIP Dual Mode Alert Timer *	1500	1500

Figure 18: CUCM Service Parameter cont.



Call Screening Timer *	<input type="text" value="4000"/>	4000
Session Resumption Await Timer *	<input type="text" value="180"/>	180
Inbound Calling Search Space for Remote Destination *	Trunk or Gateway Inbound Calling Search Space	Trunk or Gateway Inbound Calling Search Space
Enable Enterprise Feature Access *	False	False
Dial-via-Office Forward Service Access Number	<input type="text"/>	
Enable Mobile Voice Access *	False	False
Mobile Voice Access Number	<input type="text"/>	
Matching Caller ID with Remote Destination *	Complete Match	Complete Match
Number of Digits for Caller ID Partial Match *	<input type="text" value="10"/>	10
System Remote Access Blocked Numbers	<input type="text"/>	
Enable Use of Called Party Transformed Number for Mobile-terminated Calls *	False	False
Honor Gateway or Trunk Outbound Calling Party Selection for Mobile Connect Calls *	False	False

Clusterwide Parameters (System - Mobility Single Number Reach Voicemail)		
Single Number Reach Voicemail Policy *	Timer Control	Timer Control
Dial-via-Office Reverse Voicemail Policy *	Timer Control	Timer Control
User Control Delayed Announcement Timer *	<input type="text" value="1000"/>	1000
User Control Confirmed Answer Indication Timer *	<input type="text" value="10000"/>	10000

Clusterwide Parameters (Feature - Reroute Remote Destination Calls to Enterprise Number)		
Reroute Remote Destination Calls to Enterprise Number *	False	False
Ring All Shared Lines *	False	False
Ignore Call Forward All on Enterprise DN *	True	True

Clusterwide Parameters (Feature - Immediate Divert)		
Use Legacy Immediate Divert *	True	True
Allow QSIG during iDivert *	False	False
Immediate Divert User Response Timer *	<input type="text" value="5"/>	5

Clusterwide Parameters (Call Admission Control)		
Call Counting CAC Enabled *	False	False
Audio Bandwidth For Call Counting CAC *	<input type="text" value="102"/>	102
Video Bandwidth For Call Counting CAC *	<input type="text" value="500"/>	500
UCM to LBM Periodic Reservation Refresh Timer *	<input type="text" value="5"/>	5
Maximum Bandwidth Deduction Duration *	<input type="text" value="720"/>	720
Call Treatment When No LBM Available *	Allow Calls	Allow Calls
Locations Media Resource Audio Bit Rate Policy *	Lowest Bit Rate	Lowest Bit Rate
Video Call QoS Marking Policy *	Default	Default

Clusterwide Parameters (Emergency Calling for Require Off-premise Location)		
Alternate Destination for Emergency Call	<input type="text"/>	
Alternate Calling Search Space for Emergency Call	< None >	

Figure 19: CUCM Service Parameter cont.

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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.

The screenshot displays the 'Audio Codec Preference List Configuration' page. At the top, there is a navigation menu with options like System, Call Routing, Media Resources, etc. Below the navigation is a header bar with the title 'Audio Codec Preference List Configuration' and a 'Related Links' section containing 'Back To Find/List' and a 'Go' button. A toolbar below the header includes icons for Save, Delete, Copy, and Add New. The main content area is titled 'Audio Codec Preference List Information' and contains a form with the following fields:

- Name***: Vodafone codec list
- Description***: Lossy Codec List
- Codecs in List***: A list of codecs, with 'G.729a 8k' and 'G.711 A-Law 64k' highlighted by a red box. The list includes: AMR-WB (7k-24k), AMR (5k-13k), MP4A-LATM 128k, AAC-LD (MP4A Generic), MP4A-LATM 64k, MP4A-LATM 56k, L16 256k, MP4A-LATM 48k, ISAC 32k, MP4A-LATM 32k, G.722 64k, G.722.1 32k, G.722 56k, G.722.1 24k, G.722 48k, MP4A-LATM 24k, G.711 U-Law 64k, G.711 U-Law 56k, G.711 A-Law 56k, ILBC 16k, G.728 16k, GSM Enhanced Full Rate 13k, GSM Full Rate 13k, G.729b 8k, G.729ab 8k, G.729 8k, GSM Half Rate 6k, and G.723.1 7k.

At the bottom of the form, there are buttons for Save, Delete, Copy, and Add New. The status bar at the top left of the main content area shows 'Status: Ready'.

Figure 20 Audio Codec Preference List



Region

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

Region Configuration
Related Links: [Back To Find/List](#)

Region Information
 Name*

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Vodafone_Region	Vodafone codec list	Use System Default (64 kbps (G.722, G.711))	384 kbps	2147483647 kbps
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

Modify Relationship to other Regions

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
<input type="radio"/> Default <input checked="" type="radio"/> Vodafone_Region	<input type="button" value="Keep Current Setting"/>	<input checked="" type="radio"/> <input type="button" value="Keep Current Setting"/> kbps <input type="radio"/> <input type="text" value=""/> kbps	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> <input type="text" value=""/> kbps	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> <input type="text" value=""/> kbps

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.

Device Pool Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

Device Pool Information
Device Pool: Vodafone_Devicepool (6 members**)

Device Pool Settings

Device Pool Name*	Vodafone_Devicepool
Cisco Unified Communications Manager Group*	Default
Calling Search Space for Auto-registration	< None >
Adjunct CSS	< None >
Reverted Call Focus Priority	Default
Intercompany Media Services Enrolled Group	< None >

Local Route Group Settings
Standard Local Route Group < None >

Roaming Sensitive Settings

Date/Time Group*	CMLocal
Region*	Vodafone_Region
Media Resource Group List	Vodafone_MRGL
Location	< None >
Network Locale	< None >
SRST Reference*	Disable
Connection Monitor Duration***	
Single Button Barge*	Default
Join Across Lines*	Default
Physical Location	< None >
Device Mobility Group	< None >
Wireless LAN Profile Group	< None >

[View Details](#)

Device Mobility Related Information****

Device Mobility Calling Search Space	< None >
AAR Calling Search Space	< None >
AAR Group	< None >
Calling Party Transformation CSS	< None >
Called Party Transformation CSS	< None >

Figure 22 CUCM Device Pool



Geolocation Configuration

Geolocation

Geolocation Filter

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 (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text" value=""/>	<input style="width: 100px;" type="text" value=" < None > "/>
International Number	<input type="text" value="Default"/>	<input type="text" value=""/>	<input style="width: 100px;" type="text" value=" < None > "/>
Unknown Number	<input type="text" value="Default"/>	<input type="text" value=""/>	<input style="width: 100px;" type="text" value=" < None > "/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text" value=""/>	<input style="width: 100px;" type="text" value=" < None > "/>

Incoming Called Party Settings

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

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input style="width: 100px;" type="text" value=" < None > "/>
International Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input style="width: 100px;" type="text" value=" < None > "/>
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input style="width: 100px;" type="text" value=" < None > "/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input style="width: 100px;" type="text" value=" < None > "/>

Phone Settings

Caller ID For Calls From This Phone

Calling Party Transformation CSS

Connected Party Settings

Connected Party Transformation CSS

Redirecting Party Settings

Redirecting Party Transformation CSS

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.

Conference Bridge Configuration Related Links: [Back To Find/List](#)

Status

Status: Ready

Conference Bridge Information

Conference Bridge : my-Conf (my-Conf)
Registration: Registered with Cisco Unified Communications Manager clus20sub1
IPv4 Address: 10.80.10.11

IOS Conference Bridge Info

Conference Bridge Type* Cisco IOS Enhanced Conference Bridge

Device is trusted

Conference Bridge Name*

Description

Device Pool*

Common Device Configuration

Location*

Device Security Mode*

Use Trusted Relay Point*

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](#).

Media Resource Group Configuration Related Links: [Back To Find/List](#)

Status

Status: Ready

Media Resource Group Status

Media Resource Group: Vodafone_MRG (used by 11 devices)

Media Resource Group Information

Name*

Description

Devices for this Group

Available Media Resources**

- ANN_2
- ANN_3
- ANN_4
- CFB_2
- CFB_3

Selected Media Resources*

- MOH_4 (MOH)
- MTP_2 (MTP)
- MTP_3 (MTP)
- MTP_4 (MTP)
- my-Conf (CFB)

Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

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](#)

Media Resource Group List Configuration Related Links: [Back To Find/List](#)

Status

Status: Ready

Media Resource Group List Status

Media Resource Group List: Vodafone_MRGL (used by 11 devices)

Media Resource Group List Information

Name*

Media Resource Groups for this List

Available Media Resource Groups

Selected Media Resource Groups

Vodafone_MRG

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.

SIP Trunk Security Profile Configuration Related Links: [Back To Find/List](#)

Status

Status: Ready

SIP Trunk Security Profile Information

Name*	Vodafone_Non Secure SIP Trunk Profile
Description	Vodafone_Non Secure SIP Trunk Profile
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP
Outgoing Transport Type	UDP
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
X.509 Subject Name	
Incoming Port*	5060
<input type="checkbox"/> Enable Application level authorization	
<input type="checkbox"/> Accept presence subscription	
<input type="checkbox"/> Accept out-of-dialog refer**	
<input type="checkbox"/> Accept unsolicited notification	
<input type="checkbox"/> Accept replaces header	
<input type="checkbox"/> Transmit security status	
<input type="checkbox"/> Allow charging header	
SIP V.150 Outbound SDP Offer Filtering*	Use Default Filter

Figure 27 SIP Trunk Security Profile



Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	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.
Incoming Port	5060	5060 is the default port for TCP/UDP



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
Related Links: [Back To Find/List](#) ▼

Save ✖ Delete Copy Reset Apply Config Add New

Status

i Status: Ready

i All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information

Name*	Vodafone_SIP_PROFILE
Description	Vodafone_SIP_PROFILE
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled ▼
User-Agent and Server header information*	Send Unified CM Version Information as User-Ager ▼
Version in User Agent and Server Header*	Major And Minor ▼
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, an ▼
Confidential Access Level Headers*	Disabled ▼

Redirect by Application

Disable Early Media on 180

Outgoing T.38 INVITE include audio mline

Use Fully Qualified Domain Name in SIP Requests

Assured Services SIP conformance

SDP Information

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS ▼
SDP Transparency Profile	Pass all unknown SDP attributes ▼
Accept Audio Codec Preferences in Received Offer*	Default ▼

Require SDP Inactive Exchange for Mid-Call Media Change

Allow RR/RS bandwidth modifier (RFC 3556)

Parameters used in Phone

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10

Figure 28: CUCM SIP Profile



Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

Conference Join Enabled
 RFC 2543 Hold
 Semi Attended Transfer
 Enable VAD
 Stutter Message Waiting
 MLPP User Authorization

Normalization Script

Normalization Script < None >

Enable Trace

	Parameter Name	Parameter Value	
1			+

Figure 29: CUCM SIP Profile cont.



Incoming Requests FROM URI Settings

Caller ID DN

Caller Name

Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*

RSVP Over SIP*

Resource Priority Namespace List

Fall back to local RSVP

SIP Rel1XX Options*

Video Call Traffic Class*

Calling Line Identification Presentation*

Session Refresh Method*

Early Offer support for voice and video calls*

Enable ANAT

Deliver Conference Bridge Identifier

Allow Passthrough 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)*

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

Ping Retry Timer (milliseconds)*

Ping Retry Count*

SDP Information

Send send-receive SDP in mid-call INVITE

Allow Presentation Sharing using BFCP

Allow iX Application Media

Allow multiple codecs in answer SDP

Figure 30: CUCM SIP Profile cont.



Parameter	Value	Description
Default MTP Telephony Event Payload Type	101	RFC2833 DTMF payload type
SIP Rel1XX Options	Send PRACK if 1xx Contains SDP	Enable Provisional Acknowledgements (Reliable 100 messages)
Early Offer support for voice and video calls	Mandatory (insert MTP if needed)	To create a trunk that supports early offer
Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None(Default)"	Checked	
Ping Interval for In-service and Partially In-service Trunks (seconds)	60	OPTIONS message parameters- interval time
Ping Interval for Out-of-service Trunks (seconds)	120	OPTIONS message parameters- interval time



SIP Trunk Configuration

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

Trunk Configuration Related Links: [Back To Find/List](#) ▼ G

Save Delete Reset Add New

Status

Info Status: Ready

SIP Trunk Status

Service Status: Full Service
Duration: Time In Full Service: 1 day 4 hours 26 minutes

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name *	Vodafone_SIP_TRUNK
Description	
Device Pool *	Vodafone_Devicepool ▼
Common Device Configuration	< None > ▼
Call Classification *	OffNet ▼
Media Resource Group List	Vodafone_MRGL ▼
Location *	Hub_None ▼
AAR Group	< None > ▼
Tunneled Protocol *	None ▼
QSIG Variant *	No Changes ▼
ASN.1 ROSE OID Encoding *	No Changes ▼
Packet Capture Mode *	None ▼
Packet Capture Duration	0

Media Termination Point Required

Retry Video Call as Audio

Path Replacement Support

Transmit UTF-8 for Calling Party Name

Transmit UTF-8 Names in QSIG APDU

Unattended Port

SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure * When using both sRTP and TLS ▼

Route Class Signaling Enabled * Default ▼

Use Trusted Relay Point * Default ▼

PSTN Access

Figure 31: SIP Trunk to Vodafone via CUBE



Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)
 E.164 Transformation Profile

MLPP and Confidential Access Level Information
 MLPP Domain
 Confidential Access Mode
 Confidential Access Level

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

Inbound Calls
 Significant Digits*
 Connected Line ID Presentation*
 Connected Name Presentation*
 Calling Search Space
 AAR Calling Search Space
 Prefix DN
 Redirecting Diversion Header Delivery - Inbound

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

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

Figure 32: SIP Trunk to Vodafone via CUBE cont.



Incoming Called Party Settings

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

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

Connected Party Settings

Connected Party Transformation CSS

Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Calling Line ID Presentation*

Calling Name Presentation*

Calling and Connected Party Info Format*

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

Maintain Original Caller ID DN and Caller Name in Identity Headers

Figure 33: SIP Trunk to Vodafone via CUBE cont.



SIP Information

Destination

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Duration
1*	10.80.10.10		5060	up		Time Up: 0 day 4 hours 26 minutes

MTP Preferred Originating Codec*

BLF Presence Group*

SIP Trunk Security Profile*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile* [View Details](#)

DTMF Signaling Method*

Normalization Script

Normalization Script

Enable Trace

	Parameter Name	Parameter Value
1		

Recording Information

None

This trunk connects to a recording-enabled gateway

This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation

Geolocation Filter

Send Geolocation Information

Figure 34: SIP Trunk to Vodafone via CUBE cont.



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

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



Route Pattern Configuration Related Links: [Back To Find/List](#) G

Save ✖ Delete C Copy + Add New

Status
Status: Ready

Pattern Definition

Route Pattern*	9001XXXXXXXXXX
Route Partition	< None >
Description	
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	Vodafone_SIP_TRUNK (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error
Call Classification*	OffNet
External Call Control Profile	< None >
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	0
<input type="checkbox"/> Require Client Matter Code	

Calling Party Transformations

<input type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

Called Party Transformations

Discard Digits	< None >
Called Party Transform Mask	001XXXXXXXXXX
Prefix Digits (Outgoing Calls)	
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Save ✖ Delete C Copy + Add New

Figure 35 Route Pattern cont.



Parameter	Value	Description
Route Pattern	9001XXXXXXXXXX	Specify appropriate Route Pattern
Gateway/Route List	Vodafone_SIP_TRUNK	SIP Trunk name configured earlier
Calling Line ID Presentation	Default	Set to Restricted for Anonymous Calls
Called Party Transform Mask	001XXXXXXXXXX	Strip '9' on the called party number before sending it out to the CUBE



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
```

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```
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

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



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Test Results



SP_SIP_master_testp
lan_V1.xlsx



**Corporate
Headquarters**

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

**European
Headquarters**

Cisco Systems International
BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

**Americas
Headquarters**

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

**Asia Pacific
Headquarters**

Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
www.cisco.com
Tel: +65 317 7777
Fax: +65 317 7799

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