



G12 Communications SIP Trunking:

**Cisco Unified Communications Manager 11.5.1
with Cisco Unified Border Element (CUBE 12.0)
on ISR 4321/K9 [IOS-XE – 16.6.1] using SIP**

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Introduction

Service Providers today, such as G12 Communications, are offering alternative methods to connect to the PSTN via their IP networks. Most of these services utilize SIP as the primary signaling method and centralized IP to TDM POP gateways to provide on-net and off-net services.

A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and G12 Communications network, Cisco Unified Border Element (CUBE) ISR 4321/K9 running IOS-XE 16.6.1 can be used. The Cisco Unified Border Element 16.6.1 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.1 connected to G12 Communications network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco Unified Communications Manager (CUCM). Only configuration settings specifically required for G12 Communications interoperability are presented. Feature configuration and most importantly the dial plan are customer specific and need individual approach.

- This application note describes how to configure a Cisco Unified Communications Manager (CUCM) 11.5.1 and Cisco Unified Border Element (CUBE) on ISR 4321/K9 [IOS-XE 16.6.1] for connectivity to G12 Communications SIP Trunking service available in G12 Communications Business service area . The deployment model covered in this application note is CPE (Cisco UCM 11.5.1) to PSTN (G12 Communications).
- Testing was performed in accordance to G12 Communications generic SIP Trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold (MOH), unattended and attended transfers, call forward, conferences and interoperability with Cisco Unity Connection (CUC).
- The Cisco UCM configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between G12 Communications SIP network and Cisco Unified Communications. The configuration described in this document details the important configuration settings to have enabled for interoperability to be successful and care must be taken by the network administrator deploying CUCM to interoperate to G12 Communications SIP Trunking network.

This application note does not cover the use of Calling Search Spaces (CSS) or partitions on CUCM. To understand and learn how to apply CSS and partitions refer to the cisco.com link below:

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



Network Topology

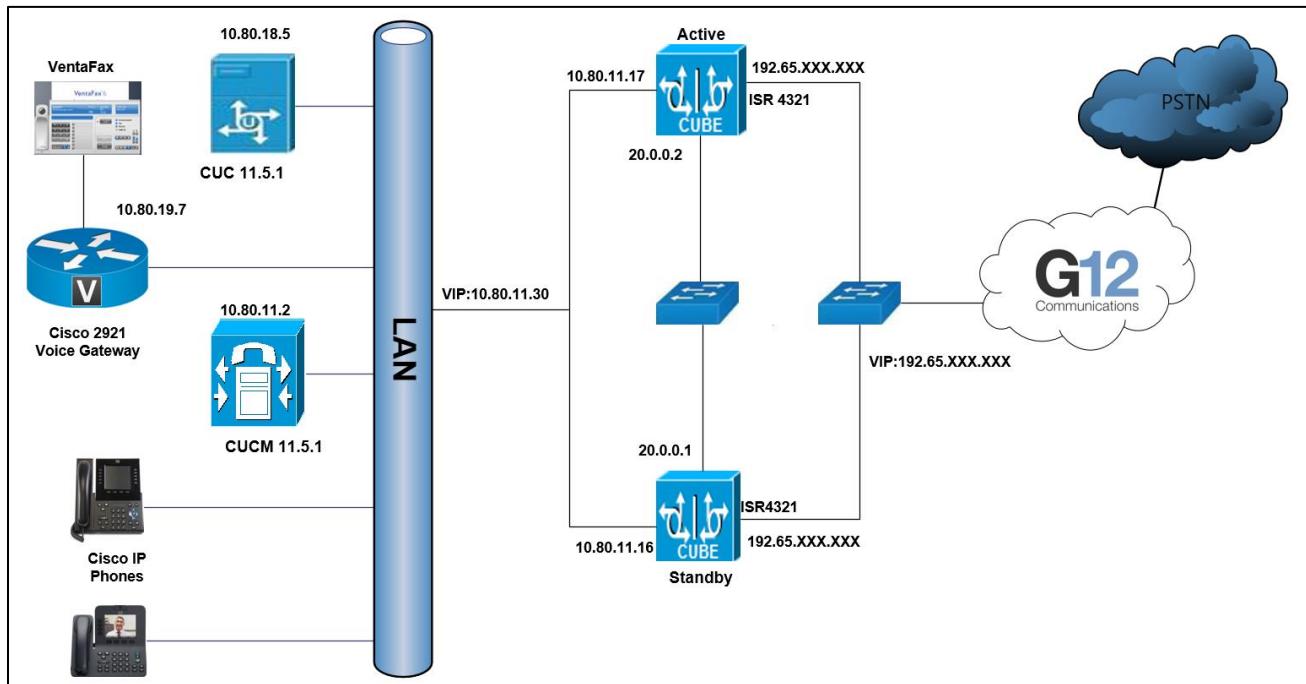


Figure 1: Network Topology

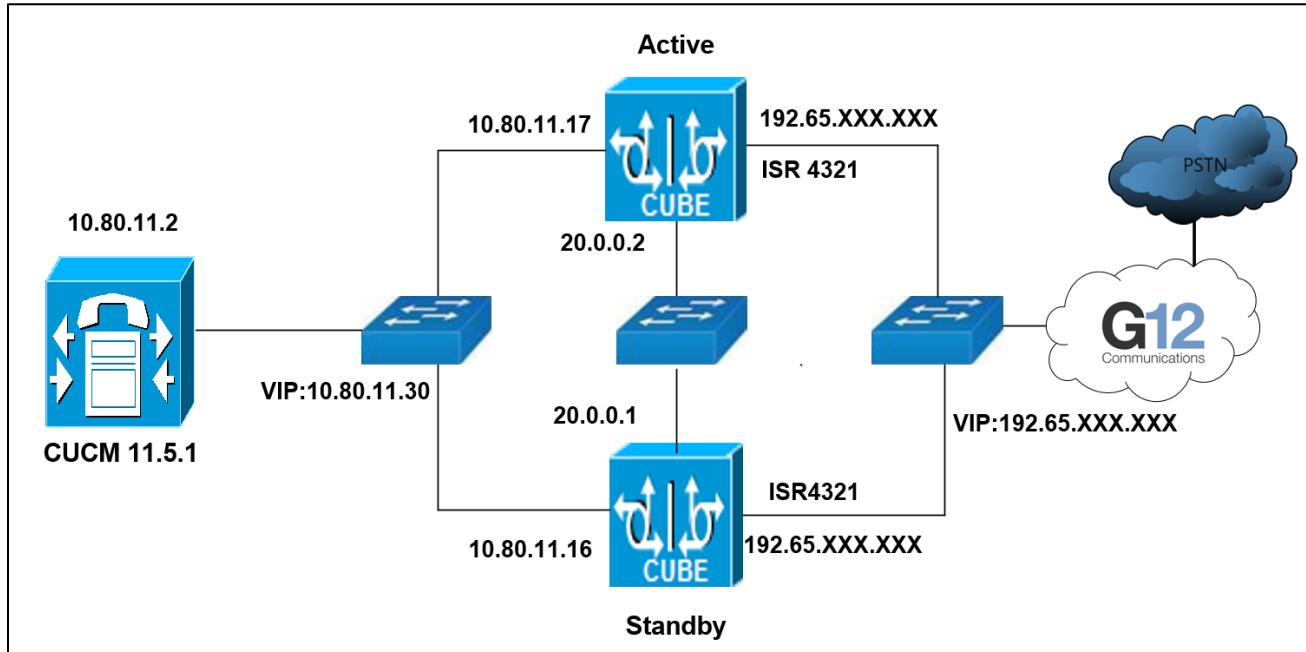


Figure 2: CUBE High Availability



System Components

Hardware Requirements

- Cisco UCSC-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR4321/K9 routers as CUBEs
- Cisco 2921 Fax Gateway
- IP Phones 9951 (SIP), 9951 (SIP) and 8945 (SCCP/SIP)

Software Requirements

- Cisco Unified Communications Manager 11.5.1
- Cisco Unity Connection 11.5.1
- IOS 16.06.01 for ISR 4321/K9 Cisco Unified Border Element
- Cisco IOS Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 16.06.01, RELEASE SOFTWARE (fc1)
- Cisco IOS XE Software, Version 16.06.01
- IOS 15.4(3)M1 for Cisco 2921 Fax Gateway

Features

Features Supported

- Incoming and Outgoing off-net calls using G711ULaw
- Call Hold
- Call Transfer (semi-attended and attended)
- Call Conference
- Call Forward (all, busy and no answer)
- Calling Line (number) Identification Presentation (CLIP)
- Calling Line (number) Identification Restriction (CLIR)
- DTMF Relay (both directions) (RFC2833)
- Media flow-through on Cisco UBE
- Fax (G.711 pass-through and T38)

Features Not Supported

- Cisco IP phones used in this test do not support blind transfer

Caveats

- Caller ID is not updated after attended or unattended transfers to off-net phones. This is due to a limitation on CUBE and will be resolved in the next release. The issue does not impact the calls.
- G12 Communications does not anchor the media for CPE to CPE loopback calls
- When putting outgoing international called party on hold, the call drops in 30 seconds



Configuration

Configuring Cisco Unified Border Element (CUBE)

Network Interface

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

```
interface GigabitEthernet0/0/0
  ip address 192.65.XXX.XXX 255.255.255.128
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 192.65.XXX.XXX exclusive
!
interface GigabitEthernet0/0/1
  ip address 10.80.11.17 255.255.255.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.11.30 exclusive
```



Global CUBE Settings

In order to enable Cisco UBE IP2IP gateway functionality, enter the following:

```
voice service voip
no ip address trusted authenticate
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through 711ulaw
sip
bind control source-interface GigabitEthernet0/0/0
bind media source-interface GigabitEthernet0/0/0
rel1xx supported "rel100"
session refresh
header-passing
asserted-id pai
early-offer forced
midcall-signaling passthru
privacy-policy passthru
g729 annexb-all
```

Explanation

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



Codecs

G711Ulaw is used as the preferred codec for this testing and changed the preferences according to the test plan description

```
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
```

Dial Peer

The CUBE uses dial-peers to route the call accordingly based on the digits

```
description Incoming from CUCM
huntstop
session protocol sipv2
incoming called-number [0-9]T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
no vad
!
dial-peer voice 20 voip
description Outgoing to G12
huntstop
destination-pattern [0-9]T
session protocol sipv2
session server-group 100
voice-class codec 1
no voice-class sip conn-reuse
voice-class sip profiles 100
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
```



```
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
no vad
!
dial-peer voice 30 voip
description Incoming from G12
huntstop
session protocol sipv2
incoming called-number 206.....
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
no vad
!
dial-peer voice 40 voip
description Outgoing to CUCM
huntstop
destination-pattern 206.....
session protocol sipv2
session target ipv4:10.80.11.2
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
no vad
!
```



Call Flow

In the sample configuration presented here, CUCM is provisioned with 4-digit directory number corresponding to the last four DID digits. No digit manipulation is performed on the CUBE.

For incoming PSTN calls, the CUBE presents the full 10-digit DID number to CUCM. The CUCM picks up the last 4 significant digits configured under SIP Trunk and routes the call based on those 4 digits. Voice calls are routed to IP phones. Fax calls are routed via a 4-digit route pattern over a SIP trunk that terminates on the Fax Gateway and in turn to the VentaFax client connected to the Fax Gateway.

CPE callers make outbound PSTN calls by dialing a “6” prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, the Cisco fax Gateway sends to the CUCM the DID with a leading access code “6”. A “6.@” route pattern strips the prefix and routes the call with the remaining digits via a SIP trunk terminating on the CUBE for Voice call or Fax. For PBX to PBX via G12, Caller dials a 6 prefix followed by the target 11-digit DID number for that extension number. 6 was stripped and the 11-digit number is sent to the CUBE. The CUBE then sends the full 11-digit DID under Dial Peer 20 and then sends it to the G12 network which will direct back to the CUBE with a 10-digit DID. This is handled the same as normal incoming PSTN call. For International calls, the same pattern 6 followed by 011 country code and the called number is used.

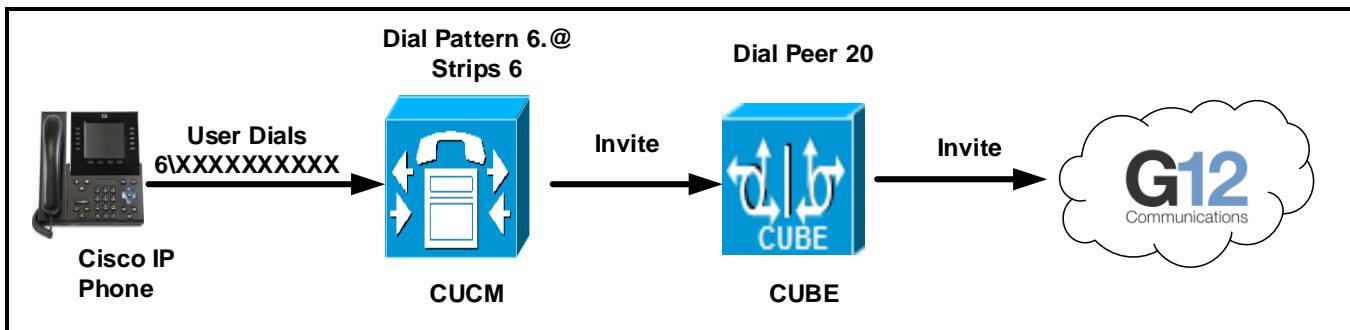


Figure 3: Outbound Voice Call

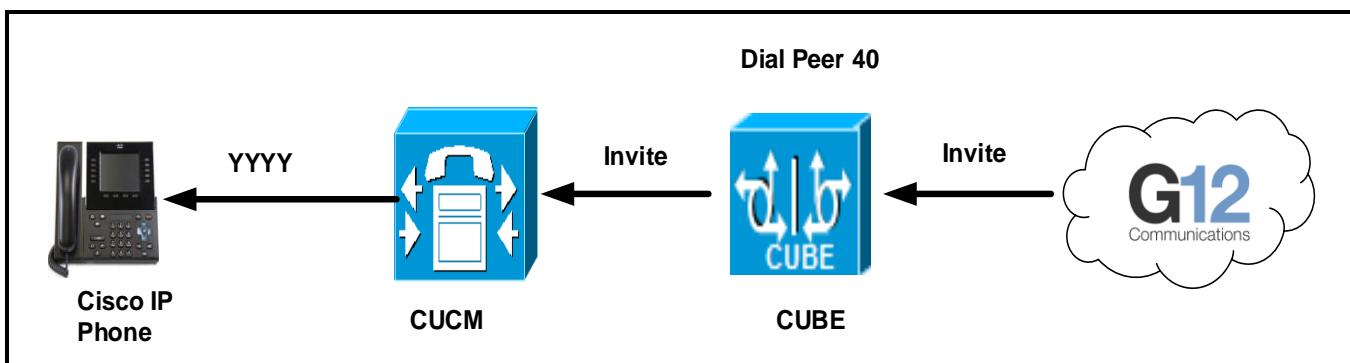


Figure 4: Inbound Voice Call

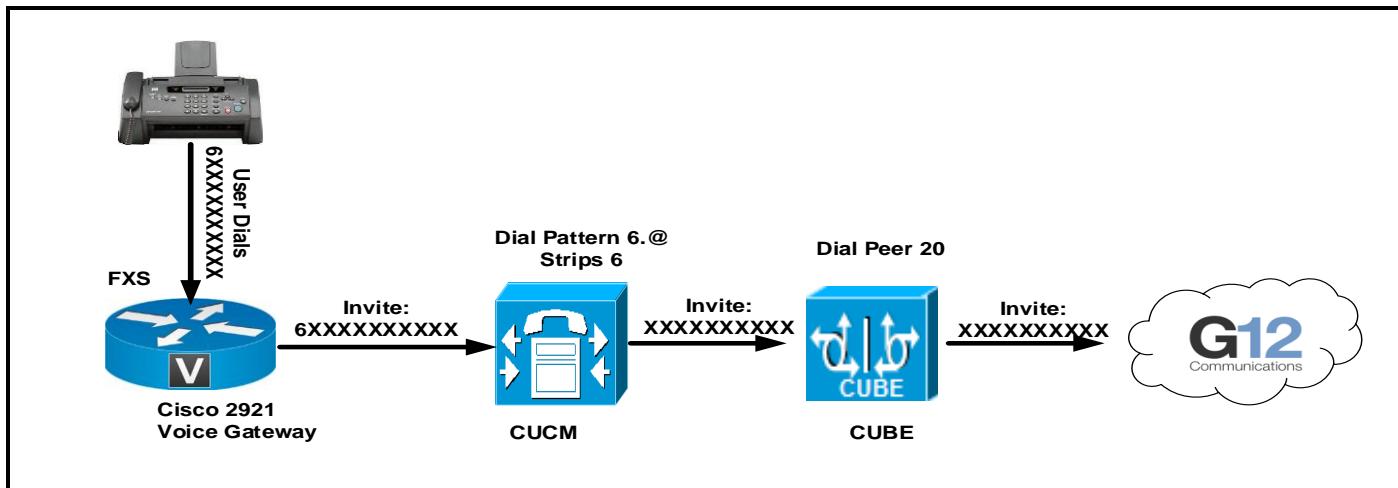


Figure 5: Outbound Fax Call

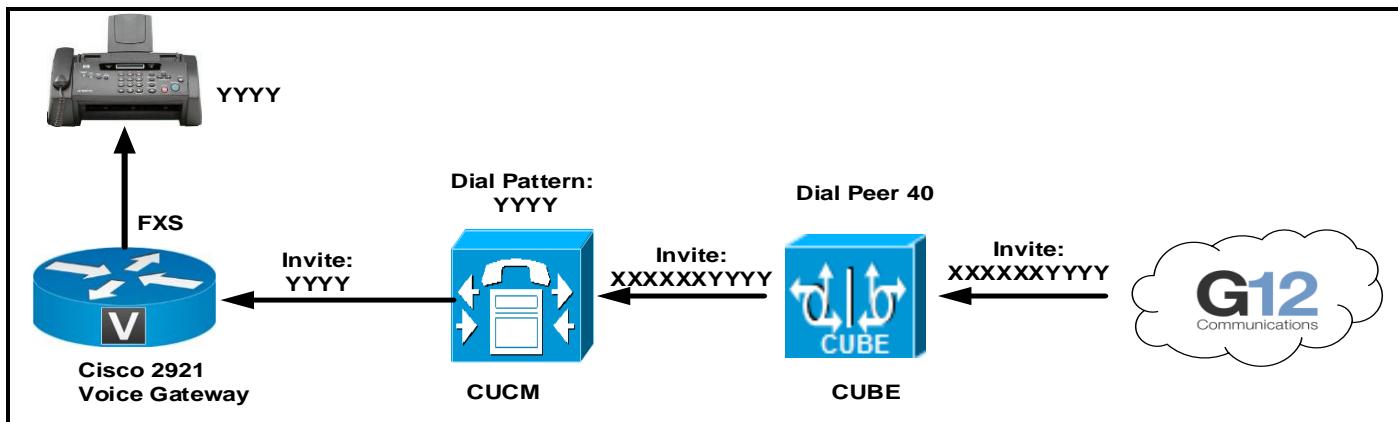


Figure 6: Inbound Fax Call

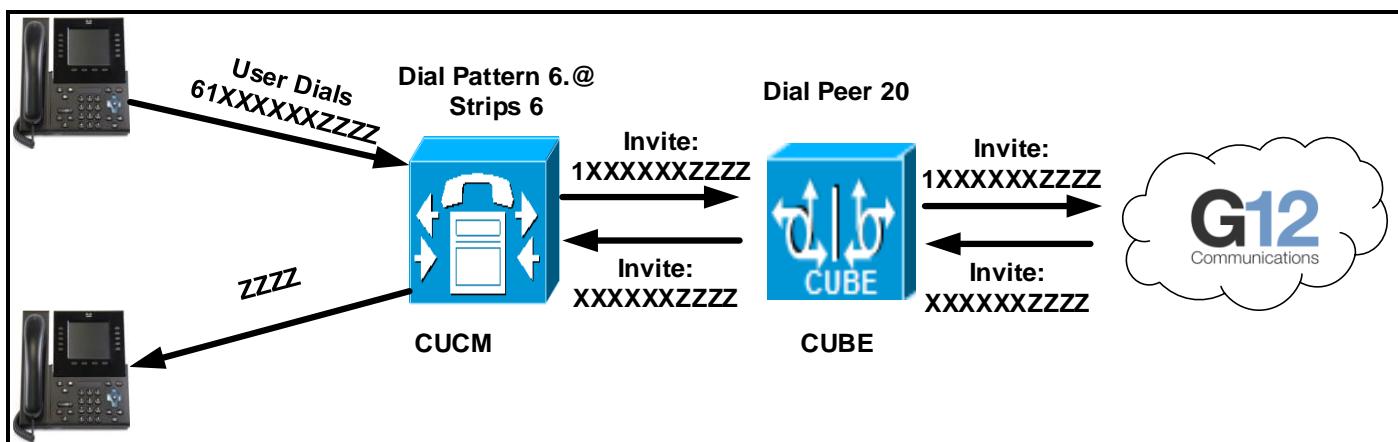


Figure 7: PBX to PBX call via G12 Communications



Configuration Example

The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously

Active Cisco UBE

```
G12_CUBE2#show run
Building configuration...

Current configuration : 7773 bytes
!
! Last configuration change at 14:51:53 UTC Wed Oct 18 2017 by cisco
!
version 16.6
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname G12_CUBE2
!
boot-start-marker
boot system flash isr4300-universalk9.16.06.01.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
enable secret 5
!
no aaa new-model
!
ip name-server 8.8.8.8
!
subscriber templating
!
multilink bundle-name authenticated
!
```



```
crypto pki trustpoint TP-self-signed-1017057749
enrollment selfsigned
subject-name cn=IOS-Self-Signed-Certificate-1017057749
revocation-check none
rsakeypair TP-self-signed-1017057749
!
crypto pki certificate chain TP-self-signed-1017057749
certificate self-signed 01
30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
69666963 6174652D 31303137 30353737 3439301E 170D3137 31303131 31333234
34365A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 30313730
35373734 39308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
0A028201 0100A957 AF2CA679 B9FD4E4C 6C231D88 7B836497 253C30D0 C8746F99
A8DD57F6 3CA0D4FD 15932BFB 00DE7FD3 436AACCE 52E3F045 E2483961 28624BFD
CB00FB53 C448CE80 58CEC4D9 AAB8DD47 EE797645 EAE5866F 7E4E89C3 B6A86EC9
BA135A50 1CA45831 F5B8C3B6 0A48C3DB D307F498 E15709BB 9FA88F6D D6066E68
3406C14C 8AE74DEB CC127E18 93A3DB47 1C05DEA5 1EAD371E C76BC53B 3ADC76FA
E87E6D70 C744562D 536B86D4 581DE290 0523CA7A 479D22B8 0D02AD54 255FDC13
13524CD1 0F0A9A10 1860D836 09ABBA03 917E0C30 ADC7C87E D461C96A F947EE37
56BEA385 BE6FF94F 8DFB9B3D 7C640CB6 4F4BF279 C948D03E 33A79940 6AC6920F
634C4E82 F9230203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
301F0603 551D2304 18301680 14E445B3 6D3F6F36 AA26B07B 43F69DFA 2D095A0F
EF301D06 03551D0E 04160414 E445B36D 3F6F36AA 26B07B43 F69DFA2D 095A0FEF
300D0609 2A864886 F70D0101 05050003 82010100 597C3005 9ABB87D0 378FE6F7
8D5984AD BD68BDD6 93538EFA 8DF317C3 AA6A608C 81FA2A45 EEEFD182 1641F44D
3CDA5BBB CDBC41F3 B215C6BB AA62402B B3F552A5 46C38875 4073F5CA 416DF24E
4748DCCF 8F31A513 A4A43762 1B08B4D2 18A8BA7D 1239EAA8 19686B0D ECDD3BB3
8B699594 73A846BD 6042C4DD 6622A1AF 8EB7C6F9 4A1F95EF 940044B6 4B31132D
C8273040 191A3F2F 9E1CC6F5 82B83C64 0357BA87 D493C319 70478CA2 0D3330B6
038D2FCE AF50DE39 89117BDD 0E191A4C D497B41F 29E99589 0D70C7E0 08CBBEF7
B22D921C E308AE92 5D21E806 B55B3F74 5E7338EB BBF7E05D 9C3EE6A5 4DA13EDE
2B8E1DDB BEE199ED 7959506F 01B682C7 D36583CE
quit
!
voice service voip
no ip address trusted authenticate
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
sip
```



```
bind control source-interface GigabitEthernet0/0/0
bind media source-interface GigabitEthernet0/0/0
rel1xx supported "rel100"
header-passing
asserted-id pai
early-offer forced
midcall-signaling passthru
privacy-policy passthru
g729 annexb-all
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g711ulaw
!
!
voice class sip-profiles 100
request INVITE sip-header Diversion modify "<sip:(.*)>@(.*)>" "<sip:2065391@12>"
!
!
voice class server-group 100
ipv4 174.127.194.15
ipv4 169.55.253.153
ipv4 169.55.93.188
description G12ServerGroup
!
license udi pid ISR4321/K9 sn FDO19220MQ9
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level uck9
license boot level securityk9
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 0
!
redundancy
mode none
application redundancy
```



```
group 1
  name voice-b2bha
  timers delay 30 reload 60
  control GigabitEthernet0/1/0 protocol 1
  data GigabitEthernet0/1/0
  track 1 shutdown
  track 2 shutdown
!
track 1 interface GigabitEthernet0/0/1 line-protocol
!
track 2 interface GigabitEthernet0/0/0 line-protocol
!
interface GigabitEthernet0/0/0
  ip address 192.65.79.187 255.255.255.128
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 192.65.79.185 exclusive
!
interface GigabitEthernet0/0/1
  ip address 10.80.11.17 255.255.255.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.11.30 exclusive
!
interface GigabitEthernet0/1/0
  ip address 20.0.0.2 255.255.255.252
  negotiation auto
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  shutdown
  negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.0.0 10.80.11.1
ip route 10.80.19.0 255.255.255.0 10.80.11.1
ip route 172.16.24.0 255.255.248.0 10.80.11.1
```



```
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
no logging trap
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 10 voip
description Incoming from CUCM
huntstop
session protocol sipv2
incoming called-number [0-9]T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
no vad
!
dial-peer voice 20 voip
description Outgoing to G12
huntstop
destination-pattern [0-9]T
session protocol sipv2
session server-group 100
voice-class codec 1
no voice-class sip conn-reuse
voice-class sip profiles 100
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
```



```
fax-relay sg3-to-g3
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
no vad
!
dial-peer voice 30 voip
description Incoming from G12
huntstop
session protocol sipv2
incoming called-number 206.....
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
no vad
!
dial-peer voice 40 voip
description Outgoing to CUCM
huntstop
destination-pattern 206.....
session protocol sipv2
session target ipv4:10.80.11.2
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
no vad
sip-ua
!
line con 0
transport input none
stopbits 1
line aux 0
!
End
```



Standby Cisco UBE

```
G12_CUBE1#show run
Building configuration...

Current configuration : 7613 bytes
!
! Last configuration change at 08:16:47 UTC Fri Oct 13 2017
!
version 16.6
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname G12_CUBE1
!
boot-start-marker
boot system flash isr4300-universalk9.16.06.01.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
enable secret 5
!
no aaa new-model
!
ip name-server 8.8.8.8
!
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-1582728230
enrollment selfsigned
subject-name cn=IOS-Self-Signed-Certificate-1582728230
revocation-check none
```



```
rsakeypair TP-self-signed-1582728230
!
crypto pki certificate chain TP-self-signed-1582728230
certificate self-signed 01
30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
69666963 6174652D 31353832 37323832 3330301E 170D3137 31303131 30363538
31375A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 35383237
32383233 30308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
0A028201 0100E31F 6900B8BC C601E015 DF198F36 1343B5FF 6EA71FB0 8DA0249C
A0FF6675 97FF283C ADFA3831 3E5D2622 AC1EFAF1 BFE3C611 7E147DD3 48BC9FC5
D3CAA2FC 0C92B5EF 18A6C5BA 037F1B9C 14469056 8F4C15FB 7520A07D 1A7F02F8
7B8F33BB FDD37EEA 47318F61 395E55F4 B544CE60 68C9CC87 65ABC151 02EDFB7C
497BA10F B11AA874 8F7F82B9 9ADF3287 77DBBCEA A28AC00C 353BE95C 585EBF7F
1410B14F 8A6400DB 92371ED3 D78117F7 235B6EED 282E8AF0 A107ED2D EC2B6D21
E74F285E E11C04BC 22A293A3 E7E91885 BEB4330C 4DBF9F37 D83292AB 2C01A364
E9C325E9 1BC51104 E250796E 6D329874 B2AAD35D 698A2981 4C33D2D0 3854DC35
AC24A11B 77C50203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
301F0603 551D2304 18301680 140B97D1 8439D048 CC281ABD BAEA237E 4F0D38FA
D6301D06 03551D0E 04160414 0B97D184 39D048CC 281ABDBA EA237E4F 0D38FAD6
300D0609 2A864886 F70D0101 05050003 82010100 C819EA60 4C307C2C 91735FA2
2234C0F1 0A6C96BD 4F86A4D0 FFF6A0B2 066BC6E9 1B487D19 7AFDDDB5 45AD813A
7A22ABE7 4FC42233 6A038BBB E649A6B6 81DB540B 78AE7018 D9D4C0BC 142E2C69
1033925C C8B08E30 DFEF1C6C 437EA288 2AF29183 B25D6B3C A507AC59 A348C946
A6055A60 AA5D9040 F831E85B 6A9E5FDE 18039682 3A5E6AD7 88030B29 3EA2C9C9
ECDFB83C AE7D5AE2 4B5630E4 6A61B151 49223E43 22200311 7B49E268 7FE48995
DAFE0697 42337C71 92967FED B5944DC2 5E97B44E A0C06E81 7BB178BF 07ED347D
07EFCB59 1F93E4A7 53285623 060ABB17 7E6AE923 26847BE2 3E5CC58E A6EEAC28
1A89ACDB 2C05B19C A4657201 7F4B60FC ABDE9BF7
    quit
!
voice service voip
no ip address trusted authenticate
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
sip
bind control source-interface GigabitEthernet0/0/0
bind media source-interface GigabitEthernet0/0/0
rel1xx supported "rel100"
header-passing
```



```
asserted-id pai
early-offer forced
midcall-signaling passthru
privacy-policy passthru
g729 annexb-all
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g711ulaw
!
voice class sip-profiles 100
request INVITE sip-header Diversion modify "<sip:(.*)>@(.*)>" "<sip:206539\1@|2>""
!
voice class server-group 100
ipv4 174.127.XXX.XX
ipv4 169.55.XXX.XXX
ipv4 169.55.XX.XXX
description G12ServerGroup
!
license udi pid ISR4321/K9 sn FDO19220MSQ
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level securityk9
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 0
!
redundancy
mode none
application redundancy
group 1
name voice-b2bha
timers delay 30 reload 60
control GigabitEthernet0/1/0 protocol 1
data GigabitEthernet0/1/0
track 1 shutdown
track 2 shutdown
```



```
!
track 1 interface GigabitEthernet0/0/1 line-protocol
!
track 2 interface GigabitEthernet0/0/0 line-protocol
!
interface GigabitEthernet0/0/0
ip address 192.65.79.186 255.255.255.128
negotiation auto
redundancy rii 2
redundancy group 1 ip 192.65.79.185 exclusive
!
interface GigabitEthernet0/0/1
ip address 10.80.11.16 255.255.255.0
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.80.11.30 exclusive
!
interface GigabitEthernet0/1/0
ip address 20.0.0.1 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.0.0 10.80.11.1
ip route 10.80.19.0 255.255.255.0 10.80.11.1
ip route 172.16.24.0 255.255.248.0 10.80.11.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
```



```
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 10 voip
description Incoming from CUCM
huntstop
session protocol sipv2
incoming called-number [0-9]T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
no vad
!
dial-peer voice 20 voip
description Outgoing to G12
huntstop
destination-pattern [0-9]T
session protocol sipv2
session server-group 100
voice-class codec 1
no voice-class sip conn-reuse
voice-class sip profiles 100
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
no vad
!
dial-peer voice 30 voip
description Incoming from G12
huntstop
session protocol sipv2
incoming called-number 206.....
voice-class codec 1
```



```
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
no vad
!
dial-peer voice 40 voip
description Outgoing to CUCM
huntstop
destination-pattern 206.....
session protocol sipv2
session target ipv4:10.80.11.2
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
no vad
!
sip-ua
!
line con 0
exec-timeout 0 0
transport input none
stopbits 1
line aux 0
stopbits 1
line vty 0 4
logging synchronous
login local
!
ntp server time-pnp.cisco.com
!
end
```



Configuring Cisco Unified Communications Manager

Cisco UCM Version

The screenshot shows the Cisco Unified CM Administration interface. At the top, there's a navigation bar with links for System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The title "Cisco Unified CM Administration" is displayed, along with the subtext "For Cisco Unified Communications Solutions". On the right side of the header, there are links for administrator, Search Documentation, About, and Logout. Below the header, a banner displays the text "Cisco Unified CM Administration" and "System version: 11.5.1.12900-21" (which is highlighted with a red box). Another banner below it provides hardware information: "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 banners, there's a small image of a server room.

Figure 8: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

1. Select **Server***: Clus21Pub--CUCM Voice/Video (Active)
2. Select **Service***: Cisco CallManager (Active)

The screenshot shows the "Service Parameter Configuration" page. The top navigation bar is identical to Figure 8. The main content area has a sub-header "Service Parameter Configuration". It includes buttons for Save, Set to Default, and Advanced. A "Status" section shows "Status: Ready". Below it is a "Select Server and Service" section where "Server*" is set to "clus21pub--CUCM Voice/Video (Active)" and "Service*" is set to "Cisco CallManager (Active)". A note states: "All parameters apply only to the current server except parameters that are in the cluster-wide group(s)." The bottom section is titled "Cisco CallManager (Active) Parameters on server clus21pub--CUCM Voice/Video (Active)". It lists several parameters under "Call Throttling":

Parameter Name	Parameter Value	Suggested Value
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

Figure 9: Service Parameters



Memory Throttling	
<u>Enable Memory Throttling</u> *	<input type="text" value="True"/> True
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.	
System	
<u>CDR Enabled Flag</u> *	<input type="text" value="False"/> False
<u>CDR Log Calls with Zero Duration Flag</u> *	<input type="text" value="False"/> False
<u>Digit Analysis Complexity</u> *	<input type="text" value="StandardAnalysis"/> StandardAnalysis
<u>Database Debounce Timer</u> *	<input type="text" value="0"/> 0
<u>Maximum Phone Fallback Queue Depth</u> *	<input type="text" value="10"/> 10
<u>Maximum Number of Registered Devices</u> *	<input type="text" value="5000"/> 5000
<u>System Initialization Timer</u> *	<input type="text" value="60"/> 60
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.	
SDL Trace	
<u>SDL Trace Data Flags</u> *	<input type="text" value="0x00000111"/> 0x00000111
<u>SDL Trace Flush Immediately</u> *	<input type="text" value="False"/> False
<u>SDL Trace Data Size</u> *	<input type="text" value="0"/> 0
<u>SDL Trace Flag</u> *	<input type="text" value="True"/> True
<u>SDL TraceType Flags</u> *	<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)	
<u>Call Diagnostics Enabled</u> *	<input type="text" value="Disabled"/> Disabled
<u>Show Line Group Member DN in finalCalledPartyNumber CDR Field</u> *	<input type="text" value="False"/> False
<u>Show Line Group Member Non Masked DN in finalCalledPartyNumber CDR Field</u> *	<input type="text" value="False"/> False
<u>CTI New Call Accept Timer</u> *	<input type="text" value="4"/> 4
<u>CTI Generate Digits Interval</u> *	<input type="text" value="250"/> 250
<u>CTI Dial Digits Interval</u> *	<input type="text" value="250"/> 250
<u>CTI Await Further Digits</u> *	<input type="text" value="False"/> False
<u>CTI Use Wildcard Pattern as calledPartyDN</u> *	<input type="text" value="False"/> False
<u>CTI Report Ringback on SIP_183 with SDP</u> *	<input type="text" value="True"/> True
<u>Retain Media on Disconnect with PI for Active Call</u> *	<input type="text" value="False"/> False
<u>Station and Backup Server KeepAlive Interval</u> *	<input type="text" value="60"/> 60
<u>Station KeepAlive Interval</u> *	<input type="text" value="30"/> 30
<u>Status Enquiry Poll Flag</u> *	<input type="text" value="False"/> False
<u>Strip # Sign from Called Party Number</u> *	<input type="text" value="True"/> True
<u>Session Handoff Alerting Timer</u> *	<input type="text" value="10"/> 10

Figure 10: Service Parameters (Cont.)



<u>T301 Timer</u> *	180000	180000
<u>T302 Timer</u> *	15000	15000
<u>T303 Timer</u> *	4000	4000
<u>T304 Timer</u> *	30000	30000
<u>T305 Timer</u> *	30000	30000
<u>T306 Timer</u> *	30000	30000
<u>T308 Timer</u> *	4000	4000
<u>T309 Timer</u> *	90000	90000
<u>T310 Timer</u> *	60000	60000
<u>T313 Timer</u> *	4000	4000
<u>T316 Timer</u> *	120000	120000
<u>T317 Timer</u> *	100000	100000
<u>T321 Timer</u> *	30000	30000
<u>T322 Timer</u> *	4000	4000
<u>Tone on Hold Timer</u> *	10	10
<u>Unknown Caller ID Flag</u> *	True	True
<u>Call Classification</u> *	OffNet	OffNet
<u>Always Display Original Dialed Number</u> *	False	False
<u>Name Display for Original Dialed Number When Translated</u> *	Show the Display Name for Original Dialed Number even if Translated	Show the Display Name for Original Dialed Number even if Translated
<u>Always Use PIs With Original Dialed Number</u> *	False	False
<u>Fail Call If Trusted Relay Point Allocation Fails</u> *	True	True
<u>Display Calling/Called ID When PI is Not Available</u> *	False	False
<u>Enable Transit Counter Processing on QSIG Trunks</u> *	False	False
<u>Egress FacilityIE Count</u> *	6	6
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		
Clusterwide Parameters (Device - Phone)		
<u>Always Use Prime Line</u> *	False	False
<u>Always Use Prime Line for Voice Message</u> *	False	False
<u>Builtin Bridge Enable</u> *	Off	Off
<u>Device Mobility Mode</u> *	Off	Off
<u>Display Device Mobility Location During Phone Registration</u> *	True	True
<u>Auto Answer Timer</u> *	1	1
<u>Extension Display on Cisco IP Phone Model 7910</u> *	False	False
<u>Alternate Idle Phone Auto-Answer Behavior Enabled</u> *	False	False
<u>Hold Type</u> *	False	False
<u>Line State Update Enabled</u> *	True	True
<u>Off-hook to First Digit Timer</u> *	15000	15000
<u>Override Auto Answer If Speaker Is Disabled</u> *	True	True
<u>Out-of-Bandwidth Text</u> *	Not Enough Bandwidth	Not Enough Bandwidth
<u>Forced Authorization Code Prompt Text</u> *	Enter Authorization Code	Enter Authorization Code

Figure 11: Service Parameters (Cont.)



<u>Client Matter Code Prompt Text</u> *	Enter Client Matter Code	Enter Client Matter Code
<u>AAR Network Congestion Rerouting Text</u> *	Network Congestion. Rerouting.	Network Congestion. Rerouting.
<u>Ring Setting of Busy Station Policy</u> *	Only Apply Ring Setting of Busy Station When Incoming Call Arrives	Only Apply Ring Setting of Busy Station When Incoming Call Arrives
<u>Transfer On-hook Enabled</u> *	False	False
<u>Ring Setting of Busy Station</u> *	Beep Only	Beep Only
<u>Ring Setting of Idle Station</u> *	Ring	Ring
<u>Call Pickup Group Audio Alert Setting of Idle Station</u> *	Ring Once	Ring Once
<u>Call Pickup Group Audio Alert Setting of Busy Station</u> *	Beep Only	Beep Only
<u>BLF Pickup Audio Alert Setting of Idle Station</u> *	Disable	Disable
<u>BLF Pickup Audio Alert Setting of Busy Station</u> *	Disable	Disable
<u>Privacy Setting</u> *	True	True
<u>Enforce Privacy Setting on Held Calls</u> *	False	False
<u>SIP Station KeepAlive Interval</u> *	120	120
<u>SIP Station Realm</u> *	ccmsipline	ccmsipline
<u>Hunt Group Logoff Notification</u> *	None	None
<u>Speed Dial Await Further Digits</u> *	False	False
<u>Display CTI Route Point Name or DN</u> *	False	False
<u>Display Original Calling Number on Transfer from Cisco Unity</u> *	False	False
<u>URI Dialing Display Preference</u> *	DN	DN
<u>Insert Hyphens in 12-Digit Numbers</u> *	False	False
<u>Allow Call Waiting During an In-Progress Outbound Analog Call</u> *	True	True
<u>Treat Foreign Domain Calls as URI in Phone Call History</u> *	False	False
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		
- Clusterwide Parameters (Device - PRI and MGCP Gateway)		
<u>Calling Party Number Screening Indicator</u> *	CallManager sets the screening indicator value - Default setting	CallManager sets the screening indicator value - Default setting
<u>Enable Outbound NetworkTrunk CallingParty Restriction</u> *	False	False
<u>Clear Calls Flag When Datalink Is Down</u> *	True	True
<u>Device Status Poll Interval</u> *	3000	3000
<u>Disable Alerting Progress Indicator</u> *	False	False
<u>Discard Non Inband Progress in Overlap Sending</u> *	False	False
<u>Disable Resume from Shared-line MGCP FXS Port</u> *	True	True

Figure 12: Service Parameters (Cont.)



<u>DTMF Silence Tone Flag</u> *	False	▼	False
<u>Enable Display IE in Codeset 6</u> *	False	▼	False
<u>Enable Sending PRI NI2 Service Message</u> *	False	▼	False
<u>Flash Hook Duration</u> *	500		500
<u>Gateway Poll Timer</u> *	10		10
<u>Location In PRI Progress Indicator IE (User Side Only)</u> *	Use the Network Side PRI progress indicator IE	▼	Use the Network Side PRI progress indicator IE
<u>Matching Calling Party with Attendant Flag</u> *	False	▼	False
<u>MGCP Database Query Delay Timer</u> *	1000		1000
<u>MGCP FXS On-Hook Pending Timer</u> *	3		3
<u>MGCP Response Timer</u> *	30		30
<u>MGCP Timer</u> *	3		3
<u>Numbering Plan Info</u> *	1		1
<u>Overlap Receiving Flag for PRI</u> *	True	▼	True
<u>Outgoing Media Connect Time for PRI</u> *	Connect ASAP	▼	Connect ASAP
<u>Port Release Timer</u> *	0		0
<u>SMDI Call Delay Timer</u> *	0		0
<u>Stable in State 4 Flag</u> *	False	▼	False
<u>Optimize MGCP Registration</u> *	True	▼	True
<u>Suppress Out-of-Channels Alarms</u> *	True	▼	True
<u>I-Frame Timer</u> *	2000		2000
<u>User-to-User IE Status</u> *	False	▼	False
<u>Convert European Progress Message to Alerting</u> *	False	▼	False
<u>Enable DMS PRI Notify Message from User to Network</u> *	True	▼	True
<u>Audit OOS Channels Interval</u> *	10		10
<u>Digital and Analog Ports Enabled</u> *	True	▼	True
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.			
- Clusterwide Parameters (Device - H323) -			
<u>Accept Unknown TCP Connection</u> *	False	▼	False
<u>BRO Enabled</u> *	False	▼	False
<u>Call Present Disconnect Flag</u> *	False	▼	False
<u>Check Progress Indicator Before Establishing Media</u> *	False	▼	False
<u>H225 Block Setup Destination</u> *	False	▼	False
<u>H225 DB Retry Timer</u> *	0		0
<u>H225 Device Connect Timer</u> *	0		0
<u>H225 DTMF Duration</u> *	100		100
<u>H225 TspReq Retry</u> *	2		2
<u>H225 Intercluster Call Throttle Timer</u> *	30	▼	30
<u>H225 T301 Timer</u> *	180000		180000
<u>H225 T302 Timer</u> *	15000		15000
<u>H225 T303 Timer</u> *	4000		4000

Figure 13: Service Parameters (Cont.)



<u>H225 T304 Timer</u> *	30000	30000
<u>H225 T305 Timer</u> *	30000	30000
<u>H225 T310 Timer</u> *	60000	60000
<u>H225 TCP Timer</u> *	5	5
<u>H245 TCS Timeout</u> *	10	10
<u>H323 Calling Party Number Screening Indicator</u> *	Calling number screened and passed	Calling number screened and passed
<u>Apply External Phone Number Mask for H.323 Calls</u> *	False	False
<u>Tone on Connect</u> *	False	False
<u>Wait Time for SDP with SR/RO Mode</u> *	3	3
<u>RAS ARQ Timer</u> *	3	3
<u>RAS BRQ Timer</u> *	3	3
<u>RAS DRQ Timer</u> *	3	3
<u>RAS RRQ Timer</u> *	3	3
<u>Ras URO Timer</u> *	3	3
<u>Retry Count for ARQ</u> *	2	2
<u>Retry Count for BRQ</u> *	2	2
<u>Retry Count for DRQ</u> *	2	2
<u>Retry Count for RRQ</u> *	2	2
<u>Retry Count for URQ</u> *	1	1
<u>Send Product ID and Version ID</u> *	False	False
<u>Send Unified CM Version as Version ID in H225Setup</u> *	False	False
<u>Send Progress Timer</u> *	3000	3000
<u>Send H225 User Info Message</u> *	User Info for Call Progress Tone	User Info for Call Progress Tone
<u>Status Enquiry Poll Timer</u> *	10000	10000
<u>Device Name of GK-controlled Trunk That Will Use Port 1720</u> *	None	None
<u>Host Name/IP Address of GK That Will Use RAS UDP Port 1719</u> *	None	None
<u>Fail Call If MTP Allocation Fails</u> *	False	False
<u>Overlap Receiving Flag for H323</u> *	False	False
<u>Allocate Transcoder for H.323 on Early Offer SIP Trunk for Calls with Early Media</u> *	False	False

Figure 14: Service Parameters (Cont.)



Clusterwide Parameters (Device - SIP)		
<u>SIP Interoperability Enabled</u> *	True	v True
<u>Retry Count for SIP Bye</u> *	10	10
<u>Retry Count for SIP Cancel</u> *	10	10
<u>Retry Count for SIP Invite</u> *	6	6
<u>Retry Count for SIP PRACK</u> *	6	6
<u>Retry Count for SIP Rel1XX</u> *	10	10
<u>Retry Count for SIP Publish</u> *	6	6
<u>Retry Count for SIP Response</u> *	6	6
<u>SIP Connect Timer</u> *	500	500
<u>SIP Disconnect Timer</u> *	500	500
<u>SIP Expires Timer</u> *	180000	180000
<u>SIP PRACK Timer</u> *	500	500
<u>SIP Rel1XX Timer</u> *	500	500
<u>SIP Trying Timer</u> *	500	500
<u>SIP Publish Timer</u> *	500	500
<u>SIP Min-SE Value</u> *	1800	1800
<u>SIPS URI Handling</u> *	Reject	v Reject
<u>SIP statistics Periodic update Timer</u> *	2	2
<u>SIP Session Expires Timer</u> *	1800	1800
<u>SIP Call Preservation Expires Timer</u> *	0	0
<u>SIP Trunk TspReq Retry</u> *	2	2
<u>SIP TCP Unused Connection Timer</u> *	14	14
<u>SIP TCP Timer</u> *	5	5
<u>SIP Station TCP Port Throttle Threshold</u> *	100	100
<u>SIP Trunk TCP Port Throttle Threshold</u> *	500	500
<u>SIP Station UDP Port Throttle Threshold</u> *	50	50
<u>SIP Trunk UDP Port Throttle Threshold</u> *	200	200
<u>SIP V.150 Outbound SDP Offer Filtering</u> *	No Filtering	v No Filtering
<u>Send SIP Multicast TTL in SDP</u> *	False	v False
<u>Default PUBLISH Expiration Timer</u> *	3600	3600
<u>Minimum PUBLISH Expiration Timer</u> *	60	60
<u>IM and Presence Publish Trunk</u>	< None >	v
<u>Send 181 Call Is Being Forwarded</u> *	False	v False
<u>Delay Sending 181 until 180/183 message is received</u> *	True	v True
<u>Fail Call Over SIP Trunk if MTP Allocation Fails</u> *	False	v False
<u>Log Call-Related REFER/NOTIFY /SUBSCRIBE SIP Messages for Session Trace</u> *	True	v True
<u>Port Received Timer for Outbound Call Setup</u> *	2	2
<u>SIP Registration Authorization Enabled</u> *	True	v True

Figure 15: Service Parameters (Cont.)



- 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	True
Call Park Reversion Timer *	<input type="text" value="60"/> 60	60
Park Monitoring Reversion Timer *	<input type="text" value="60"/> 60	60
Park Monitoring Periodic Reversion Timer *	<input type="text" value="30"/> 30	30
Park Monitoring Forward No Retrieve Timer *	<input type="text" value="300"/> 300	300
Preserve globalCallId for Parked Calls *	<input type="text" value="True"/> True	True
Maximum Call Duration Timer *	<input type="text" value="720"/> 720	720
Maximum Hold Duration Timer *	<input type="text" value="360"/> 360	360
Party Entrance Tone *	<input type="text" value="True"/> True	True
Message Waiting Lamp Policy *	<input type="text" value="Primary Line - Light and Prompt"/> Primary Line - Light and Prompt	Primary Line - Light and Prompt
Audible Message Waiting Indication Policy *	<input type="text" value="OFF"/> OFF	OFF
Message Waiting Indicator Inbound Calling Search Space	<input >="" none="" type="text" value="<"/> < None >	
Multiple Tenant MWI Modes *	<input type="text" value="False"/> False	False
MWI Non Message Center Signaling Call Duration *	<input type="text" value="0"/> 0	0
Message Waiting Indicator APDU Digit Translation CSS	<input >="" none="" type="text" value="<"/> < None >	
Block OffNet To OffNet Transfer *	<input type="text" value="False"/> False	False
Use Original Call Classification for Transferred Calls *	<input type="text" value="False"/> False	False
Use Restriction attribute of ID/Name Presentation of Transferring Party *	<input type="text" value="True"/> True	True
Local route group for redirected calls *	<input type="text" value="Local route group of calling party"/> Local route group of calling party	Local route group of calling party
Block Unencrypted Calls *	<input type="text" value="False"/> 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 *	<input type="text" value="True"/> True	True
Drop Ad Hoc Conference *	<input type="text" value="Never"/> Never	Never
Maximum Ad Hoc Conference *	<input type="text" value="4"/> 4	4
Maximum MeetMe Conference Unicast *	<input type="text" value="4"/> 4	4
Advanced Ad Hoc Conference Enabled *	<input type="text" value="True"/> True	False
Choose Encrypted Audio Conference Instead Of Video Conference *	<input type="text" value="True"/> True	True
Minimum Video Capable Participants To Allocate Video Conference *	<input type="text" value="2"/> 2	2
Enable Click-to-Conference for Third-Party Applications *	<input type="text" value="False"/> False	False
IMS Conference Factory URI *	<input type="text" value="cucm-conference-factory@cucm1.company.com"/> cucm-conference-factory@cucm1.company.com	cucm-conference-factory@cucm1.company.com
Cluster Conferencing Prefix Identifier	<input type="text"/>	

Figure 16: Service Parameters (Cont.)



Clusterwide Parameters (Feature - Call Secure Status Policy)		
Secure Call Icon Display Policy *	All media except BFCP and iX transports must be encr	v 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 *	True	v False
Forward By Reroute Enabled *	False	v False
Transform Forward by Reroute Destination *	True	v True
Always Forward Switch Voice Mail Calls *	True	v True
Forward By Reroute T1 Timer *	10	10
Include Original Called Info for Q.SIG Call Diversions *	Only after the first diversion	v Only after the first diversion
Set Private Numbering Plan for Call Forward *	False	v False
Set Type of Number for Call Forward *	Level1RegionalNumber	v Level1RegionalNumber
Max Forward UnRegistered Hops to DN *	0	0
CFA CSS Activation Policy *	With Configured CSS	v With Configured CSS
Cause Code When Maximum Forward Hop Count is Triggered *	Normal Unspecified	v 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	v False
Clusterwide Parameters (Feature - Call Pickup)		
Auto Call Pickup Enabled *	False	v 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	v Validate Except for Anonymous Users
Clusterwide Parameters (Feature - Replaces)		
Block OffNet To OffNet Replaces *	False	v False
Clusterwide Parameters (Feature - Redirection [3xx])		
Redirection Ring No Answer Reversion Timer *	24	24
Maximum Redirection Count *	70	70

Figure 17: Service Parameters (Cont.)



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
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
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

Figure 18: Service Parameters (Cont.)



- Clusterwide Parameters (Feature - Secure Tone)		
Play Tone to Indicate Secure/Non-Secure Call Status *	<input type="text" value="False"/> False	▼ False
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		
- Clusterwide Parameters (Feature - External Call Control)		
External Call Control Diversions	<input type="text" value="12"/> 12	
Maximum Hop Count *	<input type="text" value="12"/> 12	
Maximum External Call Control Diversion Hops to Pattern or DN *	<input type="text" value="12"/> 12	
External Call Control Routing Request Timer *	<input type="text" value="2000"/> 2000	2000
External Call Control Fully Qualified Role And Resource *	<input type="text" value="CISCO:UC:UCMPolicy:VoiceOrVideoCall"/> CISCO:UC:UCMPolicy:VoiceOrVideoCall	
External Call Control Initial Connection Count To PDP *	<input type="text" value="2"/> 2	
External Call Control Maximum Connection Count To PDP *	<input type="text" value="4"/> 4	
Always use External Call Control-specified Called/Calling Party Names *	<input type="text" value="True"/> True	▼ True
- Clusterwide Parameters (Route Plan)		
Stop Routing on Out of Bandwidth Flag *	<input type="text" value="False"/> False	▼ False
Stop Routing on Unallocated Number Flag *	<input type="text" value="True"/> True	▼ True
Stop Routing on User Busy Flag *	<input type="text" value="True"/> 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 *	<input type="text" value="True"/> True	▼ True
SIP Route Class Naming Authority *	<input type="text" value="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 *	<input type="text" value="False"/> False	▼ False
Use Pickup Group Of Line Group Member DN *	<input type="text" value="False"/> False	▼ False
- Clusterwide Parameters (External QoS)		
External QoS Enabled *	<input type="text" value="False"/> False	▼ False
- Clusterwide Parameters (Service)		
Default Network Hold MOH Audio Source ID *	<input type="text" value="1"/> 1	
Default User Hold MOH Audio Source ID *	<input type="text" value="1"/> 1	
Duplex Streaming Enabled *	<input type="text" value="True"/> True	▼ False
Media Exchange Interface Capability Timer *	<input type="text" value="8"/> 8	8
Send Multicast MOH in H.245 OLC Message *	<input type="text" value="True"/> True	▼ True
Media Exchange Timer *	<input type="text" value="12"/> 12	
Media Exchange Stop Streaming Timer *	<input type="text" value="8"/> 8	
Open Video Channel Response Timer for SIP Interop *	<input type="text" value="500"/> 500	500

Figure 19: Service Parameters (Cont.)



<u>Port Received Timer After Call Connection</u> *	500	500
<u>Media Resource Allocation Timer</u> *	12	12
<u>MTP and Transcoder Resource Throttling Percentage</u> *	95	95
<u>Intercluster Capabilities Mismatch Timer</u> *	1000	1000
<u>Silence Suppression</u> *	False	False
<u>Silence Suppression for Gateways</u> *	False	False
<u>Strip G.729 Annex B (Silence Suppression) from Capabilities</u> *	False	False
<u>Enable Source IP Address Verification for Software Media Devices</u> *	True	True
- Clusterwide Parameters (System - General) -		
<u>Always Use Dial Tone Setting</u> *	Default	Default
<u>Restart Cisco CallManager on Initialization Exception</u> *	True	True
<u>Digit Analysis Timer</u> *	6	6
<u>Statistics Enabled</u> *	True	True
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		
- Clusterwide Parameters (System - QoS) -		
<u>Priority Class</u> *	Normal Priority	Normal Priority
<u>DSCP for Audio Calls</u> *	46 (101110)	46 (101110)
<u>DSCP for Video Calls</u> *	34 (100010)	34 (100010)
<u>DSCP for Audio Portion of Video Calls</u> *	34 (100010)	34 (100010)
<u>DSCP for TelePresence Calls</u> *	32 (100000)	32 (100000)
<u>DSCP for Audio Portion of TelePresence Calls</u> *	32 (100000)	32 (100000)
<u>DSCP for Priority Audio Calls</u> *	45 (101101)	45 (101101)
<u>DSCP for Immediate Audio Calls</u> *	44 (101100)	44 (101100)
<u>DSCP for Flash Audio Calls</u> *	41 (101001)	41 (101001)
<u>DSCP for Flash Override Audio Calls</u> *	42 (101010)	42 (101010)
<u>DSCP for Executive Override Audio Calls</u> *	42 (101010)	42 (101010)
<u>DSCP for G.Clear Calls</u> *	46 (101110)	46 (101110)
<u>DSCP for Priority G.Clear Calls</u> *	45 (101101)	45 (101101)
<u>DSCP for Immediate G.Clear Calls</u> *	44 (101100)	44 (101100)
<u>DSCP for Flash G.Clear Calls</u> *	41 (101001)	41 (101001)
<u>DSCP for Flash Override G.Clear Calls</u> *	42 (101010)	42 (101010)
<u>DSCP for Executive Override G.Clear Calls</u> *	42 (101010)	42 (101010)
<u>DSCP for Audio Calls when RSVP Fails</u> *	0 (000000)	0 (000000)
<u>DSCP for Video Calls when RSVP Fails</u> *	0 (000000)	0 (000000)
<u>DSCP for ICCP Protocol Links</u> *	24 (011000)	24 (011000)

Figure 20: Service Parameters (Cont.)



Clusterwide Parameters (System - SDL)		
<u>SDL Listening Port Number</u> *	8002	8002
<u>SDL Max Router Latency</u> *	20	20
<u>Suppress Debug Info for Router Death</u> *	0	0
<u>Asynchronous SDL Logging Enabled</u> *	False	False
Clusterwide Parameters (System - Location and Region)		
<u>Enforce Millisecond Packet Size</u> *	True	True
<u>Locations Trace Details Enabled</u> *	False	False
<u>Preferred G.711 Millisecond Packet Size</u> *	20	20
<u>Preferred G.722 Millisecond Packet Size</u> *	20	20
<u>Preferred G.723.1 Millisecond Packet Size</u> *	30	30
<u>Preferred G.729 Millisecond Packet Size</u> *	20	20
<u>Always Use Preferred G.729 Packet Size For SIP Trunk Answers</u> *	False	False
<u>Preferred GSM EFR Bytes Packet Size</u> *	31	31
<u>G.711 A-law Codec Enabled</u> *	Enabled for All Devices	Enabled for All Devices
<u>G.711 mu-law Codec Enabled</u> *	Enabled for All Devices	Enabled for All Devices
<u>G.722 Codec Enabled</u> *	Enabled for All Devices	Enabled for All Devices
<u>iLBC Codec Enabled</u> *	Enabled for All Devices	Enabled for All Devices
<u>iSAC Codec Enabled</u> *	Enabled for All Devices	Enabled for All Devices
<u>Opus Codec Enabled</u> *	Enabled for All Devices	Enabled for All Devices
<u>Default Intraregion Max Audio Bit Rate</u> *	64 kbps (G.722, G.711)	64 kbps (G.722, G.711)
<u>Default Interregion Max Audio Bit Rate</u> *	8 kbps (G.729)	8 kbps (G.729)
<u>Default Intraregion Max Video Call Bit Rate (Includes Audio)</u> *	384	384
<u>Default Interregion Max Video Call Bit Rate (Includes Audio)</u> *	384	384
<u>Default Intraregion Max Immersive Video Call Bit Rate (Includes Audio)</u> *	2000000000	2000000000
<u>Default Interregion Max Immersive Video Call Bit Rate (Includes Audio)</u> *	2000000000	2000000000
<u>Use Video BandwidthPool for Immersive Video Calls</u> *	True	True
<u>Default Intraregion and Interregion Link Loss Type</u> *	Low Loss	Low Loss
<u>Default Audio Codec List between Regions</u> *	Factory Default low loss	Factory Default low loss
<u>Default Audio Codec List within Region</u> *	Factory Default low loss	Factory Default low loss
<u>Accept Audio Codec Preferences in Received Offer</u> *	Off	Off
<u>G.Clear Bandwidth Override</u> *	False	False
Clusterwide Parameters (System - CCM Automated Alternate Routing)		
<u>Automated Alternate Routing Enable</u> *	False	False

Figure 21: Service Parameters (Cont.)



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
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
Call Screening Timer *	4000	4000
Session Resumption Await Timer *	180	180

Figure 22: Service Parameters (Cont.)



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			
Enable Mobile Voice Access *	False	▼	False
Mobile Voice Access Number			
Matching Caller ID with Remote Destination *	Complete Match	▼	Complete Match
Number of Digits for Caller ID Partial Match *	10	10	
System Remote Access Blocked Numbers			
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 *	1000	1000	
User Control Confirmed Answer Indication Timer *	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 *	5	5	
Clusterwide Parameters (Call Admission Control)			
Call Counting CAC Enabled *	False	▼	False
Audio Bandwidth For Call Counting CAC *	102	102	
Video Bandwidth For Call Counting CAC *	500	500	
UCM to LBM Periodic Reservation Refresh Timer *	5	5	
Maximum Bandwidth Deduction Duration *	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
Deduct Audio Bandwidth Portion from Audio Pool for a Video Call *	False	▼	False
Clusterwide Parameters (Emergency Calling for Require Off-premise Location)			
Alternate Destination for Emergency Call			
Alternate Calling Search Space for Emergency Call	< None >	▼	

Figure 23: Service Parameters (Cont.)



Offnet Calls via G12 Communications SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and the G12 Communications network and calls are routed via CUBE.

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

- Name***: Non Secure SIP Trunk Profile, for example
- Description**: Non Secure SIP Trunk Profile authenticated by null String, for example

The screenshot shows the 'SIP Trunk Security Profile Information' configuration page. It includes fields for Name (Non Secure SIP Trunk Profile), Description (Non Secure SIP Trunk Profile authenticated by null String), Device Security Mode (Non Secure), Incoming Transport Type (TCP+UDP), and Outgoing Transport Type (UDP). Below these, there are several checkboxes for various SIP features, and a section for SIP V.150 Outbound SDP Offer Filtering with a dropdown menu set to 'Use Default Filter'. The 'Name' and 'Description' fields are highlighted with a red border.

Figure 24: SIP Trunk Security Profile

Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to G12 SBC should use UDP as a transport protocol for SIP. This is configured using the SIP Trunk Security profile, which is later assigned to the SIP trunk itself.



SIP Profile Configuration

SIP Profile will be later associated with the SIP trunk

Navigation: Device → Device Settings → SIP Profile

1. **Name***: G12 sip profile, for example
2. **Description**: G12 sip profile, for example

SIP Profile Information	
Name*	G12 sip profile
Description	G12 sip profile
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
User-Agent and Server header information*	Send Unified CM Version Information as User-Agent
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, and .
Confidential Access Level Headers*	Disabled
<input type="checkbox"/> Redirect by Application	
<input type="checkbox"/> Disable Early Media on 180	
<input type="checkbox"/> Outgoing T.38 INVITE include audio mline	
<input type="checkbox"/> Offer valid IP and Send/Receive mode only for T.38 Fax Relay	
<input type="checkbox"/> Use Fully Qualified Domain Name in SIP Requests	
<input type="checkbox"/> Assured Services SIP conformance	
<input type="checkbox"/> Enable External QoS**	
SDP Information	
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	< None >
Accept Audio Codec Preferences in Received Offer*	Default
<input checked="" type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change	
<input type="checkbox"/> Allow RR/RS bandwidth modifier (RFC 3556)	
Parameters used in Phone	
Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Media Port Ranges	<input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766

Figure 25: SIP Profile



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

Figure 26: SIP Profile (Cont.)



-Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*	Never
Resource Priority Namespace List	< None >
SIP Rel1XX Options*	Send PRACK if 1xx Contains SDP
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default
Session Refresh Method*	Invite
Early Offer support for voice and video calls*	Best Effort (no MTP inserted)

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
 Connect Inbound Call before Playing Queuing Announcement

-SIP OPTIONS Ping

<input checked="" type="checkbox"/> Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"
Ping Interval for In-service and Partially In-service Trunks (seconds)* 60
Ping Interval for Out-of-service Trunks (seconds)* 120
Ping Retry Timer (milliseconds)* 500
Ping Retry Count* 6

-SDP Information

<input type="checkbox"/> Send send-receive SDP in mid-call INVITE <input type="checkbox"/> Allow Presentation Sharing using BFCP <input type="checkbox"/> Allow iX Application Media <input type="checkbox"/> Allow multiple codecs in answer SDP
--

Figure 27: SIP Profile (Cont.)

Explanation

Parameter	Value	Description
Default MTP Telephony Event Payload Type	101	RFC2833 DTMF payload type
Early Offer support for voice and video calls	Best Effort (no MTP inserted)	Enable early media
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 CUBE

Navigation: Device → Trunk

Find and List Trunks												
<input type="button" value="Add New"/> <input type="button" value="Select All"/> <input type="button" value="Clear All"/> <input type="button" value="Delete Selected"/> <input type="button" value="Reset Selected"/>												
Trunks (1 - 10 of 10)												
Find Trunks where Device Name contains <input type="text" value="2"/> <input type="button" value="Find"/> <input type="button" value="Clear Filter"/>												
	Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile
<input type="checkbox"/>	Fax_GW2	fax		G711 Pool	7025				SIP Trunk	Full Service	Time In Full Service: 0 day 6 hours 17 minutes	Non Secure SIP Trunk Profile
<input type="checkbox"/>	G12_Trunk	G12_Trunk		G711 Pool	6.0				SIP Trunk	Full Service	Time In Full Service: 0 day 6 hours 17 minutes	Non Secure SIP Trunk Profile

Figure 28: SIP Trunks List

- Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	G12_Trunk
Description	G12_Trunk
Device Pool*	G711 Pool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_MTP
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default
<input checked="" type="checkbox"/> PSTN Access	
<input checked="" type="checkbox"/> Run On All Active Unified CM Nodes	

- Intercompany Media Engine (IME)

E.164 Transformation Profile	< None >
------------------------------	----------

Figure 29: SIP Trunk to CUBE



-MLPP and Confidential Access Level Information

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

-Call Routing Information

Remote-Party-Id
 Asserted-Identity
Asserted-Type* Default
SIP Privacy* Default

-Inbound Calls

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

-Incoming Calling Party Settings

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

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

-Incoming Called Party Settings

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

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

-Connected Party Settings

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

-Outbound Calls

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

-Caller Information

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

Figure 30: SIP Trunk to CUBE (Cont.)



SIP Information

Destination

Destination Address is an SRV

Destination Address	Destination Address IPv6	Destination Port	Sta
1 * 10.80.11.30		5060	u

MTP Preferred Originating Codec* 711ulaw

BLF Presence Group* Standard Presence group

SIP Trunk Security Profile* Non Secure SIP Trunk Profile

Rerouting Calling Search Space < None >

Out-Of-Dialog Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile* G12 sip profile [View Details](#)

DTMF Signaling Method* No Preference

Normalization Script

Normalization Script < None >

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 < None >

Geolocation Filter < None >

Send Geolocation Information

Figure 31: SIP Trunk to CUBE (Cont.)

Explanation

Parameter	Value	Description
Device Name	G12_Trunk	Name for the trunk
Device Pool	G711pool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_MTP	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.11.30	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	G12 sip profile	SIP Profile configured earlier



Dial Plan

Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

- Cisco IP phone dial “6”+10/11 digits number to access PSTN via Cisco UBE
 - “6” is removed before sending to Cisco UBE
- For FAX call, Access Code “6”+10/11 digits number is received from Cisco Fax Gateway
 - “6” is removed at Cisco UCM
 - The rest of the number is sent to G12 network via Cisco UBE
- Incoming fax call to 7025 will be sent to Cisco Fax gateway

Pattern	Description	Partition	Route Filter	Associated Device
6.@				G12_Trunk
7025				Fax_GW2

Figure 32: Route Patterns List

Pattern Definition	
Route Pattern*	6.@
Route Partition	< None >
Description	
Numbering Plan*	NANP
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	G12_Trunk
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error
Call Classification*	OffNet
External Call Control Profile	< None >
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	0
<input type="checkbox"/> Require Client Matter Code	
Calling Party Transformations	
<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	206395xxxx
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

Figure 33: Route Pattern for Voice



-Connected Party Transformations-

Connected Line ID Presentation* Default

Connected Name Presentation* Default

-Called Party Transformations-

Discard Digits PreDot

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type* Cisco CallManager

Called Party Numbering Plan* Cisco CallManager

-ISDN Network-Specific Facilities Information Element-

Network Service Protocol -- Not Selected --

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Figure 34: Route Pattern for Voice (Cont.)

Explanation

Setting	Value	Description
Route Pattern	6.@ for Voice Calls and 7025 for Fax Call	Specify appropriate Route Pattern
Gateway/Route List	Route Pattern 6.@ for call to G12 Communications and 7025 for SIP Trunk To Fax Gateway.	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 6. @	North American Numbering Plan
Use Calling Party's External Phone Number Mask	Checked	Send full external phone number for CLID
Calling Party Transform Mask	206395XXXX	Send the CLID for FAX caller
Discard Digits	PreDot for Route Pattern 6. @	Specifies how to modify digit before sent to G12 Communications network



Acronyms

Acronym	Definition
CPE	Customer Premise Equipment
Cisco UBE	Cisco Unified Border Element
Cisco UCM	Cisco Unified Communications Manager
MTP	Media Termination Point
POP	Point of Presence
PSTN	Public Switched Telephone Network
ESBC	Enterprise Session Border Controller
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol

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