



**Spectrum Enterprise SIP Trunking Service
Cisco Unified Communications Manager
10.5.1/Business Edition 6000 with Cisco Unified Border
Element [CUBE 10.5.0] - (IOS 15.4(3) M1)
IP PBX Configuration Guide**

About Spectrum Enterprise:

Spectrum Enterprise is a division of Charter Communications following a merger with Time Warner Cable and acquisition of Bright House Networks. Spectrum Enterprise is a national provider of scalable, fiber technology solutions. The Spectrum Enterprise portfolio includes networking and managed services solutions, including Internet access, Ethernet and Managed Network Services, Voice, TV and Cloud solutions. Our industry-leading team of experts works closely with clients to achieve greater business success.

About this document:

Spectrum Enterprise assures IP PBX compatibility by conducting interoperability testing to ensure any potential compatibility issues have been resolved prior to installation. Please review the IP PBX configuration instructions in this guide prior to your installation date.

Be advised that this document may contain references to Time Warner Cable Business Class. All references to Time Warner Cable Business Class, TWCBC or TWC should be read as Spectrum Enterprise.

Thank you,

Spectrum Enterprise



Time Warner Cable Business Class (TWCBC):

Connecting Cisco Unified Communications Manager
10.5.1/Business Edition 6000 with Cisco Unified Border Element
[CUBE 10.5.0] - (IOS 15.4(3) M1) using a TWCBC SIP Trunk

December 03, 2014



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Introduction

Time Warner Cable Business Class (TWCBC) SIP Trunks allows connection to the PSTN and offer the end customer a superior alternative to traditional PSTN connectivity. A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and TWCBC Enterprise SIP Gateway (ESG), Cisco Unified Border Element (CUBE) can be used. The Cisco Unified Border Element provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 10.5.1 connected to TWCBC IP network.

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

- This application note describes how to configure a Cisco Unified Communications Manager (Cisco UCM) 10.5.1 and Cisco Unified Border Element (Cisco UBE) 15.4(3) M1 for connectivity to TWCBC SIP Trunk service. The deployment model covered in this application note is CPE (Cisco UCM 10.5.1) to PSTN (TWCBC).
- Testing was performed using the approved Cisco test plan and among features verified were – basic calls, DTMF transport, Music on Hold, Semi-attendant and attendant transfers, call forward, conferences, and interoperability with Cisco Unity Connection
- The CUCM configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between TWCBC SIP network and Cisco Unified Communications. The configuration described in this document details the important configuration settings to have enabled for interoperability to be successful and care must be taken by the network administrator deploying Cisco UCM to interoperate to TWCBC SIP Trunking network.

This application note does not cover the use of calling search spaces (CSS) or partitions on Cisco Unified Communications Manager. To understand and learn how to apply CSS and partitions refer to the cisco.com link below: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/dialplan.html

Network Topology

Basic Call Setup

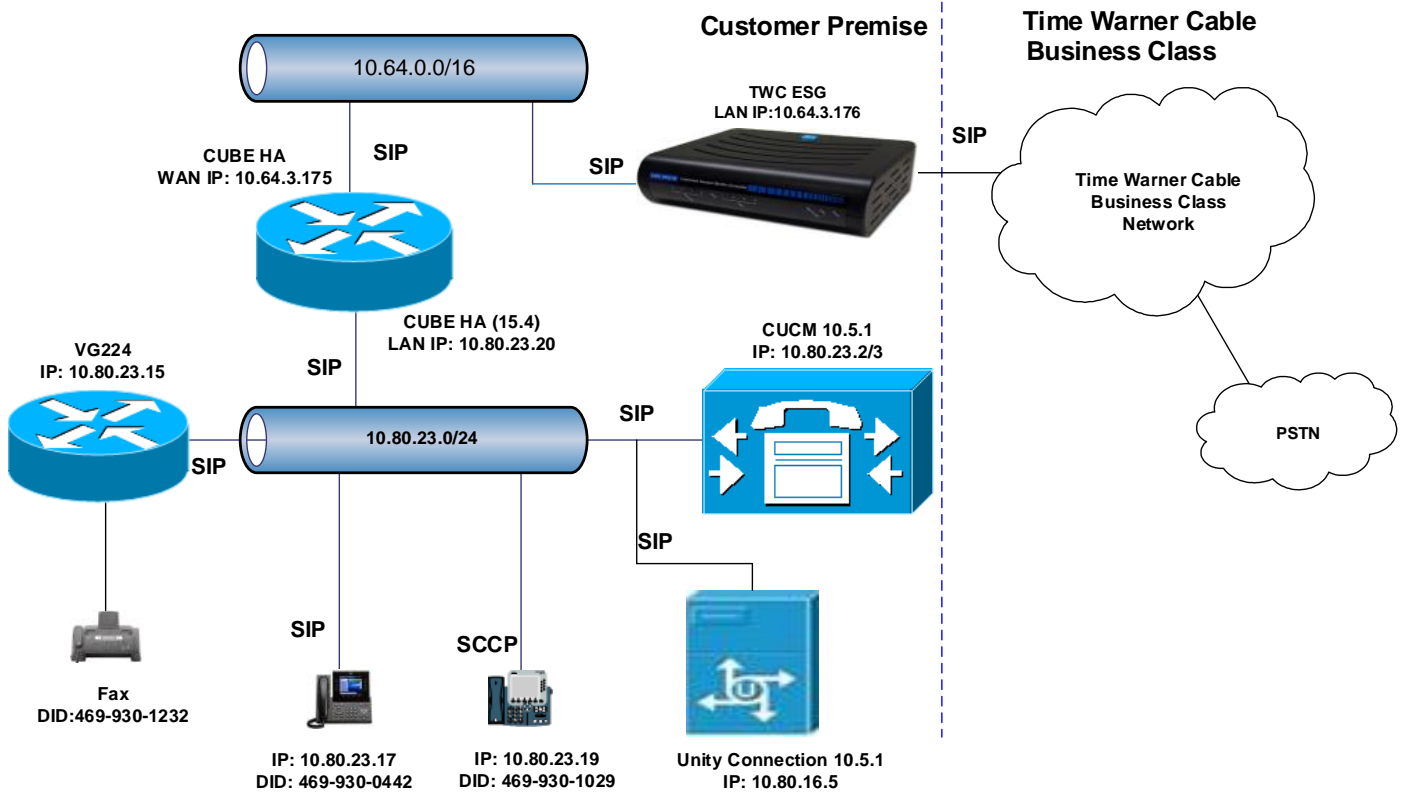


Figure 1 Network Topology

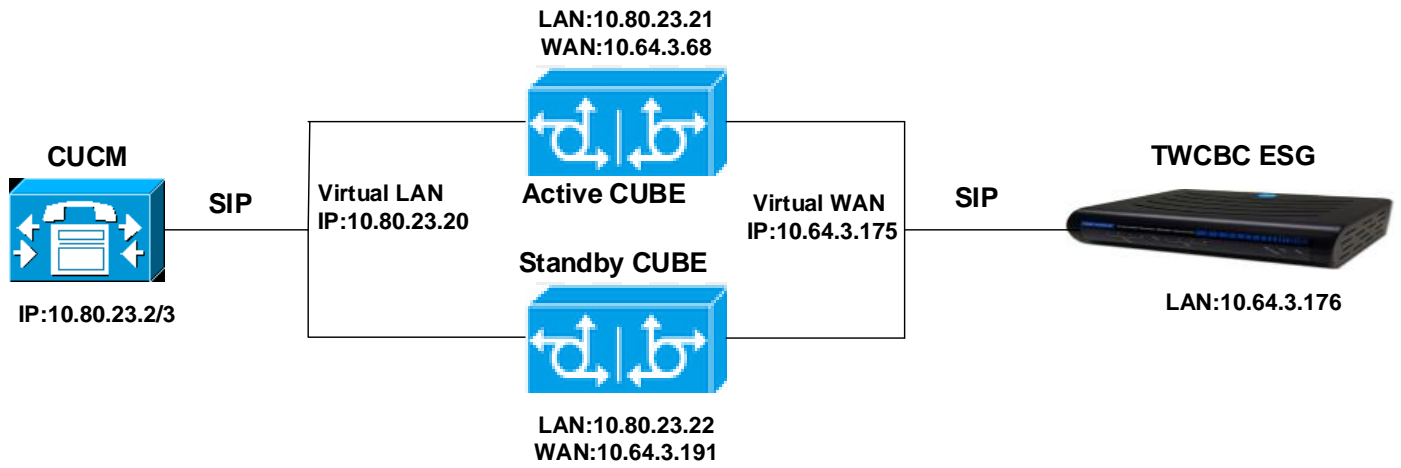


Figure 2: CUBE High Availability



System Components

Hardware Components

- Cisco UCM and Unity Connection on UCS C-240 running ESXi 5.5
- ISR G2 3945 routers (2 Routers were used for CUBE HA setup)
- IP phones 8900 and 7900 (different models, both SIP and SCCP where supported)
- Cisco Voice Gateways VG224

Software Requirements

- Cisco Unified Communications Manager 10.5.1.11901-1
- IOS 15.4(3) M1 for Cisco Unified Border Element
- IOS 15.1(3) T3 for VG224 Voice Gateways
- Cisco Unity Connection 10.5.1.10000-7

Features Supported

- Incoming and outgoing off-net calls using G711Ulaw (TWCBC only offer G711Ulaw) with 20ms packetization
- Call hold/Resume
- Call transfer (Semi-attendant and Attendant)
- Call conference
- Call forward (all, busy, no answer)
- Calling line (number) identification presentation (CLIP)
- Calling line (number) identification restriction (CLIR)
- DTMF (RFC2833)
- Media flow-through on CUBE
- Auto Attendant
- CUBE HA
- Fax G711 Pass-through
- Fax T38(Outbound only)

Features Not Supported

- Outbound SIP REFER with Replaces. Cisco UCM does not currently support generation of an outbound SIP REFER with Replaces message,
- Cisco IP phones used in this test only do Semi-attendant and Attendant transfer
- Though inbound T38 fax was successfully received during the test, TWCBC recommend to receive inbound fax with G711 pass-through only



Caveats

- It was observed that during semi-attendant transfer to off-net tests, the original caller does not hear ringback tone after PBX Extension completes the transfer and before target off-net phone answers. This is a known issue and was reported to Cisco before. This is not an issue with TWCBC SIP Trunk.
- ISR G2 will not support calls during switch over. During CUBE HA tests, existing active calls while call processing switching from Secondary CUBE back to Primary CUBE lost speech path. No issue for new calls after switchover.
- The Transcoding Profile Setting (under Telephone/Advanced section) in TWCBC ESG must be disabled to avoid audio issue in certain Hold call scenarios.
- 911 test case was not executed in this test.



Configuration

Configuring the Cisco Unified Border Element

Network Interface and CUBE HA

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—LAN and WAN.

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 15.1.2t or later are 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 10.80.23.21
remote-port 5000
remote-ip 10.80.23.22
```

Standby CUBE

```
ipc zone default
association 1
no shutdown
protocol sctp
local-port 5000
local-ip 10.80.23.22
remote-port 5000
remote-ip 10.80.23.21
```

Explanation

Command	Description
Ip 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 shutsown	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 10.64.3.176
  no ip address trusted authenticate
  address-hiding
mode border-element
allow-connections sip to sip
redundancy
!
..
redundancy inter-device
  scheme standby TWC
!
track 1 interface GigabitEthernet0/1
line-protocol
track 2 interface GigabitEthernet0/0
line-protocol

```

Standby CUBE

```

voice service voip
  ip address trusted list
    ipv4 10.64.3.176
  no ip address trusted authenticate
  address-hiding
mode border-element
allow-connections sip to sip
redundancy
!
..
redundancy inter-device
  scheme standby TWC
!
track 1 interfaceGigabitEthernet0/1
line-protocol
track 2 interfaceGigabitEthernet0/0
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	<p>Create a tracking list to track the line-protocol state of an interface</p> <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 TWC CUBE LAN
ip address 10.80.23.21 255.255.255.0
standby delay minimum 30 reload 60
standby version 2
standby 6 ip 10.80.23.20
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 GigabitEthernet0/1
description TWC CUBE WAN
ip address 10.64.3.68 255.255.0.0
standby delay minimum 30 reload 60
standby version 2
standby 1 ip 10.64.3.175
standby 1 priority 60
standby 1 preempt delay minimum 10
```

Standby CUBE

```
interface GigabitEthernet0/0
description TWC CUBE LAN
ip address 10.80.23.22 255.255.255.0
standby delay minimum 30 reload 60
standby version 2
standby 6 ip 10.80.23.20
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 GigabitEthernet0/1
description TWC CUBE WAN
ip address 10.64.3.191 255.255.0.0
standby delay minimum 30 reload 60
standby version 2
standby 1 ip 10.64.3.175
standby 1 priority 50
standby 1 preempt delay minimum 10
```



```
standby 1 name TWC
standby 1 track 1 decrement 20
standby 1 track 2 decrement 20
duplex auto
speed auto
```

```
standby 1 name TWC
standby 1 track 1 decrement 20
standby 1 track 2 decrement 20
duplex auto
speed auto
```

Explanation

Command	Description
Interface <i>type number</i>	Configures an interface type and enters interface configuration mode
Ip address <i>ip-addr mask</i>	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

Global CUBE settings

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

```
voice service voip
 ip address trusted list
  ipv4 10.64.3.176
 no ip address trusted authenticate
 address-hiding
 mode border-element
 allow-connections sip to sip
```



```
redundancy

fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback pass-
through g711ulaw

sip

rellxx supported "rell100"

session refresh

asserted-id pai

early-offer forced

midcall-signaling passthru
```

Explanation

Command	Description
ip address trusted list	Enters ip address trusted list mode and allows to manually add additional valid IP addresses
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

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

Default CUBE configuration enables CUBE to work in flow-through mode (this test use Flow-through mode). If you want to enable flow-around mode, please perform the following actions:

```
voice service voip

media flow-around
```

Codecs

TWCBC allow only G.711ulaw codec for voice calls.

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

```
voice class codec 1
```



```
codec preference 1 g711ulaw
```

Dial peer

CUCM uses dial-peer to route the call based on the digit to route the call accordingly.

```
! incoming voice call to CUCM
dial-peer voice 201 voip
  description to CUCM
  destination-pattern 469930....
  session protocol sipv2
  session target ipv4:10.80.23.3
  session transport udp
  voice-class codec 1
  voice-class sip bind control source-interface GigabitEthernet0/0
  voice-class sip bind media source-interface GigabitEthernet0/0
  dtmf-relay rtp-nte
  no vad
!
for outgoing calls to TWCBC ESG
dial-peer voice 100 voip
  description to TWC
  destination-pattern 1.....
  session protocol sipv2
  session target sip-server
  session transport udp
  voice-class codec 1
  voice-class sip bind control source-interface GigabitEthernet0/1
  voice-class sip bind media source-interface GigabitEthernet0/1
  dtmf-relay rtp-nte
  no vad
```



```
!  
dial-peer voice 300 voip  
  description to TWC-International  
  destination-pattern 011T  
  session protocol sipv2  
  session target sip-server  
  session transport udp  
  voice-class codec 1  
  voice-class sip bind control source-interface GigabitEthernet0/1  
  voice-class sip bind media source-interface GigabitEthernet0/1  
  dtmf-relay rtp-nte  
  no vad
```

```
!  
dial-peer voice 400 voip  
  description to TWC-special service  
  destination-pattern ...  
  session protocol sipv2  
  session target sip-server  
  session transport udp  
  voice-class codec 1  
  voice-class sip bind control source-interface GigabitEthernet0/1  
  voice-class sip bind media source-interface GigabitEthernet0/1  
  dtmf-relay rtp-nte  
  no vad
```

!
Call flow

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



For incoming PSTN calls, the CUBE presents the full ten-digit DID number to CUCM. The CUCM Translation Pattern strips all but the last four digits and routes the call based on those digits. Voice calls are routed to IP phones; fax calls are routed via a 4-digit route pattern to a SIP trunk that terminates on the VG224

CPE callers make outbound PSTN calls by dialing a “9” prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, VG224 sends to Cisco UCM the DID with leading access code “9”. A “9.@” 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 outbound Fax.

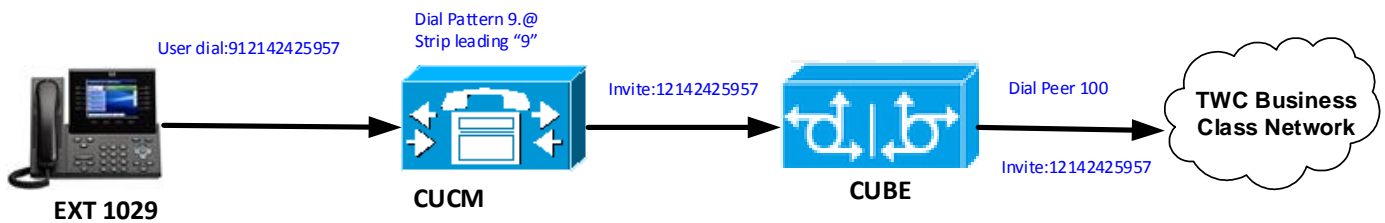


Figure 3: Outbound Voice Call

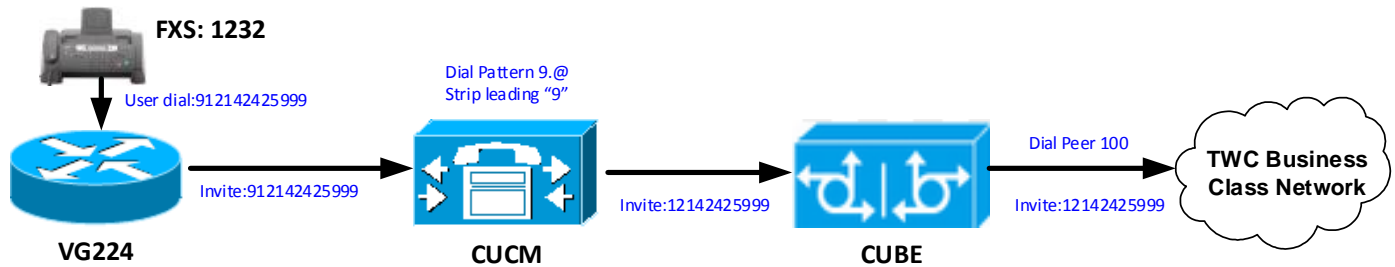


Figure 4: Outbound Fax Call

Configuration example

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

Active CUBE:

```
User Access Verification
login as: cisco
Using keyboard-interactive authentication.
Password:
```



```
TWC_CUBE1#show version
```

```
Cisco IOS Software, C3900e Software (C3900e-UNIVERSALK9-M), Version 15.4(3)M1,  
RELEASE SOFTWARE (fc1)
```

```
Technical Support: http://www.cisco.com/techsupport
```

```
Copyright (c) 1986-2014 by Cisco Systems, Inc.
```

```
Compiled Fri 24-Oct-14 22:33 by prod_rel_team
```

```
ROM: System Bootstrap, Version 15.1(1r)T5, RELEASE SOFTWARE (fc1)
```

```
TWC_CUBE1 uptime is 18 hours, 0 minutes
```

```
System returned to ROM by reload at 22:13:41 UTC Wed Dec 3 2014
```

```
System image file is "flash0:c3900e-universalk9-mz.SPA.154-3.M1.bin"
```

```
Last reload type: Normal Reload
```

```
Last reload reason: reload
```

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:
<http://www.cisco.com/wwl/export/crypto/tool/stqrg.html>



If you require further assistance please contact us by sending email to export@cisco.com.

Cisco CISCO3945-CHASSIS (revision 1.0) with C3900-SPE250/K9 with 1788928K/308224K bytes of memory.

Processor board ID FTX1744AM3X

4 Gigabit Ethernet interfaces

1 Virtual Private Network (VPN) Module

DRAM configuration is 72 bits wide with parity enabled.

256K bytes of non-volatile configuration memory.

4001760K bytes of ATA System CompactFlash 0 (Read/Write)

License Info:

License UDI:

```

-----
Device#      PID                SN
-----
*1           C3900-SPE250/K9   FOC17426ADY

```

Technology Package License Information for Module:'c3900e'

```

-----
Technology   Technology-package   Technology-package
              Current                Type                Next reboot

```



ipbase	ipbasek9	Permanent	ipbasek9
security	securityk9	EvalRightToUse	securityk9
uc	uck9	Permanent	uck9
data	None	None	None
NtwkEss	None	None	None
CollabPro	None	None	None

Configuration register is 0x2102

TWC_CUBE1#show run

Building configuration...

Current configuration : 8388 bytes

```
!  
version 15.4  
service timestamps debug datetime msec  
service timestamps log datetime msec  
service password-encryption  
!  
hostname TWC_CUBE1  
!  
boot-start-marker  
boot-end-marker  
!  
aqm-register-fnf  
!  
logging buffered 51200 warnings  
enable secret 4 tnhtc92DXBhelxjYk8LWJrPV36S2i4ntXrpb4RFmfqY
```



```
!  
!  
ipc zone default  
  association 1  
  no shutdown  
  protocol sctp  
  local-port 5000  
  local-ip 10.80.23.21  
  remote-port 5000  
  remote-ip 10.80.23.22  
!  
no aaa new-model  
!  
!  
ip name-server 10.64.1.3  
ip cef  
no ipv6 cef  
!  
!  
multilink bundle-name authenticated  
!  
password encryption aes  
cts logging verbose  
!  
crypto pki trustpoint TP-self-signed-2131491120  
  enrollment selfsigned  
  subject-name cn=IOS-Self-Signed-Certificate-2131491120  
  revocation-check none  
  rsakeypair TP-self-signed-2131491120
```



```
!  
!  
crypto pki certificate chain TP-self-signed-2131491120  
certificate self-signed 01  
3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030  
31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274  
69666963 6174652D 32313331 34393131 3230301E 170D3133 31313031 31363436  
31315A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649  
4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D32 31333134  
39313132 3030819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281  
8100AC64 457DC991 57967FE0 A1AD6097 4F1358E1 3721B264 13A1D71B 90556619  
D711054C F27B071E 91464C54 EACBD884 DC242E08 1BC34A7E 1FA49C2F 4A130BD1  
461AC476 BA1352B7 F54C4714 5990E43E 1FF4824D 8A75A247 F4AB488A 3F9EFD9C  
6CED7728 4CE96D86 B43594A1 6684B645 4302389A 99F337D9 5C04D4D6 ECD7BA8C  
1AEF0203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603  
551D2304 18301680 14B70ED5 6EF1FA77 9D2F8B0B 644BF4DE 972096BC 27301D06  
03551D0E 04160414 B70ED56E F1FA779D 2F8B0B64 4BF4DE97 2096BC27 300D0609  
2A864886 F70D0101 05050003 81810063 E882FC60 E29C53FE 5A982721 14405614  
B1A00023 124C03D7 677F2A10 178A4A9A B83448B1 EFBC136A 4080D4FC 493C3CDB  
623B6343 A3639AEB 2A7753B8 9DFB4C79 F3BF9E03 A3146AA0 11AA9FC1 9F739424  
2E4D57CB 78413BD3 10C790EE CBBBE796 A8490BE1 D0524A64 0259DC8B 91E6A14C  
6FAF8DB9 3139310F 425B3B8C 713265  
quit  
voice-card 0  
dsp services dspfarm  
!  
!  
no voice hunt unassigned-number  
!
```



```
voice service voip
  ip address trusted list
    ipv4 10.64.3.176
  no ip address trusted authenticate
  address-hiding
  mode border-element
  allow-connections sip to sip
  redundancy
  fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback pass-
through g711ulaw
  sip
    rellxx supported "rell100"
    session refresh
    asserted-id pai
    early-offer forced
    midcall-signaling passthru
!
voice class codec 1
  codec preference 1 g711ulaw
!
!
license udi pid C3900-SPE250/K9 sn FOC17426ADY
license boot module c3900e technology-package securityk9
!
!
hw-module pvdm 0/0
!
username cisco privilege 15 password 7 021201503D5715701C40
!
```



```
redundancy inter-device
  scheme standby TWC
!
!
redundancy
!
!
track 1 interface GigabitEthernet0/1 line-protocol
!
track 2 interface GigabitEthernet0/0 line-protocol
!
!
interface GigabitEthernet0/0
  description TWC CUBE LAN
  ip address 10.80.23.21 255.255.255.0
  standby delay minimum 30 reload 60
  standby version 2
  standby 6 ip 10.80.23.20
  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 GigabitEthernet0/1
  description TWC CUBE WAN
  ip address 10.64.3.68 255.255.0.0
  standby delay minimum 30 reload 60
```




```
standby version 2
standby 1 ip 10.64.3.175
standby 1 priority 60
standby 1 preempt delay minimum 10
standby 1 name TWC
standby 1 track 1 decrement 20
standby 1 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
no ip http secure-server
!
ip route 0.0.0.0 0.0.0.0 10.64.3.176
!
```



```
!  
nls resp-timeout 1  
cpd cr-id 1  
!  
!  
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 100 voip  
  description to TWC  
  destination-pattern 1.....  
  session protocol sipv2  
  session target sip-server  
  session transport udp  
  voice-class codec 1  
  voice-class sip bind control source-interface GigabitEthernet0/1  
  voice-class sip bind media source-interface GigabitEthernet0/1  
  dtmf-relay rtp-nte  
  no vad  
!  
dial-peer voice 101 voip  
  description from CUCM  
  session protocol sipv2
```



```
session target sip-server
session transport udp
incoming called-number 1.....
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 200 voip
description from TWC
session protocol sipv2
session target sip-server
session transport udp
incoming called-number 469930....
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 201 voip
description to CUCM
destination-pattern 469930....
session protocol sipv2
session target ipv4:10.80.23.3
session transport udp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0
```



```
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 300 voip
description to TWC-International
destination-pattern 011T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 400 voip
description to TWC-special service
destination-pattern ...
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 302 voip
```



```
description to TWC operator
destination-pattern 0
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
no vad
!
!
sip-ua
no remote-party-id
sip-server ipv4:10.64.3.176:5060
!
gatekeeper
shutdown
!
line con 0
exec-timeout 0 0
password 7 060506324F41
logging synchronous
login local
line aux 0
line vty 0 4
exec-timeout 0 0
password 7 1511021F0725
logging synchronous
```



```
login local
transport input all
line vty 5 15
exec-timeout 0 0
logging synchronous
login local
transport input all
!
scheduler allocate 20000 1000
ntp server 10.10.10.5
!
end
```

Standby CUBE:

User Access Verification

Username: cisco

Password:

TWC_CUBE2#show version

Cisco IOS Software, C3900e Software (C3900e-UNIVERSALK9-M), Version 15.4(3)M1, RELEASE SOFTWARE (fc1)

Technical Support: <http://www.cisco.com/techsupport>

Copyright (c) 1986-2014 by Cisco Systems, Inc.

Compiled Fri 24-Oct-14 22:33 by prod_rel_team

ROM: System Bootstrap, Version 15.1(1r)T4, RELEASE SOFTWARE (fc1)

TWC_CUBE2 uptime is 17 hours, 58 minutes

System returned to ROM by reload at 22:43:40 UTC Wed Dec 3 2014

System image file is "flash0:c3900e-universalk9-mz.SPA.154-3.M1.bin"



Last reload type: Normal Reload

Last reload reason: reload

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:
<http://www.cisco.com/wwl/export/crypto/tool/stqrg.html>

If you require further assistance please contact us by sending email to export@cisco.com.

Cisco CISCO3945-CHASSIS (revision 1.0) with C3900-SPE250/K9 with 1786880K/310272K bytes of memory.

Processor board ID FTX1541A032

4 Gigabit Ethernet interfaces

1 Virtual Private Network (VPN) Module

4 Voice FXS interfaces

DRAM configuration is 72 bits wide with parity enabled.

256K bytes of non-volatile configuration memory.

500472K bytes of ATA System CompactFlash 0 (Read/Write)



License Info:

License UDI:

```
-----  
Device#      PID                      SN  
-----  
*1           C3900-SPE250/K9         FOC15391VLH
```

Technology Package License Information for Module:'c3900e'

```
-----  
Technology    Technology-package      Technology-package  
              Current                Type                Next reboot  
-----  
ipbase        ipbasek9                Permanent           ipbasek9  
security      securityk9              RightToUse          securityk9  
uc            uck9                    Permanent           uck9  
data          None                    None                None  
NtwkEss       None                    None                None  
CollabPro     None                    None                None
```

Configuration register is 0x2102

TWC_CUBE2#show run



Building configuration...

Current configuration : 8418 bytes

```
!  
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.154-3.M1.bin  
boot-end-marker  
!  
aqm-register-fnf  
!  
logging buffered 51200 warnings  
enable secret 5 $!$q/Bk$Bu0l4yptT4JPxDeWSCcBd.  
!  
!  
ipc zone default  
  association 1  
  no shutdown  
  protocol sctp  
  local-port 5000  
  local-ip 10.80.23.22  
  remote-port 5000
```



```
remote-ip 10.80.23.21
!
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
```

```
!
```

```
!
```

```
no voice hunt unassigned-number
```

```
!
```

```
voice service voip
```

```
ip address trusted list
```

```
ipv4 10.64.3.176
```

```
no ip address trusted authenticate
```

```
address-hiding
```

```
mode border-element
```

```
allow-connections sip to sip
```

```
redundancy
```

```
fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback pass-  
through g711ulaw
```

```
sip
```

```
rellxx supported "rell100"
```

```
session refresh
```



```
asserted-id pai
early-offer forced
midcall-signaling passthru
!
voice class codec 1
  codec preference 1 g711ulaw
!
!
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 TWC
!
!
redundancy
!
track 1 interface GigabitEthernet0/1 line-protocol
!
track 2 interface GigabitEthernet0/0 line-protocol
!
!
interface GigabitEthernet0/0
```



```
description TWC CUBE LAN
ip address 10.80.23.22 255.255.255.0
standby delay minimum 30 reload 60
standby version 2
standby 6 ip 10.80.23.20
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 GigabitEthernet0/1
description TWC CUBE WAN
ip address 10.64.3.191 255.255.0.0
standby delay minimum 30 reload 60
standby version 2
standby 1 ip 10.64.3.175
standby 1 priority 50
standby 1 preempt delay minimum 10
standby 1 name TWC
standby 1 track 1 decrement 20
standby 1 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
no ip http secure-server
!
ip route 0.0.0.0 0.0.0.0 10.64.3.176
!
!
nls resp-timeout 1
cpd cr-id 1
!
!
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  
!  
!  
dial-peer voice 100 voip  
  description to TWC  
  destination-pattern 1.....  
  session protocol sipv2  
  session target sip-server  
  session transport udp  
  voice-class codec 1  
  voice-class sip bind control source-interface GigabitEthernet0/1  
  voice-class sip bind media source-interface GigabitEthernet0/1  
  dtmf-relay rtp-nte  
  no vad  
!  
dial-peer voice 101 voip  
  description from CUCM  
  session protocol sipv2  
  session target sip-server  
  session transport udp  
  incoming called-number 1.....
```



```
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 200 voip
description from TWC
session protocol sipv2
session target sip-server
session transport udp
incoming called-number 469930....
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 201 voip
description to CUCM
destination-pattern 469930....
session protocol sipv2
session target ipv4:10.80.23.3
session transport udp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
no vad
```




```
!  
dial-peer voice 300 voip  
  description to TWC-International  
  destination-pattern 011T  
  session protocol sipv2  
  session target sip-server  
  session transport udp  
  voice-class codec 1  
  voice-class sip bind control source-interface GigabitEthernet0/1  
  voice-class sip bind media source-interface GigabitEthernet0/1  
  dtmf-relay rtp-nte  
  no vad
```

```
!  
dial-peer voice 400 voip  
  description to TWC-special service  
  destination-pattern ...  
  session protocol sipv2  
  session target sip-server  
  session transport udp  
  voice-class codec 1  
  voice-class sip bind control source-interface GigabitEthernet0/1  
  voice-class sip bind media source-interface GigabitEthernet0/1  
  dtmf-relay rtp-nte  
  no vad
```

```
!  
dial-peer voice 302 voip  
  description to TWC operator  
  destination-pattern 0  
  session protocol sipv2
```



```
session target sip-server
session transport udp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
no vad
!
!
sip-ua
no remote-party-id
sip-server ipv4:10.64.3.176:5060
!
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 displays the Cisco Unified CM Administration web interface. At the top, the Cisco logo and 'Cisco Unified CM Administration' are visible, along with navigation links for 'administrator', 'Search Documentation', 'About', and 'Logout'. A menu bar includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The main content area features a blue header with 'Cisco Unified CM Administration' and a highlighted box containing 'System version: 10.5.1.11901-1'. Below this, hardware details are listed: 'VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz, disk 1: 80Gbytes, 4096Mbytes RAM, Partitions aligned'. A sidebar image shows a server room. The footer contains the last successful logon date, copyright information, and legal notices regarding cryptographic features and U.S. laws.

Figure 5 CUCM Version



Cisco CallManager Service Parameter

Go to **System > Service Parameters** .we leave all fields in the service parameter as default values for this test

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Service Parameter Configuration | Related Links: Parameters for All Servers

Save | Set to Default | Advanced

Status
Status: Ready

Select Server and Service
Server*: clus33pub--CUCM Voice/Video (Active)
Service*: Cisco CallManager (Active)
All parameters apply only to the current server except parameters that are in the cluster-wide group(s).

Cisco CallManager (Active) Parameters on server clus33pub--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.		
SDL Trace		
SDL Trace Data Flags *	0x00000111	0x00000111
SDL Trace Flush Immediately *	False	False
SDL Trace Data Size *	0	0
SDL Trace Flag *	True	True
SDL TraceType Flags *	0x8000EB15	0x8000EB15
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		

Figure 6 Service Parameter



Clusterwide Parameters (Device - General)

Call Diagnostics Enabled *	Disabled	Disabled
Show Line Group Member DN in finalCalledPartyNumber CDR Field *	False	False
Show Line Group Member Non Masked DN in finalCalledPartyNumber CDR Field *	False	False
CTI New Call Accept Timer *	4	4
CTI Generate Digits Interval *	250	250
CTI Dial Digits Interval *	250	250
CTI Await Further Digits *	False	False
CTI Use Wildcard Pattern as calledPartyDN *	False	False
Retain Media on Disconnect with PI for Active Call *	False	False
Station and Backup Server KeepAlive Interval *	60	60
Station KeepAlive Interval *	30	30
Status Enquiry Poll Flag *	False	False
Strip # Sign from Called Party Number *	True	True
Session Handoff Alerting Timer *	10	10
T301 Timer *	180000	180000
T302 Timer *	15000	15000
T303 Timer *	4000	4000
T304 Timer *	30000	30000
T305 Timer *	30000	30000
T306 Timer *	30000	30000
T308 Timer *	4000	4000
T309 Timer *	90000	90000
T310 Timer *	60000	60000
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 even if Translated	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

Figure 7 Service Parameter cont.



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
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 Incomin	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 - Defau	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
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

Figure 8 Service Parameter cont.



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
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
Send Unified CM Version as Version ID in H225Setup *	False	False
Send Progress Timer *	3000	3000

Figure 9 Service Parameter cont.



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

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

Figure 10 Service Parameter cont.



Clusterwide Parameters (Feature - General)		
Call Park Display Timer *	10	10
Caller ID Display Priority Enabled *	True	True
Call Park Reversion Timer *	60	60
Park Monitoring Reversion Timer *	60	60
Park Monitoring Periodic Reversion Timer *	30	30
Park Monitoring Forward No Retrieve Timer *	300	300
Preserve globalCallId for Parked Calls *	True	True
Maximum Call Duration Timer *	720	720
Maximum Hold Duration Timer *	360	360
Party Entrance Tone *	True	True
Message Waiting Lamp Policy *	Primary Line - Light and Prompt	Primary Line - Light and Prompt
Audible Message Waiting Indication Policy *	OFF	OFF
Message Waiting Indicator Inbound Calling Search Space	< None >	
Multiple Tenant MWI Modes *	False	False
MWI Non Message Center Signaling Call Duration *	0	0
Message Waiting Indicator APDU Digt 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		
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
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

Figure 11 Service Parameter cont.



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
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

Figure 12 Service Parameter cont.



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
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
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 *	False	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

Figure 13 Service Parameter cont.



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)
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
Preferred G.729 Millisecond Packet Size *	20	20

Figure 14 Service Parameter cont.



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
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

Figure 15 Service Parameter cont.



Enterprise Feature Access Code for Transfer *	<input type="text" value="*84"/>	*84
Enterprise Feature Access Code for Conference *	<input type="text" value="*85"/>	*85
Enterprise Feature Access Code for Session Handoff *	<input type="text" value="*74"/>	*74
Enterprise Feature Access Code for Starting Selective Recording *	<input type="text" value="*86"/>	*86
Enterprise Feature Access Code for Stopping Selective Recording *	<input type="text" value="*87"/>	*87
Smart Mobile Phone Interdigit Timer *	<input type="text" value="500"/>	500
Non-Smart Mobile Phone Interdigit Timer *	<input type="text" value="2000"/>	2000
Send Call to Mobile Menu Timer *	<input type="text" value="60"/>	60
SIP Dual Mode Alert Timer *	<input type="text" value="1500"/>	1500
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 *	<input type="text" value="Trunk or Gateway Inbound Calling Search Space"/>	Trunk or Gateway Inbound Calling Search Space
Enable Enterprise Feature Access *	<input type="text" value="False"/>	False
Dial-via-Office Forward Service Access Number	<input type="text"/>	
Enable Mobile Voice Access *	<input type="text" value="False"/>	False
Mobile Voice Access Number	<input type="text"/>	
Matching Caller ID with Remote Destination *	<input type="text" value="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 *	<input type="text" value="False"/>	False
Honor Gateway or Trunk Outbound Calling Party Selection for Mobile Connect Calls *	<input type="text" value="False"/>	False
Clusterwide Parameters (System - Mobility Single Number Reach Voicemail)		
Single Number Reach Voicemail Policy *	<input type="text" value="Timer Control"/>	Timer Control
Dial-via-Office Reverse Voicemail Policy *	<input type="text" value="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 *	<input type="text" value="False"/>	False
Ring All Shared Lines *	<input type="text" value="False"/>	False
Ignore Call Forward All on Enterprise DN *	<input type="text" value="True"/>	True
Clusterwide Parameters (Feature - Immediate Divert)		
Use Legacy Immediate Divert *	<input type="text" value="True"/>	True
Allow QSIG during iDivert *	<input type="text" value="False"/>	False
Immediate Divert User Response Timer *	<input type="text" value="5"/>	5
Clusterwide Parameters (Call Admission Control)		
Call Counting CAC Enabled *	<input type="text" value="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 *	<input type="text" value="Allow Calls"/>	Allow Calls
Locations Media Resource Audio Bit Rate Policy *	<input type="text" value="Lowest Bit Rate"/>	Lowest Bit Rate
Video Call QoS Marking Policy *	<input type="text" value="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	<input type="text" value="< None >"/>	

Figure 16 Service Parameter cont.



Off-net calls via TWCBC SIP Trunk

Off-net calls are served by SIP trunks configured between CUCM and TWCBC ESG. Calls are routed via CUBE.

SIP Trunk Security Profile

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

The screenshot shows the 'SIP Trunk Security Profile Configuration' page in Cisco Unified CM Administration. The 'Status' is 'Ready'. The 'SIP Trunk Security Profile Information' section contains the following fields:

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

Figure 17 SIP Trunk Security Profile

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to TWCBC ESG 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.



SIP Profile

SIP Profile will be later associated with the SIP trunk.

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

The screenshot shows the Cisco Unified CM Administration interface for SIP Profile Configuration. The page title is "SIP Profile Configuration" and it includes a navigation menu at the top with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The user is logged in as "administrator".

Status

- Status: Ready
- All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information

Name*	TWC SIP Trunk Profile
Description	TWC SIP Trunk 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-Agen
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, anc
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
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

Figure 18 SIP Profile



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			<input type="button" value="+"/> <input type="button" value="-"/>

Incoming Requests FROM URI Settings

Caller ID DN

Caller Name

Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on* Never

RSVP Over SIP* Local RSVP

Resource Priority Namespace List < None >

Fall back to local RSVP

SIP 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

Figure 19 SIP Profile Cont.



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

Save Delete Copy Reset Apply Config Add New

Figure 20: SIP Profile cont.

Parameter	Value	Description
Default MTP Telephony Event Payload Type	101	RFC2833 DTMF payload type
Require SDP Inactive Exchange for Mid-Call Media Change	Checked	Send SDP with Inactive when call on hold
SIP Rel1XX Options	Send PRACK for 1xx Messages	Enable Provisional Acknowledgements (Reliable 100 messages)
Early Offer support for voice and video calls	Best Effort (no MTP inserted)	Support early media
Enable OPTIONS Ping to monitor destination status for trunks with Service Type "None (Default)"	Checked	Send OPTIONS Ping to CUBE
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 TWCBC by navigating to **Device > Trunk** and clicking **Add New** button. Same apply to create SIP trunks to Cisco Unity Connection and VG224

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | Go
administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Find and List Trunks

+ Add New | Select All | Clear All | Delete Selected | Reset Selected

Status
8 records found

Trunks (1 - 8 of 8) Rows per Page: 50

Find Trunks where: Device Name | begins with | Find | 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"/>	TWC SIP Trunk	TWC SIP Trunk		G711	*769.@				SIP Trunk	Full Service	Time In Full Service: 1 day 21 hours 16 minutes	TWC SIP Trunk Profile
<input type="checkbox"/>	TWC SIP Trunk	TWC SIP Trunk		G711	9.@				SIP Trunk	Full Service	Time In Full Service: 1 day 21 hours 16 minutes	TWC SIP Trunk Profile
<input type="checkbox"/>	TWC SIP Trunk	TWC SIP Trunk		G711	9.0#				SIP Trunk	Full Service	Time In Full Service: 1 day 21 hours 16 minutes	TWC SIP Trunk Profile
<input type="checkbox"/>	Unity Connection			G711	1000				SIP Trunk	Full Service	Time In Full Service: 8 days 1 hour 49 minutes	Unity SIP Trunk Profile
<input type="checkbox"/>	VG224	FAX_VG224		G711	1232				SIP Trunk	Full Service	Time In Full Service: 1 day 21 hours 17 minutes	TWC SIP Trunk Profile

+ Add New | Select All | Clear All | Delete Selected | Reset Selected

Figure 21 SIP Trunks List



Cisco Unified CM Administration
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System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

Trunk Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status
Status: Ready

SIP Trunk Status
Service Status: Full Service
Duration: Time In Full Service: 1 day 21 hours 50 minutes

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	TWC_SIP_Trunk
Description	TWC SIP Trunk
Device Pool*	G711
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	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
 Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)
E.164 Transformation Profile < None >

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

Figure 22 SIP Trunk to CUBE



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"/>

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 23 SIP Trunk to CUBE Cont.



SIP Information

Destination

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status
1*	10.80.23.20		5060	up

MTP Preferred Originating Codec* 711ulaw

BLF Presence Group* Standard Presence group

SIP Trunk Security Profile* TWC 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* TWC SIP Trunk Profile [View Details](#)

DTMF Signaling Method* RFC 2833

Normalization Script

Normalization Script TWC_11Digits_Diversion

Enable Trace

	Parameter Name	Parameter Value
1	Diversion-Mask	1469930XXXX

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

Save Delete Reset Add New

Figure 24 SIP Trunk to CUBE Cont.

Parameter	Value	Description
Device Name	TWC_SIP_Trunk	Name for the trunk
Device Pool	G711	G711 Pool used to use 711Ulaw as preferred voice codec
Media Resource Group List	MRGL	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	All	Received all digits for incoming call and Translation Pattern strips all but the last four digits and routes the call based on those digits
Calling Party Selection	Originator	Send original caller ID
Redirecting Diversion Header Delivery - outbound	Checked	Adding Diversion Header for calls outbound from site
Destination Address	10.80.23.20	Virtual LAN IP address of the CUBE



SIP Trunk Security Profile	TWC Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	TWC SIP Trunk Profile	SIP Profile configured earlier
DTMF Signaling Method	RFC 2833	RFC 2833 is supported for DTMF transport to/from TWCBC
Normalization Script	TWC_11Digits_Diversion	Convert 4 digits EXT to 11 digits DID for Diversion header
Diversion-Mask	1469930XXXX	Used in Normalization Script

Note: Reset the trunk after the configuration is completed.

Apply same procedure to create SIP trunks to Cisco Unity Connection and VG224



SIP Normalization Script

A SIP Normalization Script is used to convert SIP Diversion Headers from 4-digit EXT to the full 11-digit E.164 telephone number, this is required for call redirecting over TWCBC SIP network.

Navigate to Device>Device Settings>SIP Normalization Script to create Normalization Script

The screenshot shows the Cisco Unified CM Administration interface for configuring a SIP Normalization Script. The page title is "SIP Normalization Script Configuration". The status is "Ready". The configuration fields are as follows:

Name*	TWC_11Digits_Diversion
Description	convert 4 digits diversion to 11 digits
Content*	<pre>M = {} local mask = scriptParameters.getValue("Diversion-Mask") -- handle the mask of the diversion header for non-911 calls function M.outbound_INVITE(message) if mask then message:applyNumberMask("Diversion", mask) end end return M</pre>
Script Execution Error Recovery Action*	Message Rollback Only
System Resource Error Recovery Action*	Disable Script
Memory Threshold*	50 kilobytes
Lua Instruction Threshold*	1000 instructions

Figure 25 Normalization Script

SIP Normalization Script (Text)

```
M = {}  
local mask = scriptParameters.getValue("Diversion-Mask")  
  
-- handle the mask of the diversion header for non-911 calls  
function M.outbound_INVITE(message)  
  if mask  
  then  
    message:applyNumberMask("Diversion", mask)  
  end  
end  
return M
```



Translation Pattern

A Translation Pattern is created to convert 10-digit Incoming Called Number to 4-digit Extension

Navigate to **Call Routing**>**Translation Pattern** and press **ADD New** button to create Translation Patterns

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Translation Pattern Configuration Related Links: [Back To Find/List](#) ▾ Go

Save Delete Copy Add New

Status: Ready

Pattern Definition

Translation Pattern	469930.XXXX
Partition	< None >
Description	
Numbering Plan	< None >
Route Filter	< None >
MLPP Precedence*	Default
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Calling Search Space	< None >
<input type="checkbox"/> Use Originator's Calling Search Space	
External Call Control Profile	< None >
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error
<input checked="" type="checkbox"/> Provide Outside Dial Tone	
<input checked="" type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Do Not Wait For Interdigit Timeout On Subsequent Hops	
<input type="checkbox"/> Route Next Hop By Calling Party Number	

Calling Party Transformations

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	PreDot
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager

Save Delete Copy Add New

Figure 26 Translation Pattern



Dialplan

Route Pattern configuration

Route patterns are configured as below, Cisco IP phones dial 9+11 digits number to access PSTN via CUBE, “9” is removed before send to CUBE; for FAX call, Access Code 9 is used at VG224, “9” is removed at UCM and 11 digits number is send to CUBE to TWCBC network. Incoming fax call to 1232 will send to VG224. 1000 is the Pilot Number for Voice mail to Unity Connection.

Navigate to **Call Routing > Route/Hunt > Route Pattern** and press **Add New** button to create Route Patterns

The screenshot shows the Cisco Unified CM Administration interface for configuring Route Patterns. The page title is "Find and List Route Patterns". Below the title, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected". A status bar indicates "7 records found". The main content area shows a table of route patterns with the following data:

Pattern	Description	Partition	Route Filter	Associated Device	Copy
*769.@	Anonymous			TWC SIP Trunk	
1000				Unity Connection	
1232	fax			VG224	
9.0#				TWC SIP Trunk	
9.@				TWC SIP Trunk	

At the bottom of the table, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

Figure 27 Route Pattern



System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

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

Save Delete Copy Add New

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan*

Route Filter

MLPP Precedence*

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

Route Class*

Gateway/Route List* (Edit)

Route Option
 Route this pattern
 Block this pattern

Call Classification*

External Call Control Profile

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>	<input type="text"/>

Figure 28 Route Pattern for Voice



System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Pattern Configuration

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

Save Delete Copy Add New

Pattern Definition

Route Pattern*	1000
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*	Unity_Connection (Edit)

Route Option

Route this pattern
 Block this pattern

Call Classification*

External Call Control Profile

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>	<input type="text"/>

Figure 29 Route Pattern Unity Connection



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

Save Delete Copy Add New

Pattern Definition

Route Pattern*	1232
Route Partition	< None >
Description	fax
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*	VG224 (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input type="text" value="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	
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 30 Route Pattern for Fax



Setting	Value	Description
Route Pattern	9.@ for outbound call	Specify appropriate Route Pattern
Gateway/Route List	TWC_SIP_Trunk	SIP Trunk name configured earlier
Require Forced Authorization Code	Checked when doing Authorization Code test	Specify if Authorization Code required when make call through this Route Pattern
Require Client Matter Code	Check when doing Account Code test	Specify if Account Code required when make call through this Route Pattern
Use Calling Party's External Phone Number Mask	Checked	Send the Calling Party information based on the configuration for each phone
Calling Party Transform mask	1469930XXXX	Specify the Calling Line ID for outgoing call through this Route Pattern, TWCBC require 11-digit Calling Line ID
Discard Digits	PreDot for RP 9.@	specifies how to modify digit before they are sending to TWCBC ESG



Configuring the Cisco Voice Gateway VG224

The following configuration snippet contains a sample configuration of Cisco Voice Gateway VG224 for fax services.

```
VG224#show run
```

```
Building configuration...
```

```
Current configuration : 2308 bytes
```

```
!  
! Last configuration change at 19:21:00 UTC Sat Mar 6 1993 by cisco  
!  
version 15.1  
no service pad  
service timestamps debug datetime msec  
service timestamps log datetime msec  
no service password-encryption  
!  
hostname VG224  
!  
boot-start-marker  
boot-end-marker  
!  
!  
enable secret 5 $1$2vXb$mom3hjaQF.cY7CZ0YP3Oo.  
!  
no aaa new-model  
crypto pki token default removal timeout 0  
!  
!  
ip source-route
```




```
ip cef
!
!
!
no ipv6 cef
!
!
voice service voip
    allow-connections sip to sip
    redirect ip2ip
    fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback pass-
through g711ulaw
    sip
        asserted-id pai
        early-offer forced
        midcall-signaling passthru
!
voice class codec 1
    codec preference 1 g711ulaw
!
!
voice-card 0
!
username cisco privilege 15 password 0 tekV1z10n
!
!
interface FastEthernet0/0
    ip address 10.80.23.15 255.255.255.0
    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 10.80.23.1
!
!
!
control-plane
!
!
voice-port 2/0
!
voice-port 2/1
description **telephone analog/fax**
!
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 1232 pots
  destination-pattern 1232
  incoming called-number [0-9]T
  no digit-strip
  port 2/1
  forward-digits 0
!
dial-peer voice 100 voip
  description outbound call
  destination-pattern 91.....
  session protocol sipv2
```



```
session target ipv4:10.80.23.3:5060
session transport udp
voice-class codec 1
dtmf-relay rtp-nte
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
CPE	Customer Premise Equipment
CUBE	Cisco Unified Border Element
CUCM	Cisco Unified Communications Manager
ESG	Enterprise SIP Gateway
MTP	Media Termination Point
POP	Point of Presence
PSTN	Public Switched Telephone Network
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol
TWCBC	Time Warner Cable Business Class



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Appendix A: Test Results (Test results will be kept on file at Cisco, but will be stripped out of the application note before publishing to Cisco.com)



SP_SIP_master_testp
lan_V1.2.xls



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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