



COLT Communications SIP Trunking: Connecting Cisco Unified Communications Manager (CUCM) 10.5.1 via the Cisco Unified Border Element (CUBE) 10.5 using SIP

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Note: Testing was conducted in tekVizion Labs.



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Introduction

Service Providers today, such as COLT Communications, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. COLT SIP Trunking is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity via either Analog or T1 lines. A demarcation device between these services and customer owned services is recommended. The Cisco Unified Border Element provides demarcation, security, interworking and session management services.

This application note describes how to configure a Cisco Unified Communications Manager (Cisco UCM) 10.5(1) with a Cisco Unified Border Element (Cisco UBE) for connectivity to the COLT SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 10.5(1)/Cisco UBE) to PSTN via COLT SIP Trunking.

Testing was performed in accordance to Cisco's Service Provider SIP Trunk Validation Test Plan and all features were verified. Key features verified are:

- Basic Calls
- Basic Calls with Calling Name and Number as allowed or restricted
- DTMF Relay
- Call Conference (Intra-site, PSTN)
- Call Transfer (Blind, Attended, Early Attended)
- Hold and Resume
- Voice Mail
- T.38 Fax G3/SG3
- Simultaneous Calls
- Auto Attendant
- International Calls
- G.711 Fax G3/SG3
- Call Forwarding – Find Me (Unconditional, Busy, No Reply)
- Codec negotiation
- Dial Plans
- PRACK with SDP early-media cut-through

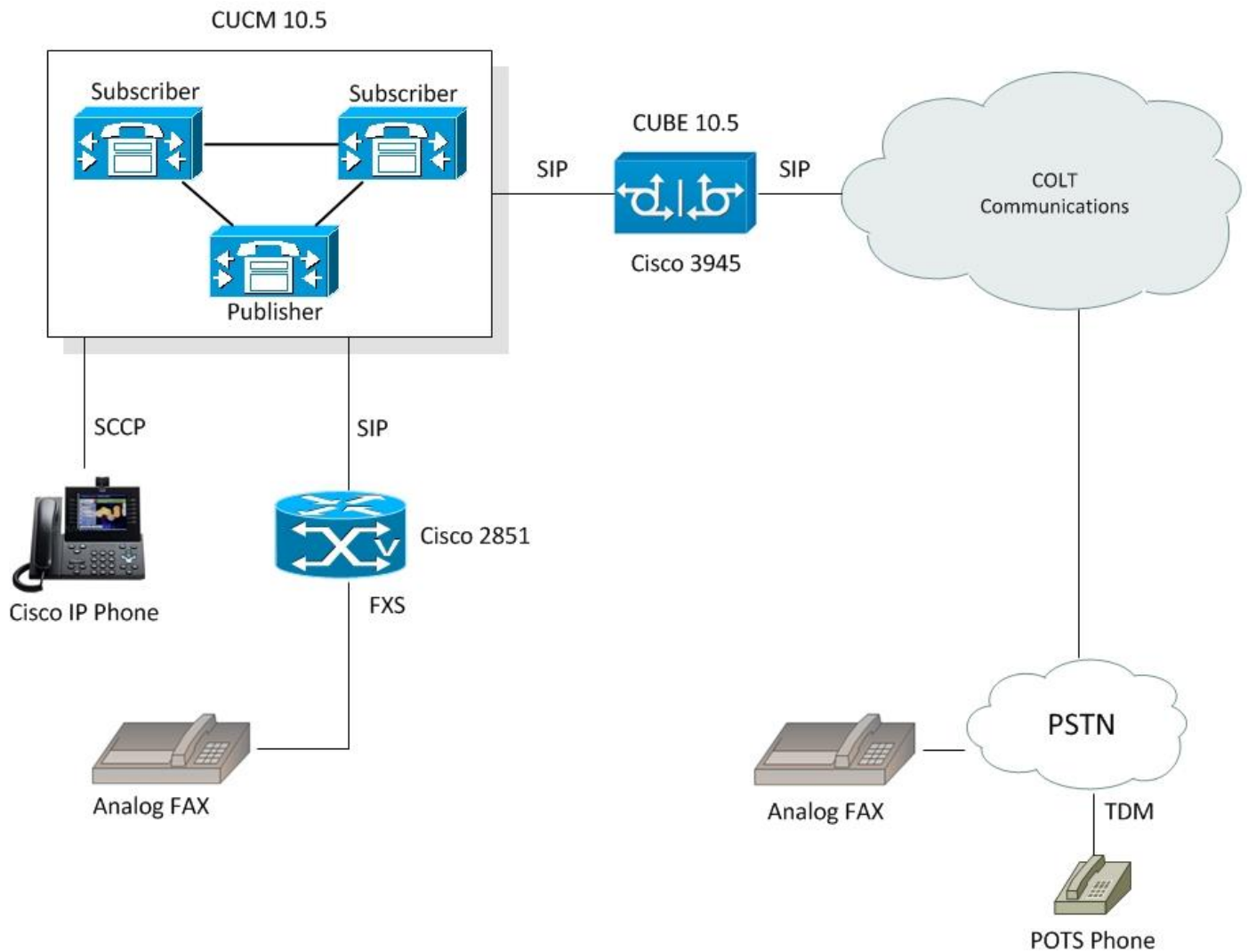
The Cisco Unified Border Element configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between COLT Communications' SIP network and Cisco Unified Communications. The configuration described in this document details the important commands to have enabled for interoperability to be successful and care must be taken, by the network administrator deploying Cisco UBE, to ensure these commands are set per each dial-peer requiring to interoperate to COLT Communications' SIP 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/en/US/docs/voice_ip_comm/cucm/admin/8_5_1/cmsys/a03ptcss.html

Network Topology

Lab Network Topology



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Note: Testing was conducted at tekVizion Labs®.



System Components

Hardware Components

Cisco 3945 (CUBE)

Cisco 2851 with 4FXS/DID (analog fax / POTS phone)

Cisco Unified Communications Manager (1 Publisher and 2 Subscriber)

Cisco IP Phones

Software Requirements

Cisco Unified Communications Manager 10.5.1.10000-7

Cisco Unified Border Element, IOS version (C3900e-UNIVERSALK9-M), Version 15.4(3)M1, RELEASE SOFTWARE (fc1)

Features

Features Supported

- Voice calls using G.729 and G.711 codecs
- RFC 3261 support
- Calling number presentation / restriction
- Call conferencing
- Call transfer (attended and unattended)
- Call hold and resume
- Call forwarding
- DTMF relay (RFC 2833)
- Early media cut-through with DTMF relay before 200 OK
- T.38 Fax G3

Features Not Supported

- Caller ID update via SIP UPDATE method
- G.711 Fax



Caveats

- COLT Communications do not support G711 FAX
- The caller-id is not updated in Call Transfer test cases using SIP UPDATE.
- The test was executed with G.729 as the preferred codec.



Call Flows

In the sample configuration presented here, CUCM is provisioned with ten-digit directory numbers corresponding to the COLT-assigned DIDs.

For incoming PSTN calls, the CUBE presents the full ten-digit DID number to CUCM. Voice calls are routed to IP phones. Fax calls are routed via SIP trunk to a Cisco 2851 router with an FXS daughter board hosting, alternately, an analog fax device or an analog phone. CPE callers make outbound PSTN calls by dialing a “9” prefix followed by the destination number. A “9.@” route pattern strips the prefix and routes the call with the remaining digits via a SIP trunk terminating on the CUBE.

Figure 2: Outbound Voice Call

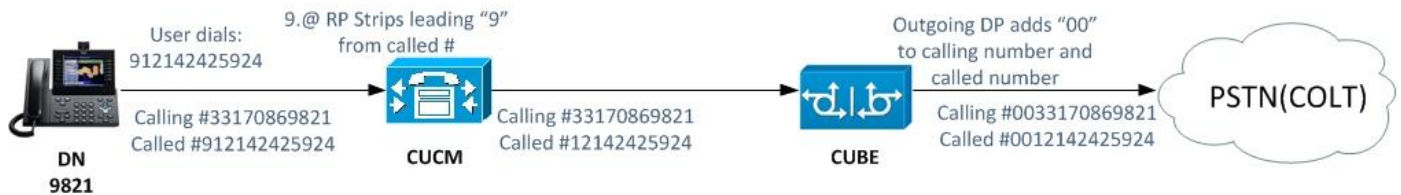


Figure 3: Inbound Voice Call

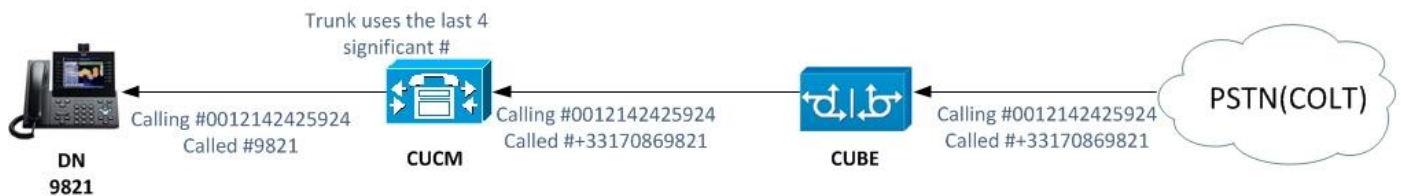


Figure 4: Outbound Fax Call

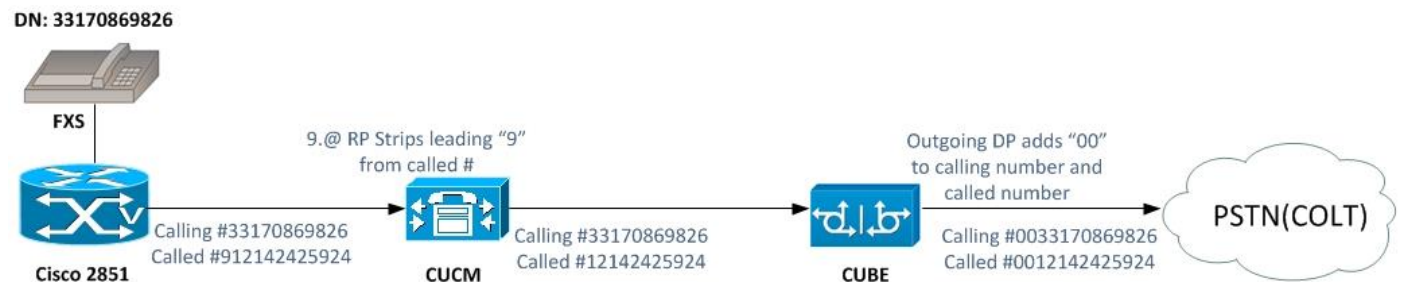


Figure 5: Inbound Fax Call





Configuration

Configuring Cisco Unified Border Element

Version Information:

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

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

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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 6 weeks, 2 days, 10 hours, 28 minutes

System returned to ROM by power-on

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

Last reload type: Normal Reload

Last reload reason: power-on

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 CISC03945-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
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	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

Running Configuration:

TWC_CUBE2#sh run
Building configuration...

```
Current configuration : 13767 bytes
!
! Last configuration change at 11:34:00 UTC Mon Feb 2 2015 by cisco
!
version 15.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname TWC_CUBE2
!
boot-start-marker
boot system flash0:c3900e-universalk9-mz.SPA.154-3.M1.bin
boot-end-marker
!
aqm-register-fnf
!
logging buffered 51200 warnings
enable secret 5 $1$q/Bk$Bu0l4yptT4JPxDewSCcBd.
!
!
!
ipc zone default
  association 1
    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
!
!
!
!
cts logging verbose
!
```

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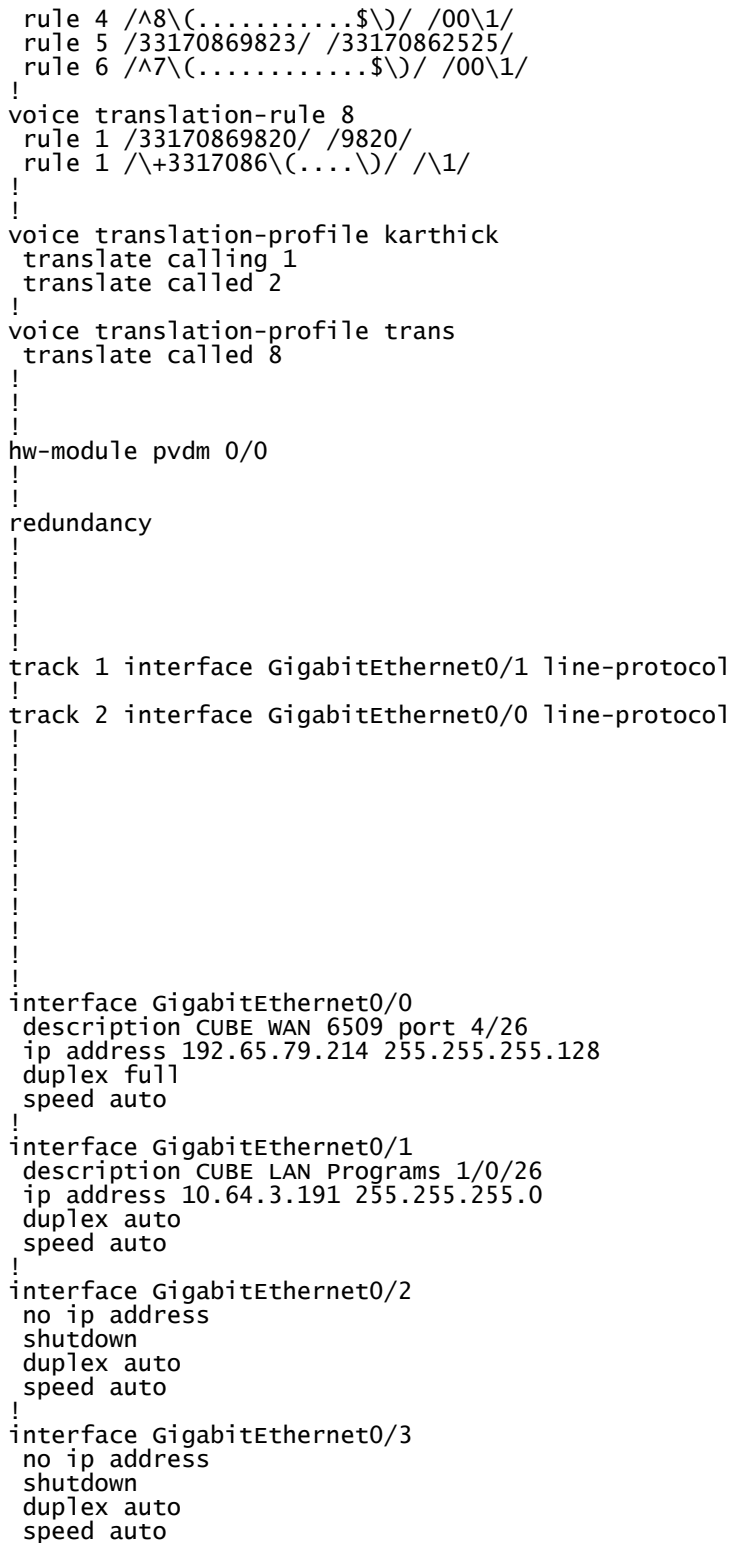
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Note: Testing was conducted at tekVizion Labs®.



```
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
!
!
voice-card 0
  dsp services dspfarm
!
!
no voice hunt unassigned-number
!
voice service voip
  ip address trusted list
    ipv4 0.0.0.0 0.0.0.0
  no ip address trusted authenticate
  address-hiding
  mode border-element
  allow-connections sip to sip
  no supplementary-service sip handle-replaces
  redirect ip2ip
  signaling forward unconditional
  fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
  sip
    rel1xx disable
    session refresh
    header-passing
    asserted-id pai
    history-info
    midcall-signaling passthru
    privacy-policy passthru
    g729 annexb-all
    sip-profiles 100
!
voice class codec 1
  no codec preference 1 g729r8
  codec preference 2 g729br8
  codec preference 3 g711ulaw
!
voice class codec 8
  codec preference 1 g729r8
  codec preference 2 g729br8
!
voice class codec 7
  codec preference 1 g711ulaw
  codec preference 2 g711alaw
!
voice class codec 6
  codec preference 1 g711ulaw
!
!
!
!
voice translation-rule 1
  rule 1 /987654/ //
  rule 2 /\(^.....$\)/ /+1\1/
  rule 3 /\(^.....$\)/ /00\1/
!
voice translation-rule 2
  rule 1 /\+1\(......\)/ /\1/
  rule 2 /\(^.....$\)/ /00\1/
  rule 3 /\(^.....$\)/ /00\1/
```





```
!  
ip forward-protocol nd  
!  
no ip http server  
no ip http secure-server  
!  
ip route 0.0.0.0 0.0.0.0 192.65.79.129  
ip route 10.0.0.0 255.0.0.0 10.64.3.31  
ip route 172.16.0.0 255.255.0.0 10.64.3.31  
!  
!  
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  
!  
!  
!  
dspfarm profile 1 transcode  
  codec g729abr8  
  codec g729ar8  
  codec g711alaw  
  codec g711ulaw  
  codec g729r8  
  maximum sessions 10  
  associate application CUBE  
!  
!  
!  
dial-peer voice 3548 voip  
  description Dial-peer to COLT from CUBE  
  translation-profile outgoing karthick  
  huntstop  
  destination-pattern 1.....  
  session protocol sipv2  
  session target ipv4:217.110.230.98:5060  
  session transport udp  
  voice-class codec 8  
  voice-class sip asserted-id pai  
  voice-class sip options-ping 60  
  voice-class sip profiles 100  
  voice-class sip options-keepalive
```



```
voice-class sip pass-thru headers 100
voice-class sip copy-list 100
voice-class sip contact-passing
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
!
dial-peer voice 3546 voip
description "incoming from network"
session transport udp
incoming called-number +3317086982[0-3]
voice-class codec 8
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
!
dial-peer voice 3545 voip
description "incoming from CUCM"
incoming called-number 1214242....
voice-class codec 7
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
!
dial-peer voice 3549 voip
description "incoming from network"
incoming called-number +3317086982[4-7]
voice-class codec 8
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
!
dial-peer voice 3550 voip
description Dial-peer to Cluster 4 for COLT testing
translation-profile outgoing karthick
destination-pattern +3317086982[4-7]
session protocol sipv2
session target ipv4:10.71.8.10:5060
session transport tcp
voice-class codec 7
voice-class sip asserted-id pai
voice-class sip options-ping 60
voice-class sip options-keepalive
dtmf-relay rtp-nte
no fax-relay sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
!
dial-peer voice 3551 voip
description Dial-peer to COLT for Loopback
translation-profile outgoing karthick
destination-pattern 833.....
session protocol sipv2
session target ipv4:217.110.230.98:5060
session transport udp
voice-class codec 8
```



```
voice-class sip asserted-id pai
voice-class sip options-ping 60
voice-class sip profiles 100
voice-class sip options-keepalive
voice-class sip referto-passing
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
!
dial-peer voice 3547 voip
description Dial-peer to Cluster 1 for COLT testing
translation-profile outgoing karthick
destination-pattern +3317086982[0-3]
session protocol sipv2
session target ipv4:10.71.1.12:5060
session transport tcp
voice-class codec 7
voice-class sip asserted-id pai
voice-class sip options-ping 60
voice-class sip options-keepalive
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 3552 voip
description Dial-peer to COLT for chennai
translation-profile outgoing karthick
huntstop
destination-pattern 7.....
session protocol sipv2
session target ipv4:217.110.230.98:5060
session transport udp
voice-class codec 8
voice-class sip options-ping 60
voice-class sip options-keepalive
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
!
sip-ua
!
!
!
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

Figure 1. SIP Trunk Security Profile

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

SIP Trunk Security Profile Configuration

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

SIP Trunk Security Profile Information

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

Save Delete Copy Reset Apply Config Add New



Figure 2. SIP Profile for CUBE

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

SIP Profile Configuration

Save Delete Copy Reset Apply Config Add New

Status

Status: Ready

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

SIP Profile Information

Name*

COLT SIP Profile

Description

OPTIONS Enabled

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, *, #, and ▾

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)

SIP OPTIONS Ping

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

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

60

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

120

Ping Retry Timer (milliseconds)*

500

Ping Retry Count*

6

SDP Information

☐ Send send-receive SDP in mid-call INVITE

☐ Allow Presentation Sharing using BFCP

☐ Allow iX Application Media

☐ Allow multiple codecs in answer SDP

Save

Delete

Copy

Reset

Apply Config

Add New



Figure 3. SIP Trunk to CUBE

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Trunk Configuration

Save Delete Reset Add New

SIP Trunk Status

Service Status: Full Service
Duration: Time In Full Service: 0 day 20 hours 50 minutes

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	CUBE
Description	CUBE for COLT
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
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

MLPP and Confidential Access Level Information

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

Call Routing Information

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

Inbound Calls

Significant Digits*	4
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	

☐ Redirecting Diversion Header Delivery - Inbound



Called Party Transformation CSS	< None >		
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS			
Calling Party Transformation CSS	< None >		
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS			
Calling Party Selection*	Last Redirect Number (External)		
Calling Line ID Presentation*	Allowed		
Calling Name Presentation*	Default		
Calling and Connected Party Info Format*	Deliver DN only in connected party		
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound			
Redirecting Party Transformation CSS	< None >		
<input checked="" type="checkbox"/> Use Device Pool Redirecting Party Transformation CSS			
Caller Information			
Caller ID DN	<input type="text"/>		
Caller Name	<input type="text"/>		
<input type="checkbox"/> Maintain Original Caller ID DN and Caller Name in Identity Headers			
SIP Information			
Destination			
<input type="checkbox"/> Destination Address is an SRV			
1*	Destination Address	Destination Address IPv6	Destination Port
	10.64.3.191		5060
Status up			
MTP Preferred Originating Codec*	711alaw		



Figure 4. Route Pattern Configuration for SIP trunk to COLT via CUBE

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Route Pattern Configuration

Save Delete Copy Add New

Status
 Status: Ready

Pattern Definition

Route Pattern* 9.@
Route Partition < None >
Description To PSTN via 3845
Numbering Plan* NANP
Route Filter < None >
MLPP Precedence* Default
☐ Apply Call Blocking Percentage
Resource Priority Namespace Network Domain < None >
Route Class* Default
Gateway/Route List* CUBE [\(Edit\)](#)
Route Option
☒ Route this pattern
☐ Block this pattern No Error
Call Classification* OffNet
External Call Control Profile < None >
☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level* 0
☐ 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* 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

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 >



Acronyms

Acronym	Definition
Cisco UBE (CUBE)	Cisco Unified Border Element
Cisco UCM (CUCM)	Cisco Unified Communications Manager
CPE	Customer Premise Equipment
CSS	Calling Search Space
DID	Direct Inward Dial
DNS	Domain Name Service
DTMF	Dual-Tone Multi-Frequency
FQDN	Fully-qualified Domain Name
FXS	Foreign Exchange Station
POTS	Plain Old Telephone Service
PSTN	Public Switched Telephone Network
PTR	DNS Pointer Record (reverse IP lookup)
RFC	Request for Comment
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SRV	DNS Service Locator Record
TDM	Time Division Multiplexing



Important Information

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Note: Testing was conducted at tekVizion Labs®.

**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.)**

Note 1 CPE refers to the Cisco Unified Communications Manager with Cisco Unified Border Element or Cisco Unified Communications Manager Session Manager Edition with Cisco Unified Border Element and the respective end-points devices.

Note 2 PSTN refers to the SIP Service Provider network

Note 3 All CPE to CPE test cases are optional and are to be completed only if it is possible to have two separate CPE environments communicating across the SP SIP core.

Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number display</i>
1.0	CPE to PSTN: codec advertisement shall be g729 and g711ulaw, with g729 codec listed as the preferred codec		
1.1	Ringback heard on Caller phone	PASS	
1.2	Two-way voice path on call answer	PASS	
1.3	Incomplete call. When caller hangs-up before callee answers, callee phone stops ringing	PASS	
1.4	Call duration: 1 hour	PASS	
1.5	DTMF relay (both directions) (RFC2833)	PASS	
1.6	Callee disconnect; Caller disconnect automatically	PASS	
1.7	Caller disconnect; Callee disconnect automatically	PASS	



Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number display</i>
2.0	PSTN to CPE: codec advertisement shall be g729 and g711ulaw, with g729 codec listed as the preferred codec		
2.1	Ringback heard on Caller phone	PASS	
2.2	Two-way voice path	PASS	
2.3	Incomplete call. When caller hangs-up before callee answers, callee phone stops ringing	PASS	
2.4	Call duration: 1 hour	PASS	
2.5	DTMF relay (both directions) (RFC2833)	PASS	
2.6	Callee disconnect; Caller disconnect automatically	PASS	
2.7	Caller disconnect; Callee disconnect automatically	PASS	

Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number display</i>
3.0	CPE to CPE (outbound to the PSTN and back) : codec advertisement shall be g729 and g711ulaw, with g729 codec listed as the preferred codec		
3.1	Ringback heard on Caller phone	PASS	
3.2	Two-way voice path	PASS	

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3.3	Incomplete call. When caller hangs-up before callee answers, callee phone stops ringing	PASS	
3.4	Call duration: 1 hour	PASS	
3.5	DTMF relay (both directions) (RFC2833)	PASS	
3.6	Callee disconnect; Caller disconnect automatically	PASS	
3.7	Caller disconnect; Callee disconnect automatically	PASS	



Test Case Details

Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling number restricted</i>
4.0	Calling number restriction		
4.1	CPE to PSTN: Set calling Number restricted	PASS	
4.2	CPE to another CPE: Set calling Number restricted	PASS	



Test Case Details

Test Case No.	Test Case	Pass / Fail	Comments:
5.0	Telephone Number Support		
5.1	CPE to PSTN: CPE must translate phone extension to 10 DID calling number	PASS	
5.2	PSTN to CPE: CPE must translate 10 digits called number to phone extension.	PASS	
5.3	CPE to CPE: CPE must translate phone extension to 10 DID calling number	PASS	

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5.4	CPE to CPE: CPE must translate 10 digits called number to phone extension.	PASS	
5.5	PSTN to CPE: CPE must translate 4 digits called number to phone extension.	PASS	
5.6	CPE to CPE: CPE must translate 4 digits called number to phone extension.	PASS	

Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name and connected name display</i>
6.0	Calling Name Presentation		
6.1	CPE to CPE: display name presentation allowed	PASS	

Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number update</i>
7.0	PSTN: Call Conference		
7.1	PSTN phone A to CPE phone A, CPE phone A conferences PSTN phone B	PASS	
7.2	CPE phone A to PSTN phone A, CPE phone A conferences PSTN phone B	PASS	



7.3	CPE phone A to CPE phone B, CPE phone A conferences PSTN phone B	PASS	
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Test Case Details

Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number update</i>
8.0	Intra-Site: Call Conference		
8.1	Phone A to Phone B. Phone A conferences Phone C at 2nd CPE site	PASS	
8.2	Phone A to 2 nd CPE Phone B. Phone A conferences PSTN	PASS	
8.3	Phone A to Phone A at 2nd CPE site. Phone A at 2 nd CPE conferences Phone B at Phone A site	PASS	
8.4	Phone A to PSTN. Phone A conferences 2 nd CPE Phone B	PASS	
8.5	Phone A at 2nd CPE site to Phone A. Phone A conferences Phone B	PASS	

Test Case Details

Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number display</i>
9.0	Attended Call Transfer		
9.1	PSTN phone A to CPE phone A, CPE phone A transfers to PSTN phone B (does caller ID update on PSTN phone B?)	PASS	The Caller id is not updated in PSTN Phone B
9.2	CPE phone A to PSTN phone A, CPE phone A transfers to PSTN phone B (does caller ID update on PSTN phone B?)	PASS	The Caller id is not updated in PSTN Phone B

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9.3	CPE Phone A to CPE Phone B. Phone A transfers to Phone A at 2nd CPE site	PASS	
9.4	CPE Phone A to CPE Phone B. CPE Phone A transfers to PSTN	PASS	
9.5	CPE Phone A to Phone A at 2nd CPE site. Phone A and 2 nd CPE transfers to Phone B at Phone A site	PASS	
9.6	CPE Phone A to PSTN. CPE Phone A transfers to CPE Phone B	PASS	
9.7	Phone A at 2nd CPE site to CPE Phone A. CPE Phone A transfers to Phone B	PASS	
9.8	PSTN to CPE Phone A. CPE Phone A transfers to CPE Phone B	PASS	
Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number display</i>
10.0	Unattended Call Transfer		
10.1	PSTN phone A to CPE phone A, CPE phone A transfers to PSTN phone B (does caller ID update on PSTN phone B?)	PASS	The Caller id is not updated in PSTN Phone B
10.2	CPE phone A to PSTN phone A, CPE phone A transfers to PSTN phone B (does caller ID update on PSTN phone B?)	PASS	The Caller id is not updated in PSTN Phone B
10.3	CPE Phone A to CPE Phone B. Phone A transfers to Phone A at 2nd CPE site	PASS	
10.4	CPE Phone A to CPE Phone B. CPE Phone A transfers to PSTN	PASS	
10.5	CPE Phone A to Phone A at 2nd CPE site. Phone A and 2 nd CPE transfers to Phone B at Phone A site	PASS	
10.6	CPE Phone A to PSTN. CPE Phone A transfers to CPE Phone B	PASS	

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10.7	Phone A at 2nd CPE site to CPE Phone A. CPE Phone A transfers to Phone B	PASS	
10.8	PSTN to CPE Phone A. CPE Phone A transfers to CPE Phone B	PASS	

Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number display</i>
11.0	Blind Call Transfer (applies if SIP blind transfer is supported)		
11.1	PSTN phone A to CPE phone A, CPE phone A transfers to PSTN phone B (does caller ID update on PSTN phone B?)	N/A	
11.2	CPE phone A to PSTN phone A, CPE phone A transfers to PSTN phone B (does caller ID update on PSTN phone B?)	N/A	
11.3	CPE Phone A to CPE Phone B. Phone A transfers to Phone A at 2nd CPE site	N/A	
11.4	CPE Phone A to CPE Phone B. CPE Phone A transfers to PSTN	N/A	
11.5	CPE Phone A to Phone A at 2nd CPE site. Phone A and 2 nd CPE transfers to Phone B at Phone A site	N/A	
11.6	CPE Phone A to PSTN. CPE Phone A transfers to CPE Phone B	N/A	
11.7	Phone A at 2nd CPE site to CPE Phone A. CPE Phone A transfers to Phone B	N/A	
11.8	PSTN to CPE Phone A. CPE Phone A transfers to CPE Phone B	N/A	

Note: Testing was conducted at tekVizion Labs®.



Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments:
12.0	Call Hold and Resume		Comments: call on hold is always perform on the CPE side. Note as of today, multicast MoH is not supported with CUBE.
12.1	CPE calls to PSTN (unicast MoH is enabled). Was MoH heard?	PASS	
12.2	CPE calls to CPE (unicast MoH is enabled). Was MoH heard?	PASS	
12.3	PSTN calls to CPE (unicast MoH is enabled). Was MoH heard?	PASS	
12.4	CPE calls to PSTN (unicast MoH is disabled). Was ToH heard?	PASS	
12.5	CPE calls to CPE (unicast MoH is disabled). Was ToH heard?	PASS	
12.6	PSTN calls to CPE (unicast MoH is disabled). Was ToH heard??	PASS	

Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments:
13.0	Voice Mail (e.g. using Unity or Unity Connection)		
13.1	PSTN to CPE: leave voice mail	PASS	
13.2	PSTN to CPE: retrieve voice mail	PASS	

Note: Testing was conducted at tekVizion Labs®.



Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments:
14.0	PSTN Voice Mail (e.g. using mobile phone voicemail)		
14.1	CPE to PSTN (mobile VM): leave voice mail	PASS	
14.2	CPE to PSTN (mobile VM): retrieve voice mail	PASS	

Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number display</i>
15.0	Find Me (Call Forward Unconditionally)		
15.1	PSTN to CPE enabled call forward unconditionally feature	PASS	
15.2	CPE to CPE enabled call forward unconditionally feature	PASS	
15.3	PSTN to CPE phone A enabled call forward unconditionally feature to PSTN	PASS	
15.4	CPE to CPE phone A enabled call forward unconditionally feature to PSTN	PASS	



Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number display</i>
16.0	T.38 FAX G3: Priority codec - G.729		
16.1	CPE FAX to PSTN FAX - G3-G3	PASS	
16.2	CPE FAX to PSTN FAX - G3-SG3	PASS	
16.3	CPE FAX from PSTN FAX - G3-G3	PASS	
16.4	CPE FAX from PSTN FAX - SG3-G3	PASS	
16.5	CPE FAX to CPE FAX - G3-G3	PASS	
16.6	CPE FAX to CPE FAX - G3-SG3	PASS	

Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number display</i>
17.0	T.38 FAX SG3 (G.729 is offered first)		
17.1	CPE FAX to PSTN FAX - SG3-G3	PASS	
17.2	CPE FAX to PSTN FAX - SG3-SG3	PASS	
17.3	CPE FAX from PSTN FAX - G3-SG3	PASS	
17.4	CPE FAX from PSTN FAX -SG3-SG3	PASS	
17.5	CPE FAX to CPE FAX - SG3-G3	PASS	
17.6	CPE FAX to CPE FAX - SG3-SG3	PASS	

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Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments:
18.0	Simultaneous Calls (Minimum 2)- Duration 90 secs		
18.1	CPE to PSTN gateway	PASS	
18.2	PSTN gateway inbound to CPE	PASS	
18.3	CPE to CPE	PASS	

Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments:
19.0	Auto Attendant (e.g. using Unity or Unity Connection AA services)		
19.1	PSTN to CPE: navigate AA menus	PASS	
19.2	PSTN to CPE: navigate AA menu to transfer to a user	PASS	

Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number display</i>
20.0	CPE to PSTN gateway international call		



20.1	Ringback heard on Caller phone	PASS	
20.2	Two-way voice path on call answerl	PASS	
20.3	Incomplete call. When caller hangs-up before callee answers, callee phone stops ringing	PASS	
20.4	DTMF relay (both directions) (RFC2833)	PASS	
20.5	Callee disconnect; Caller disconnect automatically	PASS	
20.6	Caller disconnect; Callee disconnect automatically	PASS	

Note 4 Cisco UCM does not support FAX pass-through (mid-call codec change to G711, upspeed) (Support is available on Cisco UCM 7.1(5) and above using SIP devices/gw's only). CUCM only supports Fax over G.711, where the initial call must begin as a G711 media stream. Specific configurations on Cisco gateways and CUBE are required. See appendix A for details

Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number display</i>
21.0	G.711 FAX G3	N/S	
21.1	G3 CPE to PSTN FAX - G3-G3	N/S	
21.2	G3 CPE to PSTN FAX - G3-SG3	N/S	
21.3	G3 CPE from PSTN FAX - G3-G3	N/S	
21.4	G3 CPE from PSTN FAX - SG3-G3	N/S	
21.5	G3 CPE to CPE FAX - G3-G3	N/S	
21.6	G3 CPE to CPE FAX - G3-SG3	N/S	

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Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number display</i>
22.0	G.711 FAX SG3	N/S	
22.1	FAX SG3 to PSTN FAX - SG3-G3	N/S	
22.2	FAX SG3 to PSTN FAX - SG3-SG3	N/S	
22.3	FAX SG3 from PSTN FAX - G3-SG3	N/S	
22.4	FAX SG3 from PSTN FAX -SG3-SG3	N/S	
22.5	FAX SG3 to CPE FAX - SG3-G3	N/S	
22.6	FAX SG3 to CPE FAX - SG3-SG3	N/S	

Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number display</i>
23.0	Find Me (Call Forward On Busy)		
23.1	PSTN to CPE phone A enabled Call Forward on Busy feature to phone B	PASS	
23.2	PSTN to CPE phone enabled Call Forward on Busy feature to PSTN	PASS	
23.3	CPE to CPE phone A enabled Call Forward on Busy feature to phone B	PASS	
23.4	CPE to CPE phone A enabled Call Forward on Busy feature to PSTN	PASS	



Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number display</i>
24.0	Find Me (Call Forward No Reply)		
24.1	PSTN to CPE phone A enabled Call Forward No Reply feature to phone B	PASS	
24.2	PSTN to CPE phone A enabled Call Forward No Reply feature to PSTN	PASS	
24.3	CPE to CPE phone A enabled Call Forward No Reply feature to phone B	PASS	
24.4	CPE to CPE phone A enabled Call Forward No Reply feature to PSTN	PASS	

Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments:
25.0	Codec mid-call negotiation (without transcoder)		
25.1	PSTN calls CPE phone A (G729), phone A transfers to gateway (g711u). PSTN and CPE gateway negotiate codec and call is transferred.	PASS	
25.2	CPE phone A calls PSTN (G711), PSTN transfers call to CPE phone B (G729), calls set up between CPE phone A and CPE phone B	PASS	

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Test Case Details			
Test Case No.	Test Case	Pass / Fail	Comments: <i>Note calling name/number and connected name/number display</i>
27.0	PRACK with SDP (early-media cut-through with DTMF (RFC2833) navigation before 200OK)) - call 800-864-8331 - United Airlines		
27.1	CPE phone A call 800 number, phone user navigates through AA to reach correct menu option.	PASS	

---End of the test cases-----



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