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

Feb, 2015

Note: Testing was conducted in tekVizion Labs.

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Note: Testing was conducted at tekVizion Labs®.

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

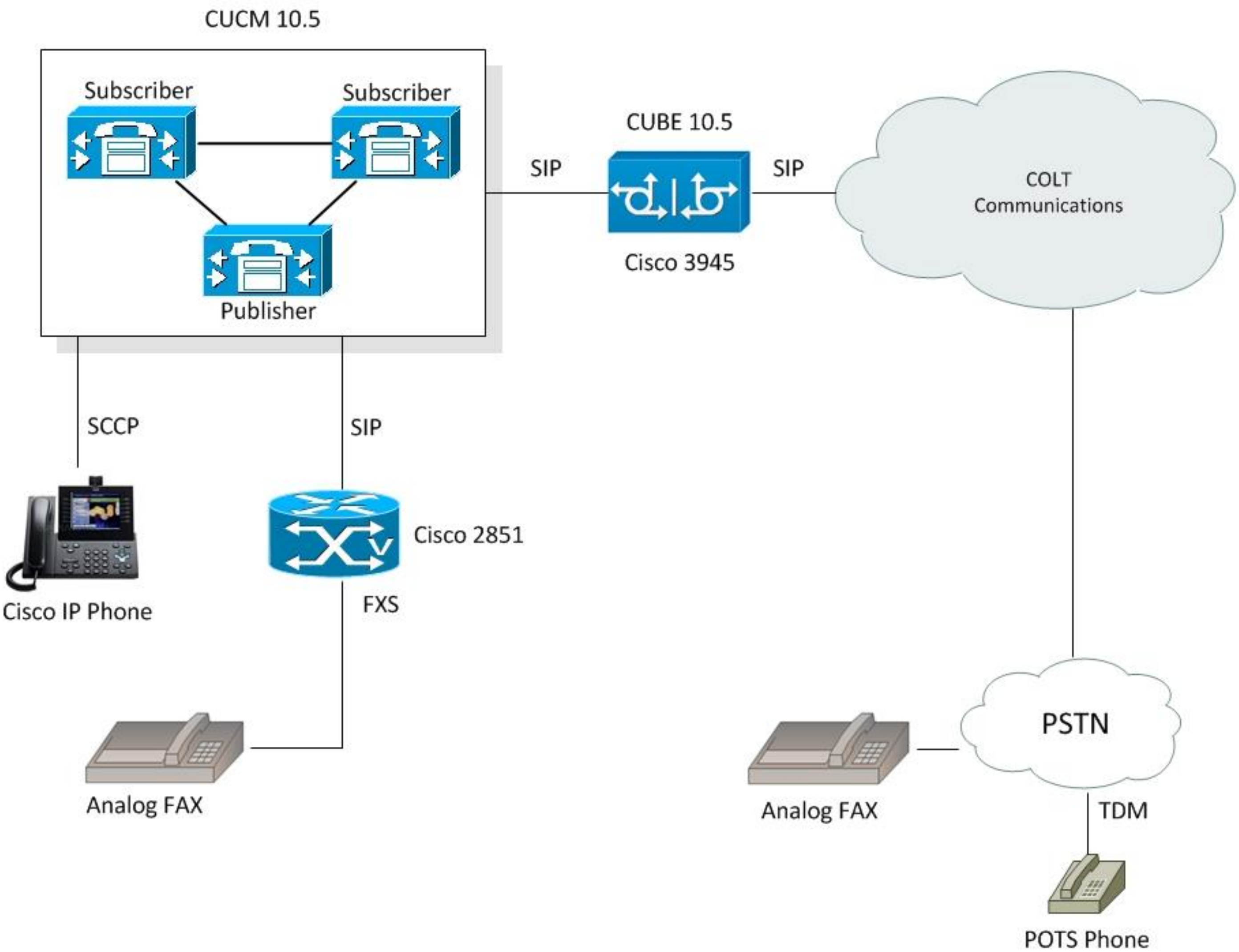
The Cisco Unified Border Element configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between COLT Communications' SIP network and Cisco Unified Communications. The configuration described in this document details the important 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/ccmsys/a03ptcss.html

Note: Testing was conducted at tekVizion Labs®.

Network Topology

Lab Network Topology



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

Note: Testing was conducted at tekVizion Labs®.

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.

Note: Testing was conducted at tekVizion Labs®.

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

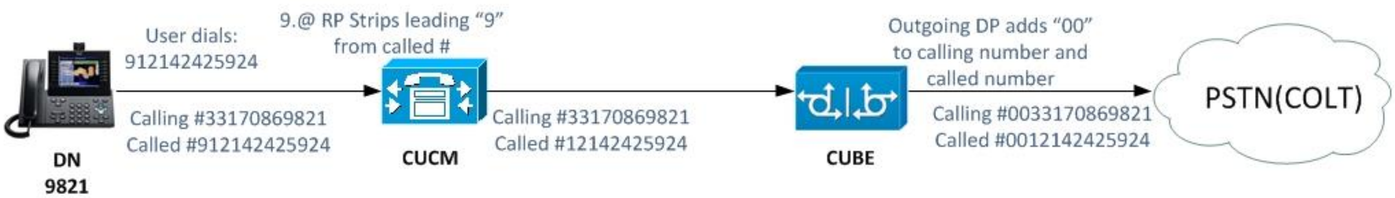


Figure 3: Inbound Voice Call

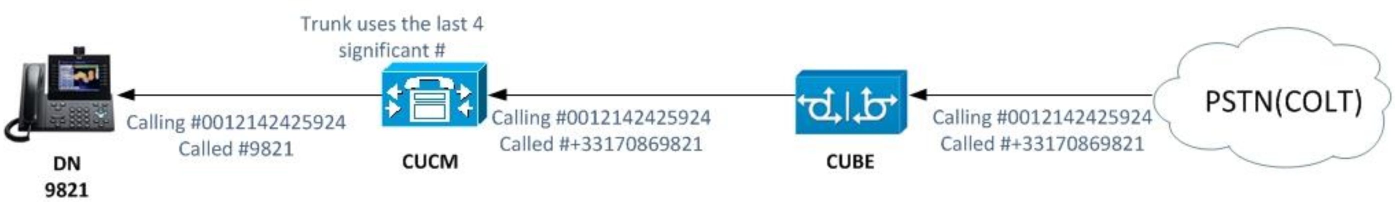


Figure 4: Outbound Fax Call

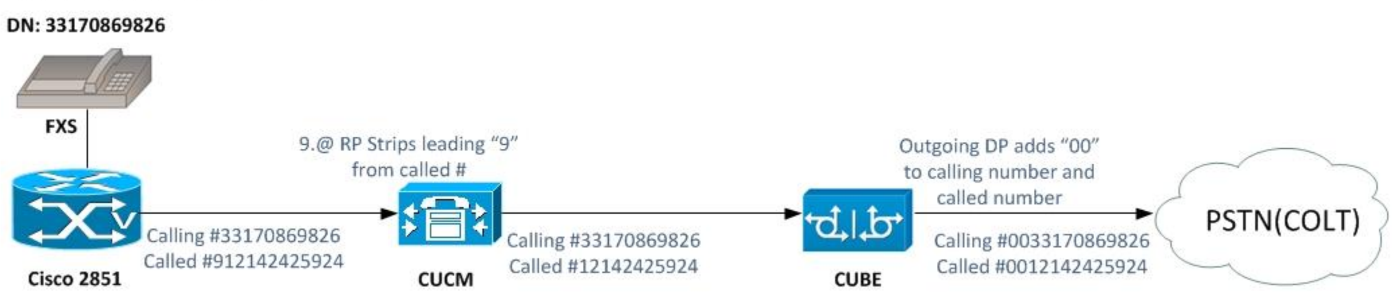


Figure 5: Inbound Fax Call

Note: Testing was conducted at tekVizion Labs®.

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
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 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 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
```

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$BuOl4yptT4JPxDeWSCcBd.
!
!
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
!
```

```
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/
```


Note: Testing was conducted at tekVizion Labs®.

```
 rule 4 /^8\(...........$\)/ /00\1/
 rule 5 /33170869823/ /33170862525/
 rule 6 /^7\(............$\)/ /00\1/
!
voice translation-rule 8
 rule 1 /33170869820/ /9820/
 rule 1 /\+3317086\(....\)/ /\1/
!
!
voice translation-profile karthick
 translate calling 1
 translate called 2
!
voice translation-profile trans
 translate called 8
!
!
!
hw-module pvdm 0/0
!
!
redundancy
!
!
!
!
!
track 1 interface GigabitEthernet0/1 line-protocol
!
track 2 interface GigabitEthernet0/0 line-protocol
!
!
!
!
!
!
!
!
!
!
!
interface GigabitEthernet0/0
 description CUBE WAN 6509 port 4/26
 ip address 192.65.79.214 255.255.255.128
 duplex full
 speed auto
!
interface GigabitEthernet0/1
 description CUBE LAN Programs 1/0/26
 ip address 10.64.3.191 255.255.255.0
 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
```


Note: Testing was conducted at tekVizion Labs®.

```
!
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
```


Note: Testing was conducted at tekVizion Labs®.

```
 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
```


Note: Testing was conducted at tekVizion Labs®.

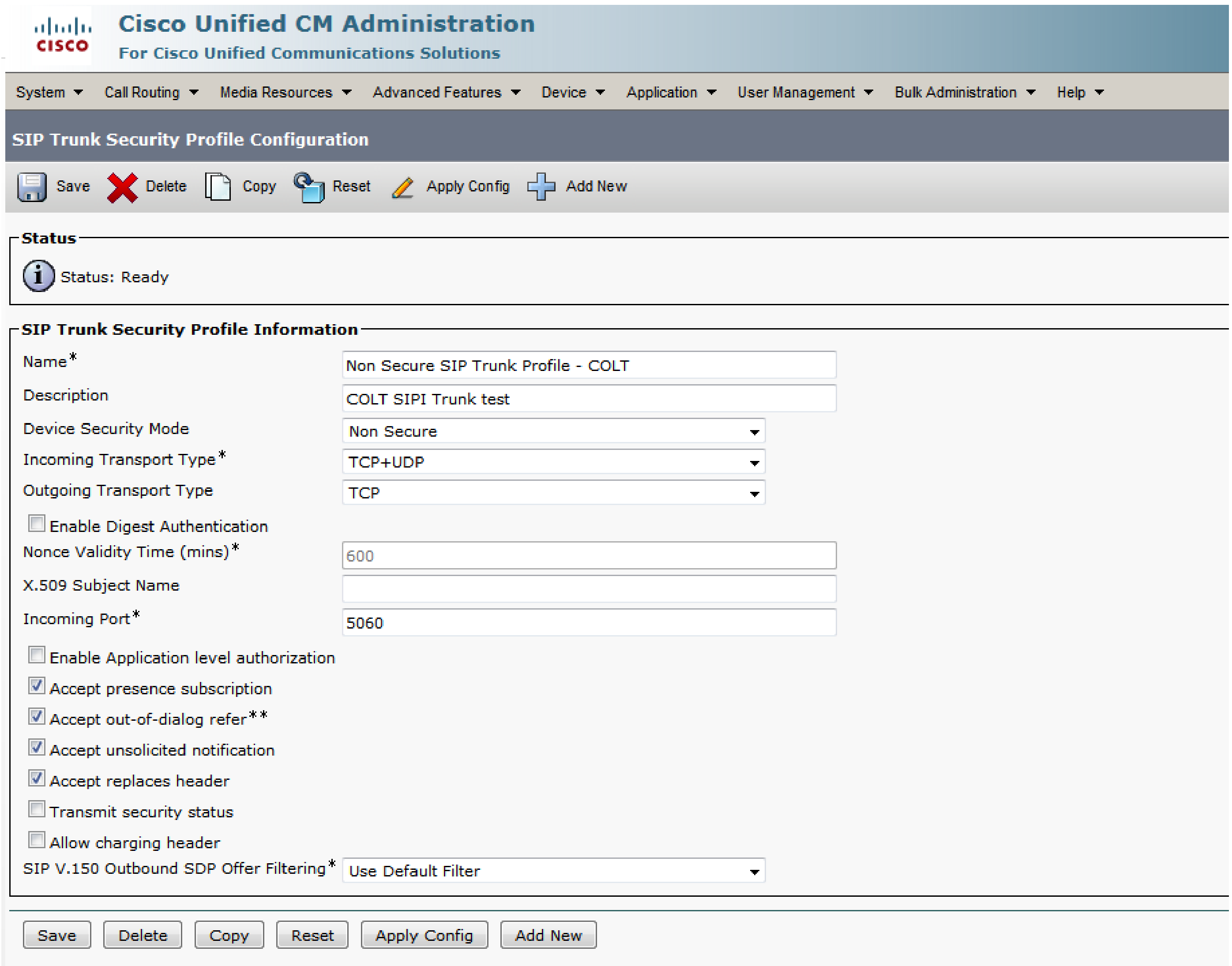
```
 voice-class sip asserted-id pai
 voice-class sip options-ping 60
 voice-class sip profiles 100
 voice-class sip options-keepalive
 voice-class sip referto-passing
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
!
dial-peer voice 3547 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
```


Note: Testing was conducted at tekVizion Labs®.

```
 login local
 transport input telnet ssh
!
scheduler allocate 20000 1000
!
end
```

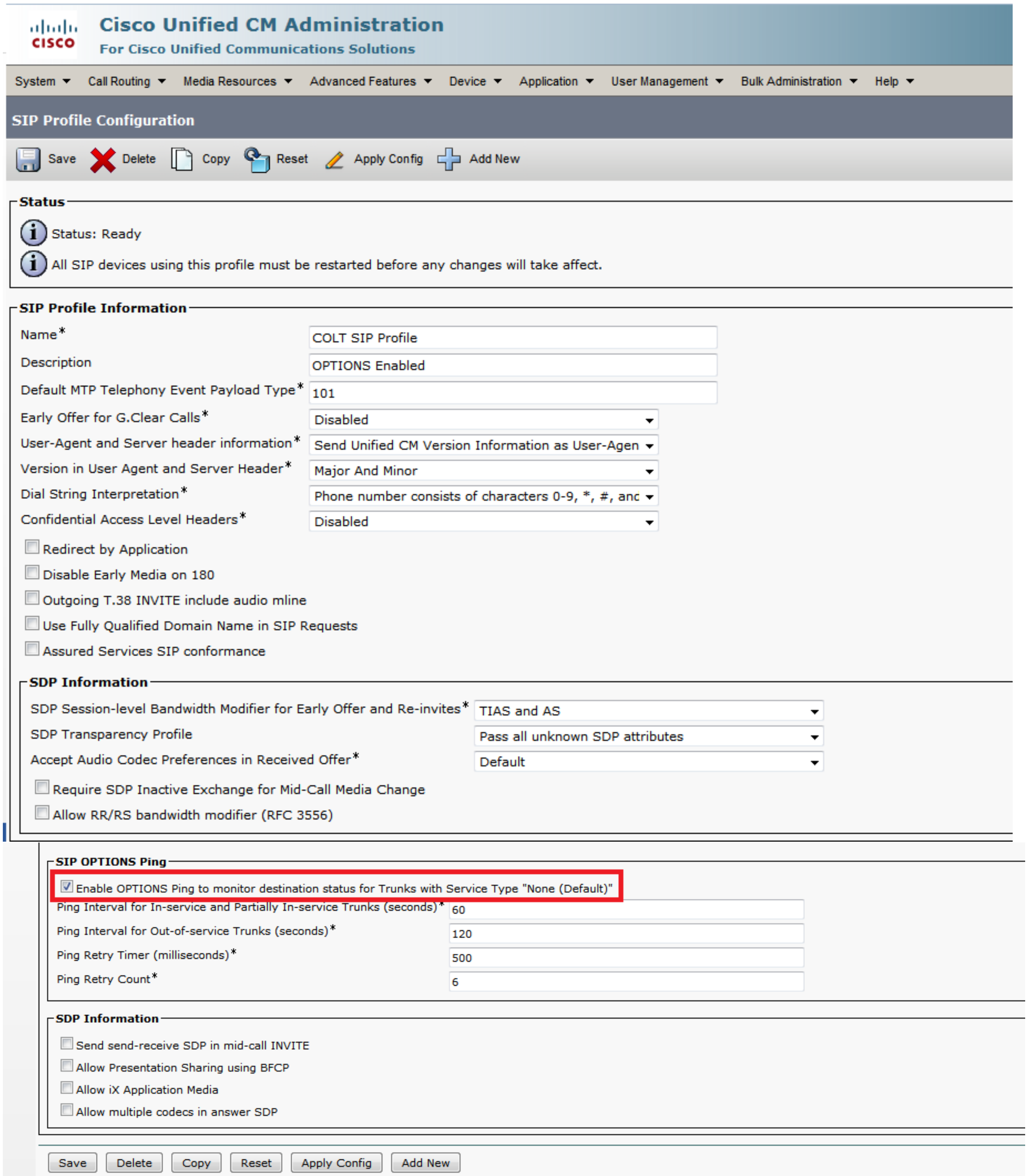
Configuring the Cisco Unified Communications Manager

Figure 1. SIP Trunk Security Profile



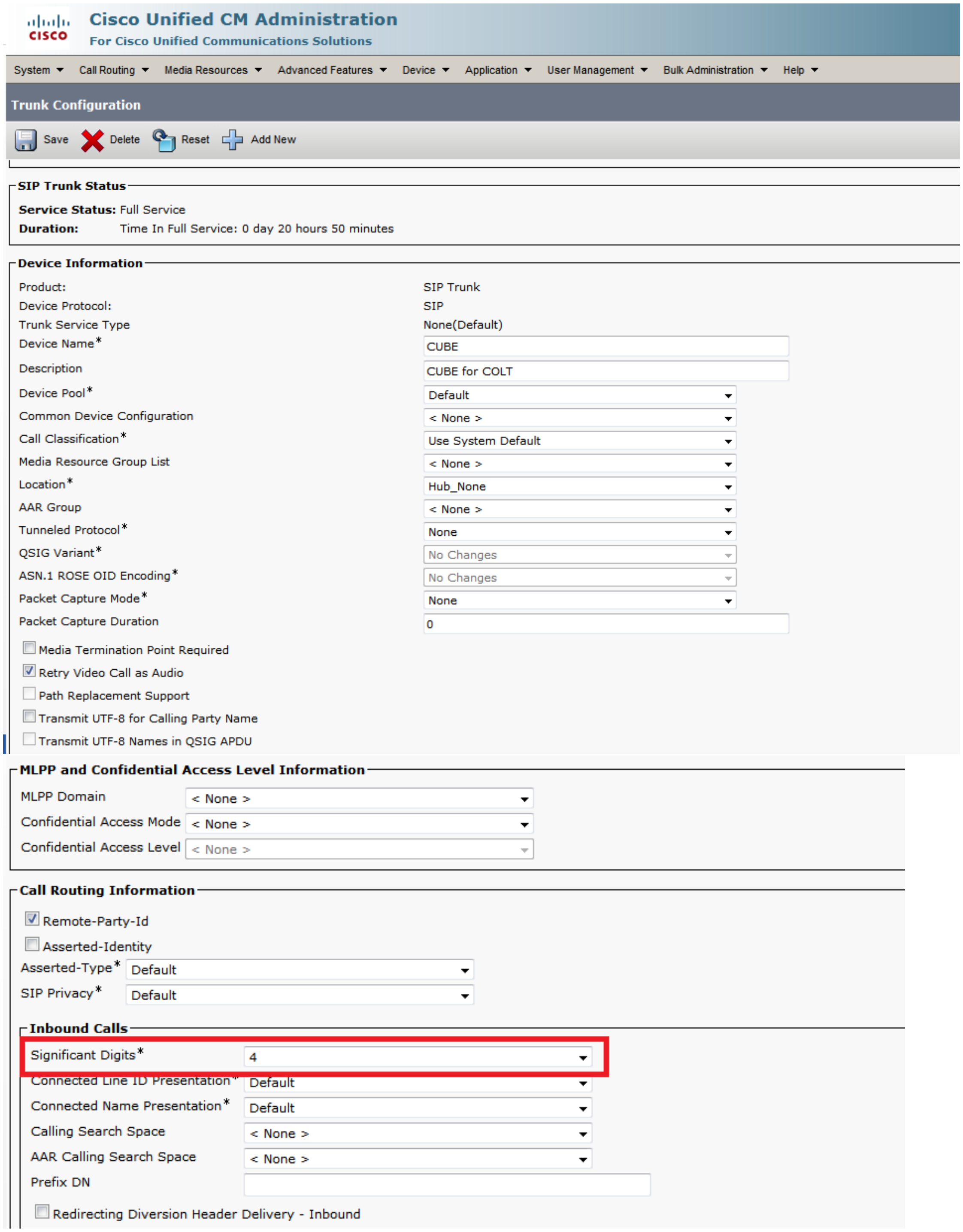
Note: Testing was conducted at tekVizion Labs®.

Figure 2. SIP Profile for CUBE

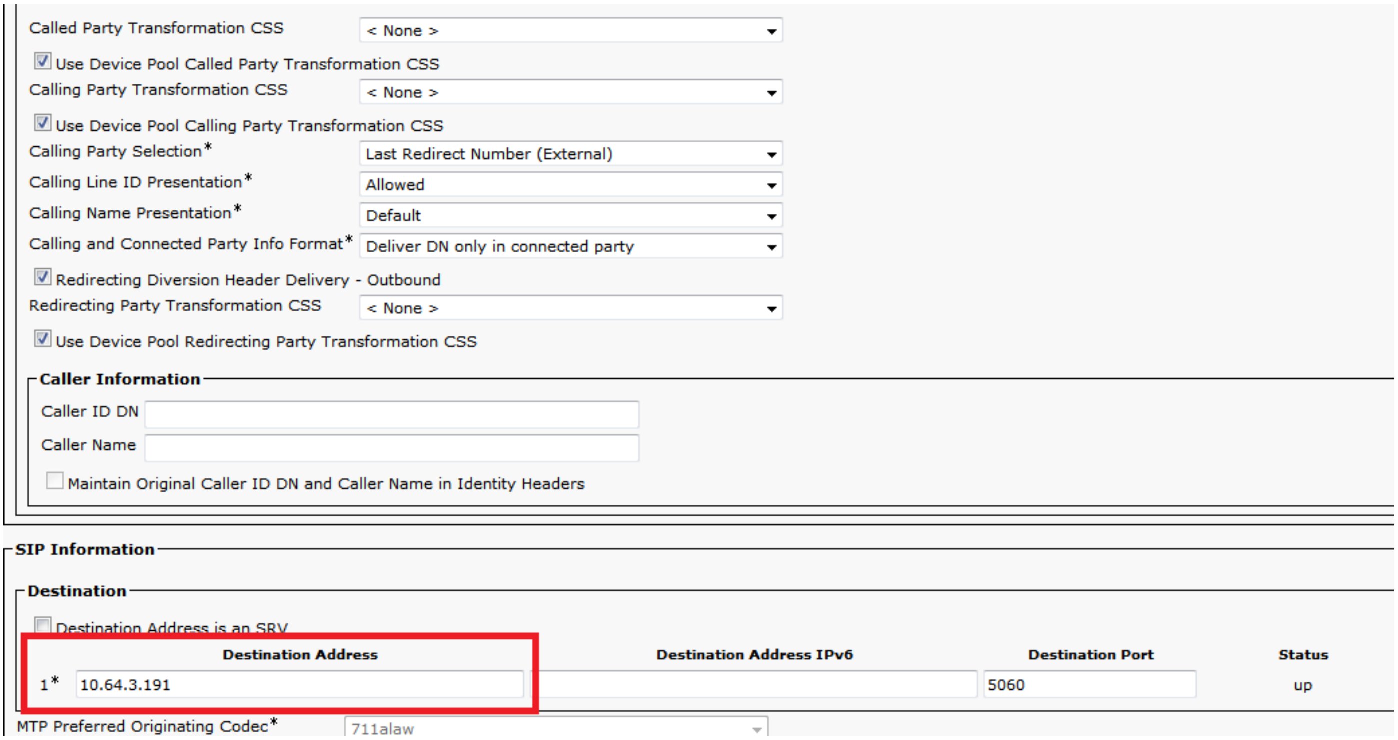


Note: Testing was conducted at tekVizion Labs®.

Figure 3. SIP Trunk to CUBE



Note: Testing was conducted at tekVizion Labs®.



Called Party Transformation CSS: < None >
[x] Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS: < None >
[x] 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
[x] Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS: < None >
[x] 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

SIP Information

Destination

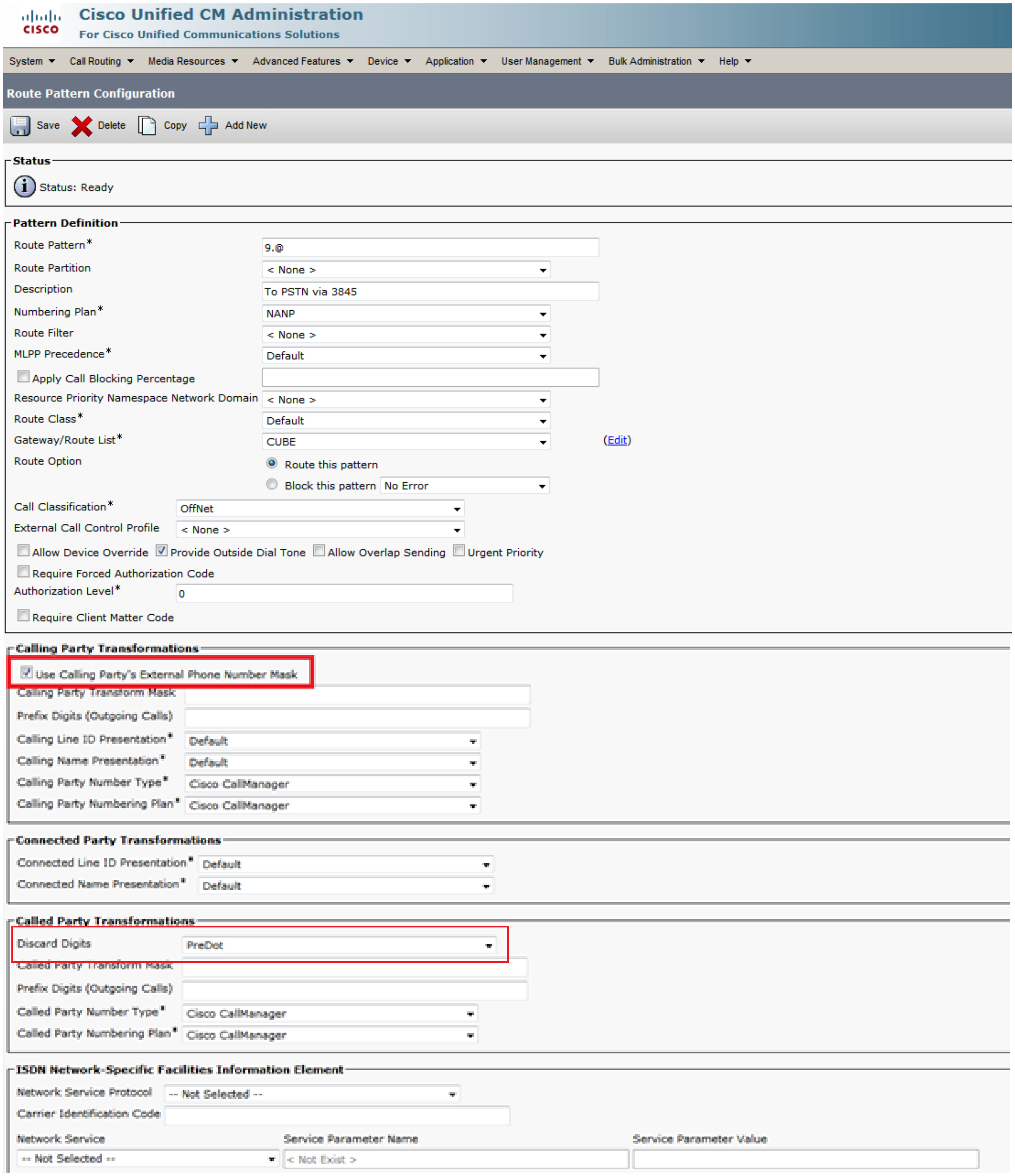
[ ] Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status
1*	10.64.3.191		5060	up

MTP Preferred Originating Codec*: 711alaw

Note: Testing was conducted at tekVizion Labs®.

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



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

Note: Testing was conducted at tekVizion Labs®.

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:** ***Note calling name/number and connected name/number display***
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**	

Note: Testing was conducted in tekVizion Labs.

Test Case Details

Test Case No.	Test Case	Pass / Fail	Comments: *Note calling name/number and connected name/number display*
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: *Note calling name/number and connected name/number display*
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	

Note: Testing was conducted at tekVizion Labs®.

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:** ***Note calling number restricted***
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	

Note: Testing was conducted at tekVizion Labs®.

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:** ***Note calling name and connected name display***
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:** ***Note calling name/number and connected name/number update***
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	

Note: Testing was conducted at tekVizion Labs®.

7.3	CPE phone A to CPE phone B, CPE phone A conferences PSTN phone B	PASS	

Test Case Details			
Test Case No.	**Test Case**	**Pass / Fail**	**Comments:** ***Note calling name/number and connected name/number update***
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 2nd CPE Phone B. Phone A conferences PSTN	PASS	
8.3	Phone A to Phone A at 2nd CPE site. Phone A at 2nd CPE conferences Phone B at Phone A site	PASS	
8.4	Phone A to PSTN. Phone A conferences 2nd 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:** ***Note calling name/number and connected name/number display***
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

Note: Testing was conducted at tekVizion Labs®.

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 2nd 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: *Note calling name/number and connected name/number display*
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 2nd 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**	

Note: Testing was conducted at tekVizion Labs®.

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: *Note calling name/number and connected name/number display*
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 2nd 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:** ***Note calling name/number and connected name/number display***
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**	

Note: Testing was conducted at tekVizion Labs®.

Test Case Details

Test Case No.	Test Case	Pass / Fail	Comments: *Note calling name/number and connected name/number display*
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: *Note calling name/number and connected name/number display*
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	

Note: Testing was conducted at tekVizion Labs®.

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: *Note calling name/number and connected name/number display*
20.0	**CPE to PSTN gateway international call**		

Note: Testing was conducted at tekVizion Labs®.

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:** ***Note calling name/number and connected name/number display***
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	

Note: Testing was conducted at tekVizion Labs®.

Test Case Details

Test Case No.	Test Case	Pass / Fail	Comments: *Note calling name/number and connected name/number display*
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: *Note calling name/number and connected name/number display*
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	

Note: Testing was conducted at tekVizion Labs®.

Test Case Details

Test Case No.	Test Case	Pass / Fail	Comments: *Note calling name/number and connected name/number display*
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	

Note: Testing was conducted at tekVizion Labs®.

Test Case Details			
Test Case No.	Test Case	Pass / Fail	**Comments:** *Note calling name/number and connected name/number display*
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-------

Note: Testing was conducted at tekVizion Labs®.

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