



tw telecom: Connecting Cisco Unified Communications Manager 9.1 via the Cisco Unified Border Element Version 9.0 using SIP

Version 1.3



6/11/13 – Version 1.3

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Introduction

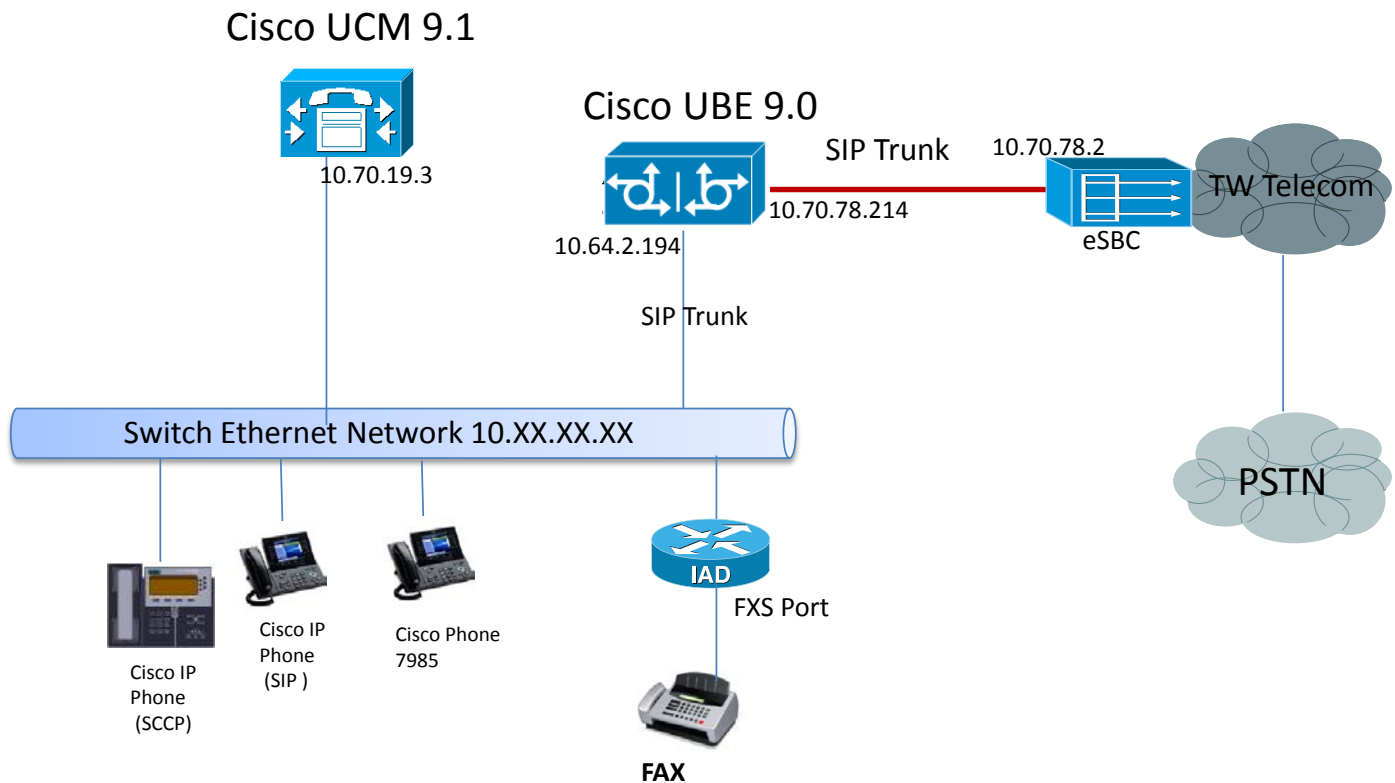
Service Providers today, such as tw telecom, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. tw telecom is a service provider offering 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 Cisco Unified Communications Manager (Cisco UCM) version 9.1 [Cisco Unified Border Element (Cisco UBE), version9.0] for connectivity to tw telecom SIP trunk provider via the eSBC (FortisVox). The deployment model covered in this application note is CPE to PSTN. This document does not address 911 emergency outbound calls. For 911 feature service details contact tw telecom, directly.
- Testing was performed in accordance to Cisco's SIP Trunk Test Plan and all features were verified. Key features verified are:
 - CPE outbound to SP Offnet gateway(PSTN) (G.729 is offered first)
 - CPE to CPE (place call out to the SP network and back) (G.729 is offered first)
 - CPE Calling number privacy
 - CPE Telephone Number Support – digit translations
 - CPE Calling Name Delivery
 - CPE offnet Call Conference
 - CPE Intra-Site Call Conference
 - CPE Intra-Site Attended Call Transfer
 - CPE Intra-Site Unattended Call Transfer
 - CPE Call Hold and Resume (call hold is always done on the IP PBX side)
 - CPE Voice Mail
 - SP Voice Mail
 - CPE Find Me (CFU)
 - Simultaneous Calls
 - CPE Auto Attendant
 - CPE to PSTN Offnet gateway international call
 - CPE G.711 FAX SG3/G3
 - CPE Find Me (Call Forward On Busy)
 - CPE Find Me (Call Forward Don't Answer)
 - Codec mid-call re-negotiation (to be tested without transcoder)
- The Cisco UCM/Cisco Unified Border Element configuration detailed in this document is based on a lab environment with simple dial-plan configurations used to ensure proper interoperability between tw telecom SIP network and Cisco Unified Communications. The configuration described in this document details the important commands that need to be 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 with the eSBC.
- 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/partner/docs/voice_ip_comm/cucm/admin/8_0_2/ccmsys/a03ptcss.html

Network Topology

The network topology includes the CUBE used in the CME mode, connected to the FortisVox SBC via a SIP trunk. tw telecom was used as the service provider. Outbound PSTN calls were ultimately routed into the tekVizion PSTN GW.

Figure 1. Basic Test Environment



System Components

Hardware Components

- CUBE – Cisco 3925
- PSTN GW – Cisco 3845
- Switch – Cisco 6509
- Cisco Phones – 7975(SCCP), 9971(SIP), 8961(SIP)
- tw telecom Trunk (Third Party SIP Trunk Provider)
- FVX eSBC

Software Requirements

- Cisco UCM version 9.1
- Cisco UBE version 9.0
- Cisco 7975 : SCCP45.9-331SR1-1S



- Cisco 9971: Sip9971.9-3-2-10
- Cisco 8961: Sip8961.9-3-2-10
- FVX eSBC 5.0.1



Features

Features Supported

- Call from/to PSTN to/from CPE – Basic and International calls , digit translations
- Hold/Resume
- DTMF
- Call transfers – attended, unattended, blind
- Call Conference
- Call Forwarding (CFU,CFB,CFNA)
- Support for early media
- Fax using G.711 Pass-through
- Basic G.711ulaw calls
- Calling Name
- Calling Party Number Presentation and Restriction

Features Not Supported

- G729 codec.
- FAX: T.38 fax protocol is not supported.
- PRACK with SDP



Caveats

- CLID updates are not observed on call transfer scenarios.
- tw telecom doesn't support G729 calls. Hence all calls were tested with G711ulaw.
- Scenarios that include Phone 3 at second PBX site were tested with the Phone 3 registered to a second Cisco Unified Communication Manager.



Configuration

Configuring Cisco Unified Border Element

Show version:

Cisco IOS Software, C3900e Software (C3900e-UNIVERSALK9-M), Version 15.2(4)M1, RELEASE SOFTWARE (fc1)
Technical Support: <http://www.cisco.com/techsupport>
Copyright (c) 1986-2012 by Cisco Systems, Inc.
Compiled Fri 27-Jul-12 00:20 by prod_rel_team

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

cube uptime is 20 minutes
System returned to ROM by power-on
System restarted at 11:55:41 CST Wed Mar 6 2013
System image file is "flash0:c3900e-universalk9-mz.SPA.152-4.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/wvl/export/crypto/tool/stqrg.html>

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

Cisco CISC03925-CHASSIS (revision 1.0) with C3900-SPE200/K9 with 752640K/295936K bytes of memory.
Processor board ID FHK1406F15E
4 FastEthernet interfaces
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.
254464K bytes of ATA System CompactFlash 0 (Read/Write)

License Info:

License UDI:

Device#	PID	SN
*0	C3900-SPE200/K9	FOC14034Z5C



Technology Package License Information for Module:'c3900e'

Technology	Technology-package Current	Technology-package Type	Technology-package Next reboot
ibase	ibasek9	Permanent	ibasek9
security	securityk9	Permanent	securityk9
uc	uck9	Permanent	uck9
data	datak9	Permanent	datak9

Configuration register is 0x2104

Running Configuration on CME:

```
CUBE#show run
```

```
Building configuration...
```

```
Current configuration : 25853 bytes
!
! Last configuration change at 08:53:53 CST Mon Feb 25 2013 by cisco
! NVRAM config last updated at 09:27:05 CST Mon Feb 25 2013 by cisco
! NVRAM config last updated at 09:27:05 CST Mon Feb 25 2013 by cisco
version 15.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname cube
!
boot-start-marker
boot system flash c3900e-universalk9-mz.SPA.152-1.T1.bin
warm-reboot
boot-end-marker
!
!
logging buffered 9999999
no logging console
logging monitor errors
enable secret 5 $1$RB6P$34wxSXEeEEeGwf9cUjcjo0
!
no aaa new-model
clock timezone CST -6 0
!
!
!
!
!
!
no ip domain lookup
ip host asr-en-us 10.64.1.37
ip host tts-en-us 10.64.1.37
ip host lync.lynclabkm2013.local 10.85.0.253
```



```
ip host cube.lyncclabkm2013.local 10.64.2.194
ip host dc.lyncclabkm2013.local 10.85.0.254
ip cef
no ipv6 cef
!
multilink bundle-name authenticated
!
!
voice-card 0
 dspfarm
 dsp services dspfarm
!
!
voice service voip
 ip address trusted list
  ipv4 10.64.1.72
  ipv4 10.64.2.195
  ipv4 174.46.0.196
  ipv4 10.64.2.196
  ipv4 10.64.1.37
  ipv4 10.70.18.17
  ipv4 10.70.18.2
  ipv4 0.0.0.0
  ipv4 0.0.0.0 0.0.0.0
  ipv4 10.70.72.10
  ipv4 10.70.12.5

 address-hiding
 mode border-element
 allow-connections sip to sip
 allow-connections sip to sip
 no supplementary-service sip moved-temporarily
 no supplementary-service sip refer
 supplementary-service media-renegotiate
 signaling forward unconditional
 fax protocol pass-through g7111
 sip
  rellxx disable
  min-se 180 session-expires 180
  session refresh
  header-passing
  registrar server expires max 600 min 60
  early-offer forced
  midcall-signaling passthru
  sip-profiles 101
!
voice class codec 42
 codec preference 1 g729br8
 codec preference 2 g711ulaw
!
!
voice class sip-profiles 101
 request INVITE sip-header Allow-Header remove
 response 200 sip-header Allow-Header remove
 response 180 sip-header Allow-Header remove
 response 183 sip-header Allow-Header remove
```

¹ Fax tested with G711u only

² G729 was offered first, all calls were established with G711



```
!  
!  
voice translation-rule 83  
  rule 1 /\(2145450305\) / /2144450305/  
  rule 2 /\(2145450308\) / /2144450308/  
  rule 3 /\(2145450309\) / /2144450309/  
!  
voice translation-rule 94  
  rule 1 /\(2144450304\) / /0304/  
  rule 2 /\(2144450305\) / /0305/  
  rule 3 /\(2144450306\) / /0306/  
  rule 4 /\(2144450307\) / /0307/  
!  
!  
translation-rule 105  
  Rule 1 0304 2654  
  Rule 2 0305 2655  
!  
!  
interface GigabitEthernet0/06  
  description To SIP service providers  
  ip address 10.70.78.214 255.255.255.0  
  duplex full  
  speed 100  
!  
interface GigabitEthernet0/17  
  description Connection to CUCM  
  ip address ip address 10.64.2.194 255.255.0.0  
  duplex full  
  speed auto  
!  
interface GigabitEthernet0/2  
  no ip address  
  shutdown  
  duplex auto  
  speed auto  
!  
interface GigabitEthernet0/3  
  no ip address  
  shutdown  
  duplex auto  
  speed auto  
!  
interface FastEthernet0/0/0  
  no ip address  
!  
interface FastEthernet0/0/1  
  no ip address  
!  
interface FastEthernet0/0/2  
  no ip address  
!  
!
```

³ Translation rule for test cases CPE to CPE through SP

⁴ Translation rule for 10 digits into 4 digit private extension

⁵ Translation rule for 4 digits (from Service Provider) to 4 digit private extensions on

⁶ Cisco UBE WAN side connection to Cisco UCM

⁷ Cisco UBE LAN side connection to tw telecomm



```
interface FastEthernet0/0/3
  no ip address
!
interface Vlan1
  no ip address
!
!
ip default-gateway 10.70.78.1
ip forward-protocol nd
!
ip http server
no ip http secure-server
!
ip route 0.0.0.0 0.0.0.0 97.79.185.129
ip route 10.0.0.0 255.0.0.0 GigabitEthernet0/1
ip route 10.70.78.0 255.255.255.0 GigabitEthernet0/0
!
!
nls resp-timeout 1
cpd cr-id 1
!
!
control-plane
!
!
voice-port 0/1/0
!
!
voice-port 0/1/1
!
!
mgcp profile default
!
sccp local GigabitEthernet0/1
sccp ccm 10.64.2.194 identifier 1 version 4.0
sccp
!
!
dial-peer voice 510 voip8
  description Incoming TW-Telecom
  session protocol sipv2
  incoming called-number 21444503..
  voice-class codec 4 offer-all
  dtmf-relay rtp-nte
  no vad
!
dial-peer voice 511 voip9
  description Incomming TM-CUBE to CUCM
  destination-pattern 214445.%
  session protocol sipv2
  session target ipv4:10.70.19.3:5060
  voice-class codec 4 offer-all
  dtmf-relay rtp-nte
  no vad
!
```

⁸ Incoming dial peer, from TW to Cisco UBE

⁹ Dial peer for routing calls towards PBX



```
dial-peer voice 512 voip10
  description from CUCM to CUBE
  session protocol sipv2
  incoming called-number .%
  voice-class codec 4 offer-all
  dtmf-relay rtp-nte
  no vad
!
!
dial-peer voice 513 voip11
  description Outbound trunk to tw telecom
  destination-pattern .....
  session protocol sipv2
  session target ipv4:10.70.78.2:5060
  voice-class codec 4 offer-all
  dtmf-relay rtp-nte
  no vad
!
!
!
dial-peer voice 515 voip12
  description to call PBX-PBX
  translation-profile outgoing TESTC3
  destination-pattern 214545...
  session protocol sipv2
  session target ipv4:10.70.78.2:5060
  voice-class codec 4 offer-all
  dtmf-relay rtp-nte
  no vad
!
!
dial-peer voice 516 voip13
  description 4 digit to private
  destination-pattern 265.
  translate-outgoing called 10
  session protocol sipv2
  session target ipv4:10.70.19.3
  incoming called-number 030.
  voice-class codec 4 offer-all
  dtmf-relay rtp-nte
!
!

dial-peer voice 517 voip14
  description Intl calls
  destination-pattern 011.T
  session protocol sipv2
  session target ipv4:10.70.78.2:5060
  voice-class codec 4 offer-all
  dtmf-relay rtp-nte
  no vad
```

¹⁰ Incoming dial peer for calls from Cisco UCM to Cisco UBE
¹¹ Incoming dial peer for routing calls towards Service Provider
¹² Outgoing dial peer for routing calls from CPE- to-CPE – via Service Provider
¹³ Incoming dial peer for 4 digits translation into 4 digit private ext.
¹⁴ Dial peer for routing international calls.



```
!  
!  
sip-ua  
!  
!  
gatekeeper  
  shutdown  
!  
!  
  
line con 0  
  login local  
line aux 0  
line vty 0 4  
  exec-timeout 0 0  
  login local  
  transport input telnet ssh  
!  
exception crashinfo dump command sh controllers g0/0  
exception crashinfo dump command sh controller g0/0  
scheduler allocate 20000 1000  
ntp server 10.10.10.5  
!  
end
```



Configuring the Cisco Unified Communications Manager

Version: 9.1

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 ▾ Help ▾

Cisco Unified CM Administration
System version: 9.1.1.10000-11



SIP Trunk Security Profile:

Figure 1 Non-Secure SIP Trunk Profile

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Trunk Security Profile. The page title is "SIP Trunk Security Profile Configuration". The status is "Ready". The configuration details are as follows:

Field	Value
Name *	Non Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null String
Device Security Mode	Non Secure
Incoming Transport Type *	TCP+UDP
Outgoing Transport Type	UDP
Enable Digest Authentication	<input type="checkbox"/>
Nonce Validity Time (mins) *	600
X.509 Subject Name	
Incoming Port *	5060
Enable Application level authorization	<input type="checkbox"/>
Accept presence subscription	<input type="checkbox"/>
Accept out-of-dialog refer **	<input type="checkbox"/>
Accept unsolicited notification	<input type="checkbox"/>
Accept replaces header	<input type="checkbox"/>
Transmit security status	<input type="checkbox"/>
Allow charging header	<input type="checkbox"/>
SIP V.150 Outbound SDP Offer Filtering *	No Filtering



SIP Profile:

Figure 2 SIP profile: Standard SIP Profile- (1/3)

The screenshot shows the Cisco Unified CM Administration interface for configuring a SIP Profile. The page title is "SIP Profile Configuration" and it includes a navigation menu with options like System, Call Routing, Media Resources, etc. The main content area is titled "SIP Profile Information" and contains the following fields and options:

Name *	Standard SIP Profile
Description	Default SIP Profile
Default MTP Telephony Event Payload Type *	101
Early Offer for G.Clear Calls *	Disabled
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites *	TIAS and AS
User-Agent and Server header information *	Send Unified CM Version Information as User-Agent Header
Accept Audio Codec Preferences in Received Offer *	Default
Dial String Interpretation *	Phone number consists of characters 0-9, *, #, and + (others treated

Below the fields, there are several checkboxes for additional configuration options:

- Redirect by Application
- Disable Early Media on 180
- Outgoing T.38 INVITE include audio mline
- Enable ANAT
- Require SDP Inactive Exchange for Mid-Call Media Change
- Use Fully Qualified Domain Name in SIP Requests
- Assured Services SIP conformance



Figure 3 SIP profile: Standard SIP Profile- (2/3)

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration administrator | Search Documentation | About | Log

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

SIP Profile Configuration Related Links: Back To Find/List Go

Copy Reset Apply Config Add New

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
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



Figure 4 SIP profile: Standard SIP Profile- (3/3)

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 Help

SIP Profile Configuration Related Links: Back To Find/List Go

Copy Reset Apply Config Add New

Normalization Script

Normalization Script < None >

Enable Trace

	Parameter Name	Parameter Value	
1	<input type="text"/>	<input type="text"/>	<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 * Disabled

Video Call Traffic Class * Mixed

Calling Line Identification Presentation * Default

Deliver Conference Bridge Identifier

Early Offer support for voice and video calls (insert MTP if needed)

Send send-receive SDP in mid-call INVITE

Allow Presentation Sharing using BFCP

Allow iX Application Media

Allow Passthrough of Configured Line Device Caller Information

Reject Anonymous Incoming Calls

Reject Anonymous Outgoing Calls

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



Trunk to Cisco UBE:

Figure 5 trunk to Cisco UBE (1/3)

The screenshot shows the Cisco Unified CM Administration interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and it includes a navigation bar with "Navigation Cisco Unified CM Administration administrator | Search Documentation | About | Log Out". Below the navigation bar is a menu with "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Help". The main content area is titled "Trunk Configuration" and includes a "Related Links: Back To Find/List" button and a "Go" button. Below this are icons for "Save", "Delete", "Reset", and "Add New".

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name *	CUBE
Description	CUBE Trunk
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

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 *

Route Class Signaling Enabled *

Use Trusted Relay Point *

PSTN Access

Run On All Active Unified CM Nodes



Figure 6 Trunk to CUBE (2/3)

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 ▾ Help ▾

Trunk Configuration Related Links: Back To Find/List Go

Save
 Delete
 Reset
 Add New

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain < None >

Call Routing Information

Remote-Party-Id

Asserted-Identity

Asserted-Type* Default

SIP Privacy* Default

Inbound Calls

Significant Digits* 4

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

Clear Prefix Settings
Default Prefix Settings

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

Connected Party Settings

Connected Party Transformation CSS < None >

Use Device Pool Connected Party Transformation CSS



Figure 7 Trunk to CUBE (3/3)

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

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

SIP Information

Destination

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	<input type="text" value=" 10.64.2.194"/>	<input type="text"/>	<input type="text" value=" 5060"/>

MTP Preferred Originating Codec*

BLF Presence Group*

SIP Trunk Security Profile*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile*

DTMF Signaling Method*

Normalization Script

Normalization Script

Enable Trace

	Parameter Name	Parameter Value
1	<input type="text"/>	<input type="text"/>

Geolocation Configuration

Geolocation

Geolocation Filter

Send Geolocation Information



Configuring Route Patterns

Route patterns are configured as below, prefixed by an 8 to identify that a call has to be routed via the CUBE.

The screenshot shows the Cisco Unified CM Administration web interface. At the top, there is a navigation bar with the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and a "Navigation" dropdown menu set to "Cisco Unified CM Administration" with a "Go" button. Below this, there are links for "administrator", "Search Documentation", "About", and "Logout". A secondary navigation bar contains dropdown menus for "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Help". The main content area is titled "Find and List Route Patterns" and includes a toolbar with icons for "Add New", "Select All", "Clear All", and "Delete Selected". Below the toolbar, a table displays a route pattern with the value "8.@" in the first column and "CUBE" in the second column. A small icon is visible in the top right corner of the table area.



Acronyms

Acronym	Definitions
Cisco UBE	Cisco Unified Border Element
Cisco UCM	Cisco Unified Communications Manager
PSTN	Public Switched Telephone Network
SIP	Session Initiation Protocol
eSBC	Enterprise Session Border Controller



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Application Note

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_Test
_Plan&Report_CUCM

Test Scenario	Description/Details	Test Results (Pass/Fail)	Comments
CPE outbound to SP Offnet gateway(PSTN) (G.729 is offered first)			
	Call ringback:	Pass	G729 not supported. Tested G711
	Voice cut through on connect	Pass	G729 not supported. Tested G711
	ring terminated on calling party disconnect	Pass	G729 not supported. Tested G711
	long duration call: 1 hour	Pass	G729 not supported. Tested G711
	DTMF relay (both directions)	Pass	G729 not supported. Tested G711
	Called party disconnect; calling party automatically disconnected	Pass	G729 not supported. Tested G711
	Calling party disconnect; called party automatically disconnected	Pass	G729 not supported. Tested G711
SP offnet gateway(PSTN) inbound to CPE (G.729 offered first)			



	Call ringback:	Pass	G729 not supported. Tested G711
	Voice cut through on connect	Pass	G729 not supported. Tested G711
	ring terminated on calling party disconnect	Pass	G729 not supported. Tested G711
	long duration call: 1 hour	Pass	G729 not supported. Tested G711
	DTMF relay	Pass	G729 not supported. Tested G711
	Called party disconnect; calling party automatically disconnected	Pass	G729 not supported. Tested G711
	Calling party disconnect; called party automatically disconnected	Pass	G729 not supported. Tested G711
CPE to CPE (place call out to the SP network and back) (G.729 is offered first)			
	Call ringback:	Pass	G729 not supported. Tested G711
	Voice cut through on connect	Pass	G729 not supported. Tested G711
	ringback terminated on caller disconnect	Pass	G729 not supported. Tested G711
	long duration call: 1 hour	Pass	G729 not supported. Tested G711
	DTMF relay	Pass	G729 not supported. Tested G711
	Called party disconnect; calling party automatically disconnected	Pass	G729 not supported. Tested G711
	Calling party disconnect; called party automatically disconnected	Pass	G729 not supported. Tested G711
CPE Calling number privacy			
	Pass calling number: IP PBX to Offnet: Mark Calling Number Private	Pass	



	Pass calling number: IP PBX to another IP PBX: Mark Calling Number Private	Pass	
CPE Telephone Number Support			
	IP PBX to offnet: translate private extension to 10 DID calling number	Pass	
	Offnet to IP PBX: IP PBX must translate 10 digit called number to private extension.	Pass	
	IP PBX to IP PBX: translate private extension to 10 DID calling number	Pass	
	IP PBX to IP PBX: IP PBX must translate 10 digit called number to private extension.	Pass	
	Offnet to IP PBX: IP PBX must translate 4 digit called number to private extension.	Pass	
	IP PBX to IP PBX: IP PBX must translate 4 digit called number to private extension.	Pass	
CPE Calling Name Delivery			
	IP PBX to IP PBX: pass display name	Pass	
CPE offnet Call Conference			
	Offnet1 to IP PBX phone 1, IP PBX phone1 conferences Offnet2	Pass	
	IP PBX phone1 to Offnet1, IP PBX phone1 conferences Offnet2	Pass	
CPE Intra-Site Call Conference			
	Phone1 to Phone2. Phone1 conferences Phone3 at 2nd IP PBX site	Pass	Conference dropped when controller dropped due to configuration on Cisco UCM
	Phone1 to Phone2. Phone1 conferences Offnet PSTN	Pass	
	Phone1 to Phone 3 at 2nd IP PBX site. Phone3 conferences Phone2 at Phone1 site	Pass	
	Phone1 to Offnet PSTN. Phone 1 conferences Phone2	Pass	
	Phone3 at 2nd IP PBX site to Phone1. Phone1 conferences Phone2	Pass	Conference dropped when controller dropped due to configuration on Cisco UCM



	Offnet PSTN to Phone1. Phone1 conferences Phone2	Pass	
CPE Intra-Site Attended Call Transfer			
	Offnet 1 to IP PBX phone 1, IP PBX phone1 transfers to Offnet2 (does caller ID update on Offnet2?)	Pass	No caller id update-both PSTN and Phone 2 have Phone1 caller id info
	IP PBX phone1 to Offnet1, IP PBX phone1 transfers to Offnet 2 (does caller ID update on Offnet2?)	Pass	No caller id update-both PSTN and Phone 2 have Phone1 caller id info
	Phone1 to Phone2. Phone1 transfers to Phone3 at 2nd IP PBX site	Pass	
	Phone1 to Phone2. Phone1 transfers to Offnet PSTN	Pass	
	Phone1 to Phone 3 at 2nd IP PBX site. Phone3 transfers to Phone2 at Phone1 site	Pass	
	Phone1 to Offnet PSTN. Phone 1 transfers to Phone2	Pass	
	Phone3 at 2nd IP PBX site to Phone1. Phone1 transfers to Phone2	Pass	
	Offnet PSTN to Phone1. Phone1 transfers to Phone2	Pass	
CPE Intra-Site Unattended Call Transfer			
	Offnet 1 to IP PBX phone 1, IP PBX phone1 transfers to Offnet2 (does caller ID update on Offnet2?)	Pass	No caller id update
	IP PBX phone1 to Offnet1, IP PBX phone1 transfers to Offnet 2 (does caller ID update on Offnet2?)	Pass	No caller id update
	Phone1 to Phone2. Phone1 transfers to Phone3 at 2nd IP PBX site	Pass	



	Phone1 to Phone2. Phone1 transfers to Offnet PSTN	Pass	
	Phone1 to Phone 3 at 2nd IP PBX site. Phone3 transfers to Phone2 at Phone1 site	Pass	
	Phone1 to Offnet PSTN. Phone 1 transfers to Phone2	Pass	
	Phone3 at 2nd IP PBX site to Phone1. Phone1 transfers to Phone2	Pass	
	Offnet PSTN to Phone1. Phone1 transfers to Phone2	Pass	
CPE Intra-Site Blind Call Transfer (if SIP blind transfer is supported these test cases apply)			
	Offnet 1 to IP PBX phone 1, IP PBX phone1 transfers to Offnet2 (does caller ID update on Offnet2?)	Pass	
	IP PBX phone1 to Offnet1, IP PBX phone1 transfers to Offnet 2 (does caller ID update on Offnet2?)	Pass	
	Phone1 to Phone2. Phone1 transfers to Phone3 at 2nd IP PBX site	Pass	
	Phone1 to Phone2. Phone1 transfers to Offnet PSTN	Pass	
	Phone1 to Phone 3 at 2nd IP PBX site. Phone3 transfers to Phone2 at Phone1 site	Pass	
	Phone1 to Offnet PSTN. Phone 1 transfers to Phone2	Pass	
	Phone3 at 2nd IP PBX site to Phone1. Phone1 transfers to Phone2	Pass	
	Offnet PSTN to Phone1. Phone1 transfers to Phone2	Pass	
CPE Call Hold and Resume (call hold is always done on the IP PBX side)			
	IP PBX to Offnet PSTN	Pass	
	IP PBX to IP PBX	Pass	
	Offnet PSTN to IP PBX	Pass	



CPE Voice Mail (e.g. using Unity or Unity Connection)			
	Offnet to IP PBX: leave voice mail	Pass	
	Offnet to IP PBX: retrieve voice mail	Pass	
SP Voice Mail (e.g. using mobile phone voicemail)			
	IP PBX to Offnet (mobile VM): leave voice mail	Pass	
	IP PBX to Offnet (mobile VM): retrieve voice mail	Pass	
CPE Find Me (Call Forward Unconditionally)			
	Offnet to IP PBX call invokes to find me feature	Pass	
	IP PBX to IP PBX call invokes to find me feature	Pass	
	Offnet to IP PBX phone1 call invokes find me feature to offnet	Pass	
	IP PBX to IP PBX phone1 call invokes find me feature to offnet	Pass	
CPE T.38 FAX G3 (G.729 is offered first)			
	G3 CPE to PSTN fax - G3-G3	N/S	
	G3 CPE to PSTN fax - G3-SG3	N/S	
	G3 CPE from PSTN fax - G3-G3	N/S	
	G3 CPE from PSTN fax - SG3-G3	N/S	
	G3 CPE to CPE fax - G3-G3	N/S	
	G3 CPE to CPE fax - G3-SG3	N/S	
CPE T.38 FAX SG3 (G.729 is offered first)			
	SG3 CPE to PSTN fax - SG3-G3	N/S	
	SG3 CPE to PSTN fax - SG3-SG3	N/S	
	SG3 CPE from PSTN fax - G3-SG3	N/S	
	SG3 CPE from PSTN fax -SG3-SG3	N/S	
	SG3 CPE to CPE fax - SG3-G3	N/S	
	SG3 CPE to CPE fax - SG3-SG3	N/S	



Simultaneous Calls (Minimum 2)			
	CPE to PSTN Offnet gateway	Pass	
	Offnet gateway inbound to CPE	Pass	
	CPE to CPE	Pass	
CPE Auto Attendant (e.g. Unity or Unity Connection AA service to transfer over to a user)			
	Offnet to IP PBX: call auto attendant	Pass	
	Offnet to IP PBX: connect to extension via auto attendant	Pass	
CPE to PSTN Offnet gateway international call			
	Call ringback	Pass	
	Voice cut through on connect	Pass	
	ringback terminated on caller disconnect	Pass	
CPE G.711 FAX G3			
	G3 CPE to PSTN fax - G3-G3	Pass	
	G3 CPE to PSTN fax - G3-SG3	Pass	
	G3 CPE from PSTN fax - G3-G3	Pass	
	G3 CPE from PSTN fax - SG3-G3	Pass	
	G3 CPE to CPE fax - G3-G3	Pass	
	G3 CPE to CPE fax - G3-SG3	Pass	
CPE G.711 FAX SG3			
	SG3 CPE to PSTN fax - SG3-G3	Pass	
	SG3 CPE to PSTN fax - SG3-SG3	Pass	
	SG3 CPE from PSTN fax - G3-SG3	Pass	
	SG3 CPE from PSTN fax -SG3-SG3	Pass	
	SG3 CPE to CPE fax - SG3-G3	Pass	
	SG3 CPE to CPE fax - SG3-SG3	Pass	



CPE Find Me (Call Forward On Busy)			
	Offnet to IP PBX phone1 call invokes find me feature to phone2	Pass	
	IP PBX to IP PBX phone1 call invokes find me feature to phone2	Pass	
	IP PBX to IP PBX phone1 call invokes find me feature to offnet	Pass	
CPE Find Me (Call Forward Don't Answer)			
	Offnet to IP PBX phone1 call invokes find me feature to phone2	Pass	
	Offnet to IP PBX phone1 call invokes find me feature to offnet	Pass	
	IP PBX to IP PBX phone1 call invokes find me feature to phone2	Pass	
	IP PBX to IP PBX phone1 call invokes find me feature to offnet	Pass	
Codec mid-call re-negotiation (to be tested without transcoder)			
	Offnet calls IP PBX phone 1 (G729), phone 1 transfers to UM/gateway (g711u). Offnet and IP PBX UM/gateway re-negotiate codec and call is transferred.	N/T	G729 not supported
	IP PBX phone 1 calls Offnet phone (call is G711), offnet phone transfers call to IP PBX phone 2 (G729 region), calls sets up between IP PBX phone1 and IP PBX phone2	N/T	G729 not supported
Dial Plans			
	Test 0, 0+10, 911, 411 1+10	Pass*	Not tested 911,411.
PRACK with SDP (early-media cut-through with DTMF (RFC2833) navigation before 200OK) - call 800-864-8331 - United Airlines			
	IP PBX phone1 call 800 number, phone user navigates through AA to reach correct menu option.	N/S	eSBC does not support/implement PRACK.



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