



Verizon IP Trunking Service: Connecting Cisco Unified Communications Manager 6.1(2) via the Cisco Unified Border Element using SIP

October 27, 2008

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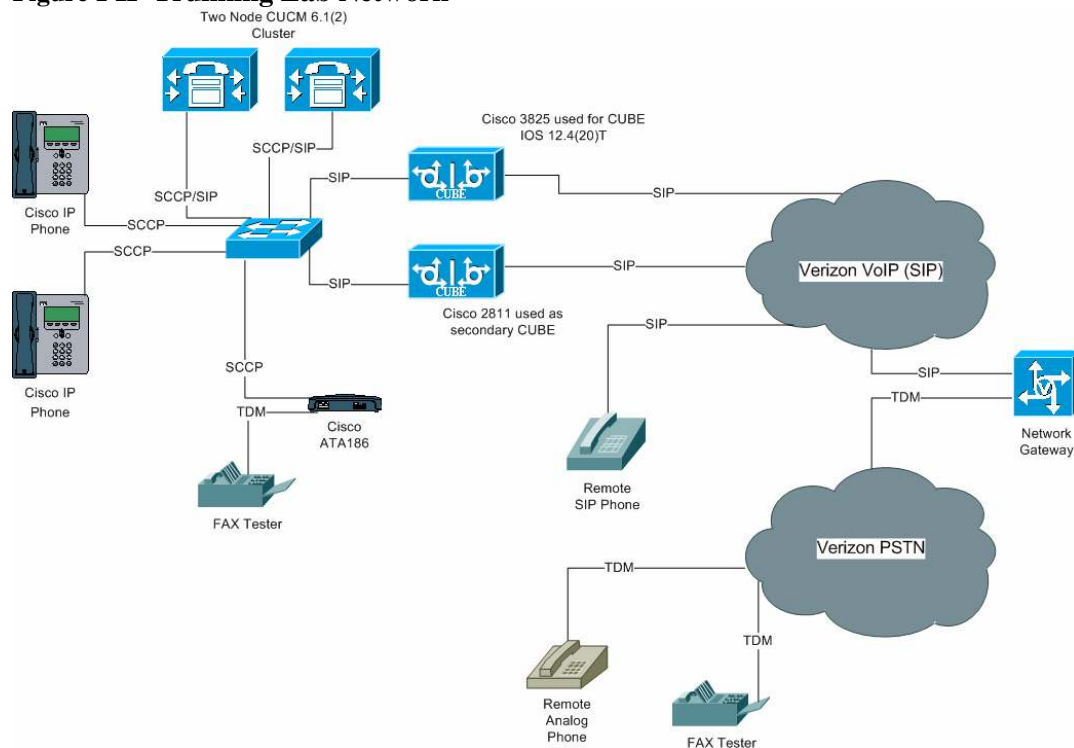
Introduction

Service Providers today, such as Verizon, 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. Verizon's IP Trunking service is an SP 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 (CUCM) 6.1.2 with a Cisco Unified Border Element (CUBE) for connectivity to Verizon's SIP trunk service. The deployment model covered in this application note is CPE (CUCM6.1.2/CUBE) to PSTN via the Verizon IP Trunking Service
- Testing was performed in accordance with the test plan for the Verizon IP trunking service, and all features were verified.
- Although this document does not detail the results of the testing performed it provides the essential commands required for interoperability between CUCM/CUBE and the Verizon IP Trunking Service.



Figure 1 IP Trunking Lab Network



System Components

Hardware Components

- Cisco CUBE 1.2 (IOS image version 12.4(20)T or later)
- Cisco Unified Border Element is an integrated Cisco IOS Software application that runs on various IOS platforms, follow the link for more details: <http://www.cisco.com/go/cube>
- Packet Voice Data Module (PVDM). You will need to install DSP modules (PVDM) on CUBE if you require MTP, Transcoding or Conference Bridge resources for codecs other than G.711.
- CUCM cluster with (2) Cisco MCS 7800 Series server (Cisco Unified Communications Manager)
- Cisco IP Phones



- Cisco ATA186 device (only needed for FAX, modem, or analog phones)

Software Requirements

- Cisco Unified Call Manager 6.1.2.1000-13 (6.1(2))
- Cisco Unified Border Element CUBE version 1.2 (IOS version 12.4(20)T or later. Advanced IP Services feature set was used for this configuration)
- The documented CUBE configuration can be supported with the following IOS feature sets: IP VOICE, SP SERVICES, ADVANCED IP SERVICES, ADVANCED ENTERPRISE SERVICES, INT VOICE/VIDEO, IPIGW, TDMIP GW,INT VOICE/VIDEO, IPIGW, TDMIP GW AES
- Consult your Cisco representative for the correct IOS image for the specific application and Device Unit License and Feature License requirements for CUCM and CUBE.

Features

Features Supported

- Voice calls using G.729 codec
- RFC3261 generic feature Support
- Locating SIP servers via DNS SRV and DNS A Records
- Early Media Cut-Through with DO to EO
- Calling party number presentation and privacy (P-Asserted Identity)
- FAX using G711ulaw passthrough
- DTMF as RFC2833 NTE (named telephone events) when a compressed audio codec is used
- CUBE: performs Delay-Offer-to-Early-Offer transcoding of an initial SIP INVITE from CUCM without SDP

Features Not Supported

- CUCM/CUBE Codec negotiation of G.726
- T.38 Fax relay is not supported by Verizon IP Trunking Service



Caveats

- A conference bridge (CFB) resource on CUBE is required in order for a CUCM IP phone to initiate a three-way conference between G729 media end-points. See configuration section for details.
- Along with SIP INVITE with early-offer Verizon requires the support of early media for outbound G729 calls. It is necessary to enable PRACK (REL1XX response) on the call leg between CUCM and CUBE. See configuration section for details.
- Verizon's IP Trunking Service only supports G711ulaw codec for FAX calls. G.711 fax passthrough is the supported FAX transport method for the Verizon IP Trunking Service. See configuration section for details
- CUCM can only support a single codec between the end device (i.e. IP Phone, ATA) and the SIP trunk. A workaround for this used during testing was to create multiple Regions and Device pools in order to control the codec selection prior to being presented to the SIP trunk. The end devices were configured with a specific Device Pool based on the codec used for offnet calls. See configuration section for details



Call Flow Overview

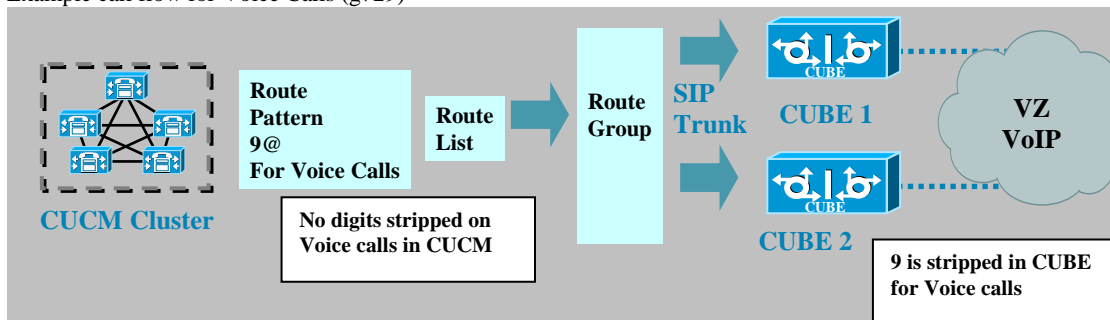
The same SIP trunks are utilized between CUCM to CUBE for both Voice and FAX offnet calls. However, the call type (i.e., Voice vs. FAX) must be differentiated to ensure the desired codec is used. This delineation is achieved by performing digit manipulation at the RL (Route List) prior to the call being delivered to the RG.

Each type of device (i.e., IP Phones vs analog devices for FAX) will have separate RPs (Route-Patterns) that belongs to their respective partition. The route patterns will then route the calls to the specified RL.

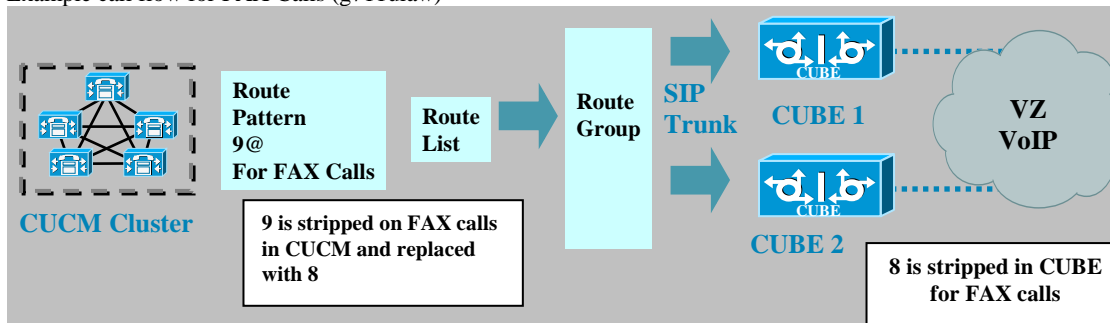
The RL is used to distinguish a Voice call from a FAX call by manipulating the called party numbers (a leading 9 for Voice and an 8 for FAX). After the digit manipulation, the Route Lists forward the calls to their respective RG and the RG routes the call to the SIP trunks.

The SIP trunks are the same for ALL calls from CUCM to CUBE.

Example call flow for Voice Calls (g729)



Example call flow for FAX Calls (g711ulaw)





Configuration

Configuration of Cisco Unified Border Element (CUBE 1.2) IOS Version 12.4(20)T

Critical commands are marked in **Bold** with footnotes at bottom of the page

```
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service sequence-numbers
!
hostname cube.1.3825
!
boot-start-marker
boot-end-marker
!
card type t1 2 1
logging message-counter syslog
logging buffered 100000
no logging console
enable password XXXX
!
no aaa new-model
no network-clock-participate slot 2
no network-clock-participate wic 0
!
dot11 syslog
ip source-route
no ip dhcp use vrf connected
ip dhcp excluded-address 192.168.0.0 192.168.0.100
!
!
!
ip dhcp pool IPPHONES1
    network 192.168.0.0 255.255.255.0
```

¹ (Optional) DHCP Service: automatically assign IP address and TFTP server configuration to Phones



```
default-router 192.168.0.10
option 150 ip 192.168.0.6
!
!
ip cef
!
!
ip domain name SIP.Domain.com
ip name-server X.X.X.X---SIP DNS Server
!
no ipv6 cef
multilink bundle-name authenticated
!
!
!
Voice-card 0
dspfarm
dsp services dspfarm
!
Voice-card 2
dspfarm
!
!
Voice service voip
  allow-connections sip to sip2
  FAX protocol pass-through g711ulaw
  h323
  sip
  early-offer forced3
  midcall-signaling passthru4
!
!
Voice class codec 1
codec preference 1 g729r8 bytes 20
codec preference 2 g711ulaw bytes 160
```

² Allow SIP to SIP call processing

³ Use this command to forcefully configure a Cisco Unified Border Element to send a SIP invite with SDP on the Out-Leg (OL), Delayed-Offer to Early-Offer for SIP calls.

⁴ Enables support for SIP Supplementary Services (Only used for SIP-to-SIP Calls)



```
!  
!  
!  
!  
!  
voice translation-rule 8  
  rule 2 /^8\(.*\)/ /\1/  
!  
voice translation-rule 9  
  rule 2 /^9\(.*\)/ /\1/  
!  
!  
voice translation-profile DIGITSTRIP-85  
  translate called 8  
!  
voice translation-profile DIGITSTRIP-96  
  translate called 9  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
username cisco privilege 15 secret <XXXX>  
archive  
  log config  
  hidekeys  
!  
!  
!  
!  
!  
!  
ip ssh version 2  
!
```

⁵ Strip the leading “8” from outgoing called number

⁶ Strip the leading “9” from outgoing called number



```
!  
!  
!  
interface GigabitEthernet0/0  
  description CUBE inside interface7  
  ip address 192.168.0.X 255.255.255.0  
  ip virtual-reassembly  
  load-interval 30  
  duplex auto  
  speed auto  
  media-type rj45  
!  
interface GigabitEthernet0/1  
  description CUBE outside interface  
  ip address X.X.X.X 255.255.255.X  
  ip virtual-reassembly  
  load-interval 600  
  duplex auto  
  speed auto  
  media-type rj45  
!  
ip route 0.0.0.0 0.0.0.0 X.X.X.X  
no ip http secure-server  
!  
!  
ip rtcp report interval 10000  
!  
!  
!  
!  
!  
!  
control-plane  
!  
call threshold global cpu-avg low 68 high 75  
call threshold global total-mem low 75 high 85  
call threshold global total-calls low 9 high 118
```

⁷ No SIP bind commands configured, SIP is sourced from both interfaces

⁸ Global Call Admission Control based on Resource utilization



```
!  
!  
!  
!  
Voice-port 0/1/09  
  no non-linear  
  playout-delay maximum 120  
  playout-delay nominal 15  
  playout-delay minimum low  
  timeouts interdigit 2  
  timeouts call-disconnect 3  
  timing digit 300  
  caller-id enable  
!  
Voice-port 0/1/1  
!  
!  
sccp local GigabitEthernet0/0  
sccp ccm 192.168.0.6 identifier 2 priority 2 version 5.0.1  
sccp ccm 192.168.0.4 identifier 5 priority 1 version 5.0.1  
sccp  
!  
sccp ccm group 10  
  associate ccm 5 priority 1  
  associate ccm 2 priority 2  
  associate profile 12 register CON001193484810  
!  
dspfarm profile 12 conference10  
  description conference bridge  
  codec g711ulaw  
  codec g711alaw  
  codec g729ar8  
  codec g729abr8  
  codec g729r8  
  codec g729br8  
  maximum sessions 10  
  associate application SCCP  
!  
!
```

⁹ **Optional FXS port for FAX devices connected directly to the CUBE**

¹⁰ **DSP resources for Conferencing registered with CUCM cluster**



```
!  
dial-peer Voice 10 pots  
  description dial '9' for outbound calls  
  preference 1  
  service session  
  destination-pattern 10XX  
  FAX rate disable  
  port 0/1/0  
!  
dial-peer Voice 100 voip  
  description OUTBOUND G729 Voice SIP calls to VzB  
  translation-profile outgoing DIGITSTRIP-911  
  destination-pattern 9T12  
  Voice-class codec 1  
  Voice-class sip early-offer forced13  
  session protocol sipv2  
  session target sip-server  
  dtmf-relay rtp-nte14  
  no vad  
!  
dial-peer Voice 101 voip  
  description INBOUND G729 Voice SIP calls from VzB  
  Voice-class codec 1  
  session protocol sipv2  
  session target sip-server  
  incoming called-number 1...15  
  dtmf-relay rtp-nte  
  no vad  
!
```

¹¹ Strip the leading “9” from outgoing called number

¹² Match on outbound calls from CUCM with leading “9”

¹³ Use this command to forcefully configure a Cisco Unified Border Element to send a SIP invite with SDP on the Out-Leg (OL), Delayed-Offer to Early-Offer for SIP calls.

¹⁴ Forwards DTMF tones by using RTP with the Named Telephone Event (NTE) payload type.

¹⁵ Enables CUBE to set configuration parameters to incoming calls based on the received called number. This ensures that both legs of the SIP call have matching CODECs



```
!  
dial-peer Voice 200 voip  
  description inbound FAX dial peer from VZ  
  session protocol sipv2  
  session target sip-server  
  incoming called-number 101816  
  codec g711ulaw17  
  FAX rate disable18  
  no vad  
!  
dial-peer Voice 201 voip  
  description outbound FAX calls to VZ  
  translation-profile outgoing DIGITSTRIP-819  
  destination-pattern 8T20  
  Voice-class sip early-offer forced  
  session protocol sipv2  
  session target sip-server  
  codec g711ulaw  
  FAX rate disable  
  no vad  
!  
  
dial-peer Voice 300 voip  
  description To/From CUCM subscriber for FAX Calls  
  preference 2  
  destination-pattern 1018  
  session protocol sipv2  
  session target ipv4:192.168.0.4  
  incoming called-number 8T21
```

¹⁶ Match on inbound call-leg for FAX calls. This ensures that both legs of the SIP call have matching CODECs.

¹⁷ CODEC is set on dial-peer to force use of g711ulaw for FAX calls.

¹⁸ Disables FAX relay transmission capability. FAX-Passthrough is the supported FAX method.

¹⁹ Strip leading “8” from outbound FAX calls before sending to VZ

²⁰ Match outbound FAX calls from CUCM cluster with leading “8”



```
codec g711ulaw
FAX rate disable
no vad
!
dial-peer Voice 301 voip
description To/From CUCM publisher for FAX Calls
preference 5
destination-pattern 1018
session protocol sipv2
session target ipv4:192.168.0.622
incoming called-number 8T
codec g711ulaw
FAX rate disable
no vad
!
dial-peer Voice 401 voip
description To/From CUCM subscriber for Voice
preference 2
destination-pattern 1...
Voice-class codec 1
session protocol sipv2
session target ipv4:192.168.0.4
incoming called-number 9T23
FAX rate disable
no vad
!
dial-peer Voice 411 voip
description To/From CUCM publisher for Voice
preference 5
destination-pattern 1...
Voice-class codec 1
session protocol sipv2
session target ipv4:192.168.0.624
```

²¹ Enables CUBE to set configuration parameters to outgoing calls based on the received called number. This ensures that both legs of the SIP call have matching CODECs

²² Multiple SIP trunks configured for redundant connections to the CUCM cluster

²³ Enables CUBE to set configuration parameters to outgoing calls based on the received called number. This ensures that both legs of the SIP call have matching CODECs

²⁴ Multiple SIP trunks configured for redundant connections to the CUCM cluster



```
incoming called-number 9T
dtmf-relay rtp-nte
no vad
!
!
sip-ua
retry invite 2
retry bye 2
retry cancel 2
sip-server dns:sip.server.com25
g729-annexb override
!
line con 0
line aux 0
line vty 0 4
!
scheduler allocate 20000 1000
end
```

Configuring Cisco Unified Communications Manager

²⁵ SIP Proxy FQDN name for outbound SIP calls to VZ



Early-Media Cut-thru : Enable PRACK on CUCM

PRACK- Provisional Acknowledgement to a Session not yet established

- Purpose is to acknowledge progress information on a requested process
- The INVITE Includes a Require header stipulating the UAC wants a reliable provisional response

SIP Rel1XX Enabled: This parameter determines whether all SIP provisional responses (other than 100 Trying messages) get sent reliably to the remote SIP endpoint. If this parameter is disabled, Cisco CallManager does not acknowledge or confirm 18X messages. Valid values specify True (acknowledge 18X messages with PRACK) or False (do not acknowledge 18X messages with PRACK).

CUCM Administrator>System>Service Parameters

Change the **SIP Rel1XX Enabled** from the default of **False** to **True**

Service Parameter Configuration		Related Links: Parameters for All Server
SIP Connect Timer *	500	500
SIP Disconnect Timer *	500	500
SIP Expires Timer *	180000	180000
SIP PRACK Timer *	500	500
SIP Rel1XX Timer *	500	500
SIP Trying Timer *	500	500
SIP Publish Timer *	500	500
SIP Rel1XX Enabled *	True	False
SIP Min-SE Value *	1800	1800
SIPS URI Handling *	Reject	Reject
SIP statistics Periodic update Timer *	2	2
SIP Session Expires Timer *	1800	1800
SIP Trunk TspReq Retry *	2	2
SIP TCP Unused Connection Timer *	14	14
SIP TCP Timer *	5	5
SIP Station UDP Port Throttle Threshold *	50	50
SIP Trunk UDP Port Throttle Threshold *	200	200

Note: No changes are required on the CUBE. The CUBE supports PRACK by default.



Media Resource Group List:

List of Media Resource Groups configured for the SIP Trunk MRGL

Media Resource Group List Configuration

Save Delete Copy Reset Add New

Status

Status: Ready

Media Resource Group List Status

Media Resource Group List: SIP_Trunk_MRGL (used by 2 devices)

Media Resource Group List Information

Name*

Media Resource Groups for this List

Available Media Resource Groups



Selected Media Resource Groups



Media Resource Group:

Configured Conference Bridge resource associated with DSP resources configured on CUBE

Media Resource Group Configuration

Save Delete Copy Reset Add New

Status
Status: Ready

Media Resource Group Status
Media Resource Group: Conf_Bridge_MRG (used by 0 devices)

Media Resource Group Information

Name*

Description

Devices for this Group

Available Media Resources **

v ^

Selected Media Resources *

Use Multicast for MOH Audio (If at least one multicast MOH resource is available)



CODEC Selection using Device Pools and Regions

All Voice calls through the SIP trunk should use G.729 and FAX devices should use G.711. Note in the configuration below, there are two regions. Calls between the “Default” and “Offsite” region will use G.729 and calls between “Default” and “Offsite” use G.711. Applying this configuration to our testbed, the SIP trunk is placed in a Device Pool with the “Offsite” region, and phone devices should be placed in a Device Pool that with the “Default” region. Devices used for analog FAX should use a Device Pool with the “Offsite” region. Devices that belong to the same region are configured to use the G.711 codec

Region Configuration

Save ~~X~~ Delete Reset Add New

Region Information

Name*

Region Relationships

Region	Audio Codec
Default	G.711
Offsite	G.729

NOTE: Region(s) not displayed Use System Default

Modify Relationship to other Regions

Regions	Audio Codec
<input type="text" value="Default"/> <input type="text" value="Offsite"/>	<input type="text" value="Keep Current Setting"/>

Save Delete Reset Add New



List of Device pools and the associated Regions

Find and List Device Pools

Add New
 Select All
 Clear All
 Delete Selected

Status

2 records found

Device Pool (1 - 2 of 2)

Find Device Pool where begins with

<input type="checkbox"/>	Name ^	Cisco Unified CM Group	Region
<input type="checkbox"/>	Default	Default	Default
<input type="checkbox"/>	Offsite_DP	Default	Offsite

List of Phones and ATA Devices:

The Device Pools selected determine the codec used for Offnet calls

Find and List Phones

Add New
 Select All
 Clear All
 Delete Selected
 Reset Selected

Status

3 records found

Phone(1 - 3 of 3)

Find Phone where begins with

<input type="checkbox"/>	Device Name(Line) ^	Description	Device Pool	Device Protocol
<input type="checkbox"/>	ATA004455552222	ATA For FAX devices	Offsite_DP	SCCP
<input type="checkbox"/>	SEP001122223333	IP Phone for G729 Voice calls	Default	SCCP
<input type="checkbox"/>	SEP001155551111	IP Phone for G729 Voice calls	Default	SCCP



SIP Trunk Configuration:

The Offsite Device Pool is configured for codec negotiation and the SIP_Trunk_MRGL is selected for Conference Bridge resources. required Not Selected

Note: MTP

Trunk Configuration
Save Delete Reset Add New

Status
Status: Ready

Device Information
Product: SIP Trunk
Device Protocol: SIP
Device Name: To_3825_CUBE
Description: To 3825 CUBE SIP Trunk
Device Pool: Offsite_DP
Common Device Configuration: < None >
Call Classification: OffNet
Media Resource Group List: SIP_Trunk_MRGL
Location: Hub_None
AAR Group: < None >
Packet Capture Mode: None
Packet Capture Duration: 0
 Media Termination Point Required
 Retry Video Call as Audio
 Transmit UTF-8 for Calling Party Name
 Unattended Port

Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain: < None >

Call Routing Information
Inbound Calls
Significant Digits: All
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
Caller ID DN:
 Redirecting Diversion Header Delivery - Outbound

SIP Information
Destination Address: 192.168.0.10
Destination Address: 192.168.0.10
 Destination Address is an SRV
Destination Port: 5060
MTP Preferred Originating Code: 711ulaw
Presence Group: Standard Presence group
SIP Trunk Security Profile: Non Secure SIP Trunk Profile
Rerouting Calling Search Space: < None >
DTMF Signaling Method: No Preference



Route Group Configuration:

Both SIP Trunks are members of the same Route Group

Route Group Configuration

Save ~~X~~ Delete + Add New

Route Group Information

Route Group Name* SIP_Trunks
Distribution Algorithm* Top Down

Route Group Member Information

Find Devices to Add to Route Group
Device Name contains Find
Available Device\$*
To_2811_CUBE
To_3825_CUBE
Port(s) None Available
Add to Route Group

Current Route Group Members

Selected Device\$**
To_3825_CUBE (All Ports)
To_2811_CUBE (All Ports)
Removed Device\$***

Route Group Members

To_3825_CUBE
To_2811_CUBE



Route List for Voice:

The previously defined RG is selected in the Voice RL

Route List Configuration

Save Delete Copy Reset Add New

Status

Status: Ready

Route List Information

Name*
Description
Cisco Unified Communications Manager Group*
 Enable this Route List (change effective on Save; no reset required)

Route List Member Information

Selected Groups**

Removed Groups***

Route List Details

[SIP_Trunks](#)



Route List Details for Voice:

No Digits are discarded for offnet Voice calls.

The leading "9" is preserved when the call is forwarded to the CUBE, this allows the CUBE to differentiate the call as Voice and use the corresponding G729 CODEC.

Route List Detail Configuration		Rel.
Save		
Status		
Status: Ready		
Route List Member Information		
Route Group SIP		
Calling Party Transformations		
Use Calling Party's External Phone Number Mask*	<input type="text" value="Default"/>	
Calling Party Transform Mask	<input type="text"/>	
Prefix Digits (Outgoing Calls)	<input type="text"/>	
Called Party Transformations		
Discard Digits	<input type="text" value="NANP:NoDigits"/>	
Called Party Transform Mask	<input type="text"/>	
Prefix Digits (Outgoing Calls)	<input type="text"/>	



Route List for FAX:

The previously defined RG is selected in the FAX RL (similar to Voice RL)

Route List Configuration

Save Delete Copy Reset Add New

Status

Status: Ready

Route List Information

Name*

Description

Cisco Unified Communications Manager Group*

Enable this Route List (change effective on Save; no reset required)

Route List Member Information

Selected Groups**

Removed Groups***

Route List Details

[SIP_Trunks](#)



Route List Details for FAX:

The “9” is stripped from the called party number and replaced with an “8”.

This dial plan configuration ensures that the user only needs to dial a “9” for Voice and FAX off-net calls.

Route List Detail Configuration

Save

Status

Status: Ready

Route List Member Information

Route Group SIP_Trunks

Calling Party Transformations

Use Calling Party\'s External Phone Number Mask*

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)



Route Pattern for Voice:

This Route Pattern is in the Voice partition and is serviced by the Voice RL (No Digits are stripped)



Route Pattern Configuration

Save Delete Copy Add New

Pattern Definition

Route Pattern* 9.@
Route Partition Voice
Description Offnet G729 calls
Numbering Plan* NANP
Route Filter < None >
MLPP Precedence Default
Resource Priority Namespace Network Domain < None >
Gateway/Route List* Voice_G729_RL (Edit)
Route Option
 Route this pattern
 Block this pattern No Error
Call Classification* OffNet
 Allow Device Override Provide Outside Dial Tone All 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 < None >
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type* Cisco CallManager
Called Party Numbering Plan* Cisco CallManager

Route Pattern for FAX:

This Route Pattern is in the FAX partition and is serviced by the FAX RL (No Digits are stripped)



System ▾ Call Routing Media Resources Voice Mail Devices Application User Management Bulk Administration Help ▾

Route Pattern Configuration

Save Delete Copy Add New

Pattern Definition

Route Pattern*
Route Partition
Description
Numbering Plan
Route Filter
MLPP Precedence*
Resource Priority Namespace Network Domain
Gateway/Route List* [\(Edit\)](#)
Route Option
 Route this pattern
 Block this pattern
Call Classification*
 Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority
 Require Forced Authorization Code
Authorization Level*
 Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation*
Calling Name Presentation*
Calling Party Number Type*
Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*
Connected Name Presentation*

Called Party Transformations

Discard Digits
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type*
Called Party Numbering Plan*



Acronyms

Acronym	Definitions
SIP	Session Initiation Protocol
SCCP	Skinny Client Control Protocol
TDM	Time Division Multiplexing
CUCM	Cisco Unified Communications Manager
CUBE	Cisco Unified Border Element

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Printed in the USA