Application Note

Bell Canada SIP Trunking Service: Connecting Cisco Unified Communications Manager v6 via the Cisco Unified Border Element using SIP

April 7, 2010

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Introduction

Service Providers today, such as Bell Canada, 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. Bell Canada 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) v.6 with a Cisco Unified Border Element (Cisco UBE) for connectivity to Bell Canada's SIP Trunking service. The deployment model covered in this application note is Cisco UCM 6/Cisco UBE to Acme Packets SBC. This document does not address 911 emergency outbound calls. For 911 feature service details contact Bell Canada directly.

- Testing was performed in accordance to Bell Canada SIP Trunking Test Plan and all features were verified. Key features verified are:
  - Basic Call using G.729 or G.711ulaw
  - Calling Party Number Presentation and Restriction
  - Calling Name
  - Intra- and Inter-site Call Transfer
  - Intra- and Inter-site Conference, see caveat section for details.
  - Call Hold and Resume
  - Call Forward All, Busy and No Answer
  - Fax using G.711 pass-through
  - TTY using G.711 pass-through
  - CUBE: performs Delayed-Offer-to-Early-Offer conversion of an initial SIP INVITE without SDP
  - Outbound calls to IP and TDM networks

- 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 Bell Canada's 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 interoperation with Bell Canada's 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/6_0_1/ccmsys/a03ptcss.html

Network Topology

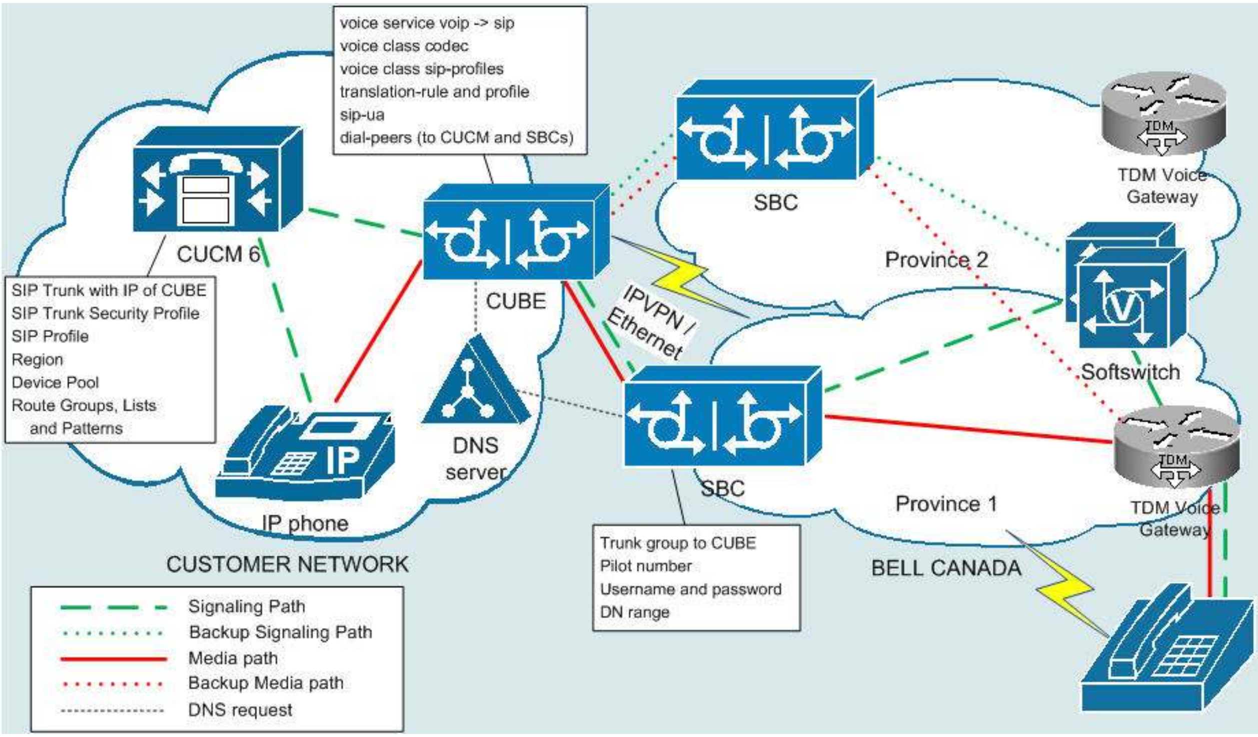


Figure 1. Basic Call Setup

System Components

Hardware Components

- CUCM v6
- CUBE

Software Requirements

Bell service is compatible with CUCM v6.0-v6.1 + CUBE on IOS 12.4.20 and above. The following list summarizes the version and patch levels of the hardware that has been validated as compatible in the Bell lab:

- CUCM v6.1.2.1000-13
- CUCM v6.0.1.2000-3
- CUBE on IOS 12.4(20)T2
- CUBE on IOS 12.4(24)T
- CUBE on IOS 15.0(1)M

Features

Features Supported

The following list summarizes the base feature set for Bell SIP Trunking that has been verified on the tested equipment.

- Basic Call using G.729 or G.711ulaw
- Calling Party Number Presentation and Restriction
- Calling Name
- Intra- and Inter-site Call Transfer
- Intra- and Inter-site Conference, see caveat section for details.
- Call Hold and Resume
- Call Forward All, Busy and No Answer
- Fax using G.711 passthru
- TTY using G.711 passthru
- CUBE: performs Delayed-Offer-to-Early-Offer conversion of an initial SIP INVITE without SDP
- Outbound calls to IP and TDM networks

Features Not Supported

- SIP REFER method
- T.38 Fax protocol
- Modem transmissions

Caveats

- Anonymous Calling

  There is a known issue with CUCM versions 6.X and 7.X that currently requires a workaround to be implemented on the CUBE to support Anonymous calling from a CUCM registered device. The CUBE must be running IOS version 15.0.1 (or later) for the workaround to be successfully applied.

  There is a fix available in the next IOS 15.1T CCO release which addresses this issue, and should be available at the end of January, 2010. With this fix, the only setup needed is to create a new trunk and route-pattern in CUCM, then have the route pattern strip the "anonymity prefix" and send the call out via the new trunk. No setup on CUBE would be necessary. In the interim, the workaround must be applied. Please contact the document authors for further details as required.

- 3-way Conferences using G.729

  When a SIP Trunking customer is using the G.729 compression codec for CUCM endpoints, a Conference Bridge (CFB) resource may be required on the CUBE in order for two CUCM-registered IP phones to initiate a three-way conference between any G.729 end-points. This requirement is not covered in this document.

- G.729 Configuration on customer CUBE

  If a customer requires G.729 support for their CUCM-registered endpoints, they must configure CUBE dial peers that have codec "g729br8" and "voice-class sip g729 annexb-all" enabled. This configuration allows the CUBE to treat the g729br8 codec as a superset of the G.729 codec family.

NOTE: The only valid ptime value for media codecs on the SIP Trunking service is 20 ms. All supported codecs must abide by this restriction. The 'ptime is specified in the SDP part of the SIP signaling, which is used to negotiate the media path for each session.

Configuration

Configuring Cisco Unified Border Element

This section covers the base configuration of the customer CUBE required to set it up as the customers interface for Bell Canada's SIP Trunking service. The CUBE acts as the demarcation point between the customer's CUCM PBX and the session border controllers that provide access to Bell Canada's SIP Trunking Service. Use the following CUBE configuration excerpt as an example of how to configure a suitable SIP interface to the SIP Trunking service.

NOTE: The recommended version for the CISCO IOS load on the customer CUBE is 15.0(1)M or later. Earlier versions might not support some of the configuration entries displayed below. Pay special attention to the bolded entries, and the reference numbers, which map to explanatory footnotes at the end of the configuration.

```
CUBE7206#show version

Cisco IOS Software, 7200 Software (C7200-ADVENTERPRISEK9-M), Version 12.4(24)T, RELEASE SOFTWARE
(fc1)
<SNIP Technical Support: http://www.cisco.com/techsupport ... SNIP>
Compiled Thu 26-Feb-09 00:31 by prod_rel_team

ROM: System Bootstrap, Version 12.2(8r)B, RELEASE SOFTWARE (fc1)
BOOTLDR: 7200 Software (C7200-KBOOT-M), Version 12.2(15)B, EARLY DEPLOYMENT RELEASE SOFTWARE
(fc1)

CUBE7206 uptime is 37 weeks, 1 day, 3 hours, 17 minutes
System returned to ROM by reload at 08:00:17 UTC Mon Jul 20 2009
System image file is "disk2:c7200-adventerprisek9-mz.124-24.T.bin"
Last reload reason: Reload Command

<SNIP This product contains cryptographic features… SNIP>

Cisco 7206VXR (NPE-G1) processor (revision A) with 491520K/32768K bytes of memory.
Processor board ID 31411490
SB-1 CPU at 700MHz, Implementation 1025, Rev 0.2, 512KB L2 Cache
6 slot VXR midplane, Version 2.7

Last reset from power-on

PCI bus mb1 (Slots 1, 3 and 5) has a capacity of 600 bandwidth points.
Current configuration on bus mb1 has a total of 400 bandwidth points.
This configuration is within the PCI bus capacity and is supported.

PCI bus mb2 (Slots 2, 4 and 6) has a capacity of 600 bandwidth points.
Current configuration on bus mb2 has a total of 700 bandwidth points.
The set of PA-2FE, PA-POS-2OC3, and I/O-2FE qualify for "half
bandwidth points" consideration, when full bandwidth point counting
results in oversubscription, under the condition that only one of the
two ports is used. With this adjustment, current configuration on bus
mb2 has a total of 500 bandwidth points.
This configuration is within the PCI bus capacity and is supported
under the above condition.

Please refer to the following document "Cisco 7200 Series Port Adaptor
Hardware Configuration Guidelines" on Cisco.com <http://www.cisco.com>
for c7200 bandwidth points oversubscription and usage guidelines.


4 FastEthernet interfaces
3 Gigabit Ethernet interfaces
1 ATM interface
509K bytes of NVRAM.

62976K bytes of ATA PCMCIA card at slot 2 (Sector size 512 bytes).
16384K bytes of Flash internal SIMM (Sector size 256K).
Configuration register is 0x2102
```

<<CUBE Configuration excerpt BEGIN>>

```
voice service voip
 address-hiding
 allow-connections sip to sip
 fax protocol pass-through g711ulaw
 sip *1
  localhost dns:company.domain.ca *2
  early-offer forced *3
  midcall-signaling passthru
  asserted-id pai
  privacy-policy passthru *4
  g729 annexb-all

!
voice class codec 1 *5
 codec preference 1 g711ulaw
 codec preference 2 g729br8
!
voice class sip-profiles 613
 request INVITE sip-header Diversion modify "@.*>" "@sipt.bell.ca;user=phone>" *6
 request INVITE sip-header Contact modify "@" ";tgrp=rate613;trunk-context=sipt.bell.ca@" *7
 request INVITE sip-header From modify ">" ";user=phone>" *8
 request INVITE sip-header P-Asserted-Identity modify ">" ";user=phone>" *9
!
voice class sip-profiles 416
 request INVITE sip-header From modify ">" ";user=phone>"
 request INVITE sip-header Diversion modify "@.*>" "@sipt.bell.ca;user=phone>"
 request INVITE sip-header P-Asserted-Identity modify ">" ";user=phone>"
 request INVITE sip-header Contact modify "@" ";tgrp=rate_416;trunk-context=sipt.bell.ca@"
!
!
!
!
!
!
```

```
voice translation-rule 1
 rule 1 /^9/ //
!
```

```
voice translation-rule 2
 rule 1 /^\+1/ //
!
voice translation-profile remove9
 translate called 1
!
voice translation-profile remove+1
 translate called 2
!
!
interface GigabitEthernet0/0
 description Public
 ip address 172.17.65.8 255.255.255.128
!
interface GigabitEthernet0/1
 description To CUCM
 ip address 172.17.71.8 255.255.255.0
!
!
!
dial-peer voice 9613 voip
 description outgoing leg to SIP trunk generic digits 613xxxxxxx
 translation-profile outgoing remove9
 destination-pattern 9613.......
 session protocol sipv2
 session target dns:sipt.bell.ca *10
 voice-class codec 1 *11
 voice-class sip profiles 613 *12
 dtmf-relay rtp-nte *13
 ip qos dscp cs5 signaling *14
!
!
!
dial-peer voice 9416 voip
 description outgoing leg to SIP trunk generic digits 416
 translation-profile outgoing remove9
 preference 1
 destination-pattern 9416.......
 session protocol sipv2
```

```
 session target dns:sipt.bell.ca
 voice-class codec 1
 voice-class sip profiles 416 *15
 dtmf-relay rtp-nte
 ip qos dscp cs5 signaling
!
dial-peer voice 94162 voip
 description overflow outgoing leg to SIP trunk for 416 dialing using trunk 613
 translation-profile outgoing remove9
 preference 2
 destination-pattern 9416.......
 session protocol sipv2
 session target dns:sipt.bell.ca
 voice-class codec 1
 voice-class sip profiles 613 *16
 dtmf-relay rtp-nte
 ip qos dscp cs5 signaling
!
dial-peer voice 9 voip *17
 description outgoing leg to SIP trunk for default trunk
 use 613 rate center for call
 translation-profile outgoing remove9
 destination-pattern 9T
 session protocol sipv2
 session target dns:sipt.bell.ca
 voice-class codec 1
 voice-class sip profiles 613
 dtmf-relay rtp-nte
 ip qos dscp cs5 signaling
!
```

```
dial-peer voice 416760126 voip
 description outgoing leg to CUCM7 for 416760126.
 destination-pattern 416760126.
 session protocol sipv2
 session target ipv4:172.17.71.4 *18
 voice-class codec 1
 dtmf-relay rtp-nte
!
```

```
dial-peer voice 100 voip *19
 description incoming leg from SIP trunk E.164 compliant
 translation-profile incoming remove+1
 answer-address +1T
 voice-class codec 1
 dtmf-relay rtp-nte
!
sip-ua
 authentication username 4167601260 password 7 075E731F1A5C4F realm sipt.bell.ca *20
 retry invite 2
 retry response 3
 retry bye 3
 retry prack 6
 timers expires 60000
 no transport tcp *21
!
end
```

<<CUBE Configuration excerpt END>>

Configuration Annotations:

*1 These settings are global SIP settings. Depending on your specific requirements, these parameters may also be provisioned specifically in dial-peer entries for more refined control granularity.

*2 The company domain agreed upon during service setup with Bell Canada. This parameter is used in the From: and P-Asserted-Identity: headers as the host part of the address URI.

*3 "early-offer-forced" is used to force an early offer of media codec support (via SDP message body) in outgoing INVITE messages. Bell Canada SIP Trunking requires this early offer.

*4 This privacy-policy setting (passthru) is used to indicate that the session target is trusted and it will maintain privacy requests. This setting is new in IOS version 15 for the CUBE.

*5 Voice codec class setup, this example shows that both G711ulaw and g729 are defined for this class. Use of this voice class will supply both codecs in the offer during media negotiation.

*6 Modification of Diversion header to allow for Call Forwarding from CUCMv6. The modification simply replaces the host portion of the URI with the Bell Canada SIP Trunking service domain. It also adds the required (mandatory) "user=phone" parameter.

*7 The contact header is modified to supply a tgrp parameter (SIP RFC 4904) to select a specific trunk for outbound calls.

*8 A further addition of mandatory user=phone parameter to the From header.

*9 A further addition of mandatory user=phone parameter to the P-Asserted Identity header.

*10 This session target DNS entry must resolve to the SIP Trunking service interface address provided by Bell Canada. The domain name is equal to the SIP Trunking domain name.

*11 Specifies the use of the codec class defined in “codec class 1” (See *5)

*12 Specifies the sip profile to use. ‘The specified profile provides trunk selection via trgp specification.

*13 Specifies DTMF signaling type for SIP Trunking service.

*14 CS5 is the QOS tagging type supported by Bell Canada's SIP Trunking service.

*15 Example of how to use a rate-centre-specific trunk (416 in this example) for calls destined to called parties in the 416 area code.

*16 This dial-peer is an ‘overflow’ dial peer used in case the capacity of a trunk (the 416 trunk in this case) is exceeded. In this example, the configuration specifies that the ‘overflow’ call will be attempted again, but using the trunk for rate centre 613. Note that the call destination is 416, but the originating trunk is for the 613 rate centre, so long distance charges will apply to any overflow calls.

*17 This Dial peer is used as a default outgoing dial peer. If a called destination does not match a more specific dial peer, the call will use this “default” dial peer.

*18 Specifies the address of the customer’s CUCMv6 PBX. This session target is the target to be used for inbound calls, which obviously need to terminate on the customer’s CUCM.

*19 This Dial peer is used as a default incoming dial peer from the customer’s SIP Trunking service. The E.164 prefix “+1” will be stripped before sending the call to the CUCMv6.

*20 Specifies the DIGEST credentials to be used for authentication challenges to outbound SIP requests. These are provided by Bell Canada.

*21 Bell Canada SIP Trunking only supports UDP transport.

Configuring the Cisco Unified Communications Manager

This section covers the base configuration of the CUCM to communicate with a Cisco Unified Border Element (CUBE). The CUBE acts as the customer interface to Bell Canada's SIP Trunking Service.

The customer administrator will use the CUCM Administration Web interface to implement the following CUCM configuration examples:

Service Parameter Settings

No changes are required to the default settings:

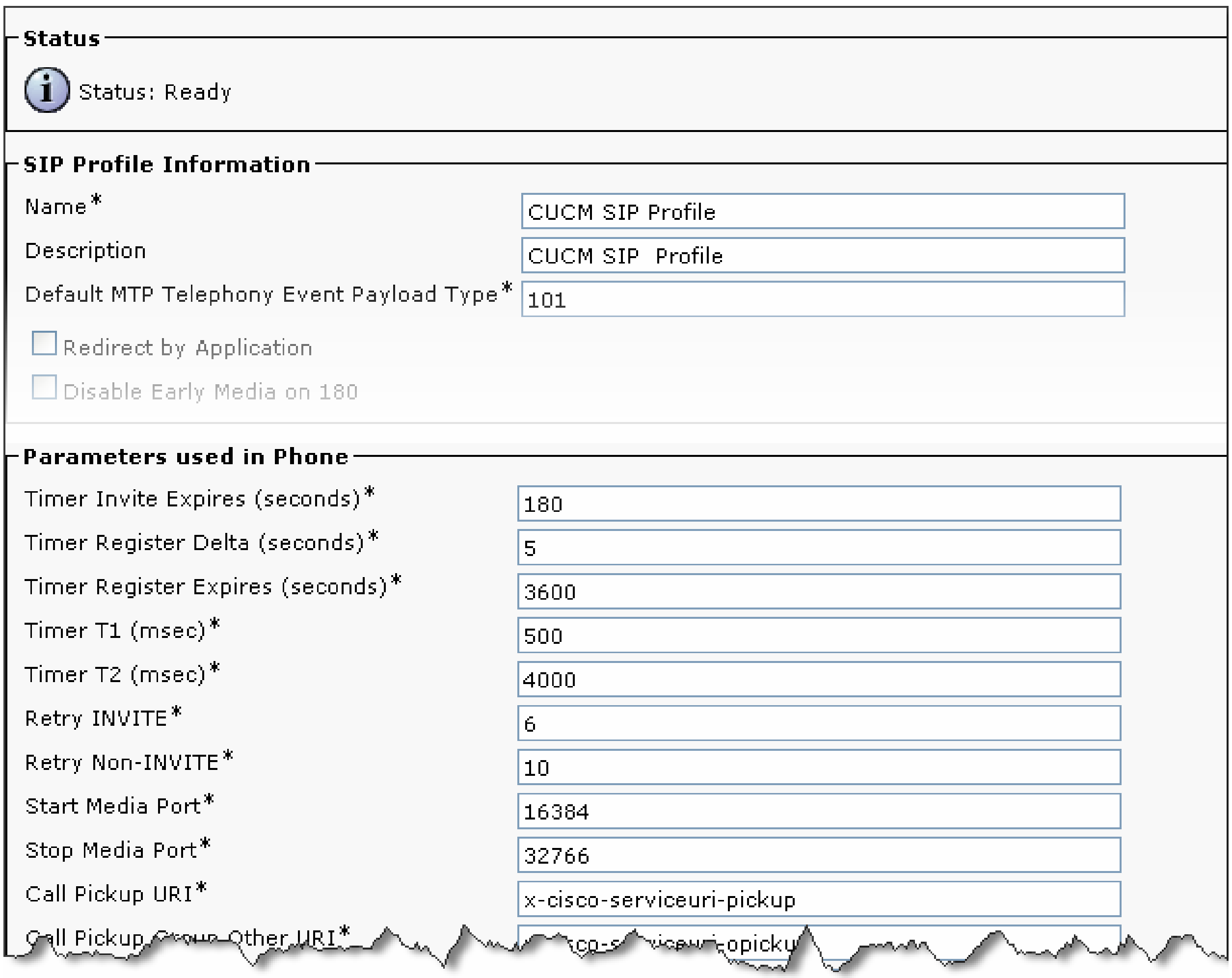
Clusterwide Parameters (Device - SIP)

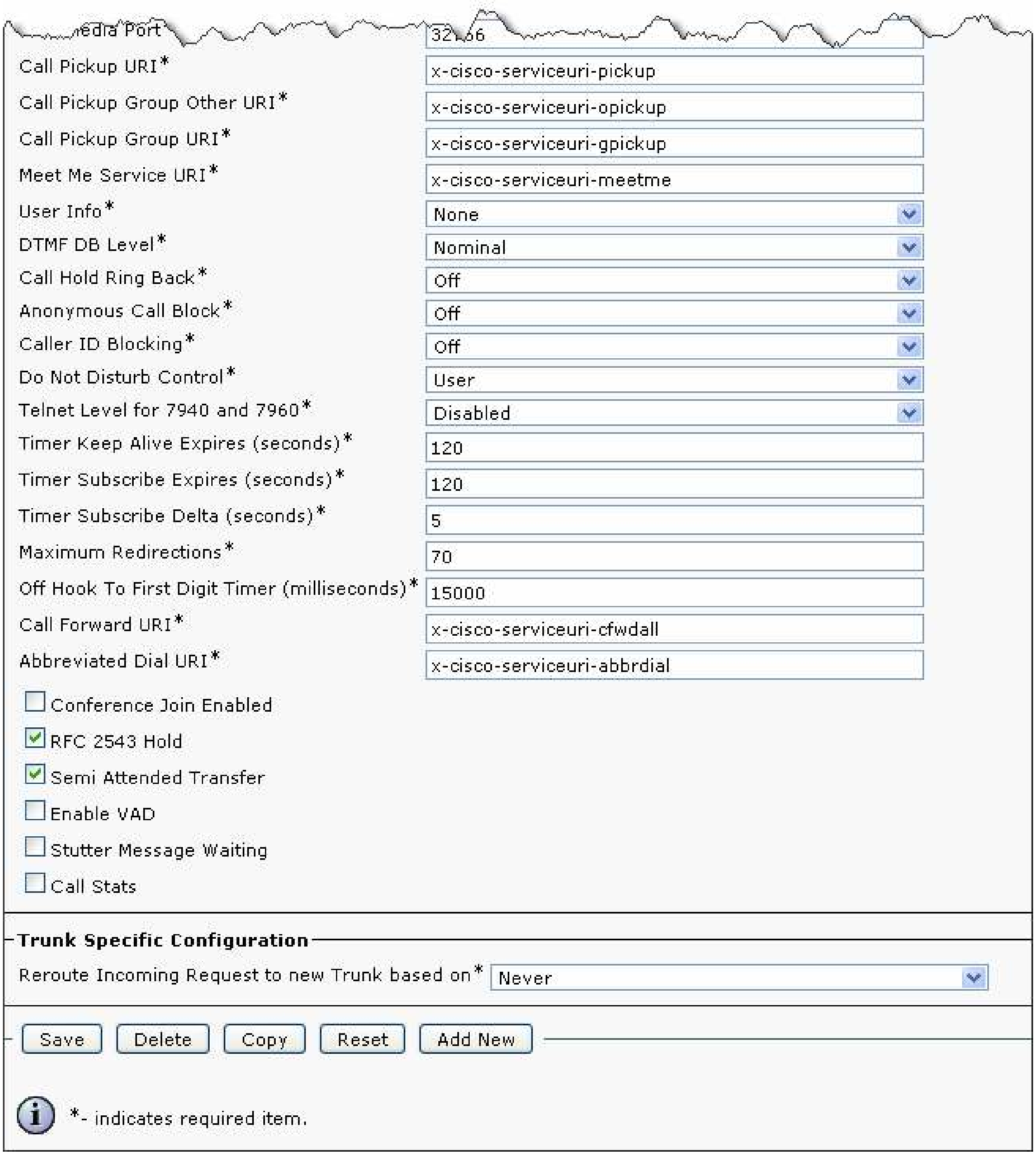
Retry Count for SIP Bye *	10	10
Retry Count for SIP Cancel *	10	10
Retry Count for SIP Invite *	6	6
Retry Count for SIP PRACK *	6	6
Retry Count for SIP Rel1XX *	10	10
Retry Count for SIP Publish *	6	6
Retry Count for SIP Response *	6	6
SIP Connect Timer *	500	500
SIP Disconnect Timer *	500	500
SIP Expires Timer *	180000	180000
SIP PRACK Timer *	500	500
SIP Rel1XX Timer *	500	500
SIP Trying Timer *	500	500
SIP Publish Timer *	500	500
SIP Rel1XX Enabled *	False	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
SIP Max Incoming Message Size *	5000	5000
SIP Max Incoming Message Headers *	100	100
Send SIP Multicast TTL in SDP *	False	False
Default PUBLISH Expiration Timer *	3600	3600
Minimum PUBLISH Expiration Timer *	60	60
CUP PUBLISH Trunk	< None >	
Multicast MOH Direction Attribute for SIP *	RecvOnly	RecvOnly

Configure a SIP Profile

Navigate to Device → Device Settings → SIP Profile

- Press the Find button to display existing SIP Profile entries.
- Copy the existing Default SIP Profile using the Copy icon on the right hand side (See NOTE below).
- Name the New SIP Profile “CUCM SIP Profile” or something similar.
- There are no changes made to the settings of the CUCM Default SIP Profile for the CUCM SIP Profile.
- Save the configuration
- NOTE: Cisco strongly suggests that each configuration profile component be created as a copy of the default version, rather than using the default version as is, or altering the default profile and re-saving it. The default profile provides a clean base reference point for future profiles, or to revert to in the event that a configuration becomes corrupted.

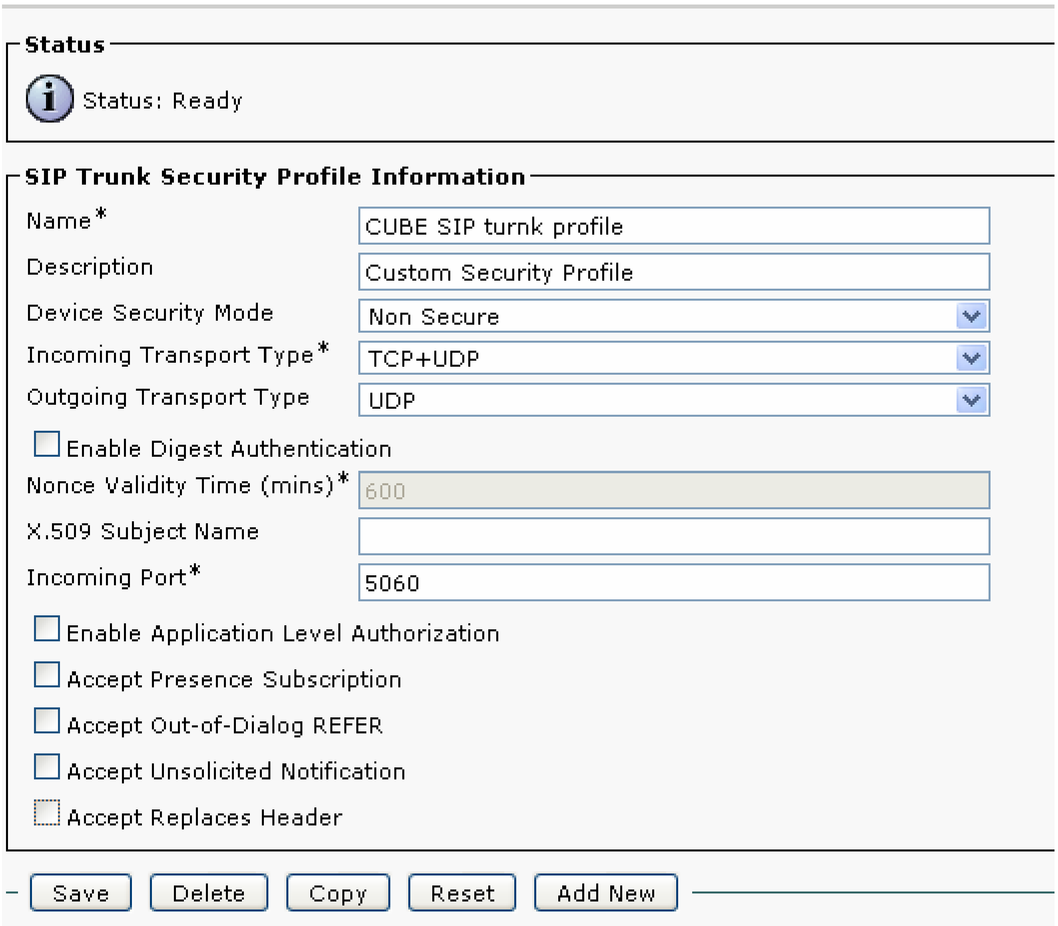



Call Pickup URI* x-cisco-serviceuri-pickup
Call Pickup Group Other URI* x-cisco-serviceuri-opickup
Call Pickup Group URI* x-cisco-serviceuri-gpickup
Meet Me Service URI* x-cisco-serviceuri-meetme
User Info* None
DTMF DB Level* Nominal
Call Hold Ring Back* Off
Anonymous Call Block* Off
Caller ID Blocking* Off
Do Not Disturb Control* User
Telnet Level for 7940 and 7960* Disabled
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
Abbreviated Dial URI* x-cisco-serviceuri-abbrdial
Conference Join Enabled
RFC 2543 Hold
Semi Attended Transfer
Enable VAD
Stutter Message Waiting
Call Stats
Trunk Specific Configuration
Reroute Incoming Request to new Trunk based on* Never
Save
Delete
Copy
Reset
Add New
*- indicates required item.

Configure a SIP Trunk Security Profile

Navigate to the System → Security Profile → SIP Trunk Security Profile page in the portal (see screen capture below):

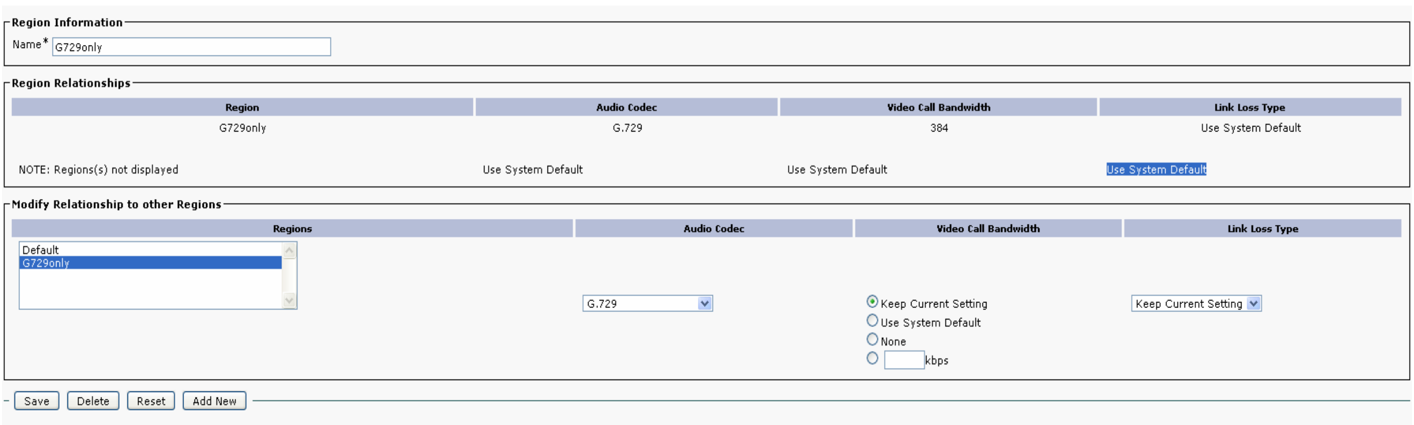
- Copy the existing Non-Secure SIP Trunk Profile.
- Name the New SIP Trunk Profile – CUBE SIP Trunk Profile or something similar.
- Make sure none of the checkboxes are selected.
- Save the configuration.



Configure a Region

Navigate to the System → Region page in the CUCM portal.

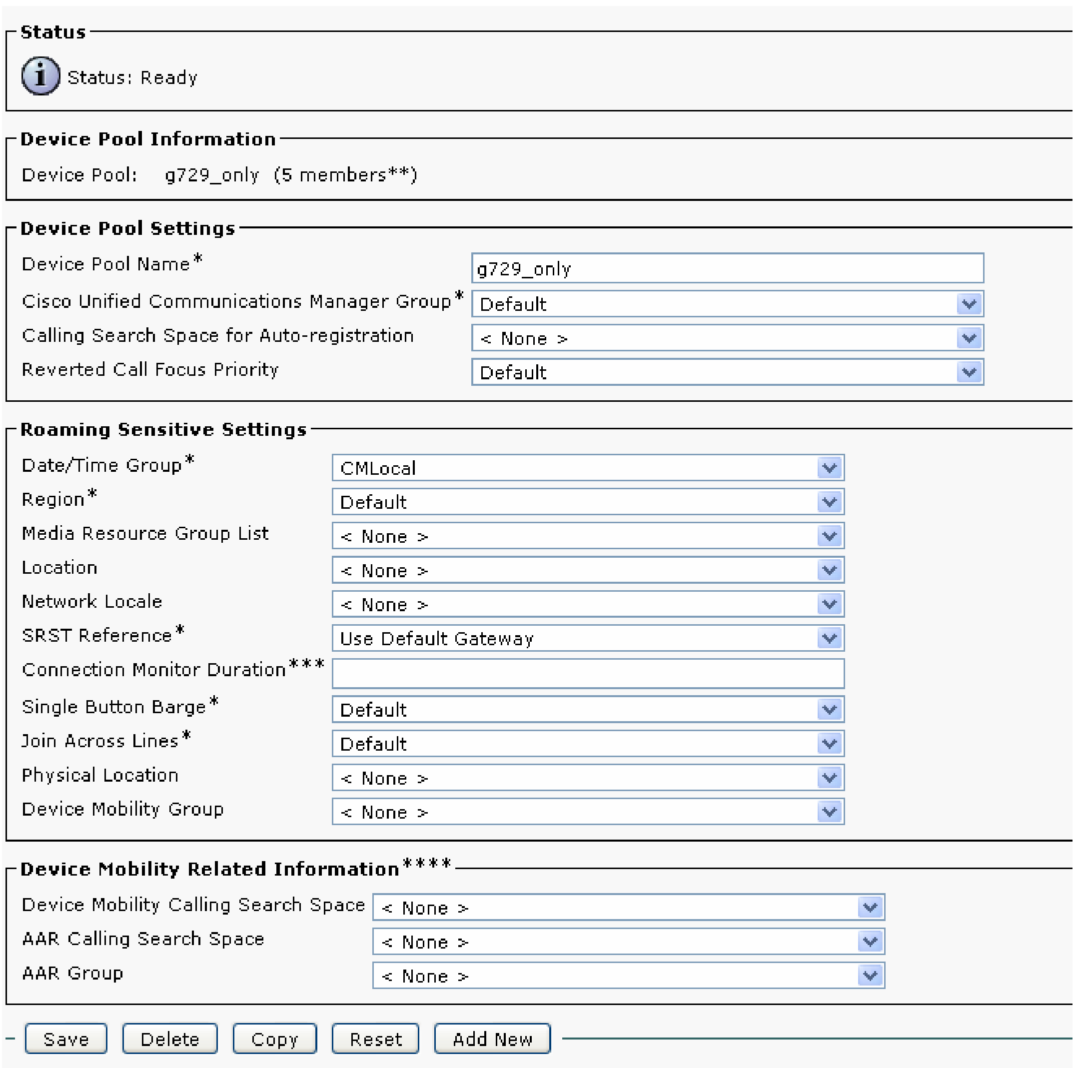
- Click on the “Add New” button
- Name the new region G-729 or something similar
- Click on the newly created region and choose an Audio Codec in the centre pane. (the example shown in the capture below, uses G.729).
- Select other settings to be consistent with the customer’s environment.
- Save the configuration



Configure a Device pool:

Navigate to the System → Device Pool page in the CUCM portal.

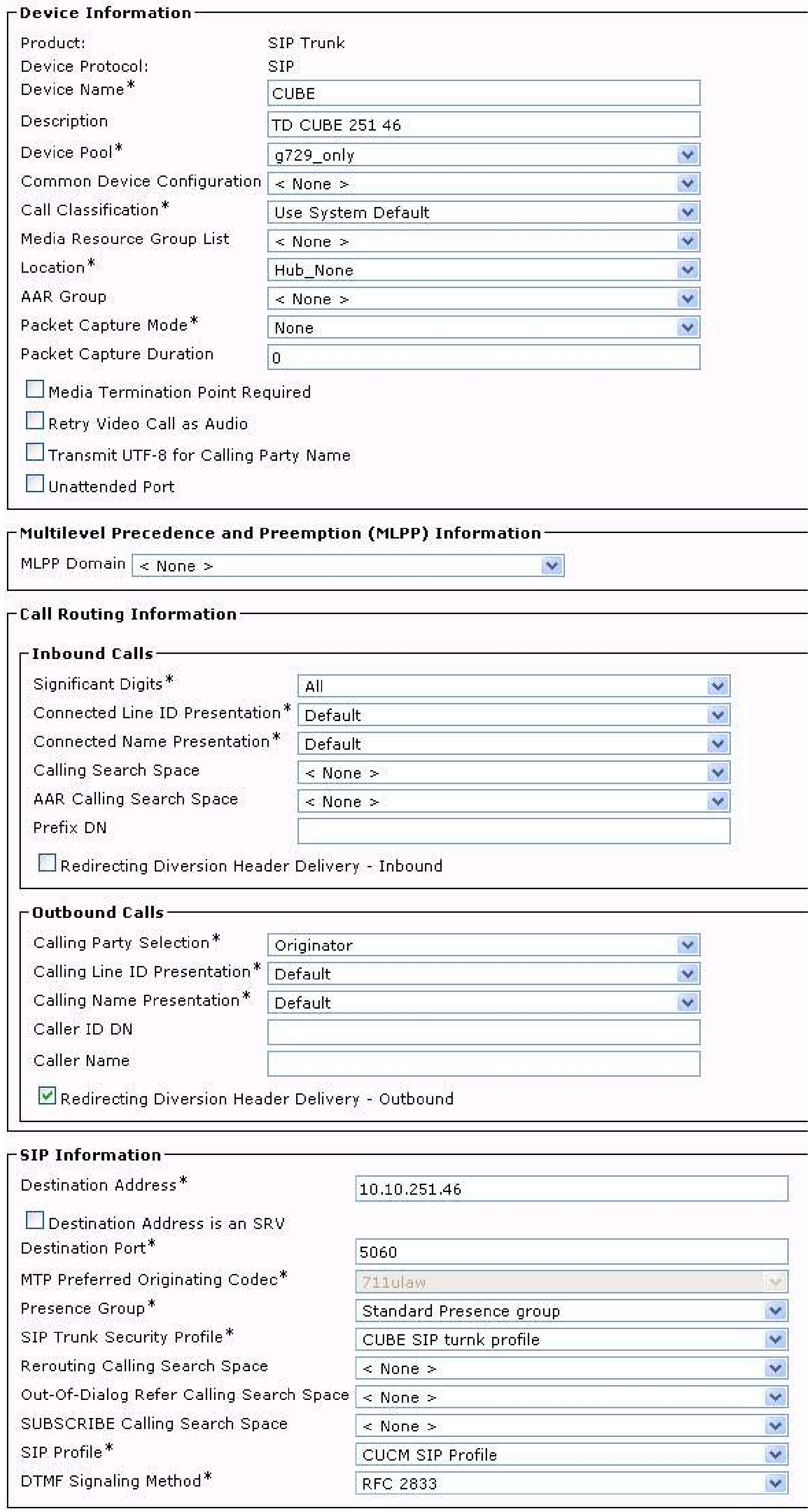
- Click on “Add New”
- Name the new Device Pool “DevicePool-G-729” or something similar.
- Select desired Region created in the previous step)
- Fill in required fields with appropriate configurations for your customer environment.
- Save the configuration



Configure a SIP Trunk

Navigate to the Device → Trunk page in the CUCM portal.

- Click on "Add New"
- Select "SIP Trunk" as the trunk type.
- The Device Protocol will automatically be populated with "SIP", just click Next
- For a G.711-only trunk, create a G.711 SIP Trunk, name it SIP-Trunk-G.711 or equivalent.
- For a G.729-only trunk, create a G.729 SIP Trunk, name it SIP-Trunk-G.729 or equivalent.
- Note: For a mixed-codec trunk, you will have to configure two SIP Trunks. This is not covered in this document.
- Once your trunk has been created and named, select the Device Pool that you previously configured in a previous step.
- The following steps are the same, regardless of the codec (G.711, G.729) you selected earlier
- Check the check box "Redirecting Diversion Header Delivery – Outbound".
- The Destination IP Address should be set to your CUBE IP address.
- Chose the SIP Trunking Security Profile you created and configured earlier.
- Choose the SIP Profile previously you created and configured earlier.
- Save the configuration.



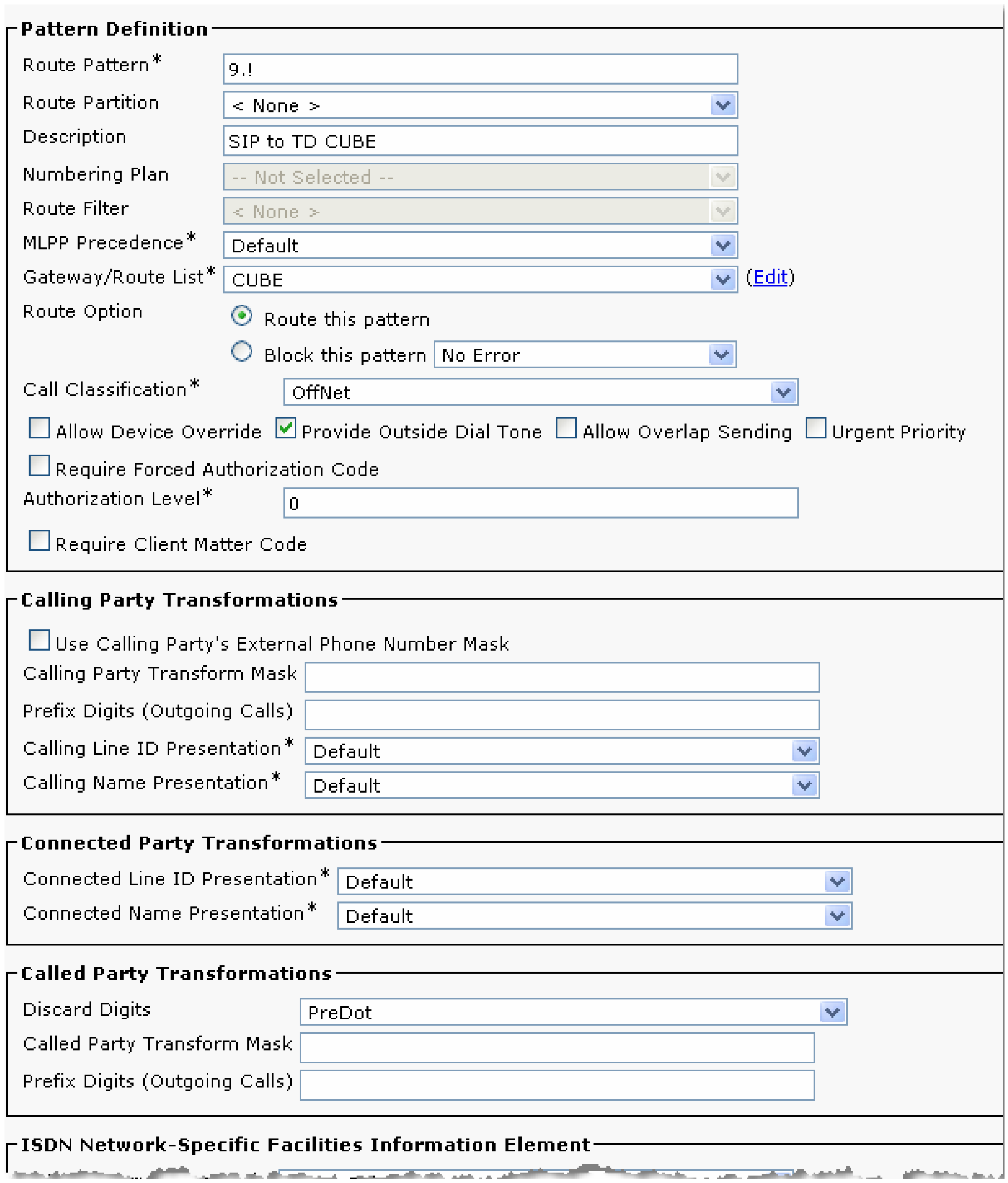
Configure Route groups, Route Lists, and Route Patterns

Due to the complexity and the wide variance in the individual needs of each enterprise, configuring enterprise route patterns requires detailed planning and consideration. For this reason, route patterns are not covered in any detail in this reference document. Please refer to the appropriate Cisco documentation for guidance in configuring Route Patterns suitable to your particular situation.

For the sake of a simple example, a simple Route Pattern that routes calls to a CUBE is defined here:

Navigate to Call routing → Route/Hunt → Route Pattern

- Click on “Add New”
- Insert route-pattern expression
- Choose a route partition
- Give it a description describing the intention of this route pattern.
- Choose the Gateway/route list as per provisioned above (SIP Trunk 729 to CUBE)
- Define any other trunk specific configurations.
- Save the configuration.



Acronyms

CUBE	Cisco Unified Border Element
CUCM	Cisco Unified Communications Manager
DNS	Domain Name Resolution
G.711	Voice Codec (Uncompressed)
G.729	Voice Codec (Compressed)
MGCP	Media Gateway Control Protocol
OTG	Originating Trunk Group
PAI	P-Asserted Identity
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
RFC	Request For Comment
RTP	Real Time Protocol
SCCP	Skinny Client Control Protocol
T.38	Fax over IP protocol
TGRP	SIP Trunk Group selection convention
URI	Uniform Resource Indicator

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