VIDEOTRON
Business

SIP Trunking Service

Configuration Guide

Cisco Unified Communications Manager PBX

Ver. 10.5

Confidentiality and copyright statement

This document was written using gender-neutral language.

The information contained herein is subject to change without prior notice.

Modification history

Edit	Date	Author	Description
1.0	2019-08-19	Pascal Beauregard	Original draft
1.1	2019-09-12	Martin Lefrançois	Review of the consistency with other manuals
1.2	2019-09-18	A. Marchard	Linguistic revision
1.3	2019-09-30	Martin Lefrançois	Validation

Table of Contents

1 Audience

The *SIP Trunking Service Configuration Guide* is intended for service users, technical managers and authorized integrators.

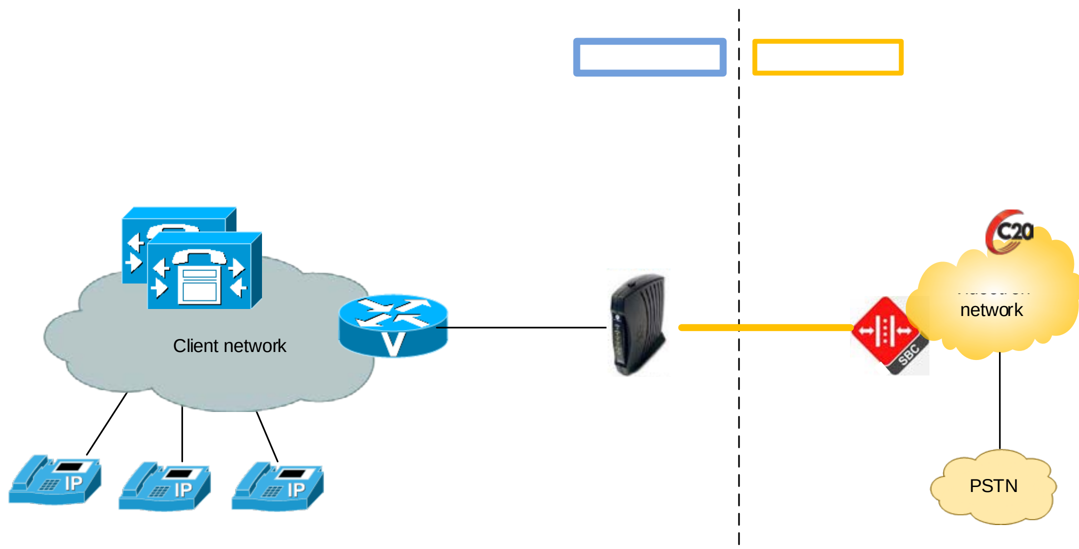
2 Introduction

The *SIP Trunking Service Configuration Guide* details the basic steps for setting up a single SIP trunk between Videotron's SBC and a Cisco Unified Border Element (CUBE) placed in front of an IP Cisco Unified Communications Manager (CUCM) PBX. Several SIP trunks may be set up, but this document does not go over the steps for doing so.

That said, this guide is not intended to help you configure PBX user/application features.

3 Network and equipment diagram

The diagram below is an overhead view of SIP trunking with a Cisco Unified Communications Manager (CUCM) PBX behind a Cisco Unified Border Element (CUBE).



The solution includes:

Customer site:

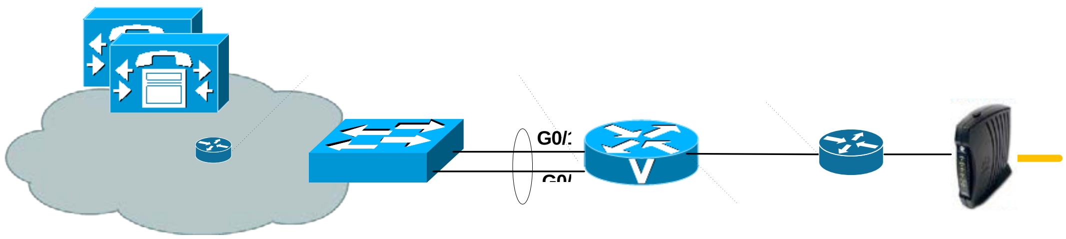
- Cisco Unified Communications Manager (CUCM) servers, version 10.5.
- CUBE: Cisco 29xx router (2901, 2921, 2951), IOS version 15.5(3)M
- IP Cisco telephones (7965, 7821, 7841)

Videotron site:

- Videotron SBC: Oracle (Acme Packet)
- Videotron Softswitch: Genband C20
- PSTN connection

3.1 Physical connection between the CUBE and the customer's Internet access

The CUBE must be linked by a 10/100Mbps network connection toward the customer's Internet. Usually, the customer has a router behind the Videotron cable modem that provides a connection to the Internet.



4 Features

4.1 Supported features

The SIP trunking service supports the following features:

Feature	Description	Limit(s)
Simultaneous calls	The simultaneous calls limit is established when the SIP trunk order is placed.	
Voice	G.711 µ-law standard used exclusively	
Fax	G.711 µ-law standard used	T.38 standard not supported
Other kinds of data (modem, alarm, etc.)	G.711 µ-law standard used	
Inbound Caller ID name and number	Inbound Caller ID name and number transmitted from the Videotron site to the PBX.	
Outbound Caller ID name	Outbound Caller ID name, as transmitted via PBX to the public network.	
Outbound Caller ID number	Outbound Caller ID number, as transmitted via PBX to the public network.	
DID display for 911 emergency call centre	DID display for 911 emergency call centre transmitted via PBX if on the predefined list of numbers.	
Direct trunk overflow	Calls are routed to another SIP trunk when the number of simultaneous calls SIP trunking can handle is exceeded.	The other SIP trunk must be on the same Videotron telephone switch as the primary SIP trunk.
Failover to another phone number	Calls are routed to another phone number when the number of simultaneous calls that the SIP trunk can handle is exceeded.	The “Redirect information” or “Original called number” field is not transmitted. The “Called number” is the actual forwarding number and not the DID.

		An overflow to another phone number requires an additional service called a “Permanent Redirect Line (PRL).” This service is billed according to the predefined number of simultaneous PRL calls. If the phone number is long distance, charges will apply. .
Failover to another SIP trunk	Calls are routed to another SIP trunk in the following three cases of failure: 1. The customer’s PBX no longer responds to calls sent to it on the SIP trunk. 2. The customer's PBX responds with the message “SIP 503 Service Unavailable.” 3. The SIP trunk is faulty.	If the PBX responds with a SIP message other than “503 Service Unavailable,” there will be no call failover.
Failover to another phone number	Calls are routed to another phone number in the same three cases as above.	If the PBX responds with a SIP message other than “503 Service Unavailable,” there will be no call failover. Same limitation as “Direct trunk overflow” with respect to the fields and the need for a Permanent Redirect Line.
“Redirect number” field (Remote Party ID)		The Videotron telephone switch transmits the original called number to the Remote-Party-ID header.
Class of restriction call blocking	No blocking for local calls, in Quebec, Canada, the United States and abroad, and for 411, 0-, 0+, 00 and 900 numbers.	1-976 calls are blocked.
Number portability	Videotron handles the transfer of a customer’s telephone number from their current service to the SIP trunking service.	The customer must provide all required documentation.
SIP-Refer	Allows you to free up lines after a call is forwarded from an external number to another external number, such as a cellphone.	If the external number is long distance in relation to the original dialled number, the call may be dropped rather than forwarded. Especially when the call is transferred through another Videotron switch. Routing between Videotron switches is subject to change without prior notice.

4.2 Unsupported or limited features

Our SIP trunking does not support the following features:

Feature	Description
Numbers outside our coverage area	Only telephone numbers in Videotron service areas will be accepted.
Fixed 911	This feature allows calls to be forwarded directly to the 911 emergency call centre in the municipality where the caller is located. Instead, the SIP trunking service uses an intermediary (“nomad”) 911 emergency call centre to forward calls. See videotron.com/ip-911 for details.
Emergency call forwarding	Allows you to forward calls to different destinations based on a predefined phone tree for emergency scenarios. This is an advanced feature reserved for the dedicated fibre optic SIP trunking service.
Authorization and billing codes	The authorization code is used to limit access to long-distance calls. The billing code is used to count calls per user for internal billing and customer billing purposes. These are advanced features reserved for the dedicated fibre optic SIP trunking service.
Equity of access	Allows you to use another long distance provider. This feature is largely irrelevant considering that Videotron offers unlimited calling plans for Canada and the United States. This feature is reserved for the dedicated local fibre optic SIP trunking service.
Occasional calls	Used to dial the 101-XXXX code in order to temporarily change long distance provider. This feature is largely irrelevant considering that Videotron offers unlimited calling plans for Canada and the United States. This feature is reserved for the dedicated local fibre optic SIP trunking service.
Signalling and voice channel encryption	Videotron does not currently support signalling encryption (SIP TLS) and voice channel encryption (SRTP). Encrypted MD5 hash password.

5 Service requirements

5.1 Registering a SIP trunk

Once the SIP trunk has been configured at the Videotron site, our technical team will send the following information to the customer:

- domain name
- username
- password

The customer PBX (in this case the customer's CUBE) must be registered with Videotron in order to connect calls via SIP trunking. The customer, or more commonly the integrator-interconnector, must configure the CUBE such as to register the SIP trunk with Videotron's switch. The Videotron team will set up a phone conference with the interconnector to complete the registration and ensure the SIP trunk is functioning properly.

The CUBE is registered by sending SIP REGISTER messages to Videotron's SBC IP address that contains a username, password and domain name.

5.2 Responding to SIP INFO messages

Videotron's telephone switch periodically sends SIP INFO messages to the customer's CUBE. If these messages do not reach the CUBE (i.e., they are blocked by the customer's firewall), or they are not answered by the CUBE, the switch will consider the CUBE out of order.

5.3 Sending the domain name in the Req URI header of SIP INVITE messages

The CUBE must be capable of sending a domain name in the Req URI of SIP INVITE messages. If the domain name is missing, any calls will be rejected.

5.4 Configuration settings overview

Table 4 provides an overview of the parameters required to set up the SIP trunking service.

Domain name	**Provided by Videotron:** <customer acronym>.sipott.v50.videotron.com
Videotron SBC address	**24.200.242.87**
SIP communication port	**UDP 5060**
Username	**Provided by Videotron: s**<last 9 numbers of primary telephone number>
Password	**Provided by Videotron:** 12 alphanumeric characters with at least 1 lowercase letter, 1 uppercase letter, and 1 number
Number of simultaneous calls on the SIP trunk	**Provided by Videotron**
Codec	**G.711 µ-law only**
Fax protocol	**In-Band (T.38 not supported)**
DTMF	**RFC2833**
SIP REFER	**The SIP REFER function must only be activated after discussion with the Videotron team. If the external number is long distance in relation to the original dialled number, the call may be dropped rather than forwarded.**

Table 1: Configuration settings overview

6 Configuration

Putting a SIP trunk in place on a CUCM phone service with CUBE requires the configuration of the CUCM and the CUBE. These two systems are highly versatile and consequently have several parameters that could affect communication on the SIP trunk. This guide provides a configuration example that we have tested and that is fully functional.

6.1 Configuring the CUBE (Cisco router 29xx)

With this proposed configuration, it is possible to test calls that will use the CUBE. The integrator will modify this configuration to meet the specific and comprehensive needs of the customer.

Step 1: Configuring the physical interfaces

Configuration that reflects the example in Section 3.1. **(The configuration must reflect the customer's network.)**

```
interface Port-channel1
  ip address 10.4.8.2 255.255.255.248

interface GigabitEthernet0/0
  description xxxxxxxx port G1/0/2
  no ip address
  duplex auto
  speed auto
 channel-group 1

interface GigabitEthernet0/1
  description xxxxxxxxxxx port G1/0/1
  no ip address
  duplex auto
  speed auto
  channel-group 1

interface GigabitEthernet0/2
  description Toward Videotron's SBC
  ip address 10.4.8.21 255.255.255.252
  duplex auto
  speed auto
```

Step 2: IP host section

This section demonstrates how to associate Videotron's SBC IP address to the domain dame that will be used for Videotron's SIP Trunk service. If the CUBE has access to a DNS server, this line is not required.

ip host **hofa01.sipott.v50.videotron.com** 24.200.242.87

Note: replace the domain name in the example with the domain name that Videotron has assigned to you.

Step 3: Voice service VoIP section

All the commands in this section must be configured.

The commands in this section define the SIP communication that enters and exits the CUBE.

```
voice service voip
  ip address trusted list
```

```
  ##List of the IP addresses that can speak SIP with the CUBE – at least have the CUCMs' and SBC's addresses.
  ipv4 24.200.242.87

## Le chiffre 100 doit être remplacé par le nombre de licences CUBE achetées
 mode border-element license capacity 100
 allow-connections sip to sip
fax protocol pass-trough g711ulaw ## Parameters for communications via fax

  sip
   registrar server expires max 3600 min 3600 ## SIP registration parameters
   no update-callerid                         ##To be copied as it is
   early-offer forced                         ##Force the SDP in the Invite SIP
   no call service stop                       ## Activate the SIP service on the router
```

Step 3: Sip-ua section

This section demonstrates how to configure the parameters for the registration to Videotron's SIP trunk service.

For this section you will need the following information:

- Username
- Password
- Domain name

Videotron's technical team will give you this information when it has programed the service on its end. A phone appointment is scheduled with Videotron's technical team and the customer/integrator.

Below is an example of programming with the following dummy parameters:

- Username: s383870001
- Password: u12Se3Rf2n53
- Domain name: hofa01.sipott.v50.videotron.com

```
sip-ua
 credentials username s383870001 password u12Se3Rf2n53 realm realm
 authentication username s383870001 password u12Se3Rf2n53
 retry invite 2
 timers keepalive active 10
 registrar 1 dns: hofa01.sipott.v50.videotron.com expires 3600
 connection-reuse
```

Step 4: Voice class sip-profiles section

Videotron would like the host part of the SIP URI in the INVITE request sent by the IP PBX to be a label that resembles a domain name rather than an IP address. Example:

Original Req URI in the SIP INVITE request sent by the CUBE toward Videotron's SBC prior to the transformation:

Req URI : : <sip:5141234567@24.200.242.87:5060>

Req URI after the transformation in the SIP INVITE request sent by the CUBE to Videotron's SBC:

Req URI : : <sip:5141234567@hofa01.sipott.v50.videotron.com:5060>

We require a voice class to replace 24.200.247.87 with "**hofa01.sipott.v50.videotron.com**" in the Req URI sent by the CUBE to Videotron.

```
voice class sip-profiles 1
     request INVITE sip-header SIP-Req-URI modify "24.200.247.87:5060"
     "hofa01.sipott.v50.videotron.com:5060"
```

To apply the voice-class, you must insert it in the outbound dial-peer to Videotron with the ***voice-class sip profiles 1*** command (see dial-peer voice 105 VoIP further in this document).

Step 5: Voice translation section (optional)

This section only applies if a 9 (for example) has been prefixed to the number dialed for outbound calls. The 9 must be removed before the called number is sent to the PSTN. The translation profile "ToPSTN" is called by the dial-peer voice 105.

```
## Is called by the voice translation-profile ToPSTN
voice translation-rule 2
 rule 1 /^9\(911\)/ /\1/
 rule 2 /^9\([2-8]11\)/ /\1/
 rule 3 /^9\([2-9]..[2-9]......\)/ /\1/
 rule 4 /^9\(1[2-9]..[2-9]......\)/ /\1/
 rule 5 /^9\(0[2-9]..[2-9]......\)/ /\1/
 rule 6 /^9\(011.*\)/ /\1/

## Called by the dial-peer voice 105 VoIP
voice translation-profile ToPSTN
  translate called 2 ## Calls voice translation-rule 2 and acts on the called number
```

Step 6: Voice class URI section

Allows you to form the list of IP addresses for which you wish to establish a match in the incoming dial-peer from Videotron's SBC (see dial-peer voice 10 VoIP).

```
voice class uri 1000 sip
 host 24.200.242.87
```

Step 7: Dial-peer section

Configuring the dial-peers allows you to route the calls when they transit via the CUBE.

The dial-peers presented in this section are only examples. The parameters in bold in the dial-peers are basic parameters to enter in all the dial-peers you configure.

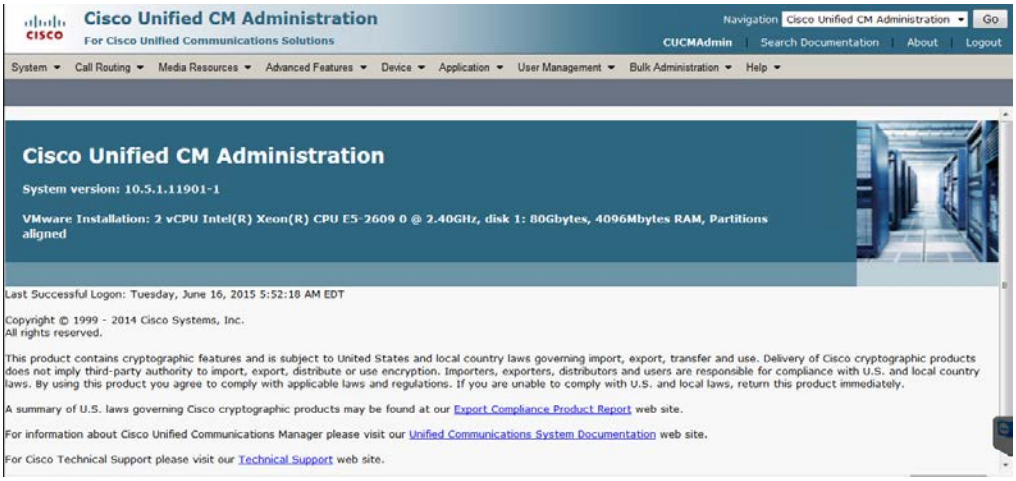
```
## Inbound dial-peer for calls from the CUCM
dial-peer voice 1 voip
  description Incoming call-leg - Calls from the CUCM
 session protocol sipv2          ## Force version 2 of SIP
 session transport udp           ##Force the SIP signaling to be used with the UDP
  incoming called-number 9T                ##To match the dial peer on 9 as first of the called
        ## The "voice-class sip bind" command associates the dial-peer to the control SIP messages and the media that
        transit on the po1 (customer network therefore SIP messages of the CUCM) these 2 commands are very important
        because SIP messages transit via the port G0/2 (to SBC) and po1 (to CUCM).
  voice-class sip bind control source-interface Port-channel1
  voice-class sip bind media source-interface Port-channel1
  dtmf-relay rtp-nte     ## Force RFC2833 for the transmission of DTMF
```

```
  codec g711ulaw        ## Force G711 voice without compression
  ip qos dscp cs3 signaling
  no vad                    ##Prevents the use of Voice Activity Detection.

## Inbound dial-peer for calls from Videotron's SBC
dial-peer voice 10 voip
  description Incoming call-leg - Inbound PSTN calls
  session protocol sipv2
  ## enables match on SIP requests from addresses that are in the voice class uri 1000 sip
  incoming uri via 1000
  voice-class sip bind control source-interface GigabitEthernet0/2
  voice-class sip bind media source-interface GigabitEthernet0/2
  dtmf-relay rtp-nte
  codec g711ulaw
  no vad

## Outbound dial-peer for local calls toward Videotron's SBC
dial-peer voice 105 voip
 description Local calls 10 digits toward the PSTN
## Command that strips the 9 before transmission to the SBC – must also configure an associated translation rule not showed in this document.
 translation-profile outgoing ToPSTN  ## Call the translation profile ToPSTN that removes the 9 as prefix (optional)
 destination-pattern 9[2-9]..[2-9]......
 session protocol sipv2
 session target ipv4: 24.200.242.87     ##Videotron's SBC at the address 24.200.242.87 is the target
 voice-class sip bind control source-interface GigabitEthernet0/2
 voice-class sip bind media source-interface GigabitEthernet0/2
 voice-class sip profiles 1     ## Call voice class sip-profiles 1 that inserts the domain in Req URI
 dtmf-relay rtp-nte
 codec g711ulaw

## Outbound dial-peer for local calls toward the CUCM
dial-peer voice 1046511 voip
  description Calls toward CUCM
  destination-pattern [2-9]..[2-9]......
  session protocol sipv2
  session target ipv4:10.4.65.11  ## The CUCM is the target
  voice-class sip bind control source-interface Port-channel1
  voice-class sip bind media source-interface Port-channel1
  dtmf-relay rtp-nte
  codec g711ulaw
  ip qos dscp cs3 signaling
  no vad
```

6.2 Configuring the CUCM

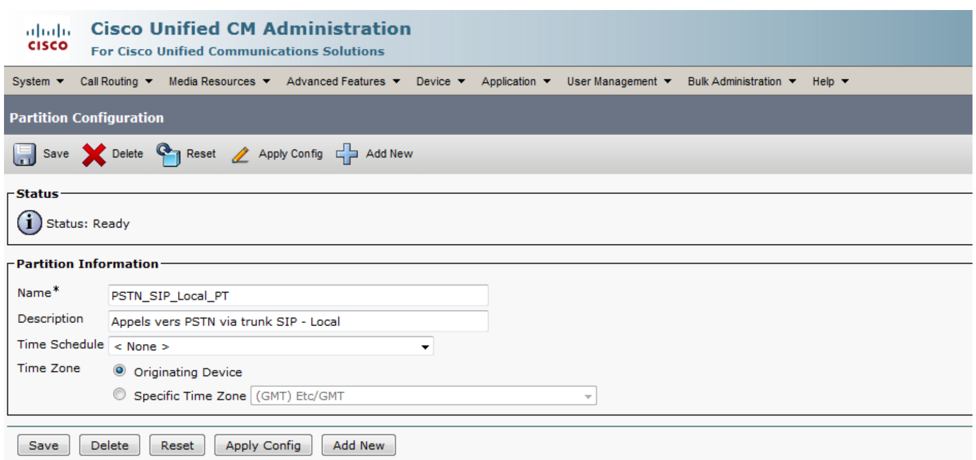
The CUCM and the CUBE are linked by a SIP trunk (a different SIP trunk than the one to Videotron). The configurations presented in this section are configuration suggestions that have been tested successfully. The integrator will modify this configuration to meet all the customer's needs.

Step 1: Login to the Publisher at Cisco Unified CM administration



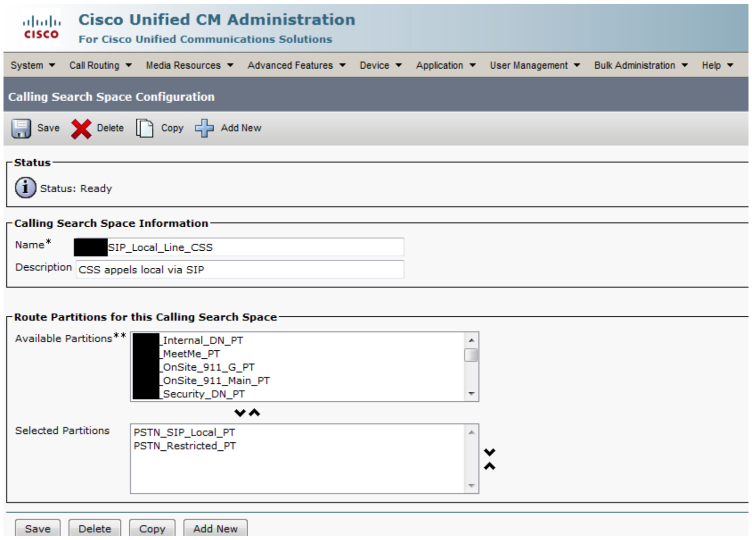
Step 2: Configuring a Partition and a Calling Search Space (outbound calls)

1. Add a partition for the routes intended for outbound calls to the SIP Trunk. Call Routing -> Class of Control -> Partition -> Add New.
2. Enter a meaningful name (e.g., PSTN_SIP_Local_PT) and a meaningful description.



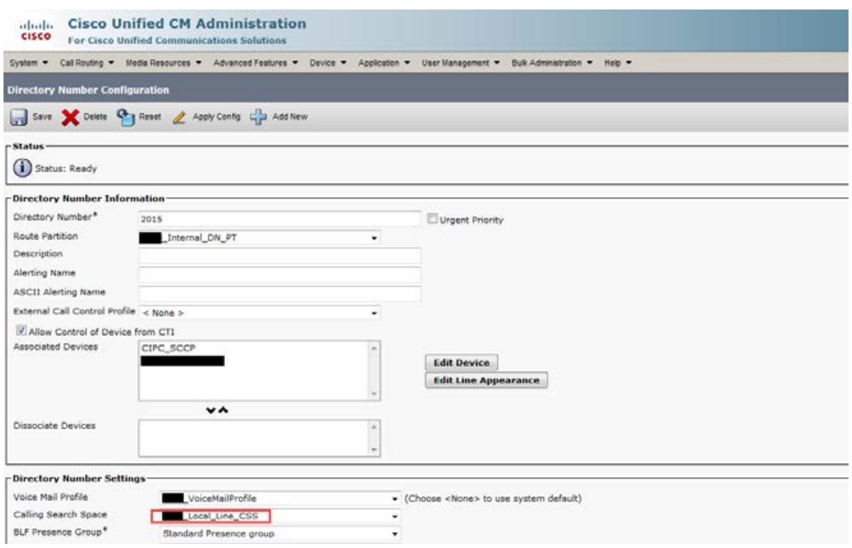
Step 3: Configuring a Calling Search Space (outbound calls)

1. Add a new CSS: Call Routing -> Class of Control -> Calling Search Space -> Add New.
2. Configure the CSS to at least add the Partition created earlier. Use a meaningful name for the CSS. E.g., XXX_SIP_Local_Line_CSS. Replace XXX with the site's acronym, and the remainder of the name provides the PSTN access level (local, long distance, etc.).



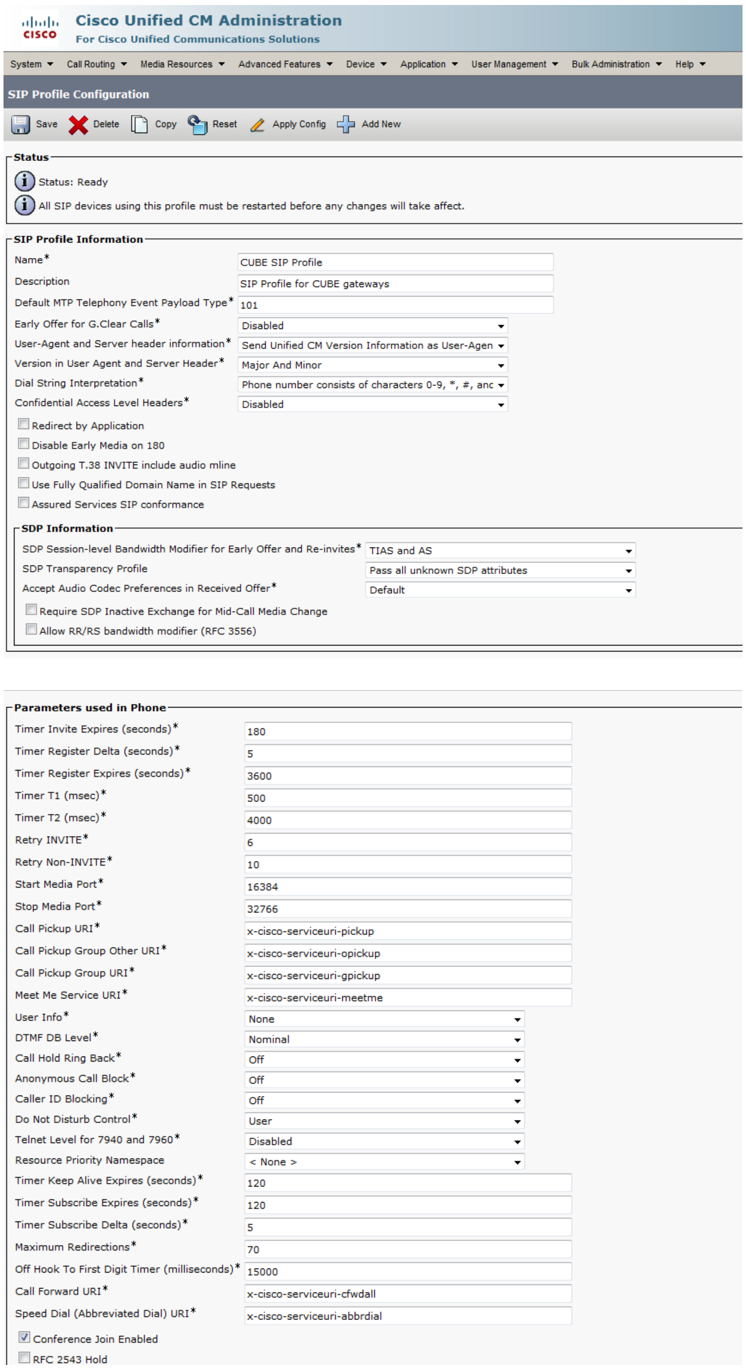
Step 4: Applying the CSS to a test telephone (outbound calls)

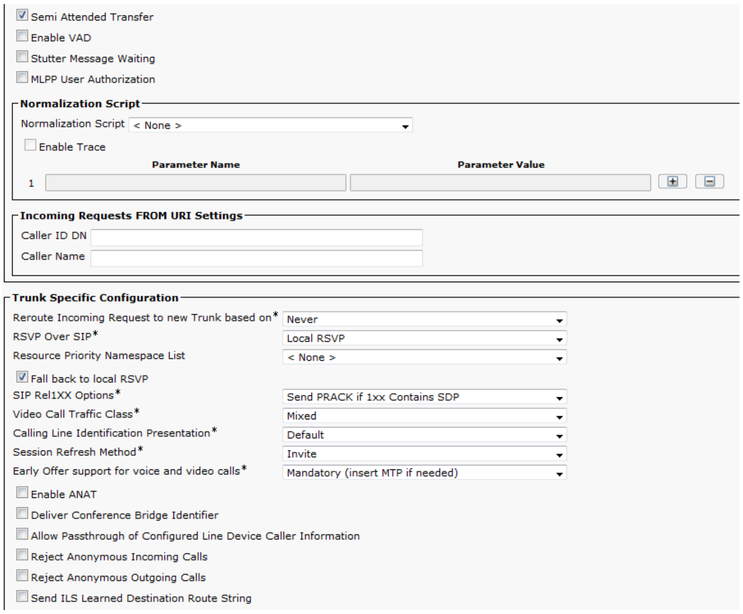
1. Go to the line of a test telephone and select the CSS created in step 3.



Step 5: Configuring a SIP Profile

1. Add a new SIP Profile: Device -> Device Settings -> SIP Profile -> Add New.
2. Configure the SIP Profile as indicated in the image below. Use a meaningful SIP Profile name.





☑ Semi Attended Transfer
☐ Enable VAD
☐ Stutter Message Waiting
☐ MLPP User Authorization

Normalization Script

Normalization Script: < None >
☐ Enable Trace

	Parameter Name	Parameter Value
1		

Incoming Requests FROM URI Settings

Caller ID DN
Caller Name

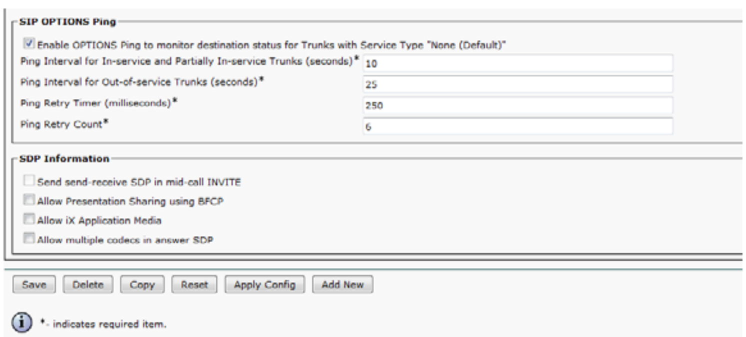
Trunk Specific Configuration

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*	Send PRACK if 1xx Contains SDP
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default
Session Refresh Method*	Invite
Early Offer support for voice and video calls*	Mandatory (insert MTP if needed)

☐ Enable ANAT
☐ Deliver Conference Bridge Identifier
☐ Allow Passthrough of Configured Line Device Caller Information
☐ Reject Anonymous Incoming Calls
☐ Reject Anonymous Outgoing Calls
☐ Send ILS Learned Destination Route String



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)*	10
Ping Interval for Out-of-service Trunks (seconds)*	25
Ping Retry Timer (milliseconds)*	250
Ping Retry Count*	6

SDP Information

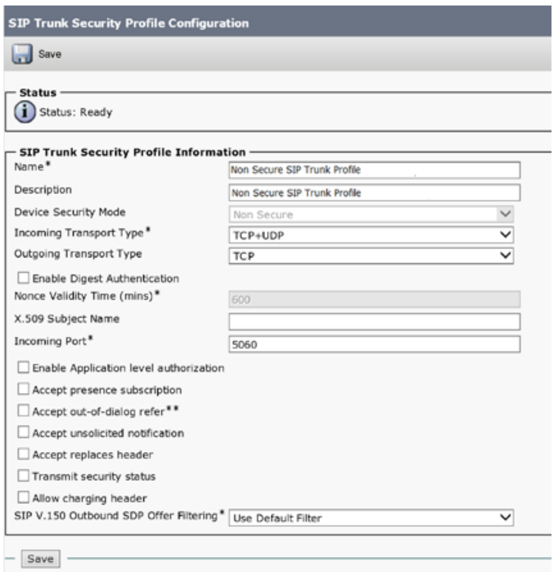
☐ Send send-receive SDP in mid-call INVITE
☐ Allow Presentation Sharing using BFCP
☐ Allow iX Application Media
☐ Allow multiple codecs in answer SDP

Save | Delete | Copy | Reset | Apply Config | Add New

*- indicates required item.

Step 6: Creating a SIP TRUNK Security Profile

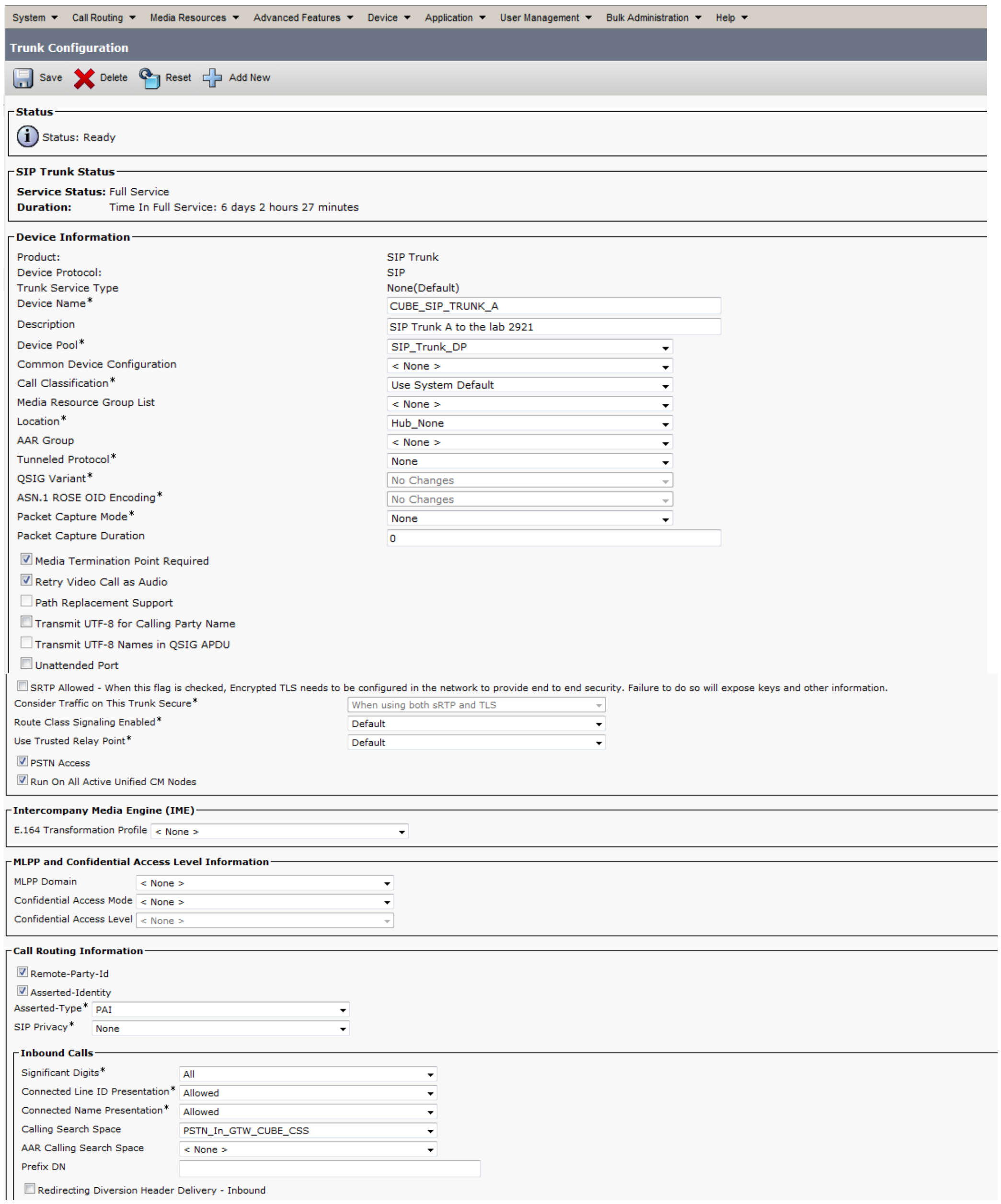
1. Add the SIP Trunk Security Profile. Go to the System menu > Security Profile > SIP Trunk Security Profile.
2. Select the Non-Secure SIP Trunk Profile and click on Copy.
3. Change the SIP Trunk Security Profile name to “PSTN SIP TRUNK Profile,” for example.
4. Save.

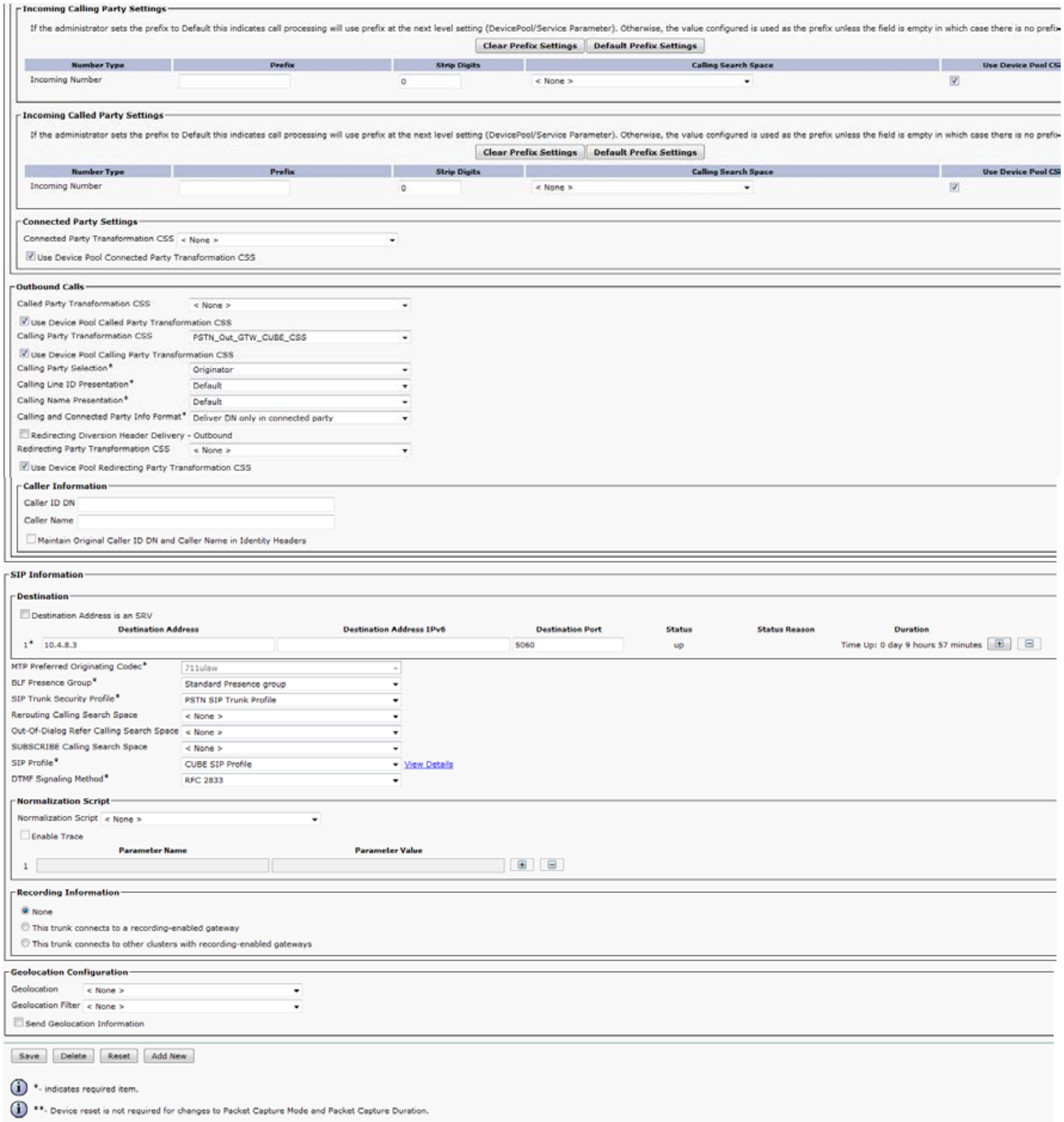


Step 7: Configuring the SIP Trunk

1. Add the Trunk SIP Device-> Trunk -> Add New.
2. Configure the Trunk SIP parameters as indicated in the image below.

Note: The configuration of the Calling Search Spaces for the “Inbound calls” section and the “Calling party transformation CSS” of the “Outbound Calls” part must have been done beforehand. The customer must define his or her call permissions for inbound calls on this SIP Trunk and the way the calling number and called number of an outbound call can be modified. The same applies for the Device Pool and Media Resource Group List.





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

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number		0	< None >	☑

Incoming Called 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

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number		0	< None >	☑

Connected Party Settings

Connected Party Transformation CSS: < None >

☑ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS: < None >

☑ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS: PSTN_Out_GTW_CUBE_CSS

☑ Use Device Pool Calling Party Transformation CSS

Calling Party Selection*: Originator

Calling Line ID Presentation*: Default

Calling Name Presentation*: Default

Calling and Connected Party Info Format*: Deliver DN only in connected party

☐ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS: < None >

☑ 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	Status Reason	Duration
1*	10.4.8.3		5060	up		Time Up: 0 day 9 hours 57 minutes

MTP Preferred Originating Codec*: 711ulaw

BLF Presence Group*: Standard Presence group

SIP Trunk Security Profile*: PSTN SIP Trunk Profile

Rerouting Calling Search Space: < None >

Out-Of-Dialog Refer Calling Search Space: < None >

SUBSCRIBE Calling Search Space: < None >

SIP Profile*: CUBE SIP Profile View Details

DTMF Signaling Method*: RFC 2833

Normalization Script

Normalization Script: < None >

☐ Enable Trace

	Parameter Name	Parameter Value
1		

Recording Information

◉ None

○ This trunk connects to a recording-enabled gateway

○ This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation: < None >

Geolocation Filter: < None >

☐ Send Geolocation Information

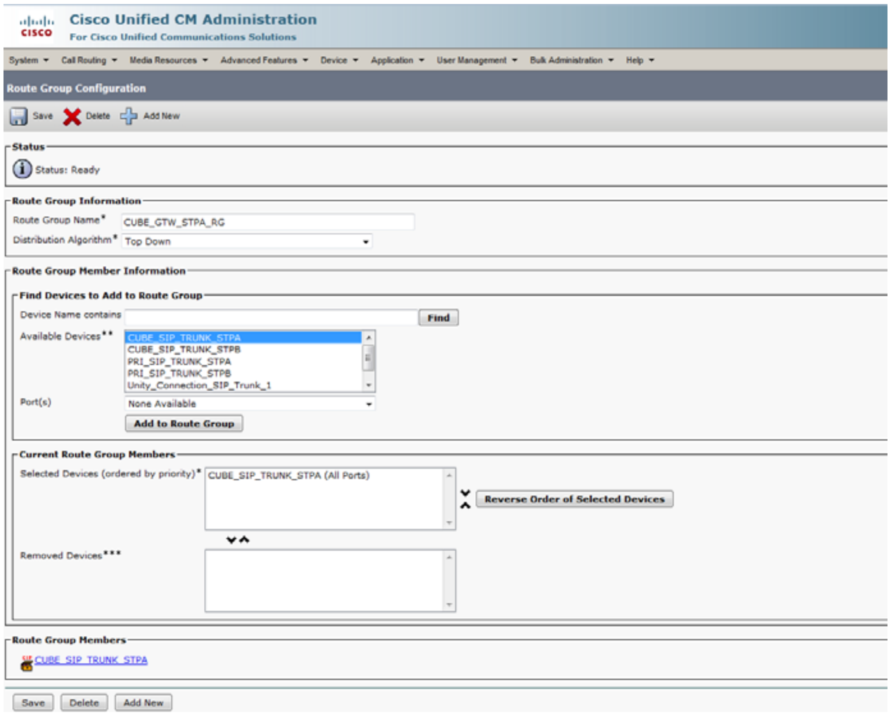
Save | Delete | Reset | Add New

*- indicates required item.

**- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

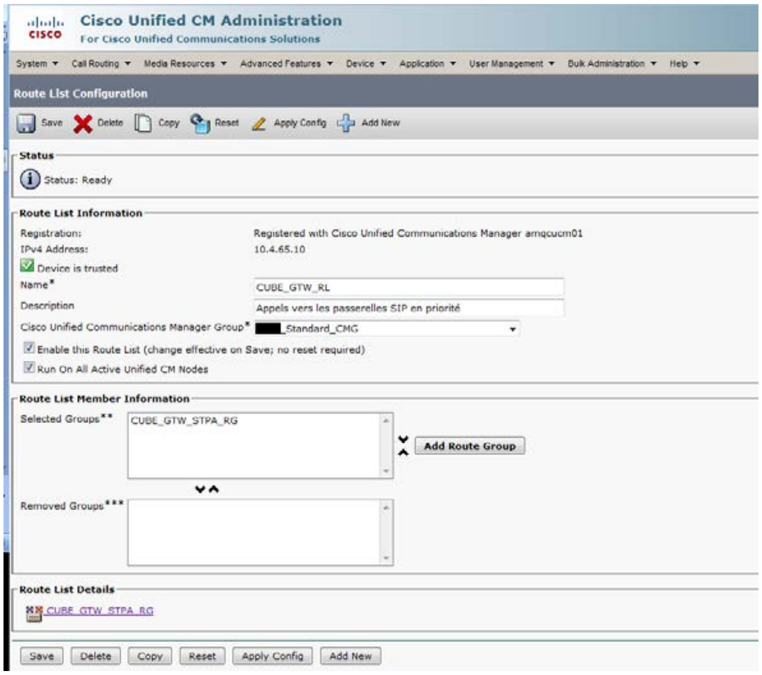
Step 8: Configuring the Route Group (outbound calls)

1. Add a Route Group: Call Routing -> Route/Hunt -> Route Group -> Add New.
2. Configure the Route Group parameters as indicated in the image below.

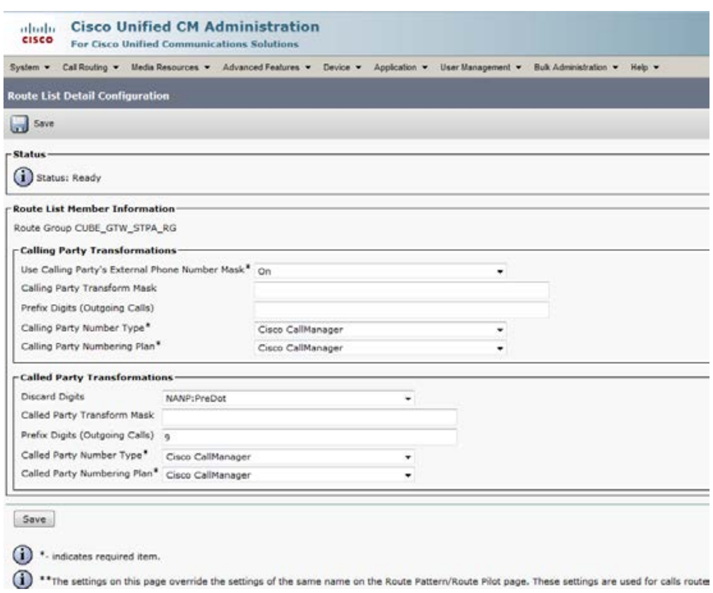


Step 9: Configuring the Route List (outbound calls)

1. Add a Route List: Call Routing -> Route/Hunt -> Route List -> Add New.
2. Configure the Route List parameters as indicated in the image below.

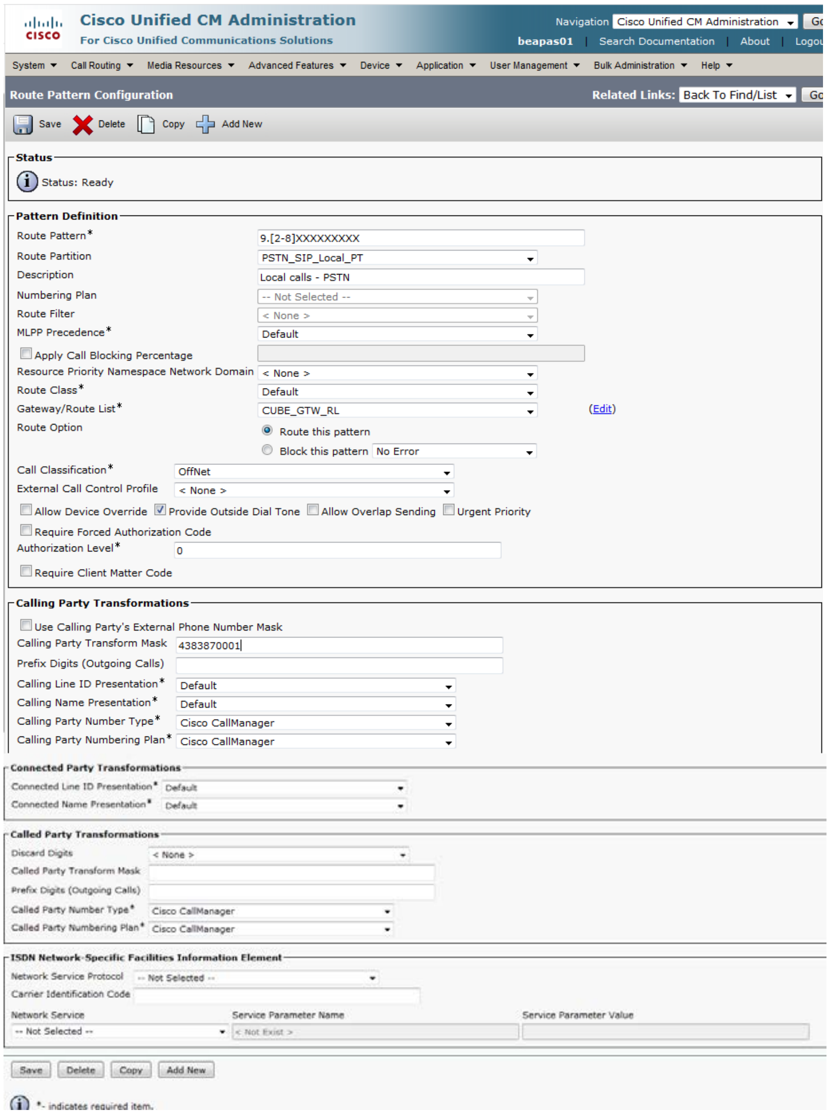


3. Click on the Route Group in the Route List to configure the Route Group parameters when it is used with this Route List.
4. Configure the Route Group parameters as indicated in the image below.



Step 10: Configuring a Route Pattern (outbound calls)

1. Add a Route Pattern: Call Routing -> Route/Hunt -> Route Pattern -> Add New.
2. Configure the Route Pattern parameters as indicated in the image below (use a different number).

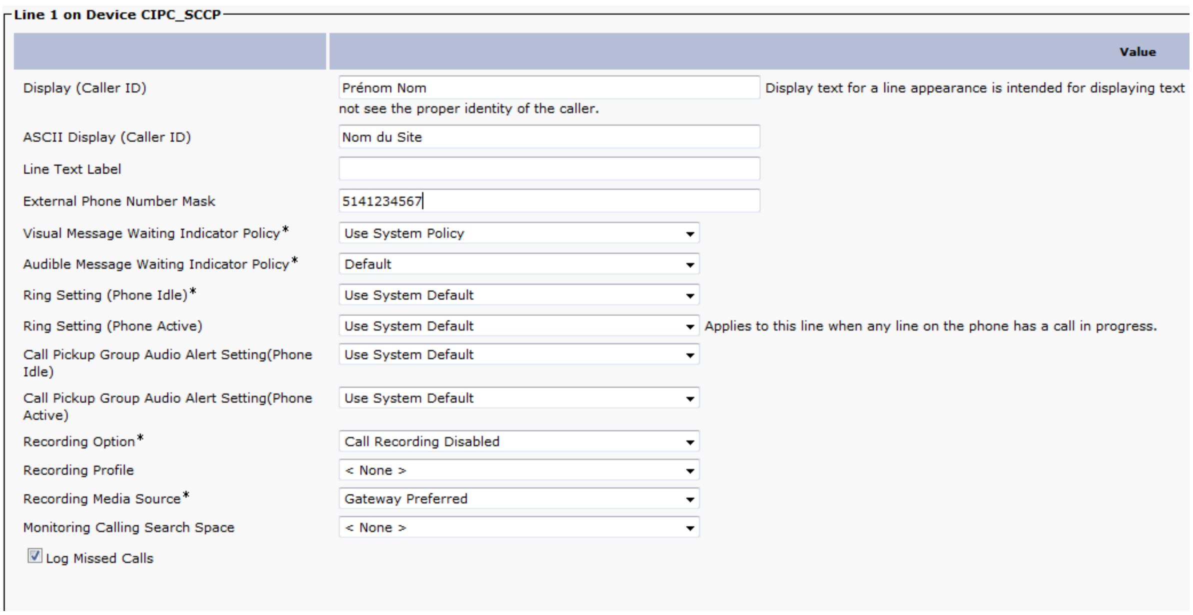


Step 11: Configuring the External Phone Number Mask (outbound calls)

Outbound display can be configured in several locations in the CUCM (e.g., Route pattern, Route-List, on a device's line).

Here is one of the methods for testing whether the name and number ID are working properly for outbound calls on the Trunk toward Videotron.

> Modify the "ASCII Display (Caller ID) field and the "External Phone Number Mask" field in the configuration of the telephone line configured in step 4.



7 Glossary

503	Service unavailable Server error code.
bursting	Feature that allows you to temporarily exceed your calling limit. Simultaneous calls are billed on a pay-per-use basis. Feature currently in development
called number	Number called or requested
called party	Person to whom a call is sent.
calling party	Person sending a call to establish communication.
C20	Videotron telephone switch
CO line	central office line Communication line that connects a PBX to a telephone service provider's switchboard.
G.711	Digital voice encoding standard
H.323	Standard for transmitting audio, data and images in real time across packet networks. Used for local networks, like an intranet, or public networks, like the Internet. Less commonly used than SIP.
IP	Internet protocol
IP-GW	IP gateway
key system	Intercom system, key telephone system Most commonly used telephone system when few additional extensions are required. Allows users to call each other directly and to communicate with public network subscribers via outbound and inbound calls.
original Called Number	
PBX	Private branch exchange A company's private telephone switch
PSTN	*public switched telephone network*
redirect information	
REFER	SIP method for transferring calls whereby the call is sent to a number indicated in the transfer request. Allows you to free up lines after a call is forwarded from an external number to another external number, such as a cellphone.
PSTN	*public switched telephone network*
SBC	session border controller A network element to monitor and protect SIP-based communications from fraud and allowing you to configure SIP trunk settings.
DID	direct inward dialling Telephone feature allowing an outbound caller to reach a subscriber directly without going through an operator or dialling an extension. DID number.
SIP	session initiation protocol Logon protocol used in IP telephony. Refers to an IP telephony service allowing a telephone switch to access the PSTN, thereby supporting the management of call signalling, over IP links using SIP trunking.

Softswitch	software switch, media gateway controller, call controller, call server Interconnection equipment that manages the operation of a media gateway that allows signals carrying voice, data or images to move from a circuit-switched public telephone network to a private packet-switched network, such as a private IP network—or to go in the reverse.
T.38	Encoding standard for sending faxes across IP networks in a real-time mode.
trunk	Circuit A line that connects switches with each other and is used to route information sequentially.
trunk group; TG	Circuitry starting from a single switch and terminating at one or more switches giving access to the same subscribers. In the specific case of the Videotron SIP trunking service, TG refers to a SIP trunk. In certain exceptional situations, there may be more than one TG or multiple SIP trunks between a PBX and Videotron.