Ribbon SBC 1000/2000 V8.0.2 IOT Cisco Unified Communication Manager PlusNet SIP Trunk TCP Application Note

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Overview

This Application Note is a configuration guide for the Ribbon SBC (Session Border Controller) 1000/2000 when connecting to Cisco Unified Communication Manager (CUCM) and PlusNet SIP Trunk.

The configuration guide supports features outlined in the Microsoft Technet web page:

- For additional information on Cisco Platform, visit http://www.cisco.com.
- For additional information on Ribbon SBC 1000/2000, visit https://ribboncommunications.com/.

Introduction

Interoperability compliance testing focuses on verifying inbound and outbound call flows between Ribbon SBC 1000/2000 and Cisco CUCM.

Audience

This technical document provides telecommunications engineers with information for configuring both the Ribbon SBC and the third-party product. Procedures in this document require navigating third-party equipment as well as applying Ribbon SBC Command Line Interface (CLI) commands. To complete the configuration and perform any troubleshooting, the engineer performing the procedures must understand the basic concepts of TCP /UDP, IP/Routing, and SIP/RTP.

> This Application Note is offered as a convenience to Ribbon customers. The specifications and information regarding the product in this document are subject to change without notice. All statements, information, and recommendations contained in this document are believed to be accurate but are presented without warranty of any kind, express or implied, and are provided “AS IS”. Users must take full responsibility for the application of the specifications and information contained here.

> The links are only internal to Ribbon partners and employees. They do not work outside of the Ribbon Network.

Requirements

The following table lists the hardware and software used in the reference configuration.

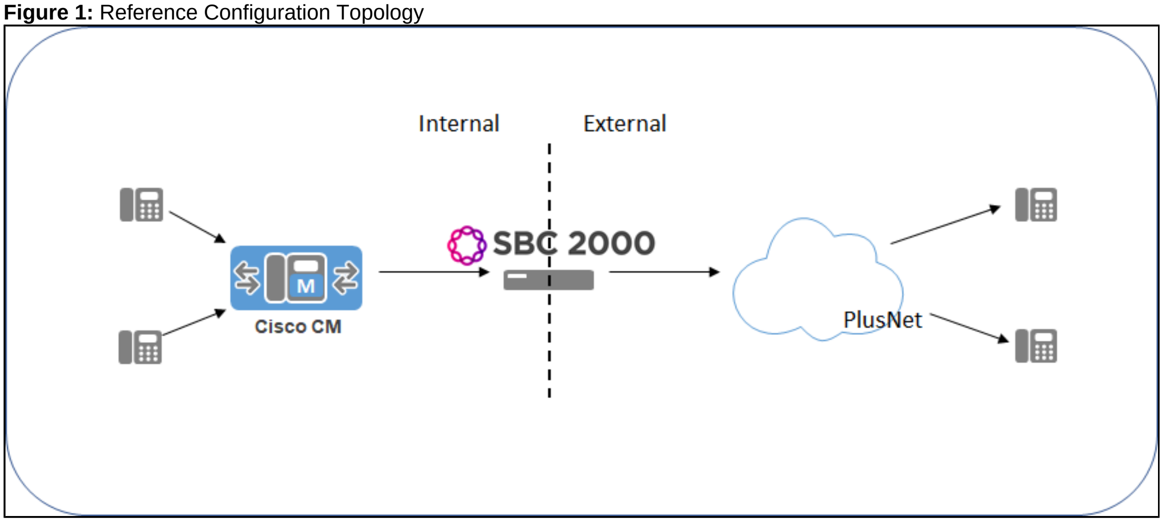
Table 1: Test Equipment and Software

Vendor	Equipment	Software Version
Ribbon Networks	SBC 2000	V8.0.2
Third-party Vendor		
Cisco	Cisco Unified CM Administration	12.0.1.21900
Cisco	Cisco SIP Phone 7841	sip78xx.11-7-1-17
VentaFax	Fax Machine VentaFax	7.6.243.616

Reference Configuration

The following figure serves as a topology for the reference configuration. The figure shows the connectivity between third-party equipment and the Ribbon SBC 1000/2000.

Figure 1: Reference Configuration Topology



Support

For questions about information in this document, contact Ribbon Support in either of the following ways:

- Global Support Assistance Center +1-978-614-8589 or +1-888-391-3434 (English language Support)
- Web: https://ribboncommunications.com/services/ribbon-support-portal-login

Verify License

The interoperability test described in this document requires no special licensing.

CUCM 12.0.1 Configuration

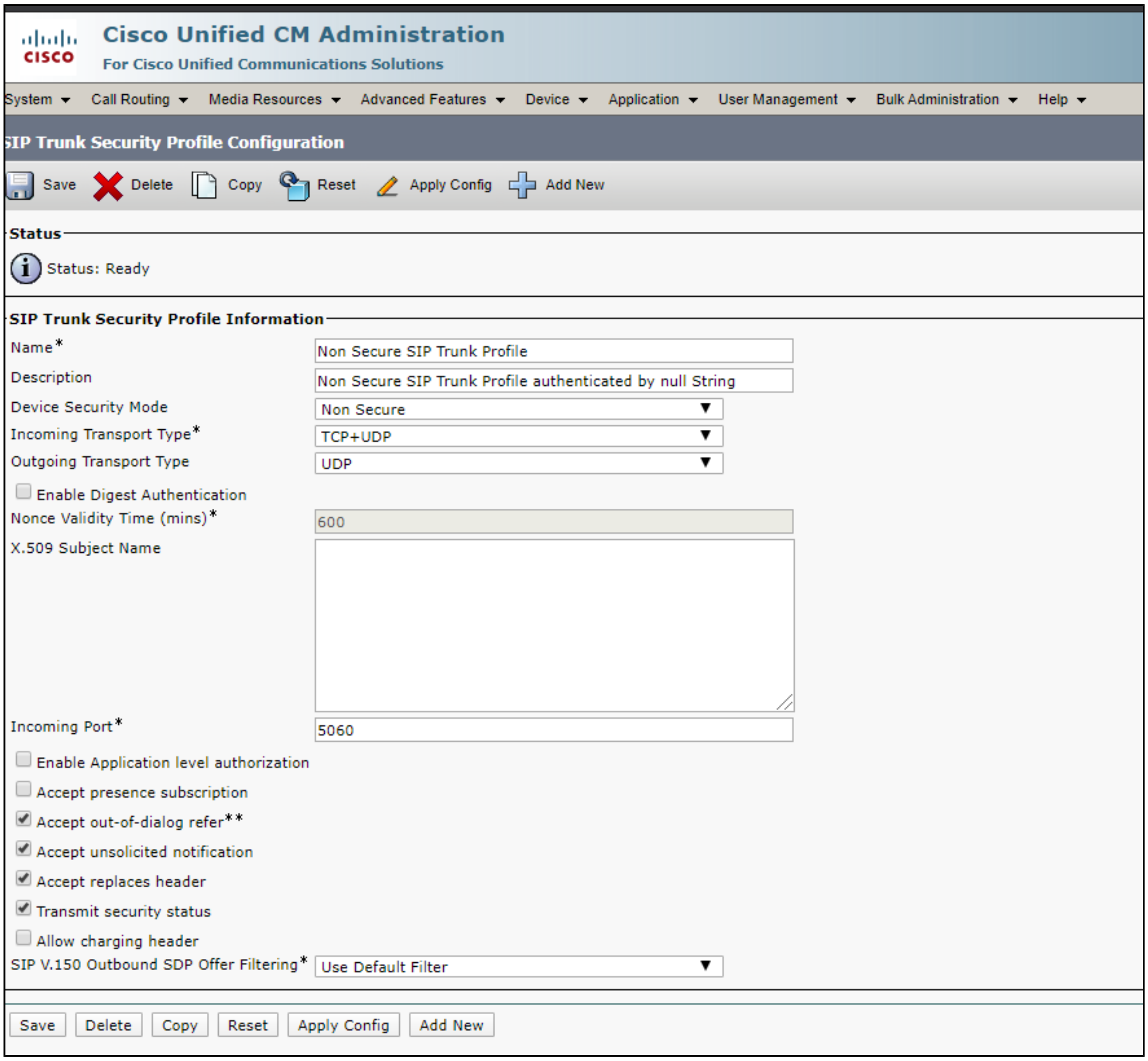
The following new configurations are included in this section:

1. Security Profile
2. SIP Profile
3. SIP Trunk
4. Route Group
5. Route List
6. Route Pattern

1. Security Profile

Select **System > Security > SIP Trunk Security Profile**

Figure 2: SIP Trunk Security Profile


Cisco Unified CM Administration
For Cisco Unified Communications Solutions
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
SIP Trunk Security Profile Configuration
Save Delete Copy Reset Apply Config Add New
Status
Status: Ready
SIP Trunk Security Profile Information
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
Nonce Validity Time (mins)*
600
X.509 Subject Name
Incoming Port*
5060
Enable Application level authorization
Accept presence subscription
Accept out-of-dialog refer**
Accept unsolicited notification
Accept replaces header
Transmit security status
Allow charging header
SIP V.150 Outbound SDP Offer Filtering*
Use Default Filter
Save Delete Copy Reset Apply Config Add New

2. SIP Profile

Select **Device > Device Settings > SIP Profile**

Figure 3: SIP Profile

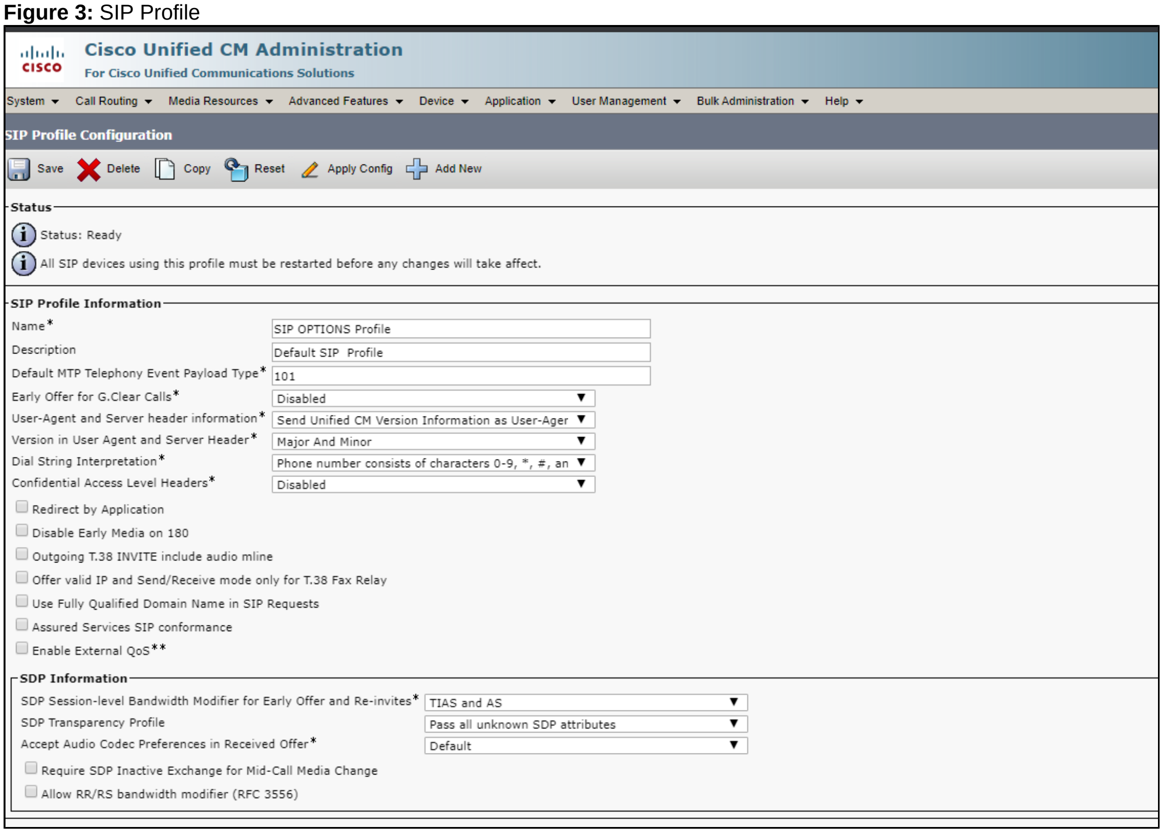


Figure 4: SIP Profile1

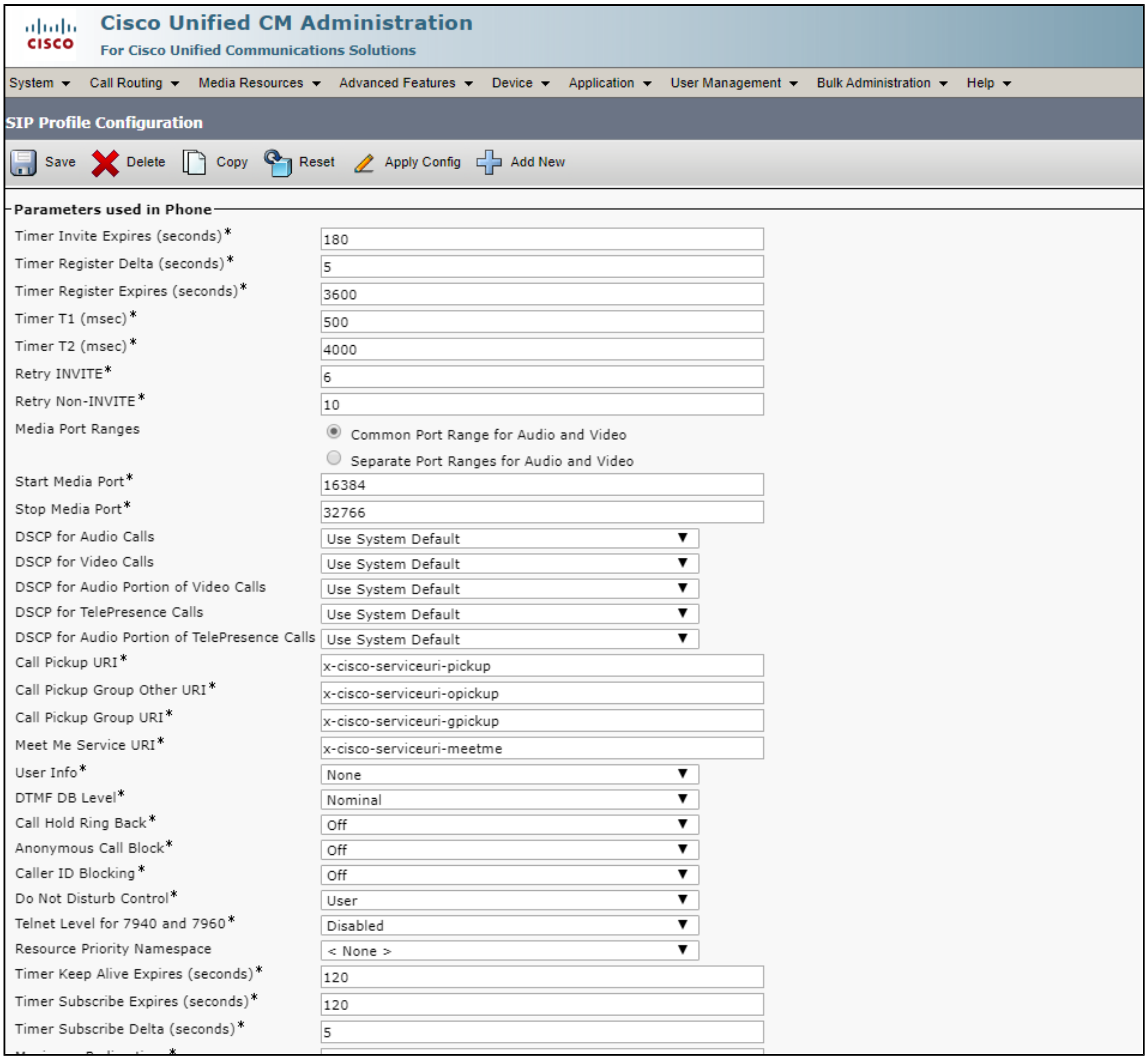


Figure 5: SIP Profile2

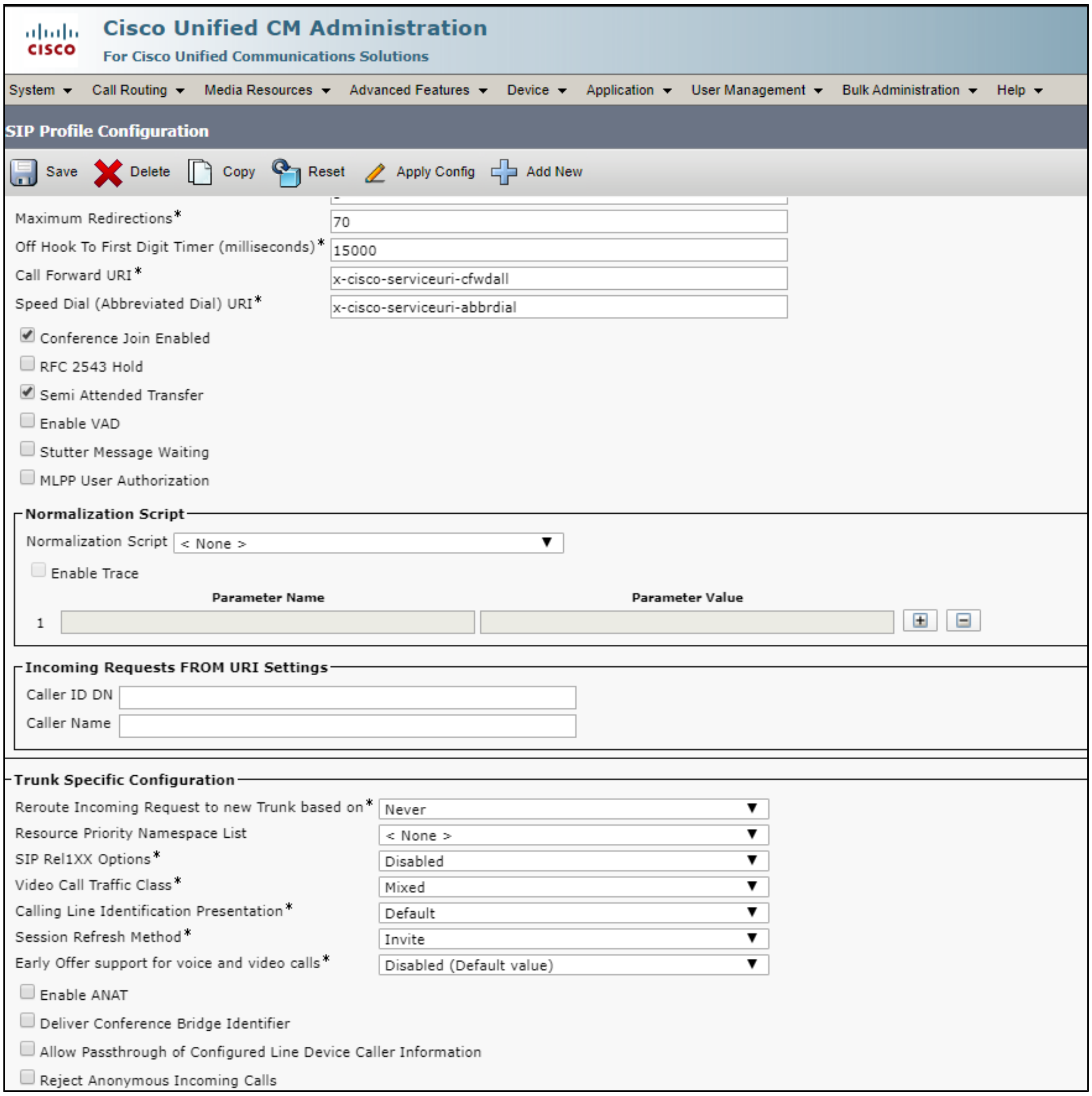
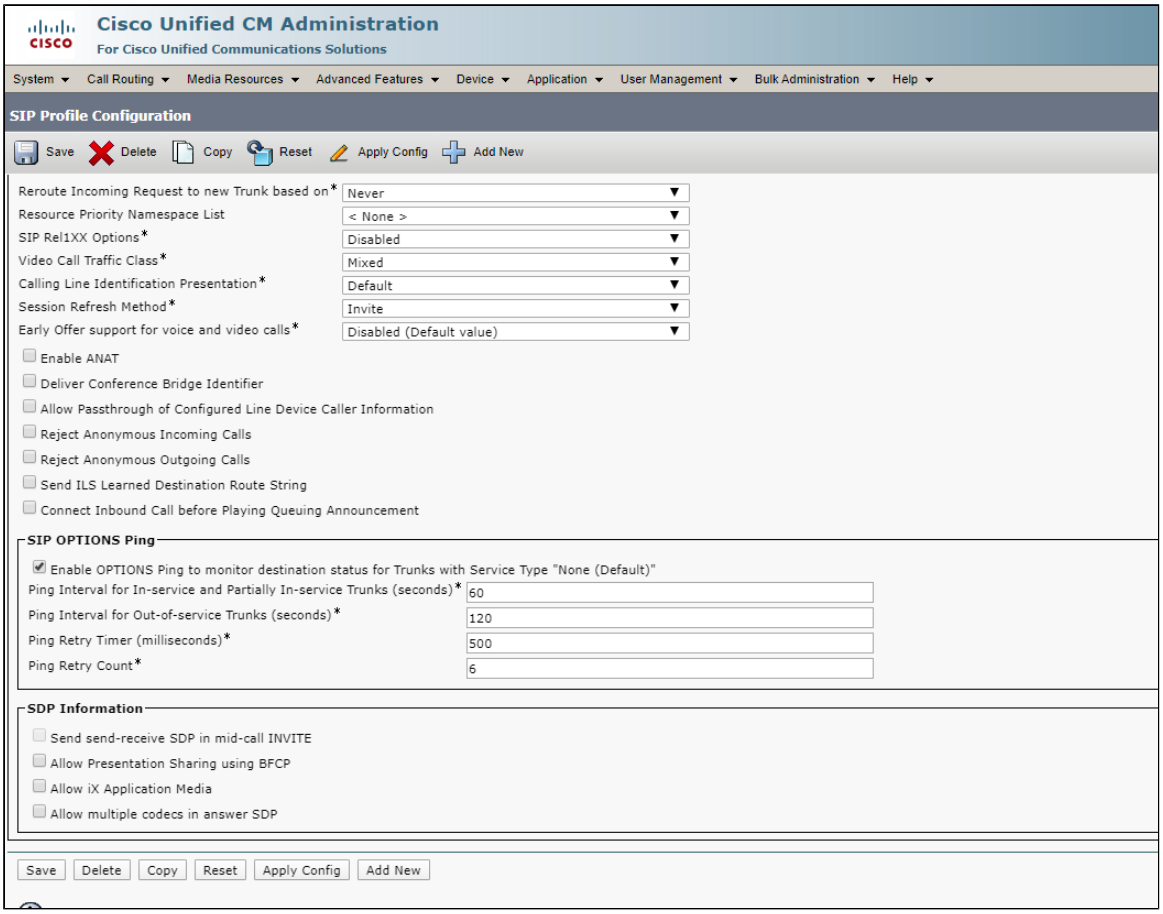


Figure 6: SIP Profile3



3. SIP Trunk

Select **Device > Trunk > Add New**

Figure 7: SIP Trunk

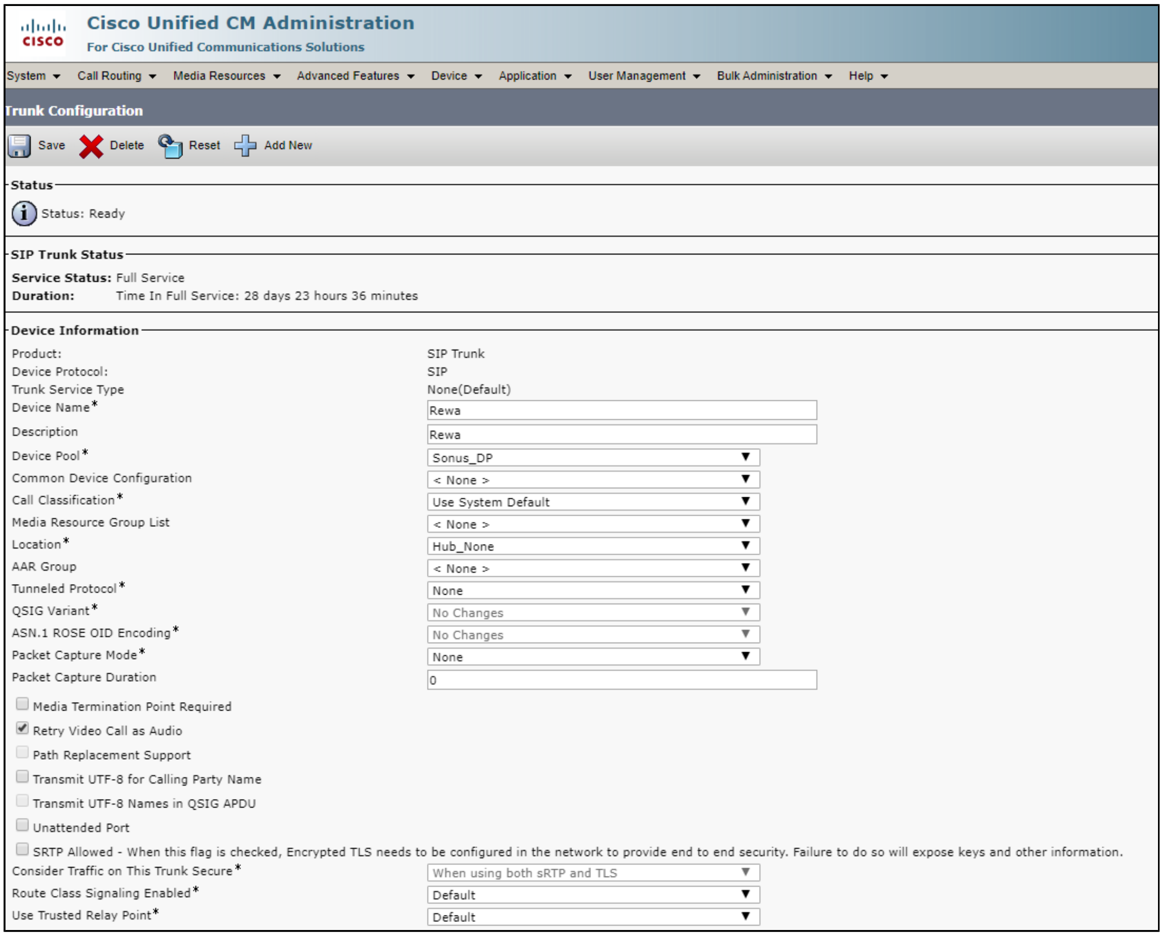


Figure 8: SIP Trunk1

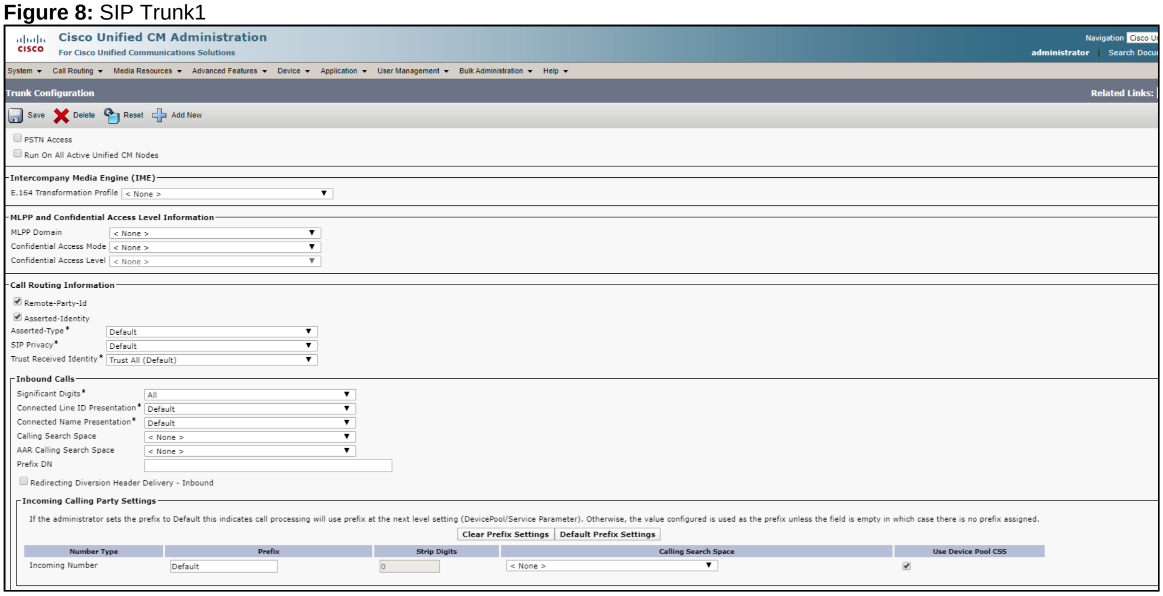


Figure 9: SIP Trunk2

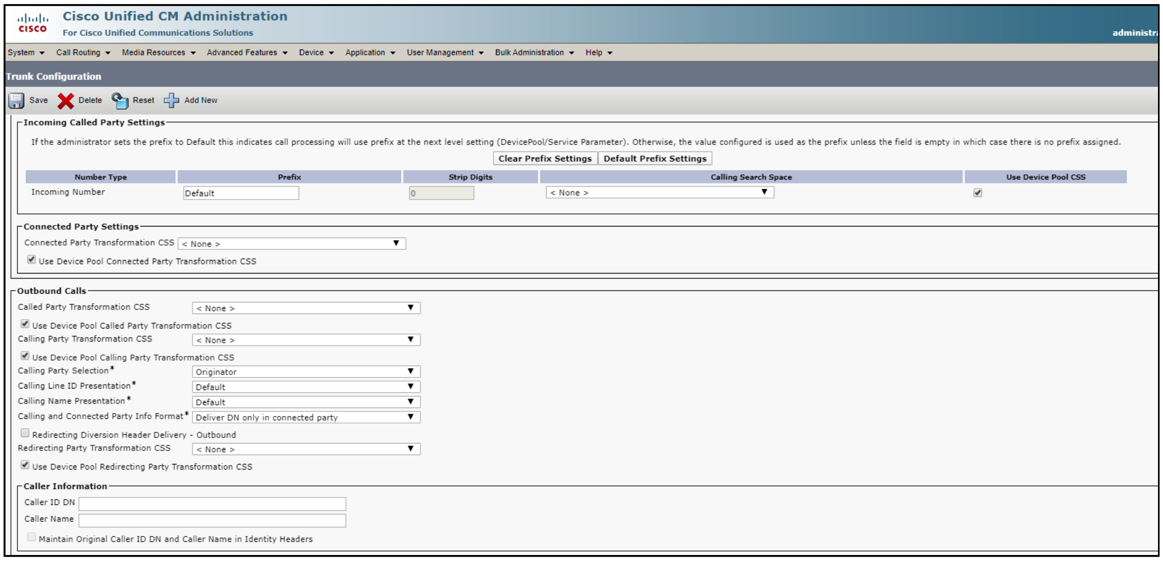
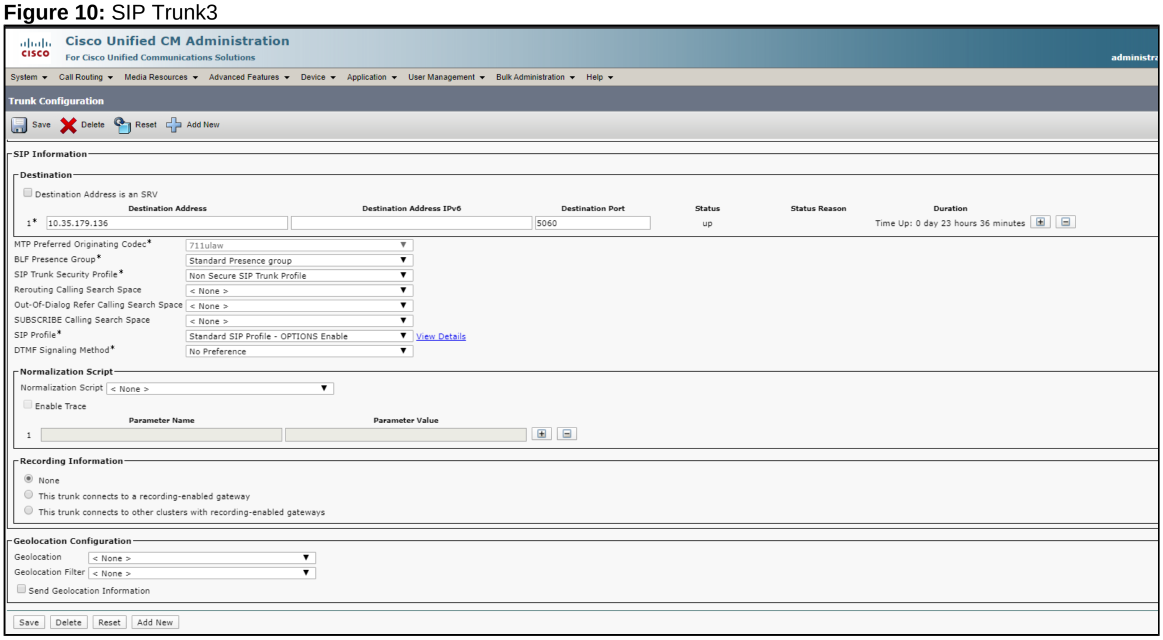


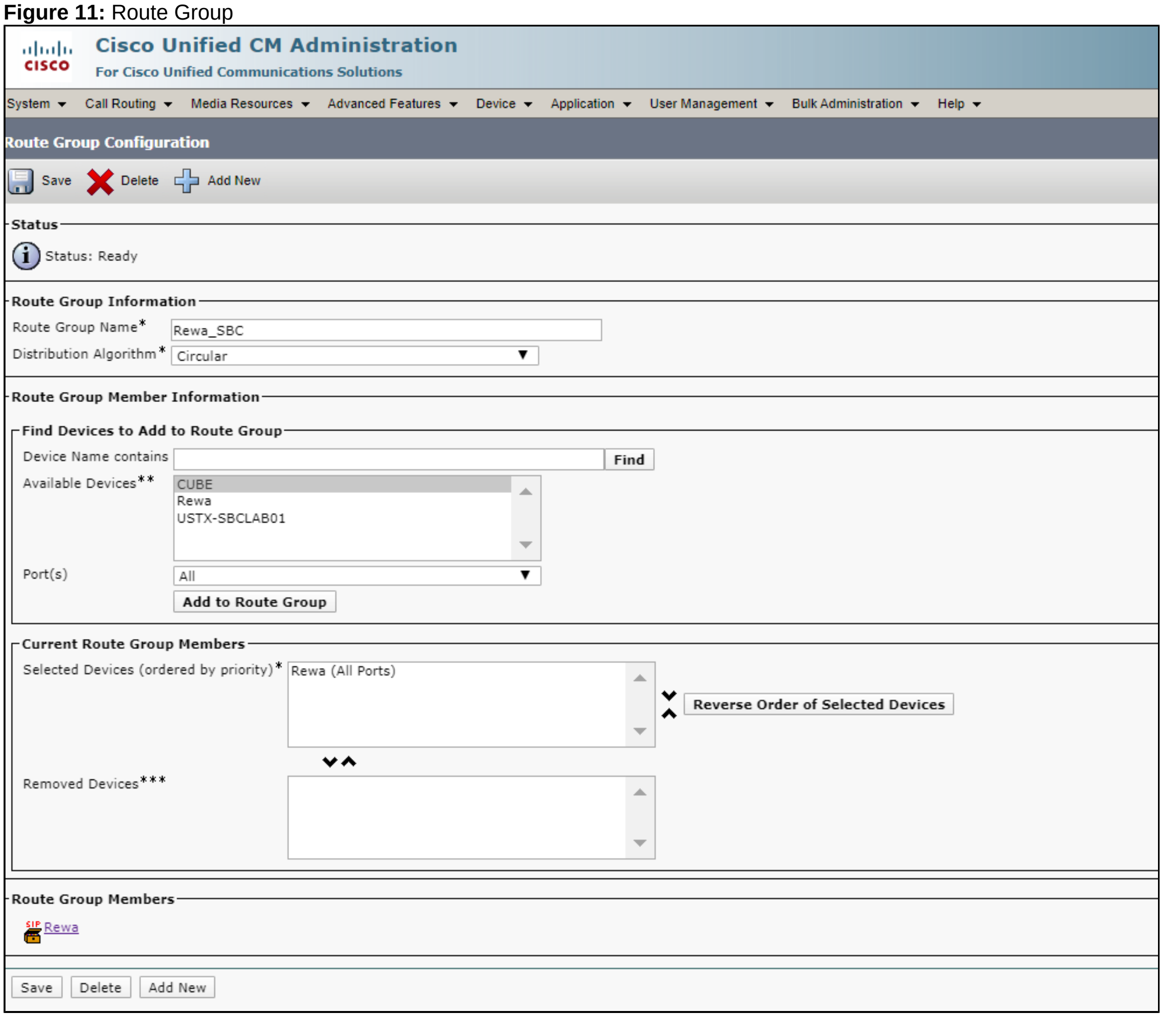
Figure 10: SIP Trunk3



4. Route Group

Select **Call Routing > Route/Hunt > Route Group > Add New**

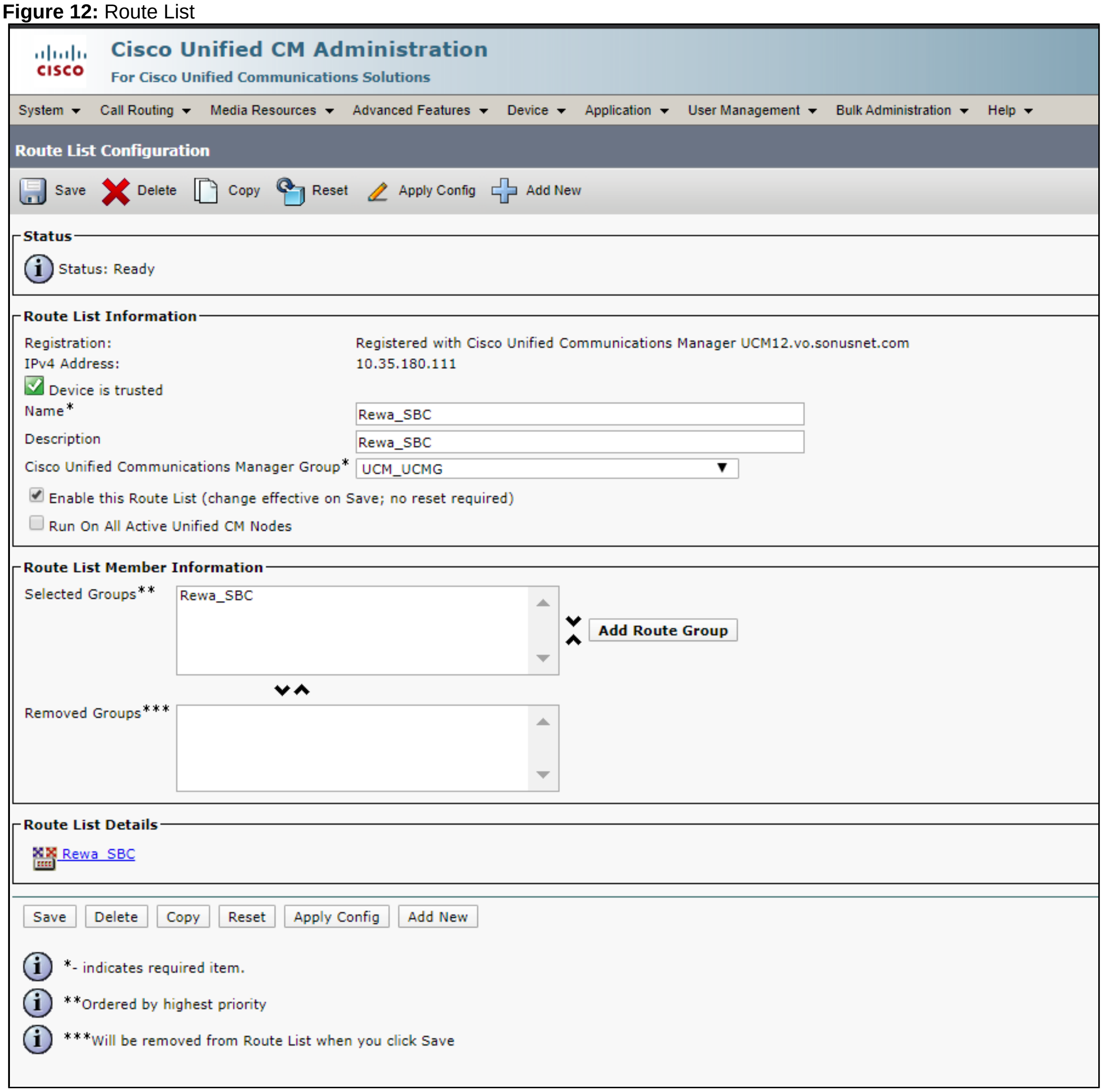
Figure 11: Route Group



5. Route List

Select **Call Routing > Route/Hunt > Route List > Add New**

Figure 12: Route List



6. Route Pattern

Select **Call Routing > Route/Hunt > Route Pattern > Add New**

Figure 13: Route Pattern

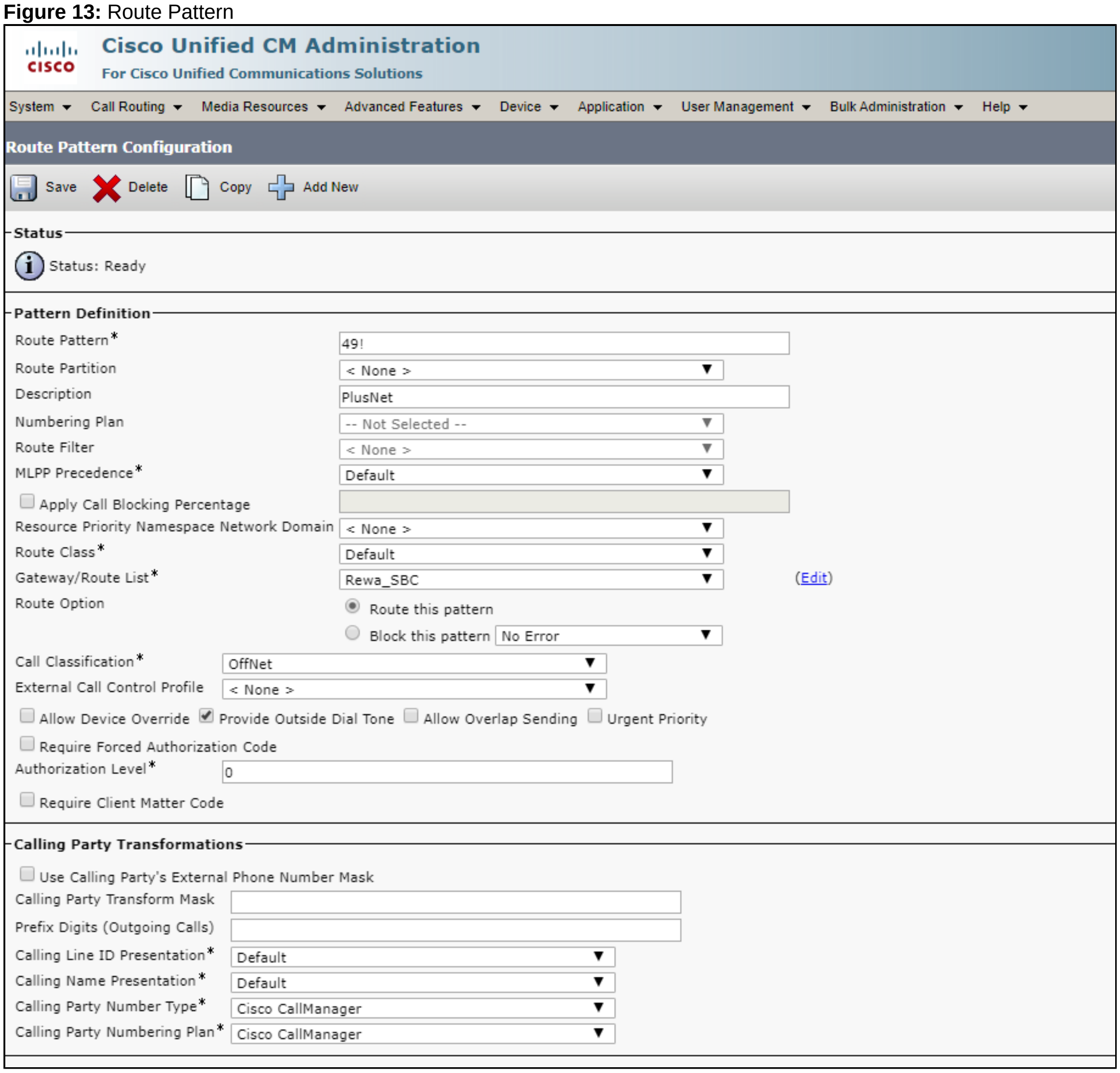
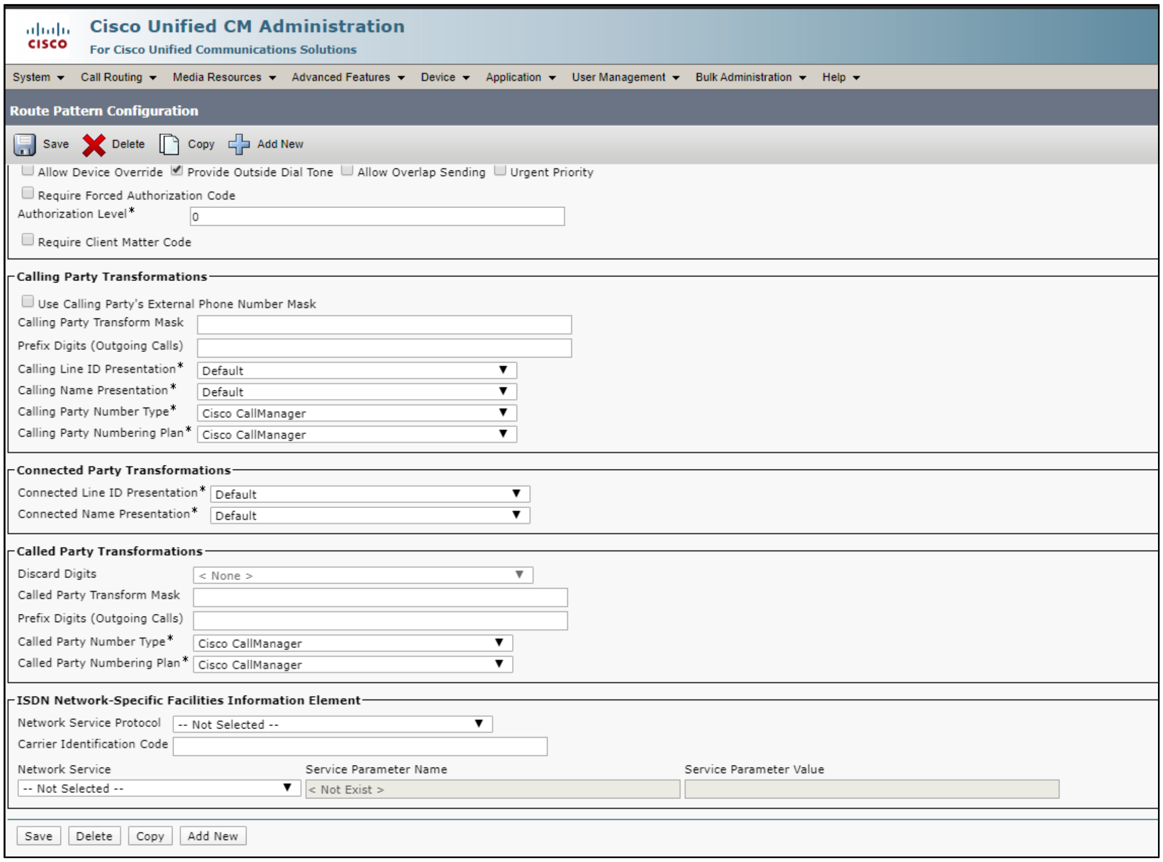


Figure 14: Route Pattern1



Ribbon SBC 1000/2000 Configuration

The following configuration steps provide an example of how to configure the Ribbon SBC 1000/2000 to interoperate with Skype 2015 and Virgin Media SIP Trunk:

1. SIP Profile
2. SIP Server
3. Media System
4. Media Profiles
5. Media List
6. Remote Authorization Tables
7. Signaling Groups
8. Transformation
9. Call Routing Table

1. SIP Profile

SIP Profiles control how the Ribbon SBC 1000/2000 communicates with SIP devices. The profiles control important characteristics such as session timers, SIP Header customization, SIP timers, MIME payloads, and option tags.

Select **Settings > SIP > SIP Profiles** to access the SIP Profile screen.

The following figures show the default SIP profile used for the Ribbon 1000/2000 used for this configuration effort.

Figure 15: PlusNet SIP Profile

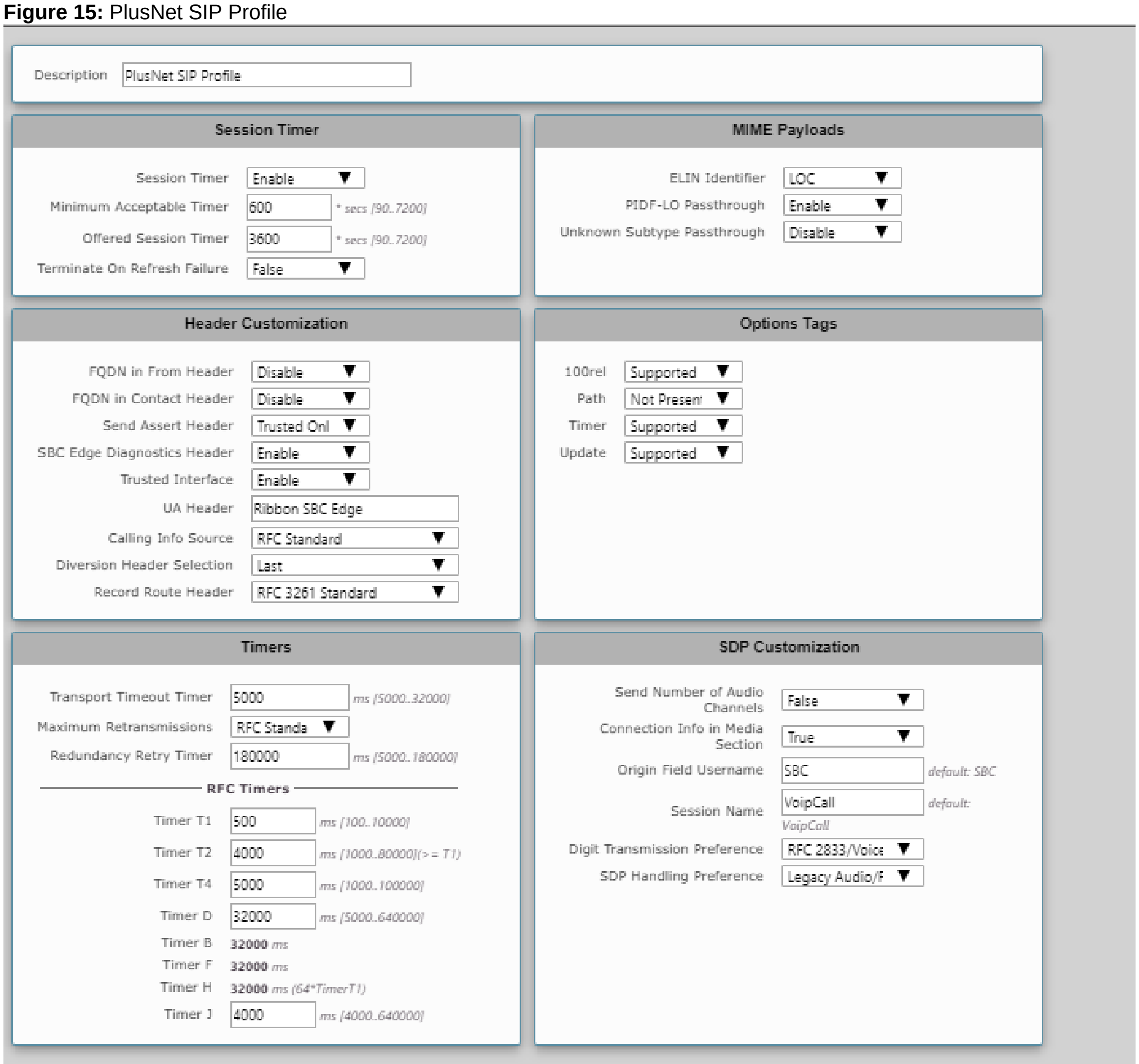


Figure 16: CUCM 12.0.1 SIP Profile

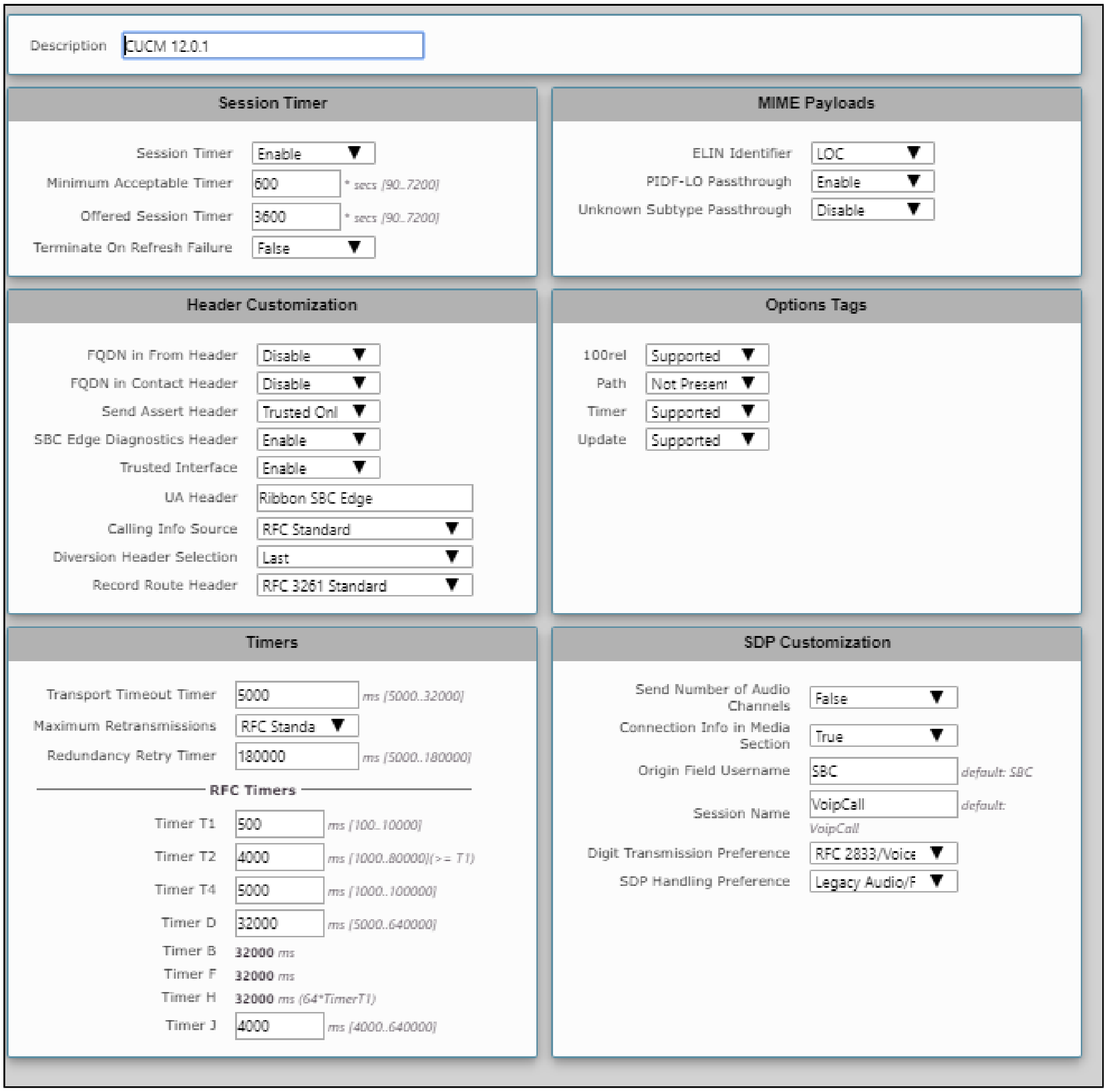


Figure 17: Fax SIP Profile


Tenor-Fax
Description
Tenor-Fax
Session Timer
Session Timer
Enable
Minimum Acceptable Timer
600
* secs [90..7200]
Offered Session Timer
3600
* secs [90..7200]
Terminate On Refresh Failure
False
MIME Payloads
ELIN Identifier
LOC
PIDF-LO Passthrough
Enable
Unknown Subtype Passthrough
Disable
Header Customization
FQDN in From Header
Disable
FQDN in Contact Header
Disable
Send Assert Header
Trusted Onl
SBC Edge Diagnostics Header
Enable
Trusted Interface
Enable
UA Header
Sonus SBC
Calling Info Source
RFC Standard
Diversion Header Selection
Last
Record Route Header
RFC 3261 Standard
Options Tags
100rel
Supported
Path
Not Present
Timer
Supported
Update
Supported
Timers
Transport Timeout Timer
5000
ms [5000..32000]
Maximum Retransmissions
RFC Standa
Redundancy Retry Timer
180000
ms [5000..180000]
RFC Timers
Timer T1
500
ms [100..10000]
Timer T2
4000
ms [1000..80000](>= T1)
Timer T4
5000
ms [1000..100000]
Timer D
32000
ms [5000..640000]
Timer B
32000 ms
Timer F
32000 ms
Timer H
32000 ms (64*TimerT1)
Timer J
4000
ms [4000..640000]
SDP Customization
Send Number of Audio Channels
True
Connection Info in Media Section
True
Origin Field Username
SBC
default: SBC
Session Name
VoipCall
default: VoipCall
Digit Transmission Preference
RFC 2833/Voice
SDP Handling Preference
Legacy Audio/F

2. SIP Server

SIP Server Tables contain information about the SIP devices connected to the Ribbon SBC 1000/2000.

Select **Settings > SIP > SIP Server Tables** to access the SIP Server Tables screen.

The entries in the tables provide information about the IP addresses, ports, and protocols used to communicate with each SIP server. The entries also contain links to counters that are useful for troubleshooting, as shown in the following figures.

Figure 18: PlusNet SIP Servers

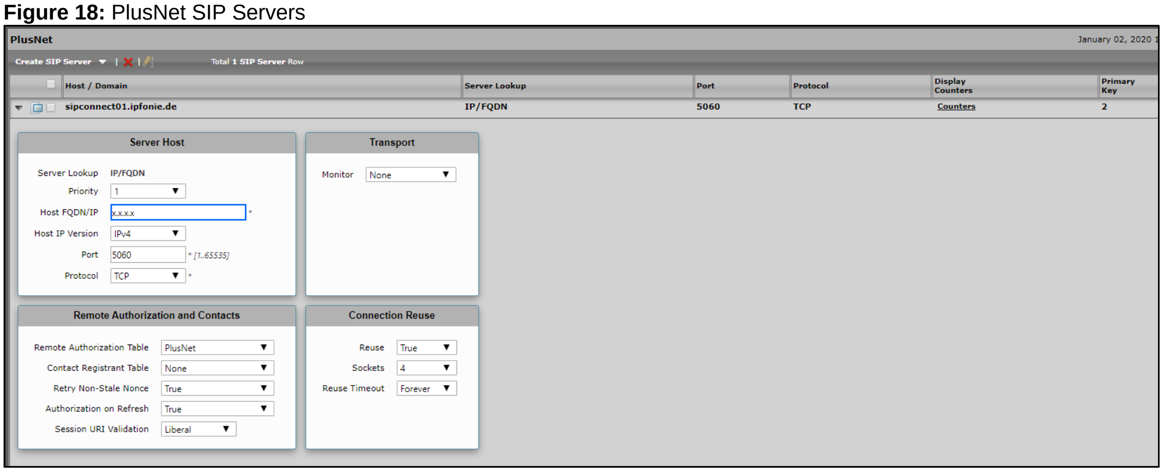


Figure 19: CUCM SIP Server

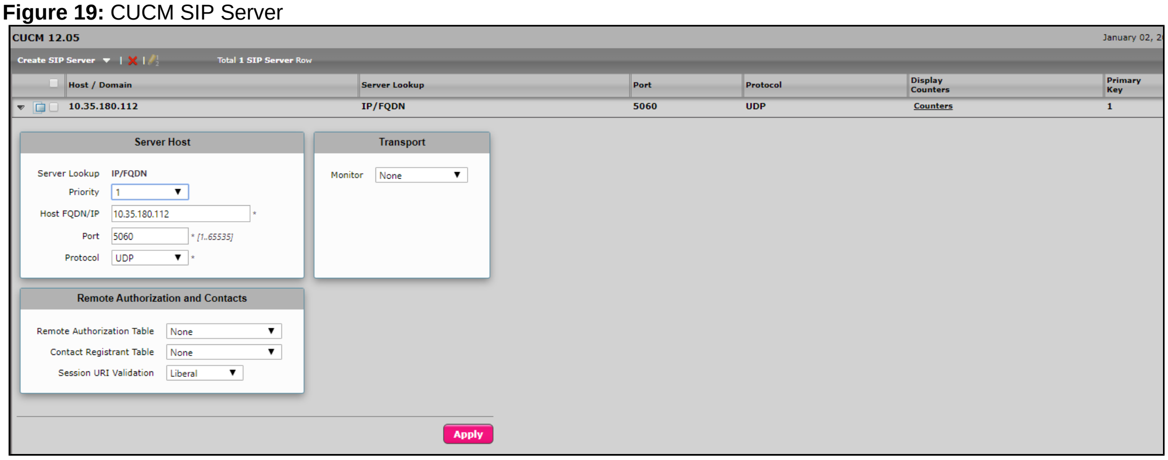
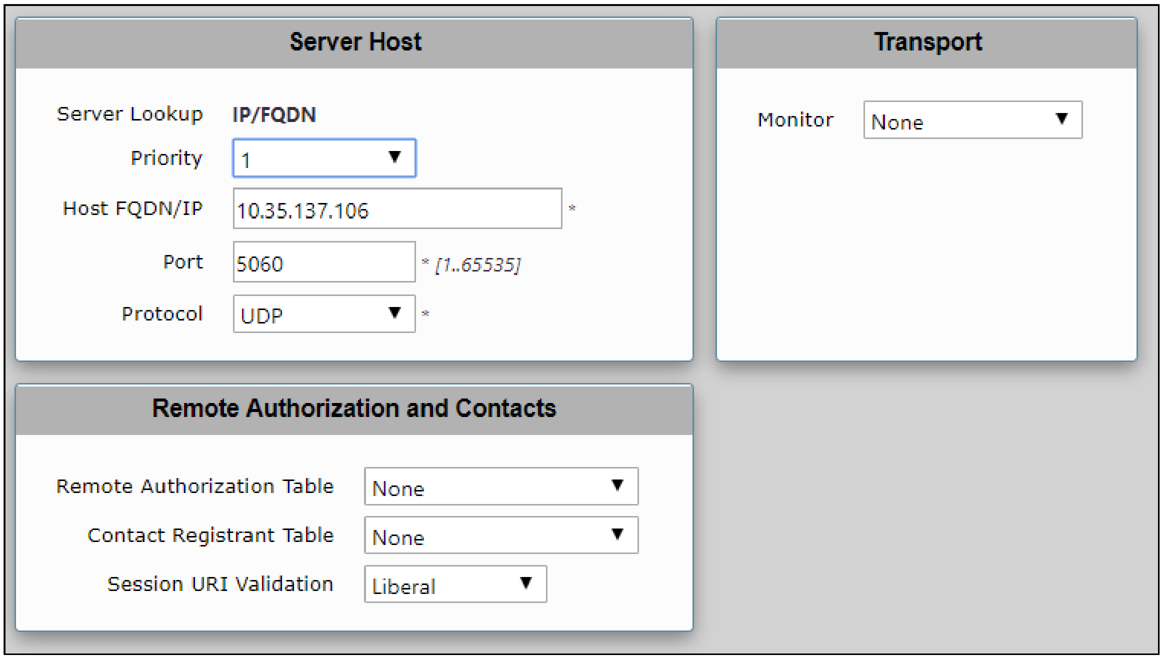


Figure 20: Fax SIP Server

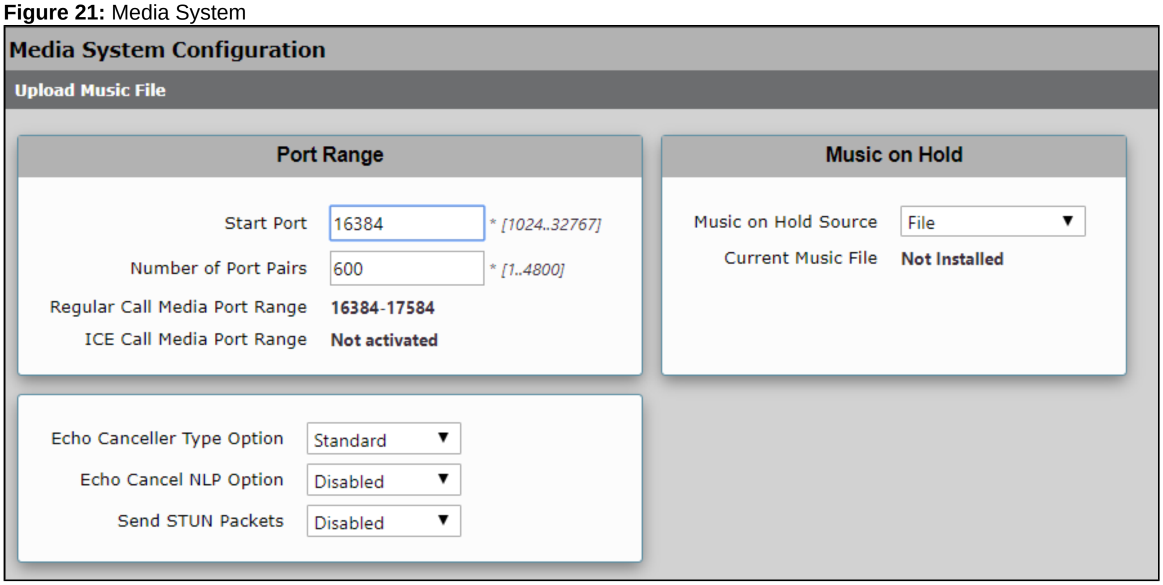

Server Host
Server Lookup
IP/FQDN
Priority
1
Host FQDN/IP
10.35.137.106
Port
5060
* [1..65535]
Protocol
UDP
Transport
Monitor
None
Remote Authorization and Contacts
Remote Authorization Table
None
Contact Registrant Table
None
Session URI Validation
Liberal

3. Media System

The Media System Configuration contains system-wide settings for the Media System. Configuring the media system means setting the number of RTP/RTCP port pairs and the starting port.

Select **Settings > Media > Media System Configuration** to access the Media System configuration screen.

Figure 21: Media System



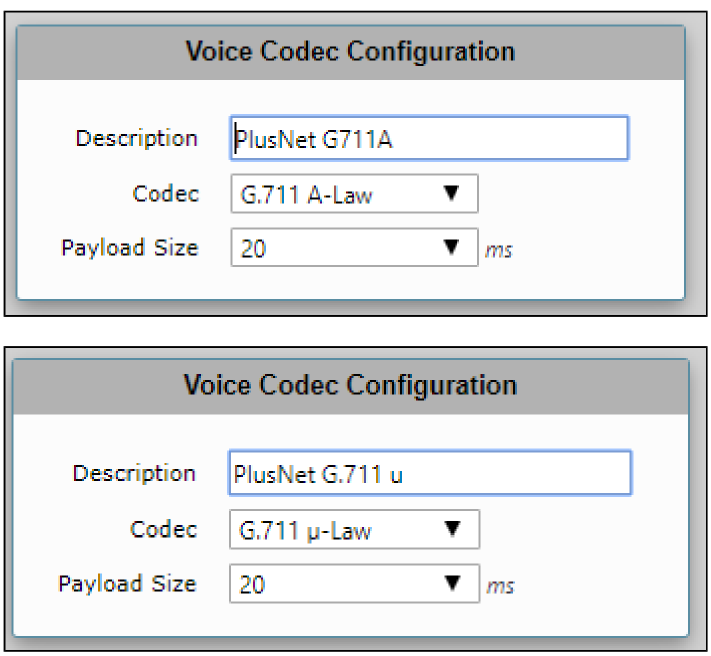
4. Media Profiles

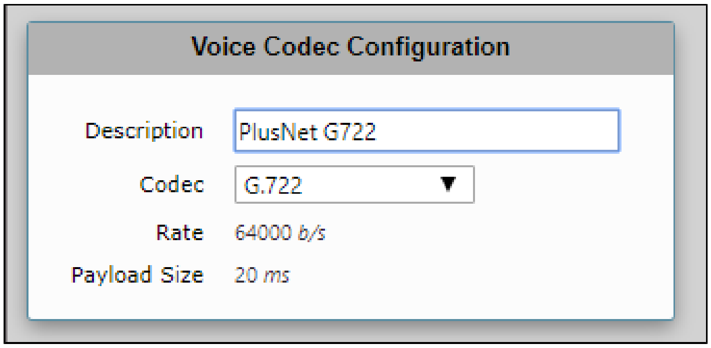
Media Profiles specify the individual voice and fax compression codecs and their associated settings for inclusion into a Media List. Different codecs provide varying levels of compression, allowing the reduction of bandwidth requirements at the expense of voice quality.

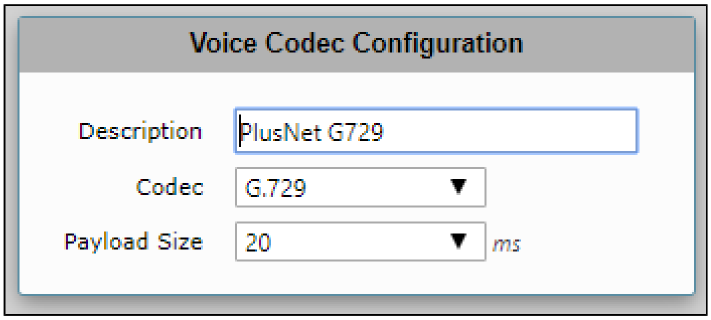
Select **Settings > Media > Media Profiles.**

The following figures illustrate possible media profiles of the voice codecs used for the SBC 1000/2000. The examples are for reference only.

Figure 22: PlusNet Media Profile







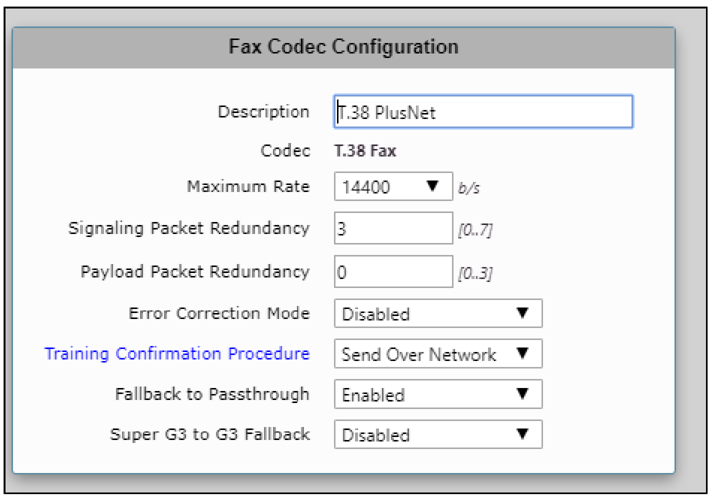


Figure 23: CUCM Media Profile

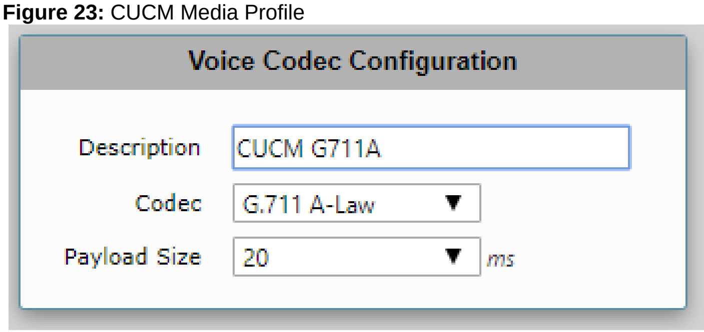
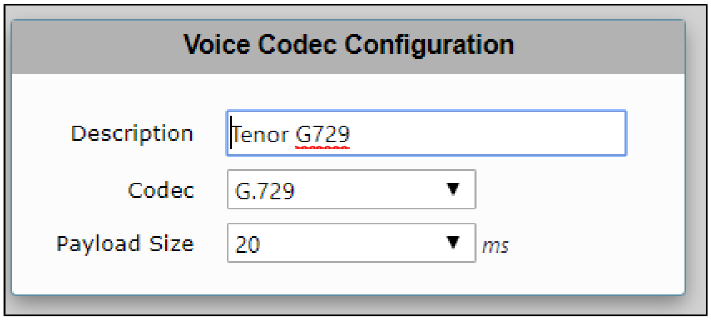
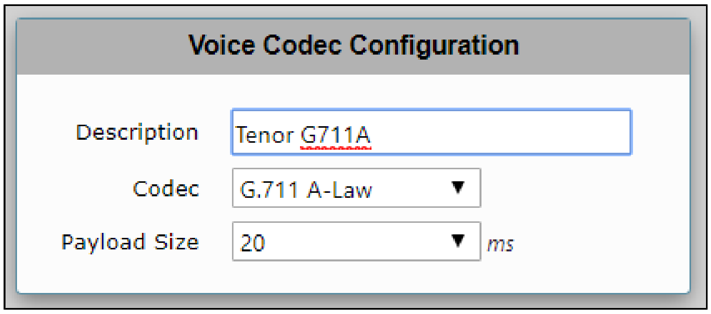
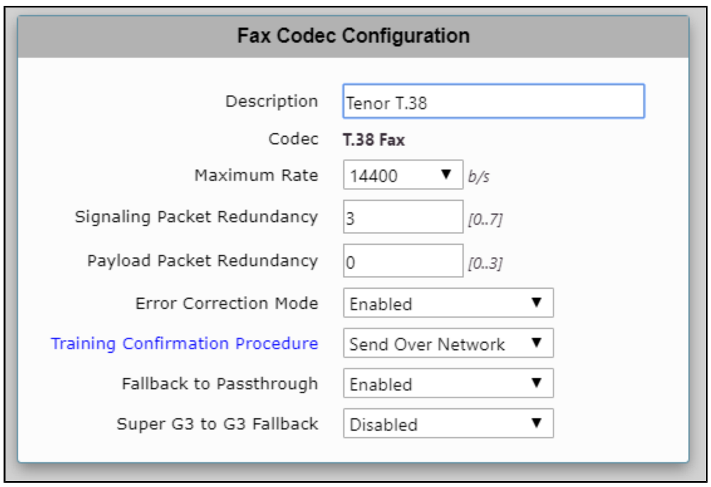


Figure 24: Fax Media Profile


Voice Codec Configuration
Description
Tenor G729
Codec
G.729
Payload Size
20
ms


Voice Codec Configuration
Description
Tenor G711A
Codec
G.711 A-Law
Payload Size
20
ms

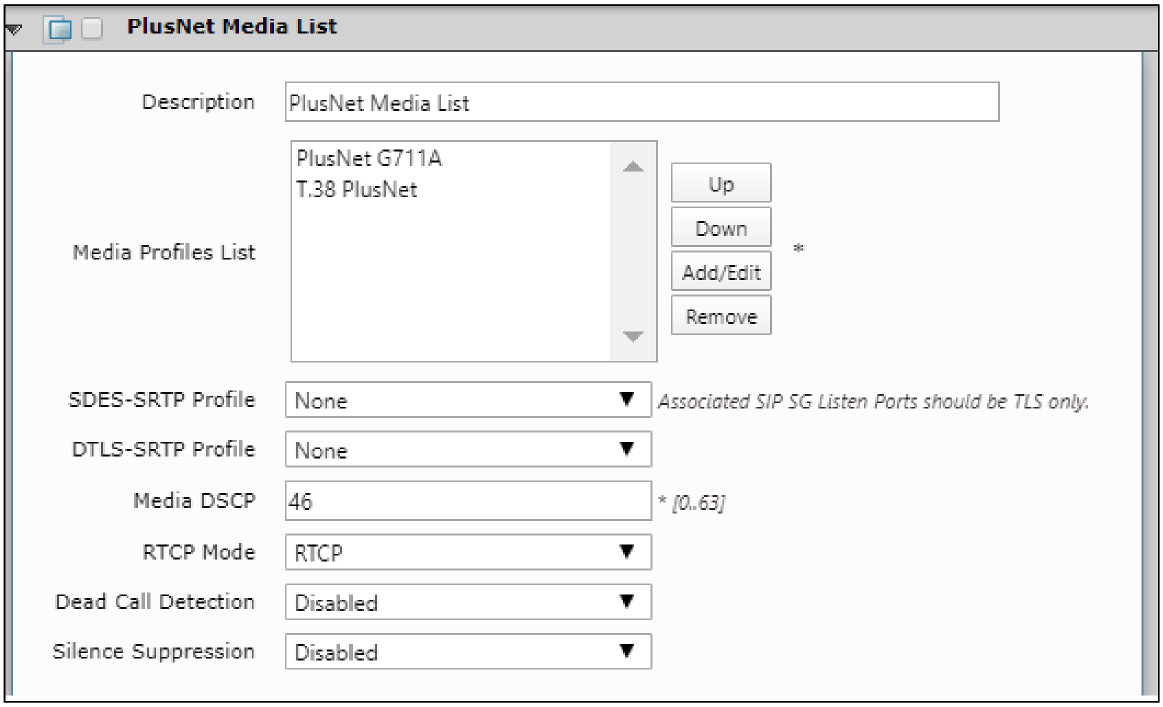

Fax Codec Configuration
Description
Tenor T.38
Codec
T.38 Fax
Maximum Rate
14400
b/s
Signaling Packet Redundancy
3
[0..7]
Payload Packet Redundancy
0
[0..3]
Error Correction Mode
Enabled
Training Confirmation Procedure
Send Over Network
Fallback to Passthrough
Enabled
Super G3 to G3 Fallback
Disabled

5. Media List

The Media List shows the selected voice and fax compression codecs and their associated settings.

Select **Settings > Media > Media List** to access the Media List configuration screen.

Figure 25: PlusNet Media List



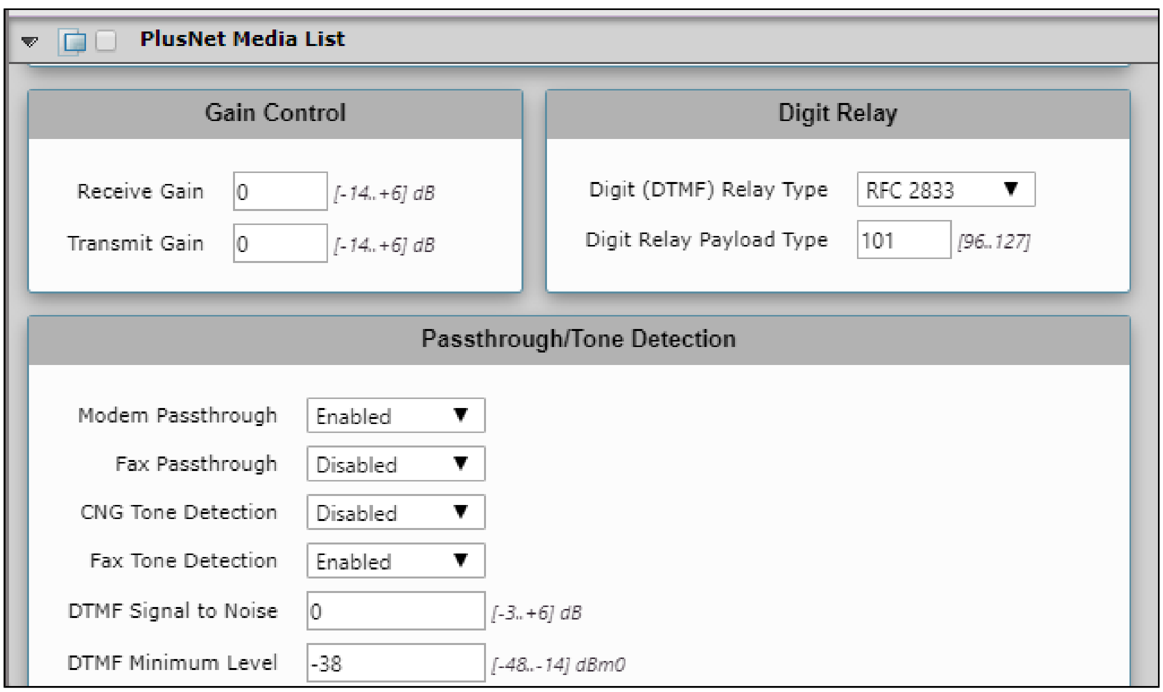


Figure 26: CUCM Media List

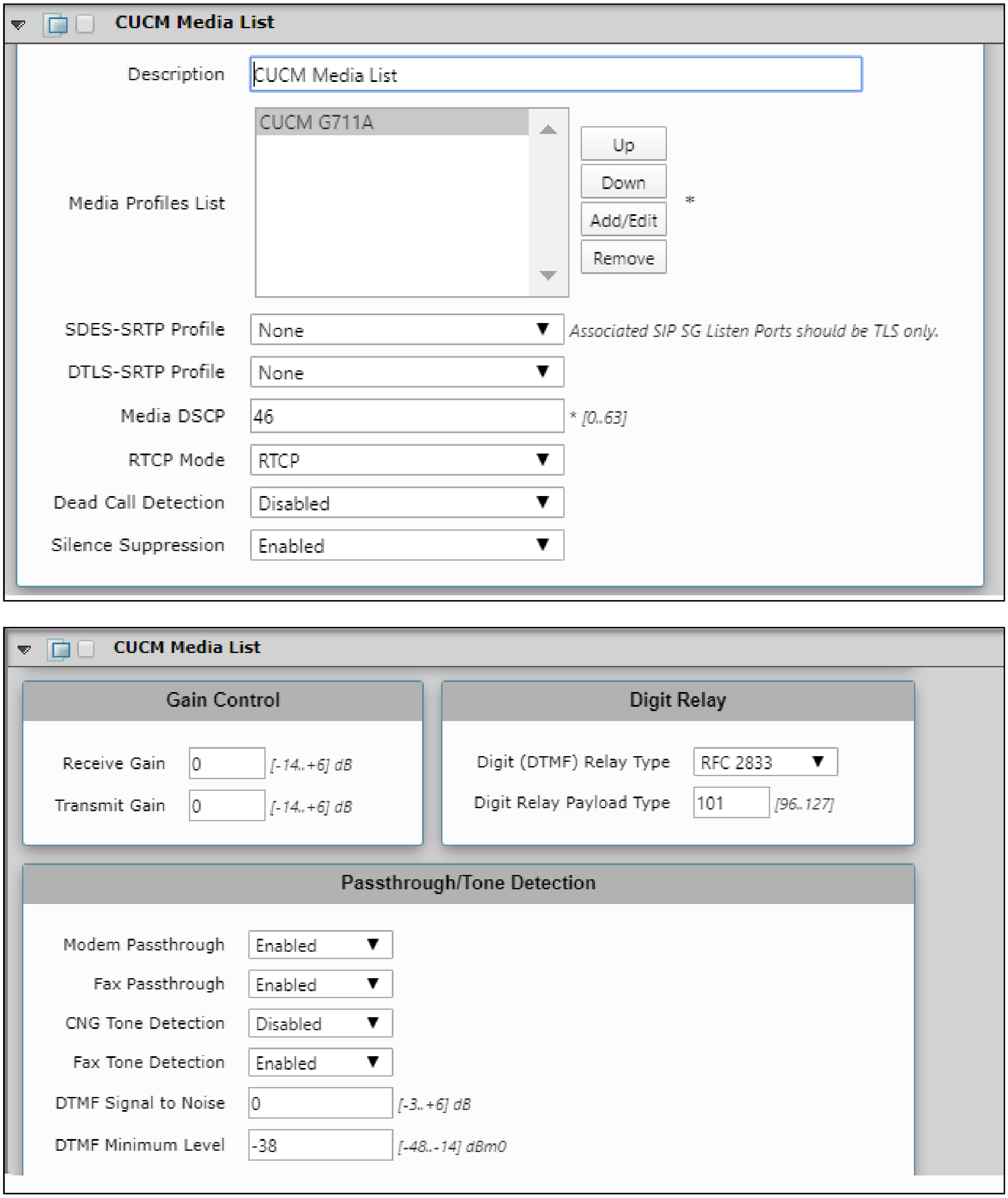
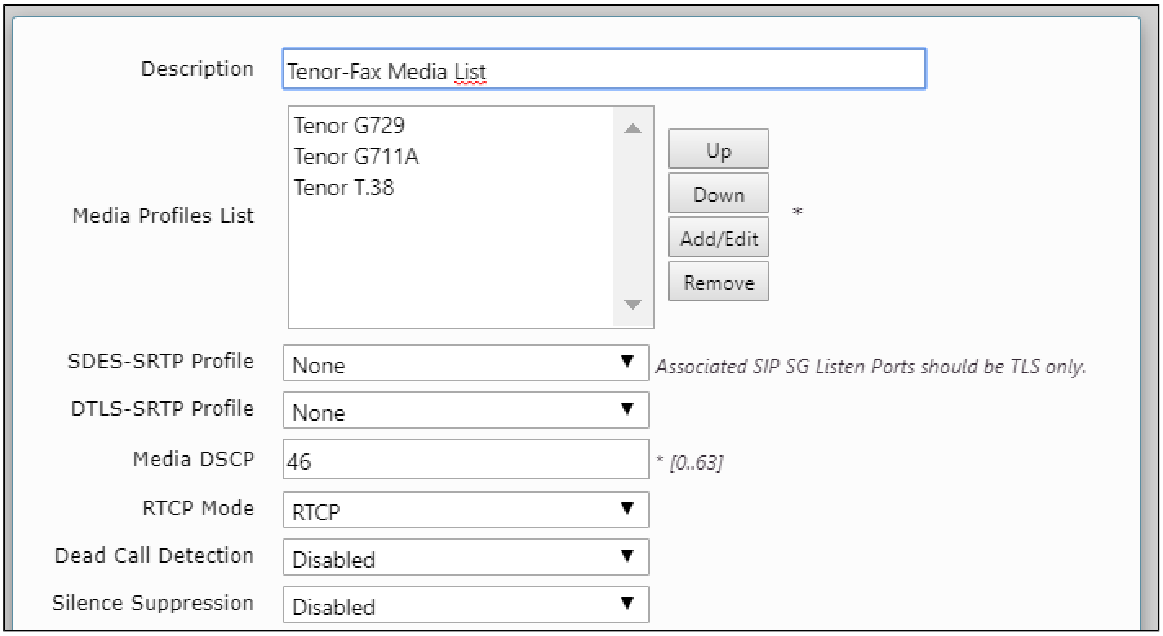
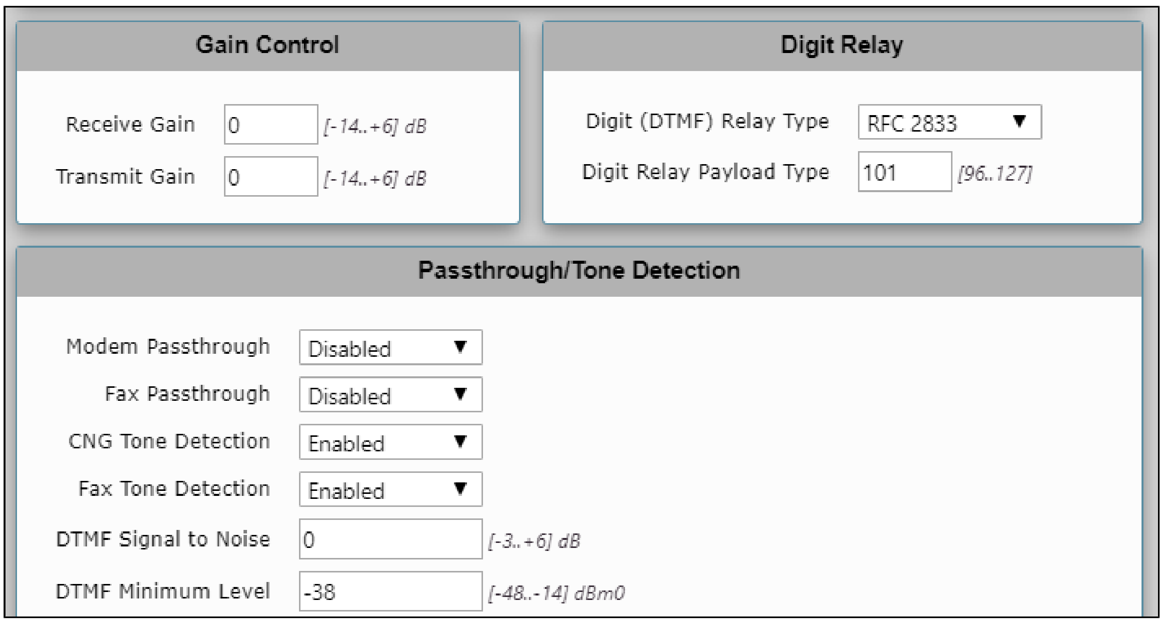


Figure 27: Fax Media List


Description
Tenor-Fax Media List
Media Profiles List
Tenor G729
Tenor G711A
Tenor T.38
Up
Down
Add/Edit
Remove
*
SDES-SRTP Profile
None
Associated SIP SG Listen Ports should be TLS only.
DTLS-SRTP Profile
None
Media DSCP
46
* [0..63]
RTCP Mode
RTCP
Dead Call Detection
Disabled
Silence Suppression
Disabled

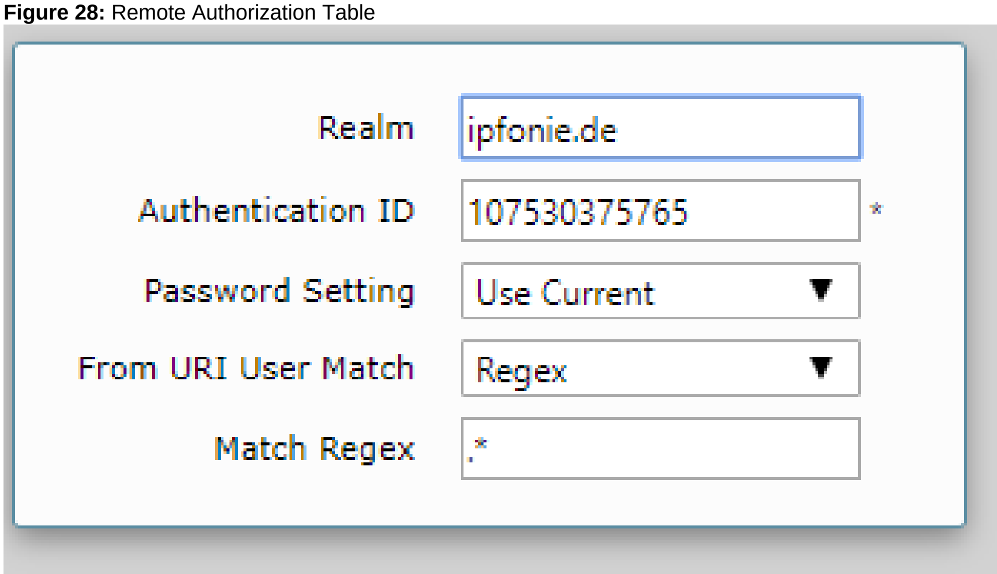

Gain Control
Receive Gain
0
[-14..+6] dB
Transmit Gain
0
[-14..+6] dB
Digit Relay
Digit (DTMF) Relay Type
RFC 2833
Digit Relay Payload Type
101
[96..127]
Passthrough/Tone Detection
Modem Passthrough
Disabled
Fax Passthrough
Disabled
CNG Tone Detection
Enabled
Fax Tone Detection
Enabled
DTMF Signal to Noise
0
[-3..+6] dB
DTMF Minimum Level
-38
[-48..-14] dBm0

6. Remote Authorization Tables

Remote Authorization Tables and their entries contain information used to respond to request message challenges by an upstream server. The Remote Authorization Tables on this page appear as options in Creating and Modifying Entries in the SIP Servers (For additional information about Remote Authorization Tables, see the Ribbon online SBC 1000/2000 documentation).

Select **Settings > SIP > Remote Authorization Tables** to access the Remote Authorization Tables configuration screen.

Figure 28: Remote Authorization Table

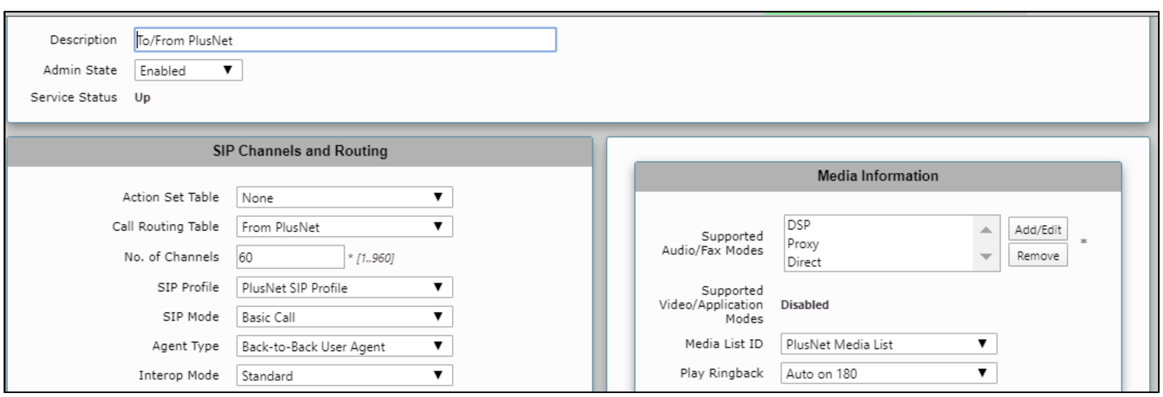


7. Signaling Groups

Signaling Groups allow telephony channels to be grouped for routing and shared configuration. These groups are the entity to which calls are routed, and the location from which Call Routes are selected. These are also the location from which Tone Tables and Action Sets are selected. In the case of SIP, Signaling Groups will specify protocol settings and links to server, media, and mapping tables.

Select **Settings > Signaling Groups** to access the Signaling Groups configuration screens.

Figure 29: PlusNet Signaling Group



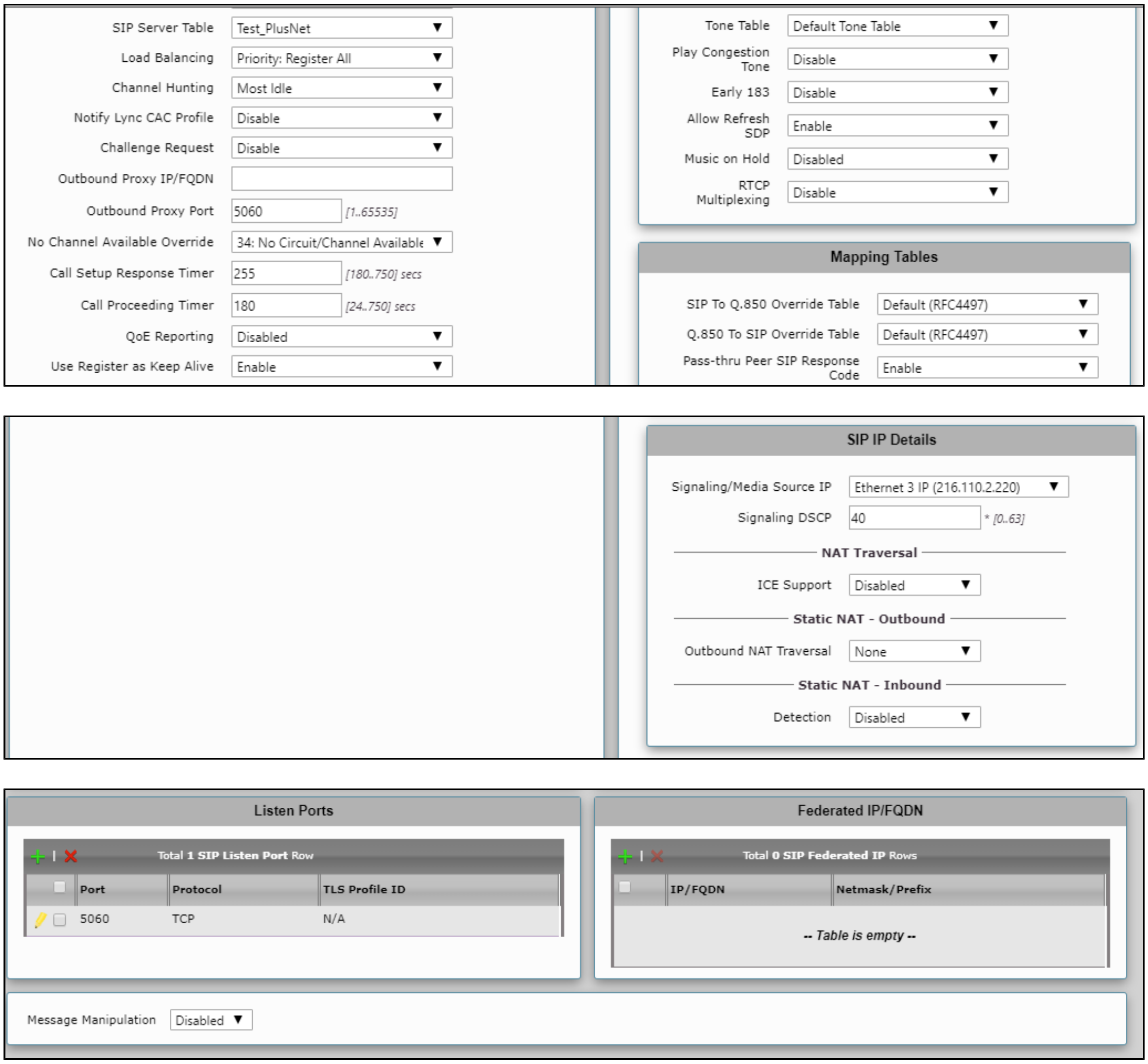
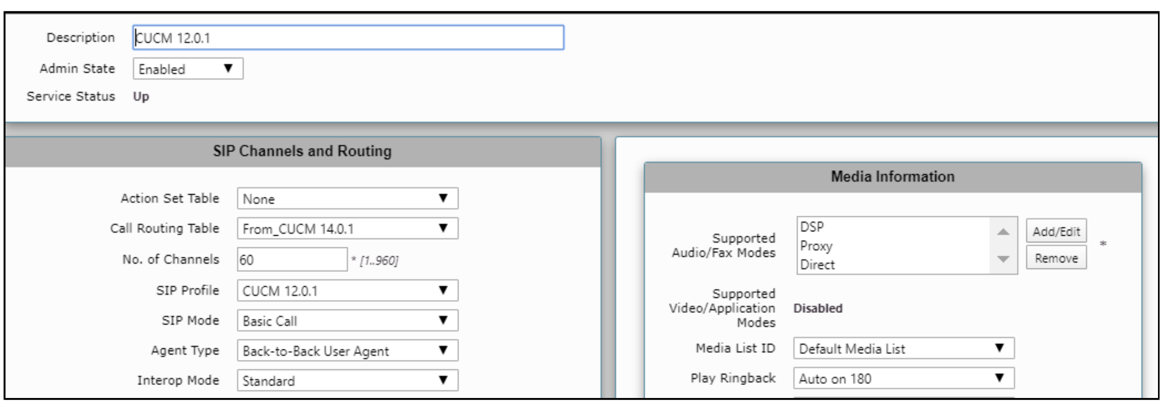


Figure 30: CUCM Signaling Group



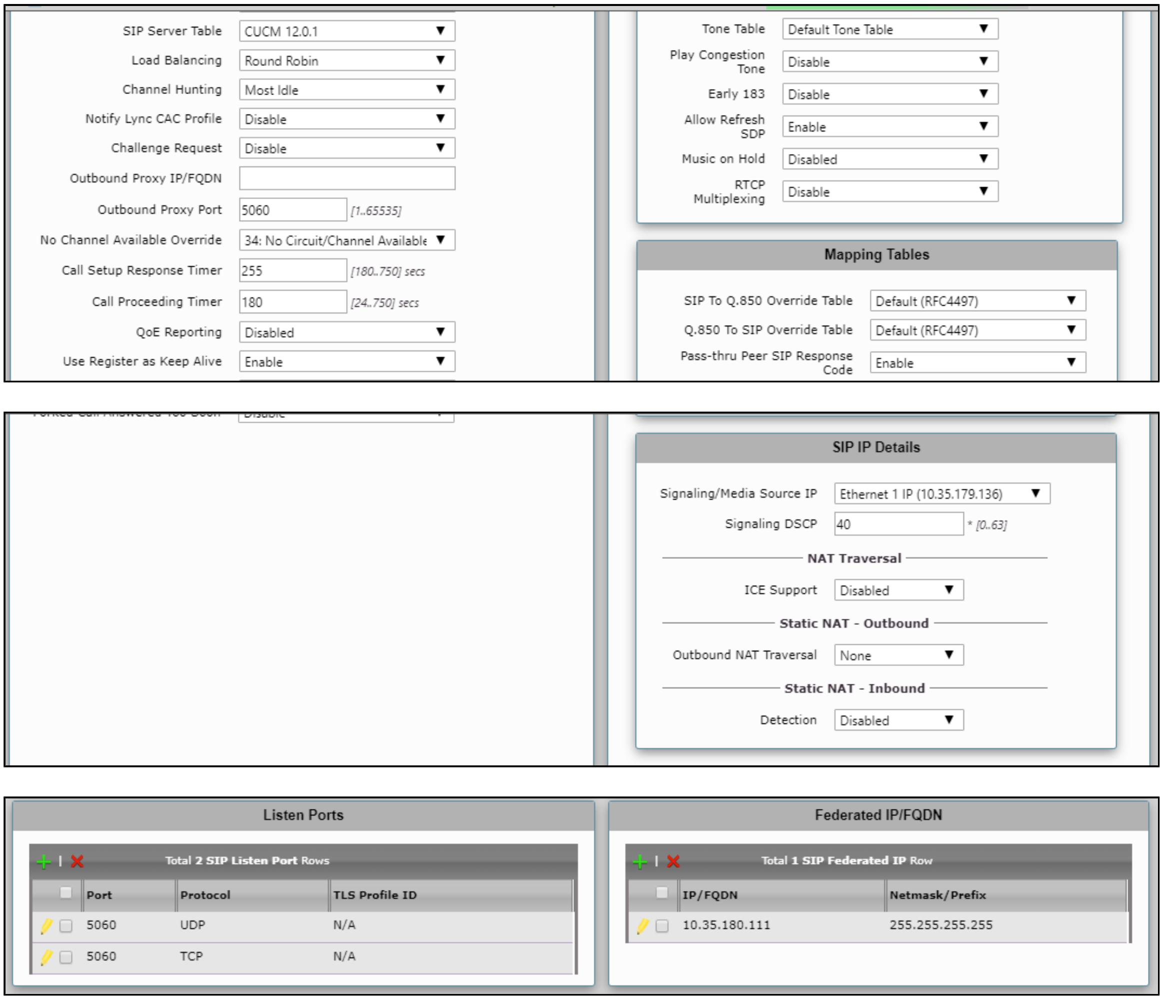
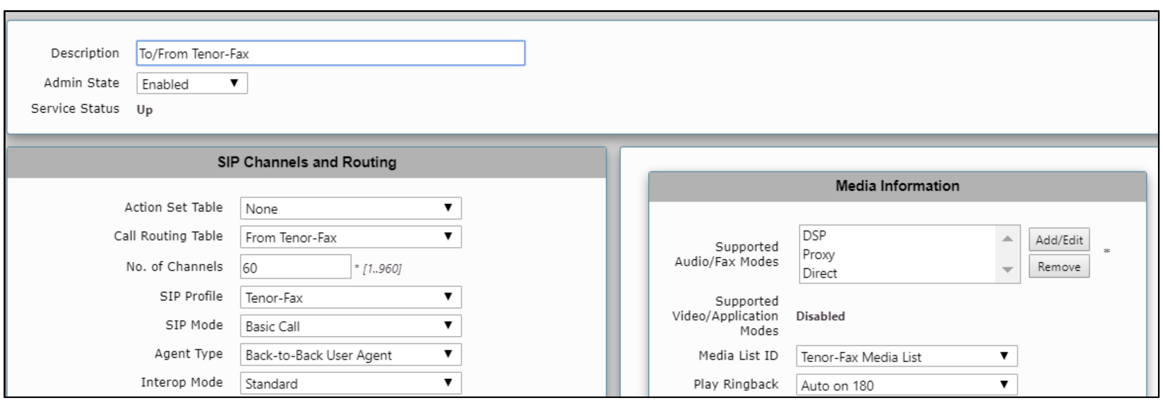
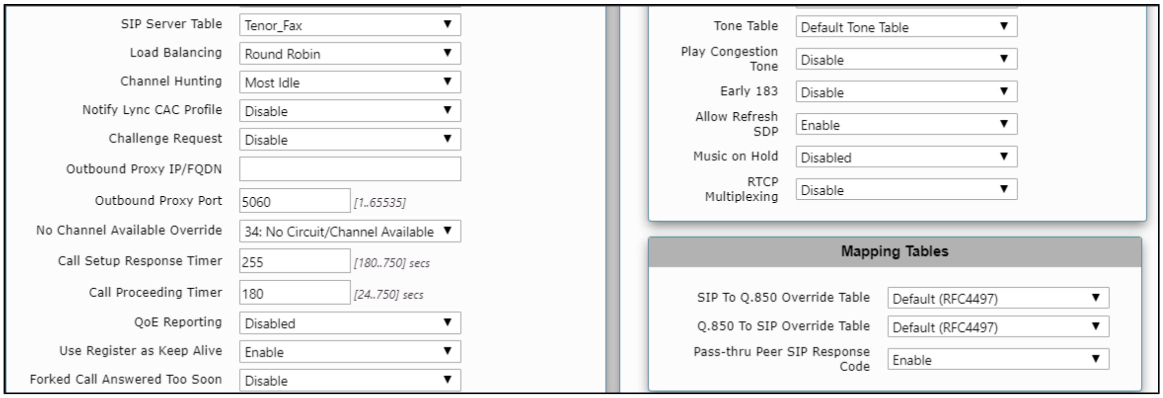
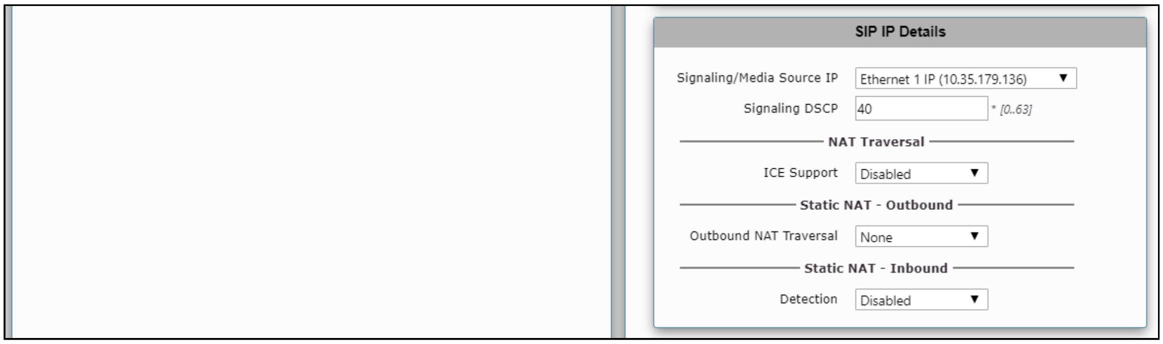
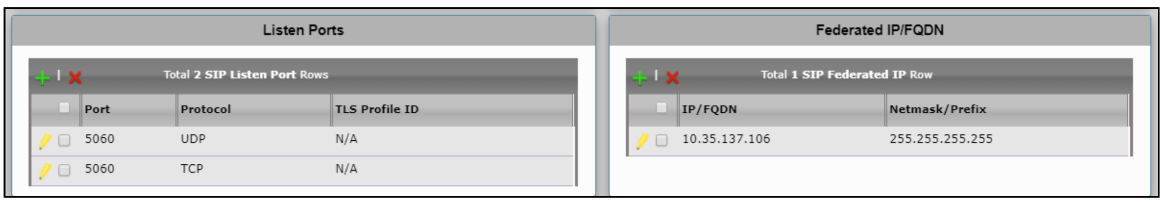


Figure 31: Fax Signaling Group







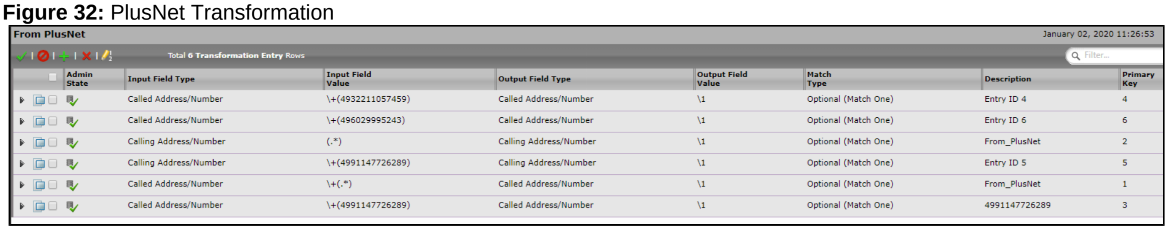


8. Transformation

Transformation Tables facilitate the conversion of names, numbers, and other fields when routing a call. For example, transformations can convert a public PSTN number into a private extension number, or a SIP address (URI). Every entry in a Call Routing Table requires a Transformation Table.

Select **Settings > Transformation** to access the Transformation configuration screen.

Figure 32: PlusNet Transformation

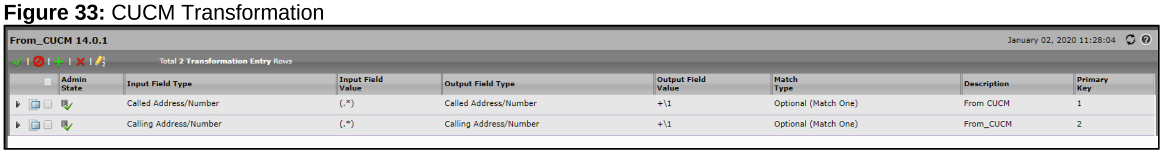


From PlusNet — January 02, 2020 11:26:53

Total 6 Transformation Entry Rows

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key
	Called Address/Number	\+(4932211057459)	Called Address/Number	\1	Optional (Match One)	Entry ID 4	4
	Called Address/Number	\+(496029995243)	Called Address/Number	\1	Optional (Match One)	Entry ID 6	6
	Calling Address/Number	(.*)	Calling Address/Number	\1	Optional (Match One)	From_PlusNet	2
	Calling Address/Number	\+(4991147726289)	Calling Address/Number	\1	Optional (Match One)	Entry ID 5	5
	Called Address/Number	\+(.*)	Called Address/Number	\1	Optional (Match One)	From_PlusNet	1
	Called Address/Number	\+(4991147726289)	Called Address/Number	\1	Optional (Match One)	4991147726289	3

Figure 33: CUCM Transformation

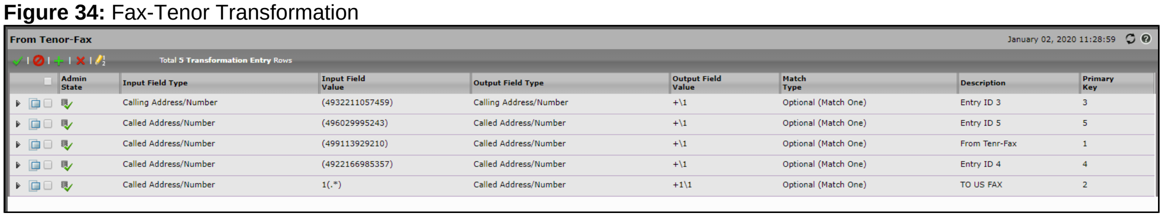


From_CUCM 14.0.1 — January 02, 2020 11:28:04

Total 2 Transformation Entry Rows

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key
	Called Address/Number	(.*)	Called Address/Number	+\1	Optional (Match One)	From CUCM	1
	Calling Address/Number	(.*)	Calling Address/Number	+\1	Optional (Match One)	From_CUCM	2

Figure 34: Fax-Tenor Transformation



From Tenor-Fax — January 02, 2020 11:28:59

Total 5 Transformation Entry Rows

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key
	Calling Address/Number	(4932211057459)	Calling Address/Number	+\1	Optional (Match One)	Entry ID 3	3
	Called Address/Number	(496029995243)	Called Address/Number	+\1	Optional (Match One)	Entry ID 5	5
	Called Address/Number	(499113929210)	Called Address/Number	+\1	Optional (Match One)	From Tenr-Fax	1
	Called Address/Number	(4922166985357)	Called Address/Number	+\1	Optional (Match One)	Entry ID 4	4
	Called Address/Number	1(.*)	Called Address/Number	+1\1	Optional (Match One)	TO US FAX	2

9. Call Routing Table

Call Routing allows calls to be carried between signaling groups, thus allowing calls to be carried between ports, and between protocols (like ISDN to SIP). Call Routing Tables define routes. The use of Call Routing Tables allows for flexible configuration of which calls will be carried, and also how the calls are translated. These tables are one of the central connection points of the system, linking Transformation Tables, Message translations, Cause Code Reroutes, Media Lists, and the three types of Signaling Groups (ISDN, SIP and, CAS).

Select **Settings > Call Routing Table** to access the Call Routing Table configuration screen.

Figure 35: PlusNet Call Routing

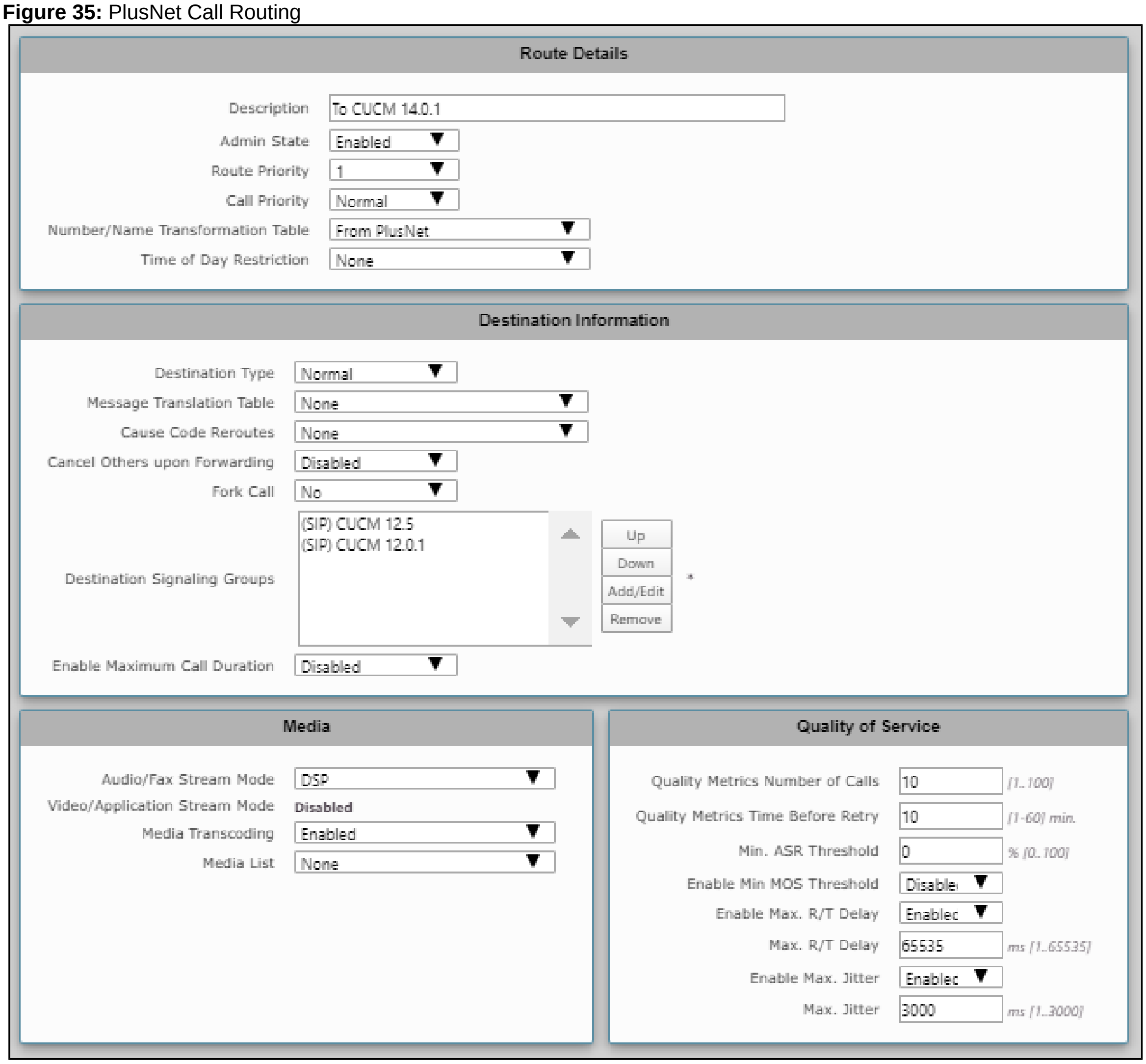


Figure 36: CUCM Call Routing

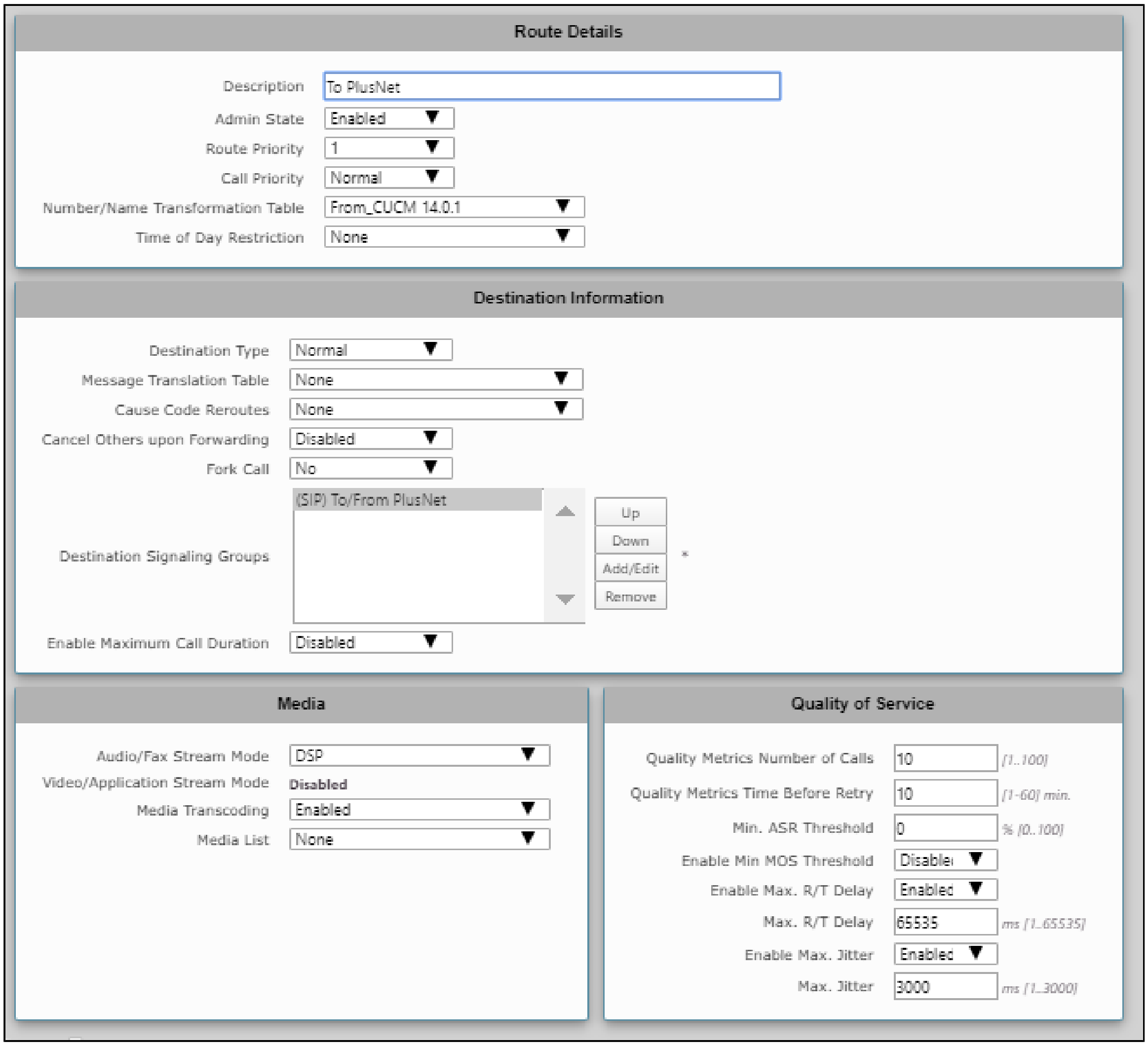
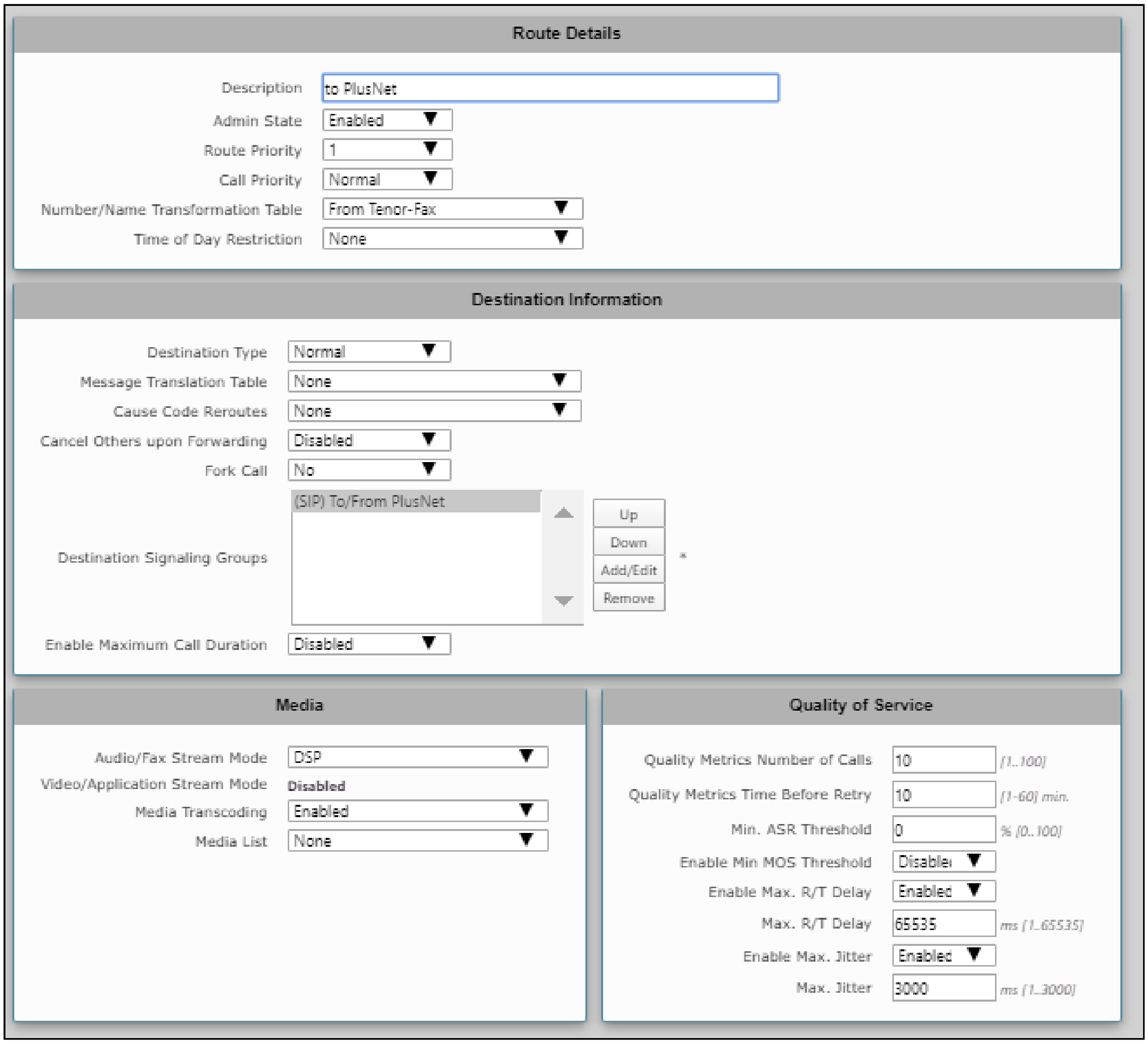


Figure 37: Fax Call Routing


Route Details
Description
to PlusNet
Admin State
Enabled
Route Priority
1
Call Priority
Normal
Number/Name Transformation Table
From Tenor-Fax
Time of Day Restriction
None
Destination Information
Destination Type
Normal
Message Translation Table
None
Cause Code Reroutes
None
Cancel Others upon Forwarding
Disabled
Fork Call
No
Destination Signaling Groups
(SIP) To/From PlusNet
Up
Down
Add/Edit
Remove
*
Enable Maximum Call Duration
Disabled
Media
Audio/Fax Stream Mode
DSP
Video/Application Stream Mode
Disabled
Media Transcoding
Enabled
Media List
None
Quality of Service
Quality Metrics Number of Calls
10
[1..100]
Quality Metrics Time Before Retry
10
[1-60] min.
Min. ASR Threshold
0
% [0..100]
Enable Min MOS Threshold
Disable
Enable Max. R/T Delay
Enabled
Max. R/T Delay
65535
ms [1..65535]
Enable Max. Jitter
Enabled
Max. Jitter
3000
ms [1..3000]

Interoperability Test Results

The following table provides test results for interoperability compliance testing between Ribbon SBC 1000/2000 and CUCM

Table 2: Interoperability Compliance Test Results

Test Number	Test Scenario	Setup / Test Results	Status	Comment
4.1	Registration and authentication (registration mode)	The PBX is able to execute the correct resolution oft he DNS SRV record	Pass	
4.3	Basic call	With the basic call tests, the standard call scenarios and the CLIP/CLIR features are tested. • Only en-bloc dialing is supported, overlap sending is not possible.	Pass	
4.3.1	Normal call	Outgoing call from PBX to PSTN • En-bloc dialling • Local area call (without area code); area code must be set by the PBX • Setting of the correct calling number with all available telephone number blocks • If two or more locations with different area codes are assigned to one SIP trunk, the number in the PAI header has to be the geographic number which belongs to the users location. This is very important in case of emergency calls, because Plusnet uses the number of the PAI header to route the emergency call to the proper emergency call center.	Pass	
4.3.1.1	Normal call	The geographic number in the PAI header corresponds to the location 11 of the user	Pass	Is it possible to configure several PAI per SIP trunk? Yes
4.3.1.2	Normal call	Display of A-number in B-party CLIP (national PSTN)	Pass	
4.3.1.3	Normal call	Display of A-number in B-party CLIP (international PSTN)	Pass	
4.3.1.4	Normal call	Display of A-number in B-party CLIP (mobile)	Pass	
4.3.1.5	Normal call	Call to mobile Outgoing call to mobile => mobile phone turned off	Pass	
4.3.1.6	Normal call	Suppression of A-number => CUR	Pass	
4.3.1.7	Normal call	Outgoing call from analog extension	Pass	
4.3.1.8	Normal call	Outgoing call (> 5 min.) => PSTN	Pass	Hold connection for 5 minutes => RTP still correct? Yes
4.3.2	Normal call	Incoming call from PSTN (national) => PBX • Test all available telephone number blocks	Pass	
4.3.2.1	Normal call	Display of A-number => CLIP	Pass	
4.3.2.2	Normal call	Incoming call from mobile => PBX Display of A-number => CLIP	Pass	
4.3.2.3	Normal call	Suppression of A-number => CLIR	Pass	
4.3.3	Normal call	Two simultaneous outgoing/incoming calls	Pass	
4.3.4	Normal call	Enabled feature DND (do not disturb)	Pass	

4.3.5	Normal call	Test call with codec G.711	Pass	
4.3.6	Normal call	Test call with codec G.722 (only SIP <=> SIP)	Fail	PlusNet didn´t support G. 722
4.3.7	Normal call	Test call with codec G.729	Pass	
4.3.2.8	Clip No Screening	Outgoing call from PBX to PSTN • With feature Clip No Screening • Test with several different A-numbers • If two or more locations with different area codes are assigned to one SIP trunk, the number in the PAI header has to be the geographic number which belongs to the users location. This is very important in case of emergency calls, because PlusNet uses the number	Pass	
4.3.2.8.1	Clip No Screening	Despite Clip No Screening the geographic number in the PAI header corresponds to the location of the user	Pass	Is it possible to configure several PAI per SIP trunk? Yes
4.3.2.8.2	Clip No Screening	Display of A-number (NoSClip) at B-party (PSTN)	Pass	
4.3.2.8.3	Clip No Screening	Display of A-number (NoSClip) at B-party (international PSTN; depending on the destination carrier, the NoSClip telephone number may not be displayed in this case!)	Pass	
4.3.2.8.3	Clip No Screening	Call made from a PSTN line to an IP-PBX line with call forward to a line within the same IP-PBX, Answer Call. • Either party terminates call.	Pass	Does the IP-PBX has configuration settings to send SIP status 181 messages to the soft switch? Yes
4.3.2.8.4	Clip No Screening	Display of A-number (NoSClip) at B-party (mobile)	Pass	
4.3.3.9	Special call situations	Outgoing call PBX => PSTN • Call is rejected by B-party	Pass	
4.3.3.10	Special call situations	Outgoing call PBX=> PSTN • B-party does not answer; clearing after timer	Pass	
4.3.3.11	Special call situations	Outgoing call PBX => PSTN • B-party busy; busy tone	Pass	
4.3.3.12	Special call situations	Outgoing call PBX=> PSTN • A-party hangs up before call is established (cancel)	Pass	
4.3.3.13	Special call situations	Incoming call PSTN => PBX • Call is rejected by PBX party	Pass	
4.3.3.14	Special call situations	Incoming call PSTN => PBX • PBX party does not answer; clearing after timer	Pass	
4.3.3.15	Special call situations	Incoming call PSTN => PBX • PBX party busy; busy tone	Pass	

4.3.3.16	Special call situations	Incoming call PSTN => PBX • A-party hangs up before call is established (cancel)	Pass	
4.3.4.17	Call clearing	Incoming / outgoing call; clearing after established call. Correct clearing on both sides • PBX party hangs up • PSTN party hangs up	Pass	
4.3.4.18	Call clearing	Interrupting the network connection of the SIP terminal device during a call • Call should be cleared correctly	Pass	
4.4.19	Hold	PBX => PSTN and PSTN => PBX • Test call in both directions	Pass	
4.4.19.1	Hold	Putting an external call on hold in the PBX	Pass	
4.4.19.2	Hold	If applicable, MoH (music on hold) at A-party (PSTN)	Pass	
4.4.19.3	Hold	HOLD RETRIEVE: retrieving the external call	Pass	
4.4.19.4	Hold	Clearing the connection of the A-party while it is put on hold	Pass	
4.4.20	Hold	PBX => PSTN and PSTN => PBX • Test call in both directions	Pass	
4.4.20.1	Hold	Putting an external call on hold in the PSTN	Pass	
4.4.20.2	Hold	If applicable, MoH (music on hold) at A-party (PBX)	Pass	
4.4.20.3	Hold	HOLD RETRIEVE: retrieving the external call	Pass	
4.4.20.4	Hold	Clearing the connection of the A-party while it is put on hold	Pass	
4.5.21	Call transfer	Internal call is transferred to external party: internal => PBX => external	Pass	
4.5.21.1	Call transfer	Call transfer from PBX party => PSTN party with announcement (attendant transfer)	Pass	
4.5.21.2	Call transfer	Call transfer from PBX party => PSTN party without announcement (blind transfer)	Pass	
4.5.21.3	Call transfer	Call transfer from PBX party => PSTN party without announcement (blind transfer) • Call is rejected or not answered	Pass	
4.5.21.4	Call transfer	Call transfer from PBX party => PSTN party without announcement (blind transfer) • PSTN party busy	Pass	
4.5.22	Call transfer	Transferred call from external party to PBX: PSTN => PBX	Pass	
4.5.22.1	Call transfer	Call transfer from PSTN => PBX party with announcement (attendant transfer)	Pass	
4.5.22.2	Call transfer	Call transfer from PSTN => PBX party without announcement (blind transfer)	Pass	
4.5.22.3	Call transfer	Call transfer from PSTN => PBX party without announcement (blind transfer) • Call is rejected or not answered	Pass	
4.5.22.4	Call transfer	Call transfer from PSTN => PBX party without announcement (blind transfer) • PBX party busy	Pass	
4.5.23	Call transfer	Call from external party transferred to another external party: external => PBX => external	Pass	
4.5.23.1	Call transfer	PSTN => PBX party => PSTN with announcement (attendant transfer)	Pass	
4.5.23.2	Call transfer	PSTN => PBX party => PSTN without announcement (blind transfer)	Pass	

4.5.23.3	Call transfer	PSTN => PBX party => PSTN without announcement (blind transfer) • Call is rejected or not answered	Pass	
4.5.23.4	Call transfer	PSTN => PBX party => PSTN without announcement (blind transfer) • PSTN party busy	Pass	
4.6.24	Call diversion	PBX party CFU to external party (PSTN)	Pass	
4.6.24.1	Call diversion	Internal call (CFU) => PSTN	Pass	
4.6.24.2	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.24.3	Call diversion	A-party clears in ring phase	Pass	
4.6.24.4	Call diversion	External => PBX party (CFU) => PSTN	Pass	
4.6.24.5	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.24.6	Call diversion	A-party clears in ring phase	Pass	
4.6.25	Call diversion	PBX party CFNR to external party	Pass	
4.6.25.1	Call diversion	Internal call (CFNR) => PSTN	Pass	
4.6.25.2	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.25.3	Call diversion	A-party clears in ring phase	Pass	
4.6.25.4	Call diversion	External => PBX party (CFNR) => PSTN	Pass	
4.6.25.5	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.25.6	Call diversion	A-party clears in ring phase	Pass	
4.6.26	Call diversion	PBX party CFB to external party	Pass	
4.6.26.1	Call diversion	Internal call (CFB) => PSTN	Pass	
4.6.26.2	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.26.3	Call diversion	A-party clears in ring phase	Pass	
4.6.26.4	Call diversion	External => PBX party (CFB) => PSTN	Pass	
4.6.26.5	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.26.6	Call diversion	A-party clears in ring phase	Pass	
4.6.27	Call diversion	External party (PSTN, mobile, etc.) CFU to PBX party:	Pass	
4.6.27.1	Call diversion	External (CFU) => PBX party	Pass	

4.6.27.2	Call diversion	PBX party busy, rejects call, does not answer	Pass	
4.6.27.3	Call diversion	A-party clears in ring phase	Pass	
4.6.27.4	Call diversion	External (CFNR) => PBX party	Pass	
4.6.27.5	Call diversion	PBX party busy, rejects call, does not answer	Pass	
4.6.27.6	Call diversion	A-party clears in ring phase	Pass	
4.6.28	Call diversion	External party (PSTN, mobile, etc.) CFB to PBX party:	Pass	
4.6.28.1	Call diversion	PBX party busy, rejects call, does not answer	Pass	
4.6.28.2	Call diversion	A-party clears in ring phase	Pass	
4.6.29	Call diversion	Call deflection: diversion during the ring phase	Pass	
4.6.29.1.1	Call diversion	Internal call PBX party CD => PBX party	Pass	
4.6.29.1.2	Call diversion	PBX party busy	Pass	
4.6.29.1.3	Call diversion	PBX party does not answer	Pass	
4.6.29.1.4	Call diversion	A-party clears in ring phase	Pass	
4.6.29.2.1	Call diversion	External call to PBX party CD => PBX party	Pass	
4.6.29.2.2	Call diversion	PBX party busy	Pass	
4.6.29.2.3	Call diversion	PBX party does not answer	Pass	
4.6.29.2.4	Call diversion	A-party clears in ring phase	Pass	
4.6.29.3.1	Call diversion	External call to PBX party CD => external PSTN party	Pass	
4.6.29.3.2	Call diversion	PSTN party busy	Pass	
4.6.29.3.3	Call diversion	PSTN party does not answer	Pass	
4.6.29.3.4	Call diversion	A-party clears in ring phase	Pass	
4.7.30	Call waiting	Incoming call during active internal call	Pass	
4.7.30.1	Call waiting	Call waiting tone	Pass	
4.7.30.2	Call waiting	Display of number of waiting party	Pass	
4.7.30.3	Call waiting	Acceptance of waiting call	Pass	
4.7.30.4	Call waiting	Putting the waiting call on hold	Pass	
4.7.30.5	Call waiting	Retrieve of waiting call	Pass	
4.7.30.6	Call waiting	Holding the 2nd call	Pass	
4.7.30.7	Call waiting	Terminating the active call	Pass	
4.7.30.8	Call waiting	Disregarding the waiting call	Pass	

4.7.30.9	Call waiting	Rejecting the waiting call	Pass	
4.8.31	3-party conference	Establishing a conference according to the operating instructions of the PBX Internal - internal - external	Pass	
4.8.31.1	3-party conference	Selecting a party (internal or external); 3rd party is put on hold • Switching to 3rd party; 2nd party is put on hold • Reactivating the conference • Clearing one party (internal or external) • Terminating the conference	Pass	
4.8.32	3-party conference	Establishing a conference according to the operating instructions of the PBX internal - external - external	Pass	
4.8.32.1	3-party conference	Selecting a party (external); 3rd party is put on hold • Switching to 3rd party; 2nd party is put on hold • Reactivating the conference • Clearing one party (external) • Terminating the conference	Pass	
4.9.33	Pick up	Picking up a call from another extension of the PBX	Pass	
4.10.34	Call list	Entries in the call list • Incoming from PSTN • Incoming from mobile • Incoming CLIR • Dialing prefix for outside line in the call list	Pass	
4.10.35	Call list	Call-back from call list • To PSTN • To mobile	Pass	
4.11.36	DTMF	DTMF support (G.711) • PSTN => PBX (SIP terminal device) • PSTN => PBX (analog or system terminal device) • PBX (SIP terminal device) => PSTN • PBX (analog or system terminal device) => PSTN	Pass	
4.11.37	DTMF	DTMF support (G.729) - PSTN => PBX (SIP terminal device) - PSTN => PBX (analog or system terminal device) - PBX (SIP terminal device) => PSTN - PBX (analog or system terminal device) => PSTN	Pass	
4.12.38	Fax	Fax reception (G.711 only)	Pass	
4.12.38.1	Fax	Network-side T.38 re-invite rejected by PBX (response 488) or only G.711 codec is confirmed • One-page fax • Multi-page fax (at least 5 pages)	Pass	
4.12.39	Fax	Fax sending (G.711 only)	Pass	
4.12.39.1	Fax	Network-side T.38 re-invite rejected by PBX (response 488) or only G.711 codec is confirmed • One-page fax • Multi-page fax (at least 5 pages)	Pass	
4.12.40	Fax	Fax reception via T.38	Pass	
4.12.40.1	Fax	Re-invite to T.38 by PBX or network - One-page fax - Multi-page fax (at least 5 pages)	Pass	
4.12.41	Fax	Fax sending via T.38 (not possible in conjunction with the encryption option)	Pass	

4.12.41.1	Fax	Re-invite to T.38 by PBX or network • T.38-only invites are not supported • One-page fax • Multi-page fax (at least 5 pages)	Pass	
5.42	Redundancy	Test of redundancy: • Only if redundant connection is possible on the PBX side • This requires at least two PBX servers to be online on the SIP trunk in registration mode, or exactly two PBX servers in peering mode.	Pass	
5.42.1	Redundancy	1. Register all available PBX systems the Plusnet SBC. 2. Calls from Plusnet => PBX are routed by round robin procedure 3. Deliberately de-register one PBX system => no more calls are routed to this PBX 4. and/or disconnect the PBX system from LAN => after the register expire period, no more calls are routed to this PBX. 5. Calls are only routed to the remaining PBX systems.	Pass	

Conclusion

This Application Notes document describes the steps required to configure the Ribbon SBC 1000/2000 to successfully interoperate with the Cisco CUCM and PlusNet SIP Trunk. All feature and serviceability test cases have been completed. The majority of test cases passed with noted exceptions and observations provided in Interoperability Test Results.

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Overview

This Application Note is a configuration guide for the Ribbon SBC (Session Border Controller) 1000/2000 when connecting to Cisco Unified Communication Manager (CUCM) and PlusNet SIP Trunk.

The configuration guide supports features outlined in the Microsoft Technet web page:

- For additional information on Cisco Platform, visit http://www.cisco.com.
- For additional information on Ribbon SBC 1000/2000, visit https://ribboncommunications.com/.

Introduction

Interoperability compliance testing focuses on verifying inbound and outbound call flows between Ribbon SBC 1000/2000 and Cisco CUCM.

Audience

This technical document provides telecommunications engineers with information for configuring both the Ribbon SBC and the third-party product. Procedures in this document require navigating third-party equipment as well as applying Ribbon SBC Command Line Interface (CLI) commands. To complete the configuration and perform any troubleshooting, the engineer performing the procedures must understand the basic concepts of TCP /UDP, IP/Routing, and SIP/RTP.

> This Application Note is offered as a convenience to Ribbon customers. The specifications and information regarding the product in this document are subject to change without notice. All statements, information, and recommendations contained in this document are believed to be accurate but are presented without warranty of any kind, express or implied, and are provided "AS IS". Users must take full responsibility for the application of the specifications and information contained here.

> The links are only internal to Ribbon partners and employees. They do not work outside of the Ribbon Network.

Requirements

The following table lists the hardware and software used in the reference configuration.

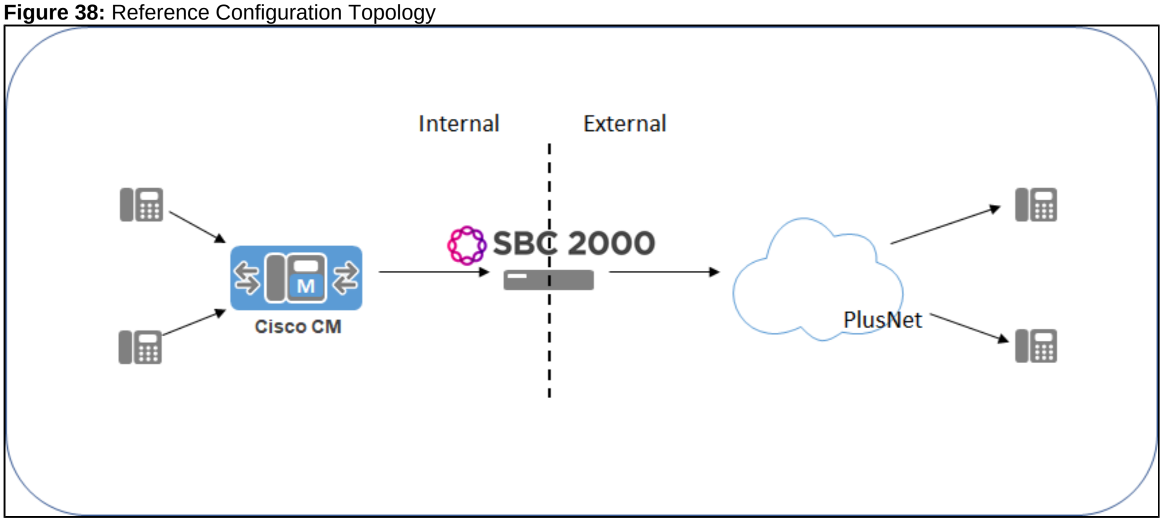
Table 3: Test Equipment and Software

Vendor	Equipment	Software Version
Ribbon Networks	SBC 2000	V8.0.2
Third-party Vendor		
Cisco	Cisco Unified CM Administration	12.0.1.21900
Cisco	Cisco SIP Phone 7841	sip78xx.11-7-1-17
VentaFax	Fax Machine VentaFax	7.6.243.616

Reference Configuration

The following figure serves as a topology for the reference configuration. The figure shows the connectivity between third-party equipment and the Ribbon SBC 1000/2000.

Figure 38: Reference Configuration Topology



Support

For questions about information in this document, contact Ribbon Support in either of the following ways:

- Global Support Assistance Center +1-978-614-8589 or +1-888-391-3434 (English language Support)
- Web: https://ribboncommunications.com/services/ribbon-support-portal-login

Verify License

The interoperability test described in this document requires no special licensing.

CUCM 12.0.1 Configuration

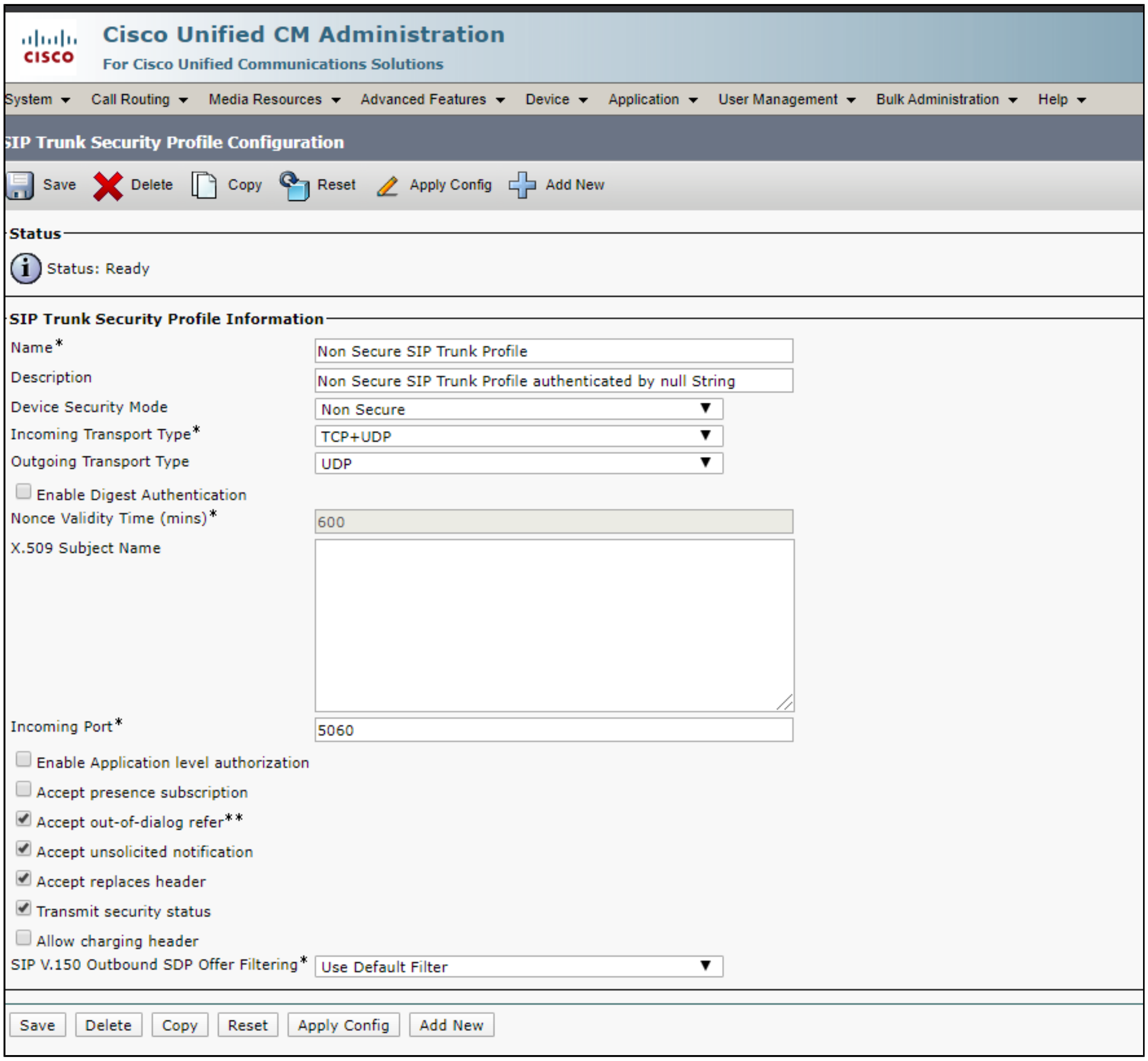
The following new configurations are included in this section:

1. Security Profile
2. SIP Profile
3. SIP Trunk
4. Route Group
5. Route List
6. Route Pattern

1. Security Profile

Select **System > Security > SIP Trunk Security Profile**

Figure 39: SIP Trunk Security Profile


Cisco Unified CM Administration
For Cisco Unified Communications Solutions
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
SIP Trunk Security Profile Configuration
Save Delete Copy Reset Apply Config Add New
Status
Status: Ready
SIP Trunk Security Profile Information
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
Nonce Validity Time (mins)*
600
X.509 Subject Name
Incoming Port*
5060
Enable Application level authorization
Accept presence subscription
Accept out-of-dialog refer**
Accept unsolicited notification
Accept replaces header
Transmit security status
Allow charging header
SIP V.150 Outbound SDP Offer Filtering*
Use Default Filter
Save Delete Copy Reset Apply Config Add New

2. SIP Profile

Select **Device > Device Settings > SIP Profile**

Figure 40: SIP Profile

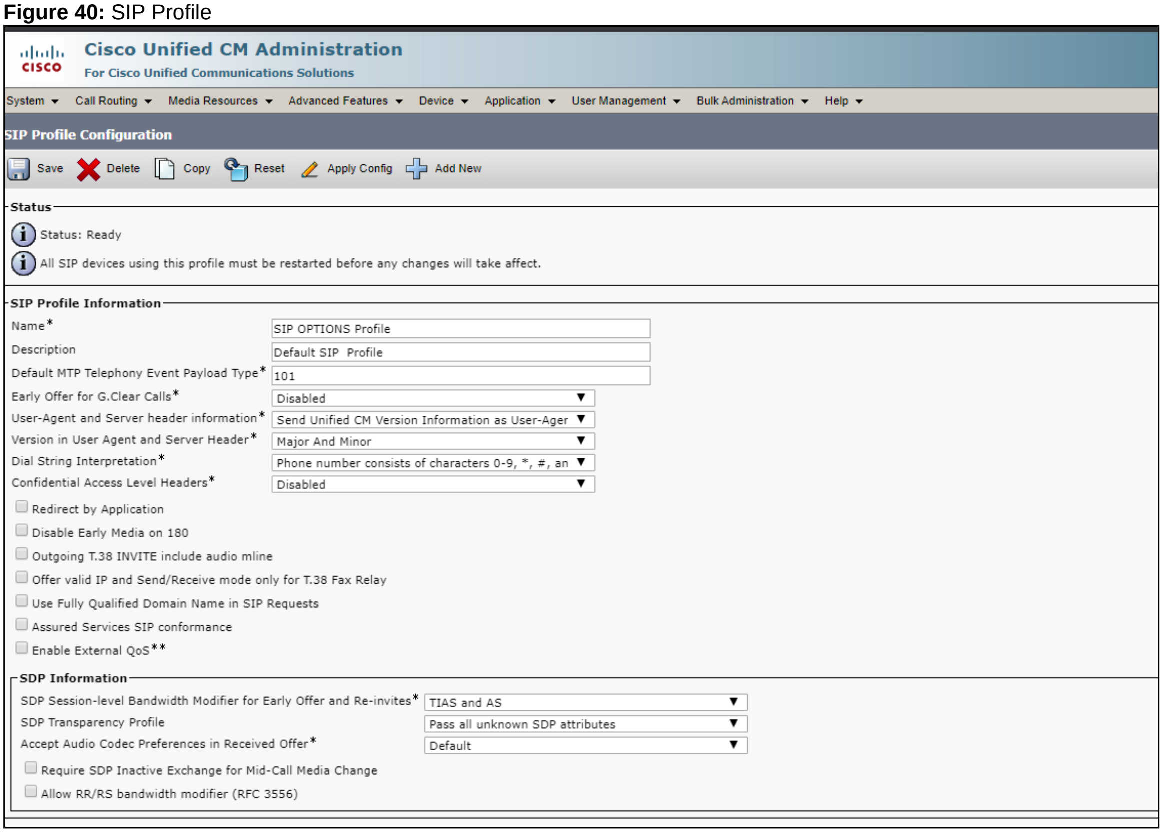


Figure 41: SIP Profile1

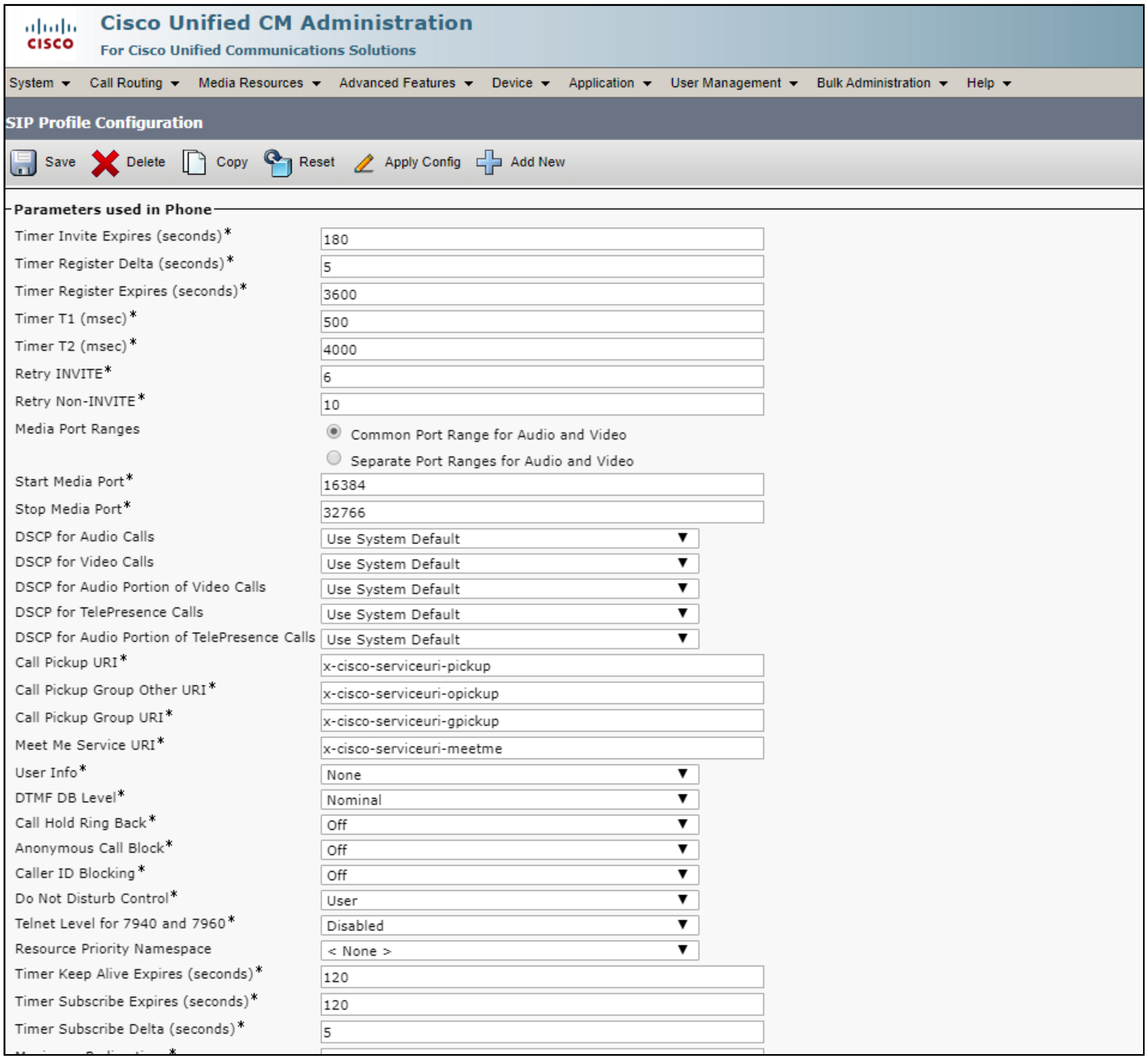


Figure 42: SIP Profile2

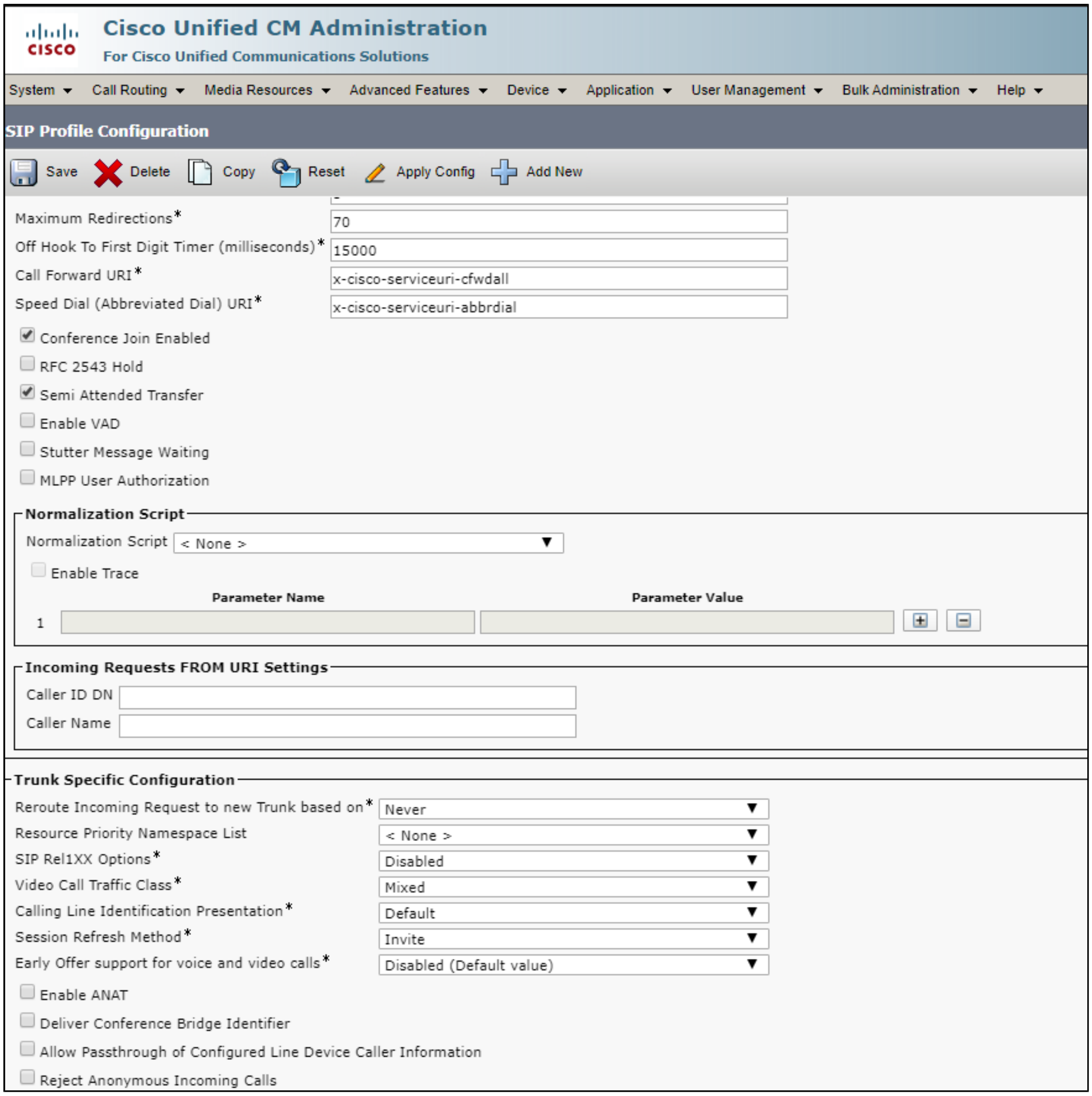
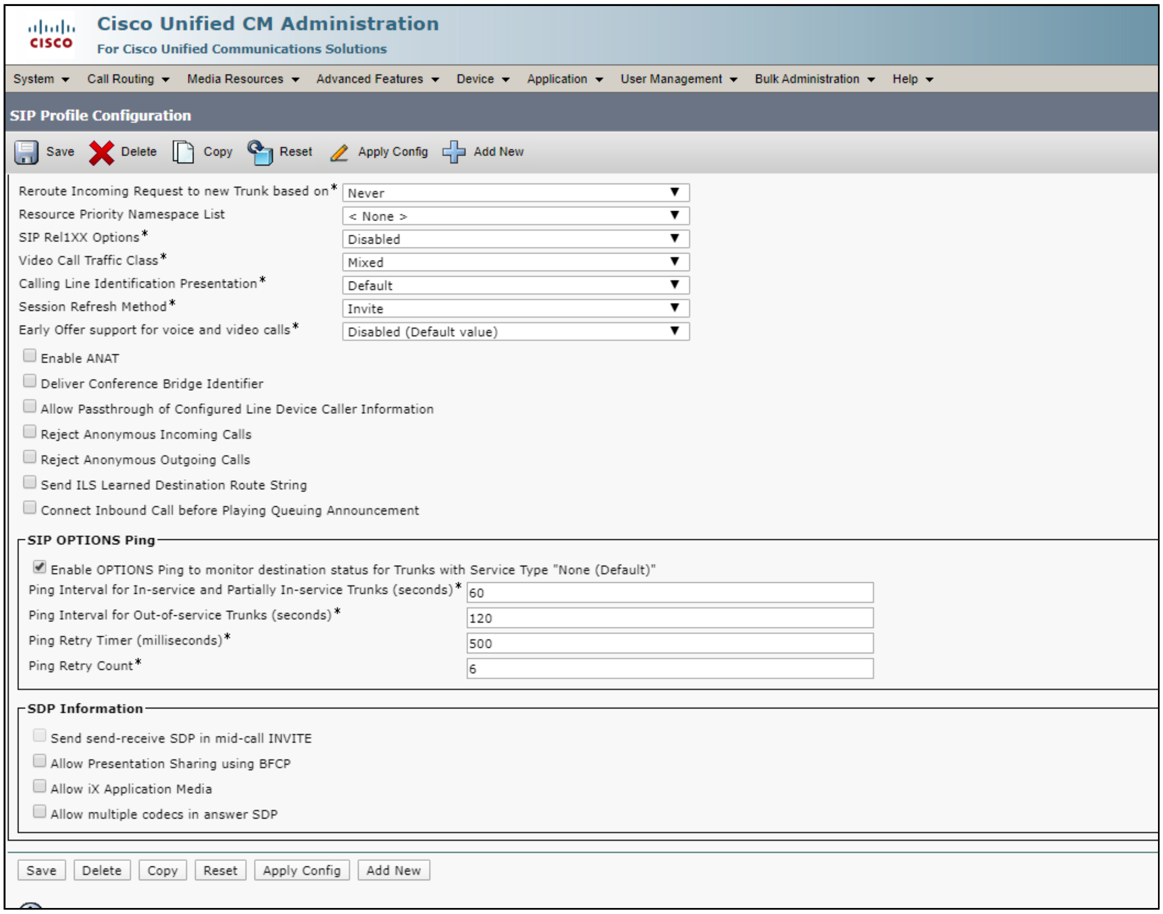


Figure 43: SIP Profile3



3. SIP Trunk

Select **Device > Trunk > Add New**

Figure 44: SIP Trunk

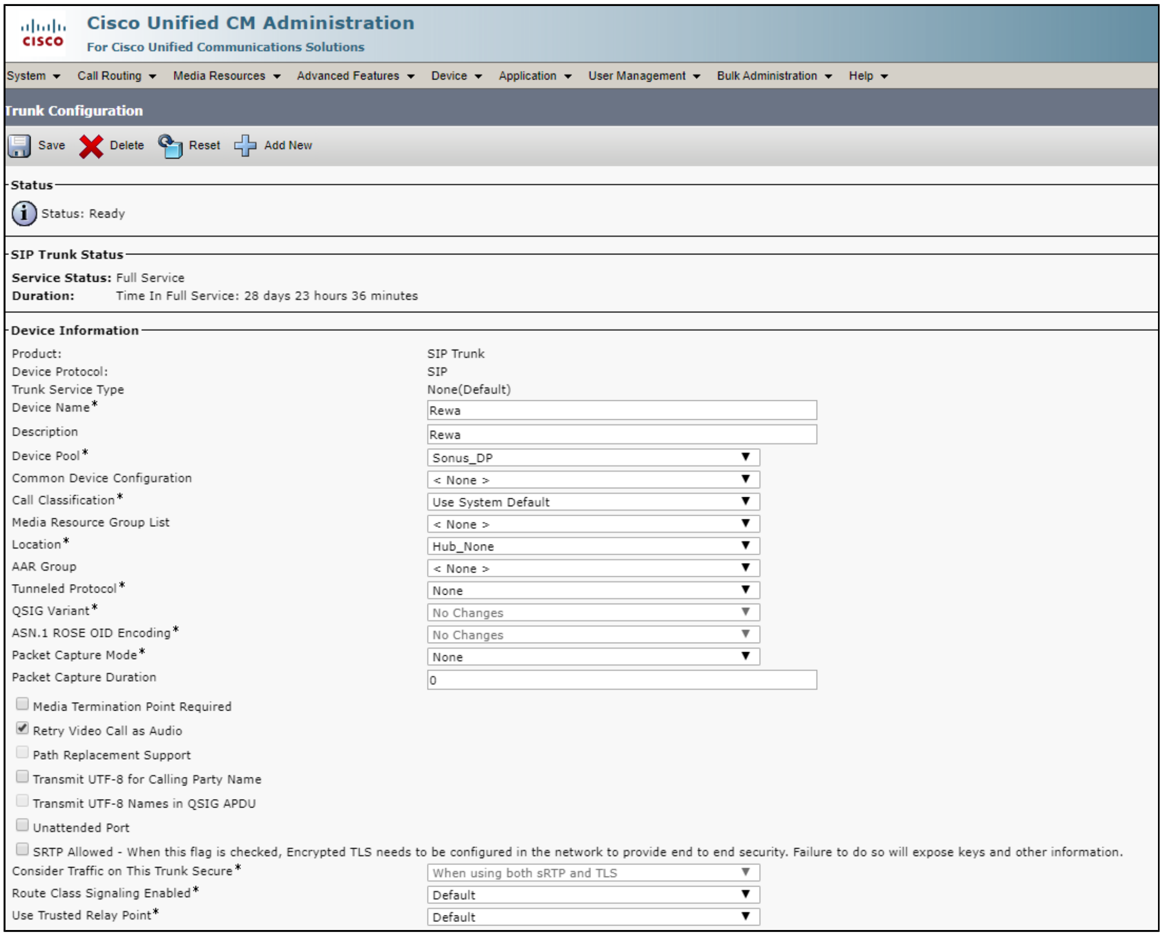


Figure 45: SIP Trunk1

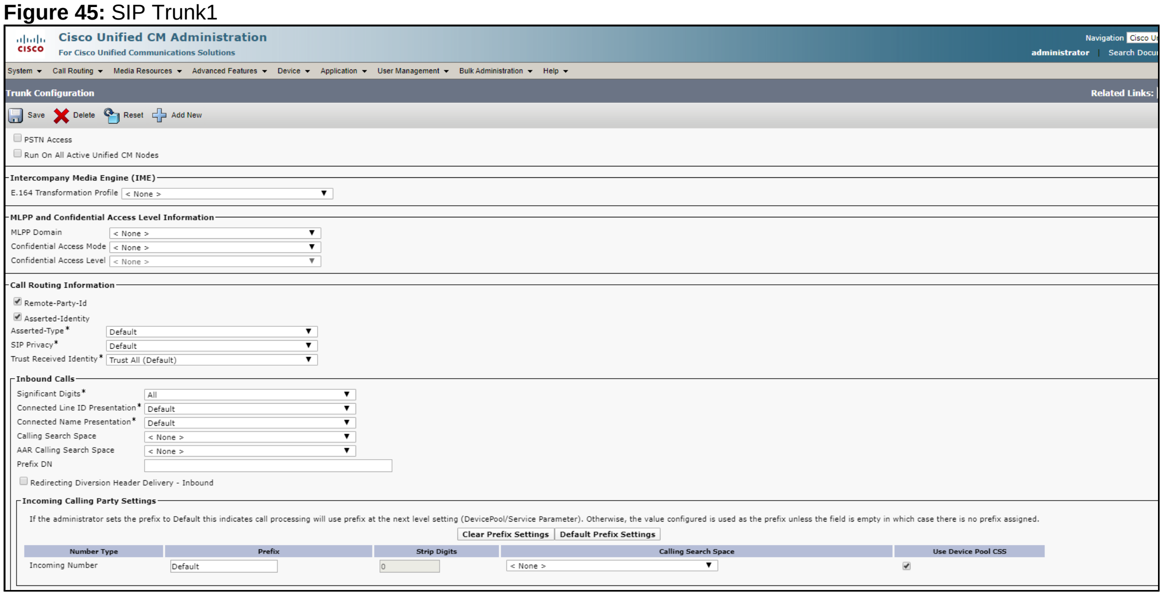


Figure 46: SIP Trunk2

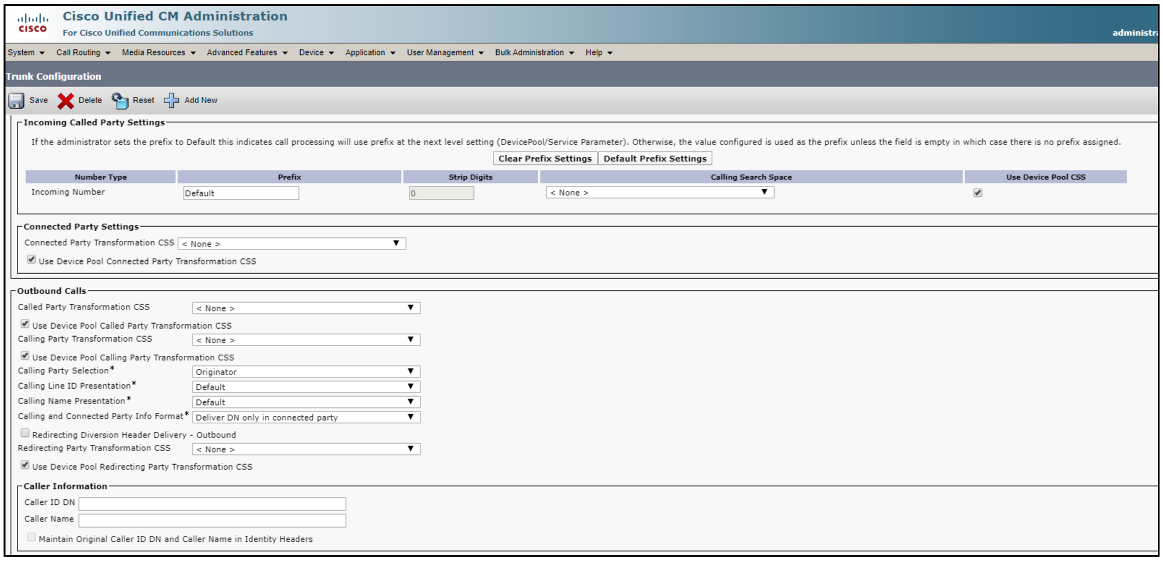
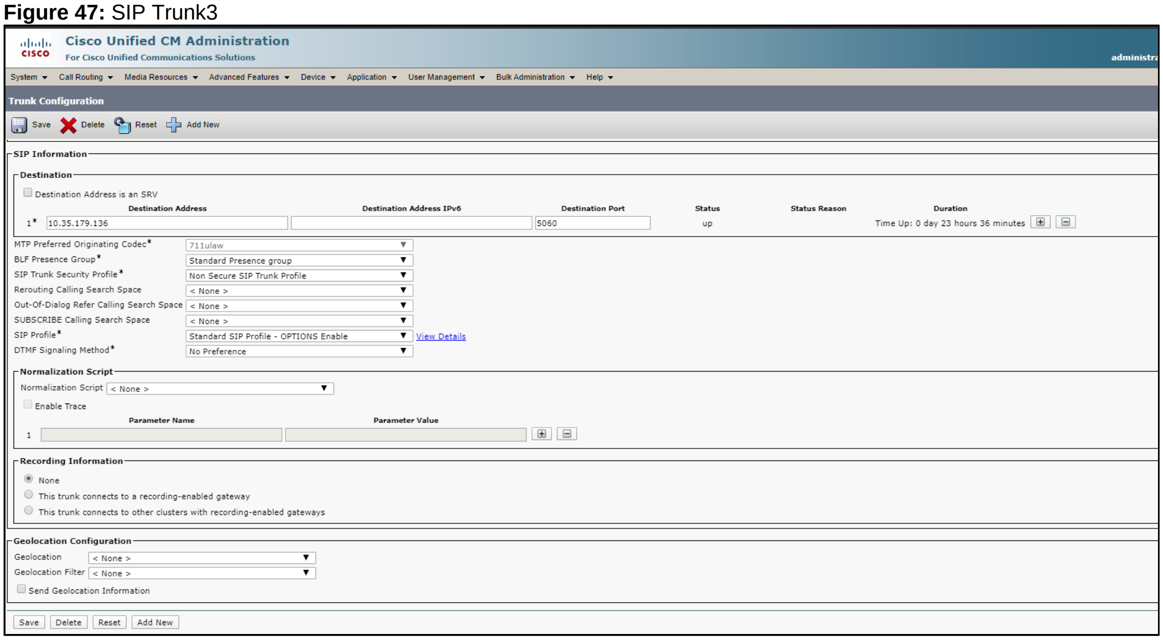


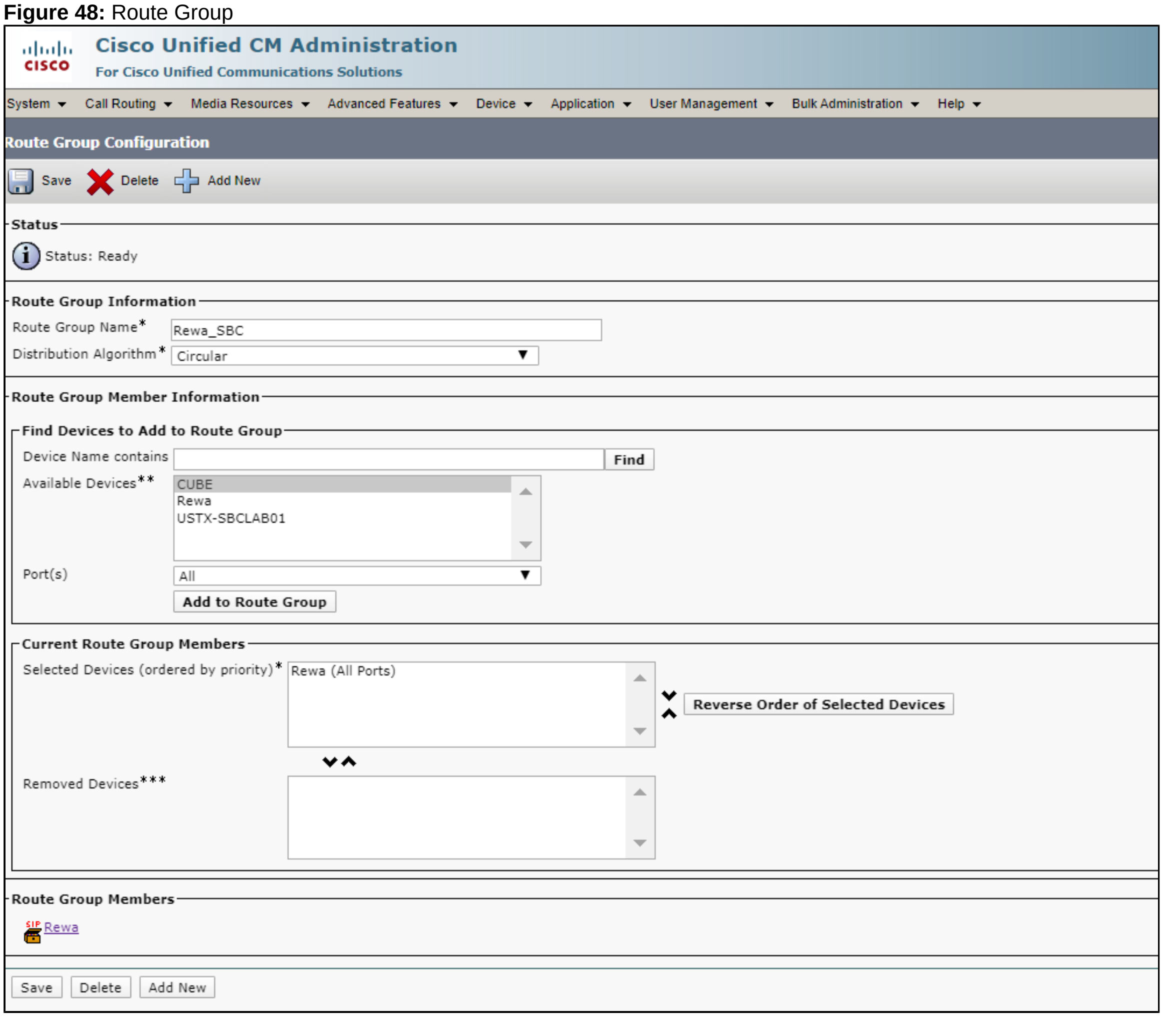
Figure 47: SIP Trunk3



4. Route Group

Select **Call Routing > Route/Hunt > Route Group > Add New**

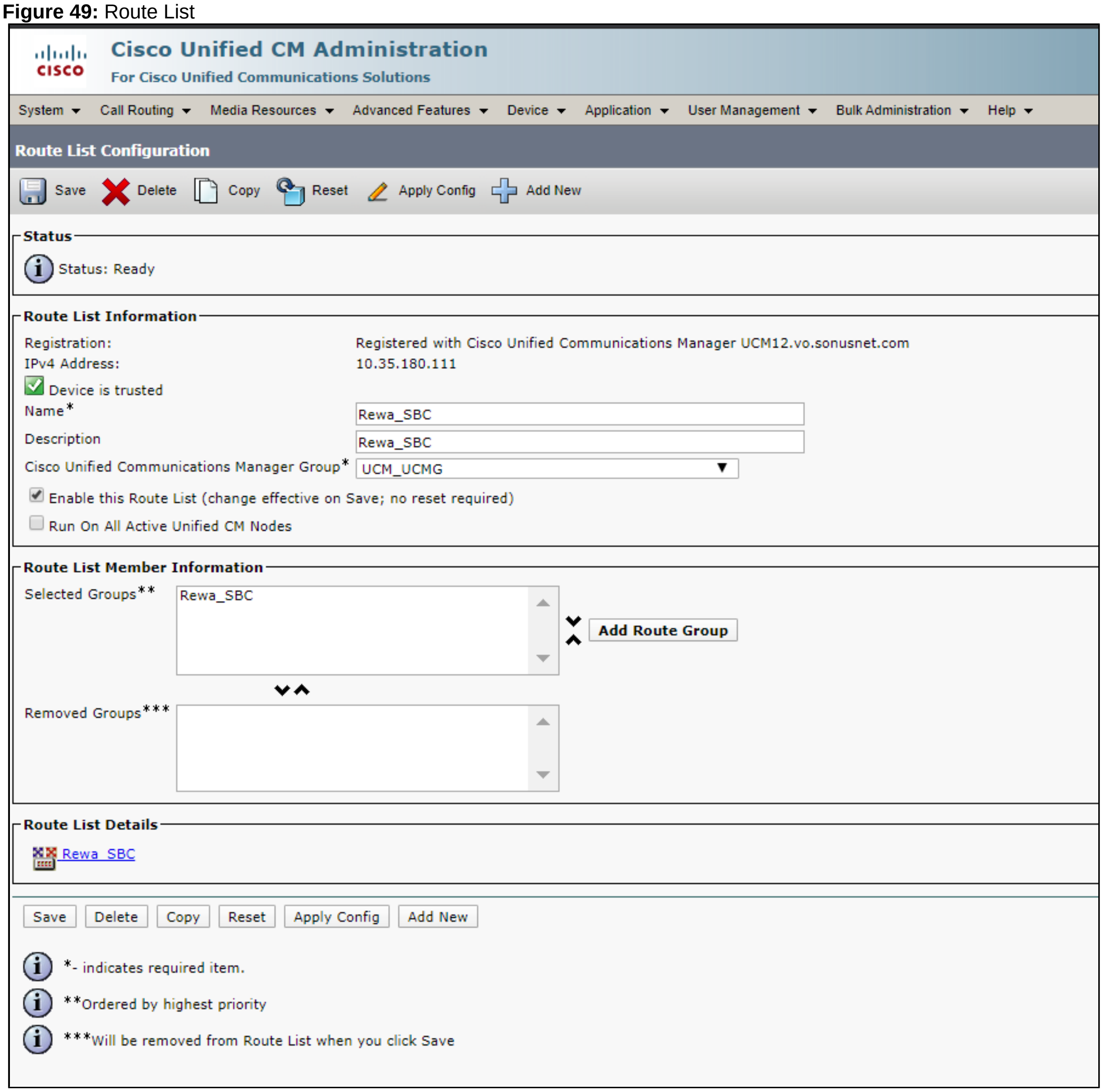
Figure 48: Route Group



5. Route List

Select **Call Routing > Route/Hunt > Route List > Add New**

Figure 49: Route List



6. Route Pattern

Select **Call Routing > Route/Hunt > Route Pattern > Add New**

Figure 50: Route Pattern

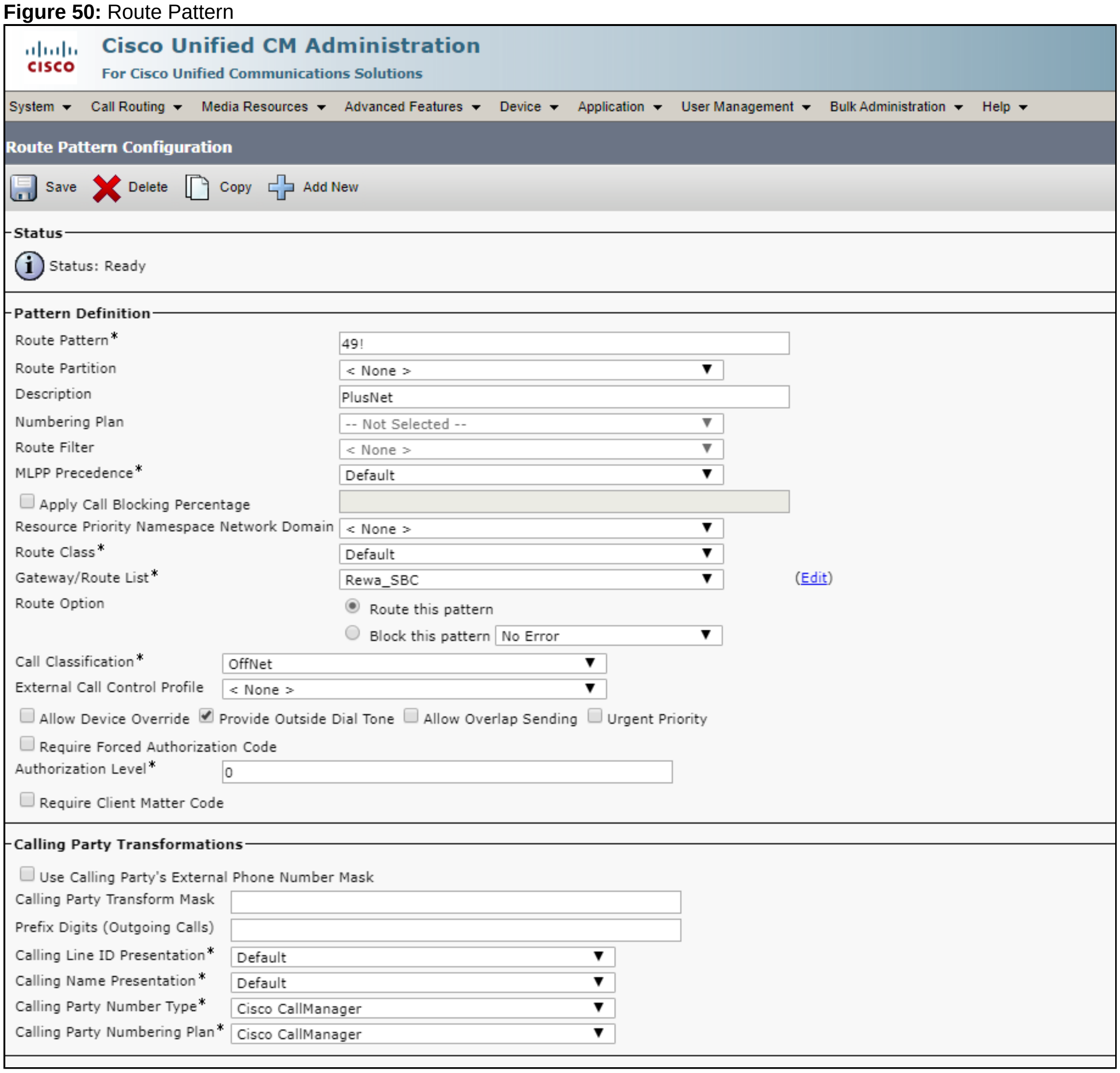
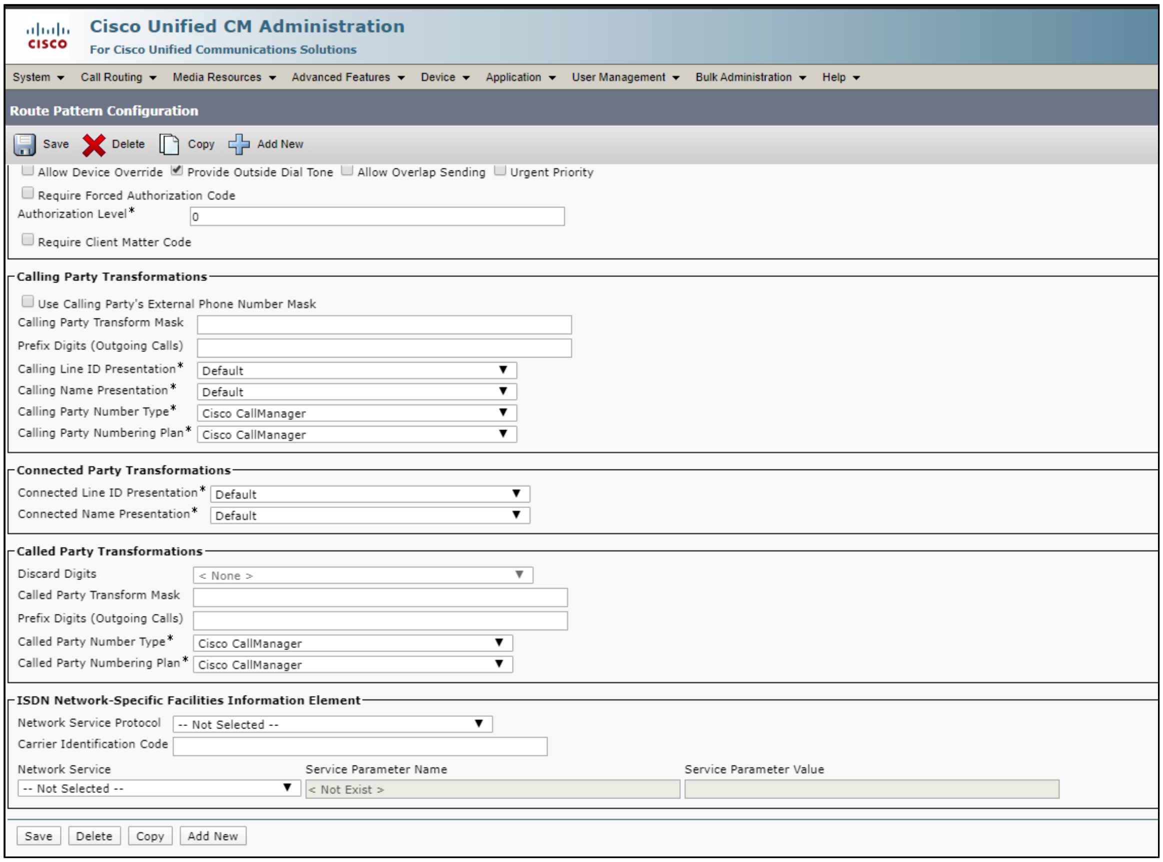


Figure 51: Route Pattern1



Ribbon SBC 1000/2000 Configuration

The following configuration steps provide an example of how to configure the Ribbon SBC 1000/2000 to interoperate with Skype 2015 and Virgin Media SIP Trunk:

1. SIP Profile
2. SIP Server
3. Media System
4. Media Profiles
5. Media List
6. Remote Authorization Tables
7. Signaling Groups
8. Transformation
9. Call Routing Table

1. SIP Profile

SIP Profiles control how the Ribbon SBC 1000/2000 communicates with SIP devices. The profiles control important characteristics such as session timers, SIP Header customization, SIP timers, MIME payloads, and option tags.

Select **Settings > SIP > SIP Profiles** to access the SIP Profile screen.

The following figures show the default SIP profile used for the Ribbon 1000/2000 used for this configuration effort.

Figure 52: PlusNet SIP Profile

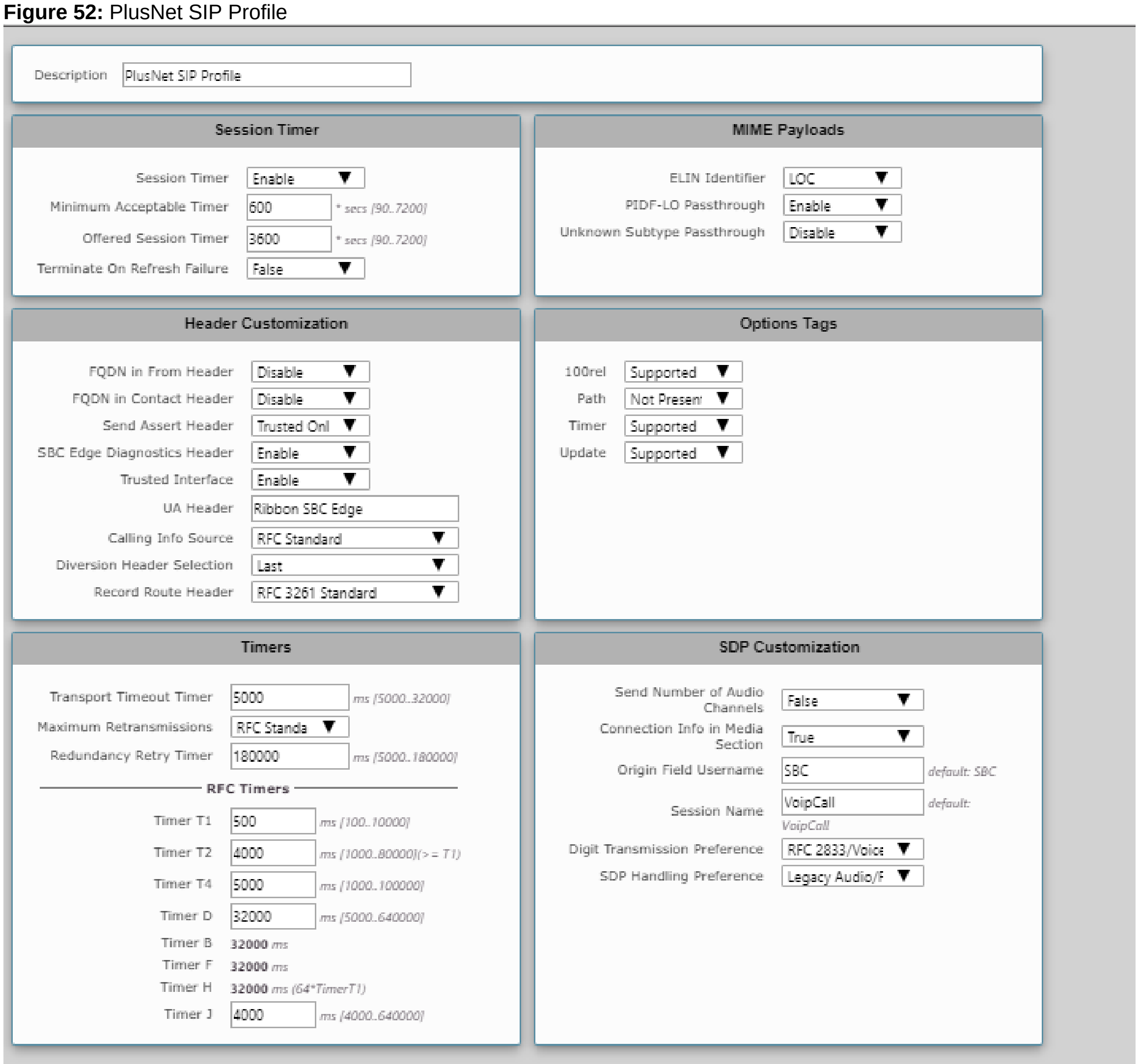


Figure 53: CUCM 12.0.1 SIP Profile

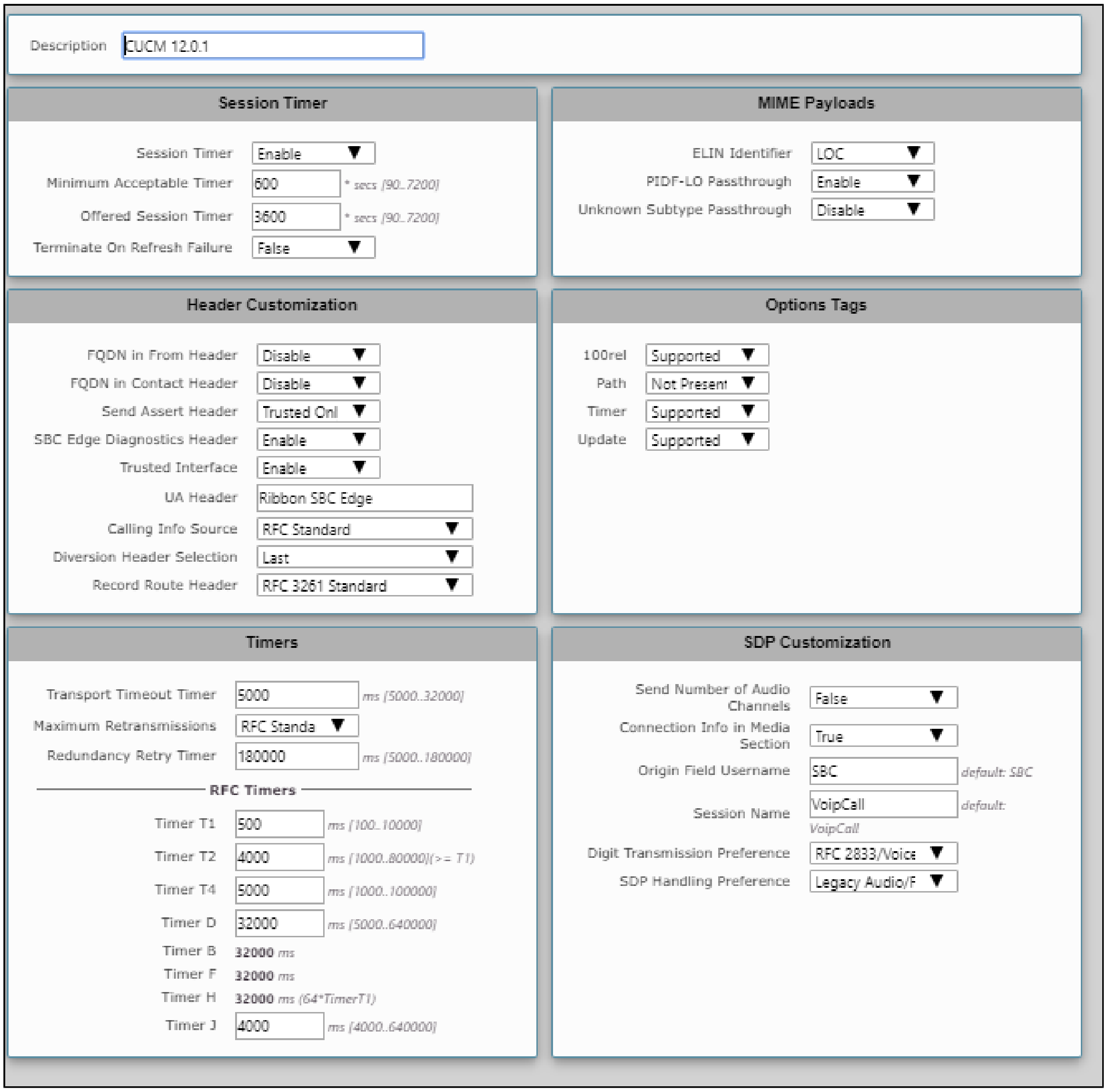


Figure 54: Fax SIP Profile


Tenor-Fax
Description Tenor-Fax
Session Timer
Session Timer Enable
Minimum Acceptable Timer 600 * secs [90..7200]
Offered Session Timer 3600 * secs [90..7200]
Terminate On Refresh Failure False
MIME Payloads
ELIN Identifier LOC
PIDF-LO Passthrough Enable
Unknown Subtype Passthrough Disable
Header Customization
FQDN in From Header Disable
FQDN in Contact Header Disable
Send Assert Header Trusted Onl
SBC Edge Diagnostics Header Enable
Trusted Interface Enable
UA Header Sonus SBC
Calling Info Source RFC Standard
Diversion Header Selection Last
Record Route Header RFC 3261 Standard
Options Tags
100rel Supported
Path Not Present
Timer Supported
Update Supported
Timers
Transport Timeout Timer 5000 ms [5000..32000]
Maximum Retransmissions RFC Standa
Redundancy Retry Timer 180000 ms [5000..180000]
RFC Timers
Timer T1 500 ms [100..10000]
Timer T2 4000 ms [1000..80000](>= T1)
Timer T4 5000 ms [1000..100000]
Timer D 32000 ms [5000..640000]
Timer B 32000 ms
Timer F 32000 ms
Timer H 32000 ms (64*TimerT1)
Timer J 4000 ms [4000..640000]
SDP Customization
Send Number of Audio Channels True
Connection Info in Media Section True
Origin Field Username SBC default: SBC
Session Name VoipCall default: VoipCall
Digit Transmission Preference RFC 2833/Voice
SDP Handling Preference Legacy Audio/F

2. SIP Server

SIP Server Tables contain information about the SIP devices connected to the Ribbon SBC 1000/2000.

Select **Settings > SIP > SIP Server Tables** to access the SIP Server Tables screen.

The entries in the tables provide information about the IP addresses, ports, and protocols used to communicate with each SIP server. The entries also contain links to counters that are useful for troubleshooting, as shown in the following figures.

Figure 55: PlusNet SIP Servers

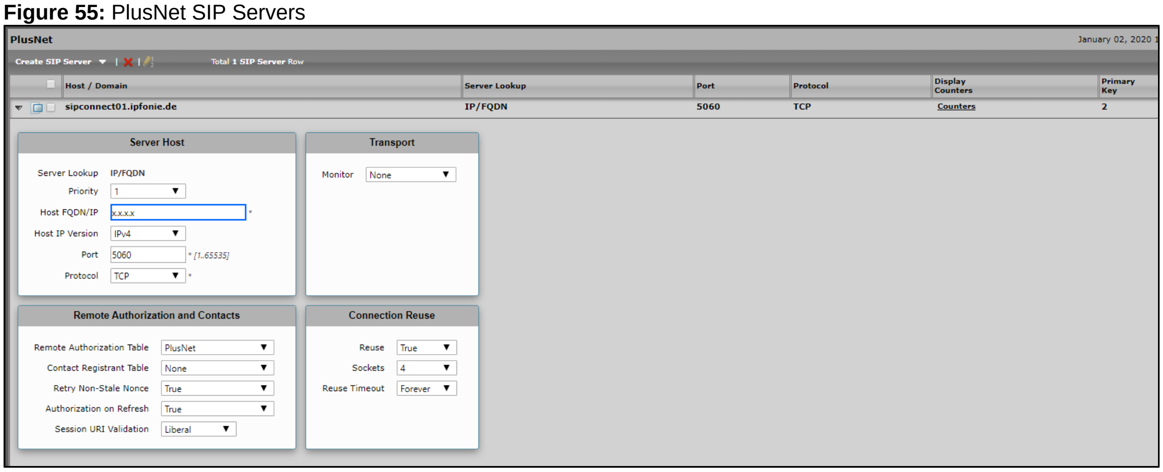


Figure 56: CUCM SIP Server

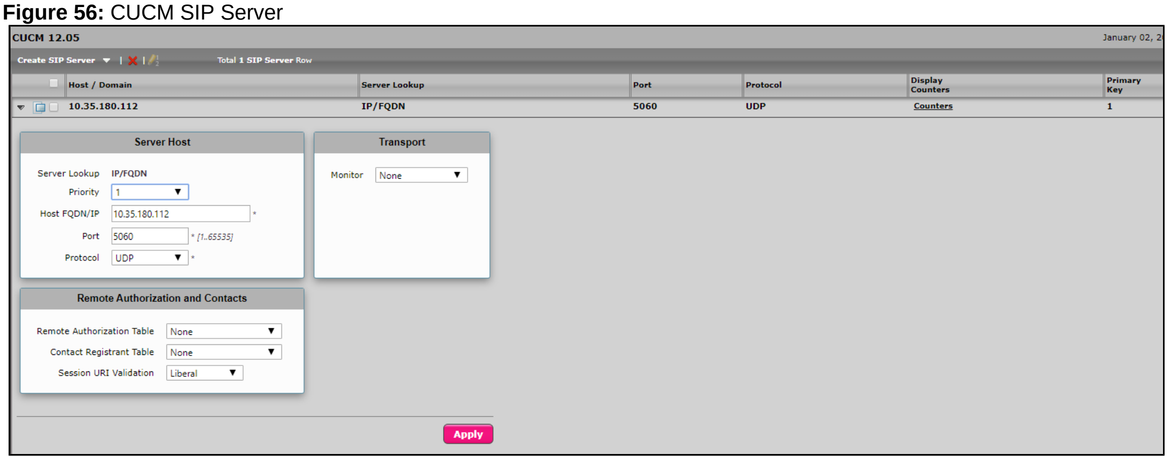
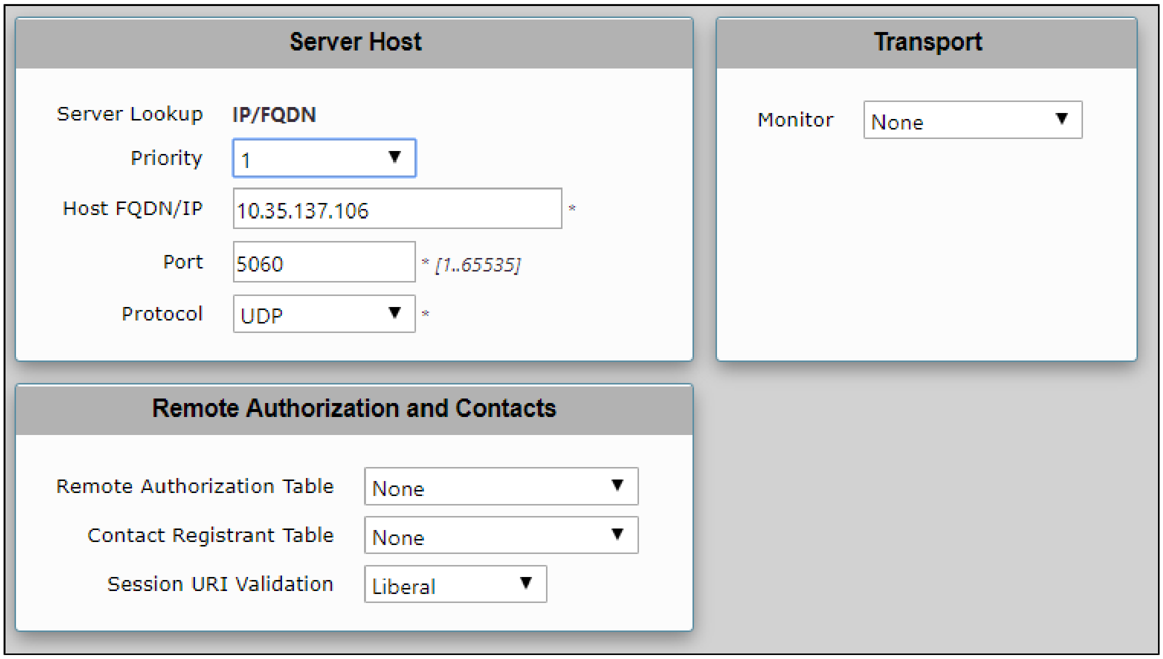


Figure 57: Fax SIP Server

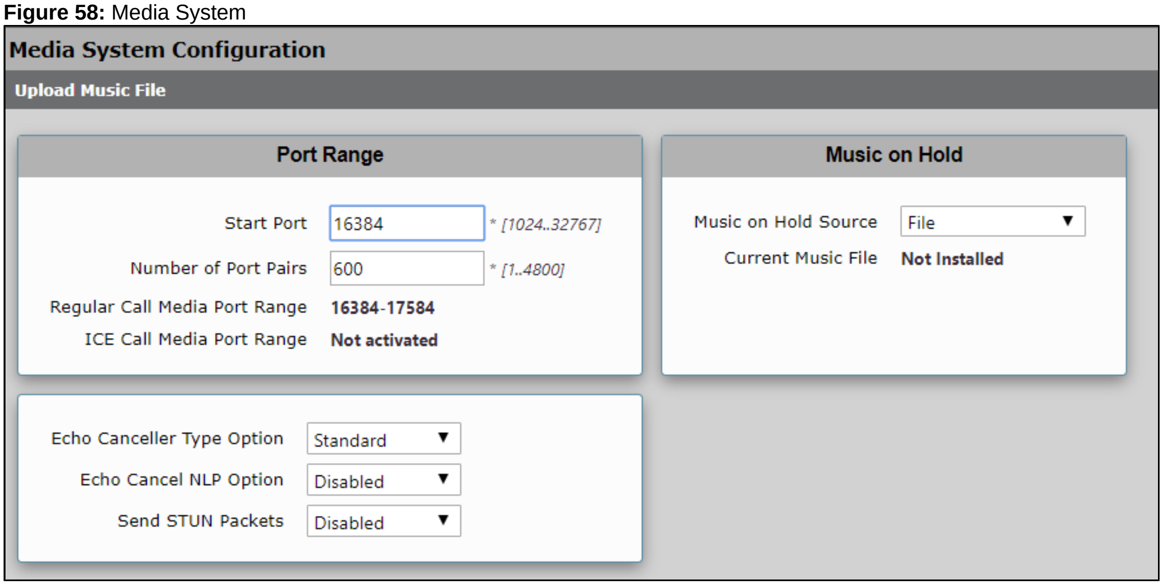

Server Host
Server Lookup
IP/FQDN
Priority
1
Host FQDN/IP
10.35.137.106
*
Port
5060
* [1..65535]
Protocol
UDP
*
Transport
Monitor
None
Remote Authorization and Contacts
Remote Authorization Table
None
Contact Registrant Table
None
Session URI Validation
Liberal

3. Media System

The Media System Configuration contains system-wide settings for the Media System. Configuring the media system means setting the number of RTP/RTCP port pairs and the starting port.

Select **Settings > Media > Media System Configuration** to access the Media System configuration screen.

Figure 58: Media System



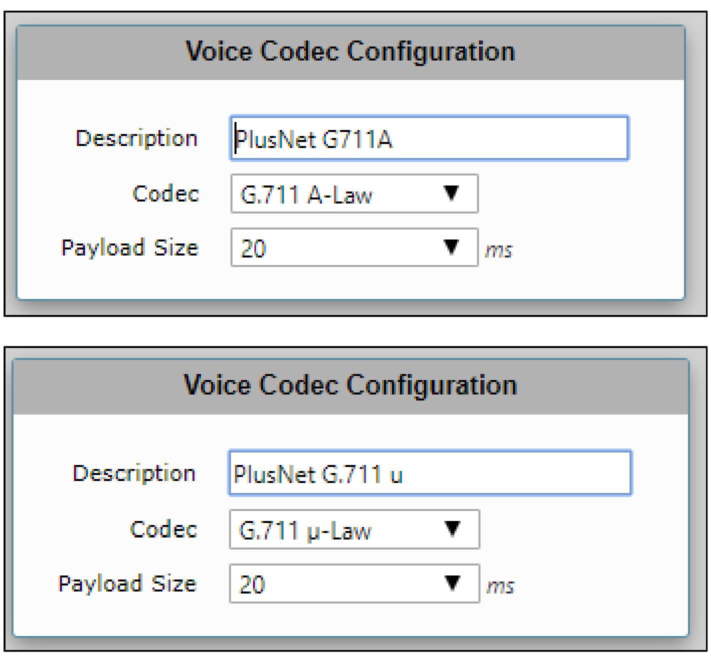
4. Media Profiles

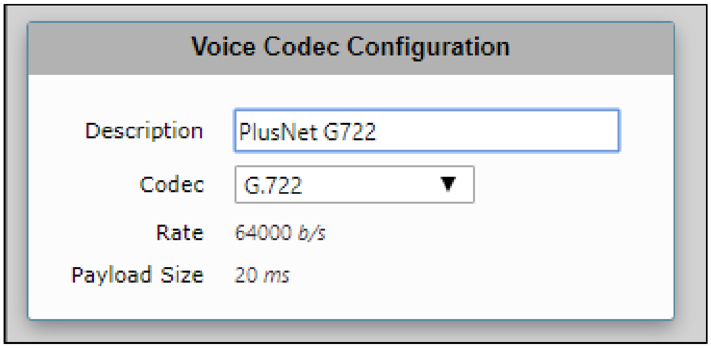
Media Profiles specify the individual voice and fax compression codecs and their associated settings for inclusion into a Media List. Different codecs provide varying levels of compression, allowing the reduction of bandwidth requirements at the expense of voice quality.

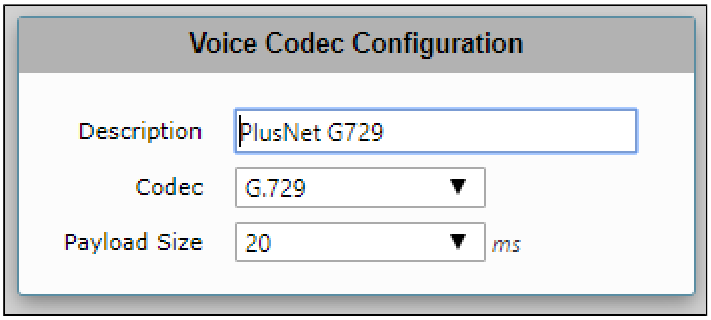
Select **Settings > Media > Media Profiles.**

The following figures illustrate possible media profiles of the voice codecs used for the SBC 1000/2000. The examples are for reference only.

Figure 59: PlusNet Media Profile







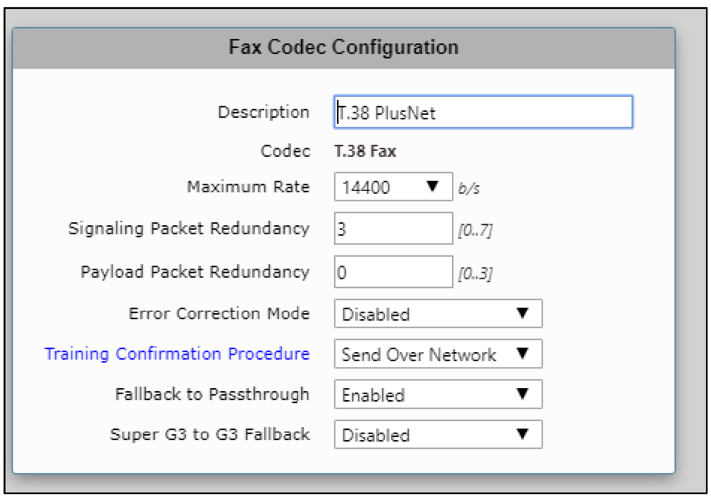


Figure 60: CUCM Media Profile

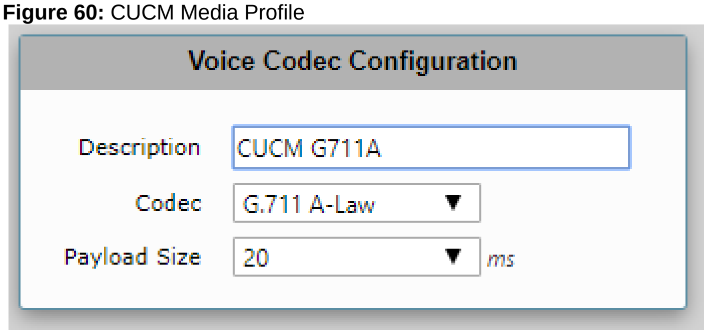
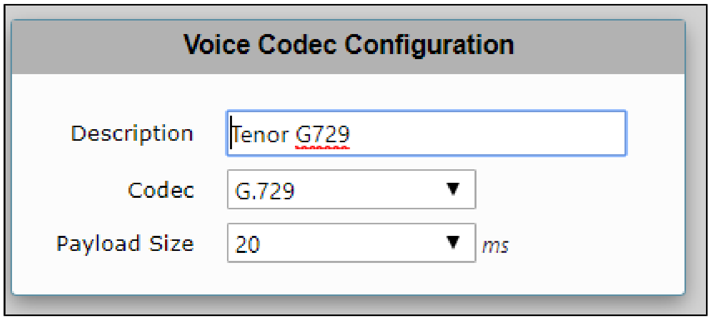
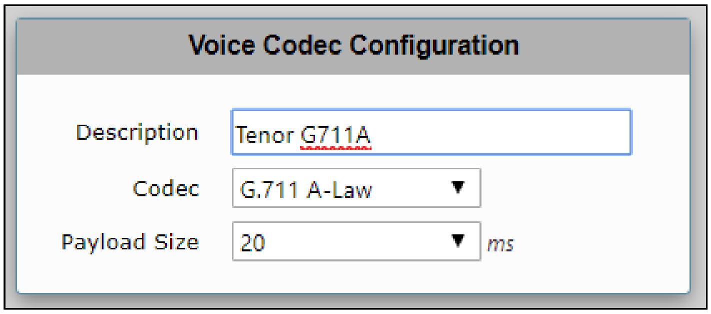
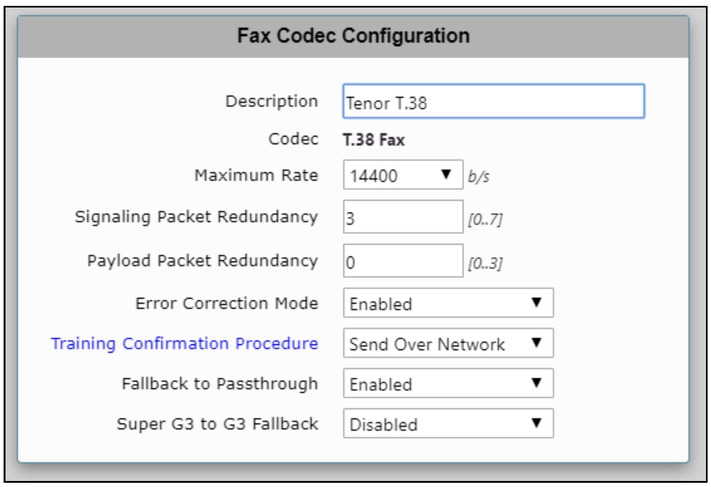


Figure 61: Fax Media Profile


Voice Codec Configuration
Description
Tenor G729
Codec
G.729
Payload Size
20
ms


Voice Codec Configuration
Description
Tenor G711A
Codec
G.711 A-Law
Payload Size
20
ms

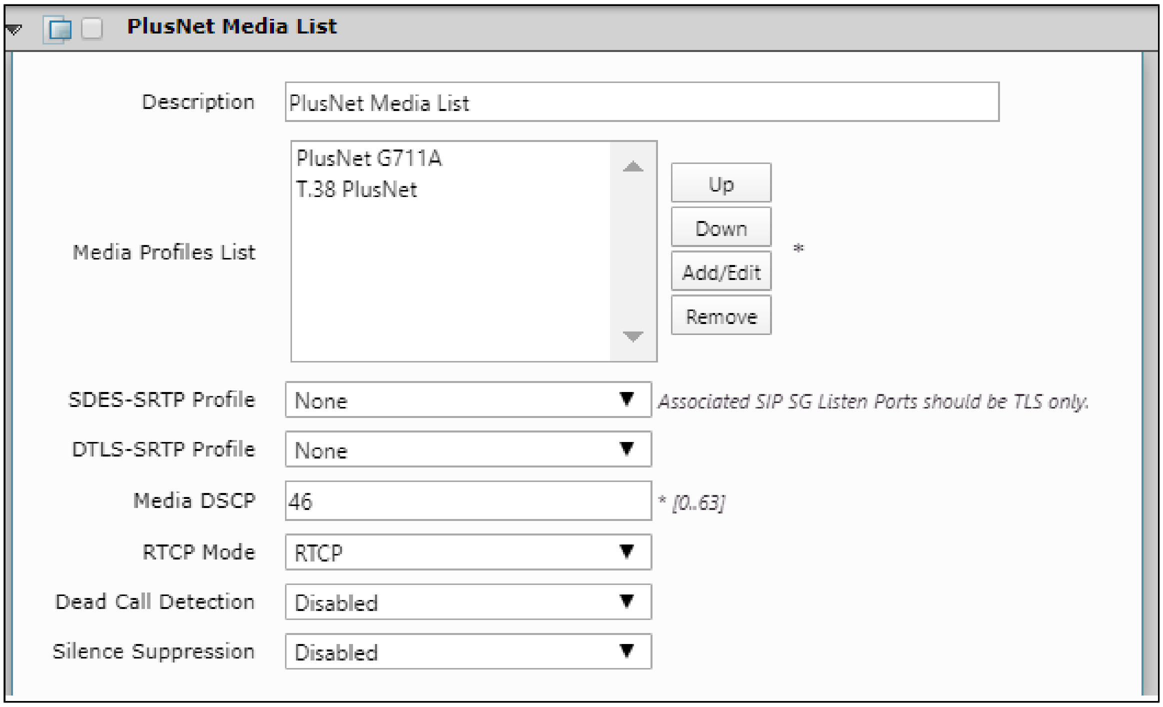

Fax Codec Configuration
Description
Tenor T.38
Codec
T.38 Fax
Maximum Rate
14400
b/s
Signaling Packet Redundancy
3
[0..7]
Payload Packet Redundancy
0
[0..3]
Error Correction Mode
Enabled
Training Confirmation Procedure
Send Over Network
Fallback to Passthrough
Enabled
Super G3 to G3 Fallback
Disabled

5. Media List

The Media List shows the selected voice and fax compression codecs and their associated settings.

Select **Settings > Media > Media List** to access the Media List configuration screen.

Figure 62: PlusNet Media List



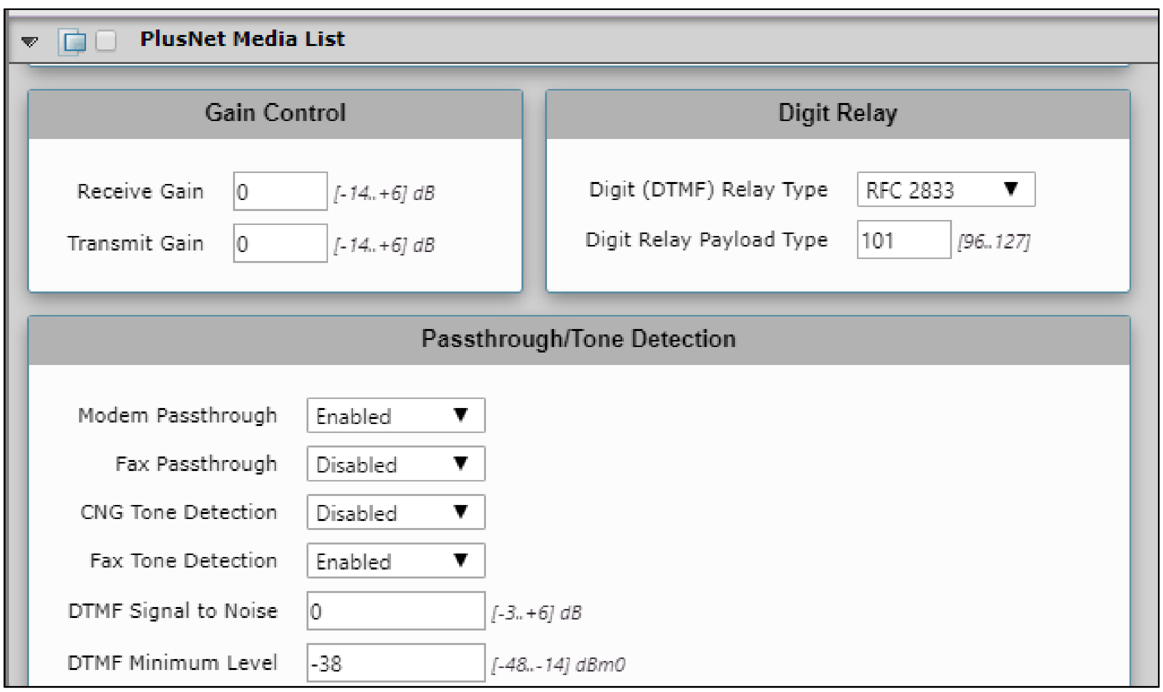


Figure 63: CUCM Media List

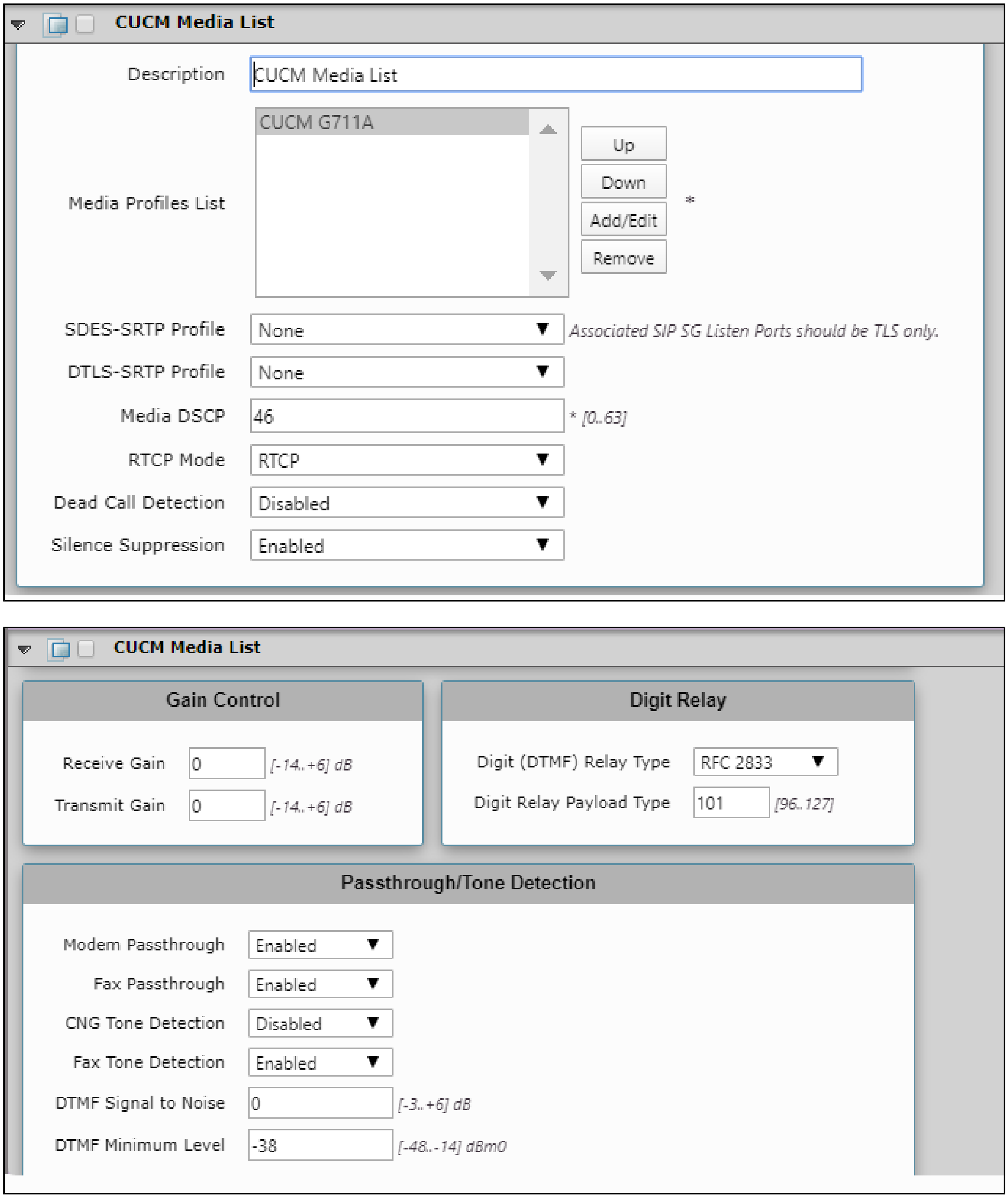
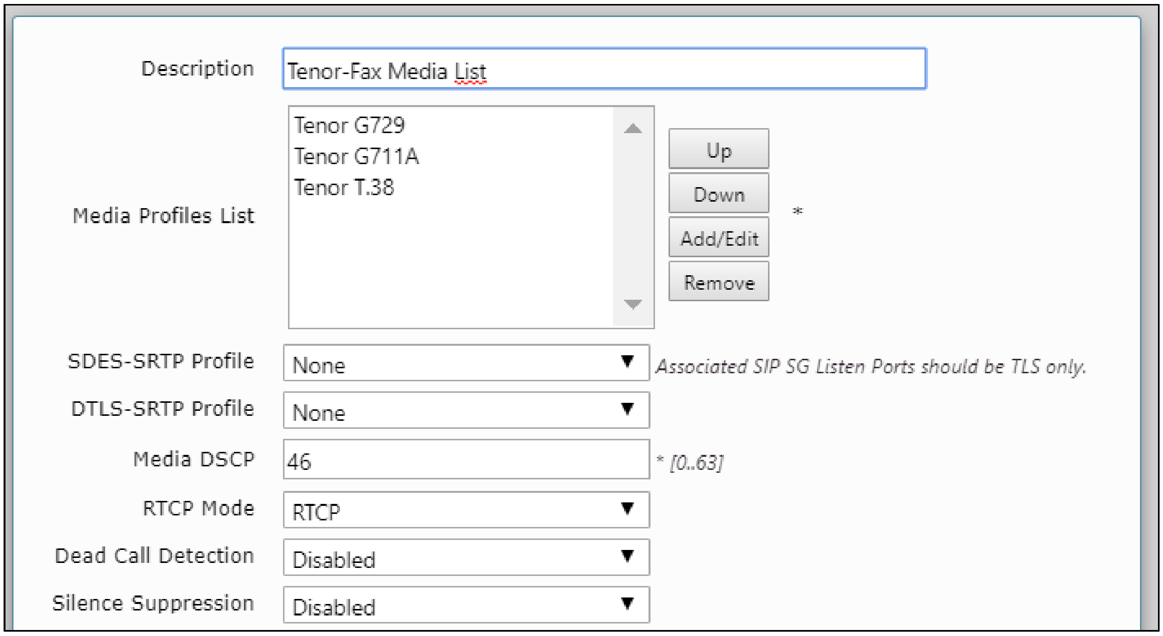
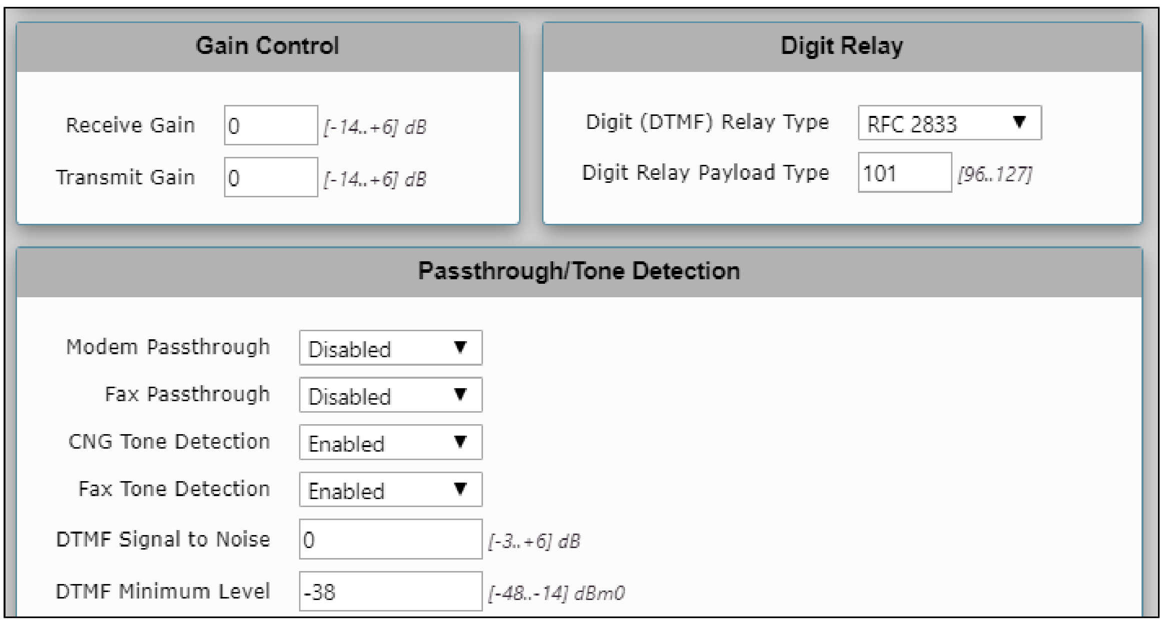


Figure 64: Fax Media List


Description
Tenor-Fax Media List
Media Profiles List
Tenor G729
Tenor G711A
Tenor T.38
Up
Down
Add/Edit
Remove
*
SDES-SRTP Profile
None
Associated SIP SG Listen Ports should be TLS only.
DTLS-SRTP Profile
None
Media DSCP
46
* [0..63]
RTCP Mode
RTCP
Dead Call Detection
Disabled
Silence Suppression
Disabled

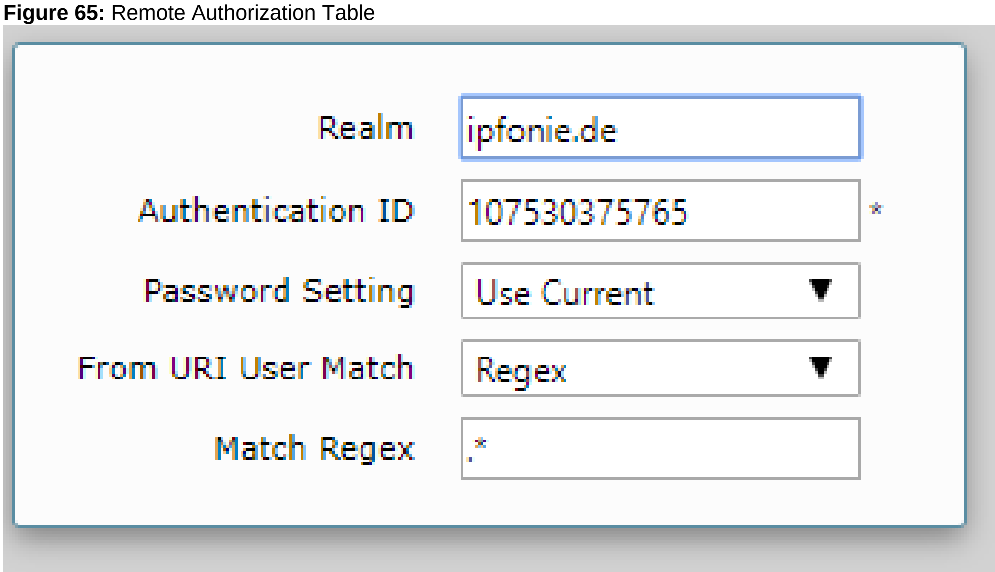

Gain Control
Receive Gain
0
[-14..+6] dB
Transmit Gain
0
[-14..+6] dB
Digit Relay
Digit (DTMF) Relay Type
RFC 2833
Digit Relay Payload Type
101
[96..127]
Passthrough/Tone Detection
Modem Passthrough
Disabled
Fax Passthrough
Disabled
CNG Tone Detection
Enabled
Fax Tone Detection
Enabled
DTMF Signal to Noise
0
[-3..+6] dB
DTMF Minimum Level
-38
[-48..-14] dBm0

6. Remote Authorization Tables

Remote Authorization Tables and their entries contain information used to respond to request message challenges by an upstream server. The Remote Authorization Tables on this page appear as options in Creating and Modifying Entries in the SIP Servers (For additional information about Remote Authorization Tables, see the Ribbon online SBC 1000/2000 documentation).

Select **Settings > SIP > Remote Authorization Tables** to access the Remote Authorization Tables configuration screen.

Figure 65: Remote Authorization Table

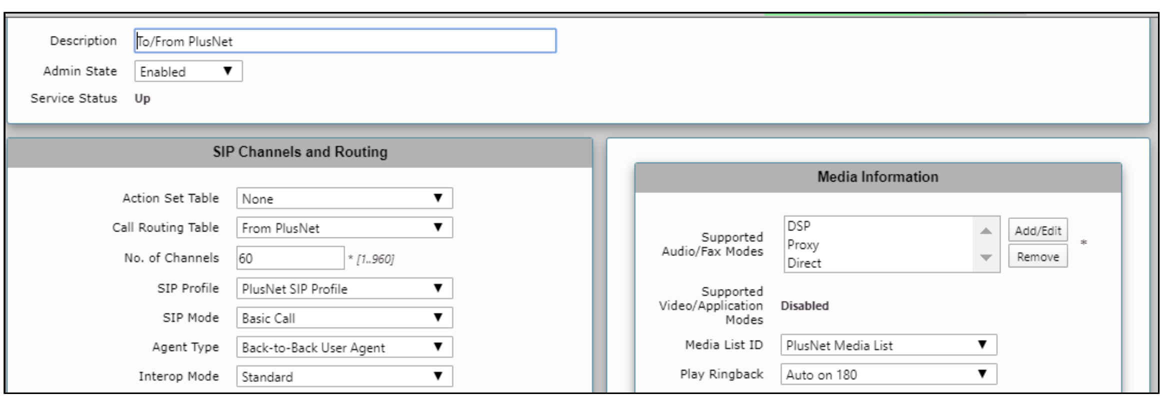


7. Signaling Groups

Signaling Groups allow telephony channels to be grouped for routing and shared configuration. These groups are the entity to which calls are routed, and the location from which Call Routes are selected. These are also the location from which Tone Tables and Action Sets are selected. In the case of SIP, Signaling Groups will specify protocol settings and links to server, media, and mapping tables.

Select **Settings > Signaling Groups** to access the Signaling Groups configuration screens.

Figure 66: PlusNet Signaling Group



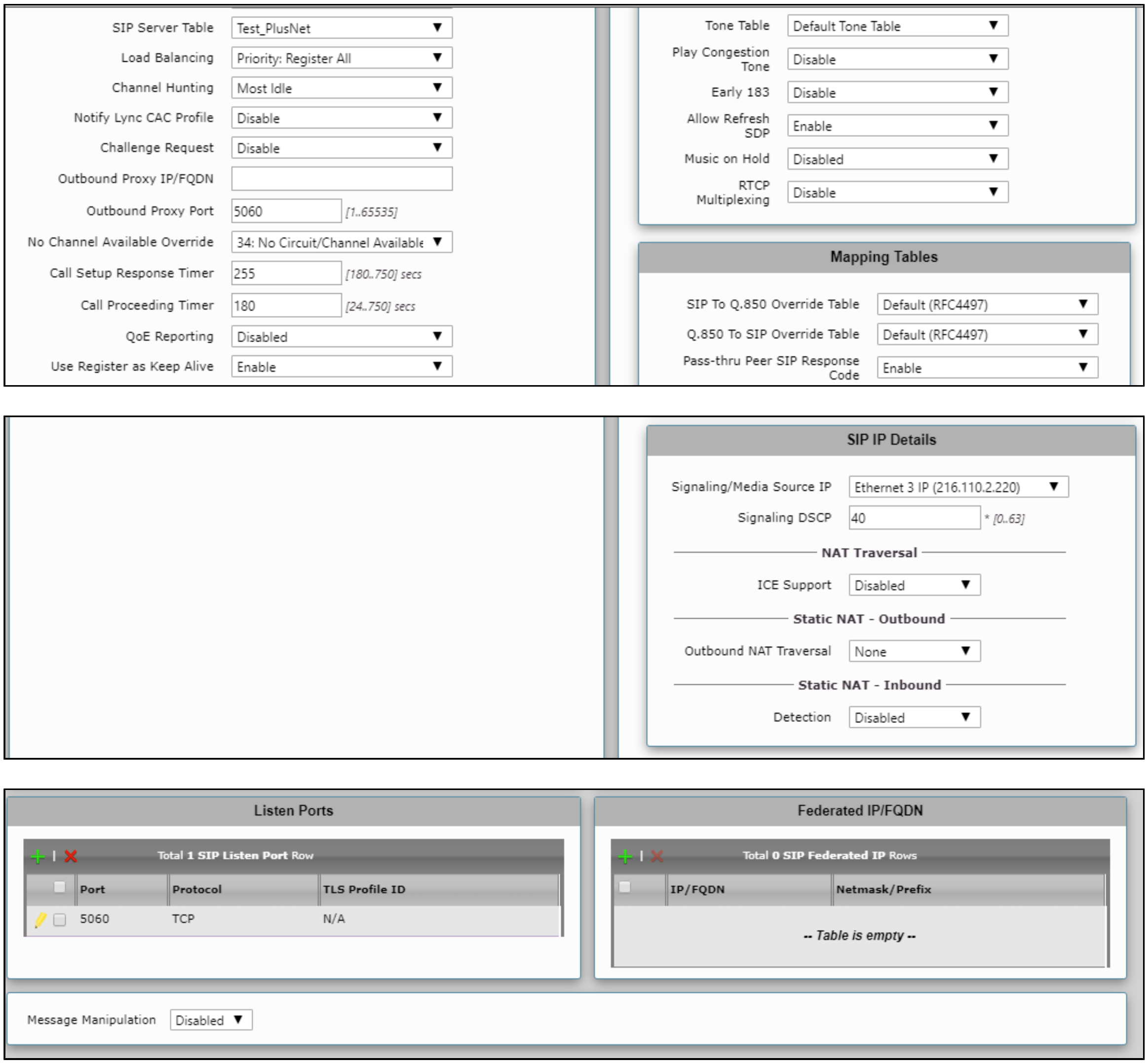
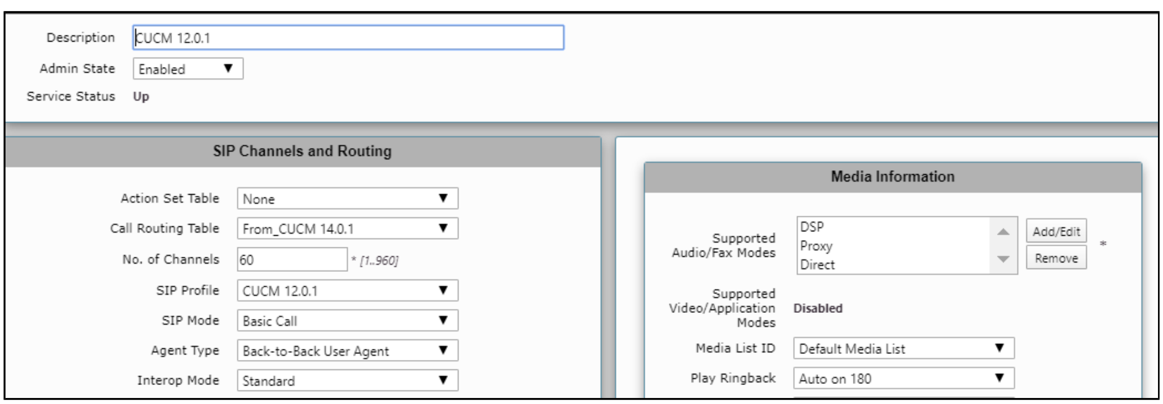


Figure 67: CUCM Signaling Group



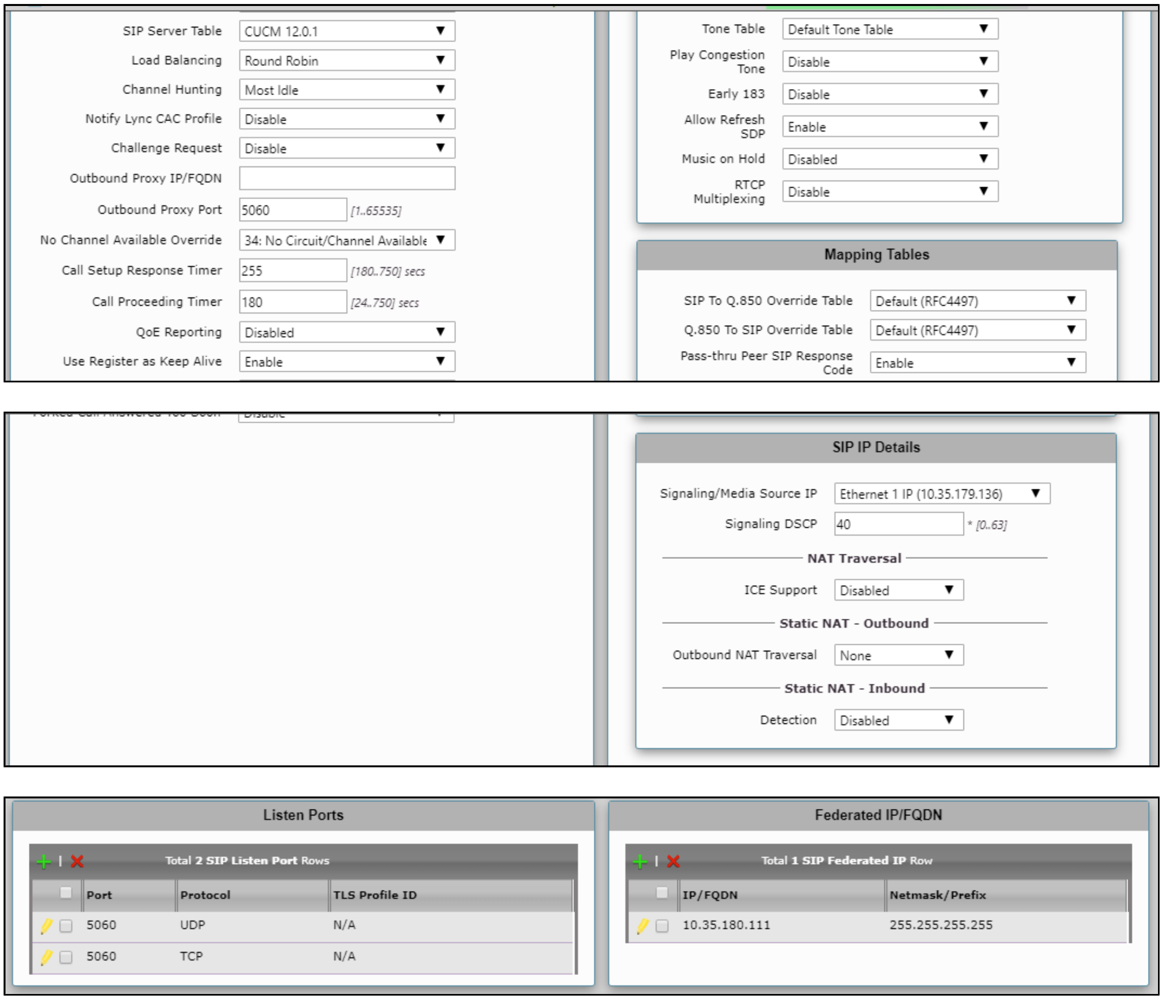
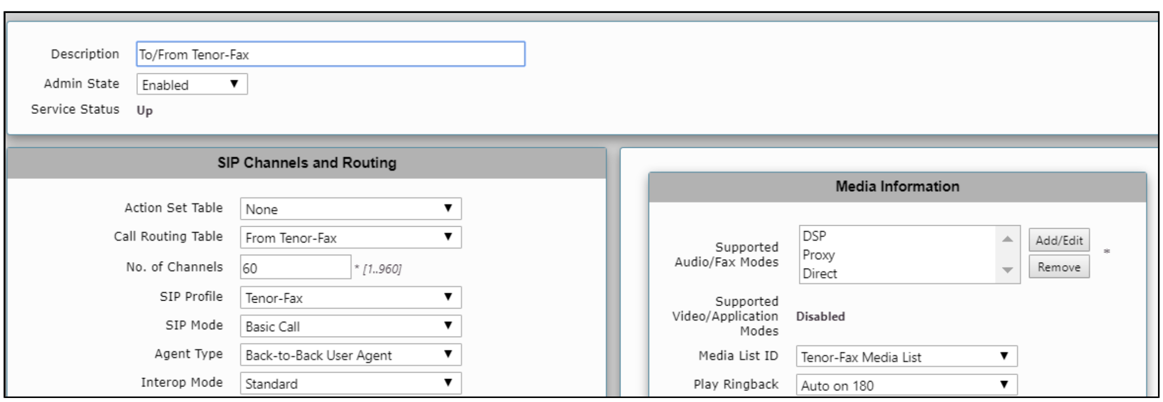
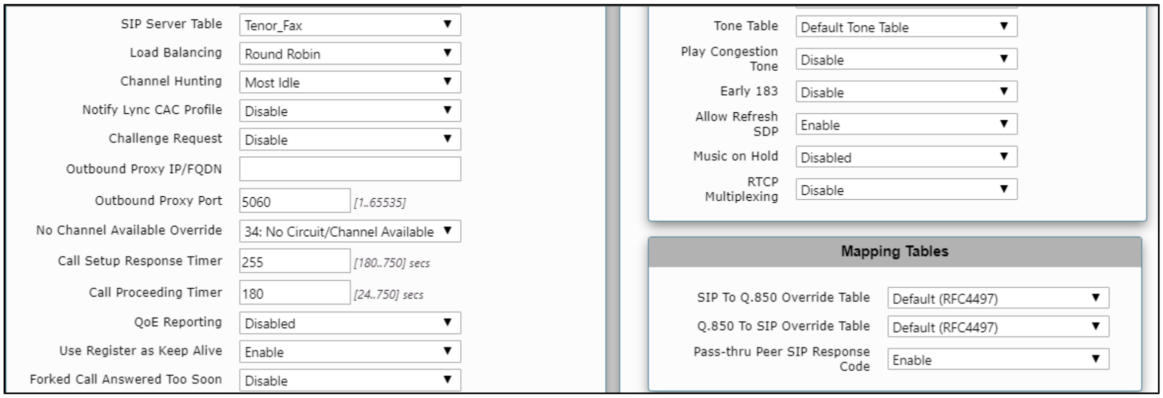
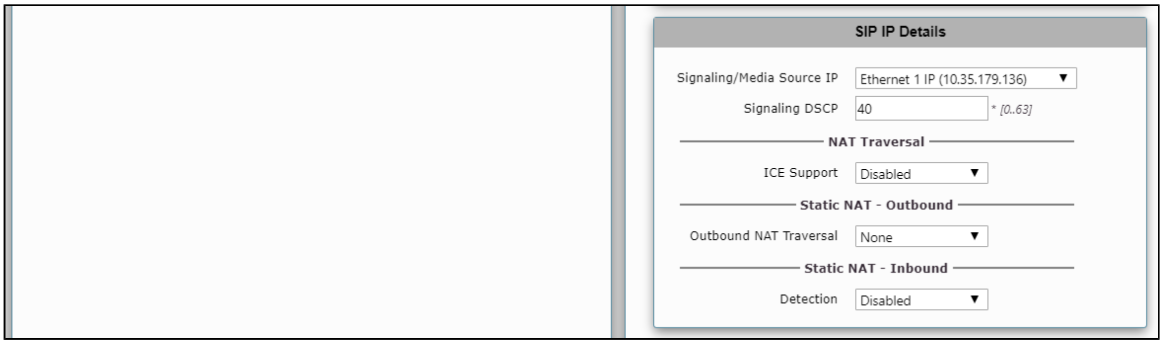
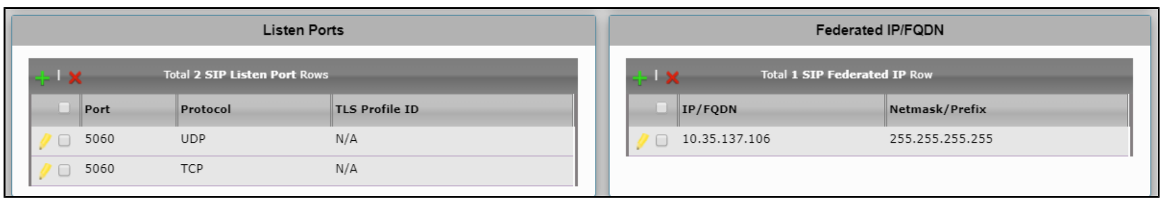


Figure 68: Fax Signaling Group







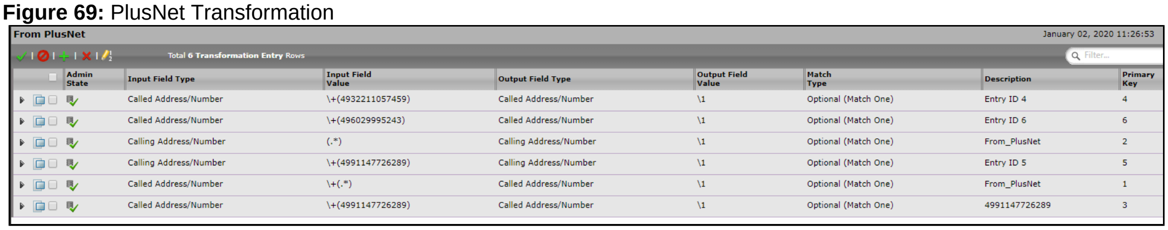


8. Transformation

Transformation Tables facilitate the conversion of names, numbers, and other fields when routing a call. For example, transformations can convert a public PSTN number into a private extension number, or a SIP address (URI). Every entry in a Call Routing Table requires a Transformation Table.

Select **Settings > Transformation** to access the Transformation configuration screen.

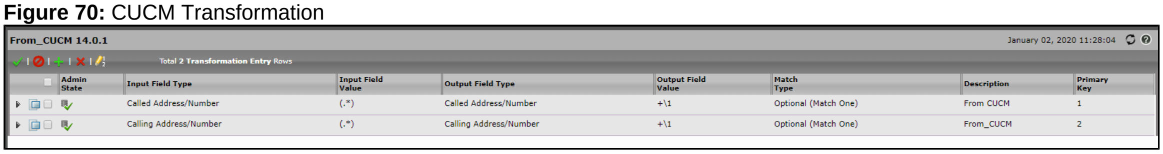
Figure 69: PlusNet Transformation



From PlusNet — January 02, 2020 11:26:53 — Total 6 Transformation Entry Rows

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key
	Called Address/Number	\+(4932211057459)	Called Address/Number	\1	Optional (Match One)	Entry ID 4	4
	Called Address/Number	\+(496029995243)	Called Address/Number	\1	Optional (Match One)	Entry ID 6	6
	Calling Address/Number	(.*)	Calling Address/Number	\1	Optional (Match One)	From_PlusNet	2
	Calling Address/Number	\+(4991147726289)	Calling Address/Number	\1	Optional (Match One)	Entry ID 5	5
	Called Address/Number	\+(.*)	Called Address/Number	\1	Optional (Match One)	From_PlusNet	1
	Called Address/Number	\+(4991147726289)	Called Address/Number	\1	Optional (Match One)	4991147726289	3

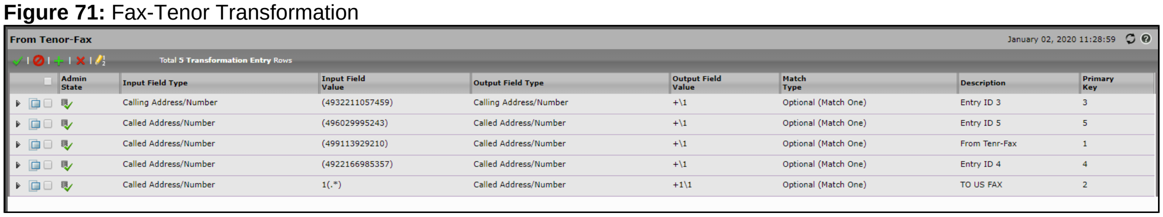
Figure 70: CUCM Transformation



From_CUCM 14.0.1 — January 02, 2020 11:28:04 — Total 2 Transformation Entry Rows

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key
	Called Address/Number	(.*)	Called Address/Number	+\1	Optional (Match One)	From CUCM	1
	Calling Address/Number	(.*)	Calling Address/Number	+\1	Optional (Match One)	From_CUCM	2

Figure 71: Fax-Tenor Transformation



From Tenor-Fax — January 02, 2020 11:28:59 — Total 5 Transformation Entry Rows

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key
	Calling Address/Number	(4932211057459)	Calling Address/Number	+\1	Optional (Match One)	Entry ID 3	3
	Called Address/Number	(496029995243)	Called Address/Number	+\1	Optional (Match One)	Entry ID 5	5
	Called Address/Number	(499113929210)	Called Address/Number	+\1	Optional (Match One)	From Tenr-Fax	1
	Called Address/Number	(4922166985357)	Called Address/Number	+\1	Optional (Match One)	Entry ID 4	4
	Called Address/Number	1(.*)	Called Address/Number	+1\1	Optional (Match One)	TO US FAX	2

9. Call Routing Table

Call Routing allows calls to be carried between signaling groups, thus allowing calls to be carried between ports, and between protocols (like ISDN to SIP). Call Routing Tables define routes. The use of Call Routing Tables allows for flexible configuration of which calls will be carried, and also how the calls are translated. These tables are one of the central connection points of the system, linking Transformation Tables, Message translations, Cause Code Reroutes, Media Lists, and the three types of Signaling Groups (ISDN, SIP and, CAS).

Select **Settings > Call Routing Table** to access the Call Routing Table configuration screen.

Figure 72: PlusNet Call Routing

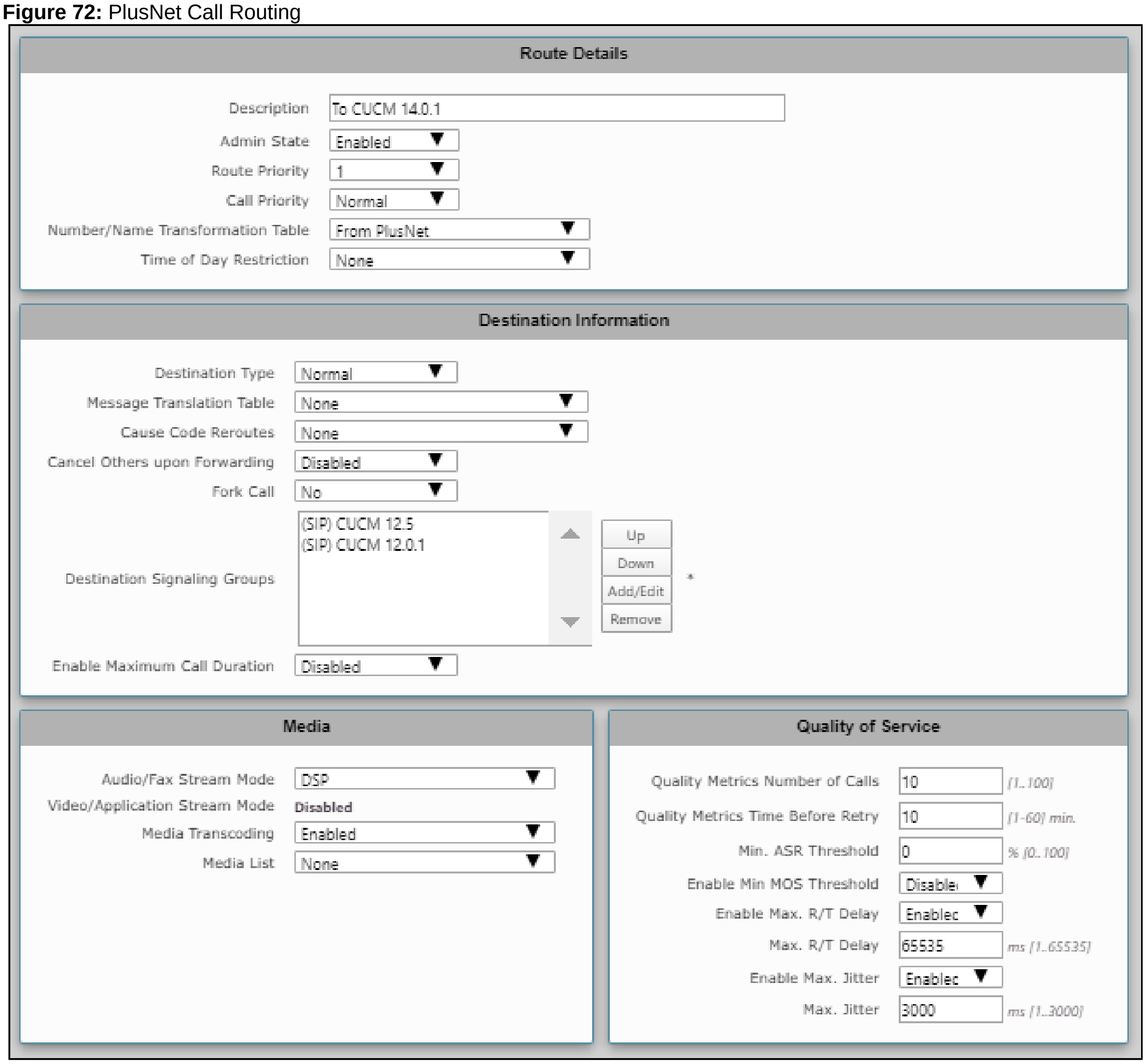


Figure 73: CUCM Call Routing

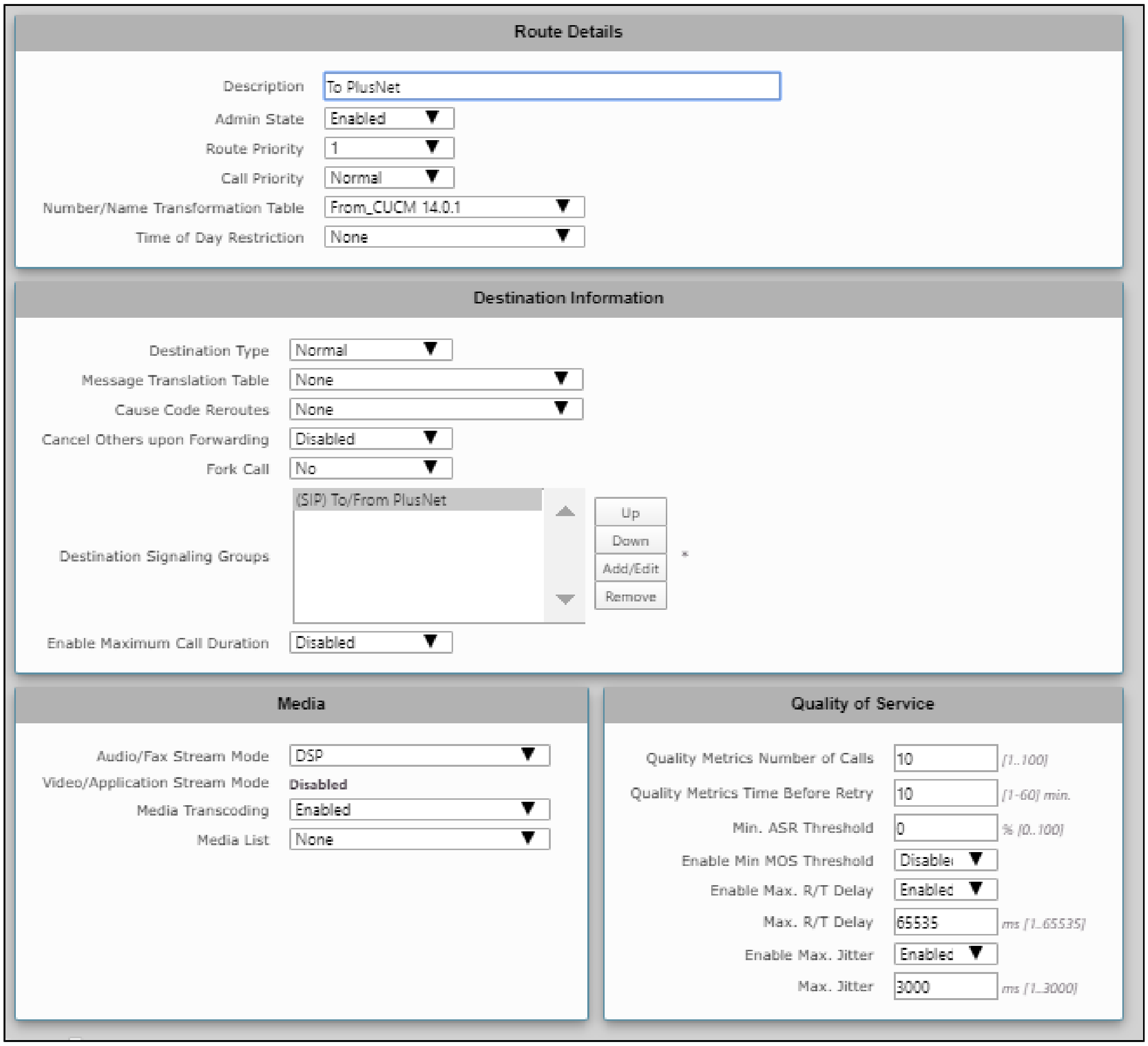
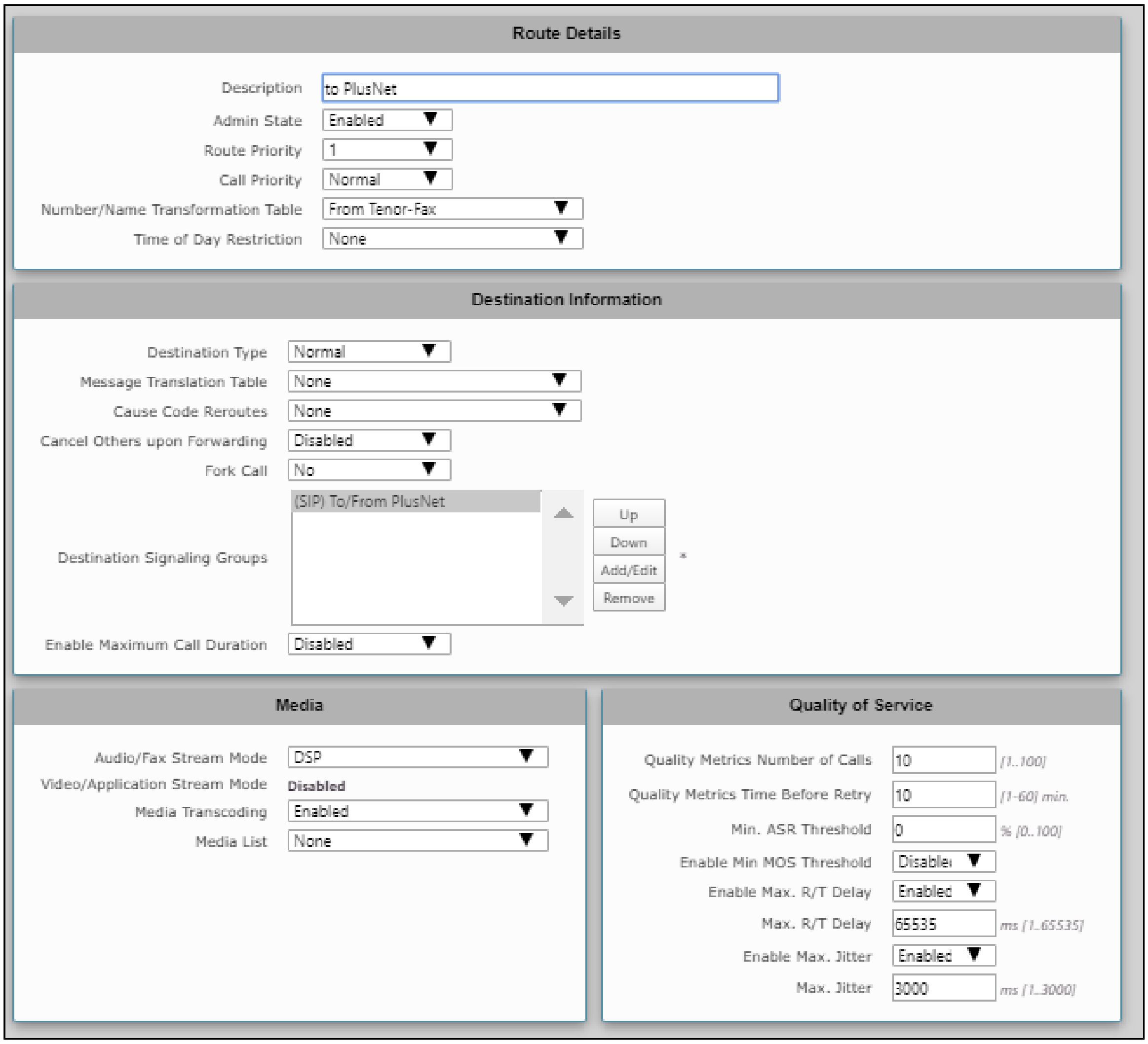


Figure 74: Fax Call Routing


Route Details
Description
to PlusNet
Admin State
Enabled
Route Priority
1
Call Priority
Normal
Number/Name Transformation Table
From Tenor-Fax
Time of Day Restriction
None
Destination Information
Destination Type
Normal
Message Translation Table
None
Cause Code Reroutes
None
Cancel Others upon Forwarding
Disabled
Fork Call
No
Destination Signaling Groups
(SIP) To/From PlusNet
Up
Down
Add/Edit
Remove
*
Enable Maximum Call Duration
Disabled
Media
Audio/Fax Stream Mode
DSP
Video/Application Stream Mode
Disabled
Media Transcoding
Enabled
Media List
None
Quality of Service
Quality Metrics Number of Calls
10
[1..100]
Quality Metrics Time Before Retry
10
[1-60] min.
Min. ASR Threshold
0
% [0..100]
Enable Min MOS Threshold
Disable
Enable Max. R/T Delay
Enabled
Max. R/T Delay
65535
ms [1..65535]
Enable Max. Jitter
Enabled
Max. Jitter
3000
ms [1..3000]

Interoperability Test Results

The following table provides test results for interoperability compliance testing between Ribbon SBC 1000/2000 and CUCM

Table 4: Interoperability Compliance Test Results

Test Number	Test Scenario	Setup / Test Results	Status	Comment
4.1	Registration and authentication (registration mode)	The PBX is able to execute the correct resolution oft he DNS SRV record	Pass	
4.3	Basic call	With the basic call tests, the standard call scenarios and the CLIP/CLIR features are tested. • Only en-bloc dialing is supported, overlap sending is not possible.	Pass	
4.3.1	Normal call	Outgoing call from PBX to PSTN • En-bloc dialling • Local area call (without area code); area code must be set by the PBX • Setting of the correct calling number with all available telephone number blocks • If two or more locations with different area codes are assigned to one SIP trunk, the number in the PAI header has to be the geographic number which belongs to the users location. This is very important in case of emergency calls, because Plusnet uses the number of the PAI header to route the emergency call to the proper emergency call center.	Pass	
4.3.1.1	Normal call	The geographic number in the PAI header corresponds to the location 11 of the user	Pass	Is it possible to configure several PAI per SIP trunk? Yes
4.3.1.2	Normal call	Display of A-number in B-party CLIP (national PSTN)	Pass	
4.3.1.3	Normal call	Display of A-number in B-party CLIP (international PSTN)	Pass	
4.3.1.4	Normal call	Display of A-number in B-party CLIP (mobile)	Pass	
4.3.1.5	Normal call	Call to mobile Outgoing call to mobile => mobile phone turned off	Pass	
4.3.1.6	Normal call	Suppression of A-number => CUR	Pass	
4.3.1.7	Normal call	Outgoing call from analog extension	Pass	
4.3.1.8	Normal call	Outgoing call (> 5 min.) => PSTN	Pass	Hold connection for 5 minutes => RTP still correct? Yes
4.3.2	Normal call	Incoming call from PSTN (national) => PBX • Test all available telephone number blocks	Pass	
4.3.2.1	Normal call	Display of A-number => CLIP	Pass	
4.3.2.2	Normal call	Incoming call from mobile => PBX Display of A-number => CLIP	Pass	
4.3.2.3	Normal call	Suppression of A-number => CLIR	Pass	
4.3.3	Normal call	Two simultaneous outgoing/incoming calls	Pass	
4.3.4	Normal call	Enabled feature DND (do not disturb)	Pass	

4.3.5	Normal call	Test call with codec G.711	Pass	
4.3.6	Normal call	Test call with codec G.722 (only SIP <=> SIP)	Fail	PlusNet didn´t support G. 722
4.3.7	Normal call	Test call with codec G.729	Pass	
4.3.2.8	Clip No Screening	Outgoing call from PBX to PSTN • With feature Clip No Screening • Test with several different A-numbers • If two or more locations with different area codes are assigned to one SIP trunk, the number in the PAI header has to be the geographic number which belongs to the users location. This is very important in case of emergency calls, because PlusNet uses the number	Pass	
4.3.2.8.1	Clip No Screening	Despite Clip No Screening the geographic number in the PAI header corresponds to the location of the user	Pass	Is it possible to configure several PAI per SIP trunk? Yes
4.3.2.8.2	Clip No Screening	Display of A-number (NoSClip) at B-party (PSTN)	Pass	
4.3.2.8.3	Clip No Screening	Display of A-number (NoSClip) at B-party (international PSTN; depending on the destination carrier, the NoSClip telephone number may not be displayed in this case!)	Pass	
4.3.2.8.3	Clip No Screening	Call made from a PSTN line to an IP-PBX line with call forward to a line within the same IP-PBX, Answer Call. • Either party terminates call.	Pass	Does the IP-PBX has configuration settings to send SIP status 181 messages to the soft switch? Yes
4.3.2.8.4	Clip No Screening	Display of A-number (NoSClip) at B-party (mobile)	Pass	
4.3.3.9	Special call situations	Outgoing call PBX => PSTN • Call is rejected by B-party	Pass	
4.3.3.10	Special call situations	Outgoing call PBX=> PSTN • B-party does not answer; clearing after timer	Pass	
4.3.3.11	Special call situations	Outgoing call PBX => PSTN • B-party busy; busy tone	Pass	
4.3.3.12	Special call situations	Outgoing call PBX=> PSTN • A-party hangs up before call is established (cancel)	Pass	
4.3.3.13	Special call situations	Incoming call PSTN => PBX • Call is rejected by PBX party	Pass	
4.3.3.14	Special call situations	Incoming call PSTN => PBX • PBX party does not answer; clearing after timer	Pass	
4.3.3.15	Special call situations	Incoming call PSTN => PBX • PBX party busy; busy tone	Pass	

4.3.3.16	Special call situations	Incoming call PSTN => PBX • A-party hangs up before call is established (cancel)	Pass	
4.3.4.17	Call clearing	Incoming / outgoing call; clearing after established call. Correct clearing on both sides • PBX party hangs up • PSTN party hangs up	Pass	
4.3.4.18	Call clearing	Interrupting the network connection of the SIP terminal device during a call • Call should be cleared correctly	Pass	
4.4.19	Hold	PBX => PSTN and PSTN => PBX • Test call in both directions	Pass	
4.4.19.1	Hold	Putting an external call on hold in the PBX	Pass	
4.4.19.2	Hold	If applicable, MoH (music on hold) at A-party (PSTN)	Pass	
4.4.19.3	Hold	HOLD RETRIEVE: retrieving the external call	Pass	
4.4.19.4	Hold	Clearing the connection of the A-party while it is put on hold	Pass	
4.4.20	Hold	PBX => PSTN and PSTN => PBX • Test call in both directions	Pass	
4.4.20.1	Hold	Putting an external call on hold in the PSTN	Pass	
4.4.20.2	Hold	If applicable, MoH (music on hold) at A-party (PBX)	Pass	
4.4.20.3	Hold	HOLD RETRIEVE: retrieving the external call	Pass	
4.4.20.4	Hold	Clearing the connection of the A-party while it is put on hold	Pass	
4.5.21	Call transfer	Internal call is transferred to external party: internal => PBX => external	Pass	
4.5.21.1	Call transfer	Call transfer from PBX party => PSTN party with announcement (attendant transfer)	Pass	
4.5.21.2	Call transfer	Call transfer from PBX party => PSTN party without announcement (blind transfer)	Pass	
4.5.21.3	Call transfer	Call transfer from PBX party => PSTN party without announcement (blind transfer) • Call is rejected or not answered	Pass	
4.5.21.4	Call transfer	Call transfer from PBX party => PSTN party without announcement (blind transfer) • PSTN party busy	Pass	
4.5.22	Call transfer	Transferred call from external party to PBX: PSTN => PBX	Pass	
4.5.22.1	Call transfer	Call transfer from PSTN => PBX party with announcement (attendant transfer)	Pass	
4.5.22.2	Call transfer	Call transfer from PSTN => PBX party without announcement (blind transfer)	Pass	
4.5.22.3	Call transfer	Call transfer from PSTN => PBX party without announcement (blind transfer) • Call is rejected or not answered	Pass	
4.5.22.4	Call transfer	Call transfer from PSTN => PBX party without announcement (blind transfer) • PBX party busy	Pass	
4.5.23	Call transfer	Call from external party transferred to another external party: external => PBX => external	Pass	
4.5.23.1	Call transfer	PSTN => PBX party => PSTN with announcement (attendant transfer)	Pass	
4.5.23.2	Call transfer	PSTN => PBX party => PSTN without announcement (blind transfer)	Pass	

4.5.23.3	Call transfer	PSTN => PBX party => PSTN without announcement (blind transfer) • Call is rejected or not answered	Pass	
4.5.23.4	Call transfer	PSTN => PBX party => PSTN without announcement (blind transfer) • PSTN party busy	Pass	
4.6.24	Call diversion	PBX party CFU to external party (PSTN)	Pass	
4.6.24.1	Call diversion	Internal call (CFU) => PSTN	Pass	
4.6.24.2	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.24.3	Call diversion	A-party clears in ring phase	Pass	
4.6.24.4	Call diversion	External => PBX party (CFU) => PSTN	Pass	
4.6.24.5	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.24.6	Call diversion	A-party clears in ring phase	Pass	
4.6.25	Call diversion	PBX party CFNR to external party	Pass	
4.6.25.1	Call diversion	Internal call (CFNR) => PSTN	Pass	
4.6.25.2	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.25.3	Call diversion	A-party clears in ring phase	Pass	
4.6.25.4	Call diversion	External => PBX party (CFNR) => PSTN	Pass	
4.6.25.5	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.25.6	Call diversion	A-party clears in ring phase	Pass	
4.6.26	Call diversion	PBX party CFB to external party	Pass	
4.6.26.1	Call diversion	Internal call (CFB) => PSTN	Pass	
4.6.26.2	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.26.3	Call diversion	A-party clears in ring phase	Pass	
4.6.26.4	Call diversion	External => PBX party (CFB) => PSTN	Pass	
4.6.26.5	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.26.6	Call diversion	A-party clears in ring phase	Pass	
4.6.27	Call diversion	External party (PSTN, mobile, etc.) CFU to PBX party:	Pass	
4.6.27.1	Call diversion	External (CFU) => PBX party	Pass	

4.6.27.2	Call diversion	PBX party busy, rejects call, does not answer	Pass	
4.6.27.3	Call diversion	A-party clears in ring phase	Pass	
4.6.27.4	Call diversion	External (CFNR) => PBX party	Pass	
4.6.27.5	Call diversion	PBX party busy, rejects call, does not answer	Pass	
4.6.27.6	Call diversion	A-party clears in ring phase	Pass	
4.6.28	Call diversion	External party (PSTN, mobile, etc.) CFB to PBX party:	Pass	
4.6.28.1	Call diversion	PBX party busy, rejects call, does not answer	Pass	
4.6.28.2	Call diversion	A-party clears in ring phase	Pass	
4.6.29	Call diversion	Call deflection: diversion during the ring phase	Pass	
4.6.29.1.1	Call diversion	Internal call PBX party CD => PBX party	Pass	
4.6.29.1.2	Call diversion	PBX party busy	Pass	
4.6.29.1.3	Call diversion	PBX party does not answer	Pass	
4.6.29.1.4	Call diversion	A-party clears in ring phase	Pass	
4.6.29.2.1	Call diversion	External call to PBX party CD => PBX party	Pass	
4.6.29.2.2	Call diversion	PBX party busy	Pass	
4.6.29.2.3	Call diversion	PBX party does not answer	Pass	
4.6.29.2.4	Call diversion	A-party clears in ring phase	Pass	
4.6.29.3.1	Call diversion	External call to PBX party CD => external PSTN party	Pass	
4.6.29.3.2	Call diversion	PSTN party busy	Pass	
4.6.29.3.3	Call diversion	PSTN party does not answer	Pass	
4.6.29.3.4	Call diversion	A-party clears in ring phase	Pass	
4.7.30	Call waiting	Incoming call during active internal call	Pass	
4.7.30.1	Call waiting	Call waiting tone	Pass	
4.7.30.2	Call waiting	Display of number of waiting party	Pass	
4.7.30.3	Call waiting	Acceptance of waiting call	Pass	
4.7.30.4	Call waiting	Putting the waiting call on hold	Pass	
4.7.30.5	Call waiting	Retrieve of waiting call	Pass	
4.7.30.6	Call waiting	Holding the 2nd call	Pass	
4.7.30.7	Call waiting	Terminating the active call	Pass	
4.7.30.8	Call waiting	Disregarding the waiting call	Pass	

4.7.30.9	Call waiting	Rejecting the waiting call	Pass	
4.8.31	3-party conference	Establishing a conference according to the operating instructions of the PBX Internal - internal - external	Pass	
4.8.31.1	3-party conference	Selecting a party (internal or external); 3rd party is put on hold • Switching to 3rd party; 2nd party is put on hold • Reactivating the conference • Clearing one party (internal or external) • Terminating the conference	Pass	
4.8.32	3-party conference	Establishing a conference according to the operating instructions of the PBX internal - external - external	Pass	
4.8.32.1	3-party conference	Selecting a party (external); 3rd party is put on hold • Switching to 3rd party; 2nd party is put on hold • Reactivating the conference • Clearing one party (external) • Terminating the conference	Pass	
4.9.33	Pick up	Picking up a call from another extension of the PBX	Pass	
4.10.34	Call list	Entries in the call list • Incoming from PSTN • Incoming from mobile • Incoming CLIR • Dialing prefix for outside line in the call list	Pass	
4.10.35	Call list	Call-back from call list • To PSTN • To mobile	Pass	
4.11.36	DTMF	DTMF support (G.711) • PSTN => PBX (SIP terminal device) • PSTN => PBX (analog or system terminal device) • PBX (SIP terminal device) => PSTN • PBX (analog or system terminal device) => PSTN	Pass	
4.11.37	DTMF	DTMF support (G.729) - PSTN => PBX (SIP terminal device) - PSTN => PBX (analog or system terminal device) - PBX (SIP terminal device) => PSTN - PBX (analog or system terminal device) => PSTN	Pass	
4.12.38	Fax	Fax reception (G.711 only)	Pass	
4.12.38.1	Fax	Network-side T.38 re-invite rejected by PBX (response 488) or only G.711 codec is confirmed • One-page fax • Multi-page fax (at least 5 pages)	Pass	
4.12.39	Fax	Fax sending (G.711 only)	Pass	
4.12.39.1	Fax	Network-side T.38 re-invite rejected by PBX (response 488) or only G.711 codec is confirmed • One-page fax • Multi-page fax (at least 5 pages)	Pass	
4.12.40	Fax	Fax reception via T.38	Pass	
4.12.40.1	Fax	Re-invite to T.38 by PBX or network - One-page fax - Multi-page fax (at least 5 pages)	Pass	
4.12.41	Fax	Fax sending via T.38 (not possible in conjunction with the encryption option)	Pass	

4.12.41.1	Fax	Re-invite to T.38 by PBX or network • T.38-only invites are not supported • One-page fax • Multi-page fax (at least 5 pages)	Pass	
5.42	Redundancy	Test of redundancy: • Only if redundant connection is possible on the PBX side • This requires at least two PBX servers to be online on the SIP trunk in registration mode, or exactly two PBX servers in peering mode.	Pass	
5.42.1	Redundancy	1. Register all available PBX systems the Plusnet SBC. 2. Calls from Plusnet => PBX are routed by round robin procedure 3. Deliberately de-register one PBX system => no more calls are routed to this PBX 4. and/or disconnect the PBX system from LAN => after the register expire period, no more calls are routed to this PBX. 5. Calls are only routed to the remaining PBX systems.	Pass	

Conclusion

This Application Notes document describes the steps required to configure the Ribbon SBC 1000/2000 to successfully interoperate with the Cisco CUCM and PlusNet SIP Trunk. All feature and serviceability test cases have been completed. The majority of test cases passed with noted exceptions and observations provided in Interoperability Test Results.