Ribbon SBC Edge 8.0.0 IOT Skype for Business 2015 Colt Application Notes

Table of Contents

Document Overview

This document provides a configuration guide for Ribbon SBC Edge Series (Session Border Controller) when connecting to Skype for Business 2015 and Colt SIP Trunking.

This configuration guide supports features described on the Microsoft Technet https://technet.microsoft.com/ web site.

- For additional information on Skype for Business 2015, please visit http://microsoft.com
- For additional information on Ribbon SBC Edge Series (Session Border Controller), please visit https://ribboncommunications.com/

Introduction

The interoperability compliance testing focuses on verifying inbound and outbound calls flows between Ribbon SBC Edge and Skype for Business 2015.

Audience

This is a technical document intended for telecommunications engineers with the purpose of configuring both the Ribbon SBC and the third-party product. There will be steps that require navigating third-party as well as the Ribbon SBC Command Line Interface (CLI). Understanding the basic concepts of TCP/UDP, IP/Routing, and SIP/RTP are also necessary to complete the configuration and for troubleshooting, if necessary.

> This configuration guide is offered as a convenience to Ribbon customers. The specifications and information regarding the product in this guide are subject to change without notice. All statements, information, and recommendations in this guide 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 in this guide.

Requirements

The following equipment and software were used for the sample configuration provided:

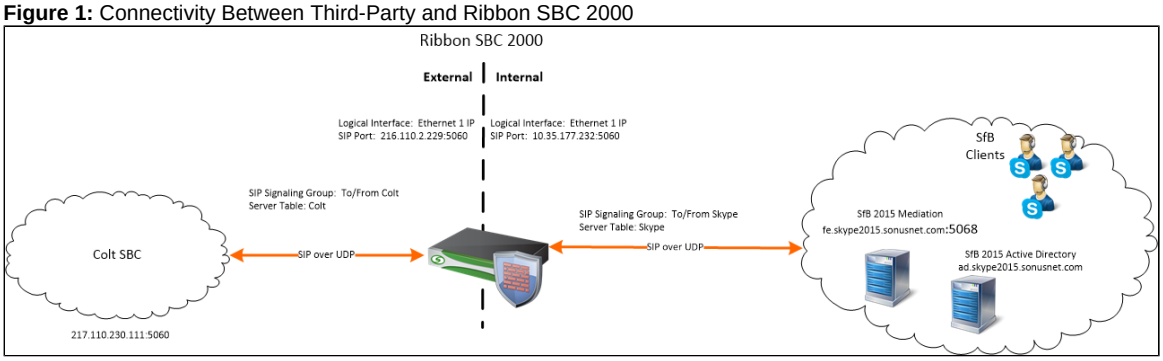
Table 1: Requirements

	Equipment	Software Version
Ribbon	SBC 2000	V8.0.0build502
	Tenor AFM200	P108-09-26
Third-party Equipment	Microsoft Skype for Business 2015 Mediation Server	6.0.9319.0
	Polycom CX600 SIP Phone	4.0.7577.44455
	VentaFax	7.6.243.597 I

Reference Configuration

The following reference configuration shows connectivity between third-party and Ribbon SBC Edge.

Figure 1: Connectivity Between Third-Party and Ribbon SBC 2000



Support

For any questions regarding this document or the content herein, please contact your maintenance and support provider.

Third-Party Product Features

The testing was executed with the Colt test plan. The following features were tested:

- Basic Calls
- Enhanced Calls
- Codec Support
- DTMF Support
- CLI Services
- Encryption

Verify License

SIP Calls

Skype for Business 2015 Configuration

The following new configurations are included in this section:

1. PSTN Gateway
2. Voice Policy
3. PSTN Usage
4. Route
5. Trunk Configuration

1. PSTN Gateway

Topology Builder > **Shared Components** > **PSTN Gateways**

Figure 2: Define a new IP/PSTN Gateway

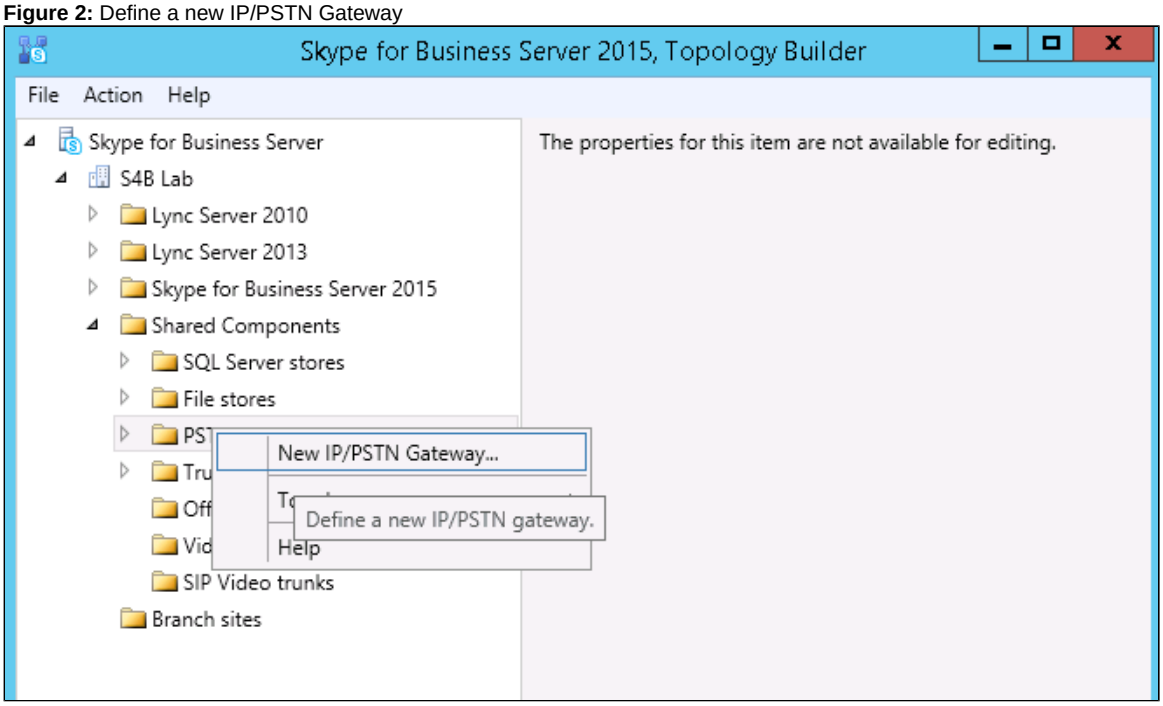


Figure 3: Define FQDN

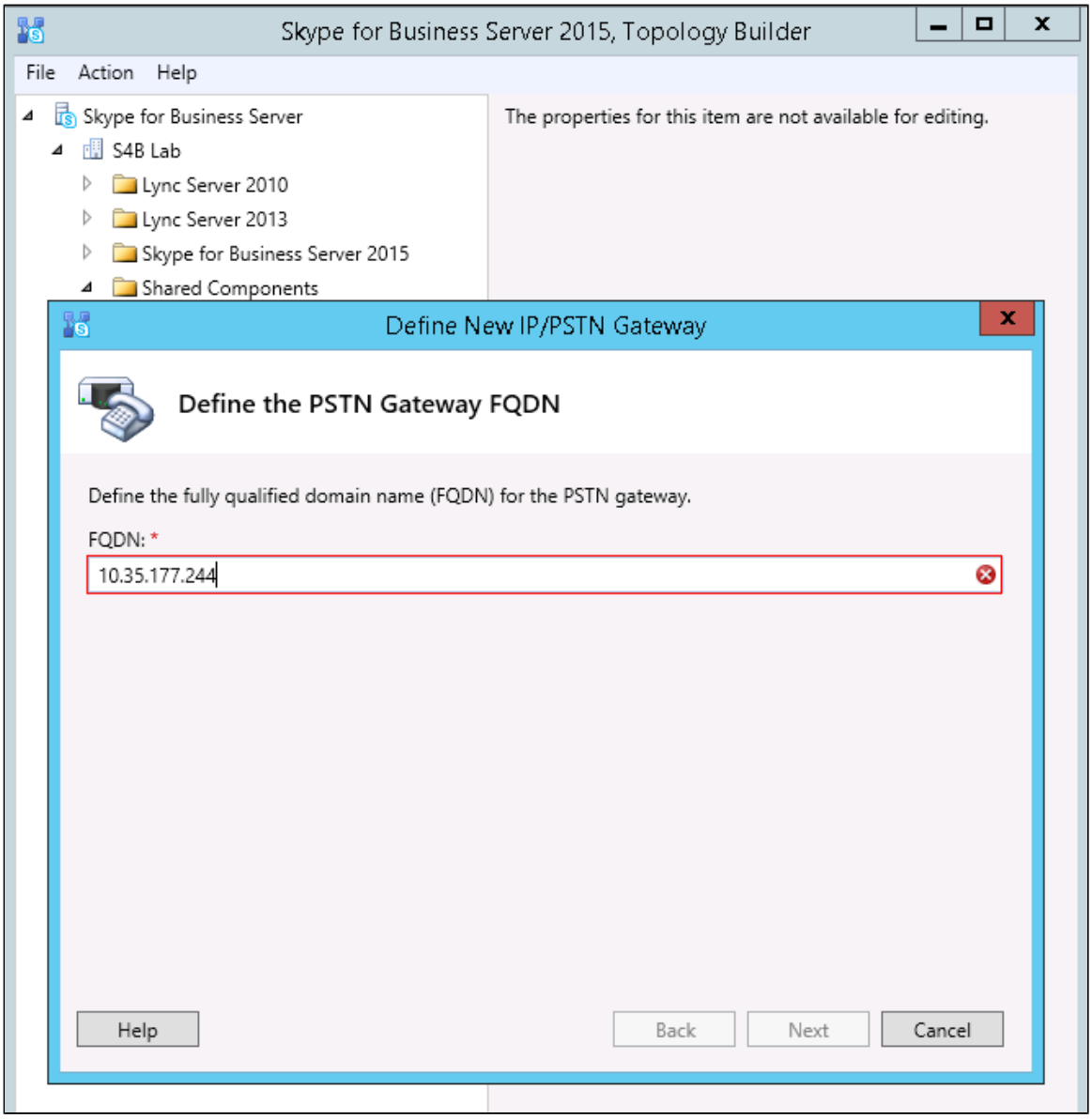


Figure 4: Define IP Address

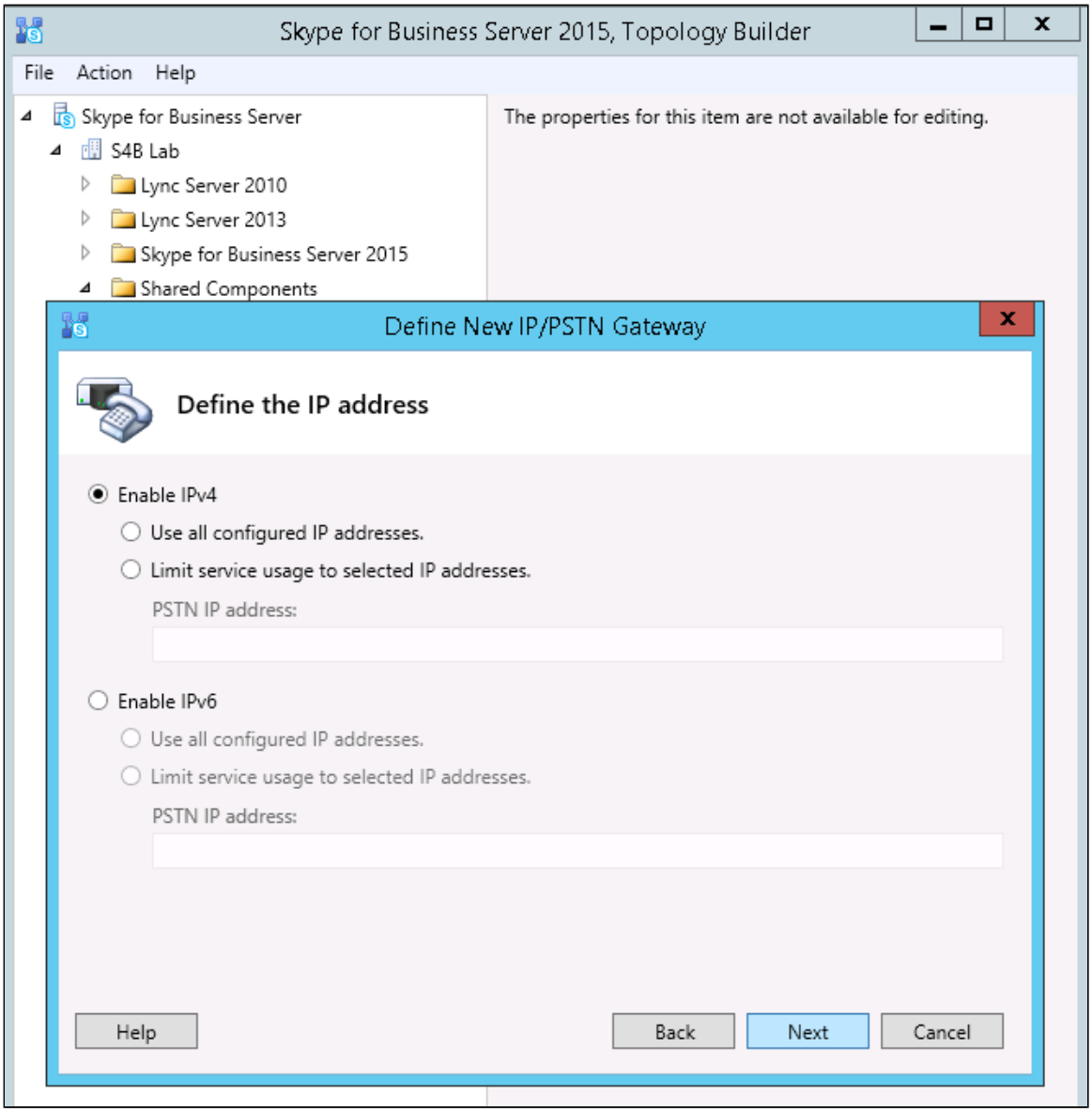
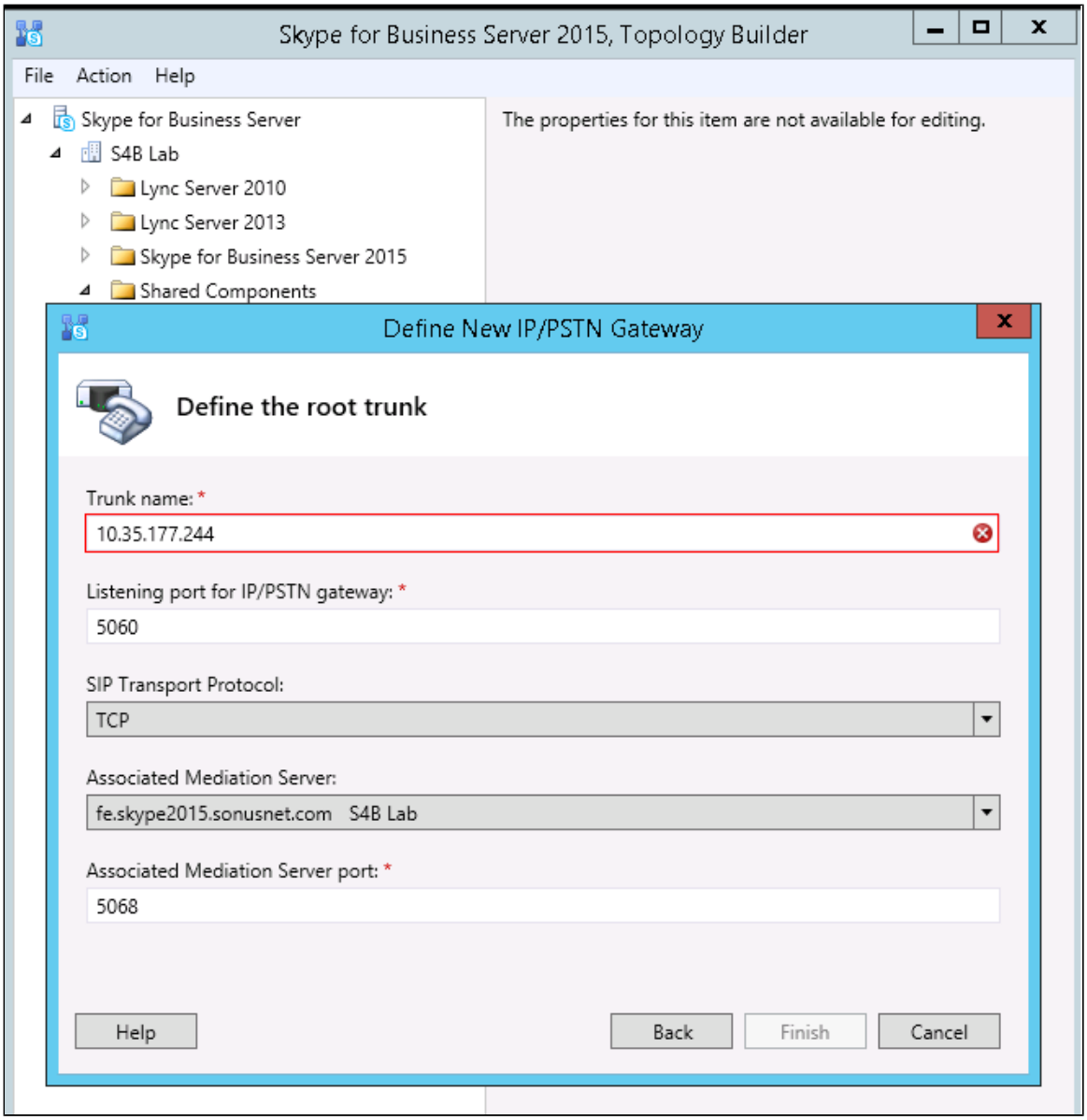


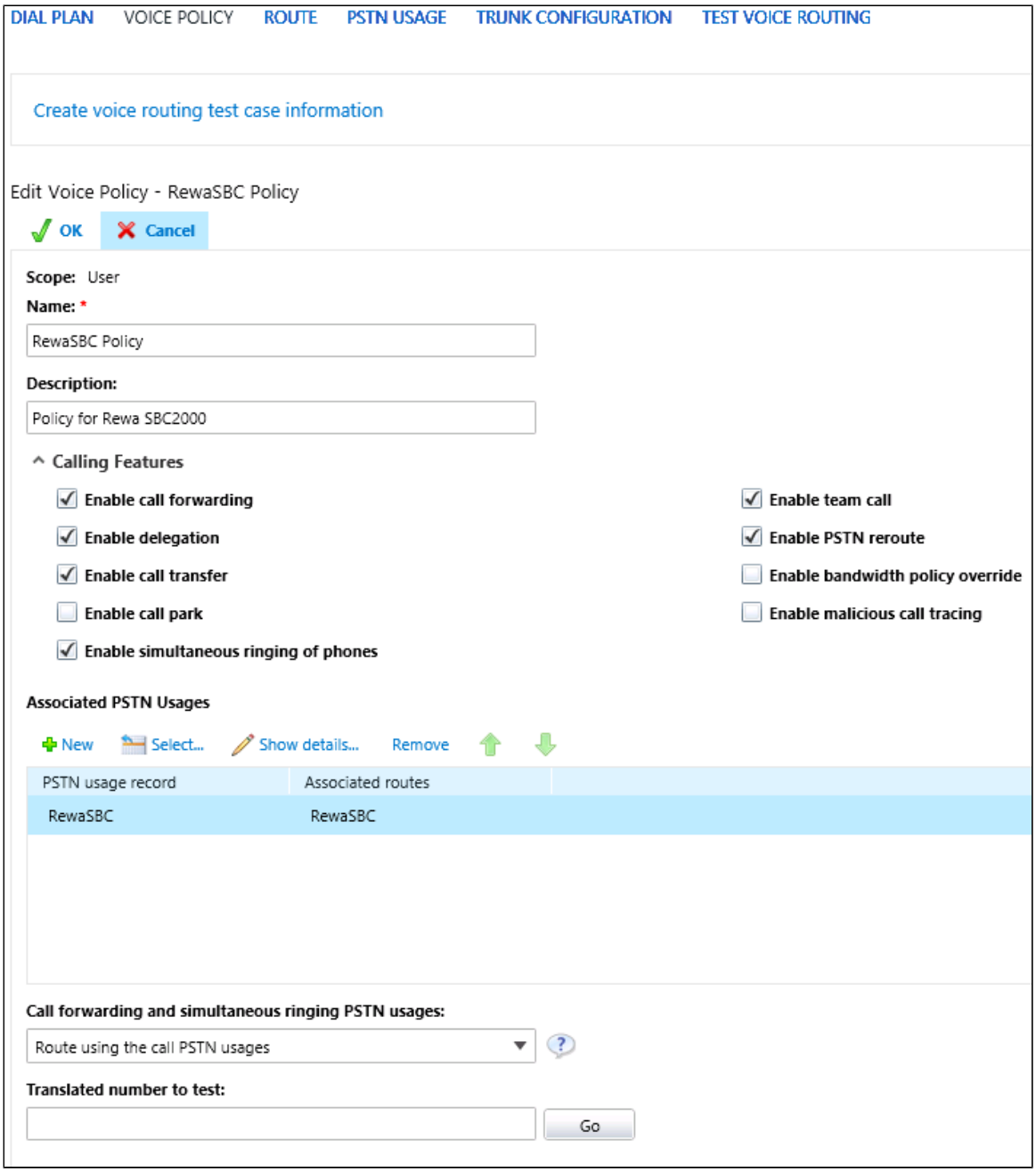
Figure 5: Define Root Trunk

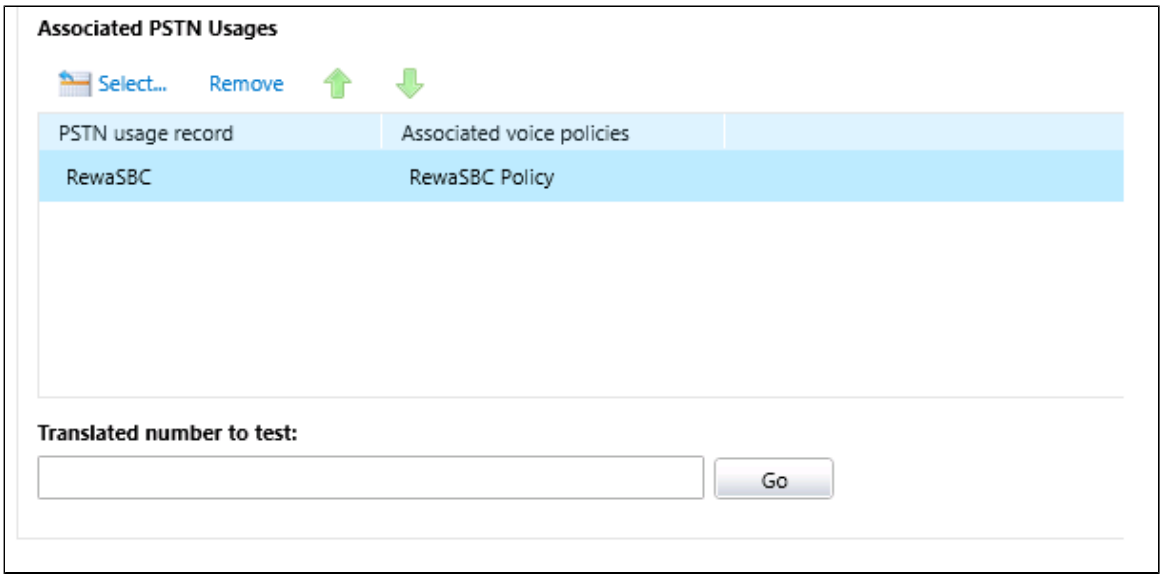


2. Voice Policy

Control Panel > **Voice Routing** > **Voice Policy**

Figure 6: Edit Voice Policy

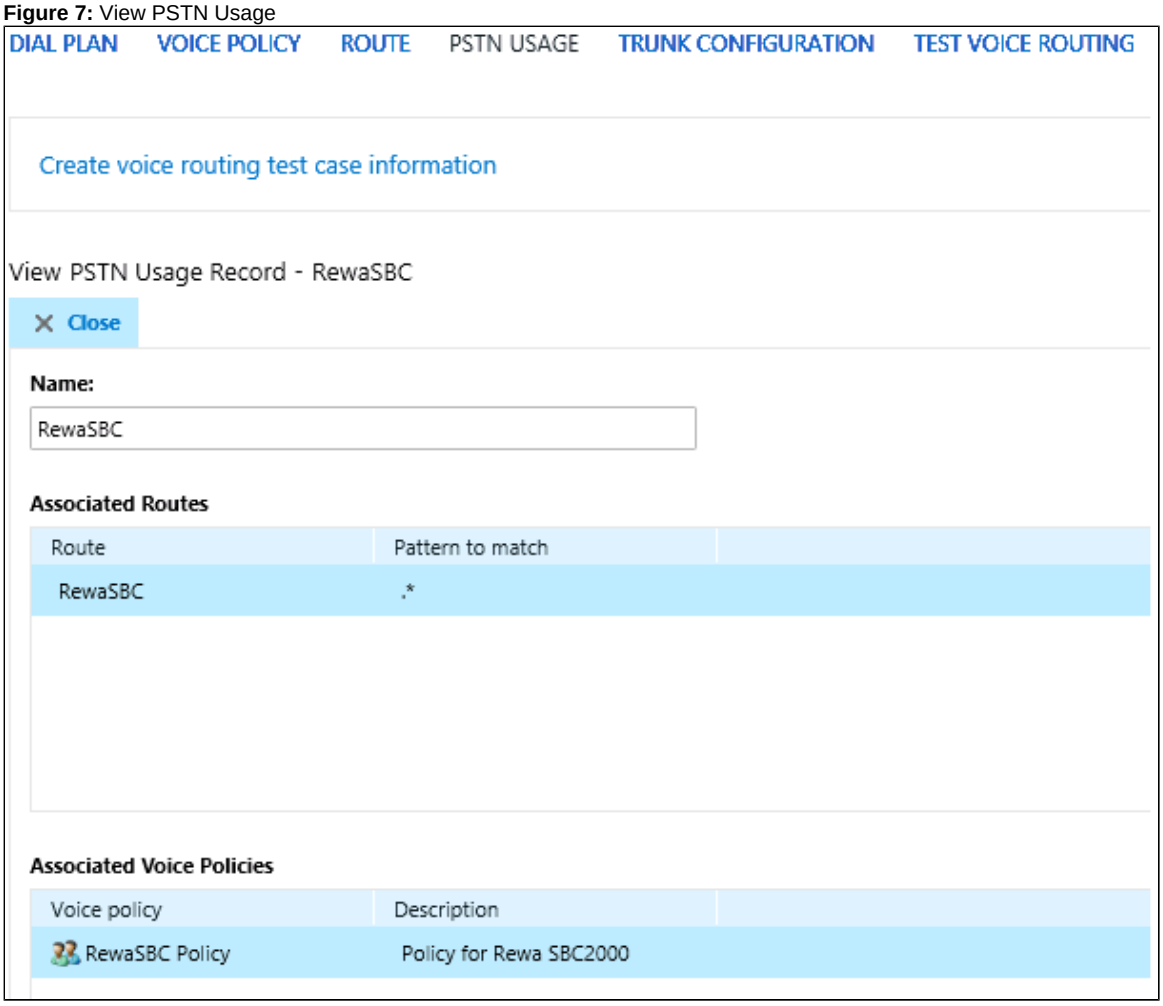




3. PSTN Usage

Control Panel > **Voice Routing** > **PSTN Usage**

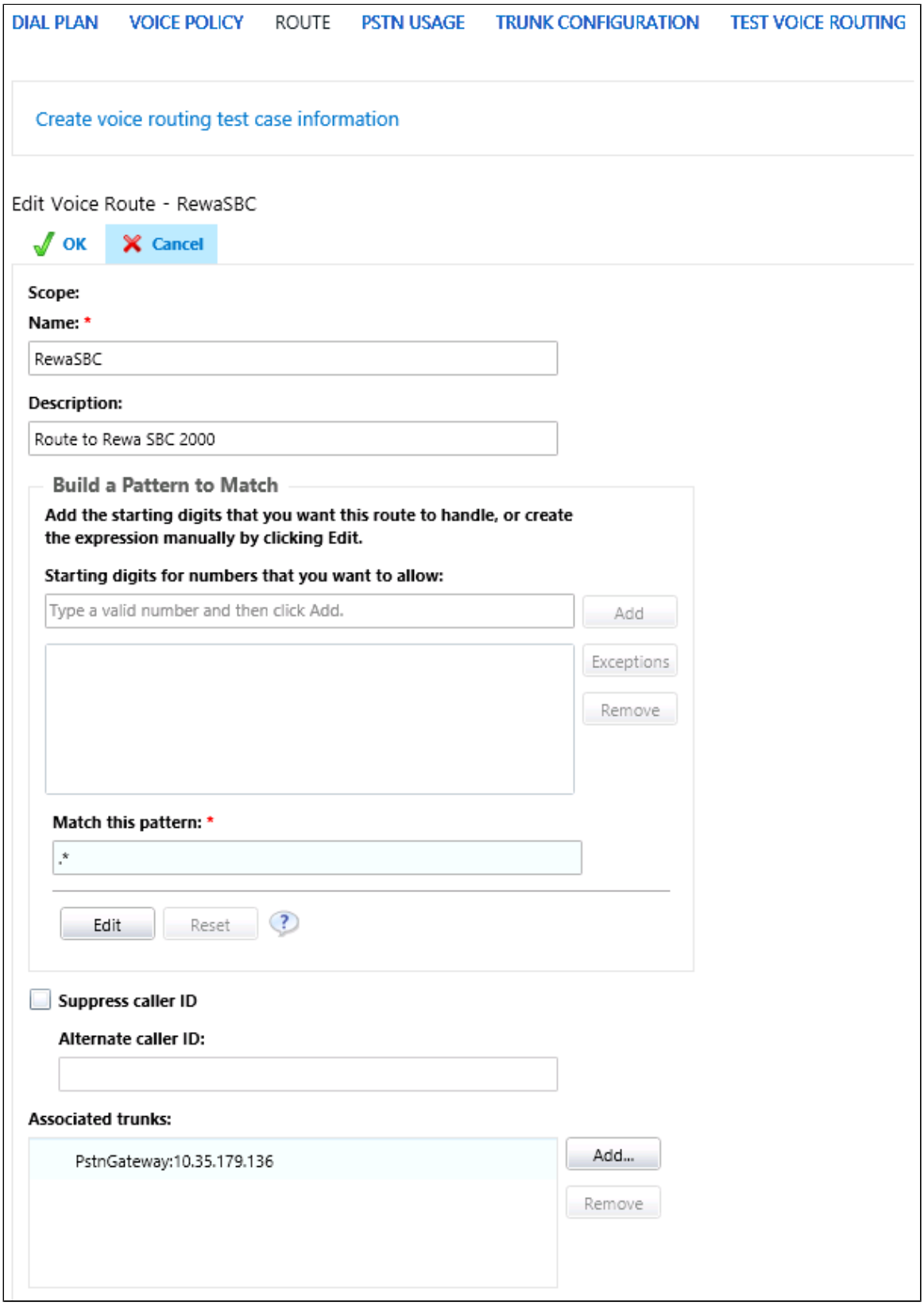
Figure 7: View PSTN Usage

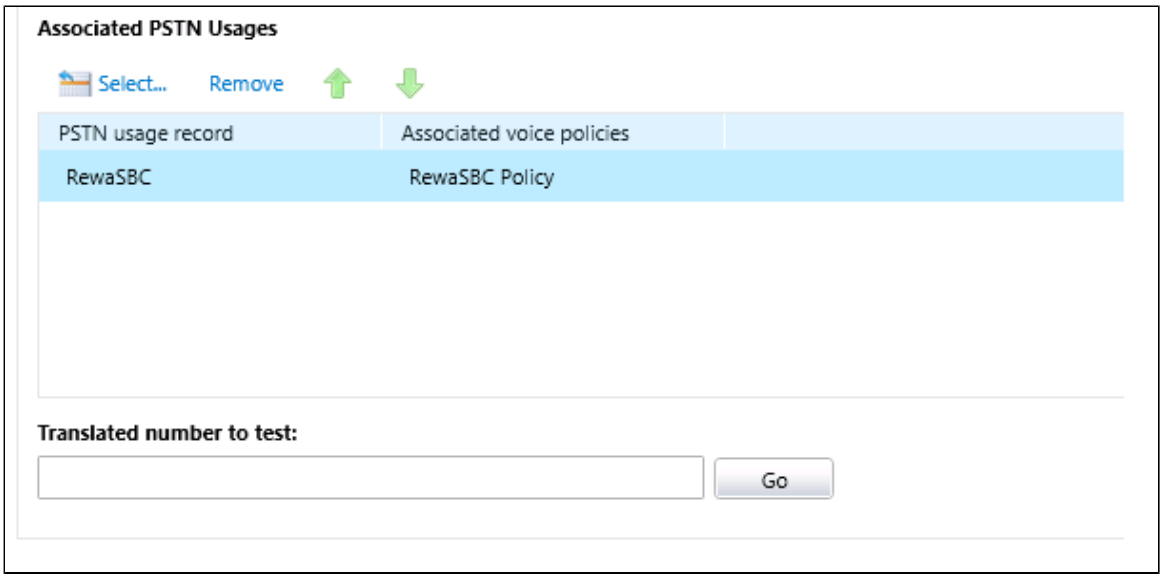


4. Route

Control Panel > **Voice Routing** > **Route**

Figure 8: Edit Voice Route

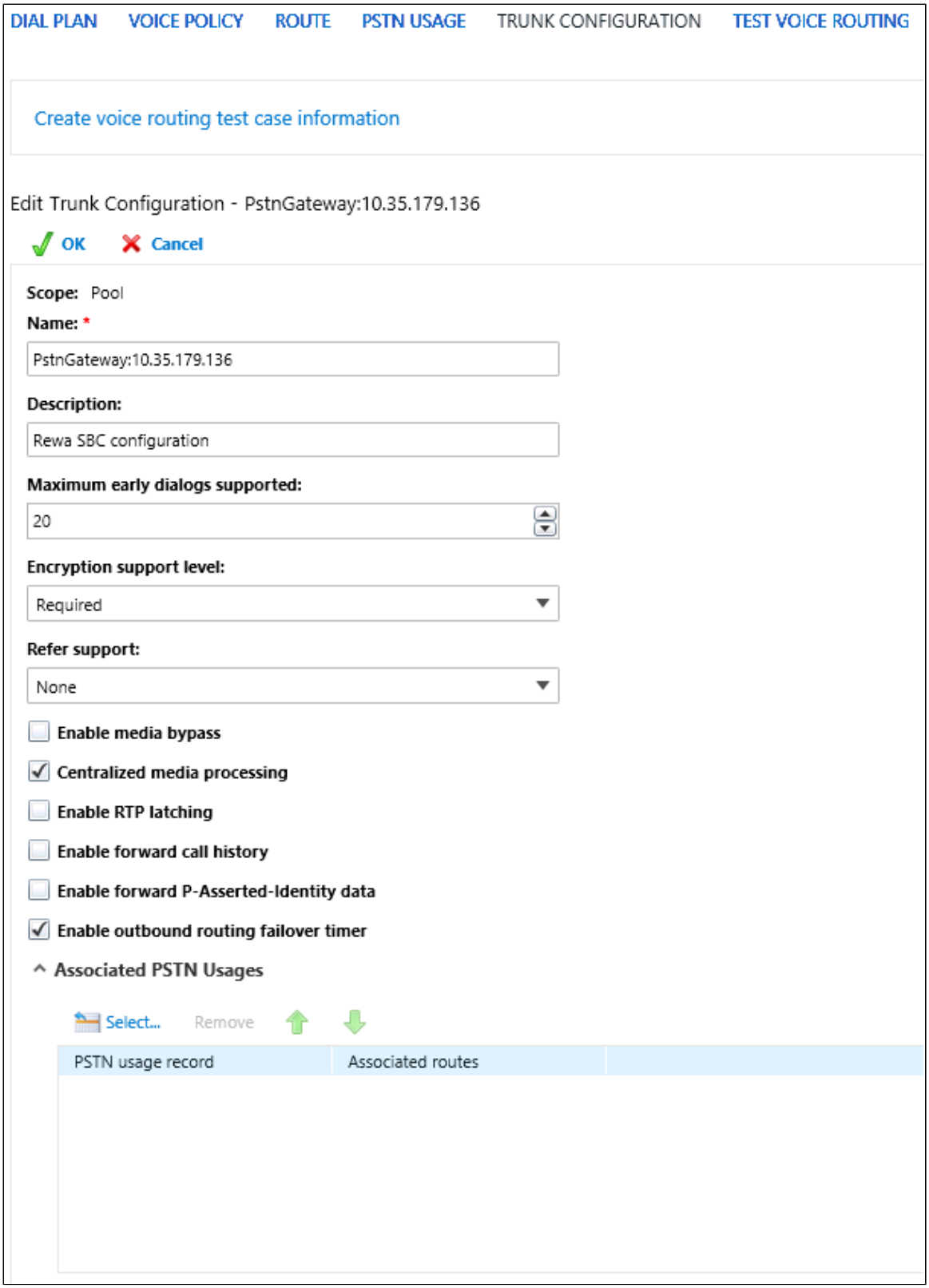




5. Trunk Configuration

Control Panel > **Voice Routing** > **Trunk Configuration**

Figure 9: Edit Trunk Configuration



Ribbon SBC 1000/2000 Configuration

The following steps provide an example of how to configure Ribbon SBC 1000/2000:

1. SIP Profile
2. SIP Server

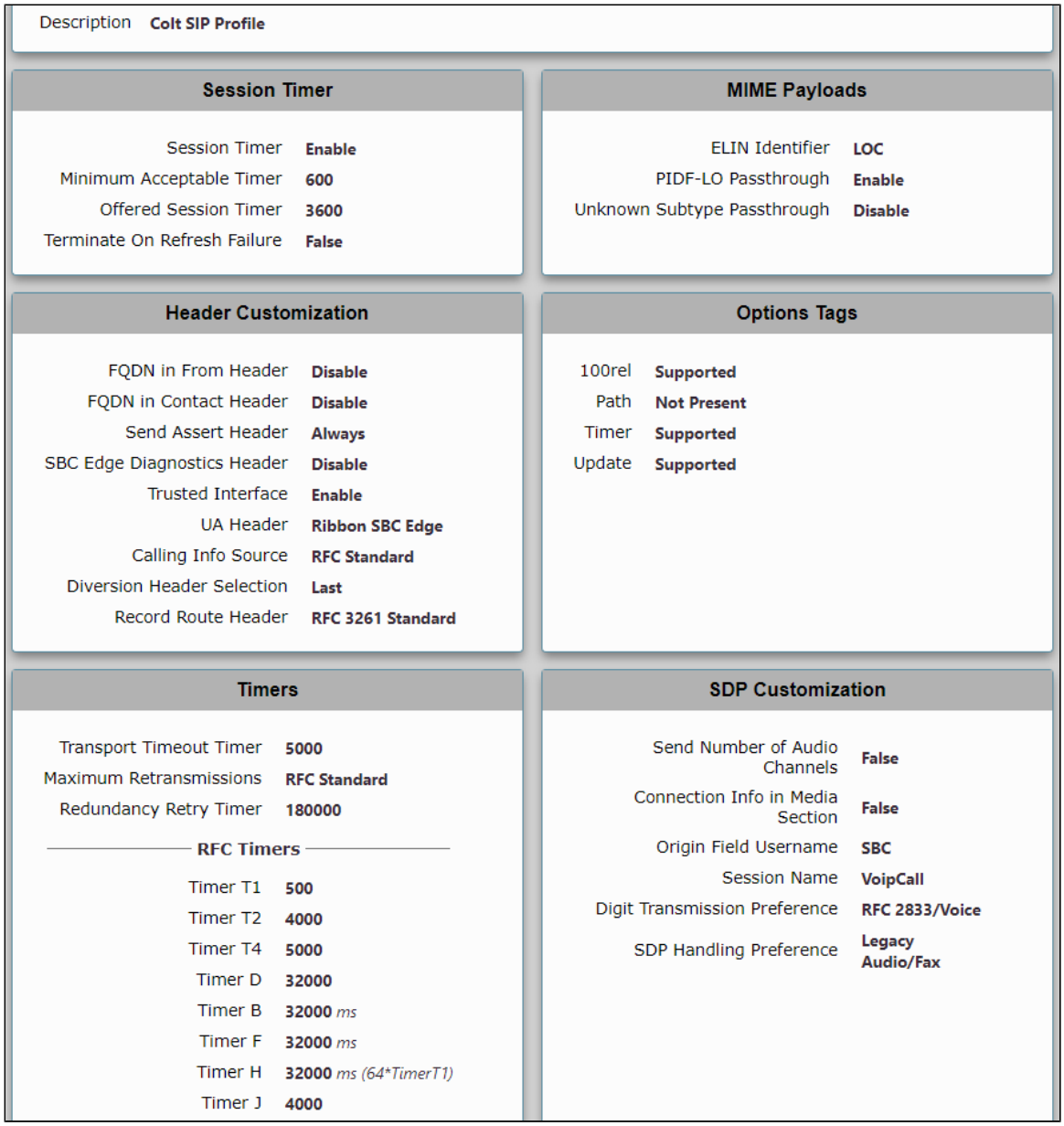
3. Media Profile
4. Media List
5. Transformation Table
6. Call Routing Table
7. Message Rule Tables
8. Signaling Groups

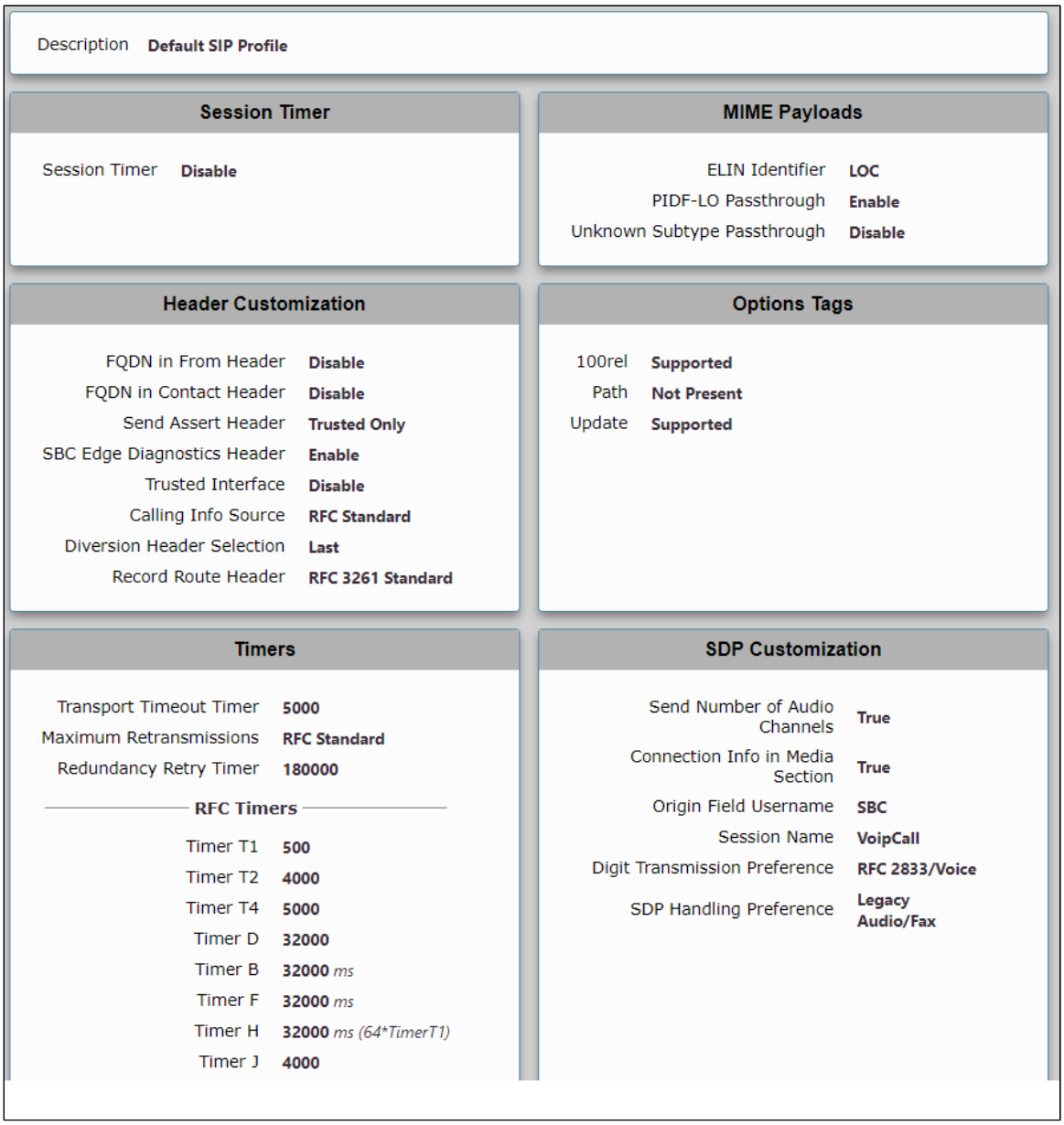
1. SIP Profile

Select Settings > **SIP** > **SIP Profiles**

SIP Profiles control how the Ribbon SBC 1000/2000 communicates with SIP devices. These control important characteristics such as session timers, SIP Header customization, SIP timers, MIME payloads, and option tags. The following figures shows the default SIP profile used for the SBC 1000 /2000 for this testing effort:

Figure 10: SIP Profiles





2. SIP Server

Select Settings > **SIP** > **SIP Server Tables**

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

Figure 11: Skype

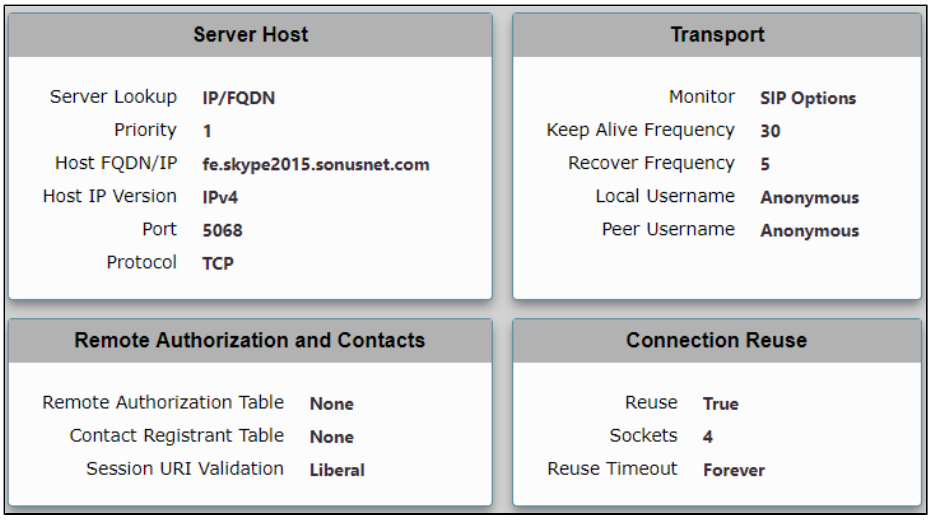


Figure 12: Fax

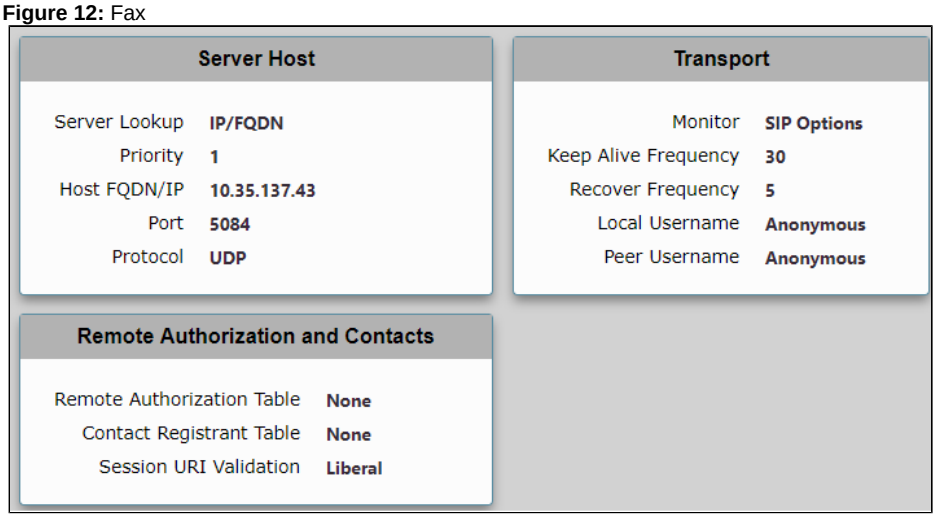
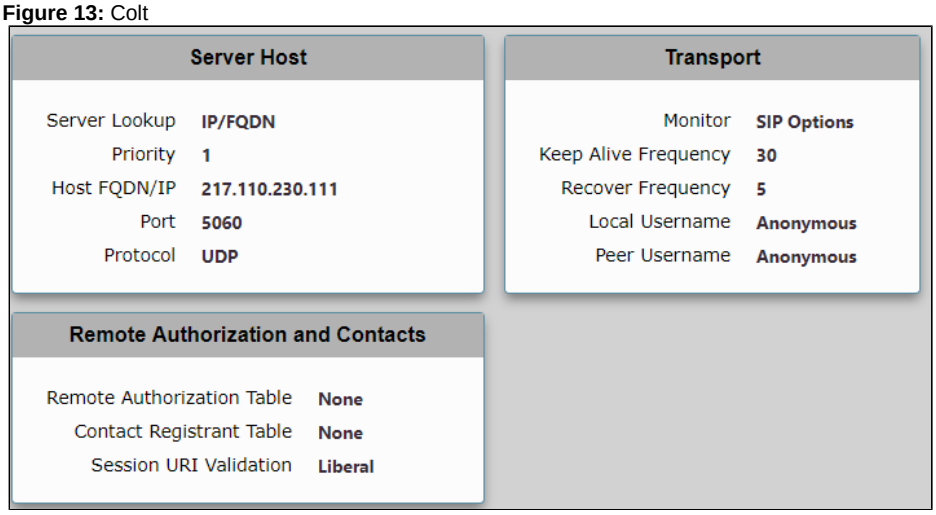


Figure 13: Colt



3. Media Profile

Select Settings > **Media** > **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. The following figures are the media profiles of the voice codecs used for the SBC 1000/2000 in this testing effort and are shown for reference only:

Figure 14: Voice Codec G711 A-Law

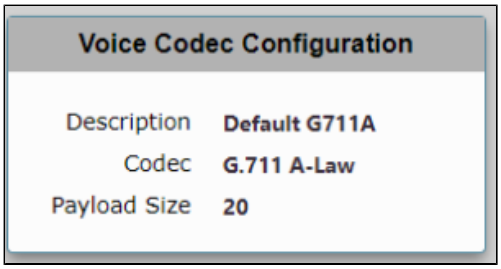


Figure 15: Voice Codec G711 U-Law

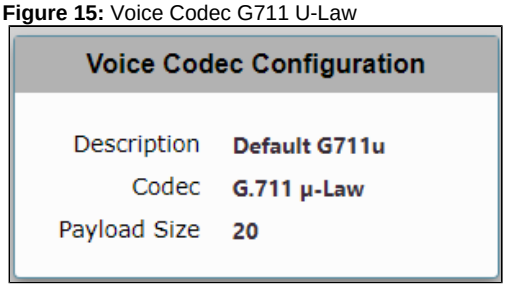


Figure 16: Voice Codec G729

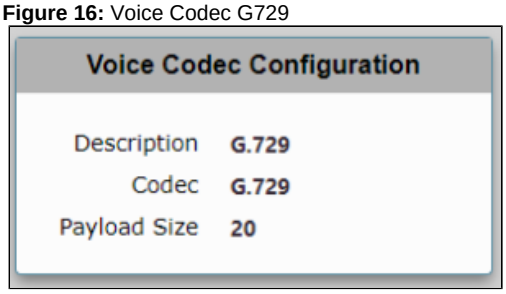
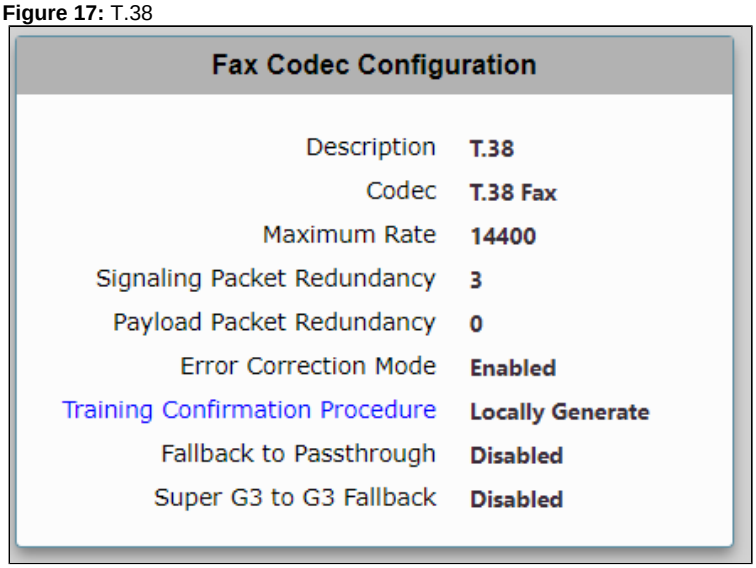


Figure 17: T.38



4. Media List

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

Figure 18: Media List

Description	**Skype Media List**
Media Profiles List	Default G711A *
SDES-SRTP Profile	**None**
DTLS-SRTP Profile	**None**
Media DSCP	**46**
RTCP Mode	**RTCP**
Dead Call Detection	**Disabled**
Silence Suppression	**Enabled**

Gain Control

Receive Gain	**0**
Transmit Gain	**0**

Digit Relay

Digit (DTMF) Relay Type	**RFC 2833**
Digit Relay Payload Type	**101**

Passthrough/Tone Detection

Modem Passthrough	**Enabled**
Fax Passthrough	**Enabled**
CNG Tone Detection	**Disabled**
Fax Tone Detection	**Enabled**
DTMF Signal to Noise	**0**
DTMF Minimum Level	**-38**

Description	**Fax Media List**
Media Profiles List	Default G711A Default G711u G.729 T.38 *
SDES-SRTP Profile	**None**
DTLS-SRTP Profile	**None**
Media DSCP	**46**
RTCP Mode	**RTCP**
Dead Call Detection	**Disabled**
Silence Suppression	**Disabled**

Gain Control

Receive Gain	**0**
Transmit Gain	**0**

Digit Relay

Digit (DTMF) Relay Type	**RFC 2833**
Digit Relay Payload Type	**101**

Passthrough/Tone Detection

Modem Passthrough	**Enabled**
Fax Passthrough	**Enabled**
CNG Tone Detection	**Enabled**
Fax Tone Detection	**Enabled**
DTMF Signal to Noise	**0**
DTMF Minimum Level	**-38**

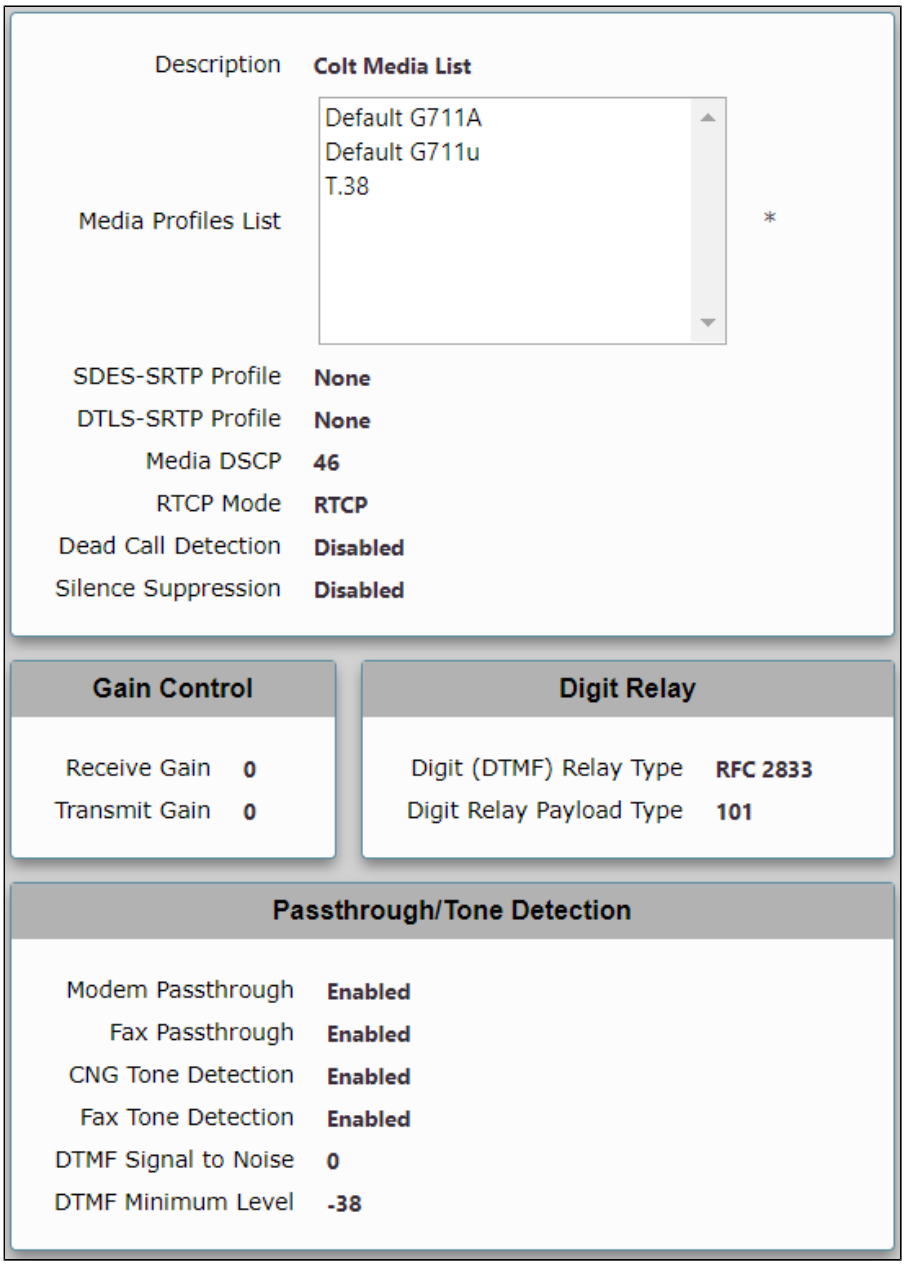
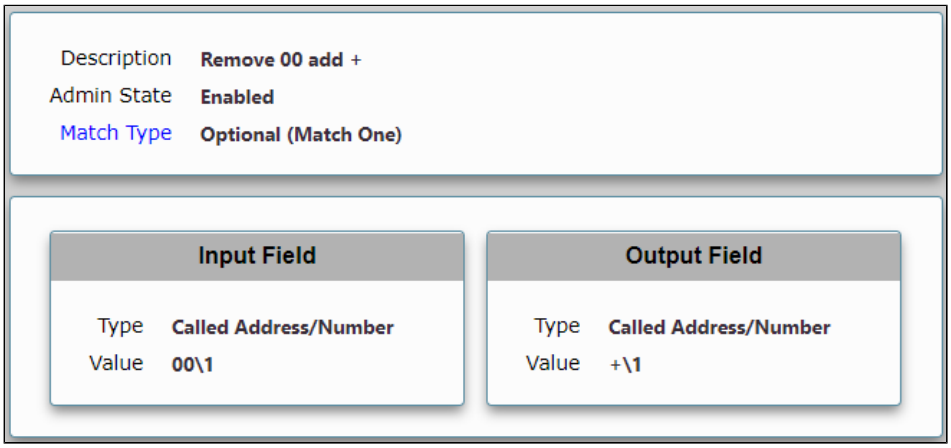
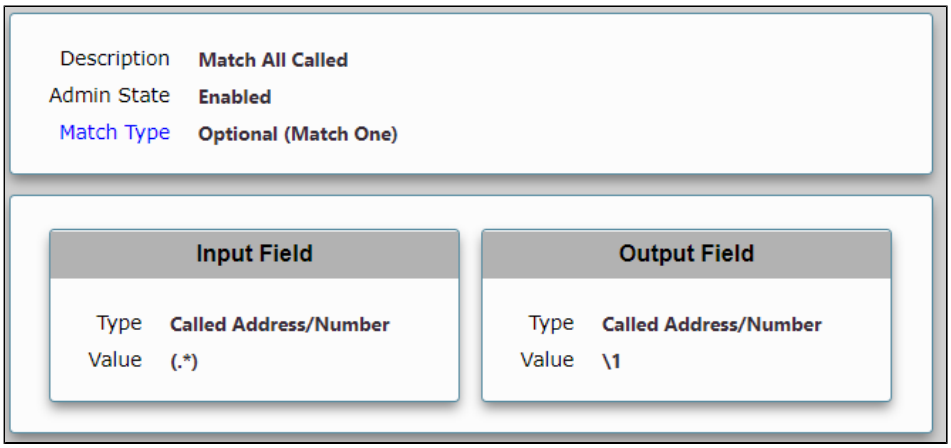
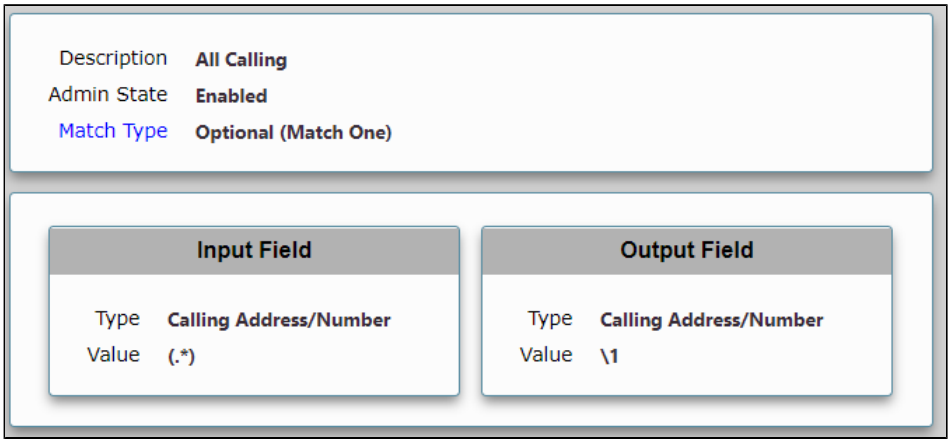


Figure 19: From Skype







6. Call Routing Table

Select Settings > **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). Routes are defined by Call Routing Tables, which allow 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).

Figure 20: From Colt

Route Details

Description	**From Colt to Fax**
Admin State	**Enabled**
Route Priority	**1**
Call Priority	**Normal**
Number/Name Transformation Table	**From Colt to 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	**Not Licensed**
Destination Signaling Groups	(SIP) To/From Fax *
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**
Quality Metrics Time Before Retry	**10**
Min. ASR Threshold	**0**
Enable Min MOS Threshold	**Disabled**
Enable Max. R/T Delay	**Enabled**
Max. R/T Delay	**65535**
Enable Max. Jitter	**Enabled**
Max. Jitter	**3000**

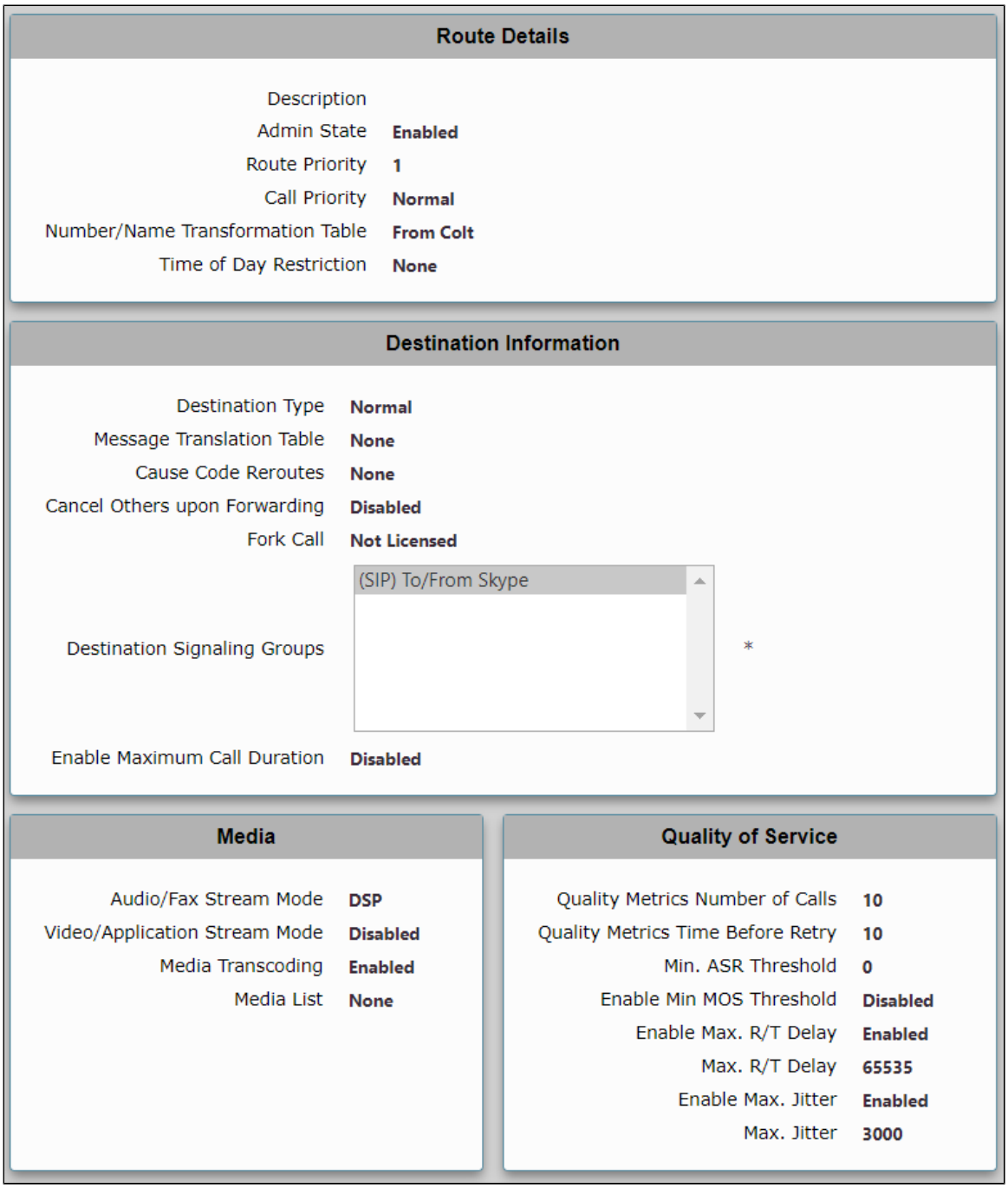


Figure 21: From Skype to Colt

Route Details

Description	
Admin State	**Enabled**
Route Priority	**1**
Call Priority	**Normal**
Number/Name Transformation Table	**From Skype to Colt**
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	**Not Licensed**
Destination Signaling Groups	(SIP) To/From Colt *
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**
Quality Metrics Time Before Retry	**10**
Min. ASR Threshold	**0**
Enable Min MOS Threshold	**Disabled**
Enable Max. R/T Delay	**Enabled**
Max. R/T Delay	**65535**
Enable Max. Jitter	**Enabled**
Max. Jitter	**3000**

7. Message Rule Tables

Select Settings > **Message Manipulation** > **Message Rule Tables**

Message Rule Tables are sets of Condition Rules and are applied in SIP Signaling Groups when Message Manipulation is enabled.

Figure 22: Colt Outbound

Description	Replace NEWPORT.lync2013.sonusnet.com with 216.110.2.229
Condition Expression	
Admin State	Enabled
Result Type	Optional

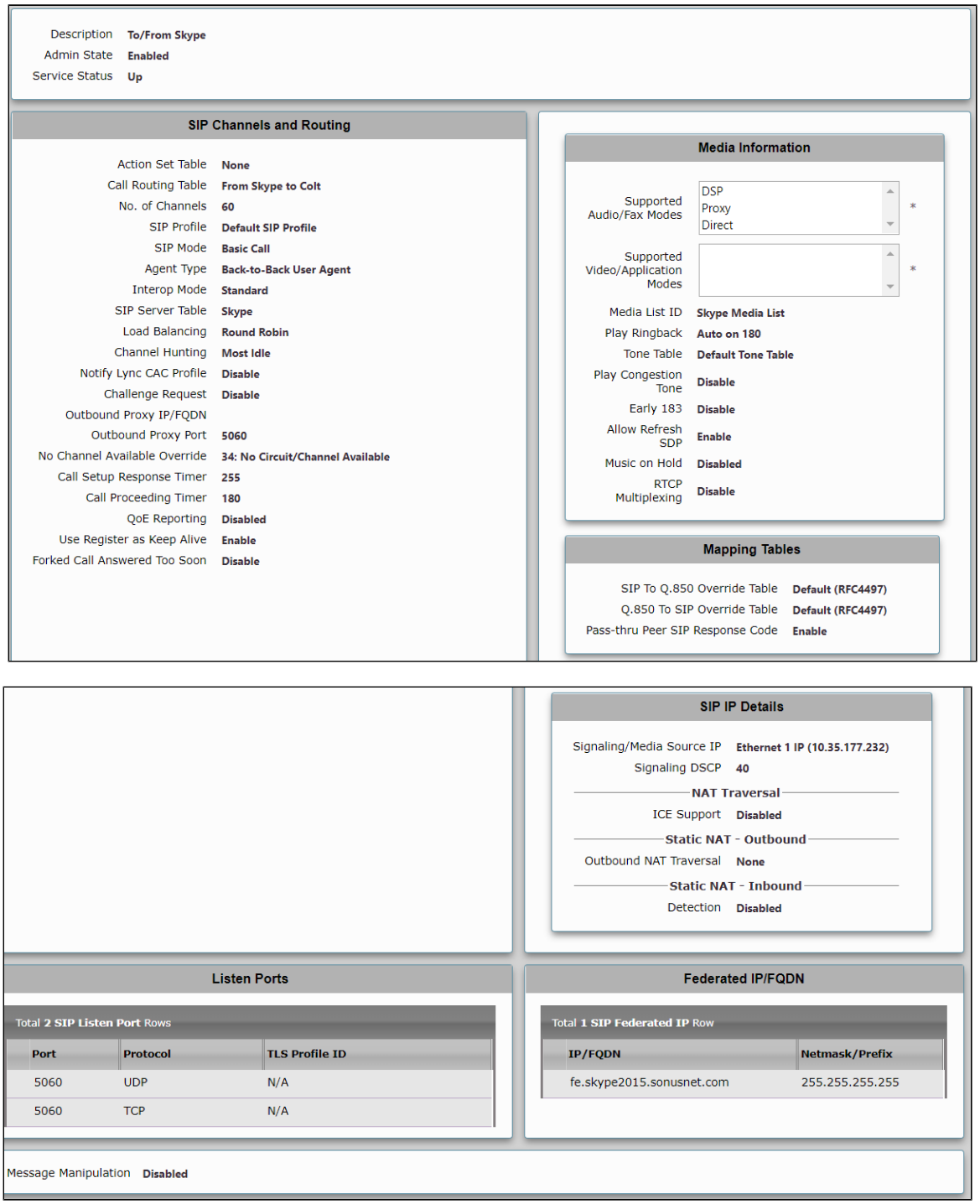
Match Regex	NEWPORT.lync2013.sonusnet.com *
Replace Regex	216.110.2.229 *

8. Signaling Groups

Select Settings > **Signaling Groups**

Signaling Groups allow telephony channels to be grouped together for the purposes of routing and shared configuration. These groups are the entity to which calls are routed, as well as 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, this will specify protocol settings and link to server, media, and mapping tables.

Figure 23: Internal Side



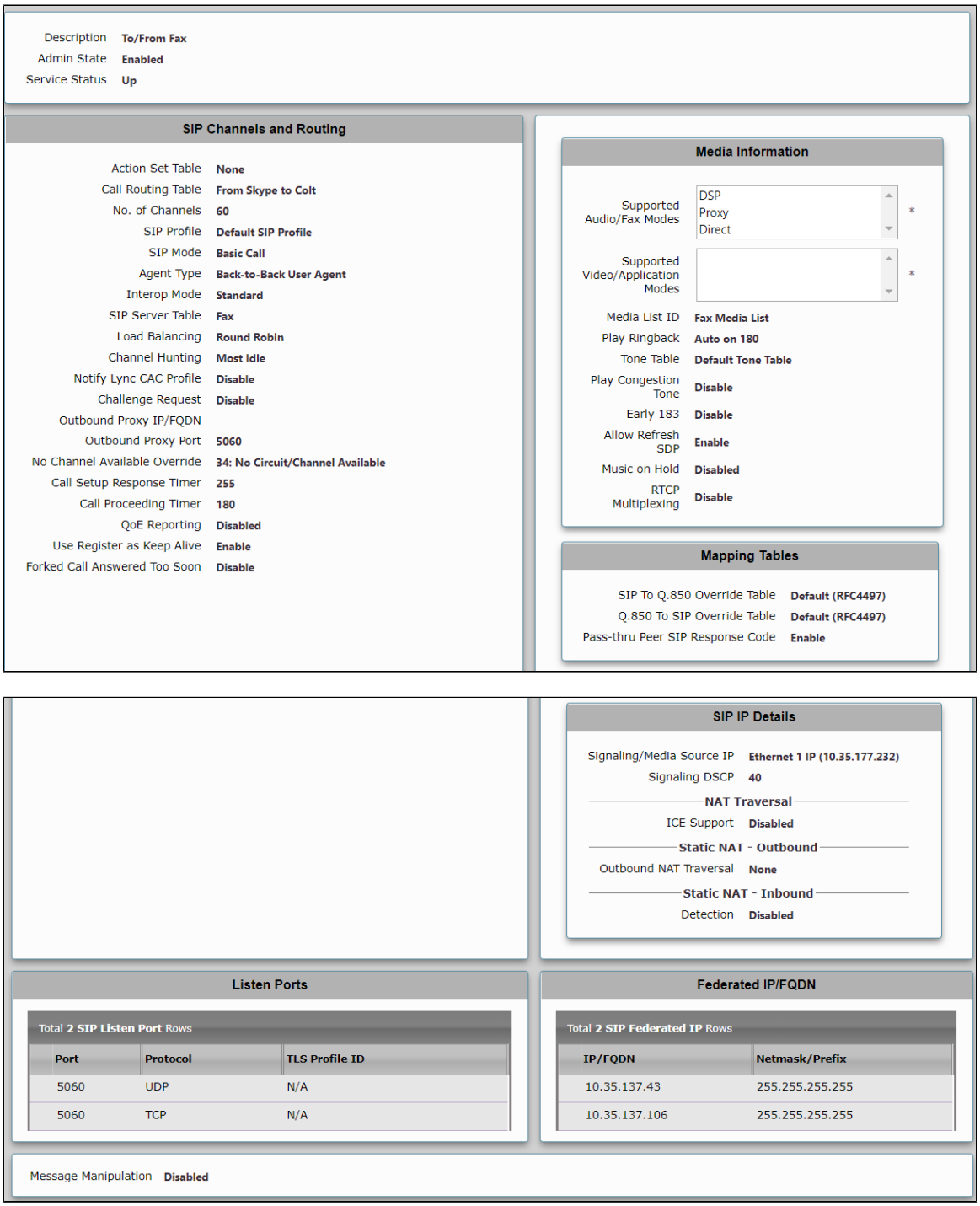
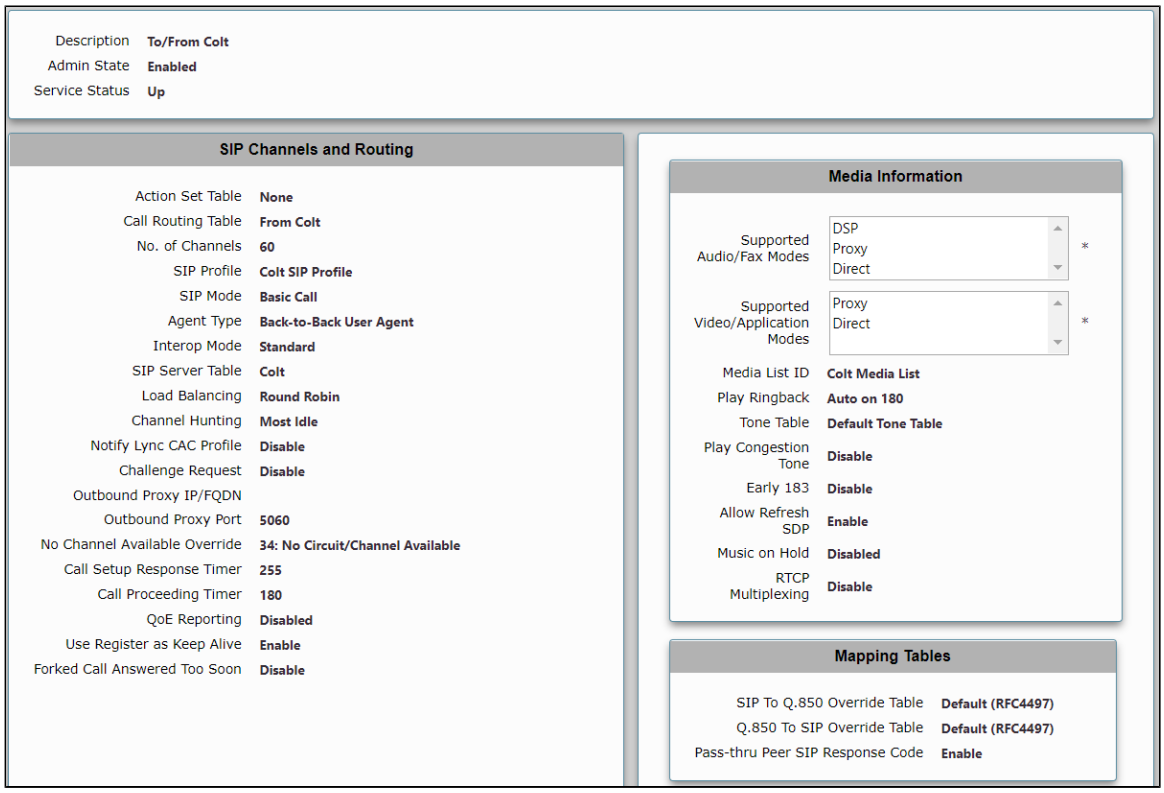
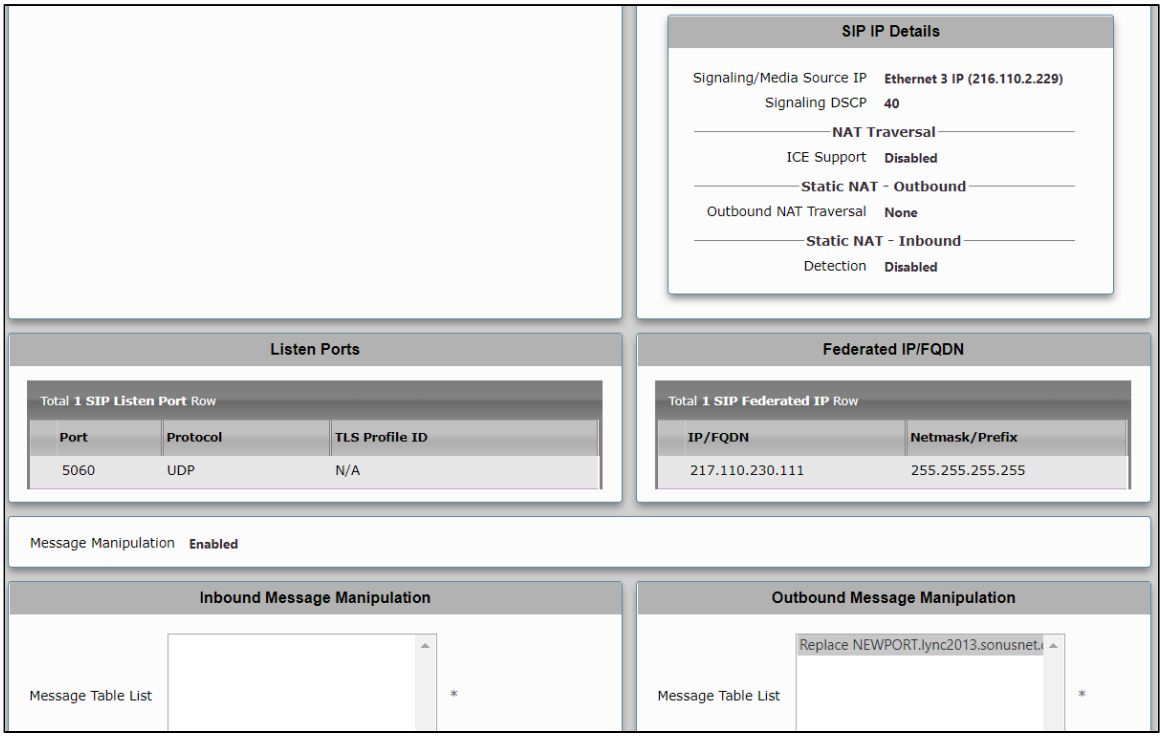


Figure 24: External Side





Test Results

Table 2: Test Results

S. No	Procedure	Observation	Result	Comment

Basic Calls				
1-1	IP Phone to PSTN Phone, IP Phone disconnect after answer		PASS	
1-2	IP Phone to PSTN Phone, PSTN Phone busy		PASS	
1-3	IP Phone to PSTN Phone, PSTN Phone no answer		PASS	
1-4	IP Phone to PSTN Phone, PSTN Phone disconnect after answer		PASS	
1-5	PSTN Phone to IP Phone, PSTN Phone disconnect after answer		PASS	
1-6	PSTN Phone to IP Phone, IP Phone busy		N/A	SfB doesn't send 486 Busy Here
1-7	PSTN Phone to IP Phone, IP Phone no answer		PASS	
1-8	PSTN Phone to IP Phone, IP Phone disconnect after answer		PASS	
1-9	PSTN Phone to IP Phone, network disconnect		PASS	
1-11	IP Phone to International Mobile		PASS	
1-12	IP Phone to International PSTN Phone, remote ringback		PASS	
1-13	IP Phone to PSTN Phone, Long Duration Call		PASS	
1-14	IP Phone to PSTN Phone, Mute both ends of call		PASS	
Enhanced Calls				
2-1	PSTN 1 to IP Phone A1, A1 blind transfers to PSTN 2		PASS	
2-2	PSTN 1 to IP Phone A1, A1 consultative transfers to PSTN 2		PASS	
2-3	PSTN 1 to IP Phone A1, A1 forwards to PSTN 2, Unconditional		PASS	
2-4	PSTN 1 to IP Phone A1, A1 forwards to busy PSTN 2, Unconditional		PASS	
2-5	PSTN 1 to IP Phone A1, A1 forwards to IP Phone A2, Unconditional		PASS	
2-6	PSTN 1 to IP Phone A1, A1 forwards to Mobile, Unconditional		PASS	
2-7	PSTN 1 to IP Phone A1, A1 forwards to PSTN 2, No Answer		PASS	
2-8	PSTN 1 to IP Phone A1, A1 forwards to IP Phone A2, No Answer		PASS	
2-9	PSTN 1 to IP Phone A1, A1 forwards to Mobile, No Answer		PASS	
2-10	IP Phone A1 to PSTN 1, A1 conference to PSTN 2, after answer		PASS	
2-11	IP Phone A1 to PSTN 1, A1 conference to IP Phone A2, after answer		PASS	
2-12	IP Phone A1 to PSTN 1, A1 conference to IP Phone A2, mixed codecs		PASS	
Codec Support				

4-1	IP Phone to PSTN Phone, G.729 codec		PASS	
4-2	IP Phone to PSTN Phone, G.711 alaw codec		PASS	
4-3	PSTN Phone to IP Phone, G.729 codec		PASS	
4-4	PSTN Phone to IP Phone, G.711 alaw codec		PASS	
4-5	IP Phone to PSTN Phone, G.726 32K codec		PASS	
4-6	PSTN Phone to IP Phone, G.726 32K codec		PASS	
4-7	IP Phone to PSTN Phone, G.711 Ulaw codec		PASS	
4-8	PSTN Phone to IP Phone, G.711 Ulaw codec		PASS	
4-9	IP Phone to PSTN Phone, iLBC codec		NOT SUPPORTED	iLBC not supported by both SBC and SfB
4-10	PSTN Phone to IP Phone, iLBC codec		NOT SUPPORTED	iLBC not supported by both SBC and SfB
4-11	IP Phone to IP Phone, G.722 codec		NOT SUPPORTED	SfB uses SILK for peer to peer if no bandwidth limitations are applied or detected, otherwise will use RTA for low bandwidth
DTMF Support				
5-1	IP Phone to PSTN Phone, DTMF using RFC2833		PASS	
5-2	PSTN Phone to IP Phone, DTMF using RFC2833		PASS	
5-3	IP Phone to PSTN Phone, DTMF using H.245 Signal		NOT SUPPORTED	SfB doesn’t support it
5-4	PSTN Phone to IP Phone, DTMF using H.245 Signal		NOT SUPPORTED	SfB doesn’t support it
5-5	IP Phone to PSTN Phone, DTMF using H.245 Alphanumeric		NOT SUPPORTED	SfB doesn’t support it
5-6	PSTN Phone to IP Phone, DTMF using H.245 Alphanumeric		NOT SUPPORTED	SfB doesn’t support it
5-7	IP Phone to PSTN Phone, DTMF Before Answer		PASS	
CLI Services				
6-1	Caller ID Presentation (CLIP) with No Screening		PASS	
6-2	Caller ID Presentation (CLIP) Screening with Correct CLI		PASS	
6-3	Caller ID Presentation (CLIP) Screening with Incorrect CLI		PASS	
6-4	IP Phone to PSTN Phone, Caller ID Restriction (CLIR)		PASS	
6-5	PSTN Phone to IP Phone, Caller ID Restriction (CLIR)		PASS	
Encryption				

7-1	IP Phone to PSTN Phone, TLS + RTP		PASS	
7-2	PSTN Phone to IP Phone, TLS + RTP		PASS	
7-3	IP Phone to PSTN Phone, TLS + SRTP		PASS	
7-4	PSTN Phone to IP Phone, TLS + SRTP		PASS	

Conclusion

These Application Notes describe the configuration steps required for the Ribbon SBC 1000/2000 to successfully interoperate with Skype for Business 2015. All feature and serviceability test cases were completed and passed with the exceptions/observations noted in Test Results.

Appendix A - TLS/SRTP Configuration

Figure 25: TLS Profile

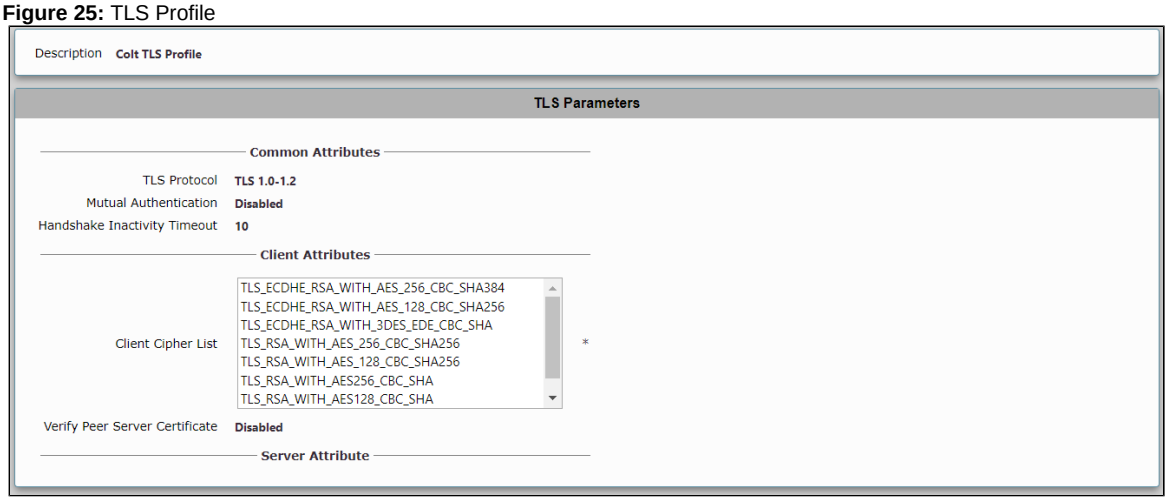


Figure 26: SRTP Profile

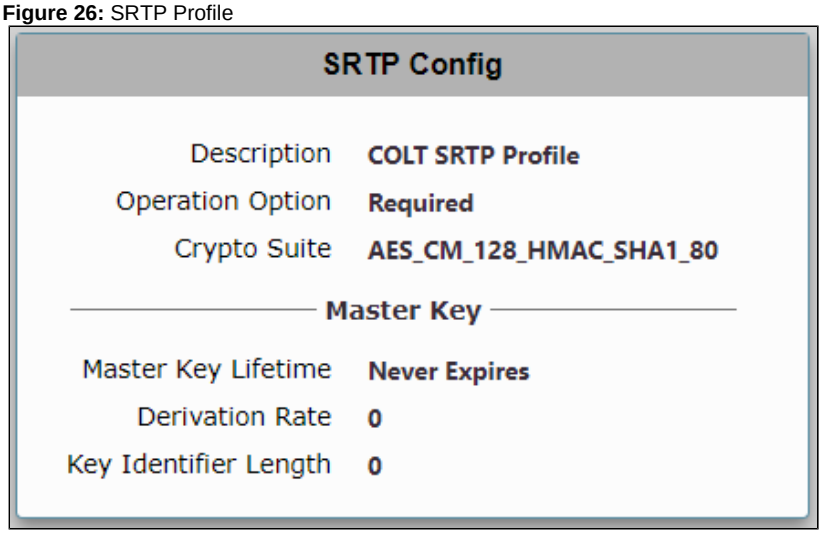


Figure 27: Colt SIP Server

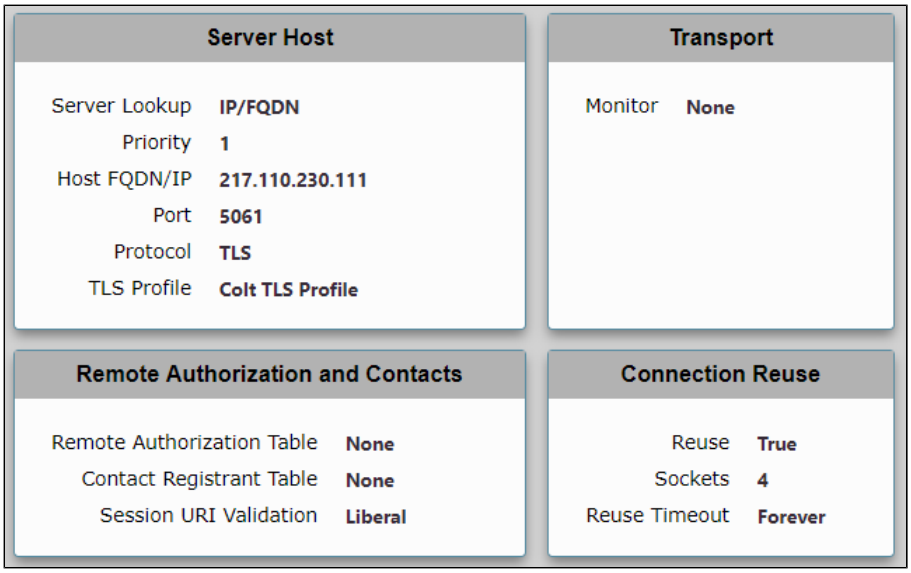


Figure 28: Colt Media List

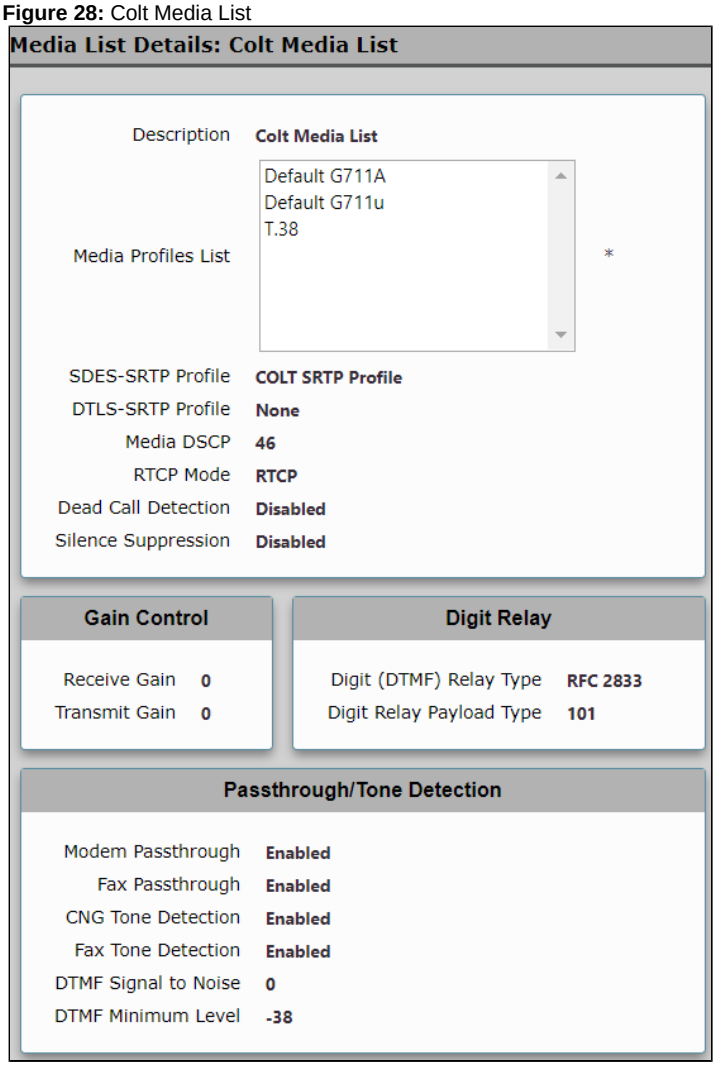


Figure 29: Colt Signaling Group

Description	To/From Colt
Admin State	Enabled
Service Status	Up

SIP Channels and Routing

Action Set Table	None
Call Routing Table	From Colt
No. of Channels	60
SIP Profile	Colt SIP Profile
SIP Mode	Basic Call
Agent Type	Back-to-Back User Agent
Interop Mode	Standard
SIP Server Table	Colt TLS
Load Balancing	Round Robin
Channel Hunting	Most Idle
Notify Lync CAC Profile	Disable
Challenge Request	Disable
Outbound Proxy IP/FQDN	
Outbound Proxy Port	5060
No Channel Available Override	34: No Circuit/Channel Available
Call Setup Response Timer	255
Call Proceeding Timer	180
QoE Reporting	Disabled
Use Register as Keep Alive	Enable
Forked Call Answered Too Soon	Disable

Media Information

Supported Audio/Fax Modes	DSP Proxy Direct *
Supported Video/Application Modes	Proxy Direct *
Media List ID	Colt Media List
Play Ringback	Auto on 180
Tone Table	Default Tone Table
Play Congestion Tone	Disable
Early 183	Disable
Allow Refresh SDP	Enable
Music on Hold	Disabled
RTCP Multiplexing	Disable

Mapping Tables

SIP To Q.850 Override Table	Default (RFC4497)
Q.850 To SIP Override Table	Default (RFC4497)
Pass-thru Peer SIP Response Code	Enable

SIP IP Details

Signaling/Media Source IP	Ethernet 3 IP (216.110.2.229)
Signaling DSCP	40
NAT Traversal	
ICE Support	Disabled
Static NAT - Outbound	
Outbound NAT Traversal	None
Static NAT - Inbound	
Detection	Disabled

Listen Ports

Total **1 SIP Listen Port** Row

Port	Protocol	TLS Profile ID
5061	TLS	Colt TLS Profile

Federated IP/FQDN

Total **1 SIP Federated IP** Row

IP/FQDN	Netmask/Prefix
217.110.230.111	255.255.255.255

Message Manipulation **Enabled**

Inbound Message Manipulation

Message Table List *

Outbound Message Manipulation

Message Table List Replace NEWPORT.lync2013.sonusnet.c *