Sonus SBC 1000/2000 V8.0.0 IOT Skype for Business 2015 British Telecom Platform SIP Trunk Application Note

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Document Overview

This document provides a configuration guide for Sonus Sonus SBC 1000/2000 Series (Session Border Controller) when connecting to Skype for Business 2015 and British Telecom Platform SIP trunk.

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

- For additional information on Skype for Business 2015, visit http://microsoft.com
- For additional information on Sonus SBC 1000/2000, visit http://sonus.net

Introduction

The interoperability compliance testing focuses on verifying inbound and outbound calls flows between Sonus SBC 1000/2000, Skype for Business 2015 and and British Telecom Platform SIP trunk.

Audience

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

> This configuration guide is offered as a convenience to Sonus 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:

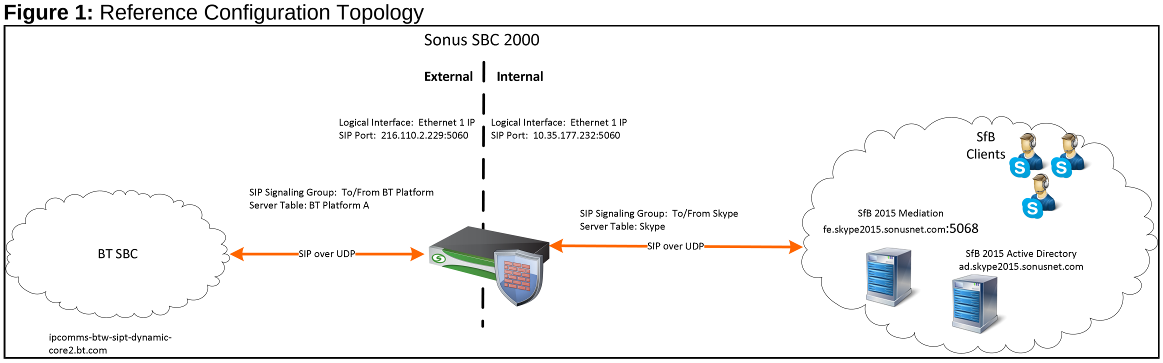
Table 1: Requirements

	Equipment	Software Version
Sonus Networks	SBC 2000	V8.0.0build480
Third-party Equipment	Microsoft Skype for Business 2015 (Skype 2015) Mediation Server	6.0.9319.0

Reference Configuration

The following reference configuration shows connectivity between BT Platform SIP Trunk, Skype 2015 and Sonus SBC 1000/2000.

Figure 1: Reference Configuration Topology



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 BT test plan. The following features were tested:

- Call Origination
- Call Termination
- Session Audit
- Session Timer
- Ringback and Early Media
- Forked Dialog
- Early-Session
- 181 Call Being Forwarded
- Dial Plan
- DTMF
- Codec Negotiation/Renegotiation
- SIP Connect Package
- SIP Connect – PBX Redirect
- SIP Connect – Calling Line ID and Privacy
- Emergency Calls Test
- PSTN Call Tests
- Conferencing
- TrunkGroupFeatures
- UserFeatures
- GroupFeatures
- ProvisioningTests
- Resilience Tests

Verify License

SBC-POL-RTU

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

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

Figure 2: Define a new IP/PSTN Gateway

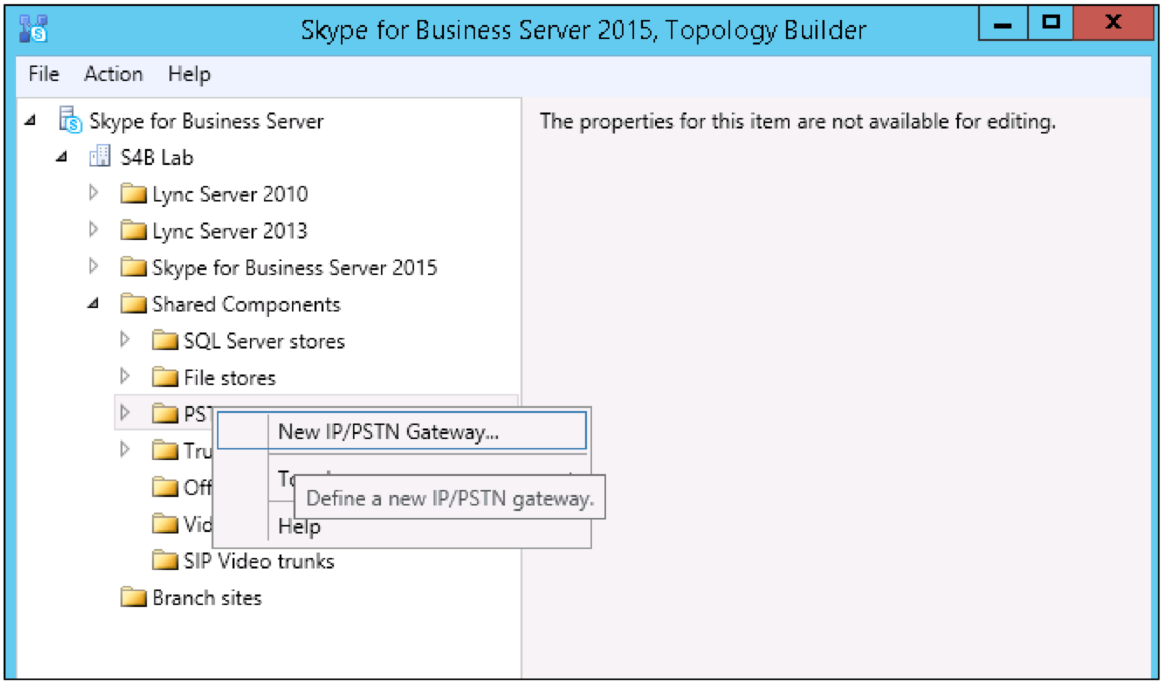


Figure 3: Define FQDN

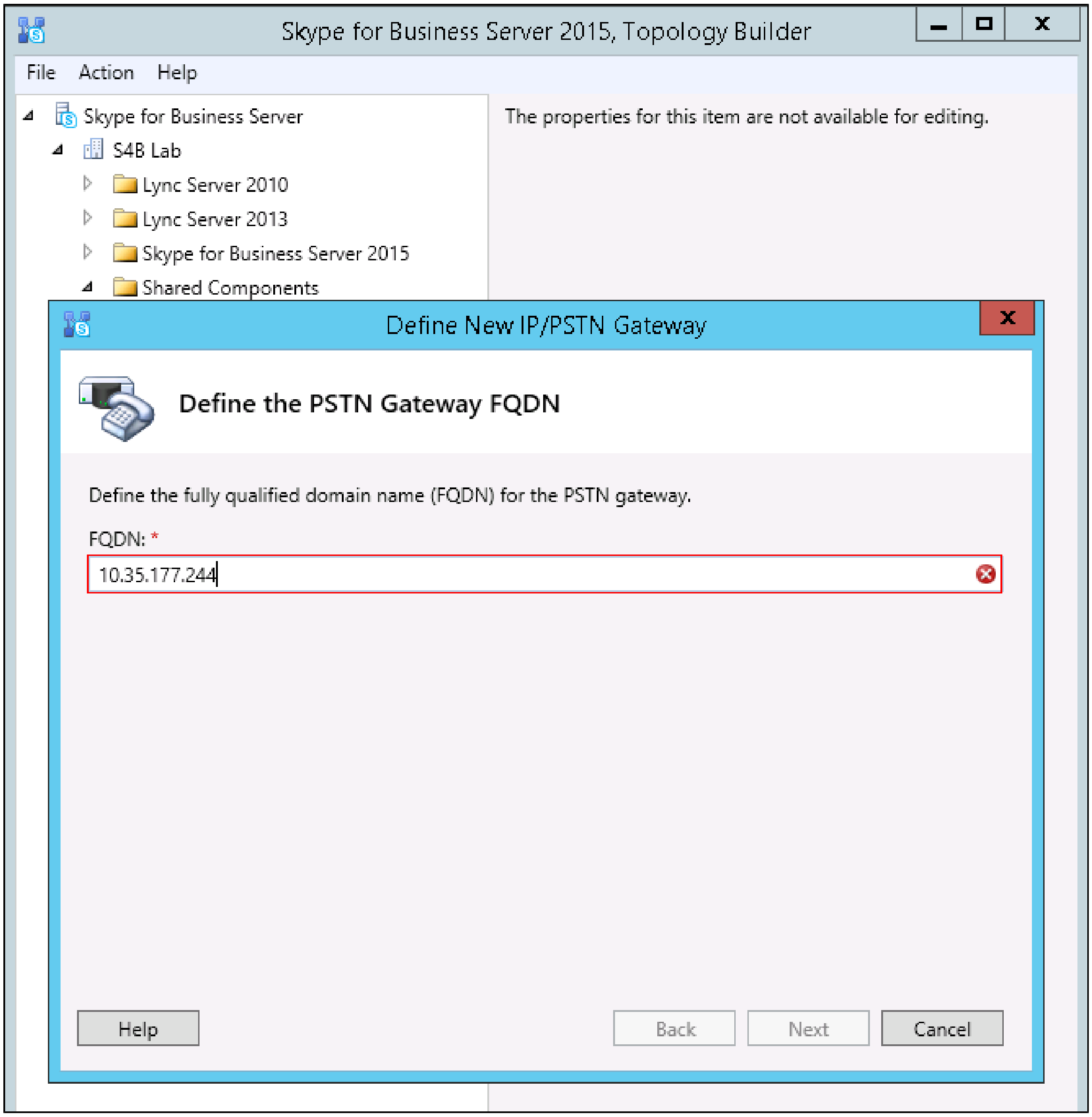


Figure 4: Define IP Address

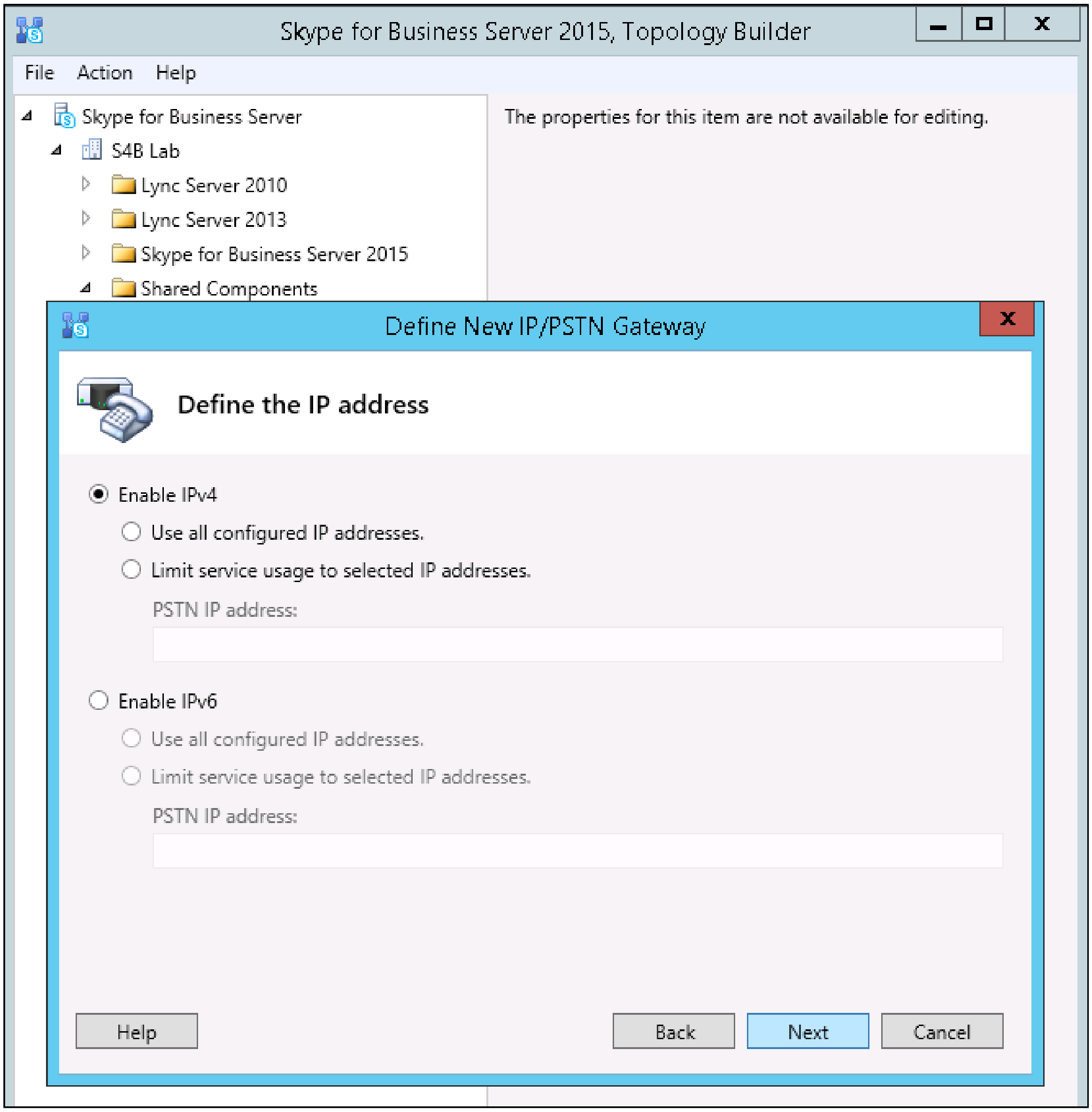
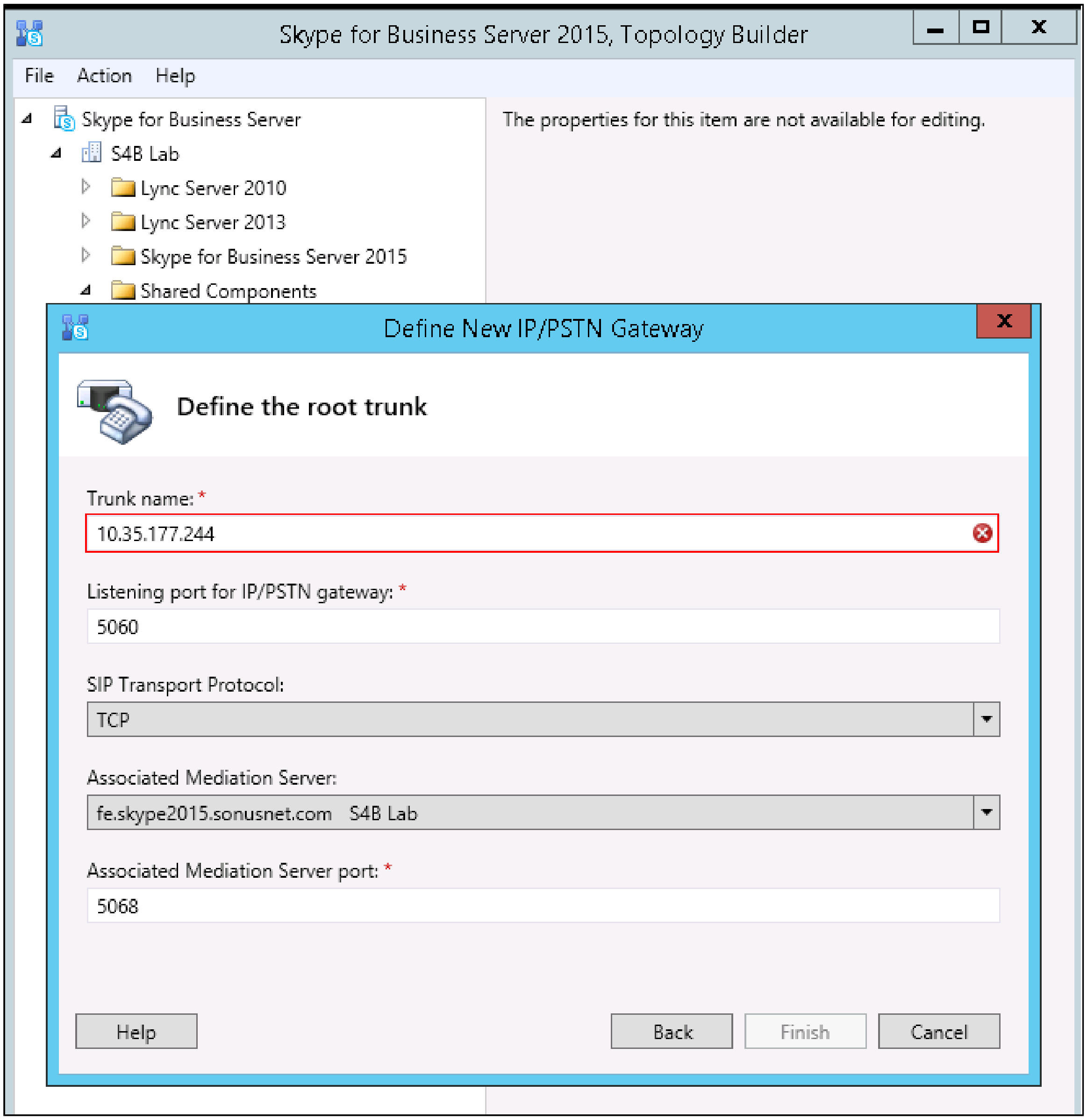


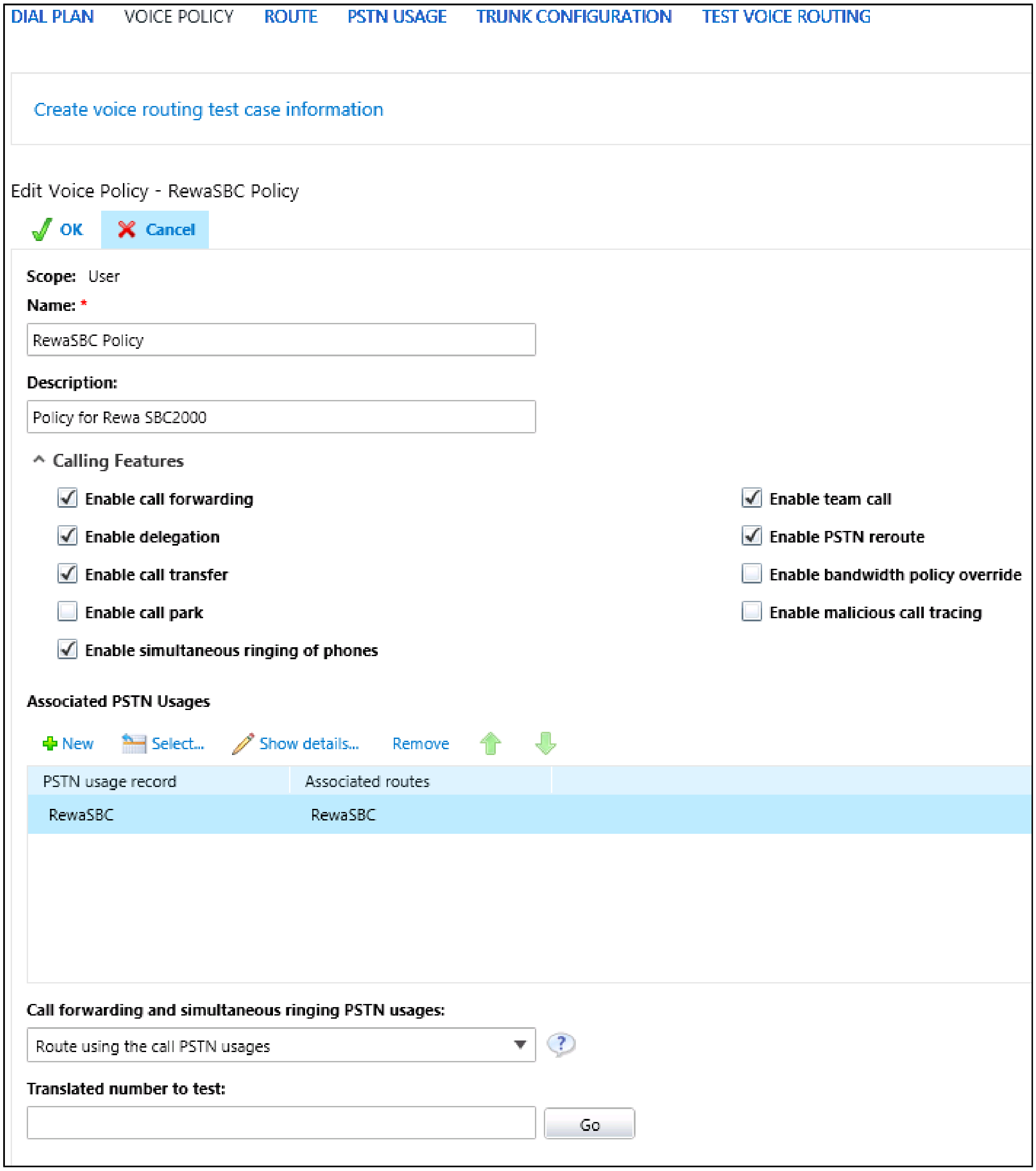
Figure 5: Define Root Trunk

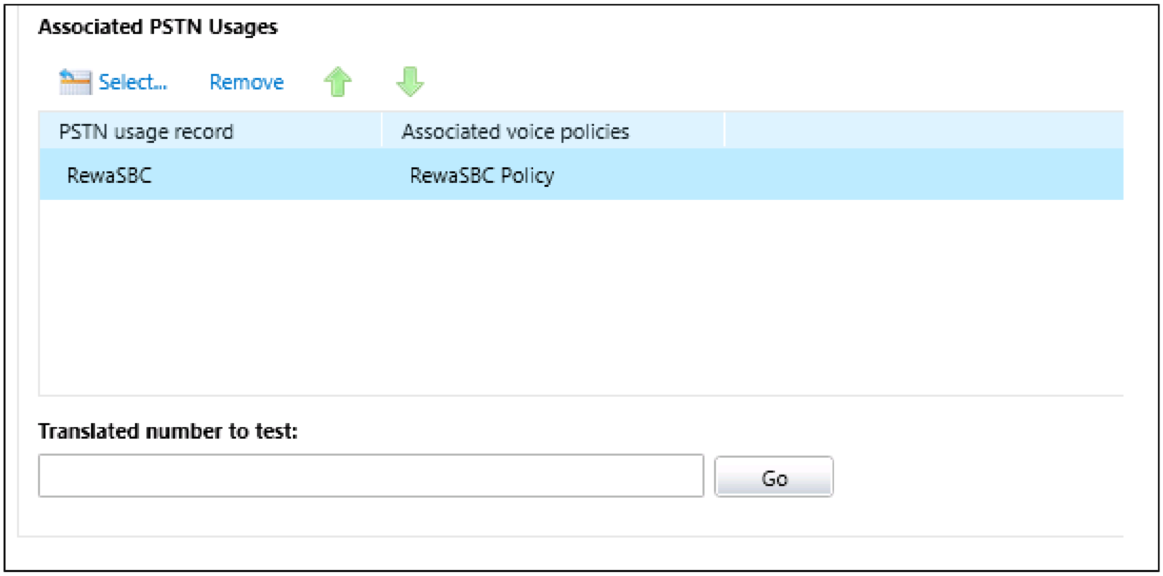


2. Voice Policy

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

Figure 6: Edit Voice Policy

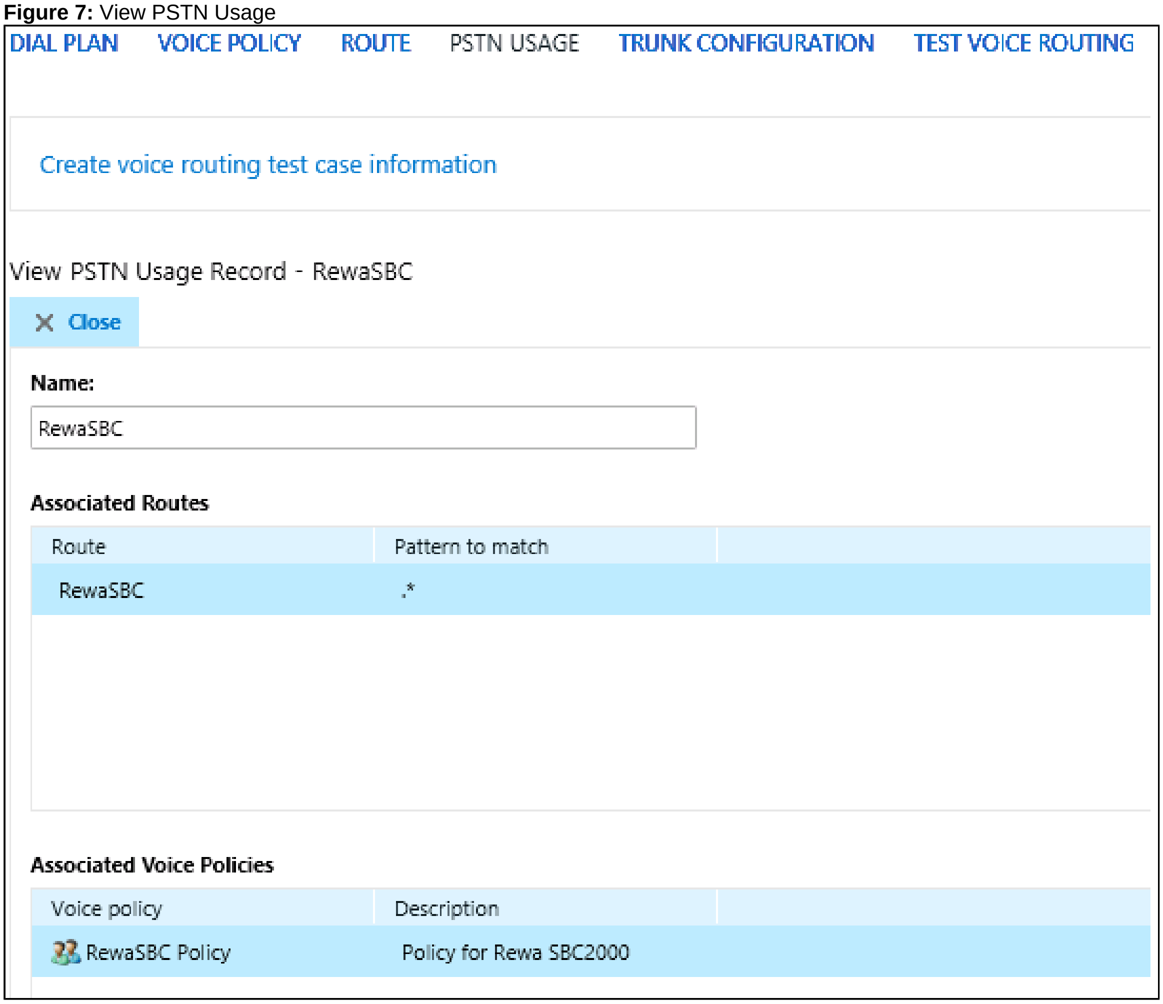

DIAL PLAN
VOICE POLICY
ROUTE
PSTN USAGE
TRUNK CONFIGURATION
TEST VOICE ROUTING
Create voice routing test case information
Edit Voice Policy - RewaSBC Policy
OK
Cancel
Scope: User
Name: *
RewaSBC Policy
Description:
Policy for Rewa SBC2000
Calling Features
Enable call forwarding
Enable delegation
Enable call transfer
Enable call park
Enable simultaneous ringing of phones
Enable team call
Enable PSTN reroute
Enable bandwidth policy override
Enable malicious call tracing
Associated PSTN Usages
New
Select...
Show details...
Remove
PSTN usage record
Associated routes
RewaSBC
RewaSBC
Call forwarding and simultaneous ringing PSTN usages:
Route using the call PSTN usages
Translated number to test:
Go



3. PSTN Usage

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

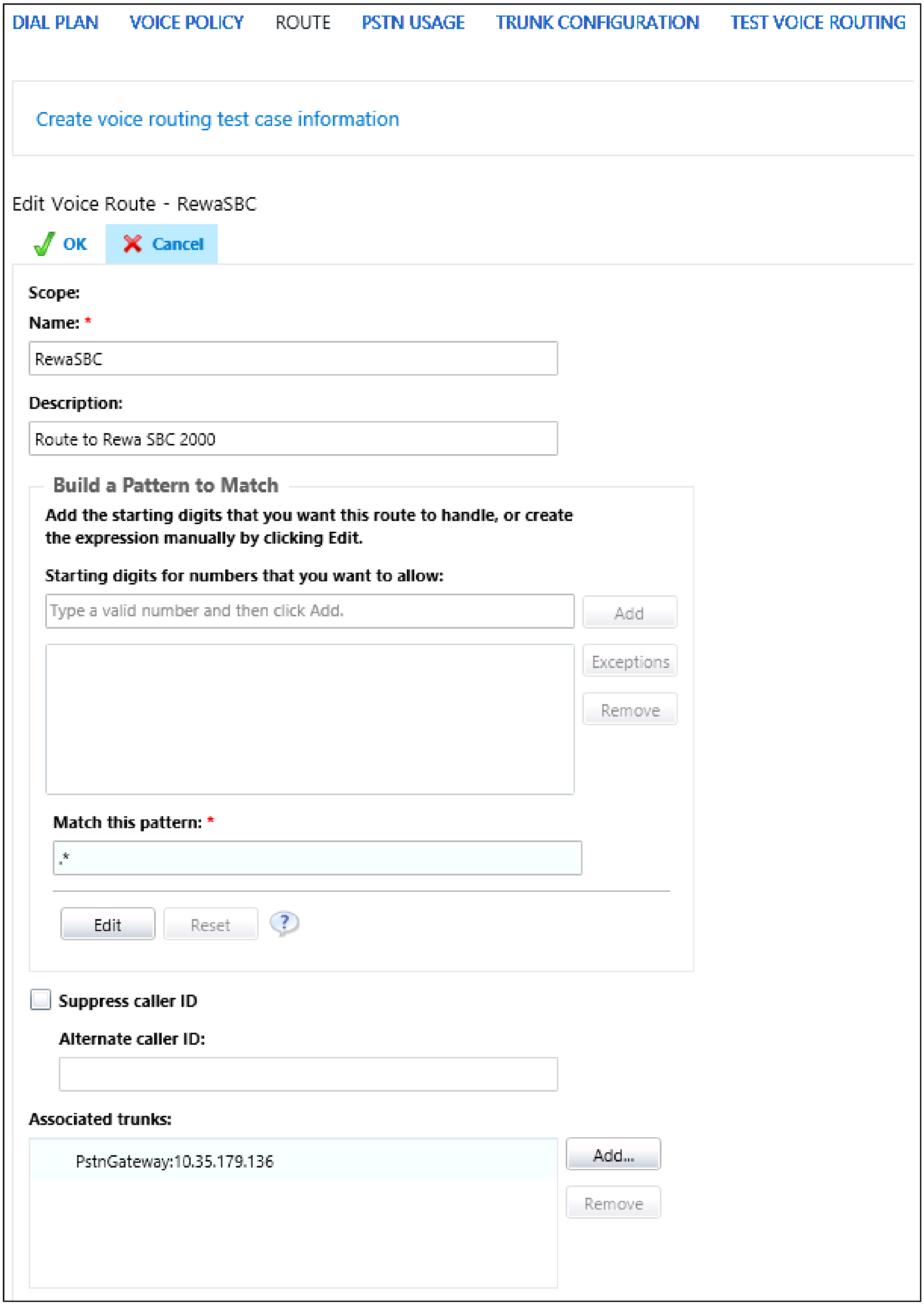
Figure 7: View PSTN Usage

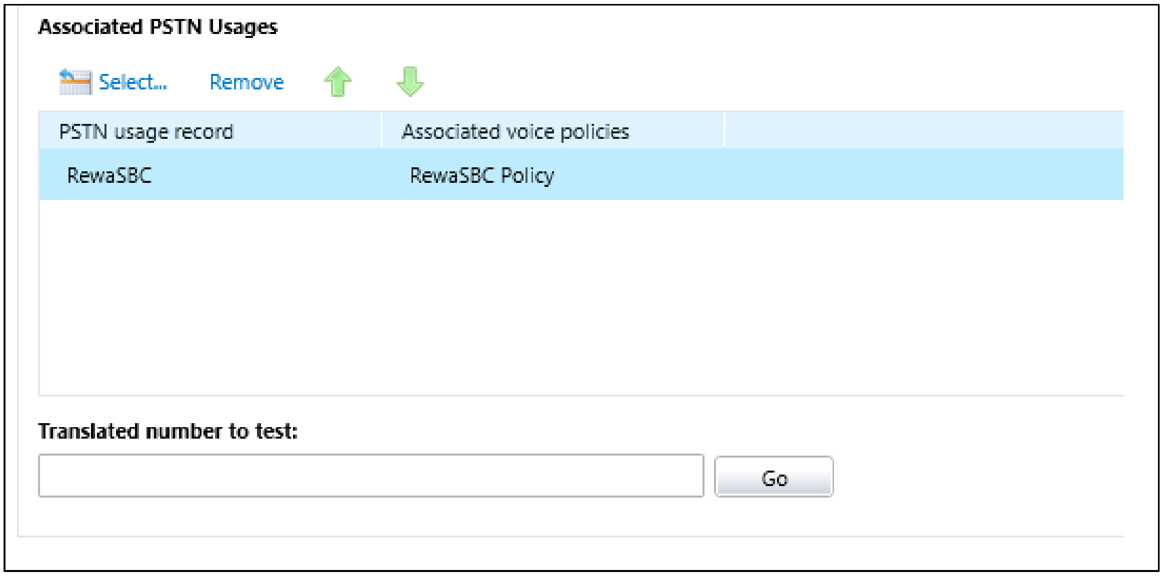


4. Route

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

Figure 8: Edit Voice Route

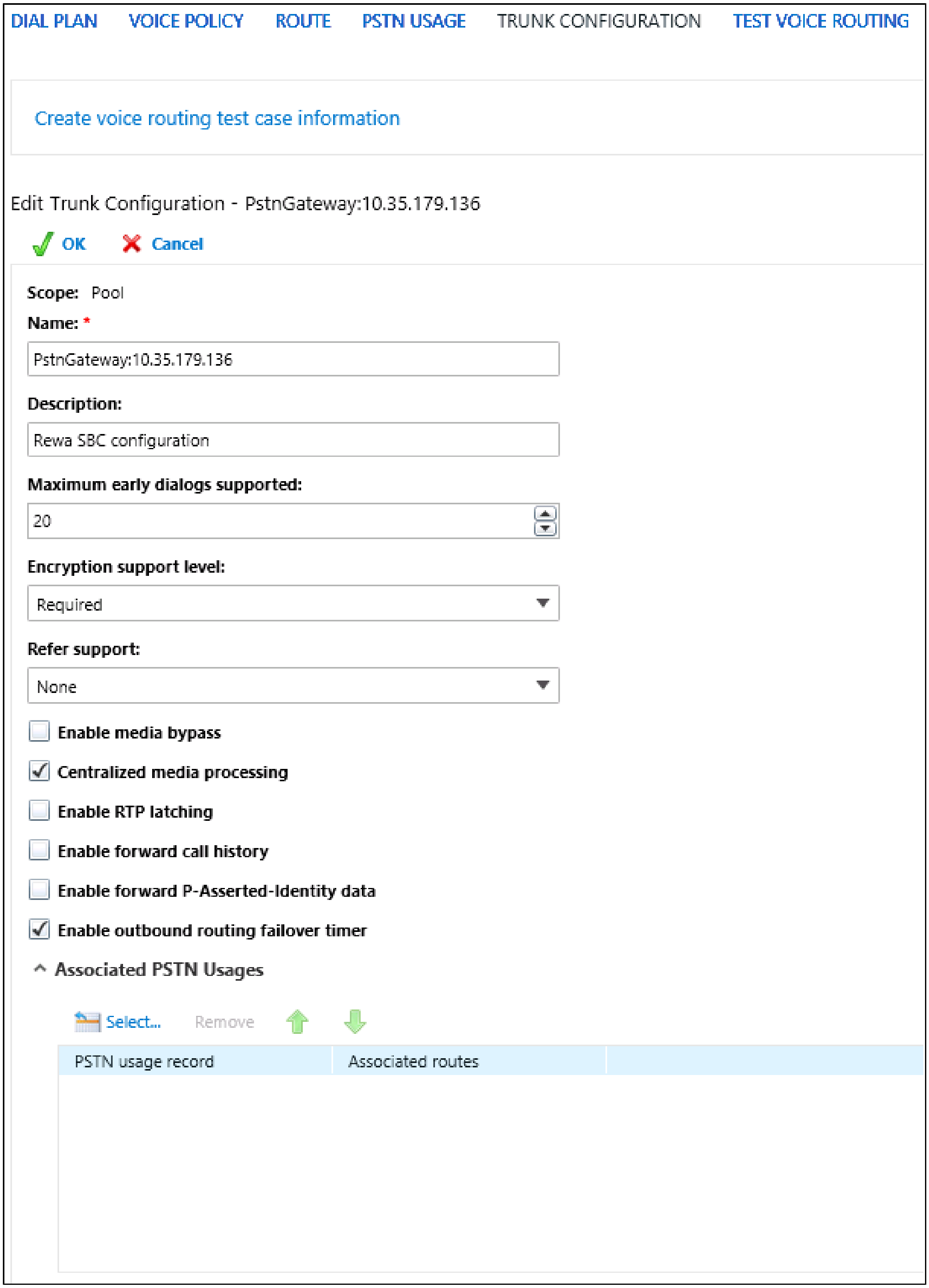




5. Trunk Configuration

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

Figure 9: Edit Trunk Configuration



Sonus SBC 1000/2000 Configuration

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

1. SIP Profile

Select **Settings > SIP > SIP Profiles**

SIP Profiles control how the Sonus 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 figure shows the default SIP profile used for the SBC 1000/2000 for this testing effort:

Figure 10: SIP Profiles

Description	Default SIP Profile

Session Timer

Session Timer	Disable

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	Never
Sonus Diagnostics Header	Enable
Trusted Interface	Enable
UA Header	Sonus SBC
Calling Info Source	RFC Standard
Diversion Header Selection	Last

Options Tags

100rel	Supported
Path	Not Present
Update	Supported

Timers

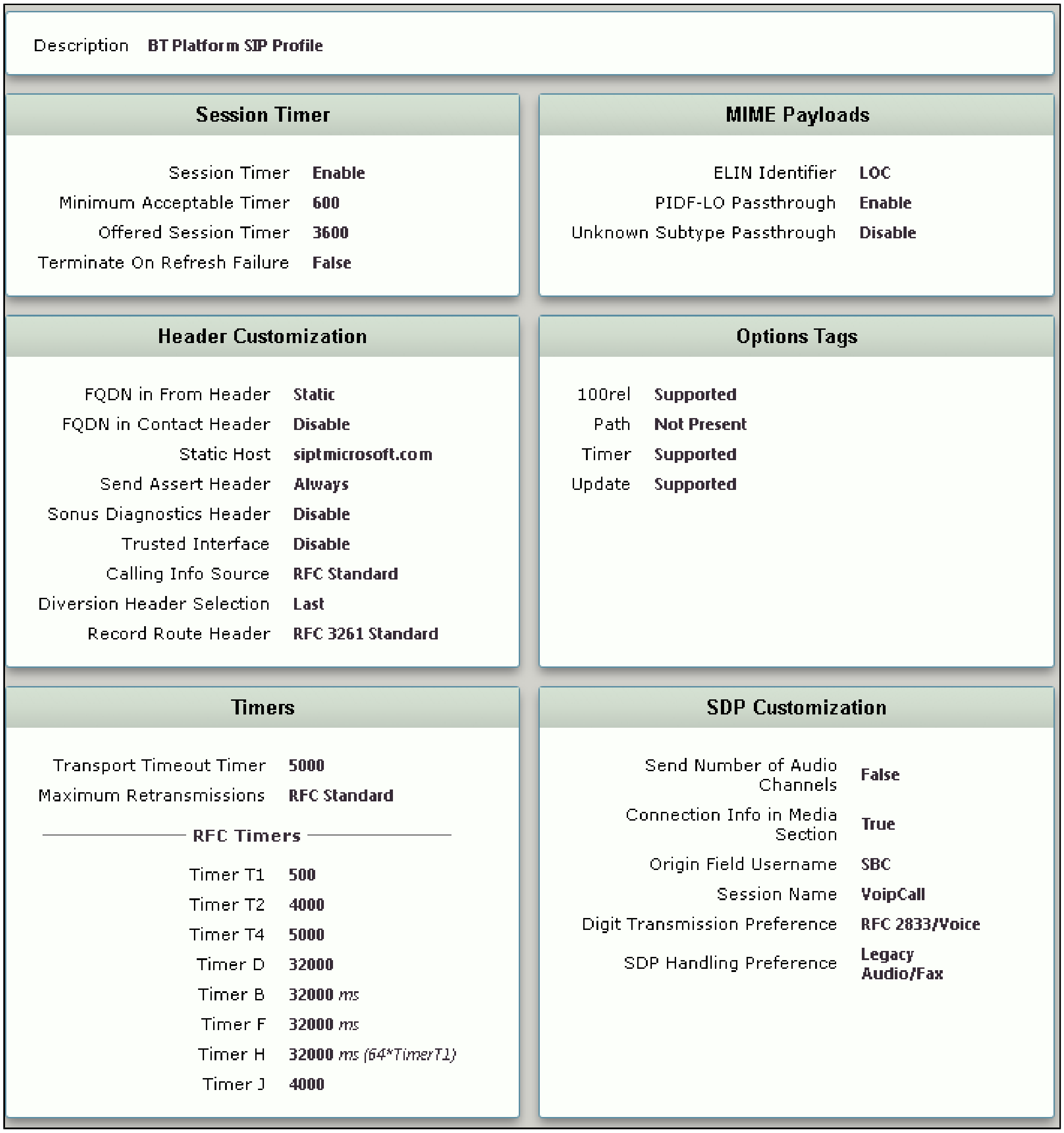
Transport Timeout Timer	5000
Maximum Retransmissions	RFC Standard

RFC timers

Timer T1	500
Timer T2	4000
Timer T4	5000
Timer D	32000
Timer B	32000 *ms*
Timer F	32000 *ms*
Timer H	32000 *ms (64*TimerT1)*
Timer J	32000 *ms (64*TimerT1)*

SDP Customization

Send Number of Audio Channels	True
Connection Info in Media Section	True
Origin Field Username	SBC
Session Name	VoipCall
Digit Transmission Preference	RFC 2833/Voice

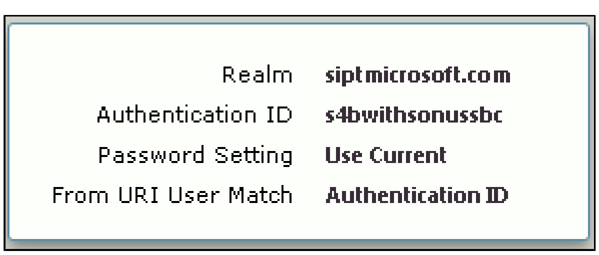


2. Remote Authorization Tables

Select **Settings > SIP > 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 defined in this document appear as options in the Remote Authorization and Contacts Panel for SIP Servers.

Figure 11: Remote Authorization Table

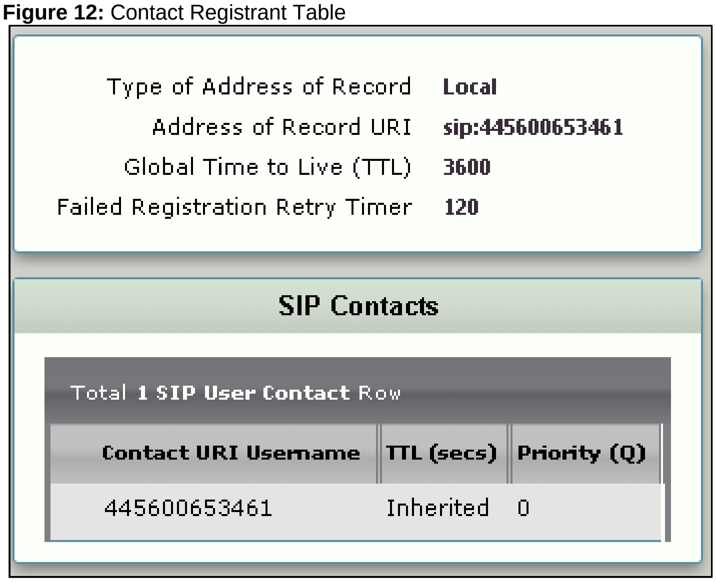


3. Contact Registrant Table

Select **Settings > SIP > Contact Registrant Table**

Contact Registrant Tables are used to manage contacts that are registered to a SIP server. The SIP Server Configuration can specify a Contact Registrant Table and the username portion of the table can be used for outbound calls.

Figure 12: Contact Registrant Table



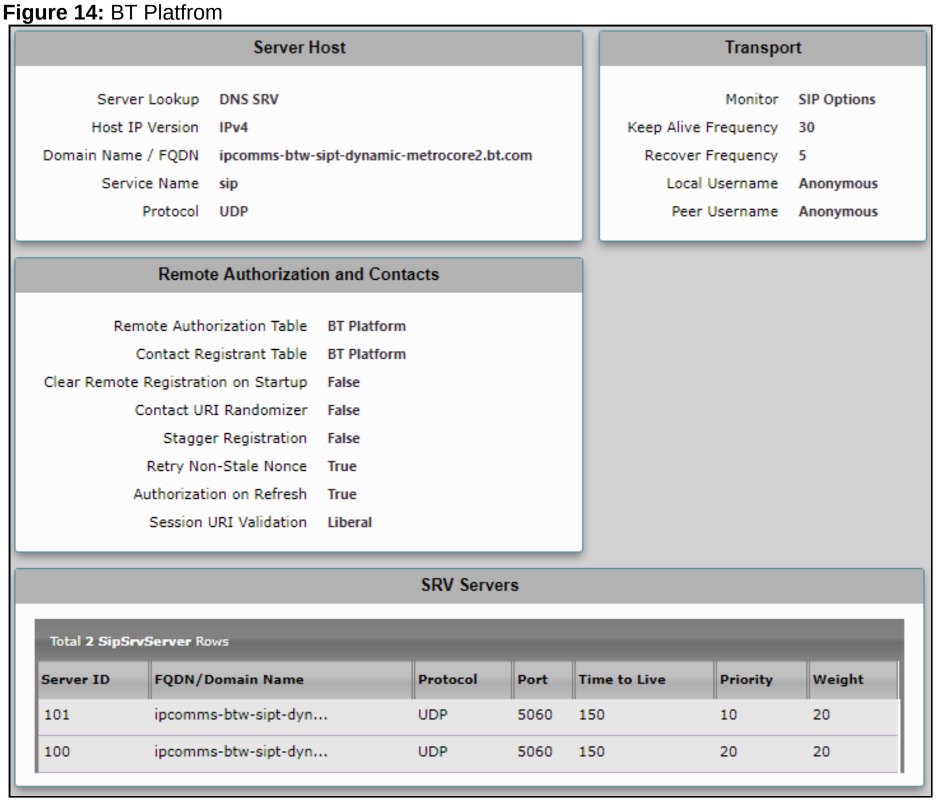
4. SIP Server

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

SIP Server Tables contain information about the SIP devices connected to the Sonus 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 13: Skype

Figure 14: BT Platfrom



5. 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 15: Voice Codec G711 A-Law

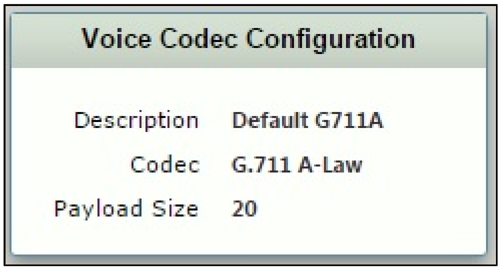


Figure 16: Voice Codec G711 U-Law

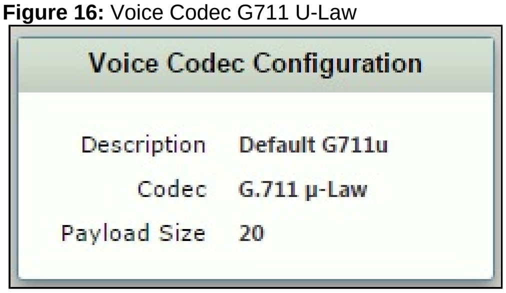


Figure 17: Voice Codec G729

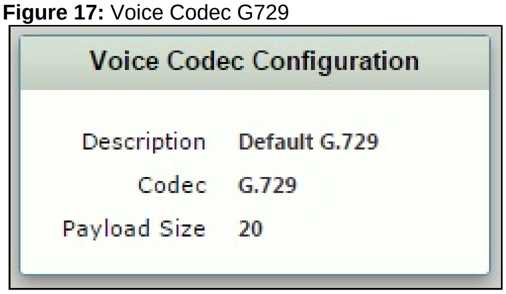
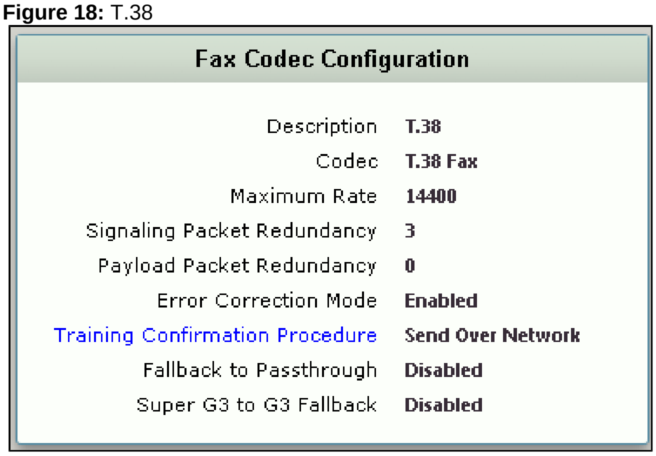


Figure 18: T.38

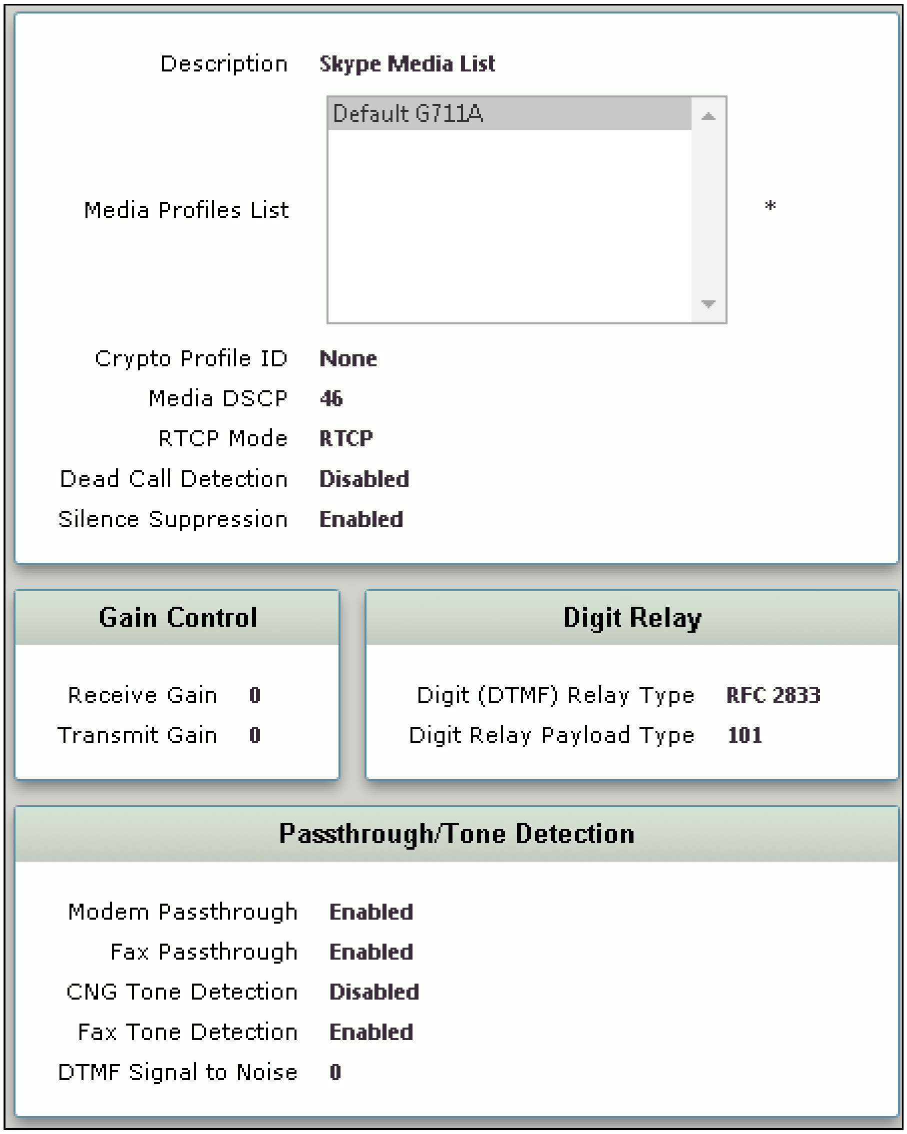


6. Media List

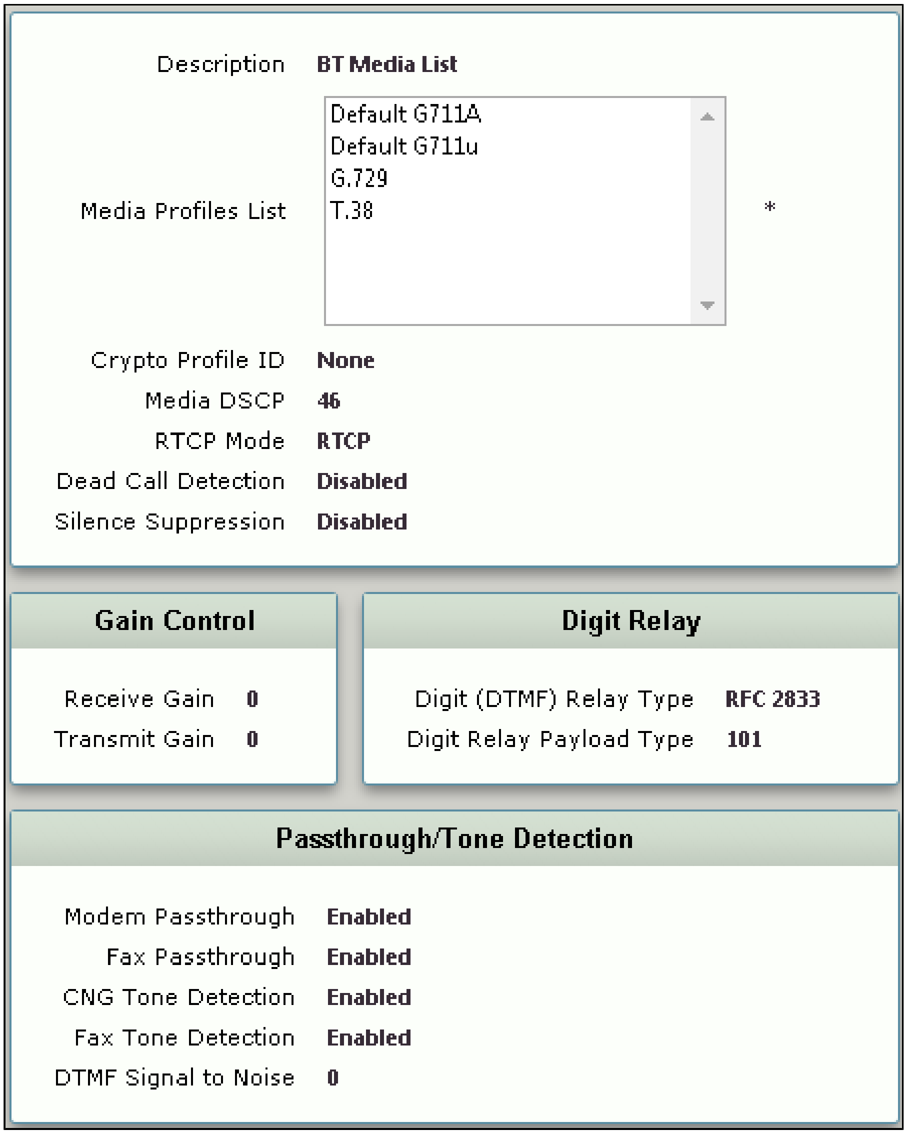
Select **Settings > Media > Media List**

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

Figure 19: Media List



Description Skype Media List
Media Profiles List
Default G711A
*
Crypto Profile ID 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



7. Transformation Table

Select **Settings > 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 into a SIP address (URI). Every entry in a Call Routing Table requires a Transformation Table and is sequentially selected from there. In addition, Transformation tables are configurable as a reusable pool that Action Sets can reference.

Figure 20: From Skype to BT Platform

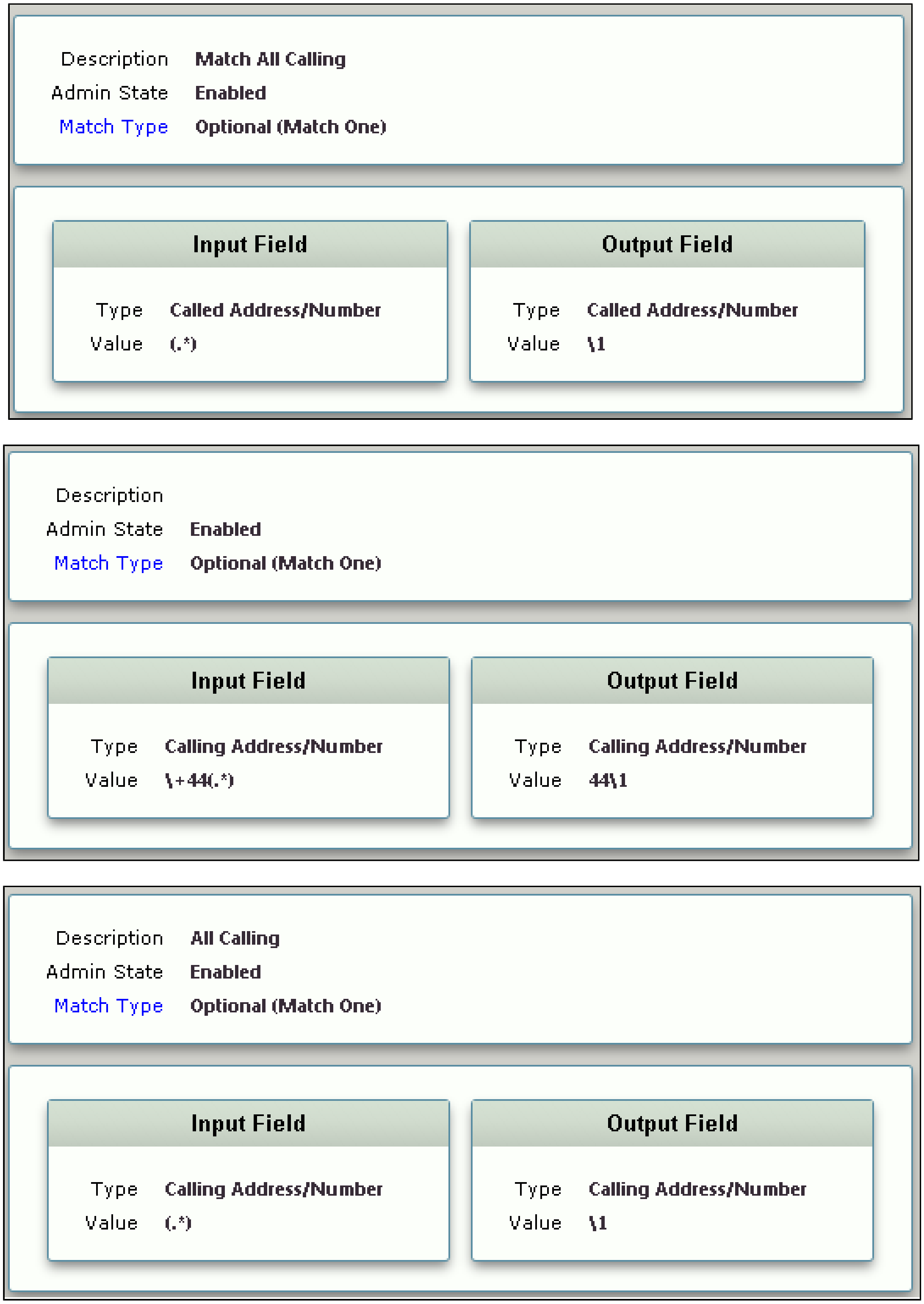
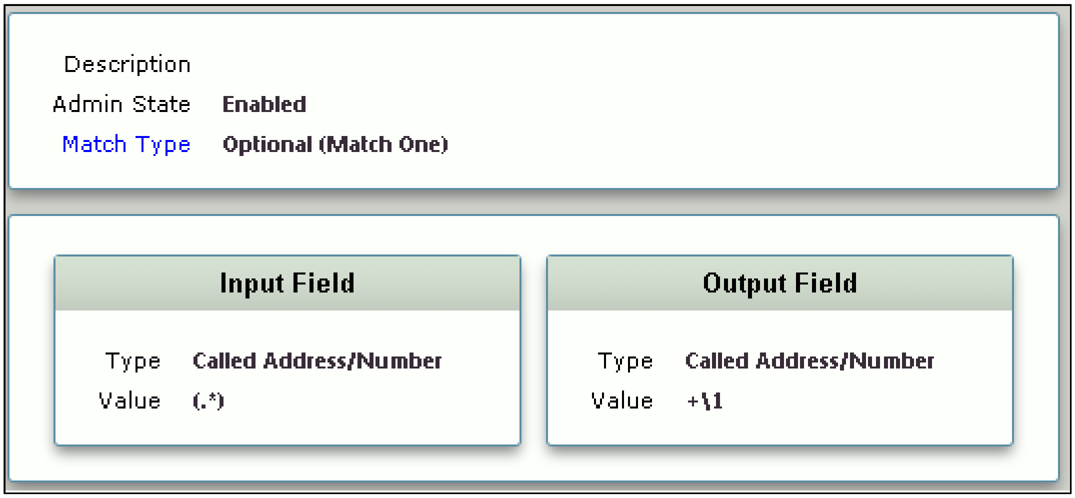


Figure 21: From BT Platfrom



8. 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 22: From Skype to BT platfrom

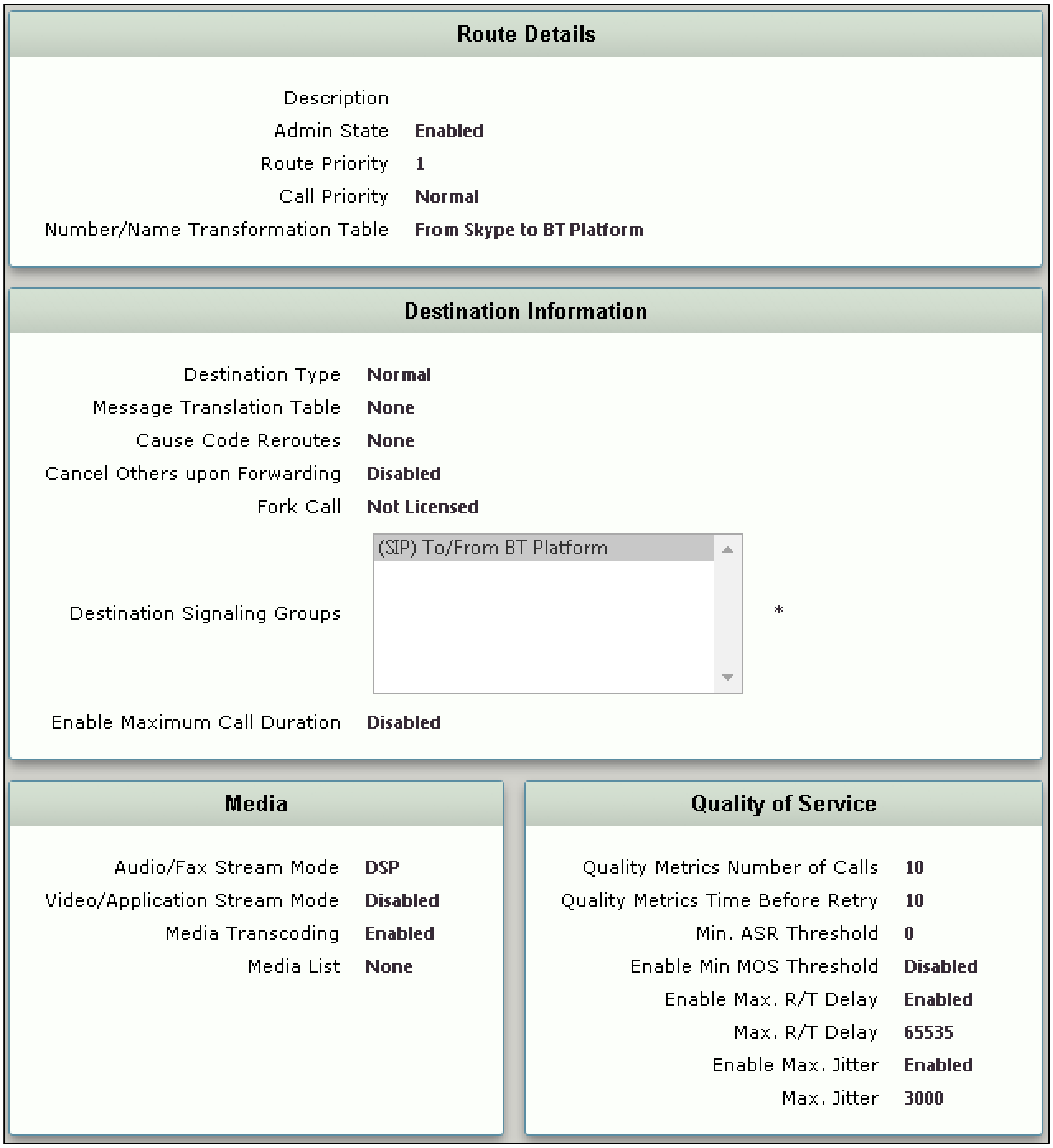
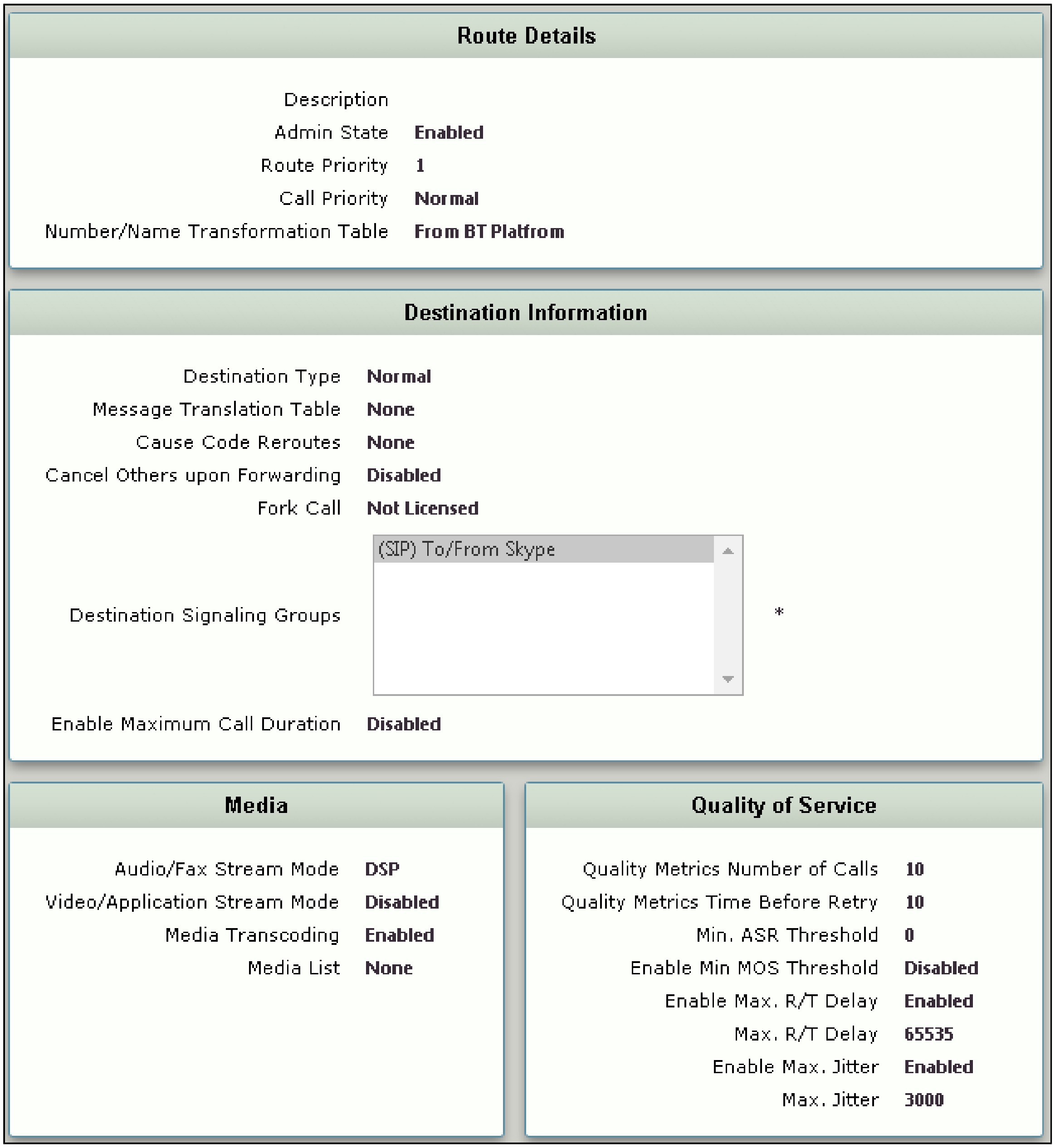


Figure 23: From BT Platfrom

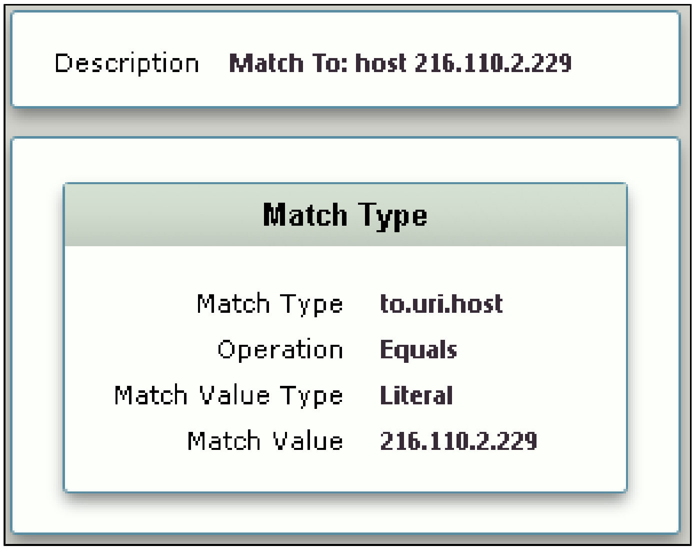


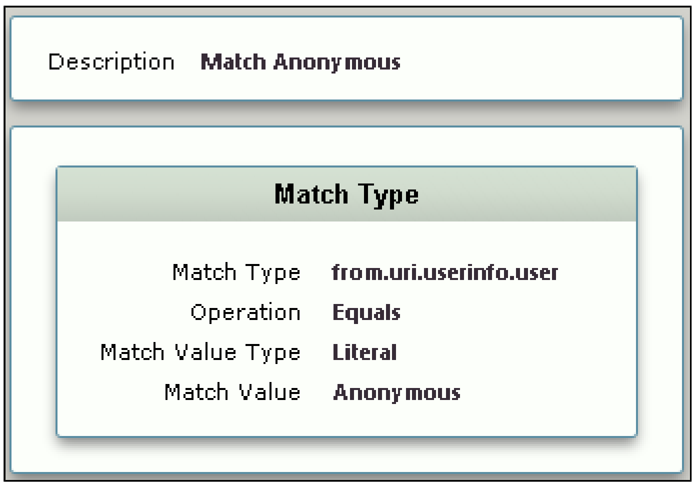
9. Condition Rule Tables

Select **Settings > SIP > Condition Rule Tables**

Condition rules are simple rules that apply to a specific component of a message (for example, diversion.uri.host, from.uri.host, etc.) the value of the field specified in the Match Type list box can be matched against a literal value, token, or REGEX.

Figure 24: Condition Rule Tables





10. Message Rule Tables

Select **Settings > SIP > Message Rule Tables**

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

Figure 25: BT Outbound MRT

Description	Remove + From
Condition Expression	'${12}'
Admin State	Enabled
Result Type	Optional
Header Action	Modify
Header Name	From *

Header Value

Display Name	Ignore
URI	

URI Scheme	Ignore
URI User Info	Modify Match: \+44560065346(.*) Replace: 44560065346\1
URI Host	Ignore
URI Port	Ignore
URI Parameters	Total **0 SPRUriParam** Rows

Name	Value	Action
-- Table is empty --		

Description	Copy from user
Condition Expression	'${12}'
Admin State	Enabled
Result Type	Optional
Header Action	Modify
Header Name	From *

Header Value

Display Name	Ignore
URI	

URI Scheme	Ignore
URI User Info	Copy Value to SG User Value 5
URI Host	Ignore
URI Port	Ignore
URI Parameters	Total **0 SPRUriParam** Rows

Name	Value	Action
-- Table is empty --		

Description	Modify PAID
Condition Expression	'${12}'
Admin State	Enabled
Result Type	Optional
Header Action	Modify
Header Name	P-Asserted-Identity *
Header Ordinal Number	1st

Header Value

Display Name Ignore

URI

URI Scheme	Ignore	
URI User Info	Modify	from.uri.userinfo.user
URI Host	Modify	'siptmicrosoft.com'
URI Port	Ignore	

URI Parameters

Total **0 SPRUriParam** Rows

Name	**Value**	**Action**
-- Table is empty --		

Description	Change diversion
Condition Expression	
Admin State	Enabled
Result Type	Optional
Header Action	Modify
Header Name	Diversion *
Header Ordinal Number	1st

Header Value

Display Name Ignore

URI

URI Scheme	Ignore	
URI User Info	Modify	'445600653461'
URI Host	Modify	'siptmicrosoft.com'
URI Port	Ignore	

URI Parameters

Total **0 SPRUriParam** Rows

Name	**Value**	**Action**
-- Table is empty --		

Description Copy Diversion into PAID
Condition Expression
Admin State Enabled
Result Type Optional
Header Action Modify
Header Name P-Asserted-Identity *
Header Ordinal Number 1st

Header Value

Display Name Ignore
URI

URI Scheme Ignore
URI User Info Modify diversion.uri.userinfo.user
URI Host Modify diversion.uri.host
URI Port Ignore
URI Parameters

Total **0 SPRUriParam** Rows

Name	Value	Action
	-- Table is empty --	

Description Copy Diversion from
Condition Expression
Admin State Enabled
Result Type Optional
Header Action Modify
Header Name From *

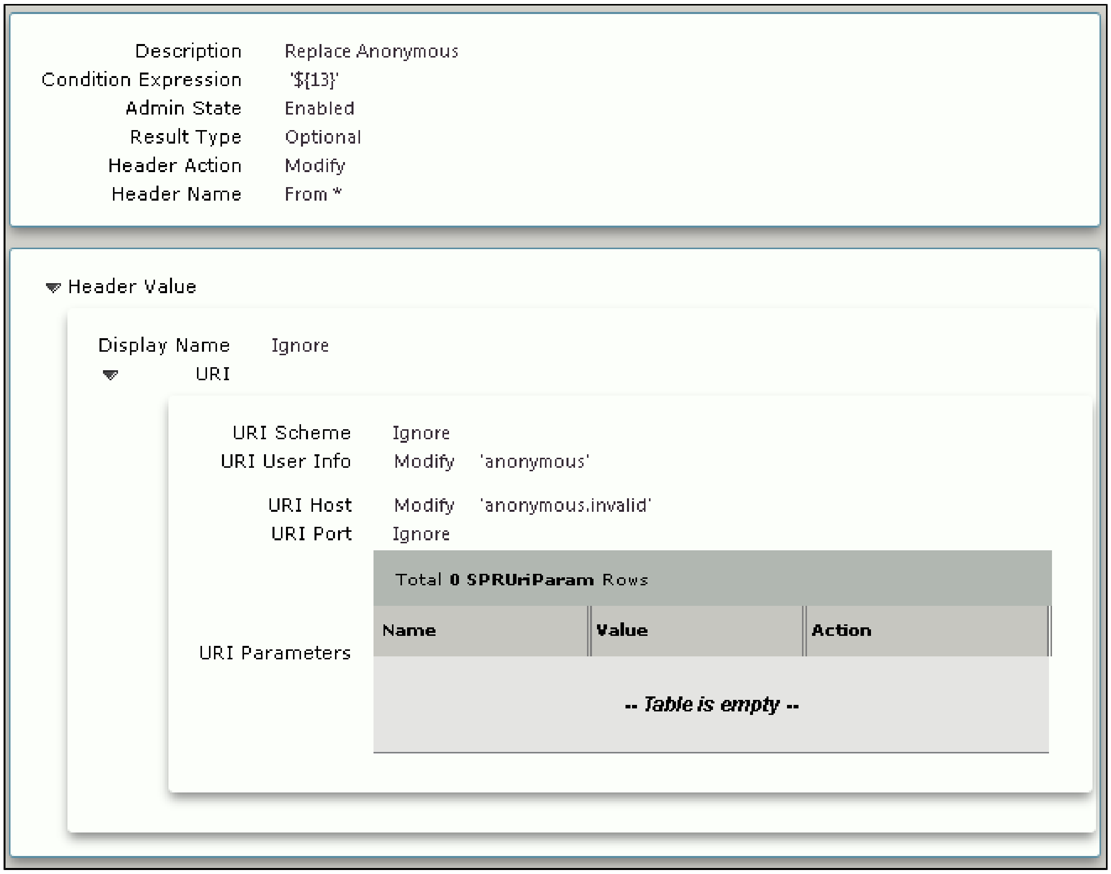
Header Value

Display Name Ignore
URI

URI Scheme Ignore
URI User Info Modify diversion.uri.userinfo.user
URI Host Ignore
URI Port Ignore
URI Parameters

Total **0 SPRUriParam** Rows

Name	Value	Action
	-- Table is empty --	



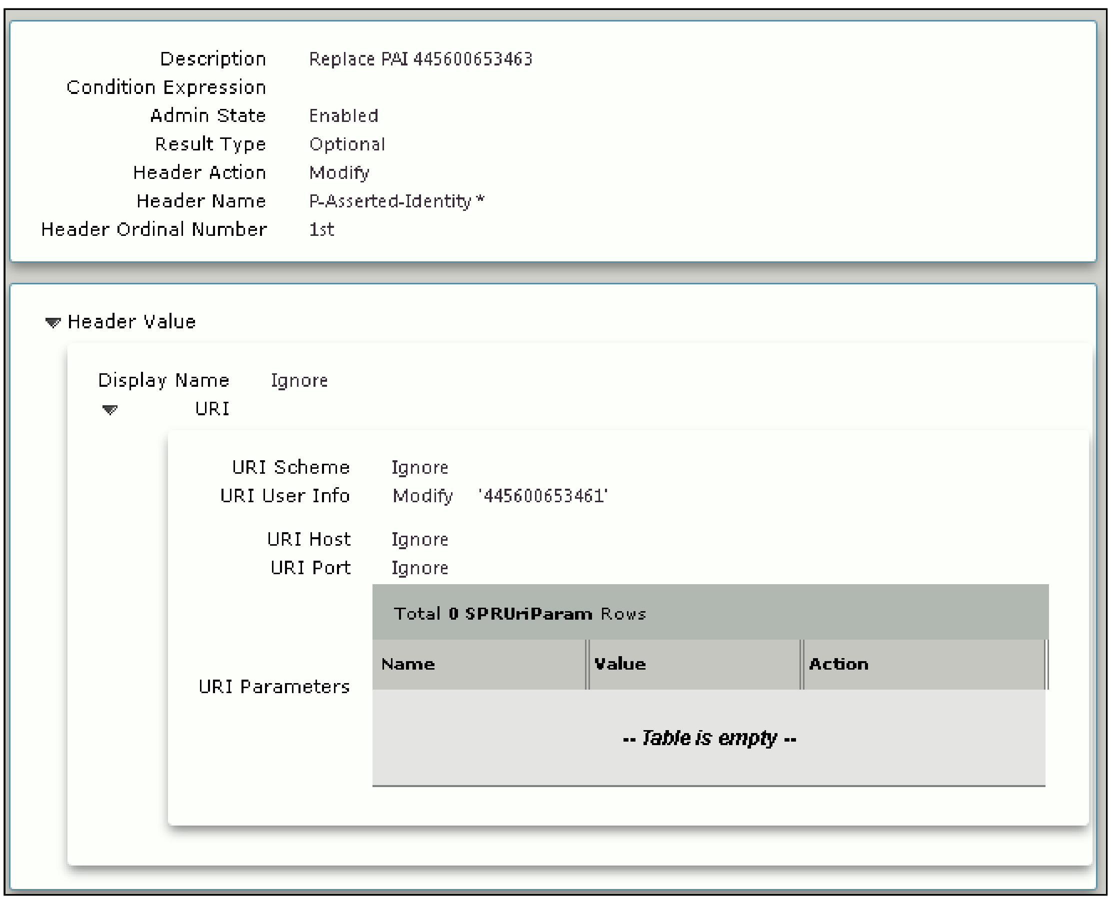


Figure 26: BT Inbound MRT

Description	Remove + From
Condition Expression	'${12}'
Admin State	Enabled
Result Type	Optional
Header Action	Modify
Header Name	From *

Header Value

Display Name	Ignore
URI	

URI Scheme	Ignore		
URI User Info	Modify	Match: \+44560065346(.*)	Replace: 44560065346\1
URI Host	Ignore		
URI Port	Ignore		

URI Parameters

Total **0 SPRUriParam** Rows

Name	Value	Action
	-- Table is empty --	

Description	Copy from user
Condition Expression	'${12}'
Admin State	Enabled
Result Type	Optional
Header Action	Modify
Header Name	From *

Header Value

Display Name	Ignore
URI	

URI Scheme	Ignore	
URI User Info	Copy Value to	SG User Value 5
URI Host	Ignore	
URI Port	Ignore	

URI Parameters

Total **0 SPRUriParam** Rows

Name	Value	Action
	-- Table is empty --	

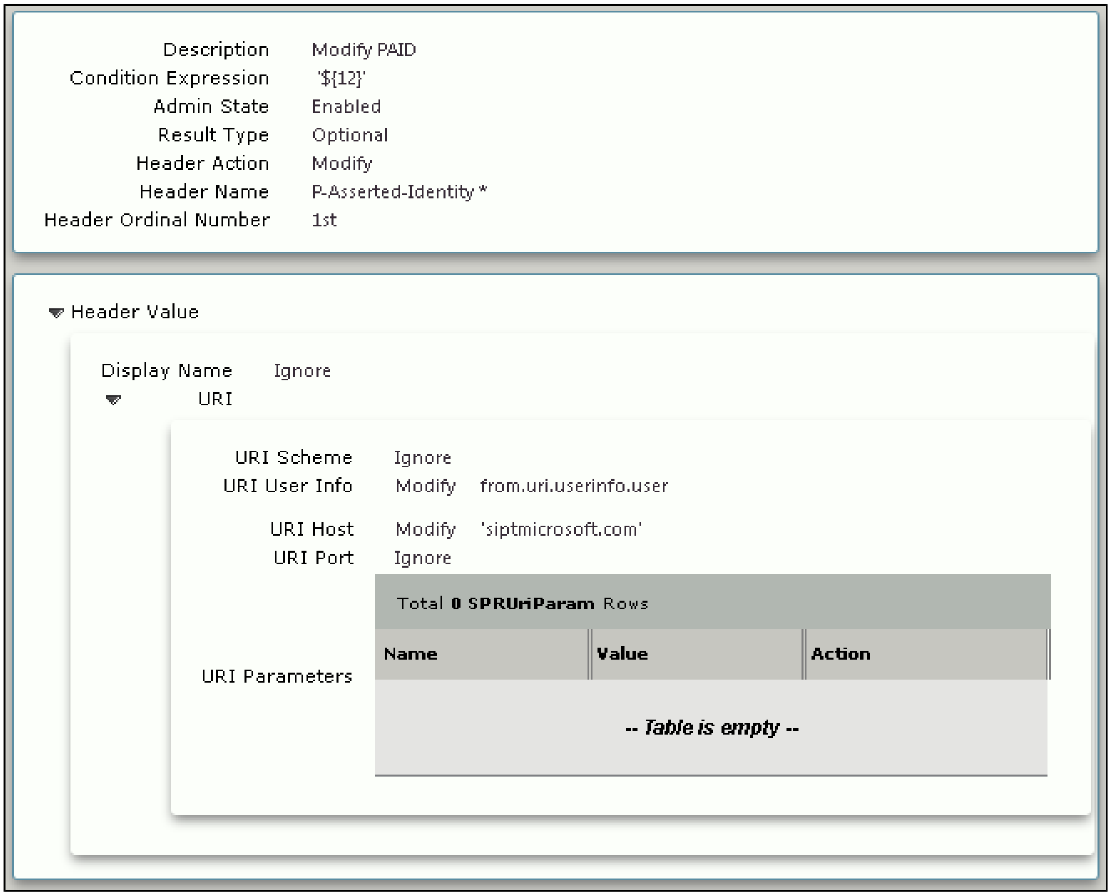
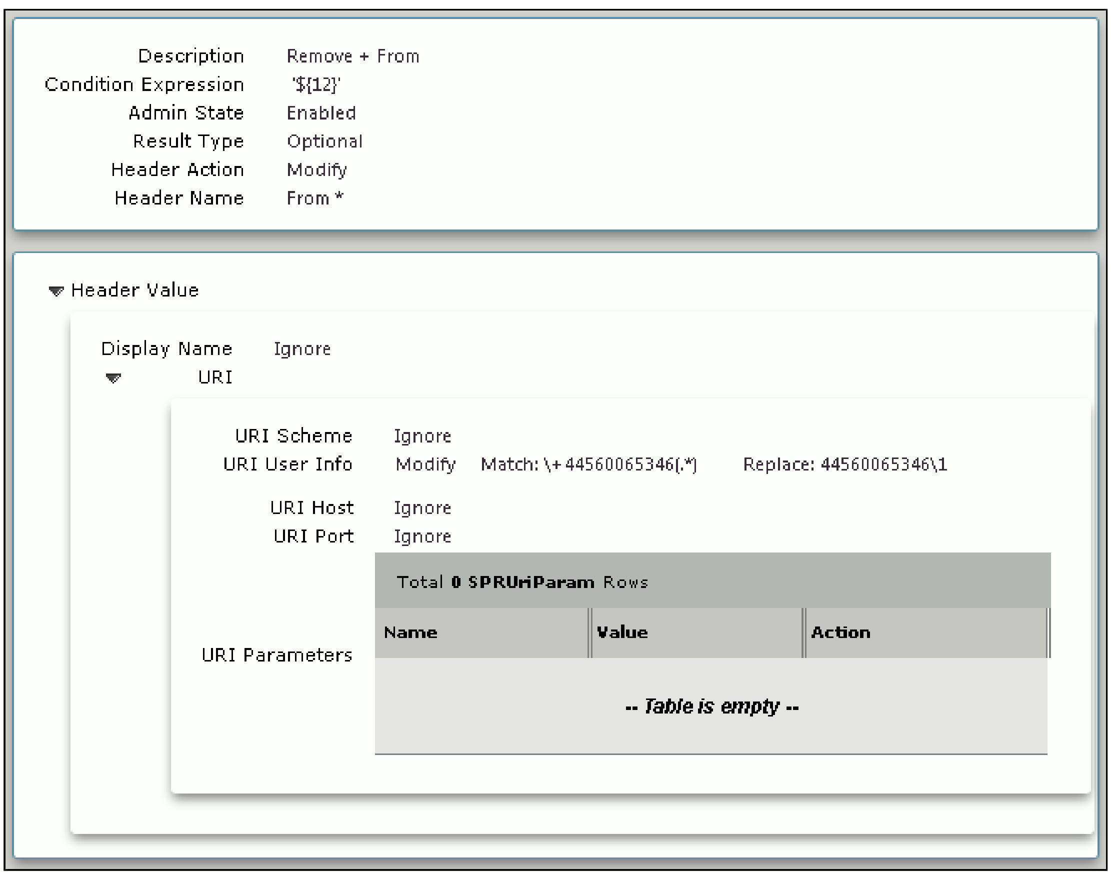
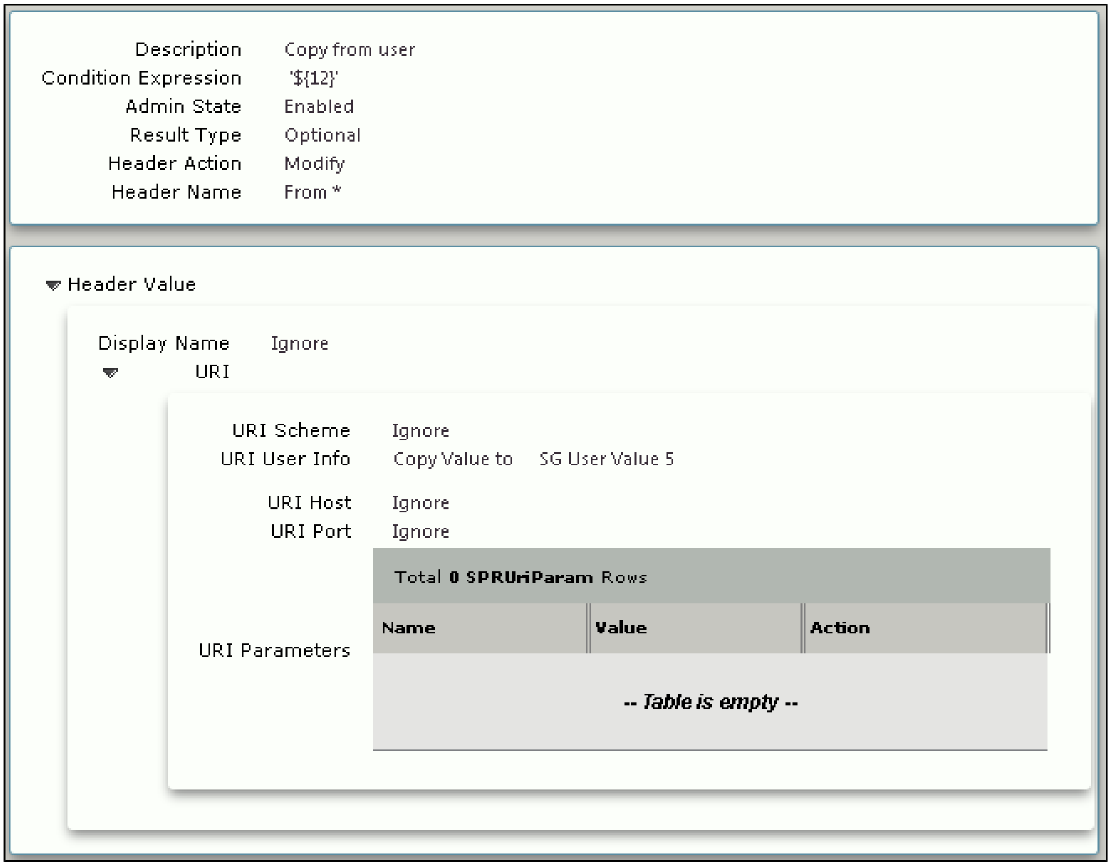
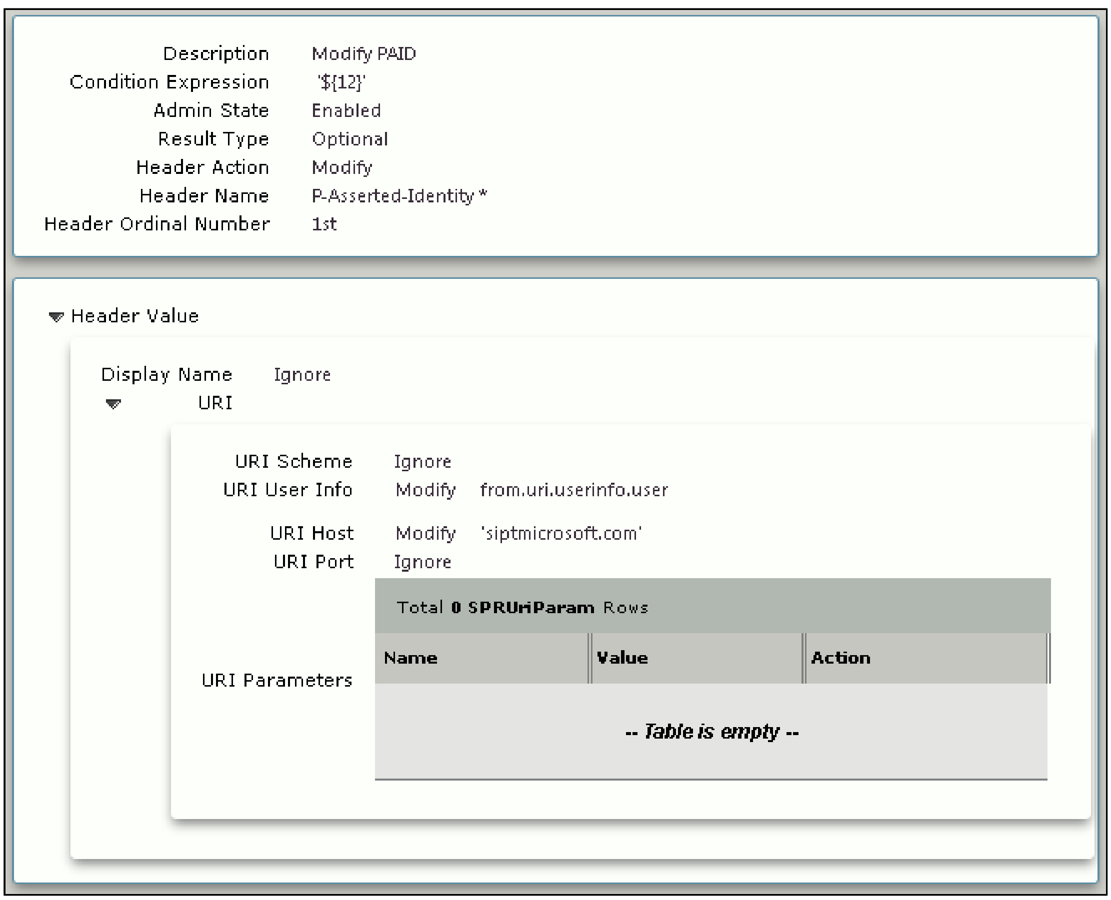


Figure 27: Modify PAID







11. 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 28: To/From Skype

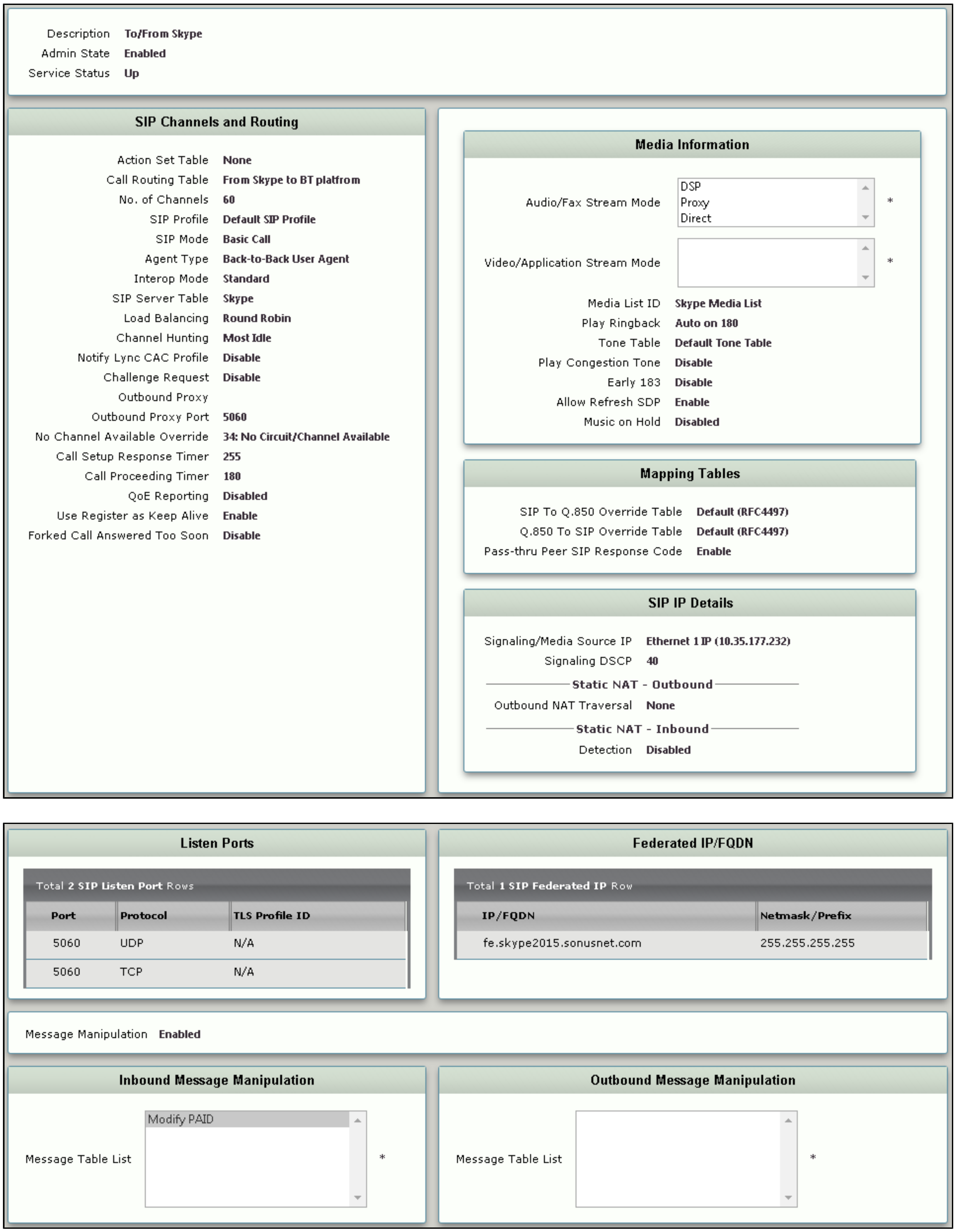
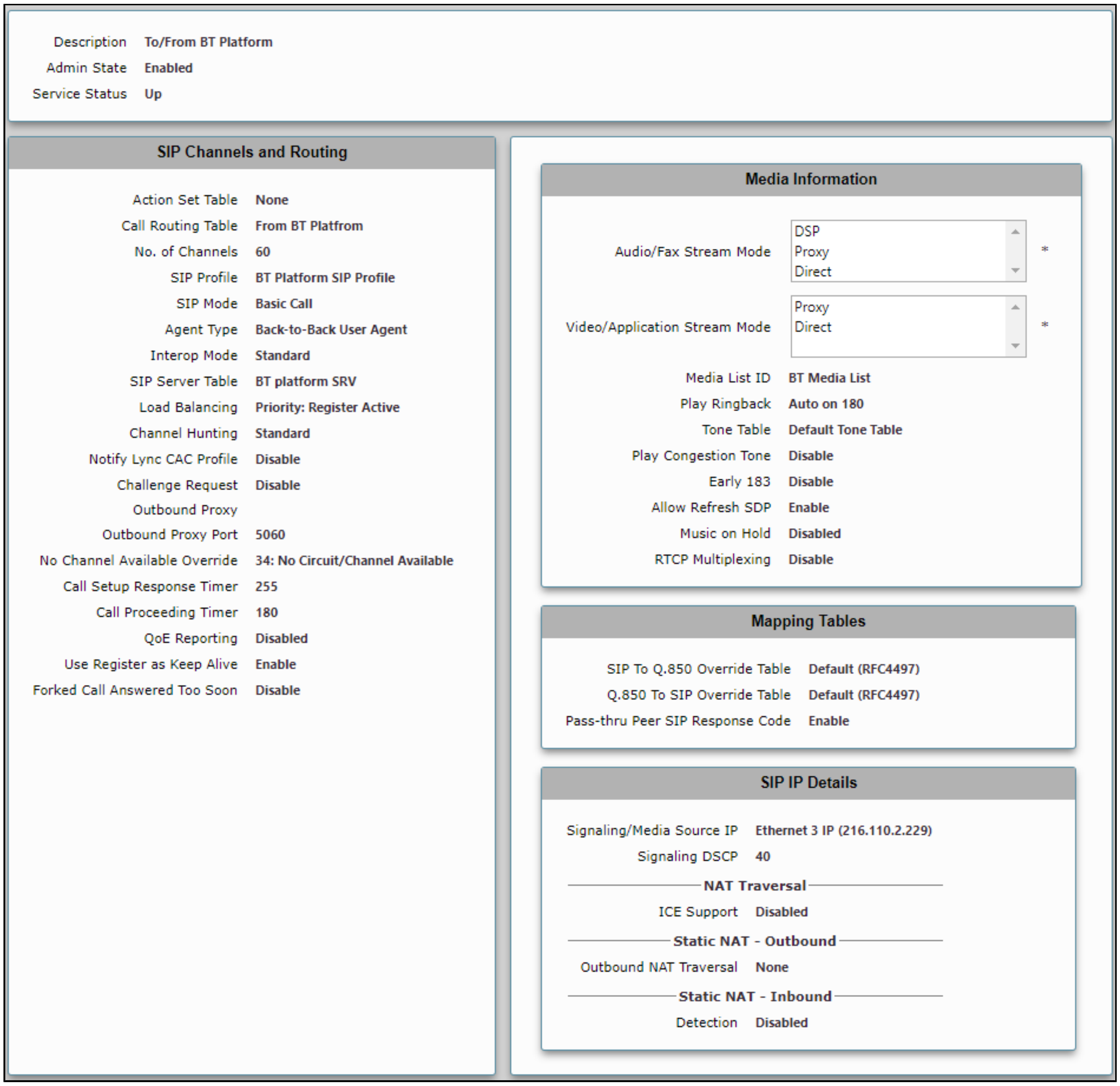
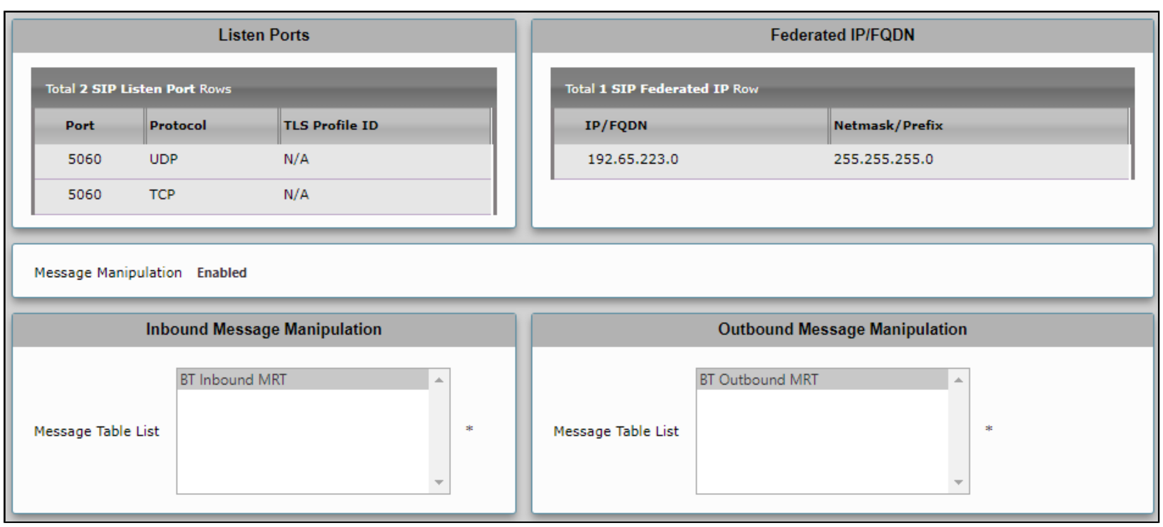


Figure 29: To/From BT Platform





Test Results

Table 2: Test Results

TCID	Test Case Name	Expected Results	Result
Call Origination			
SIP-T_Call_1.1	DUT (Pilot User) to BroadWorks; DUT Hangs Up After Answer (P0)	• DUT dials the number of BroadWorks User A. • BroadWorks User A is alerted. • DUT receives audible ringback. • BroadWorks User A answers the call. • Two-way voice path is established. • DUT hangs up. • The call is released. • Verify the SIP signaling to/from the DUT • Make sure the DUT sends all messages to the primary Application Server address.	Pass
SIP-T_Call_1.2	DUT (Pilot User) to BroadWorks; DUT Hangs Up Before Answer (P0)	• DUT dials the number of the BroadWorks User A. • BroadWorks User A is alerted. • DUT receives audible ringback. • DUT hangs up. • The call is released. • Verify the SIP signaling to/from the DUT. • Make sure the DUT sends all messages to the primary Application Server address.	Pass
SIP-T_Call_1.3	DUT (Pilot User) to BroadWorks; BroadWorks Hangs Up After Answer (P0)	• DUT dials the number of the BroadWorks User A. • BroadWorks User A is alerted. • DUT receives audible ringback. • BroadWorks User A answers the call. • Two-way voice path is established. • BroadWorks User A hangs up. • The call is released. • Verify the SIP signaling to/from the DUT. • Make sure the DUT sends all messages to the primary Application Server address.	Pass
SIP-T_Call_1.4	DUT (Pilot User) to BroadWorks; Long duration call	• DUT dials the number of the BroadWorks User A. • BroadWorks User A is alerted. • DUT receives audible ringback. • BroadWorks User A answers the call. • Two-way voice path is established. • BroadWorks User A hangs up. • The call is released. • Verify the SIP signaling to/from the DUT. Call should stay connected until released by DUT or BroadWorks user	Pass
SIP-T_Call_1.5	Repeat 1.1 - 1.4 for non-pilot user (P0)		Pass
Call Termination			
SIP-T_Call_2.1	BroadWorks to DUT (Pilot User); BroadWorks Hangs Up After Answer (P0)	• BroadWorks User A dials the DUT. • DUT is alerted. • BroadWorks User A receives audible ringback. • DUT answers the call. • Two-way voice path is established. • BroadWorks User A hangs up. • The call is released upon disconnect. • Verify the SIP signaling to/from the DUT. • Make sure the DUT sends all messages to the primary Application Server address.	Pass
SIP-T_Call_2.2	Bria Swap Internal	• BroadWorks User A dials the DUT. • DUT is alerted. • BroadWorks User A receives audible ringback. • BroadWorks User A hangs up. • The call is released upon disconnect. • Verify the SIP signaling to/from the DUT. • Make sure the DUT sends all messages to the primary Application Server address.	Pass
SIP-T_Call_2.3	BroadWorks to DUT (Pilot User); DUT Hangs Up After Answer (P0)	• BroadWorks User dials the DUT. • DUT is alerted. • BroadWorks User A receives audible ringback. • DUT answers the call. • Two-way voice path is established. • DUT hangs up. • The call is released upon disconnect. • Verify the SIP signaling to/from the DUT. • Make sure the DUT sends all messages to the primary Application Server address.	Pass

SIP-T_Call_2.4	BroadWorks to DUT (Pilot User); Long Duration Call	• DUT dials the number of the BroadWorks User A. • BroadWorks User A is alerted. • DUT receives audible ringback. • BroadWorks User A answers the call. • Two-way voice path is established. • BroadWorks User A hangs up. • The call is released. • Verify the SIP signaling to/from the DUT. Call should stay connected until released by DUT or BroadWorks user	Pass
SIP-T_Call_2.5	Repeat test 2.1 - 2.4 for a non-pilot user (P0)		Pass
Session Timer			
SIP-T_Call_4.1	BroadWorks to DUT; Wait for Session Timer (P1)	• BroadWorks User A dials the DUT. • DUT is alerted. • BroadWorks User A receives audible ringback. • DUT answers the call. • Two-way voice path is established. • The call remains up to the session timer interval. • A timer refresh request is sent. • A successful response is sent to the timer refresh request. • The call remains up and there is still a two-way voice path. • Verify the session timer signaling per the instructions at the beginning of this test section.	Pass
SIP-T_Call_4.2	BroadWorks to DUT; BroadWorks Holds, Wait for Session Timer (P1)	• BroadWorks User A dials the DUT. • DUT is alerted. • BroadWorks User A receives audible ringback. • DUT answers the call. • Two-way voice path is established. • BroadWorks User A holds call using device hold mechanism. • There is no voice path between BroadWorks User A and the DUT. • The call remains up to the session timer interval. • A timer refresh request is sent. • The call remains up passed the session timer interval. • BroadWorks User A retrieves held call by resuming the call at the device. • The call remains up and a two-way voice path is re-established. • Verify the session timer signaling per the instructions at the beginning of this test section.	Pass
SIP-T_Call_4.3	DUT to BroadWorks; Wait for Session Timer (P1)	• DUT dials BroadWorks User A. • BroadWorks User A is alerted. • DUT receives audible ringback. • BroadWorks User A answers the call. • Two-way voice path is established. • The call remains up to the session timer interval. • A timer refresh request is sent. • A successful response is sent to the timer refresh request. • The call remains up and there is still a two-way voice path. • Verify the session timer signaling per the instructions at the beginning of this test section.	Pass
SIP-T_Call_4.4	DUT to BroadWorks; DUT Holds, Wait for Session Timer (P1)	• DUT dials BroadWorks User A. • BroadWorks User A is alerted. • DUT receives audible ringback. • BroadWorks User A answers the call. • Two-way voice path is established. • DUT holds call using device hold mechanism. • There is no voice path between BroadWorks User A and the DUT. • The call remains up to the session timer interval. • A timer refresh request is sent. • The call remains up passed the session timer interval. • DUT retrieves held call by resuming the call at the device. • The call remains up and a two-way voice path is re-established.	Pass
Ringback and Early Media			
SIP-T_Call_5.1	DUT Generates Local Ringback (P0)	• DUT dials BroadWorks User A. • BroadWorks User A is alerted. • DUT audible locally supplied ringback. • Verify the signaling to the DUT. • BroadWorks sends 180 Ringing without SDP to the DUT.	Pass

SIP-T_Call_5.2	DUT Receives Remote Ringback (P0)	• DUT dials BroadWorks User A. • BroadWorks User A is alerted. • DUT receives audible ringback. • Make sure the DUT is rendering the remote ringback by removing the LAN connection from the remote endpoint. The ringback should stop. If it does not stop, then the DUT is ignoring the remote ringback and improperly generating local ringback. • Verify the signaling to the DUT. • BroadWorks sends SIP 18x with SDP.	Pass
SIP-T_Call_5.3	DUT Generates Local Ringback, Followed by Receiving Remote Ringback (P0)	• BroadWorks User A is alerted. • DUT hears audible locally supplied ringback. • After some number of rings, the call forwards and BroadWorks User B is alerted. • DUT hears a remote ringback tone. • Make sure the DUT is rendering the remote ringback by removing the LAN connection from the remote endpoint. The ringback should stop. If it does not stop, then the DUT is ignoring the remote ringback and improperly generating local ringback. • Verify the signaling to the DUT. • BroadWorks sends 180 Ringing without SDP to the DUT. • BroadWorks sends SIP 18x with SDP. To-tag may change.	Pass
SIP-T_Call_5.4	DUT Sends Remote Ringback (P1)	• BroadWorks User A dials the DUT. • DUT is alerted. • BroadWorks User A receives audible ringback. • Make sure the DUT is sending early media by removing the LAN connection from the DUT. The ringback should stop. If it does not stop, then the ringback is not being generated by the DUT. • If double ringback occurs, check the signaling to see if the DUT sends an 18x with SDP followed by an 18x without SDP. If it does, see test setup step 2. If the RFC 3398 setting is required, make a note of this in the test report. • Verify the signaling from the DUT. • DUT sends SIP 18x with SDP. • DUT may send additional SIP 18x with or without SDP.	Pass
Forked Dialog			
SIP-T_Call_5.5	Early to Confirmed Dialog Forking (P0)	• DUT dials BroadWorks User A. • BroadWorks User A is alerted. • DUT receives audible ringback. • DUT is forwarded to voice mail and hears BroadWorks User A's Voice Message No Answer greeting. • Verify the signaling to the DUT. • The TO tag in the 200 OK is different than the TO tag in 18x. • The DUT ACKs the 200 OK.	Pass
SIP-T_Call_5.6	Early to Early to Confirmed Dialog Forking (P0)	• DUT dials BroadWorks User A. • BroadWorks User A is alerted. • DUT receives audible remote ringback. • After some number of rings, BroadWorks User B is alerted. • DUT receives audible remote ringback. • BroadWorks User B answers the call. • Two-way voice path is established. • Verify the signaling to the DUT. • The TO tag in the second 18x is different from the TO tag in the first 18x. • The TO tag in the 200 OK is different than the TO tag in the second 18x. • The DUT ACKs the 200 OK.	Pass
Early UPDATE			
SIP-T_Call_5.7	DUT Receives Early UPDATE (P1)	• BroadWorks User A calls BroadWorks User B. User B answers. • Two-way voice path is established. • BroadWorks User B dials DUT via the CM. • BroadWorks User A is put on hold. • DUT is alerted. • BroadWorks User B hears ringback. • BroadWorks User B completes the transfer while DUT is ringing by clicking Transfer on the CM. • BroadWorks User A hears ringback. • DUT answers the call. • Two-way voice path is established. • Verify the signaling. • DUT receives initial INVITE. • DUT responds with 18x with SDP. • DUT receives UPDATE with OFFER SDP. • DUT responds to UPDATE with 200 OK with ANSWER SDP.	Pass

SIP-T_Call_5.8	DUT Sends Early UPDATE (P2)	• BroadWorks User A calls DUT. • DUT is alerted. • BroadWorks User A receives remote ringback. • The call CFNAs to another party or otherwise triggers UPDATE. • BroadWorks User A still hears ringback. • DUT answers the call. • Two-way voice path is established. • Verify the signaling. • DUT responds with 18x with SDP. • DUT sends UPDATE with OFFER SDP when the call forwards. • The UPDATE is sent through to BroadWorks User A's device. • BroadWorks User A's device responds to UPDATE with 200 OK with ANSWER SDP.	Pass
Early-Session			
SIP-T_Call_5.9	DUT Receives Early-Session Offer (P2)	• DUT dials BroadWorks User A. • BroadWorks User A is alerted. • DUT hears audible remote ringback. • BroadWorks User A answers the call. • Two-way voice path is established. • Verify the signaling to the DUT. • BroadWorks sends an 18x with early-offer to the DUT. • The DUT sends a PRACK with early-answer.	Pass
SIP-T_Call_5.10	DUT Sends Early-Session Offer (P2)	• BroadWorks User A dials the DUT. • DUT is alerted. • BroadWorks User A hears audible remote ringback. • DUT answers the call. • Two-way voice path is established. • Verify the signaling to the DUT. • DUT sends an 18x with early-offer. • BroadWorks sends a PRACK with early-answer.	Pass
181 Call Being Forwarded			
SIP-T_Call_5.11	181 Call Being Forwarded (P1)	• DUT dials BroadWorks User A. • BroadWorks User B is alerted. • DUT receives audible ringback. • BroadWorks User B answers the call. • Two-way voice path is established. • DUT hangs up. • The call clears. • Verify the signaling to the DUT. • BroadWorks sends a 181 Call is Being Forwarded response to the DUT. • BroadWorks sends a 180 or 183 to the DUT. • The TO tag in the 180/183 or 200 OK may be different than the TO tag in 181.	Pass
Dial Plan			
SIP-T_Call_6.1	Local Operator Dialing (0) (P0)		Pass
SIP-T_Call_6.2	International Dialing (00XXXXXXXXX) (P0)	• DUT dials an international number (prefixed by "00"). • Verify the signaling from the DUT. • DUT sends SIP INVITE with Request-URI containing the international number dialed, 00xxxxxxxxx@<sip-domain>.	Pass
SIP-T_Call_6.4	Extension Dialing (XXXX#) (P0)	• DUT dials the extension for BroadWorks User A (xxxx#). • Verify the signaling from the DUT. • DUT sends SIP INVITE with Request-URI containing the extension dialed, 1234@<sip-domain> or 1234#@<sip-domain>.	Pass
SIP-T_Call_6.5	Feature Access Code Dialing (*XX) (P0)	• DUT dials the feature access code (*xx). • Verify the signaling from the DUT. • DUT sends SIP INVITE with Request-URI containing the feature access code, *73@<sip-domain>.	Pass
SIP-T_Call_6.6	Interrogation Feature Access Code Dialing (*XX*) (P0)	• DUT dials the feature access code (*xx*). • Verify the signaling from the DUT. • DUT sends SIP INVITE with Request-URI containing the feature access code, *21*@<sip-domain>.	Pass
DTMF			

SIP-T_Call_7.1	In-band DTMF (P0)	• DUT dials the Auto Attendant. • DUT hears a greeting and a prompt provided by the Auto Attendant. • DUT responds to the Auto Attendant prompts and supplies the necessary digits to transfer the call to BroadWorks User A. • BroadWorks User A is alerted. • DUT hears audible ringback. • Verify the signaling from the DUT. • DUT sends INVITE to BroadWorks without the RFC 2833 payload type in the SDP.	Pass
SIP-T_Call_7.2	RFC 2833 DTMF Offered, In-band DTMF Negotiated (P0)	Verify the following: • DUT dials the Auto Attendant. • DUT hears a greeting and a prompt provided by the Auto Attendant. • DUT responds to the Auto Attendant prompts and supplies the necessary digits to transfer the call to BroadWorks User A. • BroadWorks User A is alerted. • DUT hears audible ringback. • Verify the signaling to/from the DUT. • DUT sends INVITE to BroadWorks with the RFC 2833 payload type in the SDP. Example: rtpmap:101 telephone-event/8000 • Verify the BroadWorks returns an SDP without the RFC 2833 payload type. • Inspect the RTP to ensure that there is no RFC 2833 DTMF sent by the DUT.	Pass
SIP-T_Call_7.3	RFC 2833 DTMF (P0)	• DUT dials the Auto Attendant. • DUT hears a greeting and a prompt provided by the Auto Attendant. • DUT responds to the Auto Attendant prompts and supplies the necessary digits to transfer the call to BroadWorks User A. • BroadWorks User A is alerted. • DUT hears audible ringback. • Verify the signaling to/from the DUT. • DUT sends INVITE to BroadWorks with the RFC 2833 payload type in the SDP. Example: rtpmap:101 telephone-event/8000 • Verify the BroadWorks returns an SDP with the RFC 2833 payload type. • Inspect the RTP to ensure that RFC 2833 DTMF is sent by the DUT.	Pass
SIP-T_Call_7.4	DTMF Relay (P2)	• DUT dials the Auto Attendant. • DUT hears a greeting and a prompt provided by the Auto Attendant. • DUT responds to the Auto Attendant prompts and supplies the necessary digits to transfer the call to BroadWorks User A. • BroadWorks User A is alerted. • DUT hears audible ringback. • Verify the signaling to/from the DUT. • DUT sends SIP INFO messages for DTMF-relay. • DUT does not send inband or RFC 2833 DTMF in addition to DTMF-relay.	Pass
Codec Negotiation/Renegotiation			
SIP-T_Call_8.1	Codec Negotiation: re-INVITE without SDP (P0)	• BroadWorks User A dials BroadWorks User B. • BroadWorks User B is alerted. • BroadWorks User A receives audible ringback. • DUT dials *98 to pick up the call. • Two-way voice path is established between the DUT and BroadWorks User A. • Verify the signaling. • Inspect the SDP in the 200 OK from DUT in response to reINVITE without SDP from BroadWorks. • The SDP must contain all of the DUT's supported and enabled codecs. • The version in the o-line must be incremented. **NOTE**: SfB rejects the call with 491 (SF00500447)	Fail
SIP-T_Call_8.2	Codec Negotiation: Initial Answer with HOLD SDP (P0)	• BroadWorks User A dials BroadWorks User B. • BroadWorks User B is alerted. • BroadWorks User A receives audible ringback. • BroadWorks User B answers the call. • Two-way voice path is established. • BroadWorks User A dials a second call to Call Park feature code *68 to park the call. • BroadWorks User A hears IVR announcement requesting a number to park the call against. • BroadWorks User A supplies the DUT extension to park the call against. • BroadWorks User A hears IVR announcement indicating the call has been parked. • BroadWorks User B is now held. • BroadWorks User A hangs up. • DUT dials *88+<DUT DN> to retrieve the parked call. • Two-way voice path is established between the DUT and BroadWorks User B. • Verify the signaling to/from the DUT. • BroadWorks sends HOLD SDP in the 200 OK answer to the initial INVITE from the DUT. • BroadWorks sends a re-INVITE with B's SDP to the DUT. • DUT sends 200 OK to BroadWorks with ANSWER SDP.	Pass

SIP-T_Call_8.3	Codec Renegotiation: Blind Transfer (P0)	• BroadWorks User A dials the DUT. • DUT is alerted. • BroadWorks User A receives audible ringback. • DUT answers the call. • Two-way voice path is established between BroadWorks User A and the DUT. • BroadWorks User A uses the CM to blind transfer the DUT to BroadWorks User B by entering BroadWorks User B's extension and selecting the Transfer button. • BroadWorks User A is released. • BroadWorks User B is alerted. • DUT receives audible ringback. • BroadWorks User B answers the call. • Two-way voice path is established between BroadWorks User B and the DUT. • Verify the SIP signaling and RTP. • Check initial 200 OK and RTP stream to determine negotiated codec between BroadWorks User A and the DUT. • During the transfer, BroadWorks sends the DUT a re-INVITE without SDP. • DUT responds with 200 OK containing OFFER SDP. • BroadWorks sends ACK containing ANSWER SDP. • Check 200 OK and RTP stream to determine negotiated codec between BroadWorks User B and the DUT. This must be different from the codec negotiated with BroadWorks User A	Pass
SIP-T_Call_8.4	Codec Renegotiation: Attended Transfer (P0)	• BroadWorks User A dials the BroadWorks User B. • BroadWorks User B is alerted. • BroadWorks User A receives audible ringback. • BroadWorks User B answers the call. • Two-way voice path is established between BroadWorks User A and BroadWorks User B. • BroadWorks User A uses the CM to call the DUT by entering the DUT's extension and selecting the Dial button. • BroadWorks User B is held. • DUT is alerted. • BroadWorks User A receives audible ringback. • DUT answers the call. • Two-way voice path is established between BroadWorks User A and the DUT. • BroadWorks User A uses the CM to transfer BroadWorks User B to the DUT by selecting Transfer. • BroadWorks User A is released. • Two-way voice path is established between BroadWorks User B and the DUT. • Verify the SIP signaling and RTP. • Check initial 200 OK and RTP stream to determine negotiated codec between BroadWorks User A and the DUT. • During the transfer, BroadWorks sends the DUT a re-INVITE without SDP. • DUT responds with 200 OK containing OFFER SDP. • BroadWorks sends ACK containing ANSWER SDP. • Check 200 OK and RTP stream to determine negotiated codec between BroadWorks User B and the DUT. This must be different from the codec negotiated with BroadWorks User A.	Pass
SIP-T_Call_8.5	Codec Renegotiation: Blind Transfer of Call on Hold (P0)	• BroadWorks User A dials the DUT. • DUT is alerted. • BroadWorks User A receives audible ringback. • DUT answers the call. • Two-way voice path is established between BroadWorks User A and the DUT. • DUT holds the call by using the DUT's hold mechanism. • The call is held (no voice path in either direction). • BroadWorks User A uses the CM to blind transfer the DUT to BroadWorks User B by entering BroadWorks User B's extension and selecting the Transfer button. • BroadWorks User A is released. • BroadWorks User B is alerted. • DUT does not receive audible ringback, as the call is still on hold. • BroadWorks User B answers the call. • Call is still held (no voice path in either direction). • DUT retrieves the held call. • Two-way voice path is established between BroadWorks User B and the DUT. • Verify the SIP signaling and RTP. • Check initial 200 OK and RTP stream to determine negotiated codec between BroadWorks User A and the DUT. • DUT sends re-INVITE with hold SDP. • DUT sends re-INVITE to resume. • Check 200 OK and RTP stream to determine negotiated codec between BroadWorks User B and the DUT. This must be different from the codec negotiated with BroadWorks User A.	Pass
SIP-T_Call_8.6	BWKS to DUT where none supported codec is offered by BWKS		
SIP Connect Package			

SIP-T_Call_9.1.1	CONNECT – GIN REGISTER (P1)	• DUT is restarted. • Verify the SIP signaling to/from the DUT. • DUT sends GIN REGISTER request to BroadWorks for the PBX main line or pilot number. Example: REGISTER sip:enterprise.com SIP/2.0 Via: SIP/2.0/UDP 10.16.145.102:5060;branch=z9hG4bK-c64fe4c From: <sip:+13441004000@enterprise.com>;tag=54321 To: <sip:+13441004000@enterprise.com> Contact: <sip:192.0.2.100;bnc>;expires=3600 Call-ID: 98765abcde CSeq: 28337 REGISTER Max-Forwards: 70 Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER Require: gin Proxy-Require: gin Supported: path Content-Length: 0 • BroadWorks responds with 200 OK.	Pass
SIP-T_Call_9.1.2	CONNECT – Call to PBX User (P1)	• BroadWorks User A dials the DUT (to a PBX user, not the PBX main line). • DUT is alerted. • BroadWorks User A receives audible ringback. • DUT answers the call. • Two-way voice path is established. • DUT hangs up the call. • The call is released. • Verify the SIP signaling to/from the DUT. • BroadWorks sends an INVITE to the DUT with the PBX user number in the Request-URI and To headers. • After answer, DUT sends 200 OK.	Pass
SIP-T_Call_9.1.3	CONNECT – Call from PBX User (P1)	• DUT (PBX user, not the PBX main line) dials BroadWorks User A. • BroadWorks User A is alerted. • DUT receives audible ringback. • BroadWorks User A answers the call. • Two-way voice path is established. • DUT hangs up the call. • The call is released. • Verify the SIP signaling to/from the DUT. • DUT sends an INVITE with the PBX user number in the From. • After answer, BroadWorks sends 200 OK.	Pass
SIP-T_Call_9.1.4	Test with (DUT) sending the bad password	BWSKS send 403 back	Pass
SIP-T_Call_9.1.5	Registering against an alphanumeric authentication username		Pass
SIP Connect – PBX Redirect			
SIP-T_Call_9.2.1	CONNECT – BroadWorks to DUT; Call Forward to BroadWorks (P1)	• BroadWorks User A dials the DUT. • The call forwards immediately to BroadWorks User B. • BroadWorks User B is alerted. • BroadWorks User A receives audible ringback. • BroadWorks User B answers the call. • Two-way voice path is established. • BroadWorks User B hangs up. • The call is released upon disconnect. • Verify the SIP signaling to/from the DUT. • The DUT sends a 302 redirect with the PBX user's number in a Diversion or History-Info header. • Alternatively, the DUT sends a new INVITE with BroadWorks User A's number in the From and the DUT's number in a Diversion or History-Info header.	Pass
SIP-T_Call_9.2.2	CONNECT – DUT to DUT; Call Forward to BroadWorks (P1)	• DUT PBX user dials the DUT PBX user with Call Forwarding enabled. • The call forwards immediately to BroadWorks User B. • BroadWorks User B is alerted. • DUT PBX user receives audible ringback. • BroadWorks User B answers the call. • Two-way voice path is established. • BroadWorks User B hangs up. • The call is released upon disconnect. • Verify the SIP signaling to/from the DUT. • The first INVITE from DUT PBX user to DUT PBX user must be internal to the PBX, not sent to BroadWorks. • For the Call Forward, the DUT sends a new INVITE with the calling DUT's number in the From and the forwarding DUT's number in a Diversion or History-Info header.	Pass

SIP-T_Call_9.2.3	CONNECT – DUT to DUT; Blind Transfer to BroadWorks (P1)	• First DUT PBX user dials second DUT PBX user. • Second DUT PBX user is alerted. • First DUT PBX user receives audible ringback. • Second DUT PBX user answers the call. • Two-way voice path is established. • Second DUT PBX blind transfers the call to BroadWorks User B. • BroadWorks User B is alerted. • First DUT PBX user receives audible ringback. • BroadWorks User B hangs up. • The call is released upon disconnect. • Verify the SIP signaling to/from the DUT. • The first INVITE from DUT PBX user to DUT PBX user must be internal to the PBX, not sent to BroadWorks. • For the Call Transfer, the DUT sends a new INVITE with the calling DUT’s number in the From and the forwarding DUT’s number in a Diversion or History-Info header.	Pass
SIP-T_Call_9.2.4	CONNECT - DUT to Broadworks; CFA set on BroadWorks UserA to DUT userB		Pass
SIP-T_Call_9.2.5	CONNECT - BroadWorks to DUT; CFA set on DUT UserA to DUT UserB		Pass
SIP Connect – Calling Line ID and Privacy			
SIP-T_Call_9.3.1	CONNECT – BroadWorks to DUT; Caller ID Presented (P1)	• BroadWorks User A dials the DUT. • DUT is alerted. • DUT endpoint displays the calling name and number for BroadWorks User A. • BroadWorks User A receives audible ringback. • DUT answers the call. • Two-way voice path is established. • BroadWorks User A hangs up. • The call is released upon disconnect. • Verify the SIP signaling to/from the DUT. • The INVITE from BroadWorks to the DUT does not contain Privacy headers (P-Asserted-Identity and Privacy) and From header contains the originator's calling name and number. Example: From:"Bill Jameson"<sip:+12408888130@as.broadworks.net;user=phone>;tag=1525954323-1216666352531	Pass
SIP-T_Call_9.3.2	CONNECT – BroadWorks to DUT; Caller ID Blocked (P1)	• BroadWorks User C dials the DUT. • DUT is alerted. • DUT endpoint displays “Anonymous” or other indication of a restricted call. The DUT endpoint must not display the calling name or number. • BroadWorks User C receives audible ringback. • DUT answers the call. • Two-way voice path is established. • BroadWorks User C hangs up. • The call is released upon disconnect. • Verify the SIP signaling to/from the DUT. • The INVITE from BroadWorks to the DUT does not contain Privacy headers (P-Asserted-Identity and Privacy). The From header should also be encrypted as anonymous. Example: From:<sip:anonymous@anonymous.invalid>;user=phone>;tag=1525954323-1216666352531	Pass
SIP-T_Call_9.3.3	CONNECT – BroadWorks to DUT; Caller ID with Unicode (non-ASCII) Characters Presented (P1)	• BroadWorks User A dials the DUT. • DUT is alerted. • DUT endpoint displays the calling name and number for BroadWorks User A. The Unicode characters in the Calling Name are properly displayed. • BroadWorks User A receives audible ringback. • DUT answers the call. • Two-way voice path is established. • BroadWorks User A hangs up. • The call is released upon disconnect. • Verify the SIP signaling to/from the DUT. • The INVITE from BroadWorks to the DUT does not contain a Privacy headers (P-Asserted-Identity and Privacy). From header contains the originator’s calling name and number. The caller name contains Unicode characters. Example: From:"Björk Guðmundsdóttir "<sip:+12408888130@as.broadworks.net;user=phone>; tag=1525954323-1216666352531	Pass

SIP-T_Call_9.3.4	CONNECT – DUT to BroadWorks; Caller ID Presented (P1)	• DUT dials BroadWorks User A. • BroadWorks User A is alerted. • BroadWorks User A endpoint displays calling name and number. • Note that BroadWorks sends the DUT's calling name and number as configured on BroadWorks to the remote endpoint. This overrides the name and number as sent in the INVITE from the DUT, so it may differ. • DUT receives audible ringback. • BroadWorks User A answers the call. • Two-way voice path is established. • DUT hangs up. • The call is released upon disconnect. • Verify the SIP signaling to/from the DUT. • The INVITE from the DUT to BroadWorks contains a Privacy header set to "none" and P-Asserted-Identity and From headers containing the originator's calling name and number. Example: From:"Scott Bean"<sip:+13018881001@as.broadworks.net;user=phone>;tag=1525954323-1216666352531 P-Asserted-Identity:"Scott Bean"<sip:+13018881001@as.broadworks.net;user=phone Privacy:none	Pass
SIP-T_Call_9.3.5	CONNECT – DUT to BroadWorks; Caller ID Blocked (P1)	• DUT dials BroadWorks User C. • BroadWorks User C is alerted. • BroadWorks User C's endpoint displays "Anonymous" or other indication of a restricted call. • DUT receives audible ringback. • BroadWorks User C answers the call. • Two-way voice path is established. • DUT hangs up. • The call is released upon disconnect. • Verify the SIP signaling to/from the DUT. • The INVITE from the DUT to BroadWorks contains a Privacy header set to "id". The P-Asserted-Identity header contains the originator's calling name and number. The From header may also be encrypted as anonymous. Example: From:<sip:anonymous@anonymous.invalid>;user=phone>;tag=1525954323-1216666352531 P-Asserted-Identity:"Scott Bean"<sip:+13018881001@as.broadworks.net;user=phone Privacy: id	Pass
Emergency Calls Test			
SIP-T_Call_10.1	DUT (Pilot User) to 999.	• DUT dials 999. • Early operator is is alerted.DUT receives audible ringback. • Early Operator answers the call and verifies the CLI • Two way voice path is established. • Emergency operator floats the call to the emergency agent. • Three way call is established. • Early operator leaves the call. • Two way call between DUT and emergency agent continues. • DUT hangs up. • The call is released. • Verify the SIP signaling to/from the DUT. • Make sure the DUT sends all messages to the primary Application Server address.	Pass
SIP-T_Call_10.2	DUT (Non Pilot User) to 999.	• DUT dials 999. • Early operator is is alerted.DUT receives audible ringback. • Early Operator answers the call and verifies the CLI. • Two way voice path is established. • Emergency operator floats the call to the emergency agent. • Three way call is established. • Early operator leaves the call. • Two way call between DUT and emergency agent continues. • DUT hangs up. • The call is released. • Verify the SIP signaling to/from the DUT. • Make sure the DUT sends all messages to the primary Application Server address.	Pass
SIP-T_Call_10.3	DUT (Pilot User) to 112.	• DUT dials 112. • Early operator is is alerted.DUT receives audible ringback. • Early Operator answers the call and verifies the CLI. • Two way voice path is established. • Emergency operator floats the call to the emergency agent. • Three way call is established. • Early operator leaves the call. • Two way call between DUT and emergency agent continues. • DUT hangs up. • The call is released. • Verify the SIP signaling to/from the DUT. • Make sure the DUT sends all messages to the primary Application Server address.	Pass

SIP-T_Call_10.4	DUT (Non Pilot User) to 112.	• DUT dials 112. • Early operator is is alerted.DUT receives audible ringback. • Early Operator answers the call and verifies the CLI. • Two way voice path is established. • Emergency operator floats the call to the emergency agent. • Three way call is established. • Early operator leaves the call. • Two way call between DUT and emergency agent continues. • DUT hangs up. • The call is released. • Verify the SIP signaling to/from the DUT. • Make sure the DUT sends all messages to the primary Application Server address.	Pass
PSTN Call Tests			
SIP-T_Call_12.1	Call from the PBX pilot to PSTN.	• DUT dials the number of PSTN User A. • PSTN User A is alerted. • DUT receives audible ringback. • PSTN User A answers the call. • Two-way voice path is established. • DUT hangs up. • The call is released. • Verify the SIP signaling to/from the DUT • Make sure the DUT sends all messages to the primary Application Server address.	Pass
SIP-T_Call_12.2	Call from PSTN to the PBX pilot.	• PSTN User A dials the DUT. • DUT is alerted. • PSTN User A receives audible ringback. • DUT answers the call. • Two-way voice path is established. • PSTN User A hangs up. • The call is released upon disconnect. • Verify the SIP signaling to/from the DUT. • Make sure the DUT sends all messages to the primary Application Server address.	Pass
SIP-T_Call_12.3	Call from the PBX non pilot user to PSTN.	• DUT dials the number of PSTN User A. • PSTN User A is alerted. • DUT receives audible ringback. • PSTN User A answers the call. • Two-way voice path is established. • DUT hangs up. • The call is released. • Verify the SIP signaling to/from the DUT • Make sure the DUT sends all messages to the primary Application Server address.	Pass
SIP-T_Call_12.4	Call from PSTN to the PBX non pilot user.	• PSTN User A dials the DUT. • DUT is alerted. • PSTN User A receives audible ringback. • DUT answers the call. • Two-way voice path is established. • PSTN User A hangs up. • The call is released upon disconnect. • Verify the SIP signaling to/from the DUT. • Make sure the DUT sends all messages to the primary Application Server address.	Pass
SIP-T_Call_12.5	Anonymous call from PSTN to PBX pilot.	• PSTN User A dials the DUT, with 141 preceding the DUTs number. • DUT is alerted. • PSTN User A receives audible ringback. • DUT answers the call. • Two-way voice path is established. • PSTN User A hangs up. • The call is released upon disconnect. • Verify the SIP signaling to/from the DUT. • Make sure the DUT sends all messages to the primary Application Server address.	Pass
SIP-T_Call_12.6	Anonymous call from PBX pilot to PSTN.	• Apply 'Calling Line ID Blocking' on business portal for the DUT. DUT dials the number of PSTN User A. • PSTN User A is alerted. • DUT receives audible ringback. • PSTN User A answers the call. • Two-way voice path is established. • DUT hangs up. • The call is released. • Verify the SIP signaling to/from the DUT • Make sure the DUT sends all messages to the primary Application Server address.	Pass
Conferencing			

SIP-T_Call_13.1	Ad-hoc conferancing, DUT to Broadworks User A and BroadWorks User B.		Pass
SIP-T_Call_13.2	Ad-hoc conferancing, Broadworks User A to DUT and BroadWorks User B.		Pass

Conclusion

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