Sonus SBC 1000/2000 V6.1.2 IOT CounterPath Bria Desktop SoftClient SIP Application Note

Table of Contents

Document Overview

This Application Note is a configuration guide for the Sonus Session Border Controller (SBC) 1000/2000 Series when connecting to CounterPath Bria SoftClient.

This configuration guide supports features described on the CounterPath http://www.counterpath.com/bria/ website.

- For additional information on CounterPath Bria SoftClient, visit http://www.counterpath.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 and CounterPath Bria SoftClient.

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:

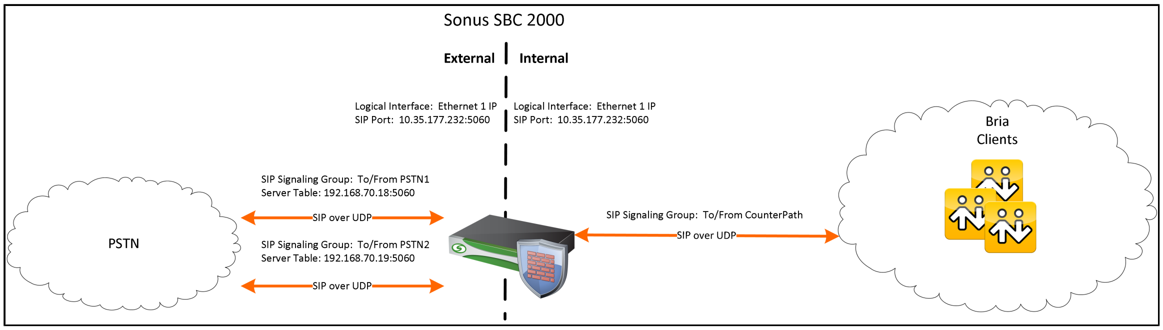
Table 1: Requirements

	Equipment	Software Version
Sonus Networks	SBC 2000	V6.1.2build471
Third-party Equipment	CounterPath Bria 4	Build 83451

Reference Configuration

The following reference configuration shows connectivity between CounterPath SoftClient 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 CounterPath test plan. The following features were tested:

- Basic Calls (Wired Ethernet Connection)
- Advanced Calls, (Wired Ethernet Connection)
- Basic Calls, Wireless Connectivity (Wireless Access Point)
- Advanced Calls, Wireless Connectivity(Wireless Access Point)
- Access Point Changes, Background Support
- Call Log, MWI
- Video Calling

Verify License

- SIP Calls
- SIP Registrations
- Video Passthrough

CounterPath Bria SoftClient Configuration

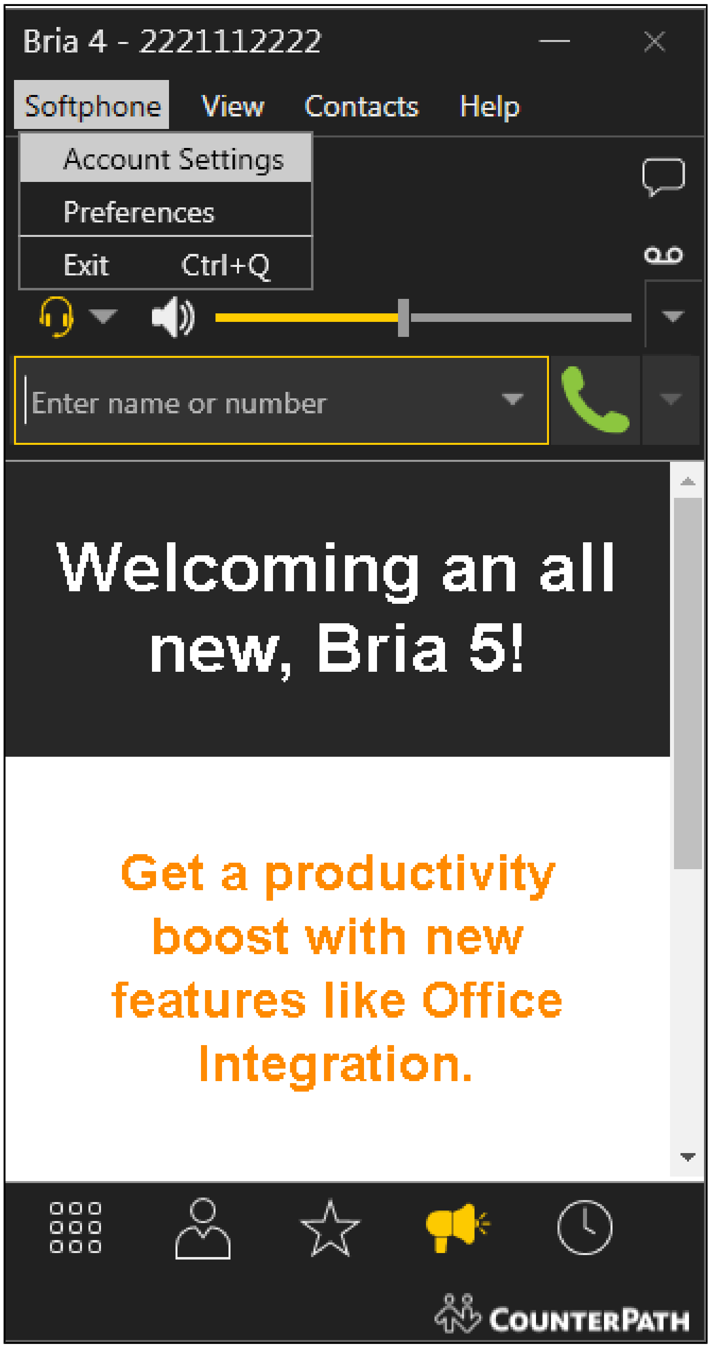
The following new configurations are included in this section:

1. Account Settings
2. Audio Codecs

1. Account Settings

Select **Softphone > Account Settings**

Figure 2: Account Settings



Select **Add > SIP Account**

Figure 3: Add SIP Account

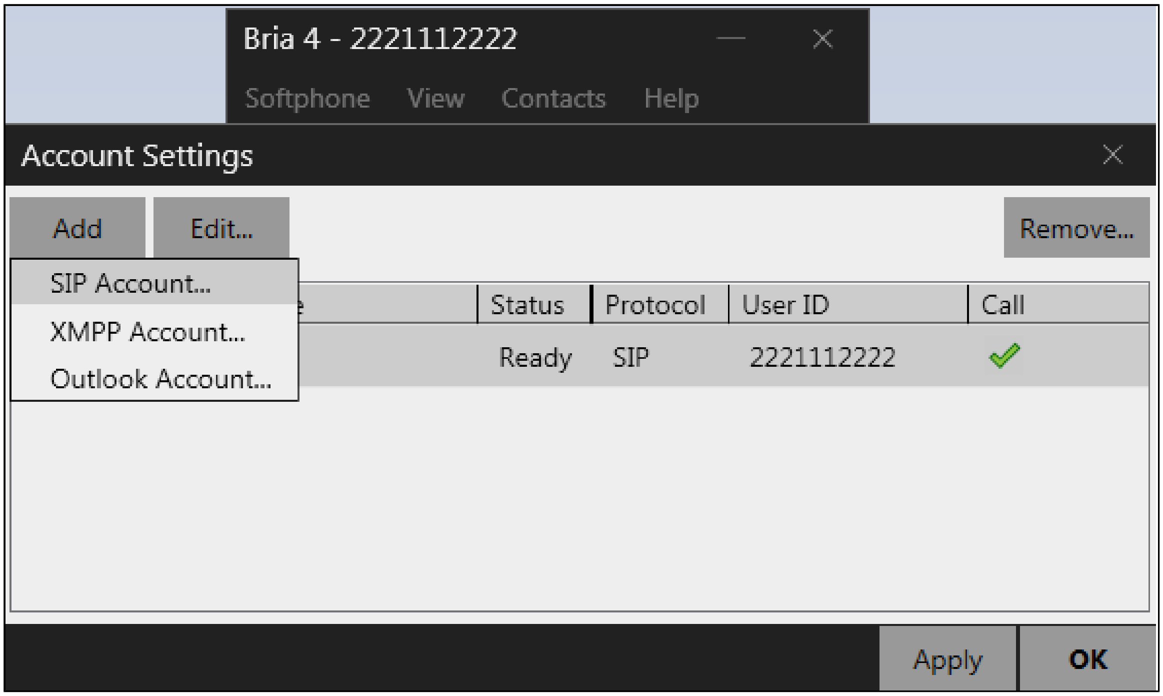
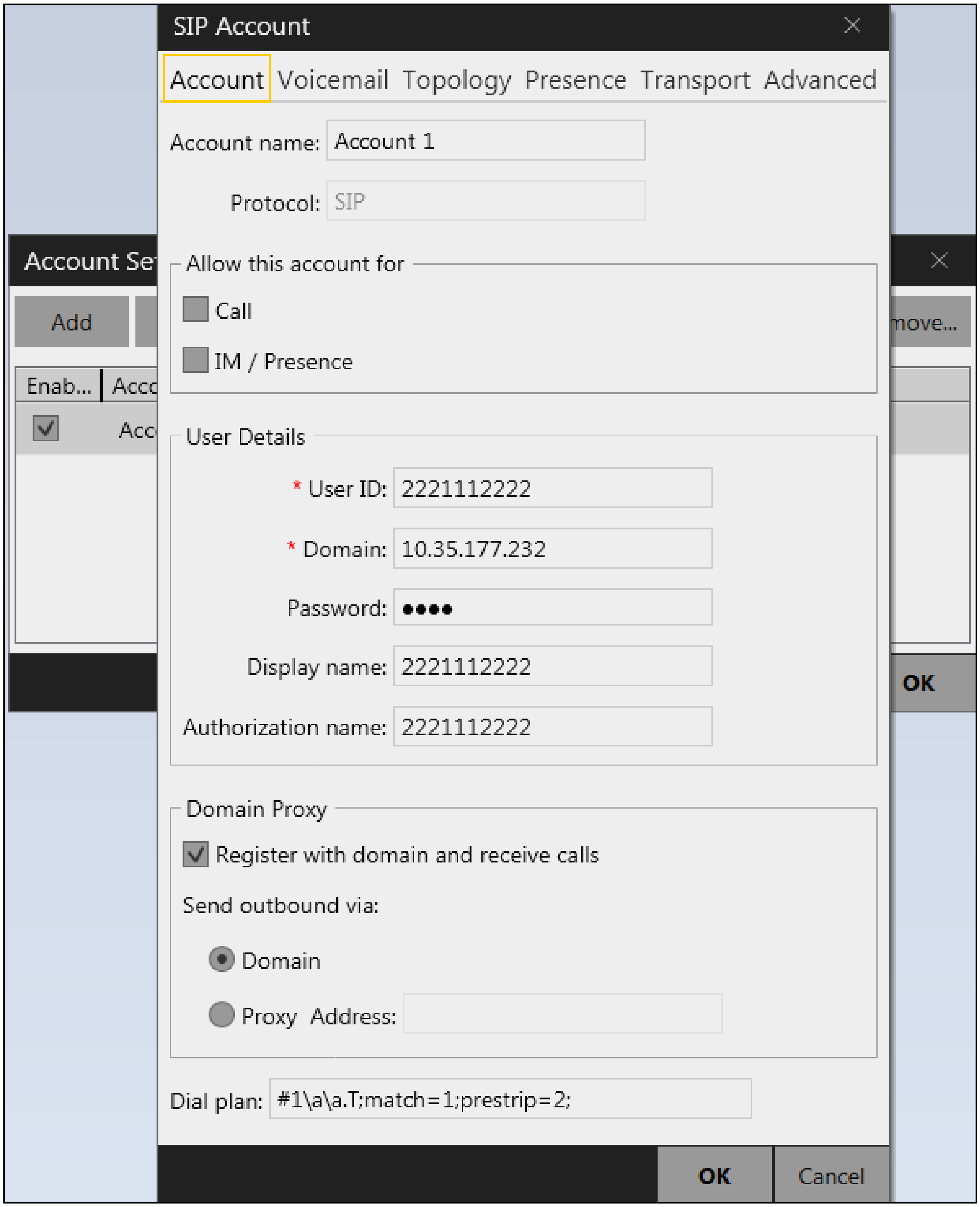


Figure 4: Add SIP Account Details



2. Audio Codecs

Select **Softphone > Preferences**

Figure 5: Preferences

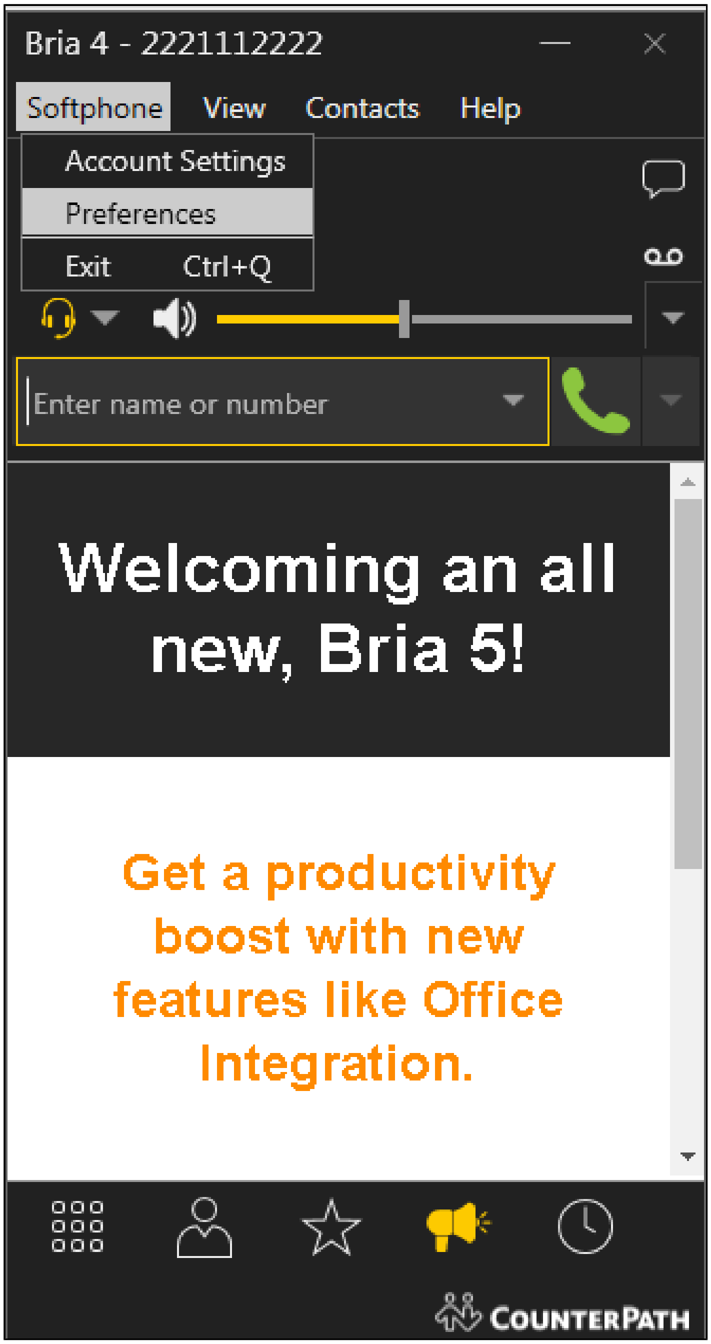
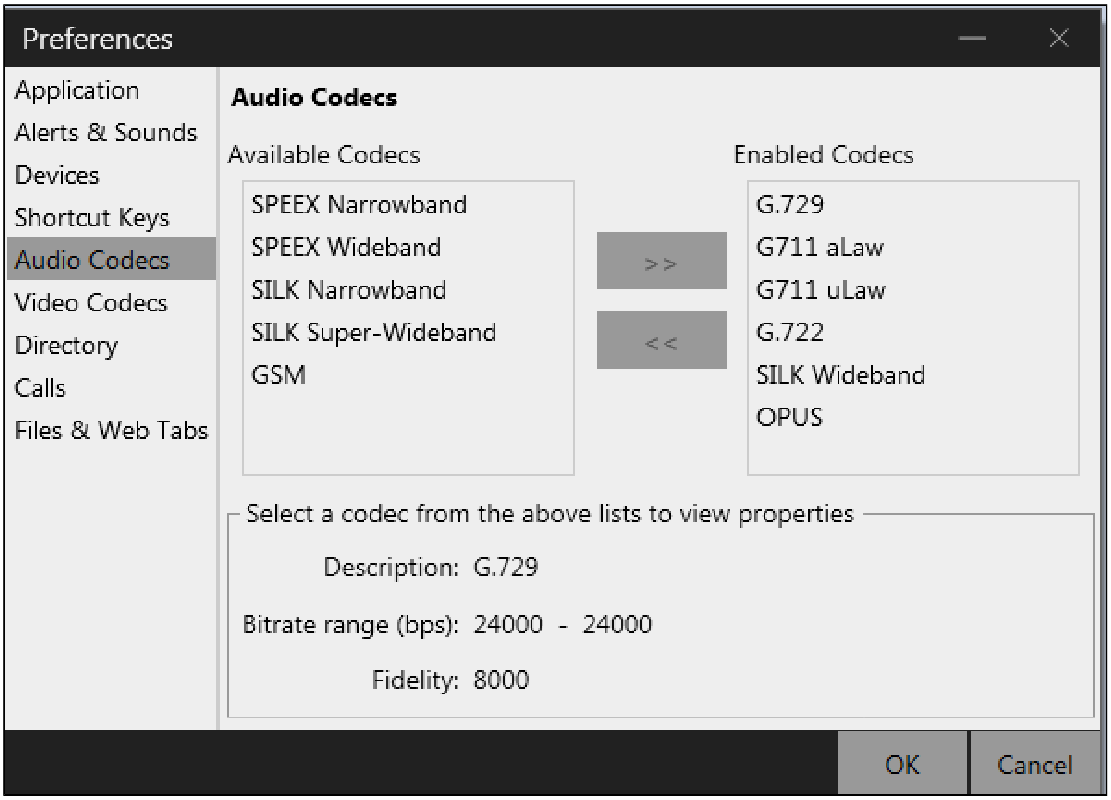


Figure 6: Add/Remove Audio Codecs



Sonus SBC 1000/2000 Configuration

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

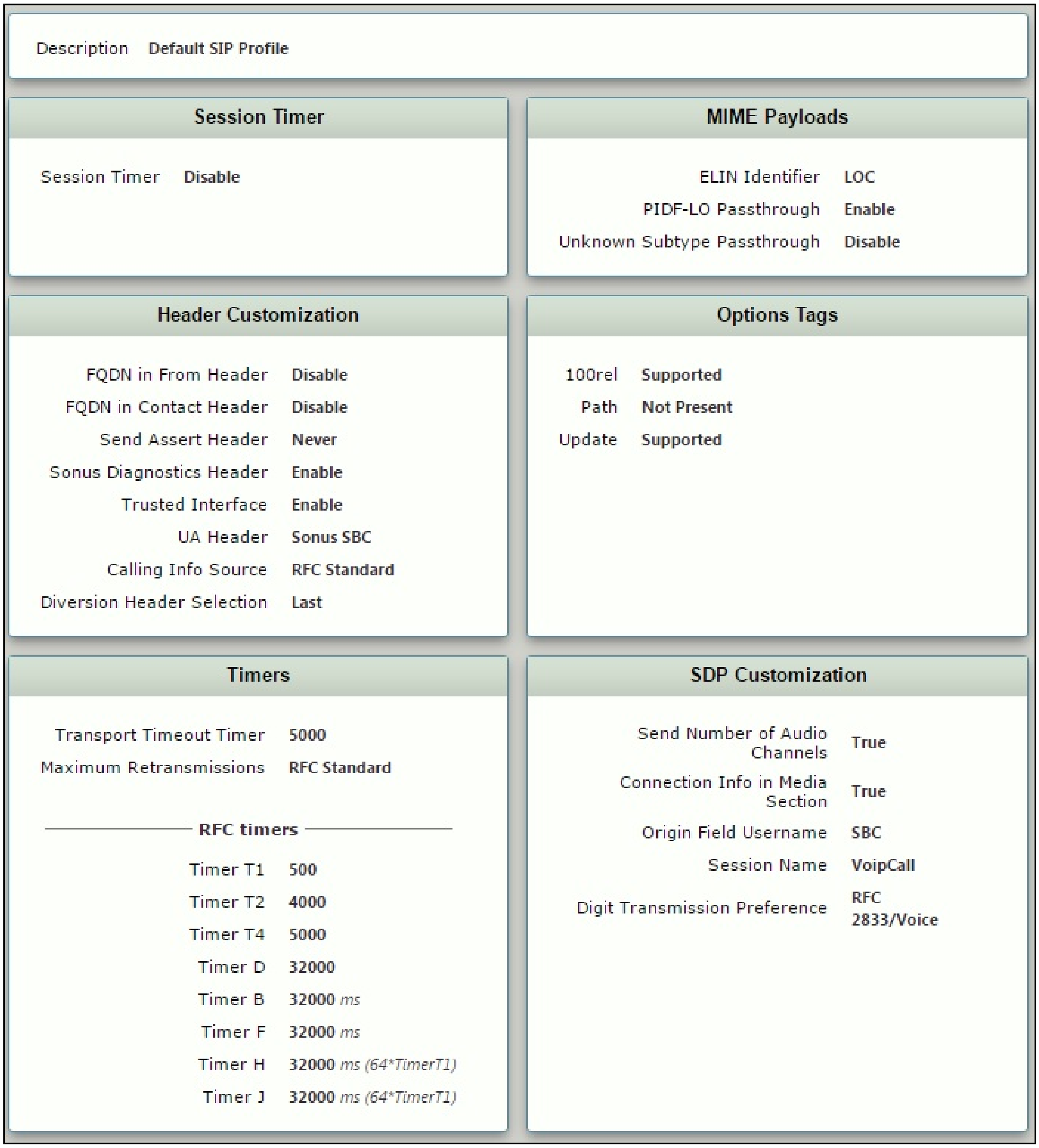
1. SIP Profile
2. SIP Server
3. Media Profile
4. Media List
5. Transformation Table
6. Call Routing Table
7. Registrar
8. Local/Pass-through Authorization Tables
9. Signaling Groups

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 7: SIP Profiles



2. 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 8: PSTN1

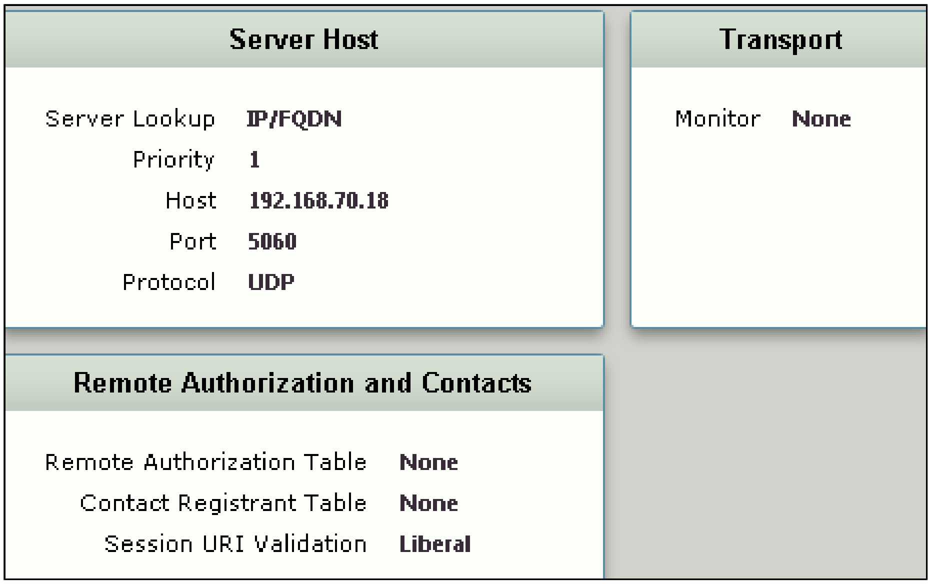
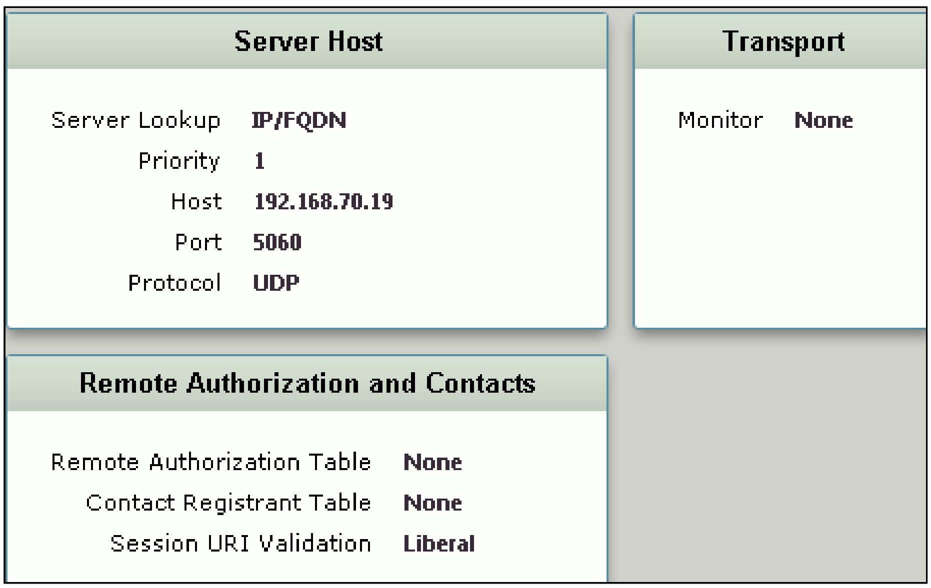


Figure 9: PSTN2



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 10: Voice Codec G711 A-Law

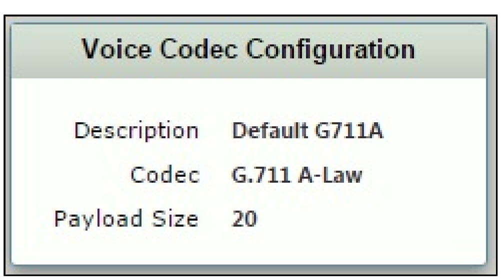


Figure 11: Voice Codec G711 U-Law

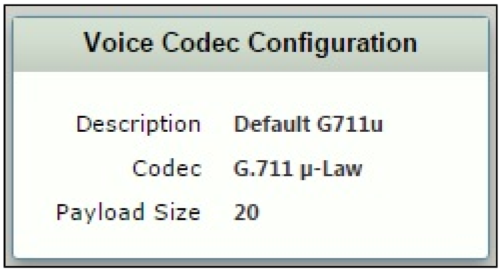


Figure 12: Voice Codec G729

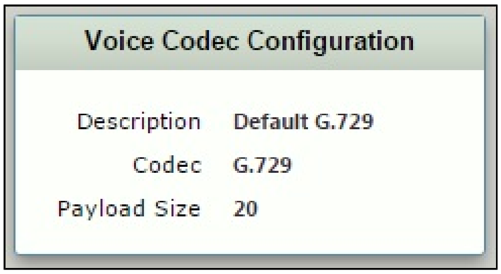
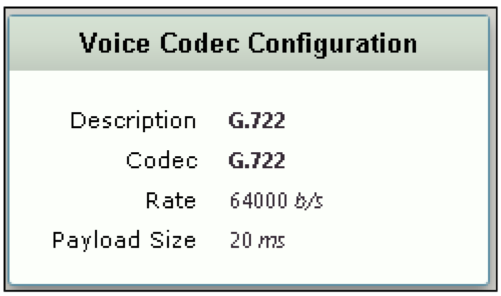


Figure 13: G.722

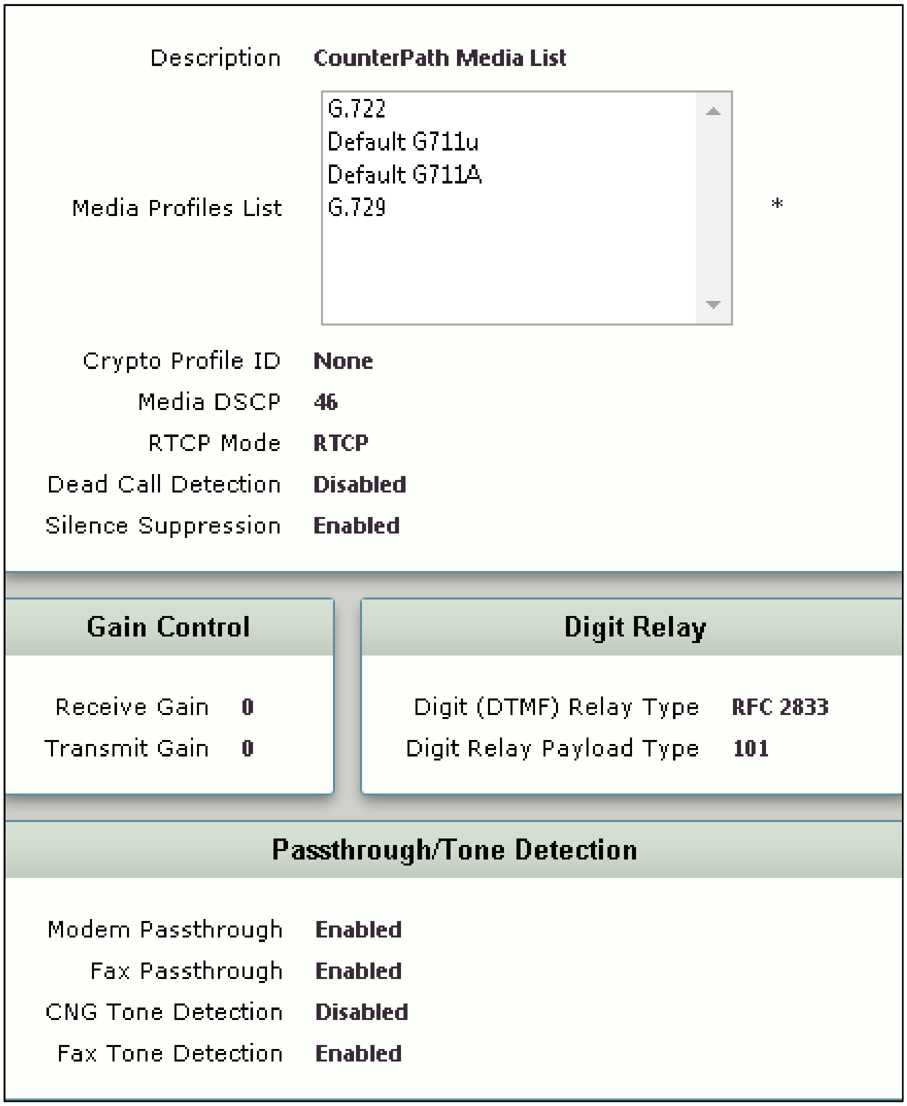


4. Media List

Select **Settings > Media > Media List**

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

Figure 14: Media List



5. 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 15: CounterPath

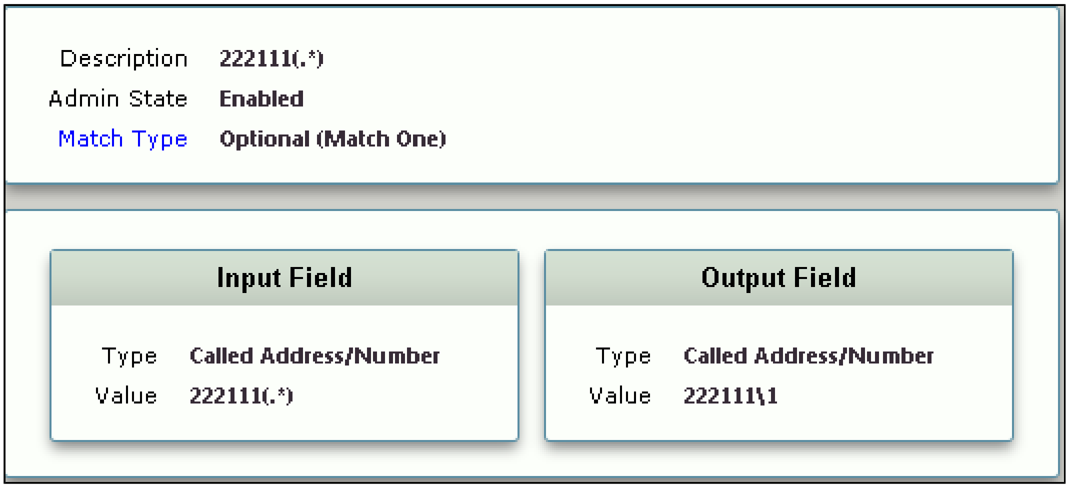


Figure 16: From CounterPath to PSTN1

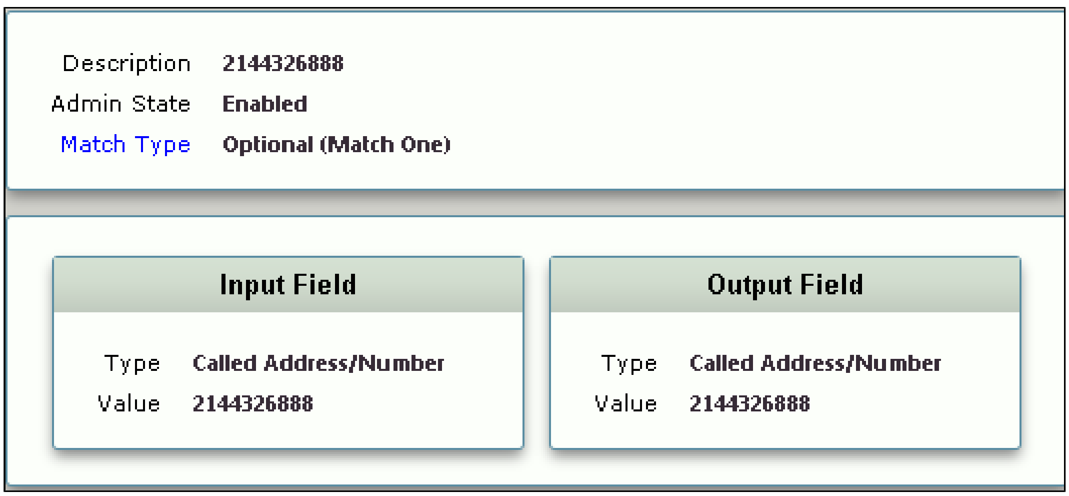
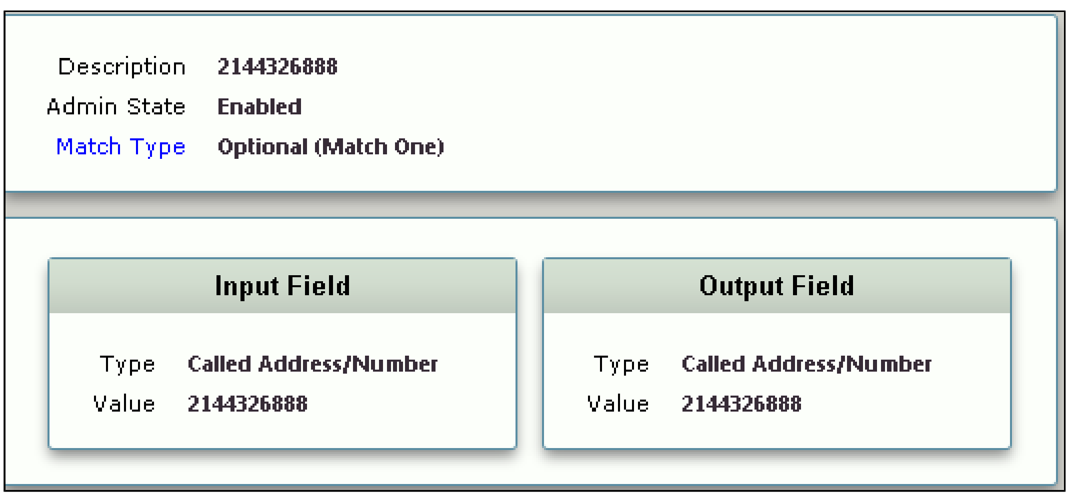


Figure 17: PSTN1



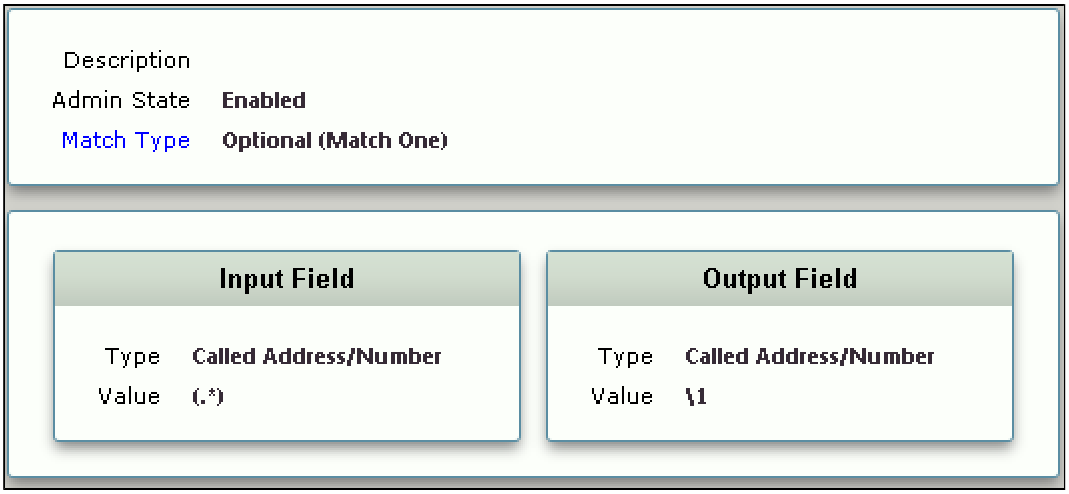


Figure 18: PSTN2

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 19: From CounterPath

Route Details

Description	**To Counterpath**
Admin State	**Enabled**
Route Priority	**1**
Call Priority	**Normal**
Number/Name Transformation Table	**CounterPath**

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) From/To CounterPath *
Enable Maximum Call Duration	**Disabled**

Media

Audio/Fax Stream Mode	**Direct**
Video/Application Stream Mode	**Direct**

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**

Route Details

Description	**To PSTN1**
Admin State	**Enabled**
Route Priority	**1**
Call Priority	**Normal**
Number/Name Transformation Table	**From CounterPath to PSTN1**

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) From/To PSTN1 *
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**

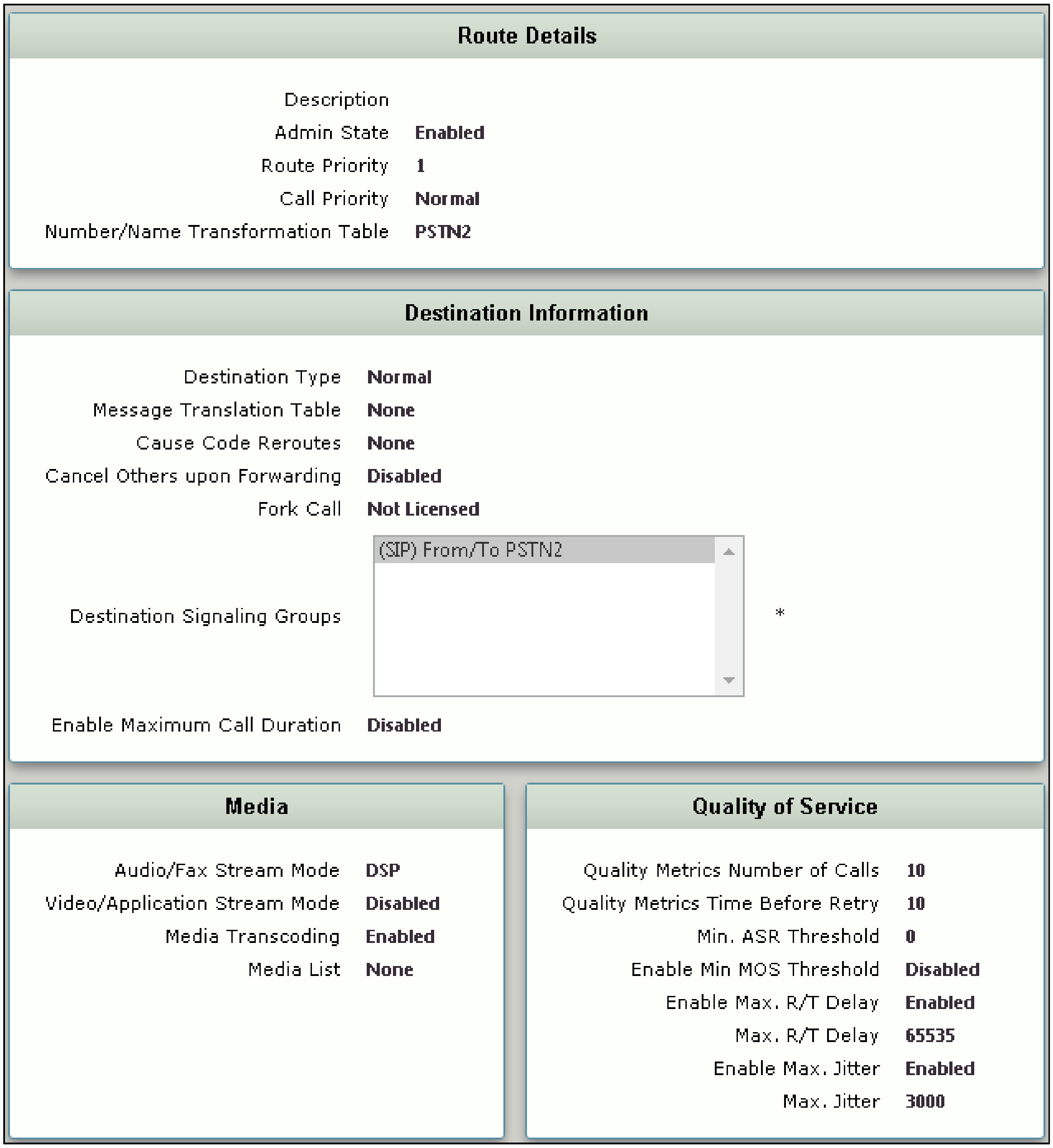
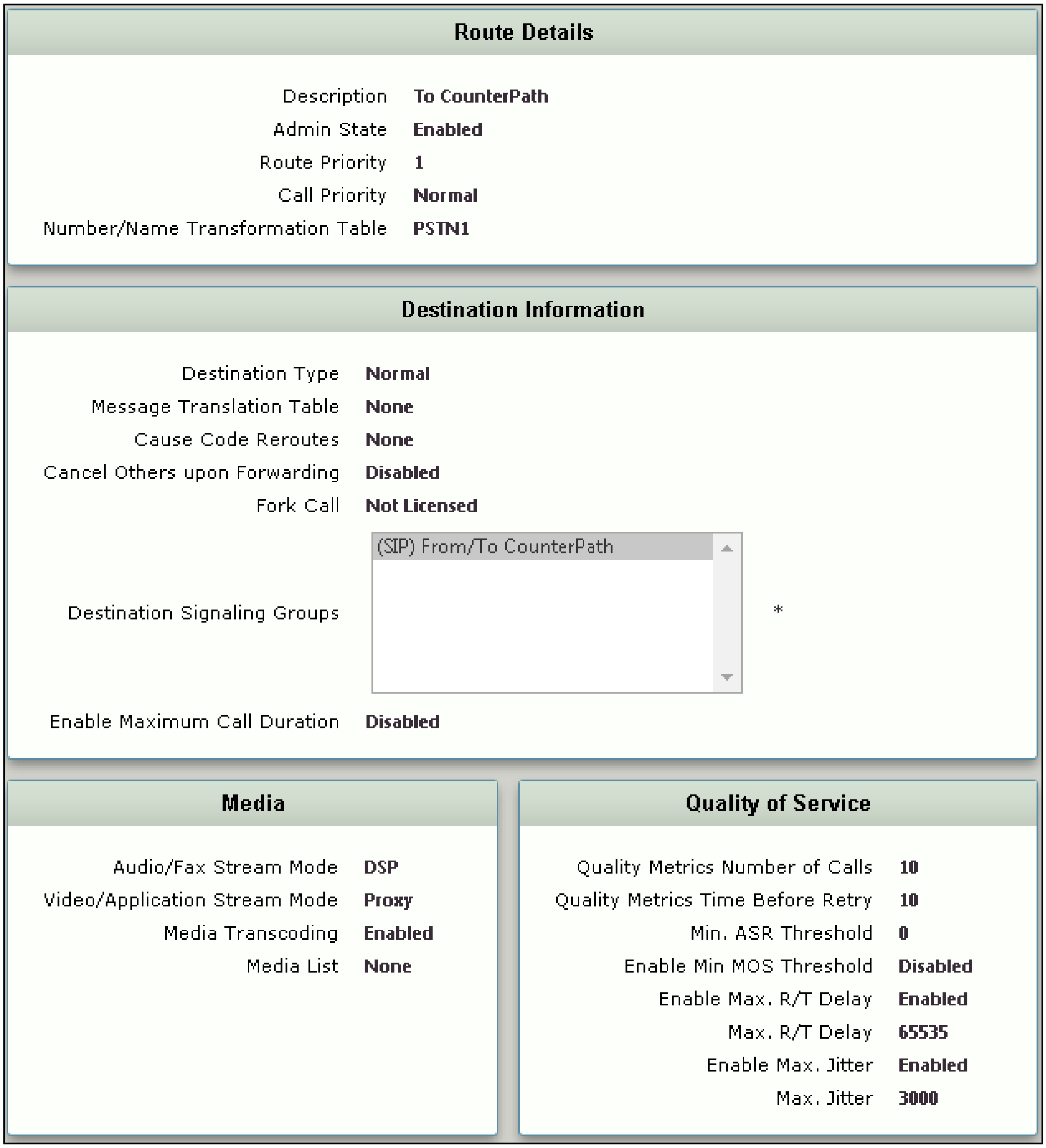


Figure 20: From PSTN1



Route Details

Description	To CounterPath
Admin State	Enabled
Route Priority	1
Call Priority	Normal
Number/Name Transformation Table	PSTN1

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) From/To CounterPath *
Enable Maximum Call Duration	Disabled

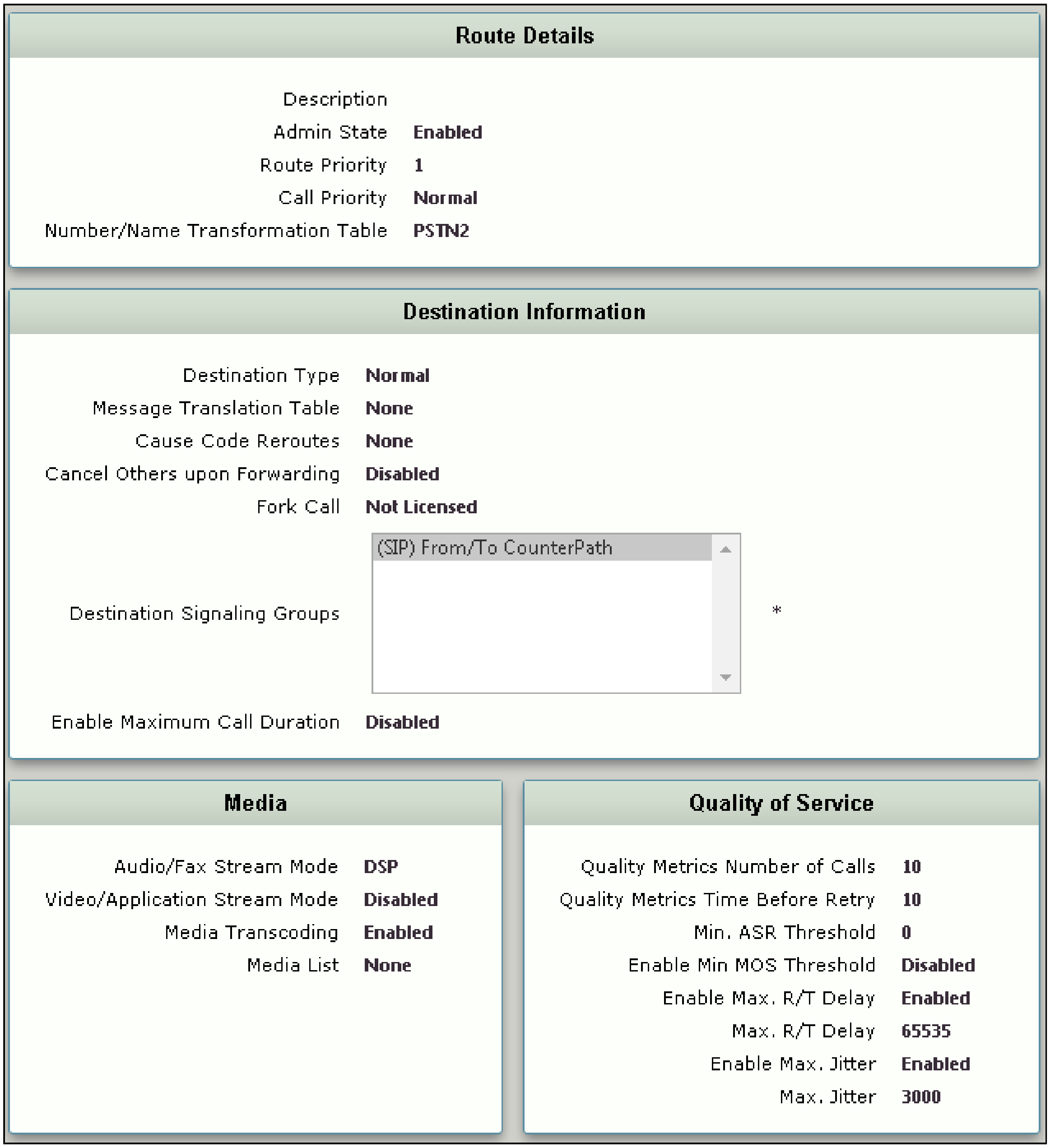
Media

Audio/Fax Stream Mode	DSP
Video/Application Stream Mode	Proxy
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 PSTN2



7. Registrar

Select **Settings > Local Registrars**

Figure 22: Registrar

8. Local/Pass-through Authorization Tables

Select **Settings > Local/Pass-through Authorization Tables**

Figure 23: Local/Pass-through Authorization Tables

Authorization Parameters

Type of Address of Record	Local
Address of Record URI	sip:2221112222

Authentication

User Name	2221112222
Password Setting	Use Current

9. 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 24: From/To CounterPath

Description	From/To CounterPath
Admin State	Enabled
Service Status	Up

SIP Channels and Routing

Action Set Table	None
Call Routing Table	From CounterPath
No. of Channels	60
SIP Profile	Default SIP Profile
SIP Mode	Local Registrar
Registrar	CounterPath
Agent Type	Back-to-Back User Agent
Registrar Min. TTL	600
Load Balancing	Round Robin
Channel Hunting	Most Idle
Notify Lync CAC Profile	Disable
Challenge Request	Enable
Authorization Realm	10.35.177.232
Local/Pass-thru Auth Table	CounterPath
Nonce Expiry	Forever
Outbound Proxy	
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
Forked Call Answered Too Soon	Disable

Media Information

Audio/Fax Stream Mode	DSP Proxy Direct *
Video/Application Stream Mode	Proxy Direct *
Media List ID	CounterPath 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

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 1 IP (10.35.177.232)
Signaling DSCP	40

Static NAT - Outbound

Outbound NAT Traversal	None

Static NAT - Inbound

Detection	Disabled

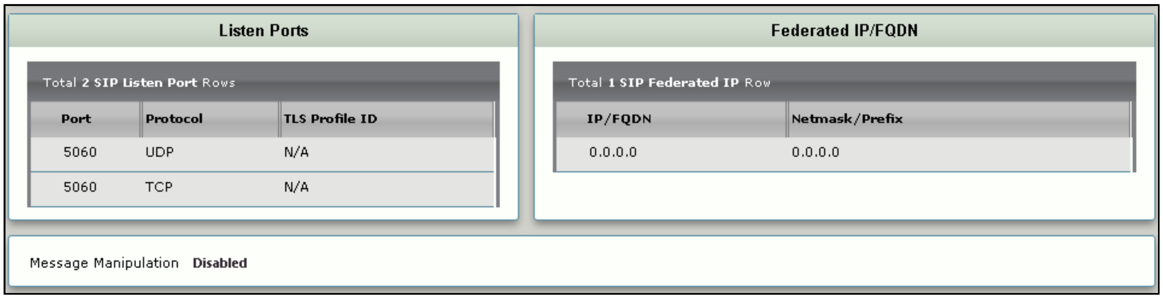


Figure 25: From/To PSTN1

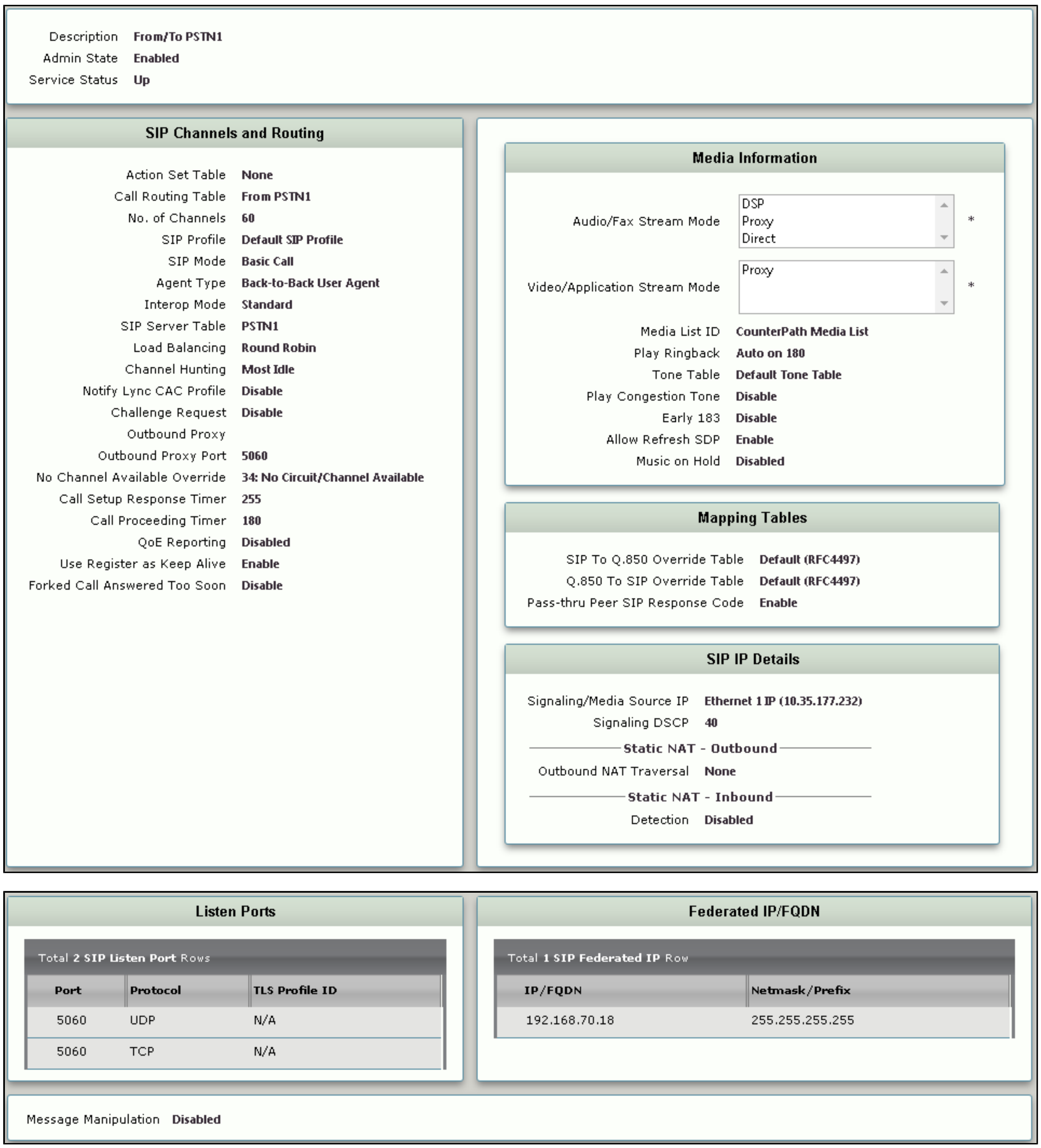
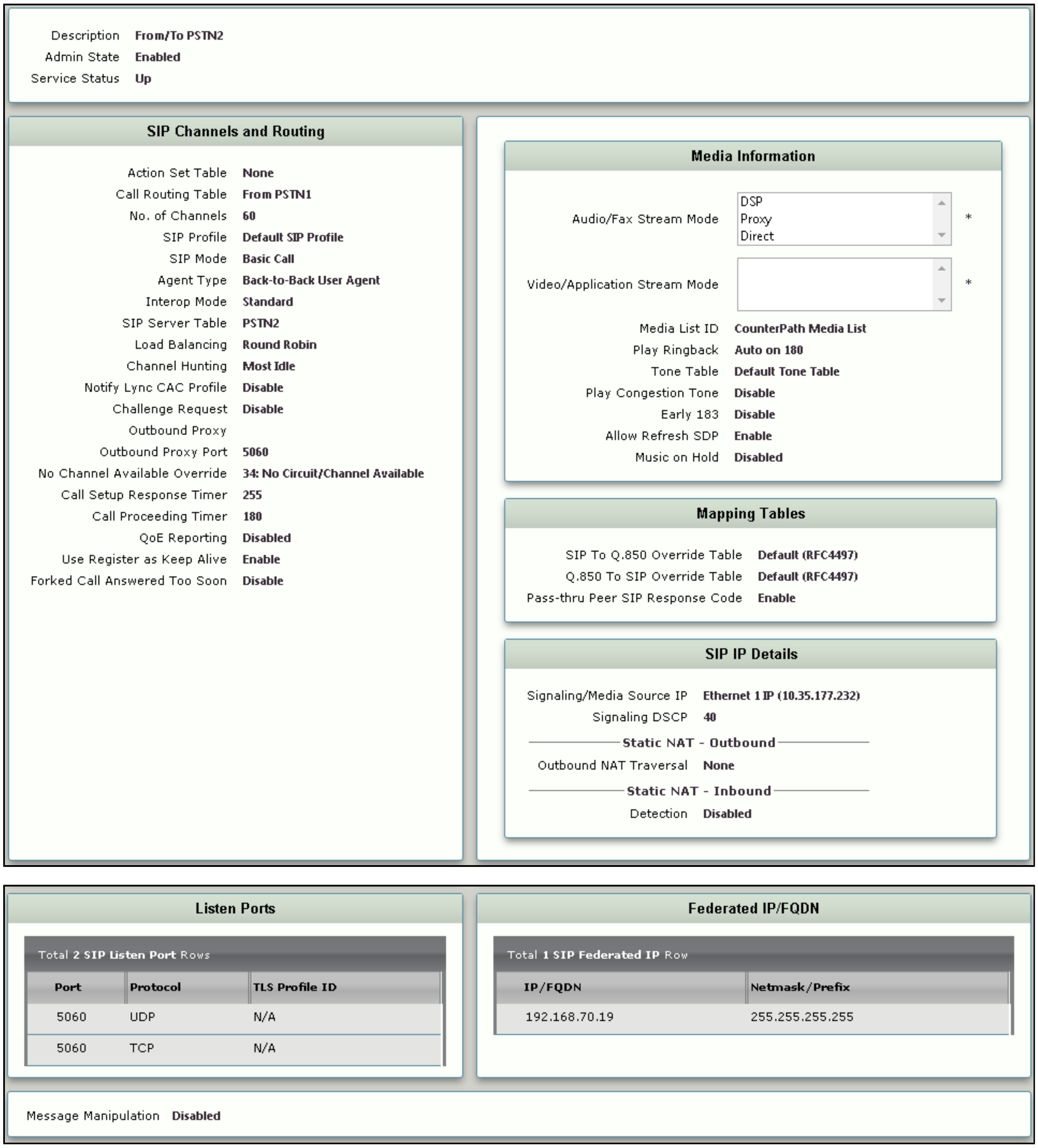


Figure 26: From/To PSTN2



Test Results

Table 2: Test Results

TCID	Test Case Name	Expected Results	Result	Comment
Section 1 - Basic Calls (Wired Ethernet Connection)				
1.01	Bria outbound call to VoIP endpoint	Call established, two-way audio.	Pass	
1.02	Bria Inbound Call from VoIP endpoint	Call established, two-way audio.	Pass	
1.03	Bria outbound call to PSTN number	Call established, two-way audio.	Pass	

1.04	Bria Inbound Call from PSTN number	Call established, two-way audio.	Pass	
1.05	Extended call with VoIP endpoint	Confirm call does not drop and 2-way audio maintained for duration of call	Pass	
1.06	Extended call with PSTN endpoint	Confirm call does not drop and 2-way audio maintained for duration of call	Pass	
1.07	Internal HOLD	Text seen: Call on Hold. Play Button appears(to allow for resuming call) and call is on Hold, no audio between call participants	Pass	
1.08	Internal Retrieve HOLD	Text Seen: Call Established. Play button removed, pause button and other audio options available , voicepath resumed	Pass	
1.09	External HOLD	Text seen: Call on Hold. Play Button appears(to allow for resuming call) and call is on Hold, no audio between call participants	Pass	
1.10	External Retrieve HOLD	Text Seen: Call Established. Play button removed, pause button and other audio options available , voicepath resumed	Pass	
Section 2 - Advanced Calls, (Wired Ethernet Connection)				
2.01	Bria Hold internal then call internal	First call goes on hold, second call established. Internal B is on hold(with play button show) Two-way audio with C.	Pass	
2.02	Bria Swap Internal	Call is swapped to first call (call with end B) with two way speech path and second call (with end C) goes on hold	Pass	
2.03	Bria Hold external then call external	First call goes on hold, second call established. Internal B is on hold(with play button show) Two-way audio with C.	Pass	
2.04	Bria Swap External	Call is swapped to first call (call with end B) with two way speech path and second call (with end C) goes on hold	Pass	
2.05	Bria Hold internal then call external	First call goes on hold, second call established. Internal B is on hold(with play button show) Two-way audio with C.	Pass	
2.06	Bria Hold external then call internal	First call goes on hold, second call established. Internal B is on hold(with play button show) Two-way audio with C.	Pass	
2.07	Bria Swap Mixed	Call is swapped to first call (call with end B) with two way speech path and second call (with end C) goes on hold(with play button show)	Pass	
2.08	Bria Swap End first call	Call with end B is terminated properly and call with C is retrieved from Hold with two way speech path.	Pass	
2.09	Bria Second call attempt cancelled	First call (call with End B) resumes when we end the second call attempt	Pass	
2.10	Bria Blind Transfer	B gets transferred to C. Two way speech path established between end B and end C.	Pass	
2.11	Bria Supervised Transfer	Bria Supervised Transfer	Pass	
2.12	Bria Conference	Conference call established successfully between Bria VOIP, end B and end C.	Pass	
2.13	Bria Split Conference	Conference split successfully resulting in independent calls with B and C, allowing swap.	Pass	

2.14	Bria Conference Hold	conference call goes on hold successfully, no audio between participants	Pass	
2.15	Bria Conference Retrieve Hold	conference call comes off hold successfully, audio restored between participants	Pass	
2.16	Receive incoming call (internal) while in active call	Confirm can answer and swap between active calls as expected	Pass	
2.17	Receive incoming call (external) while in active call	Confirm can answer and swap between active calls as expected	Pass	
Section 3 - Basic Calls, Wireless Connectivity (Wireless Access Point)				
3.01	Bria outbound call to VoIP endpoint	Call established, two-way audio.	Pass	
3.02	Bria Inbound Call from VoIP endpoint	Call established, two-way audio.	Pass	
3.03	Bria outbound call to PSTN number	Call established, two-way audio.	Pass	
3.04	Bria Inbound Call from PSTN number	Call established, two-way audio.	Pass	
3.05	Extended call with VoIP endpoint	Confirm call does not drop and 2-way audio maintained for duration of call	Pass	
3.06	Extended call with PSTN endpoint	Confirm call does not drop and 2-way audio maintained for duration of call	Pass	
3.07	Internal HOLD	Text seen: Call on Hold. Play Button appears(to allow for resuming call) and call is on Hold, no audio between call participants	Pass	
3.08	Internal Retreive HOLD	Text Seen: Call Established. Play button removed, pause button and other audio options available , voicepath resumed	Pass	
3.09	External HOLD	Text seen: Call on Hold. Play Button appears(to allow for resuming call) and call is on Hold, no audio between call participants	Pass	
3.10	External Retrieve HOLD	Text Seen: Call Established. Play button removed, pause button and other audio options available , voicepath resumed	Pass	
Section 4 - Advanced Calls, Wireless Connectivity(Wireless Access Point)				
4.01	Bria Hold internal then call internal	First call goes on hold, second call established. Internal B is on hold(with play button show) Two-way audio with C.	Pass	
4.02	Bria Swap Internal	Call is swapped to first call (call with end B) with two way speech path and second call (with end C) goes on hold	Pass	
4.03	Bria Hold external then call external	First call goes on hold, second call established. Internal B is on hold(with play button show) Two-way audio with C.	Pass	
4.04	Bria Swap External	Call is swapped to first call (call with end B) with two way speech path and second call (with end C) goes on hold	Pass	
4.05	Bria Hold internal then call external	First call goes on hold, second call established. Internal B is on hold(with play button show) Two-way audio with C.	Pass	

4.06	Bria Hold external then call internal	First call goes on hold, second call established. Internal B is on hold(with play button show) Two-way audio with C.	Pass	
4.07	Bria Swap Mixed	Call is swapped to first call (call with end B) with two way speech path and second call (with end C) goes on hold(with play button show)	Pass	
4.08	Bria Swap End first call	Call with end B is terminated properly and call with C is retrieved from Hold with two way speech path.	Pass	
4.09	Bria Second call attempt cancelled	First call (call with End B) resumes when we end the second call attempt	Pass	
4.10	Bria Blind Transfer	B gets transferred to C. Two way speech path established between end B and end C.	Pass	
4.11	Bria Supervised Transfer	A establishes call with B. B gets placed on hold while call with C is established. Call is transferred successfully and two way speech path established successfully between end B and end C.	Pass	
4.12	Bria Conference	Conference call established successfully between Bria VOIP, end B and end C.	Pass	
4.13	Bria Split Conference	Conference split successfully resulting in independent calls with B and C, allowing swap.	Pass	
4.14	Bria Conference Hold	conference call goes on hold successfully, no audio between participants	Pass	
4.15	Bria Conference Retrieve Hold	conference call comes off hold successfully, audio restored between participants	Pass	
4.16	Receive incoming call (internal) while in active call	Confirm can answer and swap between active calls as expected	Pass	
4.17	Receive incoming call (external) while in active call	Confirm can answer and swap between active calls as expected	Pass	
Section 5 - Access Point Changes, Background Support				
5.01	Switch from Wired only to Wi-Fi	Bria should maintain registration and ability to make and receive calls with 2-way audio.	Pass	
5.02	Switch from Wired only to Wi-Fi only	Bria should stay registered and be able to make and receive calls with 2-way audio.	Pass	
5.03	Switch from Wifi to Wired	Bria should maintain registration and ability to make and receive calls with 2-way audio.	Pass	
5.04	Switch from Wireless only to Wired only	Bria should re-register and be able to make and receive calls with 2-way audio.	Pass	
5.05	Background support, incoming call short	Bria should maintain registration and ability to receive calls with 2-way audio.	Pass	
5.06	Background support, incoming call long	Background support, incoming call long	Pass	

5.07	Background support, outgoing call	Bria should maintain registration and ability to make and receive calls with 2-way audio.	Pass	
Section 6 - Call Log, MWI (Advanced Voice Features)				
6.01	Bria Call Log: display correct logs for outgoing calls	Outgoing call logs displayed properly with correct name, number and icon	Pass	
6.02	Bria Call Log: display correct logs for incoming calls	Incoming call logs displayed properly with correct name, number and icon	Pass	
6.03	Bria MWI On	Verify new Message Waiting Indicator is displayed on Bria (VM icon highlighted and number of VM indicated)	N/A	
6.04	Bria Retrieve Voicemail	DTMF is sent/received allowing retrieval of voicemail	N/A	
6.05	Bria Voice mail MWI Persist on register/unregister	MWI should persist between unregister/register or Exit and re-launch of Bria VOIP Client.	N/A	
6.06	Bria New MWI Increment	MWI counter on dash board should increase by 1 after leaving new message to Bria voice mail.	N/A	
6.07	Bria MWI Decrement	MWI counter on dash board should decrease by 1 after message is deleted	N/A	
6.08	Bria MWI Off	Verify MWI turned off after all VM are retrieved by Bria	N/A	
Section 7 - Video Calling(Advanced Feature) (video codec enabled via SIP)				
7.01	Bria outbound video call to VoIP endpoint	Call established, two-way audio, two-way video	Pass	
7.02	Bria Inbound Video Call from VoIP endpoint	Call established, two-way audio, two-way video	Pass	
7.03	Bria outbound escalation from audio call to video call	Call established, two-way audio, two-way video	Pass	
7.04	Bria Inbound escalation from audio call to video call	Call established, two-way audio, two-way video	Pass	
7.05	Extended video call with VoIP endpoint	Confirm video call does not drop and 2-way audio and 2-way video maintained for duration of call	Pass	
7.06	Internal HOLD	Play/Resume button presented and call is on Hold, no audio or video between call participants	Pass	
7.07	Internal Retrieve HOLD	hold button presented back, voice and video paths resumed	Pass	
Section 8 - Messaging & Presence (Advanced Feature) (requires XMPP account)				
8.01	Change Presence Status	Confirm status change updates other clients	N/A	

8.02	Receive Presence Status	Confirm other clients status is updated on Bria	N/A	
8.03	Send IM	Message created, sent and received	N/A	
8.04	Messaging Activity	Confirm other party's messaging response activity is detected in Bria	N/A	
8.05	Receive IM	Message received.	N/A	
8.06	Group Chat	Group chat created, message sent and message received	N/A	

Conclusion

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