Configuration Note

AudioCodes Professional Services - Interoperability Lab

Connecting ShoreTel IP-PBX to BroadCloud SIP Trunk using AudioCodes Mediant™ E-SBC

Version 7.0

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Notice

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Date Published: February-3-2016

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Document Revision Record

LTRT	Description
12560	Initial document release for Version 7.0.

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1 Introduction

This Configuration Note describes how to set up AudioCodes Enterprise Session Border Controller (hereafter, referred to as *E-SBC*) for interworking between BroadCloud's SIP Trunk and IP-PBX environment.

1.1 Intended Audience

The document is intended for engineers, or AudioCodes and BroadCloud Partners who are responsible for installing and configuring BroadCloud's SIP Trunk and IP-PBX for enabling VoIP calls using AudioCodes E-SBC.

1.2 About AudioCodes E-SBC Product Series

AudioCodes' family of E-SBC devices enables reliable connectivity and security between the Enterprise's and the service provider's VoIP networks.

The E-SBC provides perimeter defense as a way of protecting Enterprises from malicious VoIP attacks; mediation for allowing the connection of any PBX and/or IP-PBX to any service provider; and Service Assurance for service quality and manageability.

Designed as a cost-effective appliance, the E-SBC is based on field-proven VoIP and network services with a native host processor, allowing the creation of purpose-built multiservice appliances, providing smooth connectivity to cloud services, with integrated quality of service, SLA monitoring, security and manageability. The native implementation of SBC provides a host of additional capabilities that are not possible with standalone SBC appliances such as VoIP mediation, PSTN access survivability, and third-party value-added services applications. This enables Enterprises to utilize the advantages of converged networks and eliminate the need for standalone appliances.

AudioCodes E-SBC is available as an integrated solution running on top of its field-proven Mediant Media Gateway and Multi-Service Business Router platforms, or as a software-only solution for deployment with third-party hardware.

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2 Component Information

2.1 IP-PBX Version

Table 2-1: IP-PBX Version

Vendor	ShoreTel
Model	ShoreGear
Software Version	14.2_Build_19.45.8701.0
Protocol	SIP/UDP
Additional Notes	None

2.2 AudioCodes E-SBC Version

Table 2-2: AudioCodes E-SBC Version

SBC Vendor	AudioCodes
Models	▪ Mediant 500 E-SBC ▪ Mediant 800 Gateway & E-SBC ▪ Mediant 1000B Gateway & E-SBC ▪ Mediant 3000 Gateway & E-SBC ▪ Mediant 2600 E-SBC ▪ Mediant 4000 E-SBC
Software Version	SIP_F7.00A.049.003
Protocol	SIP/UDP (to the both BroadCloud SIP Trunk and IP-PBX)
Additional Notes	None

2.3 BroadCloud SIP Trunking Version

Table 2-3: BroadCloud Version

Vendor/Service Provider	BroadCloud
SSW Model/Service	BroadWorks
Software Version	21
Protocol	SIP/UDP
Additional Notes	None

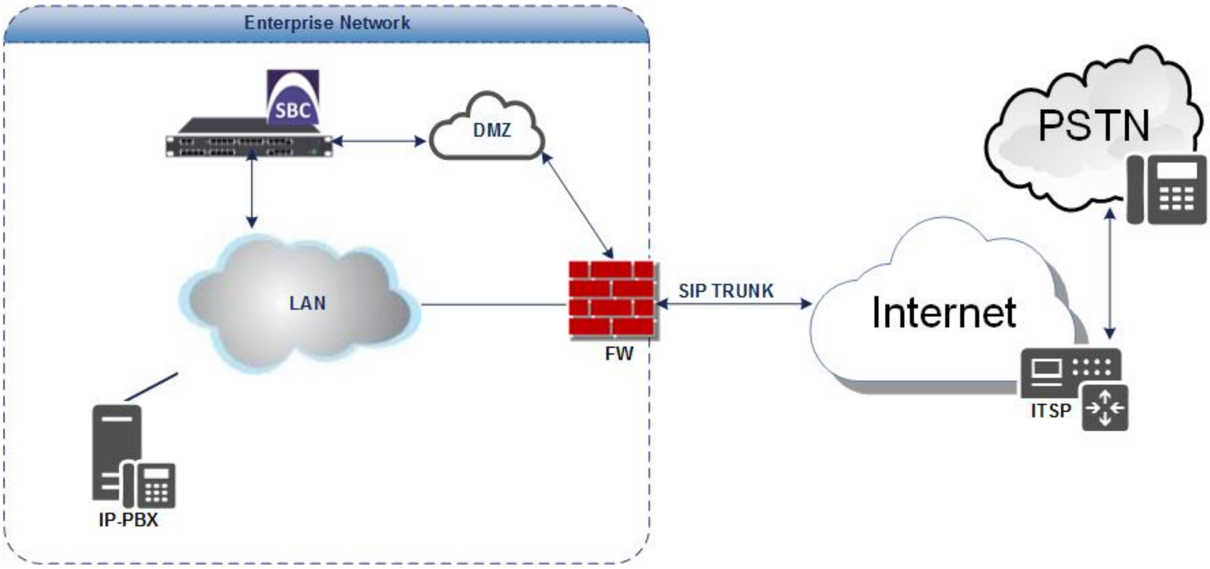
2.4 Interoperability Test Topology

The interoperability testing between AudioCodes E-SBC and BroadCloud SIP Trunk with IP-PBX was done using the following topology setup:

- Enterprise deployed with IP-PBX in its private network for enhanced communication within the Enterprise.
- Enterprise wishes to offer its employees enterprise-voice capabilities and to connect the Enterprise to the PSTN network using BroadCloud's SIP Trunking service.
- AudioCodes E-SBC is implemented to interconnect between the Enterprise LAN and the SIP Trunk.
  - **Session:** Real-time voice session using the IP-based Session Initiation Protocol (SIP).
  - **Border:** IP-to-IP network border between IP-PBX network in the Enterprise LAN and BroadCloud's SIP Trunk located in the public network.

The figure below illustrates this interoperability test topology:

Figure 2-1: Interoperability Test Topology between E-SBC and IP-PBX with BroadCloud SIP Trunk



2.4.1 Environment Setup

The interoperability test topology includes the following environment setup:

Table 2-4: Environment Setup

Area	Setup
Network	▪ IP-PBX is located on the Enterprise's LAN ▪ BroadCloud SIP Trunk is located on the WAN
Signaling Transcoding	▪ IP-PBX operates with SIP-over-UDP transport type ▪ BroadCloud SIP Trunk operates with SIP-over-UDP transport type
Codecs Transcoding	▪ IP-PBX supports G.711A-law, G.711U-law, and G.729 coder ▪ BroadCloud SIP Trunk supports G.711A-law, G.711U-law, and G.729 coder
Media Transcoding	▪ IP-PBX operates with RTP media type ▪ BroadCloud SIP Trunk operates with RTP media type

2.4.2 Known Limitations

There were no limitations observed in the interoperability tests done for the AudioCodes E-SBC interworking between IP-PBX and BroadCloud 's SIP Trunk.

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3 Configuring ShoreTel IP-PBX

This chapter describes how to configure basic parameters of the ShoreTel ShoreGear IP-PBX to operate with AudioCodes E-SBC.

Note: For more complicated configuration parameters please refer to User Manual of each IP-PBX.

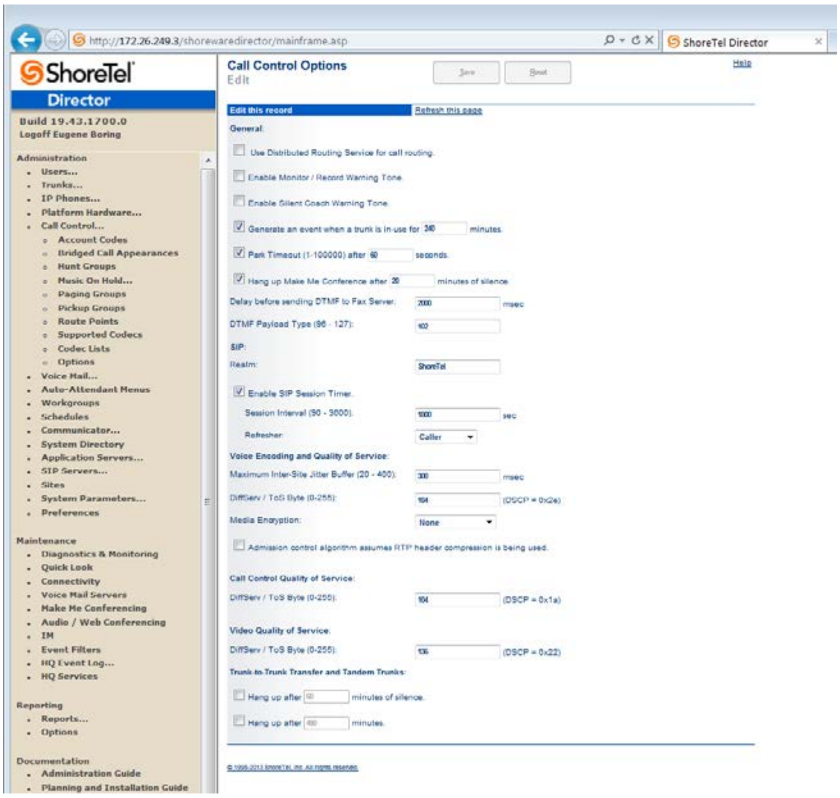
3.1 ShoreTel System Settings – General

The first settings to address within the ShoreTel system are the general system settings. These configurations include the Call Control, the Site and the Switch settings. If these items have already been configured on your system, skip this section and go on to Section 3.5 on page 19 below.

3.2 Call Control Options

The first settings to configure within ShoreTel Director are the Call Control Options. To configure these settings for the ShoreTel system, log into ShoreTel Director and select **Administration** > **Call Control** > **Options**. The Call Control Options screen appears below.

Figure 3-1: Call Control Options Screen



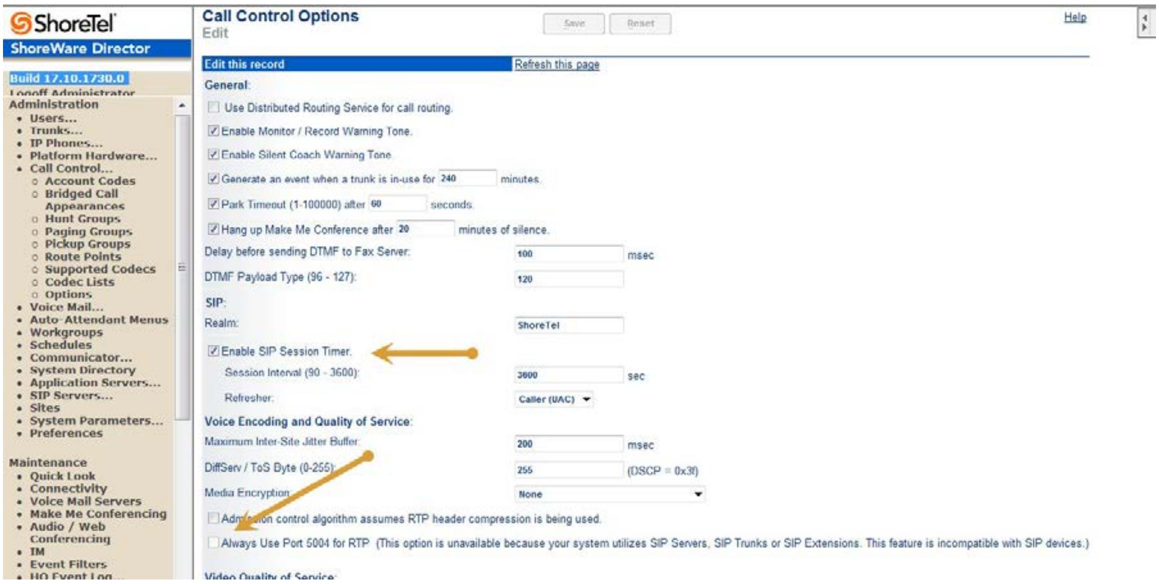
Within the Call Control Options SIP parameters, confirm that the appropriate settings are made for the Realm, Enable SIP Session Timer and Always Use Port 5004 for RTP parameters.

The 'Realm' parameter is used in authenticating all SIP devices. It is typically a description of the computer or system being accessed. Changing this value will require a reboot of all ShoreTel switches serving SIP extensions. It is not necessary to modify this parameter to get the ShoreTel IP PBX system functional with AudioCodes gateway.

➢ **To configure Call Control Options:**

1. Verify that the 'Enable SIP Session Timer' check box is selected.
2. Set the Session Interval Time to the recommended setting of 3600 seconds.
3. Select the appropriate refresher (from the pull down menu) for the SIP Session Timer. The “Refresher” field will be set either to “Caller (UAC)” [User Agent Client] or to “Callee (UAS)” [User Agent Server]. If the “Refresher” field is set to “Caller (UAC)”, the Caller’s device will be in control of the session timer refresh. If “Refresher” is set to “Callee (UAS)”, the device of the person called will control the session timer refresh.
4. Verify the “Voice Encoding and Quality of Service”, specifically the “Media Encryption” parameter. Make sure this parameter is set to “None”; otherwise you may experience one-way audio issues. Please refer to *ShoreTel Administration Guide* for additional details on media encryption and the other parameters in the “Voice Encoding and Quality of Service” area.
5. Disable (uncheck) the “Always Use Port 5004 for RTP” parameter if checked; it is required for implementing SIP trunks between ShoreTel systems only. For SIP configurations, Dynamic User Datagram Protocol (UDP) must be used for RTP Traffic. If the parameter is disabled, Media Gateway Control Protocol (MGCP) no longer uses UDP port 5004; MGCP and SIP traffic will use dynamic UDP ports **(Figure 3)**.

Figure 3-2: Call Control Options Settings



6. Once this parameter is unchecked, make sure that “everything” (IP Phones, ShoreTel Voice Switches, ShoreTel Server, Distributed Voice Mail Servers / Remote Servers, Conference Bridges and Contact Centers) is “fully” rebooted – this is a “one time only” item. By not performing a full system reboot after changing this setting, one-way audio may occur during initial testing.
7. Be sure to save your changes before leaving this screen by clicking Save at the top of the page.

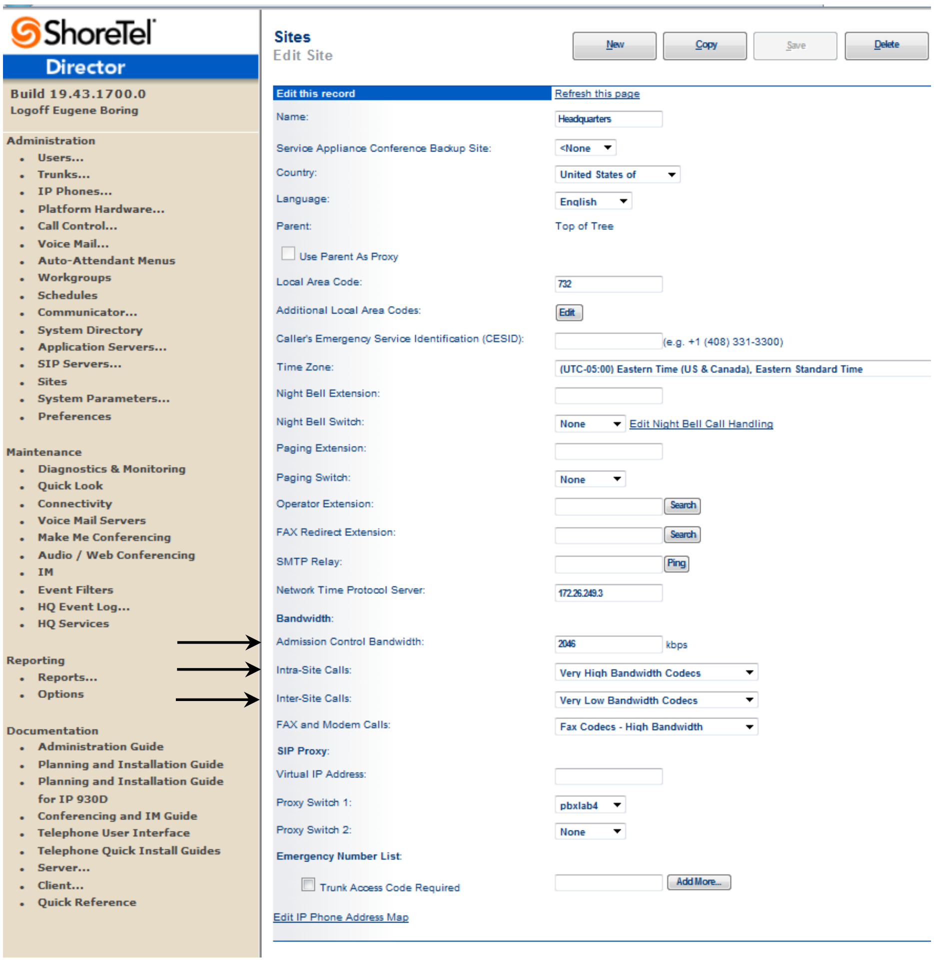
3.3 Sites Settings

The next settings to address are the administration of sites. These settings are modified under the ShoreTel Director by selecting **Administration** > **Sites**. The **Sites** screen appears.

➢ To configure Sites:

1. Within the Sites screen select the name of the site to configure. The Edit Site screen will then appear. The only changes required to the Edit Site screen are to the 'Admission Control Bandwidth', 'Intra-Site Calls' and 'Inter-Site Calls' parameters.

Figure 3-3: Site Bandwidth settings



2. Set the appropriate Admission Control Bandwidth for your network. Please refer to the *ShoreTel Planning and Installation Guide* for additional information on setting Admission Control Bandwidth for your network. Admission Control Bandwidth defines the bandwidth available to and from the site. This is important as SIP trunk calls will be counted against the site bandwidth.

Note: Bandwidth of 2046 kbps is just an example.

3. From the 'Inter-Site Calls' drop-down list, select **Very Low Bandwidth Codecs**. By default, **Very Low Bandwidth Codecs** contains two codecs - G.729 and G.711u - with G.729 being the primary codec of choice. The 'Inter-Site Calls' parameter defines which codecs will be used when establishing a call with AudioCodes – the preferred codec choice is G.729.

Note: Please do not modify the "Very Low Bandwidth Codecs" codec list.

4. Save changes before leaving this screen by clicking **Save** at the top of the page.

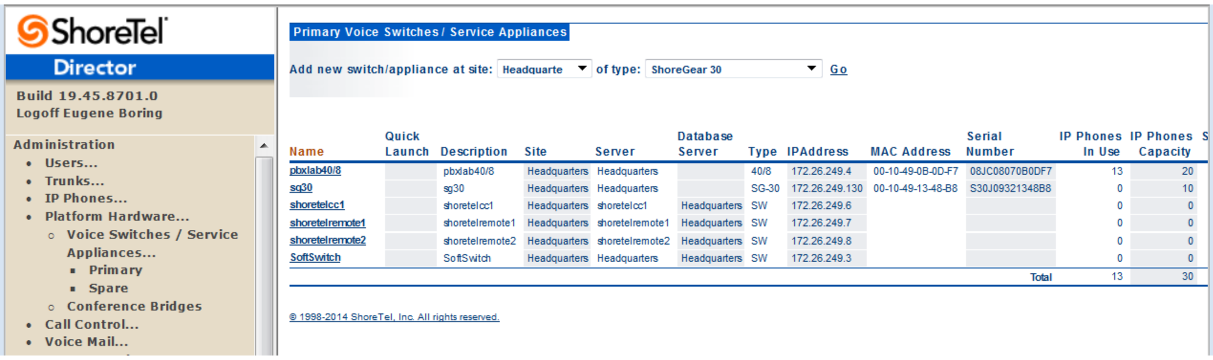
3.4 Switch Settings - Allocating Ports for SIP Trunks

The final general settings to configure are the ShoreTel Switch settings.

➢ **To configure ShoreTel Switch settings:**

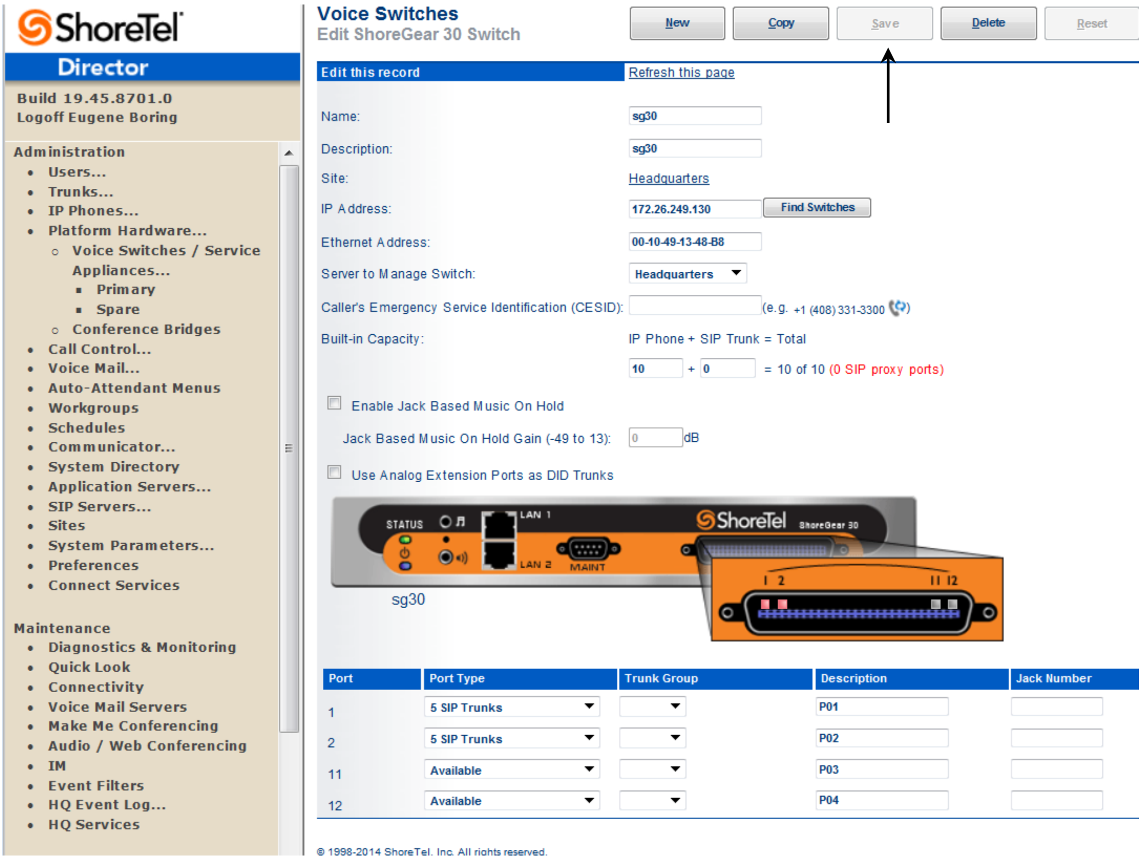
1. Navigate to the Primary Voice Switches/Service Appliances screen by selecting **Administration** > **Switches > Primary** in ShoreTel Director, as shown in the figure below.

Figure 3-4: Administration Switches



2. From the Switches screen, choose the name of the switch to configure for SIP trunks; the Edit ShoreTel Switch screen appears.
3. On the Edit ShoreTel Switch screen, select the desired number of SIP Trunks from the available ports.

Figure 3-5: ShoreTel Switch Settings



Each port designated as a Port Type of a SIP Trunk enables the support for five individual SIP trunks. Each trunk can support one concurrent call between the ShoreTel system and the BroadCloud SIP Trunk.

4. Determine the desired capacity of the interconnection between the two systems and configure the necessary resources as required, and then proceed to the next section.

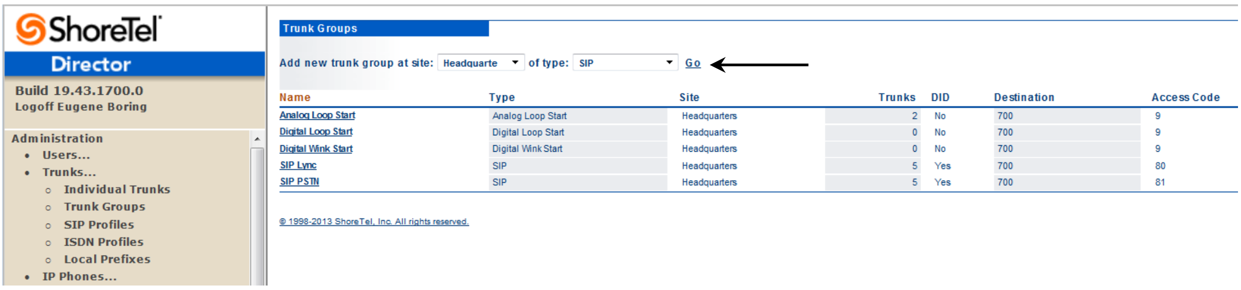
5. Be sure to save your changes before leaving this screen by clicking **Save** at the top of the screen.

3.5 ShoreTel System Settings – Trunk Groups

ShoreTel Trunk Groups only support Static IP Addresses for Individual Trunks. In trunk planning, the following needs to be considered. AudioCodes gateway interfaces should always be configured to use a “static” IP Address.

The settings for Trunk Groups are changed by selecting **Administration** > **Trunks** > **Trunk Groups** within ShoreTel Director, as shown below.

Figure 3-6: Administration Trunk Groups

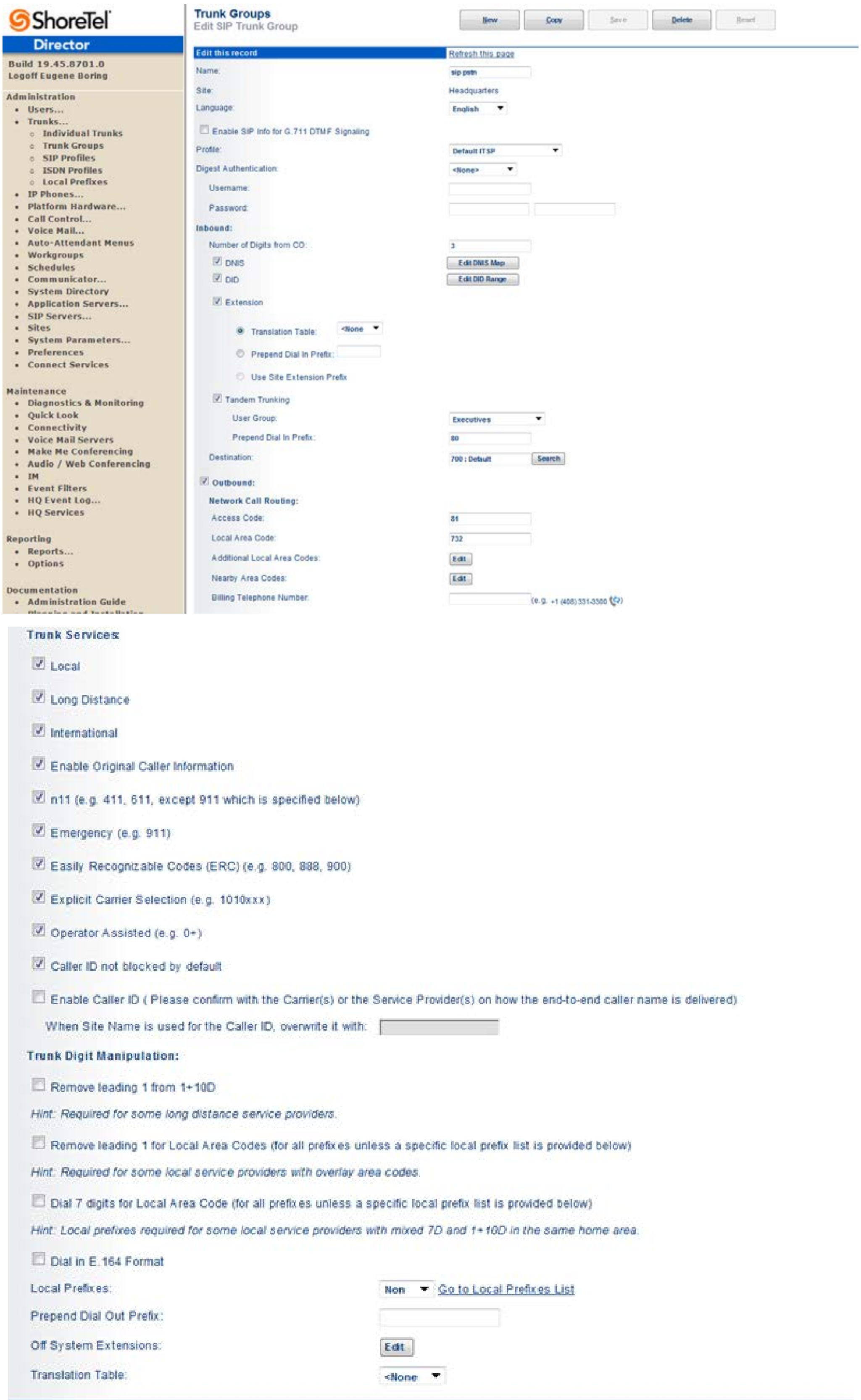


➢ **To configure Trunk Groups:**

1. From the pull down menus on the Trunk Groups screen, select the site desired and select the **SIP** trunk type to configure.
2. Click on the **Go** link from **Add new trunk group at site**. The Edit SIP Trunk Group screen appears.

3.6 SIP PSTN Trunk Group for BroadCloud

Figure 3-7: BroadCloud SIP Trunk Group (SIP PSTN)



3.7 ShoreTel System Settings – Individual Trunks

This section describes the configuration of individual trunks.

➢ **To configure individual trunks:**

1. Navigate to the Trunks Group screen by selecting **Administration** > **Trunks** > **Individual Trunks**.
2. The Trunks by Group screen is used to change the individual trunks settings that appear.

Figure 3-8: Trunks by Group

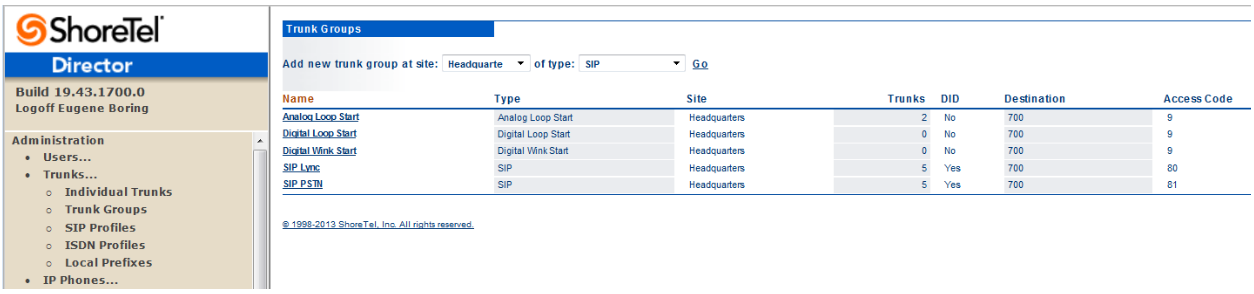


Figure 3-9: Individual Trunk Setting for BroadCloud SIP Trunk Group

3.8 Edit BroadCloud SIP Trunk Group

➢ **To edit BroadCloud SIP Trunk Group:**

1. Enter your preferred name for the new trunk group. In the example in Figure 3-7, the **SIP PSTN** has been created.
2. The 'Enable SIP Info for G.711 DTMF Signaling' parameter should not be selected. 'Enabling SIP info' is currently only used with SIP tie trunks between ShoreTel systems.
3. The 'Profile' parameter should be left at its default setting of **Default ITSP**; it is not necessary to modify this parameter when connecting to the AudioCodes SBC.
4. The 'Digest Authentication' parameter defaults to "<None>" and modification is not required when connecting to the AudioCodes SBC.
5. The next item to change in the Edit SIP Trunks Group screen is to make the appropriate settings for the 'Inbound' parameters in the figure below.

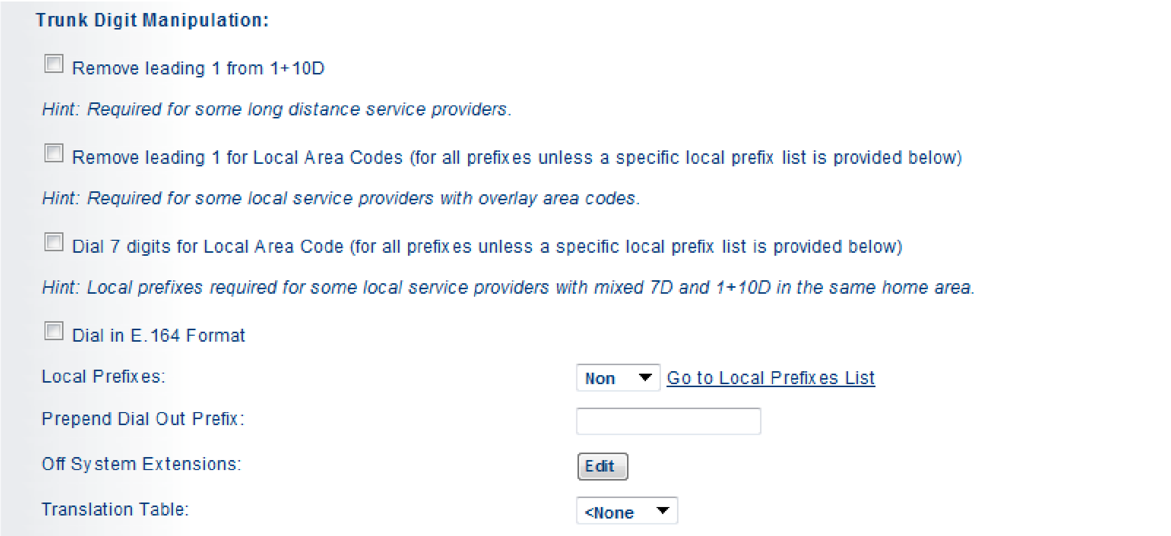
Figure 3-10: Inbound

6. Within the 'Inbound:' settings, ensure the **Number of Digits from CO** is set to match what the ShoreTel SIP trunk switch will be receiving from AudioCodes SBC and ensure that the 'DNIS', 'DID' and 'Extension' check boxes are selected.
7. It is recommended that the 'Tandem Trunking' check box should be selected. Otherwise transfers to external telephone numbers will fail via SIP trunks. For additional information on this parameter please refer to the *ShoreTel Planning and Installation Guide*.
8. Make the appropriate changes for the 'Outbound' parameters below.

Figure 3-11: Outbound and Trunk Services

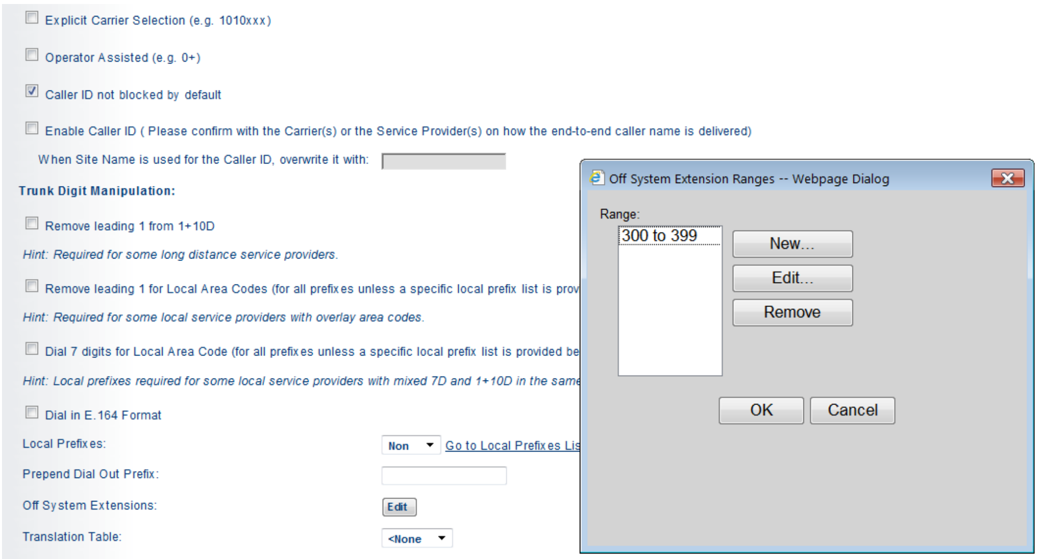
9. Select the 'Outbound' parameter and define a Trunk 'Access Code' and 'Local Area Code' as appropriate.
10. Under the **Trunk Services** group, make sure the appropriate services are enabled or disabled based on your needs. In general, we are only using this trunk group to dial the off system extensions to reach the BroadCloud audio conferencing bridge or softphone users.
11. The Caller ID not blocked by default' field determines if the call is sent out as <unknown> or with caller information (Caller ID). User DID will impact how information is passed out to the SIP Trunk group.
12. The final parameters for configuration in the Trunk Group are 'Trunk Digit Manipulation' below.

Figure 3-12: Trunk Digit Manipulation



13. Select the 'Dial in E.164 Format' parameter 'IF NEEDED' and define a Trunk 'Access Code' and 'Local Area Code' as appropriate.
14. Next you must create the Off System Extension (OSE) range that will be used to represent the BroadCloud audio conferencing bridge or BroadCloud softphone users. An OSE is required for every BroadCloud SIP Trunk endpoint that will be using the ShoreTel system.
15. Click the Edit button next to Off System Extensions; the Off Systems Extension Range dialog is displayed below.

Figure 3-13: Off System Extension Ranges



16. Click New and define the first range for the extensions that will represent the BroadCloud endpoints on the ShoreTel system.
17. Click OK to save the first range and repeat if necessary to create sufficient extensions for all your BroadCloud endpoints.
18. After all your setting changes are made to the Edit SIP Trunk Group screen, click **Save** at the top of the screen.

4 Configuring AudioCodes E-SBC

This chapter provides step-by-step procedures on how to configure AudioCodes E-SBC for interworking between IP-PBX and the BroadCloud SIP Trunk. These configuration procedures are based on the interoperability test topology described in Section 2.4 on page 10, and includes the following main areas:

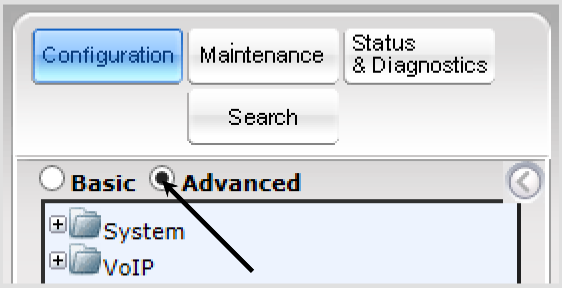
- E-SBC WAN interface - BroadCloud SIP Trunking environment
- E-SBC LAN interface - IP-PBX environment

This configuration is done using the E-SBC's embedded Web server (hereafter, referred to as *Web interface*).

Notes:

- For implementing IP-PBX and BroadCloud SIP Trunk based on the configuration described in this section, AudioCodes E-SBC must be installed with a Software License Key that includes the following software features:
  - √ **SBC**
  - √ **Security**
  - √ **DSP**
  - √ **RTP**
  - √ **SIP**

  For more information about the Software License Key, contact your AudioCodes sales representative.
- The scope of this interoperability test and document does **not** cover all security aspects for connecting the SIP Trunk to the IP-PBX environment. Comprehensive security measures should be implemented per your organization's security policies. For security recommendations on AudioCodes' products, refer to the *Recommended Security Guidelines* document.
- Before you begin configuring the E-SBC, ensure that the E-SBC's Web interface Navigation tree is in Advanced-menu display mode. To do this, select the **Advanced** option, as shown below:



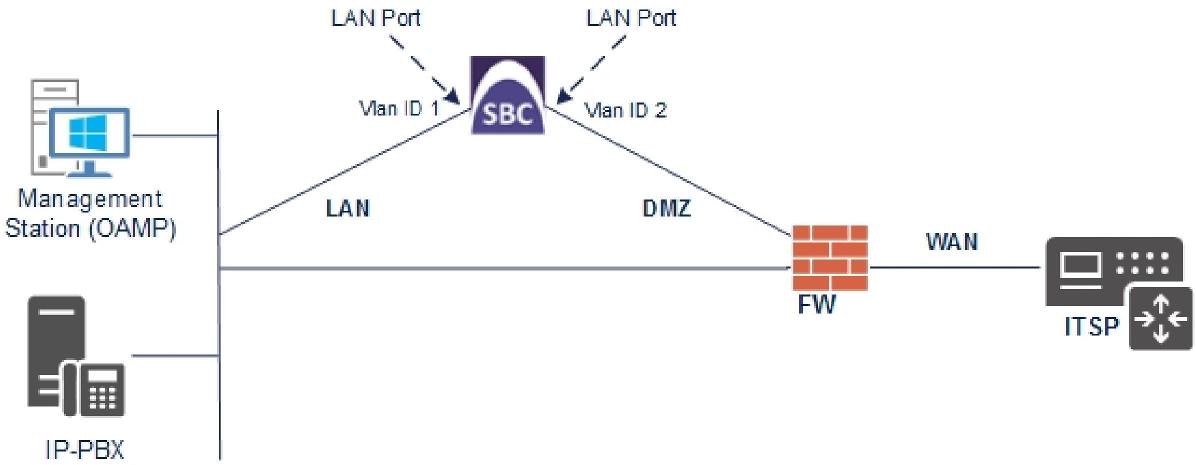
  Note that when the E-SBC is reset, the Navigation tree reverts to Basic-menu display.

4.1 Step 1: IP Network Interfaces Configuration

This step describes how to configure the E-SBC's IP network interfaces. There are several ways to deploy the E-SBC; however, this interoperability test topology employs the following deployment method:

- E-SBC interfaces with the following IP entities:
  - IP-PBX, located on the LAN
  - BroadCloud SIP Trunk, located on the WAN
- E-SBC connects to the WAN through a DMZ network
- Physical connection: The type of physical connection to the LAN depends on the method used to connect to the Enterprise's network. In the interoperability test topology, E-SBC connects to the LAN and WAN using dedicated LAN ports (i.e., two ports and two network cables are used).
- E-SBC also uses two logical network interfaces:
  - LAN (VLAN ID 1)
  - WAN (VLAN ID 2)

Figure 4-1: Network Interfaces in Interoperability Test Topology



4.1.1 Step 1a: Configure VLANs

This step describes how to define VLANs for each of the following interfaces:

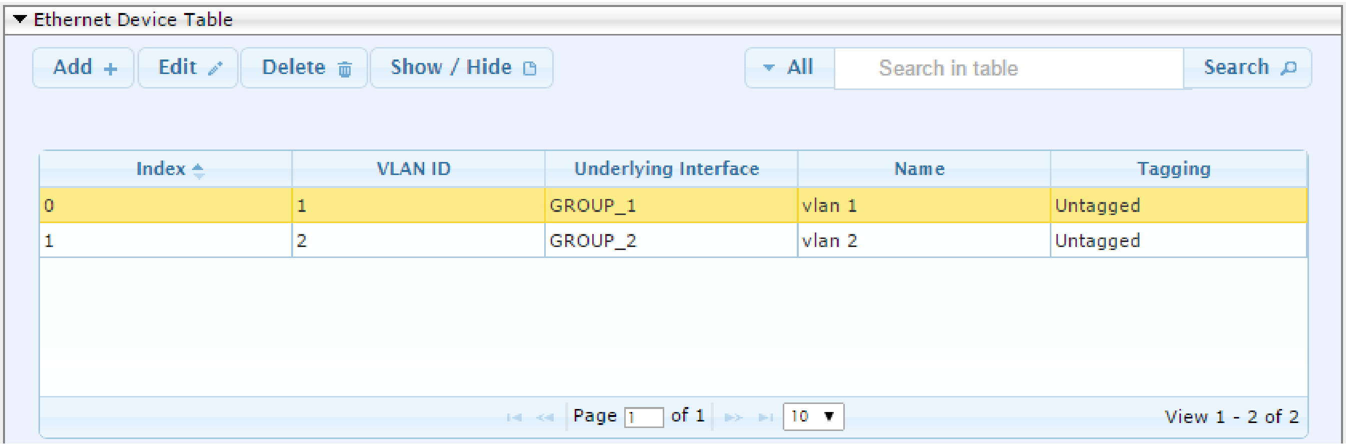
- LAN VoIP (assigned the name "Voice")
- WAN VoIP (assigned the name "WANSP")

➢ **To configure the VLANs:**

1. Open the Ethernet Device Table page (**Configuration** tab > **VoIP** menu > **Network** > **Ethernet Device Table**).
2. There will be one existing row for VLAN ID 1 and underlying interface GROUP_1.
3. Add another VLAN ID 2 for the WAN side as follows:

Parameter	Value
Index	**1**
VLAN ID	**2**
Underlying Interface	**GROUP_2** (Ethernet port group)
Name	**vlan 2**
Tagging	**Untagged**

Figure 4-2: Configured VLAN IDs in Ethernet Device Table



Index	VLAN ID	Underlying Interface	Name	Tagging
0	1	GROUP_1	vlan 1	Untagged
1	2	GROUP_2	vlan 2	Untagged

4.1.2 Step 1b: Configure Network Interfaces

This step describes how to configure the IP network interfaces for each of the following interfaces:

- LAN VoIP (assigned the name "Voice")
- WAN VoIP (assigned the name "WANSP")

➢ **To configure the IP network interfaces:**

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **Network** > **IP Interfaces Table**).

2. Modify the existing LAN network interface:
   a. Select the 'Index' radio button of the **OAMP + Media + Control** table row, and then click **Edit**.
   b. Configure the interface as follows:

Parameter	Value
IP Address	**172.26.100.169** (IP address of E-SBC)
Prefix Length	**24** (subnet mask in bits for 255.255.255.0)
Default Gateway	**172.26.100.001**
VLAN ID	**1**
Interface Name	**Voice** (arbitrary descriptive name)
Underlying Device	**vlan 1**

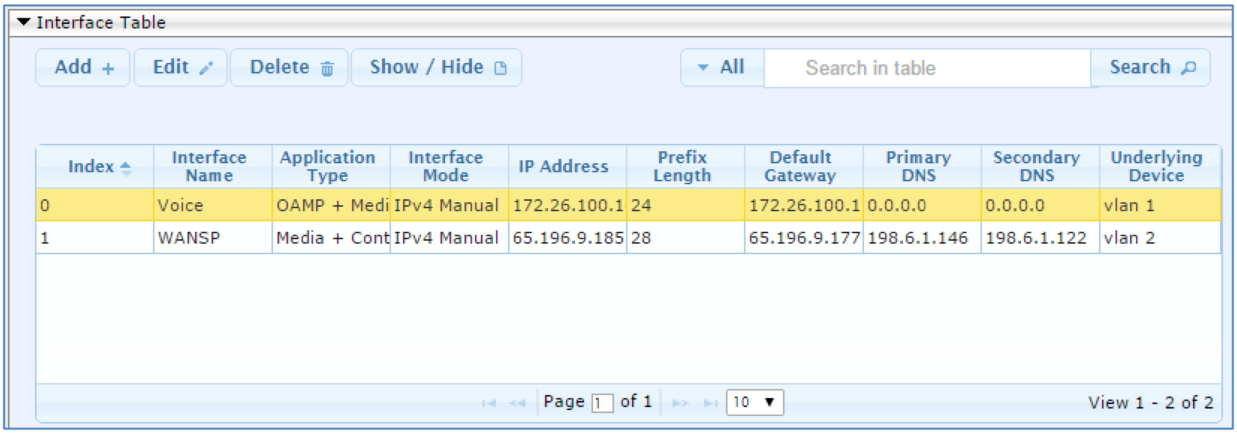
3. Add a network interface for the WAN side:
   a. Enter **1**, and then click **Add Index**.
   b. Configure the interface as follows:

Parameter	Value
Application Type	**Media + Control**
IP Address	**65.196.9.185** (WAN IP address)
Prefix Length	**28** (for 255.255.255.240)
Default Gateway	**65.196.9.177** (router's IP address)
VLAN ID	**2**
Interface Name	**WANSP**
Primary DNS Server IP Address	**198.6.1.146**
Secondary DNS Server IP Address	**198.6.1.122**
Underlying Device	**vlan 2**

4. Click **Apply**, and then **Done**.

The configured IP network interfaces are shown below:

Figure 4-3: Configured Network Interfaces in IP Interfaces Table



Index	Interface Name	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Primary DNS	Secondary DNS	Underlying Device
0	Voice	OAMP + Medi	IPv4 Manual	172.26.100.1	24	172.26.100.1	0.0.0.0	0.0.0.0	vlan 1
1	WANSP	Media + Cont	IPv4 Manual	65.196.9.185	28	65.196.9.177	198.6.1.146	198.6.1.122	vlan 2

4.2 Step 2: Enable the SBC Application

This step describes how to enable the SBC application.

➢ **To enable the SBC application:**

1. Open the Applications Enabling page (**Configuration** tab > **VoIP** menu > **Applications Enabling** > **Applications Enabling**).

Figure 4-4: Enabling SBC Application

2. From the 'SBC Application' drop-down list, select **Enable**.
3. Click **Submit**.
4. Reset the E-SBC with a burn to flash for this setting to take effect (see Section 4.13 on page 63).

4.3 Step 3: Configure Media Realms

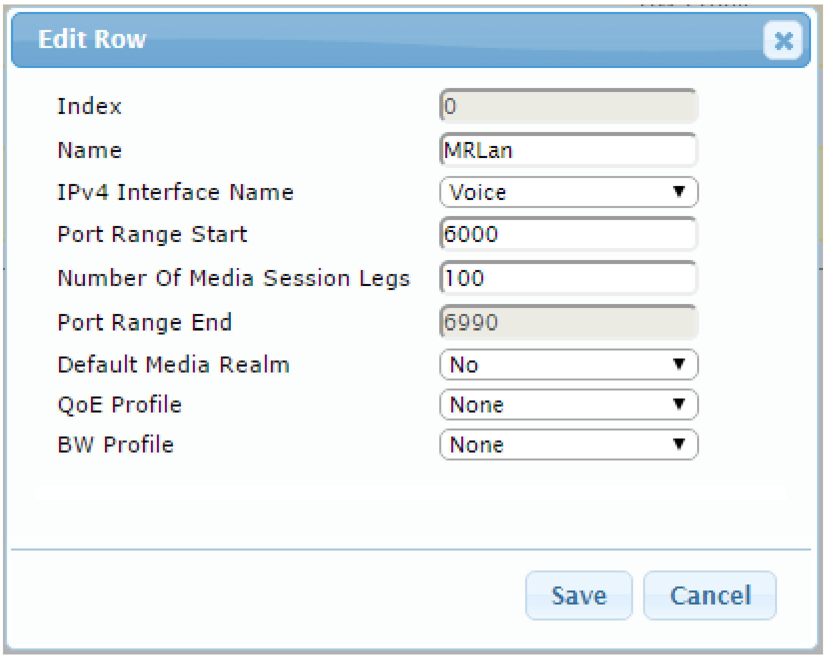
This step describes how to configure Media Realms. The simplest configuration is to create two Media Realms - one for internal (LAN) traffic and one for external (WAN) traffic.

➢ **To configure Media Realms:**

1. Open the Media Realm Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Media Realm Table**).
2. Add a Media Realm for the LAN interface. You can use the default Media Realm (Index 0), but modify it as shown below:

Parameter	Value
Index	**0**
Media Realm Name	**MRLan** (descriptive name)
IPv4 Interface Name	**Voice**
Port Range Start	**6000** (as required by IP-PBX)
Number of Media Session Legs	**100** (media sessions assigned with port range)

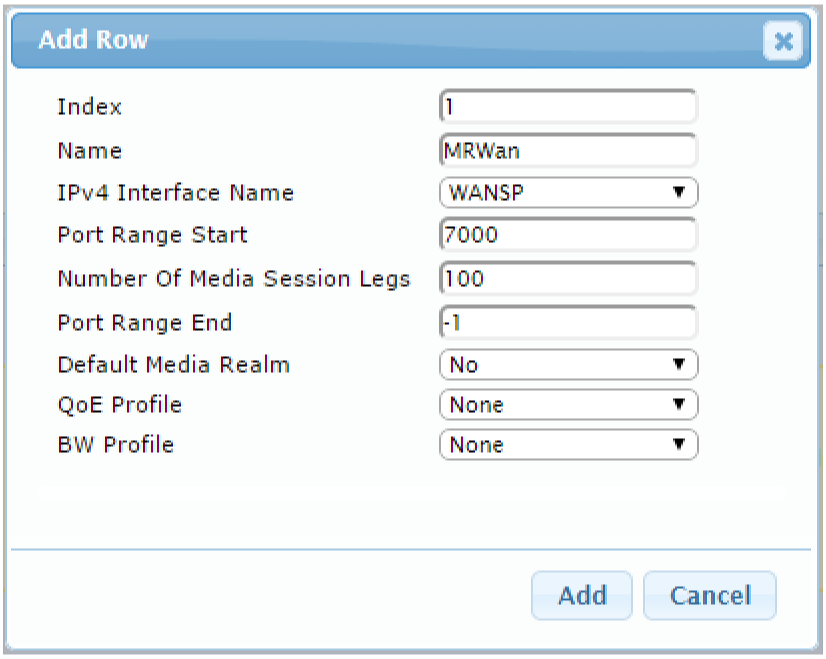
Figure 4-5: Configuring Media Realm for LAN



3. Configure a Media Realm for WAN traffic:

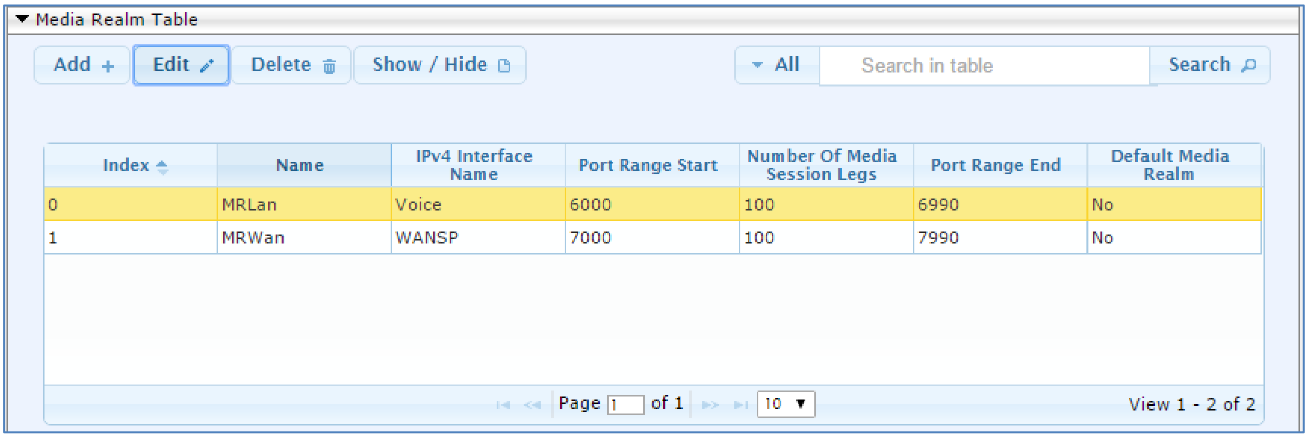
Parameter	Value
Index	**1**
Media Realm Name	**MRWan** (arbitrary name)
IPv4 Interface Name	**WANSP**
Port Range Start	**7000** (represents lowest UDP port number used for media on WAN)
Number of Media Session Legs	**100** (media sessions assigned with port range)

Figure 4-6: Configuring Media Realm for WAN



The configured Media Realms are shown in the figure below:

Figure 4-7: Configured Media Realms in Media Realm Table



4.4 Step 4: Configure SIP Signaling Interfaces

This step describes how to configure SIP Interfaces. For the interoperability test topology, an internal and external SIP Interface must be configured for the E-SBC.

- **To configure SIP Interfaces:**

1. Open the SIP Interface Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **SIP Interface Table**).
2. Add a SIP Interface for the LAN interface. You can use the default SIP Interface (Index 0), but modify it as shown below:

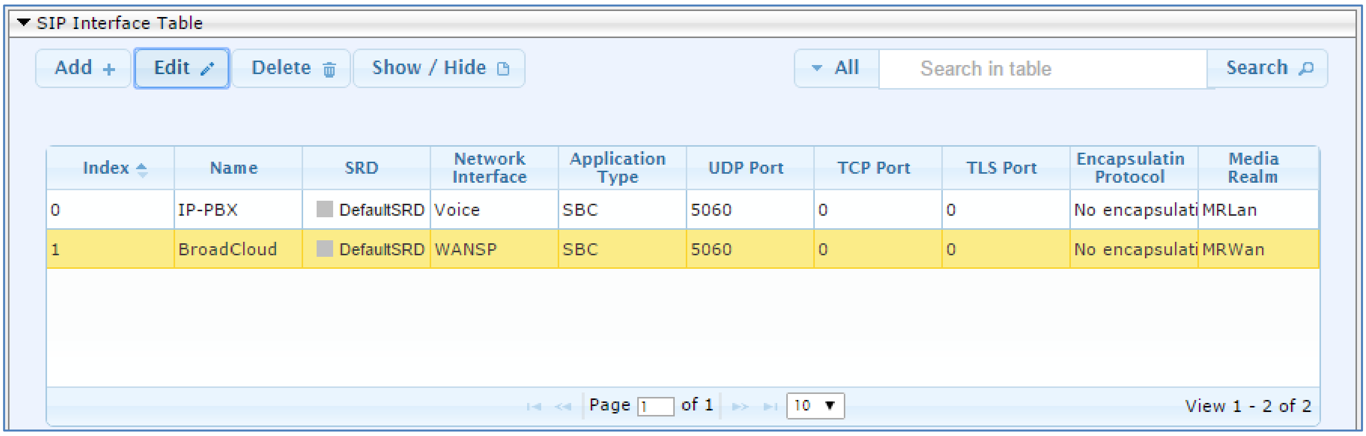
Parameter	Value
Index	**0**
Interface Name	**IP-PBX** (see Note below)
Network Interface	**Voice**
Application Type	**SBC**
UDP Port	**5060**
TCP and TLS	**0**
Media Realm	**MRLan**

3. Configure a SIP Interface for the WAN:

Parameter	Value
Index	**1**
Interface Name	**BroadCloud** (see Note below)
Network Interface	**WANSP**
Application Type	**SBC**
UDP Port	**5060**
TCP and TLS	**0**
Media Realm	**MRWan**

The configured SIP Interfaces are shown in the figure below:

Figure 4-8: Configured SIP Interfaces in SIP Interface Table



Note: Unlike in previous software releases where configuration entities (e.g., SIP Interface, Proxy Sets, and IP Groups) were associated with each other using table row indices, Version 7.0 uses the string **names** of the configuration entities. Therefore, it is recommended to configure each configuration entity with meaningful names for easy identification.

4.5 Step 5: Configure Proxy Sets

This step describes how to configure Proxy Sets. The Proxy Set defines the destination address (IP address or FQDN) of the IP entity server. Proxy Sets can also be used to configure load balancing between multiple servers.

For the interoperability test topology, two Proxy Sets need to be configured for the following IP entities:

- IP-PBX
- BroadCloud SIP Trunk

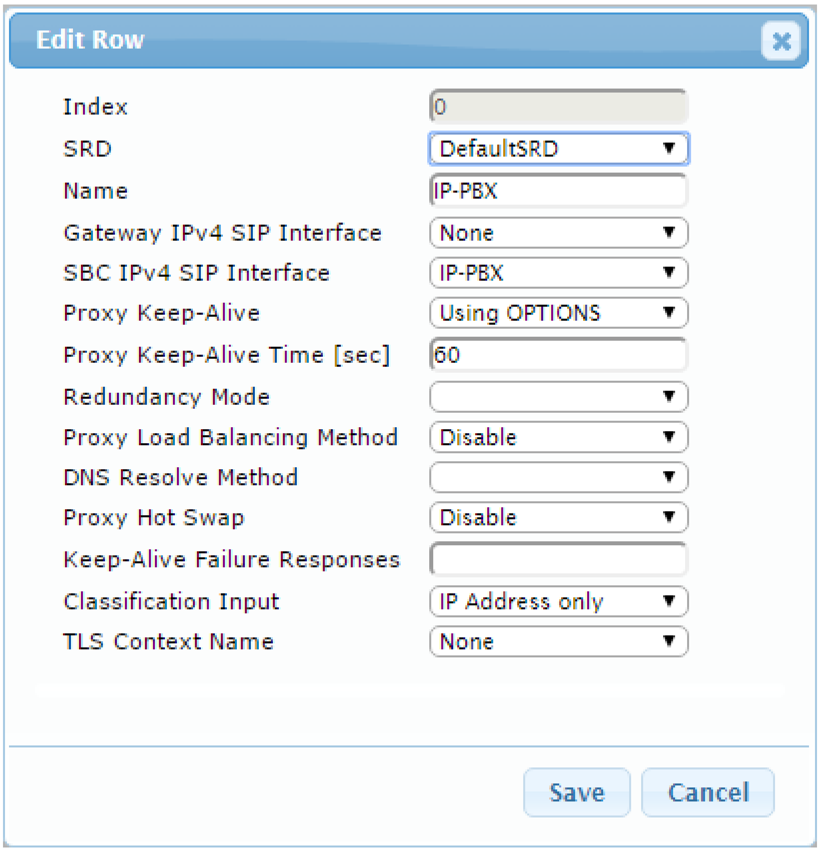
The Proxy Sets will be later applying to the VoIP network by assigning them to IP Groups.

➢ To configure Proxy Sets:

1. Open the Proxy Sets Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table**).
2. Add a Proxy Set for the IP-PBX. You can use the default Proxy Set (Index 0), but modify it as shown below:

Parameter	Value
Proxy Set ID	**0**
Proxy Name	**IP-PBX**
SBC IPv4 SIP Interface	**IP-PBX**
Proxy Keep Alive	**Using Options**

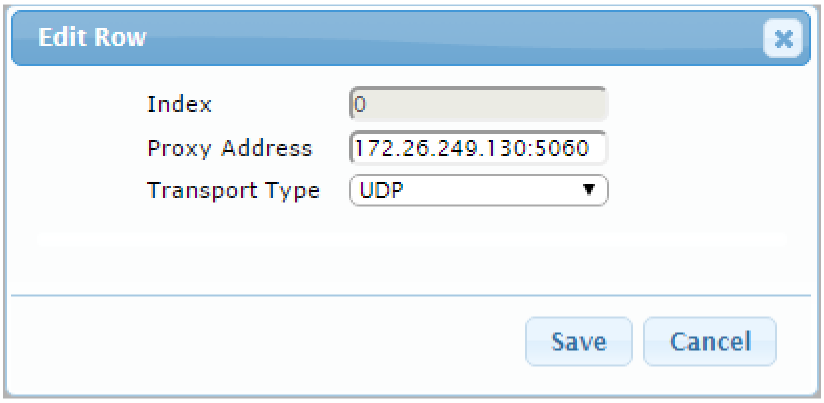
Figure 4-9: Configuring Proxy Set for IP-PBX



3. Configure a Proxy Address Table for Proxy Set for IP-PBX:
   a. Go to Configuration tab > VoIP menu > VoIP Network > Proxy Sets Table > Proxy Address Table.

Parameter	Value
Index	**0**
Proxy Address	**172.26.249.130:5060** (IP-PBX IP address / FQDN and destination port)
Transport Type	**UDP**

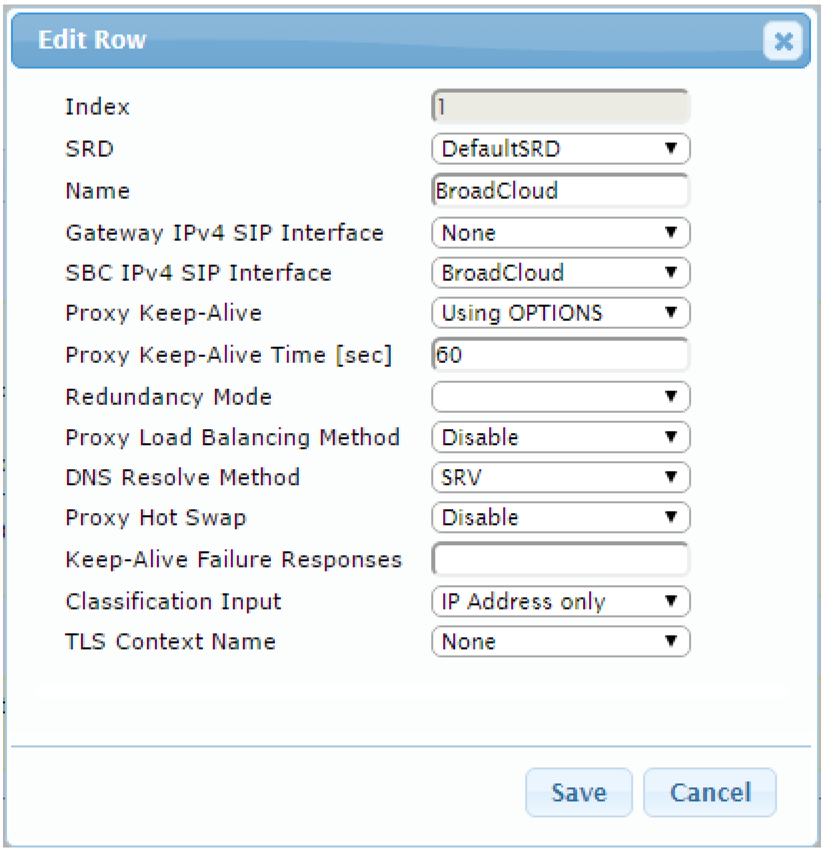
Figure 4-10: Configuring Proxy Address for IP-PBX



4. Configure a Proxy Set for the BroadCloud SIP Trunk:

Parameter	Value
Proxy Set ID	**1**
Proxy Name	**BroadCloud**
SBC IPv4 SIP Interface	**BroadCloud**
Proxy Keep Alive	**Using Options**

Figure 4-11: Configuring Proxy Set for BroadCloud SIP Trunk

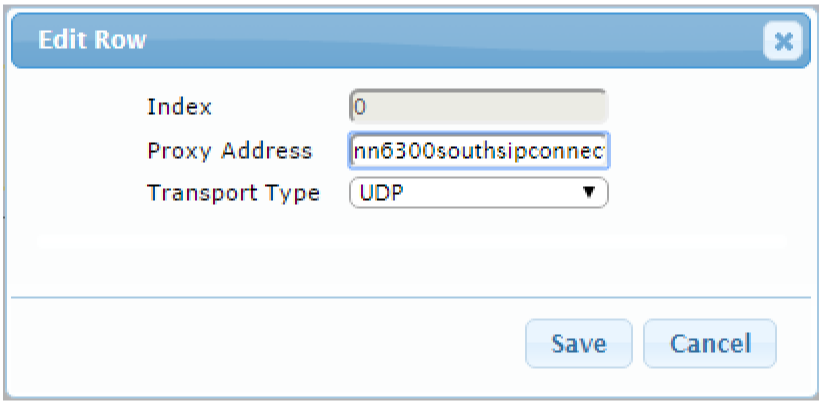


a. Configure a Proxy Address Table for Proxy Set 1:

b. Go to Configuration tab > VoIP menu > VoIP Network > Proxy Sets Table > Proxy Address Table.

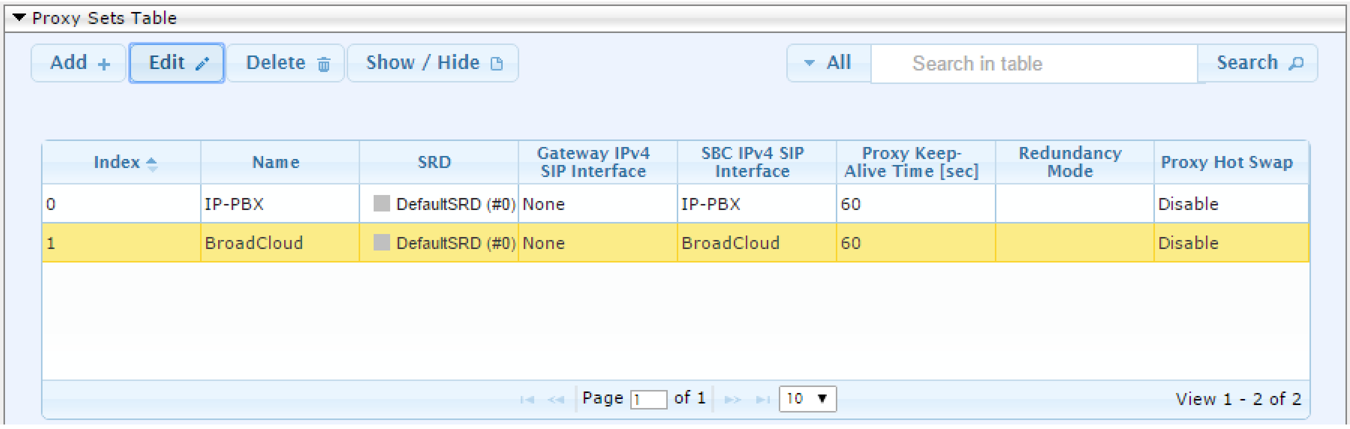
Parameter	Value
Index	**0**
Proxy Address	**nn6300southsipconnect.adpt-tech.com** (IP-PBX IP address / FQDN and destination port)
Transport Type	**UDP**

Figure 4-12: Configuring Proxy Address for



The configured Proxy Sets are shown in the figure below:

Figure 4-13: Configured Proxy Sets in Proxy Sets Table



Index	Name	SRD	Gateway IPv4 SIP Interface	SBC IPv4 SIP Interface	Proxy Keep-Alive Time [sec]	Redundancy Mode	Proxy Hot Swap
0	IP-PBX	DefaultSRD (#0)	None	IP-PBX	60		Disable
1	BroadCloud	DefaultSRD (#0)	None	BroadCloud	60		Disable

4.6 Step 6: Configure IP Profiles

This step describes how to configure IP Profiles. The IP Profile defines a set of call capabilities relating to signaling (e.g., SIP message terminations such as REFER) and media (e.g., coder and transcoding method).

In this interoperability test topology, IP Profiles need to be configured for the following IP entities:

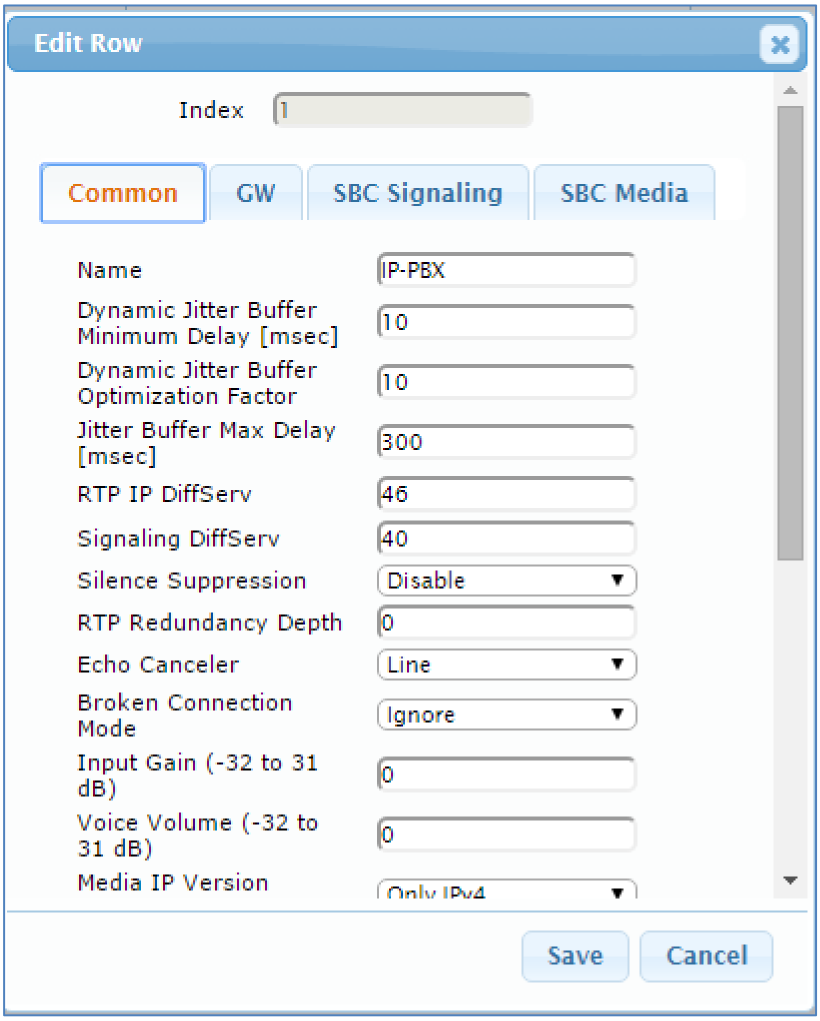
- IP-PBX - to operate in non-secure mode using RTP and UDP
- BroadCloud SIP trunk - to operate in non-secure mode using RTP and UDP

➢ **To configure IP Profile for the IP-PBX:**

1. Open the IP Profile Settings page (**Configuration** tab > **VoIP** > **Coders and Profiles** > **IP Profile Settings**).
2. Click **Add**.
3. Click the **Common** tab, and then configure the parameters as follows:

Parameter	Value
Index	**1**
Name	**IP-PBX**

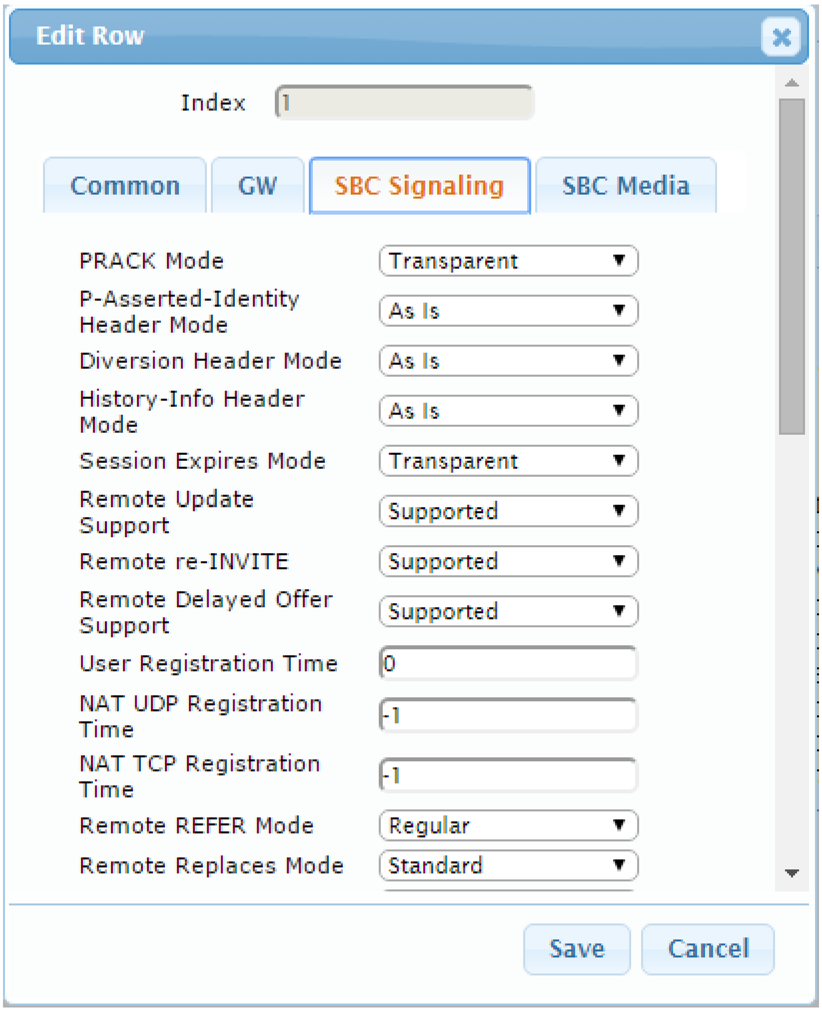
Figure 4-14: Configuring IP Profile for IP-PBX – Common Tab



4. Click the **SBC Signaling** tab, and then configure the parameters as follows:

Parameter	Value
Remote Update Support	**Supported**
Remote re-INVITE Support	**Supported**

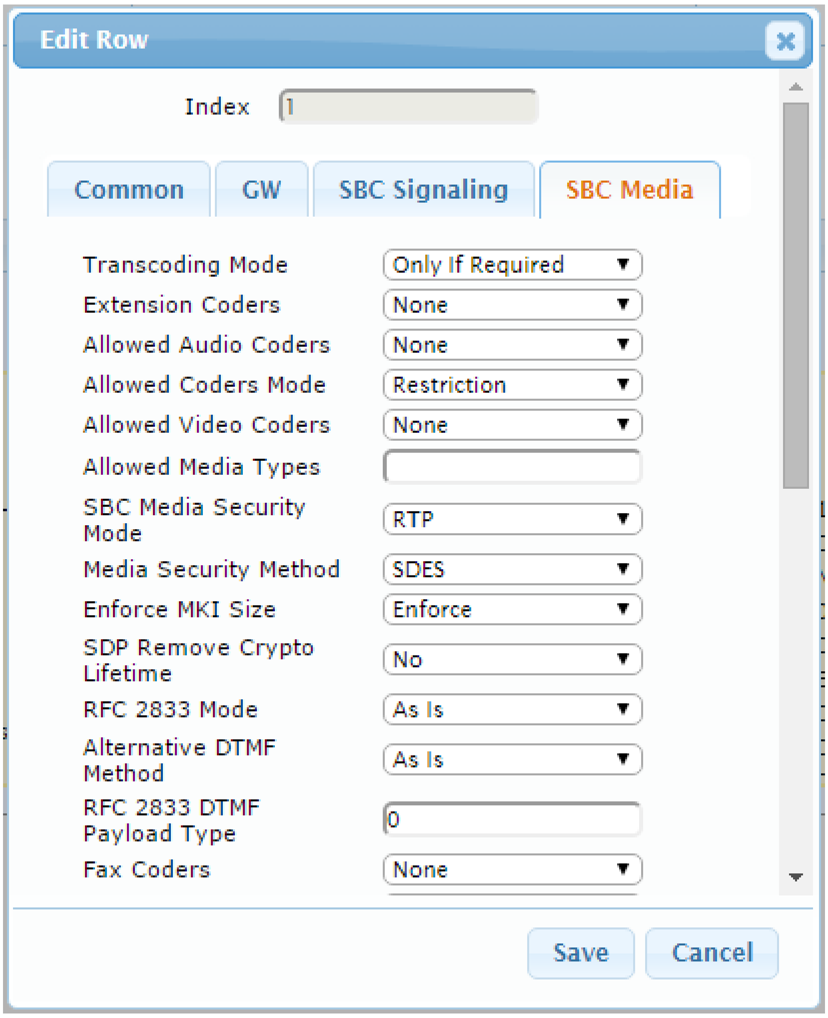
Figure 4-15: Configuring IP Profile for IP-PBX – SBC Signaling Tab



5. Click the **SBC Media** tab, and then configure the parameters as follows:

Parameter	Value
Media Security Behavior	**RTP**

Figure 4-16: Configuring IP Profile for IP-PBX – SBC Media Tab

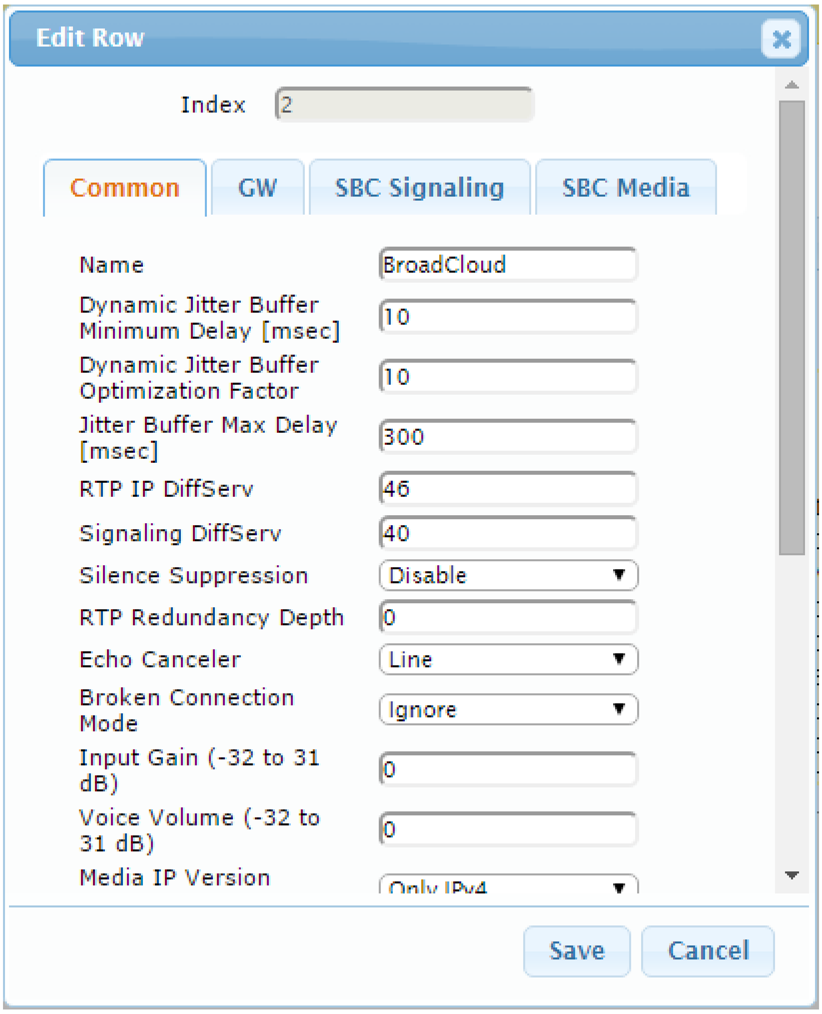


➢ **To configure an IP Profile for the BroadCloud SIP Trunk:**

1. Click **Add**.
2. Click the **Common** tab, and then configure the parameters as follows:

Parameter	Value
Index	**2**
Profile Name	**BroadCloud**

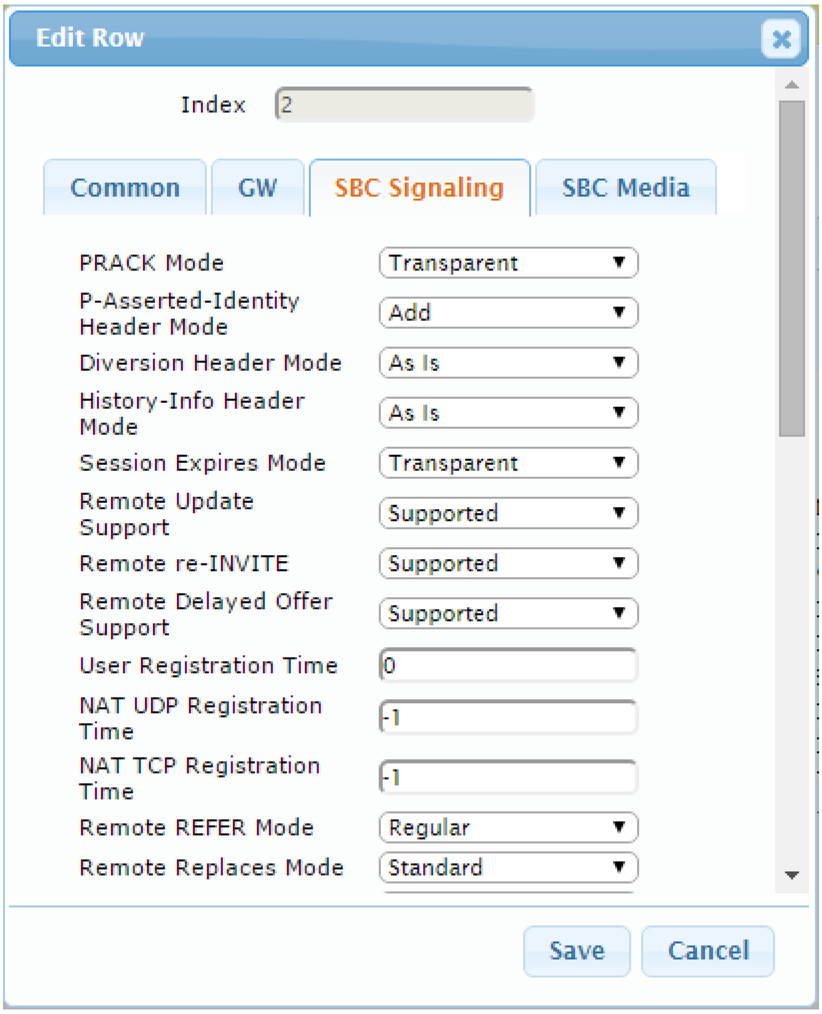
Figure 4-17: Configuring IP Profile for BroadCloud SIP Trunk – Common Tab



3. Click the **SBC Signaling** tab, and then configure the parameters as follows:

Parameter	Value
P-Asserted-Identity Header Mode	**Add** (required for anonymous calls)

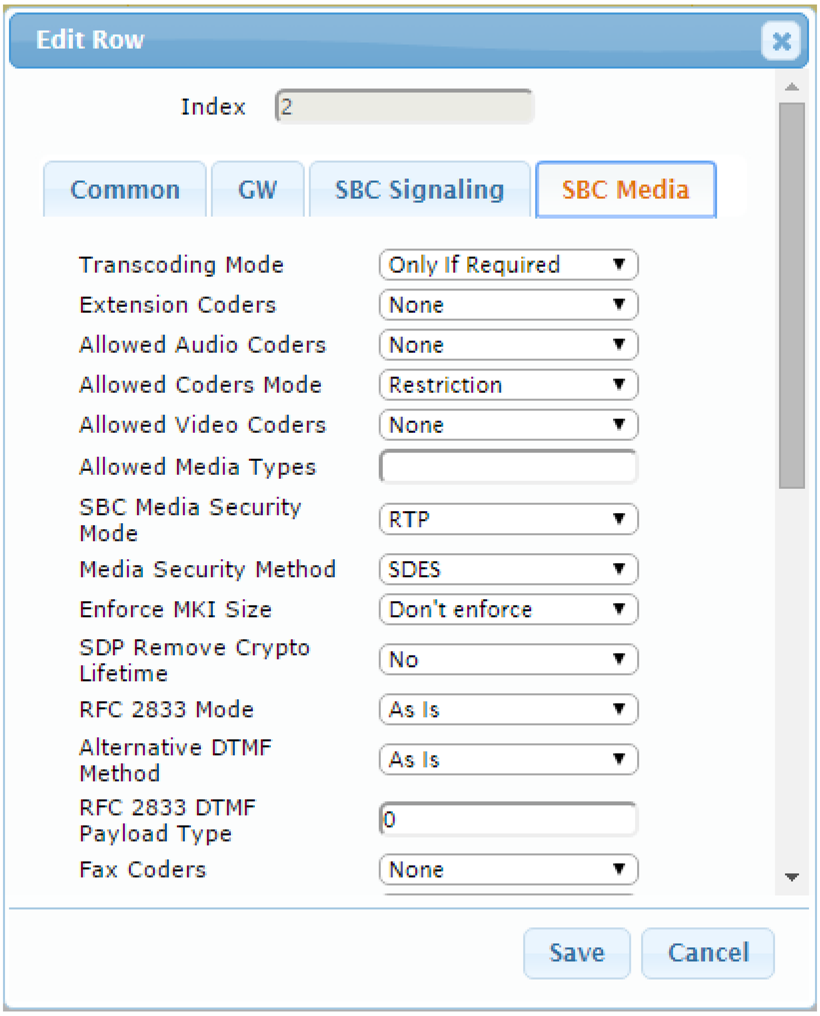
Figure 4-18: Configuring IP Profile for BroadCloud SIP Trunk – SBC Signaling Tab



4. Click the **SBC Media** tab, and then configure the parameters as follows:

Parameter	Value
Media Security Behavior	**RTP**

Figure 4-19: Configuring IP Profile for BroadCloud SIP Trunk – SBC Media Tab



4.7 Step 7: Configure IP Groups

This step describes how to configure IP Groups. The IP Group represents an IP entity on the network with which the E-SBC communicates. This can be a server (e.g., IP PBX or ITSP) or it can be a group of users (e.g., LAN IP phones). For servers, the IP Group is typically used to define the server's IP address by associating it with a Proxy Set. Once IP Groups are configured, they are used to configure IP-to-IP routing rules for denoting source and destination of the call.

In this interoperability test topology, IP Groups must be configured for the following IP entities:

- IP-PBX located on LAN
- BroadCloud SIP Trunk located on WAN

➢ **To configure IP Groups:**

1. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).
2. Add an IP Group for the IP-PBX. You can use the default IP Group (Index 0), but modify it as shown below:

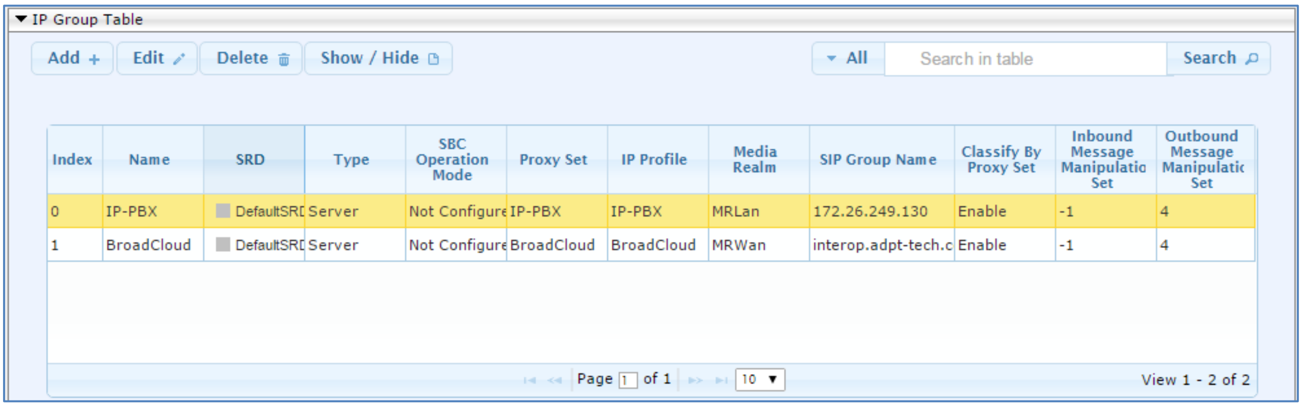
Parameter	Value
Index	**0**
Name	**IP-PBX**
Type	**Server**
Proxy Set	**IP-PBX**
IP Profile	**IP-PBX**
Media Realm	**MRLan**
SIP Group Name	**172.26.249.130** (according to IP-PBX requirement)

3. Configure an IP Group for the BroadCloud SIP Trunk:

Parameter	Value
Index	**1**
Name	**BroadCloud**
Type	**Server**
Proxy Set	**BroadCloud**
IP Profile	**BroadCloud**
Media Realm	**MRWan**
SIP Group Name	**interop.adpt-tech.com** (according to ITSP requirement)

The configured IP Groups are shown in the figure below:

Figure 4-20: Configured IP Groups in IP Group Table



Index	Name	SRD	Type	SBC Operation Mode	Proxy Set	IP Profile	Media Realm	SIP Group Name	Classify By Proxy Set	Inbound Message Manipulatio Set	Outbound Message Manipulatic Set
0	IP-PBX	DefaultSRI	Server	Not Configure	IP-PBX	IP-PBX	MRLan	172.26.249.130	Enable	-1	4
1	BroadCloud	DefaultSRI	Server	Not Configure	BroadCloud	BroadCloud	MRWan	interop.adpt-tech.c	Enable	-1	4

4.8 Step 8: Configure IP-to-IP Call Routing Rules

This step describes how to configure IP-to-IP call routing rules. These rules define the routes for forwarding SIP messages (e.g., INVITE) received from one IP entity to another. The E-SBC selects the rule whose configured input characteristics (e.g., IP Group) match those of the incoming SIP message. If the input characteristics do not match the first rule in the table, they are compared to the second rule, and so on, until a matching rule is located. If no rule is matched, the message is rejected. The routing rules use the configured IP Groups to denote the source and destination of the call. As configured in Section 4.7 on page 37, IP Group 1 represents IP-PBX, and IP Group 2 represents BroadCloud SIP Trunk.

For the interoperability test topology, the following IP-to-IP routing rules need to be configured to route calls between IP-PBX (LAN) and BroadCloud SIP Trunk (WAN):

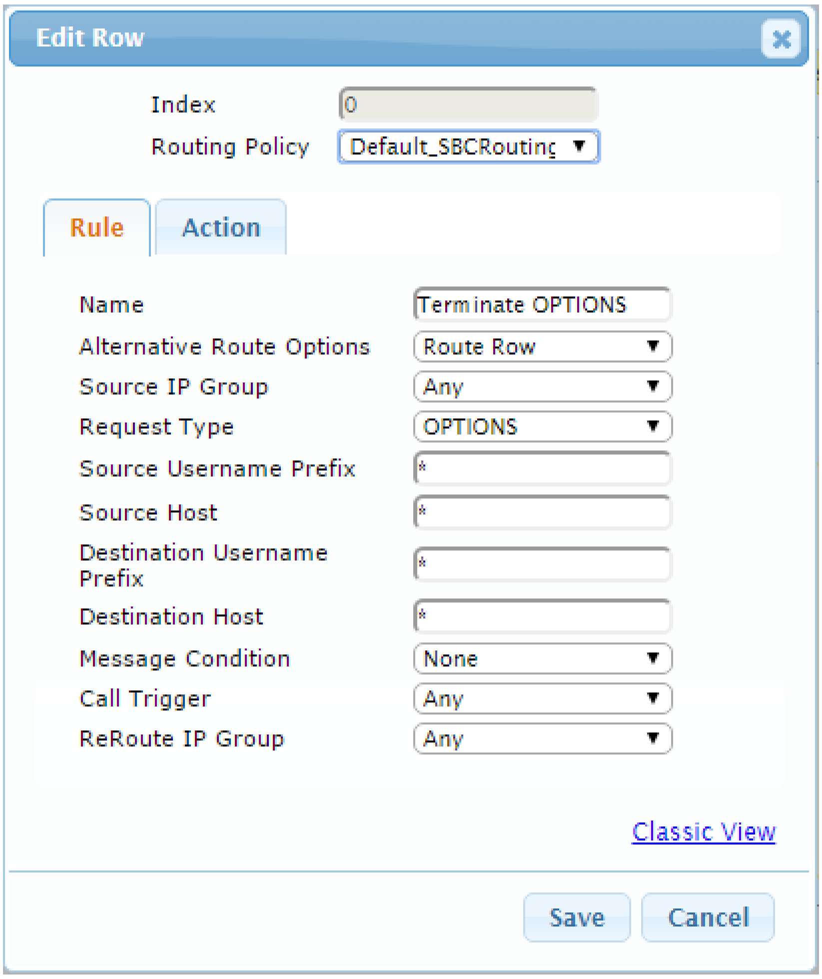
- Terminate SIP OPTIONS messages on the E-SBC
- Calls from IP-PBX to BroadCloud SIP Trunk
- Calls from BroadCloud SIP Trunk to IP-PBX

➢ **To configure IP-to-IP routing rules:**

1. Open the IP-to-IP Routing Table page (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **IP-to-IP Routing Table**).
2. Configure a rule to terminate SIP OPTIONS messages received from the LAN:
   a. Click **Add**.
   b. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**0**
Name	**Terminate OPTIONS** (arbitrary descriptive name)
Source IP Group	**Any**
Request Type	**OPTIONS**

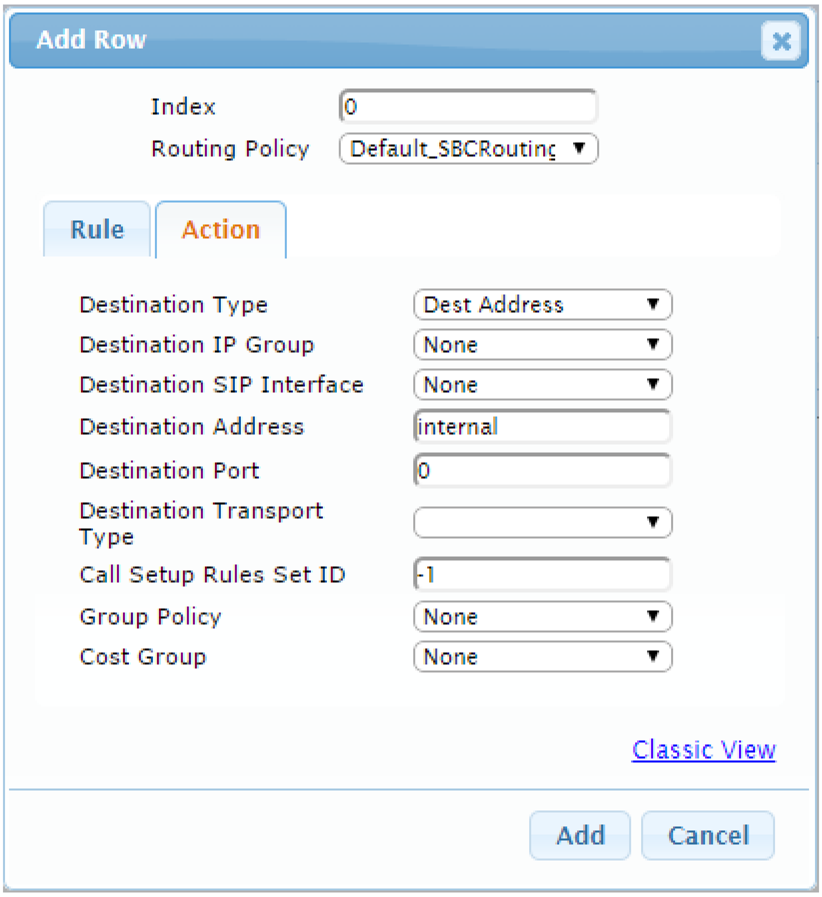
Figure 4-21: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS – Rule Tab



c. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Destination Type	**Dest Address**
Destination Address	**internal**

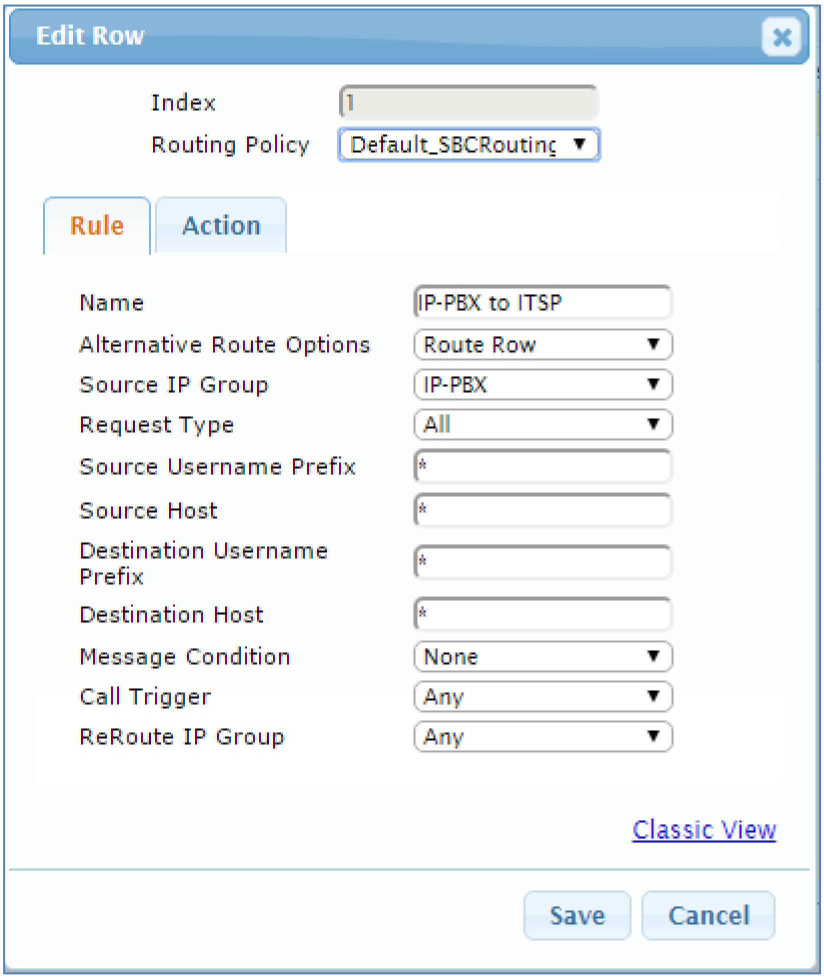
Figure 4-22: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS – Action Tab



3. Configure a rule to route calls from Skype IP-PBX to BroadCloud SIP Trunk:
   a. Click **Add**.
   b. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**1**
Route Name	**IP-PBX to ITSP** (arbitrary descriptive name)
Source IP Group	**IP-PBX**

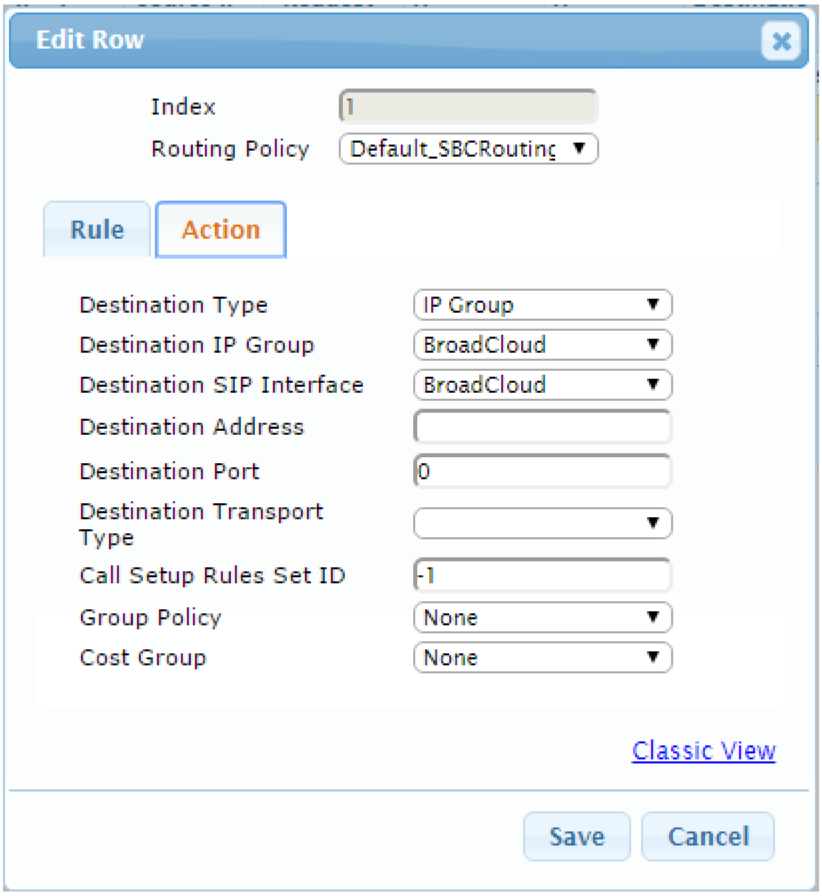
Figure 4-23: Configuring IP-to-IP Routing Rule for IP-PBX to ITSP – Rule tab



c. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Destination Type	**IP Group**
Destination IP Group	**BroadCloud**
Destination SIP Interface	**BroadCloud**

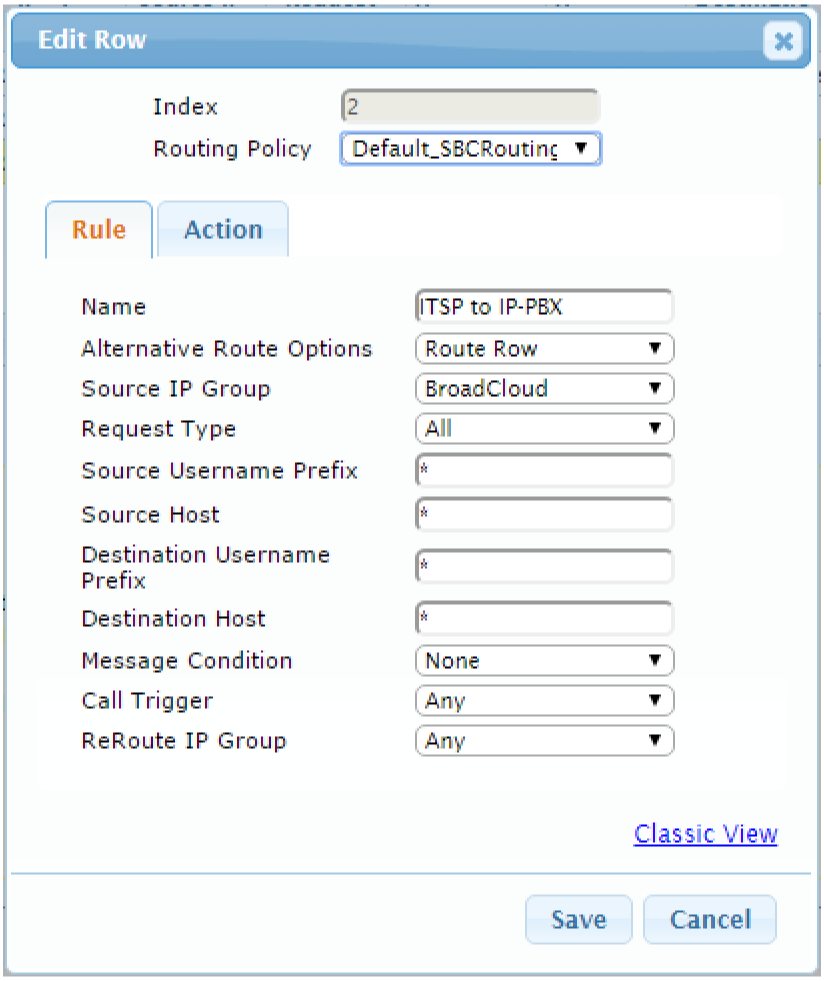
Figure 4-24: Configuring IP-to-IP Routing Rule for IP-PBX to ITSP – Action tab



4. To configure rule to route calls from BroadCloud SIP Trunk to IP-PBX:
   a. Click **Add**.
   b. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**2**
Route Name	**ITSP to IP-PBX** (arbitrary descriptive name)
Source IP Group	**BroadCloud**

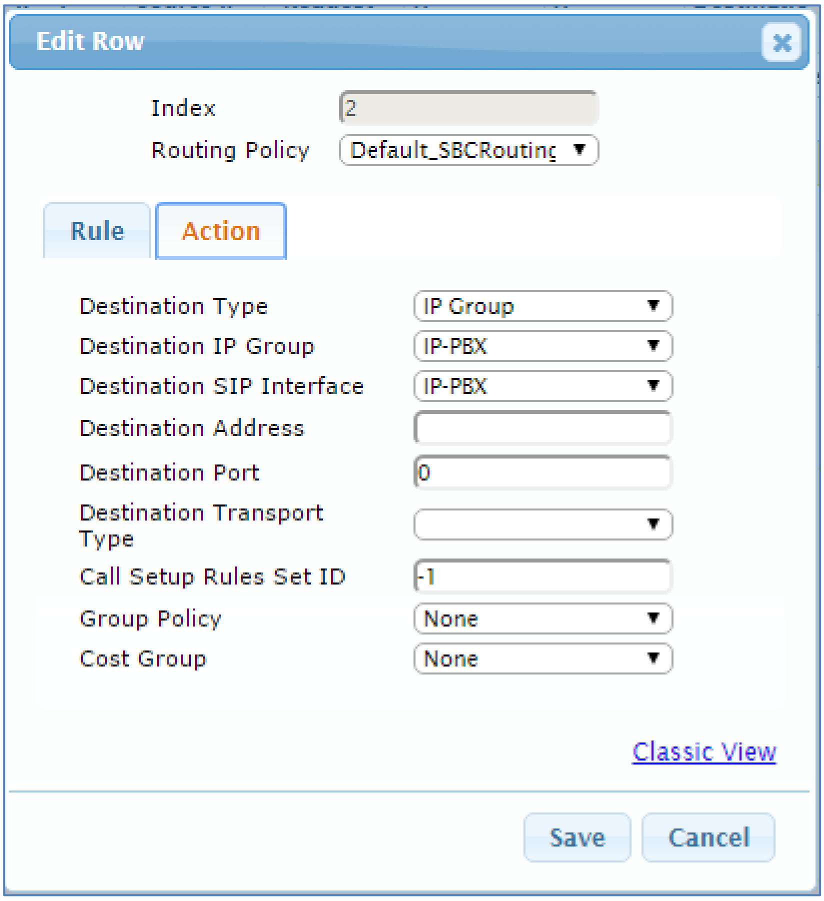
Figure 4-25: Configuring IP-to-IP Routing Rule for ITSP to IP-PBX – Rule tab



a. Click the **Action** tab, and then configure the parameters as follows:

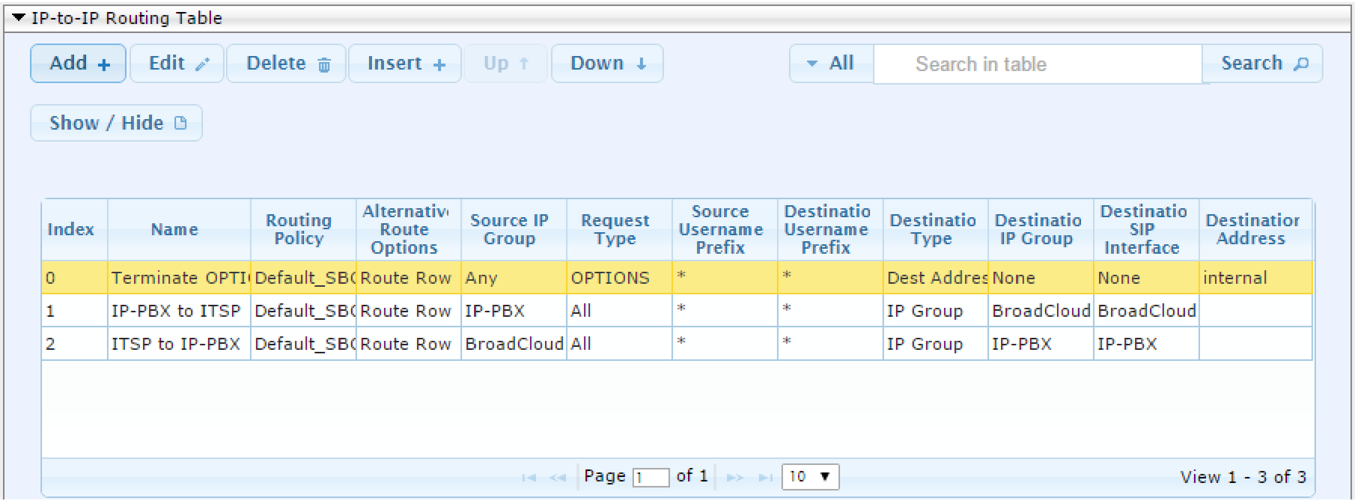
Parameter	Value
Destination Type	**IP Group**
Destination IP Group	**IP-PBX**
Destination SIP Interface	**IP-PBX**

Figure 4-26: Configuring IP-to-IP Routing Rule for ITSP to IP-PBX – Action tab



The configured routing rules are shown in the figure below:

Figure 4-27: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table



Index	Name	Routing Policy	Alternative Route Options	Source IP Group	Request Type	Source Username Prefix	Destination Username Prefix	Destination Type	Destination IP Group	Destination SIP Interface	Destination Address
0	Terminate OPTI	Default_SB(	Route Row	Any	OPTIONS	*	*	Dest Addres	None	None	internal
1	IP-PBX to ITSP	Default_SB(	Route Row	IP-PBX	All	*	*	IP Group	BroadCloud	BroadCloud	
2	ITSP to IP-PBX	Default_SB(	Route Row	BroadCloud	All	*	*	IP Group	IP-PBX	IP-PBX	

Note: The routing configuration may change according to your specific deployment topology.

4.9 Step 9: Configure IP-to-IP Manipulation Rules

This step describes how to configure IP-to-IP manipulation rules. These rules manipulate the source and / or destination number. The manipulation rules use the configured IP Groups to denote the source and destination of the call. As configured in Section 4.7 on page 37, IP Group 0 represents IP-PBX, and IP Group 1 represents BroadCloud SIP Trunk.

Note: Adapt the manipulation table according to you environment dial plan.

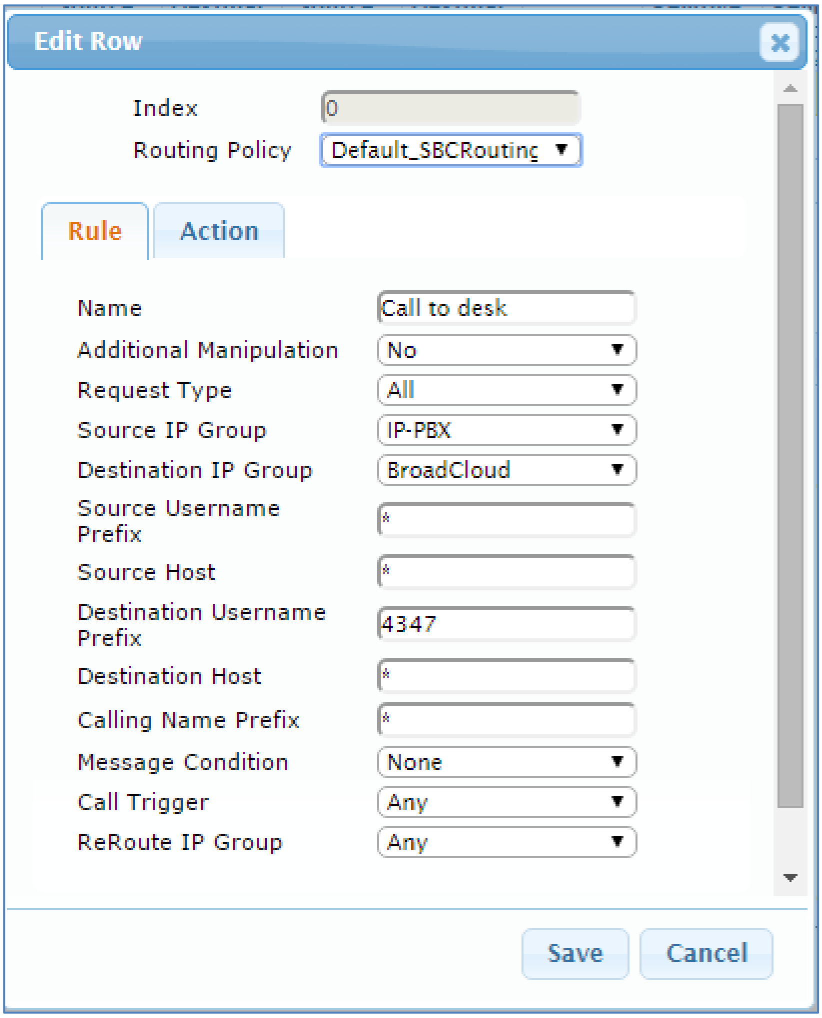
For example, for this interoperability test topology, a manipulation was configured to add the prefix to the destination number for calls from the IP-PBX IP Group to the BroadCloud SIP Trunk IP Group for specific destination username prefix.

➢ **To configure a number manipulation rule:**

1. Open the IP-to-IP Outbound Manipulation page (**Configuration** tab > **VoIP** menu > **SBC** > **Manipulations SBC** > **IP-to-IP Outbound**).
2. Click **Add**.
3. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**0**
Name	**Call to desk**
Source IP Group	**IP-PBX**
Destination IP Group	**BroadCloud**
Destination Username Prefix	**4347**

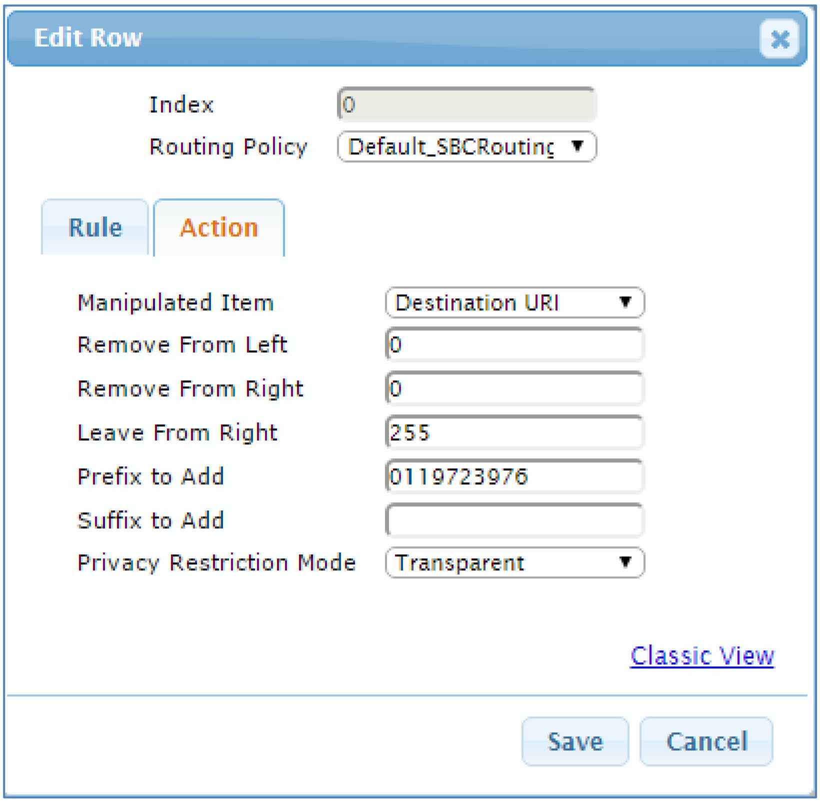
Figure 4-28: Configuring IP-to-IP Outbound Manipulation Rule – Rule Tab



4. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Manipulated Item	**Destination URI**
Prefix to Add	**0119723976**

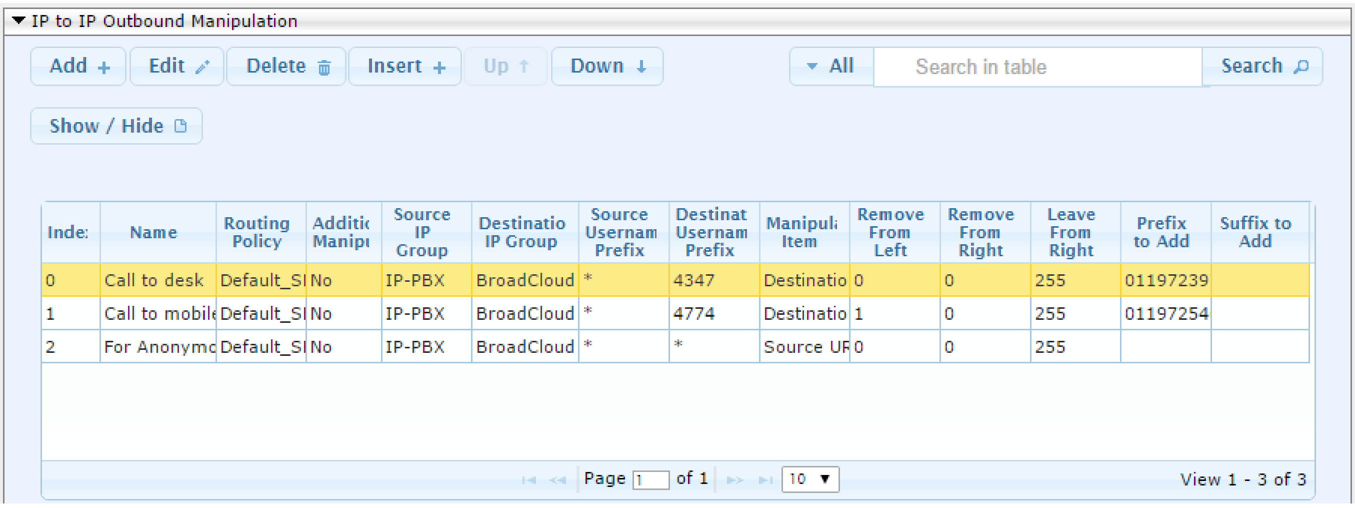
Figure 4-29: Configuring IP-to-IP Outbound Manipulation Rule - Action Tab



5. Click **Submit**.

The figure below shows an example of configured IP-to-IP outbound manipulation rules for calls between IP-PBX IP Group and BroadCloud SIP Trunk IP Group:

Figure 4-30: Example of Configured IP-to-IP Outbound Manipulation Rules



Inde:	Name	Routing Policy	Additic Manipı	Source IP Group	Destinatio IP Group	Source Usernam Prefix	Destinat Usernam Prefix	Manipulı Item	Remove From Left	Remove From Right	Leave From Right	Prefix to Add	Suffix to Add
0	Call to desk	Default_SI	No	IP-PBX	BroadCloud	*	4347	Destinatio	0	0	255	01197239	
1	Call to mobile	Default_SI	No	IP-PBX	BroadCloud	*	4774	Destinatio	1	0	255	01197254	
2	For Anonymc	Default_SI	No	IP-PBX	BroadCloud	*	*	Source UR	0	0	255		

4.10 Step 10: Configure Message Manipulation Rules

This step describes how to configure SIP message manipulation rules. SIP message manipulation rules can include insertion, removal, and/or modification of SIP headers. Manipulation rules are grouped into Manipulation Sets, enabling you to apply multiple rules to the same SIP message (IP entity).

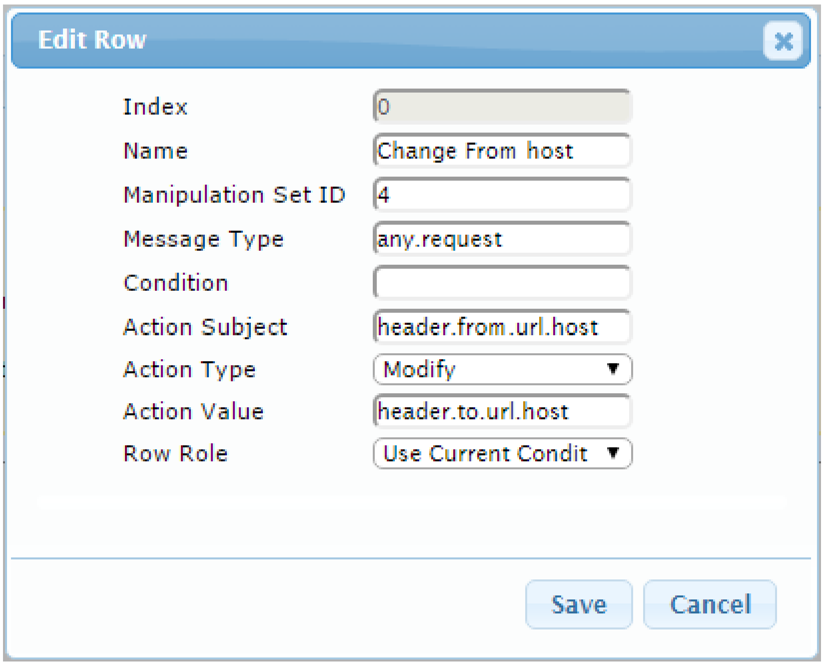
Once you have configured the SIP message manipulation rules, you need to assign them to the relevant IP Group (in the IP Group table) and determine whether they must be applied to inbound or outbound messages.

➢ **To configure SIP message manipulation rule:**

1. Open the Message Manipulations page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Msg Policy & Manipulation** > **Message Manipulations**).
2. Configure a new manipulation rule (Manipulation Set 4) for BroadCloud SIP Trunk. This rule applies to messages sent to the BroadCloud SIP Trunk IP Group. This replaces the host part of the SIP From Header with the value from the SIP To Header.

Parameter	Value
Index	**0**
Name	**Change From host**
Manipulation Set ID	**4**
Message Type	**any.request**
Action Subject	**header.from.url.host**
Action Type	**Modify**
Action Value	**header.to.url.host**

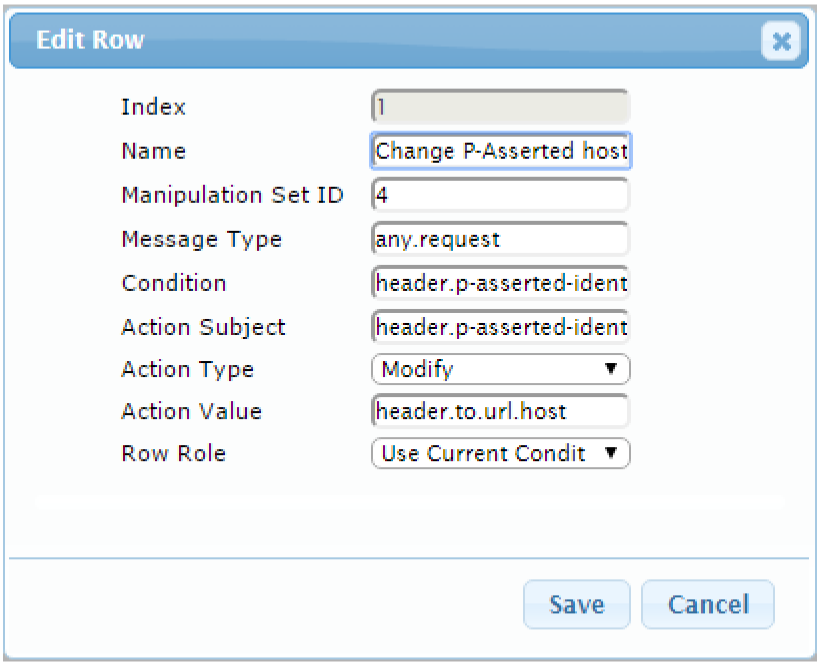
Figure 4-31: Configuring SIP Message Manipulation Rule 0 (for BroadCloud SIP Trunk)



3. Configure another manipulation rule (Manipulation Set 4) for BroadCloud SIP Trunk. This rule applies to messages sent to the BroadCloud SIP Trunk IP Group. This replaces the host part of the SIP P-Asserted-Identity Header with the value from the SIP To Header.

Parameter	Value
Index	**1**
Manipulation Name	**Change P-Asserted host**
Manipulation Set ID	**4**
Message Type	**any.request**
Condition	**header.p-asserted-identity exists**
Action Subject	**header.p-asserted-identity**
Action Type	**Modify**
Action Value	**header.to.url.host**

Figure 4-32: Configuring SIP Message Manipulation Rule 1 (for BroadCloud SIP Trunk)

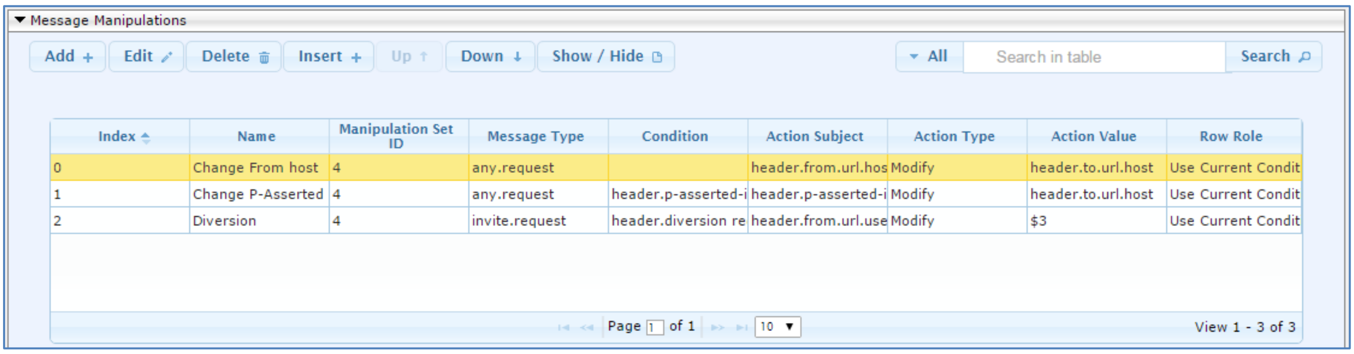


4. Configure another manipulation rule (Manipulation Set 4) for BroadCloud SIP Trunk. This rule applies to messages sent to the BroadCloud SIP Trunk IP Group in the call transfer scenario. This replaces the user part of the SIP From Header with the value from the SIP Diversion Header.

Parameter	Value
Index	**2**
Manipulation Name	**Diversion**
Manipulation Set ID	**4**
Message Type	**invite.request**
Condition	**header.diversion regex (<sip:)(..)(.*)(@)(.*)**
Action Subject	**header.from.url.user**
Action Type	**Modify**
Action Value	**$3**

Figure 4-33: Configuring SIP Message Manipulation Rule 2 (for BroadCloud SIP Trunk)

Figure 4-34: Example of Configured SIP Message Manipulation Rules

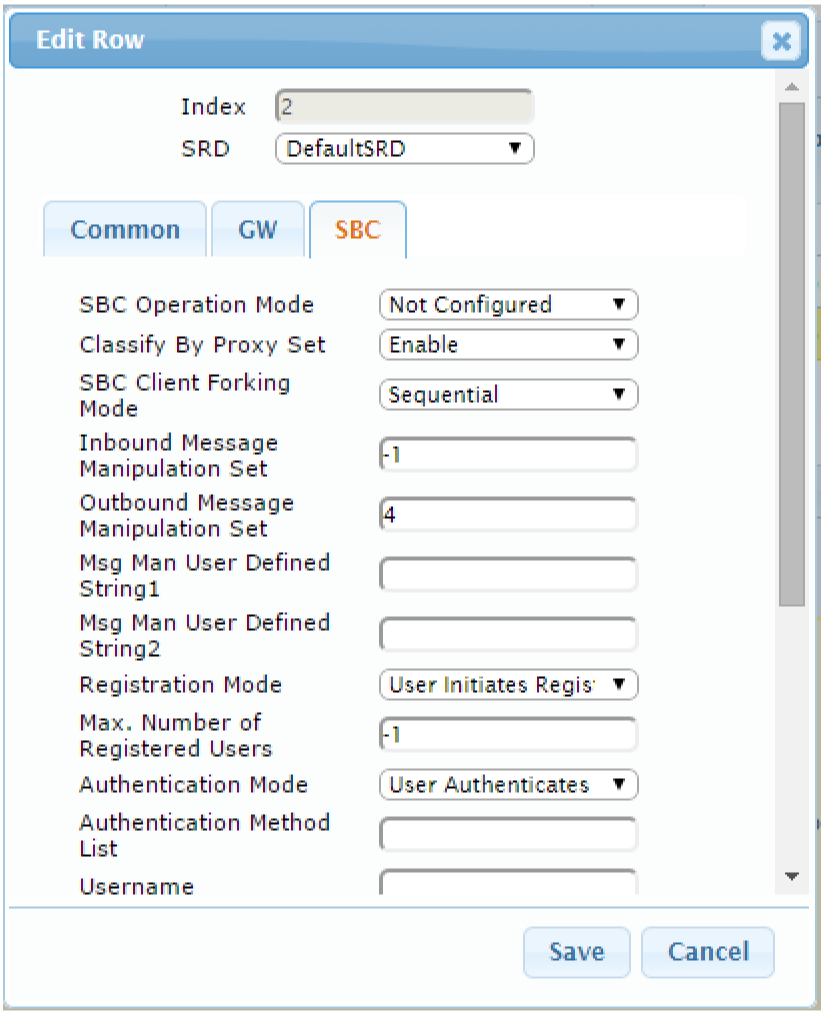


The table displayed below includes SIP message manipulation rules, which are bound together by commonality via the Manipulation Set ID 4, which are executed for messages sent to the BroadCloud SIP Trunk IP Group. These rules are specifically required to enable proper interworking between BroadCloud SIP Trunk and IP-PBX. Refer to the *User's Manual* for further details concerning the full capabilities of header manipulation.

Rule Index	Rule Description	Reason for Introducing Rule
0	This rule applies to messages sent to the BroadCloud SIP Trunk IP Group. This replaces the host part of the SIP From Header with the value from the SIP To Header.	BroadCloud SIP Trunk required that all messages should be from known hosts.
1	This rule applies to messages sent to the BroadCloud SIP Trunk IP Group. This replaces the host part of the SIP P-Asserted-Identity Header with the value from the SIP To Header.	
2	This rule applies to messages sent to the BroadCloud SIP Trunk IP Group in the call transfer scenario. This replaces the user part of the SIP From Header with the value from the SIP Diversion Header.	

5. Assign Manipulation Set ID 4 to the BroadCloud SIP trunk IP Group:
   a. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).
   b. Select the row of the BroadCloud SIP trunk IP Group, and then click **Edit**.
   c. Click the **SBC** tab.
   d. Set the 'Outbound Message Manipulation Set' field to **4**.

Figure 4-35: Assigning Manipulation Set 4 to the BroadCloud SIP Trunk IP Group



   e. Click **Submit**.

4.11 Step 11: Configure Registration Accounts

This step describes how to configure SIP registration accounts. This is required so that the E-SBC can register with the BroadCloud SIP Trunk on behalf of IP-PBX. The BroadCloud SIP Trunk requires registration and authentication to provide service.

In the interoperability test topology, the Served IP Group is IP-PBX IP Group and the Serving IP Group is BroadCloud SIP Trunk IP Group.

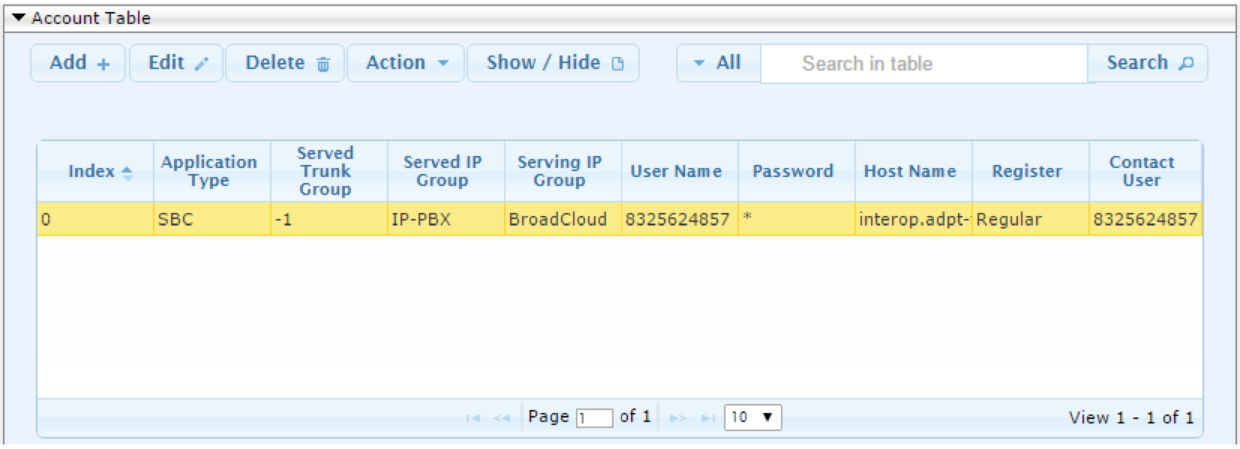
➢ **To configure a registration account:**

1. Open the Account Table page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Account Table**).
2. Enter an index number (e.g., "0"), and then click **Add**.
3. Configure the account according to the provided information from , for example:

Parameter	Value
Application Type	**SBC**
Served IP Group	**IP-PBX**
Serving IP Group	**BroadCloud**
Username	As provided by BroadCloud
Password	As provided by BroadCloud
Host Name	**interop.adpt-tech.com**
Register	**Regular**
Contact User	**8325624857** (pilot number)

4. Click **Apply**.

Figure 4-36: Configuring SIP Registration Account



4.12 Step 12: Miscellaneous Configuration

This section describes miscellaneous E-SBC configuration.

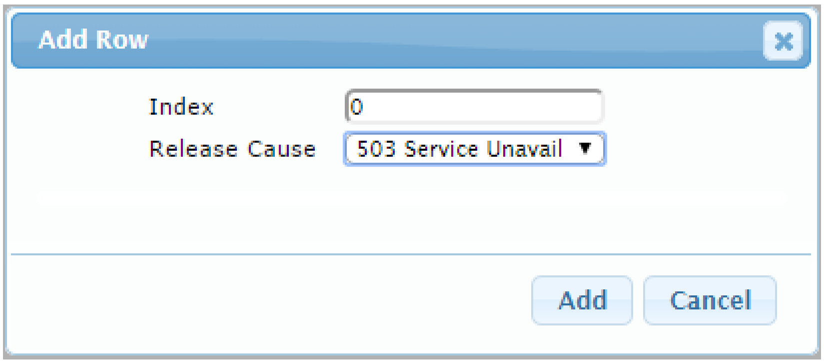
4.12.1 Step 12a: Configure SBC Alternative Routing Reasons

This step describes how to configure the E-SBC's handling of SIP 503 responses received for outgoing SIP dialog-initiating methods, e.g., INVITE, OPTIONS, and SUBSCRIBE messages. In this case E-SBC attempts to locate an alternative route for the call.

➢ **To configure SIP reason codes for alternative IP routing:**

1. Open the SBC Alternative Routing Reasons page (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **SBC Alternative Routing Reasons**).
2. Click **Add**; the following dialog box appears:

Figure 4-37: SBC Alternative Routing Reasons Table - Add Record



3. Click **Submit**.

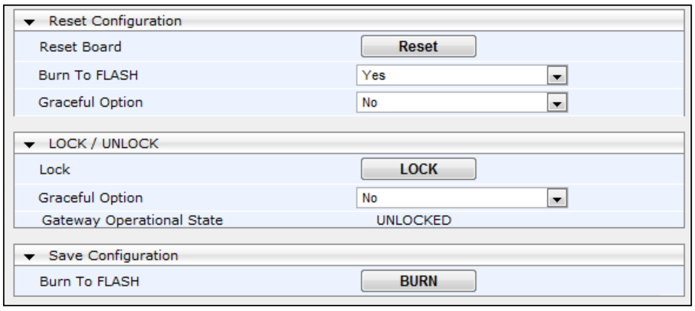
4.13 Step 13: Reset the E-SBC

After you have completed the configuration of the E-SBC described in this chapter, save ("burn") the configuration to the E-SBC's flash memory with a reset for the settings to take effect.

➢ **To save the configuration to flash memory:**

1. Open the Maintenance Actions page (**Maintenance** tab > **Maintenance** menu > **Maintenance Actions**).

Figure 4-38: Resetting the E-SBC



2. Ensure that the 'Burn to FLASH' field is set to **Yes** (default).
3. Click the **Reset** button.

This page is intentionally left blank.

A AudioCodes INI File

The *ini* configuration file of the E-SBC, corresponding to the Web-based configuration as described in Section 4 on page 25, is shown below:

Note: To load and save an ini file, use the Configuration File page (**Maintenance** tab > **Software Update** menu > **Configuration File**).

```
;**************
;** Ini File **
;**************

;Board: Mediant 800 E-SBC
;HW Board Type: 69  FK Board Type: 72
;Serial Number: 5916116
;Slot Number: 1
;Software Version: 7.00A.049.003
;DSP Software Version: 5014AE3_R => 700.44
;Board IP Address: 172.21.128.28
;Board Subnet Mask: 255.255.0.0
;Board Default Gateway: 172.21.1.1
;Ram size: 496M   Flash size: 64M   Core speed: 500Mhz
;Num of DSP Cores: 3  Num DSP Channels: 90
;Num of physical LAN ports: 4
;Profile: NONE
;;;Key features:;Board Type: 72 ;QOE features: VoiceQualityMonitoring
MediaEnhancement ;IP Media: VXML ;Channel Type: DspCh=90 ;HA ;BRITrunks=6
;DATA features: ;Security: IPSEC MediaEncryption StrongEncryption
EncryptControlProtocol ;Coders: G723 G729 G728 NETCODER GSM-FR GSM-EFR
AMR EVRC-QCELP G727 ILBC EVRC-B AMR-WB G722 EG711 MS_RTA_NB MS_RTA_WB
SILK_NB SILK_WB SPEEX_NB SPEEX_WB OPUS_NB OPUS_WB ;DSP Voice features:
RTCP-XR V150=50 ;E1Trunks=2 ;T1Trunks=2 ;E&M Ports=6 ;Control Protocols:
MSFT FEU=600 TestCall=100 MGCP SIP SASurvivability SBC=100 ;Default
features:;Coders: G711 G726;

;------  HW components------
;
; Slot # : Module type : # of ports
;----------------------------------------------
;      1 : FALC56      : 1
;      2 : Empty
;      3 : Empty
;----------------------------------------------


[SYSTEM Params]

SyslogServerIP = 172.20.22.17
EnableSyslog = 1
NTPServerUTCOffset = 7200
;VpFileLastUpdateTime is hidden but has non-default value
NTPServerIP = '0.0.0.0'
;LastConfigChangeTime is hidden but has non-default value
;PM_gwINVITEDialogs is hidden but has non-default value
```

```
;PM_gwSUBSCRIBEDialogs is hidden but has non-default value
;PM_gwSBCRegisteredUsers is hidden but has non-default value
;PM_gwSBCMediaLegs is hidden but has non-default value
;PM_gwSBCTranscodingSessions is hidden but has non-default value

[BSP Params]

PCMLawSelect = 3
UdpPortSpacing = 10
EnterCpuOverloadPercent = 99
ExitCpuOverloadPercent = 95

[Analog Params]


[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]


[MEGACO Params]

EP_Num_0 = 0
EP_Num_1 = 1
EP_Num_2 = 1
EP_Num_3 = 0
EP_Num_4 = 0

[PSTN Params]


[SS7 Params]


[Voice Engine Params]

ENABLEMEDIASECURITY = 1

[WEB Params]

UseRProductName = 'Mediant 800 E-SBC'
WebLogoText = 'BroadCloud'
UseWeblogo = 1
;UseLogoInWeb is hidden but has non-default value
UseProductName = 1
HTTPSCipherString = 'RC4:EXP'
;HTTPSPkeyFileName is hidden but has non-default value

[SIP Params]

MEDIACHANNELS = 30
GWDEBUGLEVEL = 5
;ISPRACKREQUIRED is hidden but has non-default value
ENABLESBCAPPLICATION = 1
```

```
MSLDAPPRIMARYKEY = 'telephoneNumber'
MEDIACDRREPORTLEVEL = 1
SBCFORKINGHANDLINGMODE = 1
ENERGYDETECTORCMD = 587202560
ANSWERDETECTORCMD = 10486144
;GWAPPCONFIGURATIONVERSION is hidden but has non-default value

[SCTP Params]


[IPsec Params]


[Audio Staging Params]


[SNMP Params]


[ PhysicalPortsTable ]

FORMAT PhysicalPortsTable_Index = PhysicalPortsTable_Port,
PhysicalPortsTable_Mode, PhysicalPortsTable_SpeedDuplex,
PhysicalPortsTable_PortDescription, PhysicalPortsTable_GroupMember,
PhysicalPortsTable_GroupStatus;
PhysicalPortsTable 0 = "GE_4_1", 1, 4, "User Port #0", "GROUP_1",
"Active";
PhysicalPortsTable 1 = "GE_4_2", 1, 4, "User Port #1", "GROUP_1",
"Redundant";
PhysicalPortsTable 2 = "GE_4_3", 1, 4, "User Port #2", "GROUP_2",
"Active";
PhysicalPortsTable 3 = "GE_4_4", 1, 4, "User Port #3", "GROUP_2",
"Redundant";

[ \PhysicalPortsTable ]


[ EtherGroupTable ]

FORMAT EtherGroupTable_Index = EtherGroupTable_Group,
EtherGroupTable_Mode, EtherGroupTable_Member1, EtherGroupTable_Member2;
EtherGroupTable 0 = "GROUP_1", 2, "GE_4_1", "GE_4_2";
EtherGroupTable 1 = "GROUP_2", 2, "GE_4_3", "GE_4_4";
EtherGroupTable 2 = "GROUP_3", 0, "", "";
EtherGroupTable 3 = "GROUP_4", 0, "", "";

[ \EtherGroupTable ]


[ DeviceTable ]

FORMAT DeviceTable_Index = DeviceTable_VlanID,
DeviceTable_UnderlyingInterface, DeviceTable_DeviceName,
DeviceTable_Tagging;
DeviceTable 0 = 1, "GROUP_1", "vlan 1", 0;
DeviceTable 1 = 2, "GROUP_2", "vlan 2", 0;

[ \DeviceTable ]
```

```
[ InterfaceTable ]

FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_InterfaceName, InterfaceTable_PrimaryDNSServerIPAddress,
InterfaceTable_SecondaryDNSServerIPAddress,
InterfaceTable_UnderlyingDevice;
InterfaceTable 0 = 6, 10, 172.26.249.31, 24, 172.26.249.1, "ShoreTel",
0.0.0.0, 0.0.0.0, "vlan 1";
InterfaceTable 1 = 5, 10, 65.196.9.185, 28, 65.196.9.177, "DMZ",
198.6.1.146, 198.6.1.122, "vlan 2";

[ \InterfaceTable ]


[ DspTemplates ]

;
;  *** TABLE DspTemplates ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts.
;

[ \DspTemplates ]


[ WebUsers ]

FORMAT WebUsers_Index = WebUsers_Username, WebUsers_Password,
WebUsers_Status, WebUsers_PwAgeInterval, WebUsers_SessionLimit,
WebUsers_SessionTimeout, WebUsers_BlockTime, WebUsers_UserLevel,
WebUsers_PwNonce;
WebUsers 0 = "Admin",
"$1$z/3i5+fh5+Hn5rvq4+vruby+1NDS14XdhYPQ3onZjojYiZPDw8HAxpTCnJvLw8rIxppmZ
WczZ2c+P20xOD1uOzc=", 1, 0, 2, 15, 60, 200,
"a4e40b4a1ef60fad38601e9bf6d0c1ce";
WebUsers 1 = "User",
"$1$EiUhIXBycnohfit/L3otExUbFkYcFBJMERNJGUwYGVIGV1UFB1VSDl8MAlhbDA5ydHdxd
CR/Jn15Ln11e38qMWg=", 3, 0, 2, 15, 60, 50,
"a5bdea28146076a2e00cabbb04f2139f";

[ \WebUsers ]


[ TLSContexts ]

FORMAT TLSContexts_Index = TLSContexts_Name, TLSContexts_TLSVersion,
TLSContexts_ServerCipherString, TLSContexts_ClientCipherString,
TLSContexts_OcspEnable, TLSContexts_OcspServerPrimary,
TLSContexts_OcspServerSecondary, TLSContexts_OcspServerPort,
TLSContexts_OcspDefaultResponse;
TLSContexts 0 = "default", 0, "RC4:EXP", "ALL:!ADH", 0, , , 2560, 0;

[ \TLSContexts ]


[ IpProfile ]
```

```
FORMAT IpProfile_Index = IpProfile_ProfileName, IpProfile_IpPreference,
IpProfile_CodersGroupID, IpProfile_IsFaxUsed,
IpProfile_JitterBufMinDelay, IpProfile_JitterBufOptFactor,
IpProfile_IPDiffServ, IpProfile_SigIPDiffServ, IpProfile_SCE,
IpProfile_RTPRedundancyDepth, IpProfile_RemoteBaseUDPPort,
IpProfile_CNGmode, IpProfile_VxxTransportType, IpProfile_NSEMode,
IpProfile_IsDTMFUsed, IpProfile_PlayRBTone2IP,
IpProfile_EnableEarlyMedia, IpProfile_ProgressIndicator2IP,
IpProfile_EnableEchoCanceller, IpProfile_CopyDest2RedirectNumber,
IpProfile_MediaSecurityBehaviour, IpProfile_CallLimit,
IpProfile_DisconnectOnBrokenConnection, IpProfile_FirstTxDtmfOption,
IpProfile_SecondTxDtmfOption, IpProfile_RxDTMFOption,
IpProfile_EnableHold, IpProfile_InputGain, IpProfile_VoiceVolume,
IpProfile_AddIEInSetup, IpProfile_SBCExtensionCodersGroupID,
IpProfile_MediaIPVersionPreference, IpProfile_TranscodingMode,
IpProfile_SBCAllowedMediaTypes, IpProfile_SBCAllowedCodersGroupID,
IpProfile_SBCAllowedVideoCodersGroupID, IpProfile_SBCAllowedCodersMode,
IpProfile_SBCMediaSecurityBehaviour, IpProfile_SBCRFC2833Behavior,
IpProfile_SBCAlternativeDTMFMethod, IpProfile_SBCAssertIdentity,
IpProfile_AMDSensitivityParameterSuit, IpProfile_AMDSensitivityLevel,
IpProfile_AMDMaxGreetingTime, IpProfile_AMDMaxPostSilenceGreetingTime,
IpProfile_SBCDiversionMode, IpProfile_SBCHistoryInfoMode,
IpProfile_EnableQSIGTunneling, IpProfile_SBCFaxCodersGroupID,
IpProfile_SBCFaxBehavior, IpProfile_SBCFaxOfferMode,
IpProfile_SBCFaxAnswerMode, IpProfile_SbcPrackMode,
IpProfile_SBCSessionExpiresMode, IpProfile_SBCRemoteUpdateSupport,
IpProfile_SBCRemoteReinviteSupport,
IpProfile_SBCRemoteDelayedOfferSupport, IpProfile_SBCRemoteReferBehavior,
IpProfile_SBCRemote3xxBehavior, IpProfile_SBCRemoteMultiple18xSupport,
IpProfile_SBCRemoteEarlyMediaResponseType,
IpProfile_SBCRemoteEarlyMediaSupport, IpProfile_EnableSymmetricMKI,
IpProfile_MKISize, IpProfile_SBCEnforceMKISize,
IpProfile_SBCRemoteEarlyMediaRTP, IpProfile_SBCRemoteSupportsRFC3960,
IpProfile_SBCRemoteCanPlayRingback, IpProfile_EnableEarly183,
IpProfile_EarlyAnswerTimeout, IpProfile_SBC2833DTMFPayloadType,
IpProfile_SBCUserRegistrationTime, IpProfile_ResetSRTPStateUponRekey,
IpProfile_AmdMode, IpProfile_SBCReliableHeldToneSource,
IpProfile_GenerateSRTPKeys, IpProfile_SBCPlayHeldTone,
IpProfile_SBCRemoteHoldFormat, IpProfile_SBCRemoteReplacesBehavior,
IpProfile_SBCSDPPtimeAnswer, IpProfile_SBCPreferredPTime,
IpProfile_SBCUseSilenceSupp, IpProfile_SBCRTPRedundancyBehavior,
IpProfile_SBCPlayRBTToTransferee, IpProfile_SBCRTCPMode,
IpProfile_SBCJitterCompensation,
IpProfile_SBCRemoteRenegotiateOnFaxDetection,
IpProfile_JitterBufMaxDelay,
IpProfile_SBCUserBehindUdpNATRegistrationTime,
IpProfile_SBCUserBehindTcpNATRegistrationTime,
IpProfile_SBCSDPHandleRTCPAttribute,
IpProfile_SBCRemoveCryptoLifetimeInSDP, IpProfile_SBCIceMode,
IpProfile_SBCRTCPMux, IpProfile_SBCMediaSecurityMethod,
IpProfile_SBCHandleXDetect, IpProfile_SBCRTCPFeedback,
IpProfile_SBCRemoteRepresentationMode, IpProfile_SBCKeepVIAHeaders,
IpProfile_SBCKeepRoutingHeaders, IpProfile_SBCKeepUserAgentHeader,
IpProfile_SBCRemoteMultipleEarlyDialogs,
IpProfile_SBCRemoteMultipleAnswersMode, IpProfile_SBCDirectMediaTag,
IpProfile_SBCAdaptRFC2833BWToVoiceCoderBW;
IpProfile 1 = "IP-PBX", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0,
0, -1, 1, 0, 0, -1, 0, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, "", -1, -1, 0, 2,
0, 0, 0, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 0, 2, 2, 1, 0, 0, 1, 0,
1, 0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0,
0, 0, 300, -1, -1, 0, 0, 0, 0, 0, 0, 0, -1, -1, -1, -1, -1, 0, "", 0;
IpProfile 2 = "BroadCloud", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0,
0, 0, -1, 1, 0, 0, -1, 0, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, "", -1, -1, 0,
2, 0, 0, 1, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 0, 2, 2, 1, 0, 0, 1,
0, 1, 0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 0, 0, 0,
0, 0, 0, 300, -1, -1, 0, 0, 0, 0, 0, 0, 0, -1, -1, -1, -1, -1, 0, "", 0;
```

```
[ \IpProfile ]


[ CpMediaRealm ]

FORMAT CpMediaRealm_Index = CpMediaRealm_MediaRealmName,
CpMediaRealm_IPv4IF, CpMediaRealm_IPv6IF, CpMediaRealm_PortRangeStart,
CpMediaRealm_MediaSessionLeg, CpMediaRealm_PortRangeEnd,
CpMediaRealm_IsDefault, CpMediaRealm_QoeProfile, CpMediaRealm_BWProfile;
CpMediaRealm 0 = "MRLan", "ShoreTel", "", 6000, 100, 6999, 0, "", "";
CpMediaRealm 1 = "MRWan", "DMZ", "", 7000, 100, 7999, 0, "", "";

[ \CpMediaRealm ]


[ SBCRoutingPolicy ]

FORMAT SBCRoutingPolicy_Index = SBCRoutingPolicy_Name,
SBCRoutingPolicy_LCREnable, SBCRoutingPolicy_LCRAverageCallLength,
SBCRoutingPolicy_LCRDefaultCost, SBCRoutingPolicy_LdapServerGroupName;
SBCRoutingPolicy 0 = "Default_SBCRoutingPolicy", 0, 1, 0, "";

[ \SBCRoutingPolicy ]


[ SRD ]

FORMAT SRD_Index = SRD_Name, SRD_BlockUnRegUsers, SRD_MaxNumOfRegUsers,
SRD_EnableUnAuthenticatedRegistrations, SRD_SharingPolicy,
SRD_UsedByRoutingServer, SRD_SBCOperationMode, SRD_SBCRoutingPolicyName,
SRD_SBCDialPlanName;
SRD 0 = "DefaultSRD", 0, -1, 1, 0, 0, 0, "Default_SBCRoutingPolicy", "";

[ \SRD ]


[ SIPInterface ]

FORMAT SIPInterface_Index = SIPInterface_InterfaceName,
SIPInterface_NetworkInterface, SIPInterface_ApplicationType,
SIPInterface_UDPPort, SIPInterface_TCPPort, SIPInterface_TLSPort,
SIPInterface_SRDName, SIPInterface_MessagePolicyName,
SIPInterface_TLSContext, SIPInterface_TLSMutualAuthentication,
SIPInterface_TCPKeepaliveEnable,
SIPInterface_ClassificationFailureResponseType,
SIPInterface_PreClassificationManSet, SIPInterface_EncapsulatingProtocol,
SIPInterface_MediaRealm, SIPInterface_SBCDirectMedia,
SIPInterface_BlockUnRegUsers, SIPInterface_MaxNumOfRegUsers,
SIPInterface_EnableUnAuthenticatedRegistrations,
SIPInterface_UsedByRoutingServer;
SIPInterface 0 = "IP-PBX", "ShoreTel", 2, 5060, 0, 0, "DefaultSRD", "",
"", -1, 0, 500, -1, 0, "MRLan", 0, -1, -1, -1, 0;
SIPInterface 1 = "BroadCloud", "DMZ", 2, 5060, 0, 0, "DefaultSRD", "",
"", -1, 0, 500, -1, 0, "MRWan", 0, -1, -1, -1, 0;

[ \SIPInterface ]


[ ProxySet ]
```

```
FORMAT ProxySet_Index = ProxySet_ProxyName,
ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap,
ProxySet_SRDName, ProxySet_ClassificationInput, ProxySet_TLSContextName,
ProxySet_ProxyRedundancyMode, ProxySet_DNSResolveMethod,
ProxySet_KeepAliveFailureResp, ProxySet_GWIPv4SIPInterfaceName,
ProxySet_SBCIPv4SIPInterfaceName, ProxySet_SASIPv4SIPInterfaceName,
ProxySet_GWIPv6SIPInterfaceName, ProxySet_SBCIPv6SIPInterfaceName,
ProxySet_SASIPv6SIPInterfaceName;
ProxySet 0 = "IP-PBX", 1, 60, 0, 0, "DefaultSRD", 0, "", -1, -1, "", "",
"IP-PBX", "", "", "", "";
ProxySet 1 = "BroadCloud", 1, 60, 0, 0, "DefaultSRD", 0, "", -1, 1, "",
"", "BroadCloud", "", "", "", "";

[ \ProxySet ]


[ IPGroup ]

FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Name, IPGroup_ProxySetName,
IPGroup_SIPGroupName, IPGroup_ContactUser, IPGroup_SipReRoutingMode,
IPGroup_AlwaysUseRouteTable, IPGroup_SRDName, IPGroup_MediaRealm,
IPGroup_ClassifyByProxySet, IPGroup_ProfileName,
IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet, IPGroup_OutboundManSet,
IPGroup_RegistrationMode, IPGroup_AuthenticationMode, IPGroup_MethodList,
IPGroup_EnableSBCClientForking, IPGroup_SourceUriInput,
IPGroup_DestUriInput, IPGroup_ContactName, IPGroup_Username,
IPGroup_Password, IPGroup_UUIFormat, IPGroup_QOEProfile,
IPGroup_BWProfile, IPGroup_MediaEnhancementProfile,
IPGroup_AlwaysUseSourceAddr, IPGroup_MsgManUserDef1,
IPGroup_MsgManUserDef2, IPGroup_SIPConnect, IPGroup_SBCPSAPMode,
IPGroup_DTLSContext, IPGroup_CreatedByRoutingServer,
IPGroup_UsedByRoutingServer, IPGroup_SBCOperationMode,
IPGroup_SBCRouteUsingRequestURIPort, IPGroup_SBCKeepOriginalCallID,
IPGroup_SBCDialPlanName;
IPGroup 0 = 0, "IP-PBX", "IP-PBX", "172.26.249.130", "", -1, 0,
"DefaultSRD", "MRLan", 1, "IP-PBX", -1, -1, -1, 0, 0, "", 0, -1, -1, "",
"", "$1$gQ==", 0, "", "", "", 0, "", "", 0, 0, "", 0, 0, -1, 0, 0, "";
IPGroup 1 = 0, "BroadCloud", "BroadCloud", "interop.adpt-tech.com", "", -
1, 0, "DefaultSRD", "MRWan", 1, "BroadCloud", -1, -1, 4, 0, 0, "", 0, -1,
-1, "", "", "$1$gQ==", 0, "", "", "", 0, "", "", 0, 0, "", 0, 0, -1, 0,
0, "";

[ \IPGroup ]


[ SBCAlternativeRoutingReasons ]

FORMAT SBCAlternativeRoutingReasons_Index =
SBCAlternativeRoutingReasons_ReleaseCause;
SBCAlternativeRoutingReasons 0 = 503;

[ \SBCAlternativeRoutingReasons ]


[ ProxyIp ]

FORMAT ProxyIp_Index = ProxyIp_ProxySetId, ProxyIp_ProxyIpIndex,
ProxyIp_IpAddress, ProxyIp_TransportType;
ProxyIp 0 = "0", 0, "172.26.249.130:5060", 0;
ProxyIp 1 = "1", 0, "nn6300southsipconnect.adpt-tech.com", 0;
```

```
[ \ProxyIp ]


[ Account ]

FORMAT Account_Index = Account_ServedTrunkGroup,
Account_ServedIPGroupName, Account_ServingIPGroupName, Account_Username,
Account_Password, Account_HostName, Account_Register,
Account_ContactUser, Account_ApplicationType;
Account 0 = -1, "IP-PBX", "BroadCloud", "8325624857",
"$1$SSg/LyUiDSA0NCFhZGRj", "interop.adpt-tech.com", 1, "8325624857", 2;

[ \Account ]


[ IP2IPRouting ]

FORMAT IP2IPRouting_Index = IP2IPRouting_RouteName,
IP2IPRouting_RoutingPolicyName, IP2IPRouting_SrcIPGroupName,
IP2IPRouting_SrcUsernamePrefix, IP2IPRouting_SrcHost,
IP2IPRouting_DestUsernamePrefix, IP2IPRouting_DestHost,
IP2IPRouting_RequestType, IP2IPRouting_MessageConditionName,
IP2IPRouting_ReRouteIPGroupName, IP2IPRouting_Trigger,
IP2IPRouting_CallSetupRulesSetId, IP2IPRouting_DestType,
IP2IPRouting_DestIPGroupName, IP2IPRouting_DestSIPInterfaceName,
IP2IPRouting_DestAddress, IP2IPRouting_DestPort,
IP2IPRouting_DestTransportType, IP2IPRouting_AltRouteOptions,
IP2IPRouting_GroupPolicy, IP2IPRouting_CostGroup, IP2IPRouting_DestTags,
IP2IPRouting_SrcTags;
IP2IPRouting 0 = "Terminate OPTIONS", "Default_SBCRoutingPolicy", "Any",
"*", "*", "*", "*", 6, "", "Any", 0, -1, 1, "", "", "internal", 0, -1, 0,
0, "", "", "";
IP2IPRouting 1 = "IP-PBX to ITSP", "Default_SBCRoutingPolicy", "IP-PBX",
"*", "*", "*", "*", 0, "", "Any", 0, -1, 0, "BroadCloud", "BroadCloud",
"", 0, -1, 0, 0, "", "", "";
IP2IPRouting 2 = "ITSP to IP-PBX", "Default_SBCRoutingPolicy",
"BroadCloud", "*", "*", "*", "*", 0, "", "Any", 0, -1, 0, "IP-PBX", "IP-
PBX", "", 0, -1, 0, 0, "", "", "";

[ \IP2IPRouting ]


[ IPOutboundManipulation ]

FORMAT IPOutboundManipulation_Index =
IPOutboundManipulation_ManipulationName,
IPOutboundManipulation_RoutingPolicyName,
IPOutboundManipulation_IsAdditionalManipulation,
IPOutboundManipulation_SrcIPGroupName,
IPOutboundManipulation_DestIPGroupName,
IPOutboundManipulation_SrcUsernamePrefix, IPOutboundManipulation_SrcHost,
IPOutboundManipulation_DestUsernamePrefix,
IPOutboundManipulation_DestHost,
IPOutboundManipulation_CallingNamePrefix,
IPOutboundManipulation_MessageConditionName,
IPOutboundManipulation_RequestType,
IPOutboundManipulation_ReRouteIPGroupName,
IPOutboundManipulation_Trigger, IPOutboundManipulation_ManipulatedURI,
IPOutboundManipulation_RemoveFromLeft,
IPOutboundManipulation_RemoveFromRight,
IPOutboundManipulation_LeaveFromRight, IPOutboundManipulation_Prefix2Add,
IPOutboundManipulation_Suffix2Add,
```

```
IPOutboundManipulation_PrivacyRestrictionMode,
IPOutboundManipulation_DestTags, IPOutboundManipulation_SrcTags;
IPOutboundManipulation 0 = "Clip 1", "Default_SBCRoutingPolicy", 0, "IP-
PBX", "BroadCloud", "*", "*", "[1732,1832]", "*", "*", "", 0, "Any", 0,
1, 1, 0, 255, "", "", 0, "", "";
IPOutboundManipulation 1 = "Clip +1 from source",
"Default_SBCRoutingPolicy", 0, "IP-PBX", "BroadCloud", "+", "*", "*",
"*", "*", "", 0, "Any", 0, 0, 2, 0, 255, "", "", 0, "", "";
IPOutboundManipulation 2 = "Call to desk", "Default_SBCRoutingPolicy", 0,
"IP-PBX", "BroadCloud", "*", "*", "1170", "*", "*", "", 0, "Any", 0, 1,
0, 0, 255, "1732652", "", 0, "", "";
IPOutboundManipulation 3 = "4852->118", "Default_SBCRoutingPolicy", 0,
"BroadCloud", "IP-PBX", "*", "*", "8325624852", "*", "*", "", 0, "Any",
0, 1, 10, 0, 255, "118", "", 0, "", "";
IPOutboundManipulation 4 = "4853->119", "Default_SBCRoutingPolicy", 0,
"BroadCloud", "IP-PBX", "*", "*", "8325624853", "*", "*", "", 0, "Any",
0, 1, 10, 0, 255, "119", "", 0, "", "";
IPOutboundManipulation 5 = "For Test 19", "Default_SBCRoutingPolicy", 0,
"BroadCloud", "IP-PBX", "*", "*", "*", "*", "*", "", 0, "Any", 0, 0, 0,
0, 255, "", "", 0, "", "";

[ \IPOutboundManipulation ]


[ CodersGroup0 ]

FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce,
CodersGroup0_CoderSpecific;
CodersGroup0 0 = "g711Ulaw64k", 20, 0, -1, 0, "";
CodersGroup0 1 = "g711Alaw64k", 20, 0, -1, 0, "";

[ \CodersGroup0 ]


[ CodersGroup1 ]

FORMAT CodersGroup1_Index = CodersGroup1_Name, CodersGroup1_pTime,
CodersGroup1_rate, CodersGroup1_PayloadType, CodersGroup1_Sce,
CodersGroup1_CoderSpecific;
CodersGroup1 0 = "g711Ulaw64k", 20, 0, -1, 0, "";
CodersGroup1 1 = "g711Alaw64k", 20, 0, -1, 0, "";

[ \CodersGroup1 ]


[ CodersGroup2 ]

FORMAT CodersGroup2_Index = CodersGroup2_Name, CodersGroup2_pTime,
CodersGroup2_rate, CodersGroup2_PayloadType, CodersGroup2_Sce,
CodersGroup2_CoderSpecific;
CodersGroup2 0 = "g729", 20, 0, -1, 0, "";

[ \CodersGroup2 ]


[ AllowedCodersGroup1 ]

FORMAT AllowedCodersGroup1_Index = AllowedCodersGroup1_Name;
```

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AllowedCodersGroup1 0 = "g711Ulaw64k";
AllowedCodersGroup1 1 = "g711Alaw64k";

[ \AllowedCodersGroup1 ]


[ AllowedCodersGroup2 ]

FORMAT AllowedCodersGroup2_Index = AllowedCodersGroup2_Name;
AllowedCodersGroup2 0 = "g729";
AllowedCodersGroup2 1 = "g711Alaw64k";

[ \AllowedCodersGroup2 ]


[ MessageManipulations ]

FORMAT MessageManipulations_Index =
MessageManipulations_ManipulationName, MessageManipulations_ManSetID,
MessageManipulations_MessageType, MessageManipulations_Condition,
MessageManipulations_ActionSubject, MessageManipulations_ActionType,
MessageManipulations_ActionValue, MessageManipulations_RowRole;
MessageManipulations 0 = "Change From host", 4, "any.request", "",
"header.from.url.host", 2, "header.to.url.host", 0;
MessageManipulations 1 = "Change P-Asserted host", 4, "any.request",
"header.p-asserted-identity exists", "header.p-asserted-
identity.url.host", 2, "header.to.url.host", 0;
MessageManipulations 2 = "Diversion", 4, "invite.request",
"header.diversion regex (<sip:)(..)(.*)(@)(.*)", "header.from.url.user",
2, "$3", 0;

[ \MessageManipulations ]


[ GwRoutingPolicy ]

FORMAT GwRoutingPolicy_Index = GwRoutingPolicy_Name,
GwRoutingPolicy_LCREnable, GwRoutingPolicy_LCRAverageCallLength,
GwRoutingPolicy_LCRDefaultCost, GwRoutingPolicy_LdapServerGroupName;
GwRoutingPolicy 0 = "GwRoutingPolicy", 0, 1, 0, "";

[ \GwRoutingPolicy ]


[ ResourcePriorityNetworkDomains ]

FORMAT ResourcePriorityNetworkDomains_Index =
ResourcePriorityNetworkDomains_Name,
ResourcePriorityNetworkDomains_Ip2TelInterworking;
ResourcePriorityNetworkDomains 1 = "dsn", 1;
ResourcePriorityNetworkDomains 2 = "dod", 1;
ResourcePriorityNetworkDomains 3 = "drsn", 1;
ResourcePriorityNetworkDomains 5 = "uc", 1;
ResourcePriorityNetworkDomains 7 = "cuc", 1;

[ \ResourcePriorityNetworkDomains ]
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Document #: LTRT-12560