

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

Version 7.0





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## Notice

This document describes how to connect the IP-PBX and BroadCloud SIP Trunk using AudioCodes Mediant E-SBC product series.

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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.

## Documentation Feedback

AudioCodes continually strives to produce high quality documentation. If you have any comments (suggestions or errors) regarding this document, please fill out the Documentation Feedback form on our Web site at <http://www.audiocodes.com/downloads>.

# 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**

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

### 2.2 AudioCodes E-SBC Version

**Table 2-2: AudioCodes E-SBC Version**

<b>SBC Vendor</b>	AudioCodes
<b>Models</b>	<ul style="list-style-type: none"> <li>▪ Mediant 500 E-SBC</li> <li>▪ Mediant 800 Gateway &amp; E-SBC</li> <li>▪ Mediant 1000B Gateway &amp; E-SBC</li> <li>▪ Mediant 3000 Gateway &amp; E-SBC</li> <li>▪ Mediant 2600 E-SBC</li> <li>▪ Mediant 4000 E-SBC</li> </ul>
<b>Software Version</b>	SIP_F7.00A.049.003
<b>Protocol</b>	SIP/UDP (to the both BroadCloud SIP Trunk and IP-PBX)
<b>Additional Notes</b>	None

### 2.3 BroadCloud SIP Trunking Version

**Table 2-3: BroadCloud Version**

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

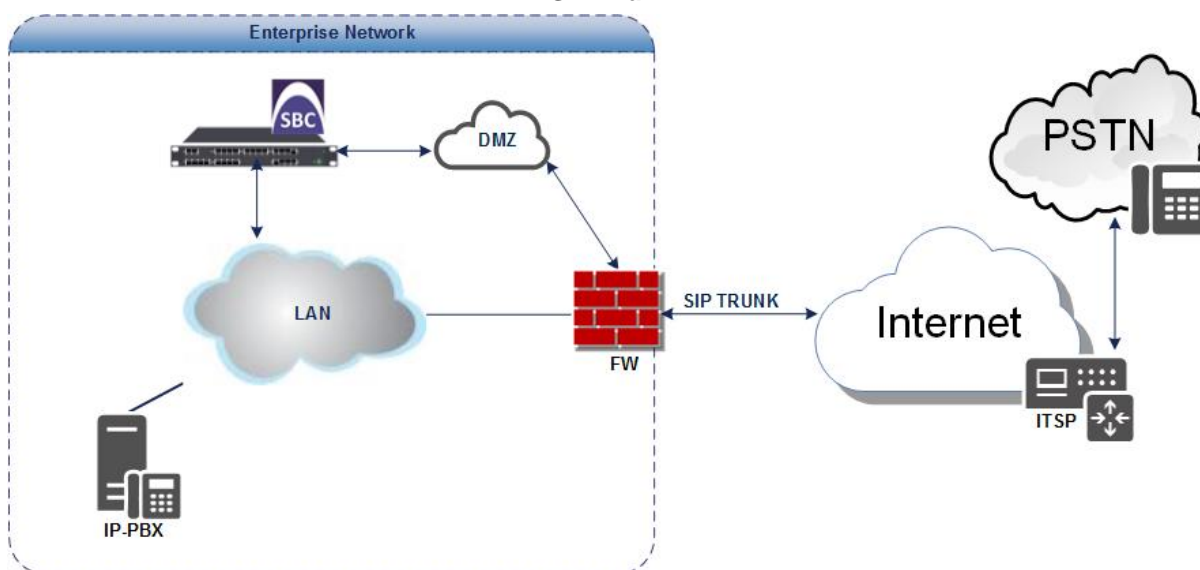
## 2.4 Interoperability Test Topology

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
<b>Network</b>	<ul style="list-style-type: none"> <li>▪ IP-PBX is located on the Enterprise's LAN</li> <li>▪ BroadCloud SIP Trunk is located on the WAN</li> </ul>
<b>Signaling Transcoding</b>	<ul style="list-style-type: none"> <li>▪ IP-PBX operates with SIP-over-UDP transport type</li> <li>▪ BroadCloud SIP Trunk operates with SIP-over-UDP transport type</li> </ul>
<b>Codecs Transcoding</b>	<ul style="list-style-type: none"> <li>▪ IP-PBX supports G.711A-law, G.711U-law, and G.729 coder</li> <li>▪ BroadCloud SIP Trunk supports G.711A-law, G.711U-law, and G.729 coder</li> </ul>
<b>Media Transcoding</b>	<ul style="list-style-type: none"> <li>▪ IP-PBX operates with RTP media type</li> <li>▪ BroadCloud SIP Trunk operates with RTP media type</li> </ul>

## 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

The screenshot displays the 'Call Control Options' configuration page in the ShoreTel Director web interface. The page is divided into several sections:

- General:**
  - Use Distributed Routing Service for call routing.
  - Enable Monitor / Record Warning Tone
  - IP Phone Silent Coach Warning Tone
  - Generate an event when a trunk is in use for 30 minutes.
  - Park Timeout (1-100000) after 60 seconds.
  - Hang up Make Me Conference after 20 minutes of silence.
  - Delay before sending DTMF to Fax Server: 200 msec
  - DTMF Payload Type (80 - 127): 60
- SIP:**
  - Realm: ShoreTel
  - Enable SIP Session Timer:
    - Session Interval (90 - 3000): 1000 sec
    - Refresher: Caller
- Voice Encoding and Quality of Service:**
  - Maximum Inter-Site Jitter Buffer (20 - 400): 30 msec
  - DiffServ / ToS Byte (0-255): 64 (DSCP = 0x16)
- Media Encryption:**
  - Admission control algorithm assumes RTP header compression is being used.
- Call Control Quality of Service:**
  - DiffServ / ToS Byte (0-255): 64 (DSCP = 0x16)
- Video Quality of Service:**
  - DiffServ / ToS Byte (0-255): 64 (DSCP = 0x22)
- Trunk to Trunk Transfer and Tandem Trunks:**
  - Hang up after 30 minutes of silence.
  - Hang up after 30 minutes.

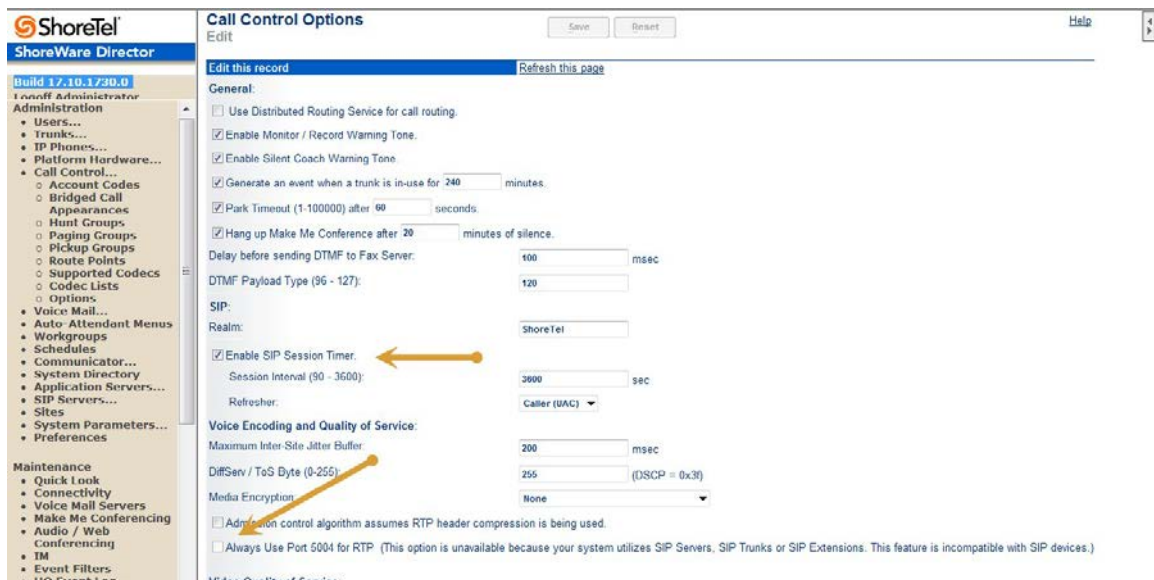
Within the Call Control Options SIP parameters, confirm that the appropriate settings are made for the Realm, Enable SIP Session Timer and Always Use Port 5004 for RTP parameters.

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**

The screenshot shows the ShoreTel Director interface for editing a site. The left sidebar contains a navigation tree with categories: Administration, Maintenance, Reporting, and Documentation. The 'Reporting' section is highlighted with three arrows pointing to the 'Admission Control Bandwidth', 'Intra-Site Calls', and 'Inter-Site Calls' fields in the main configuration area. The main area includes fields for Name (Headquarters), Country (United States of), Language (English), Parent (Top of Tree), Local Area Code (732), Additional Local Area Codes (Edit), Caller's Emergency Service Identification (CESID), Time Zone (UTC-05:00 Eastern Time), Night Bell Extension, Night Bell Switch (None), Paging Extension, Paging Switch (None), Operator Extension, FAX Redirected Extension, SMTP Relay, Network Time Protocol Server (172.26.243), Bandwidth (2046 kbps), Admission Control Bandwidth (2046 kbps), Intra-Site Calls (Very High Bandwidth Codecs), Inter-Site Calls (Very Low Bandwidth Codecs), FAX and Modem Calls (Fax Codecs - High Bandwidth), SIP Proxy, Virtual IP Address, Proxy Switch 1 (pbxlab4), Proxy Switch 2 (None), and Emergency Number List. Buttons for New, Copy, Save, and Delete are at the top right.

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**

The screenshot shows the ShoreTel Director interface. On the left is a navigation menu with 'Administration' expanded to 'Voice Switches / Service Appliances' > 'Primary'. The main content area is titled 'Primary Voice Switches / Service Appliances' and contains a table of switches. Above the table is a form to 'Add new switch/appliance at site: Headquarters of type: ShoreGear 30'. The table has columns for Name, Quick Launch, Description, Site, Server, Database Server, Type, IP Address, MAC Address, Serial Number, IP Phones In Use, and IP Phones Capacity. A 'Total' row at the bottom shows 13 IP Phones In Use and a Capacity of 30.

Name	Quick Launch	Description	Site	Server	Database Server	Type	IP Address	MAC Address	Serial Number	IP Phones In Use	IP Phones Capacity
<a href="#">pbxlab40/8</a>		pbxlab40/8	Headquarters	Headquarters		40/8	172.26.249.4	00-10-49-08-0D-F7	08JC08070B0DF7	13	20
<a href="#">sg00</a>		sg30	Headquarters	Headquarters		SG-30	172.26.249.130	00-10-49-13-48-B8	S30J09321348B8	0	10
<a href="#">shoretelcc1</a>		shoretelcc1	Headquarters	shoretelcc1	Headquarters	SW	172.26.249.6			0	0
<a href="#">shoretelremote1</a>		shoretelremote1	Headquarters	shoretelremote1	Headquarters	SW	172.26.249.7			0	0
<a href="#">shoretelremote2</a>		shoretelremote2	Headquarters	shoretelremote2	Headquarters	SW	172.26.249.8			0	0
<a href="#">SoftSwitch</a>		SoftSwitch	Headquarters	Headquarters	Headquarters	SW	172.26.249.3			0	0
<b>Total</b>										13	30

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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

**Voice Switches**  
Edit ShoreGear 30 Switch

Buttons: [New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#)

Navigation: [Edit this record](#) [Refresh this page](#)

Fields:

- Name:
- Description:
- Site:
- IP Address:  [Find Switches](#)
- Ethernet Address:
- Server to Manage Switch:
- Caller's Emergency Service Identification (CESID):  (e.g. +1 (408) 331-3300)
- Built-in Capacity: IP Phone + SIP Trunk = Total  
 +  = 10 of 10 (0 SIP proxy ports)
- Enable Jack Based Music On Hold  
 Jack Based Music On Hold Gain (-49 to 13):  dB
- Use Analog Extension Ports as DID Trunks

Image: ShoreTel ShoreGear 30 switch and a close-up of its ports.

Port	Port Type	Trunk Group	Description	Jack Number
1	5 SIP Trunks		P01	
2	5 SIP Trunks		P02	
11	Available		P03	
12	Available		P04	

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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.

- 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.
- 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**

The screenshot shows the ShoreTel Director interface. On the left is a navigation menu with 'Administration' expanded and 'Trunk Groups' selected. The main area is titled 'Trunk Groups' and contains a form to 'Add new trunk group at site: Headquarte' and 'of type: SIP'. A 'Go' link is highlighted with an arrow. Below the form is a table of existing trunk groups.

Name	Type	Site	Trunks	DID	Destination	Access Code
<a href="#">Analog Loop Start</a>	Analog Loop Start	Headquarters	2	No	700	9
<a href="#">Digital Loop Start</a>	Digital Loop Start	Headquarters	0	No	700	9
<a href="#">Digital Vtrk Start</a>	Digital Vtrk Start	Headquarters	0	No	700	9
<a href="#">SIP Vtrk</a>	SIP	Headquarters	5	Yes	700	80
<a href="#">SIP PSTN</a>	SIP	Headquarters	5	Yes	700	81

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### ➤ 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)

**ShoreTel**  
Director

Build 19.45.0701.0  
Logoff Eugene Boring

**Administration**

- Users...
- Trunks...
  - Individual Trunks
  - Trunk Groups
  - SIP Profiles
  - ISDN Profiles
  - Local Prefixes
- IP Phones...
- Platform Hardware...
- Call Control...
- Voice Mail...
- Auto-Attendant Menus
- Workgroups
- Schedules
- Communicator...
- System Directory
- Application Servers...
- SIP Servers...
- Sites
- System Parameters...
- Preferences
- Connect Services

**Maintenance**

- Diagnostics & Monitoring
- Quick Look
- Connectivity
- Voice Mail Servers
- Make Me Conferencing
- Audio / Web Conferencing
- IM
- Event Filters
- HQ Event Log...
- HQ Services

**Reporting**

- Reports...
- Options

**Documentation**

- Administration Guide

**Trunk Groups**  
Edit SIP Trunk Group

**Edit this record** Refresh this page

Name: <input type="text" value="sip.pstn"/> Site: Headquarters Language: English <input type="checkbox"/> Enable SIP Info for G.711 DTMF Signaling Profile: Default ITSP Digest Authentication: <None> Username: <input type="text"/> Password: <input type="text"/> Inbound: Number of Digits from CO: 3 <input checked="" type="checkbox"/> DNS <input type="button" value="Edit DNS Map"/> <input checked="" type="checkbox"/> DID <input type="button" value="Edit DID Range"/> <input checked="" type="checkbox"/> Extension <input checked="" type="radio"/> Translation Table: <None> <input type="radio"/> Prepend Dial In Prefix: <input type="text"/> <input type="radio"/> Use Site Extension Prefix <input checked="" type="checkbox"/> Tandem Trunking User Group: Executives Prepend Dial In Prefix: 80 Destination: 700 : Default <input type="button" value="Search"/> <input checked="" type="checkbox"/> Outbound: Network Call Routing: Access Code: 81 Local Area Code: 732 Additional Local Area Codes: <input type="button" value="Edit"/> Nearby Area Codes: <input type="button" value="Edit"/> Billing Telephone Number: <input type="text" value="(E.g. +1 (408)331-3300"/>	
---	--

**Trunk Services:**

- Local
- Long Distance
- International
- Enable Original Caller Information
- n11 (e.g. 411, 611, except 911 which is specified below)
- Emergency (e.g. 911)
- Easily Recognizable Codes (ERC) (e.g. 800, 888, 900)
- Explicit Carrier Selection (e.g. 1010xxx)
- Operator Assisted (e.g. 0+)
- Caller ID not blocked by default
- Enable Caller ID ( Please confirm with the Carrier(s) or the Service Provider(s) on how the end-to-end caller name is delivered)  
 When Site Name is used for the Caller ID, overwrite it with:

**Trunk Digit Manipulation:**

- Remove leading 1 from 1+10D  
*Hint: Required for some long distance service providers.*
- Remove leading 1 for Local Area Codes (for all prefixes unless a specific local prefix list is provided below)  
*Hint: Required for some local service providers with overlay area codes.*
- Dial 7 digits for Local Area Code (for all prefixes unless a specific local prefix list is provided below)  
*Hint: Local prefixes required for some local service providers with mixed 7D and 1+10D in the same home area.*
- Dial in E.164 Format

Local Prefixes:

Prepend Dial Out Prefix:

Off System Extensions:

Translation Table: <None>

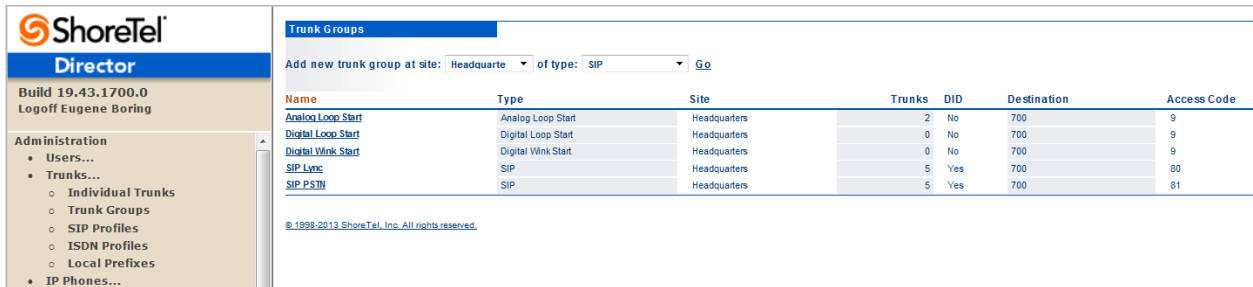
### 3.7 ShoreTel System Settings – Individual Trunks

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**

**Inbound:**

Number of Digits from CO:

DNIS

DID

Extension

Translation Table:

Prepend Dial In Prefix:

Use Site Extension Prefix

Tandem Trunking

User Group:

Prepend Dial In Prefix:

Destination:

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

**Outbound:**

**Network Call Routing:**

Access Code:

Local Area Code:

Additional Local Area Codes:

Nearby Area Codes:

Billing Telephone Number:  (e.g. +1 (408) 331-3300 )

**Trunk Services:**

Local

Long Distance

International

Enable Original Caller Information

n11 (e.g. 411, 611, except 911 which is specified below)

Emergency (e.g. 911)

Easily Recognizable Codes (ERC) (e.g. 800, 888, 900)

Explicit Carrier Selection (e.g. 1010xxx)

Operator Assisted (e.g. 0+)

Caller ID not blocked by default

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**

**Trunk Digit Manipulation:**

Remove leading 1 from 1+10D  
*Hint: Required for some long distance service providers.*

Remove leading 1 for Local Area Codes (for all prefixes unless a specific local prefix list is provided below)  
*Hint: Required for some local service providers with overlay area codes.*

Dial 7 digits for Local Area Code (for all prefixes unless a specific local prefix list is provided below)  
*Hint: Local prefixes required for some local service providers with mixed 7D and 1+10D in the same home area.*

Dial in E.164 Format

Local Prefixes: Non ▼ [Go to Local Prefixes List](#)

Prepend Dial Out Prefix:

Off System Extensions: Edit

Translation Table: <None ▼

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**

Explicit Carrier Selection (e.g. 1010xxx)

Operator Assisted (e.g. 0+)

Caller ID not blocked by default

Enable Caller ID ( Please confirm with the Carrier(s) or the Service Provider(s) on how the end-to-end caller name is delivered)  
 When Site Name is used for the Caller ID, overwrite it with:

**Trunk Digit Manipulation:**

Remove leading 1 from 1+10D  
*Hint: Required for some long distance service providers.*

Remove leading 1 for Local Area Codes (for all prefixes unless a specific local prefix list is provided below)  
*Hint: Required for some local service providers with overlay area codes.*

Dial 7 digits for Local Area Code (for all prefixes unless a specific local prefix list is provided below)  
*Hint: Local prefixes required for some local service providers with mixed 7D and 1+10D in the same home area.*

Dial in E.164 Format

Local Prefixes: Non ▼ [Go to Local Prefixes List](#)

Prepend Dial Out Prefix:

Off System Extensions: Edit

Translation Table: <None ▼

**Off System Extension Ranges -- Webpage Dialog**

Range:

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*).

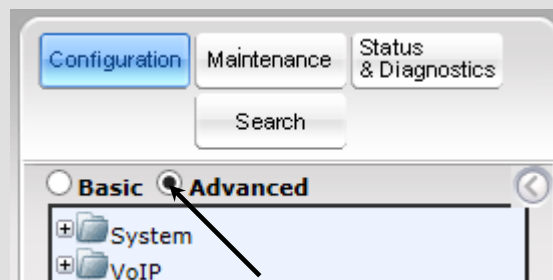
### Notes:

- 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:



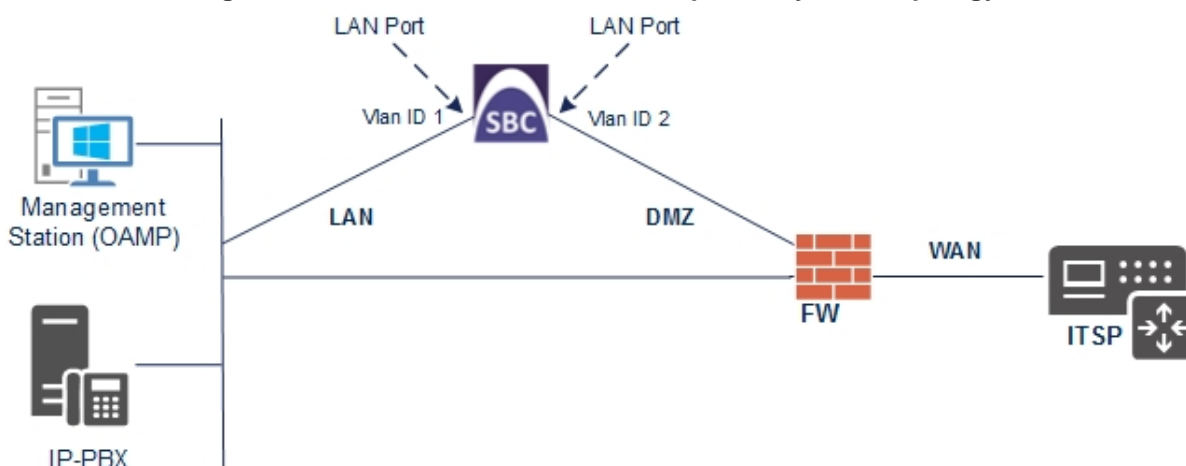
Note that when the E-SBC is reset, the Navigation tree reverts to Basic-menu display.

## 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:

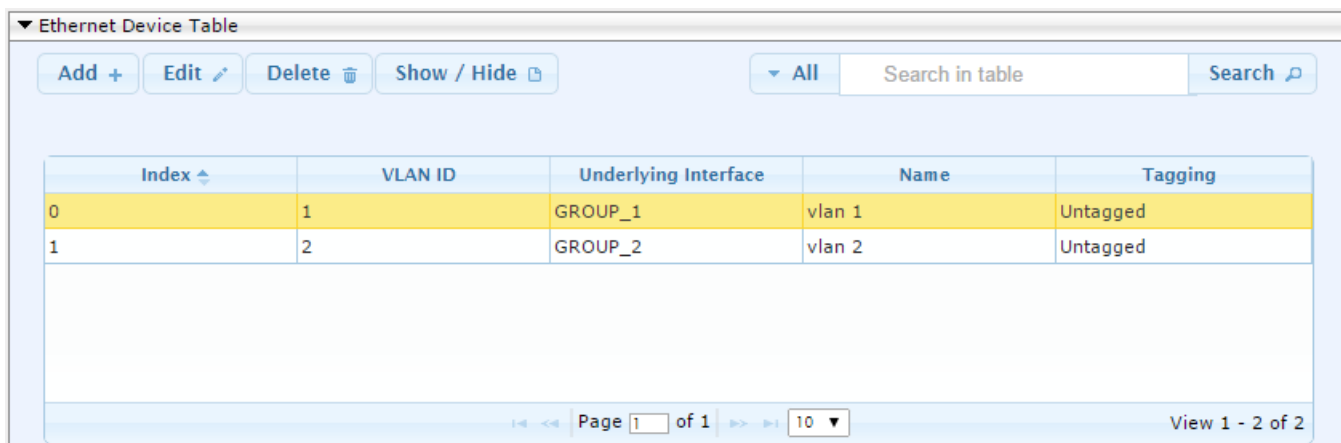
- LAN VoIP (assigned the name "Voice")
- WAN VoIP (assigned the name "WANSP")

➤ **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



### 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	<b>172.26.100.169</b> (IP address of E-SBC)
Prefix Length	<b>24</b> (subnet mask in bits for 255.255.255.0)
Default Gateway	<b>172.26.100.001</b>
VLAN ID	<b>1</b>
Interface Name	<b>Voice</b> (arbitrary descriptive name)
Underlying Device	<b>vlan 1</b>

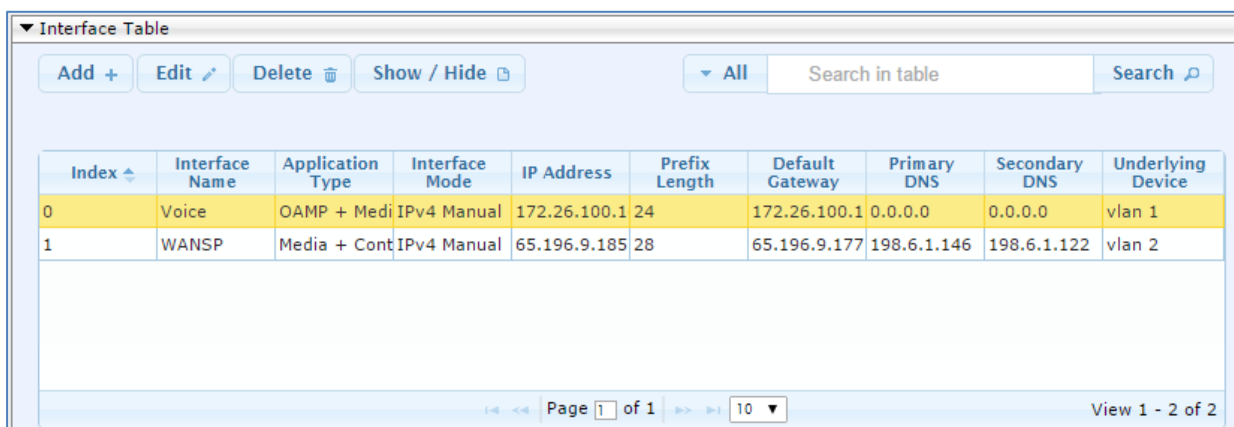
3. Add a network interface for the WAN side:
  - a. Enter **1**, and then click **Add Index**.
  - b. Configure the interface as follows:

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

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

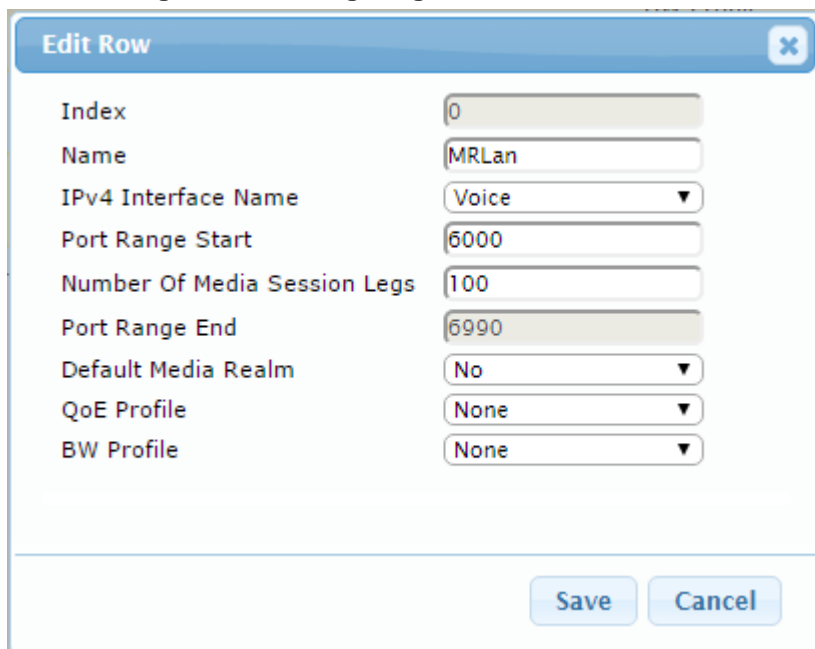
This step describes how to configure Media Realms. The simplest configuration is to create two Media Realms - one for internal (LAN) traffic and one for external (WAN) traffic.

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

**Figure 4-5: Configuring Media Realm for LAN**



Edit Row
✕

Index	<input type="text" value="0"/>
Name	<input type="text" value="MRLan"/>
IPv4 Interface Name	<input type="text" value="Voice"/>
Port Range Start	<input type="text" value="6000"/>
Number Of Media Session Legs	<input type="text" value="100"/>
Port Range End	<input type="text" value="6990"/>
Default Media Realm	<input type="text" value="No"/>
QoE Profile	<input type="text" value="None"/>
BW Profile	<input type="text" value="None"/>

- 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

Index	Name	IPv4 Interface Name	Port Range Start	Number Of Media Session Legs	Port Range End	Default Media Realm
0	MRLan	Voice	6000	100	6990	No
1	MRWan	WANSP	7000	100	7990	No

## 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	<b>0</b>
Interface Name	<b>IP-PBX</b> (see Note below)
Network Interface	<b>Voice</b>
Application Type	<b>SBC</b>
UDP Port	<b>5060</b>
TCP and TLS	<b>0</b>
Media Realm	<b>MRLan</b>

3. Configure a SIP Interface for the WAN:

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



The configured SIP Interfaces are shown in the figure below:

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

Index	Name	SRD	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	Encapsulating Protocol	Media Realm
0	IP-PBX	DefaultSRD	Voice	SBC	5060	0	0	No encapsulation	MRLan
1	BroadCloud	DefaultSRD	WANSP	SBC	5060	0	0	No encapsulation	MRWan



**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**

Edit Row
✕

Index	<input type="text" value="0"/>
SRD	<input style="border: 1px solid #ccc;" type="text" value="DefaultSRD"/>
Name	<input type="text" value="IP-PBX"/>
Gateway IPv4 SIP Interface	<input style="border: 1px solid #ccc;" type="text" value="None"/>
SBC IPv4 SIP Interface	<input style="border: 1px solid #ccc;" type="text" value="IP-PBX"/>
Proxy Keep-Alive	<input style="border: 1px solid #ccc;" type="text" value="Using OPTIONS"/>
Proxy Keep-Alive Time [sec]	<input type="text" value="60"/>
Redundancy Mode	<input style="border: 1px solid #ccc;" type="text"/>
Proxy Load Balancing Method	<input style="border: 1px solid #ccc;" type="text" value="Disable"/>
DNS Resolve Method	<input style="border: 1px solid #ccc;" type="text"/>
Proxy Hot Swap	<input style="border: 1px solid #ccc;" type="text" value="Disable"/>
Keep-Alive Failure Responses	<input style="border: 1px solid #ccc;" type="text"/>
Classification Input	<input style="border: 1px solid #ccc;" type="text" value="IP Address only"/>
TLS Context Name	<input style="border: 1px solid #ccc;" type="text" value="None"/>

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	<b>0</b>
Proxy Address	<b>172.26.249.130:5060</b> (IP-PBX IP address / FQDN and destination port)
Transport Type	<b>UDP</b>

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

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

Parameter	Value
Proxy Set ID	<b>1</b>
Proxy Name	<b>BroadCloud</b>
SBC IPv4 SIP Interface	<b>BroadCloud</b>
Proxy Keep Alive	<b>Using Options</b>

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

Index	1
SRD	DefaultSRD
Name	BroadCloud
Gateway IPv4 SIP Interface	None
SBC IPv4 SIP Interface	BroadCloud
Proxy Keep-Alive	Using OPTIONS
Proxy Keep-Alive Time [sec]	60
Redundancy Mode	
Proxy Load Balancing Method	Disable
DNS Resolve Method	SRV
Proxy Hot Swap	Disable
Keep-Alive Failure Responses	
Classification Input	IP Address only
TLS Context Name	None

Save Cancel

- 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

Index	0
Proxy Address	nn6300southsipconnec
Transport Type	UDP

Save Cancel

The configured Proxy Sets are shown in the figure below:

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

The screenshot shows a web interface titled "Proxy Sets Table". At the top, there are buttons for "Add +", "Edit", "Delete", and "Show / Hide". To the right, there is a search bar with a dropdown menu set to "All" and a "Search" button. Below these elements is a table with the following data:

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

At the bottom of the interface, there is a pagination control showing "Page 1 of 1" and a dropdown menu set to "10". On the far right, it says "View 1 - 2 of 2".

## 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:

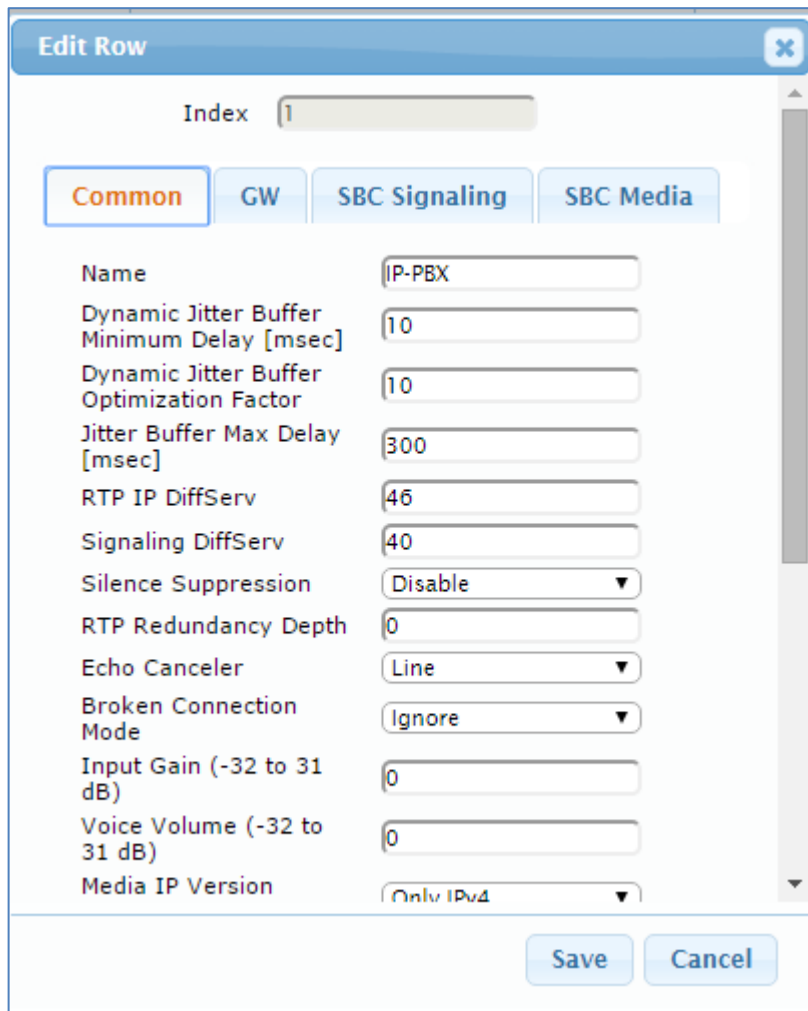
- IP-PBX - to operate in non-secure mode using RTP and UDP
- BroadCloud SIP trunk - to operate in non-secure mode using RTP and UDP

➤ **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**



✕
Edit Row

Index

Common
GW
SBC Signaling
SBC Media

Name	<input style="width: 90%;" type="text" value="IP-PBX"/>
Dynamic Jitter Buffer Minimum Delay [msec]	<input style="width: 90%;" type="text" value="10"/>
Dynamic Jitter Buffer Optimization Factor	<input style="width: 90%;" type="text" value="10"/>
Jitter Buffer Max Delay [msec]	<input style="width: 90%;" type="text" value="300"/>
RTP IP DiffServ	<input style="width: 90%;" type="text" value="46"/>
Signaling DiffServ	<input style="width: 90%;" type="text" value="40"/>
Silence Suppression	<input style="width: 90%;" type="text" value="Disable"/>
RTP Redundancy Depth	<input style="width: 90%;" type="text" value="0"/>
Echo Canceler	<input style="width: 90%;" type="text" value="Line"/>
Broken Connection Mode	<input style="width: 90%;" type="text" value="Ignore"/>
Input Gain (-32 to 31 dB)	<input style="width: 90%;" type="text" value="0"/>
Voice Volume (-32 to 31 dB)	<input style="width: 90%;" type="text" value="0"/>
Media IP Version	<input style="width: 90%;" type="text" value="Only IPv4"/>

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

Parameter	Value
Remote Update Support	<b>Supported</b>
Remote re-INVITE Support	<b>Supported</b>

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

The screenshot shows a configuration window titled "Edit Row" with a close button (X) in the top right corner. Below the title bar, there is an "Index" field containing the value "1". There are four tabs: "Common", "GW", "SBC Signaling" (which is selected and highlighted in orange), and "SBC Media". The "SBC Signaling" tab contains the following parameters and their values:

- PRACK Mode: Transparent
- P-Asserted-Identity Header Mode: As Is
- Diversion Header Mode: As Is
- History-Info Header Mode: As Is
- Session Expires Mode: Transparent
- Remote Update Support: Supported
- Remote re-INVITE: Supported
- Remote Delayed Offer Support: Supported
- User Registration Time: 0
- NAT UDP Registration Time: -1
- NAT TCP Registration Time: -1
- Remote REFER Mode: Regular
- Remote Replaces Mode: Standard

At the bottom of the window, there are "Save" and "Cancel" buttons.

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**

The screenshot shows the 'Edit Row' configuration window for the SBC Media tab. The window has a title bar with 'Edit Row' and a close button. Below the title bar is an 'Index' field with the value '1'. There are four tabs: 'Common', 'GW', 'SBC Signaling', and 'SBC Media', with 'SBC Media' being the active tab. The configuration parameters and their values are as follows:

Transcoding Mode	Only If Required
Extension Coders	None
Allowed Audio Coders	None
Allowed Coders Mode	Restriction
Allowed Video Coders	None
Allowed Media Types	
SBC Media Security Mode	RTP
Media Security Method	SDES
Enforce MKI Size	Enforce
SDP Remove Crypto Lifetime	No
RFC 2833 Mode	As Is
Alternative DTMF Method	As Is
RFC 2833 DTMF Payload Type	0
Fax Coders	None

At the bottom right of the window are 'Save' and 'Cancel' buttons.



➤ **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**

The screenshot shows a configuration window titled "Edit Row" with a close button (X) in the top right corner. Below the title bar, there is a field for "Index" containing the value "2". Below this, there are four tabs: "Common" (selected), "GW", "SBC Signaling", and "SBC Media". The "Common" tab contains the following parameters and values:

- Name: BroadCloud
- Dynamic Jitter Buffer Minimum Delay [msec]: 10
- Dynamic Jitter Buffer Optimization Factor: 10
- Jitter Buffer Max Delay [msec]: 300
- RTP IP DiffServ: 46
- Signaling DiffServ: 40
- Silence Suppression: Disable
- RTP Redundancy Depth: 0
- Echo Canceler: Line
- Broken Connection Mode: Ignore
- Input Gain (-32 to 31 dB): 0
- Voice Volume (-32 to 31 dB): 0
- Media IP Version: Only IPv4

At the bottom right of the window, there are "Save" and "Cancel" buttons.

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

Parameter	Value
P-Asserted-Identity Header Mode	<b>Add</b> (required for anonymous calls)

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

The screenshot shows the 'Edit Row' configuration window for the SBC Signaling tab. The window title is 'Edit Row' and it has a close button (X) in the top right corner. Below the title bar, there is an 'Index' field with the value '2'. There are four tabs: 'Common', 'GW', 'SBC Signaling' (which is selected and highlighted in orange), and 'SBC Media'. The 'SBC Signaling' tab contains the following parameters and values:

- PRACK Mode: Transparent
- P-Asserted-Identity Header Mode: Add
- Diversion Header Mode: As Is
- History-Info Header Mode: As Is
- Session Expires Mode: Transparent
- Remote Update Support: Supported
- Remote re-INVITE: Supported
- Remote Delayed Offer Support: Supported
- User Registration Time: 0
- NAT UDP Registration Time: -1
- NAT TCP Registration Time: -1
- Remote REFER Mode: Regular
- Remote Replaces Mode: Standard

At the bottom of the window, there are 'Save' and 'Cancel' buttons.

- 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**

The screenshot shows a configuration window titled "Edit Row" with a close button (X) in the top right corner. Below the title bar, there is an "Index" field containing the number "2". There are four tabs: "Common", "GW", "SBC Signaling", and "SBC Media", with "SBC Media" being the active tab. The configuration parameters and their values are as follows:

- Transcoding Mode: Only If Required
- Extension Coders: None
- Allowed Audio Coders: None
- Allowed Coders Mode: Restriction
- Allowed Video Coders: None
- Allowed Media Types: (empty text box)
- SBC Media Security Mode: RTP
- Media Security Method: SDES
- Enforce MKI Size: Don't enforce
- SDP Remove Crypto Lifetime: No
- RFC 2833 Mode: As Is
- Alternative DTMF Method: As Is
- RFC 2833 DTMF Payload Type: 0
- Fax Coders: None

At the bottom right of the window, there are "Save" and "Cancel" buttons.

## 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	<b>0</b>
Name	<b>IP-PBX</b>
Type	<b>Server</b>
Proxy Set	<b>IP-PBX</b>
IP Profile	<b>IP-PBX</b>
Media Realm	<b>MRLan</b>
SIP Group Name	<b>172.26.249.130</b> (according to IP-PBX requirement)

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

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

The configured IP Groups are shown in the figure below:

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

The screenshot shows a web interface titled "IP Group Table". At the top, there are buttons for "Add +", "Edit", "Delete", and "Show / Hide". To the right, there is a search bar with a dropdown menu set to "All" and a "Search" button. Below this is a table with the following data:

Index	Name	SRD	Type	SBC Operation Mode	Proxy Set	IP Profile	Media Realm	SIP Group Name	Classify By Proxy Set	Inbound Message Manipulation Set	Outbound Message Manipulation Set
0	IP-PBX	<input type="checkbox"/> DefaultSRV Server		Not Configured	IP-PBX	IP-PBX	MRLan	172.26.249.130	Enable	-1	4
1	BroadCloud	<input checked="" type="checkbox"/> DefaultSRV Server		Not Configured	BroadCloud	BroadCloud	MRWan	interop.adpt-tech.c	Enable	-1	4

At the bottom of the table, there is a pagination control showing "Page 1 of 1" and a dropdown menu set to "10". On the far right, it says "View 1 - 2 of 2".

## 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	<b>0</b>
Name	<b>Terminate OPTIONS</b> (arbitrary descriptive name)
Source IP Group	<b>Any</b>
Request Type	<b>OPTIONS</b>

**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	<b>Dest Address</b>
Destination Address	<b>internal</b>

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:

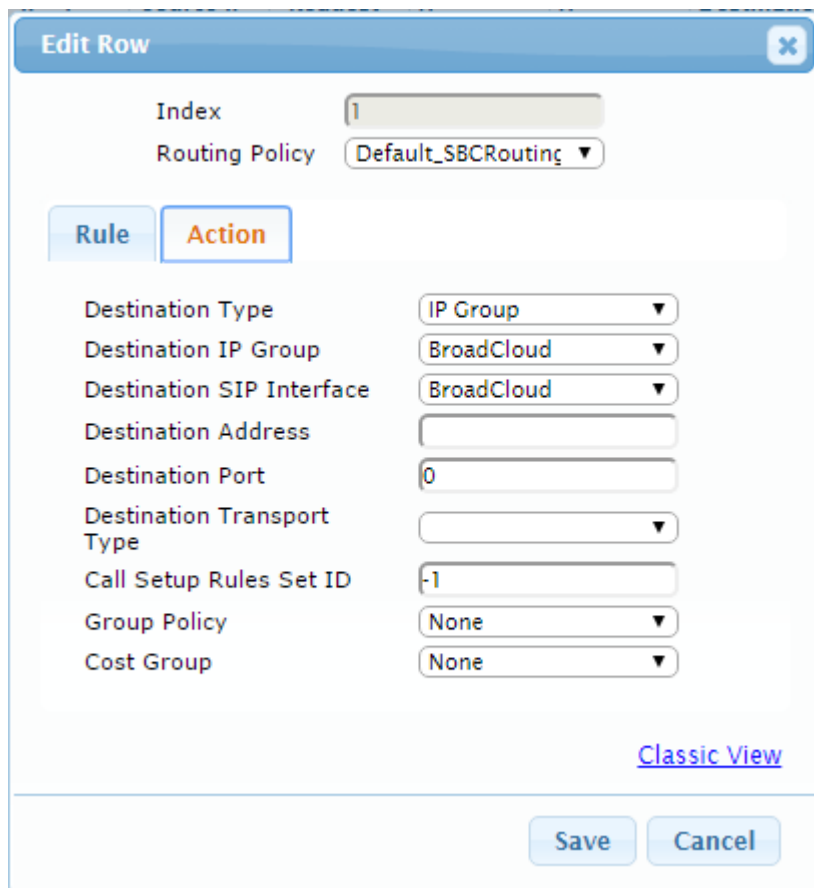
Parameter	Value
Index	1
Route Name	IP-PBX to ITSP (arbitrary descriptive name)
Source IP Group	IP-PBX



**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	<b>IP Group</b>
Destination IP Group	<b>BroadCloud</b>
Destination SIP Interface	<b>BroadCloud</b>

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


**Edit Row** [X]

Index:

Routing Policy:

**Rule** | **Action**

Destination Type:

Destination IP Group:

Destination SIP Interface:

Destination Address:

Destination Port:

Destination Transport Type:

Call Setup Rules Set ID:

Group Policy:

Cost Group:

[Classic View](#)

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	<b>2</b>
Route Name	<b>ITSP to IP-PBX</b> (arbitrary descriptive name)
Source IP Group	<b>BroadCloud</b>

**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	<b>IP Group</b>
Destination IP Group	<b>IP-PBX</b>
Destination SIP Interface	<b>IP-PBX</b>

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

✕
Edit Row

Index   
 Routing Policy Default\_SBCRouting

Rule
Action

Destination Type IP Group ▾  
 Destination IP Group IP-PBX ▾  
 Destination SIP Interface IP-PBX ▾  
 Destination Address   
 Destination Port   
 Destination Transport Type   ▾  
 Call Setup Rules Set ID   
 Group Policy None ▾  
 Cost Group None ▾

[Classic View](#)

Save
Cancel

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

▼ IP-to-IP Routing Table

Add + Edit ✎ Delete 🗑 Insert + Up ↑ Down ↓
▼ All
Search in table
Search 🔍

Show / Hide 📄

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 Address	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	

Page 1 of 1
10 ▾
View 1 - 3 of 3



**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	<b>0</b>
Name	<b>Call to desk</b>
Source IP Group	<b>IP-PBX</b>
Destination IP Group	<b>BroadCloud</b>
Destination Username Prefix	<b>4347</b>

**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	<b>Destination URI</b>
Prefix to Add	<b>0119723976</b>

**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**

▼ IP to IP Outbound Manipulation

Index	Name	Routing Policy	Additio Manipi	Source IP Group	Destinatio IP Group	Source Usernam Prefix	Destinat Usernam Prefix	Manipuli Item	Remove From Left	Remove From Right	Leave From Right	Prefix to Add	Suffix to Add
0	Call to desk	Default_S	No	IP-PBX	BroadCloud	*	4347	Destinatio	0	0	255	01197239	
1	Call to mobile	Default_S	No	IP-PBX	BroadCloud	*	4774	Destinatio	1	0	255	01197254	
2	For Anonymo	Default_S	No	IP-PBX	BroadCloud	*	*	Source UR	0	0	255		

Page  of 1
 


View 1 - 3 of 3

## 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).

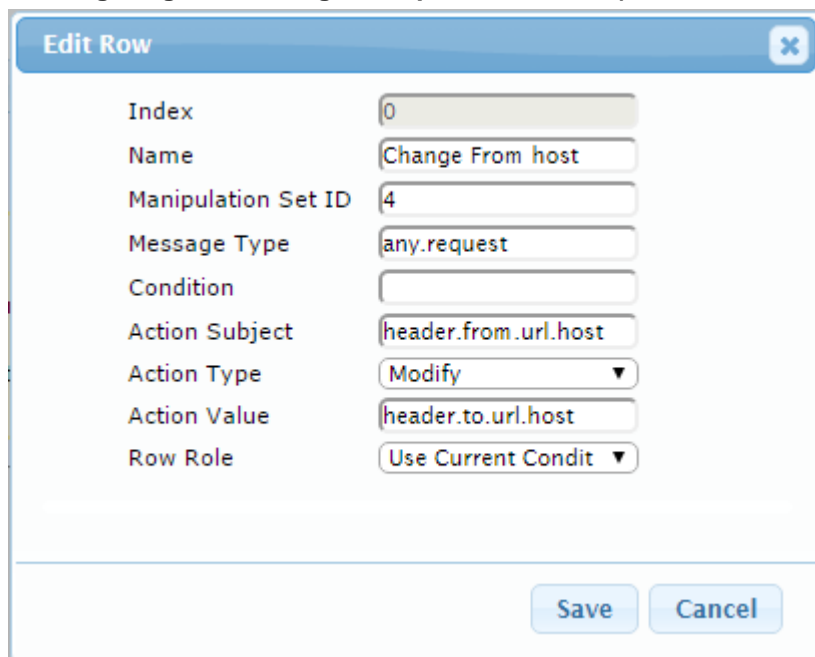
Once you have configured the SIP message manipulation rules, you need to assign them to the relevant IP Group (in the IP Group table) and determine whether they must be applied to inbound or outbound messages.

➤ **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	<b>0</b>
Name	<b>Change From host</b>
Manipulation Set ID	<b>4</b>
Message Type	<b>any.request</b>
Action Subject	<b>header.from.url.host</b>
Action Type	<b>Modify</b>
Action Value	<b>header.to.url.host</b>

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



Edit Row
✕

Index	<input type="text" value="0"/>
Name	<input type="text" value="Change From host"/>
Manipulation Set ID	<input type="text" value="4"/>
Message Type	<input type="text" value="any.request"/>
Condition	<input type="text" value=""/>
Action Subject	<input type="text" value="header.from.url.host"/>
Action Type	<input type="text" value="Modify"/>
Action Value	<input type="text" value="header.to.url.host"/>
Row Role	<input type="text" value="Use Current Condit"/>



- 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	<b>Change P-Asserted host</b>
Manipulation Set ID	4
Message Type	<b>any.request</b>
Condition	<b>header.p-asserted-identity exists</b>
Action Subject	<b>header.p-asserted-identity</b>
Action Type	<b>Modify</b>
Action Value	<b>header.to.url.host</b>

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

The screenshot shows a web-based configuration interface for editing a SIP message manipulation rule. The dialog box is titled "Edit Row" and contains the following fields:

- Index:** 1
- Name:** Change P-Asserted host
- Manipulation Set ID:** 4
- Message Type:** any.request
- Condition:** header.p-asserted-ident
- Action Subject:** header.p-asserted-ident
- Action Type:** Modify
- Action Value:** header.to.url.host
- Row Role:** Use Current Condit

At the bottom of the dialog, there are "Save" and "Cancel" buttons.



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**

The screenshot shows the 'Edit Row' configuration window for an IP Group. The window has a title bar with 'Edit Row' and a close button. Below the title bar, there are two input fields: 'Index' with the value '2' and 'SRD' with a dropdown menu showing 'DefaultSRD'. Below these fields are three tabs: 'Common', 'GW', and 'SBC'. The 'SBC' tab is selected. The 'SBC' tab contains several configuration fields: 'SBC Operation Mode' (dropdown menu showing 'Not Configured'), 'Classify By Proxy Set' (dropdown menu showing 'Enable'), 'SBC Client Forking Mode' (dropdown menu showing 'Sequential'), 'Inbound Message Manipulation Set' (text input field with '-1'), 'Outbound Message Manipulation Set' (text input field with '4'), 'Msg Man User Defined String1' (text input field), 'Msg Man User Defined String2' (text input field), 'Registration Mode' (dropdown menu showing 'User Initiates Regis'), 'Max. Number of Registered Users' (text input field with '-1'), 'Authentication Mode' (dropdown menu showing 'User Authenticates'), 'Authentication Method List' (text input field), and 'Username' (text input field). At the bottom right of the window are two buttons: 'Save' and 'Cancel'.

- 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	<b>SBC</b>
Served IP Group	<b>IP-PBX</b>
Serving IP Group	<b>BroadCloud</b>
Username	As provided by BroadCloud
Password	As provided by BroadCloud
Host Name	<b>interop.adpt-tech.com</b>
Register	<b>Regular</b>
Contact User	<b>8325624857</b> (pilot number)

4. Click **Apply**.

**Figure 4-36: Configuring SIP Registration Account**

The screenshot shows the 'Account Table' interface. At the top, there are buttons for 'Add +', 'Edit', 'Delete', 'Action', and 'Show / Hide'. Below these is a search bar with 'All' selected and a search icon. The table below has the following columns: Index, Application Type, Served Trunk Group, Served IP Group, Serving IP Group, User Name, Password, Host Name, Register, and Contact User. A single row is visible with the following values: Index: 0, Application Type: SBC, Served Trunk Group: -1, Served IP Group: IP-PBX, Serving IP Group: BroadCloud, User Name: 8325624857, Password: \*, Host Name: interop.adpt-, Register: Regular, Contact User: 8325624857. At the bottom, there is a pagination control showing 'Page 1 of 1' and a 'View 1 - 1 of 1' indicator.

Index	Application Type	Served Trunk Group	Served IP Group	Serving IP Group	User Name	Password	Host Name	Register	Contact User
0	SBC	-1	IP-PBX	BroadCloud	8325624857	*	interop.adpt-	Regular	8325624857

## 4.12 Step 12: Miscellaneous Configuration

This section describes miscellaneous E-SBC configuration.

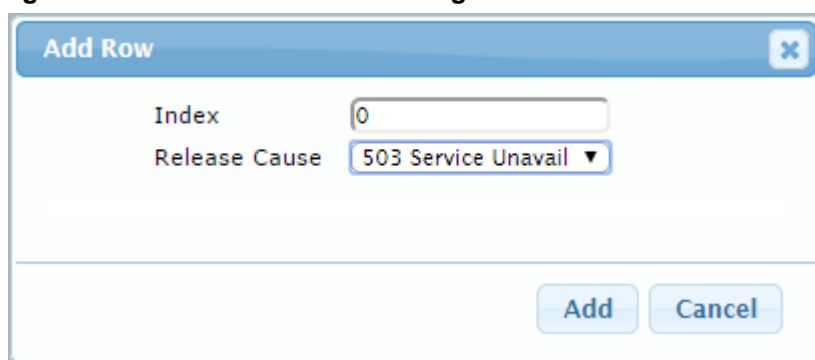
### 4.12.1 Step 12a: Configure SBC Alternative Routing Reasons

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**



Add Row	
Index	<input type="text" value="0"/>
Release Cause	<input type="text" value="503 Service Unavail"/>
<input type="button" value="Add"/> <input type="button" value="Cancel"/>	

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**

The screenshot displays a web-based configuration interface for the E-SBC. It is organized into three main sections, each with a dropdown arrow on the left:

- Reset Configuration:** This section contains three rows. The first row is 'Reset Board' with a 'Reset' button. The second row is 'Burn To FLASH' with a dropdown menu set to 'Yes'. The third row is 'Graceful Option' with a dropdown menu set to 'No'.
- LOCK / UNLOCK:** This section contains three rows. The first row is 'Lock' with a 'LOCK' button. The second row is 'Graceful Option' with a dropdown menu set to 'No'. The third row is 'Gateway Operational State' with the text 'UNLOCKED'.
- Save Configuration:** This section contains one row: 'Burn To FLASH' with a 'BURN' button.

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
HTTPSPCipherString = '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+1NDS14XdhYPQ3onZjojYiZPDw8HAXpTCnJvLw8rIxpmmZ
WczZ2c+P20xODluOzc=", 1, 0, 2, 15, 60, 200,
"a4e40b4a1ef60fad38601e9bf6d0c1ce";
WebUsers 1 = "User",
"$1$EiUhIXBycnohfit/L3otExUbFkYcFBJMERNJGUwYGVIGV1UFB1VSD18MA1hbDA5ydHdx
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 ]
    
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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_SBCDiversionsMode, 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_EnableEarlyl183,
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_SBCAdaptRFC2833BWTtoVoiceCoderBW;
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, 300, -1, -1, 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, 300, -1, -1, 0, 0, 0, 0, 0, 0, -1, -1, -1, -1, -1, 0, "", 0;

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```

[ \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_TCPSPort, 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 ]

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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;

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[ \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,

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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