Configuration Note

AudioCodes Professional Services - Interoperability Lab

Connecting Mitel 3300 IP-PBX To BroadCloud SIP Trunk using AudioCodes Mediant™ E-SBC

Version 7.0

Mitel

broadsoft broadcloud

HD VoIP Sounds Better

AudioCodes

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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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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 Mitel 3300 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	Mitel
Model	3300
Software Version	11.0.2.79
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

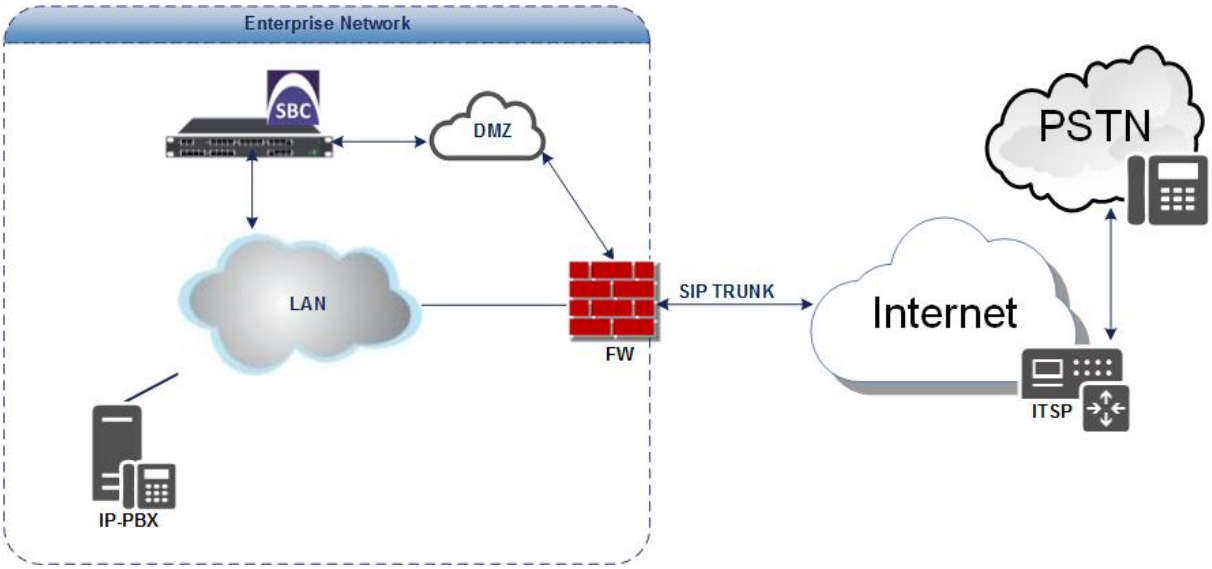
2.4 Interoperability Test Topology

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

- Enterprise deployed with Mitel 3300 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 Mitel 3300 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	▪ Mitel 3300 IP-PBX is located on the Enterprise's LAN ▪ BroadCloud SIP Trunk is located on the WAN
Signaling Transcoding	▪ Mitel 3300 IP-PBX operates with SIP-over-UDP transport type ▪ BroadCloud SIP Trunk operates with SIP-over-UDP transport type
Codecs Transcoding	▪ Mitel 3300 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	▪ Mitel 3300 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 Mitel 3300 IP-PBX and BroadCloud 's SIP Trunk.

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3 Configuring Mitel

This chapter describes how to configure the basic parameters of the Mitel 3300 IP-PBX to operate with the AudioCodes E-SBC.

Note: For more complicated configuration parameters, refer to *User's Manual* for each IP-PBX.

3.1 Basic Configuration Parameters

The screenshots below show the main parameters which should be configured for the Mitel 3300 to operate with the AudioCodes E-SBC.

➢ **To configure Mitel 3300:**

1. Connect to the Mitel via the Web Interface (**License and Option Selection SAT > License >**).

   License and Option Selection Ensure that the MCD is equipped with sufficient SIP Trunking licenses for the connection to BroadSoft. This can be verified within the License and Option Selection form. The total number of licenses in the SIP Trunk Licenses field is the maximum number of SIP Trunk sessions that can be configured in the MCD to be used with all service providers and applications. The figure below shows that the system has three SIP Trunks available within the assigned Application Record.

Figure 3-1: License and Option Selection

License and Option Selection

Licensed Options	Locally Consumed	Locally Allocated	Available for Allocation	Purchased	Local Limits: Licenses Allowed	Local Limits: Can be Over Allocated
Users						
IP Users	1	16	0	16	Unrestricted	Yes
External Hot Desk Users	0	10	0	10	Unrestricted	Yes
ACD Active Agents	0	10	0	10	Unrestricted	No
HTML Applications	0	0	20	0	Unrestricted	Yes
Analog Lines	0	16	0	16	Unrestricted	Yes
IP Console Active Operators	0	0	0	0	0	No
Multi-device Users	0	5	0	5	Unrestricted	Yes
Multi-device Suites	0	0	5	5	0	No
Messaging						
Embedded Voice Mail	0	16	0	16	Unrestricted	Yes
Embedded Voice Mail PMS	1	Yes	0	1	Unrestricted	Yes
Trunking/Networking						
Digital Links	0	1	0	1	Unrestricted	Yes
Compression		8	0	8	Unrestricted	Yes
FAX Over IP (T.38)		4	0	4	Unrestricted	Yes
SIP Trunks	3	3	0	3	Unrestricted	Yes
Others						
MCD IDS Connection	1	Yes	0	1	Unrestricted	Yes
MLPP	0	No	0	0	Unrestricted	No

2. Open the Class of Service Options are configured in the Class of Service Options page (**SAT** > **System Properties** > **System Feature Settings** > **Class of Service Options**).

   The following table describes the set/selected for best service for the classes for the devices which use the BroadCloud SIP Trunk and for the SIP Trunk itself.

Table 3-1: Class of Service Options

Setting	Value
Busy Override Security	Yes
Call Forwarding (External)	Yes
Call Forwarding Accept	Yes
Disable Call Reroute Chaining On	No
Campon Tone Security / FAX Machine	Yes
Public Network Access via	Yes

3. Open the Network Element page (**Voice Network** > **Network Elements**).

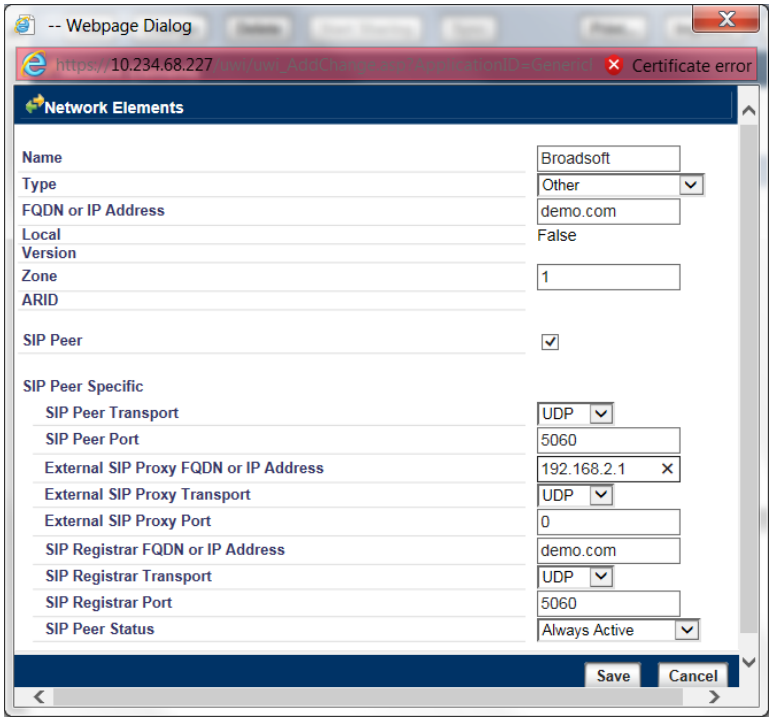
   A new Network Element needs to be created which defines the connection point on the BroadSoft platform for the MCD.

   The following parameters should be set in the available fields.

Table 3-2: Network Elements

Setting	Value
Name	Give the Network Element a meaningful name e.g. BroadCloud.
Type	Other
FQDN or IP Address	This should be set to the Domain name set for the customer.
Zone	Select the desired zone for which the Network Element will reside within.
SIP Peer	Checked
SIP Peer Transport	UDP
External SIP Proxy FQDN or IP Address	The external SIP proxy should be set to the IP of the AudioCodes device.
SIP Peer Port	5060
SIP Registrar FQDN or IP Address	This should match the value you have entered for the FQDN or IP address
SIP Registrar Transport	UDP
SIP Registrar Port	5060
SIP Peer Status	Always Active

Figure 3-2: Network Elements



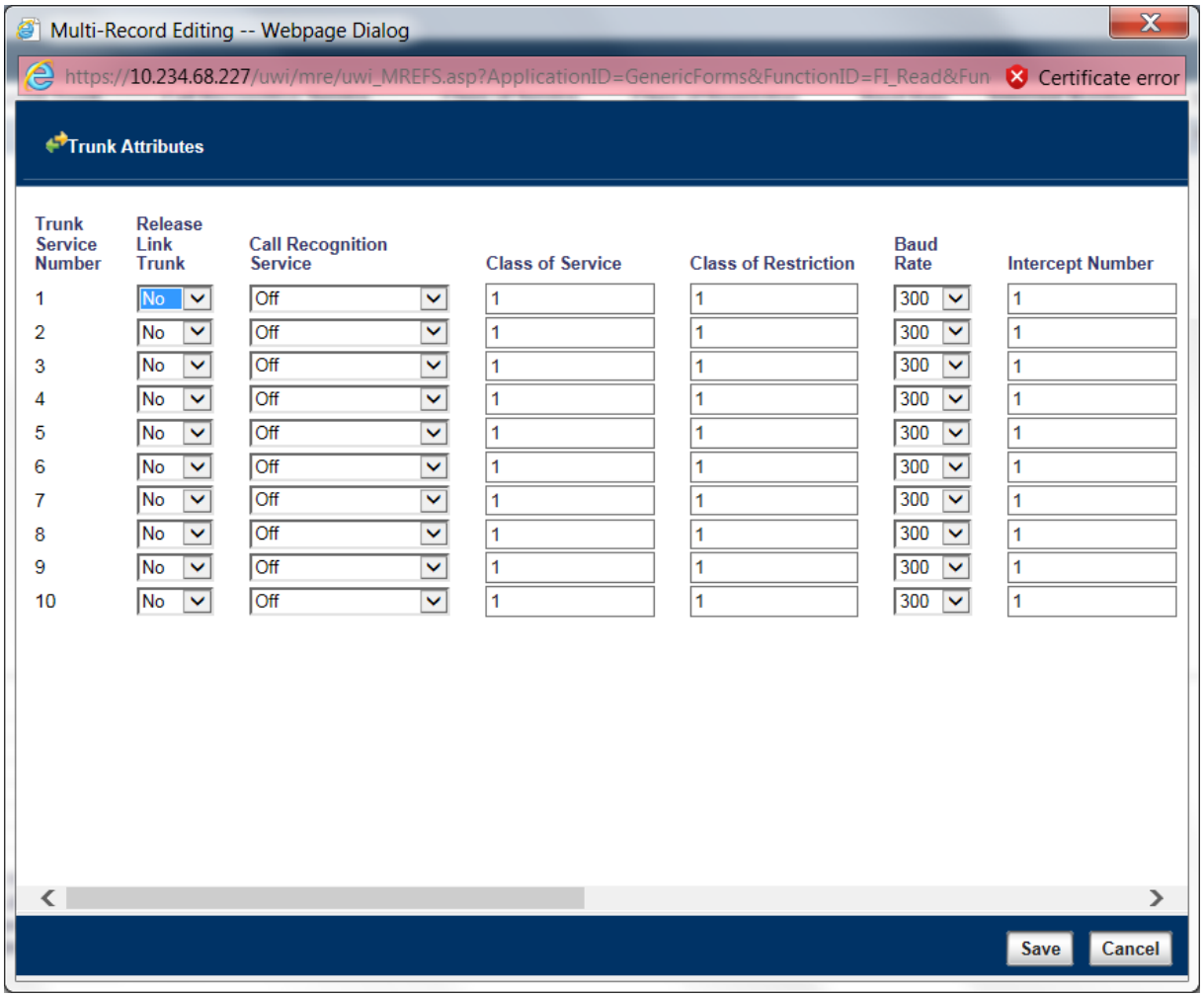
4. Open the Trunk Attributes page (**Trunks** > **Trunk Attributes**).

In addition to the standard function of directing incoming calls to an answer point, we have used the Trunk Attributes page to modify the incoming called number to provide a match with the internally configured users.

In our example deployment, the users on the MCD make use of the last four digits of their full DDI as configured on the BroadCloud platform; as such the Trunk Attributes Service profile we setup removes the first eight digits.

The below image shows the setup of the Trunk Attributes Service profile in our example setup.

Figure 3-3: Trunk Attributes



5. Open the SIP Peer Profile page (**Trunks** > **SIP** > **SIP Peer Profile**).

   The SIP Peer Profile defines the settings used by the MCD when communicating with the previously configured Network element.

   Within the configuration for the new SIP Peer Profile, set as shown in the table below. Where a value is not discussed it should be left as the default value.

   Earlier versions of MCD do not use a tab structure and instead all values are displayed in a single configuration page. In such scenarios, map the values where applicable.

Table 3-3:SIP Peer Profile

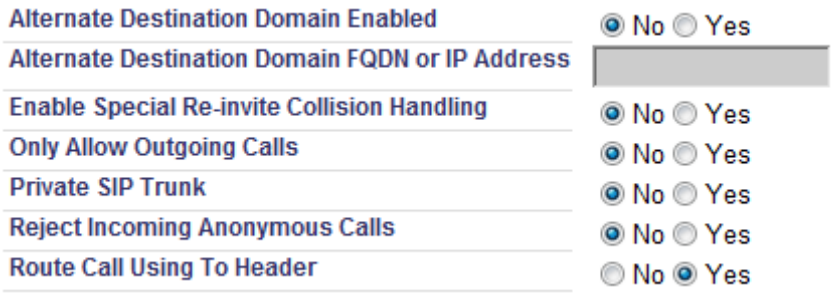
Setting	Value
SIP Peer Profile Label	Give the Peer Profile a meaningful label e.g. BroadCloud.
Network Element	Select the Name of the Network Element created previously.
Local Account Information	
Address Type	IP Address
Administration Options	
Trunk Service	This should match the Trunk Attributes Service setup previously.

Setting	Value
Maximum Simultaneous Calls	This should be set to a value matching that which is set for the Trunk Group in the Service Provider Portal
Authentication Options	
Authentication Option for Incoming Calls	This should always be set to No Authentication.

6. Select the **Call Routing** tab.

   Unless advised otherwise, we recommend that the settings under this tab are left as their default as shown below.

Figure 3-4: Call Routing



7. Select the **Calling Line ID** tab.

   If in your site deployment, you wish all outbound calls to display a single number regardless of their source in the system. This tab allows you to define the number that is used.

   In our deployment, each user was set to display its own DDI; therefore this page was not configured.

   Values other than the **Default CPN** fields should be left as default as shown below.

Figure 3-5: Default CPN

8. Select the **SDP Options** tab.

   Unless advised otherwise, make sure that the settings under this tab match those shown below.

Figure 3-6: SDP Options Tab

Allow Peer To Use Multiple Active M-Lines	No Yes
Allow Using UPDATE For Early Media Renegotiation	No Yes
Avoid Signaling Hold to the Peer	No Yes
Enable Mitel Proprietary SDP	No Yes
Force sending SDP in initial Invite message	No Yes
Force sending SDP in initial Invite - Early Answer	No Yes
Limit to one Offer/Answer per INVITE	No Yes
NAT Keepalive	No Yes
Prevent the Use of IP Address 0.0.0.0 in SDP Messages	No Yes
Renegotiate SDP To Enforce Symmetric Codec	No Yes
Repeat SDP Answer If Duplicate Offer Is Received	No Yes
RTP Packetization Rate Override	No Yes
RTP Packetization Rate	20ms
Special handling of Offers in 2XX responses (INVITE)	No Yes
Suppress Use of SDP Inactive Media Streams	No Yes

9. Select the **Signalling and Header Manipulation** tab.

   Unless advised otherwise, make sure that the settings under this tab match those shown in the figure below.

Figure 3-7: Signalling and Header Manipulation Tab

Trunk Group Label	
Allow Display Update	No Yes
Build Contact Using Request URI Address	No Yes
De-register Using Contact Address not *	No Yes
Disable Reliable Provisional Responses	No Yes
Disable Use of User-Agent and Server Headers	No Yes
E.164: Enable sending '+'	No Yes
E.164: Add '+' if digit length > N digits	0
E.164: Do not add '+' to Emergency Called Party	No Yes
E.164: Do not add '+' to Called Party	No Yes
Force Max-Forward: 70 on Outgoing Calls	No Yes
Ignore Incoming Loose Routing Indication	No Yes
Only use SDP to decide 180 or 183	No Yes
Require Reliable Provisional Responses on Outgoing Calls	No Yes
Use Privacy: none	No Yes
Use P-Asserted Identity Header	No Yes
Use P-Preferred Identity Header	No Yes
Use Restricted Character Set For Authentication	No Yes
Use To Address in From Header on Outgoing Calls	No Yes
Use user=phone	No Yes

10. Select the **Key Press Event** tab.

Unless advised otherwise, make sure that the settings under this tab match those shown below.

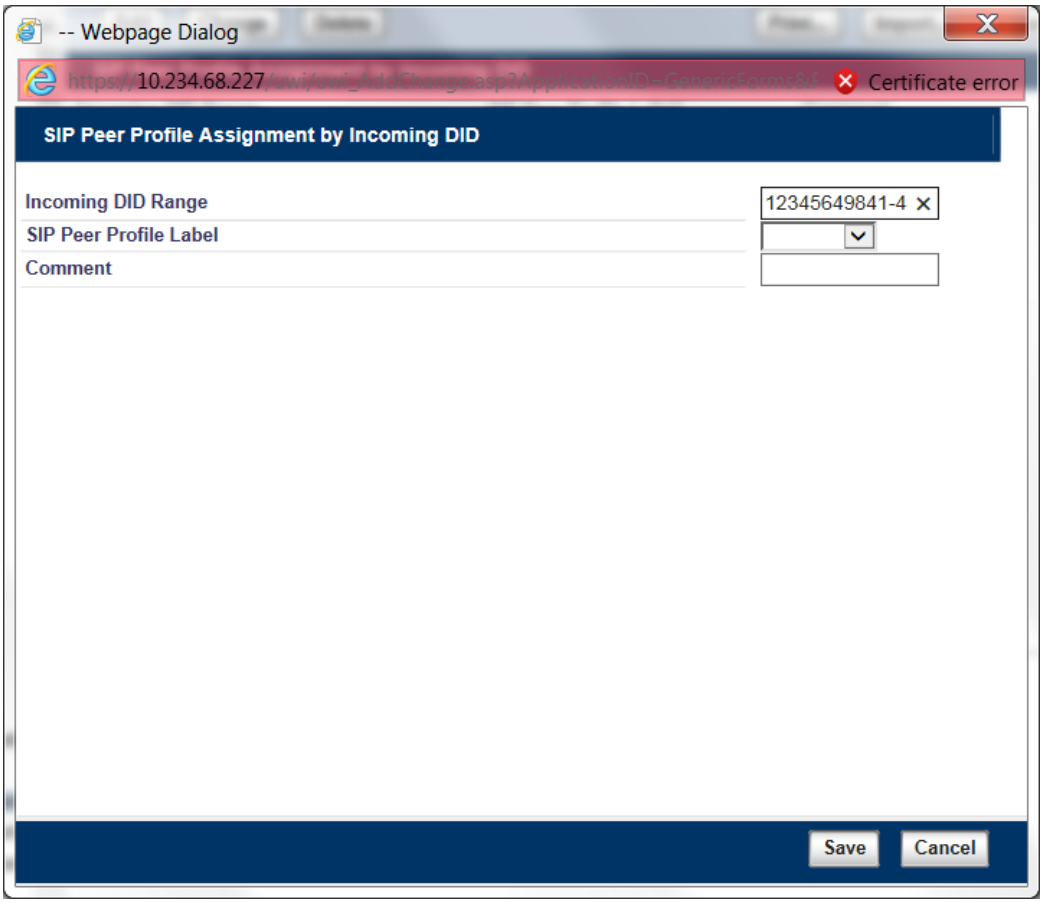
Figure 3-8: Key Press Event Tab

Allow Inc Subscriptions for Local Digit Monitoring	No Yes
Allow Out Subscriptions for Remote Digit Monitoring	No Yes
Force Out Subscriptions for Remote Digit Monitoring	No Yes
Request Outbound Proxy to Handle Out Subscriptions	No Yes
KPML Transport	default
KPML Port	0

11. Open the SIP Peer Profile Assignment by Incoming DID page (**Trunks** > **SIP** > **SIP Peer Profile Assignment by Incoming DID**).

The rules defined in this page are used to associate incoming calls to the appropriate SIP Peer Profile based on the received DDI.

Figure 3-9: SIP Interconnection Setup



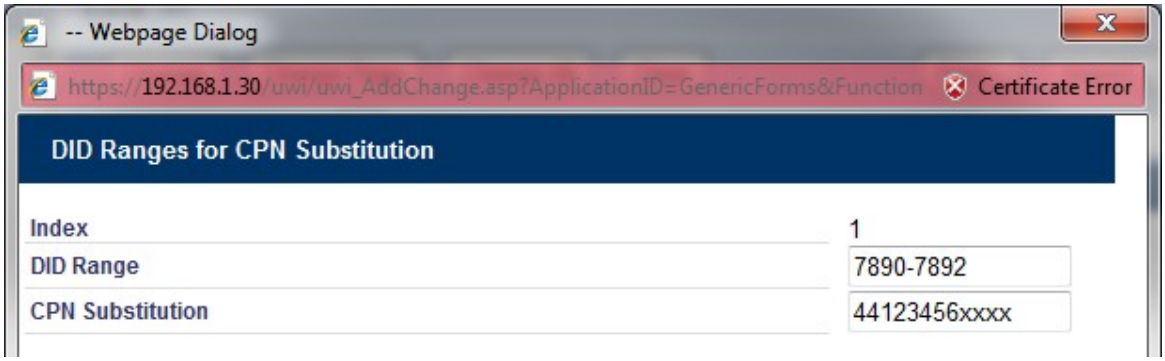
12. Open the DID Ranges for CPN Substitution page (**Trunks** > **SIP** > **DID Ranges for CPN Substitution**).

You can use the profiles on this page to define DDI's displayed as the source number for calls from users on the MCD, map them to DDI's of the users setup on the BroadCloud platform i.e. these profiles can display numbers to the PSTN.

The transformation should mean that the number presented to BroadCloud is the number of the Trunk Group or associated Trunk User on the BroadCloud platform.

The example below shows a profile where the DID of each user is a direct mapping to the last 4 digits of that user's DDI on the BroadCloud platform.

Figure 3-10: DID Ranges for CPN Substitution



13. Open the ARS Digit Modification Plans (Optional) page (**Call Routing** > **Automatic Route Selection (ARS)** > **ARS Digit Modification Plans**).

    Depending on the configuration of your MCD system and the dialing plan that is employed, it may be necessary to modify the called number for outbound calls to the BroadCloud platform i.e. dialing "9" for an outside line is used.

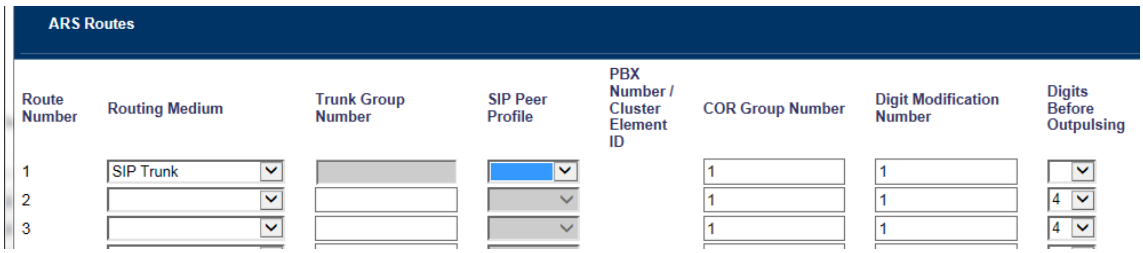
    The below profile is an example where "9" is dialed to trigger an Outbound call to the SIP Trunk, i.e. "9" is removed before the call is passed to BroadCloud.

Figure 3-11: ARS Digit Modification Plans

14. Open the ARS Routes page (**Call Routing** > **Automatic Route Selection (ARS)** > **ARS Routes**).

    Each ARS Routes profile allows you to define an Outbound route for calls that will be referenced in the dialing plan of the MCD. The profile you create for calls to the BroadCloud platform should consist of the following.

Figure 3-12: ARS Routes

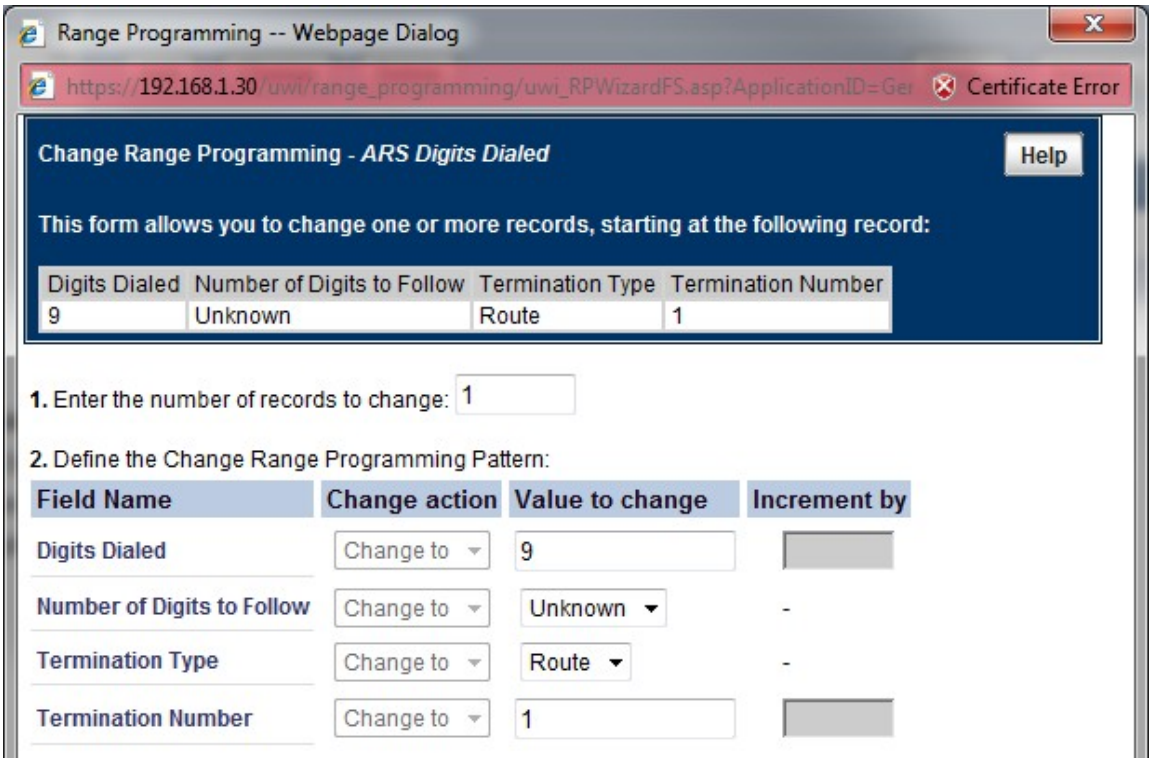


15. Open the ARS Digits Dialed page (**Call Routing > Automatic Route Selection (ARS) > ARS Digits Dialed**).

    The ARS Digits Dialed profiles form the dial plan for the MCD system. In an existing MCD system, most of the required profiles should already be in place, in which case you will just need to change the Termination Number for each profile to the previously configured ARS Routes profile ID.

    The below example shows a system where "9" must be dialed followed by any number of digits, the call is then routed to ARS Routes profile 1, which is setup to send calls to the BroadCloud SIP Peer.

Figure 3-13: ARS Digits Dialed



When configuring a new MCD system for the sole purpose of working with the BroadCloud platform, carefully consider all required routes and the associated ARS Digits Dialed profiles.

16. Open the Codec Filter Settings page (**Maintenance and Diagnostics** > **Maintenance Commands**).

    By default, the MCD system allows the use of several codecs by the IP handsets. These codecs are generally not supported on the BroadCloud platform.

    This can cause certain call flows to fail due to the manner upon which Mitel answers specific SIP requests. To prevent this from occurring, you must disable use of the unsupported codecs using the **ADDCODECFILTER** command.

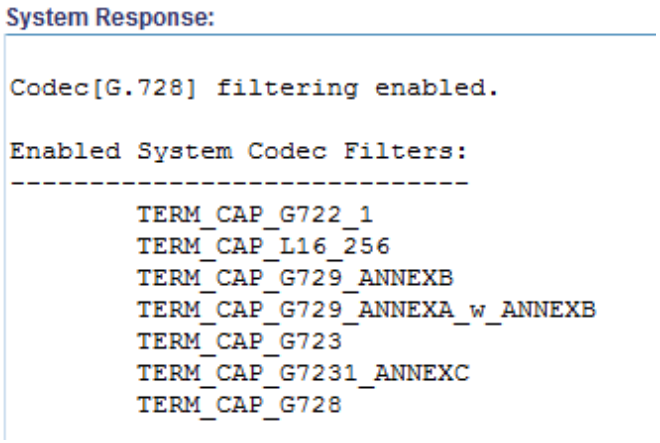
    We recommend filtering the following codecs on the MCD:

Figure 3-14: Codec Filter Settings

G.722.1
L16-256
G.729b
G.729ab
G.723
G.723.1c
G.728

When you have added filters for all the above codecs, the command entry confirmation should appear as displayed in the figure below:

Figure 3-15: System Response



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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:

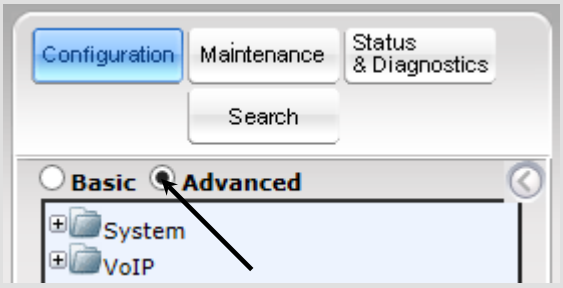
- E-SBC WAN interface – BroadCloud SIP Trunking environment
- E-SBC LAN interface – Mitel 3300 IP-PBX environment

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:



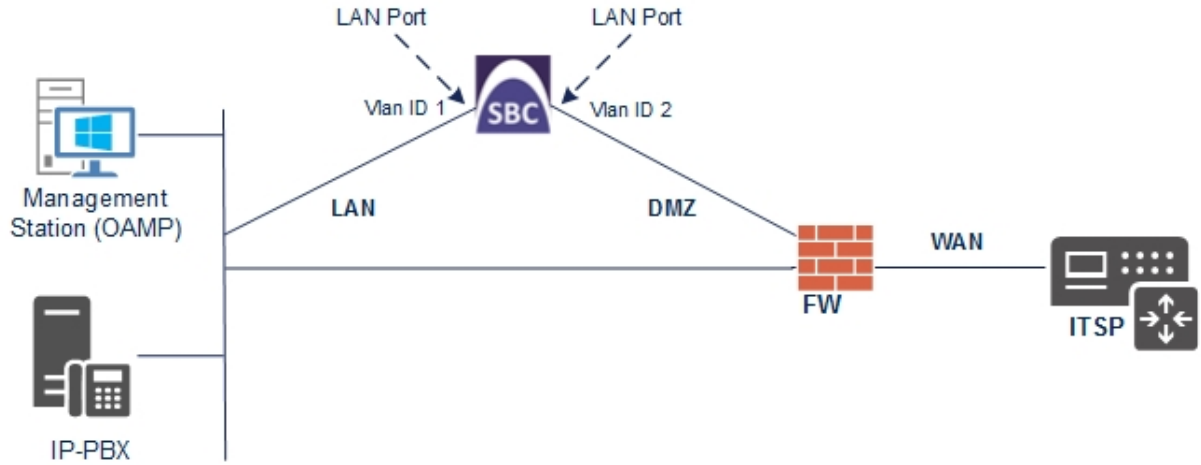
Note that when the E-SBC is reset, the Navigation tree reverts to Basic-menu display.

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:
  - Mitel 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:

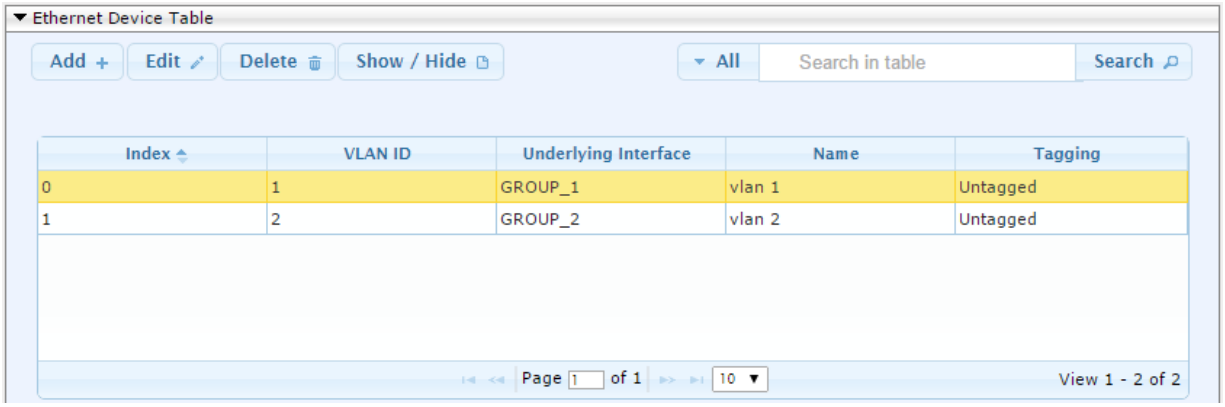
- LAN VoIP (assigned the name "Voice")
- WAN VoIP (assigned the name "WANSP")

➢ **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	**10.234.68.240** (IP address of E-SBC)
Prefix Length	**15** (subnet mask in bits for 255.255.255.0)
Default Gateway	**10.234.68.1**
VLAN ID	**1**
Interface Name	**Voice** (arbitrary descriptive name)
Primary DNS Server IP Address	**10.234.35.20**
Underlying Device	**vlan 1**

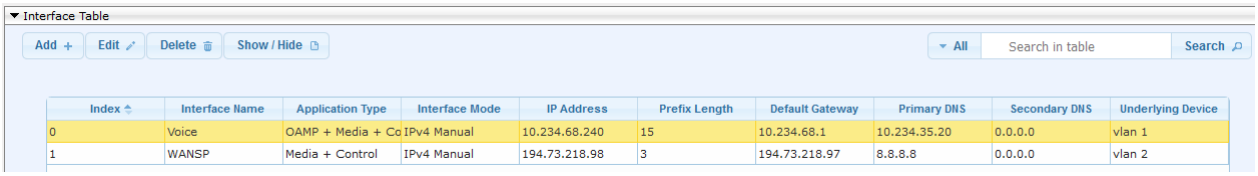
3. Add a network interface for the WAN side:
   a. Enter **1**, and then click **Add Index**.
   b. Configure the interface as follows:

Parameter	Value
Application Type	**Media + Control**
IP Address	**194.73.218.98** (WAN IP address)
Prefix Length	**3** (for 255.255.255.128)
Default Gateway	**194.73.218.97** (router's IP address)
VLAN ID	**2**
Interface Name	**WANSP**
Primary DNS Server IP Address	**8.8.8.8**
Secondary DNS Server IP Address	**0.0.0.0**
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



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 62).

4.3 Step 3: Configure Media Realms

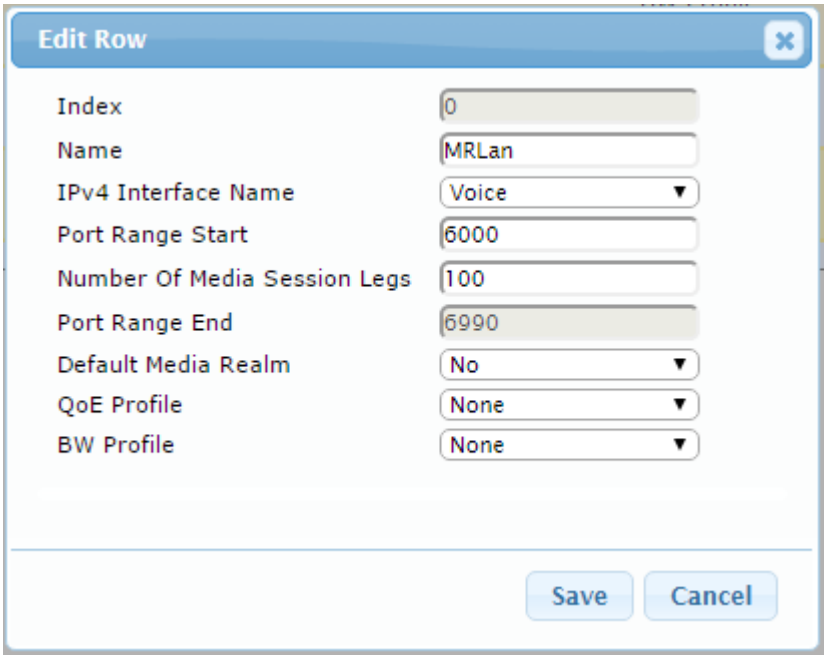
This step describes how to configure Media Realms. The simplest configuration is to create two Media Realms - one for internal (LAN) traffic and one for external (WAN) traffic.

➢ **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)

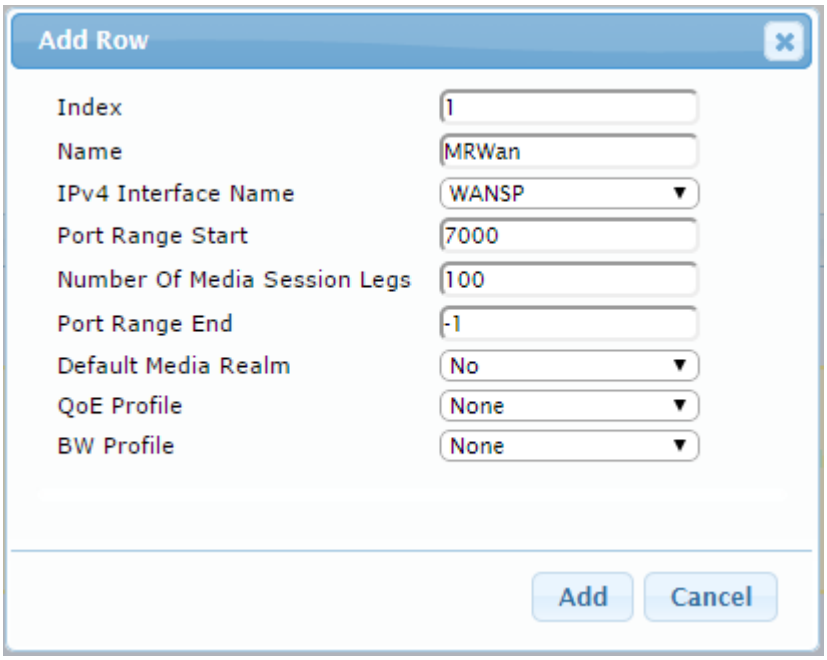
Figure 4-5: Configuring Media Realm for LAN



3. Configure a Media Realm for WAN traffic:

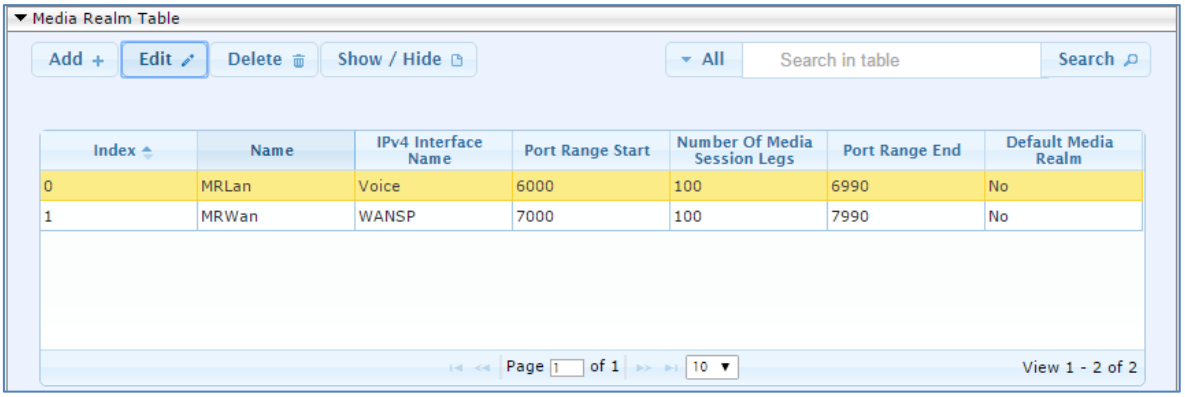
Parameter	Value
Index	**1**
Media Realm Name	**MRWan** (arbitrary name)
IPv4 Interface Name	**WANSP**
Port Range Start	**7000** (represents lowest UDP port number used for media on WAN)
Number of Media Session Legs	**100** (media sessions assigned with port range)

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); however, modify it as shown below:

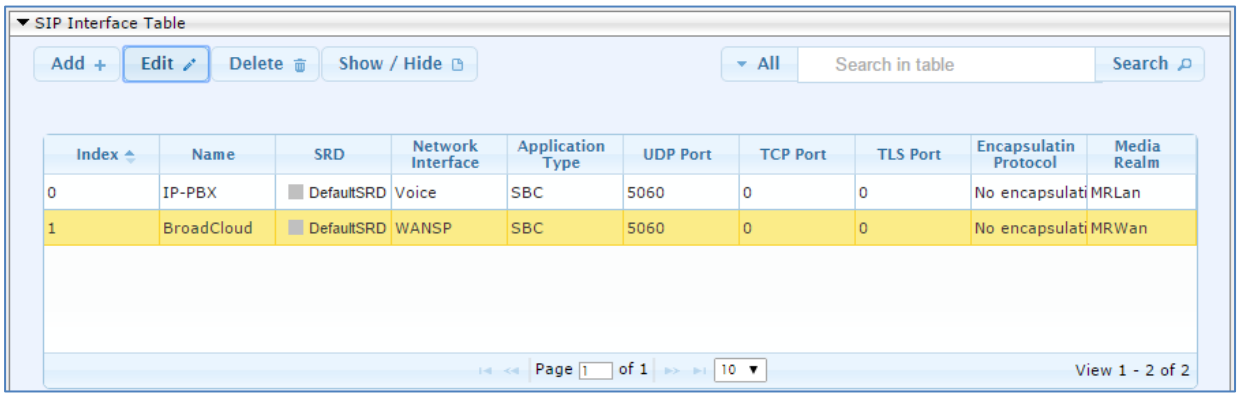
Parameter	Value
Index	**0**
Interface Name	**IP-PBX** (see Note below)
Network Interface	**Voice**
Application Type	**SBC**
UDP Port	**5060**
TCP and TLS	**0**
Media Realm	**MRLan**

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:

- Mitel 3300 IP-PBX
- BroadCloud SIP Trunk

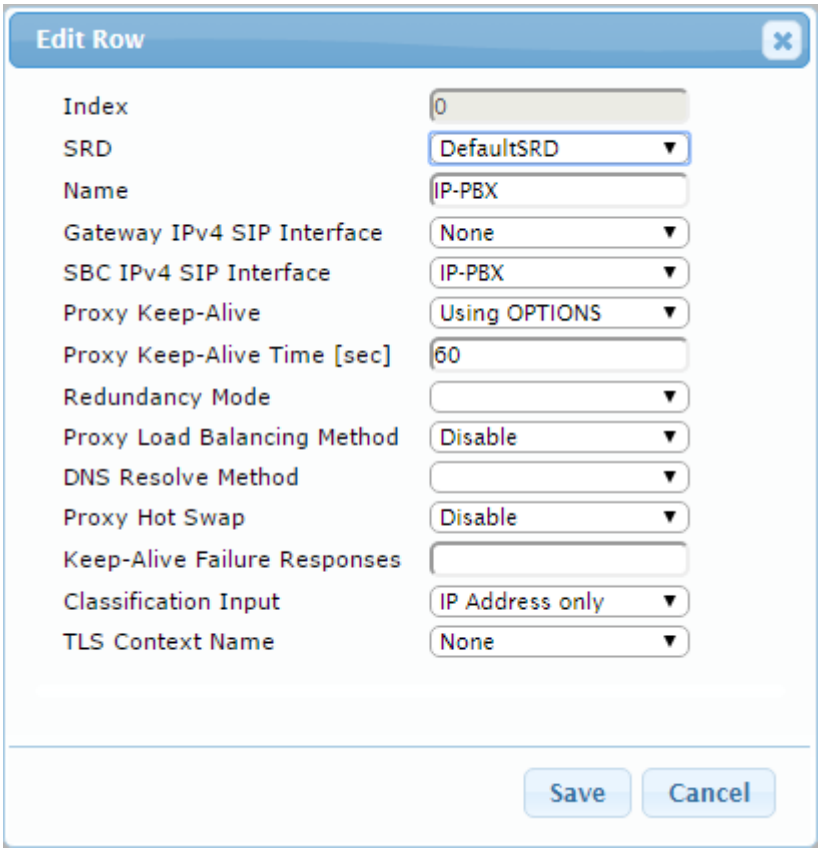
The Proxy Sets will be later applied to the VoIP network by assigning them to IP Groups.

➢ **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** (see Note in Section 4.4)
SBC IPv4 SIP Interface	**IP-PBX**
Proxy Keep Alive	**Using Options**

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

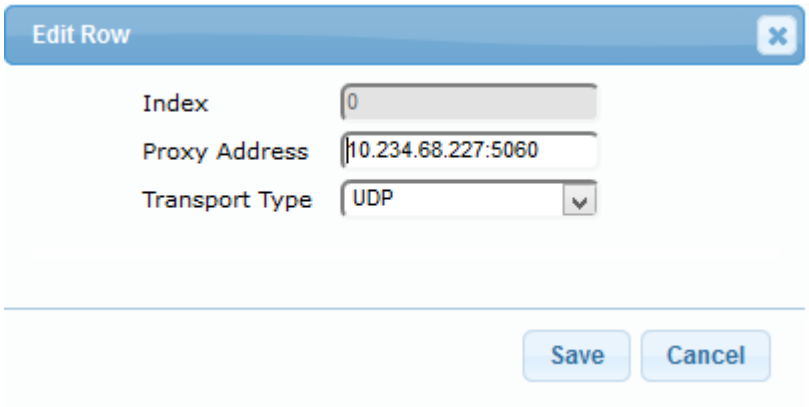


3. Configure a Proxy Address Table for Proxy Set for Mitel 3300 IP-PBX:

   a. Open the Proxy Address table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table** > **Proxy Address Table**).

Parameter	Value
Index	**0**
Proxy Address	**10.234.68.227:5060** (IP-PBX IP address / FQDN and destination port)
Transport Type	**UDP**

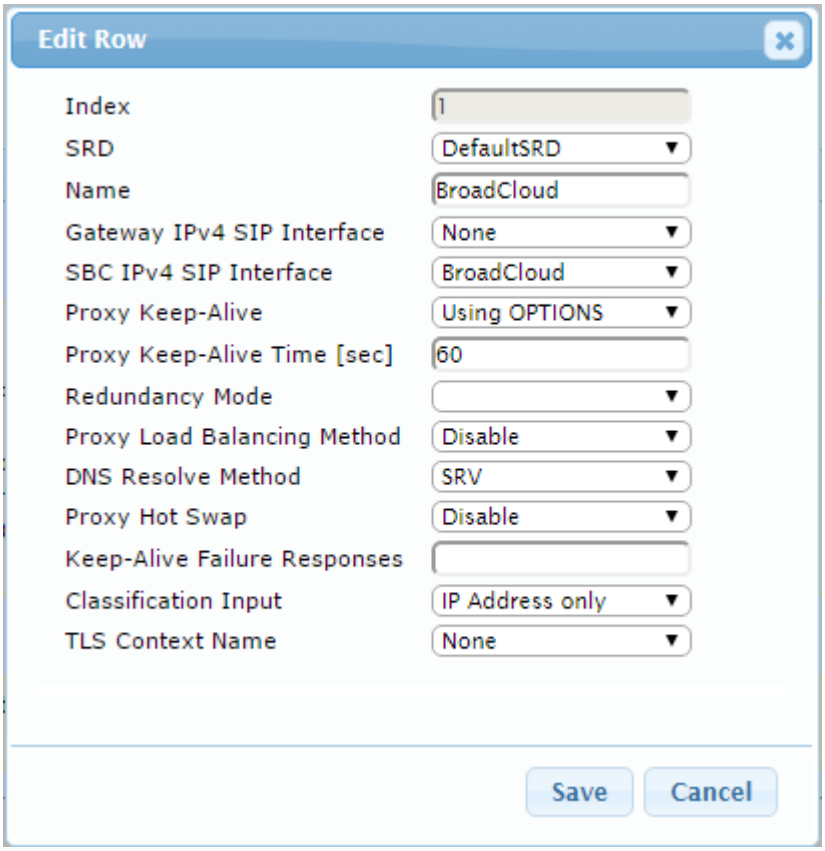
Figure 4-10: Configuring Proxy Address for IP-PBX



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

Parameter	Value
Proxy Set ID	**1**
Proxy Name	**BroadCloud** (see Note in Section 4.4)
SBC IPv4 SIP Interface	**BroadCloud**
Proxy Keep Alive	**Using Options**

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

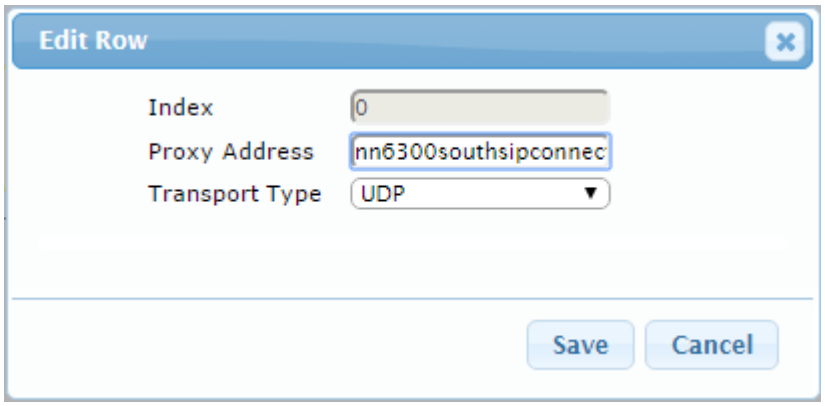


5. Configure a Proxy Address Table for BroadCloud SIP Trunk:

   a. Open the Proxy Address Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table** > **Proxy Address Table**).

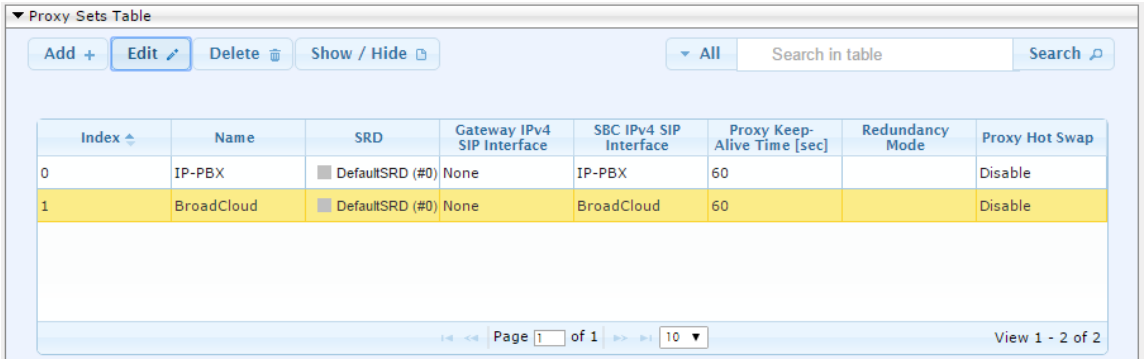
Parameter	Value
Index	**0**
Proxy Address	**nn6300southsipconnect.adpt-tech.com** (IP-PBX IP address / FQDN and destination port)
Transport Type	**UDP**

Figure 4-12: Configuring Proxy Address BroadCloud SIP Trunk:



The configured Proxy Sets are shown in the figure below:

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



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:

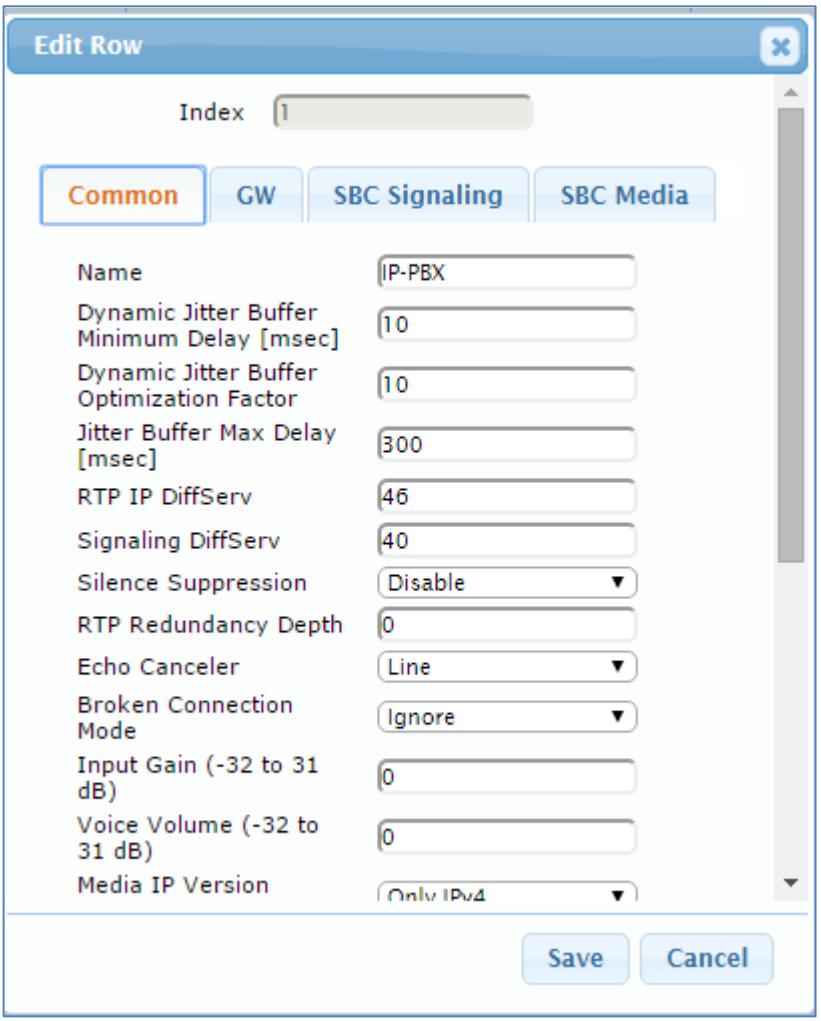
- Mitel 3300 IP-PBX - to operate in non-secure mode using RTP and UDP
- BroadCloud SIP trunk - to operate in non-secure mode using RTP and UDP

➢ **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** (see Note in Section 4.4)

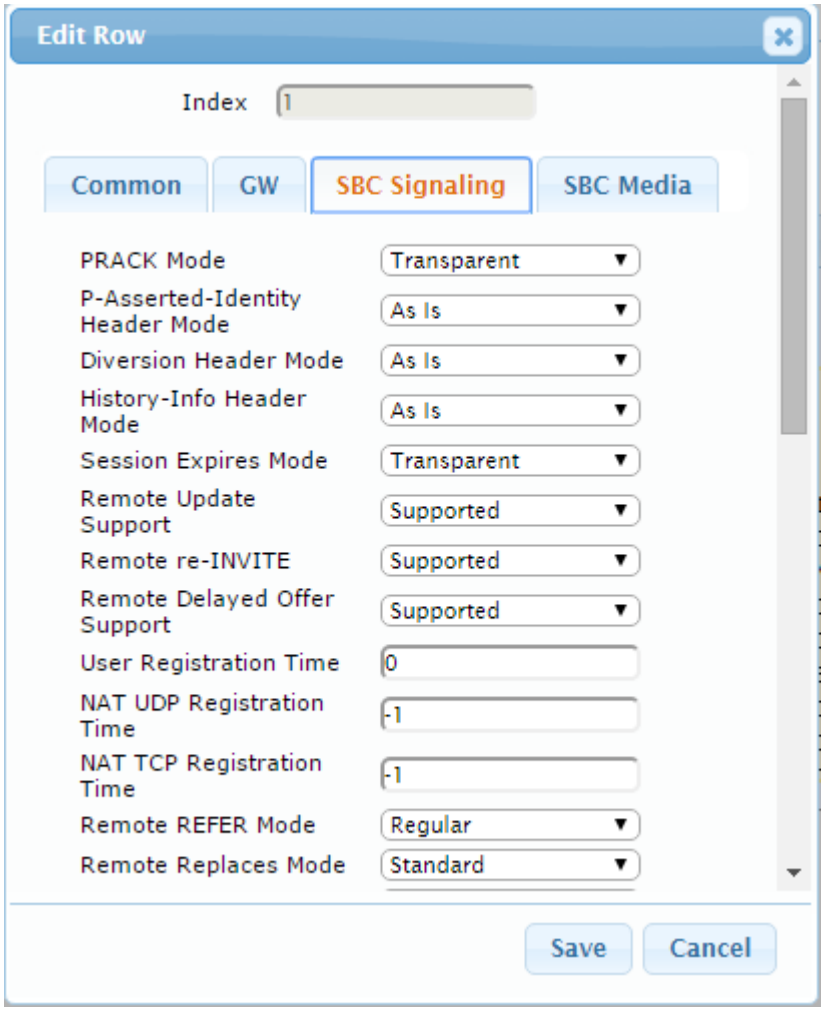
Figure 4-14: Configuring IP Profile for IP-PBX – Common Tab



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

Parameter	Value
Remote Update Support	**Supported**
Remote re-INVITE Support	**Supported**
Remote Can Play Ringback	**No**

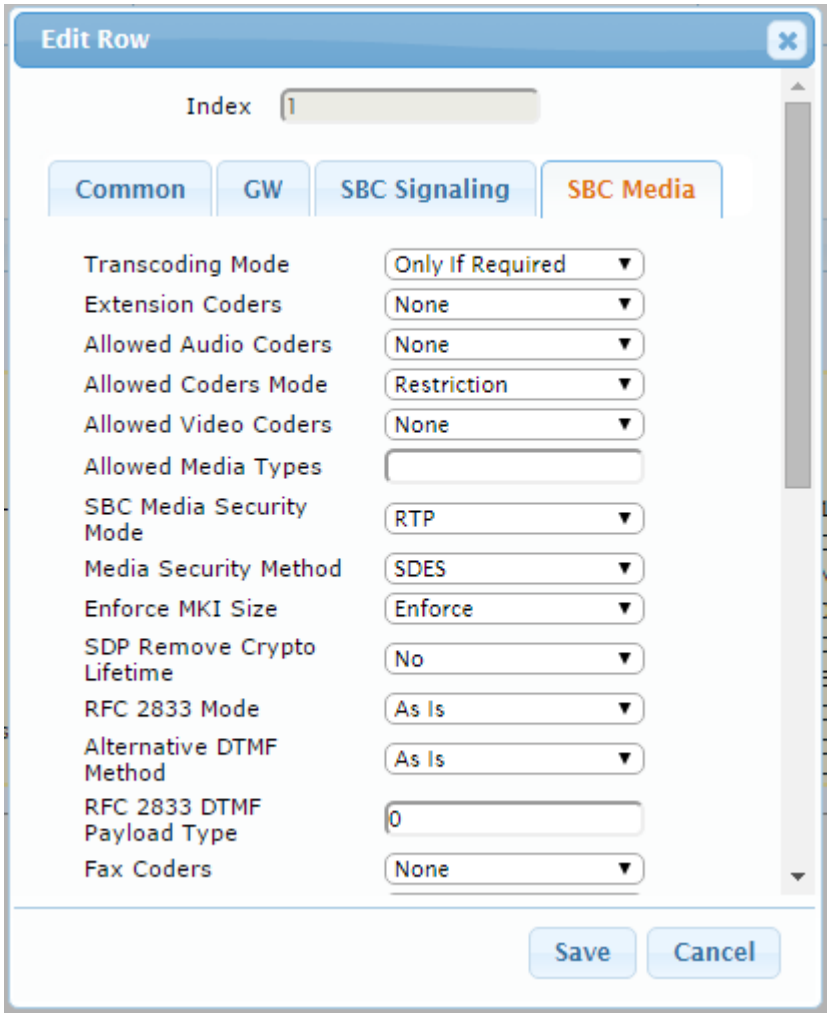
Figure 4-15: Configuring IP Profile for IP-PBX – SBC Signaling Tab



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

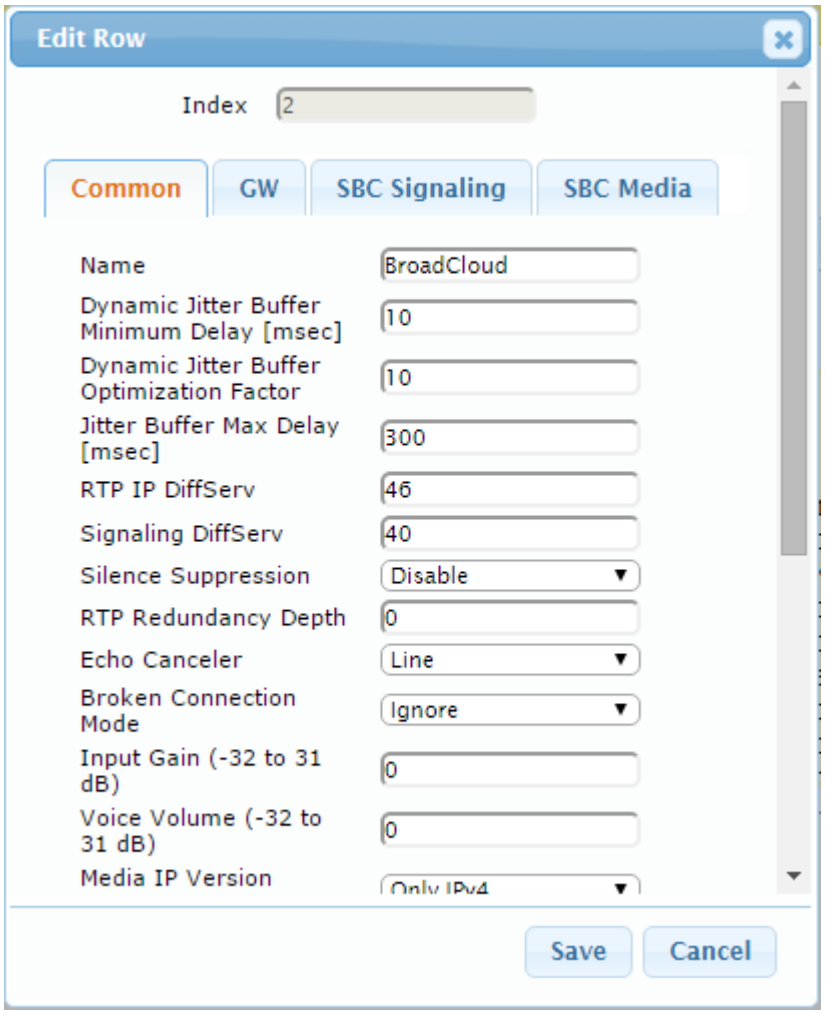


➢ **To configure an IP Profile for the BroadCloud SIP Trunk:**

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

Parameter	Value
Index	**2**
Profile Name	**BroadCloud** (see Note in Section 4.4)

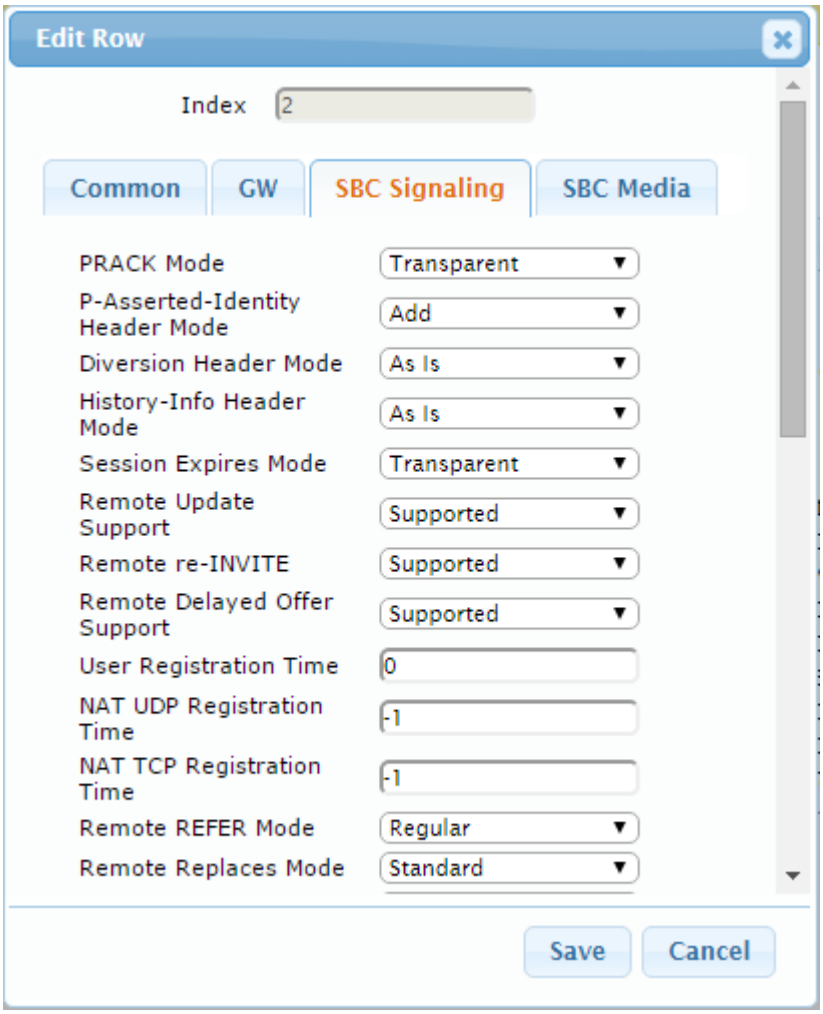
Figure 4-17: Configuring IP Profile for BroadCloud SIP Trunk – Common Tab



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)

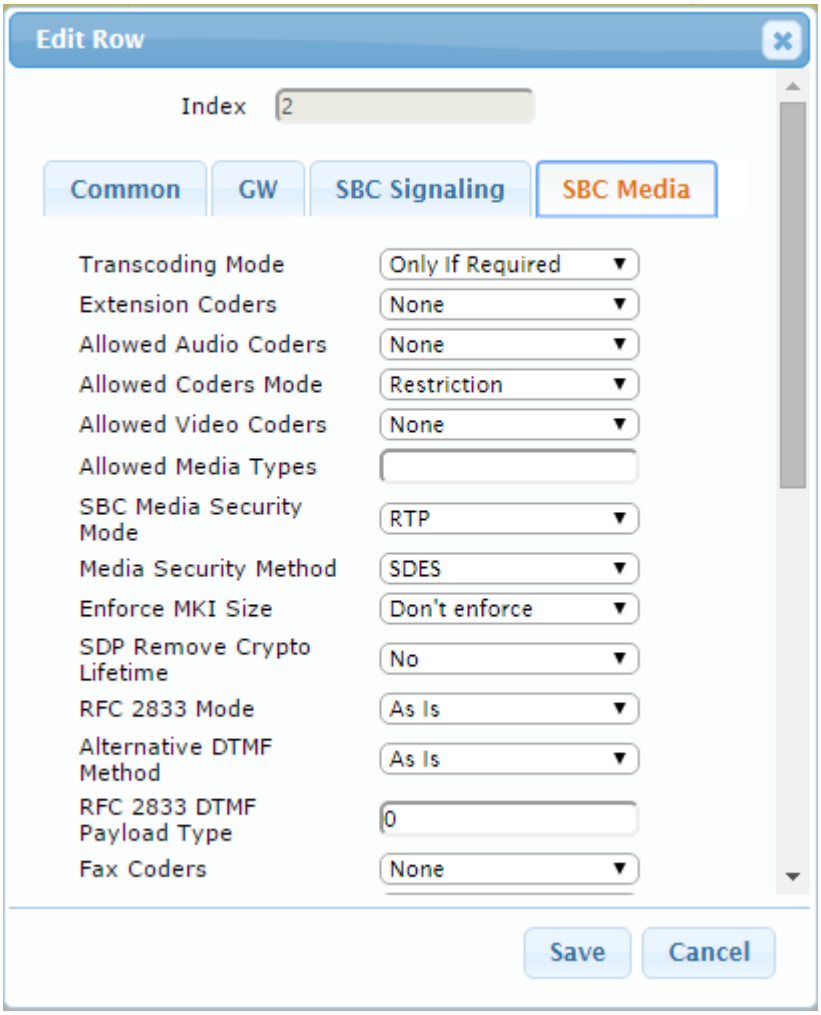
Figure 4-18: Configuring IP Profile for BroadCloud SIP Trunk – SBC Signaling Tab



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:

- Mitel 3300 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 Mitel 3300 IP-PBX. You can use the default IP Group (Index 0), but modify it as shown below:

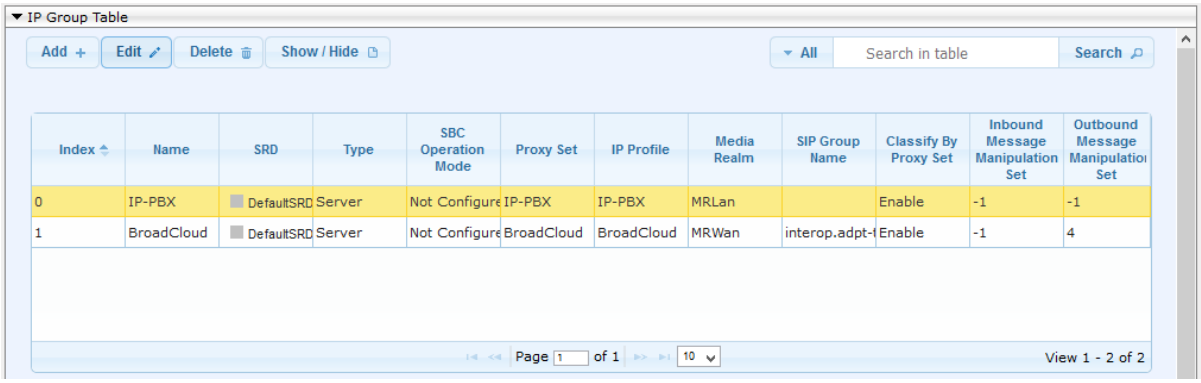
Parameter	Value
Index	**0**
Name	**IP-PBX** (see Note in Section 4.4)
Type	**Server**
Proxy Set	**IP-PBX**
IP Profile	**IP-PBX**
Media Realm	**MRLan**
SIP Group Name	

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

Parameter	Value
Index	**1**
Name	**BroadCloud** (see Note in Section 4.4)
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



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):

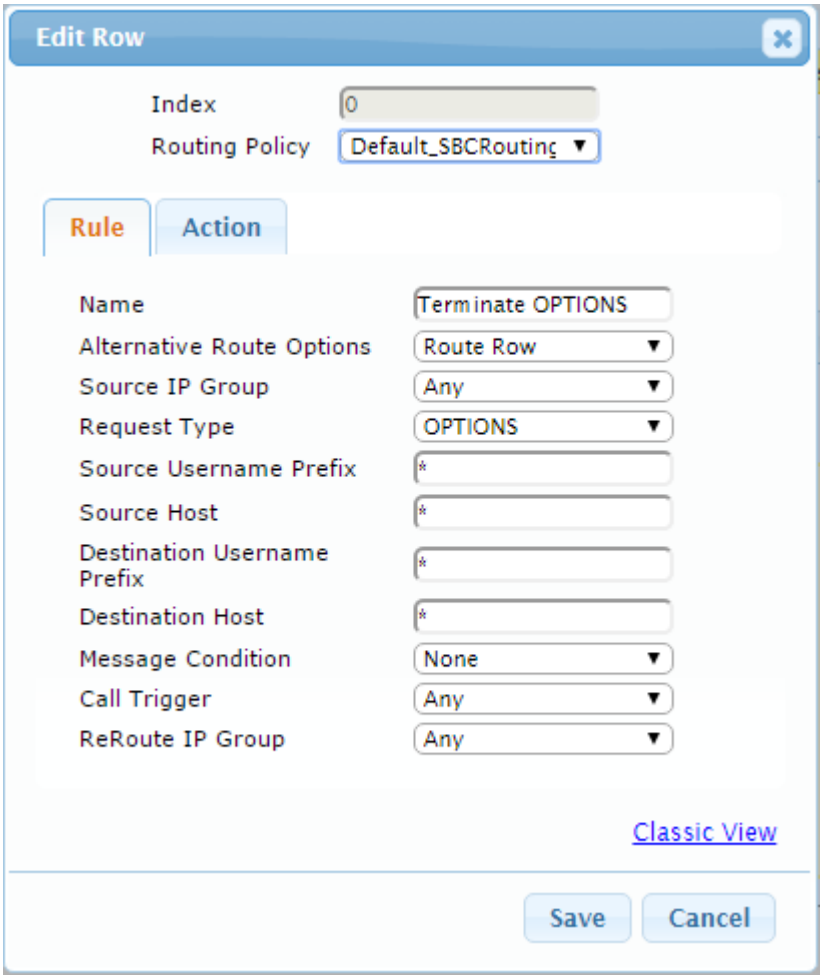
- Terminate SIP OPTIONS messages on the E-SBC
- Calls from IP-PBX to BroadCloud SIP Trunk
- Calls from BroadCloud SIP Trunk to Mitel 3300 IP-PBX

➢ **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**

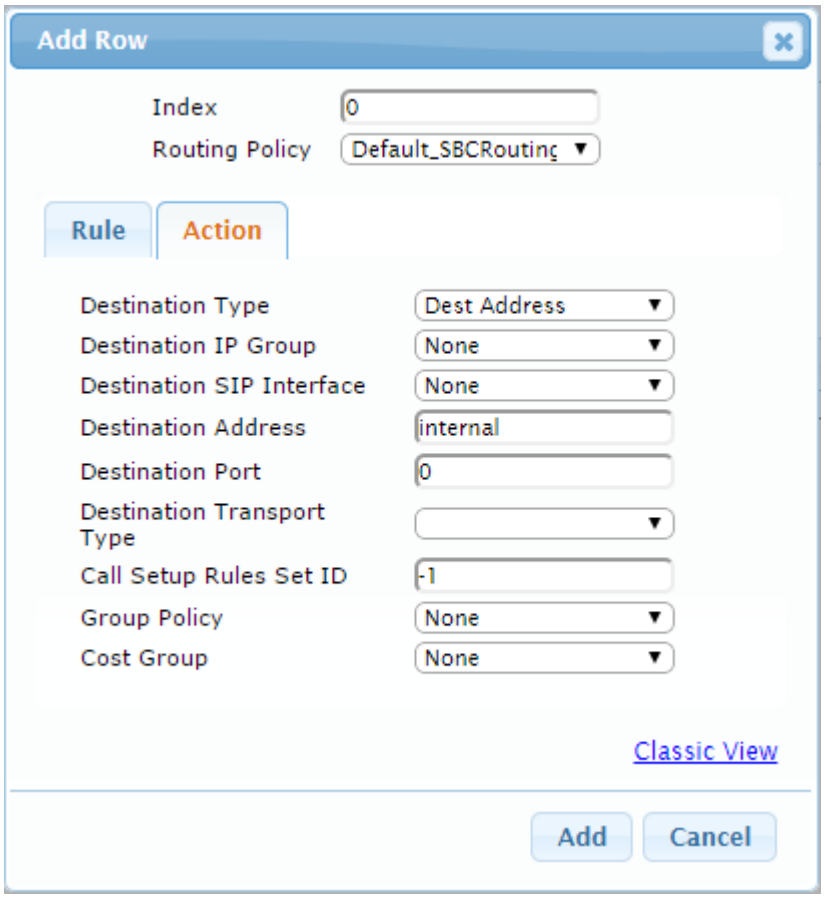
Figure 4-21: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS – Rule Tab



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

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

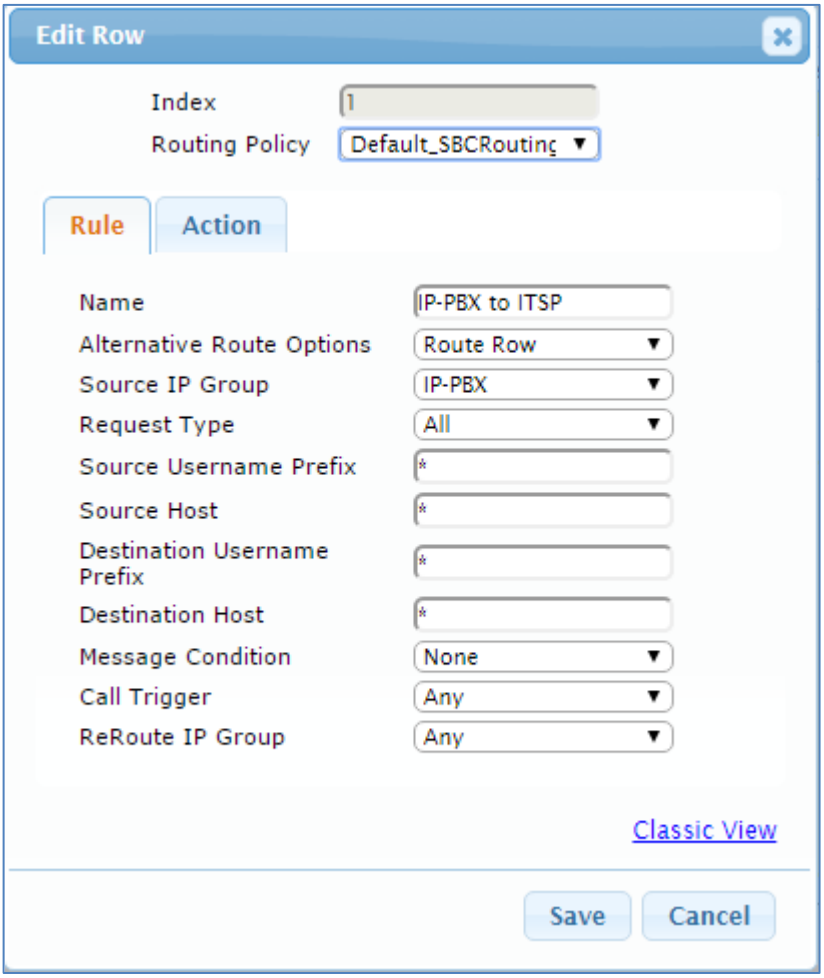
Figure 4-22: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS – Action Tab



3. Configure a rule to route calls from Mitel 3300 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**

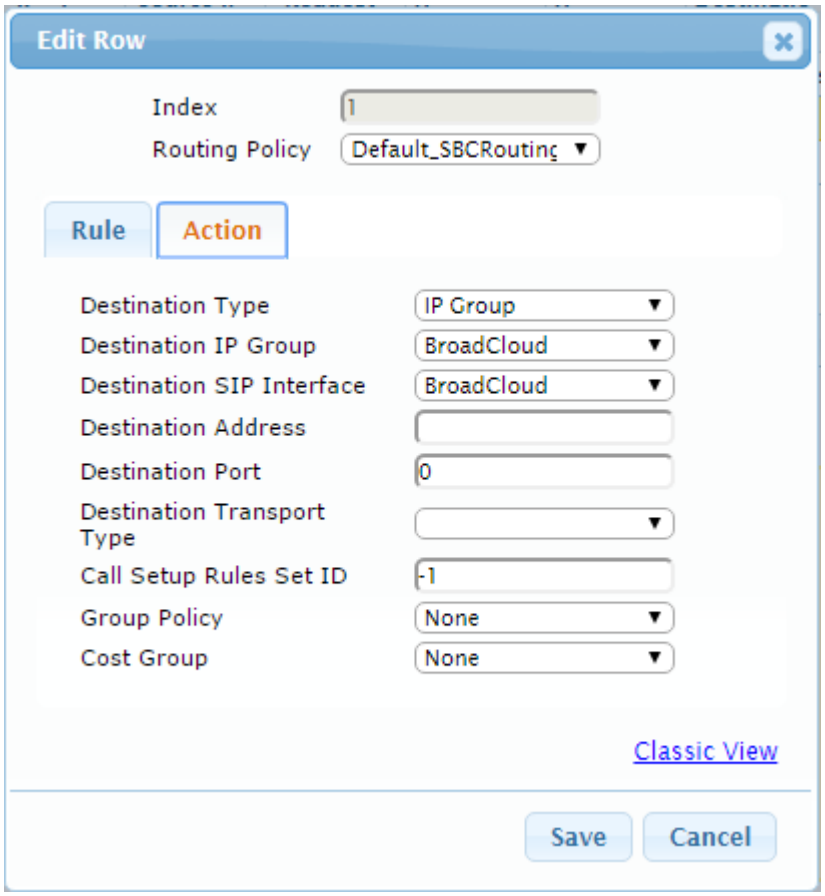
Figure 4-23: Configuring IP-to-IP Routing Rule for IP-PBX to ITSP – Rule tab



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**

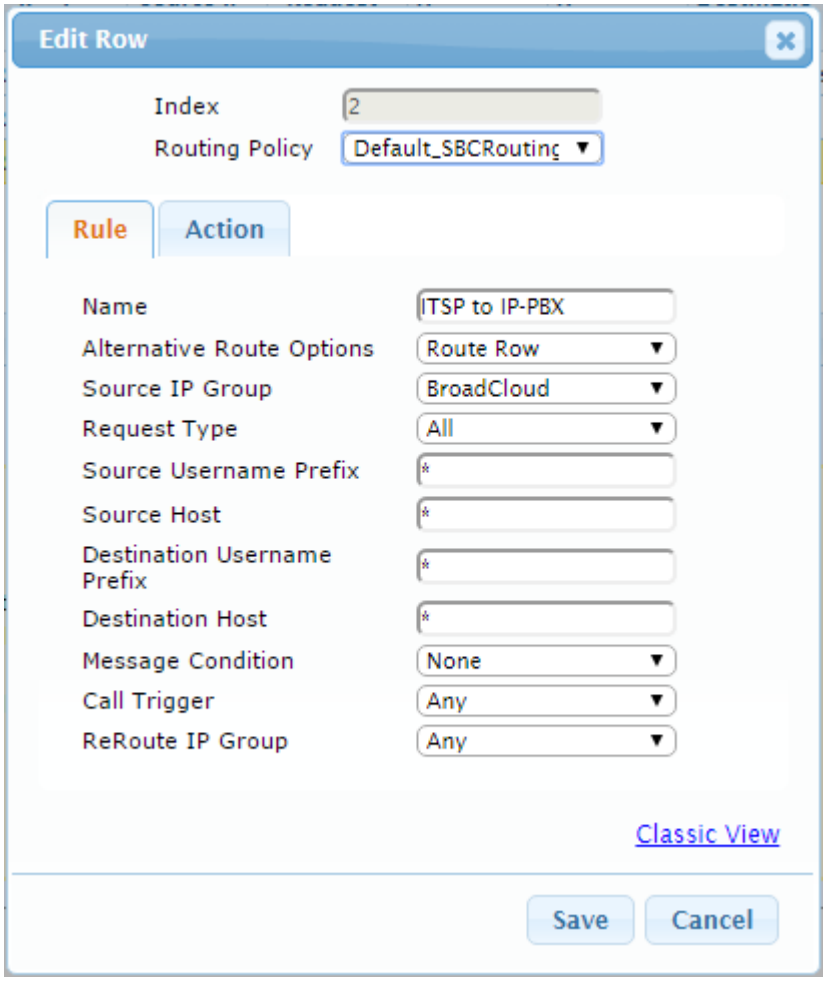
Figure 4-24: Configuring IP-to-IP Routing Rule for IP-PBX to ITSP – Action tab



4. To configure rule to route calls from BroadCloud SIP Trunk to Mitel 3300 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**

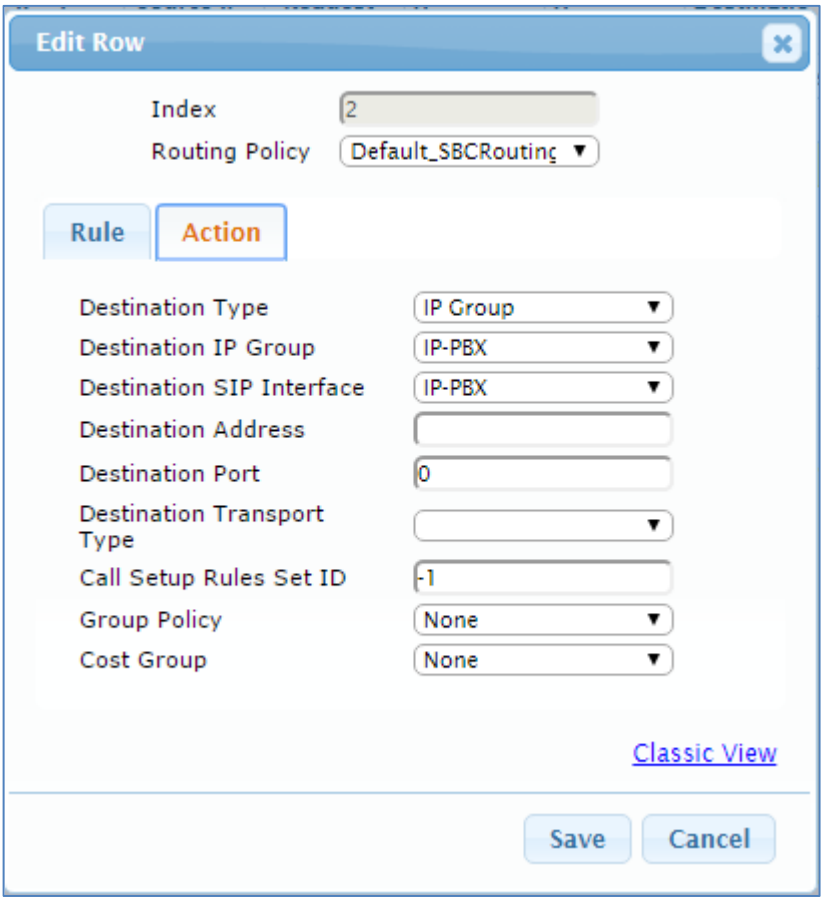
Figure 4-25: Configuring IP-to-IP Routing Rule for ITSP to IP-PBX – Rule tab



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

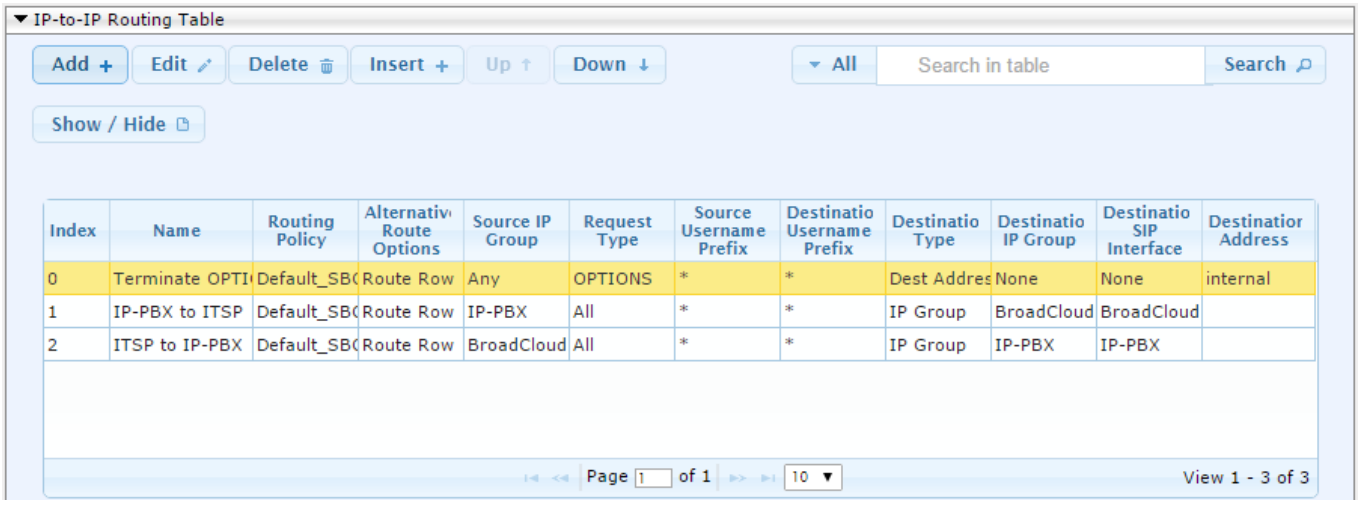
Parameter	Value
Destination Type	**IP Group**
Destination IP Group	**IP-PBX**
Destination SIP Interface	**IP-PBX**

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



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 Mitel 3300 IP-PBX, and IP Group 1 represents BroadCloud SIP Trunk.

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

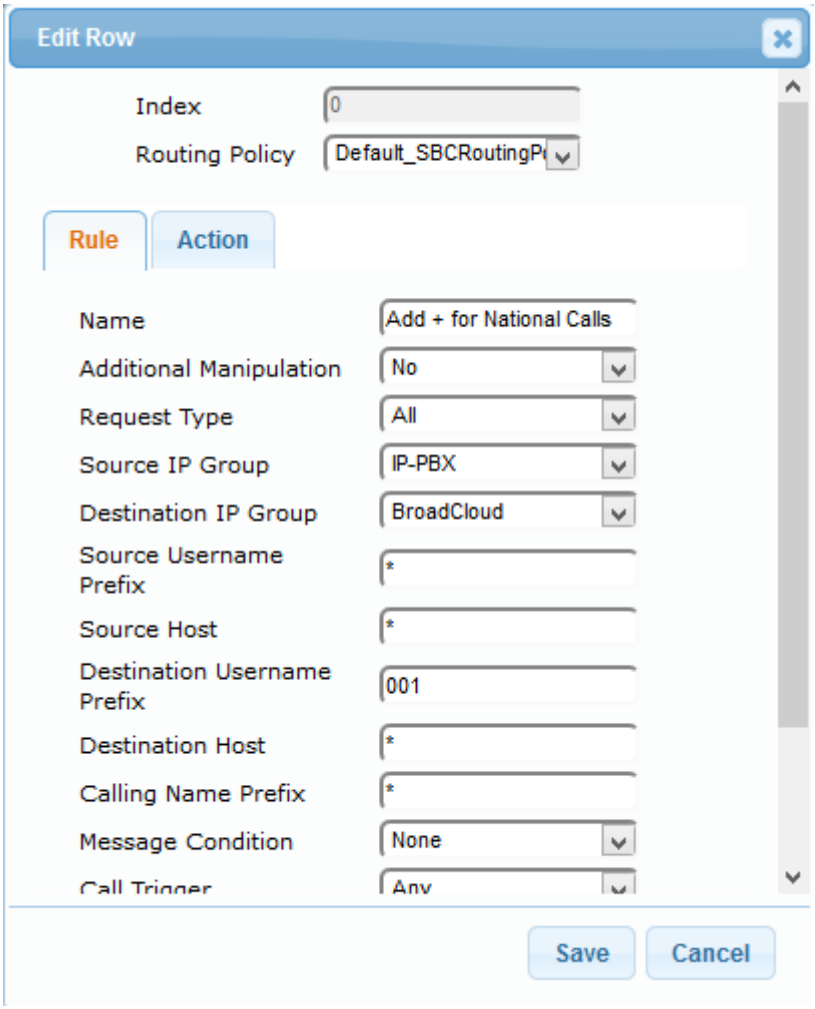
For example, for this interoperability test topology, a manipulation was configured to add the prefix to the destination number for calls from the IP-PBX IP Group to the BroadCloud SIP Trunk IP Group for specific destination username prefix.

➢ **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	**Add + for National Calls**
Source IP Group	**IP-PBX**
Destination IP Group	**BroadCloud**
Destination Username Prefix	**001**

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



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

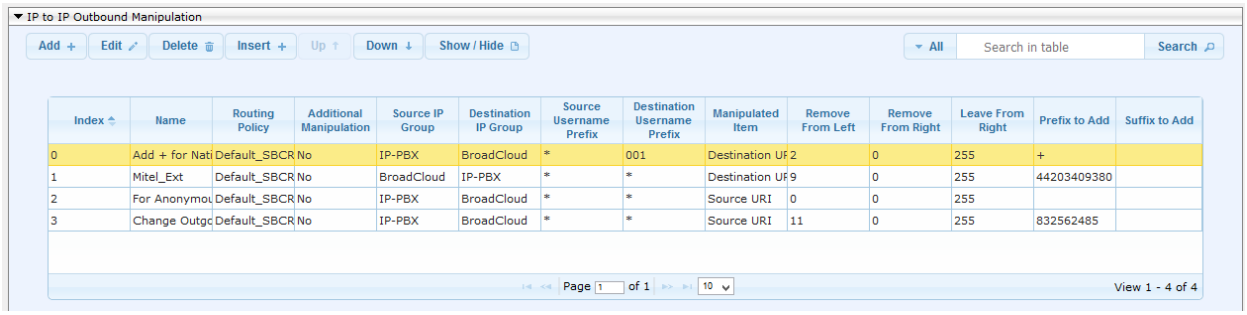
Parameter	Value
Manipulated Item	**Destination URI**
Remove From Left	**2**
Prefix to Add	**+**

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 Mitel 3300 IP-PBX IP Group and BroadCloud SIP Trunk IP Group:

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



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).

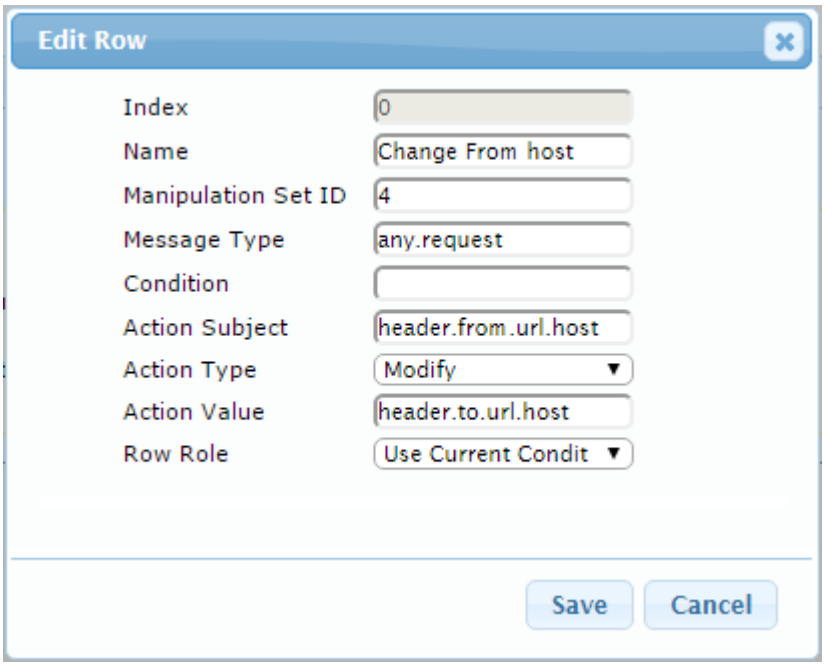
Once you have configured the SIP message manipulation rules, you need to assign them to the relevant IP Group (in the IP Group table) and determine whether they must be applied to inbound or outbound messages.

➢ **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)

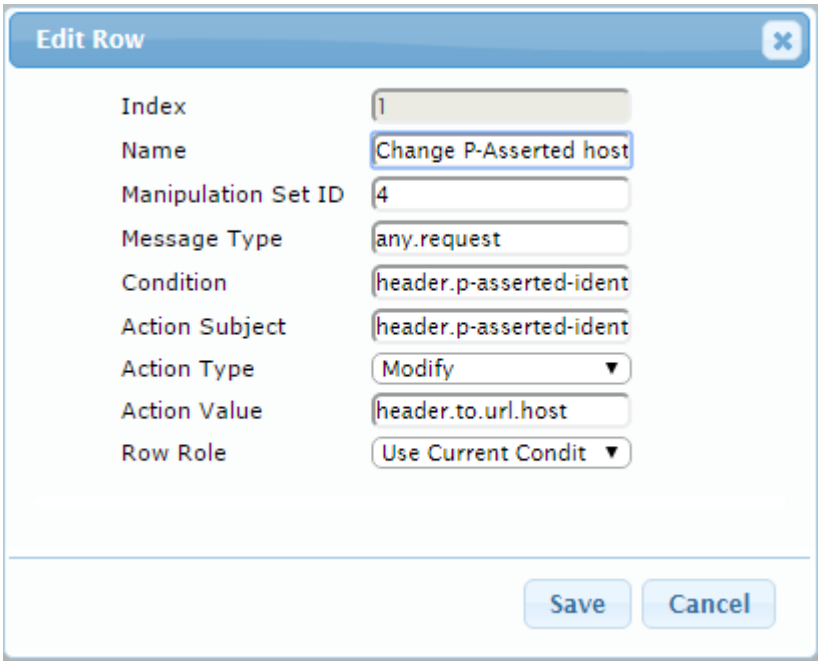
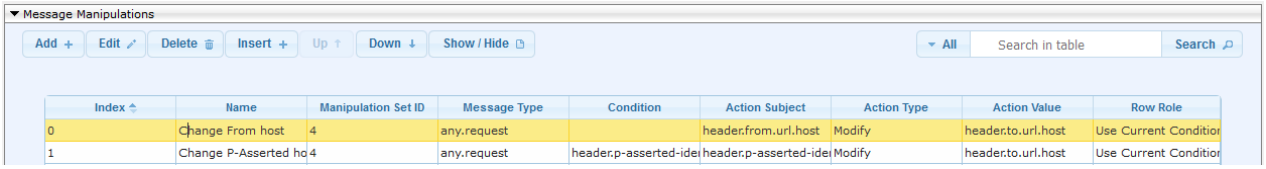


Figure 4-33: Example of Configured SIP Message Manipulation Rules

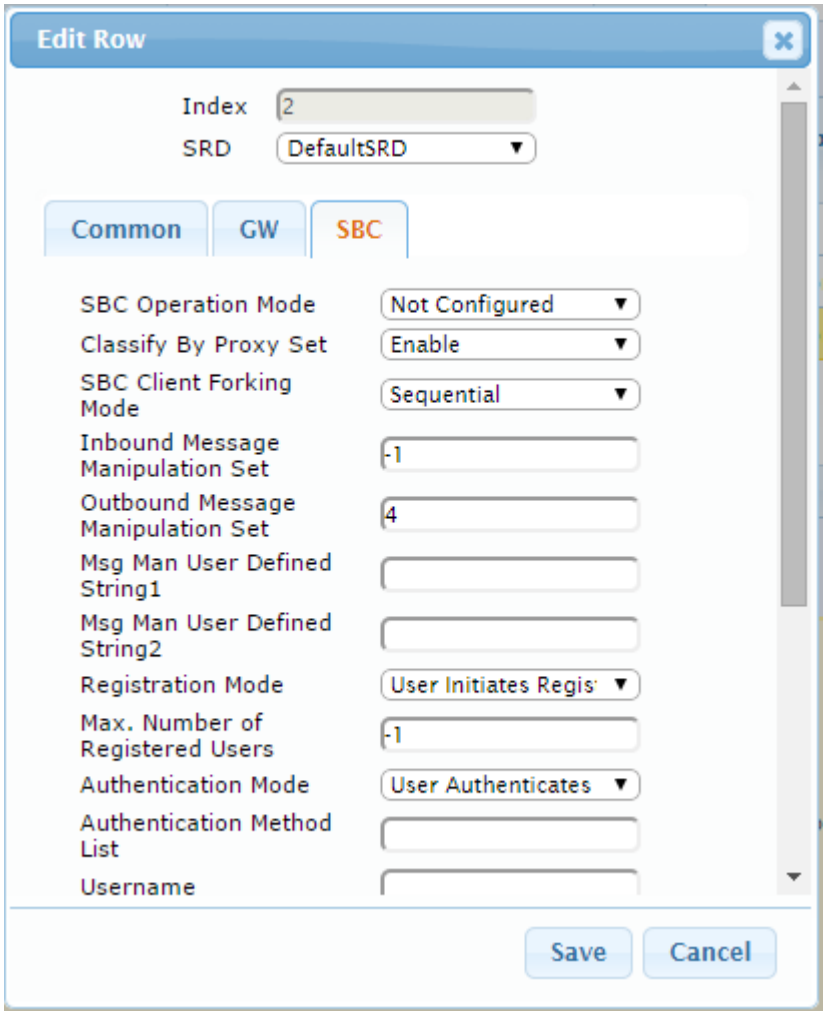


The table displayed below includes SIP message manipulation rules, which are bound together by commonality via the Manipulation Set ID 4, which are executed for messages sent to the BroadCloud SIP Trunk IP Group. These rules are specifically required to enable proper interworking between BroadCloud SIP 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.	

4. 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 4-34: 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.

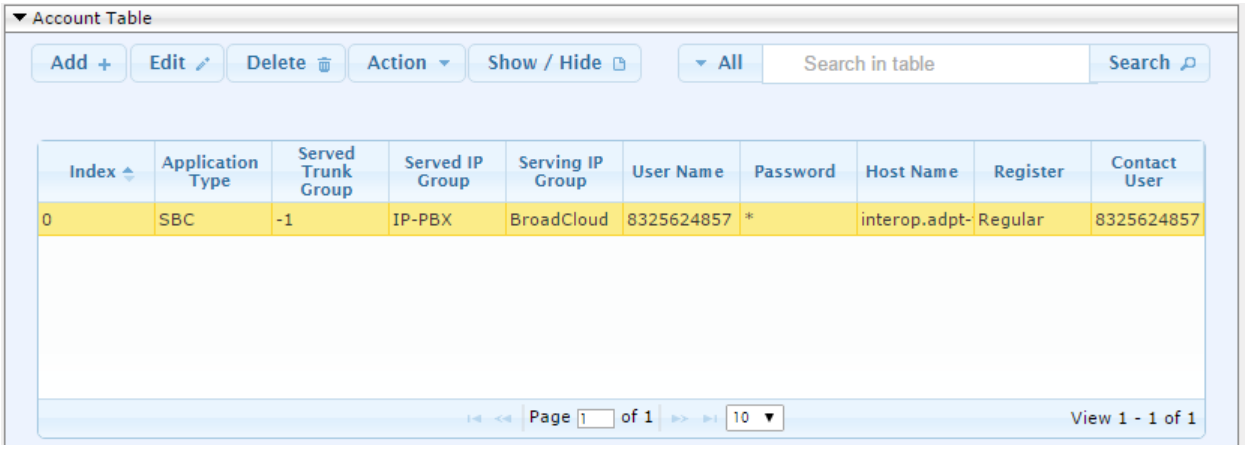
➢ **To configure a registration account:**

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 4-35: Configuring SIP Registration Account



4.12 Step 12: Miscellaneous Configuration

This section describes miscellaneous E-SBC configuration.

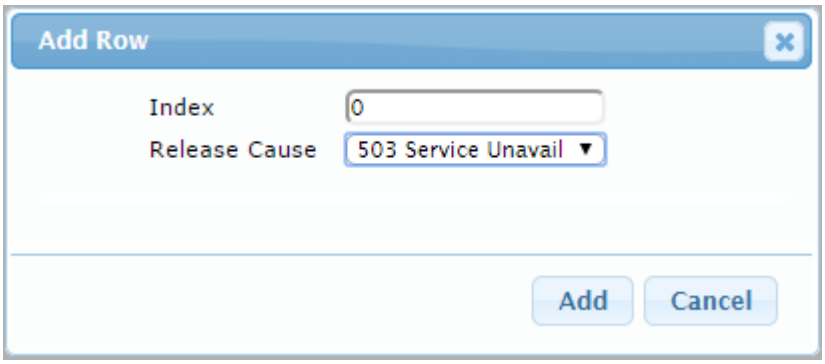
4.12.1 Step 12a: Configure SBC Alternative Routing Reasons

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 4-36: SBC Alternative Routing Reasons Table - Add Record



3. Click **Submit**.

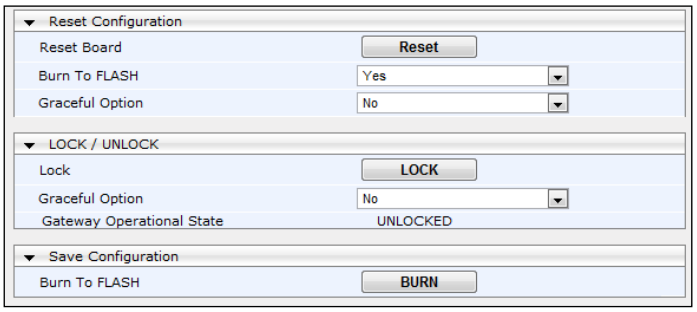
4.13 Step 13: Reset the E-SBC

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 4-37: Resetting the E-SBC



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

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 500 E-SBC
;HW Board Type: 69  FK Board Type: 85
;Serial Number: 8951962
;Slot Number: 1
;Software Version: 7.00A.049.003
;DSP Software Version: 5011AE3_R => 700.44
;Board IP Address: 10.234.68.240
;Board Subnet Mask: 255.254.0.0
;Board Default Gateway: 10.234.68.1
;Ram size: 496M   Flash size: 128M   Core speed: 300Mhz
;Num of DSP Cores: 0  Num DSP Channels: 0
;Num of physical LAN ports: 4
;Profile: NONE
;;;Key features:;Board Type: 85 ;Security: IPSEC MediaEncryption StrongEncryption
EncryptControlProtocol ;DSP Voice features: ;DATA features: ;Channel Type: RTP
DspCh=60 IPMediaDspCh=60 ;HA ;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 ;Control Protocols: CLI TRANSCODING=60 CODER-
TRANSCODING=60 SBC-SIGNALING=60 SBC-MEDIA=60 WebRTC MGCP SIP SBC=60 ;Default
features:;Coders: G711 G726;

;------  HW components------
;
; Slot # : Module type : # of ports
;-----------------------------------------------
;-----------------------------------------------

[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]

CallProgressTonesFilename = 'usa_tones_13.dat'

[WEB Params]

UseRProductName = 'Mediant 500 E-SBC'
LogoWidth = '145'
UseProductName = 1
HTTPSCipherString = 'RC4:EXP'

[SIP Params]

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, 10.234.68.240, 15, 10.234.68.1, "Voice", 10.234.35.20,
0.0.0.0, "vlan 1";
InterfaceTable 1 = 5, 10, 194.73.218.98, 3, 194.73.218.97, "WANSP", 8.8.8.8,
0.0.0.0, "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",
"$1$juq+86Wi9veioKH7/aj4q/z8rcWUk5LBl5TBnJqbmJrPzJmAh9HQhofSj4mOjImJiIyIpKD1oPWm8vL
58P749Ks=", 1, 0, 2, 15, 60, 200, "5defa220d918d4a93d2ecba9436e766a";
WebUsers 1 = "User",
"$1$lKT0pqWhra6p/6Wmq8aZkZTFkpaTnJ+YyJibzJnW04GHhIPVgdyJiIjdjtqM+ffy9fP28PKq/qvz9PS
o/ejp5bY=", 1, 0, 2, 15, 60, 50, "11bb2bd9dc4e498f75e0277a53e49d5c";

[ \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, 0, 0, 0, 0, 0, 8, 300,
400, 0, 0, 0, -1, 1, 0, 1, 3, 0, 2, 2, 1, 0, 0, 1, 0, 1, 1, 1, 1, 0, 0, 0, 0, 0, 0,
0, 1, 0, 1, 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, 1, 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, 1, 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", "Voice", "", 6000, 100, 6999, 1, "", "";
CpMediaRealm 1 = "MRWan", "WANSP", "", 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", "Voice", 2, 5060, 0, 0, "DefaultSRD", "", "default", -1,
0, 500, -1, 0, "MRLan", 0, -1, -1, -1, 0;
SIPInterface 1 = "BroadCloud", "WANSP", 2, 5060, 0, 0, "DefaultSRD", "", "default",
-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", "10.234.68.227", "", -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, "10.234.68.227:5060", 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 = "Add + for National Calls", "Default_SBCRoutingPolicy",
0, "IP-PBX", "BroadCloud", "*", "*", "001", "*", "*", "", 0, "Any", 0, 1, 2, 0,
255, "+", "", 0, "", "";
IPOutboundManipulation 1 = "Mitel_Ext", "Default_SBCRoutingPolicy", 0,
"BroadCloud", "IP-PBX", "*", "*", "*", "*", "*", "", 0, "Any", 0, 1, 9, 0, 255,
"44203409380", "", 0, "", "";
IPOutboundManipulation 2 = "For Anonymous", "Default_SBCRoutingPolicy", 0, "IP-
PBX", "BroadCloud", "*", "*", "*", "*", "*", "", 0, "Any", 0, 0, 0, 0, 255, "", "",
0, "", "";
IPOutboundManipulation 3 = "Change Outgoing Source", "Default_SBCRoutingPolicy", 0,
"IP-PBX", "BroadCloud", "*", "*", "*", "*", "*", "", 0, "Any", 0, 0, 11, 0, 255,
"832562485", "", 0, "", "";

[ \IPOutboundManipulation ]


[ CodersGroup0 ]

FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce,
CodersGroup0_CoderSpecific;
CodersGroup0 0 = "g711Alaw64k", 20, 255, -1, 0, "";

[ \CodersGroup0 ]


[ CodersGroup1 ]

FORMAT CodersGroup1_Index = CodersGroup1_Name, CodersGroup1_pTime,
CodersGroup1_rate, CodersGroup1_PayloadType, CodersGroup1_Sce,
CodersGroup1_CoderSpecific;
CodersGroup1 0 = "g711Alaw64k", 20, 0, -1, 0, "";
CodersGroup1 1 = "g711Ulaw64k", 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 1 = "g711Alaw64k", 20, 0, -1, 0, "";
CodersGroup2 2 = "g711Ulaw64k", 20, 0, -1, 0, "";
CodersGroup2 3 = "g7231", 30, 0, -1, 0, "";

[ \CodersGroup2 ]


[ CodersGroup4 ]

FORMAT CodersGroup4_Index = CodersGroup4_Name, CodersGroup4_pTime,
CodersGroup4_rate, CodersGroup4_PayloadType, CodersGroup4_Sce,
CodersGroup4_CoderSpecific;
CodersGroup4 0 = "t38fax", 255, 255, -1, 255, "";

[ \CodersGroup4 ]


[ AllowedCodersGroup1 ]

FORMAT AllowedCodersGroup1_Index = AllowedCodersGroup1_Name;
AllowedCodersGroup1 0 = "g711Ulaw64k";
AllowedCodersGroup1 1 = "g711Alaw64k";

[ \AllowedCodersGroup1 ]


[ AllowedCodersGroup2 ]

FORMAT AllowedCodersGroup2_Index = AllowedCodersGroup2_Name;
AllowedCodersGroup2 0 = "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 ]


[ 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 ]
```

International Headquarters
1 Hayarden Street,
Airport City
Lod 7019900, Israel
Tel: +972-3-976-4000
Fax: +972-3-976-4040

AudioCodes Inc.
27 World's Fair Drive,
Somerset, NJ 08873
Tel: +1-732-469-0880
Fax: +1-732-469-2298

Contact us: www.audiocodes.com/info
Website: www.audiocodes.com

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