

Connecting PBX to BroadSoft SIP Trunk using AudioCodes Mediant PRI Gateway

Version 7.2



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1 Introduction

This document describes how to set up AudioCodes' PRI Gateway (hereafter, referred to as *Gateway*) for interworking between BroadSoft's SIP Trunk and PBX environment. For detailed information on each AudioCodes Gateway, refer to the corresponding *User's Manual* and *Hardware Installation Manual*.

1.1 Component Information

Table 1-1: Component Information

AudioCodes Gateway Version	
Gateway Vendor	AudioCodes
Models	Mediant 500; Mediant 800B; Mediant 1000B
Software Version	7.20A.104.001
Protocol	<ul style="list-style-type: none">▪ SIP/UDP (to the BroadSoft SIP Trunk)▪ PRI (to the PBX)
BroadSoft SIP Trunking Version	
Vendor/Service Provider	
SSW Model/Service	
Software Version	
Protocol	SIP/UDP

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2 Obtain Software Files

Download the certified firmware file (*firmware_xxx.cmp*), configuration file (*configuration_xxx.ini*) and Call Progress Tones file (*call_progress_uk.dat*), of the specific AudioCodes PRI Gateway (referred as “xxx”), from Support Centre.

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3 Cable Device for Initial Access

The device's factory default IP address for operations, administration, maintenance, and provisioning (OAMP) is **192.168.0.2/24** (default gateway 192.168.0.1).

1. Change your PC's IP address and subnet mask to correspond with the device's default IP address.
2. Cable as follows:
 - Connect the PC to the device's Ethernet port labelled **Port 1** (left-most port).
 - Ground the device using the grounding lug.
 - Using the supplied AC power cable, connect the device's AC port to a standard electrical wall outlet.

Figure 3-1: Mediant 500 – Front and Rear Panels

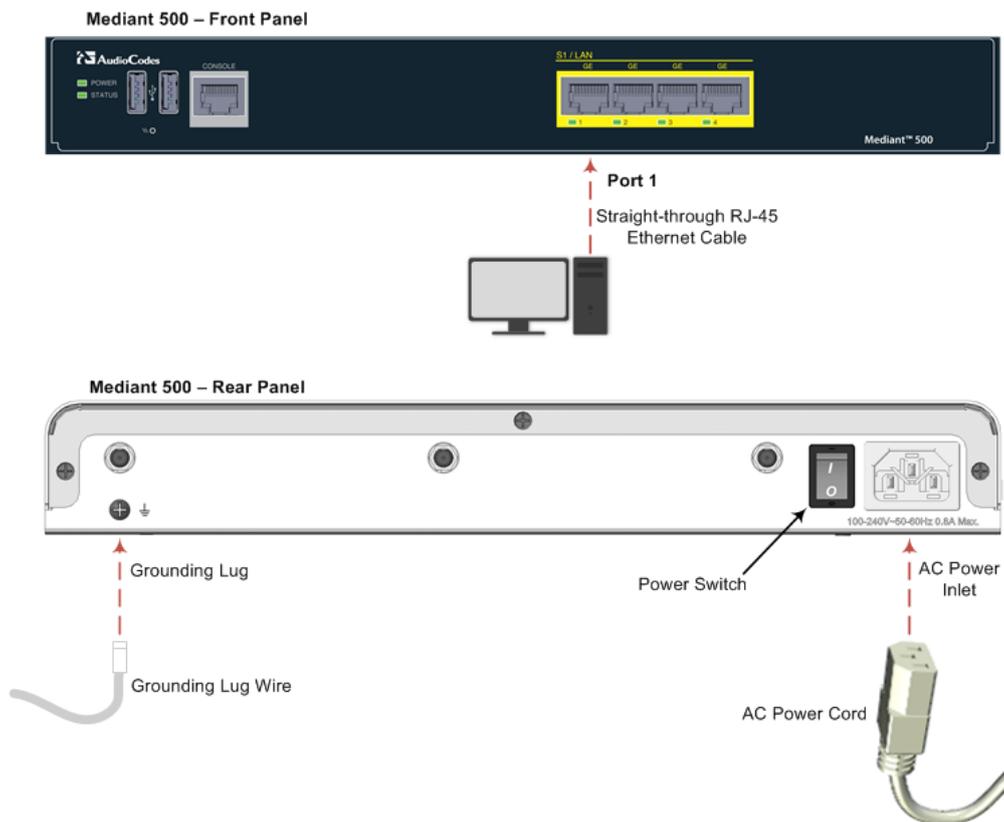


Figure 3-2: Mediant 800 – Front and Rear Panels

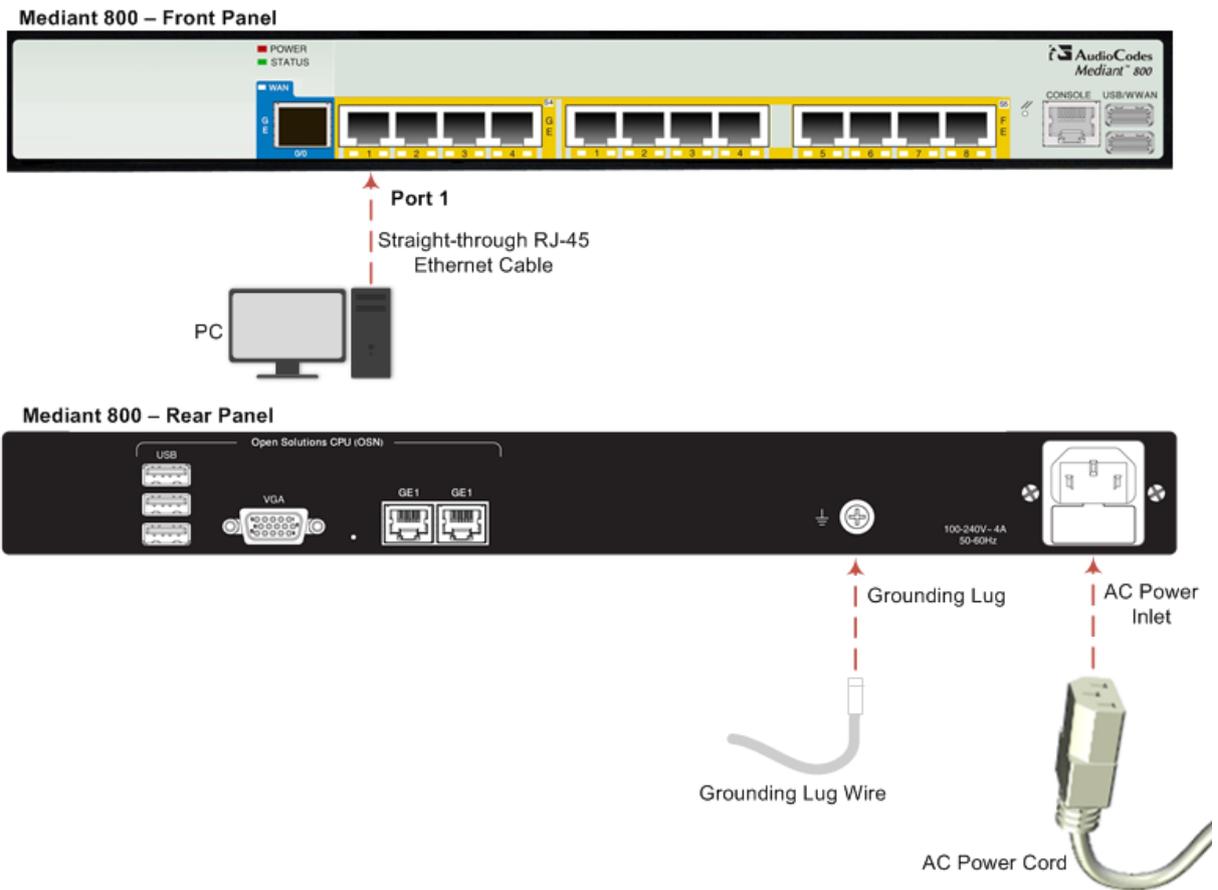
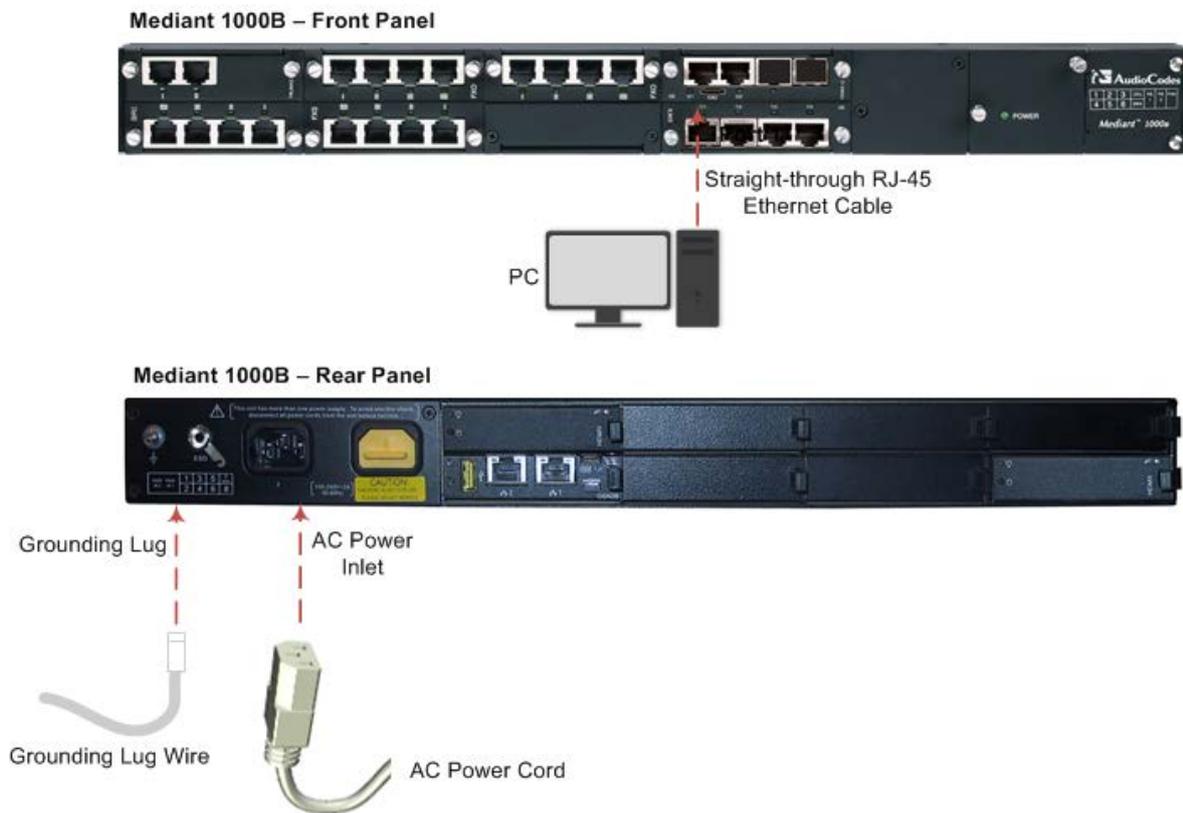


Figure 3-3: Mediant 1000B – Front and Rear Panels



3. Access the device's Web-based management interface:
 - a. On your PC, start your Web browser and then in the URL address field, enter the device's default IP address; the following appears:

Figure 3-4: Web Login

Web Login

Username

Password

Remember Me

- b. In the 'Username' and 'Password' fields, enter the default login username ("Admin") and password ("Admin"), and then click **Login**.

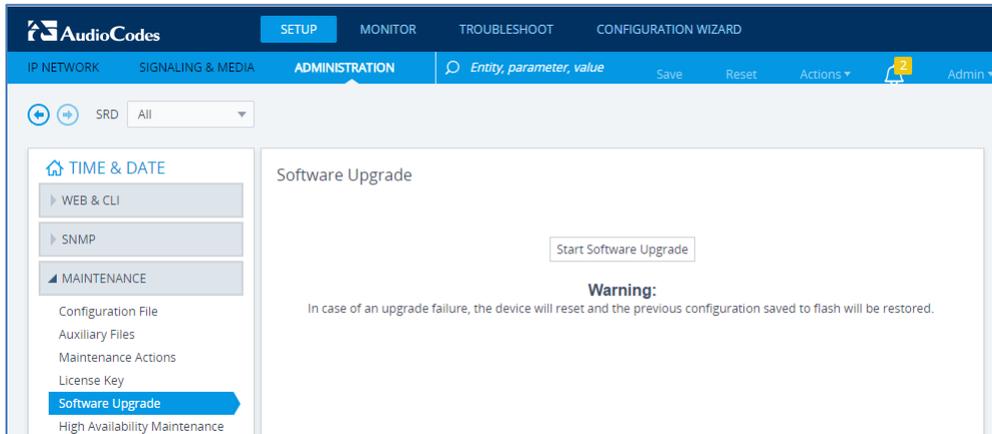
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4 Upload Software to Device

Upload the certified software files, which you downloaded in Section [Obtain Software Files](#), to the device:

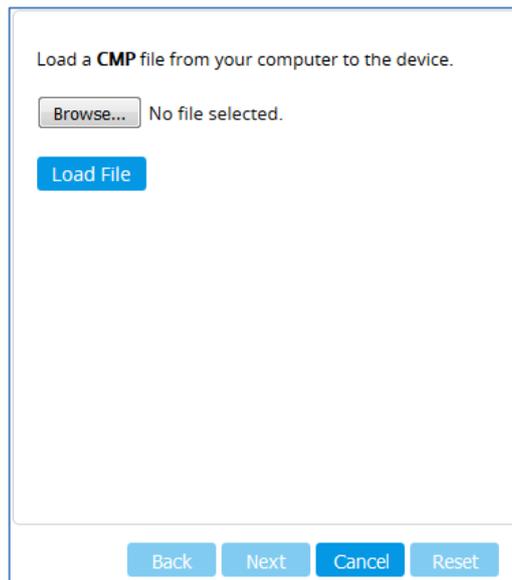
- In the Web interface, open the Software Upgrade Wizard:
 - Toolbar:** From the **ACTIONS** drop-down menu, choose **Software Upgrade**.
 - Navigation tree:** **Setup** menu > **Administration** tab > **Maintenance** folder > **Software Upgrade**.

Figure 4-1: Software Upgrade Page



- Click **Start Software Upgrade**; the wizard starts, prompting you to load a .cmp file:

Figure 4-2: Loading CMP File in Software Upgrade Wizard

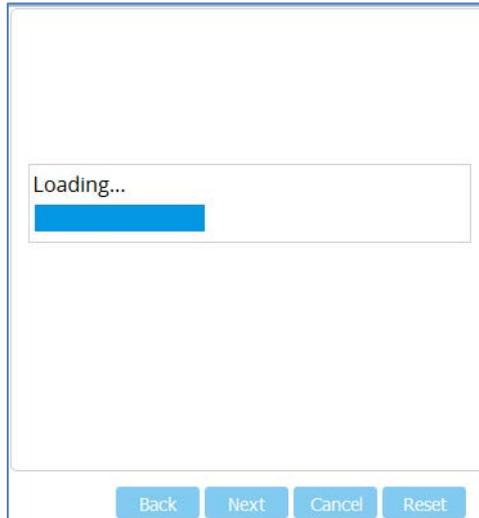


Note: At this stage, you can quit the Software Upgrade wizard without having to reset the device, by clicking **Cancel**. However, if you continue with the wizard and start loading the CMP file, the upgrade process must be completed with a device reset.

- Click **Browse**, and then navigate to and select the .cmp file.

- Click **Load File**; the device begins to install the .cmp file and a progress bar displays the status of the loading process:

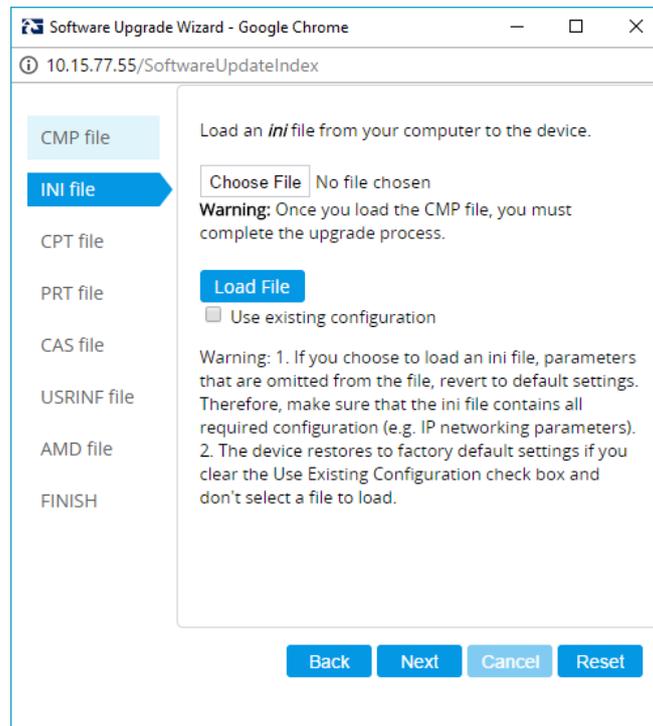
Figure 4-3: CMP File Loading Progress Bar



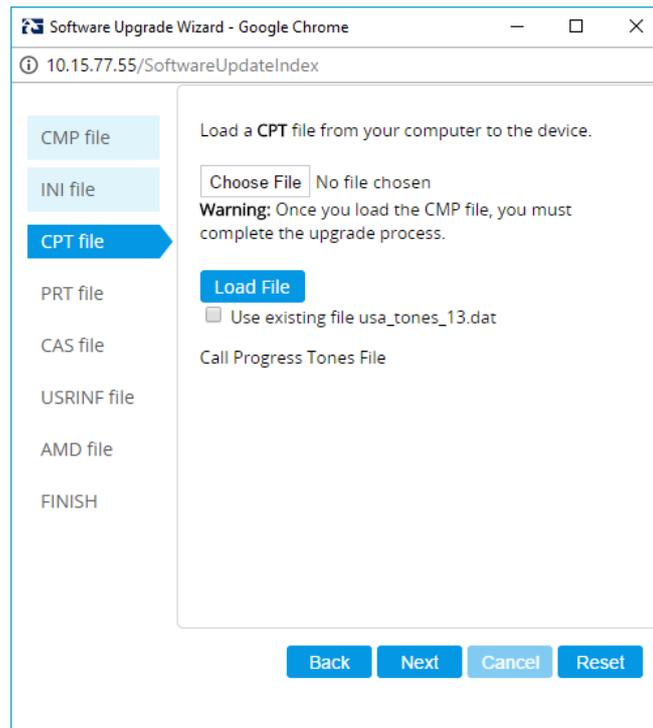
A message is displayed to inform you that the file has been loaded.

- When successfully loaded, click **Next** to access the wizard page for loading the *ini* file.
- Clear the **Use existing configuration** option, click **Browse** to select the configuration file (.ini) on your PC, and then click **Load File** to load the file:

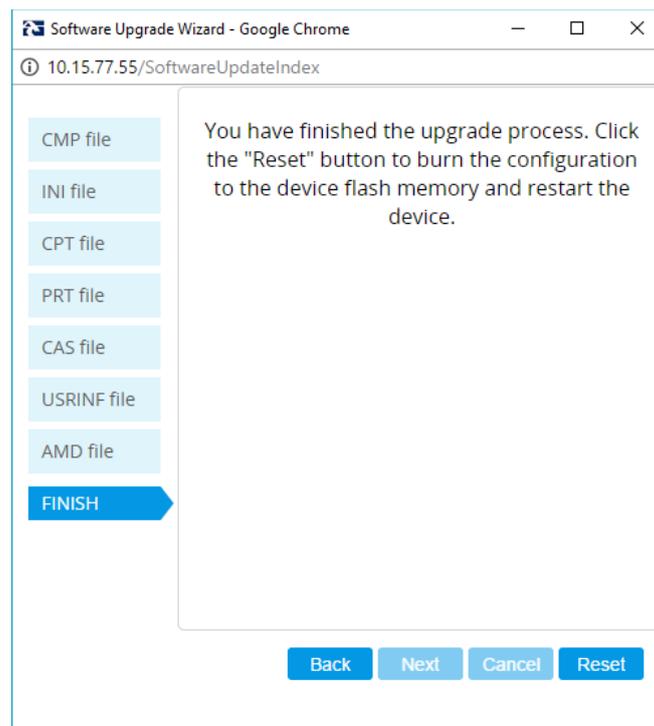
Figure 4-4: Load an INI File in the Software Upgrade Wizard



- Click **Next** to access the wizard page for loading the Call Progress Tones (CPT) file.
- Click **Browse** to select the **CPT** file on your PC, and then click **Load File** to load the file:

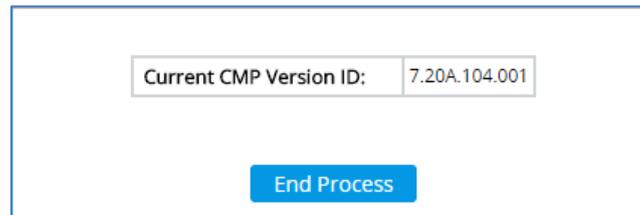
Figure 4-5: Load an CPT File in the Software Upgrade Wizard

9. Keep clicking **Next** until the last Wizard page appears (the **FINISH** button is highlighted in the left pane) and the following message appears:

Figure 4-6: Software Upgrade Wizard – Google Chrome

10. Click **Reset** to install the files by saving them on the device's flash memory with a device. Once complete, the following is displayed:

Figure 4-7: Current CMP Version ID



11. Click **End Process** to close the wizard, and then log in again to the Web interface.
12. Enter your login username and password (**Admin, Admin** respectively), and then click **Login**; a message box appears informing you of the new .cmp file version.
13. Click **OK**; the Web interface becomes active, reflecting the upgraded device.

5 Configure Device

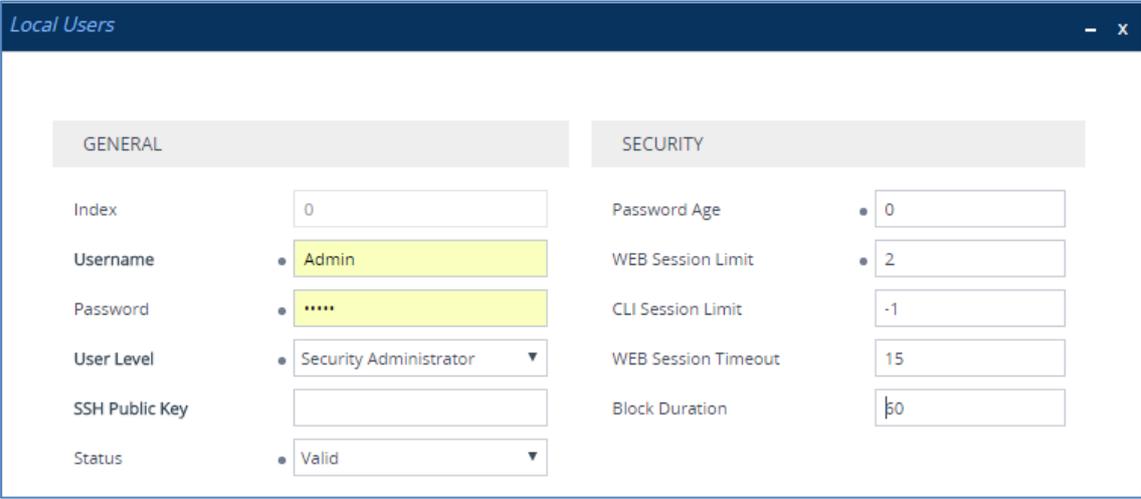
This section describes device configuration.

5.1 Change Default Management User Login Passwords

To secure access to the device's Web management interface, follow these guidelines:

- The device is shipped with a default **Security Administrator** access-level user account – username 'Admin' and password 'Admin'. This user has full read-write access privileges to the device. It is recommended to change the default password to a hard-to-hack string. The login username and password are configured in the Web Interface's Local Users page (**Setup** menu > **Administration** tab > **Web & CLI** folder > **Local Users**) using the 'Password' and 'Apply' fields:

Figure 5-1: Changing Password of Default Security Administrator User



The screenshot shows the 'Local Users' configuration window with two tabs: 'GENERAL' and 'SECURITY'. The 'GENERAL' tab is active, showing the following fields:

Field	Value
Index	0
Username	Admin
Password
User Level	Security Administrator
SSH Public Key	
Status	Valid

The 'SECURITY' tab is also visible, showing the following fields:

Field	Value
Password Age	0
WEB Session Limit	2
CLI Session Limit	-1
WEB Session Timeout	15
Block Duration	60

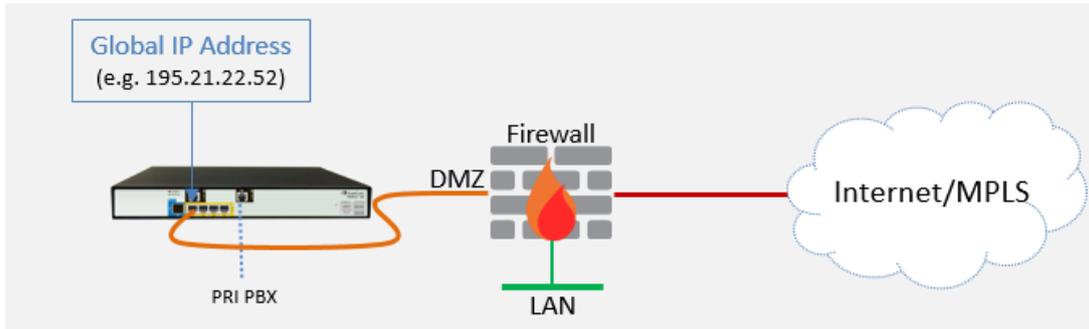
- The device is shipped with a default **Monitor** access-level user account - username and password: 'User' who has read access only and page viewing limitations but can view certain SIP settings such as proxy server addresses. Therefore, to prevent an attacker from obtaining sensitive SIP settings that could result in possible call theft etc., change its default login password to a hard-to-hack string.

5.2 Configure a Network Interface for the Device

You can connect the device to the DMZ network using one of the following methods:

- **Method A:** (Preferred method) A global IP address is provided to the device (**without NAT**):

Figure 5-2: Method A

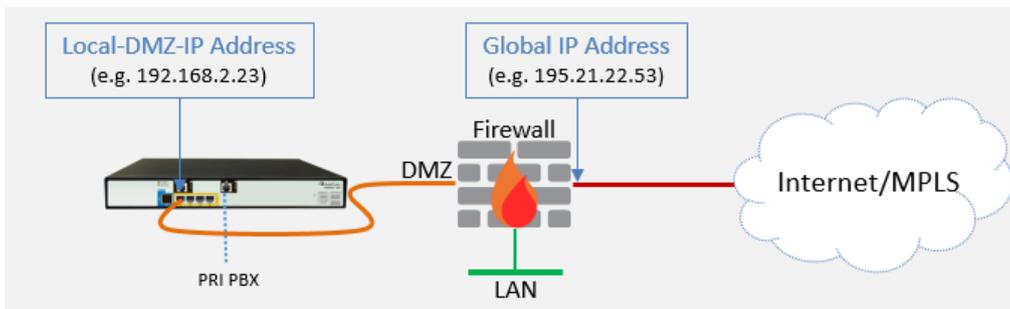


The Enterprise firewall is configured with rules, for example:

Original		
Source	Destination	Ports/Service
<any> (e.g. ITSP)	Global IP Address (public address)	SIP service: 5060 / UDP RTP service: 6000-8500 / UDP

- **Method B:** A local DMZ IP address **behind NAT**:

Figure 5-3: Method B



The firewall is configured with rules, for example:

Original			Translated		
Source	Destination	Ports/Service	Source	Destination	Ports/Service
<any> (e.g. ITSP)	Global IP Address (public address)	SIP service: 5060 / UDP RTP service: 6000-8500 / UDP	<any> (e.g. ITSP)	Local DMZ IP Address	<as original>

NAT rules (port forwarding):

Source	Destination	Ports/Service	Source	Destination	Ports/Service
<any> (e.g. ITSP)	Global IP Address (public address)	SIP service: 5060 / UDP RTP service: 6000-8500 / UDP	<any> (e.g. ITSP)	Local DMZ IP Address	<as original>
Local DMZ IP Address	<any> (e.g. ITSP)	SIP service: 5060 / UDP RTP service: 6000-8500 / UDP	Global IP Address (public address)	<any> (e.g. ITSP)	<as original>

5.2.1 Configure Network Interface

Configure network interface, as described below:

1. Open the IP Interfaces table (**Setup** menu > **IP Network** tab > **Core Entities** folder > **IP Interfaces**).
2. Modify the existing network interface ('Voice'):
 - 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
Name	Voice (arbitrary descriptive name, you may change it)
Application Type	OAMP + Media + Control (leave as is)
Ethernet Device	vlan 1
IP Address	If working in <u>Method A</u> : Global-IP-Address (public address) If working in <u>Method B</u> : Local-DMZ-IP-Address
Prefix Length	Subnet mask in bits, e.g.28 (for 255.255.255.240)
Default Gateway	The default gateway IP address (In Method B: router's IP address)
Primary DNS Server IP Address	Primary DNS IP address
Secondary DNS Server IP Address	Secondary DNS IP address (optional)

3. Click **Apply**.

The figure below shows an example of a configured IP network interface.

Figure 5-4: Example of a Configured Network Interface in IP Interfaces Table

The screenshot shows a web interface for managing IP interfaces. At the top, it says 'IP Interfaces (1)'. Below this are buttons for '+ New', 'Edit', and a trash icon. There is also a pagination control showing 'Page 1 of 1' and a 'Show 10 records per page' dropdown. The main table has the following data:

INDEX	NAME	APPLICATION TYPE	INTERFACE MODE	IP ADDRESS	PREFIX LENGTH	DEFAULT GATEWAY	PRIMARY DNS	SECONDARY DNS	ETHERNET DEVICE
0	Voice	OAMP + Med	IPv4 Manual	10.15.77.77	16	10.15.0.1	10.15.27.1		vlan 1

5.2.2 Configure NAT



Note:

- NAT configuration is applicable only if you are behind a firewall NAT (see [Method B](#)).
- The NAT IP Address is the Global-IP-address used in front of the firewall facing the BroadSoft service. If the DMZ holds the global-IP-address (no NAT is performed by the firewall) and the Gateway is already assigned the Global-IP-address as its address, skip this NAT configuration.

Configure the global IP address as follows:

1. Open the NAT Translation table (**Setup** menu > **IP Network** tab > **Core Entities** folder > **NAT Translation**).
2. Click **New**; the following dialog appears:

Figure 5-5: NAT Translation Table - Dialog Box

3. Use the table below as reference when configuring a NAT translation rule.

Figure 5-6: NAT Translation Table Parameter Descriptions

Parameter	Description
Index	0
Source Interface	Voice (the interface to apply this rule to)
Target IP Address	Global-IP-address . Defines the global (public) IP address.
Source Start Port	(leave empty)
Source End Port	(leave empty)
Target Start Port	(leave empty)
Target End Port	(leave empty)

4. Click **Apply**.

5.3 Configure General SIP Parameters

This step identifies the device configuration needed to support the SIP General Parameters configuration.

5.3.1 Configure SIP General Settings

This step shows how to configure the SIP General Settings.

➤ **To configure the SIP General Settings parameters:**

1. Open the SIP Proxy & Registration Parameters page (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **SIP Definitions General Settings**).
2. Configure following parameters:

Parameter	Value
Gateway Settings	
Source Header For Called Number	Use To header
Gateway Session Expires	
Session Expires Method	Update
Disconnect Supervision	
Broken Connection Mode	Ignore

Figure 5-7: Configuring SIP General Settings Parameters

The screenshot shows the 'SIP Definitions General Settings' configuration page in the AudioCodes Mediant Gateway. The page is organized into four main sections:

- GENERAL:** Includes 'Send Reject (503) upon Overload' (Enable), 'Retry-After Time' (0), 'Fake Retry After' (0), and 'X-Channel Header' (Disable).
- GATEWAY SESSION EXPIRES:** Includes 'Session-Expires Time' (0), 'Minimum Session-Expires' (90), 'Session Expires Method' (UPDATE), and 'Session Expires Disconnect Time' (32).
- GATEWAY SETTINGS:** Includes 'PRACK Mode' (Supported), 'Early 183' (Disable), '183 Message Behavior' (Progress), '3xx Behavior' (Forward), 'Call Transfer using re-INVITEs' (Disable), 'First Call Ringback Tone ID' (-1), 'Enable Delayed Offer' (Disable), 'Source Header For Called Number' (Use To header), 'Verify Received VIA' (Disable), and 'Reject Cancel after Connect' (Disable).
- DISCONNECT SUPERVISION:** Includes 'Broken Connection Mode' (Ignore) and 'Broken Connection Timeout [100 msec]' (100).

Arrows in the screenshot point to the 'Source Header For Called Number' (Use To header), 'Session Expires Method' (UPDATE), and 'Broken Connection Mode' (Ignore) settings.

3. Click **Apply**.

5.3.2 Configure SIP Message Structure Parameters

This step shows how to add SIP P-Asserted Header.

- **To configure the SIP Message Structure parameters:**
 1. Open the SIP Proxy & Registration Parameters page (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **Message Structure**).
 2. From the 'Asserted Identity Mode' dropdown, select **Add P-Asserted-Identity**.

Figure 5-8: Configuring SIP Message Structure Parameters

The screenshot shows the 'Message Structure' configuration page in the AudioCodes management interface. The page is divided into two main sections: 'GENERAL' and 'GATEWAY SETTINGS'. The 'Asserted Identity Mode' dropdown in the 'GATEWAY SETTINGS' section is highlighted with a red arrow, indicating that 'Add P-Asserted-Identity' has been selected.

GENERAL	GATEWAY SETTINGS
Display Default SIP Port: <input type="text" value="5060"/>	Enable Remote Party ID: <input type="text" value="Disable"/>
Enable Reason Header: <input type="text" value="Enable"/>	Enable P-Associated-URI Header: <input type="text" value="Disable"/>
Use Tel URI for Asserted Identity: <input type="text" value="Disable"/>	Subject: <input type="text" value=""/>
Skype Capabilities Header: <input type="text" value="Disable"/>	Enable History-Info Header: <input type="text" value="Disable"/>
	Enable Contact Restriction: <input type="text" value="Disable"/>
	User-Agent Information: <input type="text" value="AudioCodes M800 Gateway"/>
	X-Channel Header: <input type="text" value="Disable"/>
	Asserted Identity Mode: <input type="text" value="Add P-Asserted-Identity"/>
	Enable P-Charging Vector: <input type="text" value="Disable"/>

3. Click **Apply**.

5.3.3 Configure Registration Parameters

This step shows how to configure the SIP Proxy and Registration parameters, including configuring a Proxy Name, Registrar Name, Registration and Subscription modes.

➤ **To configure the SIP Proxy & Registration parameters:**

1. Open the SIP Proxy & Registration Parameters page (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **Proxy & Registration**).
2. Configure following parameters:

Parameter	Value
Gateway Name	uk.ic.sipconnect.hipcom.co.uk
Use Gateway Name for OPTIONS	Yes
Use Default Proxy	Use Proxy
Proxy Name	Per SIP Trunk requirement
Always Use Proxy	Enable
Gateway Authentication	
User Name	Trunk Group Pilot User
Password	Trunk Group Pilot User Password
Authentication Mode	Per Gateway
Gateway Registration	
Enable Registration	Enable
Registrar Name	uk.ic.sipconnect.hipcom.co.uk

Figure 5-9: Configuring Proxy & Registration Parameters

The screenshot shows the 'Proxy & Registration' configuration page in the AudioCodes management console. The left sidebar contains a 'TOPOLOGY VIEW' menu with categories like CORE ENTITIES, GATEWAY, MEDIA, CODERS & PROFILES, SIP DEFINITIONS, MESSAGE MANIPULATION, INTRUSION DETECTION, and SIP RECORDING. The 'SIP DEFINITIONS' section is expanded, showing 'Proxy & Registration' as the active configuration.

The main configuration area is divided into several sections:

- GENERAL:**
 - Redundancy Mode: Parking
 - Proxy IP List Refresh Time: 60
 - Proxy DNS Query Type: A-Record
 - Number of RTX Before Hot-Swap: 3
 - Use Proxy IP as Host: Disable
 - Enable User-Information Usage: Disable
 - Add Empty Authorization Header: Disable
 - Gateway Name: uk.ic.sipconnect.hipcom.ci
 - Use Gateway Name for OPTIONS: Yes
 - Challenge Caching Mode: None
- GATEWAY PROXY:**
 - Use Default Proxy: Use Proxy
 - Proxy Set Table: uk.ic.sipconnect
 - Prefer Routing Table: No
 - Use Routing Table for Host Names and Profiles: Disable
 - Always Use Proxy: Enable
 - Enable Fallback to Routing Table: Disable
- REGISTRATION:**
 - Registration Time: 180
 - Re-registration Timing [%]: 50
 - Registration Retry Time: 30
 - Registration Time Threshold: 0
 - Re-register On INVITE Failure: Disable
 - Re-register On Connection Failure: Disable
- GATEWAY AUTHENTICATION:**
 - User Name: 1234567890
 - Password: *
 - Cnonce: Default_Cnonce
 - Authentication Mode: Per Gateway
- GATEWAY REGISTRATION:**
 - Enable Registration: Enable
 - Registrar Name: uk.ic.sipconnect
 - Registrar IP Address: (empty)

At the bottom of the configuration area, there are 'Cancel' and 'APPLY' buttons.

5.3.4 Configure the SIP Trunk IP Address

This step shows how to configure the Proxy Set toward SIP Trunk. 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.

➤ **To configure Proxy Set:**

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**).
2. Edit the Proxy Set 0 (you can identify it by the 'Proxy Name' field).

Parameter	Value
Index	0
Proxy Keep-Alive	Using Options

- a. Click the **Proxy Address** link located below the table; the Proxy Address table opens.
- b. Click **New**; the following dialog box appears:

Figure 5-10: Configuring Proxy Address for SIP Trunk

- c. Configure the address of the Proxy Set per the parameters described in the table below.

Parameter	Value
Index	0
Proxy Address	uk.ic.sipconnect.hipcom.co.uk:5060 (SIP Trunk IP address / FQDN and destination port)
Transport Type	UDP (Network transport type for the SIP Trunk)

- d. Click **Apply**.

5.3.5 Configure Message Manipulation Rules

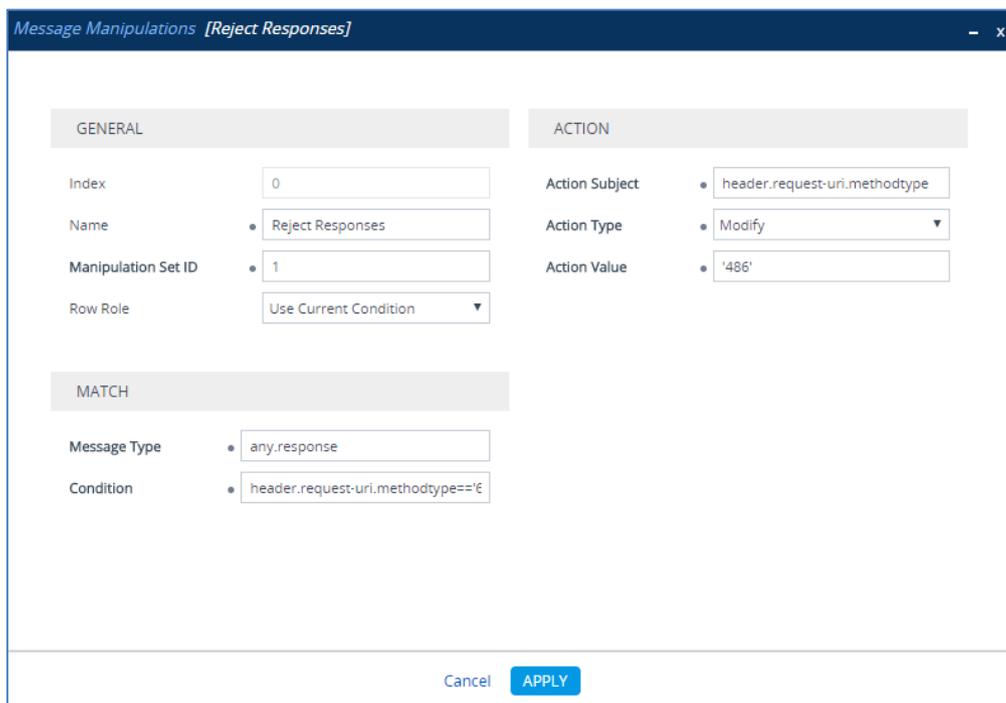
This step describes how to configure SIP message manipulation rules, which can include insertion, removal, and/or modification of SIP headers. Manipulation rules are grouped into Manipulation Sets, enabling you to apply multiple rules to the same SIP message (IP entity). Configured SIP message manipulation rules will be assigned as gateway outbound message manipulation set and will be applied to all outbound messages.

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

1. Open the Message Manipulations page (**Setup** menu > **Signaling & Media** tab > **Message Manipulation** folder > **Message Manipulations**).
2. Configure a new manipulation rule (Manipulation Set 1) for BroadSoft SIP Trunk. This rule applies to response messages sent to the BroadSoft SIP Trunk for Rejected Calls initiated by the PBX. This replaces the '503' and '603' method types with the value '486', because BroadSoft SIP Trunk not recognizes these method types.

Parameter	Value
Index	0
Name	Reject Responses
Manipulation Set ID	1
Message Type	any.response
Condition	header.request-uri.methodtype=='603' OR header.request-uri.methodtype=='503'
Action Subject	header.request-uri.methodtype
Action Type	Modify
Action Value	'486'

Figure 5-11: Configuring SIP Message Manipulation Rule 0 (for BroadSoft SIP Trunk)



The screenshot shows the configuration window for a SIP message manipulation rule named "Reject Responses". The window is divided into three main sections: GENERAL, MATCH, and ACTION.

- GENERAL:**
 - Index: 0
 - Name: Reject Responses
 - Manipulation Set ID: 1
 - Row Role: Use Current Condition
- MATCH:**
 - Message Type: any.response
 - Condition: header.request-uri.methodtype=='603' OR header.request-uri.methodtype=='503'
- ACTION:**
 - Action Subject: header.request-uri.methodtype
 - Action Type: Modify
 - Action Value: '486'

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

- Configure another manipulation rule (Manipulation Set 1) for BroadSoft SIP Trunk. This rule applies to messages sent to the BroadSoft SIP Trunk in a call forward scenario. This add prefix to the user part of SIP Diversion Header to complete to the full number.

Parameter	Value
Index	1
Name	Full # in Diversion
Manipulation Set ID	1
Message Type	any.request
Condition	header.diversion.url.user len== '4'
Action Subject	header.diversion.url.user
Action Type	Add Prefix
Action Value	'44203621'

Figure 5-12: Configuring SIP Message Manipulation Rule 1 (for BroadSoft SIP Trunk)

The screenshot shows a configuration window titled "Message Manipulations [Full # in Diversion]". The window is divided into three main sections: GENERAL, ACTION, and MATCH.

GENERAL Section:

- Index: 1
- Name: Full # in Diversion
- Manipulation Set ID: 1
- Row Role: Use Current Condition

ACTION Section:

- Action Subject: header.diversion.url.user
- Action Type: Add Prefix
- Action Value: '44203621'

MATCH Section:

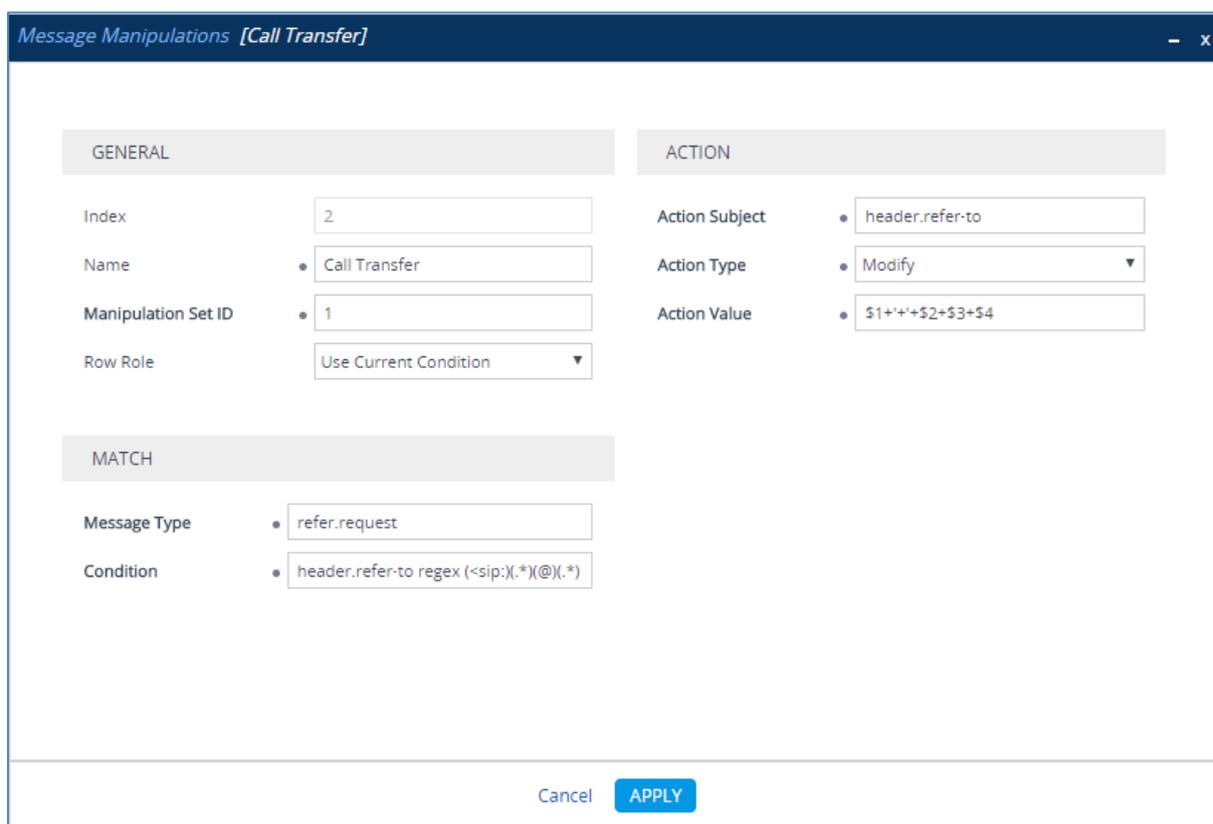
- Message Type: any.request
- Condition: header.diversion.url.user len== '4'

At the bottom of the window, there are two buttons: "Cancel" and "APPLY".

4. Configure another manipulation rule (Manipulation Set 1) for BroadSoft SIP Trunk. This rule applies to messages sent to the BroadSoft SIP Trunk in a call transfer scenario. This will add '+' prefix to the user part of SIP Refer-To Header to complete the BroadSoft SIP Trunk number conversion.

Parameter	Value
Index	2
Name	Call Transfer
Manipulation Set ID	1
Message Type	refer.request
Condition	header.refer-to regex (<sip:)(.*)((@)(.))
Action Subject	header.refer-to
Action Type	Modify
Action Value	\$1+'+'\$2+\$3+\$4

Figure 5-13: Configuring SIP Message Manipulation Rule 2 (for BroadSoft SIP Trunk)



Message Manipulations [Call Transfer]

GENERAL		ACTION	
Index	2	Action Subject	header.refer-to
Name	Call Transfer	Action Type	Modify
Manipulation Set ID	1	Action Value	\$1+'+'\$2+\$3+\$4
Row Role	Use Current Condition		

MATCH	
Message Type	refer.request
Condition	header.refer-to regex (<sip:)(.*)((@)(.))

Cancel **APPLY**

- Configure another manipulation rule (Manipulation Set 1) for BroadSoft SIP Trunk. This rule applies to messages sent to the BroadSoft SIP Trunk in a call forward scenario. This replaces the user part of the SIP From Header with the value from the SIP Diversion Header.

Parameter	Value
Index	3
Name	Call Forward
Manipulation Set ID	1
Message Type	invite
Condition	header.diversion exists
Action Subject	header.from.url.user
Action Type	Modify
Action Value	header.diversion.url.user

Figure 5-14: Configuring SIP Message Manipulation Rule 3 (for BroadSoft SIP Trunk)

The screenshot shows a configuration window titled "Message Manipulations [Call Forward]". The window is divided into three main sections: GENERAL, ACTION, and MATCH. At the bottom, there are "Cancel" and "APPLY" buttons.

GENERAL

- Index: 3
- Name: Call Forward
- Manipulation Set ID: 1
- Row Role: Use Current Condition

ACTION

- Action Subject: header.from.url.user
- Action Type: Modify
- Action Value: header.diversion.url.user

MATCH

- Message Type: invite
- Condition: header.diversion exists

Figure 5-15: Example of Configured SIP Message Manipulation Rules

INDEX	NAME	MANIPULATION SET ID	MESSAGE TYPE	CONDITION	ACTION SUBJECT	ACTION TYPE	ACTION VALUE	ROW ROLE
0	Reject Responses	1	any.response	header.request-u	header.request-u	Modify	'486'	Use Current Con
1	Full # in Diversion	1	any.request	header.diversion.	header.diversion.	Add Prefix	'44203621'	Use Current Con
2	Call Transfer	1	refer.request	header.refer-to re	header.refer-to	Modify	\$1+'+'\$2+\$3+\$4	Use Current Con
3	Call Forward	1	invite	header.diversion	header.from.url	Modify	header.diversion.	Use Current Con

The table displayed below includes SIP message manipulation rules which are grouped together under Manipulation Set ID 1 and which are executed for messages sent to the BroadSoft SIP Trunk. These rules are specifically required to enable proper interworking between BroadSoft SIP Trunk and 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 response messages sent to the BroadSoft SIP Trunk for Rejected Calls initiated by the PBX. This replaces the '503' and '603' method types with the value '486'.	The BroadSoft SIP Trunk not recognizes these method types and continue to try to setup call to the PBX.
1	This rule applies to messages sent to the BroadSoft SIP Trunk in a call forward scenario. This add prefix to the user part of SIP Diversion Header to complete to the full number.	If the PBX is configured with endpoints in 4-digits format in the Call Forward scenario, SIP Diversion Header will present a 4-digit number, that will cause a problem in the Forward Call setup.
2	This rule applies to messages sent to the BroadSoft SIP Trunk in a call transfer scenario. This will add '+' prefix to the user part of SIP Refer-To Header.	For complete the BroadSoft SIP Trunk number conversion.
3	This rule applies to messages sent to the BroadSoft SIP Trunk in a call forward scenario. This replaces the user part of the SIP From Header with the value from the SIP Diversion Header.	For Call Forward scenarios, BroadSoft SIP Trunk needs that User part in SIP From Header will be defined number. In order to do this, User part of the SIP From Header replaced with the value from the SIP Diversion Header.

5.4 Configure Coders

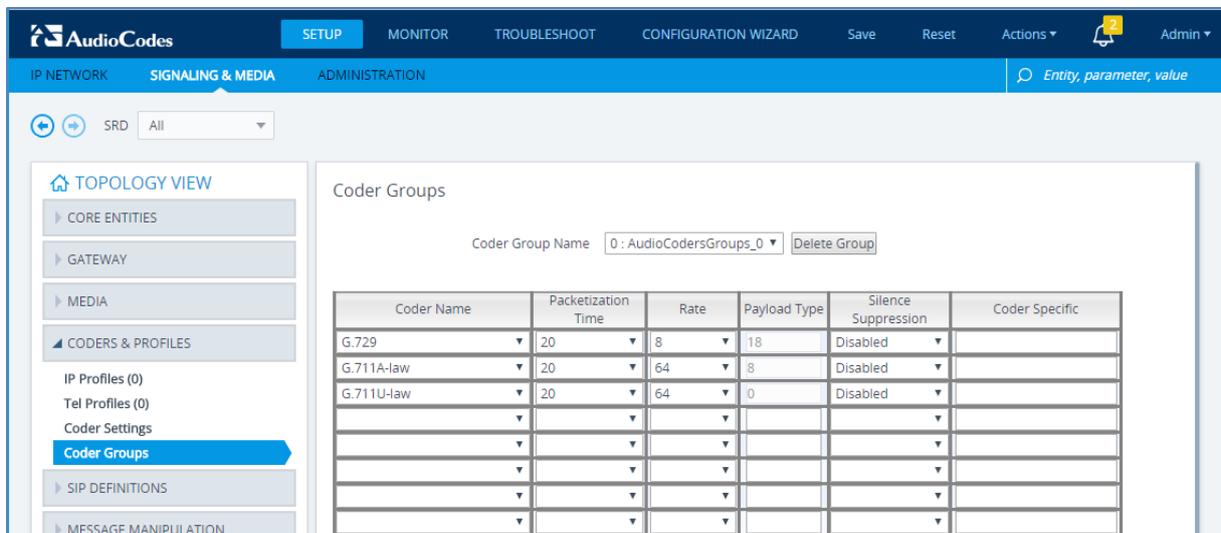
This step describes how to configure coders (termed *Coder Group*) per BroadSoft SIP Trunk requirement.

➤ **To configure coders:**

1. Open the Coder Groups table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **Coder Groups**).
2. Configure a Default Coder Group:

Parameter	Value
Coder Group Name	AudioCodersGroups_0
Coder Name	G.729
Coder Name	G.711A-law
Coder Name	G.711U-law

Figure 5-16: Configuring a Default Coder Group



5.5 Configure PSTN Interface

This section describes the configuration of the public switched telephone network (PSTN) related parameters.

5.5.1 Configure PRI Trunk Settings

This step shows how to configure the PRI Trunk.

➤ **To configure the PRI PSTN interface:**

1. Open the Trunk Settings page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Trunks & Groups** > **Trunks**).
2. Configure following parameters per PSTN network:

Parameter	Value
Protocol Type	E1 EURO ISDN
Clock Master	Generated (The device is clock master) Recovered (The device slaves from the line clock)
Framing Method	E1 Framing MFF CRC4 Ext (per remote side, PBX or PSTN, definitions)
ISDN Termination Side	Network side or User side (per remote side definitions)
Local ISDN Ringback Tone Source	Gateway

Figure 5-17: Configuring the PRI PSTN Interface

The screenshot shows the 'Trunk Settings' page in the AudioCodes configuration tool. The left sidebar contains a 'TOPOLOGY VIEW' with a tree structure including 'CORE ENTITIES', 'GATEWAY', 'Trunks & Groups', 'Routing', 'Manipulation', 'DTMF & Supplementary', 'Analog Gateway', 'Digital Gateway', 'MEDIA', 'CODERS & PROFILES', and 'SIP DEFINITIONS'. The main area is divided into several sections:

- GENERAL:** Module ID (1), Trunk ID (1), Trunk Configuration State (Active), Protocol Type (E1 EURO ISDN).
- TRUNK CONFIGURATION:** Clock Master (Generated), Auto Clock Trunk Priority (0), Line Code (HDB3), Line Build Out Loss (0 dB), Trace Level (No Trace), Line Build Out Overwrite (OFF), Framing Method (E1 FRAMING MFF CRC4).
- ISDN CONFIGURATION:** ISDN Termination Side (Network side), Q931 Layer Response Behavior (0x0).
- ADVANCED SETTINGS:** PSTN Alert Timeout (-1), Transfer Mode (Disable), Local ISDN Ringback Tone Source (Gateway), Set PI in Rx Disconnect Message (Not Configured), ISDN Transfer Capabilities (Not Configured), Progress Indicator to ISDN (Not Configured), Select Receiving of Overlap Dialing (None), B-channel Negotiation (Not Configured), Out-Of-Service Behavior (Not Configured), Remove Calling Name (Use Global Parame), Play Ringback Tone to Trunk (Not Configured), Call Rerouting Mode (None), ISDN Duplicate Q931 BuffMode (0), Trunk Name.

3. Repeat for another PRI trunks if applicable (Mediant 800B and Mediant 1000B)
4. **Reset the device with a save-to-flash for your settings to take effect.**

5.5.2 Configure Trunk Group Parameters

This step shows how to configure the device's channels, which includes assigning them to Trunk Groups. A Trunk Group is a logical group of physical trunks and channels. A Trunk Group can include multiple trunks and ranges of channels. To enable and activate the device's channels, Trunk Groups must be configured. Channels not configured in this table are disabled. After configuring Trunk Groups, use them to route incoming IP calls to the Tel side, represented by a specific Trunk Group (ID). You can also use Trunk Groups for routing Tel calls to the IP side.

➤ **To configure the PRI PSTN interface:**

1. Open the Trunk Group table (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Trunks & Groups** > **Trunk Groups**).

Figure 5-18: Configuring PRI Trunk Group Table

Group Index	Module	From Trunk	To Trunk	Channels	Phone Number	Trunk Group ID	Tel Profile Name
1	Module 1 PRI	1	1	1-31		1	None
2							None
3							None

2. Configure each Trunk Group as required. If more than one PRI port is available, on line 1 of the table above, set 'To Trunk' to the last PRI port (2).

5.5.3 Configure Inbound IP Routing

This section shows configuring Mediant PRI Gateway Inbound (IP-to-Tel) Routing. When having more than one TDM interface, you can choose to route calls based on incoming IP SIP call message to a specific TDM port i.e., Trunk Group.

➤ **To configure IP-to-Tel or Inbound IP Routing Rules:**

1. Open the IP-to-Tel Routing table (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Routing** > **IP** > **Tel Routing**).

Figure 5-19: Configuring Inbound IP Routing Rules

INDEX	NAME	SOURCE IP GROUP	SOURCE SIP INTERFACE	SOURCE IP ADDRESS	SOURCE PHONE PREFIX	DESTINATION PHONE PREFIX	TRUNK GROUP ID
0	To PBX	..	Any			*	1

2. Configure a rule for all incoming IP calls, route them to **Trunk Group ID 1** (connected to the PBX).
3. Click **Apply**.

5.6 Miscellaneous Configuration

This section describes miscellaneous Mediant gateway configuration.

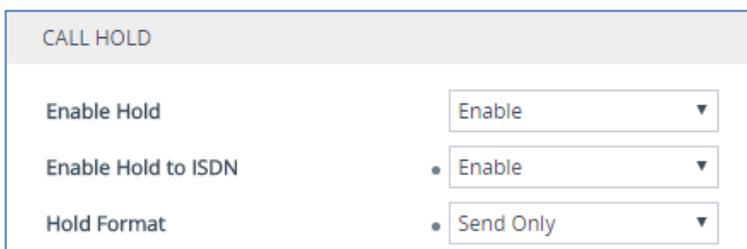
5.6.1 Configure Supplementary Services

This step describes how to configure Hold Format.

➤ **To configure Hold Format:**

1. Open the Gateway Supplementary Services Settings page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **DTMF & Supplementary** > **Supplementary Services Settings**).
2. From the 'Enable Hold to ISDN' drop-down list, select **Enable**.
3. From the 'Hold Format' drop-down list, select **Send Only**.

Figure 5-20: Configuring Hold Format



CALL HOLD	
Enable Hold	Enable
Enable Hold to ISDN	• Enable
Hold Format	• Send Only

4. Click **Apply**.

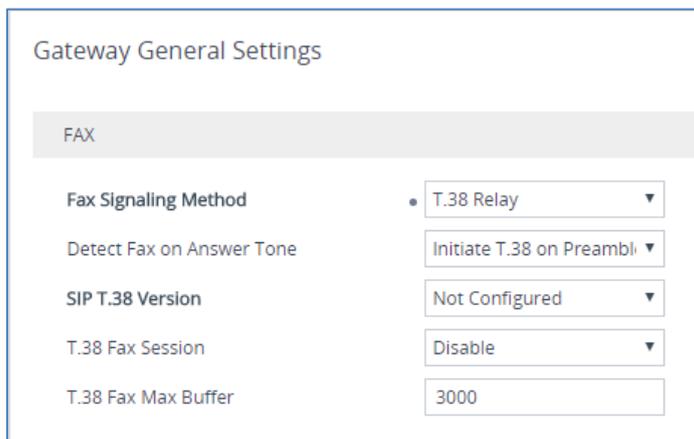
5.6.2 Configure Gateway General Settings

This step describes how to configure the Mediant Gateway to enable T.38 Fax Signaling Method and to play ring-back tone to PRI trunk.

➤ **To configure Gateway General Settings:**

1. Open the Gateway General Settings page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Gateway General Settings**).
2. From the 'Fax Signaling Method' drop-down list, select **T.38 Relay**.

Figure 5-21: Configuring Fax Signaling Method



Gateway General Settings	
FAX	
Fax Signaling Method	• T.38 Relay
Detect Fax on Answer Tone	Initiate T.38 on Preambl
SIP T.38 Version	Not Configured
T.38 Fax Session	Disable
T.38 Fax Max Buffer	3000

3. From the 'Play Ringback Tone to Tel' drop-down list, select **Play on Local**.

Figure 5-22: Configuring to play ringback tone to PSTN

BEHAVIOR	
NAT IP Address	<input type="text" value="::"/>
Channel Select Mode	<input type="text" value="Cyclic Ascending"/>
Tel to IP No Answer Timeout	<input type="text" value="180"/>
Play Ringback Tone to IP	<input type="text" value="Don't Play"/>
Play Ringback Tone to Tel	<input checked="" type="text" value="Play on Local"/>

4. Click **Apply**.

5.6.3 Configure Early Media

This step describes how to configure the Mediant Gateway to enable Early Media.

➤ **To configure Early Media:**

1. Open the Media Settings page (**Setup** menu > **Signaling & Media** tab > **Media** folder > **Media Settings**).
2. From the 'Enable Early Media' drop-down list, select **Enable**.

Figure 5-23: Configuring Early Media

GATEWAY SETTINGS	
Enable Early Media	<input checked="" type="text" value="Enable"/>
Multiple Packetization Time Format	<input type="text" value="None"/>

3. Click **Apply**.

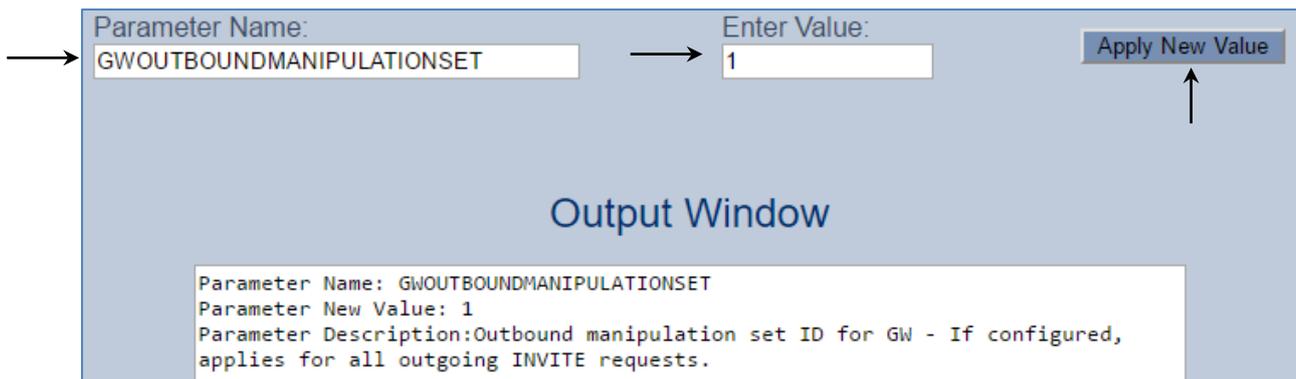
5.6.4 Configure Gateway Manipulation Set

This step describes how to configure the Mediant Gateway outbound manipulation set number.

➤ **To configure Gateway Outbound Manipulation Set number:**

1. Open the Admin page.
2. Append the case-sensitive suffix 'AdminPage' to the device's IP address in your Web browser's URL field (e.g., <http://10.15.77.55/AdminPage>).
3. In the left pane of the page that opens, click *ini* Parameters.

Figure 5-24: Configuring Gateway Outbound Manipulation Set number in AdminPage



4. Enter these values in the 'Parameter Name' and 'Enter Value' fields:

Parameter	Value
GWOUTBOUNDMANIPULATIONSET	1

5. Click the **Apply New Value** button for each field.

5.7 Check the SIP Trunk Registration Status

- To check if the device successfully registered with BroadSoft service:
 1. Open the Registration Status page (**Status & Diagnostics** tab > **VoIP Status** > **Registration Status**).
 2. Check the **Proxy Sets Status**. A successful registration will show as **ONLINE** (see the figure below).

Figure 5-25: Successful SIP Trunk Registration

The screenshot shows the AudioCodes Mediant Gateway MONITOR page. The 'Proxy Sets Status' section is active, displaying a table with the following data:

PROXY SET ID	MODE	KEEP ALIVE	ADDRESS	PRIORITY	WEIGHT	SUCCESS COUNT	FAILURE COUNT	STATUS
0	Parking	Enabled	lab.ic.sipconnect.hipcom.co.uk(85.119.61.20) (*)	-	-	11	0	ONLINE

The 'STATUS' column for the first row is circled in red, indicating a successful registration.



Note: If the status of the Proxy Sets shows OFFLINE, check your WAN connectivity:

- Check Ethernet wiring
- DMZ configuration may not be correct on the enterprise firewall
- Check IP address configuration (**Setup** menu > **IP Network** tab > **Core Entities** folder > **Static Routes**)
- Check proxy (SIP Trunk) configuration (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **Proxy & Registration**)

5.8 Secure Device Access



Note: Due to the vast number of potential attacks (such as DDoS), security of your VoIP network should be your paramount concern. The AudioCodes device provides a wide range of security features to support perimeter defense. For recommended security configuration for your AudioCodes device, refer to AudioCodes' *Security Guidelines* document.

It's recommended that when leaving the device at the end customer's premises, its management interface will be accessible by remote, **only when required**. If not required, request the end customer's IT administrator to disable the following ports:

- Port 80 - HTTP Web interface access
- Port 443 - HTTPS Web interface access
- Port 22 - SSH access
- Port 23 - Telnet access
- Ports 161 - SNMP access

If future remote management is required, first ask the end customer's IT administrator to open the appropriate port (e.g., HTTP or HTTPS port) to manage the device.

5.9 Save Configuration



Note: Firewall settings for the DMZ must be in place before resetting the device. After the device is reset, its new IP configuration is applied and it is no longer available for management from the LAN. Therefore, make sure the firewall allows the ports required for call handling. See Section 5.2 for more information.

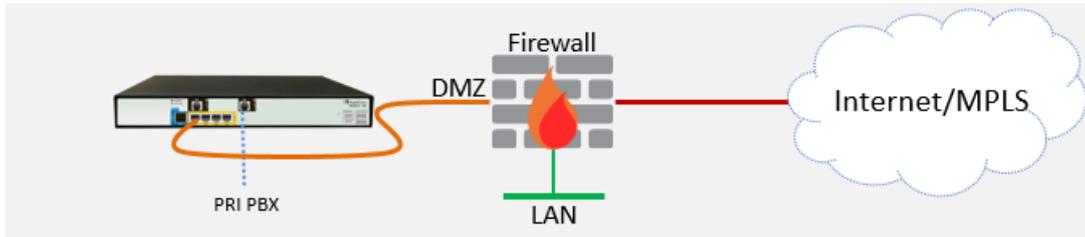
Save configuration as follows:

1. On the toolbar, click **Save** button:



6 Cable Device to DMZ

Once you the device has reset with your new configuration (as described in the previous section), its IP address changes to your newly configured address. You can now cable the device to your DMZ network:



1. Disconnect the cable connecting the device to your PC.
2. Cable to the DMZ Network:
 - a. Connect one end of a straight-through RJ-45 Ethernet cable (Cat 5e or Cat 6) to Port 1.
 - b. Connect the other end of the cable to your DMZ network.
3. Connect the E1/T1 trunk interface:
 - a. Connect the E1/T1 trunk cable to the device's E1/T1 port.
 - b. Connect the other end of the trunk cable to your PBX switch.

Figure 6-1: Mediant 500 Cabling E1/T1 Port

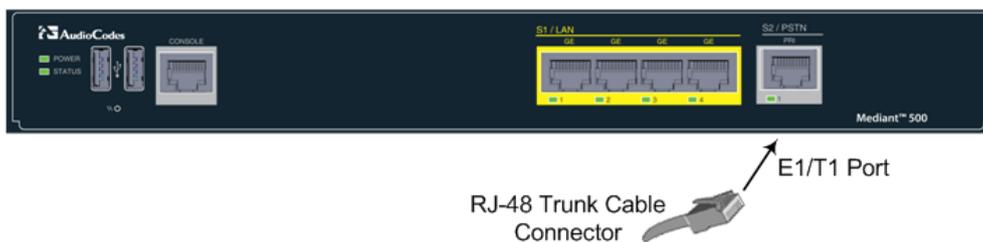


Figure 6-2: Mediant 800B Cabling E1/T1 Port

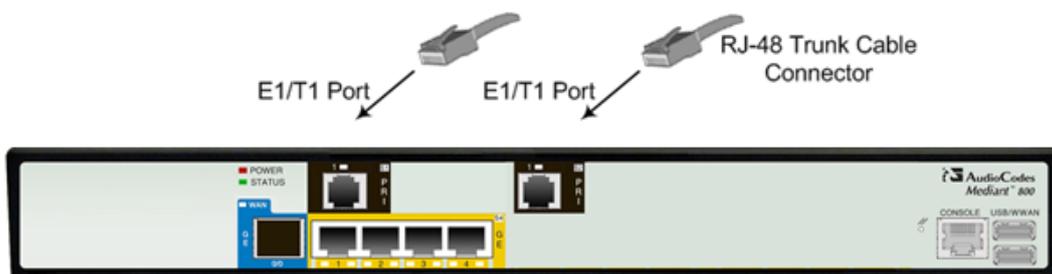


Figure 6-2: Mediant 1000B Cabling E1/T1 Port



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A Troubleshooting

This section describes issues that can be encountered and shows how to solve them.

A.1 Connecting to CLI

Connect to the device's serial port labeled CONSOLE connecting a standard RJ-45 to DB-9 female serial cable to a PC (sold separately). Connect to the console CLI and then:

1. Establish a serial communication (e.g., Telnet) with the device using a terminal emulator program such as HyperTerminal, with the following communication port settings:
 - Baud Rate: 115,200 bps
 - Data Bits: 8
 - Parity: None
 - Stop Bits: 1
 - Flow Control: None
2. At the CLI prompt, type the username (default is **Admin** - case sensitive):
Username: Admin
3. At the prompt, type the password (default is **Admin** - case sensitive):
Password: Admin
4. At the prompt, type the following:
enable
5. At the prompt, type the password again:
Password: Admin

A.2 Enabling Logging on CLI

To enable the device to send the error messages (e.g. Syslog messages) to the CLI console, use the following commands:

1. Start the syslog on the screen by typing:
debug log
2. Enable SIP call debugging
debug sip 5
3. Stop Syslog on the screen by typing:
no debug log

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