

**SIP**

**Mediant 1000**

## **Configuration Note**

**AudioCodes' Mediant VoIP Gateway and  
Avaya's Modular Messaging**

*with*

**Definity G3 Prologix and S87x0/S8x00  
using E1 QSIG**

**AVAYA**

 **AudioCodes**  
Connecting Networks



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## Reader's Notes

## Disclaimer

This PBX Configuration Note is designed to be a general guide reflecting AudioCodes and Avaya experience in configuring their systems. These notes cannot anticipate every configuration possibility, given the inherent variations in hardware and software products. Therefore, if you experience a problem not detailed in this document, please notify AudioCodes' Technical Support at [support@audiocodes.com](mailto:support@audiocodes.com), and if appropriate, we will include it in our next document revision. AudioCodes Ltd. and Avaya Inc. accept no responsibility for errors or omissions contained herein.

This document is subject to change without notice.

Date Published: March-25-2009

## Version Information

Version	Date of Modification	Details of Modification
01	March 2009	Initial version by AudioCodes

## Overview

This document describes the configuration required to setup Avaya Definity G3 and AudioCodes' Mediant 1000 gateway, using E1 QSIG as the telephony signaling protocol to an Avaya Modular Messaging system connected to the gateway using SIP.

## Targeted Audience

This document is intended for Avaya Installation Engineers or Avaya Business Partners who are installing Modular Messaging using AudioCodes gateway.

**Reader's Notes**

# 1 Components Information

## 1.1 PBX or IP-PBX

<b>PBX Vendor</b>	Avaya
<b>Model</b>	Definity G3 Prologix & S87x0/S8x00
<b>Software Version</b>	Definity G3 & Prologix: G3V10.1 Load 43 S8x00/S87x0: V1.1, V2.0, V3.x, V4.0
<b>Telephony Signaling</b>	E1 QSIG
<b>Additional Notes</b>	None

## 1.2 AudioCodes Gateway

<b>Gateway Vendor</b>	AudioCodes
<b>Model</b>	Mediant 1000
<b>Software Version</b>	5.60A.007.004
<b>VoIP Protocol</b>	SIP
<b>Additional Notes</b>	None

## 1.3 Avaya Modular Messaging Version

<b>Version</b>	Avaya Modular Messaging Release 5.0
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**Reader's Notes**



## **2 Prerequisites**

### **2.1 Gateway Prerequisites**

None

### **2.2 PBX Prerequisites**

Refer to Section 3.1.

### **2.3 Cabling Requirements**

Refer to Section 3.1.

**Reader's Notes**

## **3 PBX Setup Notes**

### **3.1 PBX Configuration**

Configure the PBX as specified in the Avaya Modular Messaging PBX Configuration Note "CN88004.pdf" Section 5.0.

You can obtain this Configuration Note from **Avaya** Web sites <http://support.avaya.com>

### **3.2 Special Instructions for PBX Configuration**

Refer to Section 3.1.

### **3.3 Other Comments**

Refer to Section 3.1.

## Reader's Notes

## 4 Gateway Setup Notes

This section describes the configuration of AudioCodes' gateway required for integration with both the PBX and the Avaya Modular Messaging System.

You can configure the gateway using one of the following methods:

- Uploading an *ini* configuration file (\*.ini file) – refer to Section 4.1
- Configuring the gateway via the Web interface – refer to Section 4.2

### 4.1 Configuration Files

For initial setup and configuration, you can upload an *ini* file (\*.ini) to AudioCodes gateway that includes the template *ini* file settings shown in Appendix A. Simply create a new text file (e.g., using Microsoft Notepad) with the file extension \*.ini, copy and paste the *ini* file settings from Appendix A into the text file, and then upload the file to the gateway.

Typically, for interoperability with the deployed PBX interfaces and Avaya Modular Messaging, it's sufficient that you use the this *ini* file template. However, due to specificity of site deployment, you may need to modify or define certain parameters (such as IP addresses and Trunk settings) after uploading the *ini* file.

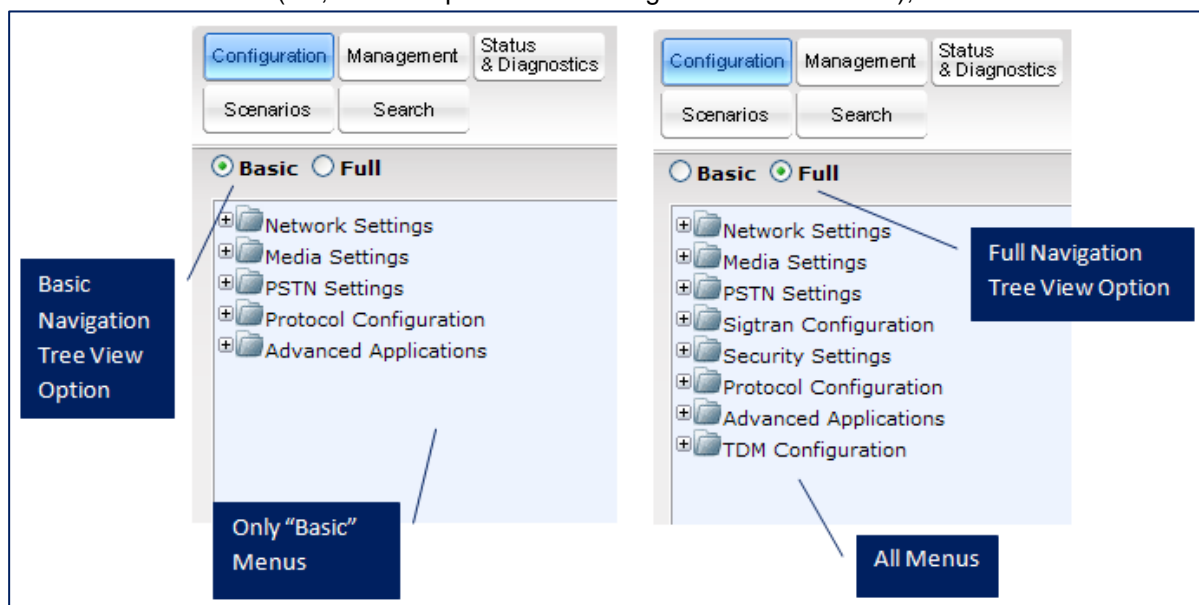
### 4.2 Configuring AudioCodes Gateway

This section provides step-by-step procedures for configuring AudioCodes' gateway, using the Web interface. Ensure that you configure the gateway according to the configuration settings displayed in the screenshots provided in this section.

The procedures describe how to setup Avaya Modular Messaging with the gateway implementing SIP over TLS **with** and **without** SRTP.

Note the following Web interface guidelines:

- When making configuration changes for each procedure, ensure that you click the **Submit** button to save your changes; unless otherwise instructed.
- Some of the changes may require a gateway reset for these changes to take effect. Therefore, (and to save time), reset the gateway only after you complete all of the gateway configurations.
- The procedures described in this section are performed using the gateway's Web-based management tool (i.e., embedded Web server). Before you begin configuring the gateway, ensure that the Web interface's Navigation tree is in full menu display mode (i.e., the Full option on the Navigation bar is selected), as shown below:



**Step 1: Trunk Setting Setup**

Open the 'Trunk Settings' page (**Configuration Tab: PSTN Settings > Trunk Settings**).

The screenshot displays the configuration interface for a trunk. At the top, there is a navigation bar with tabs 1 and 2, and a progress indicator. Below this, the configuration is organized into sections:

- General Settings:**
  - Module ID: 1
  - Trunk ID: 1
  - Trunk Configuration State: **Inactive**
  - Protocol Type: **E1 QSIG** (indicated by an arrow)
- Trunk Configuration:**
  - Clock Master: Recovered
  - Auto Clock Trunk Priority: 0
  - Line Code: HDB3
  - Framing Method: **E1 FRAMING MFF CRC4 EXT** (indicated by an arrow)
- ISDN Configuration:**
  - ISDN Termination Side: User side
  - Q931 Layer Response Behavior: 0x40000000
  - Outgoing Calls Behavior: 0x400
  - Incoming Calls Behavior: 0x0
  - General Call Control Behavior: 0x20
  - NFAS Group Number: 0
  - IUA Interface ID: -1
  - NFAS Interface ID: 255
  - D-channel Configuration: PRIMARY
- Additional Settings (bottom section):**
  - PSTN Alert Timeout: -1
  - QSIG Transfer Mode: **Path Replacement Transfer** (indicated by an arrow)
  - Local ISDN Ringback Tone Source: PBX
  - Set PI in Rx Disconnect Message: Not Configured
  - ISDN Transfer Capabilities: Not Configured
  - Progress Indicator to ISDN: Not Configured
  - Enable Receiving of Overlap Dialing: Enable
  - B-channel Negotiation: Not Configured
  - Out-Of-Service Behavior: Default
  - Play Ringback Tone to Trunk: Don't Play

Before you can modify parameters on this page, you need to click the **Stop Trunk** button to de-activate the trunk.

After you modify the parameters, click the **Apply Trunk Settings** button, and then wait for the trunk settings to be applied. Once the trunk settings are applied, the trunk status icons at the top of the page change to green for all trunks that are connected to the PBX.

In case of more than one trunk connection between the PBX and gateway, repeat Step 1 for each of the trunks, or click the **Apply to All Trunks** button.

## Step 2: SIP Environment Setup

Open the 'SIP General Parameters' page (Configuration Tab: Protocol Configuration > Protocol Definition > SIP General Parameters).

SIP General	
PRACK Mode	Supported
Channel Select Mode	Ascending
Enable Early Media	Disable
Session-Expires Time	0
Minimum Session-Expires	90
Session Expires Method	Re-INVITE
Asserted Identity Mode	Disabled
Fax Signaling Method	T.38 Relay
SIP Transport Type	TLS
SIP UDP Local Port	5060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Enable SIPS	Disable
SIP Destination Port	5061
Enable Remote Party ID	Disable
Enable History-Info Header	Disable
Play Ringback Tone to IP	Don't Play
Play Ringback Tone to Tel	Play According to Early Media
Enable Reason Header	Enable

Retransmission Parameters	
---------------------------	--

It is recommended that you configure the gateway and Avaya's Modular Messaging to use TLS. If you prefer to use TCP, then ensure that you configure the gateway settings relating to TLS (in the screen above) to use TCP:

- **SIP Transport Type:** "TCP"
- **SIP TCP Local Port:** "5060"
- **SIP Destination Port:** "5060"

### Step 3: Configuring SRTP

Open the 'Media Security' page (**Configuration** Tab: **Media Settings** > **Media Security**).

▼ General Media Security Settings	
Media Security	Enable
Media Security Behavior	Mandatory
Disable Authentication On Transmitted RTP Packets	0
Disable Encryption On Transmitted RTP Packets	0
Disable Encryption On Transmitted RTCP Packets	0
▼ SRTP Setting	
Master Key Identifier (MKI) Size	0

If you are using SIP over TLS **with** SRTP, then set the SRTP Setting parameters as follows:

- **Media Security:** "Enable"
- **Media Security Behavior:** "Mandatory"
- **Disable Encryption On Transmitted RTCP Packets:** "1"


If you are using SIP over TLS **without** SRTP, then set the SRTP Setting parameters as follows:

- **Media Security:** "Disabled"
- **Media Security Behavior:** "Mandatory"
- **Disable Encryption On Transmitted RTCP Packets:** "0"



**Step 4: Routing, PBX-to-IP Routing, SIP Environment and Gateway Name Setup**

Open the 'Proxy & Registration' page (Configuration Tab: Protocol Configuration > Protocol Definition > Proxy & Registration).

Use Default Proxy	Yes
Proxy Set Table	
Proxy Name	
Redundancy Mode	Parking
Proxy IP List Refresh Time	60
Enable Fallback to Routing Table	Disable
Prefer Routing Table	No
Use Routing Table for Host Names and Profiles	Disable
Always Use Proxy	Disable
Enable Registration	Disable
Gateway Name	
Gateway Registration Name	
User Name	
Password	Default_Passwd
Cnonce	Default_Cnonce
Authentication Mode	Per Gateway

Assign an FQDN name to the gateway (for example, mygateway.mynet1.mynet.com). Any gateway name that corresponds to your network environment is applicable, but it must meet requirements for FQDNs.

Proxy Set ID: 0

	Proxy Address	Transport Type
1	10.15.10.11	TLS
2		
3		
4		
5		

Enable Proxy Keep Alive: Disable  
 Proxy Keep Alive Time: 60  
 Proxy Load Balancing Method: Round Robin  
 Is Proxy Hot Swap: No

1. In the 'Proxy Address' field, enter either the IP address or FQDN of the Avaya Modular Messaging MAS. If your Avaya Modular Messaging system includes multiple MAS's, then enter multiple IP addresses or FQDNs for the MAS's - one MAS per table row. It is recommended that you use FQDNs.
2. From the 'Transport Type' drop-down list, select the transport type for each MAS. **Note:** When not configured, the value of the parameter 'SIPTransportType' is used.
3. From the 'Proxy Load Balancing Method' drop-down, select "Round Robin" to load balance the calls across all MAS's in your Avaya Modular Messaging System.

**Step 5: Coder Setup**

Open the 'Coders Table' page (Configuration Tab: Protocol Configuration > Protocol Definition > Coders).

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711U-law	20	64	0	Disabled

Configure the Coders table to contain only G.711U-law.

**Step 6: Digit Collection Setup**

Open the 'DTMF & Dialing' page (Configuration Tab: Protocol Configuration > Protocol Definition > DTMF & Dialing).

Max Digits In Phone Num	30
Inter Digit Timeout for Overlap Dialing [sec]	4
Declare RFC 2833 in SDP	Yes
1st Tx DTMF Option	RFC 2833
2nd Tx DTMF Option	
RFC 2833 Payload Type	96
⚡ Digit Mapping Rules	
Default Destination Number	serveduser
Special Digit Representation	Special

Set the following fields to the value indicated:

- **Default Destination Number:** "serveduser"

## Step 7: General Setup

Open the 'Advanced Parameters' page (Configuration Tab: Protocol Configuration > SIP Advanced Parameters > Advanced Parameters).

▼ General	
IP Security	Disable
Filter Calls to IP	Don't Filter
⚡ Enable Digit Delivery to Tel	Disable
⚡ Enable Digit Delivery to IP	Disable
RTP Only Mode	Disable
PSTN Alert Timeout	180
▼ Disconnect and Answer Supervision	
Disconnect on Broken Connection	No
Broken Connection Timeout [100 msec]	3
Disconnect Call on Silence Detection	No
⚡ Silence Detection Period [sec]	120
⚡ Silence Detection Method	Voice/Energy Detectors
Enable Fax Re-Routing	Disable
▼ CDR and Debug	
CDR Server IP Address	
CDR Report Level	None
Debug Level	5
▼ Misc. Parameters	
Progress Indicator to IP	Not Configured
Enable X-Channel Header	Disable
Enable Busy Out	Disable

From the 'Disconnect on Broken Connection' drop-down list, select 'No'.

**Step 8: Trunk Group Setup**

Open the 'Trunk Group Table' page (Configuration Tab: Protocol Configuration > Trunk/IP Group > Trunk Group).

Add Phone Context As Prefix		Disable	
Trunk Group Index		1-12	

Group Index	Module	From Trunk	To Trunk	Channels	Phone Number	Trunk Group ID	IP Profile ID
1	Module 1 PRI	1	1	1-24	2000		0
2							
3							
4							
5							
6							
7							
8							
9							
10							
11							
12							

The 'Phone Number' field must match the pilot number of the QSIG trunk.  
 If more than one trunk is used, in the 'To Trunk' field, enter the last trunk number (e.g., 2) pertaining to the Trunk Group and then in the 'Channel' field, enter the number of channels (e.g., 1-48) accordingly.

**Step 9: Voice Mail Settings**

Open the 'Voice Mail Settings' page (**Configuration** Tab: **Advanced Applications** > **Trunk Group**).

▼ General	
Voice Mail Interface	→ QSIG
▼ Digit Patterns	
Forward on Busy Digit Pattern (Internal)	<input type="text"/>
Forward on No Answer Digit Pattern (Internal)	<input type="text"/>
Forward on Do Not Disturb Digit Pattern (Internal)	<input type="text"/>
Forward on No Reason Digit Pattern (Internal)	<input type="text"/>
Forward on Busy Digit Pattern (External)	<input type="text"/>
Forward on No Answer Digit Pattern (External)	<input type="text"/>
Forward on Do Not Disturb Digit Pattern (External)	<input type="text"/>
Forward on No Reason Digit Pattern (External)	<input type="text"/>
Internal Call Digit Pattern	<input type="text"/>
External Call Digit Pattern	<input type="text"/>
Disconnect Call Digit Pattern	<input type="text"/>
Digit To Ignore Digit Pattern	<input type="text"/>
▼ Message Waiting Indication (MWI)	
MWI Off Digit Pattern	<input type="text"/>
MWI On Digit Pattern	<input type="text"/>
MWI Suffix Pattern	<input type="text"/>
MWI Source Number	<input type="text"/>
▼ SMDI	
⚡ Enable SMDI	Disable
SMDI Timeout [msec]	2000

From the 'Voice Mail Interface' drop-down list, select 'QSIG'.

### Step 10: TDM BUS Settings

Open the 'TDM Bus Settings' page (Configuration Tab: TDM Configuration > TDM Bus Settings).

PCM Law Select	ALaw
TDM Bus Type	Framers
Idle PCM Pattern	255
Idle ABCD Pattern	0x0F
TDM Bus Local Reference	1
TDM Bus PSTN Auto Clock	Enable
TDM Bus Clock Source	Network

From the 'PCM Law Select' drop-down list, select 'ALaw'.

From the 'TDM Bus Clock Source' drop-down list, select 'Network'.

### Step 11: Application Settings

Open the 'Application Settings' page (**Configuration** Tab: **Network Settings** > **Application Settings**).

DNS Settings	
DNS Primary Server IP	10.1.1.11
DNS Secondary Server IP	10.1.1.10

Set the following fields to the value indicated:

- **DNS Primary Server IP:** set the IP address of the first DNS server.
- **DNS Secondary Server IP:** set the IP address of the second DNS server.

**Step 12: CNG Detector Mode**

Open the 'Fax/Modem/CID Settings' page (**Configuration** Tab: **Media Settings** > **Fax/Modem/CID Settings**).

Fax Transport Mode	RelayEnable	▼
Caller ID Transport Type	Mute	▼
Caller ID Type	Standard Bellcore	▼
V.21 Modem Transport Type	Disable	▼
V.22 Modem Transport Type	Enable Bypass	▼
V.23 Modem Transport Type	Enable Bypass	▼
V.32 Modem Transport Type	Enable Bypass	▼
V.34 Modem Transport Type	Enable Bypass	▼
Fax Relay Redundancy Depth	0	
Fax Relay Enhanced Redundancy Depth	4	
Fax Relay ECM Enable	Enable	▼
Fax Relay Max Rate (bps)	14400bps	▼
Fax/Modem Bypass Coder Type	G711Mulaw	▼
Fax/Modem Bypass Packing Factor	1	
Fax Bypass Output Gain	0	
Modem Bypass Output Gain	0	
Fax CNG Mode	Disable	▼
CNG Detector Mode	Disable	▼

From the 'CNG Detector Mode' drop-down list, select 'Disable'.

**Step 13: Add Internal DNS Table**

Open the 'Internal DNS Table' page (Configuration Tab: Protocol Configuration > Routing Tables > Internal DNS Table).

	Domain Name	First IP Address	Second IP Address	Third IP Address	Fourth IP Address
1	anonymous.invalid	10.15.10.7	0.0.0.0	0.0.0.0	0.0.0.0
2	stockleyg3.com	10.15.10.7	0.0.0.0	0.0.0.0	0.0.0.0
3					
4					
5					
6					
7					

Set the following fields to the value indicated:

- **Domain Name '1':** "anonymous.invalid"
- **Domain Name '2':** "stockleyg3.com"
- **First IP Address:** set the IP address of the Mediant 1000 (e.g., 10.15.10.7) in both lines.



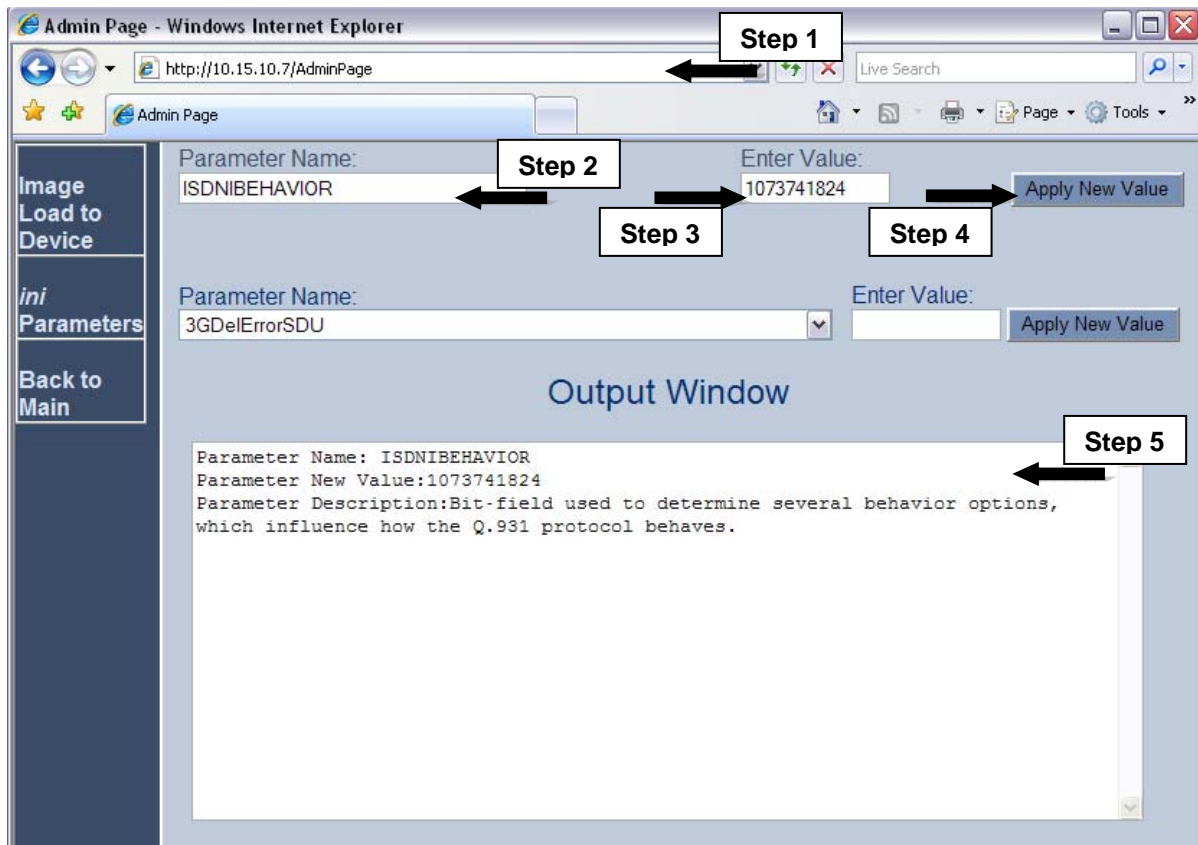
### Step 14: Modify Parameters in the AdminPage

The following changes need to be made in the AdminPage:

- **ISDNIBehavior:** "1073741824"
- **EnableMWI:** "1"
- **ECNLPMODE:** "1"
- **SubscriptionMode:** "1"
- **TrunkTransferMode\_x:** "0" (where, x denotes the trunk number - for example, for the first trunk, set TrunkTransferMode\_0 = 0)

#### ➤ To modify parameters:

1. Open the 'AdminPage' page at the following URL (case-sensitive):  
**http://<gateway's IP address>/AdminPage**
2. In the 'Parameter Name', enter the parameter's name.
3. In the 'Enter Value', enter the parameter's value.
4. Click the **Apply New Value** button.
5. Check the output.



### Step 15: Reset the Mediant 1000 Gateway

After you have completed the gateway configuration as described in the steps above, burn the configuration to the gateway's flash memory and reset the gateway.

Click the **Reset** button to burn the configuration to flash and reset the gateway (ensure that the 'Burn to FLASH' field is set to "Yes").

▼ Reset Configuration	
Reset Board	<b>Reset</b>
Burn To FLASH	Yes
Graceful Option	No
▼ LOCK / UNLOCK	
Lock	<b>LOCK</b>
Graceful Option	No
Current Admin State	UNLOCKED
▼ Save Configuration	
Burn To FLASH	<b>BURN</b>

**For Reset Board :**  
If you choose not to burn the device's configuration into flash memory, all changes made since the last time the configuration was burned will be lost after the device is reset.

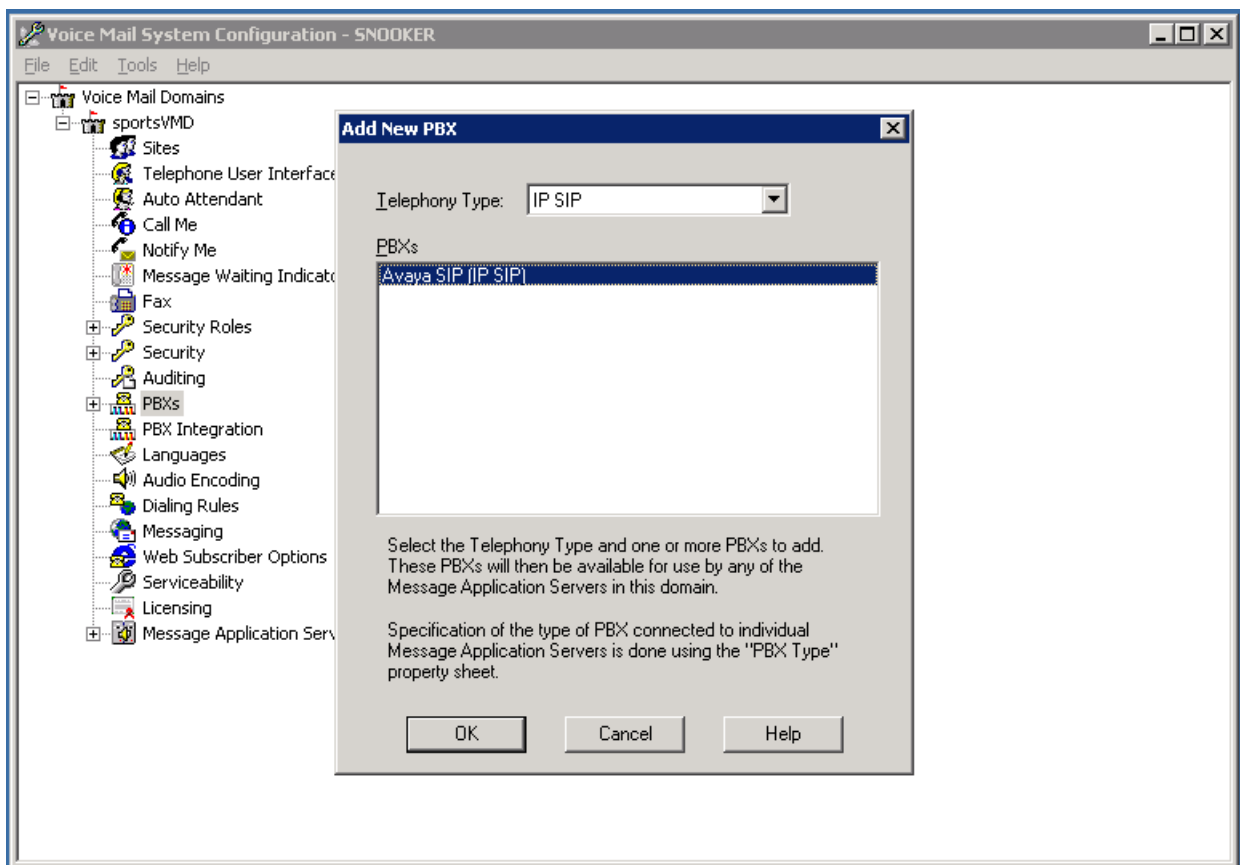
**For Save Configuration:** Saving configuration to flash memory may cause some temporary degradation in voice quality, therefore, it is recommended to perform this during low-traffic periods

## 5 Avaya Modular Messaging Configuration

Complete the following steps to configure Avaya's Modular Messaging MAS for integration with AudioCodes' gateway.

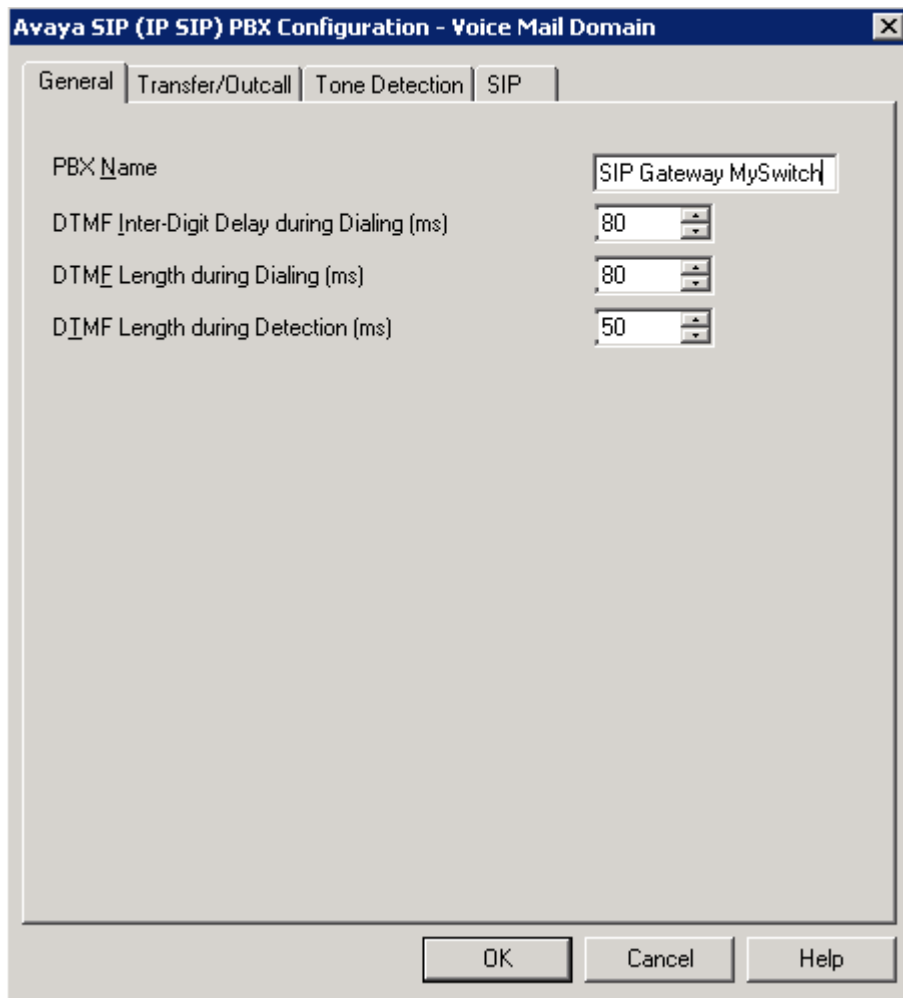
If required, use Voice Mail System Configuration (VMSC) to create a new SIP PBX for the Avaya Modular Messaging Voice Mail Domain (VMD):

1. Launch VMSC.
2. Expand your VMD tree.
3. Right-click **PBXs**, and then select **Add New PBX**.
4. From the drop-down list, select "IP SIP".



5. From the list of available PBXs, select "Avaya SIP (IP SIP)", and then click **OK**.

6. Open the Properties for the newly added SIP PBX.



The screenshot shows a dialog box titled "Avaya SIP (IP SIP) PBX Configuration - Voice Mail Domain" with a close button (X) in the top right corner. The dialog has four tabs: "General", "Transfer/Outcall", "Tone Detection", and "SIP". The "SIP" tab is selected. The configuration fields are as follows:

Field	Value
PBX Name	SIP Gateway MySwitch
DTMF Inter-Digit Delay during Dialing (ms)	80
DTMF Length during Dialing (ms)	80
DTMF Length during Detection (ms)	50

At the bottom of the dialog, there are three buttons: "OK", "Cancel", and "Help".

7. Change the PBX Name to a unique name that reflects the AudioCodes gateway you are installing.

8. Select the **SIP** tab.

The screenshot shows the 'Avaya SIP (IP SIP) PBX Configuration - Voice Mail Domain' dialog box with the 'SIP' tab selected. The 'Gateways' section contains a table with one entry:

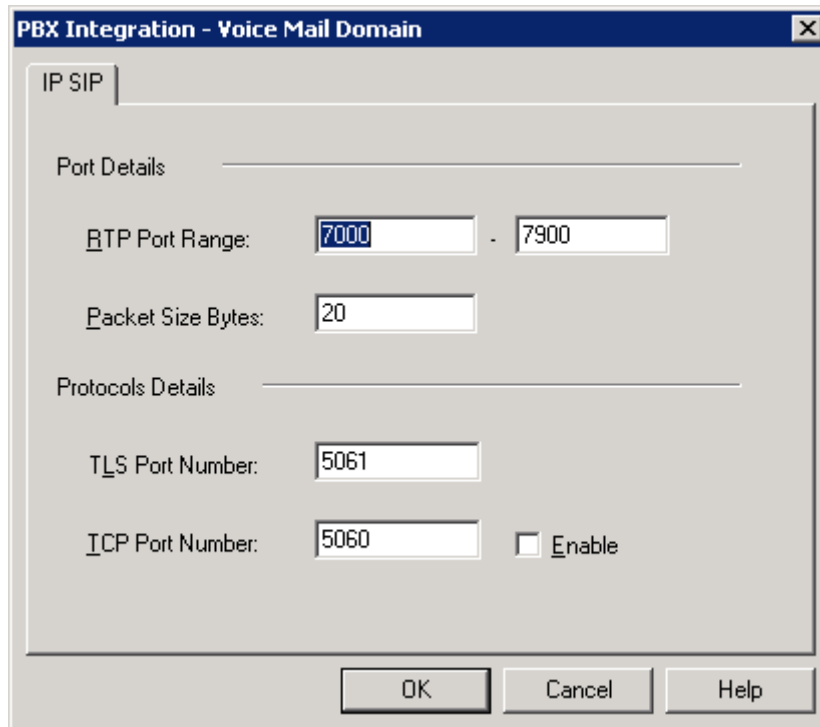
	Address/FQDN	Protocol	MWI	SRTP
<input checked="" type="checkbox"/>		TLS	<input checked="" type="checkbox"/>	None

Below the table are input fields for 'SIP Domain' and 'P-Asserted-Identity'. The 'Phone Number Translation Rules' section includes a 'Configure...' button and a note: 'Translation rules are effective only after MultiSite has been enabled.' At the bottom are 'OK', 'Cancel', and 'Help' buttons.

9. Enter the following details on the SIP configuration page:
- Enter the FQDN or IP address of the AudioCodes gateway you are installing.
  - Select the protocol for the AudioCodes gateway.
 

**Note:** It is recommended to use TLS.
  - Select "MWI" if the AudioCodes gateway handles MWI requests.
  - Select the SRTP configuration required for the AudioCodes gateway:
    - ◆ If using SIP over TLS with SRTP, then select the required SRTP level.
    - ◆ If using SIP over TLS without SRTP, then select "None".
  - Enter a unique SIP Domain to be used with the AudioCodes gateway.
  - Click **OK**.

10. Select the VMD PBX Integration.



The screenshot shows a dialog box titled "PBX Integration - Voice Mail Domain" with a close button (X) in the top right corner. The "IP SIP" tab is selected. The dialog is divided into two sections: "Port Details" and "Protocols Details".

**Port Details:**

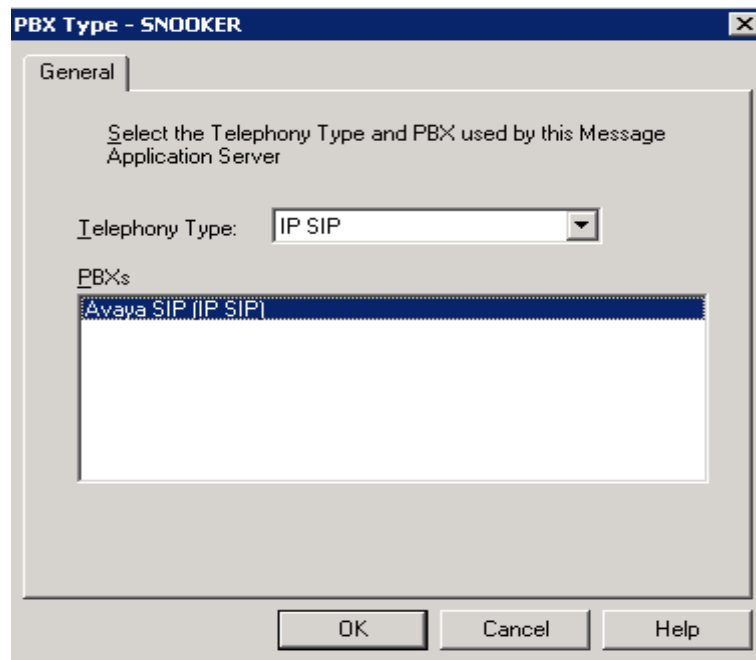
- RTP Port Range: 7000 - 7900
- Packet Size Bytes: 20

**Protocols Details:**

- TLS Port Number: 5061
- ICP Port Number: 5060  Enable

At the bottom of the dialog are three buttons: "OK", "Cancel", and "Help".

11. Configure the IP SIP settings as required, and then click **OK**, (the figure above shows the default settings).  
**Note:** If you are using SIP over TCP, then select the 'TCP Enable' check box.
12. Expand the Message Application Servers (MAS) item.
13. From the drop-down list, select the Telephony Type IP SIP.



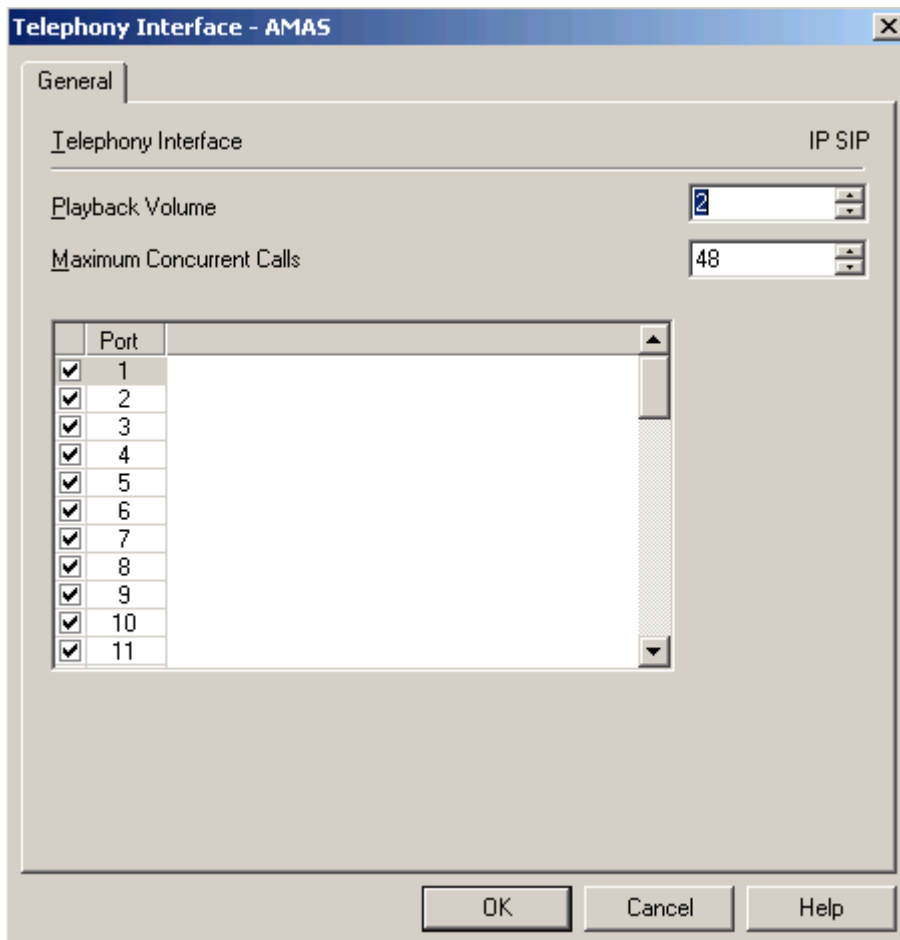
The screenshot shows a dialog box titled "PBX Type - SNOOKER" with a close button (X) in the top right corner. The "General" tab is selected. The dialog contains the following elements:

- Instruction: "Select the Telephony Type and PBX used by this Message Application Server"
- Telephony Type: IP SIP (selected in a dropdown menu)
- PBXs: A list box containing "Avaya SIP (IP SIP)" which is currently selected.

At the bottom of the dialog are three buttons: "OK", "Cancel", and "Help".

14. From the list of IP SIP PBXs, select the new PBX, and then click **OK**.

15. Select the Telephony Interface Item.

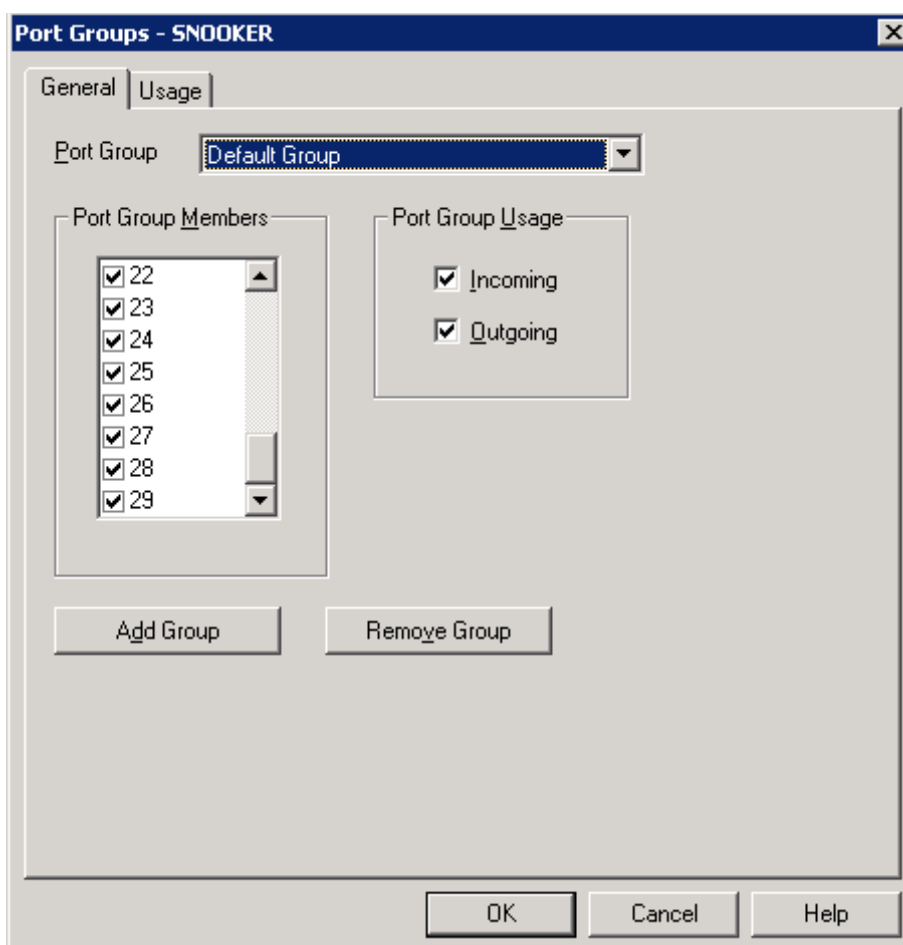


16. Set the Maximum Concurrent Calls to match the number of Channels that you want on this MAS, and then click **OK**.

**Note:** If your Avaya Modular Messaging system is running in a non-Multisite environment, then the Maximum Concurrent Calls value should match the number of telephony channels on your AudioCodes gateway. Therefore, if you have one QSIG E1 Trunk connected to the gateway, then enter 29 in the Maximum Concurrent Calls field.

If your Modular Messaging system is running in a Multisite environment, then you should configure the Maximum Concurrent Calls value to 48 (this is the default value for SIP integrations with Modular Messaging).

17. Select the Port Groups Item.



18. Ensure that all the Port Group Members for the Default Group are selected and Incoming and Outgoing usages are selected.
19. Add a new Port Group called 'MWI Group'.
20. Add just the upper most port to this group and ensure the Incoming and Outgoing usages are selected.
21. Click **OK**.
22. Restart the Message Application Server Service using the Services Applet to apply these changes to the MAS.



## 6 Summary and Limitations

### 6.1 Detailed Description of Limitations

<b>Failure Point</b>	None
<b>Phone Type (if phone-specific)</b>	Analog Phone
<b>Call Scenarios Associated with Failure Point</b>	None
<b>List of UM Features Affected by Failure Point</b>	None
<b>Additional Comments</b>	None

**Reader's Notes**

## 7 Troubleshooting

The tools used for debugging include network sniffer applications (such as Wireshark) and AudioCodes' Syslog protocol.

### 7.1 Configuring AudioCodes Gateway for Syslog Server

The Syslog client, embedded in the AudioCodes gateway sends error reports/events generated by the gateway application to a Syslog server, using IP/UDP protocol.

➤ **To activate the Syslog client on the AudioCodes gateways:**

1. Set the parameter 'Enable Syslog' to "Enable".
2. Use the parameter 'Syslog Server IP Address' to define the IP address of the Syslog server you use.

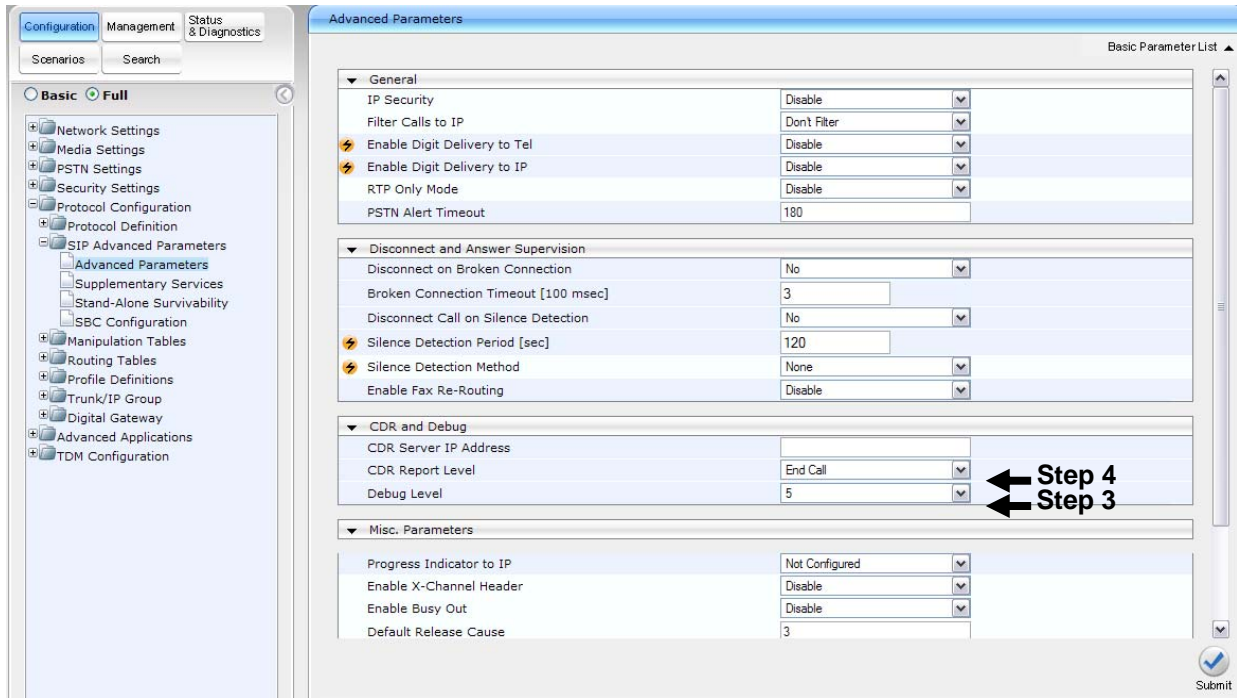
**Note:** The Syslog Server IP address must be one that corresponds with your network environment in which the Syslog server is installed (for example, 10.15.2.5).

The screenshot displays the 'Management Settings' configuration page. The left sidebar shows a tree view with 'Management Configuration' expanded, containing 'Management Settings', 'Regional Settings', 'Maintenance Actions', and 'Software Update'. The main content area is divided into three sections:

- Syslog Settings:** Contains fields for 'Syslog Server IP Address' (10.15.2.5), 'Syslog Server Port' (514), 'Enable Syslog' (set to 'Enable'), and 'Trunks Filter' (-1). Two black arrows point to these fields, labeled 'Step 2' and 'Step 1' respectively.
- SNMP Settings:** Includes 'SNMP Trap Destinations', 'SNMP Community String', 'SNMP V3 Table', 'SNMP Trusted Managers', 'Disable SNMP' (set to 'No'), and 'Trap Manager Host Name'.
- Activity Types to Report via 'Activity Log' Messages:** A list of checkboxes for various events, including 'Parameters Value Change', 'Auxiliary Files Loading', 'Device Reset', 'Flash Memory Burning', 'Device Software Update', 'Access to Restricted Domains', 'Non-Authorized Access', and 'Sensitive Parameters Value Change'.

A 'Submit' button is located at the bottom right of the configuration area.

3. To determine the Syslog logging level, use the parameter 'Debug Level' and set this parameter to "5".
4. Change the 'CDR Report Level' to "End Call" to enable additional call information.



AudioCodes has also developed the following advanced diagnostic tools for high-level troubleshooting:

- **PSTN Trace:** used for monitoring and tracing PSTN elements (E1/T1) in AudioCodes digital gateways (Mediant 1000). These utilities are designed to convert PSTN trace binary files into textual form.
- **DSP Recording:** used for monitoring the DSP operation (e.g., RTP packets and events).

## A AudioCodes ini File

```

.*****
;
.** Ini File **
.*****
;

;Board: Mediant 1000
;Serial Number: 469831
;Slot Number: 1
;Software Version: 5.60A.007.004
;DSP Software Version: 624AE3 => 560.10
;Board IP Address: 10.15.10.7
;Board Subnet Mask: 255.255.0.0
;Board Default Gateway: 10.15.0.1
;Ram size: 128M Flash size: 32M
;Num DSPs: 12 Num DSP channels: 72
;Profile: NONE
;Key features;;Board Type: Mediant 1000;DSP Voice features: ;Channel Type: RTP ATM
PCI DspCh=120 ;Coders: G723 G729 G728 NETCODER GSM-FR GSM-EFR AMR
EVRC-QCELP G727 ILBC EVRC-B AMR-WB G722 H263 H264 MPEG4 EG711 ;Security:
IPSEC MediaEncryption StrongEncryption EncryptControlProtocol ;IP Media: VXML
;E1Trunks=4;T1Trunks=4;Control Protocols: MGCP SIP ;Default features;;Coders: G711
G726;

;----- Mediant-1000 HW components-----
;
; Slot # : Module type : # of ports
;-----
; 1 : FALC56 : 2
; 2 : Empty
; 3 : Empty
; 4 : Empty
; 5 : Empty
; 6 : Empty
;-----

[SYSTEM Params]

ENABLEPARAMETERSMONITORING = 1
TLSVersion = 1

[BSP Params]

PCMLawSelect = 1
TDMBusClockSource = 4
StorageServerNetworkAddress = 255.255.255.255

```

[ATM Params]

[Analog Params]

[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]

[MEGACO Params]

EP\_Num\_0 = 0

EP\_Num\_1 = 1

EP\_Num\_2 = 0

EP\_Num\_3 = 0

EP\_Num\_4 = 0

[PSTN Params]

TraceLevel = 1

TDMBusPSTNAutoClockEnable = 1

ProtocolType\_0 = 21

ProtocolType\_1 = 0

FramingMethod\_0 = c

FramingMethod\_1 = 0

LineCode\_0 = 0

LineCode\_1 = 2

ISDNIBehavior = 1073741824

ISDNGeneralCCBehavior = 32

PSTNReserved3 = 8

[SS7 Params]

[Voice Engine Params]

ECNLPMMode = 1

BrokenConnectionEventTimeout = 3

FarEndDisconnectSilenceMethod = 0

DTMFDetectorSensitivity = 1

AggressiveDTMFErasure = 770

TTYTRANSPORTTYPE = 1

```
[WEB Params]

HTTPSCipherString = 'RC4:EXP'

[SIP Params]

ISPROXYUSED = 1
SIPDESTINATIONPORT = 5061
CHANNELSELECTMODE = 2
GWDEBUGLEVEL = 5
ISDNRXOVERLAP_0 = 1
ISDNRXOVERLAP_1 = 0
ISDNRXOVERLAP_2 = 0
ISDNRXOVERLAP_3 = 0
ISDNRXOVERLAP_4 = 0
ISDNRXOVERLAP_5 = 0
ISDNRXOVERLAP_6 = 0
ISDNRXOVERLAP_7 = 0
ISDNRXOVERLAP_8 = 0
ISDNRXOVERLAP_9 = 0
ISDNRXOVERLAP_10 = 0
ISDNRXOVERLAP_11 = 0
ISDNRXOVERLAP_12 = 0
ISDNRXOVERLAP_13 = 0
ISDNRXOVERLAP_14 = 0
ISDNRXOVERLAP_15 = 0
ISDNRXOVERLAP_16 = 0
ISDNRXOVERLAP_17 = 0
ISDNRXOVERLAP_18 = 0
ISDNRXOVERLAP_19 = 0
DEFAULTNUMBER = 'serveduser'
DISCONNECTONBROKENCONNECTION = 0
ENABLEMWI = 1
ISFAXUSED = 1
TRUNKTRANSFERMODE_0 = 2
TRUNKTRANSFERMODE_1 = 0
TRUNKTRANSFERMODE_2 = 0
TRUNKTRANSFERMODE_3 = 0
TRUNKTRANSFERMODE_4 = 0
TRUNKTRANSFERMODE_5 = 0
TRUNKTRANSFERMODE_6 = 0
TRUNKTRANSFERMODE_7 = 0
TRUNKTRANSFERMODE_8 = 0
TRUNKTRANSFERMODE_9 = 0
TRUNKTRANSFERMODE_10 = 0
TRUNKTRANSFERMODE_11 = 0
```

```
TRUNKTRANSFERMODE_12 = 0
TRUNKTRANSFERMODE_13 = 0
TRUNKTRANSFERMODE_14 = 0
TRUNKTRANSFERMODE_15 = 0
TRUNKTRANSFERMODE_16 = 0
TRUNKTRANSFERMODE_17 = 0
TRUNKTRANSFERMODE_18 = 0
TRUNKTRANSFERMODE_19 = 0
VoiceMailInterface = 3
SIPTRANSPORTTYPE = 2
SUBSCRIPTIONMODE = 1
MEDIASECURITYPEHAVIOUR = 1
PLAYRBTONE2TRUNK_0 = 0
PLAYRBTONE2TRUNK_1 = -1
PLAYRBTONE2TRUNK_2 = -1
PLAYRBTONE2TRUNK_3 = -1
PLAYRBTONE2TRUNK_4 = -1
PLAYRBTONE2TRUNK_5 = -1
PLAYRBTONE2TRUNK_6 = -1
PLAYRBTONE2TRUNK_7 = -1
PLAYRBTONE2TRUNK_8 = -1
PLAYRBTONE2TRUNK_9 = -1
PLAYRBTONE2TRUNK_10 = -1
PLAYRBTONE2TRUNK_11 = -1
PLAYRBTONE2TRUNK_12 = -1
PLAYRBTONE2TRUNK_13 = -1
PLAYRBTONE2TRUNK_14 = -1
PLAYRBTONE2TRUNK_15 = -1
PLAYRBTONE2TRUNK_16 = -1
PLAYRBTONE2TRUNK_17 = -1
PLAYRBTONE2TRUNK_18 = -1
PLAYRBTONE2TRUNK_19 = -1

[SCTP Params]
[VXML Params]
[IPsec Params]
[Audio Staging Params]

; *** TABLE CoderName ***
[ CoderName ]
FORMAT CoderName_Index = CoderName_Type, CoderName_PacketInterval,
CoderName_rate, CoderName_PayloadType, CoderName_Sce;
CoderName 0 = g711Ulaw64k, 20, 0, 255, 0;
[ \CoderName ]
```



```
;
; *** TABLE TrunkGroup ***
[ TrunkGroup ]
FORMAT TrunkGroup_Index = TrunkGroup_TrunkGroupNum, TrunkGroup_FirstTrunkId,
TrunkGroup_FirstBChannel, TrunkGroup_LastBChannel,
TrunkGroup_FirstPhoneNumber, TrunkGroup_ProfileId, TrunkGroup_LastTrunkId,
TrunkGroup_Module;
TrunkGroup 0 = 0, 0, 1, 24, 2000, 0, 0, 1;
[ \TrunkGroup ]

;
; *** TABLE Dns2Ip ***
[ Dns2Ip ]
FORMAT Dns2Ip_Index = Dns2Ip_DomainName, Dns2Ip_FirstIpAddress,
Dns2Ip_SecondIpAddress, Dns2Ip_ThirdIpAddress, Dns2Ip_FourthIpAddress;
Dns2Ip 0 = anonymous.invalid, 10.15.10.7, 0.0.0.0, 0.0.0.0, 0.0.0.0;

[ \Dns2Ip ]
; *** TABLE ProxyIp ***
[ ProxyIp ]
FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = 10.15.10.11, -1, 0;
[ \ProxyIp ]

;
; *** TABLE TxDtmfOption ***
[ TxDtmfOption ]
FORMAT TxDtmfOption_Index = TxDtmfOption_Type;
TxDtmfOption 0 = 4;

[ \TxDtmfOption ]

;
; *** TABLE ProxySet ***
[ ProxySet ]
FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive,
ProxySet_ProxyKeepAliveTime, ProxySet_ProxyLoadBalancingMethod,
ProxySet_IsProxyHotSwap;
ProxySet 0 = 0, 60, 0, 0;
[ \ProxySet ]
```

**SIP**

**Mediant 1000**

## **Configuration Note**

**AudioCodes' Mediant VoIP Gateway and  
Avaya's Modular Messaging**

*with*

**Definity G3 Prologix and S87x0/S8x00  
using E1 QSIG**