

SIP

Mediant 1000

Configuration Note

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

with

Siemens HiPath 4000 using E1 QSIG Interface

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, 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.

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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 Siemens HiPath 4000 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

| | |
|----------------------------|-----------------------|
| PBX Vendor | Siemens |
| Model | HiPath 4000 |
| Software Version | Version 2.0 SA09 RL00 |
| Telephony Signaling | E1 QSIG |
| Additional Notes | None |

1.2 AudioCodes Gateway

| | |
|-------------------------|---------------|
| Gateway Vendor | AudioCodes |
| Model | Mediant 1000 |
| Software Version | 5.60A.012.007 |
| VoIP Protocol | SIP |
| Additional Notes | None |

1.3 Avaya Modular Messaging Version

| | |
|----------------|-------------------------------------|
| Version | Avaya Modular Messaging Release 5.0 |
|----------------|-------------------------------------|

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 "CN88514.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.

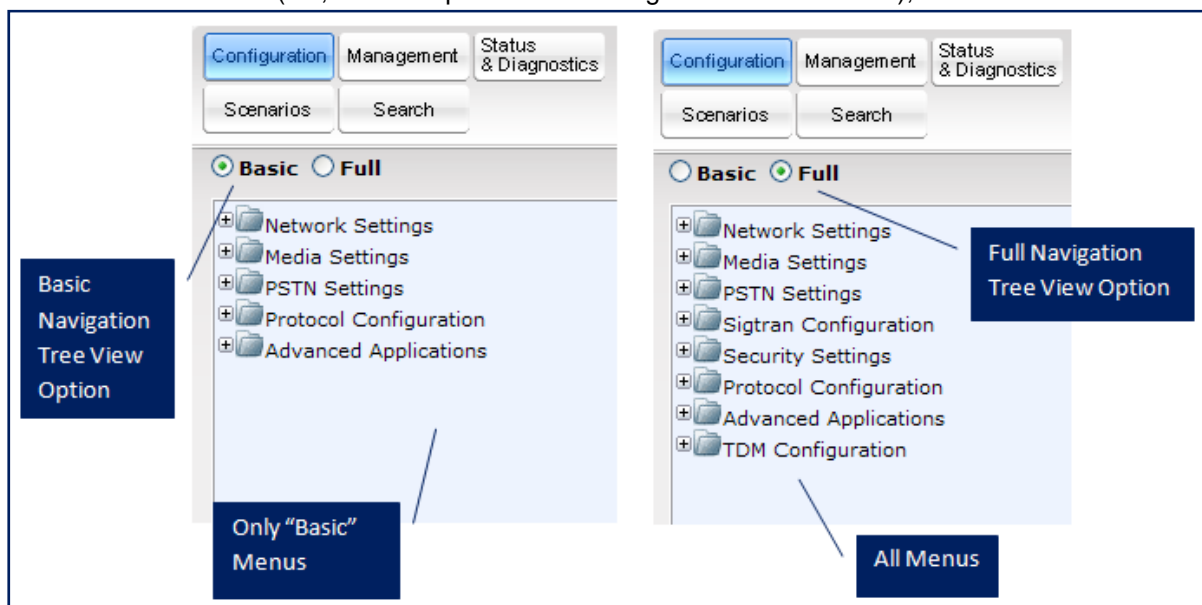
4.2 Configuring AudioCodes Gateway

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**).

| General Settings | |
|-------------------------------|-------------------------|
| Module ID | 1 |
| Trunk ID | 1 |
| Trunk Configuration State | Inactive |
| Protocol Type | E1 QSIG |
| Trunk Configuration | |
| Clock Master | Recovered |
| Auto Clock Trunk Priority | 0 |
| Line Code | HDB3 |
| Framing Method | E1 FRAMING MFF CRC4 EXT |
| 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 |

| | |
|-------------------------------------|---------------------------|
| PSTN Alert Timeout | -1 |
| QSIG Transfer Mode | Path Replacement Transfer |
| 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: 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 10: 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 11: 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 12: 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 13: Modify Parameters in the AdminPage

The following changes need to be made in the AdminPage:

- **ISDNIBehavior:** "1073741824"
- **EnableMWI:** "1"
- **ECNLPMODE:** "1"
- **SubscriptionMode:** "1"
- **MWISIGMSGCENTRELDIDPARTYNUMBER:** "209849"
- **VoiceMailInterface:** "6"
- **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.

The screenshot shows the 'Admin Page' in Internet Explorer. The browser address bar displays `http://10.15.10.7/AdminPage`. The page contains a sidebar with navigation options: 'Image Load to Device', 'ini Parameters', and 'Back to Main'. The main content area features two parameter configuration forms. The first form is for the 'ISDNIBEHAVIOR' parameter, with the 'Parameter Name' field containing 'ISDNIBEHAVIOR' and the 'Enter Value' field containing '1073741824'. The 'Apply New Value' button is visible to the right. The second form is for the '3GDelErrorSDU' parameter, with the 'Parameter Name' field containing '3GDelErrorSDU' and the 'Enter Value' field empty. Below the forms is an 'Output Window' displaying the following text: 'Parameter Name: ISDNIBEHAVIOR', 'Parameter New Value:1073741824', and 'Parameter Description:Bit-field used to determine several behavior options, which influence how the Q.931 protocol behaves.' Arrows and boxes labeled 'Step 1' through 'Step 5' indicate the sequence of actions: Step 1 (URL), Step 2 (Parameter Name), Step 3 (Value), Step 4 (Apply button), and Step 5 (Output window).

Step 14: 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 | <input type="button" value="Reset"/> |
| Burn To FLASH | <input type="text" value="Yes"/> |
| Graceful Option | <input type="text" value="No"/> |
| ▼ LOCK / UNLOCK | |
| Lock | <input type="button" value="LOCK"/> |
| Graceful Option | <input type="text" value="No"/> |
| Current Admin State | UNLOCKED |
| ▼ Save Configuration | |
| Burn To FLASH | <input type="button" value="BURN"/> |

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

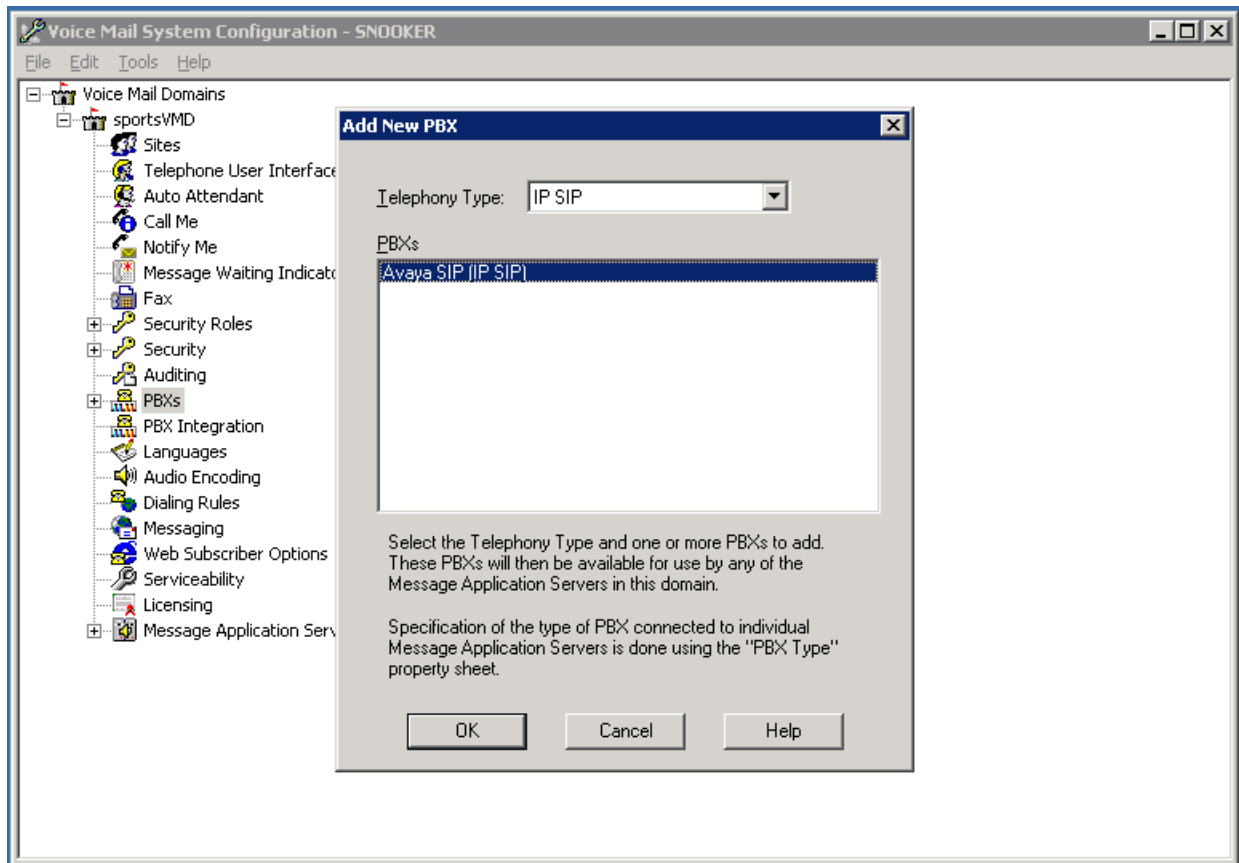
Reader Notes

5 Avaya Modular Messaging Configuration

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

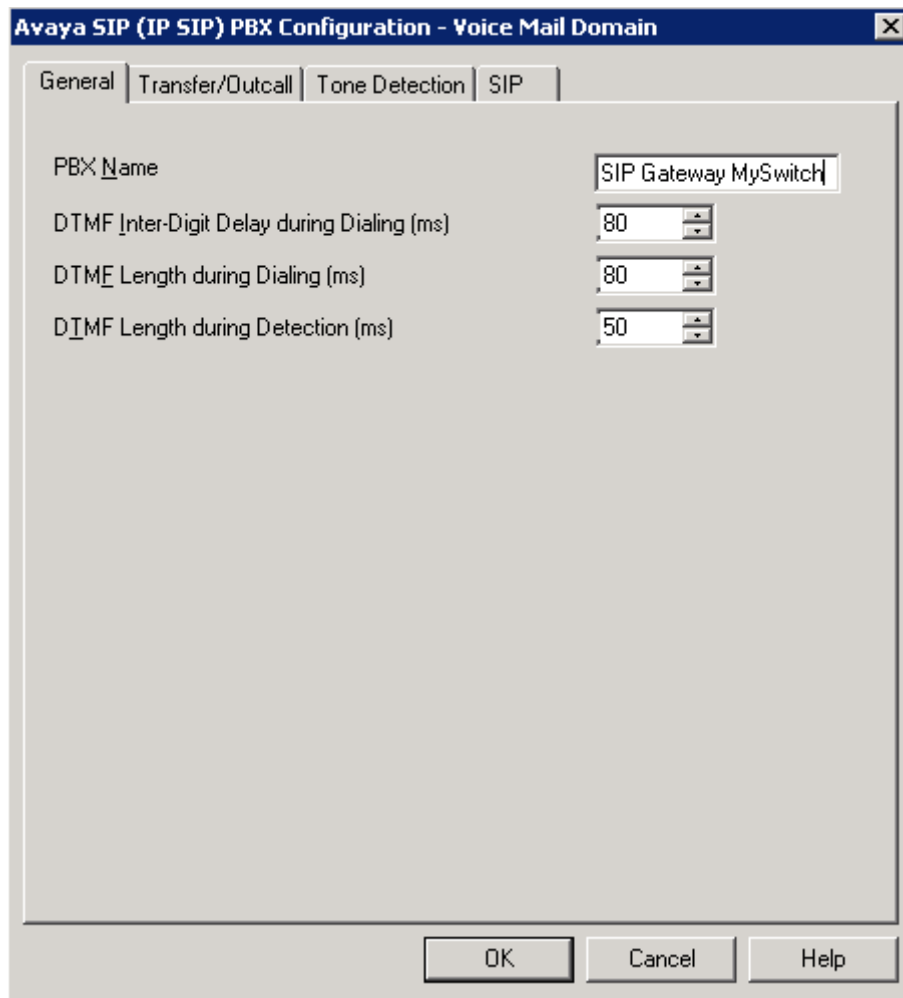
If required, use Voice Mail System Configuration (VMSC) to create a new SIP PBX for the Avaya Modular Messaging Voice Mail Domain (VMD):

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.



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.

PBX Integration - Voice Mail Domain

IP SIP

Port Details

RTP Port Range: 7000 - 7900

Packet Size Bytes: 20

Protocols Details

TLS Port Number: 5061

ICP Port Number: 5060 Enable

OK Cancel 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.

PBX Type - SNOOKER

General

Select the Telephony Type and PBX used by this Message Application Server

Telephony Type: IP SIP

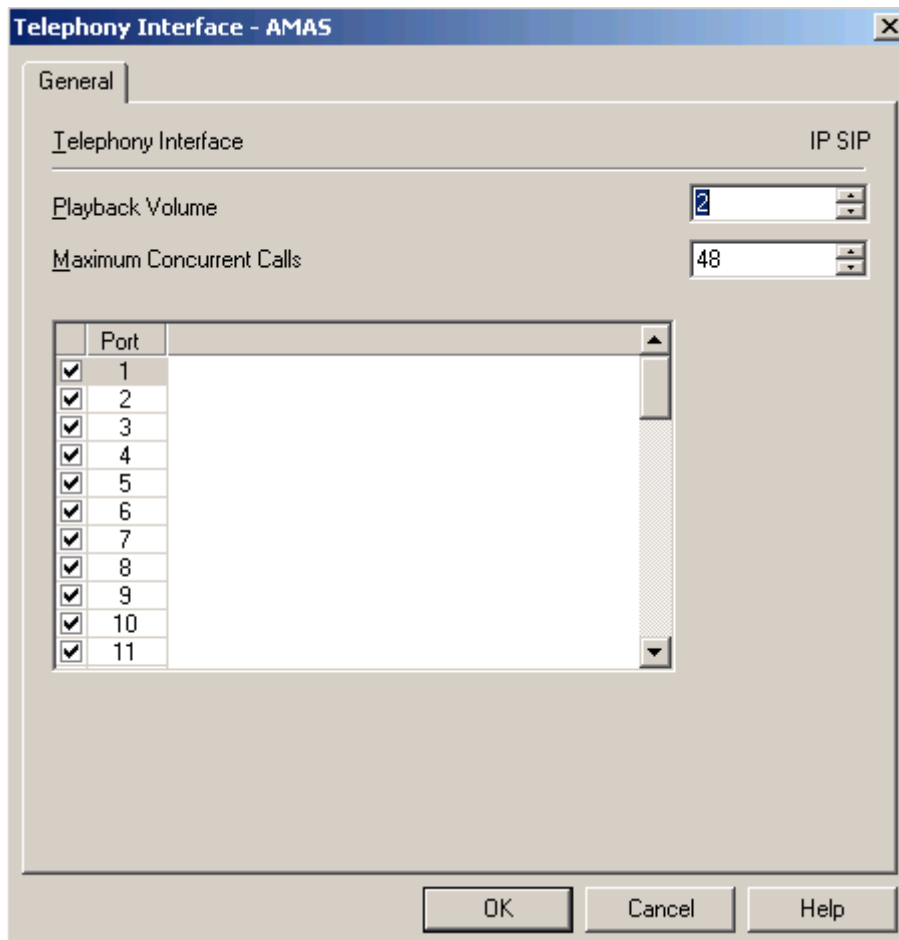
PBXs

Avaya SIP (IP SIP)

OK Cancel Help

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

15. Select the Telephony Interface Item.

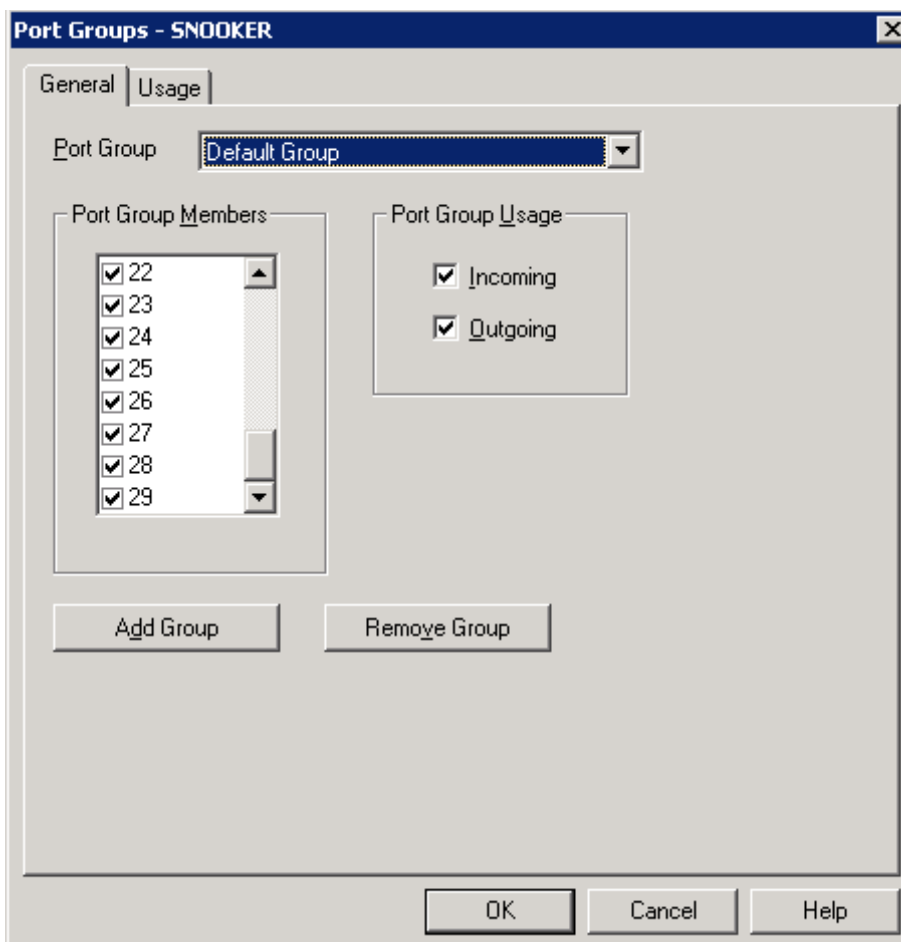


16. Set the Maximum Concurrent Calls to match the number of Channels that you want on this MAS, and then click **OK**.

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

| | |
|--|------|
| Failure Point | None |
| Phone Type (if phone-specific) | None |
| Call Scenarios Associated with Failure Point | None |
| List of UM Features Affected by Failure Point | None |
| Additional Comments | 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 shows the 'Management Settings' page in the AudioCodes gateway configuration interface. The left sidebar shows a tree view with 'Management Configuration' expanded to 'Management Settings'. 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' (a dropdown menu set to 'Enable'), and 'Trunks Filter' (-1). Two black arrows point to the 'Syslog Server IP Address' field (labeled 'Step 2') and the 'Enable Syslog' dropdown (labeled 'Step 1').
- SNMP Settings:** Contains fields for 'SNMP Trap Destinations', 'SNMP Community String', 'SNMP V3 Table', 'SNMP Trusted Managers', 'Disable SNMP' (a dropdown menu set to 'No'), and 'Trap Manager Host Name'.
- Activity Types to Report via 'Activity Log' Messages:** A list of activity types with checkboxes: '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 in the bottom right corner of the interface.

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.

The screenshot shows the 'Advanced Parameters' configuration page for the AudioCodes Mediant 1000 Gateway. The left sidebar contains a tree view with 'Full' selected under 'Basic'. The main area is divided into several sections:

- 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: None
 - Enable Fax Re-Routing: Disable
- CDR and Debug:**
 - CDR Server IP Address: [Empty]
 - CDR Report Level: End Call
 - Debug Level: 5
- Misc. Parameters:**
 - Progress Indicator to IP: Not Configured
 - Enable X-Channel Header: Disable
 - Enable Busy Out: Disable
 - Default Release Cause: 3

Two black arrows point to the 'CDR Report Level' and 'Debug Level' fields, labeled 'Step 4' and 'Step 3' respectively. A 'Submit' button is located at the bottom right.

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.012.007
;DSP Software Version: 624AE3 => 560.11
;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]

TLSVersion = 1

[BSP Params]

PCMLawSelect = 1
TDMBusClockSource = 4
TDMBusLocalReference = 1
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

ProtocolType_0 = 21

ProtocolType_1 = 0

FramingMethod_0 = c

FramingMethod_1 = 0

LineCode_0 = 0

LineCode_1 = 2

ISDNBehavior = 1073741824

ISDNInCallsBehavior = 0

ISDNOutCallsBehavior = 1024

ISDNGeneralCCBehavior = 0

ISDNNFASInterfaceID = 255

IUAInterfaceID = -1

NFASGroupNumber = 0

PSTNReserved3 = 8

DIGITALPORTINFO = "

AutoClockTrunkPriority = 0

DPNSSBehavior = 12

CasTrunkDialPlanName = "

BriLayer2Mode = 0

[SS7 Params]

[Voice Engine Params]

IdlePCMPattern = 85
ECNLPMMode = 1
BrokenConnectionEventTimeout = 3
DTMFDetectorSensitivity = 1
AggressiveDTMFErasure = 770
TTYTRANSPORTTYPE = 1

[WEB Params]

HTTPSCipherString = 'RC4:EXP'

[SIP Params]

ISPROXYUSED = 1
SIPDESTINATIONPORT = 5061
MAXACTIVECALLS = 29
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
SUBSCRIPTIONMODE = 1
SIPTRANSPORTTYPE = 2
PROGRESSINDICATOR2ISDN = -1
LOCALISDNRBSOURCE = 0
ISDNTRANSFERCAPABILITY = -1
PIFORDISCONNECTMSG = -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
MEDIASECURITYPEHAVIOUR = 1
TRUNKPSTNALERTTIMEOUT_0 = -1
TRUNKPSTNALERTTIMEOUT_1 = -1
TRUNKPSTNALERTTIMEOUT_2 = -1
```



```
TRUNKPSTNALERTTIMEOUT_3 = -1
TRUNKPSTNALERTTIMEOUT_4 = -1
TRUNKPSTNALERTTIMEOUT_5 = -1
TRUNKPSTNALERTTIMEOUT_6 = -1
TRUNKPSTNALERTTIMEOUT_7 = -1
TRUNKPSTNALERTTIMEOUT_8 = -1
TRUNKPSTNALERTTIMEOUT_9 = -1
TRUNKPSTNALERTTIMEOUT_10 = -1
TRUNKPSTNALERTTIMEOUT_11 = -1
TRUNKPSTNALERTTIMEOUT_12 = -1
TRUNKPSTNALERTTIMEOUT_13 = -1
TRUNKPSTNALERTTIMEOUT_14 = -1
TRUNKPSTNALERTTIMEOUT_15 = -1
TRUNKPSTNALERTTIMEOUT_16 = -1
TRUNKPSTNALERTTIMEOUT_17 = -1
TRUNKPSTNALERTTIMEOUT_18 = -1
TRUNKPSTNALERTTIMEOUT_19 = -1
MWISOURCENUMBER = '209898'
RTPONLYMODEFORTRUNK = -1
BCHANNELNEGOTIATIONFORTRUNK = -1
DIGITALOOSBEHAVIORFORTRUNK_0 = 0
DIGITALOOSBEHAVIORFORTRUNK_1 = -1
DIGITALOOSBEHAVIORFORTRUNK_2 = -1
DIGITALOOSBEHAVIORFORTRUNK_3 = -1
DIGITALOOSBEHAVIORFORTRUNK_4 = -1
DIGITALOOSBEHAVIORFORTRUNK_5 = -1
DIGITALOOSBEHAVIORFORTRUNK_6 = -1
DIGITALOOSBEHAVIORFORTRUNK_7 = -1
DIGITALOOSBEHAVIORFORTRUNK_8 = -1
DIGITALOOSBEHAVIORFORTRUNK_9 = -1
DIGITALOOSBEHAVIORFORTRUNK_10 = -1
DIGITALOOSBEHAVIORFORTRUNK_11 = -1
DIGITALOOSBEHAVIORFORTRUNK_12 = -1
DIGITALOOSBEHAVIORFORTRUNK_13 = -1
DIGITALOOSBEHAVIORFORTRUNK_14 = -1
DIGITALOOSBEHAVIORFORTRUNK_15 = -1
DIGITALOOSBEHAVIORFORTRUNK_16 = -1
DIGITALOOSBEHAVIORFORTRUNK_17 = -1
DIGITALOOSBEHAVIORFORTRUNK_18 = -1
DIGITALOOSBEHAVIORFORTRUNK_19 = -1
VoiceMailInterface = 6
MWIQSIGMSGCENTRELDIDPARTYNUMBER = '209849'
```

[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

Siemens HiPath 4000 using E1 QSIG Interface