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.

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 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.

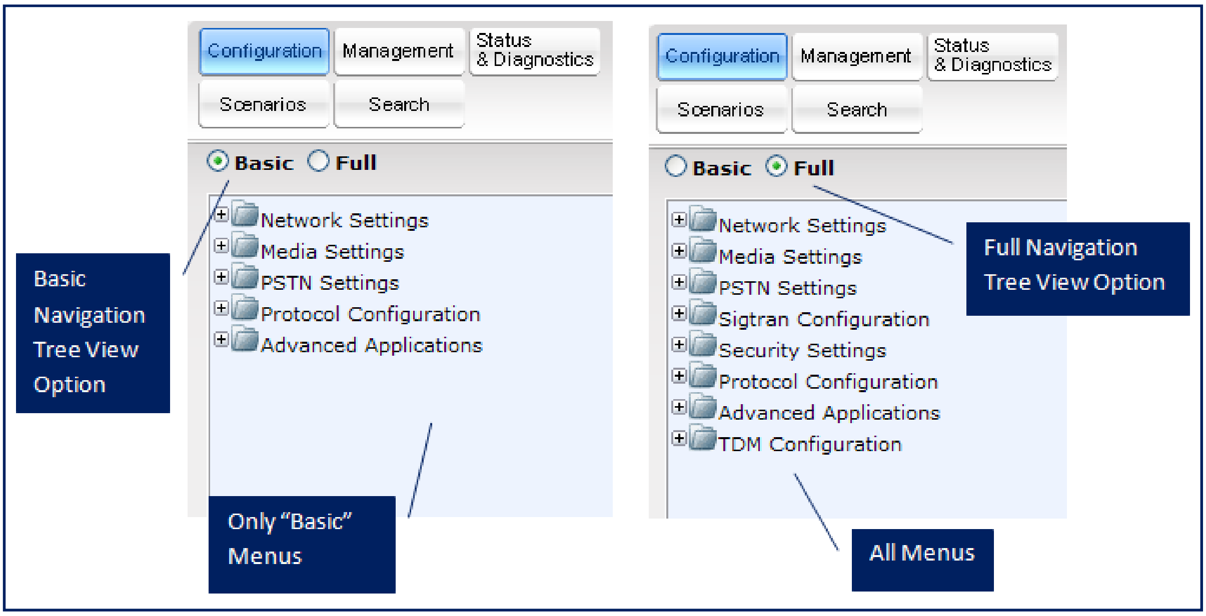
4.2 Configuring AudioCodes Gateway

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.

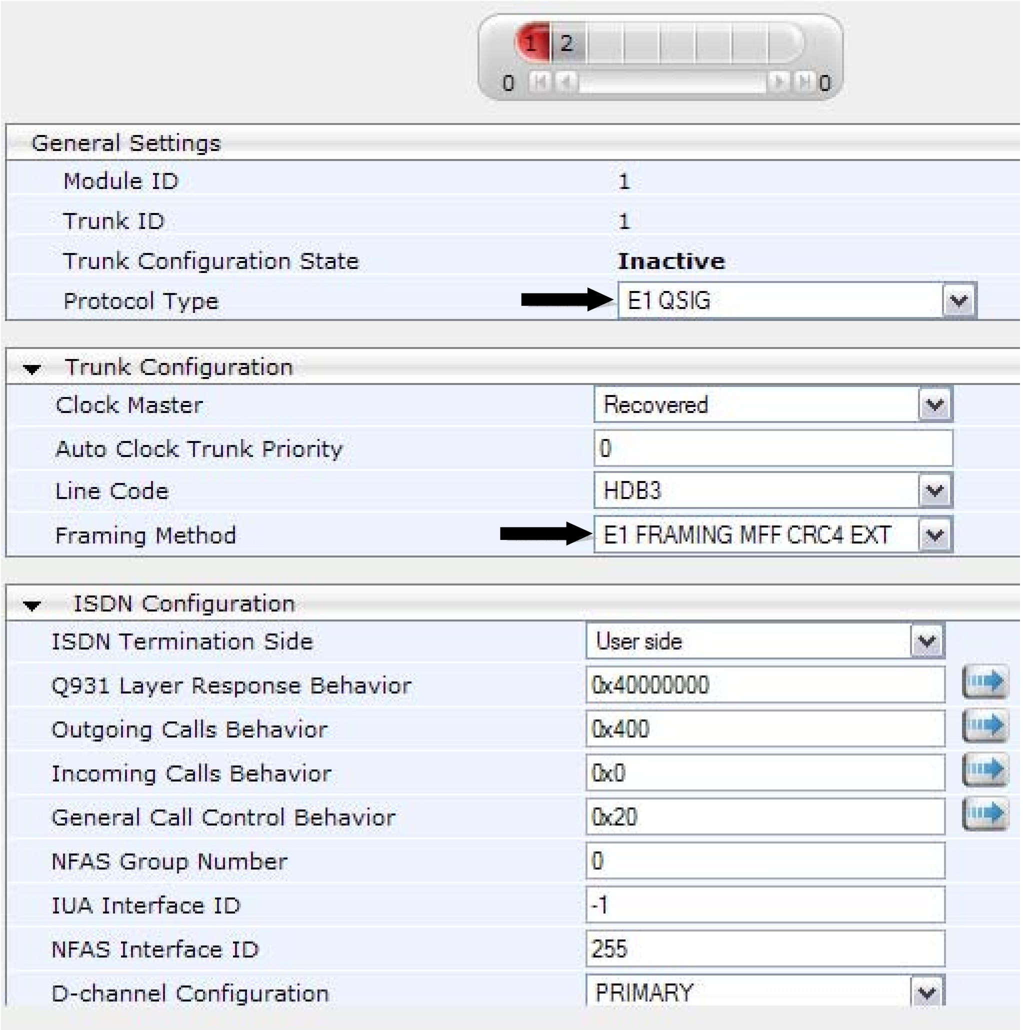
Note the following Web interface guidelines:

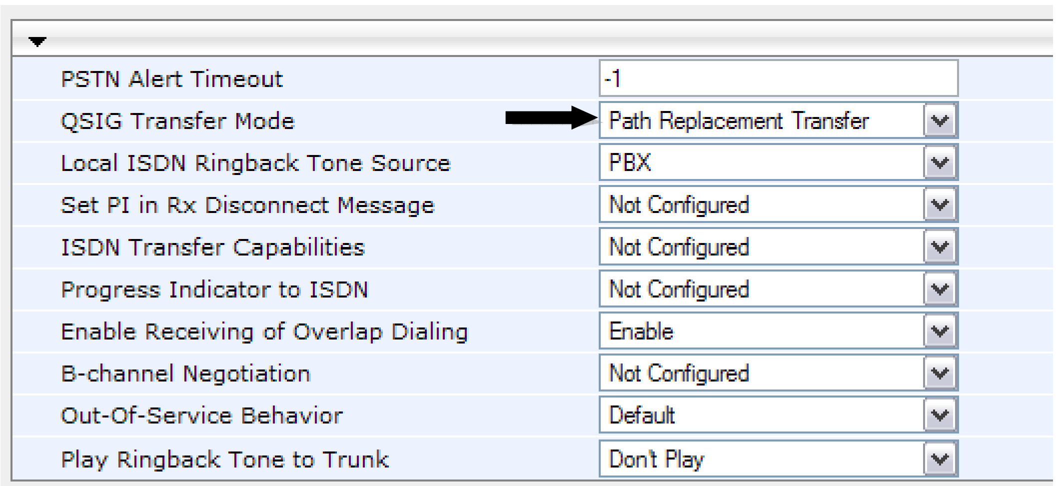
- When making configuration changes for each procedure, ensure that you click the **Submit** button to save your changes; unless otherwise instructed.
- Some of the changes may require a gateway reset for these changes to take effect. Therefore, (and to save time), reset the gateway only after you complete 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**).





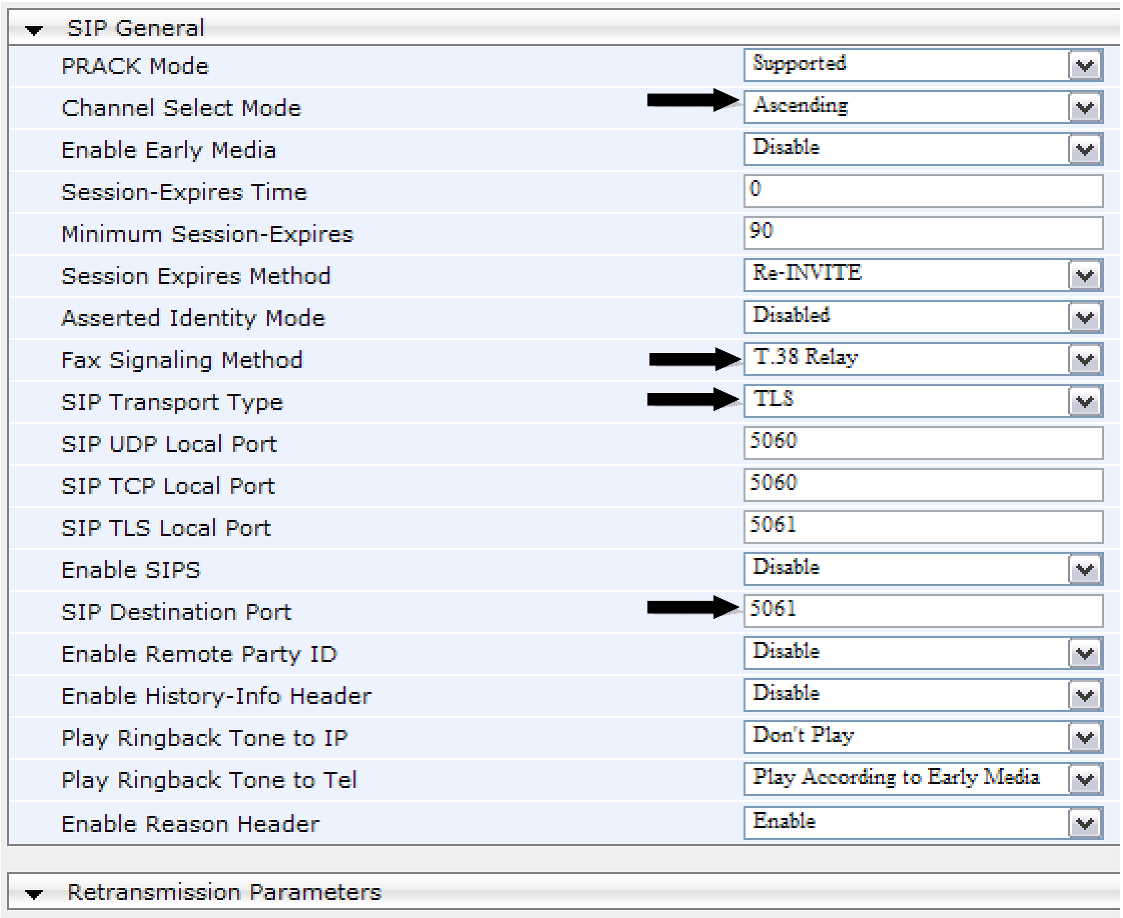
Before you can modify parameters on this page, you need to click the **Stop Trunk** button to de-activate the trunk.

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**).

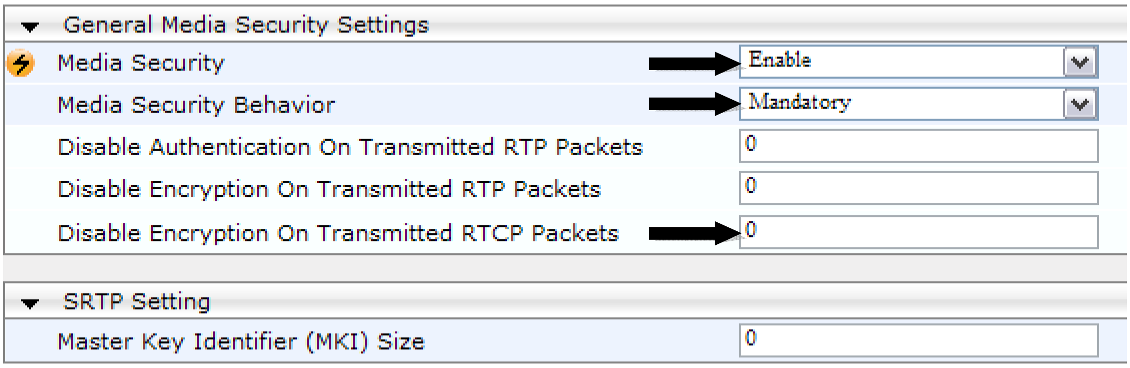


It is recommended that you configure the gateway and Avaya's Modular Messaging to use TLS. If you prefer to use TCP, then ensure that you configure the gateway settings relating to TLS (in the screen above) to use TCP:

- **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**).



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

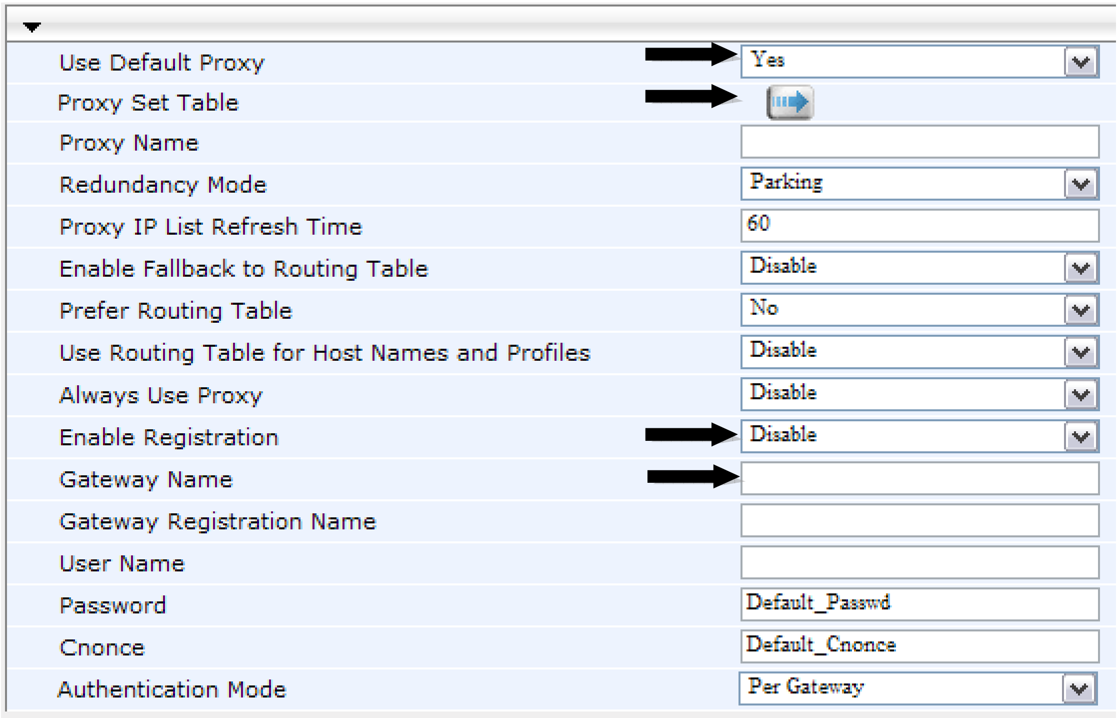
- **Media Security:** "Enable"
- **Media Security Behavior:** "Mandatory"
- **Disable Encryption On Transmitted RTCP Packets:** "1"

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

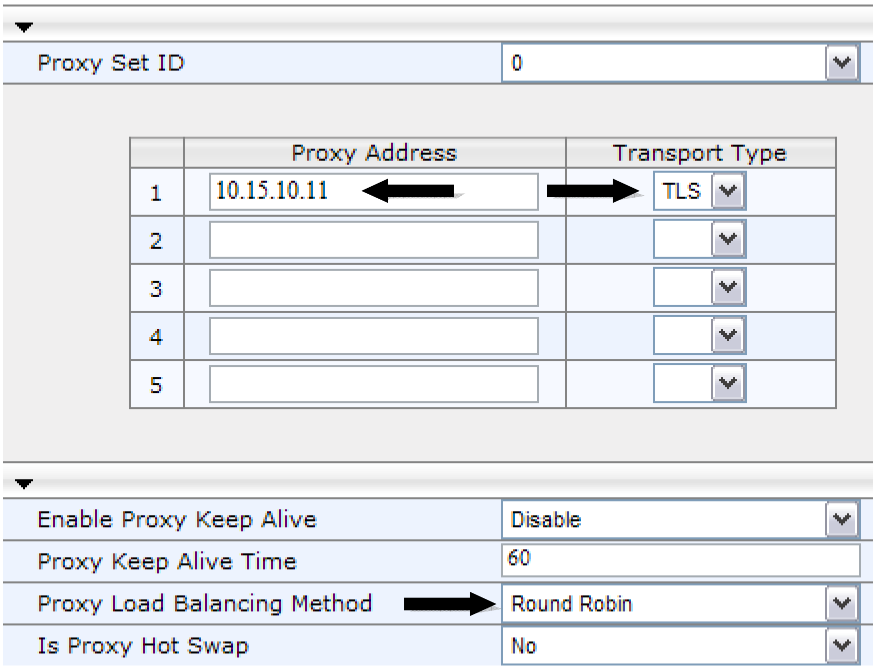
- **Media Security:** "Disabled"
- **Media Security Behavior:** "Mandatory"
- **Disable Encryption On Transmitted RTCP Packets:** "0"

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**).



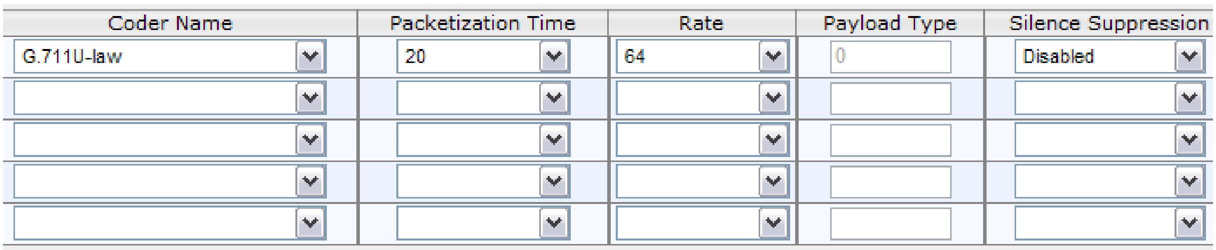
Assign an FQDN name to the gateway (for example, mygateway.mynet1.mynet.com). Any gateway name that corresponds to your network environment is applicable, but it must meet requirements for FQDNs.



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

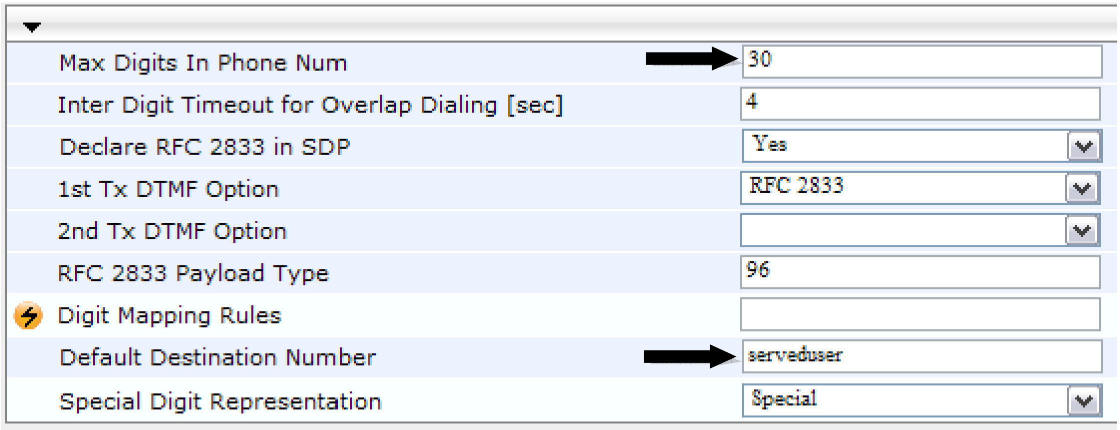
Open the 'Coders Table' page (**Configuration** tab: **Protocol Configuration** > **Protocol Definition** > **Coders**).



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**).

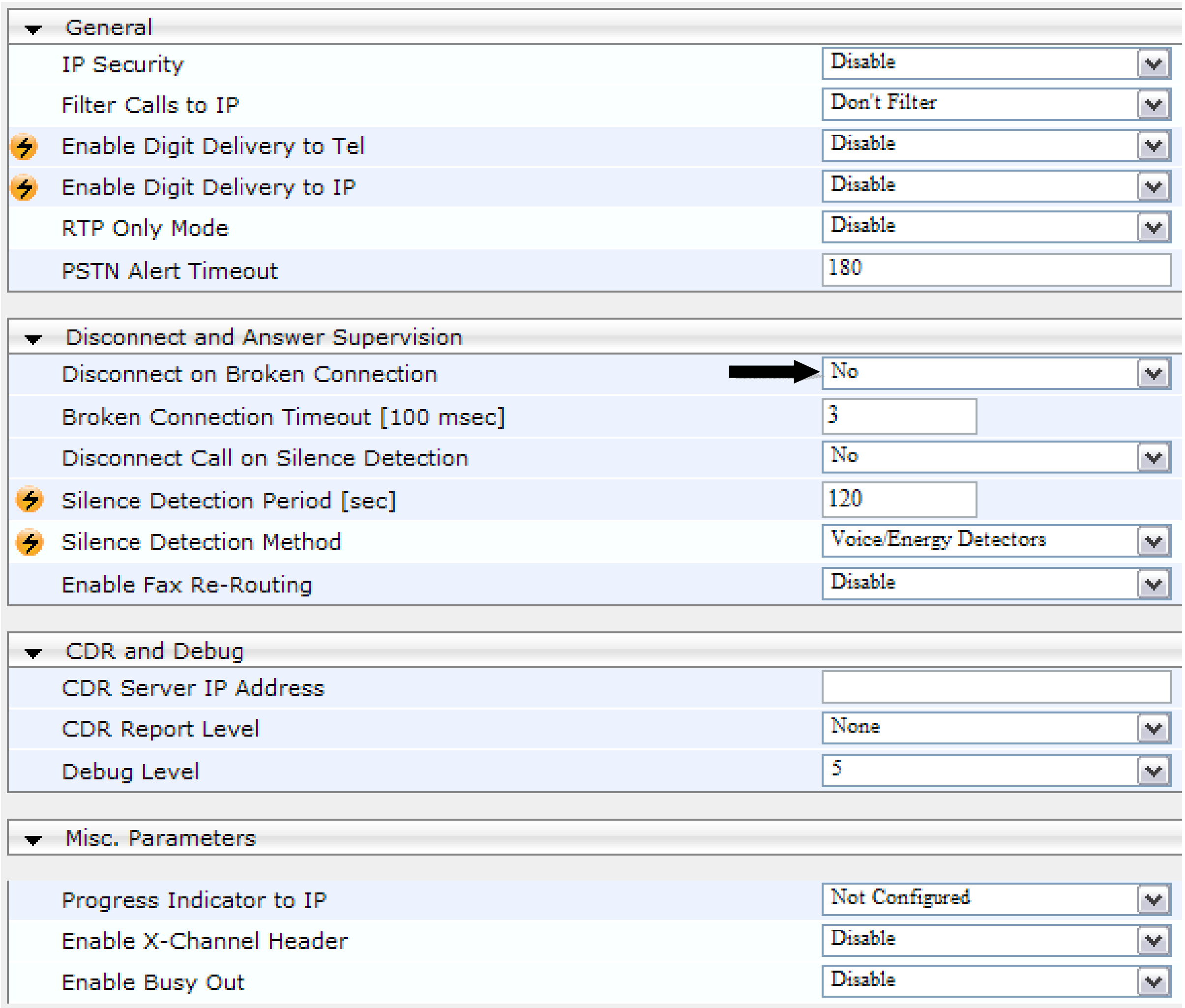


Set the following fields to the value indicated:

- **Default Destination Number:** “serveduser”

Step 7: General Setup

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



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

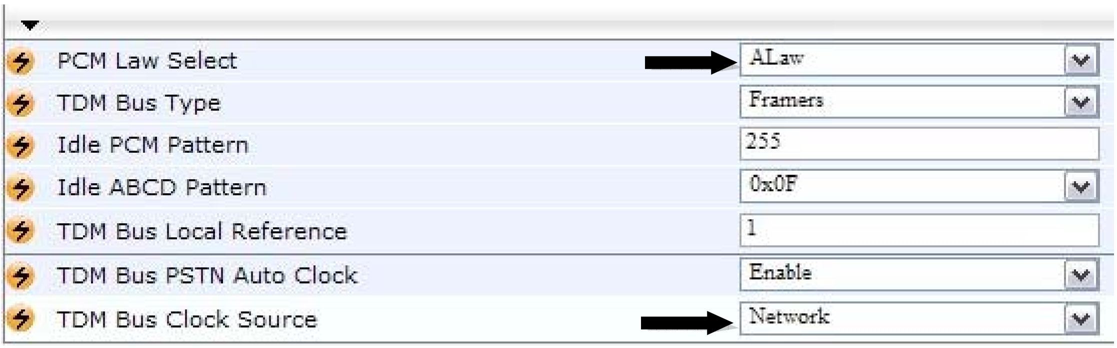
Group Index	Module	From Trunk	To Trunk	Channels	Phone Number	Trunk Group ID	IP Profile ID
1	Module 1 PRI	1	1	1-24	2000		0
2							
3							
4							
5							
6							
7							
8							
9							
10							
11							
12							

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

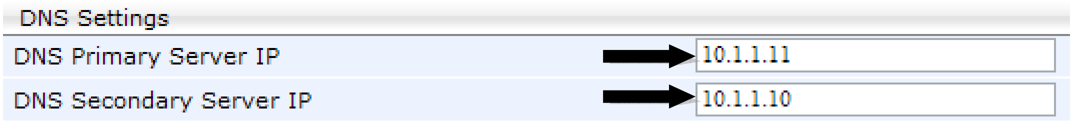
Open the 'TDM Bus Settings' page (**Configuration** tab: **TDM Configuration** > **TDM Bus Settings**).



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**).



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.

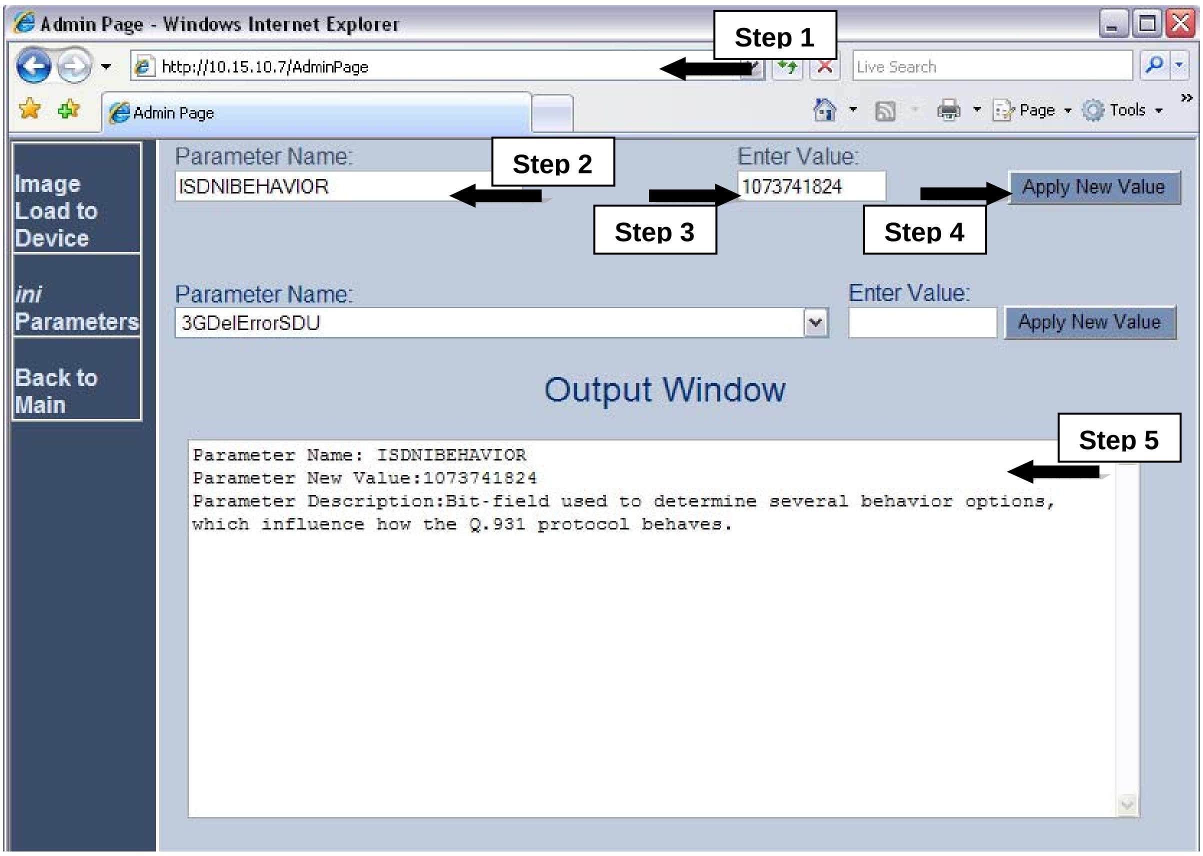
Step 13: Modify Parameters in the AdminPage

The following changes need to be made in the AdminPage:

- **ISDNIBehavior:** "1073741824"
- **EnableMWI:** "1"
- **ECNLPMode:** "1"
- **SubscriptionMode:** "1"
- **MWIQSIGMSGCENTRELDIDPARTYNUMBER:** "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:

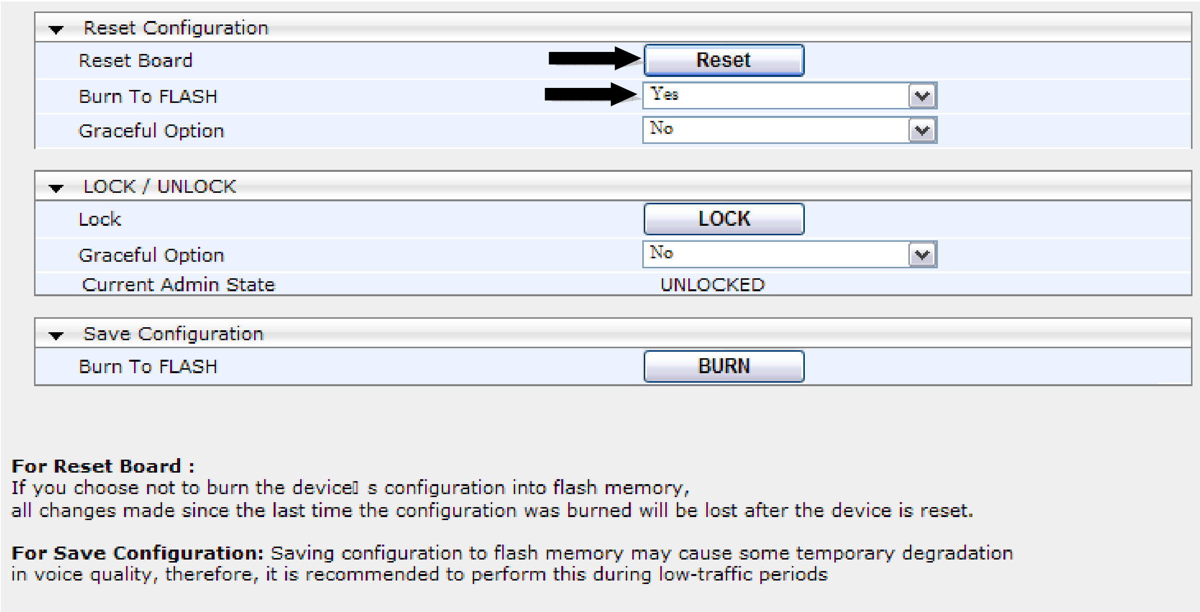
1. Open the 'AdminPage' page at the following URL (case-sensitive):
   `http://<gateway's IP address>/AdminPage`
2. In the 'Parameter Name', enter the parameter's name.
3. In the 'Enter Value', enter the parameter's value.
4. Click the **Apply New Value** button.
5. Check the output.



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").



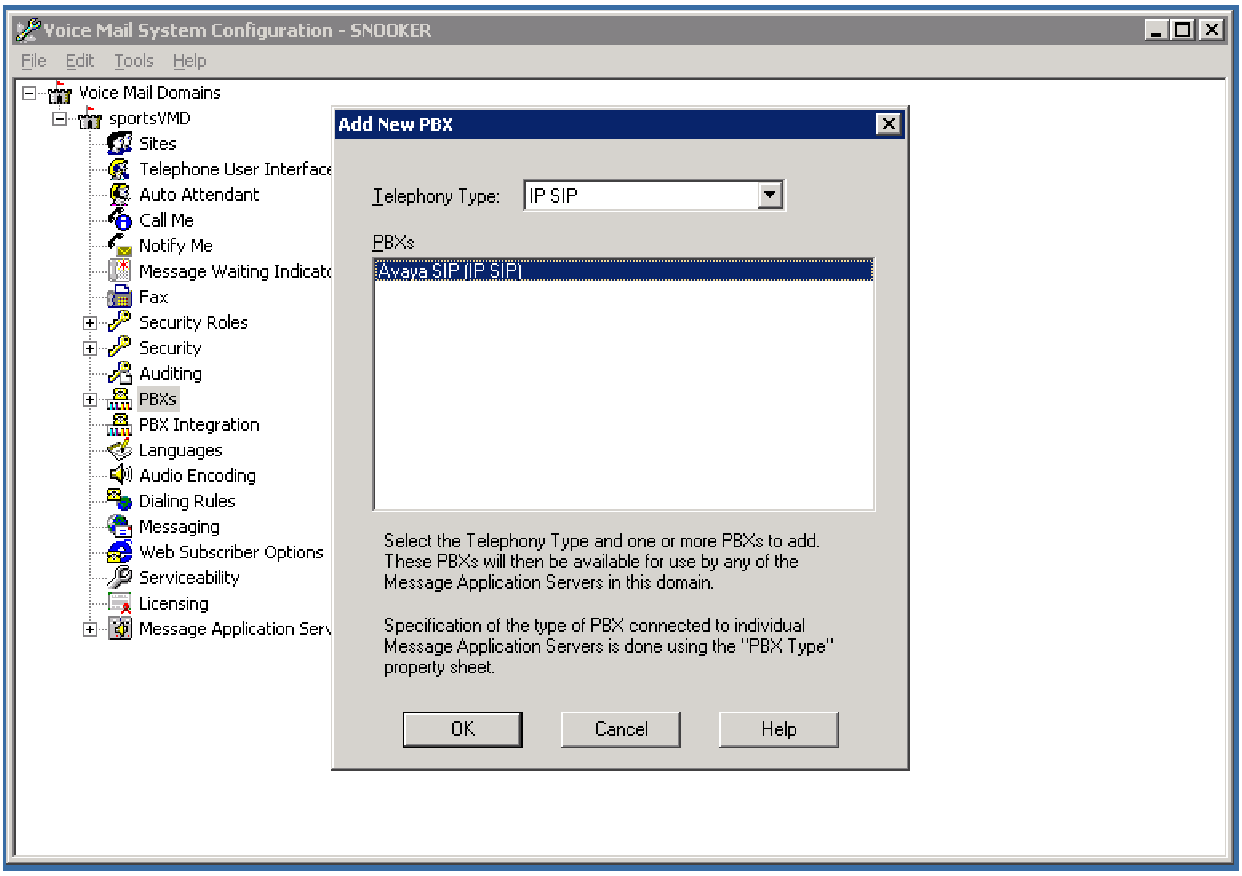
Reader Notes

5 Avaya Modular Messaging Configuration

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

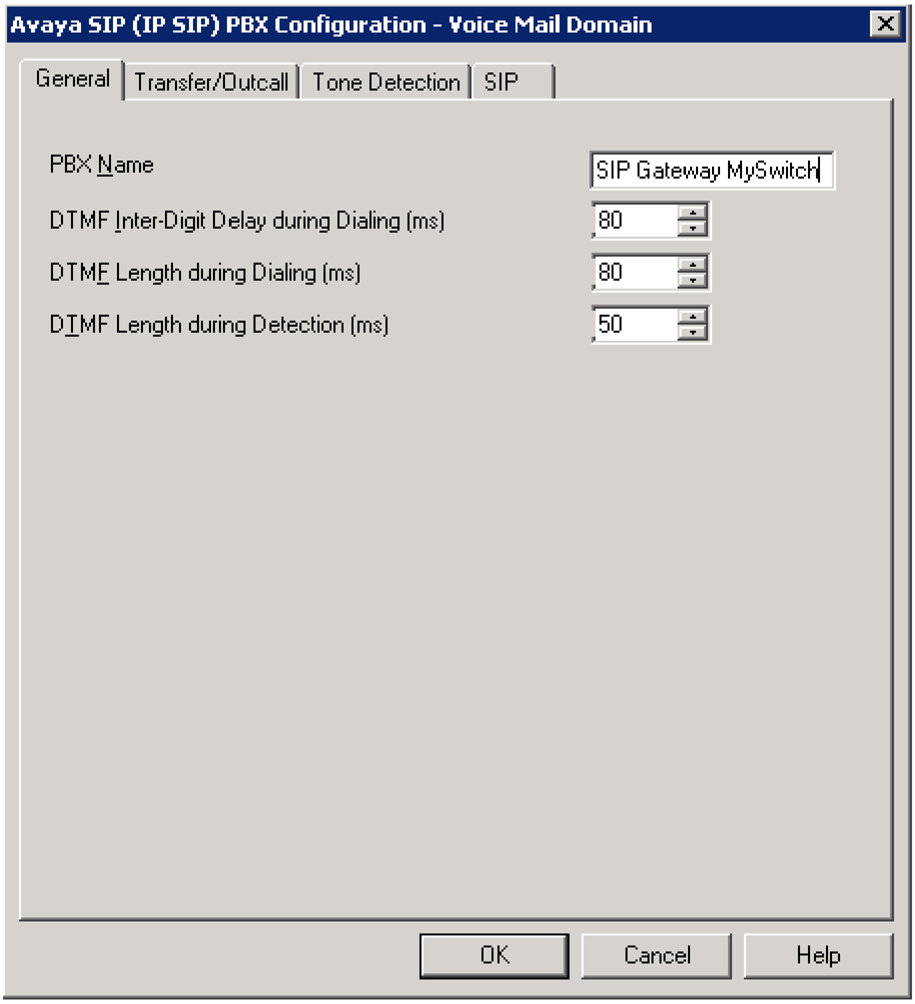
If required, use Voice Mail System Configuration (VMSC) to create a new SIP PBX for the Avaya Modular Messaging Voice Mail Domain (VMD):

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".



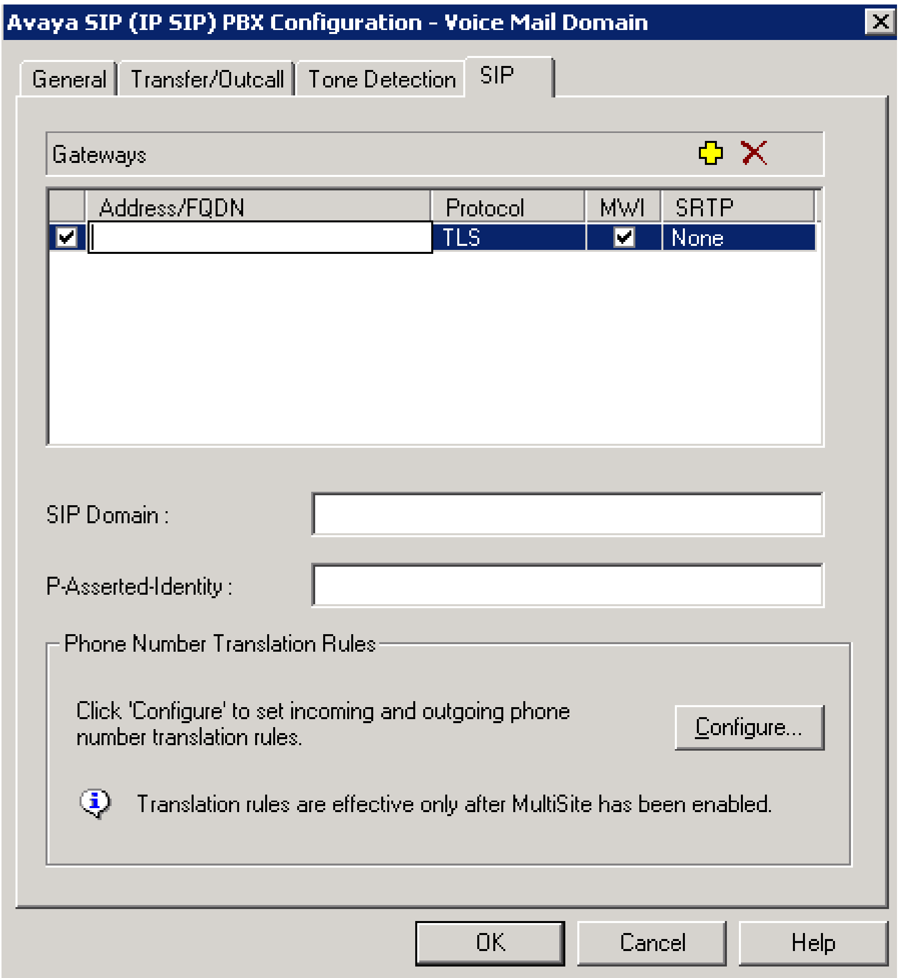
5. From the list of available PBXs, select "Avaya SIP (IP SIP)", and then click **OK**.

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.



9. Enter the following details on the SIP configuration page:

a. Enter the FQDN or IP address of the AudioCodes gateway you are installing.

b. Select the protocol for the AudioCodes gateway.

Note: It is recommended to use TLS.

c. Select "MWI" if the AudioCodes gateway handles MWI requests.

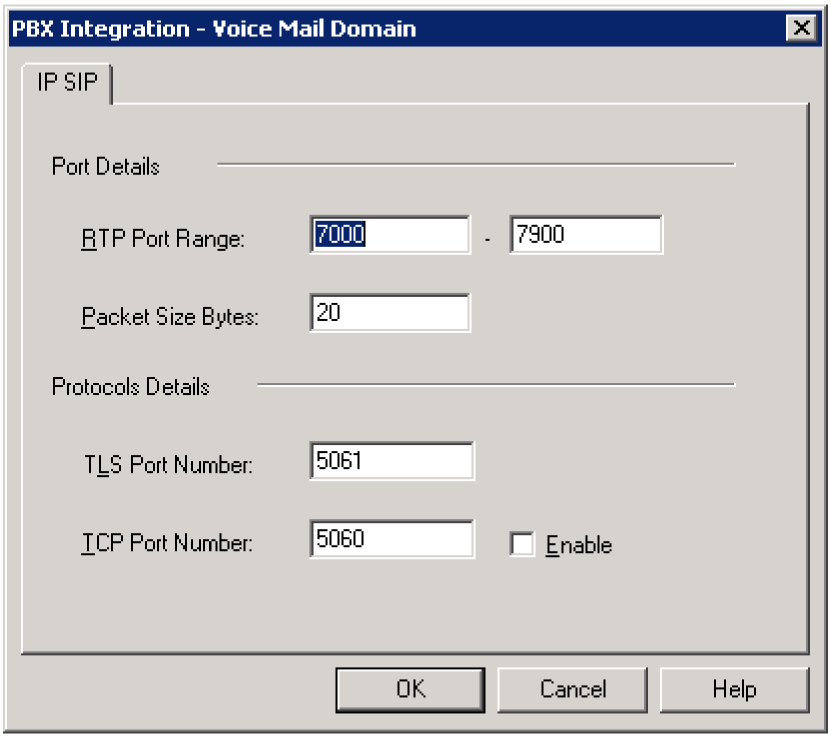
d. Select the SRTP configuration required for the AudioCodes gateway:

- If using SIP over TLS with SRTP, then select the required SRTP level.
- If using SIP over TLS without SRTP, then select "None".

e. Enter a unique SIP Domain to be used with the AudioCodes gateway.

f. Click **OK**.

10. Select the VMD PBX Integration.

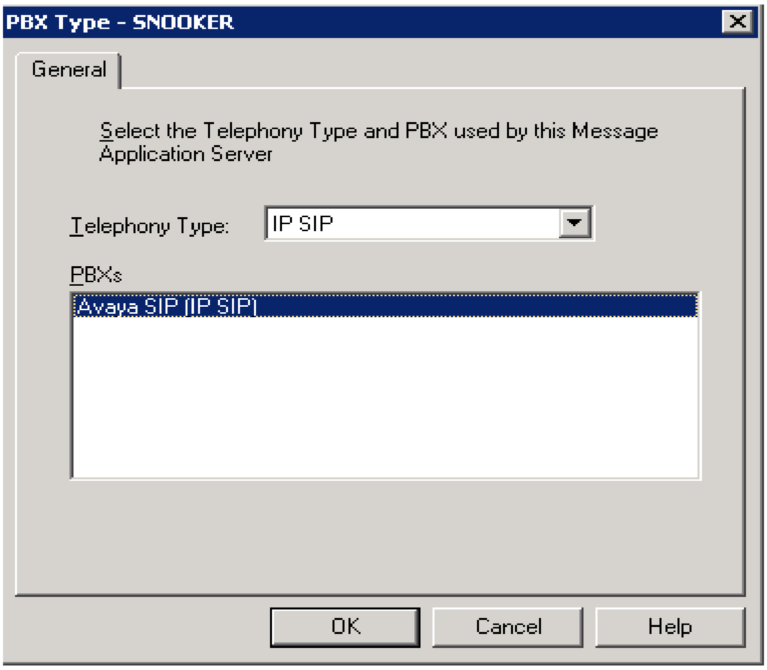


11. Configure the IP SIP settings as required, and then click **OK**, (the figure above shows the default settings).

    Note: If you are using SIP over TCP, then select the 'TCP Enable' check box.

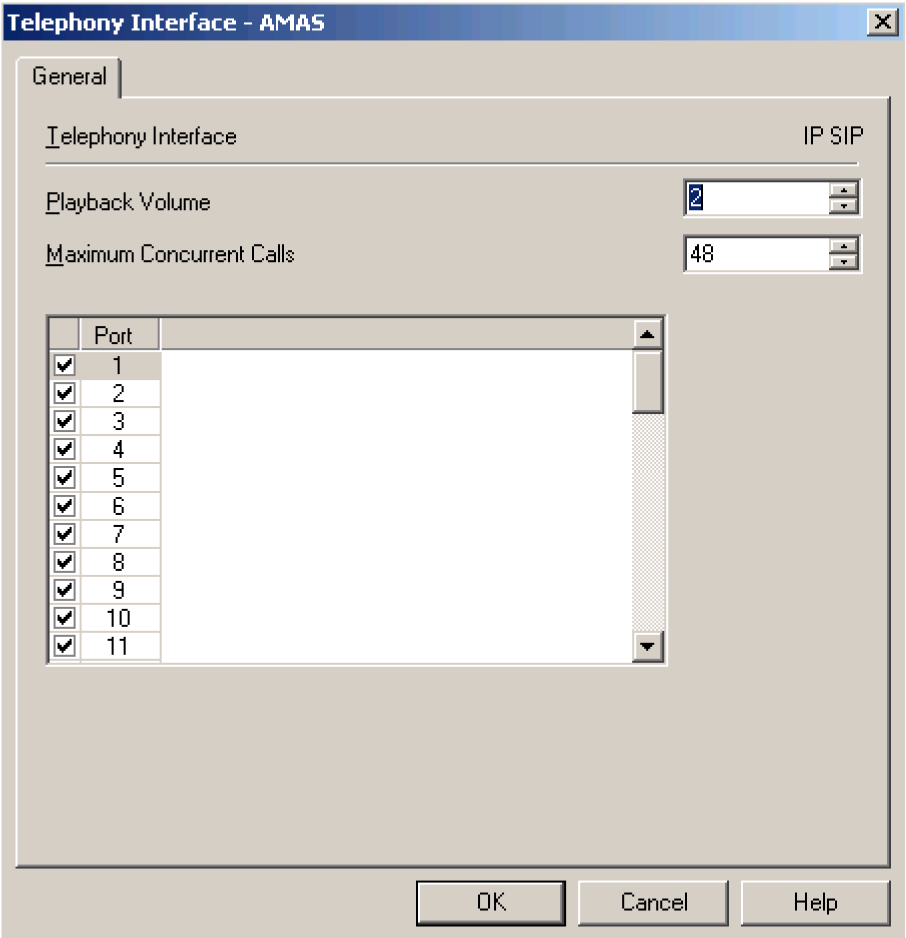
12. Expand the Message Application Servers (MAS) item.

13. From the drop-down list, select the Telephony Type IP SIP.



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

15. Select the Telephony Interface Item.

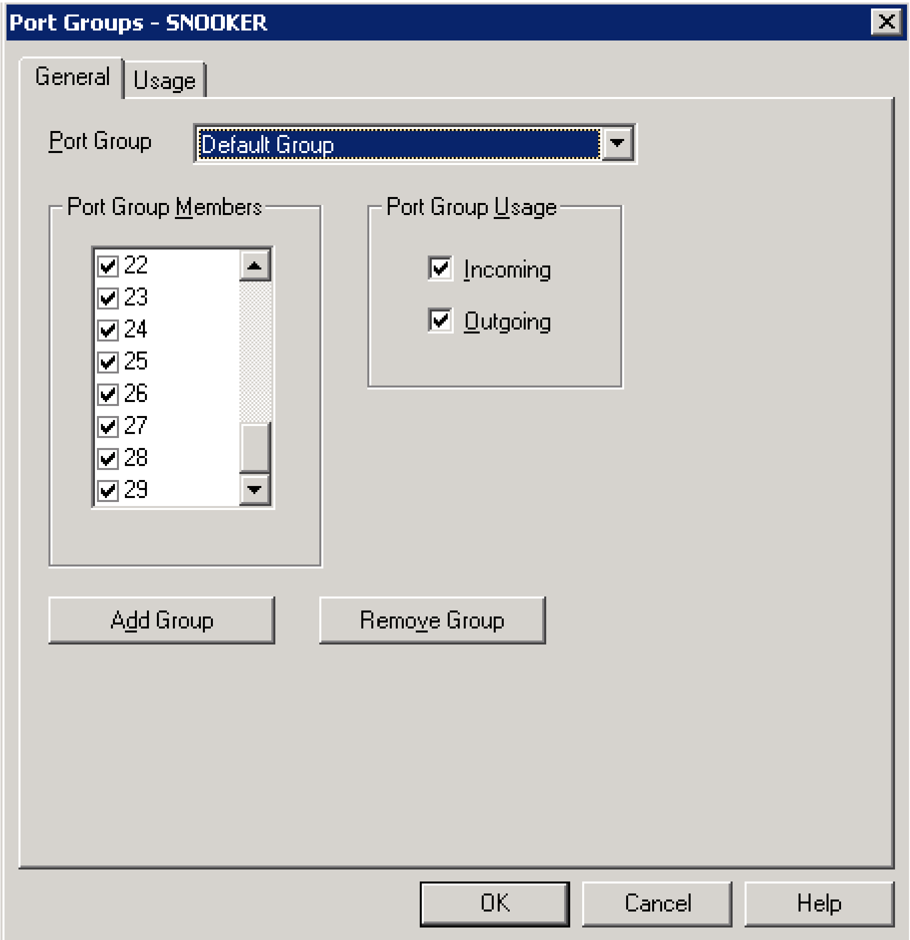


16. Set the Maximum Concurrent Calls to match the number of Channels that you want on this MAS, and then click **OK**.

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.

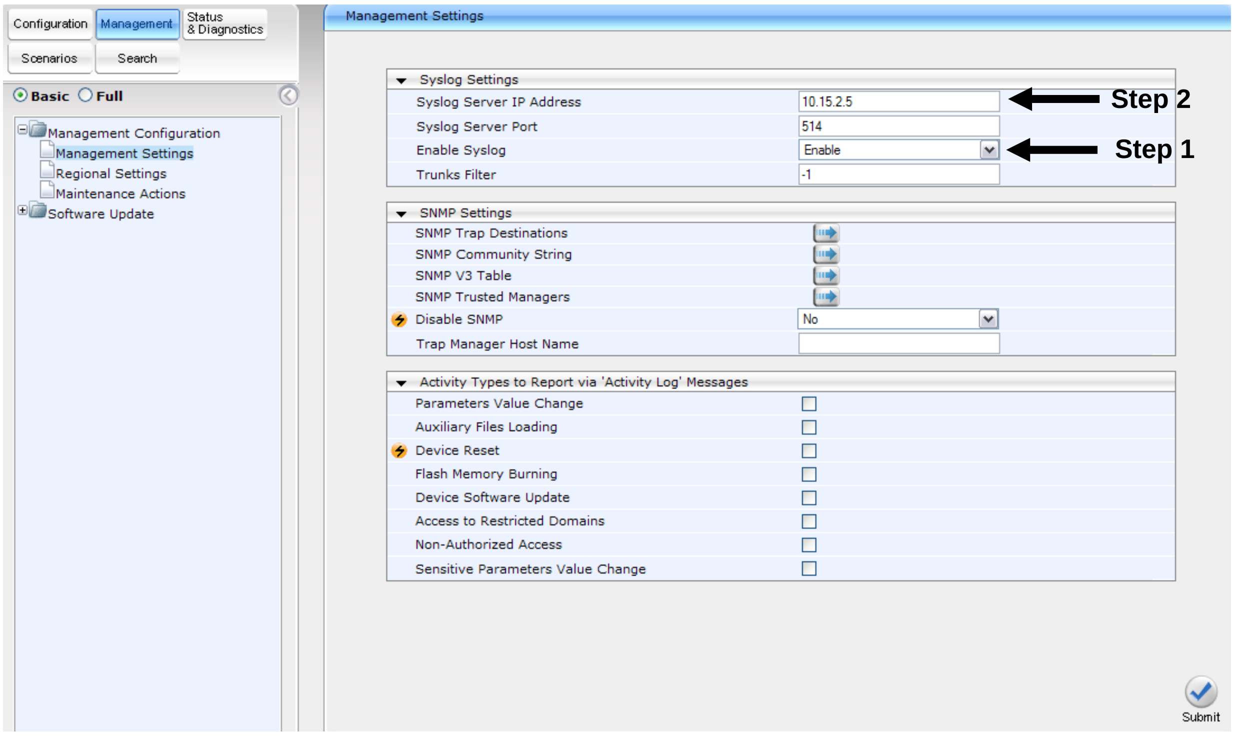
7.1 Configuring AudioCodes Gateway for Syslog Server

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.

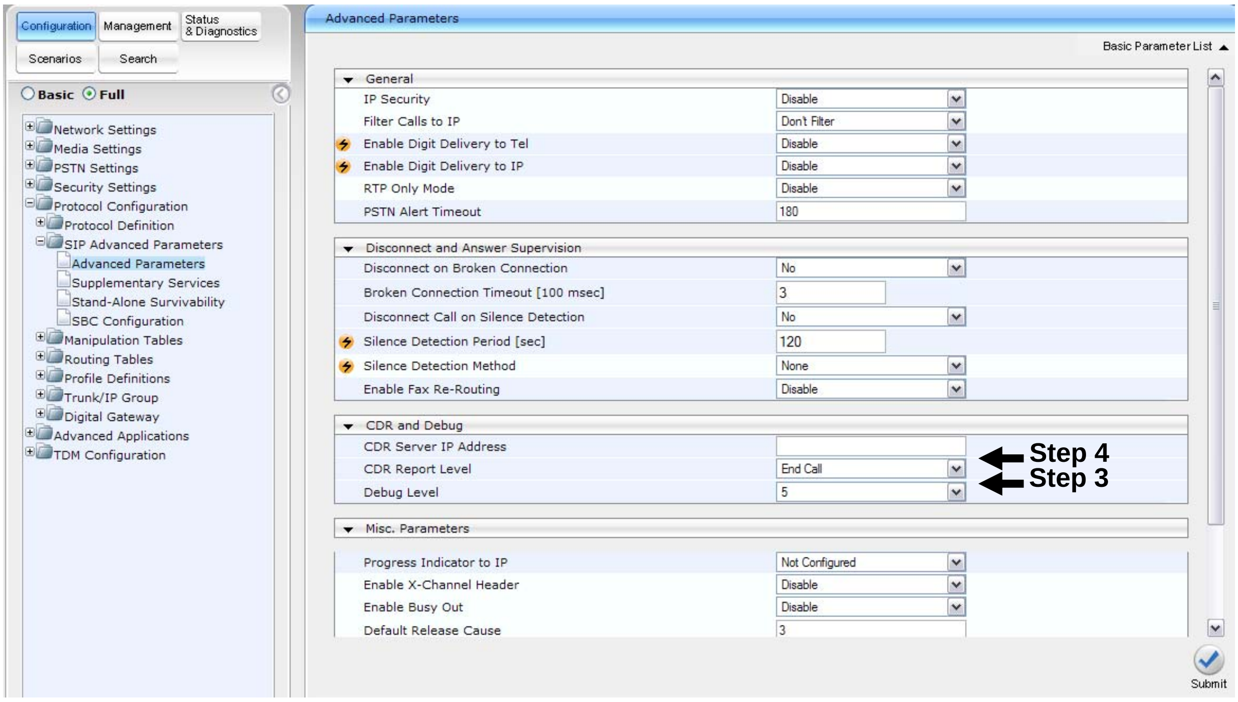
➢ To activate the Syslog client on the AudioCodes gateways:

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).



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.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
ISDNIBehavior = 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
ECNLPMode = 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
MEDIASECURITYBEHAVIOUR = 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