

SIP

Mediant 1000

Configuration Note

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

with

Avaya's Definity G3 PBX using T1 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	April 2009	Initial version by AudioCodes

Overview

This document describes the configuration required to setup Avaya Definity and AudioCodes' Mediant 1000 gateway, using T1 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	Avaya
Model	Definity G3
Software Version	R013i.01.1.628.7
Telephony Signaling	T1 QSIG
Additional Notes	None

1.2 AudioCodes Gateway

Gateway Vendor	AudioCodes
Model	Mediant 1000
Software Version	5.60A.007.004
VoIP Protocol	SIP
Additional Notes	

1.3 Avaya Modular Messaging Version

Version	Avaya Modular Messaging Release 5.0
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Reader's Notes

2 Prerequisites

2.1 Gateway Prerequisites

None.

2.2 PBX Prerequisites

Enter the command **display system-parameters customer-options**, and then ensure that the QSIG OPTIONAL FEATURES parameters (on screen 3 of the PBX configuration tool, page 28) are set to “y”.

PBX hardware must include an installed Trunk Card Module TN464F.

2.3 Cabling Requirements

This integration uses standard RJ-48c cables to connect digital trunks (T1) between TN464F and Mediant 1000 trunk interfaces.

Reader's Notes

3 PBX Setup Notes

3.1 Special Instructions for PBX Configuration

Information used for this test case:

- Digital VoiceMail ports: T1 QSIG trunk.
- VoiceMail Hunt Group Pilot: *03
- VoiceMail User Phone: ext. 1001, ext. 2001, ext. 2002 and ext. 2003
- Coverage Path for ext. 1001, 2001, 2002 and 1003 is 8
- User Test Phone Avaya 8410D and Analog phone

Step 1: Display System Parameters and Customer Options

Use the **Display System Parameters Customer Option** command to ensure all required software features are enabled on the PBX. Below is an example of the forms required for QSIG integration.

```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display system-parameters customer-options Page 4 of 10
OPTIONAL FEATURES
Emergency Access to Attendant? y IP Stations? y
  Enable 'dadmin' Login? y
  Enhanced Conferencing? y ISDN Feature Plus? y
  Enhanced EC500? y ISDN Network Call Redirection? n
  ISDN-BRI Trunks? y
  ISDN-PRI? y
  Local Survivable Processor? n
  Malicious Call Trace? y
  Media Encryption Over IP? n
  Mode Code for Centralized Voice Mail? y
  Extended Cvg/Fwd Admin? y
  External Device Alarm Admin? y
  Five Port Networks Max Per MCC? n
  Flexible Billing? n
  Forced Entry of Account Codes? y
  Global Call Classification? y
  Hospitality <Basic>? y
  Hospitality <G303 Enhancements>? y
  IP Trunks? y
  Multimedia Appl. Server Interface <MASI>? n
  Multimedia Call Handling <Basic>? y
  Multimedia Call Handling <Enhanced>? y
  IP Attendant Consoles? y
  (NOTE: You must logoff & login to effect the permission changes.)
  
```

Note: The 'ISDN-PRI' field must be set to 'y'.

```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display system-parameters customer-options Page 5 of 10
OPTIONAL FEATURES

Multiple Level Precedence & Preemption? n      Station and Trunk MSP? n
Multiple Locations? y                          Station as Virtual Extension? y

Personal Station Access (PSA)? y              System Management Data Transfer? n
Posted Messages? y                            Tenant Partitioning? y
Port Network Support? y                       Terminal Trans. Init. (TTI)? y
                                                Time of Day Routing? y
Processor and System MSP? n                   Usage Allocation Enhancements? y
Private Networking? y                         TN2501 UAL Maximum Capacity? y

                                                Wideband Switching? n
                                                Wireless? n

Remote Office? n
Restrict Call Forward Off Net? y
Secondary Data Module? y

<NOTE: You must logoff & login to effect the permission changes.>

```

Notes:

- The 'Uniform Dialing Plan' field must be set to 'y'.
- The 'Private Networking' field must be set to 'y'.

```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display system-parameters customer-options Page 8 of 10
QSIG OPTIONAL FEATURES

Basic Call Setup? y
Basic Supplementary Services? y
Centralized Attendant? y
Interworking with DCS? y
Supplementary Services with Rerouting? y
Transfer into QSIG Voice Mail? y
Value-Added (VALU)? y

<NOTE: You must logoff & login to effect the permission changes.>

```

Notes:

- The 'Basic Call Setup' field must be set to 'y'.
- The 'Supplementary Services' field must be set to 'y'.
- The 'Supplementary Services with Rerouting' field must be set to 'y'.
- The 'Transfer into QSIG Voice Mail' field must be set to 'y'.

Step 2: Change System Parameter Features

Use the **Change System-Parameters Features** command to define the QSIG TSC Extension, QSIG Path Replacement Extension (use any unused Extensions number) and MWI Number of Digits per Voice Mail Subscriber. This parameter must match the number of digits used for mailbox/extension length.

```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display system-parameters features Page 8 of 17
FEATURE-RELATED SYSTEM PARAMETERS

ISDN PARAMETERS
Send Non-ISDN Trunk Group Name as Connected Name? n
Display Connected Name/Number for ISDN DCS Calls? n
Send ISDN Trunk Group Name on Tandem Calls? n
Send Custom Messages Through QSIG? n

                QSIG TSC Extension: 1999
MWI - Number of Digits Per Voice Mail Subscriber: 4
                Feature Plus Ext: 1997
                National CPN Prefix:
                International CPN Prefix:
                Pass Prefixed CPN to ASAI? n
Unknown Numbers Considered Internal for AUDIX? n
USNI Calling Name for Outgoing Calls? n
Path Replacement with Measurements? y
                QSIG Path Replacement Extension: 1998
Path Replace While in Queue/Vectoring? y
  
```

Step 3: Change Feature Access Codes

Change the feature and assign your private network access codes, in this example we assigned *17.

```

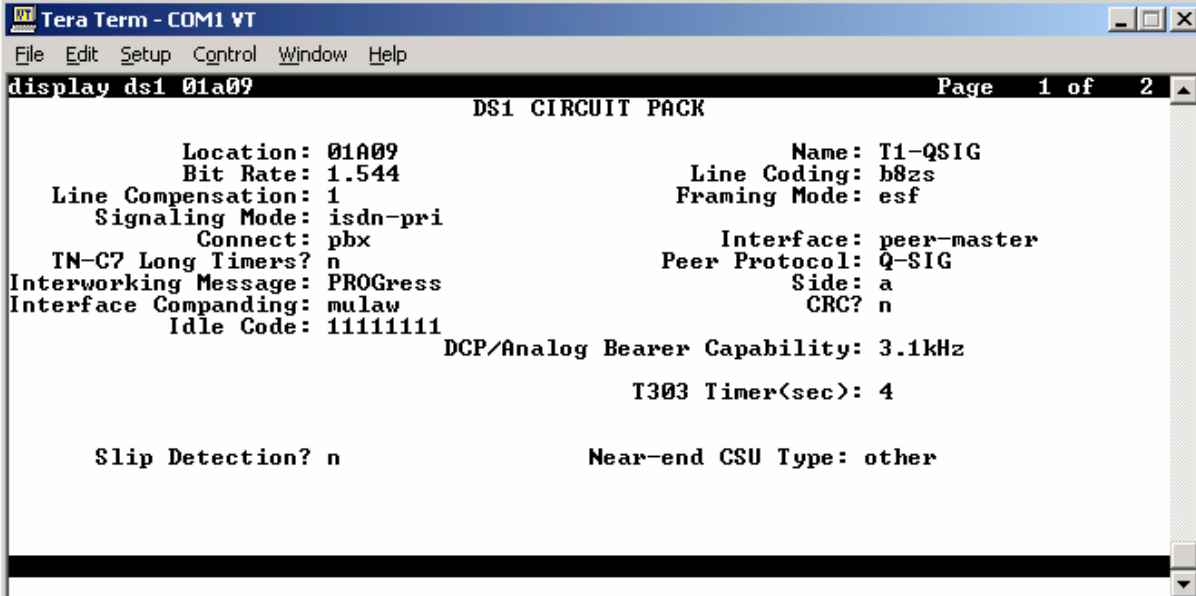
Tera Term - COM1 VT
File Edit Setup Control Window Help
display feature-access-codes Page 1 of 7
FEATURE ACCESS CODE <FAC>
Abbreviated Dialing List1 Access Code: *11
Abbreviated Dialing List2 Access Code: *12
Abbreviated Dialing List3 Access Code: *13
Abbreviated Dial - Prgm Group List Access Code: #12
Announcement Access Code: *27
Answer Back Access Code: #25

Auto Alternate Routing <AAR> Access Code: *17
Auto Route Selection <ARS> - Access Code 1: 9 Access Code 2:
Automatic Callback Activation: *41 Deactivation: #41
Call Forwarding Activation Busy/DA: #02 All: *14 Deactivation: #14
Call Park Access Code: *25
Call Pickup Access Code: *77
CAS Remote Hold/Answer Hold-Unhold Access Code: *22
CDR Account Code Access Code: *24
Change COR Access Code:
Change Coverage Access Code:
  
```

Step 4: T1 Setting using E1/T1 Card

Define T1 using the E1/T1 card that is connected to the Mediant 1000, using the **Add DS1** command. In the example below, the T1 is configured in slot no. 7 (location: 01A07).

Configure the E1/T1 card as follows:



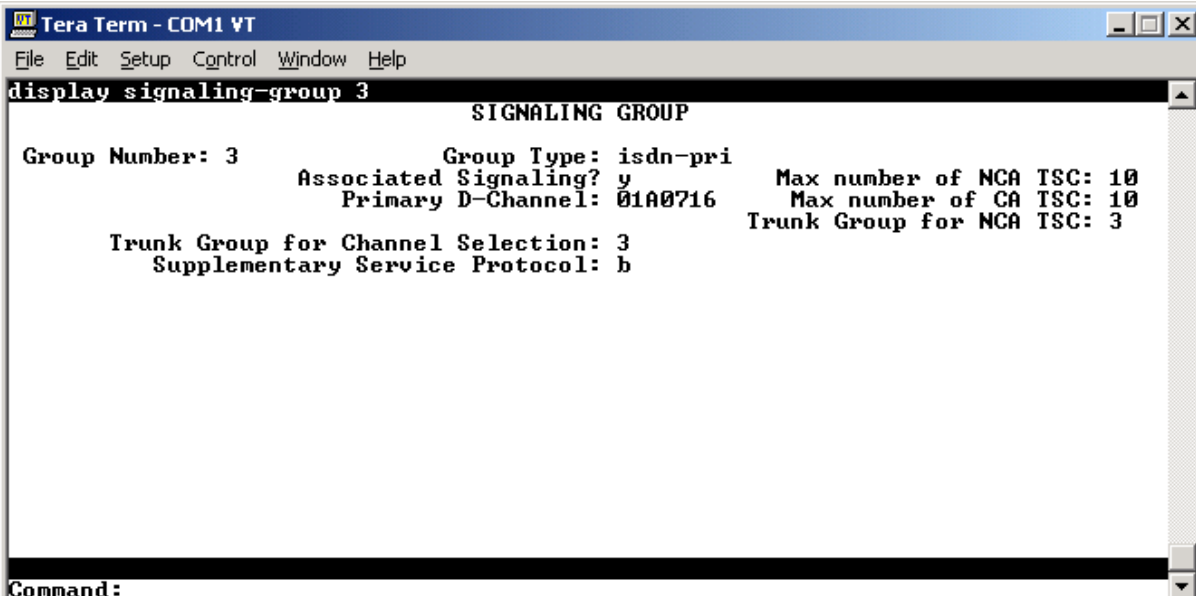
```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display ds1 01a07 Page 1 of 2
DS1 CIRCUIT PACK
      Location: 01A09                      Name: T1-QSIG
      Bit Rate: 1.544                      Line Coding: b8zs
Line Compensation: 1                      Framing Mode: esf
      Signaling Mode: isdn-pri             Interface: peer-master
      Connect: pbx                         Peer Protocol: Q-SIG
      TN-C7 Long Timers? n                 Side: a
Interworking Message: PROGRESS            CRC? n
Interface Companding: mulaw
      Idle Code: 1111111                  DCP/Analog Bearer Capability: 3.1kHz
                                           T303 Timer(sec): 4
Slip Detection? n                         Near-end CSU Type: other

```

Step 5: Add a Signaling Group

Use the **Add Signaling Group** command to configure a Signaling Group that will be assigned to the DS1 channels. To attach the signaling group to the appropriate DS1, the 'Primary D-Channel' field value must be the DS1 Location (i.e., slot no.) and the D-Channel time slot (e.g., 01A0716). The Signaling Group should be configured as follows:



```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display signaling-group 3
SIGNALING GROUP
Group Number: 3      Group Type: isdn-pri
Associated Signaling? y      Max number of NCA TSC: 10
Primary D-Channel: 01A0716  Max number of CA TSC: 10
Trunk Group for Channel Selection: 3      Trunk Group for NCA TSC: 3
Supplementary Service Protocol: b
Command:

```

Note: The Max number of NCA TSC (Non Carrier Associated Temporary Signaling Channels) must be 1 or higher. This allows a path (1 or more) for MWI to operate.

The MWI trunk group for NCA TSC must also be defined.

Step 6: Add Trunk Group

Use the **Add Trunk Group** command to create a trunk group and assign the newly created DS1 channels to it (except channel 16). The trunk group must be configured as follows:

```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display trunk-group 3 Page 1 of 11
TRUNK GROUP
Group Number: 3          Group Type: isdn          CDR Reports: y
  Group Name: Qsig          COR: 1          TN: 1          TAC: *03
  Direction: two-way      Outgoing Display? n      Carrier Medium: PRI/BRI
  Dial Access? y          Busy Threshold: 99      Night Service:
Queue Length: 0
Service Type: tie          Auth Code? n          TestCall ITC: rest
TestCall BCC: 4          Far End Test Line No:

```

```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display trunk-group 3 Page 2 of 11
  Group Type: isdn
TRUNK PARAMETERS
  Codeset to Send Display: 6      Codeset to Send National IEs: 6
  Max Message Size to Send: 260  Charge Advice: none
  Supplementary Service Protocol: b  Digit Handling <in/out>: enbloc/enbloc
  Trunk Hunt: cyclical          QSIG Value-Added? n
  Incoming Calling Number - Delete:  Insert:          Digital Loss Group: 13
  Bit Rate: 1200              Synchronization: async  Duplex: full
  Disconnect Supervision - In? y  Out? y
  Answer Supervision Timeout: 0

```

```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display trunk-group 3 Page 3 of 11
TRUNK FEATURES
  ACA Assignment? n
    Measured: none
    Wideband Support? n
    Internal Alert? n
    Maintenance Tests? y
    Data Restriction? n
    NCA-TSC Trunk Member: 29
    Send Name: y
    Send Calling Number: y
    Hop Dgt? n
    Send EMU Visitor CPN? n
    Used for DCS? n
    Suppress # Outpulsing? n
    Format: unk-pvt
    Outgoing Channel ID Encoding: exclusive
    UII IE Treatment: service-provider
    Replace Restricted Numbers? n
    Replace Unavailable Numbers? n
    Send Connected Number: y
    Hold/Unhold Notifications? y
    Modify Tandem Calling Number? n
    Send UUI IE? y
    Send UCID? n
    Send Codeset 6/7 LAI IE? y
    Ds1 Echo Cancellation? n
    Network <Japan> Needs Connect Before Disconnect? n
    Apply Local Ringback? n

```

Note: NCA-TSC Trunk Member should have the highest timeslot from the upper T1 span in the trunk group. For example, if there is one T1 span in the MAS then it should be 29, and if there are 2 spans then it should be 59.

```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display trunk-group 3 Page 4 of 11
      QSIG TRUNK GROUP OPTIONS
    Diversion by Reroute? y
    Path Replacement? y
    Path Replacement with Retention? n
    Path Replacement Method: always
    SBS? n
    Display Forwarding Party Name? y
    Character Set for QSIG Name: eurofont

```



```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display trunk-group 3                                     Page 5 of 11
TRUNK GROUP
Administered Members (min/max): 1/30
Total Administered Members: 30
GROUP MEMBER ASSIGNMENTS
  Port      Code Sfx Name      Night      Sig Grp
1: 01A0701  TN464 F
2: 01A0702  TN464 F
3: 01A0703  TN464 F
4: 01A0704  TN464 F
5: 01A0705  TN464 F
6: 01A0706  TN464 F
7: 01A0707  TN464 F
8: 01A0708  TN464 F
9: 01A0709  TN464 F
10: 01A0710 TN464 F
11: 01A0711 TN464 F
12: 01A0712 TN464 F
13: 01A0713 TN464 F
14: 01A0714 TN464 F
15: 01A0715 TN464 F

```

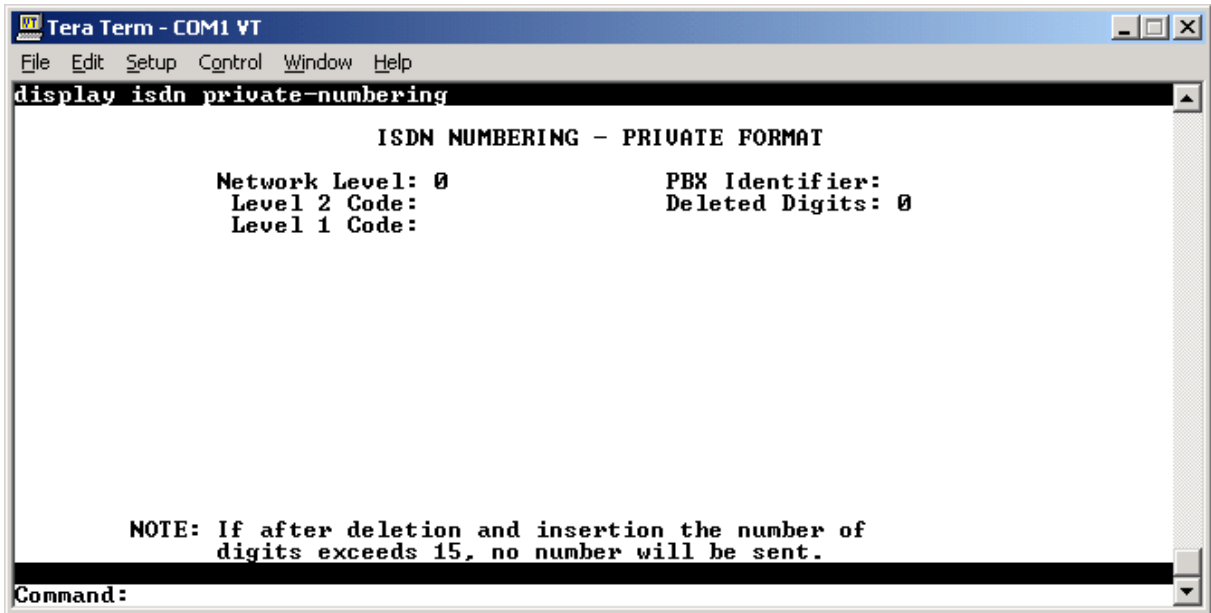
```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display trunk-group 3                                     Page 6 of 11
TRUNK GROUP
Administered Members (min/max): 1/30
Total Administered Members: 30
GROUP MEMBER ASSIGNMENTS
  Port      Code Sfx Name      Night      Sig Grp
16: 01A0717 TN464 F
17: 01A0718 TN464 F
18: 01A0719 TN464 F
19: 01A0720 TN464 F
20: 01A0721 TN464 F
21: 01A0722 TN464 F
22: 01A0723 TN464 F
23: 01A0724 TN464 F
24: 01A0725 TN464 F
25: 01A0726 TN464 F
26: 01A0727 TN464 F
27: 01A0728 TN464 F
28: 01A0729 TN464 F
29: 01A0730 TN464 F
30: 01A0731 TN464 F

```

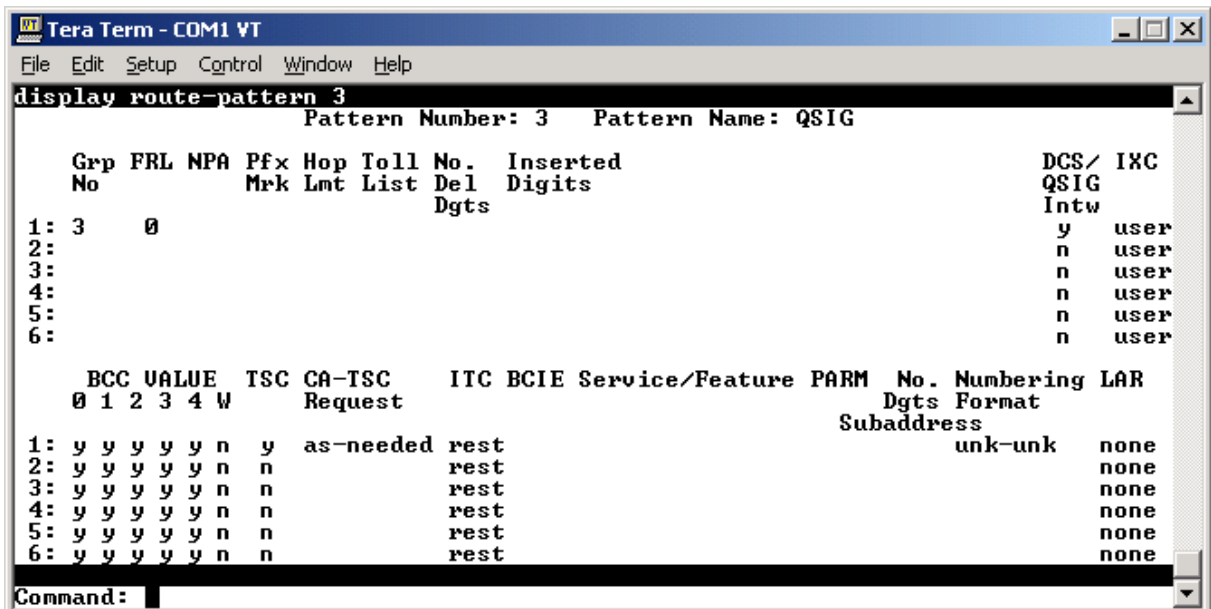
Step 7: Change ISDN Private Numbering

Change the ISDN numbering – private network form, to configure and ensure the PBX for proper Network Level to be used.



Step 8: Create a Route Pattern

Use the **Change Route-Pattern** command to create a Route Pattern for the trunk group that was previously created for the DS1 channels. The Route Pattern must be configured as follows:



Step 9: Create AAR Digit Analysis

Within the AAR Digit Analysis Table, create a dialed string to map calls to the newly created Route Pattern. For example, dial string 4000 with length 4 to match route pattern 3.

The screenshot shows a terminal window titled 'Tera Term - COM1 VT' with a menu bar (File, Edit, Setup, Control, Window, Help). The prompt is 'display aar analysis 4'. The output is 'AAR DIGIT ANALYSIS TABLE' with 'Page 1 of 2' and 'Percent Full: 1'. The table contains the following data:

Dialed String	Total		Route Pattern	Call Type	Node Num	ANI Reqd
	Min	Max				
4000	4	4	3	aar		n
5	7	7	254	aar		n
6	7	7	254	aar		n
7	7	7	254	aar		n
8	7	7	254	aar		n
9	7	7	254	aar		n

Step 10: Create a Hunt Group

Configure a Hunt Group to be used as the Call Coverage Point for the Call Coverage Path assigned to the UM subscribers. This hunt group's extension number is used as the UM Pilot Number.

```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display hunt-group 8                                     Page 1 of 10
HUNT GROUP
Group Number: 8                                         ACD? n
Group Name: QSIG                                       Queue? n
Group Extension: 4000                                   Vector? n
Group Type: ucd-mia                                    Coverage Path:
IN: 1                                                  Night Service Destination:
COR: 1                                                MM Early Answer? n
Security Code:                                         Local Agent Preference? n
ISDN Caller Display:

```

```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display hunt-group 8                                     Page 2 of 10
HUNT GROUP
LWC Reception: none
Message Center: qsig-mwi
Send Reroute Request: n
Voice Mail Number: 4000
Routing Digits (e.g. AAR/ARS Access Code): *17       Provide Ringback? n

```

Notes:

- Enter the dialed string created previously in the AAR Digit Analysis Table in the "Voice Mail Number" field.
- In the "Routing Digit (e.g. AAR/ARS Access Code)" field of this form, enter your PBX's AAR Access Code as defined on the Feature Access Codes form (Step 3 of this section).

Step 11: Add Station with Coverage Path

Add a station using the **Add Station** command, and then configure the station using the below settings (in this example, it's a digital station, but it can also be an analog station). Define the Coverage Path for this station, for example, as 8.

```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display station 1001                                     Page 1 of 4
STATION
Extension: 1001                                         Lock Messages? n      BCC: 0
Type: 8410D                                           Security Code:         TN: 1
Port: 01A0801                                         Coverage Path 1: 8    COR: 1
Name: DIGITAL-01                                       Coverage Path 2:      COS: 1
                                                         Hunt-to Station:
STATION OPTIONS
    Loss Group: 2                                       Personalized Ringing Pattern: 1
    Data Module? n                                     Message Lamp Ext: 1001
    Speakerphone: 2-way                               Mute Button Enabled? y
    Display Language: english
                                                         Media Complex Ext:
                                                         IP SoftPhone? n
  
```

Step 12: Define the Coverage Path Properties

Add a coverage path using the **Add Coverage Path** command. Configure the coverage path properties using the below settings.

Note: Ensure that you define the coverage destination point.

```

Tera Term - COM1 VT
File Edit Setup Control Window Help
display coverage path 8                                COVERAGE PATH
                                                         Coverage Path Number: 8
                                                         Next Path Number:
                                                         Hunt after Coverage? n
                                                         Linkage
COVERAGE CRITERIA
    Station/Group Status   Inside Call   Outside Call
    Active?                n             n
    Busy?                  y             y
    Don't Answer?         y             y
    All?                   n             n
    DND/SAC/Goto Cover?   y             y
    Holiday Coverage?     n             n
    Number of Rings: 2
COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
    Point1: h8            Rng:         Point2:
    Point4:                Point5:      Point3:
                                                         Point6:
Command:
  
```

The coverage path example configured in the figure above, means that when the station does not answer the call.

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.

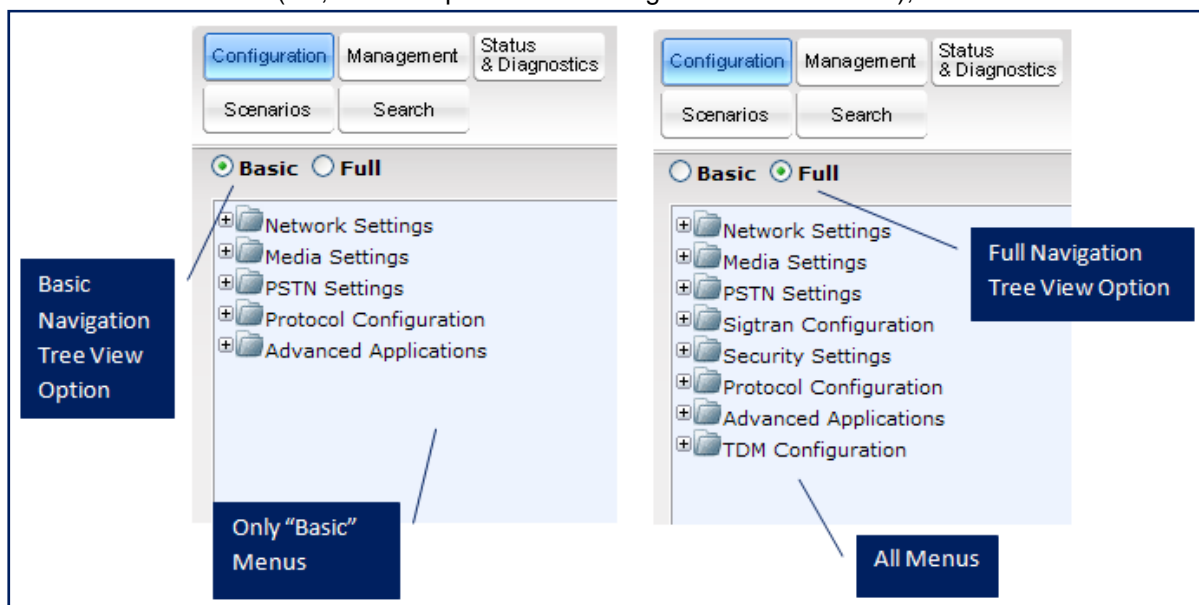
4.2 Configuring AudioCodes Gateway

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 'Trunk Settings' configuration page. At the top, there is a navigation bar with a yellow tab labeled '1' and a '2' tab. Below this, the configuration is organized into several sections:

- General Settings:**
 - Module ID: 1
 - Trunk ID: 1
 - Trunk Configuration State: **Inactive**
 - Protocol Type: T1 QSIG (indicated by a black arrow)
- Trunk Configuration:**
 - Clock Master: Recovered
 - Auto Clock Trunk Priority: 0
 - Line Code: B8ZS
 - Framing Method: T1 FRAMING ESF CRC6 (indicated by a black 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
- PSTN Alert Timeout:**
 - PSTN Alert Timeout: -1
 - QSIG Transfer Mode: Disable (indicated by a black 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: Disable
 - B-channel Negotiation: Not Configured
 - Out-Of-Service Behavior: Default
 - Play Ringback Tone to Trunk: Not Configured

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	MuLaw
TDM Bus Type	Framers
Idle PCM Pattern	85
Idle ABCD Pattern	0x0F
TDM Bus Local Reference	2
TDM Bus PSTN Auto Clock	Disable
TDM Bus Clock Source	Network

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

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.

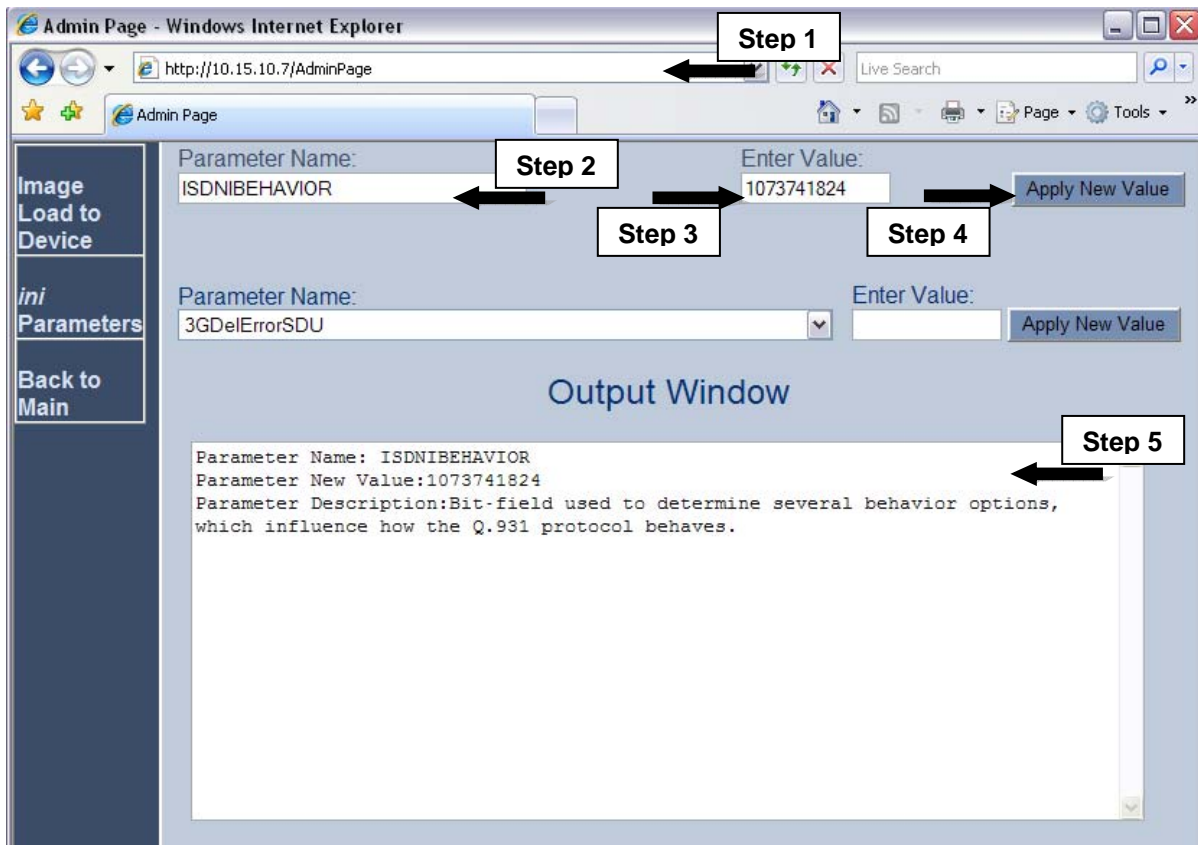
Step 14: Modify Parameters in the AdminPage

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	<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.

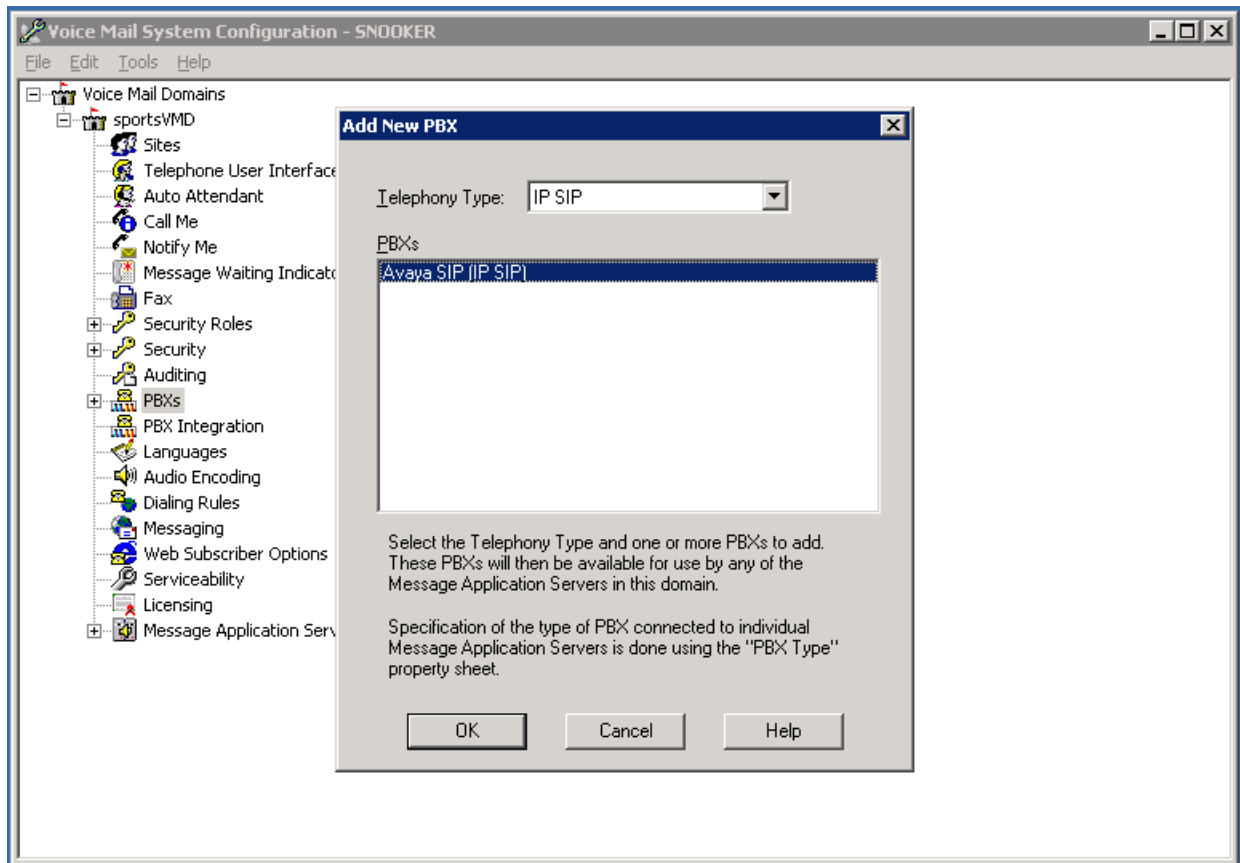
For Save Configuration: Saving configuration to flash memory may cause some temporary degradation in voice quality, therefore, it is recommended to perform this during low-traffic periods

5 Avaya Modular Messaging Configuration

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

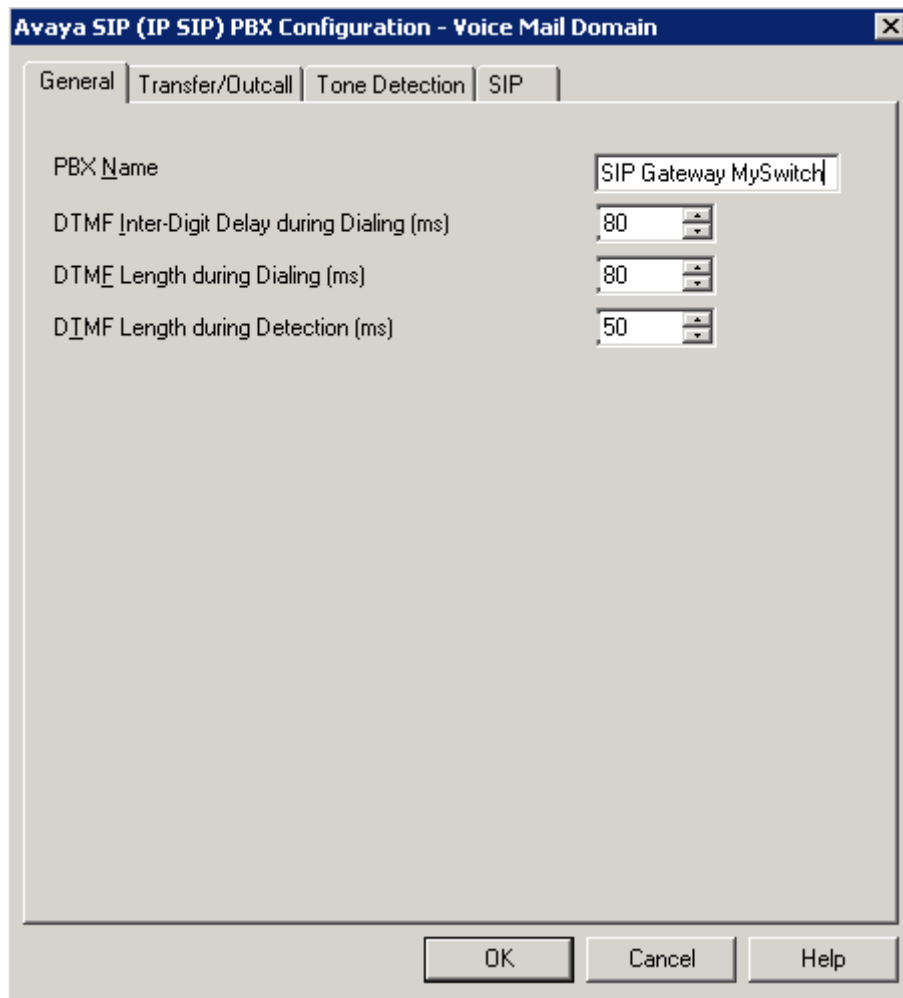
If required, use Voice Mail System Configuration (VMSC) to create a new SIP PBX for the Avaya Modular Messaging Voice Mail Domain (VMD):

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

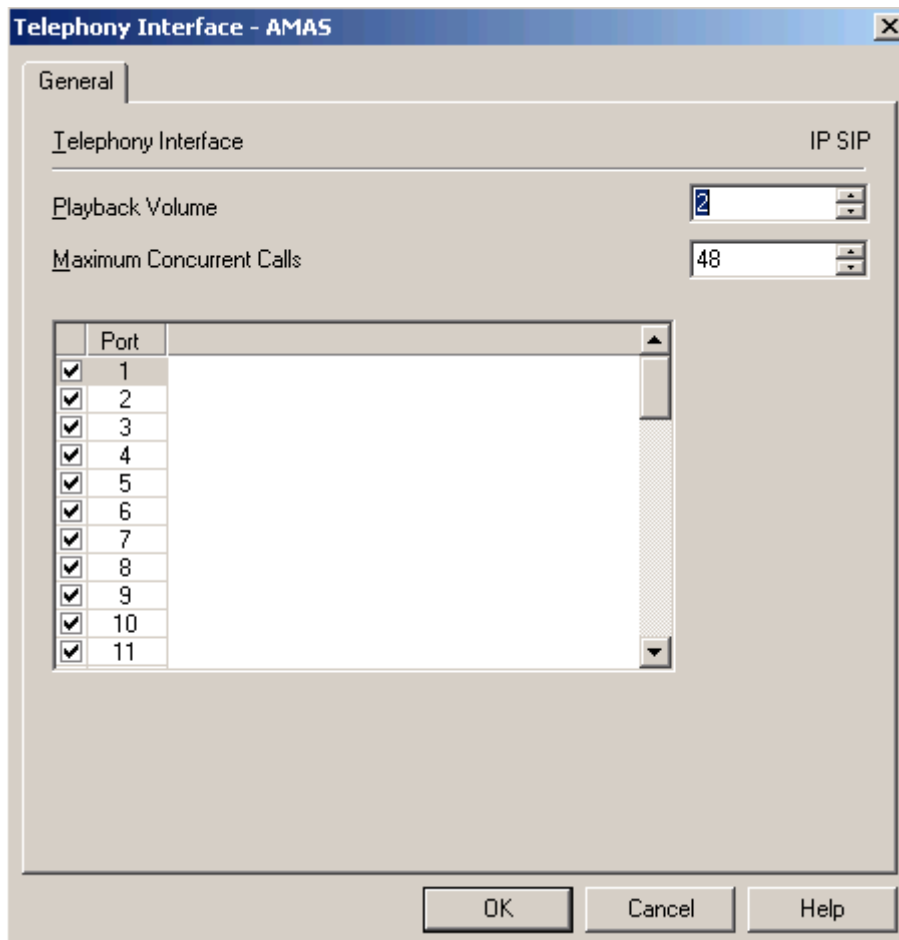
PBXs

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.

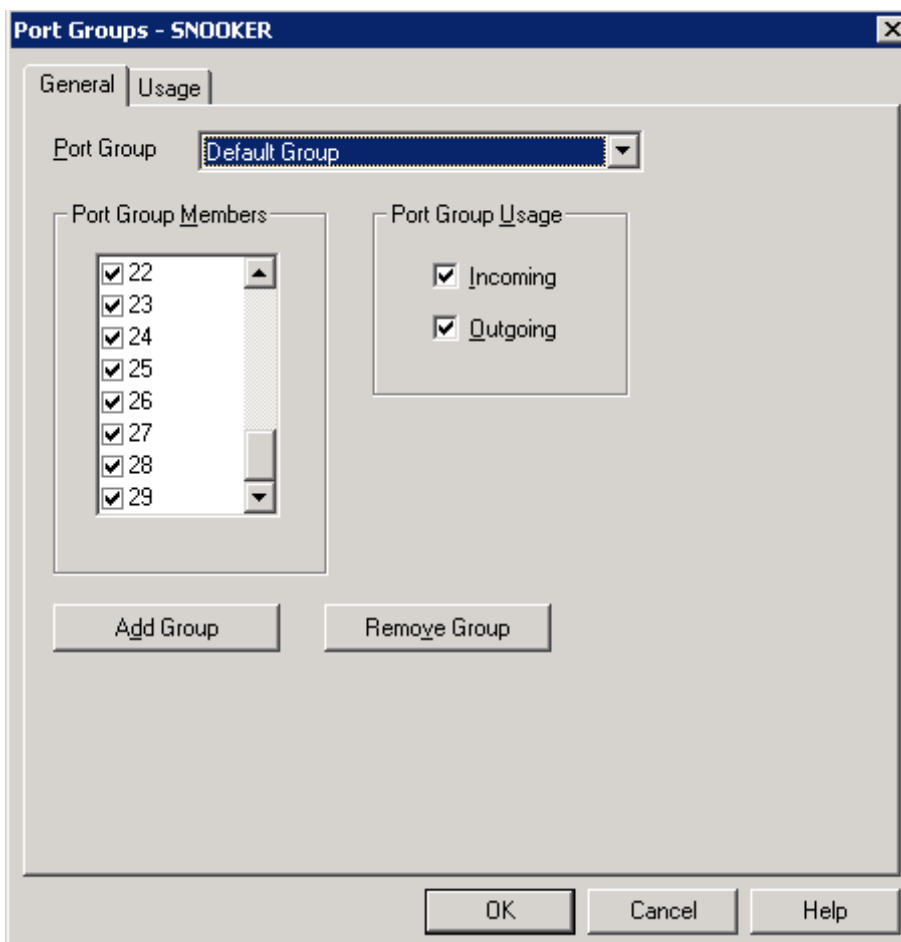


16. Set the Maximum Concurrent Calls to match the number of Channels that you want on this MAS, and then click **OK**.

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 T1 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)	
Call Scenarios Associated with Failure Point	
List of UM Features Affected by Failure Point	
Additional Comments	

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). Arrows labeled 'Step 1' and 'Step 2' point to the 'Enable Syslog' dropdown and the 'Syslog Server IP Address' field, respectively.
- SNMP Settings:** Includes 'SNMP Trap Destinations', 'SNMP Community String', 'SNMP V3 Table', and 'SNMP Trusted Managers' (each with a plus icon), and a 'Disable SNMP' dropdown set to 'No'.
- Activity Types to Report via 'Activity Log' Messages:** A list of checkboxes for various events: '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 with a checkmark icon 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.

The screenshot shows the 'Advanced Parameters' configuration page for the AudioCodes Mediant 1000 Gateway. The left sidebar contains a tree view with categories like 'Basic' and 'Full'. 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). This section is annotated with 'Step 4' and 'Step 3' pointing to the CDR Report Level and Debug Level fields respectively.
- Misc. Parameters:** Progress Indicator to IP (Not Configured), Enable X-Channel Header (Disable), Enable Busy Out (Disable), Default Release Cause (3).

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

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]

TLSVersion = 1

[BSP Params]

PCMLawSelect = 3
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 = 23

ProtocolType_1 = 0

FramingMethod_0 = D

FramingMethod_1 = 0

LineCode_0 = 0

LineCode_1 = 2

ISDNIBehavior = 1073741824

ISDNGeneralCCBehavior = 32

PSTNReserved3 = 8

[SS7 Params]

[Voice Engine Params]

IdlePCMPattern = 85

ECNLPMMode = 1

BrokenConnectionEventTimeout = 3

FaxModemBypassCoderType = 1

DTMFDetectorSensitivity = 1

AggressiveDTMFErase = 770

TTYTRANSPORTTYPE = 1

RTCPEncryptionDisableTx = 0

[WEB Params]

HTTPSCipherString = 'RC4:EXP'

[SIP Params]

ENABLEMWI = 1

ISPROXYUSED = 1

SIPDESTINATIONPORT = 5061

CHANNELSELECTMODE = 2

GWDEBUGLEVEL = 5

DEFAULTNUMBER = 'serveduser'

DISCONNECTONBROKENCONNECTION = 0

ISFAXUSED = 1

VoiceMailInterface = 3

SIPTRANSPORTTYPE = 2

MEDIASECURITYPEHAVIOUR = 0

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

[SCTP Params]

[VXML Params]

[IPsec Params]

[Audio Staging Params]

VideoEnableTestPattern = 1

VideoRateControlType = 0

;

; *** TABLE DspTemplates ***

; This table contains hidden elements and will not be exposed.

; This table exists on board and will be saved during restarts

;

;

; *** TABLE CoderName ***

;

;

;

[CoderName]

; ** NOTE: Changes were made to active configuration.

; ** The data below is different from current values.

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

Avaya's Definity G3 PBX using T1 QSIG Interface