

Mediant 1000B

VoIP Gateway and Avaya Aura Messaging With Nortel CS1000 using T1 QSIG

Version 6.8



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Notice

This document describes how to connect the AudioCodes Mediant 1000B Gateway with Avaya Aura Messaging Nortel Communication Server 1000 using T1 QSIG.

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Each abbreviation, unless widely used, is spelled out in full when first used.

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Documentation Feedback

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1 Introduction

This document describes how to connect the AudioCodes Mediant 1000B Gateway with Avaya Aura Messaging Nortel Communication Server 1000 using T1 QSIG.

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2 Components Information

2.1 PBX or IP-PBX

PBX Vendor	Nortel
Model	Communication Server 1000
Software Version	Version 7.6
Telephony Signaling	T1 QSIG
Additional Notes	None

2.2 AudioCodes Gateway

Gateway Vendor	AudioCodes
Model	Mediant 1000B
Software Version	6.80A.231.002
VoIP Protocol	SIP
Additional Notes	Note

2.3 Avaya Aura Messaging Server Version

Version	Avaya Aura Messaging Server Release 6.3.x or later
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3 Prerequisites

3.1 Gateway Prerequisites

None

3.2 PBX Prerequisites

Refer to Section 4.1.

3.3 Cabling Requirements

Refer to Section 4.1.

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4 PBX Setup Notes

4.1 PBX Configuration

Configure the PBX as specified in Section 5 of the *Avaya Aura Messaging PBX Configuration Note (cn88024 – Nortel M1 T1 QSIG.pdf)* at https://downloads.avaya.com/elmodocs2/Octel/mm_r2_0/cn88024.pdf.

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5 Gateway Configuration

The procedures below describe the configuration of AudioCodes' gateway required for integration with both the PBX and the Avaya Aura Messaging System.

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

- Uploading an *ini* configuration file (*.ini file) – see Section 5.1
- Configuring the gateway via the Web interface – see Section 5.2

5.1 Configuring the ini File

For initial setup and configuration, you can upload an *ini* file (*.ini) to the AudioCodes gateway that includes the template *ini* file settings shown in Appendix A.

➤ **To upload an ini file:**

1. Create a new text file (e.g., using Microsoft Notepad) with the file extension *.ini.
2. Copy the *ini* file settings from Appendix A and paste them into the text file.
3. Upload the file to the gateway.

Typically, for interoperability with the deployed PBX interfaces and Avaya Aura Messaging, it is sufficient that you use 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.

5.2 Configuring AudioCodes Gateway

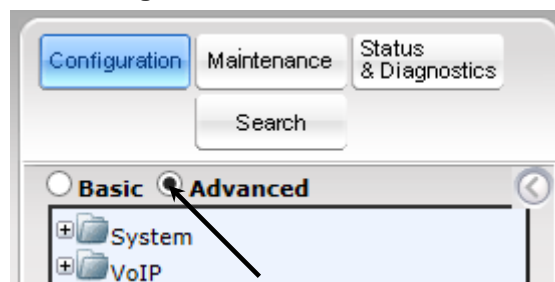
The procedures below provide step-by-step instructions for configuring the AudioCodes gateway, using the Web interface. Ensure that you configure the gateway according to the configuration settings displayed in the screenshots provided below.

The instructions describe how to setup Avaya Aura 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.
- These procedures 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 **Advanced** menu display mode (i.e., the **Advanced** option on the Navigation bar is selected), as shown below:

Figure 5-1: Advanced Mode



5.3 Step 1: Configure IP Network Interfaces

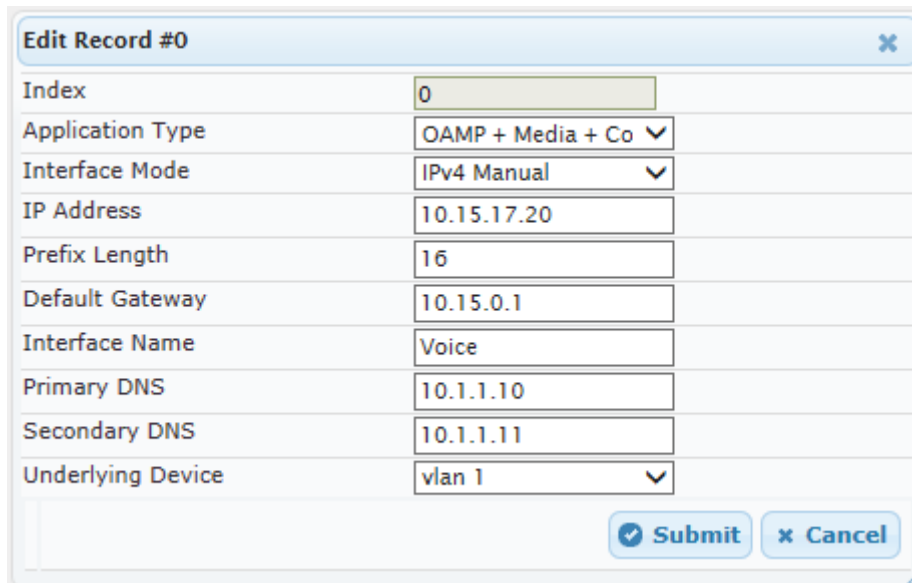
This step describes how to configure the IP network interfaces.

➤ **To configure the IP network interfaces:**

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** > **Network** > **IP Interfaces Table**).
2. Modify the existing network interface:
 - a. Select the 'Index' radio button of the **OAMP + Media + Control** table row, and then click **Edit**.
 - b. Configure the interface as follows:

Parameter	Value
IP Address	10.15.17.20 (IP address of the gateway)
Prefix Length	16 (subnet mask in bits for 255.255.0.0)
Default Gateway	10.15.0.1
Interface Name	Voice (arbitrary descriptive name)
Primary DNS Server IP Address	10.1.1.10
Secondary DNS Server IP Address	10.1.1.11
Underlying Device	vlan 1

Figure 5-2: Edit Record



The screenshot shows a dialog box titled "Edit Record #0" with a close button (X) in the top right corner. The dialog contains the following fields and values:

Index	0
Application Type	OAMP + Media + Co
Interface Mode	IPv4 Manual
IP Address	10.15.17.20
Prefix Length	16
Default Gateway	10.15.0.1
Interface Name	Voice
Primary DNS	10.1.1.10
Secondary DNS	10.1.1.11
Underlying Device	vlan 1

At the bottom right of the dialog, there are two buttons: "Submit" (with a checkmark icon) and "Cancel" (with an X icon).

3. Click **Submit**.

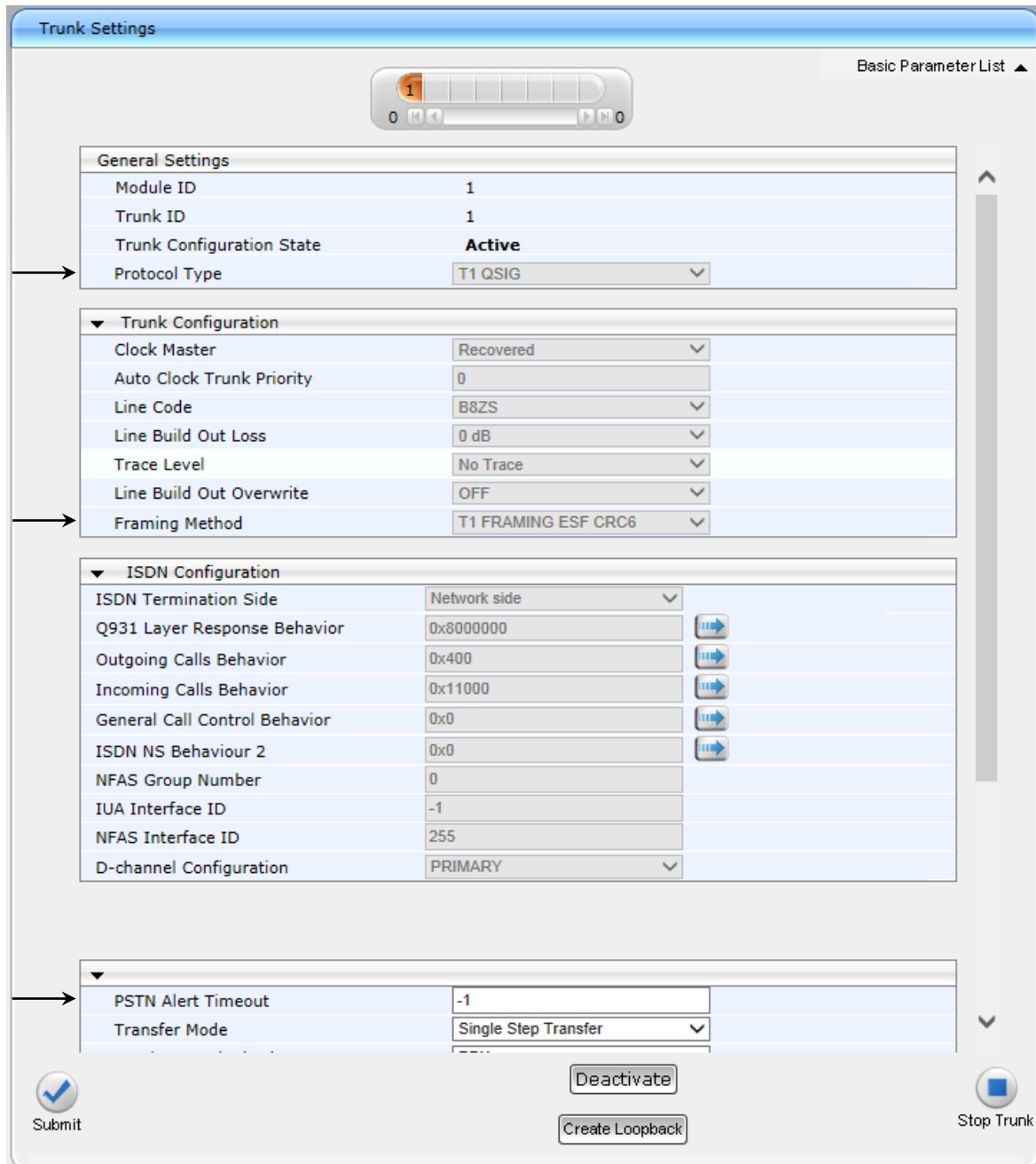
5.4 Step 2: Configure Trunk Settings

This step describes how to configure Trunk settings.

➤ **To set up Trunk settings:**

1. Open the 'Trunk Settings' page (**Configuration** tab > **VoIP** > **PSTN** > **Trunk Settings**).

Figure 5-3: Trunk Settings



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

3. Configure the relevant values to your setup for the following:
 - Protocol Type
 - Framing Method
 - Transfer Mode
4. After you configure 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.
5. If there is more than one trunk connection between the PBX and gateway, repeat this step for each of the trunks, or click the **Apply to All Trunks** button.

5.5 Step 3: Configure TDM BUS Settings

This step describes how to configure TDM Bus settings.

➤ **To configure TDM Bus settings:**

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

Figure 5-4: TDM Bus Settings

▼ TDM Bus Settings		
→ ⚡ PCM Law Select	MuLaw	▼
→ ⚡ TDM Bus Clock Source	Network	▼
⚡ TDM Bus PSTN Auto FallBack Clock	Disable	▼
⚡ TDM Bus PSTN Auto Clock Reverting	Disable	▼
⚡ Idle PCM Pattern	255	
⚡ Idle ABCD Pattern	0x0F	▼
TDM Bus Local Reference	1	
⚡ TDM Bus Type	Framers	▼

2. From the 'PCM Law Select' drop-down list, select **'MuLaw'**.
3. From the 'TDM Bus Clock Source' drop-down list, select **'Network'**.
4. Click **Submit**.

5.6 Step 4: Configure the SIP Environment

This step describes how to configure the SIP environment.

➤ **To configure the SIP environment:**

1. Open the 'SIP General Parameters' page (**Configuration** tab > **VoIP** > **SIP Definition** > **General Parameters**).

Figure 5-5: SIP General Settings for TLS

SIP General	
NAT IP Address	0.0.0.0
PRACK Mode	Supported
Channel Select Mode	Ascending
Enable Early Media	Disable
183 Message Behavior	Progress
Session-Expires Time	0
Minimum Session-Expires	90
Session Expires Method	re-INVITE
Asserted Identity Mode	Disabled
Fax Signaling Method	T.38 Relay
Detect Fax on Answer Tone	Initiate T.38 on Preamble
SIP Transport Type	TLS
SIP UDP Local Port	5060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Display Default SIP Port	Disable
Enable SIPS	Disable
Enable TCP Connection Reuse	Enable
TCP Timeout	0
SIP Destination Port	5061
Use user=phone in SIP URL	Yes
Use user=phone in From Header	No
Use Tel URI for Asserted Identity	Disable
Tel to IP No Answer Timeout	180

2. It is recommended that you configure the gateway and Avaya Aura Messaging to use **TLS**. If you prefer to use TCP, then ensure that you configure the following gateway settings (in the screen above) for **TCP**:

➤ **To configure the gateway and Avaya Aura Messaging to use TLS:**

1. From the **SIP Transport Type** drop-down list, select **TLS**.
2. In the **SIP TCP Local Port** field, enter "5061".
3. In the **SIP Destination Port** field, enter "5061".
4. Click **Submit**.

➤ **To configure the gateway and Avaya Aura Messaging to use TCP:**

1. From the **SIP Transport Type** drop-down list, select **TCP**.
2. In the **SIP TCP Local Port** field, enter "5060".
3. In the **SIP Destination Port** field, enter "5060".
4. Click **Submit**.

5.7 Step 5: Configure SRTP

This step describes how to configure SRTP.

➤ **To configure SRTP:**

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

Figure 5-6: General Media Security Settings - SIP over TLS with SRTP

General Media Security Settings	
Media Security	Enable
Media Security Behavior	Mandatory
Authentication On Transmitted RTP Packets	Active
Encryption On Transmitted RTP Packets	Active
Encryption On Transmitted RTCP Packets	Active
SRTP Tunneling Authentication for RTP	Disable
SRTP Tunneling Authentication for RTCP	Disable

➤ **To configure SIP over TLS with SRTP:**

1. From the 'Media Security' drop-down list, select **Enable**.
2. From the 'Media Security Behavior' drop-down list, select **Mandatory**.
3. From the 'Encryption On Transmitted RTCP Packets' drop-down list, select **Active**.
4. Click **Submit**.

➤ **To configure SIP over TLS without SRTP:**

1. From the 'Media Security' drop-down list, select **Disabled**.
2. From the 'Media Security Behavior' drop-down list, select **Mandatory**.
3. From the "Encryption On Transmitted RTCP Packets" drop-down list, select **Inactive**.
4. Click **Submit**.

5.8 Step 6: Configure Trunk Group

This step describes how to configure the Trunk group.

➤ **To configure the Trunk group:**

1. Open the 'Trunk Group Table' page (**Configuration Tab: VoIP > GW and IP to IP > Trunk Group > Trunk Group**).

Figure 5-7: Trunk Group Table

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

3. Match the 'Phone Number' field with the pilot number of the QSIG trunk.
4. 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.
5. Click **Submit**.

5.9 Step 7: Configure SIP Environment and Gateway Name

This step describes how to configure the SIP Environment and Gateway Name.

➤ **To configure the SIP Environment and Gateway Name:**

Open the 'Proxy & Registration' page (**Configuration** tab > **VoIP** > **SIP Definitions** > **Proxy & Registration**).

1. In the 'Gateway Name' field, enter an FQDN name to the gateway (for example, SIP-GW.com). Any gateway name that corresponds to your network environment is applicable, but it must meet requirements for FQDNs.

Figure 5-8: Proxy & Registration

Use Default Proxy	No
Proxy Name	
Redundancy Mode	Parking
Proxy IP List Refresh Time	60
Enable Fallback to Routing Table	Disable
Prefer Routing Table	No
Always Use Proxy	Disable
Redundant Routing Mode	Routing Table
SIP ReRouting Mode	Use Routing Table
Enable Registration	Disable
Gateway Name	SIP-GW.com
Gateway Registration Name	

2. Open the 'Proxy & Registration' page (**Configuration** tab > **VoIP** > **VoIP Network** > **Proxy Sets Table**).
3. From the 'Proxy Set ID' drop-down list, select 1.

Figure 5-9: Proxy Set ID

Proxy Set ID	1
--------------	---

4. In the 'Proxy Address' field, enter either the IP address or FQDN of the Avaya Aura Messaging (AAM). If your Avaya Aura Messaging system includes multiple AAM's, then enter multiple IP addresses or FQDNs for the MAS's - one AAM per table row. It is recommended that you use FQDNs.

- From the 'Transport Type' drop-down list, select the transport type for each AAM.

Figure 5-10: Proxy Address and Transport Type

	Proxy Address	Transport Type
1	10.15.10.11	TLS ▼
2		▼
3		▼



Note: When not configured, the value of the 'SIPtransportType' parameter is used.

- If your Avaya Aura Messaging System includes multiple AAM's, from the 'Proxy Load Balancing Method' drop-down, select **Round Robin** to load balance the calls across all AAM's in your Avaya Aura Messaging System.

Figure 5-11: Proxy Sets Table

Enable Proxy Keep Alive	Disable ▼
Proxy Keep Alive Time	60
KeepAlive Failure responses	
DNS Resolve Method	Not Configured ▼
Proxy Load Balancing Method	Round Robin ▼
Is Proxy Hot Swap	No ▼
Proxy Redundancy Mode	Not Configured ▼
SRD Index	0
Classification Input	IP only ▼

- Open the 'IP Group Table' page (**Configuration** tab > **VoIP** > **VoIP Network** > **IP Group Table**).
- Click **Add** to add IP Group 1.
- Configure the 'Contact User' field if necessary.

Figure 5-12: Add IP Group 1

Common		GW	SBC
Index	1		
Description			
Proxy Set ID	1		
SIP Group Name			
Contact User	7060		
SRD	0		
Media Realm Name	None ▼		
IP Profile ID	0		
Local Host Name			
UUI Format	Disable ▼		
QoS Profile	None ▼		
Bandwidth Profile	None ▼		
Media Enhancement Profile	None ▼		
Always Use Source Address	No ▼		

10. Click **Submit**.

5.10 Step 8: Configure Routing

This step describes how to configure routing.

➤ **To configure routing:**

1. Open the 'Outbound IP Routing Table' page (**Configuration tab > VoIP > GW and IP to IP > Routing > Tel to IP Routing**).
2. Configure routing from PBX (Tel) to IP. Route all messages from the PSTN to IP Group 1.

Figure 5-13: Outbound IP Routing Table

	Dest Host Prefix	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Prefix	->	Dest. IP Address	Port	Transport Type	Dest. IP Group ID
1		*	*	*				Not Configured	1
2								Not Configured	-1
3								Not Configured	-1
4								Not Configured	-1
5								Not Configured	-1
6								Not Configured	-1
7								Not Configured	-1
8								Not Configured	-1
9								Not Configured	-1
10								Not Configured	-1

3. Open the 'Inbound IP Routing Table' page (**Configuration tab > VoIP > GW and IP to IP > Routing > IP to Trunk Group Routing**).
4. Configure routing from IP to PBX. Route all messages from the IP to Trunk Group 1.

Figure 5-14: Inbound IP Routing Table

	Route Name	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	Source SRD ID	->	Trunk Group ID	Source IP Group ID
1		*	*		-1		1	-1
2					-1			
3					-1			
4					-1			
5					-1			
6					-1			
7					-1			
8					-1			
9					-1			
10					-1			
11					-1			
12					-1			

5. Click **Submit**.

5.11 Step 9: Configure Coders

This step describes how to configure coders.

➤ **To configure coders:**

1. Open the 'Coders Table' page (**Configuration** tab > **VoIP** > **Coders and Profiles** > **Coders**).

Figure 5-15: Coders Table

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

2. From the 'Coder Name' drop-down list, select **G.711U-law**.



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

3. Click **Submit**.

5.12 Step 10: Configure Digit Collection

This step describes how to configure Digit Collection.

➤ **To configure digit collection:**

1. Open the 'DTMF & Dialing' page (**Configuration** Tab: **VoIP > GW and IP to IP > DTMF and Supplementary > DTMF & Dialing**).

Figure 5-16: 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
Hook-Flash Option	Not Supported ▼
Digit Mapping Rules	
Dial Plan Index	-1
Min Routing Overlap Digits	1
ISDN Overlap IP-to-Tel Dialing	Disable ▼
Default Destination Number	serveduser
Special Digit Representation	Special ▼

2. In the 'Max Digits In Phone Num' field, enter "30".
3. In the 'Default Destination Number' field, enter "serveduser".
4. Click **Submit**.

5.13 Step 11: Configure General Settings

This step describes how to configure the General settings.

➤ **To configure General settings:**

1. Open the 'Advanced Parameters' page (**Configuration** tab > **VoIP** > **SIP Definitions** > **Advanced Parameters**).

Figure 5-17: General Settings

▼ General	
IP Security	Disable ▼
Filter Calls to IP	Don't Filter ▼
⚡ Enable Digit Delivery to Tel	Disable ▼
⚡ Enable Digit Delivery to IP	Disable ▼
PSTN Alert Timeout	180
QoS Statistics in Release Msg	Disable ▼
▼ Disconnect and Answer Supervision	
Disconnect on Broken Connection	No ▼
Amd Mode	Don't disconnect ▼
Broken Connection Timeout [100 msec]	100
⚡ 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 ▼
Media CDR Report Level	None ▼
▼ Misc. Parameters	
Progress Indicator to IP	Not Configured ▼
X-Channel Header	Disable ▼
Early 183	Disable ▼
Enable Busy Out	Disable ▼
Graceful Busy Out Timeout [sec]	0
Default Release Cause	3
Max Number of Active Calls	800
Max Call Duration [min]	0

2. From the 'Disconnect on Broken Connection' drop-down list, select **No**.
3. Click **Submit**.

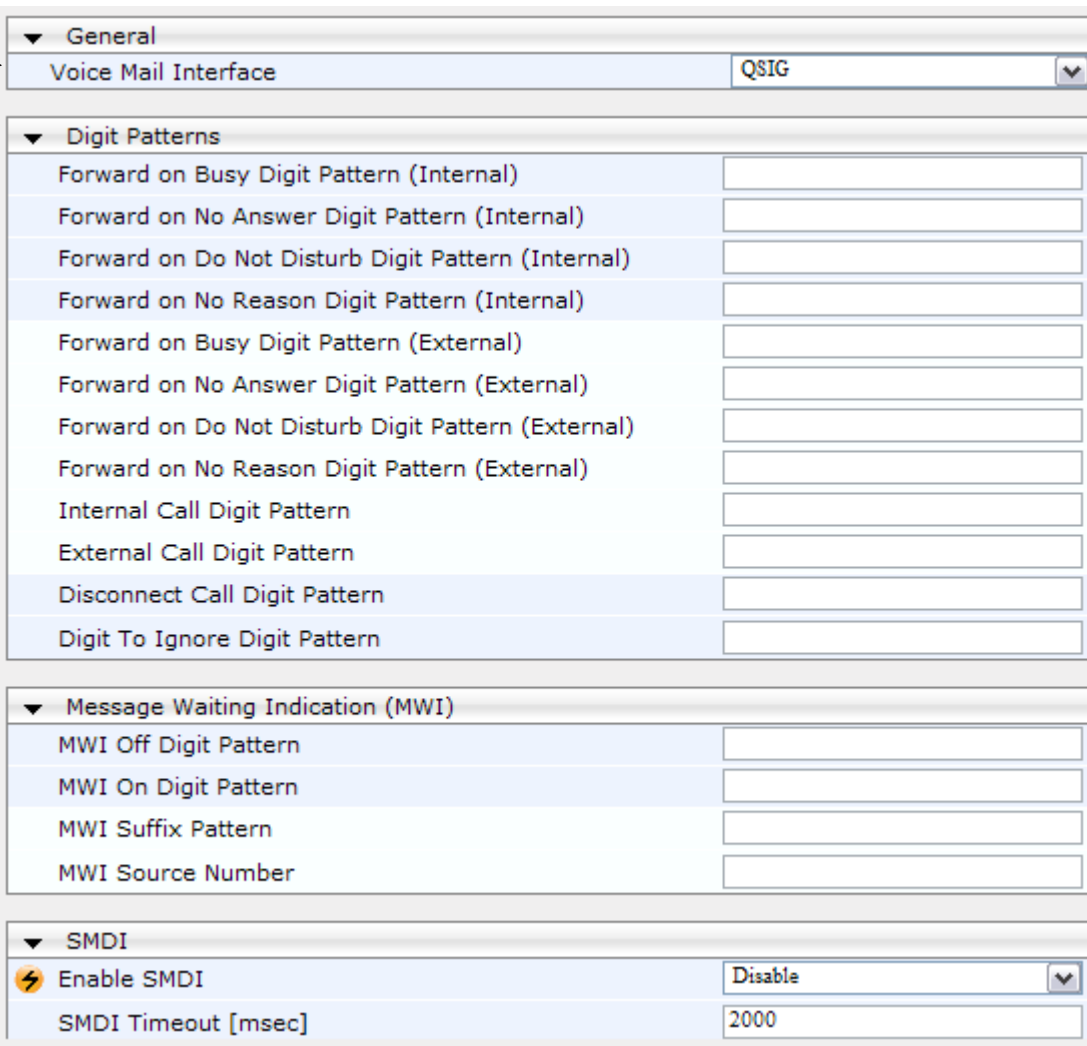
5.14 Step 12: Configure Voice Mail Settings

This step describes how to configure Voice Mail settings.

➤ **To configure Voice Mail Settings:**

1. Open the 'Voice Mail Settings' page (**Configuration** Tab: **VoIP** > **Services** > **Voice Mail Settings**).

Figure 5-18: Voce Mail Settings



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

2. From the 'Voice Mail Interface' drop-down list, select 'QSIG'.
3. Click **Submit**.

5.15 Step 13: Configure CNG Detector Mode

This step describes how to configure CNG Detector Mode.

➤ **To configure CNG Detector Mode:**

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

Figure 5-19: General Settings

▼ General Settings	
Fax Transport Mode	T.38 Relay
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 CNG Mode	Doesn't send T.38 re-INVITE
CNG Detector Mode	Disable



2. From the 'CNG Detector Mode' drop-down list, select '**Disable**'.
3. Click **Submit**.

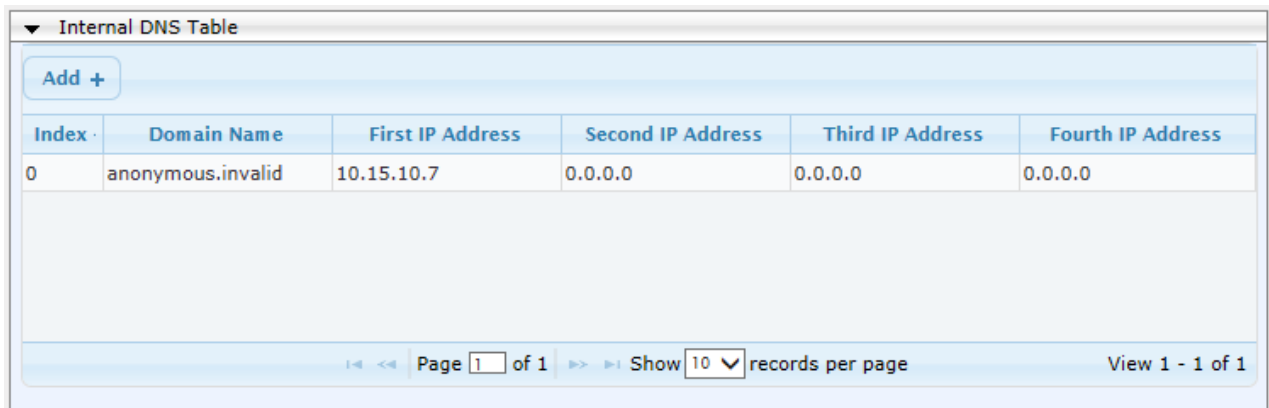
5.16 Step 14: Add Internal DNS Table

This step describes how to add an internal DNS table.

➤ **To add an internal DNS Table:**

1. Open the 'Internal DNS Table' page (**Configuration Tab: VoIP > Network > DNS > Internal DNS Table**).

Figure 5-20: Internal DNS Table



Index	Domain Name	First IP Address	Second IP Address	Third IP Address	Fourth IP Address
0	anonymous.invalid	10.15.10.7	0.0.0.0	0.0.0.0	0.0.0.0

Page 1 of 1 Show 10 records per page View 1 - 1 of 1

2. In the 'Domain Name' field, enter "anonymous.invalid".
3. In the 'First IP Address' field enter the IP address of the Mediant 1000 (e.g., 10.15.10.7).
4. Click **Submit**.

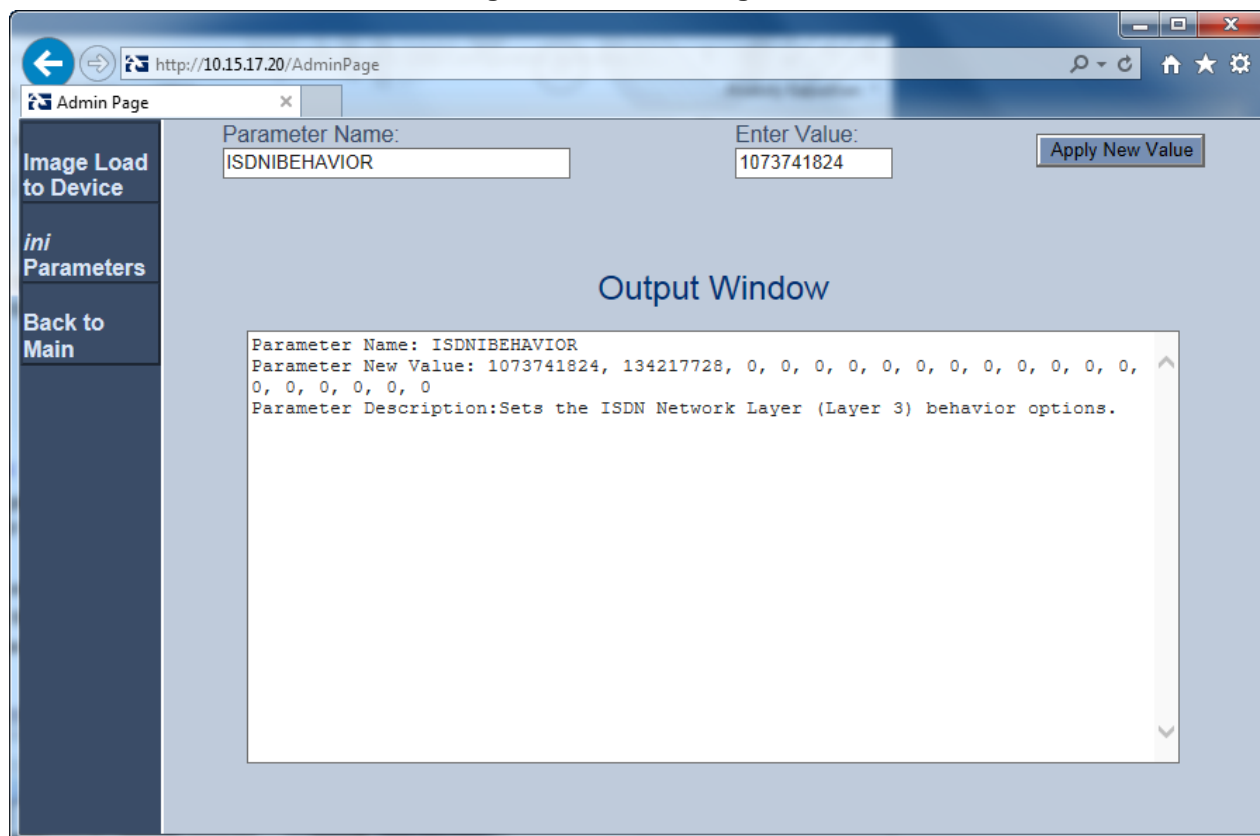
5.17 Step 15: Modify Parameters in the AdminPage

This step describes how to modify parameters on the AdminPage.

➤ **To modify parameters on the AdminPage:**

1. Open the 'AdminPage' page at the following URL (case-sensitive):
`http://<gateway's IP address>/AdminPage`
2. Click on the *ini* Parameters menu option.
3. In the 'Parameter Name' field, enter " ISDNIBehavior ".
2. In the 'Enter Value', enter "1073741824".
4. Click **Apply New Value**.
5. In the Output Window, confirm that all the fields have been correctly updated.
6. Repeat Steps 3 to 5 for the following Parameter Names and appropriate Enter Values.
 - ECNLPMODE: "1"
 - EnableMWI: "1"
 - SubscriptionMode: "1"

Figure 5-21: AdminPage



5.18 Step 16: Reset the Mediant 1000B Gateway

This step describes how to reset the Mediant 1000B 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").

Figure 5-22: Reset the Gateway

▼ Reset Configuration	
Reset Board	<input type="button" value="Reset"/>
Burn To FLASH	Yes ▼
Graceful Option	No ▼
▼ LOCK / UNLOCK	
Lock	<input type="button" value="LOCK"/>
Graceful Option	No ▼
Gateway Operational State	UNLOCKED
▼ Save Configuration	
Burn To FLASH	<input type="button" value="BURN"/>

For Reset Board :
If you choose not to save the device's configuration to flash memory, all changes made since the last time the configuration was saved 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

6 Avaya Aura Messaging Server Configuration

The following describes how to configure the Avaya Aura Messaging Server with the AudioCodes' gateway.

➤ **To configure the Avaya Aura Messaging Server with the AudioCodes' gateway:**

1. Log in to the Storage server Messaging SMI page.
2. Navigate to the Telephony Domain Administration page.

Figure 6-1: Avaya – Telephony Domain Administration Example

The screenshot shows the Avaya Aura Messaging System Management Interface (SMI) for a server named QINC27699AAR1. The main content area is titled 'Telephony Domain Administration' and contains the following information:

Far-end Domains (1 selected):

Delete	Telephony Profile Name	Gateway ID	Messaging SIP Domain	Far-end SIP Domain
<input type="checkbox"/>	default	1	itservices.sbc.com	itservices.sbc.com

Far-end Connections (1 selected):

Delete	Gateway ID	IP	Transport	Port	Monitor Interval
<input type="checkbox"/>	1	130.5.175.60	TCP	5060	0

Buttons: Save, Help

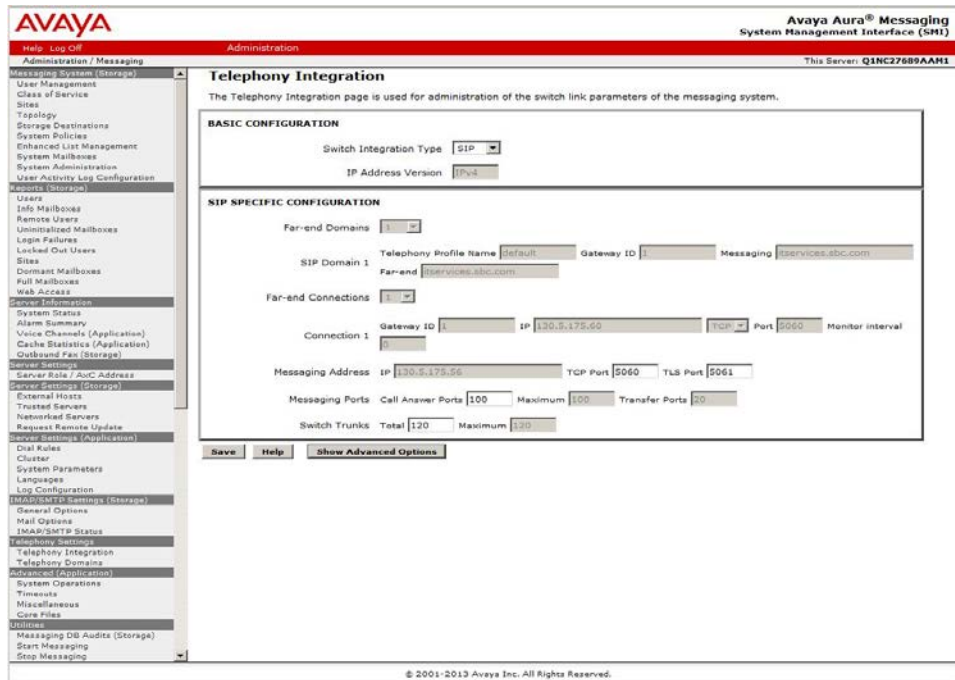
Telephony Topology Reports: None

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- a. Under the Far-End Domain group, add the following fields:
 - ◆ Telephony Profile Name
 - ◆ Gateway ID
 - ◆ Messaging SIP Domain
 - ◆ Far-end SIP Domain
- b. Under the Far-End Connections group, enter the Gateway ID that was configured in the Far-End domain and add the following:
 - ◆ Corresponding SIP Mediant 1000B IP address
 - ◆ Transport (TCP/TLS)
 - ◆ Port number
 - ◆ Monitor Interval (defaults to 0).
- c. Save the Telephony Domain Administration screen.
- d. Log in to the Network server page and select the correct Mailbox Number Length.

3. Log in to the Messaging Application server Messaging SMI page.

Figure 6-2: Avaya – Telephony Domain Administration Example



AVAYA Avaya Aura® Messaging System Management Interface (SMI) This Server: Q1NC27689AAH1

Administration / Messaging Administration

Telephony Integration
The Telephony Integration page is used for administration of the switch link parameters of the messaging system.

BASIC CONFIGURATION

Switch Integration Type:

IP Address Version:

SIP SPECIFIC CONFIGURATION

Far-end Domains:

SIP Domain 1: Telephone Profile Name: Gateway ID: Messaging:
Far-end:

Far-end Connections:

Connection 1: Gateway ID: IP: TCP Port: Monitor interval:

Messaging Address: IP: TCP Port: TLS Port:

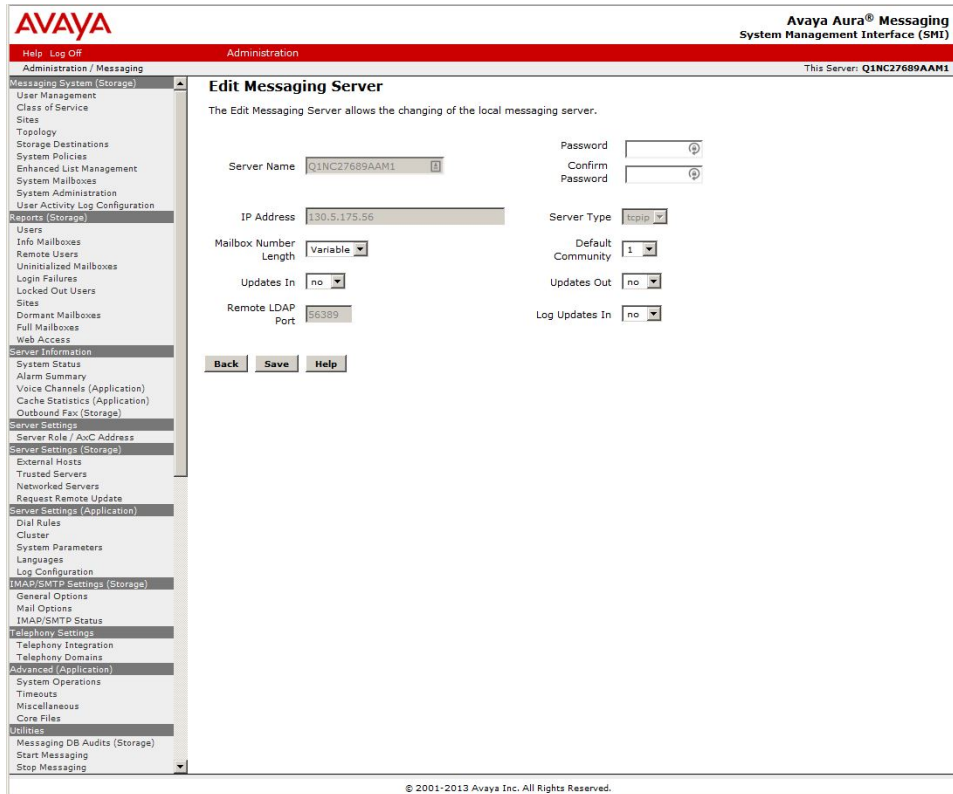
Messaging Ports: Call Answer Ports: Maximum: Transfer Ports:

Switch Trunks: Total: Maximum:

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- a. Select the Telephony Integration page.
 - b. Verify that the Switch Integration Type is set to 'SIP'.
 - c. Verify that the TCP port defaults to 5060.
 - d. Verify that the TLS port defaults to 5061.
 - e. In the 'Messaging Ports Call Answer Ports' field, enter the number of messaging ports configured on the CS1K.
 - f. In the Switch Trunks Total' field, enter the number of switch ports configured on the CS1K.
 - g. Click **Save**.
4. The system prompts you to restart messaging after these changes have been made.
 5. Continue administering the Aura Messaging servers using the *Administering the Aura Messaging Guide*.

Figure 6-3: Avaya – Telephony Integration and Networked Server Example



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7 Troubleshooting

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

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. Open the Syslog Settings page (**Configure** tab > **System** > **Syslog Settings**).
2. From the 'Enable Syslog' drop-down list, select **Enable**.
3. 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 (e.g., 10.15.17.100).

Figure 7-1: Syslog Settings

The screenshot displays the 'Syslog Settings' page in the AudioCodes Mediant 1000 web interface. The page is divided into two main sections: 'Syslog Settings' and 'Activity Types to Report via 'Activity Log' Messages'.

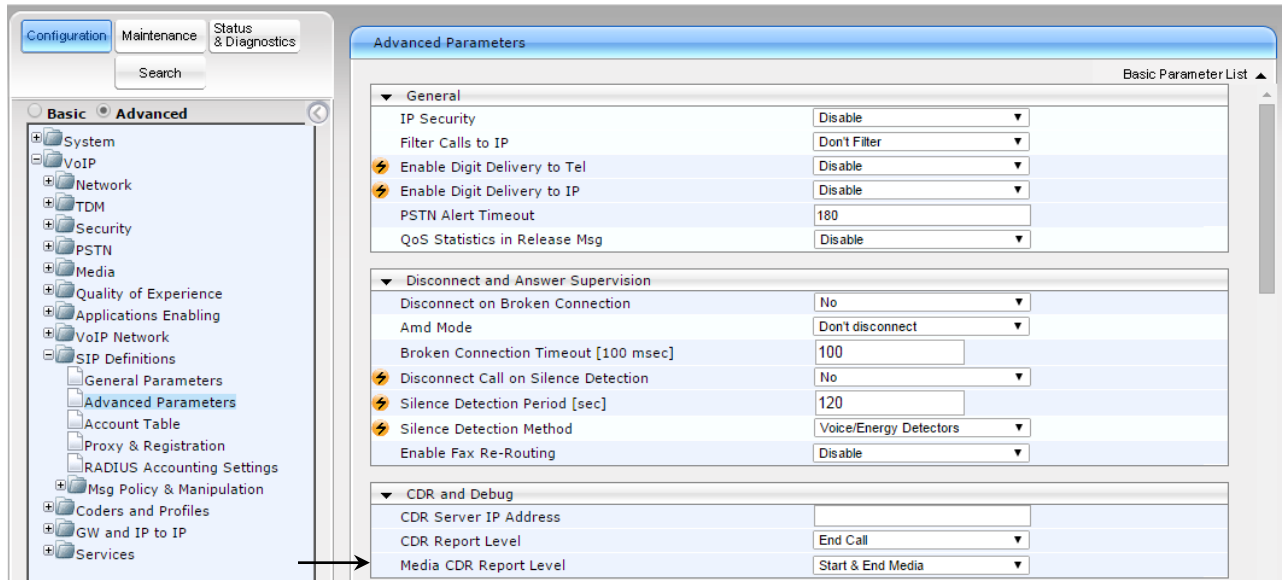
Syslog Settings	
Enable Syslog	Enable
Syslog Server IP Address	10.15.17.100
Syslog Server Port	514
CDR Session ID	Enable
Debug Level	5

Activity Types to Report via 'Activity Log' Messages	
Parameters Value Change	<input type="checkbox"/>
Auxiliary Files Loading	<input type="checkbox"/>
Device Reset	<input type="checkbox"/>
Flash Memory Burning	<input type="checkbox"/>
Device Software Update	<input type="checkbox"/>
Access to Restricted Domains	<input type="checkbox"/>
Non-Authorized Access	<input type="checkbox"/>
Sensitive Parameters Value Change	<input type="checkbox"/>
Login and Logout	<input type="checkbox"/>

The interface includes a navigation menu on the left with 'Basic' and 'Advanced' tabs, and a 'Submit' button at the bottom right.

4. From the 'Debug Level' drop-down list, select **5**, to determine the Syslog logging level.

5. Open the Advanced Parameters page (**Configure** tab > **VoIP** > **SIP Definitions** > **Advanced Parameters**).
6. From the 'Media CDR Report Level' drop-down list, select **Start & End Media** to enable additional call information.

Figure 7-2: Advanced Parameters


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

The *ini* configuration file of the Mediant 1000B, corresponding to the existing customer configuration, is shown below:



Note: To load and save an ini file, use the Configuration File page (**Maintenance** tab > **Software Update** menu > **Configuration File**).

```

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

;Board: Mediant 1000
;HW Board Type: 47  FK Board Type: 71
;Serial Number: 8680944
;Slot Number: 1
;Software Version: 6.80A.231.002
;DSP Software Version: 620AE3=> 660.11
;Board IP Address: 130.5.175.60
;Board Subnet Mask: 255.255.255.0
;Board Default Gateway: 130.5.175.3
;Ram size: 497M  Flash size: 64M
;Num of DSP Cores: 8  Num DSP Channels: 40
;Num of physical LAN ports: 3
;Profile: NONE
;;Key features:;Board Type: Mediant 1000 ;IP Media: Conf VXML
VoicePromptAnnounc(H248.9) ;E1Trunks=8 ;T1Trunks=8 ;Channel Type:
RTP DspCh=240 IPMediaDspCh=240 ;Security: IPSEC MediaEncryption
StrongEncryption EncryptControlProtocol ;Coders: G723 G729
NETCODER GSM-FR G727 ILBC G722 ;DSP Voice features: IpmDetector
;PSTN Protocols: ISDN IUA=4 CAS ;Control Protocols: MSFT MGCP
MEGACO SIP ;Default features:;Coders: G711 G726;

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

[SYSTEM Params]

```

```
SyslogServerIP = 155.168.209.142
EnableSyslog = 1
;NTPServerIP_abs is hidden but has non-default value
NTPServerUTCOffset = -18000
ENABLEPARAMETERSMONITORING = 1
ActivityListToLog = 'pvc', 'spc'
DebugRecordingDestIP = 155.168.209.142
;VpFileLastUpdateTime is hidden but has non-default value
DebugRecordingStatus = 0
NTPServerIP = '135.203.69.232'

[BSP Params]

PCMLawSelect = 3
TDMBusClockSource = 4
EnableLANWatchdog = 0
Mediant1000DualPowerSupplySupported = 2
UdpPortSpacing = 10
EnterCpuOverloadPercent = 99
ExitCpuOverloadPercent = 95

[Analog Params]

FarEndDisconnectType = 7

[ControlProtocols Params]

AdminStateLockControl = 0

[PSTN Params]

TraceLevel = 0
ProtocolType = 23
ClockMaster = 0
TerminationSide = 1
FramingMethod = D
LineCode = 0
LineBuildOut.LOSS = 0
LineBuildOut.OVERWRITE = 0
LineBuildOut.XPM0 = 0
LineBuildOut.XPM1 = 0
LineBuildOut.XPM2 = 0
DCHConfig = 0
ISDNIBehavior = 134217728
ISDNInCallsBehavior = 69632
ISDNOutCallsBehavior = 1024
ISDNGeneralCCBehavior = 0
ISDNNFASInterfaceID = 255
NFASGroupNumber = 0
ISDNDuplicateQ931BuffMode = 0
DIGITALPORTINFO = ''
AutoClockTrunkPriority = 0
```

```
ISDNNSBehaviour2 = 0

[Voice Engine Params]

ECNLPMODE = 1
BrokenConnectionEventTimeout = 100
FaxTransportMode = 0
V22ModemTransportType = 0
V23ModemTransportType = 0
V32ModemTransportType = 0
V34ModemTransportType = 0
RFC2833TxPayloadType = 127
RFC2833RxPayloadType = 127
EnabledDSIPMDetectors = 1
EnableIPMediaChannels = 1
RTPAuthenticationDisableTx = 0
RTCPEncryptionDisableTx = 0
RTPEncryptionDisableTx = 0
FarEndDisconnectSilenceMethod = 2
FarEndDisconnectSilencePeriod = 120
CallProgressTonesFilename = 'usa_tones_13.dat'

[WEB Params]

LogoWidth = '145'
HTTPSONly = 0
HTTPSCipherString = 'RC4:EXP'
DenyAuthenticationTimer = 60
DenyAccessOnFailCount = 3
DisplayLoginInformation = 0
;WebSessionTimeout is hidden but has non-default value

[SIP Params]

MAXDIGITS = 15
TIMEBETWEENDIGITS = 4
;ISUSEFREECHANNEL is hidden but has non-default value
MEDIACHANNELS = 48
ISPROXYUSED = 0
ISREGISTERNEEDED = 0
AUTHENTICATIONMODE = 1
ROUTEMODEIP2TEL = 1
ROUTEMODETEL2IP = 1
CHANNELSELECTMODE = 2
GWDEBUGLEVEL = 5
;ISPRACKREQUIRED is hidden but has non-default value
ISDNRXOVERLAP = 0
DEFAULTNUMBER = 'serveduser'
SIPGATEWAYNAME = 'Q1NC27689GWY1.itservices.sbc.com'
PROGRESSINDICATOR2IP = -1
```

```

;SHOULDREGISTER is hidden but has non-default value
DISCONNECTONBROKENCONNECTION = 0
CDRSYSLOGSERVERIP = 0.0.0.0
ENABLEMWI = 1
PSTNALERTTIMEOUT = 180
ISFAXUSED = 1
TRUNKTRANSFERMODE = 4
VoiceMailInterface = 3
SUBSCRIPTIONMODE = 1
SIPTRANSPORTTYPE = 1
PROGRESSINDICATOR2ISDN = -1
LOCALISDNRBSOURCE = 0
ISDNTRANSFERCAPABILITY = -1
PIFORDISCONNECTMSG = -1
PLAYRBTONE2TRUNK = 0
MEDIASECURITYPEBEHAVIOUR = 0
ENABLEHISTORYINFO = 1
ADDPHONECONTEXTASPREFIX = 1
TRUNKPSTNALERTTIMEOUT = -1
ENABLEVMURI = 1
EmergencyNumbers = '', '', '', ''
BCHANNELNEGOTIATIONFORTRUNK = -1
DIGITALOOSBEHAVIORFORTRUNK = -1
SBCREGISTRATIONTIME = 0
SIPREROUTINGMODE = 2
RemoveCallingNameForTrunk = -1
MSLDAPPRIMARYKEY = 'telephoneNumber'
CALLREROUTINGMODE = 0
ENABLESYMMETRICMKI = 0
QSIGCALLTRANSFERREVERSEENDDDESIGNATION = 1
ENERGYDETECTORCMD = 587202560
ANSWERDETECTORCMD = 10485760

[ PhysicalPortsTable ]

FORMAT PhysicalPortsTable_Index = PhysicalPortsTable_Port,
PhysicalPortsTable_Mode, PhysicalPortsTable_NativeVlan,
PhysicalPortsTable_SpeedDuplex,
PhysicalPortsTable_PortDescription,
PhysicalPortsTable_GroupMember, PhysicalPortsTable_GroupStatus;
PhysicalPortsTable 0 = "GE_0_1", 1, 1, 3, "User Port #0",
"GROUP_1", "Active";
PhysicalPortsTable 1 = "GE_0_2", 0, 1, 4, "User Port #1", "None",
" ";

[ \PhysicalPortsTable ]

[ EtherGroupTable ]
    
```

```
FORMAT EtherGroupTable_Index = EtherGroupTable_Group,
EtherGroupTable_Mode, EtherGroupTable_Member1,
EtherGroupTable_Member2;
EtherGroupTable 0 = "GROUP_1", 1, "GE_0_1", "";
EtherGroupTable 1 = "GROUP_2", 0, "", "";

[ \EtherGroupTable ]

[ DeviceTable ]

FORMAT DeviceTable_Index = DeviceTable_VlanID,
DeviceTable_UnderlyingInterface, DeviceTable_DeviceName;
DeviceTable 0 = 1, "GROUP_1", "vlan 1";

[ \DeviceTable ]

[ InterfaceTable ]

FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_VlanID, InterfaceTable_InterfaceName,
InterfaceTable_PrimaryDNSServerIPAddress,
InterfaceTable_SecondaryDNSServerIPAddress,
InterfaceTable_UnderlyingDevice;
InterfaceTable 0 = 6, 10, 130.5.175.60, 24, 130.5.175.3, 1,
"O+M+C", 135.200.124.232, 135.201.95.232, "vlan 1";

[ \InterfaceTable ]

[ DspTemplates ]

;
; *** TABLE DspTemplates ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts.
;

[ \DspTemplates ]

[ CpMediaRealm ]

FORMAT CpMediaRealm_Index = CpMediaRealm_MediaRealmName,
CpMediaRealm_IPv4IF, CpMediaRealm_IPv6IF,
CpMediaRealm_PortRangeStart, CpMediaRealm_MediaSessionLeg,
CpMediaRealm_PortRangeEnd, CpMediaRealm_IsDefault,
CpMediaRealm_QoeProfile, CpMediaRealm_BWProfile;
CpMediaRealm 0 = "DefaultRealm", "O+M+C", "", 6000, 330, 9290, 1,
"", "";
```

```

[ \CpMediaRealm ]

[ PREFIX ]

FORMAT PREFIX_Index = PREFIX_RouteName, PREFIX_DestinationPrefix,
PREFIX_DestAddress, PREFIX_SourcePrefix, PREFIX_ProfileId,
PREFIX_MeteringCode, PREFIX_DestPort, PREFIX_SrcIPGroupID,
PREFIX_DestHostPrefix, PREFIX_DestIPGroupID, PREFIX_SrcHostPrefix,
PREFIX_TransportType, PREFIX_SrcTrunkGroupID, PREFIX_DestSRD,
PREFIX_CostGroup, PREFIX_ForkingGroup, PREFIX_CallSetupRulesSetId;
PREFIX 0 = "", "*", "", "*", 0, 255, 0, -1, "", 1, "", -1, -1, -1,
"", -1, -1;
PREFIX 1 = "", "*", "", "*", 0, 255, 0, -1, "", 1, "", -1, -1, -1,
"", -1, -1;

[ \PREFIX ]

[ TrunkGroup ]

; ** NOTE: Changes were made to active configuration.
; **      The data below is different from current values.
FORMAT TrunkGroup_Index = TrunkGroup_TrunkGroupNum,
TrunkGroup_FirstTrunkId, TrunkGroup_FirstBChannel,
TrunkGroup_LastBChannel, TrunkGroup_FirstPhoneNumber,
TrunkGroup_ProfileId, TrunkGroup_LastTrunkId, TrunkGroup_Module;
TrunkGroup 0 = 1, 0, 1, 23, "7060", 0, 0, 1;

[ \TrunkGroup ]

[ NumberMapIp2Tel ]

FORMAT NumberMapIp2Tel_Index = NumberMapIp2Tel_ManipulationName,
NumberMapIp2Tel_DestinationPrefix, NumberMapIp2Tel_SourcePrefix,
NumberMapIp2Tel_SourceAddress, NumberMapIp2Tel_SrcHost,
NumberMapIp2Tel_DestHost, NumberMapIp2Tel_NumberType,
NumberMapIp2Tel_NumberPlan, NumberMapIp2Tel_RemoveFromLeft,
NumberMapIp2Tel_RemoveFromRight, NumberMapIp2Tel_LeaveFromRight,
NumberMapIp2Tel_Prefix2Add, NumberMapIp2Tel_Suffix2Add,
NumberMapIp2Tel_IsPresentationRestricted,
NumberMapIp2Tel_SrcIPGroupID;
NumberMapIp2Tel 0 = "LD OUT", "9", "*", "*", "*", "*", 0, 9, 0, 0,
255, "", "", 255, -1;
NumberMapIp2Tel 1 = "PBX ONLY", "*", "*", "*", "*", "*", 255, 255,
0, 0, 255, "", "", 255, -1;

[ \NumberMapIp2Tel ]

[ NumberMapTel2Ip ]
    
```

```

FORMAT NumberMapTel2Ip_Index = NumberMapTel2Ip_ManipulationName,
NumberMapTel2Ip_DestinationPrefix, NumberMapTel2Ip_SourcePrefix,
NumberMapTel2Ip_NumberType, NumberMapTel2Ip_NumberPlan,
NumberMapTel2Ip_RemoveFromLeft, NumberMapTel2Ip_RemoveFromRight,
NumberMapTel2Ip_LeaveFromRight, NumberMapTel2Ip_Prefix2Add,
NumberMapTel2Ip_Suffix2Add,
NumberMapTel2Ip_IsPresentationRestricted,
NumberMapTel2Ip_SrcTrunkGroupID, NumberMapTel2Ip_SrcIPGroupID,
NumberMapTel2Ip_DestIPGroupID;
NumberMapTel2Ip 1 = "", "*", "*", 255, 255, 0, 0, 255, "", "",
255, 1, -1, -1;

[ \NumberMapTel2Ip ]

[ SourceNumberMapIp2Tel ]

FORMAT SourceNumberMapIp2Tel_Index =
SourceNumberMapIp2Tel_ManipulationName,
SourceNumberMapIp2Tel_DestinationPrefix,
SourceNumberMapIp2Tel_SourcePrefix,
SourceNumberMapIp2Tel_SourceAddress,
SourceNumberMapIp2Tel_SrcHost, SourceNumberMapIp2Tel_DestHost,
SourceNumberMapIp2Tel_NumberType,
SourceNumberMapIp2Tel_NumberPlan,
SourceNumberMapIp2Tel_RemoveFromLeft,
SourceNumberMapIp2Tel_RemoveFromRight,
SourceNumberMapIp2Tel_LeaveFromRight,
SourceNumberMapIp2Tel_Prefix2Add,
SourceNumberMapIp2Tel_Suffix2Add,
SourceNumberMapIp2Tel_IsPresentationRestricted,
SourceNumberMapIp2Tel_SrcIPGroupID;
SourceNumberMapIp2Tel 0 = "LD OUT", "9", "*", "*", "*", "*", 255,
255, 4, 0, 255, "7045107000", "", 255, -1;
SourceNumberMapIp2Tel 1 = "PBX ONLY", "*", "*", "*", "*", "*",
255, 255, 0, 0, 255, "", "", 255, -1;

[ \SourceNumberMapIp2Tel ]

[ PstnPrefix ]

FORMAT PstnPrefix_Index = PstnPrefix_RouteName,
PstnPrefix_DestPrefix, PstnPrefix_TrunkGroupId,
PstnPrefix_SourcePrefix, PstnPrefix_SourceAddress,
PstnPrefix_ProfileId, PstnPrefix_SrcIPGroupID,
PstnPrefix_DestHostPrefix, PstnPrefix_SrcHostPrefix,
PstnPrefix_SrcSRDID, PstnPrefix_TrunkId,
PstnPrefix_CallSetupRulesSetId;
PstnPrefix 0 = "", "*", 1, "*", "", 0, -1, "*", "", "", -1, -1;

[ \PstnPrefix ]

[ Dns2Ip ]

```

```

FORMAT Dns2Ip_Index = Dns2Ip_DomainName, Dns2Ip_FirstIpAddress,
Dns2Ip_SecondIpAddress, Dns2Ip_ThirdIpAddress,
Dns2Ip_FourthIpAddress;
Dns2Ip 0 = "anonymous.invalid", 130.5.175.60, 0.0.0.0, 0.0.0.0,
0.0.0.0;
Dns2Ip 1 = "itservices.sbc.com", 130.5.175.60, 0.0.0.0, 0.0.0.0,
0.0.0.0;

[ \Dns2Ip ]

[ ProxyIp ]

; ** NOTE: Changes were made to active configuration.
; **      The data below is different from current values.
FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 1 = "130.5.175.56", 1, 1;

[ \ProxyIp ]

[ TxDtmfOption ]

FORMAT TxDtmfOption_Index = TxDtmfOption_Type;
TxDtmfOption 0 = 4;

[ \TxDtmfOption ]

[ TrunkGroupSettings ]

FORMAT TrunkGroupSettings_Index = TrunkGroupSettings_TrunkGroupId,
TrunkGroupSettings_ChannelSelectMode,
TrunkGroupSettings_RegistrationMode,
TrunkGroupSettings_GatewayName, TrunkGroupSettings_ContactUser,
TrunkGroupSettings_ServingIPGroup,
TrunkGroupSettings_MWIInterrogationType,
TrunkGroupSettings_TrunkGroupName;
TrunkGroupSettings 0 = 1, 1, 255, "", "", -1, 255, "";

[ \TrunkGroupSettings ]

[ ProxySet ]

; ** NOTE: Changes were made to active configuration.
; **      The data below is different from current values.
FORMAT ProxySet_Index = ProxySet_ProxyName,
ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap,
ProxySet_SRD, ProxySet_ClassificationInput, ProxySet_TLSContext,
    
```



```

ProxySet_ProxyRedundancyMode, ProxySet_DNSResolveMethod,
ProxySet_KeepAliveFailureResp;
ProxySet 1 = "", 0, 60, 0, 0, 0, 0, "-1", -1, -1, "";

[ \ProxySet ]

[ IPGroup ]

FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Description,
IPGroup_ProxySetId, IPGroup_SIPGroupName, IPGroup_ContactUser,
IPGroup_EnableSurvivability, IPGroup_ServingIPGroup,
IPGroup_SipReRoutingMode, IPGroup_AlwaysUseRouteTable,
IPGroup_RoutingMode, IPGroup_SRD, IPGroup_MediaRealm,
IPGroup_ClassifyByProxySet, IPGroup_ProfileId,
IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet,
IPGroup_OutboundManSet, IPGroup_RegistrationMode,
IPGroup_AuthenticationMode, IPGroup_MethodList,
IPGroup_EnableSBCCClientForking, IPGroup_SourceUriInput,
IPGroup_DestUriInput, IPGroup_ContactName, IPGroup_Username,
IPGroup_Password, IPGroup_UIFormat, IPGroup_QOEProfile,
IPGroup_BWProfile, IPGroup_MediaEnhancementProfile,
IPGroup_AlwaysUseSourceAddr, IPGroup_MsgManUserDef1,
IPGroup_MsgManUserDef2;
IPGroup 1 = 0, "", 1, "", "7060", 0, -1, 1, 0, -1, 0, "", 1, 0, -
1, -1, -1, 0, 0, "", 0, -1, -1, "", "", "$1$gQ==", 0, "", "", "",
0, "", "";

[ \IPGroup ]

[ TLSContexts ]

FORMAT TLSContexts_Index = TLSContexts_Name,
TLSContexts_TLSVersion, TLSContexts_ServerCipherString,
TLSContexts_ClientCipherString, TLSContexts_OcspEnable,
TLSContexts_OcspServerPrimary, TLSContexts_OcspServerSecondary,
TLSContexts_OcspServerPort, TLSContexts_OcspDefaultResponse;
TLSContexts 0 = "default", 0, "RC4:EXP", "ALL:!ADH", 0, 0.0.0.0,
0.0.0.0, 2560, 0;

[ \TLSContexts ]

[ CodersGroup0 ]

FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce;
CodersGroup0 0 = "g711Ulaw64k", 20, 0, -1, 0;

[ \CodersGroup0 ]

[ RoutingRuleGroups ]

```

```
FORMAT RoutingRuleGroups_Index = RoutingRuleGroups_LCREnable,  
RoutingRuleGroups_LCRAverageCallLength,  
RoutingRuleGroups_LCRDefaultCost;  
RoutingRuleGroups 0 = 0, 0, 1;
```

```
[ \RoutingRuleGroups ]
```

```
[ LoggingFilters ]
```

```
FORMAT LoggingFilters_Index = LoggingFilters_FilterType,  
LoggingFilters_Value, LoggingFilters_Syslog,  
LoggingFilters_CaptureType;  
LoggingFilters 1 = 2, "1", -1, 4;  
LoggingFilters 2 = 2, "1", -1, 3;  
LoggingFilters 3 = 12, "7060", -1, 3;  
LoggingFilters 4 = 12, "917192149026", -1, 3;
```

```
[ \LoggingFilters ]
```

```
[ ResourcePriorityNetworkDomains ]
```

```
FORMAT ResourcePriorityNetworkDomains_Index =  
ResourcePriorityNetworkDomains_Name,  
ResourcePriorityNetworkDomains_Ip2TelInterworking;  
ResourcePriorityNetworkDomains 1 = "dsn", 0;  
ResourcePriorityNetworkDomains 2 = "dod", 0;  
ResourcePriorityNetworkDomains 3 = "drsn", 0;  
ResourcePriorityNetworkDomains 5 = "uc", 1;  
ResourcePriorityNetworkDomains 7 = "cuc", 0;
```

```
[ \ResourcePriorityNetworkDomains ]
```

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