



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring AudioCodes Mediant 1000 VoIP Media Gateway to Interoperate with Avaya Voice Portal using Line Side T1 Connectivity to Avaya Aura™ Communication Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for the AudioCodes Mediant 1000 VoIP Media Gateway to interoperate with Avaya Voice Portal (using a SIP trunking interface) and Avaya Aura™ Communication Manager (using a line side T1 interface).

The AudioCodes Mediant 1000 VoIP Media Gateway serves as a gateway between TDM and IP networks. AudioCodes Mediant 1000 supports multiple hardware interfaces and control protocols. Capacity can be scaled upward by adding additional interface modules. During compliance testing, AudioCodes Mediant 1000 was configured as a SIP to line side T1 gateway connecting Avaya Voice Portal to Avaya Aura™ Communication Manager. The AudioCodes CAS tables used during compliance testing support the LoopStart FXO interface and OPS signaling. The CAS tables are interoperable with the Avaya Line Side T1 and E1 interfaces, as configured in this document, and should be compatible with a third party PBX that supports the same interfaces.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the procedure for configuring the AudioCodes Mediant 1000 VoIP Media Gateway to interoperate with Avaya Voice Portal (via SIP) and Avaya Aura™ Communication Manager (via line side T1). The AudioCodes Mediant 1000 VoIP Media Gateway serves as a gateway between TDM and IP networks. AudioCodes Mediant 1000 supports multiple hardware interfaces and control protocols. Capacity can be scaled upward by adding additional interface modules. During compliance testing, AudioCodes Mediant 1000 was configured as a SIP to line side T1 gateway connecting Avaya Voice Portal to a simulated PSTN network through Avaya Aura™ Communication Manager. This solution allows Avaya Voice Portal to receive calls from the PSTN and transfer calls back to the PSTN or PBX call center agent. Refer to **Figure 1** for details of the test configuration.

1.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. The feature testing focused on verifying access to Avaya Voice Portal and exercising interactive voice response functions through the AudioCodes Mediant 1000 VoIP Media Gateway. Areas tested included:

- Basic calls from the PSTN to Avaya Voice Portal
- Call transfers by Avaya Voice Portal to the PSTN, including blind, consultative, and bridged transfers
- Call transfers by Avaya Voice Portal to a Call Center agent, including blind, consultative, and bridged transfers
- DTMF tones / RFC 2833 support
- G.711 mu-law and G.711 a-law codec support
- Line side T1 connectivity between AudioCodes Mediant 1000 and Avaya Aura™ Communication Manager
- SIP connectivity between AudioCodes Mediant 1000 and Avaya Voice Portal

The serviceability testing focused on verifying the ability of AudioCodes Mediant 1000 to recover from adverse conditions, such as disconnecting/reconnecting the IP and line side T1 cables to simulate network failures, and stopping/starting AudioCodes Mediant 1000 to simulate power outages.

1.2. Support

For technical support on the AudioCodes Mediant 1000 VoIP Media Gateway, contact AudioCodes via the support link at www.audiocodes.com.

2. Reference Configuration

Figure 1 illustrates the configuration used during compliance testing. In the reference configuration, the AudioCodes Mediant 1000 VoIP Media Gateway connects to Voice Portal through a SIP trunking interface on the one side, and to a Communication Manager through a line side T1 interface on the other side. The Communication Manager in turn has an ISDN-PRI connection to a simulated PSTN.

Inbound calls from the PSTN to Voice Portal will be routed across the line side T1 connection to Mediant 1000 through Communication Manager. Mediant 1000 will then route the calls from its line side T1 interface to its SIP interface to be terminated on the Voice Portal MPP (Media Processing Platform) server. Outbound calls to PSTN (as a result of transferring the inbound call to another PSTN user or call center agent) follow the same path in the reverse order.

In the reference configuration below, Voice Portal consists of two MPP (Media Processing Platform) servers and a VPMS (Voice Portal Management System) server. A Nuance speech server providing ASR (Automatic Speech Recognition) and TTS (Text To Speech) functions, as well as an application server hosting the voice application, are also used in the reference configuration.

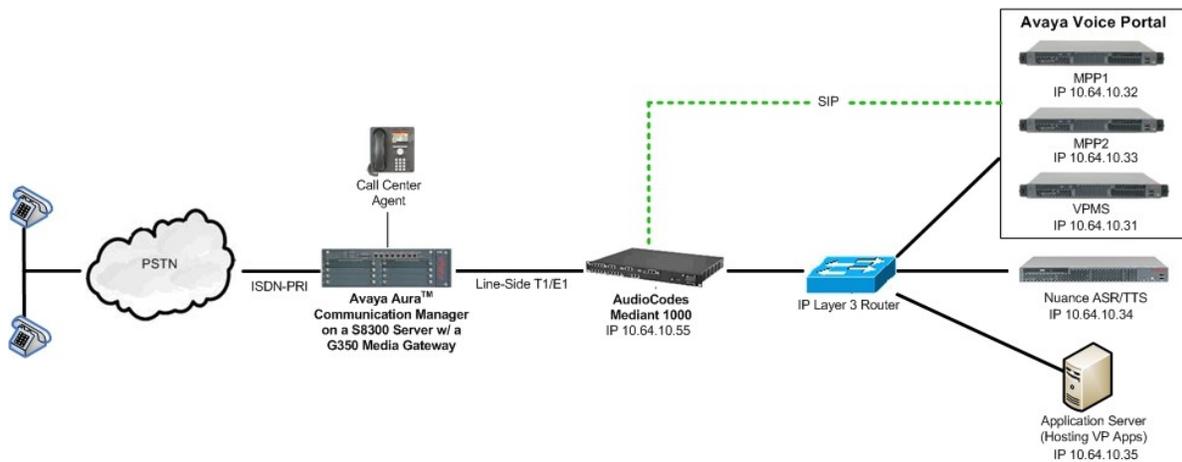


Figure 1: AudioCodes Mediant 1000 VoIP Media Gateway with Avaya Voice Portal

3. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software
Avaya Voice Portal <ul style="list-style-type: none">Voice Portal Management System (VPMS)Media Processing Platform (MPP)	5.0 SP2
Application Server – HTTP Server running in Windows	Microsoft Windows 2003 Server Service Pack 2
Nuance Speech Server <ul style="list-style-type: none">Nuance OpenSpeech RecognizerNuance RealSpeak	3.0 4.0
Avaya Aura™ Communication Manager - Avaya S8300 Server	5.2.1 Service Pack 2
Avaya G450 Media Gateway	-
Call Center Agent Avaya 9600 Series IP Telephone (H.323)	3.0
Analog and Digital Telephones	-
AudioCodes Mediant 1000 VoIP Media Gateway	6.00A.009.002

4. Configure Line Side T1 on Avaya Aura™ Communication Manager

This section provides the procedures for configuring Communication Manger for line side T1 connectivity to the AudioCodes Mediant 1000 VoIP Media Gateway. The procedures include the following areas:

- Configure DS1 (to be used for line side T1 connectivity to AudioCodes)
- Configure DS1FD stations (line side ports)
- Configure VDN (to route calls to a vector)
- Configure Vector (to route calls to a hunt group)
- Configure Hunt group (to route calls to an available Agent)
- Configure Agents (mapped to line side ports)

Note that in the reference configuration, a Call Center agent phone is shown for receiving calls transferred by Voice Portal. The configuration of this agent phone is standard and therefore is not covered in these Application Notes. Similarly, the configuration of the ISDN-PRI connection from Communication Manager to the simulated PSTN is not included since it is beyond the scope of these Application Notes. The configuration of Communication Manager was performed using the System Access Terminal (SAT). After the completion of the configuration, enter the **save translation** command to make the changes permanent.

It is assumed that Communication Manager is enabled with feature licenses for Vectoring and Expert Agent Selection. The general configuration and call flow for PSTN calls to Voice Portal are outlined below:

- Line side ports are configured as DS1FD Stations (5501-5524).
- Agent LoginIDs are created and are mapped one-to-one to each DS1FD station/line side port. The Agent LoginIDs are administered to automatically log into hunt group/skill 4 (Agent LoginIDs 54101-54124).
- Inbound calls from the PSTN are routed to VDN 53500, which then invokes Vector 2.
- Vector 2 queues the call to skill 4, thus selecting an available Agent/line-side port to be used to route the call to the AudioCodes Mediant 1000 VoIP Media Gateway. AudioCodes then routes the call on to Voice Portal.

4.1. Configure DS1

Configure a DS1 board to provide T1 connectivity to the AudioCodes Mediant 1000 VoIP Media Gateway. Use the **add ds1 n** command, where **n** is a valid board number.

- Enter a descriptive **Name** (optional)
- Set **Bit Rate** to **1.544**.
- Set **Line Coding** to **b8zs**.
- Set **Framing Mode** to **esf**.
- Set **Signaling Mode** to **robbed-bit**.
- Set **Interface Companding** to **mulaw**.
- Use the default values for the remaining fields.

```
add ds1 001v3                                     Page 1 of 2
                                                DS1 CIRCUIT PACK
      Location: 001V3                               Name: line side T1
      Bit Rate: 1.544                               Line Coding: b8zs
Line Compensation: 1                               Framing Mode: esf
      Signaling Mode: robbed-bit

Interface Companding: mulaw
      Idle Code: 11111111

Slip Detection? n                               Near-end CSU Type: other
```

4.2. Configure DS1FD Stations

Use the **add station n** command, where **n** is a valid extension, to configure each line side port as a station with the **Type** field set to **DS1FD**. Repeat this configuration for each port. During compliance testing, 24 ports were configured with an extension range of 5501 to 5524.

For each station created:

- Set **Type** to **DS1FD**.
- Set **Port** to an available port on the DS1 configured in **Section 4.1**.
- Enter a descriptive **Name** (optional).

Station 5510 is shown as an example below.

```
add station 5510                                     Page 1 of 4
                                                    STATION
Extension: 5510                                     Lock Messages? n      BCC: 0
  Type: DS1FD                                       Security Code:         TN: 1
  Port: 001V310                                     Coverage Path 1:      COR: 1
  Name: line-side port 10                          Coverage Path 2:      COS: 1
                                                    Hunt-to Station:     Tests? y
STATION OPTIONS
                                                    Time of Day Lock Table:
  Loss Group: 4
  Off Premises Station? y
  R Balance Network? n
  Survivable COR: internal
  Survivable Trunk Dest? y
```

4.3. Configure VDN

Use the **add vdn n** command, where **n** is an unused VDN number, to create the Vector Director Number (VDN) that will handle all incoming calls. Configure the following fields:

- Set **Extension** to an available extension (e.g. **53500**).
- Enter a descriptive **Name*** (optional).
- Set **Destination** to an available vector (e.g. **Vector Number 2**).

```
add vdn 53500                                       Page 1 of 3
                                                    VECTOR DIRECTORY NUMBER
  Extension: 53500
  Name*: Voice Portal
  Destination: Vector Number 2
  Allow VDN Override? n
  COR: 1
  TN*: 1
  Measured: none
```

4.4. Configure Vector

Use the **change vector n** command, where **n** is an unused vector number, to configure the vector. VDN 53500, configured above, will invoke vector 2 which will queue the call to skill 4 via the **queue-to skill** step. Configure the vector as shown below.

```
change vector 2                                     Page 1 of 6
                                           CALL VECTOR
Number: 2                                           Name: Line Side
                                           Lock? n
Basic? y      EAS? y      G3V4 Enhanced? y      ANI/II-Digits? n      ASAI Routing? y
Prompting? y  LAI? n      G3V4 Adv Route? n      CINFO? n      BSR? n      Holidays? n
Variables? n  3.0 Enhanced? y
01 wait-time  2 secs hearing ringback
02 queue-to   skill 4 pri m
03
```

4.5. Configure Hunt Group

Enter the **add hunt-group n** command, where **n** is an unused hunt group number. Agents associated with the line side ports (DS1FD stations) will automatically log into this hunt group.

- Set the **Group Extension** field to a valid extension.
- Set **ACD** to **y**.
- Set **Vector** to **y**.

```
add hunt-group 4                                     Page 1 of 3
                                           HUNT GROUP
Group Number: 4                                     ACD? y
Group Name: Voice Portal Prompts App              Queue? y
Group Extension: 5552                             Vector? y
Group Type: ucd-mia
TN: 1
COR: 1                                             MM Early Answer? n
Security Code:                                     Local Agent Preference? n
ISDN/SIP Caller Display:
Queue Limit: unlimited
Calls Warning Threshold: Port:
Time Warning Threshold: Port:
```

On **Page 2** of the Hunt Group form,

- Set **Skill** to **y**.
- Set **AAS** to **y** (the AAS option will allow the agents to automatically log into the hunt group via the corresponding Agent LoginIDs administration).

```
add hunt-group 4                                     Page 2 of 3
                                                    HUNT GROUP

Skill? y
AAS? y
Measured: none
Supervisor Extension:

Controlling Adjunct: none

Interruptible Aux Threshold: none
Redirect on No Answer (rings):
Redirect to VDN:
Forced Entry of Stroke Counts or Call Work Codes? n
```

4.6. Configure Agent Login ID

Use the add **agent-loginID n** command, where **n** is a valid extension, to add an agent. Add an Agent LoginID for each line side port.

- Set **AAS** to **y**.
- Set **Port Extension** to an available DS1FD station extension (configured in **Section 4.2**).
- Repeat this configuration for each DS1FD station.

During compliance testing, agent login IDs 54101 to 54124 were created.

```
add agent-loginID 54110                             Page 1 of 2
                                                    AGENT LOGINID

Login ID: 54110                                     AAS? y
Name: Voice Portal port 10                         AUDIX? n
TN: 1                                               LWC Reception: spe
COR: 1                                              LWC Log External Calls? n
Coverage Path:                                     AUDIX Name for Messaging:
Security Code:
Port Extension: 5510                               LoginID for ISDN/SIP Display? n

Auto Answer: station
MIA Across Skills: system
```

On **Page 2** of the Agent LoginID form,

- Set Skill Number (SN) to **4** (the hunt group number created in **Section 4.5**)
- Set Skill Level (SL) to **1**.

```
add agent-loginID 54110 Page 2 of 2
                                AGENT LOGINID
    Direct Agent Skill:
Call Handling Preference: skill-level Service Objective? n
Local Call Preference? n

    SN   RL SL           SN   RL SL
1:  4   1           16:
2:
3:
4:
5:
6:
7:
8:
9:
10:
11:
12:
13:
14:
15:
```

5. Configure Avaya Voice Portal

This section covers the administration of Voice Portal. The Voice Portal configuration required for interoperating with the AudioCodes Mediant 1000 VoIP Media Gateway includes following areas:

- Install certificates for TLS authentication
- Configure SIP connection
- Add MPP server
- Configure VoIP audio format
- Add speech server
- Add voice application
- Start MPP server

Voice Portal is configured via the Voice Portal Management System (VPMS) web interface. To access the web interface, enter `http://<ip-addr>/VoicePortal` as the URL in an Internet browser, where `<ip-addr>` is the IP address assigned to the VPMS server. Log in using the Administrator user role. The initial Voice Portal screen after login is shown below.

The screenshot shows the Avaya Voice Portal Management System (VPMS) web interface. At the top left is the Avaya logo. At the top right, it says "Welcome, admin" and "Last logged in today at 11:21:52 AM EDT". Below the logo is a red navigation bar with "Voice Portal 5.0 (VoicePortal)" and links for "Home", "Help", and "Logoff". On the left is a navigation menu with categories like "User Management", "Real-Time Monitoring", "System Maintenance", "System Management", "System Configuration", "Security", and "Reports". The main content area shows "You are here: Home" and "Voice Portal Management System Version 5.0.0.2.0104". Below this is a "Legal Notice" section with a scrollable text area containing copyright information and a disclaimer. At the bottom, it shows "Last Login: 6/10/10 11:21:52 AM EDT".

Note: All of the screens in this section are shown after Voice Portal had already been configured. The navigation sequence to each screen is displayed at the top of each screen.

5.1. Install Certificate for TLS Authentication

Voice Portal was configured to use TCP for the SIP interface to the AudioCodes Mediant 1000 VoIP Media Gateway (to facilitate debugging). A production environment is more likely to use TLS authentication over the SIP interface between Voice Portal and Mediant 1000. To install the certificate for TLS authentication, navigate to **Security** → **Certificates** and select the **Root Certificate** tab. Specify the directory path where the certificate is located, enter the appropriate password, and click **Install**. The screen below shows a certificate that has already been installed.

The screenshot shows the Avaya Voice Portal 5.0 (VoicePortal) interface. The top navigation bar includes the Avaya logo, a welcome message for 'admin', and a logoff button. The left sidebar contains a menu with categories like User Management, Real-Time Monitoring, System Maintenance, System Management, System Configuration, Security, and Reports. The main content area is titled 'Certificates' and shows the 'Root Certificate' tab selected. The certificate details are as follows:

```
Owner: CN=vpms,OU=SIP CA,O=Avaya
Issuer: CN=vpms,OU=SIP CA,O=Avaya
Serial Number: 4be97135
Valid from: Tue May 11 11:01:09 EDT 2010 until: Fri May 08 11:01:09 EDT 2020
Certificate fingerprints
MD5: 7a:b4:f6:4e:ea:e1:c1:f1:ad:1a:b2:9c:07:c1:20:4c
SHA: 33:06:2c:7c:15:9d:28:fd:a0:85:40:ed:6d:90:a6:f0:84:7c:2f:91
```

Below the details is the 'Install New Security Certificate' section, which includes a text input for 'Enter Security Certificate Path:', a 'Browse...' button, a 'Password:' input field, and three buttons: 'Install', 'Cancel', and 'Help'.

5.2. Configure SIP Connection

To configure a SIP connection to the AudioCodes Mediant 1000 VoIP Media Gateway, navigate to **System Configuration** → **VoIP Connections**, and click on the **SIP** tab. Click the **Add** button to add a new connection. On the resulting screen, configure the parameters as follows:

- Enter a descriptive text for **Name**.
- Select the **Yes** radio button for **Enable**.
- Select **TCP** as the **Proxy Transport**.
- Specify the IP address assigned to Mediant 1000 for **Proxy Server Address** and specify **5060** for **Proxy Server Port**.
- Set the **Listener Port** field to **5060** for TCP.
- Specify the IP address assigned to Mediant 1000 for the **SIP Domain**.
- Set the **Maximum Simultaneous Calls**. In this example, a maximum of **20** calls is specified.
- Accept the default values for the other fields.

The screenshot shows the Avaya Voice Portal 5.0 interface. The top navigation bar includes the Avaya logo, the user name 'Welcome, admin', and the login time 'Last logged in today at 11:21:52 AM EDT'. The main navigation menu on the left is expanded to show 'System Configuration' > 'VoIP Connections'. The main content area is titled 'Change SIP Connection' and contains the following configuration fields:

- Name:** AudioCodes Mediant 1000
- Enable:** Yes No
- Proxy Transport:** TCP
- Proxy Servers:** A table with columns for Address, Port, and Administration. One entry is shown: Address: 10.64.10.29, Port: 5060, Administration: Administration (with a Remove button).
- Additional Proxy Server:** (Empty text field)
- Listener Port:** 5060
- SIP Domain:** 10.64.10.29
- P-Asserted-Identity:** (Empty text field)
- Call Capacity:** Maximum Simultaneous Calls: 20
- Call Capacity Options:** All Calls can be either inbound or outbound; Configure number of inbound and outbound calls allowed

At the bottom of the form are buttons for Save, Apply, Cancel, and Help.

5.3. Add MPP server

Add a Media Processing Platform (MPP) server by navigating to **System Configuration** → **MPP Servers**. Click the **Add** button to add a new MPP Server. In the MPP Server

configuration page, specify a descriptive name and the **Host Address** of the MPP server. Also, specify the **Maximum Simultaneous Calls** supported on this MPP server. The screen below shows the configuration for the first MPP server used in the reference configuration. Although two MPP servers were configured in the reference configuration, only 1 was used. Repeat these steps to configure additional MPP servers as necessary.

AVAYA Welcome, admin
Last logged in today at 11:21:52 AM EDT

Voice Portal 5.0 (VoicePortal) Home Help Logoff

Expand All | Collapse All

You are here: [Home](#) > [System Configuration](#) > [MPP Servers](#) > [Change MPP Server](#)

Change MPP Server

Use this page to change the configuration of an MPP. Take care when changing the MPP Trace Logging Thresholds. Do not set Trace Levels to Finest if your Voice Portal system has heavy call traffic. The system might experience performance issues if Trace Levels are set to Finest. Set Trace Levels to Finest only when you are troubleshooting the system.

Name:

Host Address:

Network Address (VoIP):

Network Address (MRCP):

Network Address (AppSvr):

Maximum Simultaneous Calls:

Restart Automatically: Yes No

MPP Certificate

```

Owner: CN=mpp1,O=Avaya,OU=MPP
Issuer: CN=mpp1,O=Avaya,OU=MPP
Serial Number: bd9dbf79c96e701b
Valid from: Thu May 06 13:19:32 EDT 2010 until: Sun May 03 13:19:32 EDT 2020
Certificate fingerprints
MD5: 3f:ba:f6:63:67:37:0e:b8:52:ba:e3:64:cc:b2:7b:1f
SHA: 2a:3f:97:9f:a8:29:f7:37:ce:9d:2a:fc:36:de:5e:d1:27:05:3b:66
    
```

Categories and Trace Levels ▾

5.4. Configure VoIP Audio Format

The **VoIP Audio Format** for the MPP servers is configured in the **VoIP Settings** screen, accessible from **System Configuration** → **MPP Servers**. The AudioCodes Mediant 1000 VoIP Media Gateway supports both G.711 mu-law and G.711 a-law. The **MPP Native Format** field in the screen below is set to *audio/basic* for mu-law.

AVAYA Welcome, admin
Last logged in today at 11:21:52 AM EDT

Voice Portal 5.0 (VoicePortal) Home Help Logoff

You are here: Home > System Configuration > MPP Servers > VoIP Settings

VoIP Settings

Voice over Internet Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols such as H.323 and Real-time Transfer Protocol (RTP). Use this page to configure parameters that affect how voice data is transferred through the network. Note that if you make any changes to this page, you must restart all MPPs.

Port Ranges

	Low	High
UDP:	<input type="text" value="23000"/>	<input type="text" value="30999"/>
TCP:	<input type="text" value="31000"/>	<input type="text" value="31999"/>
MRCP:	<input type="text" value="32000"/>	<input type="text" value="32999"/>
H.323 Station:	<input type="text" value="35000"/>	<input type="text" value="50000"/>

RTCP Monitor Settings

Host Address:

Port:

VoIP Audio Formats

MPP Native Format:

QoS Parameters

	VLAN	Diffserv
H.323:	<input type="text" value="6"/>	<input type="text" value="46"/>
SIP:	<input type="text" value="6"/>	<input type="text" value="46"/>
RTSP:	<input type="text" value="6"/>	<input type="text" value="46"/>

Out of Service Threshold (% of VoIP Resources)

	Trigger	Reset
Warn:	<input type="text" value="10"/>	<input type="text" value="0"/>
Error:	<input type="text" value="20"/>	<input type="text" value="10"/>
Fatal:	<input type="text" value="70"/>	<input type="text" value="50"/>

5.5. Add Speech Server

Adding a speech server for providing ASR (Automatic Speech Recognition) and/or TTS (Text To Speech) services is part of the standard configuration for Voice Portal. This configuration is not directly related to achieving interoperability between the AudioCodes Mediant 1000 VoIP Media Gateway and Voice Portal. It is included here for completeness.

To configure the ASR server, navigate to **System Configuration** → **Speech Servers**, select the **ASR** tab, and then click **Add**. The screen below shows the configuration for the ASR server used during compliance testing. Set the **Engine Type** to the appropriate value. In the reference configuration, a Nuance ASR server was used so the engine type was set to *Nuance*. Set the **Network Address** field to the IP address assigned to the speech server and select the desired **Languages** to be supported. The other fields were set to their default values.

The screenshot shows the Avaya Voice Portal 5.0 configuration interface. The top navigation bar includes the Avaya logo, the user name 'admin', and the last login time '11:21:52 AM EDT'. The main navigation menu on the left lists various system management and configuration options, with 'System Configuration' expanded to show 'Speech Servers'. The main content area is titled 'Change ASR Server' and contains the following configuration fields:

- Name:** Nuance ASR
- Enable:** Yes No
- Engine Type:** Nuance (dropdown menu)
- Network Address:** 10.64.10.34
- Base Port:** 4900
- Total Number of Licensed ASR Resources:** 4
- New Connection per Session:** Yes No
- Languages:** A list of languages including Dutch(Netherlands) nl-nl, English(Australia) en-au, English(UK) en-gb, English(India) en-in, English(Singapore) en-SG, and English(USA) en-us (selected).
- MRCP:**
 - Ping Interval:** 15 second(s)
 - Response Timeout:** 4 second(s)
 - Protocol:** MRCP V1 (dropdown menu)
 - RTSP URL:** 10.64.10.34/media/speechrecognizer

At the bottom of the form are four buttons: **Save**, **Apply**, **Cancel**, and **Help**.

To configure the TTS server, navigate to **System Configuration** → **Speech Servers**, select the **TTS** tab, and then click **Add**. The screen below shows the configuration for the TTS server used during compliance testing. In this configuration, a Nuance TTS server was used so the engine type was set to **Nuance**. Set the **Network Address** field to the IP address assigned to the speech server and select the desired **Languages** to be supported. The other fields were set to their default values.

AVAYA Welcome, admin
Last logged in today at 11:21:52 AM EDT

Voice Portal 5.0 (VoicePortal) Home Help Logoff

You are here: [Home](#) > [System Configuration](#) > [Speech Servers](#) > Change TTS Server

Change TTS Server

Use this page to change the configuration of a TTS server.

Name: Nuance TTS

Enable: Yes No

Engine Type: Nuance

Network Address: 10.64.10.34

Base Port: 4900

Total Number of Licensed TTS Resources: 4

New Connection per Session: Yes No

Voices:

- English(Irish) en-EI Moira F
- English(South_African) af-ZA Tessa F
- English(Scottish) en-SC Fiona F
- English(USA) en-US Donna F
- English(USA) en-US Erica F
- English(USA) en-US Jennifer F

MRCP

Ping Interval: 15 second(s)

Response Timeout: 4 second(s)

Protocol: MRCP V1

RTSP URL: 10.64.10.34/media/speechsynthesizer

Save Apply Cancel Help

5.6. Add Voice Application

Adding a voice application for Voice Portal is part of Voice Portal's standard administration. This configuration is not directly related to achieving interoperability between the AudioCodes Mediant 1000 VoIP Media Gateway and Voice Portal. It is included here for completeness.

Navigate to **System Configuration** → **Applications**, and then click **Add**. Specify a **Name** for the application, select the **Yes** radio button for **Enable**, set the **MIME Type** field to the appropriate value (e.g., **VoiceXML**), and set the **VoiceXML URL** field to point to a VoiceXML application on the application server. Next, specify the type of **ASR** and **TTS** servers to be used by the application and the number that invokes the application. The configuration for the voice application used in the compliance test is shown in the screen below.

The screenshot displays the Avaya Voice Portal 5.0 administration interface. The top navigation bar includes the Avaya logo, the user name 'admin', and the login time 'Last logged in today at 11:33:04 AM EDT'. The main content area is titled 'Change Application' and provides configuration options for a VoiceXML or CCXML application. The application name is 'Intro', and it is currently enabled. The MIME Type is set to 'VoiceXML', and the VoiceXML URL is 'http://10.64.10.35/mpp/misc/avptestapp/intro.vxml'. Under 'Speech Servers', the ASR is set to 'Nuance' and the TTS is set to 'Nuance'. The selected language is 'English(USA) en-us' and the selected voice is 'English(USA) en-US Jennifer F'. The 'Application Launch' section shows the application is configured for 'Inbound' calls, with a 'Number' selected as the launch type. The 'Called Number' field contains '5511', '5220 - 5221', and '5510'. The interface also includes sections for 'Speech Parameters', 'Reporting Parameters', and 'Advanced Parameters', along with 'Save', 'Apply', 'Cancel', and 'Help' buttons.

5.7. Start MPP Server

Start the MPP server from **System Management** → **MPP Manager** as shown below. Select the MPP(s) for use and then click the **Start** button. The **Mode** of the started MPP should be **Online** and the **State** should be **Running**.

The screenshot displays the Avaya Voice Portal 5.0 MPP Manager interface. The top navigation bar includes the Avaya logo, user information (Welcome, admin), and navigation links (Home, Help, Logoff). The left sidebar contains a tree view of system management options, with 'System Management' expanded to show 'MPP Manager'. The main content area shows the MPP Manager page for 6/10/10 at 11:38:22 AM EDT. A table lists two MPP servers, mmp1 and mmp2, both in 'Online' mode and 'Running' state. Below the table are sections for 'State Commands' (Start, Stop, Restart, Reboot, Halt, Cancel), 'Mode Commands' (Offline, Test, Online), and 'Restart/Reboot Options' (One server at a time, All selected servers at the same time). A 'Help' button is located at the bottom left of the main content area.

	Server Name	Mode	State	Config	Auto Restart	Restart Schedule		Active Calls	
						Today	Recurring	In	Out
<input type="checkbox"/>	mmp1	Online	Running	OK	No	No	None	0	0
<input type="checkbox"/>	mmp2	Online	Running	OK	No	No	None	0	0

6. Configure AudioCodes Mediant 1000 VoIP Media Gateway

This section provides the procedures for configuring the AudioCodes Mediant 1000 VoIP Media Gateway version 6.0 to interoperate with Voice Portal and Communication Manager. It is assumed that Mediant 1000 has been properly installed with the initial configuration following Mediant 1000 standard installation procedures.

The Mediant 1000 configuration procedures include the following areas:

- Network IP settings
- PSTN trunk settings
- SIP General parameters
- SIP Advanced parameters
- SIP Proxy and Registration
- Proxy Sets table
- Coders
- DTMF and Dialing
- Trunk Group
- IP to trunk group routing
- Media voice settings

The configuration of the AudioCodes Mediant 1000 VoIP Media Gateway is performed via a Web browser. To access the device, enter the IP address of the gateway as the URL, then log in with the proper credentials. The main Mediant 1000 screen after login is shown below.

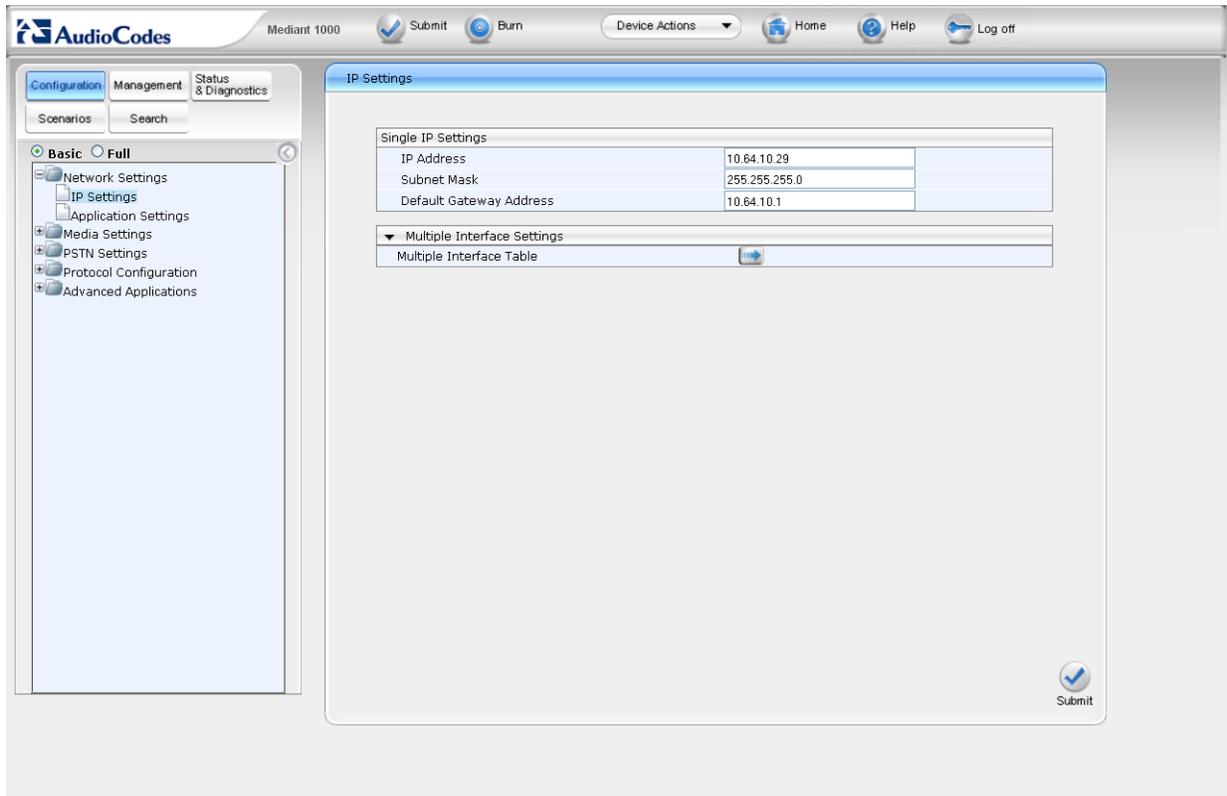
The screenshot displays the AudioCodes Mediant 1000 web interface. At the top, there is a navigation bar with the AudioCodes logo, the device name 'Mediant 1000', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. Below this is a secondary navigation bar with 'Configuration', 'Management', and 'Status & Diagnostics' tabs. A left-hand navigation pane shows a tree view with 'Basic' selected and sub-items like 'Network Settings', 'Media Settings', 'PSTN Settings', 'Protocol Configuration', and 'Advanced Applications'. The main content area, titled 'Mediant 1000 Home Page', features a dashboard with six status indicators: 1 Digital (OK), 2 BRI (OK), 3 FXO (Warning), CPU (OK), 4 FXS (OK), and 5 (OK). Below the dashboard are two tables: 'General Information' and 'Trunk (Digital Modules) / Channel (Analog Modules)'. The 'General Information' table lists IP Address (10.64.10.29), Subnet Mask (255.255.255.0), Default Gateway (10.64.10.1), Digital Port Number (1), BRI Port Number (4), Analog Port Number (8), Firmware Version (6.00A.009.002), and Protocol Type (SIP). The 'Trunk / Channel' table shows various alarm and connection statuses for digital and analog modules.

General Information	
IP Address	10.64.10.29
Subnet Mask	255.255.255.0
Default Gateway	10.64.10.1
Digital Port Number	1
BRI Port Number	4
Analog Port Number	8
Firmware Version	6.00A.009.002
Protocol Type	SIP

Trunk (Digital Modules)		Channel (Analog Modules)	
Disable		Not Connected	
Active - OK		Inactive	
RAI Alarm		Handset Offhook	
LOS / LOF Alarm		Call Connected	
AIS Alarm			
D-Channel Alarm			

6.1. Network IP settings

The network settings that were configured during installation can be viewed by navigating to **Network Settings** → **IP Settings** in the left pane. If necessary, changes can be made to the settings on this page followed by clicking the **Submit** icon button at the bottom of the screen. For compliance testing, the **IP Address**, **Subnet Mask** and **Default Gateway Address** were set to values consistent with the test configuration shown in **Figure 1**.



6.2. PSTN trunk setting

Navigate to **PSTN Settings** → **Trunk Settings** to configure the line side T1 interface to Communication Manager. These settings must be consistent with the DS1 settings on Communication Manager (**Section 4.1**). Configure the following parameters.

- Set **Protocol Type** to **T1 CAS**.
- Set **Line Code** to **B8ZS**.
- Set **Framing Method** to **T1 FRAMING ESF CRC6**.
- Enable **CAS Table per trunk** and select the appropriate CAS table from the pull down menu

Note: The names of the CAS tables used during compliance testing were changed after testing was completed. The CAS table chosen impacts the behavior of calls. See **Section 7** for the updated CAS table names and for a description of the behaviors observed using each CAS table.

- Enter **100** (seconds) for **PSTN Alert Timeout**. This timeout setting on the trunk is used for disconnecting unanswered calls on the PSTN side.

Default values may be retained for all other fields. Click the **Apply Trunk Settings** icon button at the bottom of the screen (not shown)

The screenshot displays the 'Trunk Settings' configuration page for a Mediant 1000 device. The interface includes a navigation menu on the left with options like 'Basic' and 'Full'. The main content area is divided into several sections:

- General Settings:** Module ID (1), Trunk ID (1), Trunk Configuration State (Active), and Protocol Type (T1 CAS).
- Trunk Configuration:** Clock Master (Recovered), Auto Clock Trunk Priority (0), Line Code (B8ZS), Line Build Out Loss (0 dB), Line Build Out Overwrite (OFF), and Framing Method (T1 FRAMING ESF CRC6).
- CAS Configuration:** CAS Table per Trunk (loopstarttable_bxo_2.dat (201.005)) and CAS Table per Channel (empty).
- PSTN Alert Timeout:** Set to 100 seconds.
- Out-Of-Service Behavior:** Set to Default.

At the bottom of the page, there are 'Submit' and 'Stop Trunk' buttons, along with a 'Deactivate' button.

6.3. SIP General Parameters

Navigate to **Protocol Configuration** → **Protocol Definition** → **SIP General Parameters**.

Configure the parameters as described below.

- For the **Enable Early Media** field, select **Enabled**. If enabled, the Mediant 1000 sends Session Description Protocol (SDP) information in the 18x SIP responses allowing the media stream to be set-up prior to answering the call.
- Select **TCP** for the **SIP Transport Type** field.
- Verify the correct port numbers are set for **SIP UDP Local Port (5060)**, **SIP TCP Local Port (5060)**, **SIP TLS Local Port (5061)**, **SIP Destination Port (5060)**.

Default values may be retained for all other fields.

The screenshot displays the AudioCodes Mediant 1000 configuration interface. The top navigation bar includes 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The left sidebar shows a tree view of configuration options, with 'SIP General Parameters' selected under 'Protocol Definition'. The main content area is titled 'SIP General Parameters' and contains a table of configuration fields. A 'Submit' button is located at the bottom right of the configuration area.

SIP General	
NAT IP Address	0.0.0.0
PRACK Mode	Supported
Channel Select Mode	Ascending
Enable Early Media	Enable
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	TCP
SIP UDP Local Port	5060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Enable SIPs	Disable
Enable TCP Connection Reuse	Enable
SIP Destination Port	5060
Enable Remote Party ID	Disable
Enable History-Info Header	Enable
Play Ringback Tone to IP	Don't Play
Play Ringback Tone to Tel	Play According to Early Media
3xx Behavior	Forward
Enable Reason Header	Enable

Retransmission Parameters

6.4. SIP Advanced Parameters

Click the **Full** radio button above the navigation pane on the left, and then navigate to **Protocol Configuration** → **SIP Advanced Parameters** → **Advanced Parameters**. Configure the parameters as described below.

- Specify **100** (seconds) for **PSTN Alert Timeout**. This timeout setting on the gateway is for disconnecting unanswered calls on the PSTN side.
- Set the **Max Number of Active Calls** field to an appropriate value.

Default values may be retained for all other fields.

The screenshot displays the AudioCodes Mediant 1000 configuration interface. The left navigation pane shows the 'SIP Advanced Parameters' section expanded, with 'Advanced Parameters' selected. The main content area shows the 'Advanced Parameters' configuration page. The 'Full' radio button is selected in the navigation pane. The configuration table is as follows:

Parameter	Value
Resilience Time	0
PSTN Alert Timeout	100
Disconnect and Answer Supervision	
Send Digit Pattern on Connect	
Enable Polarity Reversal	Enable
Enable Current Disconnect	Enable
Disconnect on Broken Connection	No
Broken Connection Timeout [100 msec]	100
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	
CDR Report Level	None
Debug Level	5
Misc. Parameters	
Progress Indicator to IP	Not Configured
Enable X-Channel Header	Disable
Enable Busy Out	Disable
Graceful Busy Out Timeout [sec]	0
Default Release Cause	34
Max Number of Active Calls	150

6.5. SIP Proxy and Registration

Click the **Basic** radio button above the navigation pane on the left to return to the Basic configuration menu tree, and then navigate to **Protocol Configuration → Proxies, Registration, IP Groups → Proxy & Registration**. Select **Yes** for the **Use Default Proxy** field. Default values may be retained for all other fields.

The screenshot displays the AudioCodes Mediant 1000 configuration interface. The top navigation bar includes the AudioCodes logo, the device name 'Mediant 1000', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The left sidebar shows a configuration tree with 'Basic' selected. The main content area is titled 'Proxy & Registration' and contains a table of settings. The 'Use Default Proxy' field is set to 'Yes'. Below the table are buttons for 'Register', 'Un-Register', and 'Submit'.

Advanced Parameter List	
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
Registration Time	3600
Re-registration Timing [%]	50
Registration Retry Time	30
Registration Time Threshold	0
Re-register On INVITE Failure	Disable
ReRegister On Connection Failure	Disable
Gateway Name	
Gateway Registration Name	
Subscription Mode	Per Gateway
User Name	
Password	
Cnonce	Default_Cnonce
Registration Mode	Per Gateway

Register Un-Register
Submit

6.6. Proxy Sets Table

Navigate to **Protocol Configuration** → **Proxies, Registration, IP Groups** → **Proxy Sets Table** to reach the Proxy Sets Table configuration page. Enter the IP address assigned to the Voice Portal MPP server for **Proxy Address**, and **TCP** for **Transport Type**. Default values may be retained for all other fields.

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The top navigation bar includes the AudioCodes logo, the device name 'Mediant 1000', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The left sidebar contains a tree view of configuration categories: Configuration, Management, and Status & Diagnostics. Under Configuration, there are sub-sections for Scenarios and Search. The main content area is titled 'Proxy Sets Table' and features a dropdown menu for 'Proxy Set ID' set to '0'. Below this is a table with 5 rows and 2 columns: 'Proxy Address' and 'Transport Type'. The first row is populated with '10.64.10.32' and 'TCP'. Below the table is another table with 5 rows and 2 columns: 'Enable Proxy Keep Alive', 'Proxy Keep Alive Time', 'Proxy Load Balancing Method', 'Is Proxy Hot Swap', and 'SRD Index'. The values are 'Disable', '60', 'Disable', 'No', and '0' respectively. A 'Submit' button is located in the bottom right corner of the configuration area.

	Proxy Address	Transport Type
1	10.64.10.32	TCP
2		
3		
4		
5		

Enable Proxy Keep Alive	Disable
Proxy Keep Alive Time	60
Proxy Load Balancing Method	Disable
Is Proxy Hot Swap	No
SRD Index	0

6.7. Coders

Navigate to **Protocol Configuration → Coders and Profile Definitions → Coders**. In the screen below, select the list of preferred codecs to be used by the AudioCodes Mediant 1000 VoIP Media Gateway with the most preferred codec at the top and working downward to the least preferred. This list must have an overlap with the VoIP audio format as configured for Voice Portal in **Section 5.4**. The codec is selected from the pull-down menu under the **Coder Name** field.

The codec list used during compliance testing is shown in the example below. *G.711U-law* was selected as the most preferred codec. Default values were retained for all other fields.

The screenshot displays the AudioCodes Mediant 1000 configuration web interface. The top navigation bar includes the AudioCodes logo, the device name 'Mediant 1000', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. On the left, a navigation tree is visible with 'Basic' selected and 'Full' unselected. The tree includes categories like Network Settings, Media Settings, PSTN Settings, Protocol Configuration, Applications Enabling, Trunk Group, Protocol Definition, Proxies, Registration, IP Groups, Coders And Profile Definitions, and Advanced Applications. The 'Coders' sub-menu under 'Coders And Profile Definitions' is expanded. The main content area is titled 'Coders Table' and contains a table with the following data:

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711U-law	20	64	0	Disabled
G.729	20	8	18	Enable

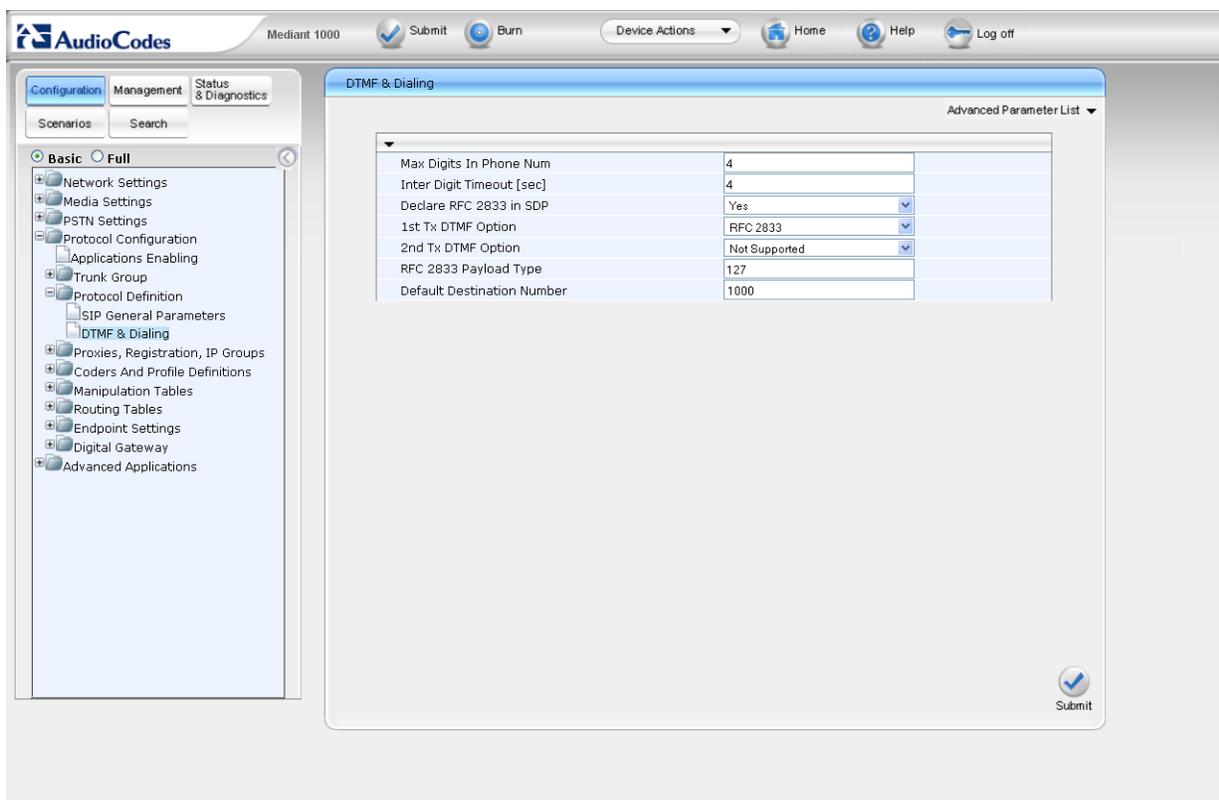
A 'Submit' button is located at the bottom right of the table area.

6.8. DTMF and Dialing

Navigate to **Protocol Configuration** → **Protocol Definition** → **DTMF & Dialing**. Configure the parameters as described below.

- In the **Max Digits in Phone Num** field, enter the maximum number of digits that can be dialed.
- For the **Declare RFC 2833 in SDP** field, select **Yes**.
- For the **1st Tx DTMF Option** field, select **RFC 2833**. This selects RFC 2833 as the preferred DTMF transmission method.
- Enter **127** as the **RFC 2833 Payload Type**.

Default values may be retained for all other fields.



The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar displays a tree view of configuration options, with 'DTMF & Dialing' selected under 'Protocol Definition'. The main content area shows the 'DTMF & Dialing' configuration page with an 'Advanced Parameter List' table. The table contains the following parameters and values:

Parameter	Value
Max Digits In Phone Num	4
Inter Digit Timeout [sec]	4
Declare RFC 2833 in SDP	Yes
1st Tx DTMF Option	RFC 2833
2nd Tx DTMF Option	Not Supported
RFC 2833 Payload Type	127
Default Destination Number	1000

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

6.9. Trunk Group

Navigate to **Protocol Configuration** → **Trunk Group** → **Trunk Group**. The Trunk Group Table maps a particular trunk channel to a trunk group. In the **From Trunk** and **To Trunk** columns, enter the starting and ending trunks to be assigned. In the **Channel(s)** column, enter the range of channels on those trunks to be assigned. The setting **1-24** means 24 channels are assigned to each trunk as defined in the **From Trunk** and **To Trunk** columns. A phone number may be entered in the **Phone Number** column or it may be left blank. If a number is entered, this number will be used as the originating calling party if no calling party information is received from the originating PSTN trunk. Each channel is assigned a unique number starting

with the value in the **Phone Number** column and incrementing for each subsequent channel. If the **Phone Number** column is left blank, the Mediant 1000 will use a default value (1000) for the originating calling party if no calling party information is received from the originating PSTN trunk. In the **Trunk Group ID** column, enter the trunk group that will contain these channels. The default value may be used for the **Tel Profile ID** column.

In the example below, the table entry assigns channels **1 – 24** of trunk **1** to Trunk Group **1**. A range of numbers arbitrary chosen to start at **5501** will be used for the originating calling party number if no calling party information is received from the originating PSTN trunk.

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar contains a tree view with categories like Network Settings, Media Settings, PSTN Settings, Protocol Configuration, Trunk Group, and Trunk Group Settings. The main area displays the 'Trunk Group Table' configuration page. At the top, there are settings for 'Add Phone Context As Prefix' (set to 'Disable') and 'Trunk Group Index' (set to '1-10'). Below this is a table with the following data:

Group Index	Module	From Trunk	To Trunk	Channels	Phone Number	Trunk Group ID	Tel Profile ID
1	Module 1 PRI	1	1	1-24	5501	1	0
2							
3							
4							
5							
6							
7							
8							
9							
10							

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

6.10. Trunk Group Settings

Navigate to **Protocol Configuration** → **Trunk Group** → **Trunk Group Settings**. Configure the parameters as described below.

- For **Trunk Group ID**, enter **1** as configured for Trunk Group (**Section 6.9**).
- Select the **Channel Select Mode** as **Cyclic Ascending**. The channels in this trunk group are treated as a pool, and each will be selected in cyclic ascending order.

The screenshot shows the AudioCodes Mediant 1000 web interface. The left sidebar contains a navigation tree with 'Basic' selected and 'Full' unselected. The main content area is titled 'Trunk Group Settings' and features an 'Advanced Parameter List' dropdown. Below this is a table with 12 rows, each representing a channel. The first row is pre-filled with '1' in the 'Trunk Group ID' column, 'Cyclic Ascending' in the 'Channel Select Mode' column, and 'Don't Register' in the 'Registration Mode' column. The remaining rows are empty. A 'Submit' button is located at the bottom right of the main content area.

	Trunk Group ID	Channel Select Mode	Registration Mode
1	1	Cyclic Ascending	Don't Register
2			
3			
4			
5			
6			
7			
8			
9			
10			
11			
12			

6.11. IP to Trunk Group Routing

Navigate to **Protocol Configuration** → **Routing Tables** → **IP to Trunk Group Routing**. The Inbound IP Routing Table defines the mapping of IP calls to the trunk group created in **Section 6.9**. The **Dest. Phone Prefix**, **Source Phone Prefix** and **Source IP Address** columns define which calls are mapped to the trunk group in the **Trunk Group ID** column. In the example below, the table entry maps calls from any destination prefix, or any source prefix or any source IP address to trunk group 1.

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar contains a navigation tree with 'IP to Trunk Group Routing' selected. The main area displays the 'IP To Trunk Group Routing Table' configuration page. At the top, there are dropdown menus for 'Routing Index' (set to 1-12) and 'IP To Tel Routing Mode' (set to 'Route calls before manipulation'). Below these is a table with 12 rows and 7 columns: 'Dest. Phone Prefix', 'Source Phone Prefix', 'Source IP Address', 'Trunk Group ID', and 'IP Profile ID'. The first row contains asterisks in the first three columns and the number '1' in the 'Trunk Group ID' column. The rest of the table is empty. A 'Submit' button is located at the bottom right of the configuration area.

	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Trunk Group ID	IP Profile ID
1	*	*	*		1	0
2						
3						
4						
5						
6						
7						
8						
9						
10						
11						
12						

Note: The Tel to IP Routing Table was not configured for compliance testing. This is because Voice Portal's MPP IP address was configured as the proxy in the Proxy Sets Table (**Section 6.6**); therefore, all calls from the Tel (line-side) side will be sent to the Voice Portal MPP on the IP side. In a configuration where no default proxy was defined, the Tel to IP Routing Table would need to be configured.

6.12. Media Voice Settings

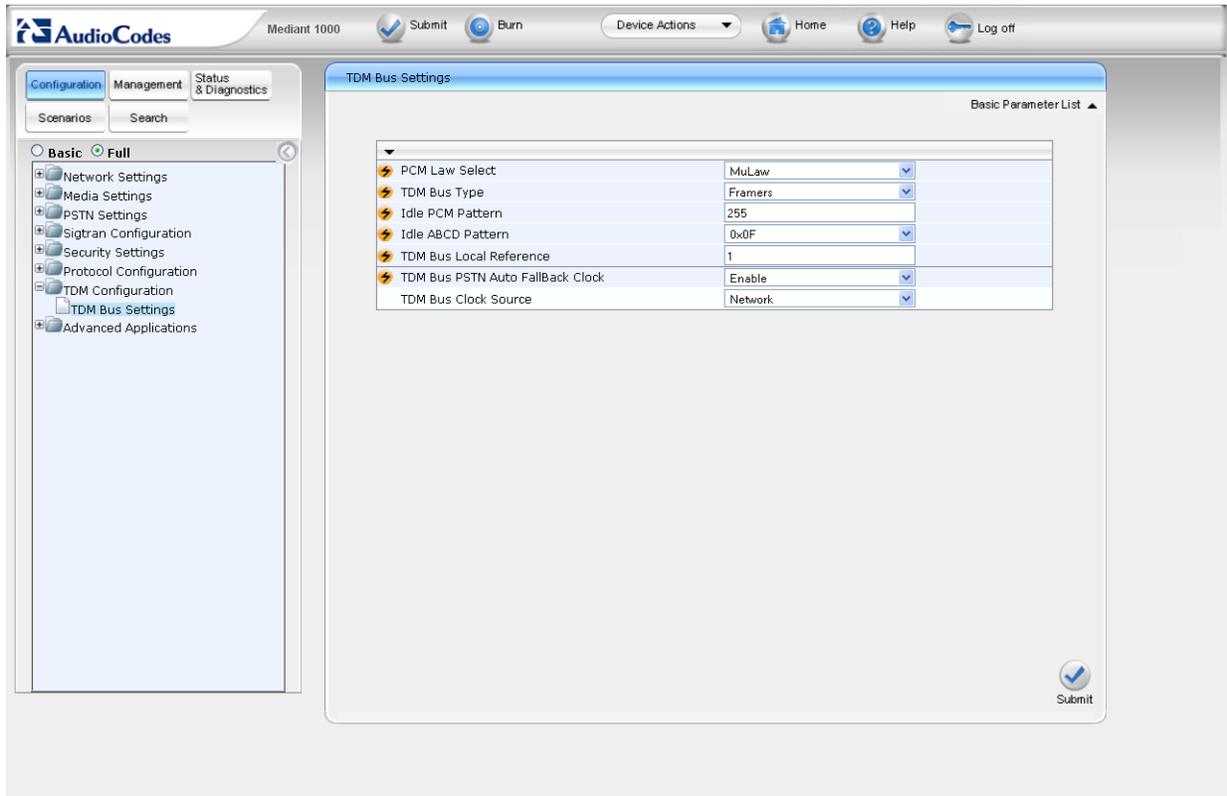
Navigate to **Media Settings** → **Voice Settings**. For **DTMF Transport Type**, select **RFC2833 Relay DTMF**. Default values may be retained for all other fields.

The screenshot displays the AudioCodes Mediant 1000 configuration interface. The left sidebar shows a navigation tree with 'Basic' selected. The main content area is titled 'Voice Settings' and contains a table of parameters. The 'DTMF Transport Type' is set to 'RFC2833 Relay DTMF'. A 'Submit' button is located at the bottom right of the settings area.

Parameter	Value
Voice Volume (-32 to 31 dB)	0
Input Gain (-32 to 31 dB)	0
Silence Suppression	Disable
DTMF Transport Type	RFC2833 Relay DTMF
DTMF Volume (-31 to 0 dB)	-11
Enable Answer Detector	Disable
Answer Detector Activity Delay	0
Answer Detector Silence Time	10
Answer Detector Redirection	0
Answer Detector Sensitivity	0
CAS Transport Type	CASEventsOnly
Echo Canceller	Enable

6.13. TDM Bus Settings

Click the **Full** radio button above the navigation pane on the left to return to the Full configuration menu tree, and then navigate to **TDM Configuration** → **TDM Bus Settings**. For **PCM Law Select**, select **MuLaw**. Default values may be retained for all other fields.



6.14. IP Media Settings and Answer Supervision

Enable voice detection/answer supervision.

Navigate to **Media Settings** → **IP Media Settings**. For **IPMedia Detectors**, select **Enable**. The “bolt” symbol to the left of this parameter indicates that a restart is required for the change to take effect. Default values may be retained for all other fields.

The screenshot displays the AudioCodes Mediant 1000 web interface. The top navigation bar includes the AudioCodes logo, the device name 'Mediant 1000', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The main interface is divided into two panes. The left pane shows a tree view of configuration categories: Configuration, Management, and Status & Diagnostics. Under Configuration, there are sub-categories for Scenarios and Search. The 'Basic' and 'Full' configuration modes are selected. The 'Full' mode is expanded to show a list of settings: Network Settings, Media Settings, Voice Settings, Fax/Modem/CID Settings, RTP/RTCP Settings, IP Media Settings (highlighted), General Media Settings, Analog Settings, Media Security, PSTN Settings, Sigtran Configuration, Security Settings, Protocol Configuration, TDM Configuration, and Advanced Applications. The right pane is titled 'IPMedia Settings' and contains a table of parameters. The 'IPMedia Detectors' parameter is set to 'Enable' and has a bolt icon to its left. Other parameters include 'Enable Answer Detector' (Disable), 'Answer Detector Activity Delay' (0), 'Answer Detector Silence Time' (10), 'Answer Detector Redirection' (0), 'Answer Detector Sensitivity' (0), 'Answer Machine Detector Sensitivity Resolution' (Normal), 'Answer Machine Detector Sensitivity' (3), 'Answer Machine Detector Beep Detection Timeout' (200), 'Answer Machine Detector Beep Detection Sensitivity' (0), 'Enable AGC' (Disable), 'AGC Slope' (3), 'AGC Redirection' (0), 'AGC Target Energy' (19), 'Enable Energy Detector' (Disable), 'Energy Detector Quality Factor' (4), 'Energy Detector Threshold' (3), 'Enable Pattern Detector' (Disable), 'Active Speakers Min Interval' (20), and 'Number of Media Channels' (60). Below the table are sections for 'Configure Audio Playback' (Playback Audio Format: PCMA) and 'Configure Audio Recording' (End Of Record Time: 60). A 'Submit' button is located at the bottom right of the settings pane.

Parameter	Value
IPMedia Detectors	Enable
Enable Answer Detector	Disable
Answer Detector Activity Delay	0
Answer Detector Silence Time	10
Answer Detector Redirection	0
Answer Detector Sensitivity	0
Answer Machine Detector Sensitivity Resolution	Normal
Answer Machine Detector Sensitivity	3
Answer Machine Detector Beep Detection Timeout	200
Answer Machine Detector Beep Detection Sensitivity	0
Enable AGC	Disable
AGC Slope	3
AGC Redirection	0
AGC Target Energy	19
Enable Energy Detector	Disable
Energy Detector Quality Factor	4
Energy Detector Threshold	3
Enable Pattern Detector	Disable
Active Speakers Min Interval	20
Number of Media Channels	60
Configure Audio Playback	
Playback Audio Format	PCMA
Configure Audio Recording	
End Of Record Time	60

Click the **Basic** radio button above the navigation pane on the left to return to the Basic configuration menu tree, and then navigate to **Advanced Applications** → **FXO Settings**. For **Answer Supervision**, select **Yes**.

The screenshot shows the AudioCodes Mediant 1000 web interface. The top navigation bar includes the AudioCodes logo, the device name 'Mediant 1000', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The left navigation pane has tabs for 'Configuration', 'Management', and 'Status & Diagnostics'. Under 'Configuration', there are 'Scenarios' and 'Search' buttons. The configuration tree shows 'Basic' selected, with sub-items: Network Settings, Media Settings, PSTN Settings, Protocol Configuration, Advanced Applications, Voice Mail Settings, and FXO Settings. The main content area is titled 'FXO Settings' and contains a table of configuration parameters:

Dialing Mode	One Stage
Waiting for Dial Tone	No
Time to Wait before Dialing [msec]	1000
Ring Detection Timeout [sec]	8
Reorder Tone Duration [sec]	255
Answer Supervision	Yes
Rings before Detecting Caller ID	1
Send Metering Message to IP	No
Disconnect Call on Busy Tone Detection (CAS)	Disable
Disconnect On Dial Tone	Disable
Guard Time Between Calls	1
FXO AutoDial Play BusyTone	Disable

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

7. General Test Approach and Test Results

The general test approach was to make calls from the PSTN through Communication Manager and the AudioCodes Mediant 1000 VoIP Media Gateway to reach Voice Portal. Using Voice Portal voice prompts, various Voice Portal functions were exercised and verified, particularly the 3 kinds of call transfers by Voice Portal (Blind, Consultative and Bridge) to either a second PSTN user or a Call Center agent.

The serviceability test cases were performed by disconnecting/reconnecting the line side and/or IP cables (to simulate network failures) and powering down then restarting the Mediant 1000 gateway (to simulate power outage).

The Mediant 1000 gateway passed compliance testing. The following observations were noted during compliance testing:

- If a PSTN call is placed to Voice Portal while the connection between the AudioCodes gateway and Voice Portal is down, or if no ports are available on Voice Portal, the caller will hear ringback, instead of a busy signal.
- CAS Table selection: Two CAS table were used during compliance testing. Depending on which CAS table is selected, different behaviors can be expected for calls transferred from Voice Portal. Consider the scenario where a call is placed from the PSTN and it arrives at Voice Portal. The call is then transferred from Voice Portal back to the PSTN. When the call is answered at the transferred-to party and voice is detected (due to the **IPMedia Detectors** field being enabled in **Section 6.14**), the AudioCodes gateway will notify Voice Portal that the call has been completed, via a 200 OK.

Now, consider the scenario where no voice is detected, (for example, if the call is unanswered, busy, or in queue hearing silence). Using CAS table **loopstarttable_fxo_Avaya**, approximately 24 seconds after the second leg call is initiated (from Voice Portal to the PSTN), AudioCodes gateway to notify Voice Portal of a request timeout (SIP 408 Request Timeout) and the second leg call for the transfer will be dropped. If the transfer was a bridged or consultative transfer, then the original PSTN call into Voice Portal will still be connected. The caller and Voice Portal will then have the option to reroute the call.

In the same scenario, where no voice is detected and CAS table **loopstarttable_fxo_Avaya_AutoConnect** is used, the AudioCodes gateway will automatically notify Voice Portal that the second leg call for the transfer has been completed, via a 200 OK. The 200 OK is sent 20 seconds after initiating the second leg call. As a result, the transferred call will not be dropped. The caller will continue to hear ringback, a busy signal, silence, etc. However, since Voice Portal has been notified that the transfer call has been completed, the option for Voice Portal to reroute the call is no longer available.

It is important for the AudioCodes administrator to understand the consequences of choosing each CAS table, and then select the appropriate table for the desired behavior.

8. Verification Steps

This section provides the verification steps that may be performed to verify that a PSTN call can reach Voice Portal through Communication Manager and the AudioCodes Mediant 1000 VoIP Media Gateway.

1. From VPMS (Voice Portal Management System) web interface, verify that the MPP server in use is online and running as shown below.

The screenshot displays the Avaya Voice Portal 5.0 MPP Manager web interface. The top navigation bar includes the Avaya logo, user information (Welcome, admin), and navigation links (Home, Help, Logoff). The left sidebar contains a tree view of system management options, with 'MPP Manager' selected under 'System Management'. The main content area shows the MPP Manager status page, including a breadcrumb trail, a title, a refresh button, and a description of the page's function. A table lists the current state of two MPP servers, mpp1 and mpp2. Below the table are sections for State Commands and Restart/Reboot Options.

	Server Name	Mode	State	Config	Auto Restart	Restart Schedule		Active Calls	
						Today	Recurring	In	Out
<input type="checkbox"/>	mpp1	Online	Running	OK	No	No	None	0	0
<input type="checkbox"/>	mpp2	Online	Running	OK	No	No	None	0	0

State Commands: Start, Stop, Restart, Reboot, Halt, Cancel

Restart/Reboot Options:

- One server at a time
- All selected servers at the same time

Mode Commands: Offline, Test, Online

Help

2. Make a PSTN call to access Voice Portal. Verify that
 - The Avaya Voice Portal voice greeting as defined by the configured voice application is provided
 - VPMS web interface shows that one port is in Connected state as shown below

The screenshot shows the Avaya Voice Portal 5.0 (VoicePortal) web interface. The page title is "Port Distribution (6/11/10 11:33:44 AM EDT)". The page content includes a table of port distribution information. The table has the following columns: Port, Mode, State, Port Group, Protocol, Current Allocation, and Base Allocation. The table shows 24 ports in total. The first port (5220) is in a "Connected" state, while all other ports (5221-5223 and 1-10) are in "In service" states. The interface also includes a navigation menu on the left and a "Help" button at the bottom.

Port	Mode	State	Port Group	Protocol	Current Allocation	Base Allocation
5220	Online	Connected	AudioCodes Mediant 1000 SIP_Trunk	H323	mpp1	
5221	Online	In service	8300	H323	mpp2	
5222	Online	In service	8300	H323	mpp1	
5223	Online	In service	8300	H323	mpp2	
1	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp1	
2	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp1	
3	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp1	
4	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp1	
5	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp1	
6	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp1	
7	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp1	
8	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp1	
9	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp1	
10	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp1	
1	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp2	
2	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp2	
3	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp2	
4	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp2	
5	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp2	
6	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp2	
7	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp2	
8	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp2	
9	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp2	
10	Online	In service	AudioCodes Mediant 1000 SIP_Trunk		mpp2	

3. Select the voice prompt selection to transfer the call to another user on the PSTN. Verify that two-way audio is established between the two PSTN users.

9. Conclusion

The AudioCodes Median 1000 VoIP Media Gateway version 6.0 passed compliance testing. These Application Notes describe the configurations required for AudioCodes Median 1000 VoIP Media Gateway to successfully interoperate with Avaya Voice Portal (via a SIP trunking interface) and Avaya Aura™ Communication Manager (via a line side T1 interface). Most of the feature and serviceability test cases passed, the failed test cases did not block compliance testing (See **Section 7** for observations noted).

10. Additional References

This section references the product documentation relevant to these Application Notes.

The following Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Administering Avaya Aura™ Communication Manager*, Doc ID 03-300509, May 2009.
- [2] *Feature Description and Implementation for Avaya Communication Manager*, Doc ID555-245-205, Issue 7, Release 5.2, May 2009
- [3] *Administering Voice Portal*, June 2010.

Product documentation for the AudioCodes Mediant 1000 VoIP Media Gateway can be obtained from AudioCodes at the following web sites: <http://www.audiocodes.com/products/mediant-1000> and <http://audiocodes.com/support>.

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