



## **Application Notes for Configuring AudioCodes Mediant 1000 VoIP Media Gateway to Interoperate with Avaya Voice Portal using Line Side E1 Connectivity to Avaya Aura<sup>TM</sup> 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<sup>TM</sup> Communication Manager (using a line side E1 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 E1 gateway connecting Avaya Voice Portal to Avaya Aura<sup>TM</sup> 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 E1). 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 E1 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 E1 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 E1 cables to simulate network failures, and stopping/starting AudioCodes Mediant 1000 to simulate power outages.

## 1.2. Support

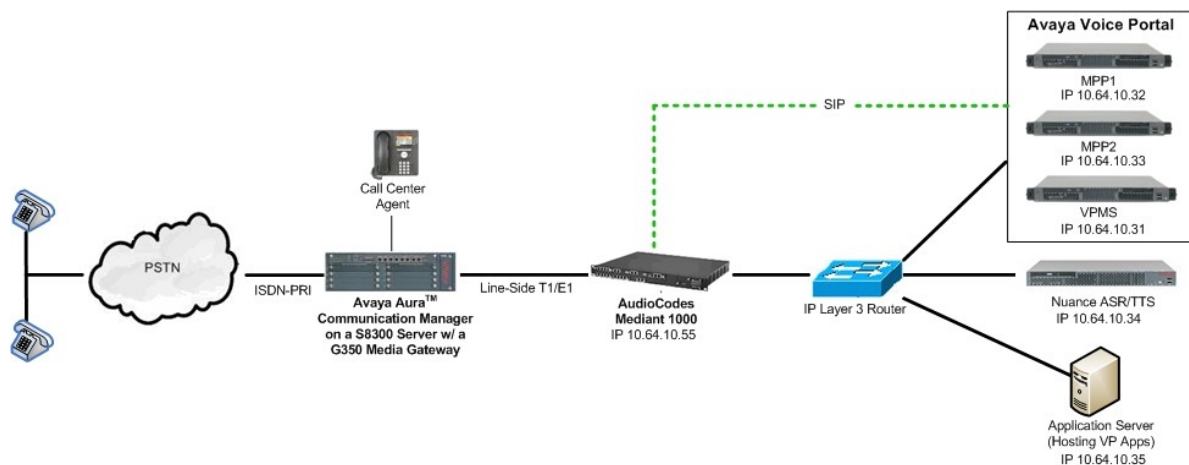
For technical support on the AudioCodes Mediant 1000 VoIP Media Gateway, contact AudioCodes via the support link at [www.audiocodes.com](http://www.audiocodes.com).

## 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 E1 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 E1 connection to Mediant 1000 through Communication Manager. Mediant 1000 will then route the calls from its line side E1 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 a 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"><li>Voice Portal Management System (VPMS)</li><li>Media Processing Platform (MPP)</li></ul>	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"><li>Nuance OpenSpeech Recognizer</li><li>Nuance RealSpeak</li></ul>	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 E1 on Avaya Aura™ Communication Manager

This section provides the procedures for configuring Communication Manager for line side E1 connectivity to the AudioCodes Mediant 1000 VoIP Media Gateway. The procedures include the following areas:

- Configure DS1 (to be used for line side E1 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 (5501-5530).
- 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 54101-54130).
- 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 E1 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 **2.048**.
- Set **Line Coding** to **hdb3**.
- Set **Signaling Mode** to **CAS**.
- Set **Interconnect** to **pbx**.
- Set **Country Protocol** to **1**.
- Set **CRC** to **y**.
- Set **Interface Companding** to **alaw**.
- Use the default values for the remaining fields.

<b>add ds1 001v6</b>		Page 1 of 1
DS1 CIRCUIT PACK		
Location: 001V6	Name: line side E1	
Bit Rate: 2.048	Line Coding: hdb3	
Signaling Mode: CAS		
Interconnect: pbx	Country Protocol: 1	
Interface Companding: alaw	CRC? y	
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, 30 ports were configured with an extension range of 5501 to 5530.

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.

<b>add station 5510</b>		Page 1 of 4
STATION		
Extension: 5510	Lock Messages? n	BCC: 0
<b>Type: DS1FD</b>	Security Code:	TN: 1
<b>Port: 001V610</b>	Coverage Path 1:	COR: 1
<b>Name: line-side port 10</b>	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**).

<b>add vdn 53500</b>		Page 1 of 3
VECTOR DIRECTORY NUMBER		
<b>Extension: 53500</b>		
<b>Name*: Voice Portal</b>		
<b>Destination: Vector Number 2</b>		
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	HUNT GROUP	Page 2 of 3
<b>Skill? y</b>		
<b>AAS? y</b>		
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 54130 were created.

add agent-loginID 54110	AGENT LOGINID	Page 1 of 2
Login ID: 54110		
<b>AAS? y</b>		
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:		
<b>Port Extension: 5510</b>		
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**.

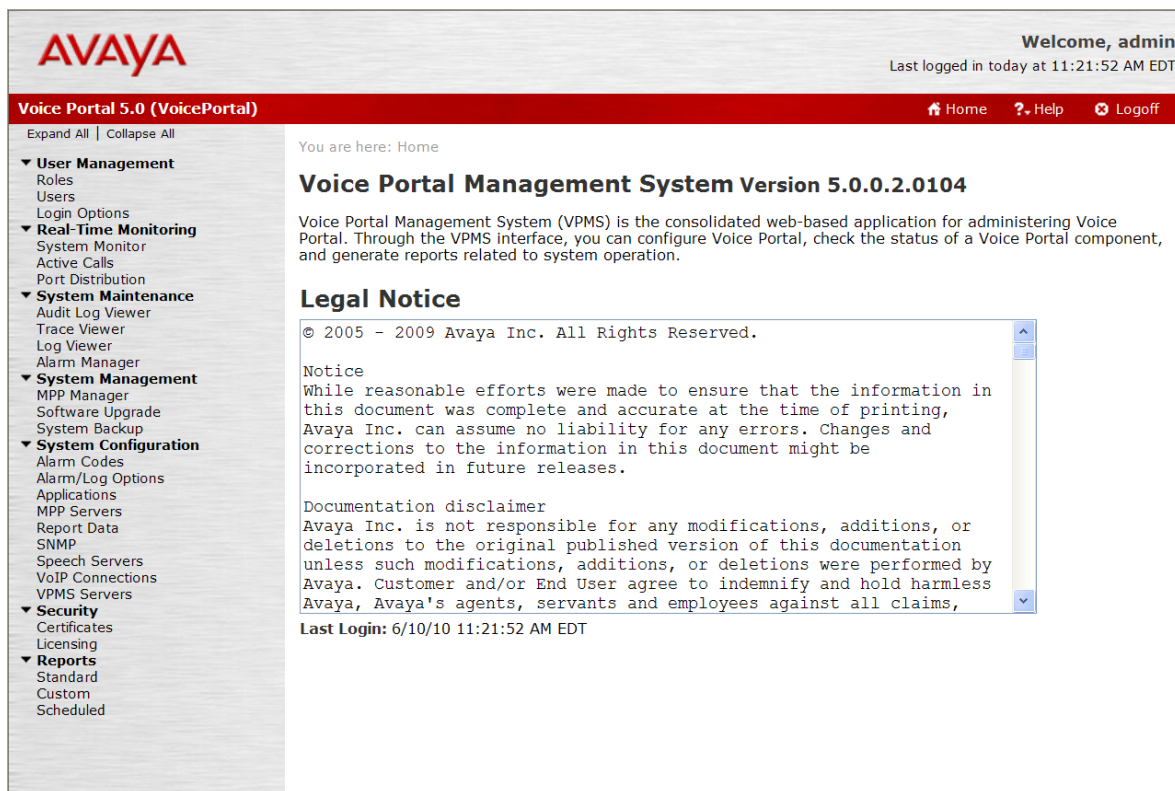
add agent-loginID 54110										Page 2 of 2	
AGENT LOGINID											
Direct Agent Skill:										Service Objective? n	
Call Handling Preference: skill-level										Local Call Preference? n	
	SN	RL	SL			SN	RL	SL			
1:	4		1			16:					
2:						17:					
3:						18:					
4:						19:					
5:						20:					
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.



**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.

**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

- ▼ **User Management**
  - Roles
  - Users
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- ▼ **Real-Time Monitoring**
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  - Applications
  - MPP Servers
  - Report Data
  - SNMP
  - Speech Servers
  - VoIP Connections
  - VPMS Servers
- ▼ **Security**
  - Certificates
  - Licensing
- ▼ **Reports**
  - Standard
  - Custom
  - Scheduled


You are here: [Home](#) > [Security](#) > [Certificates](#)

### Certificates

This page displays the Voice Portal root certificate and application certificate that are currently in effect.

Root Certificate Application Certificates Speech Server Certificates

#### Security Certificate

 [Export](#)

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

#### Install New Security Certificate

Enter Security Certificate Path:  [Browse...](#)

Password:

[Install](#) [Cancel](#) [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.

**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](#) > [VoIP Connections](#) > Change SIP Connection

### Change SIP Connection

Use this page to change the configuration of a SIP connection.

Name: AudioCodes Mediant 1000

Enable: ☒ Yes ☐ No

Proxy Transport: TCP

#### Proxy Servers

Address	Port	Administration
10.64.10.29	5060	<b>Administration</b> Remove

[Additional Proxy Server](#)

Listener Port: 5060

SIP Domain: 10.64.10.29

P-Asserted-Identity:

#### Call Capacity

Maximum Simultaneous Calls: 20

☒ All Calls can be either inbound or outbound

☐ Configure number of inbound and outbound calls allowed

**Save** **Apply** **Cancel** **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

- ▼ User Management
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  - Users
  - Login Options
- ▼ Real-Time Monitoring
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  - Active Calls
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- ▼ System Maintenance
  - Audit Log Viewer
  - Trace Viewer
  - Log Viewer
  - Alarm Manager
- ▼ System Management
  - MPP Manager
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- ▼ System Configuration
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  - MPP Servers
  - Report Data
  - SNMP
  - Speech Servers
  - VoIP Connections
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  - Licensing
- ▼ Reports
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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: mpp1

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: 3f1ba1f6163:67:37:0e: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)

Expand All | Collapse All

User Management

Users

Login Options

Real-Time Monitoring

System Monitor

Active Calls

Port Distribution

System Maintenance

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Trace Viewer

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Alarm Manager

System Management

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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"/>

Save

Apply

Cancel

Help

## 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 (VoicePortal) interface. The top navigation bar includes the Avaya logo, a welcome message for 'admin', and links for Home, Help, and Logout. The left sidebar contains a tree view of system configuration options, with 'System Configuration' expanded. The main content area is titled 'Change ASR Server' and provides a form for configuring an ASR server. The form includes fields for Name, Enable, Engine Type, Network Address, Base Port, Total Number of Licensed ASR Resources, New Connection per Session, Languages, MRCP settings (Ping Interval, Response Timeout, Protocol), and RTSP URL. The 'Languages' dropdown is open, showing a list of language options including Dutch, English (Australia, UK, India, Singapore), and English (USA).

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

**Voice Portal 5.0 (VoicePortal)** Home Help Logout

Expand All Collapse All

▼ **User Management**  
Roles  
Users  
Login Options

▼ **Real-Time Monitoring**  
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▼ **System Maintenance**  
Audit Log Viewer  
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▼ **Security**  
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Licensing

▼ **Reports**  
Standard  
Custom  
Scheduled

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

### Change ASR Server

Use this page to change the configuration of an ASR server.

Name: Nuance ASR

Enable: ☒ Yes ☐ No

Engine Type: Nuance

Network Address: 10.64.10.34

Base Port: 4900

Total Number of Licensed ASR Resources: 4

New Connection per Session: ☐ Yes ☒ No

Languages: Dutch(Netherlands) nl-nl  
English(Australia) en-au  
English(UK) en-gb  
English(India) en-in  
English(Singapore) en-SG  
English(USA) en-us

**MRCP**

Ping Interval: 15 second(s)

Response Timeout: 4 second(s)

Protocol: MRCP V1

RTSP URL: 10.64.10.34/media/speechrecognizer

**Save Apply Cancel 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

Expand All | Collapse All

- ▼ **User Management**
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  - Users
  - Login Options
- ▼ **Real-Time Monitoring**
  - System Monitor
  - Active Calls
  - Port Distribution
- ▼ **System Maintenance**
  - Audit Log Viewer
  - Trace Viewer
  - Log Viewer
  - Alarm Manager
- ▼ **System Management**
  - MPP Manager
  - Software Upgrade
  - System Backup
- ▼ **System Configuration**
  - Alarm Codes
  - Alarm/Log Options
  - Applications
  - MPP Servers
  - Report Data
  - SNMP
  - Speech Servers
  - VoIP Connections
  - VPMS Servers
- ▼ **Security**
  - Certificates
  - Licensing
- ▼ **Reports**
  - Standard
  - Custom
  - Scheduled

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-El 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 (VoicePortal) administration interface. The top navigation bar includes the Avaya logo, the text 'Voice Portal 5.0 (VoicePortal)', and user information: 'Welcome, admin' and 'Last logged in today at 11:33:04 AM EDT'. A red banner at the top right contains links for 'Home', 'Help', and 'Logoff'. The left sidebar shows a tree view of system components, with 'System Configuration' expanded to show 'Applications'. The main content area is titled 'Change Application' and includes the following fields and sections:

- Name:** Intro
- Enable:** ☒ Yes ☐ No
- MIME Type:** VoiceXML (selected from a dropdown)
- VoiceXML URL:** http://10.64.10.35/mpp/misc/avptestapp/intro.vxml (with a 'Verify' button)
- Speech Servers:**
  - ASR:** Nuance (selected from a dropdown)
  - TTS:** Nuance (selected from a dropdown)
  - Languages:** English(USA) en-us (selected from a dropdown)
  - Voices:** English(USA) en-US Jennifer F (selected from a dropdown)
- Application Launch:**
  - Type:** ☒ Inbound ☐ Inbound Default ☐ Outbound
  - Number:** ☒ Number ☐ Number Range ☐ URI
  - Called Number:** 5511, 5220 - 5221, 5510 (with an 'Add' button and a 'Remove' button)
- Speech Parameters:** (expandable section)
- Reporting Parameters:** (expandable section)
- Advanced Parameters:** (expandable section)
- Buttons:** Save, Apply, Cancel, Help

## 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**.

**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

▼ **User Management**  
Roles  
Users  
Login Options

▼ **Real-Time Monitoring**  
System Monitor  
Active Calls  
Port Distribution

▼ **System Maintenance**  
Audit Log Viewer  
Trace Viewer  
Log Viewer  
Alarm Manager

▼ **System Management**  
**MPP Manager**  
Software Upgrade  
System Backup

▼ **System Configuration**  
Alarm Codes  
Alarm/Log Options  
Applications  
MPP Servers  
Report Data  
SNMP  
Speech Servers  
VoIP Connections  
VPMS Servers

▼ **Security**  
Certificates  
Licensing

▼ **Reports**  
Standard  
Custom  
Scheduled

You are here: [Home](#) > System Management > MPP Manager

**MPP Manager (6/10/10 11:38:22 AM EDT)** Refresh

This page displays the current state of each MPP in the Voice Portal system. To enable the state and mode commands, select one or more MPPs. To enable the mode commands, the selected MPPs must also be stopped.

Last Poll: 6/10/10 11:38:16 AM EDT

	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

**State Commands**

Start Stop Restart Reboot Halt Cancel

**Mode Commands**

Offline Test Online

**Restart/Reboot Options**

☐ One server at a time

☒ All selected servers at the same time

Help

## 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. The top navigation bar includes the AudioCodes logo, 'Mediant 1000', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The left sidebar contains tabs for 'Configuration', 'Management', and 'Status & Diagnostics', with a search bar and a tree view showing categories like 'Basic' and 'Full'.

The main content area, titled 'Mediant 1000 Home Page', features a status dashboard with six modules: 1 Digital, 2 BRI, 3 FXO, CPU, 4 FXS, and 5. Below this, there are two tables: 'General Information' and 'Channel (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

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**.

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 sidebar contains a tree view with categories: Configuration, Management, and Status & Diagnostics. Under Configuration, there are sub-items: Scenarios, Search, Basic (selected), and Full. The Basic section is expanded, showing Network Settings, IP Settings (selected), Application Settings, Media Settings, PSTN Settings, Protocol Configuration, and Advanced Applications. The main content area is titled 'IP Settings' and contains two sections: 'Single IP Settings' and 'Multiple Interface Settings'. The 'Single IP Settings' section has three input fields: 'IP Address' (10.64.10.29), 'Subnet Mask' (255.255.255.0), and 'Default Gateway Address' (10.64.10.1). The 'Multiple Interface Settings' section has a dropdown menu labeled 'Multiple Interface Table' and a right-pointing arrow button. A 'Submit' button with a checkmark icon is located at the bottom right of the main content area.

Single IP Settings	
IP Address	10.64.10.29
Subnet Mask	255.255.255.0
Default Gateway Address	10.64.10.1

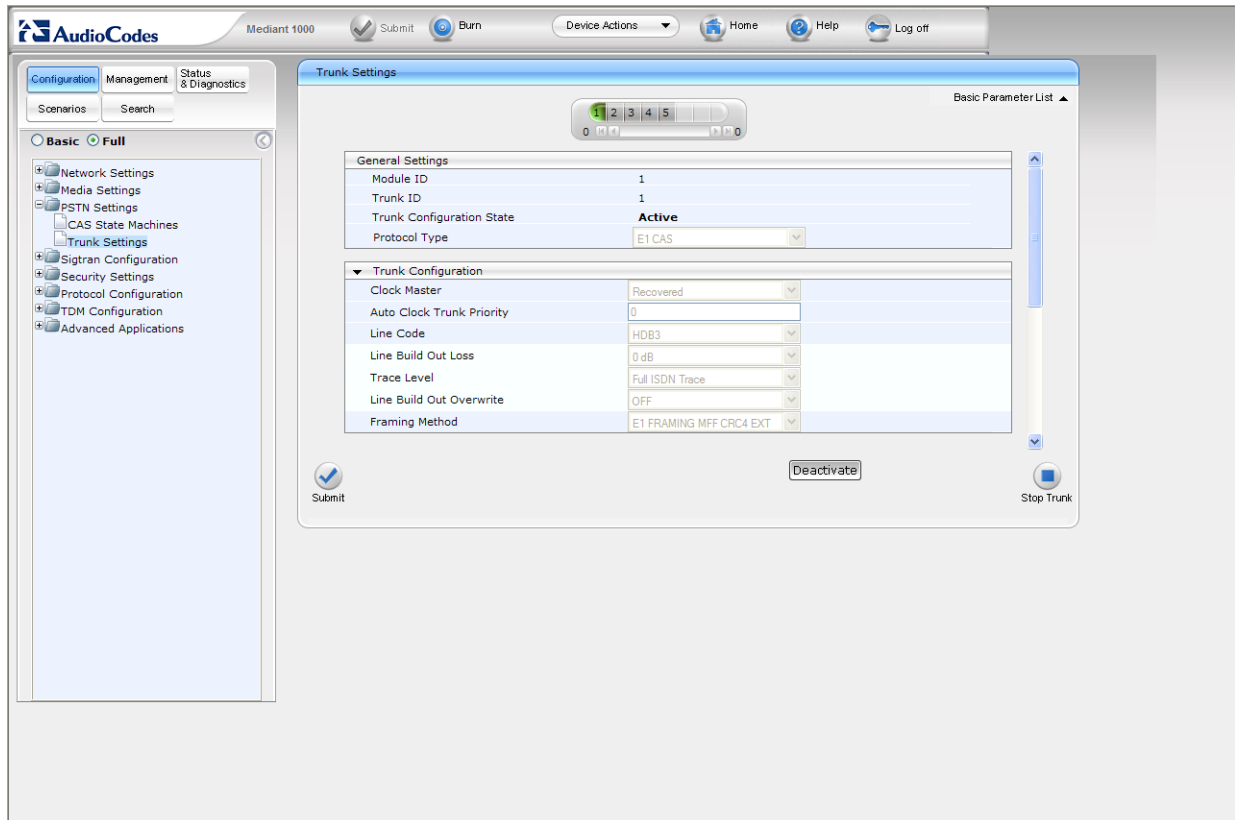
  

Multiple Interface Settings	
Multiple Interface Table	

## 6.2. PSTN trunk setting

Navigate to **PSTN Settings** → **Trunk Settings** to configure the line side E1 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 *E1 CAS*.
- Set **Line Code** to *HDB3*.
- Set **Framing Method** to *E1 FRAMING MFF CRC4 EXT*.



Scroll down, then configure following parameters as described:

- 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.

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, and Trunk Settings. The main area is titled 'Trunk Settings' and contains a 'Basic Parameter List' section. The 'Framing Method' is set to 'E1 FRAMING MFF CRC4 EXT'. Under 'CAS Configuration', 'CAS Table per Trunk' is selected with a value of 'loopstarttable\_fw\_2.dat (2010-5)'. Under 'PSTN Alert Timeout', the value is set to '100'. Other settings include 'Out-Of-Service Behavior' (Default), 'Remove Calling Name' (Use Global Parameter), and 'Play Ringback Tone to Trunk' (Not Configured). At the bottom, there are 'Submit', 'Deactivate', and 'Stop Trunk' buttons.

After all the parameters are properly specified, click the **Apply Trunk Settings** button at the bottom of the screen (not shown).



### 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 web interface. The top navigation bar includes the AudioCodes logo, 'Mediant 1000', and buttons for '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.
- Scroll down and set the **Max Number of Active Calls** field to an appropriate value.

Default values may be retained for all other fields.

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left navigation pane is expanded to 'SIP Advanced Parameters' under 'Protocol Configuration'. The 'Full' radio button is selected. The main area displays the 'Advanced Parameters' configuration page. The 'PSTN Alert Timeout' is set to 100. The 'Disconnect and Answer Supervision' section includes settings for 'Send Digit Pattern on Connect', 'Enable Polarity Reversal', 'Enable Current Disconnect', 'Disconnect on Broken Connection', 'Broken Connection Timeout [100 msec]', 'Disconnect Call on Silence Detection', 'Silence Detection Period [sec]', 'Silence Detection Method', and 'Enable Fax Re-Routing'. The 'CDR and Debug' section includes 'CDR Server IP Address', 'CDR Report Level', and 'Debug Level'. The 'Misc. Parameters' section includes 'Progress Indicator to IP', 'Enable X-Channel Header', 'Enable Busy Out', 'Graceful Busy Out Timeout [sec]', 'Default Release Cause', 'Max Number of Active Calls' (set to 150), and 'Max Call Duration [min]'.

Advanced Parameters	
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
Max Call Duration [min]	0

## 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 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 sidebar shows a configuration tree with 'Basic' selected. The main panel is titled 'Proxy & Registration' and contains a list of parameters with their current values. At the bottom of the panel 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 left sidebar contains a tree view with the following categories: Configuration, Management, and Status & Diagnostics. Under Configuration, there are sub-categories: Scenarios, Search, Basic, and Full. The Basic category is selected, and the tree view shows the following items: Network Settings, Media Settings, PSTN Settings, Protocol Configuration, Applications Enabling, Trunk Group, Protocol Definition, Proxies, Registration, IP Groups, Proxy & Registration, Proxy Sets Table, Coders And Profile Definitions, Manipulation Tables, Routing Tables, Endpoint Settings, Digital Gateway, and Advanced Applications. The Proxy Sets Table item is selected. The main content area displays the Proxy Sets Table configuration page. At the top, there is a dropdown menu for Proxy Set ID, currently set to 0. Below this is a table with 5 rows and 2 columns: Proxy Address and Transport Type. The first row is pre-filled with 10.64.10.32 and TCP. The other rows are empty. Below the table is a section with several configuration options: Enable Proxy Keep Alive (Disable), Proxy Keep Alive Time (60), Proxy Load Balancing Method (Disable), Is Proxy Hot Swap (No), and SRD Index (0). A Submit button is located at the bottom right 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 sidebar menu shows the configuration tree with 'Coders' selected under 'Coders And Profile Definitions'. The main area is titled 'Coders Table' and contains a table with the following columns: Coder Name, Packetization Time, Rate, Payload Type, and Silence Suppression. The table lists two codecs: G.711U-law and G.729, with their respective settings. A 'Submit' button is located at the bottom right of the table area.

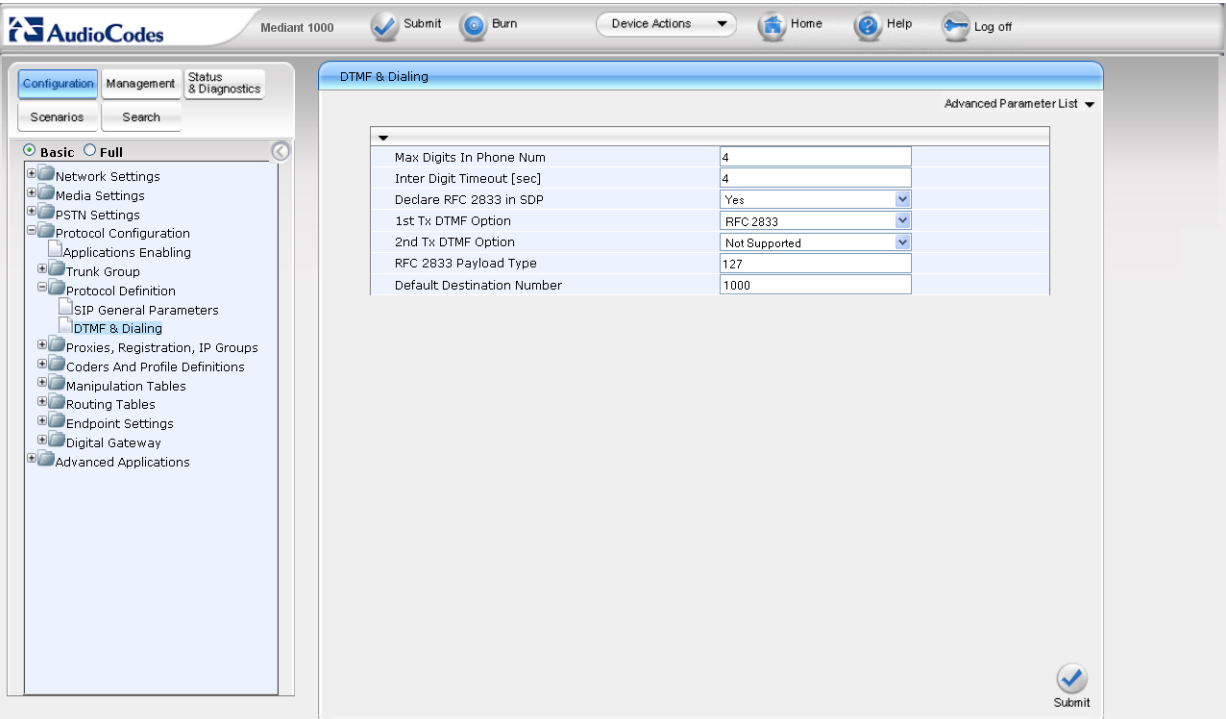
Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711U-law	20	64	0	Disabled
G.729	20	8	18	Enable

## 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 **1<sup>st</sup> 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.



DTMF & Dialing	
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

## 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 **1-30** means 30 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 – 30** 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 Configuration, Management, and Status & Diagnostics. Under Configuration, there are sub-items for Network Settings, Media Settings, PSTN Settings, Protocol Configuration, Applications Enabling, Trunk Group, Trunk Group Settings, Protocol Definition, Proxies, Registration, IP Groups, Coders And Profile Definitions, Manipulation Tables, Routing Tables, Endpoint Settings, Digital Gateway, and Advanced Applications. The 'Trunk Group' item is selected.

The main area displays the 'Trunk Group Table' configuration. At the top, there are two dropdown menus: 'Add Phone Context As Prefix' (set to 'Disable') and 'Trunk Group Index' (set to '1-10'). Below these is a table with the following columns: Group Index, Module, From Trunk, To Trunk, Channels, Phone Number, Trunk Group ID, and Tel Profile ID.

Group Index	Module	From Trunk	To Trunk	Channels	Phone Number	Trunk Group ID	Tel Profile ID
1	Module 1 PRI	1	1	1-30	5501	1	0
2							
3							
4							
5							
6							
7							
8							
9							
10							

At the bottom right of the table area is a 'Submit' button.

## 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.

AudioCodes Mediant 1000 Submit Burn Device Actions Home Help Log off

Configuration Management Status & Diagnostics

Scenarios Search

Basic Full

- Network Settings
- Media Settings
- PSTN Settings
- Protocol Configuration
  - Applications Enabling
  - Trunk Group
    - Trunk Group Settings
- Protocol Definition
- Proxies, Registration, IP Groups
- Coders And Profile Definitions
- Manipulation Tables
- Routing Tables
- Endpoint Settings
- Digital Gateway
- Advanced Applications

Trunk Group Settings

Advanced Parameter List

Index 1-12

	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			

Submit



## 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 tree view with categories like Network Settings, PSTN Settings, Protocol Configuration, and Routing Tables. The 'IP to Trunk Group Routing' option is selected under Routing Tables. The main area displays the 'IP To Trunk Group Routing Table' with a table of 12 rows. The first row is pre-filled with asterisks in the first three columns and the value '1' in the 'Trunk Group ID' column. The table has columns for 'Dest. Phone Prefix', 'Source Phone Prefix', 'Source IP Address', 'Trunk Group ID', and 'IP Profile ID'. Above the table, there are dropdowns for 'Routing Index' (set to 1-12) and 'IP To Tel Routing Mode' (set to 'Route calls before manipulation'). A 'Submit' button is at the bottom right.

	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 shows the AudioCodes Mediant 1000 web interface. The top navigation bar includes the AudioCodes logo, 'Mediant 1000', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. On the left, a sidebar contains tabs for 'Configuration', 'Management', and 'Status & Diagnostics'. Under 'Configuration', there are sub-tabs for 'Scenarios' and 'Search'. A tree view shows the following structure: Basic (selected), Full, Network Settings, Media Settings (expanded), Voice Settings (selected), Fax/Modem/CID Settings, PSTN Settings, Protocol Configuration, and Advanced Applications. The main content area is titled 'Voice Settings' and features an 'Advanced Parameter List' dropdown. The list contains the following parameters and their values:

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

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

## 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 **ALaw**. Default values may be retained for all other fields.

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'. On the left, a navigation pane has tabs for 'Configuration', 'Management', and 'Status & Diagnostics'. Under 'Configuration', there are sub-tabs for 'Scenarios' and 'Search'. The 'Full' radio button is selected. The navigation tree on the left lists various settings categories, with 'TDM Bus Settings' highlighted. The main content area is titled 'TDM Bus Settings' and contains a table of parameters. A 'Basic Parameter List' link is visible in the top right of the content area. A 'Submit' button with a checkmark icon is located in the bottom right corner of the configuration area.

Basic Parameter List ▲	
PCM Law Select	ALaw
TDM Bus Type	Frame
Idle PCM Pattern	213
Idle ABCD Pattern	0x0F
TDM Bus Local Reference	1
TDM Bus PSTN Auto FallBack Clock	Enable
TDM Bus Clock Source	Network

## 6.14. IP Media Settings

Enable voice detection/answer supervision.

Navigate to **Media Settings → IP Media Settings**. For **IPMedia Detectors**, select **Enable**. This will enable Voice Detection. Default values may be retained for all other fields.

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar contains a tree view with the following categories: Configuration, Management, Status & Diagnostics, Scenarios, and Search. Under Configuration, the 'Full' tab is selected, and the tree view shows: Network Settings, Media Settings (selected), Voice Settings, Fax/Modem/CID Settings, RTP/RTCP Settings, IP Media Settings (selected), General Media Settings, Analog Settings, Media Security, PSTN Settings, Sigtran Configuration, Security Settings, Protocol Configuration, TDM Configuration, and Advanced Applications. The main content area is titled 'IPMedia Settings' and contains a 'Basic Parameter List' table. The table has the following rows:

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	100

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

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 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 pane shows a tree structure with 'Basic' and 'Full' radio buttons. The 'Basic' radio button is selected. The tree includes 'Network Settings', 'Media Settings', 'PSTN Settings', 'Protocol Configuration', 'Advanced Applications', 'Voice Mail Settings', and 'FXO Settings'. The 'FXO Settings' page is displayed in the main area, featuring a table of configuration parameters. A 'Submit' button is located at the bottom right of the settings area.

Parameter	Value
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

## 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.

**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 Management > MPP Manager

### MPP Manager (6/10/10 11:38:22 AM EDT)

This page displays the current state of each MPP in the Voice Portal system. To enable the state and mode commands, select one or more MPPs. To enable the mode commands, the selected MPPs must also be stopped.

Last Poll: 6/10/10 11:38:16 AM EDT

	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**

**Mode Commands**

**Restart/Reboot Options**

☐ One server at a time

☒ All selected servers at the same time

2. Make a PSTN call to access Voice Portal. Verify that:
  - The 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 top navigation bar includes the Avaya logo, a welcome message for 'admin', and a 'Last logged in yesterday at 4:24:30 PM EDT' timestamp. The left sidebar contains a menu with categories like User Management, Real-Time Monitoring, System Maintenance, System Configuration, Security, and Reports. The main content area is titled 'Port Distribution (6/11/10 11:33:44 AM EDT)' and includes a 'Refresh' button. Below the title, a message states: 'This page displays information about how the telephony resources have been distributed to the MPPs. You configure the telephony resources on the VoIP Connections page.' A table titled 'Total Ports: 24' shows port status and allocation details. The table has columns for Port, Mode, State, Port Group, Protocol, Current Allocation, and Base Allocation. The data shows 24 ports, all in 'Online' and 'In service' states, with various port groups and protocols (H323, SIP\_Trunk) and current allocations (mpp1, mpp2).

Port	Mode	State	Port Group	Protocol	Current Allocation	Base Allocation
5220	Online	In service	8300	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	Connected	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 Mediant 1000 VoIP Media Gateway to successfully interoperate with Avaya Voice Portal (via a SIP trunking interface) and Avaya Aura<sup>TM</sup> Communication Manager (via a line side E1 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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