

Avaya Solution & Interoperability Test Lab

# Application Notes for Configuring AudioCodes Mediant 1000 VoIP Media Gateway with Avaya Voice Portal Using SIP Trunks – Issue 1.0

## Abstract

These Application Notes describe the configurations required for the AudioCodes Mediant 1000 VoIP Media Gateway to interoperate with Avaya Voice Portal using SIP trunking interface.

The AudioCodes Mediant 1000 VoIP Media Gateway serves as a gateway between TDM and IP networks. The Mediant 1000 supports multiple hardware interfaces and control protocols. Capacity can be scaled upward by adding additional interface modules. The compliance test configured the Mediant 1000 as a SIP to ISDN-PRI gateway connecting Avaya Voice Portal to PSTN.

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.

The AudioCodes Mediant 1000 VoIP Media Gateway serves as a gateway between TDM and IP networks. The Mediant 1000 supports multiple hardware interfaces and control protocols. Capacity can be scaled upward by adding additional interface modules. The compliance test configured the Mediant 1000 as a SIP to ISDN-PRI gateway connecting Avaya Voice Portal to the PSTN network through a simulated third party PBX. This solution allows Avaya Voice Portal to receive calls from the PSTN and transfer calls to the PSTN or to a third party 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:

- Basic calls from PSTN to reach Avaya Voice Portal
- Call transfers by Avaya Voice Portal to PSTN, including blind, consultative, and bridged transfers
- Call transfers by Avaya Voice Portal to a Call Center agent on simulated third party PBX, including blind, consultative, and bridged transfers
- DTMF tones / RFC 2833 support
- Sending UUI from Avaya Voice Portal to PSTN
- G.711 mu-law and G.711 a-law codec support
- T1/ISDN network interface between Mediant 1000 and simulated third party PBX
- SIP trunking interface between Mediant 1000 and Avaya Voice Portal

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

## 1.2. Support

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

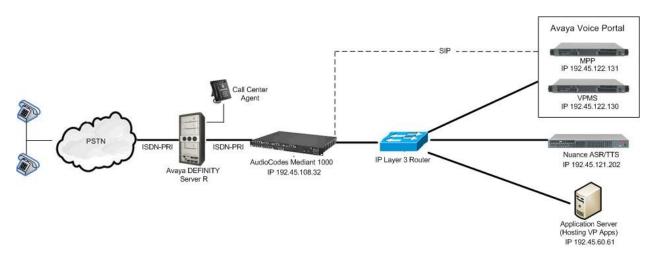
## 2. Reference Configuration

**Figure 1** illustrates the configuration used in the compliance test. In the sample configuration, the AudioCodes Mediant 1000 VoIP Media Gateway connects to Avaya Voice Portal through a SIP trunking interface on the one side, and to an Avaya DEFINITY Server R via an ISDN-PRI trunk on the other side. The Avaya DEFINITY Server R in turn has an ISDN-PRI connection to the PSTN. In this configuration, the Avaya DEFINITY Server R simulates a third party PBX with one Call Center agent phone configured directly on the PBX (for receiving PSTN calls transferred by Avaya Voice Portal).

Inbound calls from PSTN to Avaya Voice Portal will be routed across the ISDN-PRI connection to the Median 1000 through the Avaya DEFINITY Server R. The Mediant 1000 then routes the calls from its ISDN-PRI interface to its SIP interface to be terminated on the Avaya Voice Portal MPP (Media Processing Platform) server. Outbound calls to PSTN (transferred inbound call to another PSTN user on request from the original caller) follow the same path in the reverse order. Transferred calls to a Call Center agent on request from the original PSTN caller terminates on the agent phone connected directly to the PBX.

The incoming PSTN number of the ISDN-PRI trunk is mapped to the Avaya Voice Portal access number on the Avaya DEFINITY Server R.

In the compliance test, the Avaya Voice Portal consists of an MPP (Media Processing Platform) server 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 used in the test are also included in the test configuration.



#### Figure 1: AudioCodes Mediant 1000 VoIP Media Gateway with Avaya Voice Portal Using SIP Trunking Interface

# 3. Equipment and Software Validated

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

Equipment	Software
Avaya Voice Portal	5.0
• Voice Portal Management System (VPMS)	
Media Processing Platform (MPP)	
Application Server – HTTP Server running in	Microsoft Windows 2003
Windows	Server Service Pack 2
Nuance Speech Server	
Nuance OpenSpeech Recognizer	3.0
Nuance RealSpeak	4.0
Avaya DEFINITY Server R	R011r.03.1.635.0
Call Center Agent	Avaya one-X® Deskphone
Avaya 1600 Series IP Telephone (H.323)	Value Edition Release 1.100
Analog Telephones	-
AudioCodes Mediant 1000 VoIP Media Gateway	5.60A.024.003

# 4. Configure ISDN-PRI on Avaya DEFINITY Server R

This section provides the procedures for configuring Avaya DEFINITY Server R for the ISDN-PRI connection to the AudioCodes Mediant 1000 VoIP Media Gateway. The procedures include the following areas:

- Verify ISDN-PRI and Private Networking enablement
- DS1 circuit pack configuration
- Administer ISDN-PRI signaling group
- Administer ISDN-PRI trunk group
- Associate ISDN-PRI trunk group with ISDN-PRI signaling group
- Configure inbound and outbound routing for ISDN-PRI trunks

Note that the Avaya DEFINITY Server R was used in the compliance test to simulate a 3<sup>rd</sup> party PBX that supports ISDN-PRI interface to Mediant 1000. The specific ISDN-PRI configuration on the PBX is vendor-specific. The configurations on the Avaya DEFINITY Server R are given in these application notes as an example; similar configurations must be performed and tested if a different PBX is used.

Note also that in the configuration of the compliance test a Call Center agent phone is configured on the Avaya DEFINITY Server R (for receiving transferred PSTN calls by Avaya Voice Portal). The configuration of this agent phone is standard per PBX in use and therefore is not covered in these application notes. Similarly, configuration of the ISDN-PRI connection from the PBX to the PSTN is not included since it is beyond the scope of these application notes.

The configuration of Avaya DEFINITY Server R was performed using the System Access Terminal (SAT). After the completion of the configuration, perform a **save translation** command to make the changes permanent.

### 4.1. Verify ISDN-PRI Enablement

Use the **display system-parameters customer-options** command to verify that the ISDN-PRI feature is enabled on Page 3. If the feature is not enabled, contact an authorized Avaya sales representative to make the appropriate changes.

```
display system-parameters customer-options
                                                          Page
                                                                  3 of 10
                                                                             SPE B
                                 OPTIONAL FEATURES
   Emergency Access to Attendant? y
                                                            ISDN Feature Plus? y
                                      ISDN Network Call Redirection? y
           Enable 'dadmin' Login? n
           Enhanced Conferencing? n
                                                               ISDN-BRI Trunks? y
                  Enhanced EC500? n
                                                                      ISDN-PRI? y
         Extended Cvg/Fwd Admin? y
                                                        Local Spare Processor? n
    External Device Alarm Admin? y
                                                         Malicious Call Trace? n
  Five Port Networks Max Per MCC? n
                                                    Media Encryption Over IP? n
               Flexible Billing? n Mode Code for Centralized Voice Mail? n
   Forced Entry of Account Codes? n
     Global Call Classification? n
                                                     Multifrequency Signaling? y
            Hospitality (Basic)? y Multimedia Appl. Server Interface (MASI)? y
Hospitality (G3V3 Enhancements)? y
IP Trunks? y
Multimedia Call Handling (Basic)? y
Multimedia Call Handling (Enhanced)? y
Multiple Locations? n
                                                          Multiple Locations? n
           IP Attendant Consoles? n Personal Station Access (PSA)? y
                     IP Stations? y
                                                               Posted Messages? n
```

On Page 4, verify that Private Networking is enabled.

display system-parameters customer-options OPTIONAL FE	Page 4 of 10 SPE B
OPTIONAL FE	AIURES
	The sector Development of the sector of the
PNC Duplication? n	Tenant Partitioning? n
Port Network Support? y	Terminal Trans. Init. (TTI)? y
	Time of Day Routing? n
Processor and System MSP? y	Uniform Dialing Plan? y
Private Networking? y	Usage Allocation Enhancements? y
	TN2501 VAL Maximum Capacity? y
Remote Office? y	Wideband Switching? y
Restrict Call Forward Off Net? y	Wireless? n
Secondary Data Module? y	
Station and Trunk MSP? y	
Station as Virtual Extension? n	
beacton as virtual Extension. In	
System Management Data Transfer? Y	

## 4.2. DS1 Circuit Pack Configuration

An ISDN-PRI trunk requires the use of a DS1 circuit pack. To configure the DS1 circuit pack, use the **add ds1** n command where n is the location in the chassis taken by the DS1 circuit pack to be used. In the example below, the location is 1c07. The Name field can be any descriptive name. All other fields in bold in the example below should be set to the value shown. The combination of Country Protocol (1) and Protocol Version (b) determine which version of ISDN-PRI will be used, specifically National ISDN 2. The Connect setting of *line-side* will match the ISDN termination side setting (*User side*) on Mediant 1000. Default values may be retained for all other fields.

```
add ds1 1c07
                                                                  2
                                                                      SPE B
                                                    Page
                                                           1 of
                               DS1 CIRCUIT PACK
           Location: 01C07
                                                     Name: GW DS1
           Bit Rate: 1.544
                                              Line Coding: b8zs
  Line Compensation: 1
                                             Framing Mode: esf
     Signaling Mode: isdn-pri
           Connect: line-side
  TN-C7 Long Timers? n
                                         Country Protocol: 1
Interworking Message: PROGress
                                        Protocol Version: b
Interface Companding: mulaw
                                                      CRC? n
         Idle Code: 11111111
                             DCP/Analog Bearer Capability: 3.1kHz
      Slip Detection? n
                                       Near-end CSU Type: other
                         Alarm When PRI Endpoint Detached? y
                                Block Progress Indicator? N
```

### 4.3. Administer ISDN-PRI Signaling Group

Use the **add signaling-group** n command, where n is the number of an unused signaling group to be added. Set the fields in **bold** to the values shown below. The **Primary D-Channel** field is set to the 24th channel of the DS1 board in slot 1c07. This board was added to the configuration in the previous step. The **Trunk Group for Channel Selection** field will be populated at a later step after the trunk group has been created.

add signaling-group			Page	1 of	5	SPE B	
	SIGNALING	GROUP					
Group Number: 12	Group Type:	isdn-pri	i				
	Associated Signaling?	У	Max nu	mber of	NCA	TSC: (	C
	Primary D-Channel:	01C0724	Max n	umber o:	f CA	TSC: (	C
			Trunk Gr	oup for	NCA	TSC:	
Trunk Group	for Channel Selection:	12	X-Mobility	y/Wirele	ess :	Гуре: 1	NONE
Supplemen	tary Service Protocol:	a	Networ	k Call 🖞	Irans	sfer? n	n

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#### 4.4. Administer ISDN-PRI Trunk Group

Use the **add trunk-group** *n* command, where *n* is the number of an unused trunk group, to be added. Set the fields in bold to the values shown below. The **Group Name** can be any descriptive name. The **TAC** (Trunk Access Code) must be chosen to be consistent with the existing dial plan.

add trunk-group 12 Page 1 of 22 SPE B TRUNK GROUP Group Type: isdn Group Number: 12 CDR Reports: y COR: 1 TN: 1 TAC: 112 Outgoing Display? n Carrier Medium: PRI/BRI Busy Threshold: 255 Night Service: Group Name: GW PRI Direction: two-way Dial Access? n Queue Length: 0 Service Type: tie Auth Code? n TestCall ITC: rest Far End Test Line No: TestCall BCC: 4 TRUNK PARAMETERS Codeset to Send Display: 6 Codeset to Send National IEs: 6 Max Message Size to Send: 260 Charge Advice: none Supplementary Service Protocol: a Digit Handling (in/out): enbloc/enbloc Trunk Hunt: cyclical QSIG Value-Added? n Digital Loss Group: 13 Calling Number - Delete: Insert: Numbering Format: Bit Rate: 1200 Synchronization: async Duplex: full Disconnect Supervision - In? y Out? n Answer Supervision Timeout: 0

On Page 2, set the fields in bold to the values shown below.

add trunk-group 12 Page 2 of 22 SPE B TRUNK FEATURES ACA Assignment? n Measured: internal Wideband Support? n Internal Alert? n Maintenance Tests? y Data Restriction? n NCA-TSC Trunk Member: Send Name: y Send Calling Number: y Used for DCS? n Suppress # Outpulsing? n Numbering Format: public Outgoing Channel ID Encoding: preferred UUI IE Treatment: service-provider Replace Restricted Numbers? n Replace Unavailable Numbers? n Send Connected Number: n Network Call Redirection: none Send UUI IE? y Send UCID? n Send Codeset 6/7 LAI IE? y Ds1 Echo Cancellation? n US NI Delayed Calling Name Update? n SBS? n Network (Japan) Needs Connect Before Disconnect? n

Solution & Interoperability Test Lab Application Notes ©2009 Avaya Inc. All Rights Reserved. On Page 3, define the incoming call handling treatment for calls coming from the Mediant 1000 on this trunk. The entry in bold below specifies that all incoming calls of *11* digits in length (e.g., 17325551234) will have \*9 inserted at the beginning of the dial string. \*9 is the feature access code for Automatic Route Selection (ARS) on the Avaya DEFINITY Server R for routing calls out to PSTN.

add trunk-group	b 12			Page	3 03	= 2	22	SPE B
		INCOMING CALL	HANDLIN	G TREATMENT				
Service/	Called	Called	Del	Insert	Pe	er (	Call	Night
Feature	Len	Number			CI	PN/E	BN	Serv
tie	11			*9				

On Page 6, enter the trunk group members. For each DS1 port to be added as a member of the trunk group, enter the port number in the **Port** field and the corresponding signaling group for that port in the **Sig Grp** field. The **Code** field is filled in automatically. In the compliance test, each of the 23 bearer channels of the DS1 board added in **Section 4.2** were added to this group. Only the first 15 members are shown below. The signaling group for each of these ports is the signaling group added in Section **4.3**.

add trunk-grou	up 12	TRUNK GROUP	Page	6 of	22	SPE B
GROUP MEMBER 2	ASSIGNMENTS	Administered M Total Admi			'	1/23 23
Port 1: 01C0701 2: 01C0702 3: 01C0703 4: 01C0704 5: 01C0705 6: 01C0706 7: 01C0707 8: 01C0708 9: 01C0709 10: 01C0710 11: 01C0711 12: 01C0712 13: 01C0713 14: 01C0714 15: 01C0715	Code         Sfx         Name           TN464         F           TN464         F	Night	Sig Grp 12 12 12 12 12 12 12 12 12 12 12 12 12			

#### 4.5. Associate ISDN-PRI trunk group with ISDN-PRI signaling group

Use the **change signaling-group 12** command to return to the **Signaling Group** form shown in **Section 4.3**. Set the **Trunk Group for Channel Selection** field to the number of the trunk group created in **Section 4.4**.

change signaling-group 12		Page 1 of 5 SPE B
SIGNALING	GROUP	
Group Number: 12 Group Type:	isdn-pri	i
Associated Signaling?	У	Max number of NCA TSC: 0
Primary D-Channel:	01C0724	Max number of CA TSC: 0
		Trunk Group for NCA TSC:
Trunk Group for Channel Selection:	12	X-Mobility/Wireless Type: NONE
Supplementary Service Protocol:	a	Network Call Transfer? n

#### 4.6. Configure Inbound and Outbound Routing for ISDN-PRI trunks

The compliance testing used Automatic Alternate Routing (AAR) to define route pattern used for routing calls to access Avaya Voice Portal. The AAR table as shown below specifies that calls to *2122960* (as used in the compliance test) containing exactly 7 digits will use **Route Pattern** *12* for routing. Note that the Avaya DEFINITY Server R has already processed the called number from the PSTN to reduce the called digits to 7 and designated the call for AAR. This standard incoming-call treatment is beyond the scope of these application notes and therefore not included here.

display aar analysis 21	22960				P	Page 1 of 2 SPE B
	A	AR DI	GIT ANALYS	SIS TABI	LE	
						Percent Full: 7
Dialed	Tot	al	Route	Call	Node	ANI
String	Min	Max	Pattern	Туре	Num	Reqd
2122960	7	7	12	aar		n
2122961	7	7	14	aar		n
2122962	7	7	14	aar		n
2122963	7	7	14	aar		n
2122964	7	7	14	aar		n
2122965	7	7	22	aar		n
2122967	7	7	10	aar		n
2122968	7	7	10	aar		n
2122969	7	7	10	aar		n
2123358	7	7	10	aar		n
2123359	7	7	10	aar		n
2123360	7	7	10	aar		n
2123361	7	7	10	aar		n
2123362	7	7	10	aar		n

The example below shows the route pattern used in the compliance test for inbound calls to access Avaya Voice Portal (via the Mediant 1000). The **Pattern Name** can be any descriptive name. The **Grp No**. is set to the trunk-group number for the trunk to be used for routing calls to. In this case trunk group *12* is the ISDN-PRI trunk group already configured on the Avaya DEFINITY Server R to connect to the Mediant 1000 (see Section 4.4). The **FRL** field defines the facilty restriction level for this route pattern. The value of  $\theta$  is the least restrictive. The **No**. **Del Dgts** field is set to *3* and the *Inserted Digits* field is set to *2*. With 3 digits deleted and "2" prefixed to the called number 2122960 (see above screen) to access Avaya Voice Portal, the final

access number becomes 22960. This final access number will be used in the Avaya Voice Portal configuration (see Section 5.6). The Default values for all other fields can be retained.

change route-pattern 12 Page 1 of 3 SPE B Pattern Number: 12 Pattern Name: To Mediant100 Grp FRL NPA Pfx Hop Toll No. Inserted DCS/ IXC No Mrk Lmt List Del Digits QSIG Dgts Intw 1:12 0 3 2 n user 2: n user 3: n user 4: user n 5: n user 6: n user BCC VALUE TSC CA-TSC ITC BCIE Service/Feature BAND No. Numbering LAR 0 1 2 3 4 W Request Dgts Format Subaddress bothept rest 1: yyyyyn n none 2: yyyyyn n none 2: y y y y y n n 3: y y y y y n n 4: y y y y y n n rest none rest none 5: y y y y y n n 6: y y y y y n n rest none rest none

The compliance testing used Automatic Route Selection (ARS) to define route pattern 2 as the route for all outbound calls to the PSTN from the Avaya DEFINITY Server R. The ARS table as shown below specifies that destination numbers starting with *1* and containing exactly *11* digits will use route pattern *2* for routing. The **Call Type** *fnpa* means this is 10-digit North American Numbering Plan (NANP) call (11 digits with Prefix Digit "1").

display ars analysis O					P	Page 1 of 2 SPE B
	A	RS DI	GIT ANALYS	SIS TABI	LE	
			Location:	all		Percent Full: 7
Dialed	Tot	al	Route	Call	Node	ANI
String	Min	Max	Pattern	Type	Num	Reqd
0	1	1	deny	op		n
0	8	8	deny	op		n
0	11	11	deny	op		n
00	2	2	deny	op		n
01	9	17	deny	iop		n
011	10	18	2	intl		n
01133	5	28	2	intl		n
1	11	11	2	fnpa		n
10xxx0	6	6	deny	op		n
10xxx0	16	16	deny	op		n
10xxx01	14	22	deny	iop		n
10xxx1	16	16	deny	fnpa		n
120	11	11	2	fnpa		n
1200	11	11	2	fnpa		n

Solution & Interoperability Test Lab Application Notes ©2009 Avaya Inc. All Rights Reserved. The example below shows the route pattern used in the compliance test for outbound calls to the PSTN. The **Pattern Name** can be any descriptive name. The **Grp No**. is set to the trunk-group number for the trunk to be used. In this case trunk group I is the trunk group already configured on the Avaya DEFINITY Server R to connect to the PSTN (its configuration is standard and therefore not included in these application notes). The **FRL** field defines the facilty restriction level for this route pattern. The value of  $\theta$  is the least restrictive. The **Pfx Mrk** field is set to I. The Prefix Mark sets the requirement for sending a prefix digit 1. Setting the **Pfx Mrk** field to a 1 results in a 1 being prefixed to any 10-digit number. An 11-digit number, presumably already with a 1, is left unchanged. Default values for all other fields can be used.

cha	nge :	route	e-pat	terr	n 2					Page	e 1	of	3	SPE B	
					Pattern N	Jumber	·· 2	Pattern N	lame•	G3R1-T	nHse	Sw			
					raccern r	vanio e i	• •	raccorn n	anic .	00111 1		0 11			
	Grp	FRL	NPA	Pfx	Hop Toll	No.	Inser	ted					DCS	/ IXC	
	No			Mrk	Lmt List	Del	Digit	s					QSI	G	
						Dqts	2						Int		
1	-	•				Dyts									
1:	1	0		1									n	user	
2:													n	user	
3:													n	user	
4:													n	user	
5:													n	user	
6:													n	user	
	BC	C VAI	LUE	TSC	CA-TSC	ITC	BCIE	Service/Fe	ature	e BAND	No.	Numb	ering	LAR	
			4 W		Request							Form	-		
											addr				
1										Suc	auur	200			
⊥:	УУ	УУ	y n	n		both	lept							none	
2:	УУ	У У	y n	n		rest	-							none	
3:	УУ	У У	y n	n		rest	-							none	
4:	У У	УУ	y n	n		rest	:							none	
5:	УУ	УУ	y n	n		rest	5							none	
6:	УУ	УУ	y n	n		rest	5							none	

# 5. Configure Avaya Voice Portal

This section covers the administration of Avaya Voice Portal. Avaya 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

Avaya 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.

Address 🙆 https://192.45.122.1	130/VoicePortal/Faces/main.jsf	🔽 🔁 Go
Αναγα	Wel Last logged in today at 9	come, admi :11:12 AM ED
Voice Portal 5.0 (VoicePor	tal) 👫 Home 🖓 Help	😆 Logoff
Expand All   Collapse All • User Management	You are here: Home	
Roles Users	Voice Portal Management System Version 5.0.0.0.4602	
Login Options <b>Real-Time Monitoring</b> System Monitor Active Calls Port Distribution <b>System Maintenance</b>	Voice Portal Management System (VPMS) is the consolidated web-based application for administer Portal. Through the VPMS interface, you can configure Voice Portal, check the status of a Voice Po component, and generate reports related to system operation.	
Audit Log Viewer	Legal Notice	
Trace Viewer Log Viewer Alarm Manager • System 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	<ul> <li>Q 2005 - 2009 Avaya Inc. All Rights Reserved.</li> <li>Notice</li> <li>While reasonable efforts were made to ensure that the information in this document was complete and accurate at the time of printing, Avaya Inc. can assume no liability for any errors. Changes and corrections to the information in this document might be incorporated in future releases.</li> <li>Documentation disclaimer Avaya Inc. is not responsible for any modifications, additions, or deletions to the original published version of this documentation unless such modifications, additions, or deletions were performed by Avaya. Customer and/or End User agree to indemnify and hold harmless Avaya, Avaya's agents, servants and employees against all claims,</li> </ul>	
Standard Custom Scheduled	Last Login: 6/30/09 9:11:12 AM EDT	

**Note:** All of the screens in this section are shown after Avaya Voice Portal had been configured. In addition, the navigation sequence to each screen is displayed at the top of each screen.

## 5.1. Install Certificate for TLS Authentication

In the compliance test, Avaya Voice Portal was configured to use TCP on 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 Avaya Voice Portal and Mediant 1000. To install the certificate for TLS authentication, navigate to **Security**  $\rightarrow$  **Certificates** and select the **Root Certificate** tab. Specify the directory path where the certificate is located and the password, and then click **Install**. The screen below shows a certificate that has already been installed.

🗿 Avaya Voice Portal Managem	ent System - Microsoft Inte	ernet Explorer			×
File Edit View Favorites To	ols Help				1
Address 🕘 https://192.45.122.130/	/oicePortal/faces/main.jsf			💌 🛃 G	io
AVAYA				Welcome, adm Last logged in today at 9:11:12 AM E	10-00-00
Voice Portal 5.0 (VoicePortal	)			📅 Home 📪 Help 🛽 Logoff	F
Expand All   Collapse All	You are here: <u>Home</u> > Sec	urity > Certificates			
▼ User Management Roles Users	Certificates				
Login Options <b>Real-Time Monitoring</b> System Monitor Active Calls Port Distribution	This page displays the Vo	ice Portal root certif	cate and application certifica	ate that are currently in effect.	
<ul> <li>System Maintenance Audit Log Viewer</li> </ul>	Root Certificate Appli	ication Certificates	Speech Server Certificates		ñ
Trace Viewer Log Viewer Alarm Manager ▼ System Management	Security Certificate			Export	
MPP Manager Software Upgrade System Backup <b>System Configuration</b> Alarm Codes Alarm/Log Options Applications MPP Servers Report Data SNMP Speech Servers VoIP Connections VPMS Servers	Certificate fingerpri MD5: fb:ce:o	IP CÀ,0=Avaya fcc 13 10:44:28 KDT 2 ints dd:94:f6:90:e4:93	009 until: Tue Apr 13 1 :0f:3a:4e:4e:a0:02:30:b :c5:66:11:e7:1b:79:ae:5	56	
<ul> <li>✓ Security Certificates Licensing</li> <li>✓ Reports</li> <li>Standard</li> <li>Custom</li> <li>Scheduled</li> </ul>	Install New Security Ce Enter Security Certificate Password: Install Cancel			Browse	

## **5.2. Configure SIP Connection**

To configure a SIP connection to the AudioCodes Mediant 1000 VoIP Media Gateway, navigate to **System Configuration**  $\rightarrow$  **VoIP Connections**, click on the **SIP** tab. The SIP tab is displayed as shown below. 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 Listener Port fields to 5060 for TCP
- Specify the IP address assigned to Mediant 1000 for 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 Voice Portal Manageme	ent System - Microsoft Internet Explorer
File Edit View Favorites Too	is Help 🥂
Address 🙆 https://192.45.122.130/V	lojcePortal/faces/main.jsf 💽 🔂 Go
· · · · · · · · · · · · · · · · · · ·	loicePortal/faces/main.jsf
MPP Servers Report Data SNMP Speech Servers VOIP Connections VPMS Servers <b>Security</b> Certificates Licensing <b>Reports</b> Standard Custom Scheduled	Listener Port: 5060 SIP Domain: 192.45.108.32 P-Asserted-Identity: Call Capacity Maximum Simultaneous Calls: 20 O All Calls can be either inbound or outbound O 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  $\rightarrow$  MPP Servers. 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 MPP server used in the compliance test. Only one MPP server was used in the compliance test. Repeat these steps to configure additional MPP servers if necessary.

🗿 Avaya Voice Portal Managem	ent System - Microsoft Internet	Explorer 📃 🗖 🛛
File Edit View Favorites To	ools Help	N1
Address 🕘 https://192.45.122.130/	/VoicePortal/faces/main.jsf	💌 🔁 Go
AVAYA		Welcome, admin Last logged in today at 9:11:12 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
<ul> <li>User Management Roles Users</li> <li>Login Options</li> <li>Real-Time Monitoring System Monitor</li> <li>Active Calls Port Distribution</li> <li>System Maintenance Audit Log Viewer Trace Viewer Log Viewer Alarm Manager</li> <li>System Mantenance Audit Log Viewer Alarm Manager</li> <li>System Management MPP Manager Software Upgrade System Backup</li> <li>System Configuration Alarm Codes Alarm/Log Options Applications MPP Servers Report Data SNMP Speech Servers VoIP Connections VPMS Servers</li> <li>Secutity Certificates Licensing</li> <li>Reports Standard Custom Scheduled</li> </ul>	Change MPP Server Use this page to change the co Thresholds. Do not set Trace Li might experience performance you are troubleshooting the sy: Name: Host Address: Network Address (VoIP): Network Address (MRCP): Network Address (MRCP): Network Address (AppSvr): Maximum Simultaneous Calls: Restart Automatically: MPP Certificate Owner: CN=mpp1,0=Avaya,00 Serial Number: aefa96a715 Valid from: Mon Apr 13 10 Certificate fingerprints	r nfiguration of an MPP. Take care when changing the MPP Trace Logging evels to Finest if your Voice Portal system has heavy call traffic. The system issues if Trace Levels are set to Finest. Set Trace Levels to Finest only when stem. mpp1 192.45.122.131 <default> <default> 20 ○ Yes ● No =MPP U=MPP</default></default>
	Categories and Trace Level	

## 5.4. Configure VoIP Audio Format

The VoIP Audio Format for MPP servers is configured in the VoIP Settings screen accessible from System Configuration  $\rightarrow$  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 Voice Portal Managem	ient System - Microsoft Internet Explorer	
File Edit View Favorites To	ools Help	<b></b>
Address 🗃 https://192.45.122.130/	(VoicePortal/faces/main.jsf	💌 🔁 Go
AVAYA		Welcome, admin Last logged in today at 9:11:12 AM EDT
Voice Portal 5.0 (VoicePortal	0	🛱 Home 📪 Help 🛽 Logoff
Voice Portal S.D (VoicePortal Expand All   Collapse All Verser Management Roles Users Login Options Verser Monitoring System Monitoring System Monitoring System Monitoring Collabor System Manager Addit Log Viewer Trace Viewer Log Viewer Alarm Manager Software Upgrade System Manager Software Upgrade System Manager Software Upgrade System Configuration Alarm Codes Alarm/Log Options Applications MPP Servers Report Data SNMP Speech Servers VoIP Connections VPMS Servers Standard Custom Scheduled	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 netwo protocols such as H.323 and Real-time Transfer Protocol (RTP). Use this page to configur data is transferred through the network. Note that if you make any changes to this page. Port Ranges UDP: 2000 30999 TCP: 31000 31999 MRCP: 32000 32999 H.323 35000 50000 RTCP Monitor Settings Host Address: Port: VOIP Audio Formats MPP Native Format: GoS Parameters	k who using one or more standard re parameters that affect how voice
	VUAN       Diffserv         H.323:       6         SIP:       6         46         RTSP:       6         46         Out of Service Threshold (% of VoIP Resources)         Trigger       Reset         Warn:       10         10       0         Error:       20         20       10         Fatal:       70         Sove       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 Avaya Voice Portal; this configuration is not directly related to achieving interoperability between AudioCodes Mediant 1000 VoIP Media Gateway and Avaya Voice Portal. It is included here for completeness.

To configure the ASR server, navigate to **System Configuration**  $\rightarrow$  **Speech Servers**, select the **ASR** tab, and then click **Add**. The screen below shows the configuration for the ASR server used in the compliance test. Set the **Engine Type** to the appropriate value. In the test 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.

🗿 Avaya Voice Portal Manage	ement System - Microsoft Internet Explore	ir	
File Edit View Favorites	Tools Help		A.
Address 🙆 https://192.45.122.1	30/VoicePortal/faces/main.jsf		💌 🄁 Go
AVAYA			Welcome, admin Last logged in today at 9:11:12 AM EDT
Voice Portal 5.0 (VoicePort	tal)		ft Home 📪 Help 📊 😡 Logoff
Expand All   Collapse All	You are have Hame > Susters Configure	ation > <u>Speech Servers</u> > Change ASR Server	
▼ User Management Roles Users Login Options	Change ASR Server Use this page to change the configuration		
<ul> <li>Real-Time Monitoring System Monitor</li> </ul>			
Active Calls	Name:	Nuance ASR	
Port Distribution  System Maintenance Audit Log Viewer	Enable:	⊙ Yes ○ No	
Trace Viewer	Engine Type:	Nuance 👻	
Log Viewer Alarm Manager	Network Address:	192.45.121.202	]
<ul> <li>System Management MPP Manager</li> </ul>	Base Port:	4900	
Software Upgrade System Backup	Total Number of Licensed ASR Resourc	es: 4	
▼ System Configuration	New Connection per Session:	🔿 Yes 💿 No	
Alarm Codes Alarm/Log Options Applications MPP Servers Report Data SNMP Speech Servers VoIP Connections VPMS Servers <b>VoIP Servers</b> <b>Vertificates</b> Licensing	Languages:	Dutch(Netherlands) nI-nI English(Australia) en-au English(UK) en-in English(India) en-in English(Singapore) en-SG English(USA) en-us	
▼ Reports	Ping Interval: 15 second(s)		
Standard Custom	Response Timeout: 4 second(s)		
Scheduled	Protocol: MRCP V1 V		
	RTSP URL: 192.45.121.202/media/spec	echrecognize	
	Save Apply Cancel Help	0	

To configure the TTS server, navigate to System Configuration  $\rightarrow$  Speech Servers, select the TTS tab, and then click Add. The screen below shows the configuration for the TTS server used in the compliance test. 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 Voice Portal Managem	nent System - Microsoft Internet Explore	r	
File Edit View Favorites To	ools Help		<u>A</u> *
Address 🗃 https://192.45.122.130	/VoicePortal/faces/main.jsf		💌 🄁 Go
AVAYA	La La		Welcome, admin Last logged in today at 9:11:12 AM EDT
Voice Portal 5.0 (VoicePorta	1)		📅 Home 📪 Help 😆 Logoff
Expand All   Collapse All	You see have Users a Custom Conference	ation > <u>Speech Servers</u> > Change TTS Server	
▼ User Management Roles Users Login Options ▼ Real-Time Monitoring	Change TTS Server		2
System Monitor Active Calls	Name:	Nuance TTS	
Port Distribution  System Maintenance	Enable:	⊙ Yes ○ No	
Audit Log Viewer Trace Viewer	Engine Type:	Nuance 👻	
Log Viewer Alarm Manager	Network Address:	192.45.121.202	3. U
<ul> <li>System Management MPP Manager</li> </ul>	Base Port:	4900	
Software Upgrade System Backup	Total Number of Licensed TTS Resource	s: 4	
<ul> <li>System Configuration Alarm Codes</li> </ul>	New Connection per Session:	🔘 Yes 💿 No	
Alarm/Log Options Applications MPP Servers Report Data SIMP Speech Servers VoIP Connections VPMS Servers <b>VERS</b> Servers <b>Escurity</b> Certificates Licensing	Voices: MRCP	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	
▼ Reports	Ping Interval: 15 second(s)		
Standard Custom	Response Timeout: 4 second(s)		
Scheduled			
	Protocol: MRCP V1 V		
	RTSP URL: 192.45.121.202/media/spee	echsynthesiz	
	Save Apply Cancel Help	1	

## 5.6. Add Voice Application

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

Navigate to System Configuration  $\rightarrow$  Applications to add a Voice Portal application. 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 called number that invokes the application (22960 as configured in Section 4.6). The configuration for the voice application used in the compliance test is shown in the screen below.

🗿 Avaya Voice Portal Managen	nent System - Microsoft Internet Explorer	
File Edit View Favorites To	ools Help	A¥
Address 🙋 https://192.45.122.130	/VoicePortal/faces/main.jsf	🔽 🎦 Go
AVAYA	Last	Welcome, admin t logged in today at 9:11:12 AM EDT
Voice Portal 5.0 (VoicePorta	1) (1) (1) (1) (1) (1) (1) (1) (1) (1) (	📅 Home 🛛 🤉 Help 😆 Logoff
Voice Portal S.U (VoicePorta Expand All   Collapse All	You are here: Home > System Configuration > Applications > Change Application   Change Application   Use this page to change the configuration of a VoiceXML or CCXML application.   Name: Intro   Enable: • Yes • No   MIME Type: VoiceXML •   VoiceXML URL: http://192.45.60.61/avptestapp/dtmf/intro.vxml   Speech Servers   ASR: • Nuance •   English(USA) en-us   Voices: • English(USA) en-US Jennifer F   Languages: • English(USA) en-us   Voices: • English(USA) en-US JIF   Application Launch   Type: • Inbound • Inbound Default • Outbound   • Number • Number Range • URI   Called Number: • Add   \$Speech Parameters >	Verify
	Advanced Parameters > Save Apply Cancel Help	×

#### 5.7. Start MPP Server

Start the MPP server from System Management  $\rightarrow$  MPP Manager as shown below. Select the MPP for use and then click the Start button (the compliance test used only one MPP server; the other one shown in the screen was used for other purposes). The Mode of the started MPP should be *Online* and the State should be *Running*.

🚰 Avaya Voice Portal Managem	ent System - Microsof	t Internet Ex	plorer						
File Edit View Favorites To	ools Help	R							<b></b>
Address 🙆 https://192.45.122.130/	/VoicePortal/faces/main.jsf								💌 🄁 Go
AVAYA						Las	st logged in to		ome, admin 11:12 AM EDT
Voice Portal 5.0 (VoicePortal	)						n Home	?- Help	😢 Logoff
Expand All   Collapse All	You are here: <u>Home</u>	> System Ma	nagement	> MPP Mai	nager				
<ul> <li>✓ User Management Roles Users Login Options</li> <li>✓ Real-Time Monitoring System Monitor Active Calls Port Distribution</li> </ul>	MPP Manage This page displays th commands, select or stopped.	ne current sta	te of each	MPP in the	e Voice Porta				
▼ System Maintenance Audit Log Viewer Trace Viewer						Last F	Poll: 6/30/09	11:13:40	AM EDT
Log Viewer Alarm Manager ▼ <b>System Management</b>	Server Nam	ie Mode	State	Config	Auto Restart	<b>Restart</b> Today	Schedule Recurring	Active In	Calls Out
MPP Manager Software Upgrade	mpp1	Online	Running	ок	No 🥒	No 🥒	None 🥓	0	0
System Backup  System Configuration	mpp2	Online	Running	ок	No 🥒	No 🥒	None 🖋	0	0
Alarm Codes Alarm/Log Options Applications	State Commands				A	lestart/Re	boot Option	15	
MPP Servers Report Data SNMP Speech Servers	Start Stop I	Restort   f	leboot l	-lalt Ca	incei		rver at a time cted servers -		ne time
VoIP Connections VPMS Servers • Security Certificates Licensing • Reports Standard Custom Scheduled	Mode Commands Offline Tes	t Online	[						

## 6. Configure Mediant 1000

This section provides the procedures for configuring the AudioCodes Mediant 1000 VoIP Media Gateway to interoperate with Avaya Voice Portal. It is assumed that the Mediant 1000 has been properly installed with the initial configuration following the 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 Mediant 1000 is performed via a Web browser. To access the device, enter the IP address of the Mediant 1000 as access URL, then log in with the proper credentials. The main Mediant 1000 screen after login is shown below.

AudioCodes	nt 1000 Subinit i Burn Device Actions	🔹 💼 Home 🔞	Help 🔶 Log off
onfiguration Management Status & Diagnostics Scenarios Search	Mediant 1000 Home Page		
Basic Full C Network Settings Media Settings PSTN Settings Protocol Configuration Advanced Applications	1     Digital     2     BRI     3       1     1     1     1     1       1     1     1	FX0 CPU	□ 
	General Information IP Address 192,45108.32	Trunk (Digital Modules) Disable	Channel (Analog Modules
	Subnet Mask 255.255.0	Active - OK	Inactive W
	Default Gateway Address 192,45.108.1	RAI Alarm	Handset Offhook
	Digital Port Number 1	LOS / LOF Alarm	Call Connected
	BRI Port Number 4	AIS Alarm	
	Analog Port Number 8	D-Channel Alarm	
	Firmware Version 5.60A.024.003		
	Protocol Type SIP		

## 6.1. Network IP settings

The network settings that were configured during installation can be viewed by navigating to **Network Settings**  $\rightarrow$  **IP Settings** in the right 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 the compliance test, the **IP Address**, **Subnet Mask** and **Default Gateway Address** were set to values consistent with the test configuration shown in **Figure 1**. Default values may be retained for all other fields.

AudioCodes       Mediant 1000       Submit       Device Actions       More       Here       Here       Here       Here       Here       L         Configuration       Management       Status & Diagnostics         Scenarios       Search         Pasic       Full         P Settings       IP Settings         IP Settings       IP Address       192.45.108.32         Submit       Submit Mask       255.255.255.0         Default Gateway Address       192.45.108.1         Protocol Configuration       Multiple Interface Settings         Protocol Configurations       Multiple Interface Settings         VLAN Mode       Image: Native VLAN ID         VLAN Mode       Image: Native VLAN ID         VLAN ID       1         Oxdard VLAN ID       2         Media VLAN ID       3	ress 🐻 http://192.45.108.32/Index			R.	💌 🄁 G
Scinarios Search     Basic Full     IP Settings     IP Address     IP Address     IP Settings     IP Address     IP Address <t< th=""><th>AudioCodes Mediant 100</th><th>0 🐼 Submit 🙆 Burn 🛛 Devic</th><th>e Actions 🔹 💼 Home</th><th>() Help</th><th>🖢 Log</th></t<>	AudioCodes Mediant 100	0 🐼 Submit 🙆 Burn 🛛 Devic	e Actions 🔹 💼 Home	() Help	🖢 Log
Basic Full     IP Settings   IP Settings   IP Settings   IP Settings   IP Address		IP Settings			
Basic Full   IP Settings   IP Settings   IP Settings IP Address   IP Address <tr< td=""><td>Scenarios</td><td>IP Settings</td><td></td><td></td><td></td></tr<>	Scenarios	IP Settings			
IP settings   Application Settings   PSTN Settings   Protocol Configuration   Multiple Interface Settings   Multiple Interface Settings   Multiple Interface Settings   Multiple Interface Settings   VLAN Mode   VLAN ID Settings   Native VLAN ID   1   OAM VLAN ID   3	Basic 🔾 Full		Single IP Network	*	
IP Settings   Application Settings   PSTN Settings   Protocol Configuration   Multiple Interface Settings   Multiple Interface Table   VLAN Mode   VLAN Mode   VLAN ID Settings   Native VLAN ID   Native VLAN ID   Control VLAN ID   Media VLAN ID   Media VLAN ID   VLAN ID   Media VLAN ID   Media VLAN ID   Media VLAN ID   Media VLAN ID   Ometric VLAN ID   Media VLAN ID   Multiple Interface	Matwork Sattings	Single IP Settings			
Application Settings Media Settings PSTN Settings Advanced Applications ULAN ID Settings VLAN ID 1 OAM VLAN ID 1 Control VLAN ID 1 Control VLAN ID 2 Media VLAN ID 3 VLAT Settings VLAT Settings VLAT Settings VLAT IP Address 0.0.0 V		IP Address	192.45.108.32		
Protocol Configuration Advanced Applications VLAN Mode VLAN Mode VLAN ID Settings Native VLAN ID 1 Control VLAN ID 1 Control VLAN ID 2 Media VLAN ID 3 VLAT Settings 0.0.0 V		Subnet Mask	255.255.255.0		
Protocol Configuration Advanced Applications VLAN Mode VLAN ID Settings Native VLAN ID 1 Control VLAN ID 1 Control VLAN ID 2 Media VLAN ID 3 • NAT Settings • NAT Settings • NAT IP Address 0.0.0 • • • • • • • • • • • • • • • • • • •		Default Gateway Address	192.45.108.1		
Advanced Applications          Multiple Interface Table         VLAN Mode         VLAN ID Settings         Native VLAN ID         OAM VLAN ID         1         Control VLAN ID         Media VLAN ID         Media VLAN ID         NAT Settings         NAT IP Address         0.0.0		Multiple Interface Settings			
✓ VLAN Mode   ✓ VLAN ID Settings   Native VLAN ID   1   OAM VLAN ID   1   Control VLAN ID   2   Media VLAN ID   3     ✓ NAT Settings   Ø NAT IP Address     0.0.0	Advanced Applications		[m]		
✓ VLAN ID Settings   Native VLAN ID   1   OAM VLAN ID   1   Control VLAN ID   2   Media VLAN ID   3     ✓ NAT Settings   Ø NAT IP Address   0.0.0					
VLAN ID Settings     Native VLAN ID     1     OAM VLAN ID     1     Control VLAN ID     2     Media VLAN ID     3     VLAN ID     VLAN ID     Address     0.0.0     VLAN			0		
Native VLAN ID 1 OAM VLAN ID 1 Control VLAN ID 2 Media VLAN ID 3 ✓ NAT Settings ✓ NAT IP Address 0.0.00 ✓		VLAN Mode	U		
OAM VLAN ID 1 Control VLAN ID 2 Media VLAN ID 3 ✓ NAT Settings ✓ NAT IP Address 0.0.0.0 ✓		✓ VLAN ID Settings			
Control VLAN ID 2 Media VLAN ID 3 VAT Settings NAT IP Address 0.0.0.0		Native VLAN ID	1		
Media VLAN ID 3 NAT Settings MAT IP Address 0.0.0.0		OAM VLAN ID	1,		
<ul> <li>▼ NAT Settings</li> <li>Ø NAT IP Address</li> <li>0.0.0.0</li> <li>✓</li> </ul>		Control VLAN ID	2		
Ø NAT IP Address 0.0.0.0		Media VLAN ID	3		
Ø NAT IP Address 0.0.0.0		▼ NAT Settings			
			0.0.00		-
Submit					
Submit					
					Submit
					Submit

### 6.2. PSTN trunk setting

Navigate to **PSTN Settings**  $\rightarrow$  **Trunk Settings** to configure the ISDN-PRI interface to Avaya DEFINITY Server R. These configuration parameters will vary based on the trunk settings provided by the far-end. For the compliance test, these parameters must be compatible with the settings used on Avaya DEFINITY Server R in Section 4. The parameters were configured as described below.

- The **Protocol Type** was set to *T1NI2ISDN*. This setting must be consistent with the settings on Avaya DEFINITY Server R through the proper selection of the **Country Protocol** and **Protocol Version** fields in the **DS1 Circuit Pack** form on the Avaya DEFINITY Server R.
- The Line Code was set to *B8ZS*. This must match the corresponding value in the DS1 Circuit Pack form on Avaya DEFINITY Server R.
- The **Framing Method** was set to *T1 Framing ESF CRC6*. This must match the corresponding value in the **DS1 Circuit Pack** form on Avaya DEFINITY Server R.

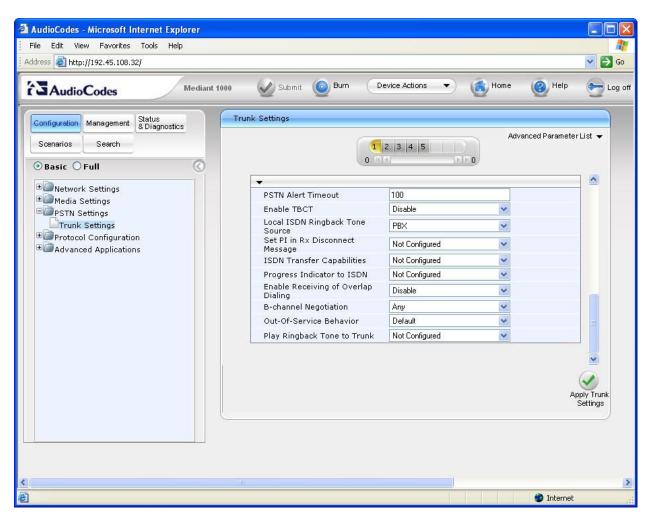
AMC; Reviewed: SPOC 8/18/2009 • The **ISDN Termination Side** was set to *User side*. This setting means the clock for the T1 trunk synchronization will be recovered from the trunk. The Avaya DEFINITY Server R side of the link was set to *line-side* (see Section 4.2).

SAudioCodes Mediant 1000 Configuration Management Status & Diagnostics	Submit 💿 Burn	De	vice Actions 🔹 🧃	Home 🙆 He	lp 😏 Li
Configuration Management Status .	runk Settings				
Scenarios Search		-		Basic Param	eter List 🔺
Basic 💿 Full					
Detwork Settings	General Settings Module ID		1		_ ^
💵 💷 Media Settings	Trunk ID		1		
PERSTN Settings	Trunk Configuration Sta	te	Inactive		
Trunk Settings	Protocol Type		T1 NI2 ISDN	~	
CAS State Machines	The second spectrum second sec			40.568	
Configuration					
Protocol Configuration	Clock Master		Recovered	*	
Democrations	Auto Clock Trunk Priorit	:y	0		=
E TDM Configuration	Line Code		B8ZS	~	
	Line Build Out Loss		0 dB	~	
	Trace Level		No Trace	*	
	Line Build Out Overwrite		OFF	*	
	Framing Method		T1 FRAMING ESF CRC6	~	
	ISDN Termination Side	User s	ide 🔽		
	Q931 Layer Response Behavior	0x0			
	Outgoing Calls Behavior	0x0			
	Incoming Calls Behavior	0x0			
	General Call Control	0x20			
	Behavior				
	NFAS Group Number	0			
	IUA Interface ID	-1			
	NFAS Interface ID	255			1000
	D-channel Configuration	PRIMA	ARY 💌		
					Apply Trunk
					Settings

Scroll down in the Trunk Settings display area to the bottom, then configure following parameters as described:

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

AMC; Reviewed: SPOC 8/18/2009 Solution & Interoperability Test Lab Application Notes ©2009 Avaya Inc. All Rights Reserved. Default values may be retained for all other fields.



After all the parameters are properly specified, click the **Apply Trunk Settings** icon button at the bottom of the screen.

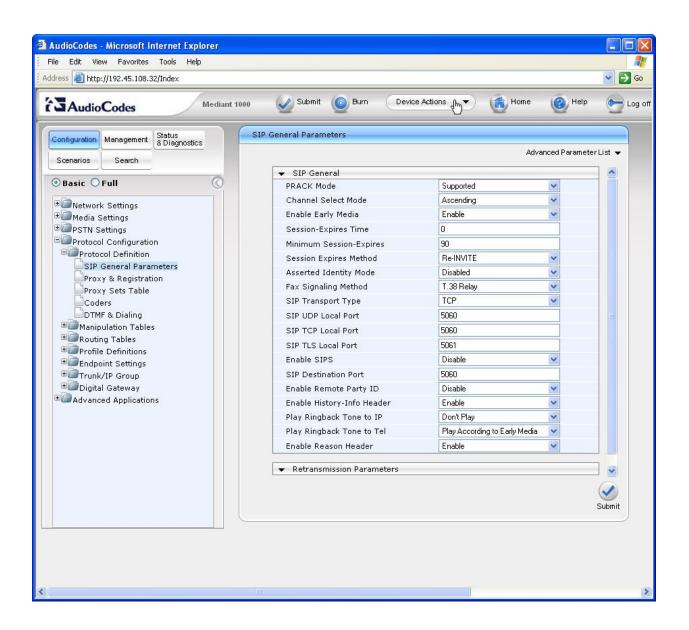
#### 6.3. SIP General Parameters

Navigate to **Protocol Configuration**  $\rightarrow$  **Protocol Definition**  $\rightarrow$  **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 number for SIP UDP Local Port (5060), SIP TCP Local Port (5060), SIP TLS Local Port (5061), SIP Destination Port (5060). Correct if necessary.

Default values may be retained for all other fields.

AMC; Reviewed: SPOC 8/18/2009

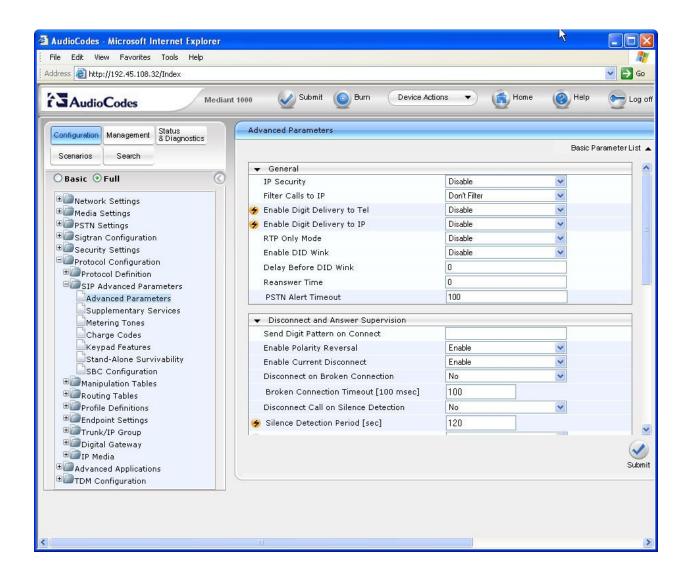


### 6.4. SIP Advanced Parameters

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

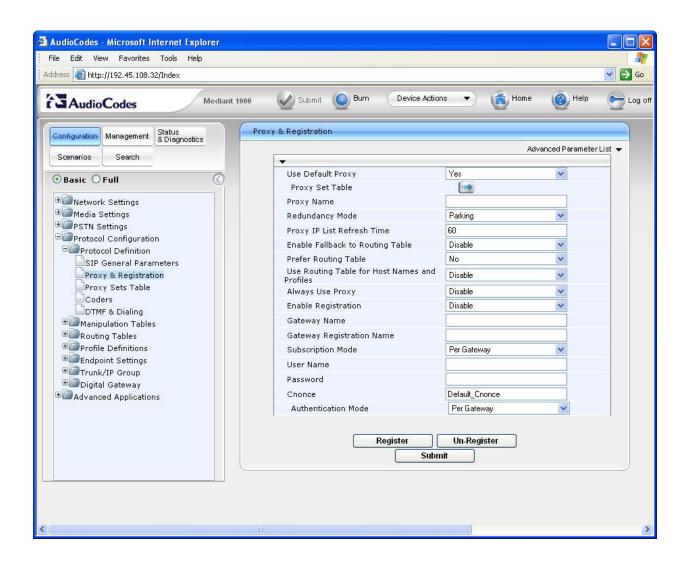
- Specify *100* (seconds) for **PSTN Alert Timeout**. This timeout setting on the gateway is for disconnecting unanswered calls on the PSTN side.
- Scroll down and set the Max Number of Active Calls field to an appropriate value (not shown).

Default values may be retained for all other fields.



## 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, then navigate to **Protocol Definition**  $\rightarrow$  **Proxy & Registration**. Select *Yes* for the **Use Default Proxy** field. Default values may be retained for all other fields.



#### 6.6. Proxy Sets Table

Click the right arrow icon button in the upper part of the **Proxy & Registration** page above to reach the Default Proxy Sets Table configuration page. Enter the IP address assigned to the Avaya Voice Portal MPP server for **Proxy Address**, and *TCP* for **Transport Type**. Default values may be retained for all other fields.

AudioCodes - Microsoft Internet Explorer File Edit View Favorites Tools Help		
Address a http://192.45.108.32/Index	<u>&gt;</u>	🖌 🄁 Go
AudioCodes Mediant 1000	ubmit 🧕 Burn Device Actions 🔻 💼 Home 🔞 Help	😁 Log off
Configuration Management Status Default Proxy	y Sets Table	
Scenarios Search     Search     Basic O Full	Proxy Set ID 0	
Network Settings     Media Settings     PSTN Settings     Protocol Configuration     SIP General Parameters     Proxy & Registration     Proxy Sets Table     Coders     DTMF & Dialing     Manipulation Tables     Torofile Definitions     Profile Definitions     Pont Settings	Proxy Address       Transport Type         1       192.45.122.131         2       Image: Comparison of the second se	
⊕©Trunk/IP Group ⊕©Digital Gateway ⊕©Advanced Applications	Proxy Load Balancing Disable Method Is Proxy Hot Swap No	Submi
		>

### 6.7. Coders

Navigate to **Protocol Configuration**  $\rightarrow$  **Protocol Definition**  $\rightarrow$  **Coders**. In the screen below, select the list of preferred codecs to be used by the Mediant 1000 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 Avaya Voice Portal in Section 5.4. The codec is selected from the pull-down menu under the **Coder Name** field.

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

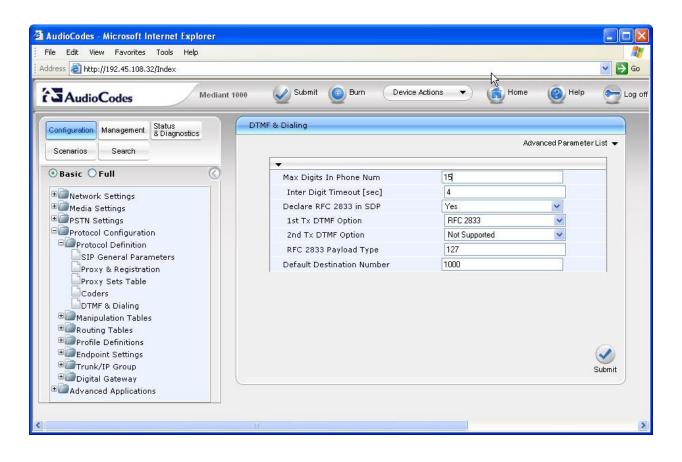
🗃 街 http://192.45.108.32/Index						~
AudioCodes Mediant 10	00 🥑 Submit 🧿 Burn	De	evice Actions	💌 🔞 Hor	ne 📀	Help 💽
iguration Management Status & Diagnostics	Coders Table					
enarios Search						
asic 🔾 Full	Coder Name	Pa	acketization Time	Rate	Payload Type	Siler Suppre
	G.711U-law	20	*	64 👻	0	Disabled
Network Settings Media Settings	G.729	✓ 20	*	8 🗸	18	Enable
PSTN Settings		~	~	~		1
Protocol Configuration		~	~			
Protocol Definition			14000	~		
SIP General Parameters		*	*	~		
Proxy Sets Table Coders DTMF & Dialing Manipulation Tables						
Routing Tables	<					
Profile Definitions     Endpoint Settings     Trunk/IP Group						Submi
Digital Gateway Advanced Applications						Gabini

### 6.8. DTMF and Dialing

Navigate to **Protocol Configuration**  $\rightarrow$  **Protocol Definition**  $\rightarrow$  **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.



### 6.9. Trunk Group

Navigate to **Protocol Configuration**  $\rightarrow$  **Trunk/IP Group**  $\rightarrow$  **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. A maximum of 24 channels can be assigned per trunk. 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. 1000 is the default value. 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 for the originating calling party if no calling 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 1000 will be used for the originating calling party number if no calling party information is received from the originating PSTN trunk.

Tru	nk Group Tabl	е							
-									
-	Add Phone Ci	ontex	t As Pref	ix		Disable	*		^
						1-12	~		
Group Index	Module		From Trunk	To Trunk	Channels	Phone Number	Trunk Group	Tel Profile ID	
1	Module 1 PRI	*	1 🗸	1	1-24	1000	1	0	1
2		~	Y	~					
3		*	~						
4		~	Y	2			43		
5		*	~	2					
6		*	Y	6					
7			Y	1					
		The second second							
		_							
12		~	X						~
	Group Index 1 2 3 4 5 6	Group Index         Module           1         Module 1 PRI           2	Group Index     Module       1     Module 1 PRI       2     ✓       3     ✓       4     ✓       5     ✓       6     ✓       7     ✓       8     ✓       9     ✓       10     ✓	Trunk Group Index       Group Index     Module     From Trunk       1     Module 1 PRI     1       2     •     •       3     •     •       4     •     •       5     •     •       6     •     •       7     •     •       8     •     •       9     •     •       10     •     •	Group Index     Module     From Trunk     To Trunk       1     Module 1 PRI     1     1       2     Image: Comparison of the second secon	Trunk Group Index         Group Index       Module       From To Trunk       Channels         1       Module 1 PRI       1       1       1-24         2             3             4             5             6             7             9             10	Trunk Group Index       1-12         Group Index       Module       From Trunk       To Trunk       Channels       Phone Number         1       Module 1 PRI       1       1       1-24       1000         2       Image: Second S	Trunk Group Index     1.12       Group Index     Module     From Trunk     To Trunk     Channels     Phone Number     Trunk Group ID       1     Module 1 PRI     1     1     1-24     1000     1       2     Image: Comparison of the second of	Trunk Group Index       1-12         Group Index       Module       From Trunk       To Trunk       Channels       Phone Number       Group Group       Tel Profile ID         1       Module 1 PRI ×       1 ×       1 ×       1-24       1000       1       0         2       ×       ×               3       ×       ×                4       ×       ×

### 6.10. Trunk Group Settings

Navigate to **Protocol Configuration**  $\rightarrow$  **Trunk/IP Group**  $\rightarrow$  **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.

		Submit 🧕 Burn	Device Actions	🕘 Help 😁 Log
ation Management Status & Diagnostics	Trunk Grou	p Settings		
ios Search				Advanced Parameter Li
c O Full	▼ R	outing Index	1-12 😵	
etwork Settings	-			
edia Settings		Trunk Group ID	Channel Select Mode	Registration Mode
GTN Settings otocol Configuration	1	1	Cyclic Ascending	Don't Register 🔽
Protocol Definition     Manipulation Tables     Profile Definitions     Endpoint Settings     Trunk/IP Group     Trunk Group     Trunk Group     Drunk Group			~	~
			~	~
			×	×
			*	~
			~	~
			¥	¥
dvanced Applications	8		¥	~
	9		~	~
	10		~	~
	11		~	~
	12		×	~

## 6.11. IP to Trunk Group Routing

Navigate to Protocol Configuration  $\rightarrow$  Routing Tables  $\rightarrow$  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.

SAudioCodes Mediant	1000 🕢 Submit	Burn     Device Actions	🔹 😭 Home 🏼 🔞	Help 🏊	Log off
Configuration Management Status & Diagnostics	Inbound IP Routin	2			
Scenarios Search				Advanced P	arameter Li
Basic O Full		▼ Routing Index IP To Tel Routing Mode Ro	1-12 💌 ute calls before manipulation 💌		
■ ■Media Settings ■ ■ PSTN Settings	Dest. Phon	e Prefix Source Phone Prefix	Source IP Address	- Trunk > Group ID	IP Profil
Protocol Configuration	1 *	*	*	1	0
Protocol Definition     Manipulation Tables	2				
Contracting Tables	3				
IP to Trunk Group Routing	4				
	5				
™@Trunk/IP Group ™@Digital Gateway	6				1
E Advanced Applications	8				
	9				
	10				
	11				
	12				
					SI

Note that the Tel to IP Routing Table was not configured for the compliance test. This is because Avaya 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 (ISDN-PRI) side will be sent to the Avaya Voice Portal MPP on the IP side.

### 6.12. Media Voice Settings

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

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st <b>▼</b>

# 7. General Test Approach and Test Results

The general test approach was to make calls from the PSTN through the Audio Codes Mediant 1000 VoIP Media Gateway to reach Avaya Voice Portal. Using Voice Portal voice prompts, various Voice Portal functions are 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 on the Avaya DEFINITY Server R.

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

The Mediant 1000 passed compliance testing. The following issue was identified in the compliance test:

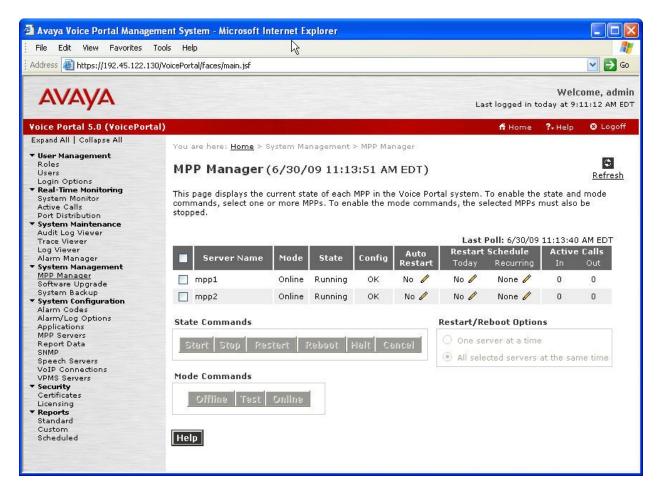
• Mediant 1000 does not forward User-to-User Information (UUI) received from Avaya Voice Portal over the ISDN interface to the far end.

The above problem is due to the Median 1000 not supporting UUI in the SIP REFER message in its current implementation. This problem is not compliance-blocking.

## 8. Verification Steps

This section provides the verification steps that may be performed to verify that a PSTN call can reach Avaya Voice Portal through 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.



- 2. Make a PSTN call to access Avaya 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

File Edit View Favorites	Tools Help				
Address 🙆 https://192.45.122.	.130/VoicePortal/faces/ma	in. jsf			🔽 🄁 Go
Αναγα			Las	Wel t logged in 6/30/09 at 10	come, admi :27:17 AM ED
loice Portal 5.0 (VoicePo	rtal)			ff Home 🛛 🔭 Help	🛚 Logoff
Expand All   Collapse All	You are here: H	lome > Real-Time Monitoring > Port Distrib	oution		
Viser Management		·			-
Roles	Port Dich	ribution (7/2/09 9:17:47 AM			\$
Users	POILDISC	IDUUUII (772709 9:17:47 AM			Refresh
Login Options					
Real-Time Monitoring	This page displ	ays information about how the telephony (	resources have b	een distributed to the MP	Ps. You
System Monitor Active Calls	configure the te	elephony resources on the VoIP Connectio	ons page.		
Port Distribution					
System Maintenance	Total Ports: 1	n	Last Poll	: 7/2/09 9:17:48 AM EDT	
Audit Log Viewer	Port \$ Mode :				
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Log Viewer	<u>1</u> Online	Connected AudioCodes M1K SIP_Trunk	mpp1		
Alarm Manager	2 Online	In service AudioCodes M1K SIP_Trunk	mpp1		
System Management MPP Manager	3 Online	In service AudioCodes M1K SIP_Trunk	mpp1		
Software Upgrade		In service AudioCodes M1K SIP Trunk	1 M (1 M		
System Backup					
System Configuration		In service AudioCodes M1K SIP_Trunk	mpp1		
Alarm Codes	<u>1</u> Online	In service AudioCodes M1K SIP_Trunk	mpp2		
Alarm/Log Options	2 Online	In service AudioCodes M1K SIP_Trunk	mpp2		
Applications MPP Servers		In service AudioCodes M1K SIP_Trunk	1		
Report Data					
SNMP		In service AudioCodes M1K SIP_Trunk			
Speech Servers	<u>5</u> Online	In service AudioCodes M1K SIP_Trunk	mpp2		
VoIP Connections					
VPMS Servers					
Security	Help				
Certificates Licensing					
Reports					
Standard					
Custom					

- 3. Verify that the Message Log (under Status & Diagnostics) in the Mediant 1000 web interface shows a SIP INVITE message with Headers containing correct information:
  - From: calling PSTN phone number with Mediant 1000's IP address
  - To: access number to Avaya Voice Portal with IP address of MPP server
  - Via: IP address of Mediant 1000

dress 🕘 http://192.45.108.32/Index	R.	<b></b>
SAudioCodes №	ediant 1000 🖉 Submit 🥥 Burn Device Actions 🔻 👘 Hor	ne 🕜 Help 🕁 Log
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⊙ Basic	6d:19h:1m:49s ( lgr_flow)(135317 )	#61:SIP
Call Status & Diagnostics	6d:19h:1m:49s ( lgr_stk_ses)(135318 ) <session #<="" td=""><td>61&gt; SendToCall - even</td></session>	61> SendToCall - even
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	<	

4. 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 passed compliance testing. These Application Notes describe the configurations required for Median 1000 to successfully interoperate with Avaya Voice Portal using SIP trunking interface. Most of the feature and serviceability test cases passed, the failed test cases did not block compliance (See Section 7 for problem identified).

## 10. Additional References

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

- 1. Administering Voice Portal, March 2009.
- 2. Avaya Voice Portal 5.0 Release Notes (Issue 2), March 2009.

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- **3.** *Mediant 1000 & Mediant 600 SIP Release Notes Ver. 5.6,* January 2009, Document #: LTRT-83104.
- **4.** *Mediant 1000 & Mediant 600 SIP User's Manual Ver. 5.6,* January 2009, Document #: LTRT-83304.

Product documentation for Avaya products can be found at <u>http://support.avaya.com</u>.

Product documentation for Mediant 1000 can be obtained from AudioCodes support web site <u>http://audiocodes.com/support</u>.

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