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](image-url)
### 3. Equipment and Software Validated

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

<table>
<thead>
<tr>
<th>Equipment</th>
<th>Software</th>
</tr>
</thead>
<tbody>
<tr>
<td>Avaya Voice Portal</td>
<td></td>
</tr>
<tr>
<td>• Voice Portal Management System (VPMS)</td>
<td></td>
</tr>
<tr>
<td>• Media Processing Platform (MPP)</td>
<td>5.0</td>
</tr>
<tr>
<td>Application Server – HTTP Server running in</td>
<td>Microsoft Windows 2003 Server Service Pack 2</td>
</tr>
<tr>
<td>Windows</td>
<td></td>
</tr>
<tr>
<td>Nuance Speech Server</td>
<td></td>
</tr>
<tr>
<td>• Nuance OpenSpeech Recognizer</td>
<td>3.0</td>
</tr>
<tr>
<td>• Nuance RealSpeak</td>
<td>4.0</td>
</tr>
<tr>
<td>Avaya DEFINITY Server R</td>
<td>R011r.03.1.635.0</td>
</tr>
<tr>
<td>Call Center Agent</td>
<td></td>
</tr>
<tr>
<td>Avaya 1600 Series IP Telephone (H.323)</td>
<td>Avaya one-X® Deskphone Value Edition Release 1.100</td>
</tr>
<tr>
<td>Analog Telephones</td>
<td>-</td>
</tr>
<tr>
<td>AudioCodes Mediant 1000 VoIP Media Gateway</td>
<td>5.60A.024.003</td>
</tr>
</tbody>
</table>
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 3rd 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.

<table>
<thead>
<tr>
<th>OPTIONAL FEATURES</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Emergency Access to Attendant?</td>
<td>y</td>
</tr>
<tr>
<td>Enable 'dadmin' Login?</td>
<td>n</td>
</tr>
<tr>
<td>Enhanced Conferencing?</td>
<td>n</td>
</tr>
<tr>
<td>Enhanced EC500?</td>
<td>n</td>
</tr>
<tr>
<td>Extended Cfg/Fxd Admin?</td>
<td>y</td>
</tr>
<tr>
<td>External Device Alarm Admin?</td>
<td>y</td>
</tr>
<tr>
<td>Five Port Networks Max Per MCC?</td>
<td>n</td>
</tr>
<tr>
<td>Flexible Billing?</td>
<td>n</td>
</tr>
<tr>
<td>Forced Entry of Account Codes?</td>
<td>n</td>
</tr>
<tr>
<td>Global Call Classification?</td>
<td>n</td>
</tr>
<tr>
<td>Hospitality (Basic)?</td>
<td>y</td>
</tr>
<tr>
<td>Hospitality (G3V3 Enhancements)?</td>
<td>y</td>
</tr>
<tr>
<td>ISDN Feature Plus?</td>
<td>y</td>
</tr>
<tr>
<td>ISDN-BRI Trunks?</td>
<td>y</td>
</tr>
<tr>
<td>ISDN-PRI?</td>
<td>y</td>
</tr>
<tr>
<td>Local Spare Processor?</td>
<td>n</td>
</tr>
<tr>
<td>Malicious Call Trace?</td>
<td>n</td>
</tr>
<tr>
<td>Media Encryption Over IP?</td>
<td>n</td>
</tr>
<tr>
<td>Mode Code for Centralized Voice Mail?</td>
<td>n</td>
</tr>
<tr>
<td>Multimedia Call Handling (Basic)?</td>
<td>y</td>
</tr>
<tr>
<td>Multimedia Call Handling (Enhanced)?</td>
<td>y</td>
</tr>
<tr>
<td>Multiple Locations?</td>
<td>n</td>
</tr>
<tr>
<td>Personal Station Access (PSA)?</td>
<td>y</td>
</tr>
<tr>
<td>Posted Messages?</td>
<td>n</td>
</tr>
</tbody>
</table>

On Page 4, verify that **Private Networking** is enabled.

<table>
<thead>
<tr>
<th>OPTIONAL FEATURES</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>PNC Duplication?</td>
<td>n</td>
</tr>
<tr>
<td>Port Network Support?</td>
<td>y</td>
</tr>
<tr>
<td>Processor and System MSP?</td>
<td>y</td>
</tr>
<tr>
<td><strong>Private Networking?</strong></td>
<td>y</td>
</tr>
<tr>
<td>Tenant Partitioning?</td>
<td>n</td>
</tr>
<tr>
<td>Terminal Trans. Init. (TTI)?</td>
<td>y</td>
</tr>
<tr>
<td>Time of Day Routing?</td>
<td>n</td>
</tr>
<tr>
<td>Uniform Dialing Plan?</td>
<td>y</td>
</tr>
<tr>
<td>Usage Allocation Enhancements?</td>
<td>y</td>
</tr>
<tr>
<td>TN2501 VAL Maximum Capacity?</td>
<td>y</td>
</tr>
<tr>
<td>Wireless?</td>
<td>n</td>
</tr>
</tbody>
</table>

Remote Office? y
Restrict Call Forward Off Net? y
Secondary Data Module? y
Station as Virtual Extension? n
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

DS1 CIRCUIT PACK

Location: 01C07
Bit Rate: 1.544
Line Compensation: 1
Signaling Mode: isdn-pri
Connect: line-side
TN-C7 Long Timers? n
Interworking Message: PROGress
Interface Companing: mulaw
Idle Code: 11111111

Name: GW DS1
Line Coding: b8zs
Framing Mode: esf
Country Protocol: 1
Protocol Version: b
DPC/Analog Bearer Capability: 3.1kHz
```

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 12

SIGNALING GROUP

Group Number: 12
Group Type: isdn-pri
Associated Signaling? y
Primary D-Channel: 01C0724
Trunk Group for Channel Selection: 12
Supplementary Service Protocol: a

Max number of NCA TSC: 0
Max number of CA TSC: 0
Trunk Group for NCA TSC:
Trunk Group for CA TSC:
X-Mobility/Wireless Type: NONE
Network Call Transfer? n
```
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

TRUNK GROUP

Group Number: 12  Group Name: GW PRI
Group Type: isdn  Carrier Medium: PRI/BRI
CDR Reports: y
Direction: two-way  TAC: 112
Outgoing Display? n
Busy Threshold: 255
Dial Access? n
Night Service:
Queue Length: 0
Service Type: tie
Far End Test Line No:
Auth Code? n
TestCall ITC: rest

TestCall BCC: 4

TRUNK PARAMETERS

Codeset to Send Display: 6
Max Message Size to Send: 260
Supplementary Service Protocol: a
Trunk Hunt: cyclical
Charge Advice: none
Digit Handling (in/out): enbloc/enbloc
 Antique Group: cyclical
QSIG Value-Added? n
Digital Loss Group: 13
Calling Number - Delete: Insert:
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

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

Send UUI IE? y
Send UCID? n
Send Codeset 6/7 LAI IE? y
UUI IE Treatment: service-provider
Replace Restricted Numbers? n
Replace Unavailable Numbers? n
Send Connected Number: n
Network Call Redirection: none
UUI NI Delayed Calling Name Update? n
US NI Delayed Calling Name Update? n
```

SBS? n
Network (Japan) Needs Connect Before Disconnect? n
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.

<table>
<thead>
<tr>
<th>Service/Feature</th>
<th>Called Len</th>
<th>Called Del</th>
<th>Insert</th>
<th>Per Call CPN/BN</th>
<th>Night Serv</th>
</tr>
</thead>
<tbody>
<tr>
<td>tie</td>
<td>11</td>
<td></td>
<td>*9</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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.

<table>
<thead>
<tr>
<th>Port</th>
<th>Code Sfx Name</th>
<th>Night</th>
<th>Sig Grp</th>
</tr>
</thead>
<tbody>
<tr>
<td>1: 01C0701</td>
<td>TN464 F</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>2: 01C0702</td>
<td>TN464 F</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>3: 01C0703</td>
<td>TN464 F</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>4: 01C0704</td>
<td>TN464 F</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>5: 01C0705</td>
<td>TN464 F</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>6: 01C0706</td>
<td>TN464 F</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>7: 01C0707</td>
<td>TN464 F</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>8: 01C0708</td>
<td>TN464 F</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>9: 01C0709</td>
<td>TN464 F</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>10: 01C0710</td>
<td>TN464 F</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>11: 01C0711</td>
<td>TN464 F</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>12: 01C0712</td>
<td>TN464 F</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>13: 01C0713</td>
<td>TN464 F</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>14: 01C0714</td>
<td>TN464 F</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>15: 01C0715</td>
<td>TN464 F</td>
<td>12</td>
<td></td>
</tr>
</tbody>
</table>

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.
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.

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 facility restriction level for this route pattern. The value of 0 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.

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").
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 facility restriction level for this route pattern. The value of *0* is the least restrictive. The **Pfx Mrk** field is set to *1*. 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.

<table>
<thead>
<tr>
<th>Grp</th>
<th>FRL</th>
<th>NPA</th>
<th>Pfx</th>
<th>Hop</th>
<th>Toll No.</th>
<th>Inserted</th>
<th>DCS/IXC</th>
</tr>
</thead>
<tbody>
<tr>
<td>1:</td>
<td>1</td>
<td>0</td>
<td>1</td>
<td>n</td>
<td>user</td>
<td>n user</td>
<td></td>
</tr>
<tr>
<td>2:</td>
<td>n</td>
<td>n</td>
<td>n</td>
<td>n</td>
<td>user</td>
<td>n user</td>
<td></td>
</tr>
<tr>
<td>3:</td>
<td>n</td>
<td>n</td>
<td>n</td>
<td>n</td>
<td>user</td>
<td>n user</td>
<td></td>
</tr>
<tr>
<td>4:</td>
<td>n</td>
<td>n</td>
<td>n</td>
<td>n</td>
<td>user</td>
<td>n user</td>
<td></td>
</tr>
<tr>
<td>5:</td>
<td>n</td>
<td>n</td>
<td>n</td>
<td>n</td>
<td>user</td>
<td>n user</td>
<td></td>
</tr>
<tr>
<td>6:</td>
<td>n</td>
<td>n</td>
<td>n</td>
<td>n</td>
<td>user</td>
<td>n user</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>BCC VALUE</th>
<th>TSC CA-TSC</th>
<th>ITC BCIE Service/Feature</th>
<th>BAND No.</th>
<th>Numbering</th>
<th>LAR Dgts Format</th>
<th>Subaddress</th>
</tr>
</thead>
<tbody>
<tr>
<td>1: yyyyy y n n</td>
<td>both</td>
<td>none</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>2: yyyyy y n n</td>
<td>rest</td>
<td>none</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>3: yyyyy y n n</td>
<td>rest</td>
<td>none</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>4: yyyyy y n n</td>
<td>rest</td>
<td>none</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>5: yyyyy y n n</td>
<td>rest</td>
<td>none</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>6: yyyyy y n n</td>
<td>rest</td>
<td>none</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
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.
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 ➔ 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.
5.2. Configure SIP Connection

To configure a SIP connection to the AudioCodes Mediant 1000 VoIP Media Gateway, navigate to **System Configuration → 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
5.3. Add MPP server

Add a Media Processing Platform (MPP) server by navigating to System Configuration ➔ 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.
5.4. Configure VoIP Audio Format

The **VoIP Audio Format** for 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.
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 → 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.

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 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.
### 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 ➔ 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.
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**.
MPP Manager (6/30/09 11:13:51 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.

<table>
<thead>
<tr>
<th>Server Name</th>
<th>Mode</th>
<th>State</th>
<th>Config</th>
<th>Auto Restart</th>
<th>Restart Schedule</th>
<th>Active Calls</th>
</tr>
</thead>
<tbody>
<tr>
<td>mpp1</td>
<td>Online</td>
<td>Running</td>
<td>OK</td>
<td>No</td>
<td>No</td>
<td>0</td>
</tr>
<tr>
<td>mpp2</td>
<td>Online</td>
<td>Running</td>
<td>OK</td>
<td>No</td>
<td>No</td>
<td>0</td>
</tr>
</tbody>
</table>

**State Commands**

- Start
- Stop
- Restart
- Reboot
- Halt
- Cancel

**Restart/Reboot Options**

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

**Mode Commands**

- Offline
- Test
- Online
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.
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 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.
6.2. PSTN trunk setting

Navigate to PSTN Settings → 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.
- 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).

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.
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 → 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 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.
6.4. SIP Advanced Parameters

Click the **Full** radio button above the navigation pane on the left, then navigate to **Protocol Configuration → 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.
6.7. Coders

Navigate to Protocol Configuration → Protocol Definition → 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.
6.8. DTMF and Dialing

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

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

Default values may be retained for all other fields.
6.9. Trunk Group

Navigate to Protocol Configuration → Trunk/IP 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. 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 party information is received from the originating PSTN trunk. In the Trunk Group ID column, enter the trunk group that will contain these channels. The default value may be used for the Tel Profile ID column.

In the example below, the table entry assigns channels 1 – 24 of trunk 1 to Trunk Group 1. A range of numbers arbitrary chosen to start at 1000 will be used for the originating calling party number if no calling party information is received from the originating PSTN trunk.
6.10. Trunk Group Settings

Navigate to Protocol Configuration → Trunk/IP 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.
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.
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 → Voice Settings. For DTMF Transport Type, select RFC2833 Relay DTMF. Default values may be retained for all other fields.
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 Mediant 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.

![VPMS web interface screenshot]

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
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
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.


Product documentation for Avaya products can be found at [http://support.avaya.com](http://support.avaya.com).

Product documentation for Mediant 1000 can be obtained from AudioCodes support web site [http://audiocodes.com/support](http://audiocodes.com/support).