Avaya Solution & Interoperability Test Lab

Application Notes for Configuring AudioCodes Mediant 1000 VoIP Media Gateway to Interoperate with Avaya Voice Portal using Line Side E1 Connectivity to Avaya Aura™ Communication Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for the AudioCodes Mediant 1000 VoIP Media Gateway to interoperate with Avaya Voice Portal (using a SIP trunking interface) and Avaya Aura™ Communication Manager (using a line side E1 interface).

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

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

1. Introduction

These Application Notes describe the procedure for configuring the AudioCodes Mediant 1000 VoIP Media Gateway to interoperate with Avaya Voice Portal (via SIP) and Avaya Aura™ Communication Manager (via line side E1). The AudioCodes Mediant 1000 VoIP Media Gateway serves as a gateway between TDM and IP networks. AudioCodes Mediant 1000 supports multiple hardware interfaces and control protocols. Capacity can be scaled upward by adding additional interface modules. During compliance testing, AudioCodes Mediant 1000 was configured as a SIP to line side E1 gateway connecting Avaya Voice Portal to a simulated PSTN network through Avaya Aura™ Communication Manager. This solution allows Avaya Voice Portal to receive calls from the PSTN and transfer calls back to the PSTN or PBX call center agent. Refer to **Figure 1** for details of the test configuration.

1.1. Interoperability Compliance Testing

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

- Basic calls from the PSTN to Avaya Voice Portal
- Call transfers by Avaya Voice Portal to the PSTN, including blind, consultative, and bridged transfers
- Call transfers by Avaya Voice Portal to a Call Center agent, including blind, consultative, and bridged transfers
- DTMF tones / RFC 2833 support
- G.711 mu-law and G.711 a-law codec support
- Line side E1 connectivity between AudioCodes Mediant 1000 and Avaya Aura™ Communication Manager
- SIP connectivity between AudioCodes Mediant 1000 and Avaya Voice Portal

The serviceability testing focused on verifying the ability of AudioCodes Mediant 1000 to recover from adverse conditions, such as disconnecting/reconnecting the IP and line side E1 cables to simulate network failures, and stopping/starting AudioCodes Mediant 1000 to simulate power outages.

1.2. Support

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

2. Reference Configuration

Figure 1 illustrates the configuration used during compliance testing. In the reference configuration, the AudioCodes Mediant 1000 VoIP Media Gateway connects to Voice Portal through a SIP trunking interface on the one side, and to a Communication Manager through a line side E1 interface on the other side. The Communication Manager in turn has an ISDN-PRI connection to a simulated PSTN.

Inbound calls from the PSTN to Voice Portal will be routed across the line side E1 connection to Mediant 1000 through Communication Manager. Mediant 1000 will then route the calls from its line side E1 interface to its SIP interface to be terminated on the Voice Portal MPP (Media Processing Platform) server. Outbound calls to PSTN (as a result of a transferring the inbound call to another PSTN user or call center agent) follow the same path in the reverse order.

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

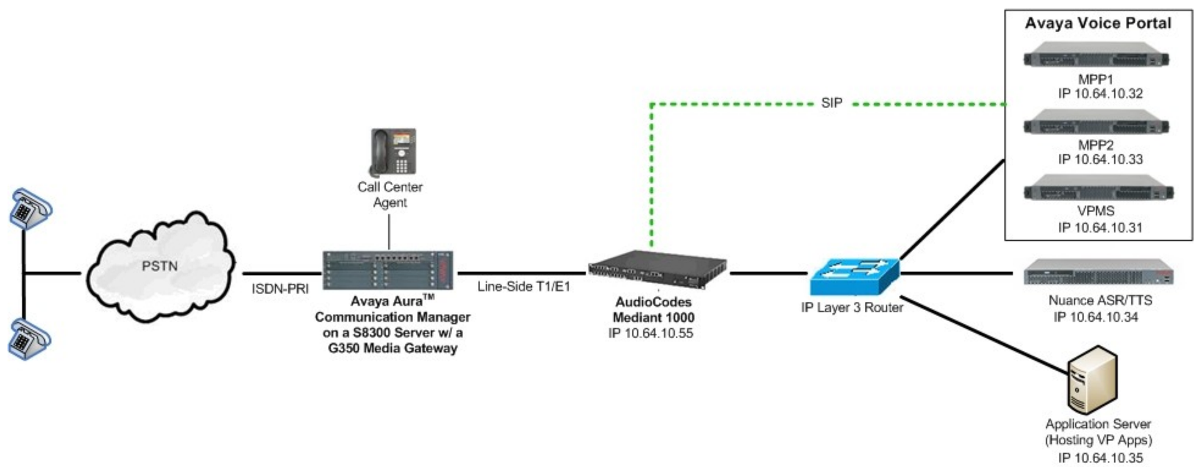


Figure 1: AudioCodes Mediant 1000 VoIP Media Gateway with Avaya Voice Portal

3. Equipment and Software Validated

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

Equipment	Software
Avaya Voice Portal • Voice Portal Management System (VPMS) • Media Processing Platform (MPP)	5.0 SP2
Application Server – HTTP Server running in Windows	Microsoft Windows 2003 Server Service Pack 2
Nuance Speech Server • Nuance OpenSpeech Recognizer • Nuance RealSpeak	 3.0 4.0
Avaya Aura™ Communication Manager - Avaya S8300 Server	5.2.1 Service Pack 2
Avaya G450 Media Gateway	-
Call Center Agent Avaya 9600 Series IP Telephone (H.323)	 3.0
Analog and Digital Telephones	-
AudioCodes Mediant 1000 VoIP Media Gateway	6.00A.009.002

4. Configure Line Side E1 on Avaya Aura™ Communication Manager

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

- Configure DS1 (to be used for line side E1 connectivity to AudioCodes)
- Configure DS1FD stations (line side ports)
- Configure VDN (to route calls to a vector)
- Configure Vector (to route calls to a hunt group)
- Configure Hunt group (to route calls to an available Agent)
- Configure Agents (mapped to line side ports)

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

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

- Line side ports are configured as DS1FD Stations (5501-5530).
- Agent LoginIDs are created and are mapped one-to-one to each DS1FD station/line side port. The Agent LoginIDs are administered to automatically log into hunt group/skill 4 (Agent LoginIDs 54101-54130).
- Inbound calls from the PSTN are routed to VDN 53500, which then invokes Vector 2.
- Vector 2 queues the call to skill 4, thus selecting an available Agent/line-side port to be used to route the call to the AudioCodes Mediant 1000 VoIP Media Gateway. AudioCodes then routes the call on to Voice Portal.

4.1. Configure DS1

Configure a DS1 board to provide E1 connectivity to the AudioCodes Mediant 1000 VoIP Media Gateway. Use the **add ds1 n** command, where **n** is a valid board number.

- Enter a descriptive **Name** (optional)
- Set **Bit Rate** to **2.048**.
- Set **Line Coding** to **hdb3**.
- Set **Signaling Mode** to **CAS**.
- Set **Interconnect** to **pbx**.
- Set **Country Protocol** to **1**.
- Set **CRC** to **y**.
- Set **Interface Companding** to **alaw**.
- Use the default values for the remaining fields.

```
add ds1 001v6                                                Page   1 of   1
                                DS1 CIRCUIT PACK

            Location: 001V6                           Name: line side E1
            Bit Rate: 2.048                    Line Coding: hdb3

      Signaling Mode: CAS

        Interconnect: pbx                 Country Protocol: 1

Interface Companding: alaw                            CRC? y
           Idle Code: 11111111




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

4.2. Configure DS1FD Stations

Use the **add station n** command, where **n** is a valid extension, to configure each line side port as a station with the **Type** field set to **DS1FD**. Repeat this configuration for each port. During compliance testing, 30 ports were configured with an extension range of 5501 to 5530.
For each station created:

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

Station 5510 is shown as an example below.

```
add station 5510                                                Page   1 of   4
                                     STATION

Extension: 5510                         Lock Messages? n               BCC: 0
     Type: DS1FD                        Security Code:                  TN: 1
     Port: 001V610                    Coverage Path 1:                 COR: 1
     Name: line-side port 10          Coverage Path 2:                 COS: 1
                                      Hunt-to Station:               Tests? y
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 4
    Off Premises Station? y
       R Balance Network? n


          Survivable COR: internal
   Survivable Trunk Dest? y
```

4.3. Configure VDN

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

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

```
add vdn 53500                                                   Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 53500
                                 Name*: Voice Portal
                           Destination: Vector Number        2


                    Allow VDN Override? n
                                  COR: 1
                                  TN*: 1
                             Measured: none
```

4.4. Configure Vector

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

```
change vector 2                                                      Page   1 of   6
                                  CALL VECTOR

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

4.5. Configure Hunt Group

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

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

```
add hunt-group 4                                                     Page   1 of   3
                                  HUNT GROUP

            Group Number: 4                                 ACD? y
              Group Name: Voice Portal Prompts App        Queue? y
         Group Extension: 5552                            Vector? y
              Group Type: ucd-mia
                      TN: 1
                     COR: 1                    MM Early Answer? n
           Security Code:              Local Agent Preference? n
 ISDN/SIP Caller Display:

             Queue Limit: unlimited
 Calls Warning Threshold:      Port:
  Time Warning Threshold:      Port:
```

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

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

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

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


      Controlling Adjunct: none








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

4.6. Configure Agent Login ID

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

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

During compliance testing, agent login IDs 54101 to 54130 were created.

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

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


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

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

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

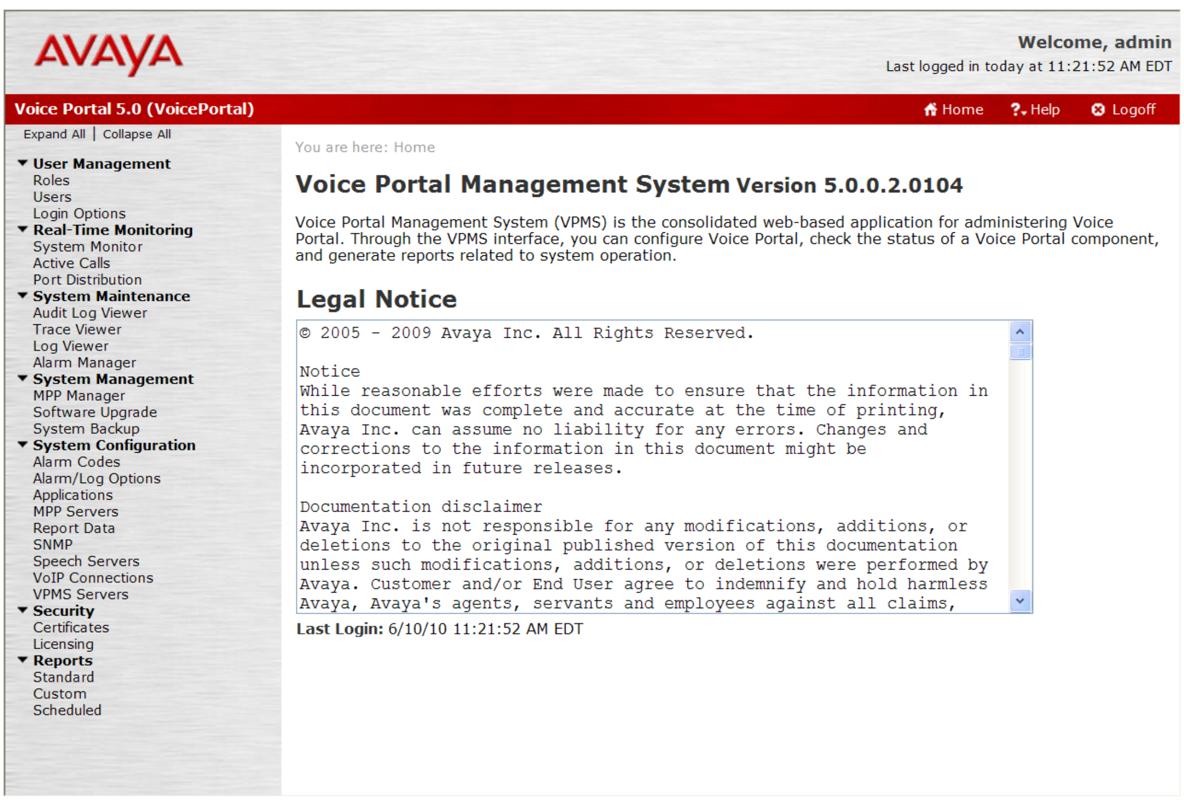
```
add agent-loginID 54110                                         Page   2 of   2
                                 AGENT LOGINID
      Direct Agent Skill:                                  Service Objective? n
Call Handling Preference: skill-level                  Local Call Preference? n

    SN   RL SL          SN   RL SL
 1: 4        1      16:
 2:                 17:
 3:                 18:
 4:                 19:
 5:                 20:
 6:
 7:
 8:
 9:
10:
11:
12:
13:
14:
15:
```

5. Configure Avaya Voice Portal

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

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

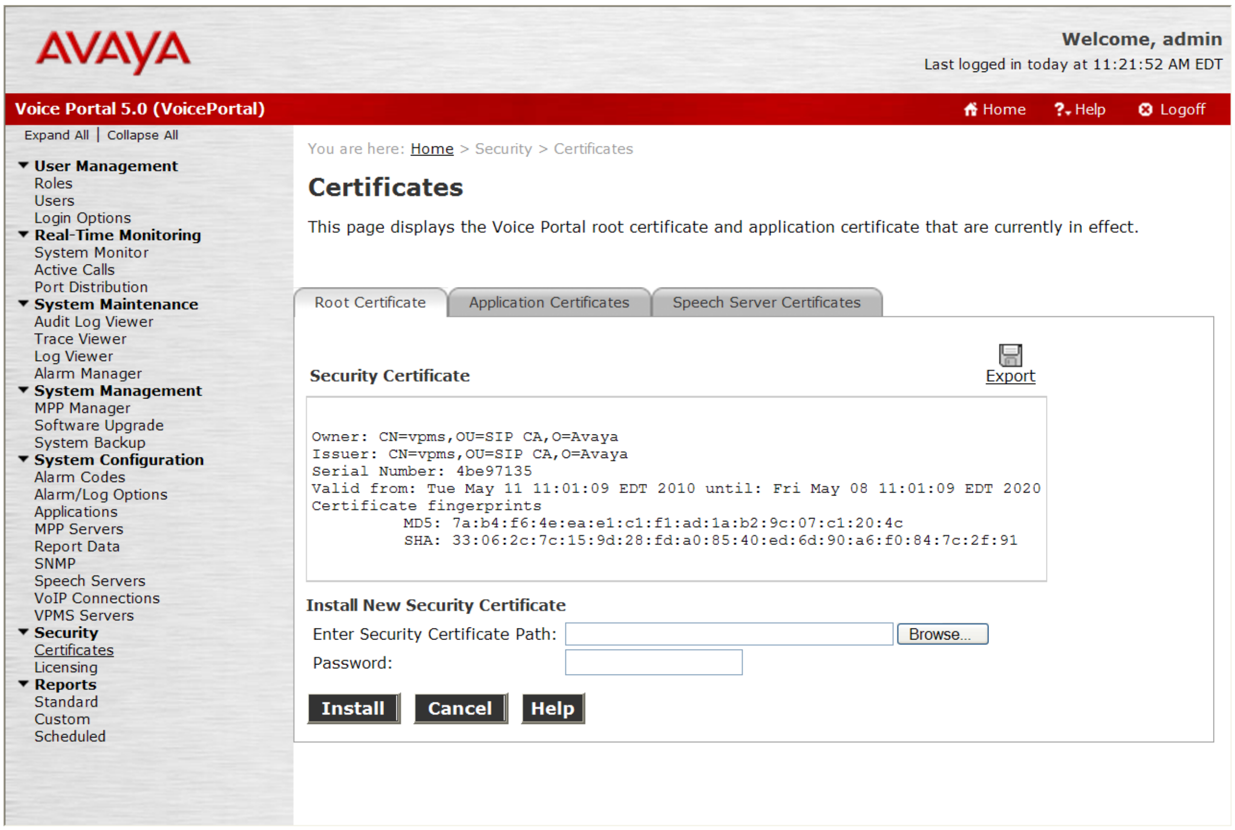
Voice Portal is configured via the Voice Portal Management System (VPMS) web interface. To access the web interface, enter `http://<ip-addr>/VoicePortal` as the URL in an Internet browser, where `<ip-addr>` is the IP address assigned to the VPMS server. Log in using the Administrator user role. The initial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



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

5.1. Install Certificate for TLS Authentication

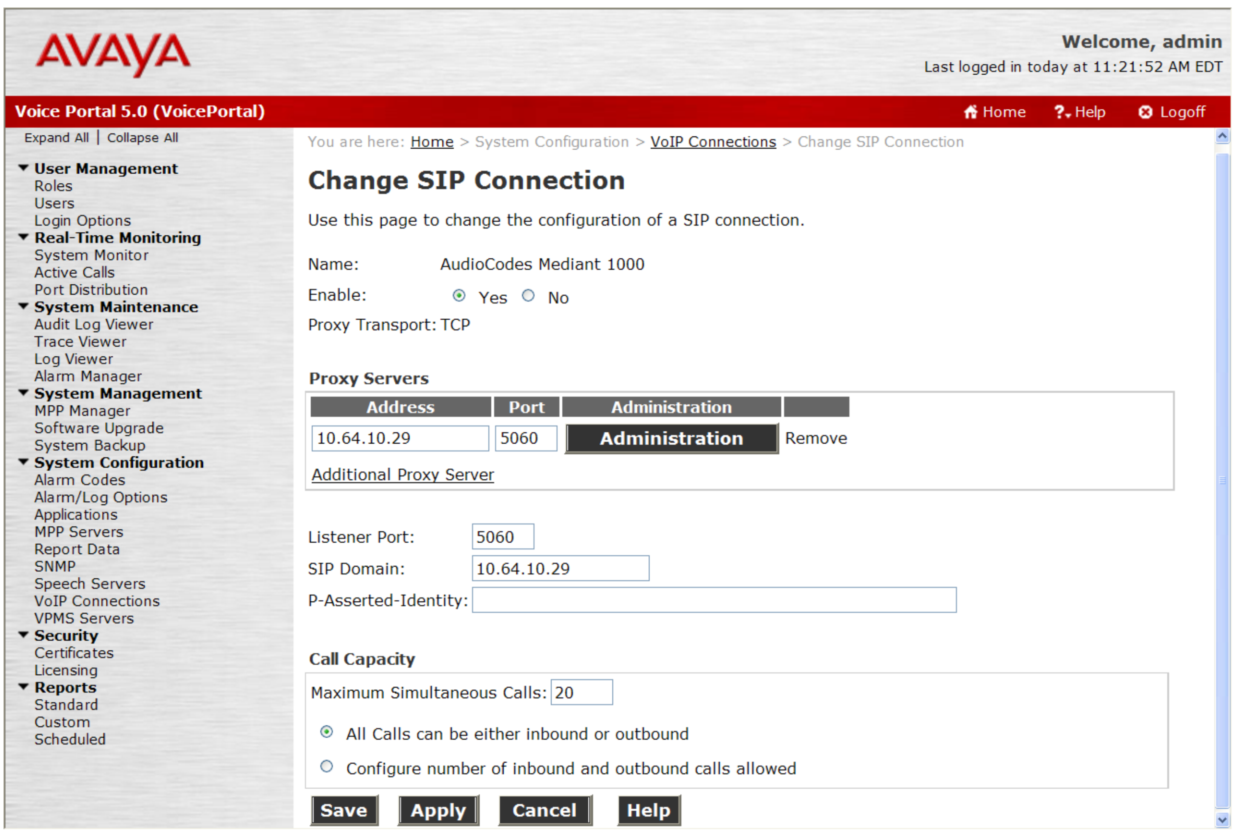
Voice Portal was configured to use TCP for the SIP interface to the AudioCodes Mediant 1000 VoIP Media Gateway (to facilitate debugging). A production environment is more likely to use TLS authentication over the SIP interface between Voice Portal and Mediant 1000. To install the certificate for TLS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



5.2. Configure SIP Connection

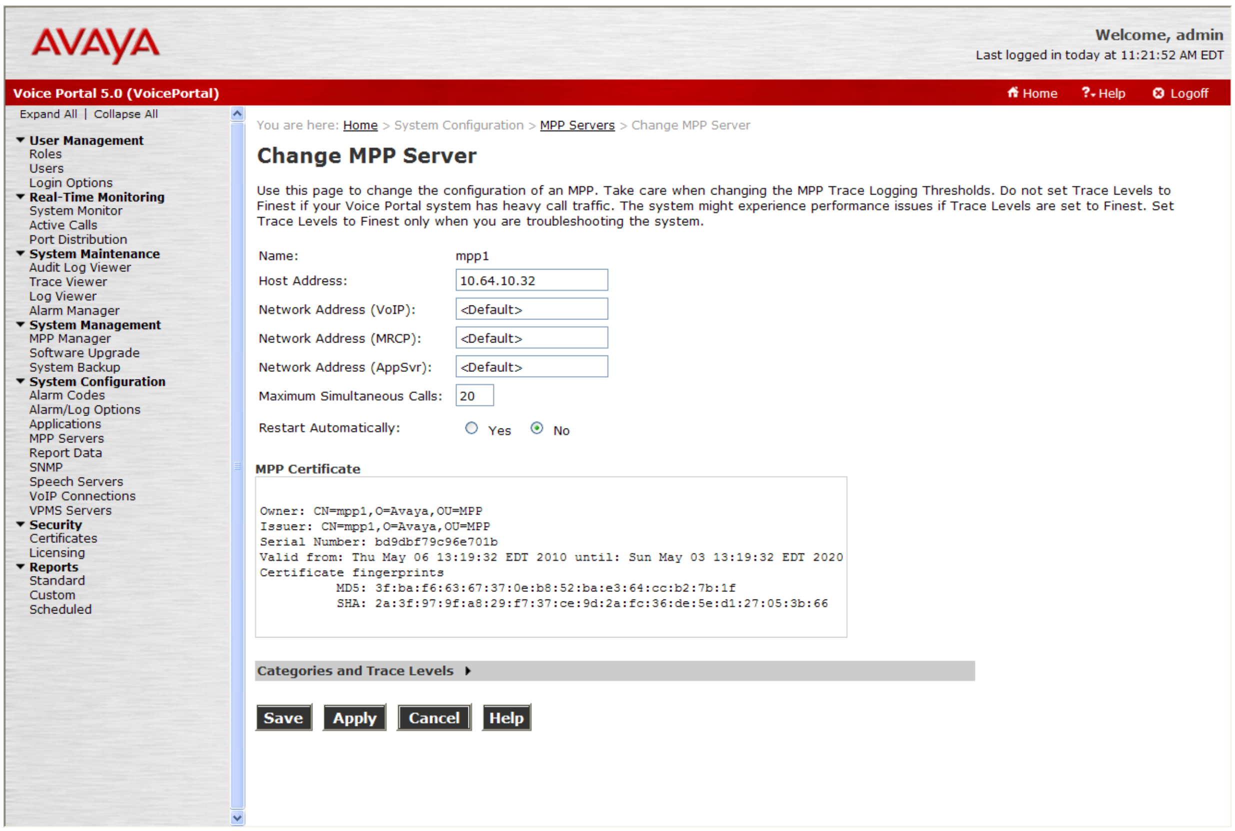
To configure a SIP connection to the AudioCodes Mediant 1000 VoIP Media Gateway, navigate to **System Configuration → VoIP Connections**, and click on the **SIP** tab. Click the **Add** button to add a new connection. On the resulting screen, configure the parameters as follows:

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



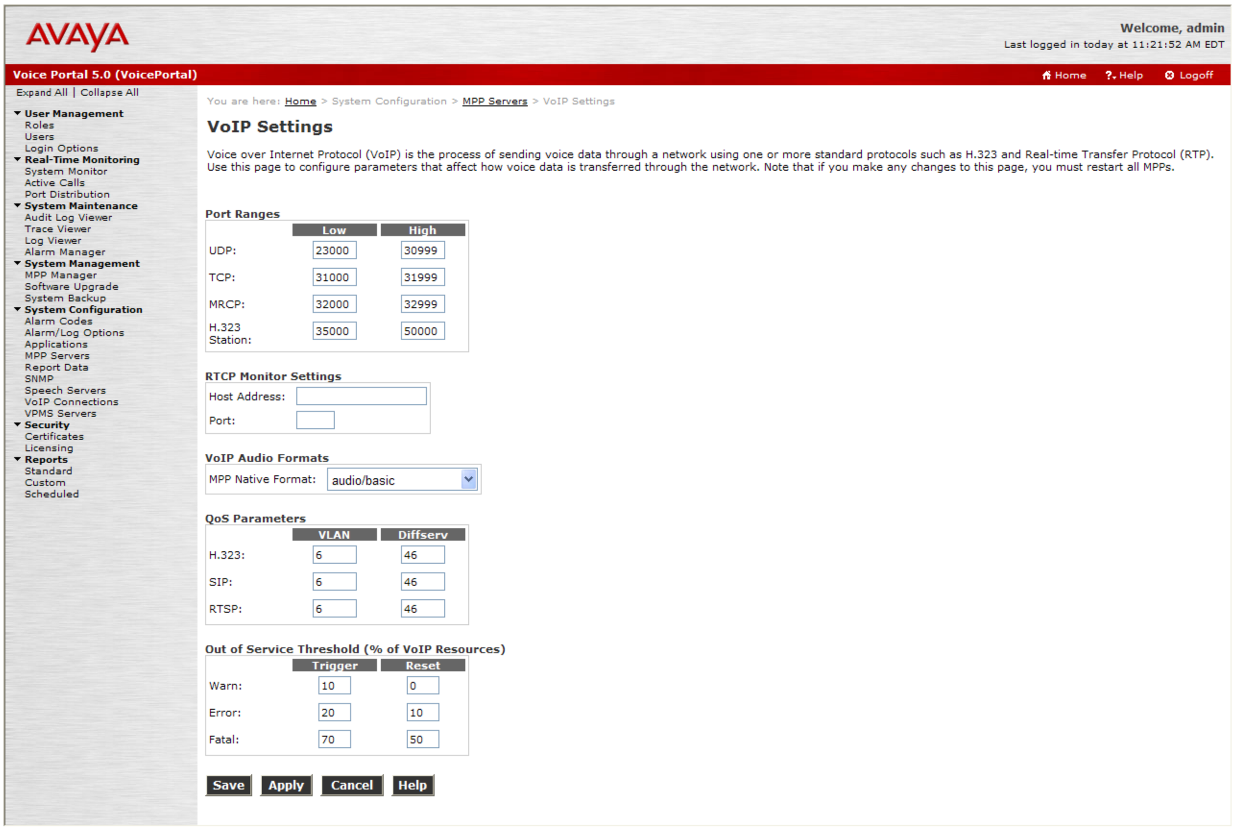
5.3. **Add MPP server**

Add a Media Processing Platform (MPP) server by navigating to **System Configuration → MPP Servers**. Click the **Add** button to add a new MPP Server. In the MPP Server configuration page, specify a descriptive name and the **Host Address** of the MPP server. Also, specify the **Maximum Simultaneous Calls** supported on this MPP server. The screen below shows the configuration for the first MPP server used in the reference configuration. Although two MPP servers were configured in the reference configuration, only 1 was used. Repeat these steps to configure additional MPP servers as necessary.



5.4. Configure VoIP Audio Format

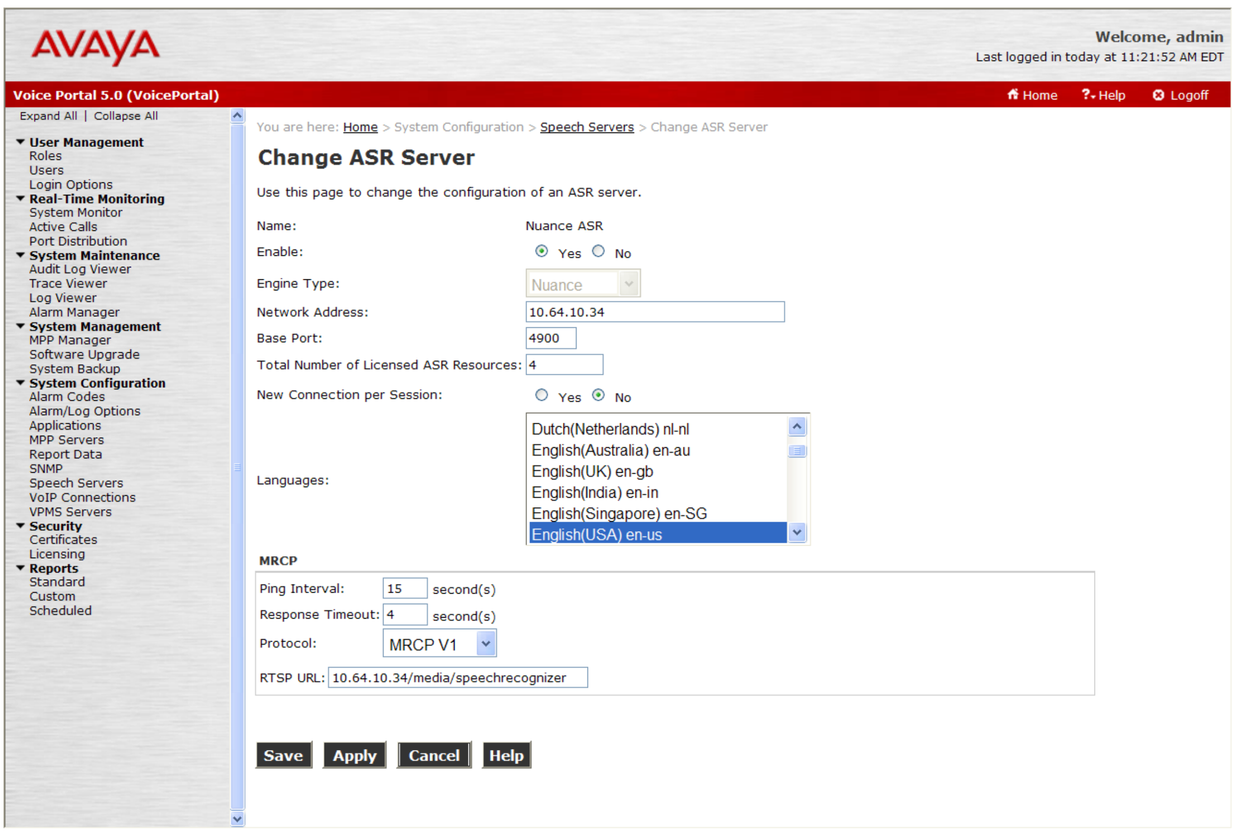
The **VoIP Audio Format** for the MPP servers is configured in the **VoIP Settings** screen, accessible from **System Configuration → MPP Servers**. The AudioCodes Mediant 1000 VoIP Media Gateway supports both G.711 mu-law and G.711 a-law. The **MPP Native Format** field in the screen below is set to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



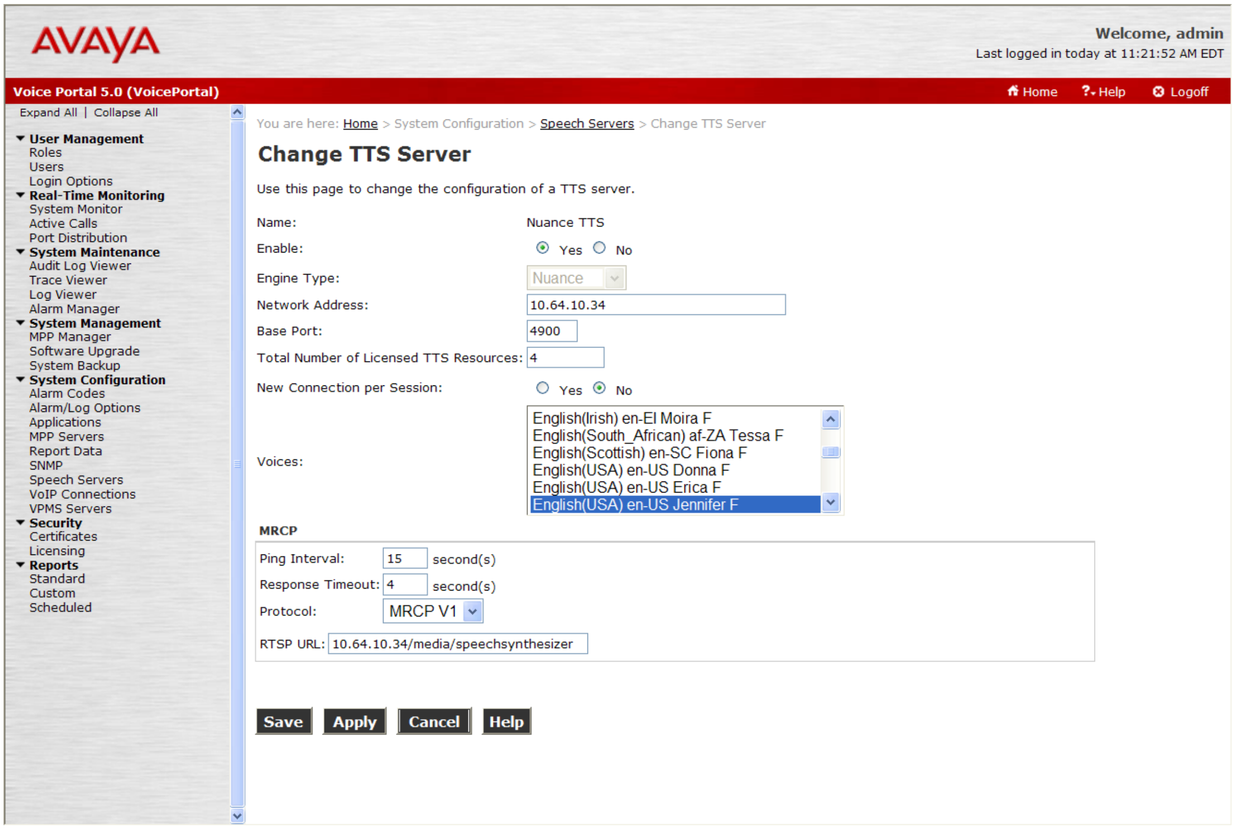
5.5. **Add Speech Server**

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

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



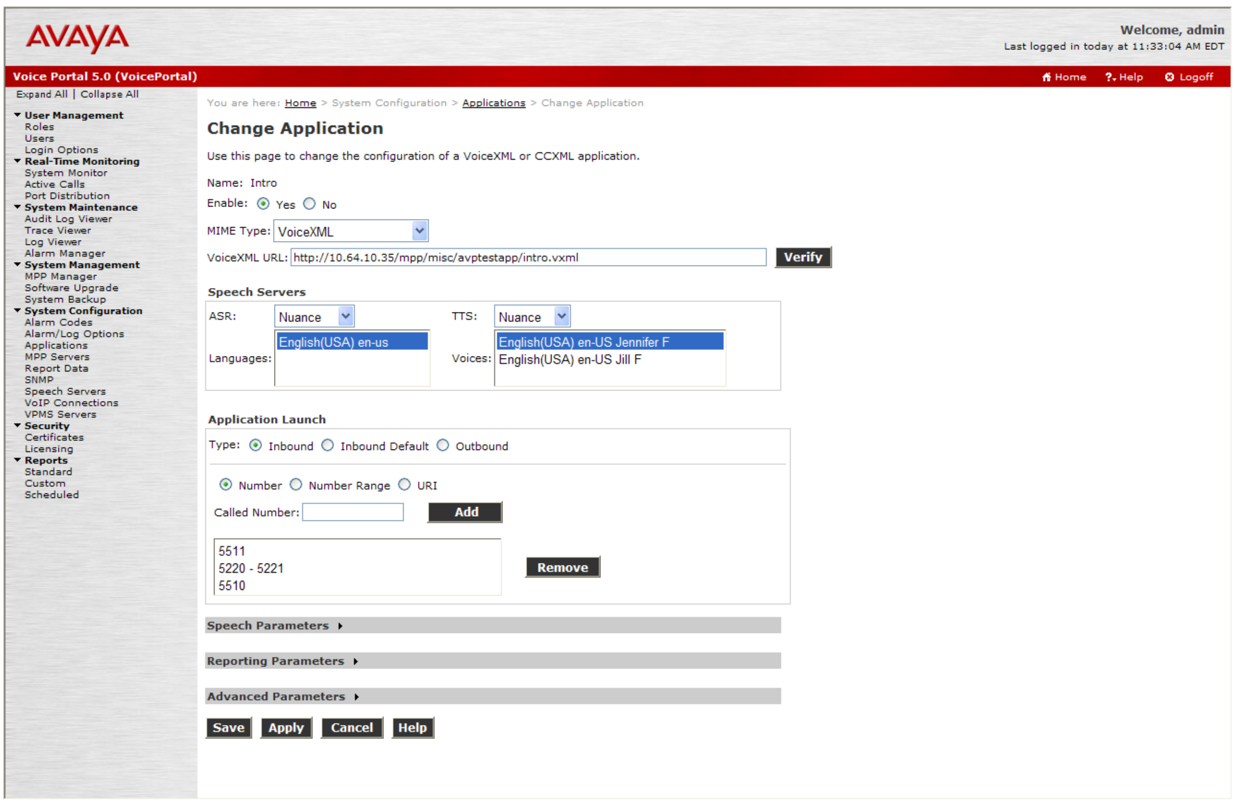
To configure the TTS server, navigate to **System Configuration → Speech Servers**, select the **TTS** tab, and then click **Add**. The screen below shows the configuration for the TTS server used during compliance testing. In this configuration, a Nuance TTS server was used so the engine type was set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



5.6. Add Voice Application

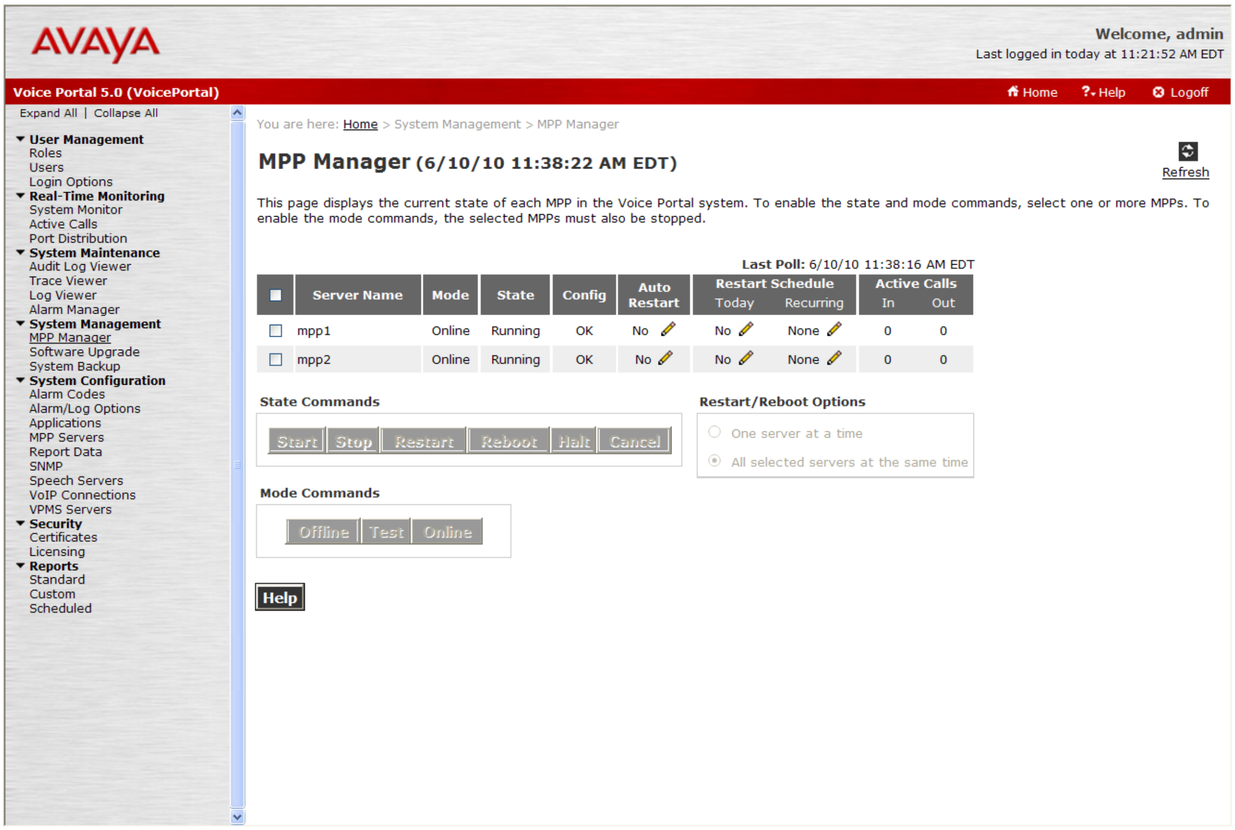
Adding a voice application for Voice Portal is part of Voice Portal's standard administration. This configuration is not directly related to achieving interoperability between the AudioCodes Mediant 1000 VoIP Media Gateway and Voice Portal. It is included here for completeness.

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



5.7. Start MPP Server

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



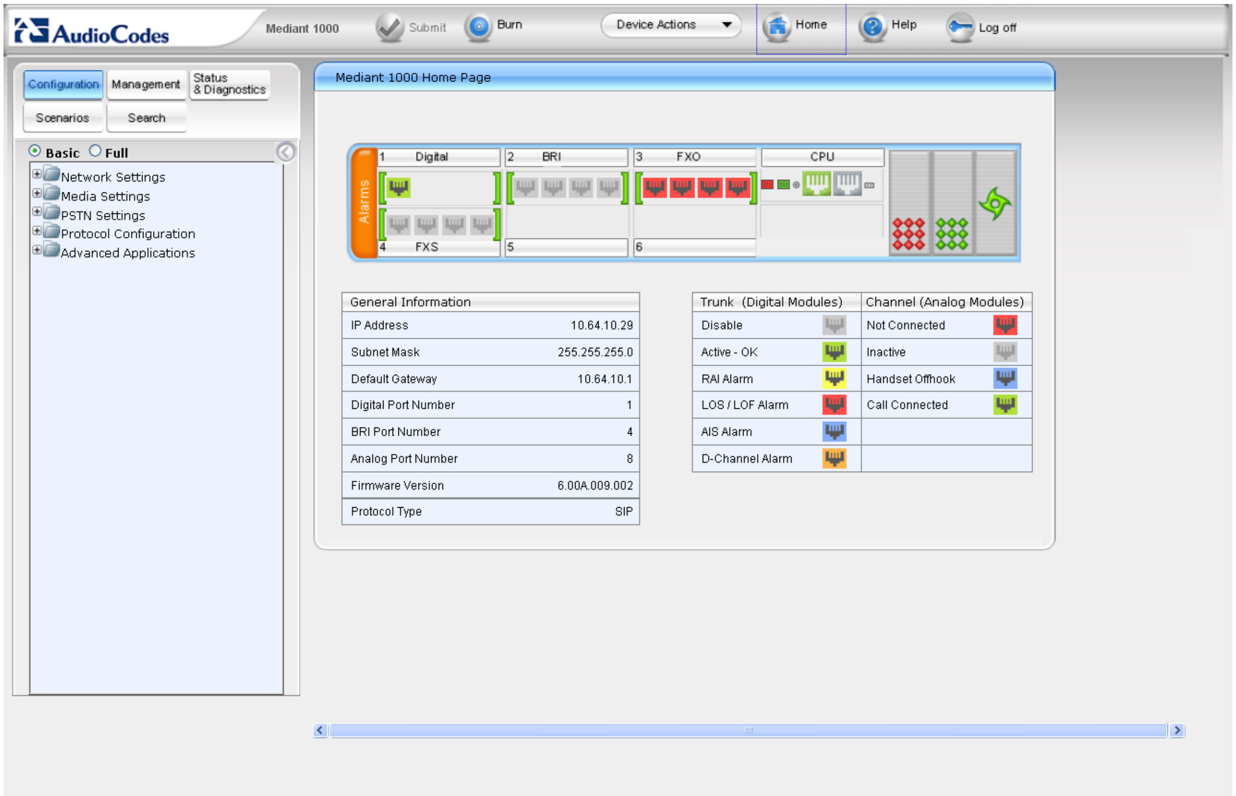
6. Configure AudioCodes Mediant 1000 VoIP Media Gateway

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

The Mediant 1000 configuration procedures include the following areas:

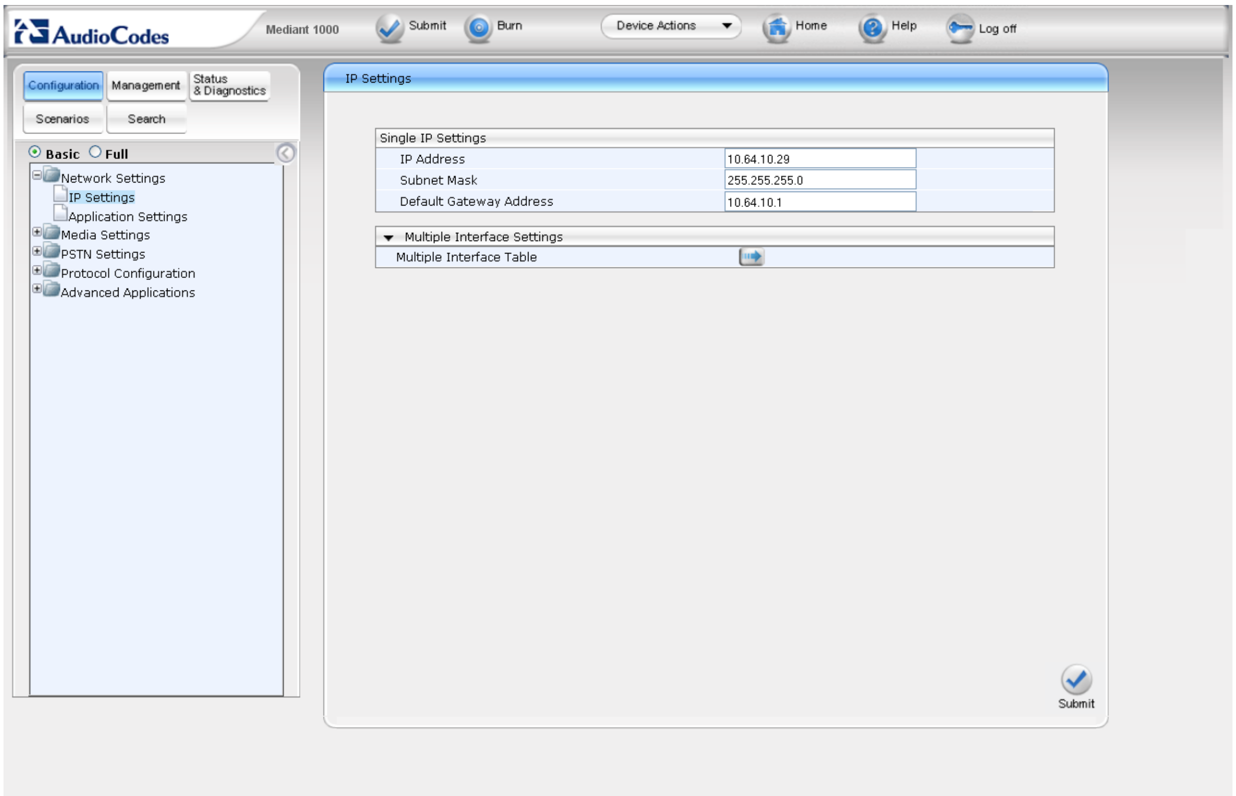
- Network IP settings
- PSTN trunk settings
- SIP General parameters
- SIP Advanced parameters
- SIP Proxy and Registration
- Proxy Sets table
- Coders
- DTMF and Dialing
- Trunk Group
- IP to trunk group routing
- Media voice settings

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



6.1. **Network IP settings**

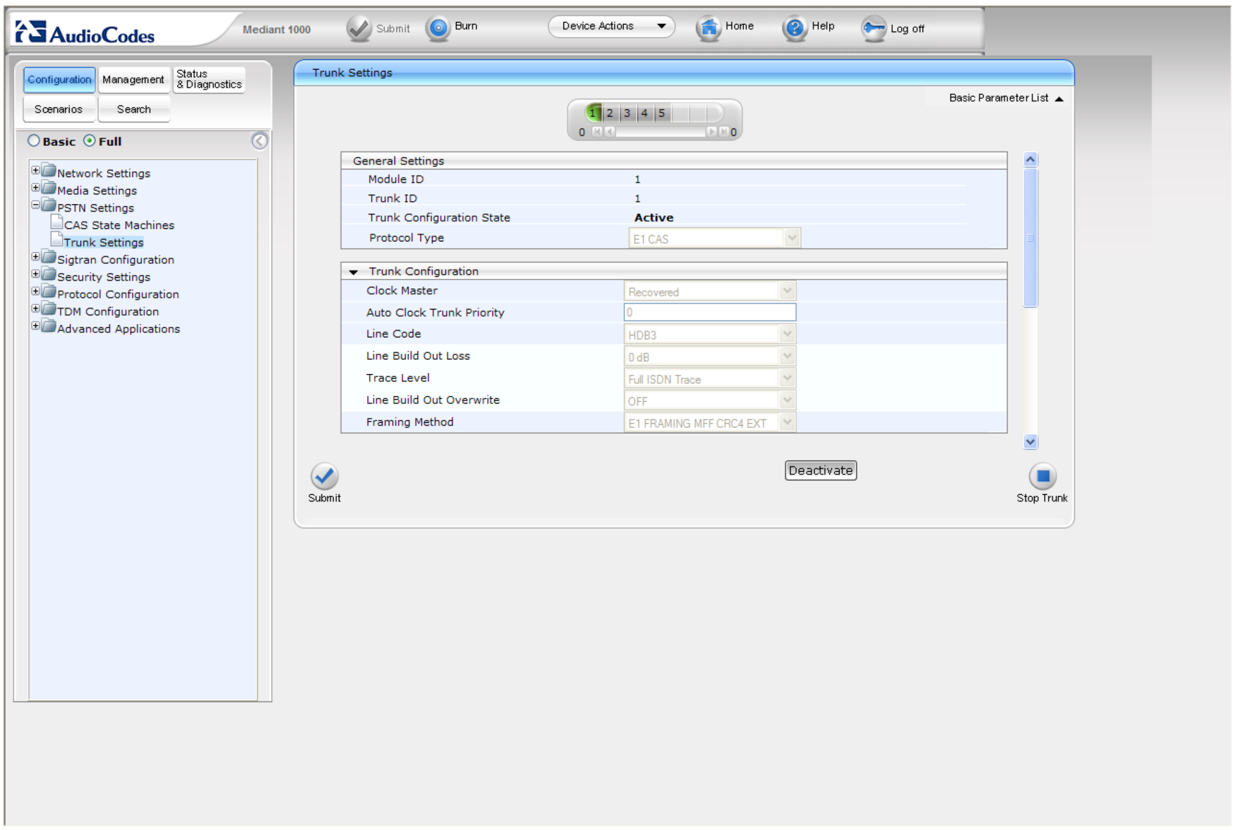
The network settings that were configured during installation can be viewed by navigating to **Network Settings → IP Settings** in the left pane. If necessary, changes can be made to the settings on this page followed by clicking the **Submit** icon button at the bottom of the screen. For compliance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



6.2. PSTN trunk setting

Navigate to **PSTN Settings → Trunk Settings** to configure the line side E1 interface to Communication Manager. These settings must be consistent with the DS1 settings on Communication Manager (**Section 4.1**). Configure the following parameters.

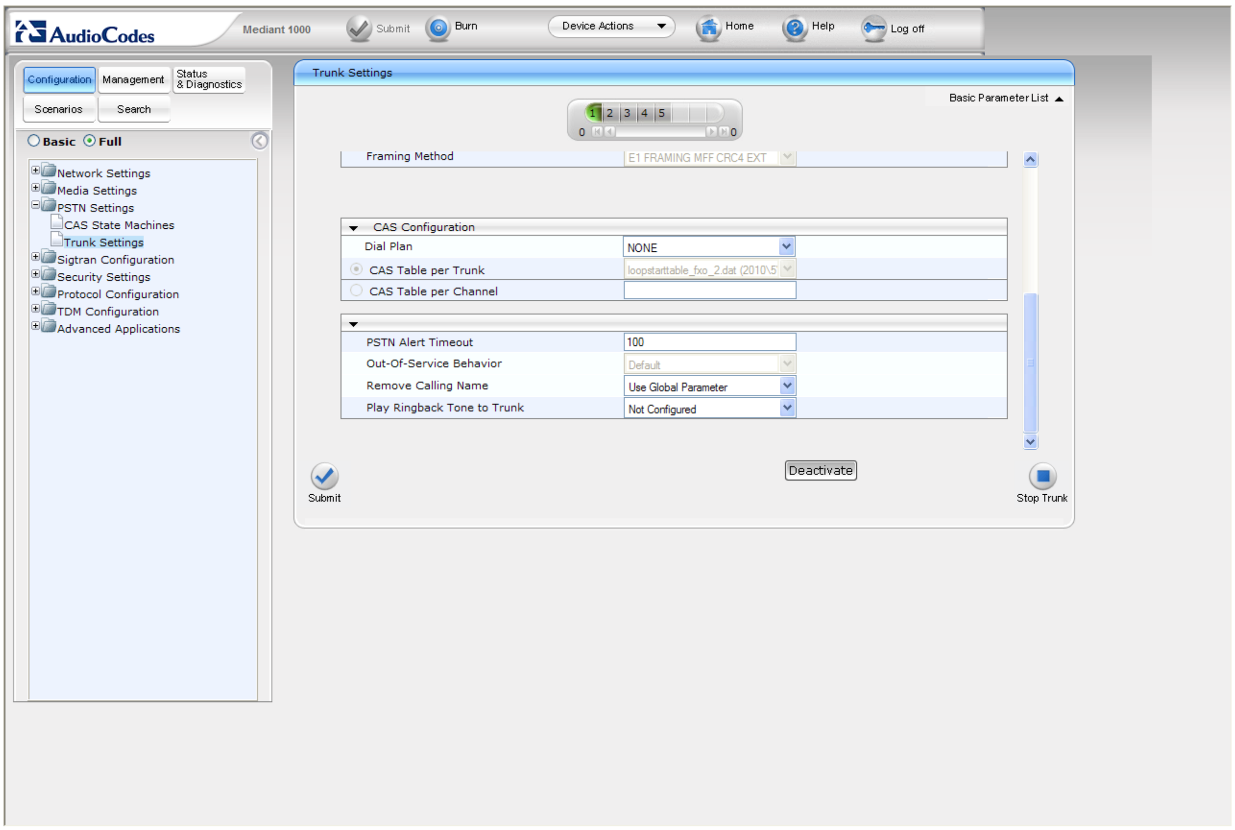
- Set **Protocol Type** to ***E1 CAS***.
- Set **Line Code** to ***HDB3***.
- Set **Framing Method** to ***E1 FRAMING MFF CRC4 EXT***.



Scroll down, then configure following parameters as described:

- Enable **CAS Table per trunk** and select the appropriate CAS table from the pull down menu
  Note: The names of the CAS tables used during compliance testing were changed after testing was completed. The CAS table chosen impacts the behavior of calls. See **Section 7** for the updated CAS table names and for a description of the behaviors observed using each CAS table.
- Enter ***100*** (seconds) for **PSTN Alert Timeout**. This timeout setting on the trunk is used for disconnecting unanswered calls on the PSTN side.

Default values may be retained for all other fields.



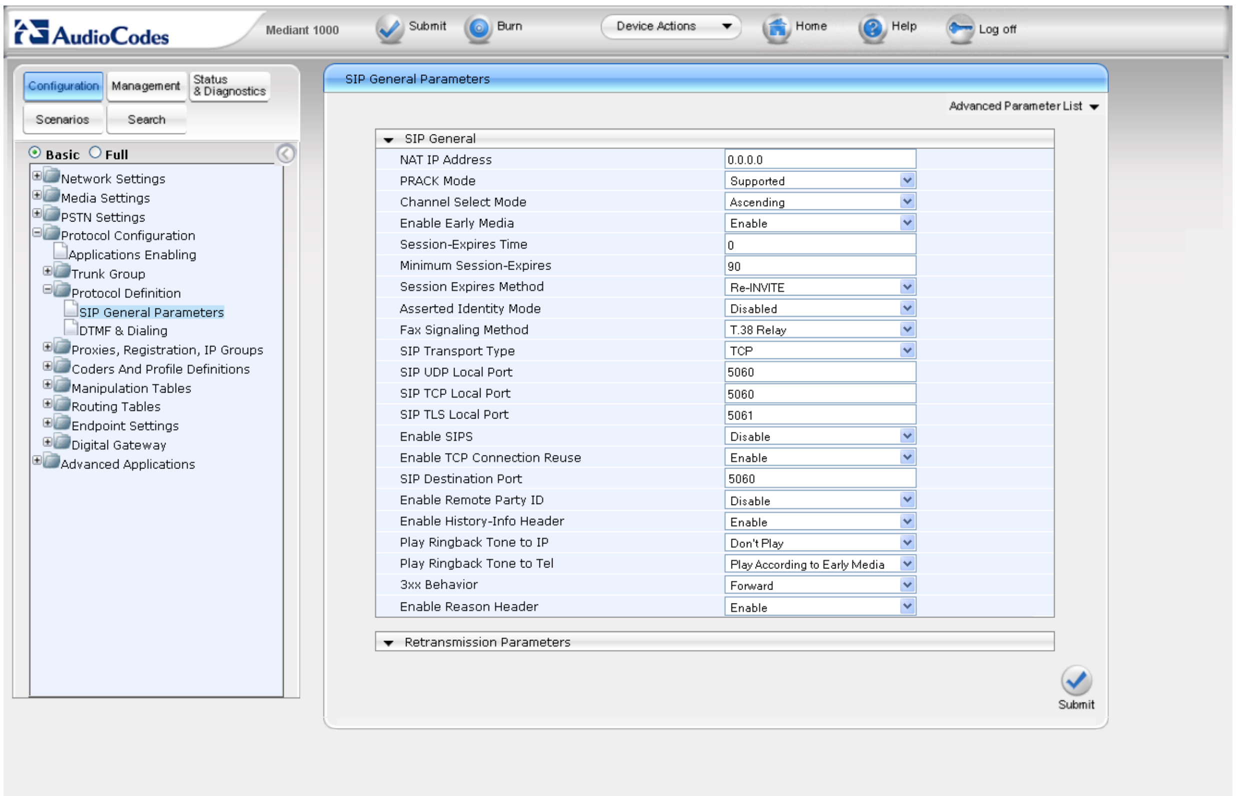
After all the parameters are properly specified, click the **Apply Trunk Settings** button at the bottom of the screen (not shown).

6.3. SIP General Parameters

Navigate to **Protocol Configuration → Protocol Definition → SIP General Parameters**. Configure the parameters as described below.

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

Default values may be retained for all other fields.

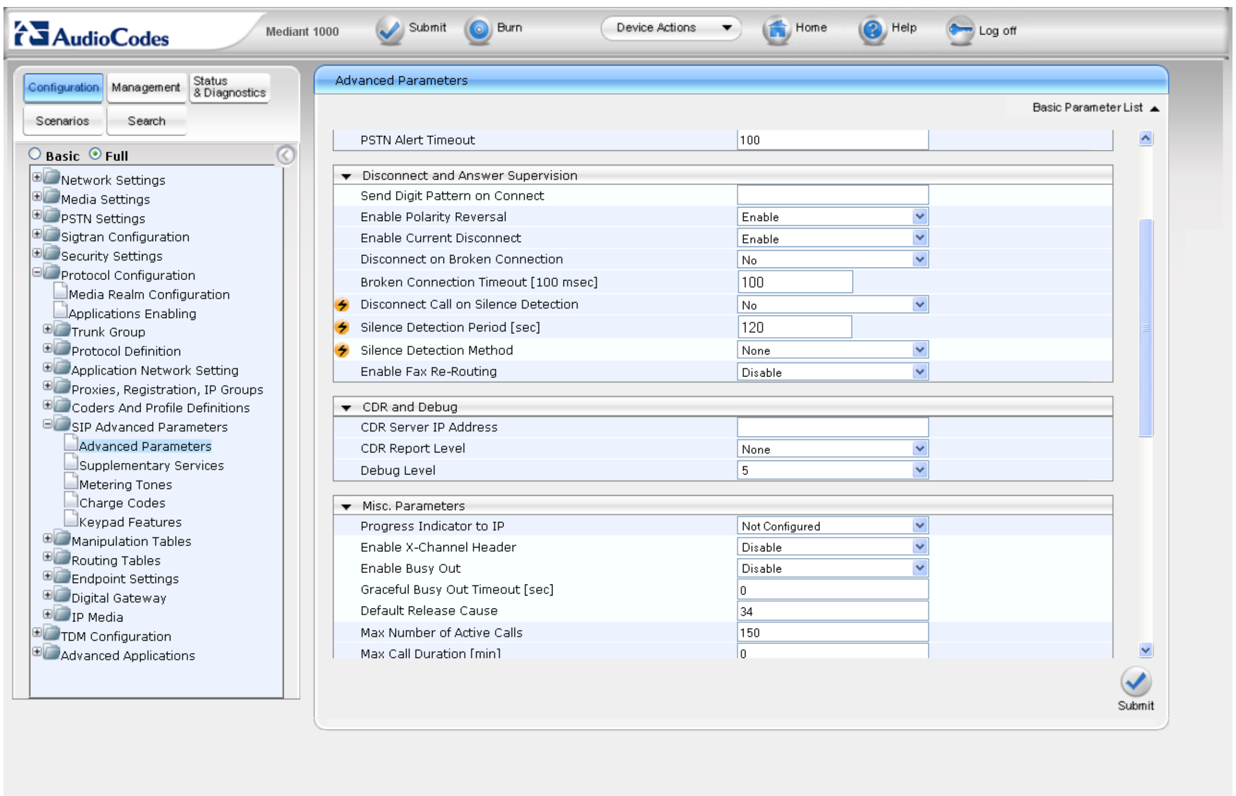


6.4. SIP Advanced Parameters

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

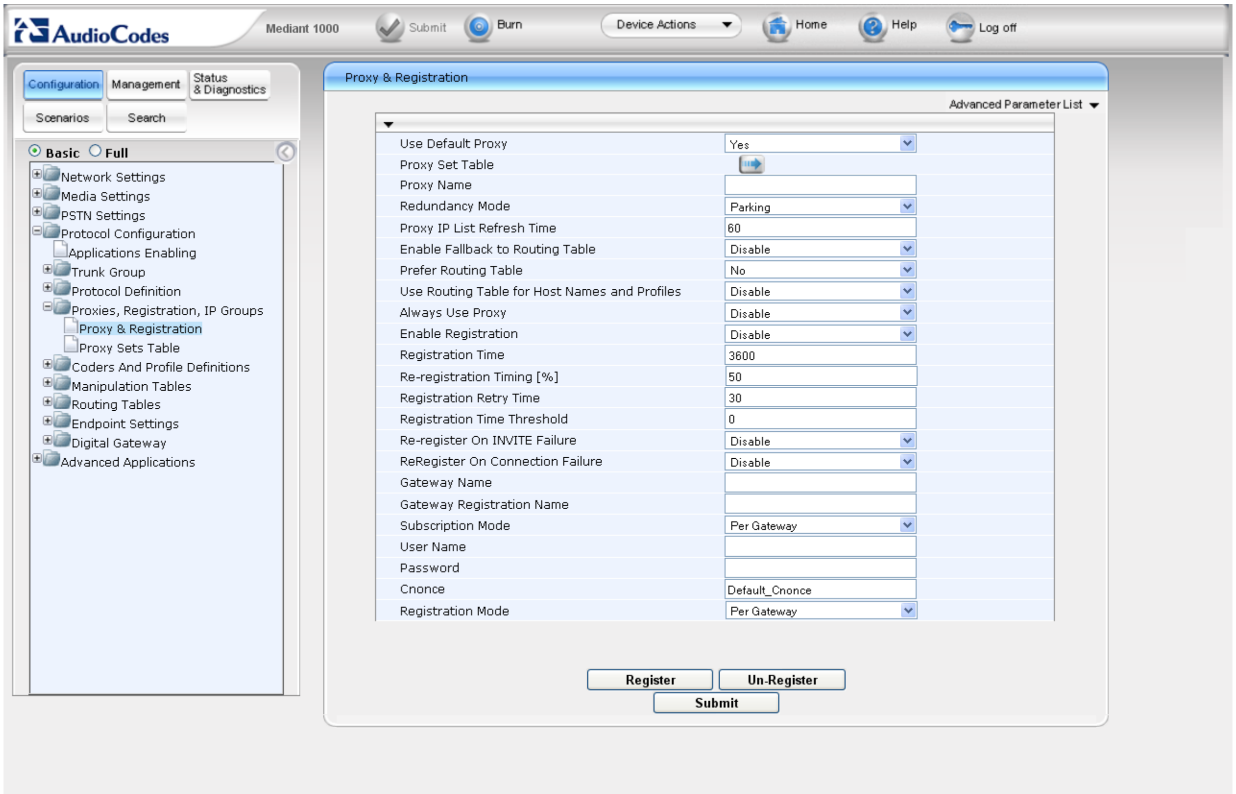
- Specify ***100*** (seconds) for **PSTN Alert Timeout**. This timeout setting on the gateway is for disconnecting unanswered calls on the PSTN side.
- Scroll down and set the **Max Number of Active Calls** field to an appropriate value.

Default values may be retained for all other fields.



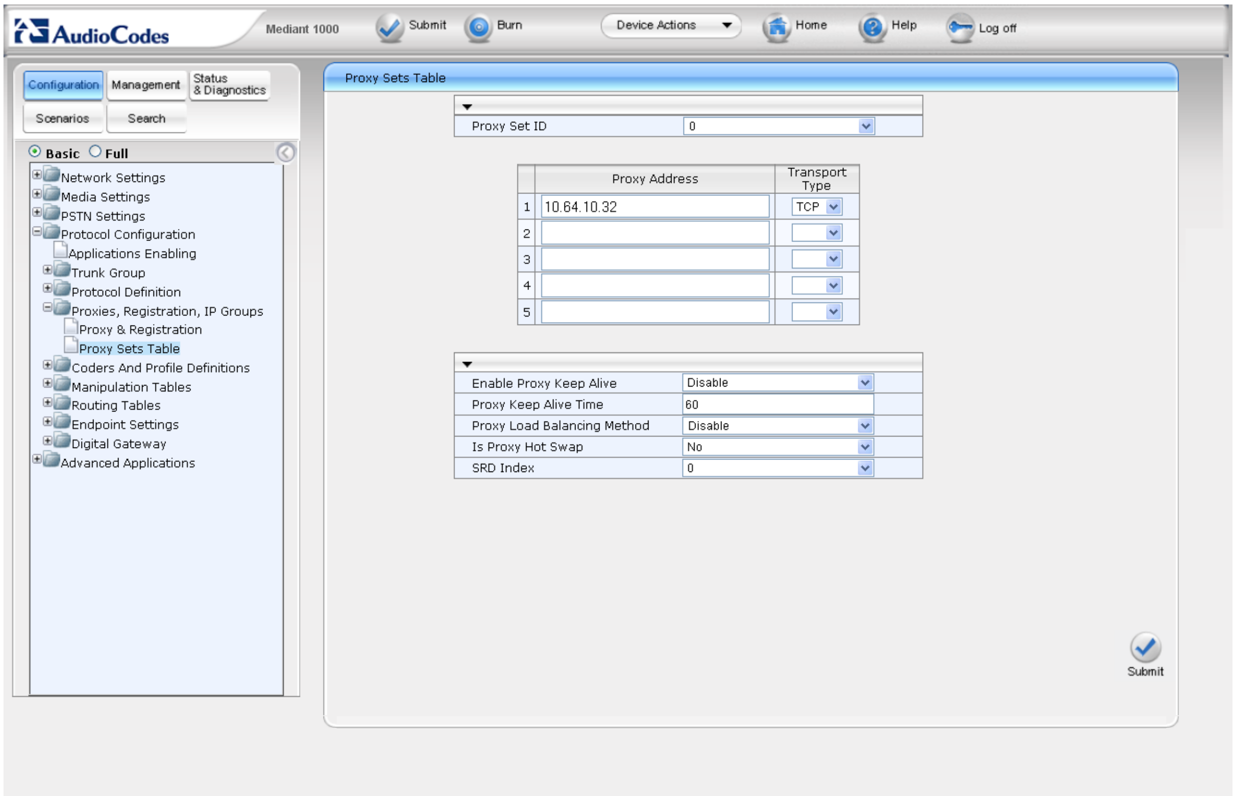
6.5. **SIP Proxy and Registration**

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



6.6. Proxy Sets Table

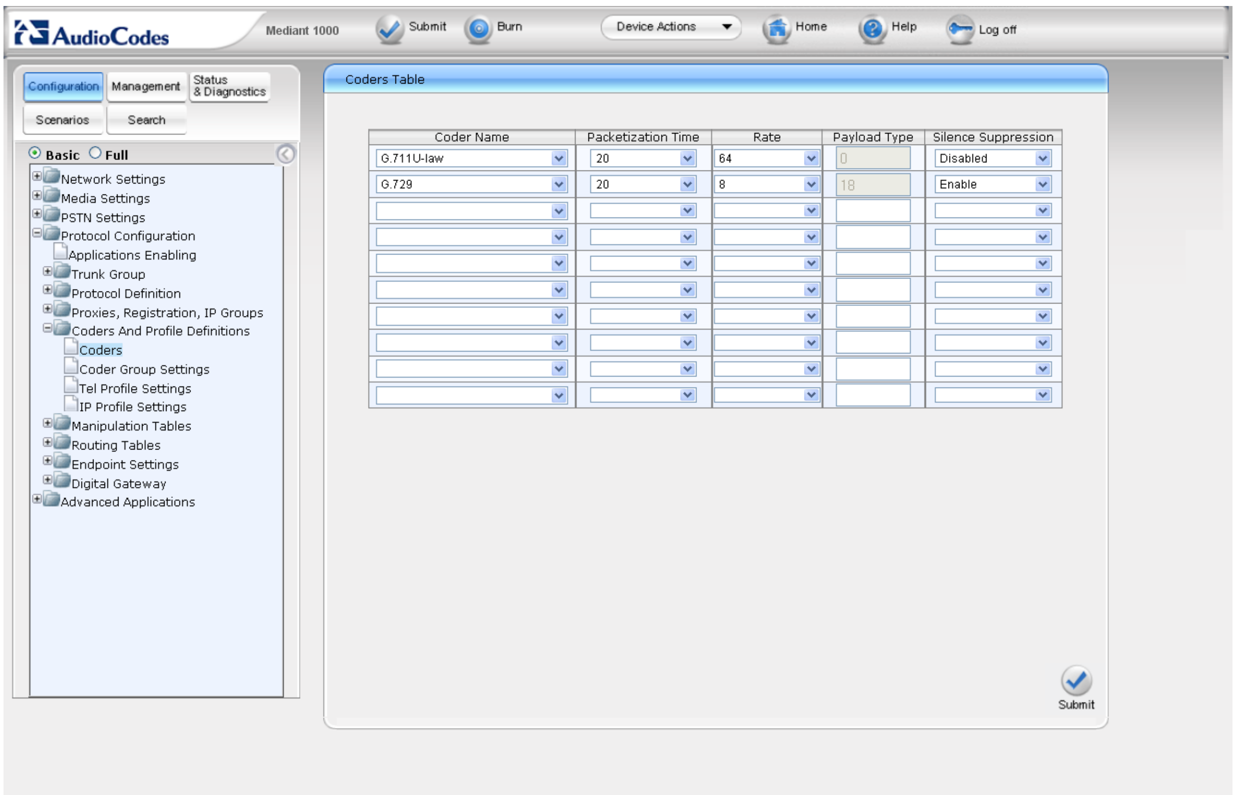
Navigate to **Protocol Configuration → Proxies, Registration, IP Groups → Proxy Sets Table** to reach the Proxy Sets Table configuration page. Enter the IP address assigned to the Voice Portal MPP server for **Proxy Address**, and ***TCP*** for **Transport Type**. Default values may be retained for all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



6.7. Coders

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

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

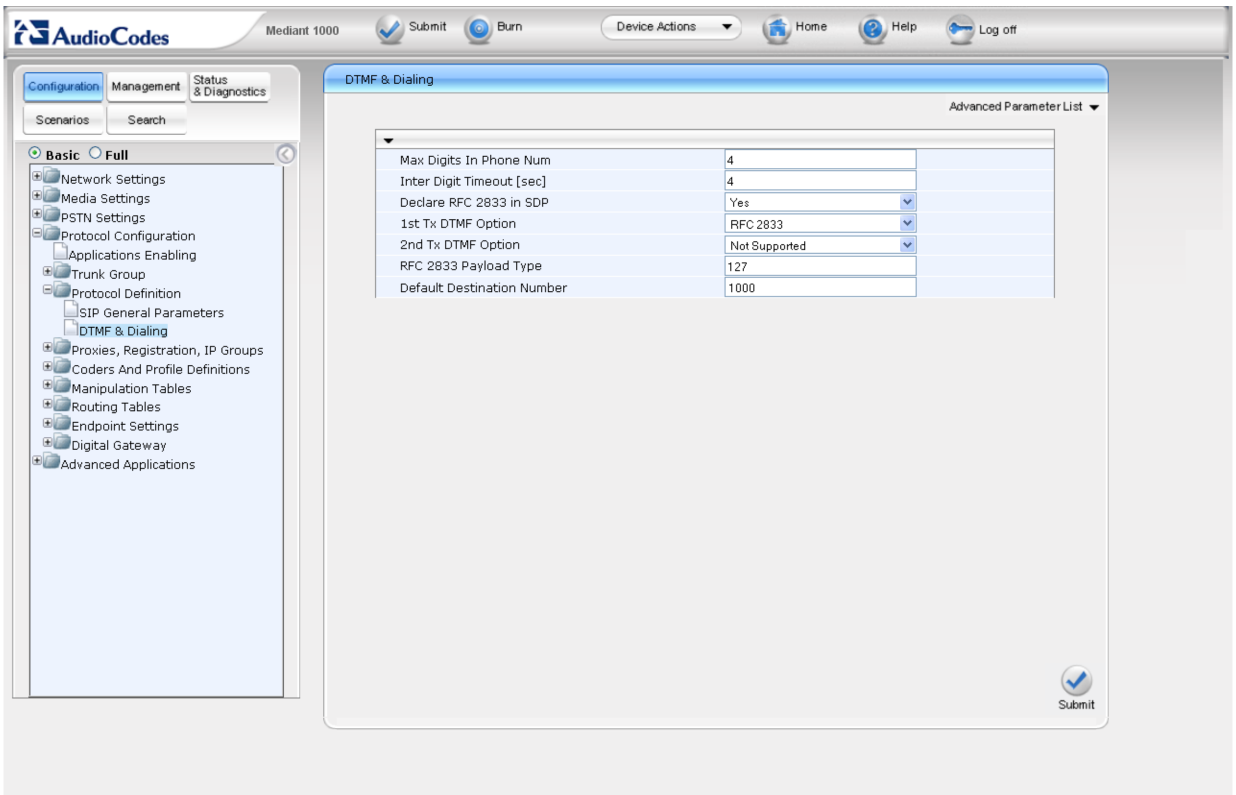


6.8. DTMF and Dialing

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**.

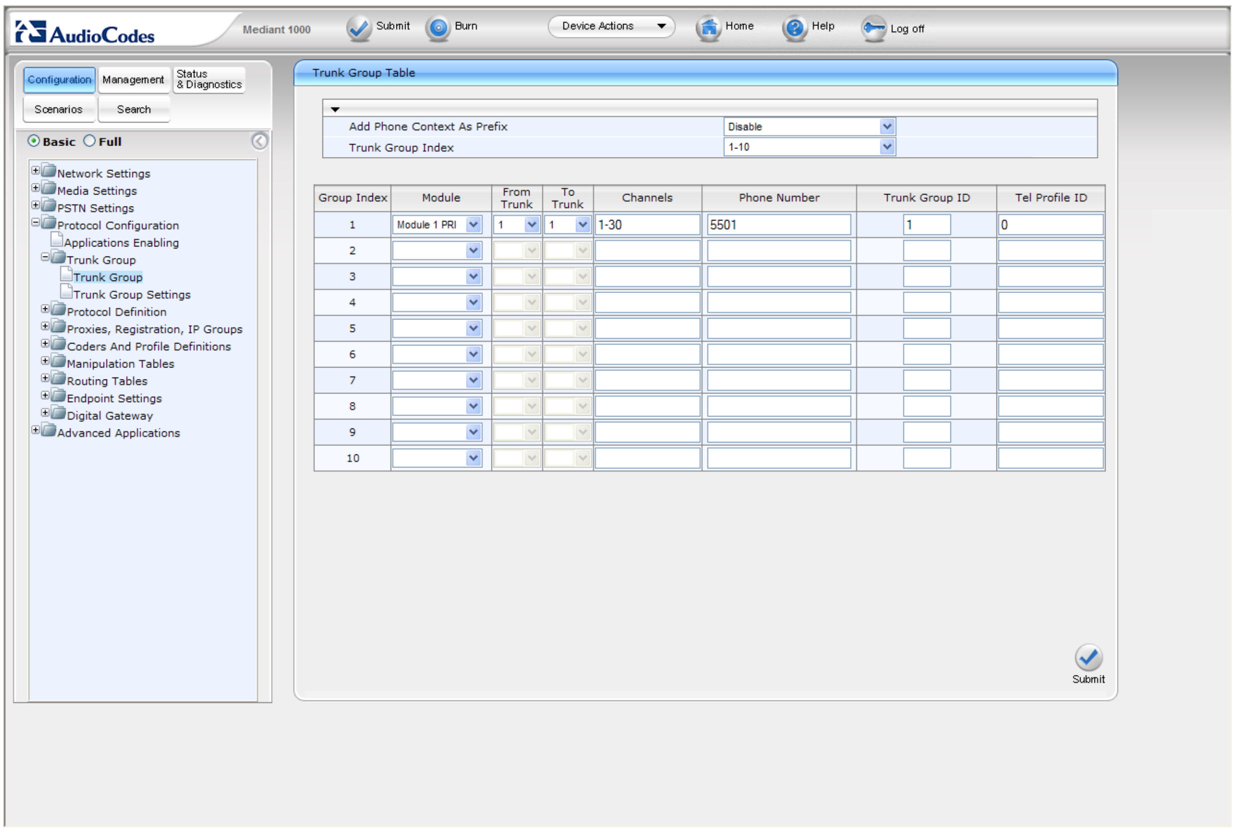
Default values may be retained for all other fields.



6.9. Trunk Group

Navigate to **Protocol Configuration → Trunk Group → Trunk Group.** The Trunk Group Table maps a particular trunk channel to a trunk group. In the **From Trunk** and **To Trunk** columns, enter the starting and ending trunks to be assigned. In the **Channel(s)** column, enter the range of channels on those trunks to be assigned. The setting ***1-30*** means 30 channels are assigned to each trunk as defined in the **From Trunk** and **To Trunk** columns. A phone number may be entered in the **Phone Number** column or it may be left blank. If a number is entered, this number will be used as the originating calling party if no calling party information is received from the originating PSTN trunk. Each channel is assigned a unique number starting with the value in the **Phone Number** column and incrementing for each subsequent channel. If the **Phone Number** column is left blank, the Mediant 1000 will use a default value (1000) for the originating calling party if no calling party information is received from the originating PSTN trunk. In the **Trunk Group ID** column, enter the trunk group that will contain these channels. The default value may be used for the **Tel Profile ID** column.

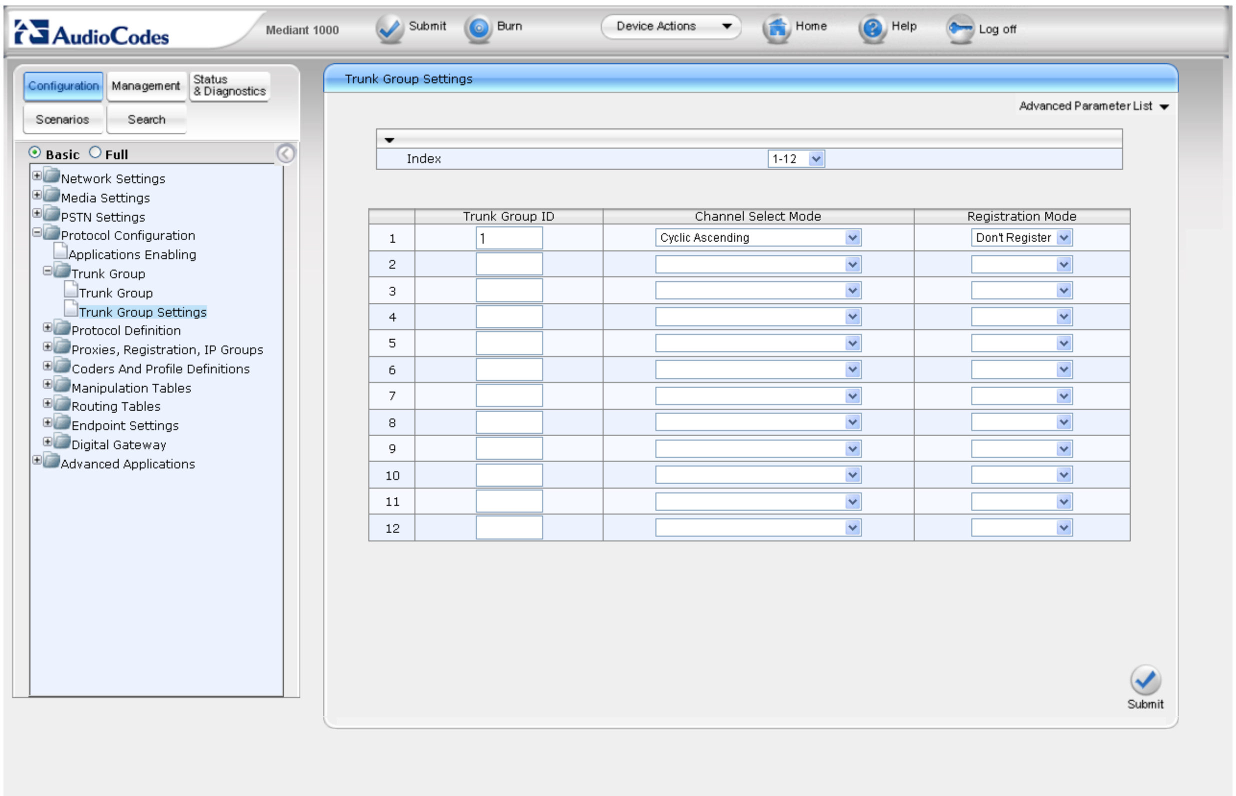
In the example below, the table entry assigns channels ***1 – 30*** of trunk ***1*** to Trunk Group ***1.*** A range of numbers arbitrary chosen to start at ***5501*** will be used for the originating calling party number if no calling party information is received from the originating PSTN trunk.



6.10. **Trunk Group Settings**

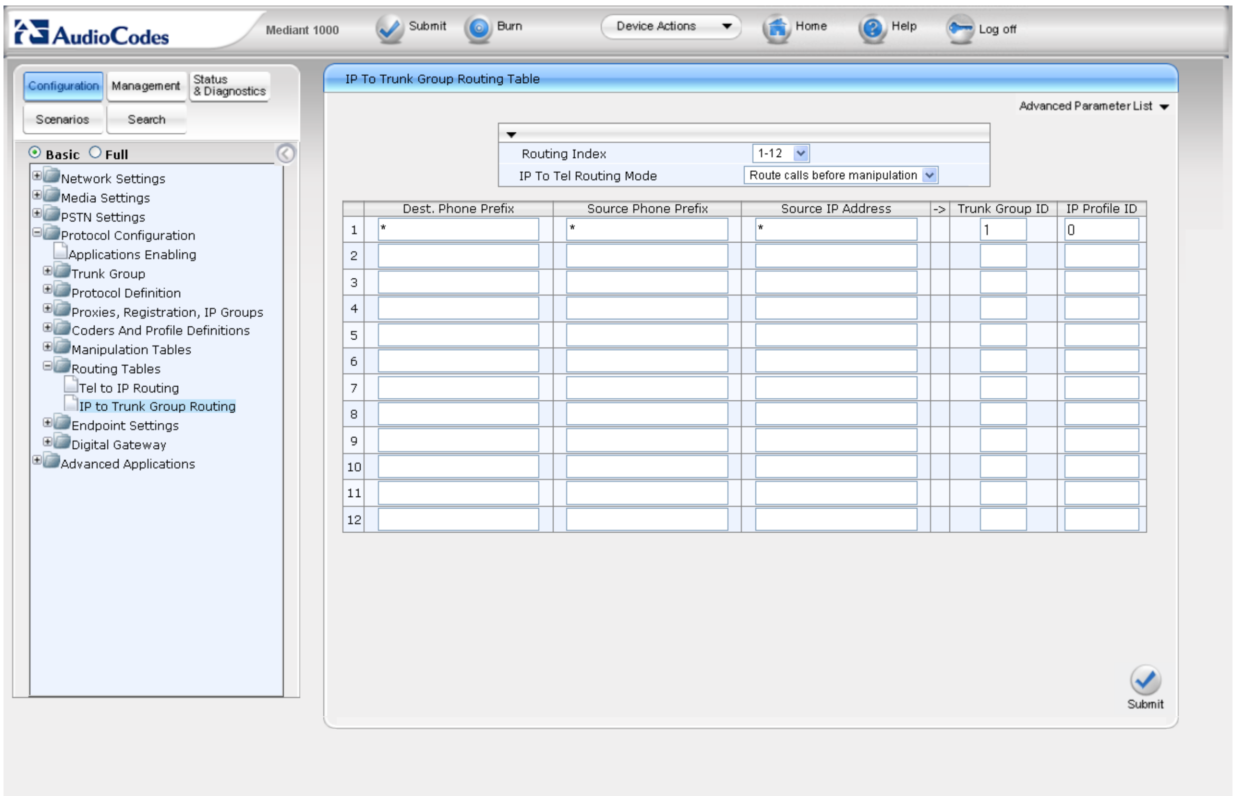
Navigate to **Protocol Configuration → Trunk Group → Trunk Group Settings**. Configure the parameters as described below.

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



6.11. **IP to Trunk Group Routing**

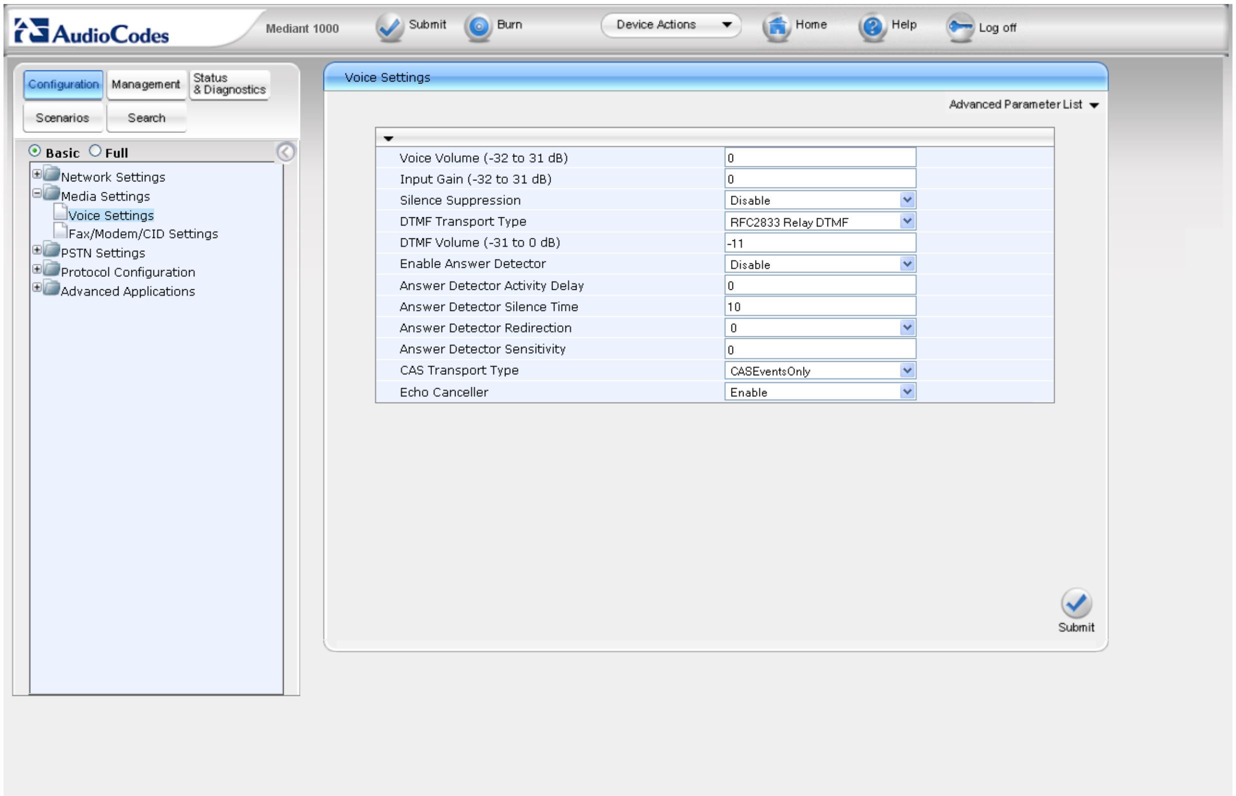
Navigate to **Protocol Configuration → Routing Tables → IP to Trunk Group Routing**. The Inbound IP Routing Table defines the mapping of IP calls to the trunk group created in **Section 6.9**. The **Dest. Phone Prefix**, **Source Phone Prefix** and **Source IP Address** columns define which calls are 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: The Tel to IP Routing Table was not configured for compliance testing. This is because Voice Portal's MPP IP address was configured as the proxy in the Proxy Sets Table (**Section 6.6**); therefore, all calls from the Tel (line-side) side will be sent to the Voice Portal MPP on the IP side. In a configuration where no default proxy was defined, the Tel to IP Routing Table would need to be configured.

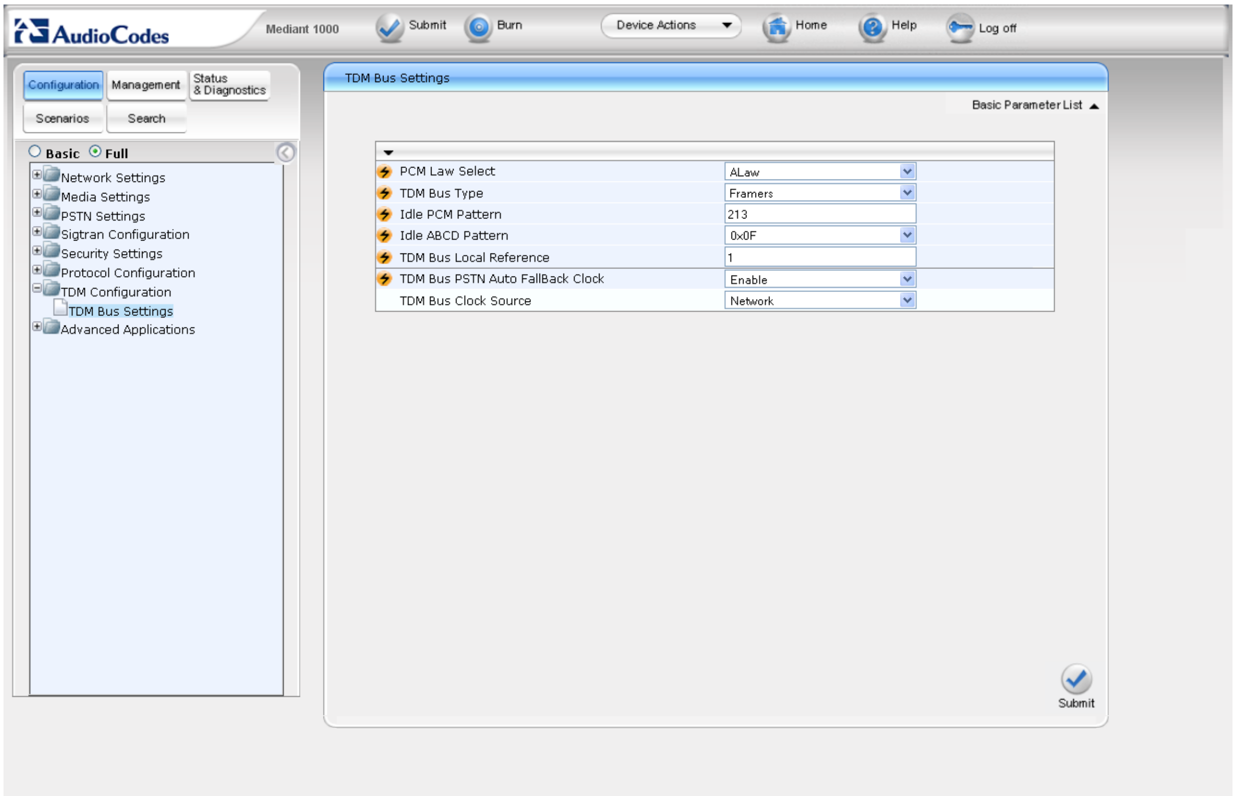
6.12. **Media Voice Settings**

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



6.13. **TDM Bus Settings**

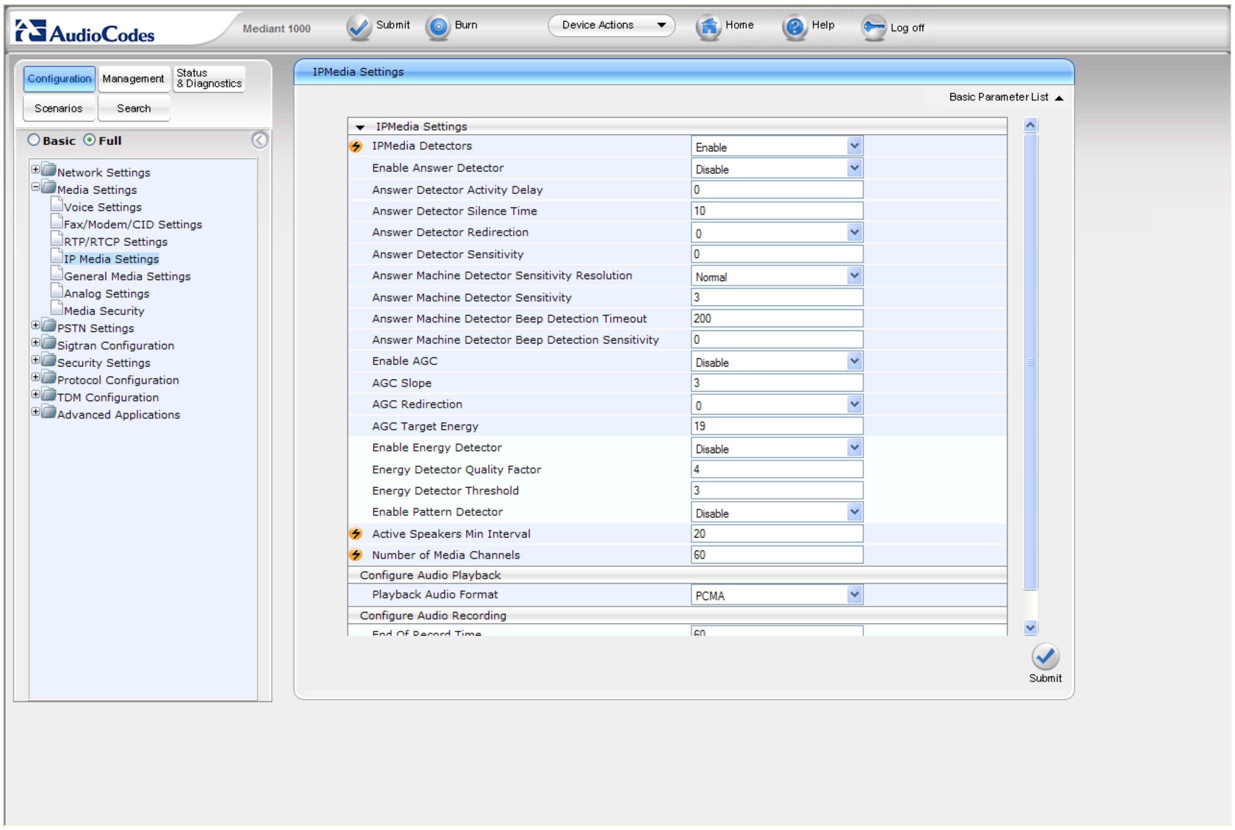
Click the **Full** radio button above the navigation pane on the left to return to the Full configuration menu tree, and then navigate to **TDM Configuration → TDM Bus Settings**. For **PCM Law Select**, select ***ALaw***. Default values may be retained for all other fields.



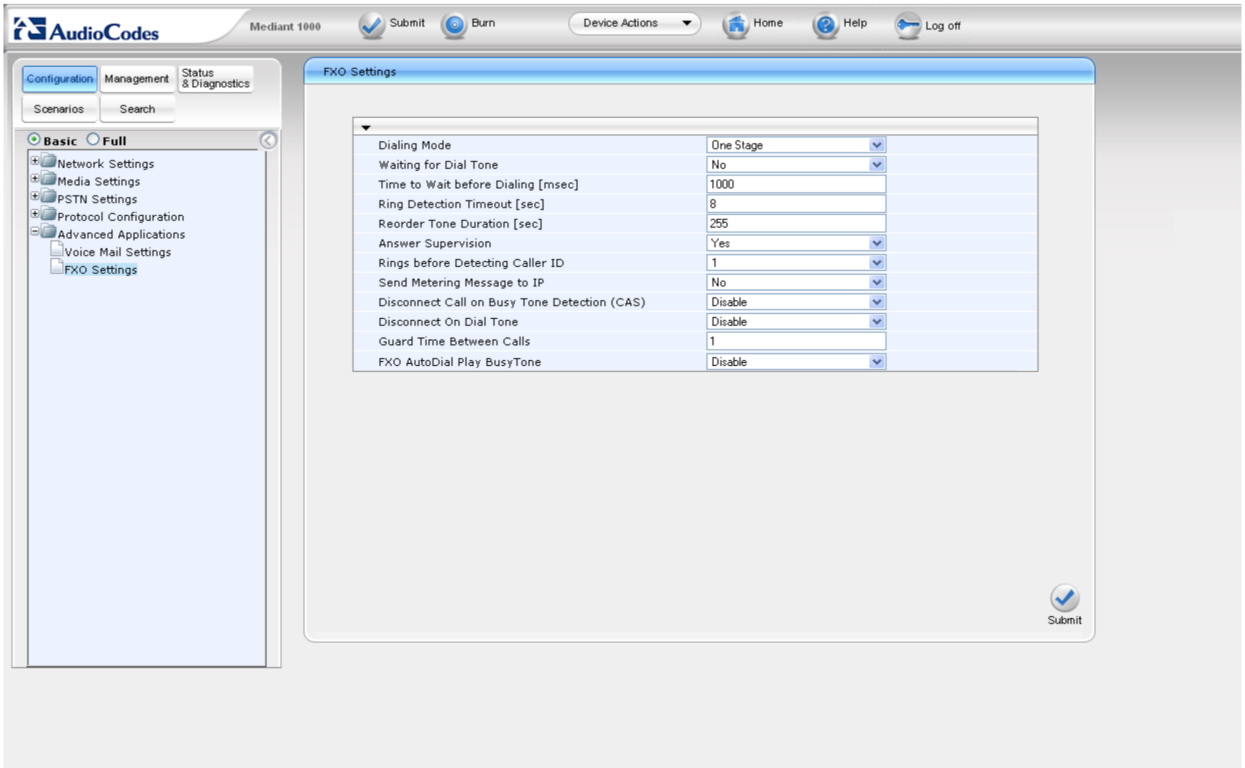
6.14. IP Media Settings

Enable voice detection/answer supervision.

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



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



7. General Test Approach and Test Results

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

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

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

- If a PSTN call is placed to Voice Portal while the connection between the AudioCodes gateway and Voice Portal is down, or if no ports are available on Voice Portal, the caller will hear ringback, instead of a busy signal.

- CAS Table selection: Two CAS table were used during compliance testing. Depending on which CAS table is selected, different behaviors can be expected for calls transferred from Voice Portal. Consider the scenario where a call is placed from the PSTN and it arrives at Voice Portal. The call is then transferred from Voice Portal back to the PSTN. When the call is answered at the transferred-to party and voice is detected (due to the **IPMedia Detectors** field being enabled in **Section 6.14**), the AudioCodes gateway will notify Voice Portal that the call has been completed, via a 200 OK.

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

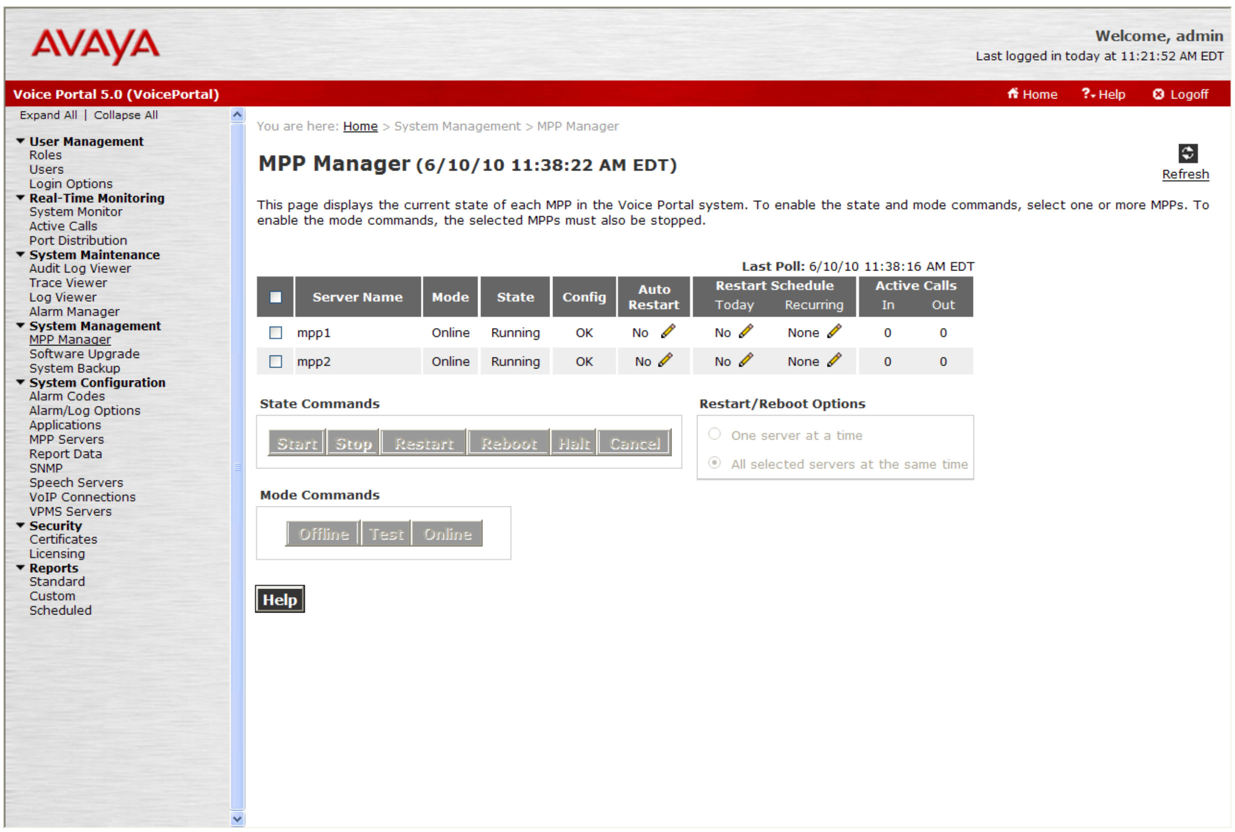
  In the same scenario, where no voice is detected and CAS table **loopstarttable_fxo_Avaya_AutoConnect** is used, the AudioCodes gateway will automatically notify Voice Portal that the second leg call for the transfer has been completed, via a 200 OK. The 200 OK is sent 20 seconds after initiating the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

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

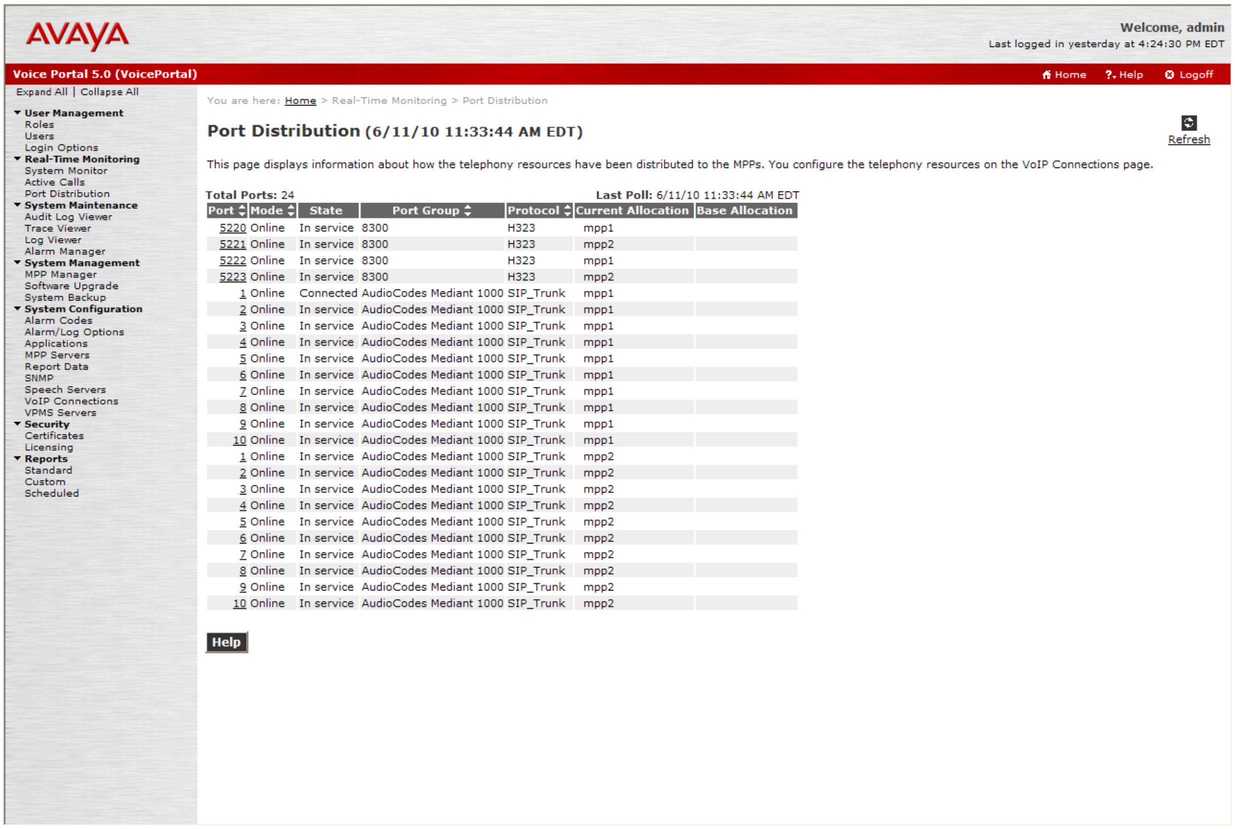
8. Verification Steps

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

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



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



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

9. Conclusion

The AudioCodes Median 1000 VoIP Media Gateway version 6.0 passed compliance testing. These Application Notes describe the configurations required for AudioCodes Mediant 1000 VoIP Media Gateway to successfully interoperate with Avaya Voice Portal (via a SIP trunking interface) and Avaya Aura^{TM} Communication Manager (via a line side E1 interface). Most of the feature and serviceability test cases passed, the failed test cases did not block compliance testing (See **Section 7** for observations noted).

10. Additional References

This section references the product documentation relevant to these Application Notes. The following Avaya product documentation is available at http://support.avaya.com.

[1] *Administering Avaya Aura™ Communication Manager*, Doc ID 03-300509, May 2009.

[2] *Feature Description and Implementation for Avaya Communication Manage*r, Doc ID555-245-205, Issue 7, Release 5.2, May 2009

[3] *Administering Voice Portal,* June 2010.

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

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.