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.

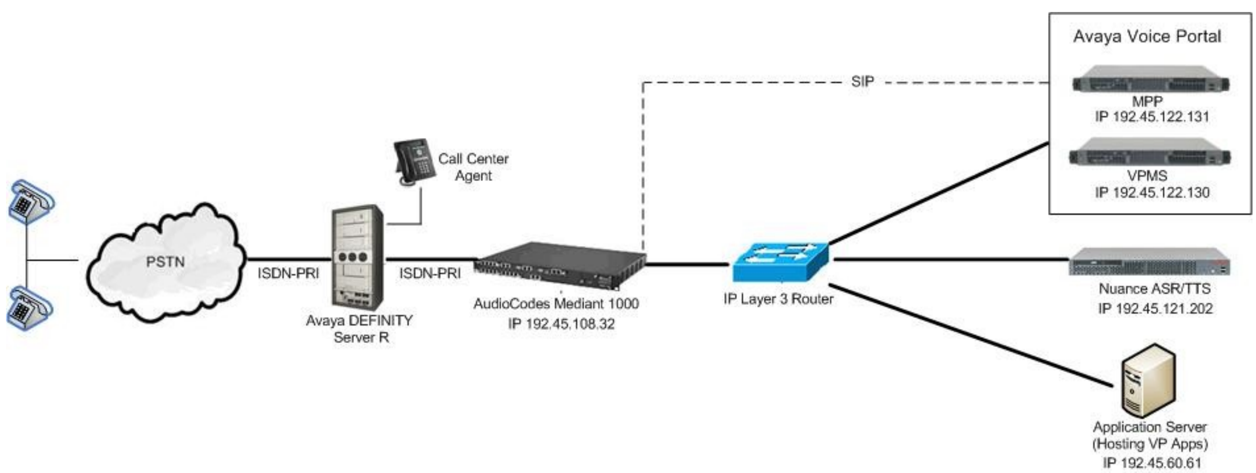


Figure 1: AudioCodes Mediant 1000 VoIP Media Gateway with Avaya Voice Portal Using SIP Trunking Interface

3. Equipment and Software Validated

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

Equipment	Software
Avaya Voice Portal • Voice Portal Management System (VPMS) • Media Processing Platform (MPP)	5.0
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 DEFINITY Server R	R011r.03.1.635.0
Call Center Agent Avaya 1600 Series IP Telephone (H.323)	Avaya one-X® Deskphone 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^{rd} 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
           Enable 'dadmin' Login? n            ISDN Network Call Redirection? y
           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        Multimedia Call Handling (Basic)? y
                       IP Trunks? y     Multimedia Call Handling (Enhanced)? y
                                                      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                Page   4 of  10   SPE B
                                OPTIONAL FEATURES


                    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

    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                                         Page   1 of   2   SPE B
                                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 12                               Page   1 of   5   SPE B
                                SIGNALING GROUP

 Group Number: 12              Group Type: isdn-pri
                    Associated Signaling? y          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
```

4.4. 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 Number: 12                Group Type: isdn          CDR Reports: y
  Group Name: GW PRI                 COR: 1        TN: 1         TAC: 112
   Direction: two-way        Outgoing Display? n        Carrier Medium: PRI/BRI
 Dial Access? n                Busy Threshold: 255       Night Service:
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
```

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 12                                         Page   3 of  22   SPE B
                       INCOMING CALL HANDLING TREATMENT
 Service/       Called    Called        Del  Insert           Per Call  Night
 Feature         Len      Number                              CPN/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-group 12                                         Page   6 of  22   SPE B
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/23
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  23

       Port    Code Sfx Name        Night           Sig Grp
  1: 01C0701  TN464  F                                12
  2: 01C0702  TN464  F                                12
  3: 01C0703  TN464  F                                12
  4: 01C0704  TN464  F                                12
  5: 01C0705  TN464  F                                12
  6: 01C0706  TN464  F                                12
  7: 01C0707  TN464  F                                12
  8: 01C0708  TN464  F                                12
  9: 01C0709  TN464  F                                12
 10: 01C0710  TN464  F                                12
 11: 01C0711  TN464  F                                12
 12: 01C0712  TN464  F                                12
 13: 01C0713  TN464  F                                12
 14: 01C0714  TN464  F                                12
 15: 01C0715  TN464  F                                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
                    Associated Signaling? y          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 2122960                             Page   1 of   2   SPE B
                             AAR DIGIT ANALYSIS TABLE
                                                            Percent Full:    7

          Dialed            Total     Route     Call   Node   ANI
          String          Min  Max   Pattern    Type   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 ***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.

```
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:                                                                        n   user
 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
 1: y y y y y n  n            bothept                                        none
 2: y y y y y n  n            rest                                           none
 3: y y y y y n  n            rest                                           none
 4: y y y y y n  n            rest                                           none
 5: y y y y y n  n            rest                                           none
 6: y y y y y n  n            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 0                                    Page   1 of   2   SPE B
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    7

          Dialed           Total     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
```

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 ***1*** 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 ***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.

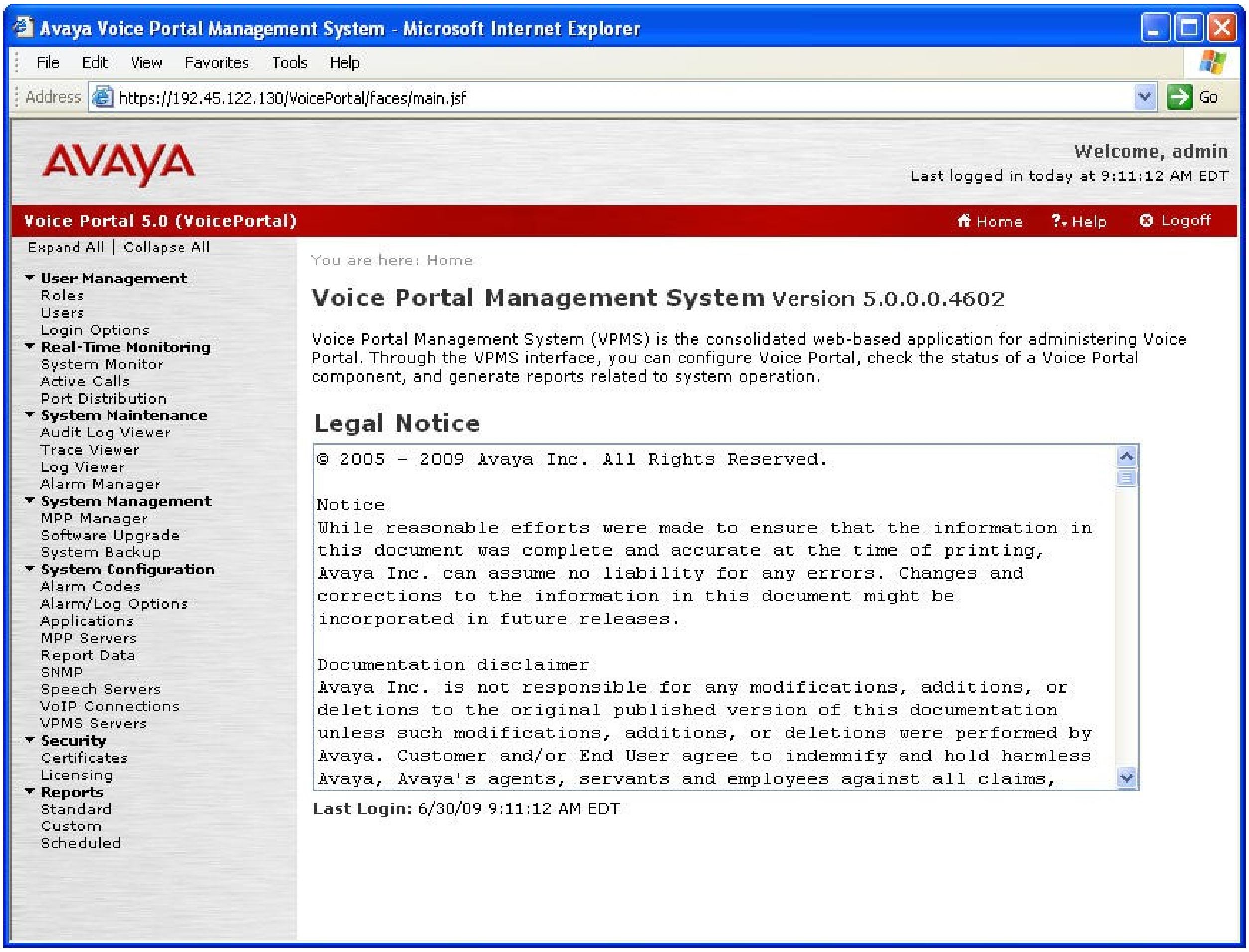
```
change route-pattern 2                                         Page   1 of   3   SPE B
                    Pattern Number: 2   Pattern Name: G3R1-InHseSw

    Grp FRL NPA Pfx Hop Toll No.  Inserted                                 DCS/ IXC
    No          Mrk Lmt List Del  Digits                                   QSIG
                             Dgts                                          Intw
 1: 1    0       1                                                          n   user
 2:                                                                         n   user
 3:                                                                         n   user
 4:                                                                         n   user
 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
 1: y y y y y n  n            bothept                                          none
 2: y y y y y n  n            rest                                             none
 3: y y y y y n  n            rest                                             none
 4: y y y y y n  n            rest                                             none
 5: y y y y y n  n            rest                                             none
 6: y y y y y n  n            rest                                             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

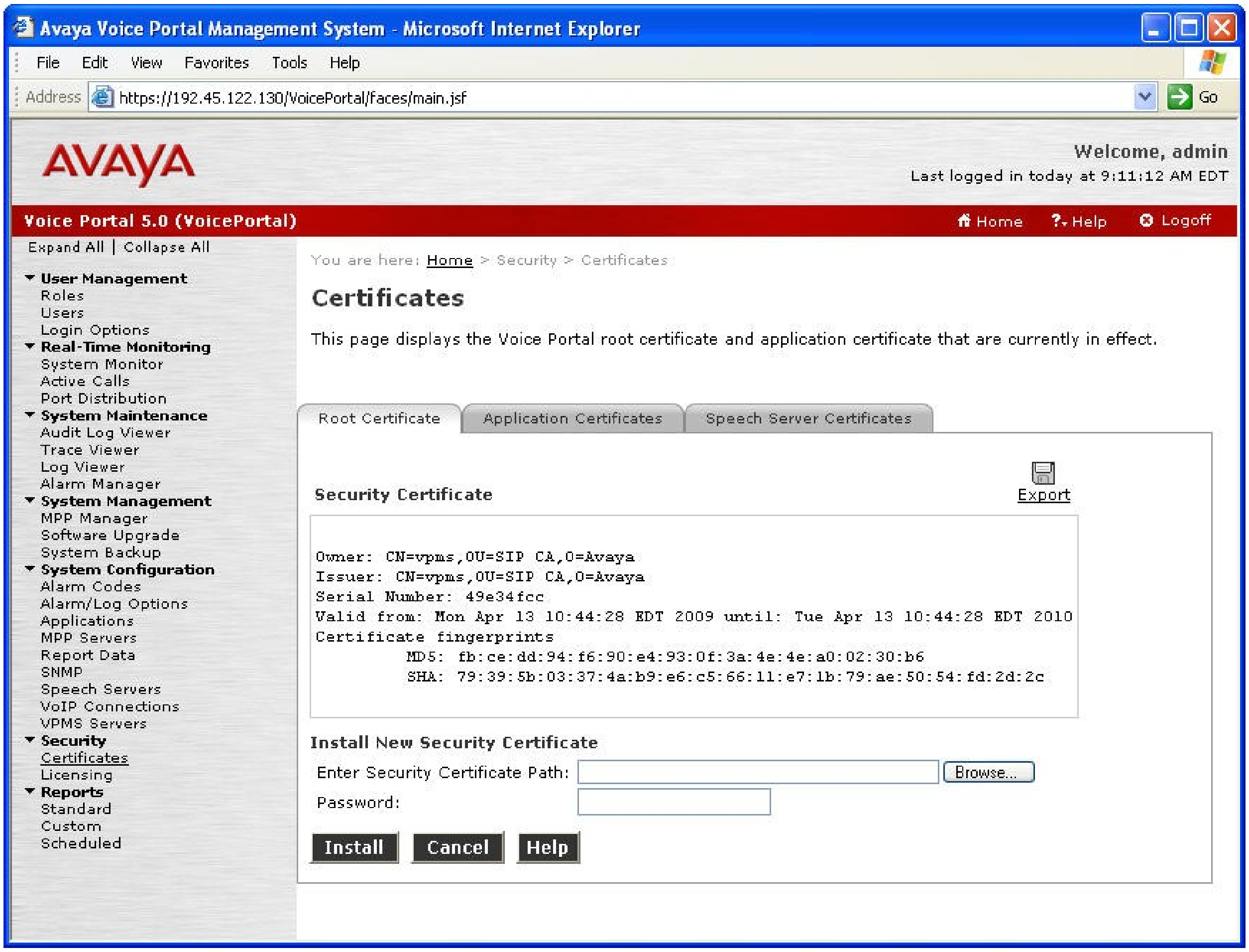
Avaya Voice Portal is configured via the Voice Portal Management System (VPMS) web interface. To access the web interface, enter `http://<ip-addr>/VoicePortal` as the URL in an Internet browser, where `<ip-addr>` is the IP address assigned to the VPMS server. Log in using the Administrator user role.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



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

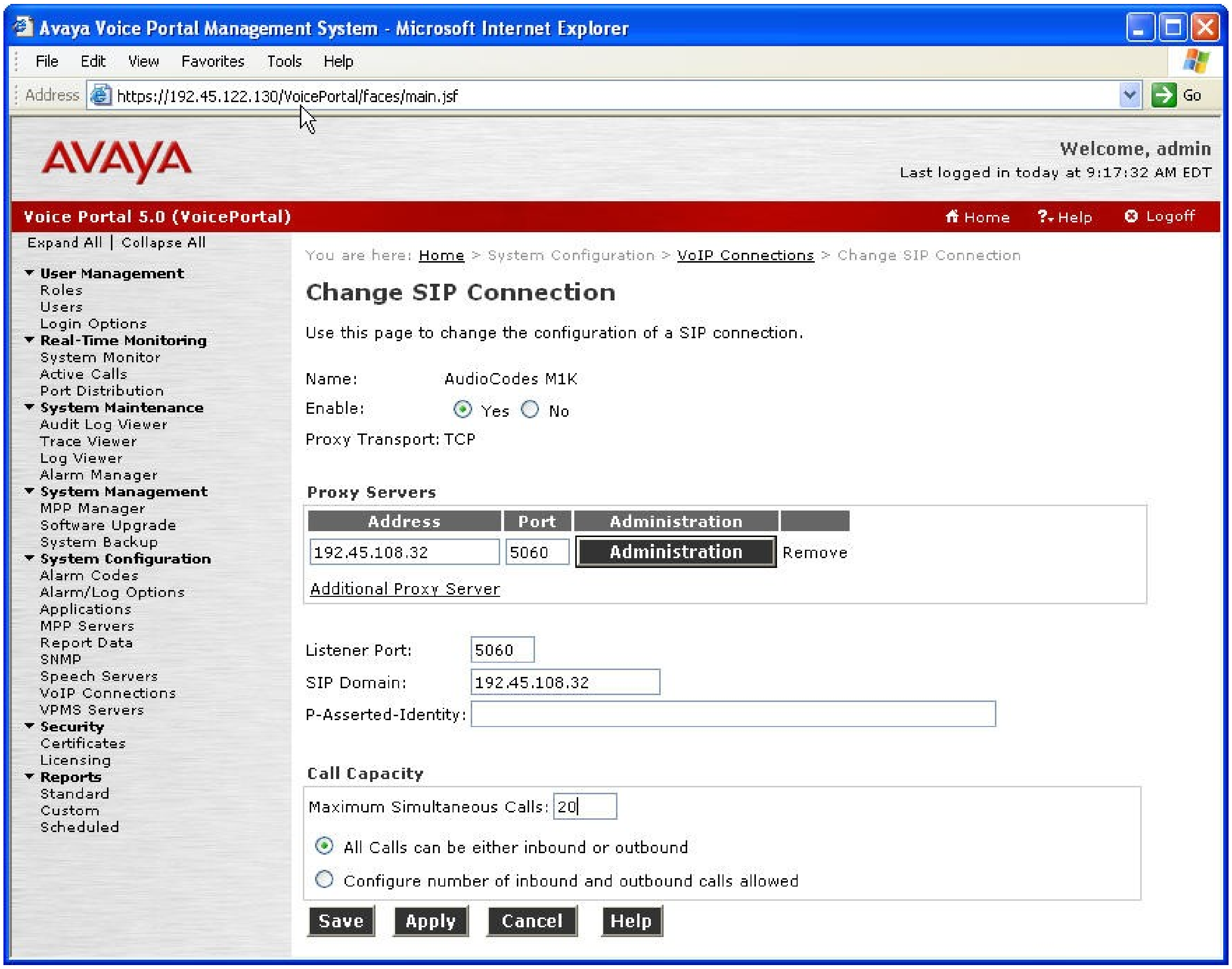
In the compliance test, Avaya Voice Portal was configured to use TCP on SIP interface to the AudioCodes Mediant 1000 VoIP Media Gateway (to facilitate debugging). A production environment is more likely to use TLS authentication over the SIP interface between Avaya Voice Portal and Mediant 1000. To install 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

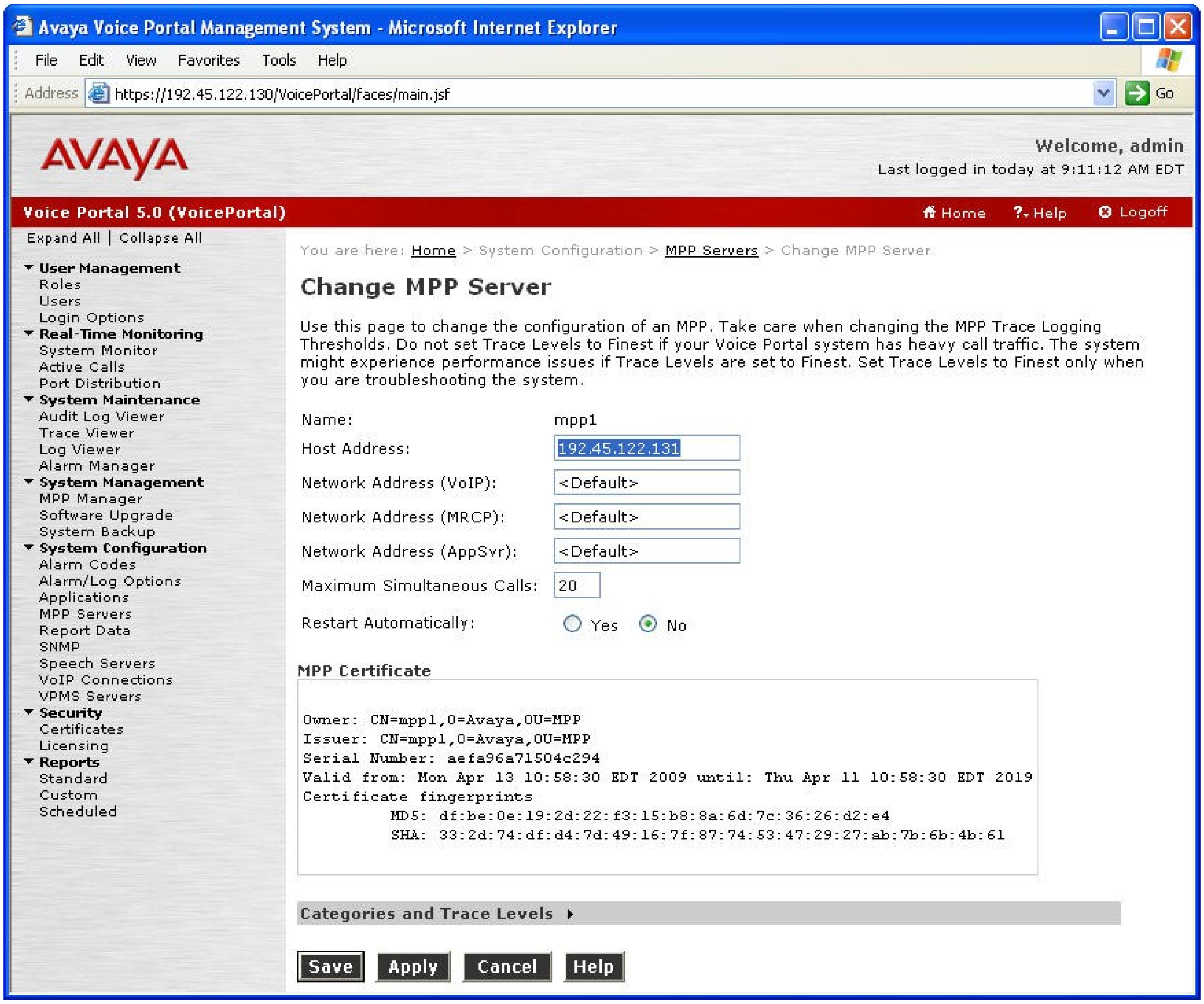
To configure a SIP connection to the AudioCodes Mediant 1000 VoIP Media Gateway, navigate to **System Configuration → VoIP Connections**, click on the **SIP** tab. The SIP tab is displayed as shown below. Configure the parameters as follows:

- 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



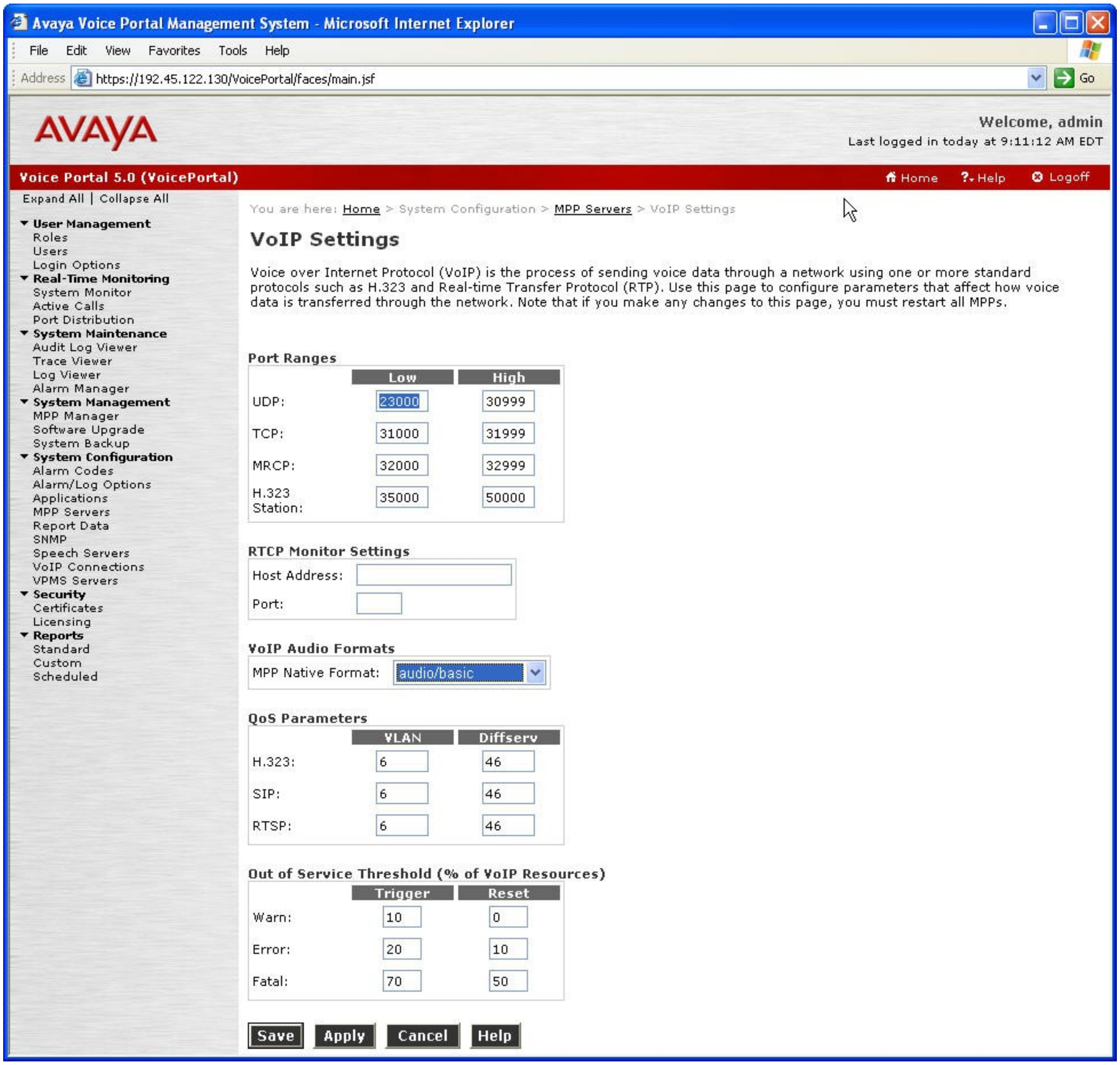
5.3. Add MPP server

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..

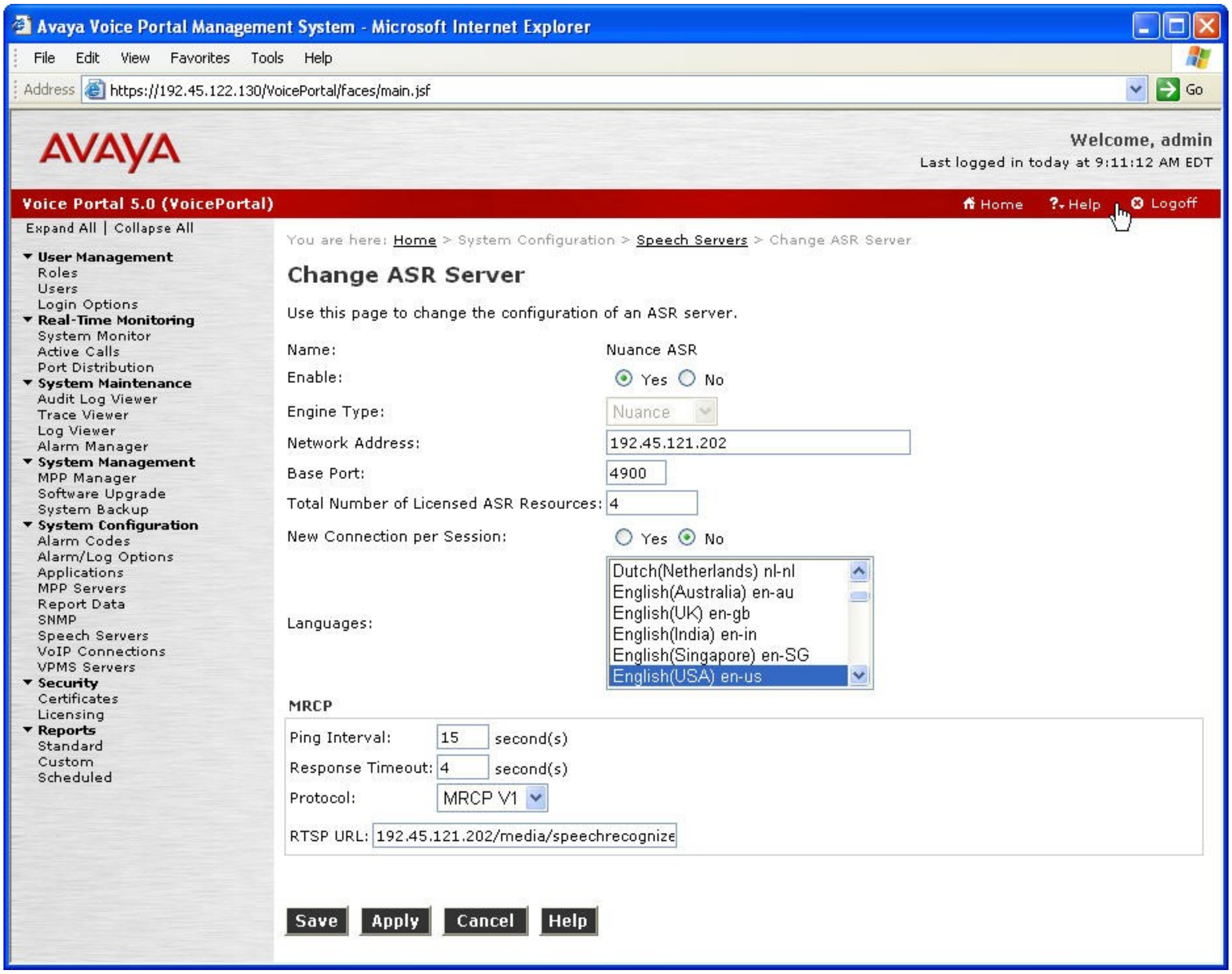


5.5. Add Speech Server

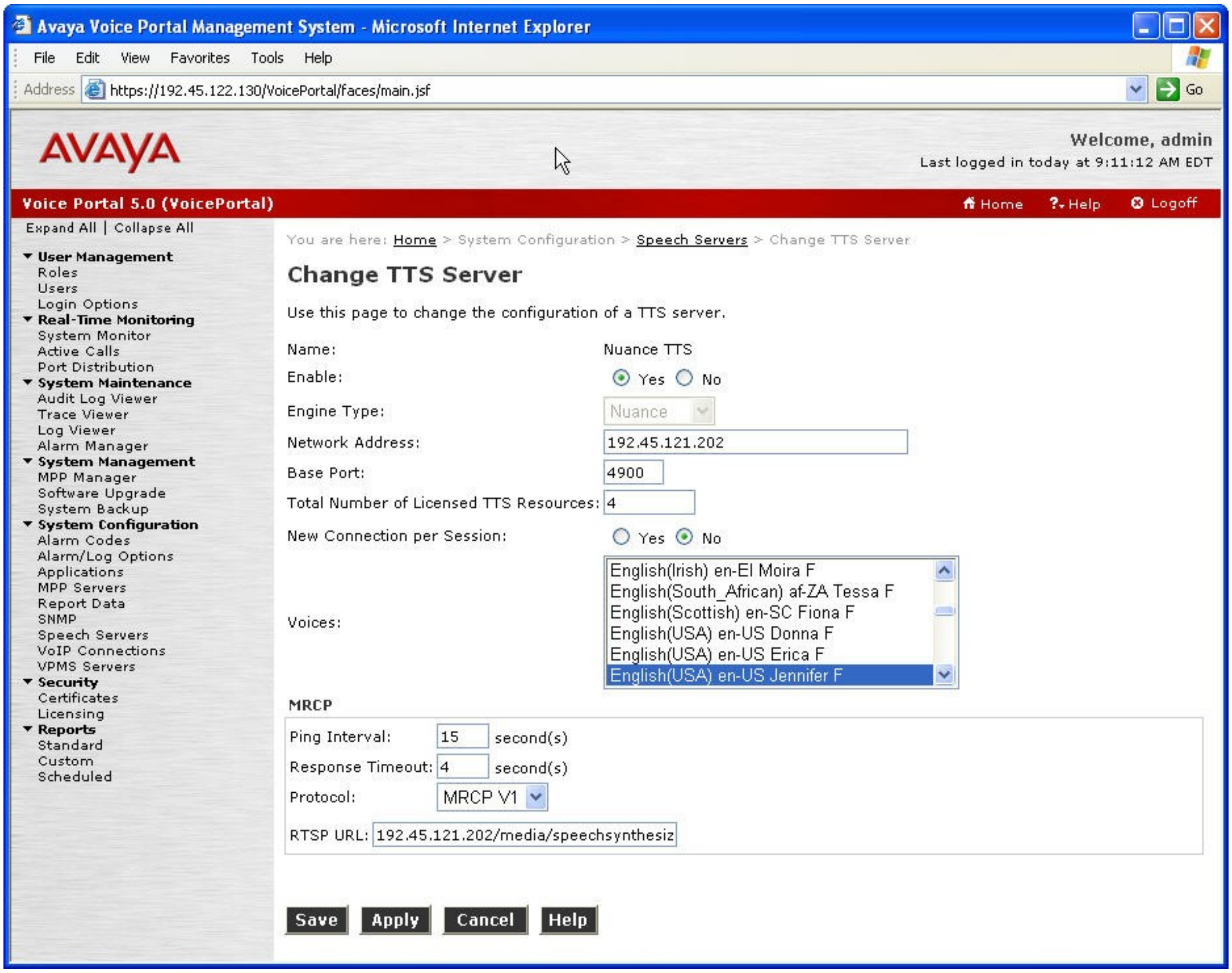
Adding a speech server for providing ASR (Automatic Speech Recognition) and/or TTS (Text To Speech) services is part of the standard configuration for Avaya Voice Portal; this configuration is not directly related to achieving interoperability between AudioCodes Mediant 1000 VoIP Media Gateway and Avaya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



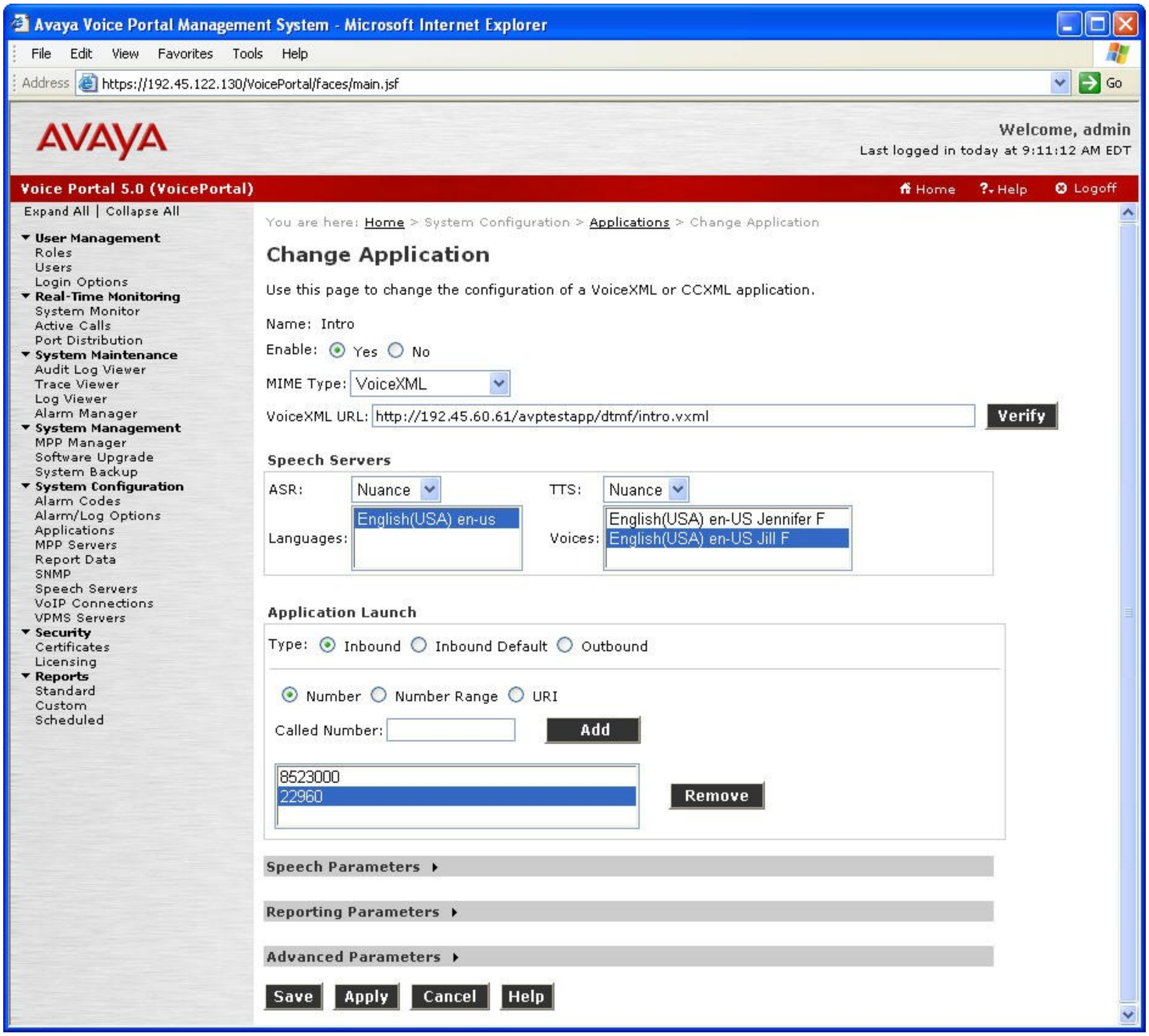
To configure the TTS server, navigate to **System Configuration → Speech Servers**, select the **TTS** tab, and then click **Add**. The screen below shows the configuration for the TTS server used in the compliance test.. In this configuration, a Nuance TTS server was used so the engine type was set 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

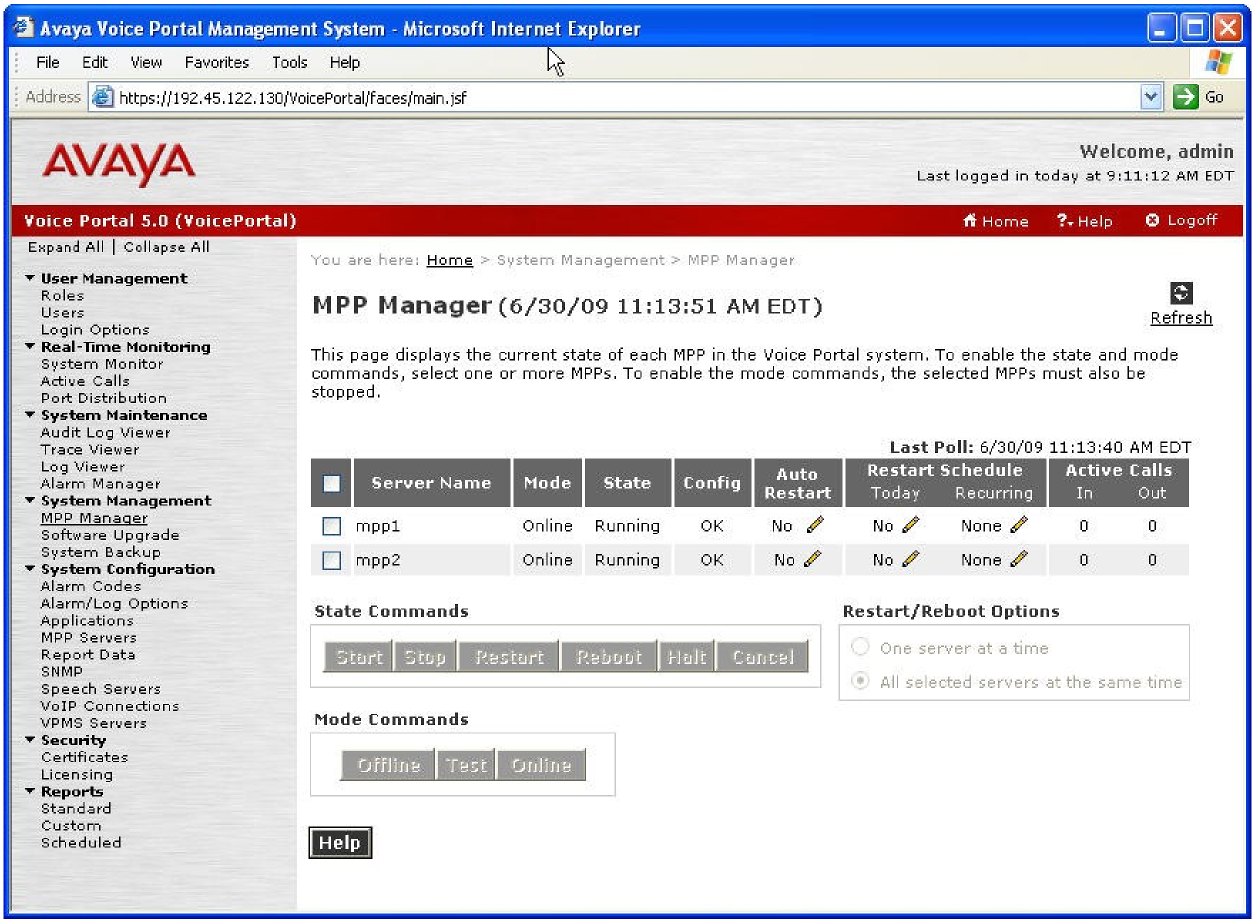
Adding a voice application for Avaya Voice Portal is part of Voice Portal's standard administration; this configuration is not directly related to achieving interoperability between AudioCodes Mediant 1000 VoIP Media Gateway and Avaya Voice Portal. It is included here for completeness.

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 → 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 Management System - Microsoft Internet Explorer
File Edit View Favorites Tools Help
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
User Management
Roles
Users
Login Options
Real-Time Monitoring
System Monitor
Active Calls
Port Distribution
System Maintenance
Audit Log Viewer
Trace Viewer
Log Viewer
Alarm Manager
System Management
MPP Manager
Software Upgrade
System Backup
System Configuration
Alarm Codes
Alarm/Log Options
Applications
MPP Servers
Report Data
SNMP
Speech Servers
VoIP Connections
VPMS Servers
Security
Certificates
Licensing
Reports
Standard
Custom
Scheduled
You are here: Home > System Management > MPP Manager
MPP Manager (6/30/09 11:13:51 AM EDT)
Refresh
This page displays the current state of each MPP in the Voice Portal system. To enable the state and mode commands, select one or more MPPs. To enable the mode commands, the selected MPPs must also be stopped.
Last Poll: 6/30/09 11:13:40 AM EDT
Server Name | Mode | State | Config | Auto Restart | Restart Schedule Today | Recurring | Active Calls In | Out
mpp1 | Online | Running | OK | No | No | None | 0 | 0
mpp2 | Online | Running | OK | No | No | None | 0 | 0
State Commands
Start Stop Restart Reboot Halt Cancel
Restart/Reboot Options
One server at a time
All selected servers at the same time
Mode Commands
Offline Test Online
Help

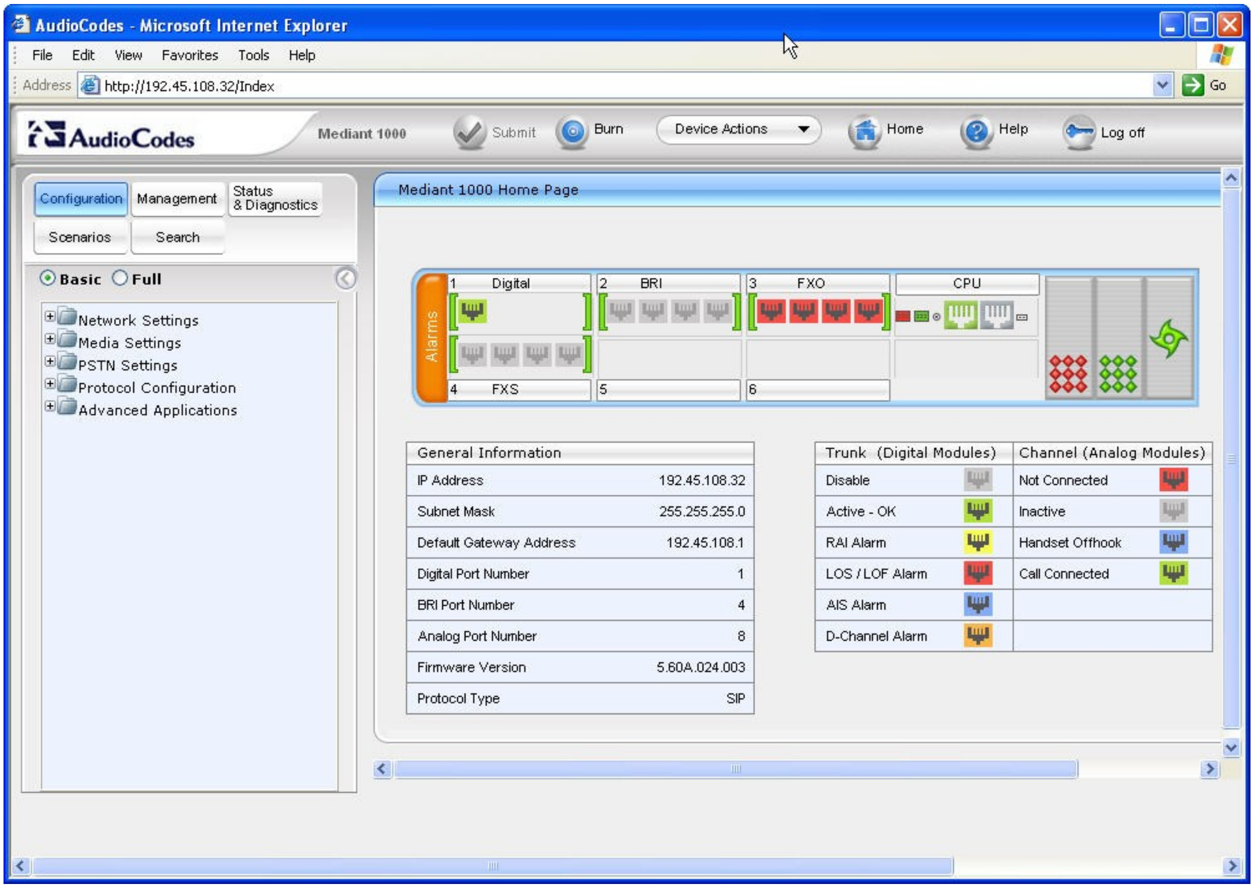
6. Configure Mediant 1000

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:

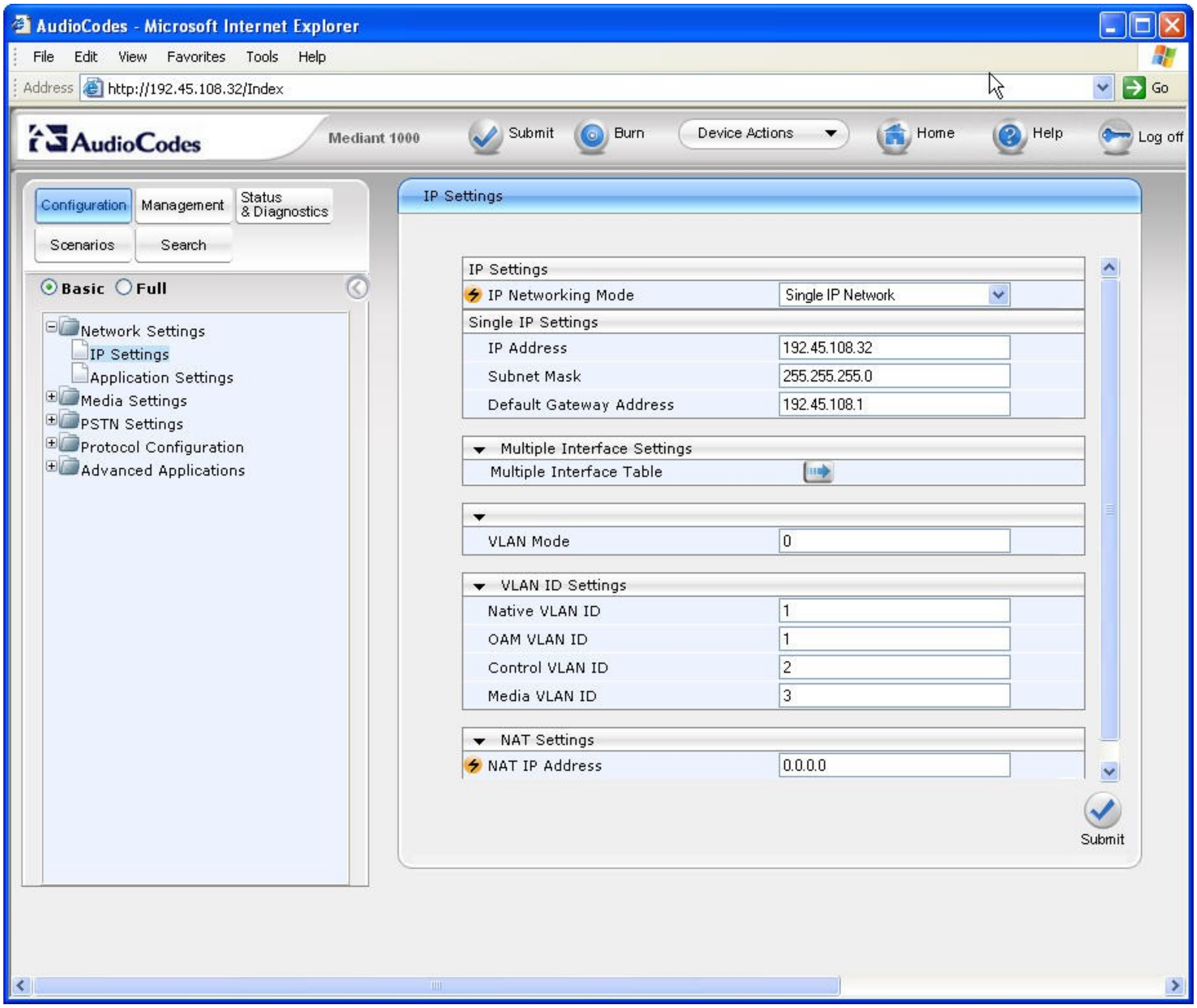
- Network IP settings
- PSTN trunk settings
- SIP General parameters
- SIP Advanced parameters
- SIP Proxy and Registration
- Proxy Sets table
- Coders
- DTMF and Dialing
- Trunk Group
- IP to trunk group routing
- Media voice settings

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.

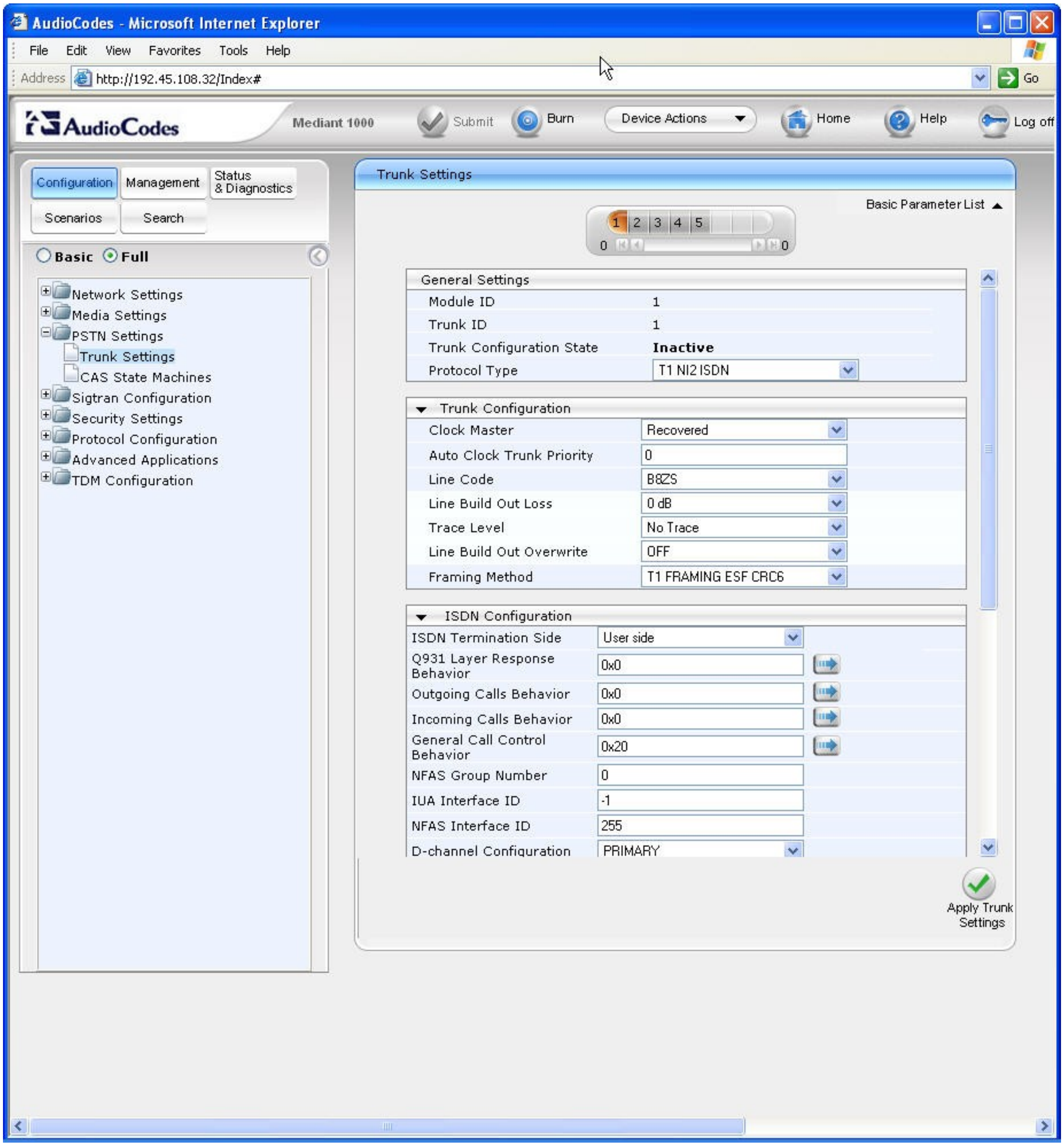


6.2. PSTN trunk setting

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.

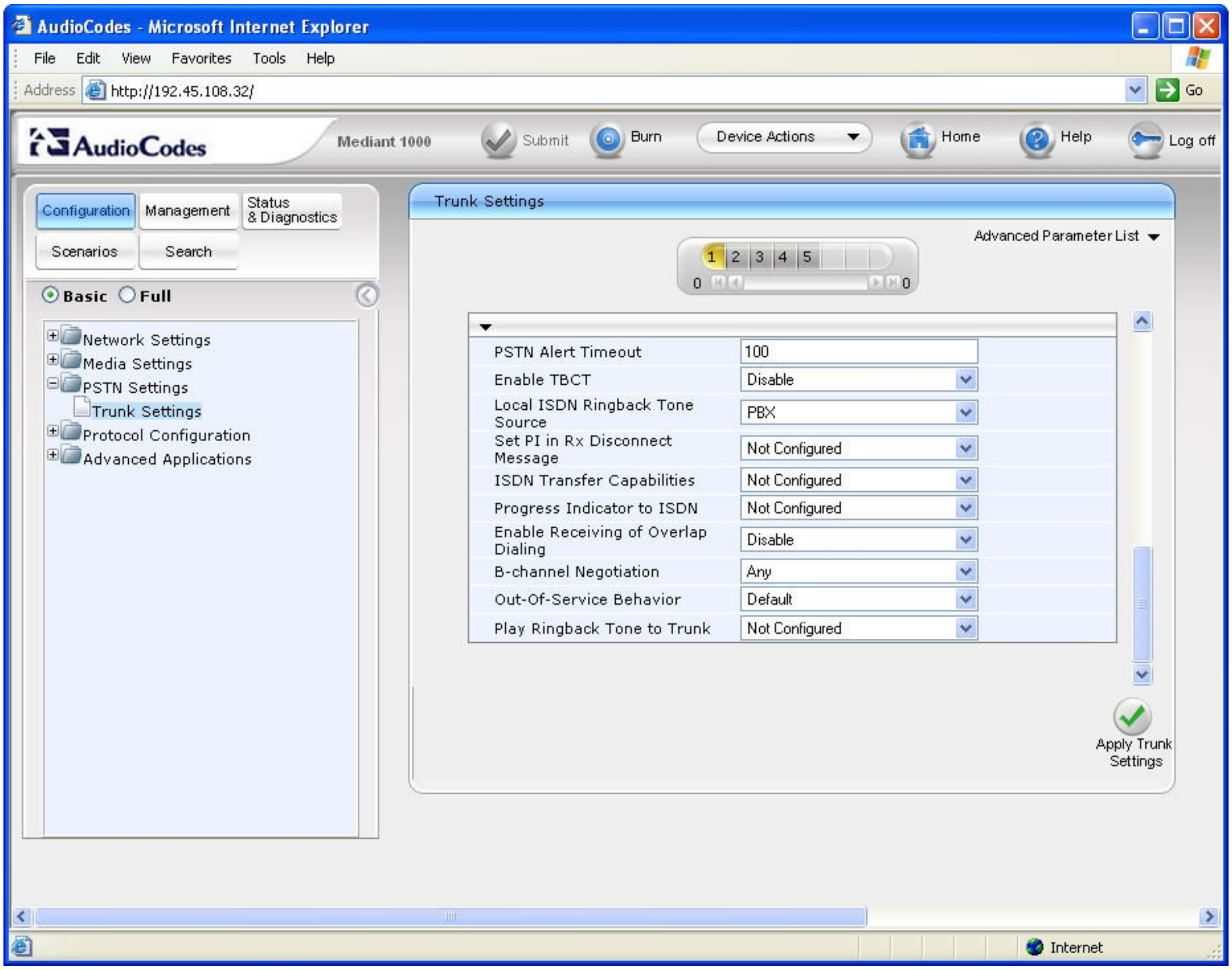
- The **ISDN Termination Side** was set to ***User side***. This setting means the clock for the T1 trunk synchronization will be recovered from the trunk. The Avaya DEFINITY Server R side of the link was set to ***line-side*** (see **Section 4.2**).



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.



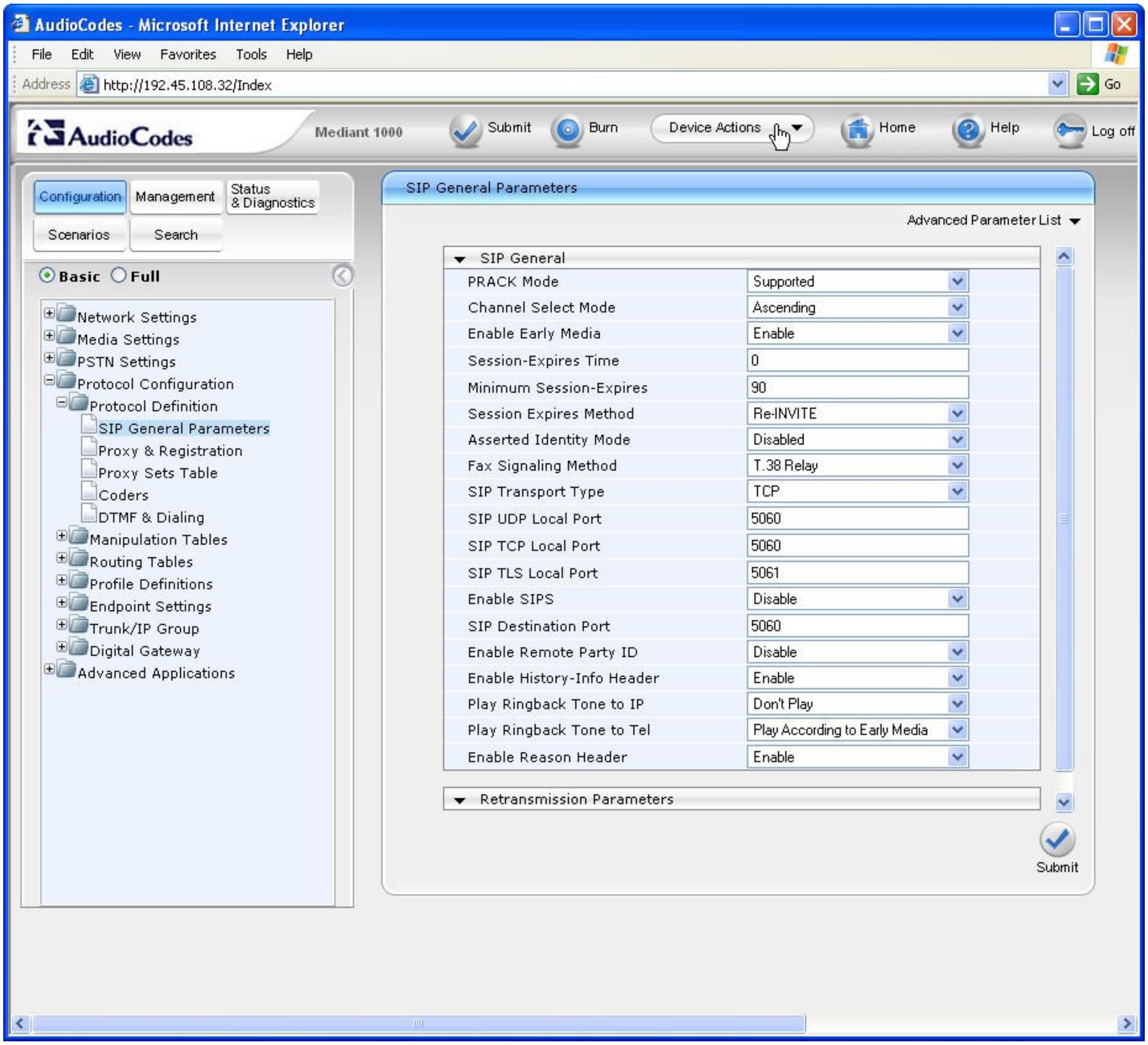
After all the parameters are properly specified, click the **Apply Trunk Settings** icon button at the bottom of the screen.

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.

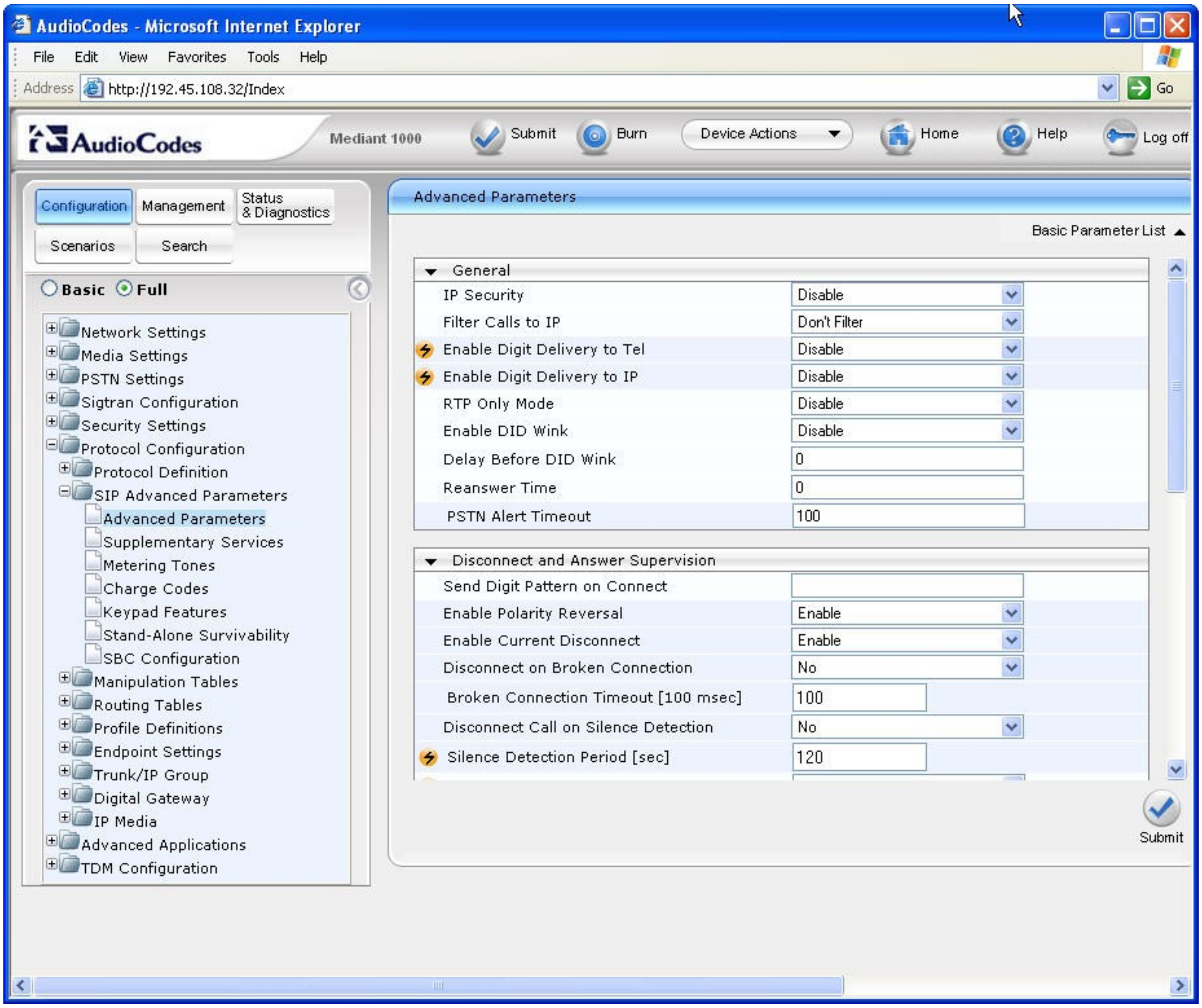


6.4. SIP Advanced Parameters

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.

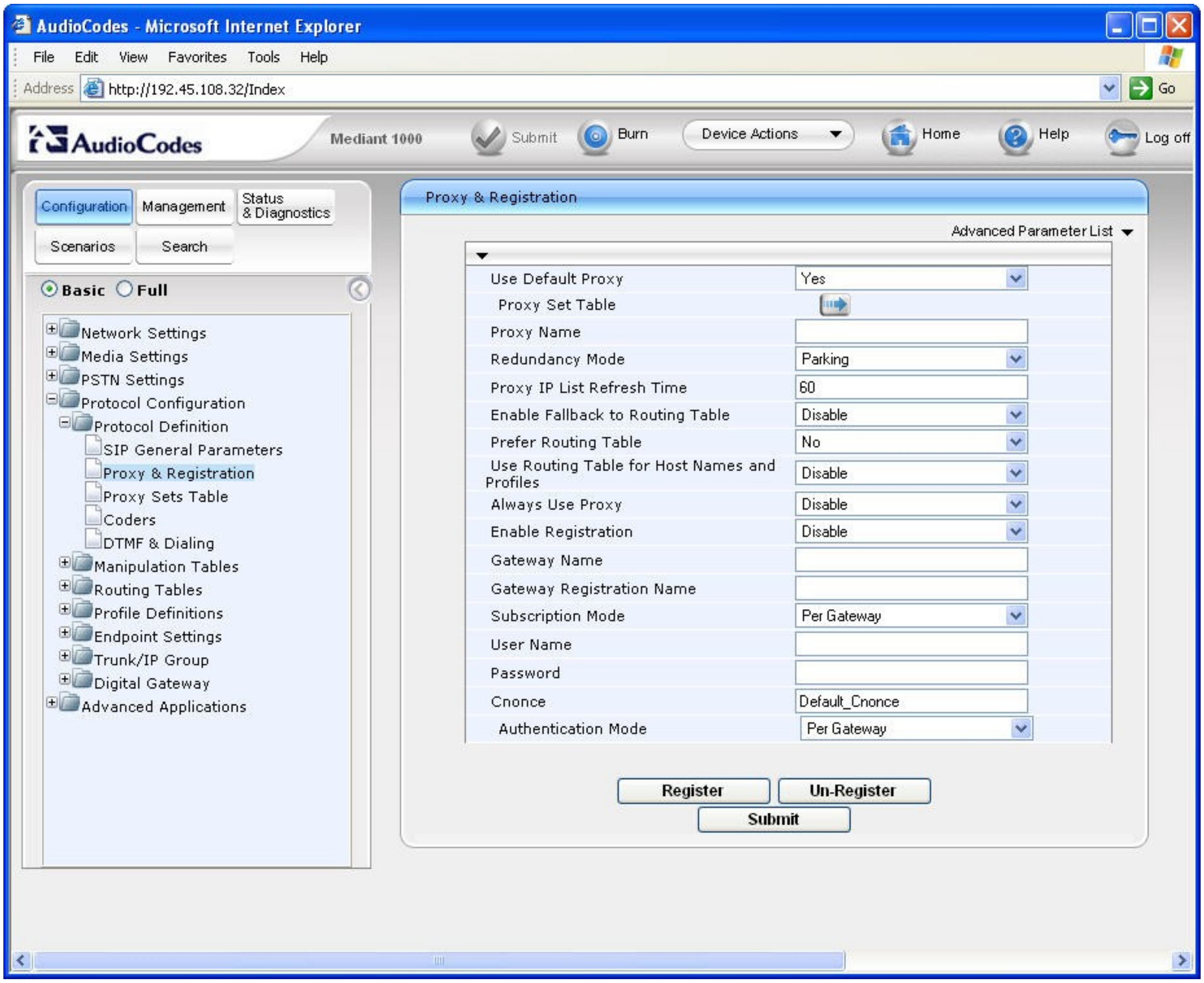
- Specify ***100*** (seconds) for **PSTN Alert Timeout**. This timeout setting on the gateway is for disconnecting unanswered calls on the PSTN side.
- Scroll down and set the **Max Number of Active Calls** field to an appropriate value (not shown).

Default values may be retained for all other fields.



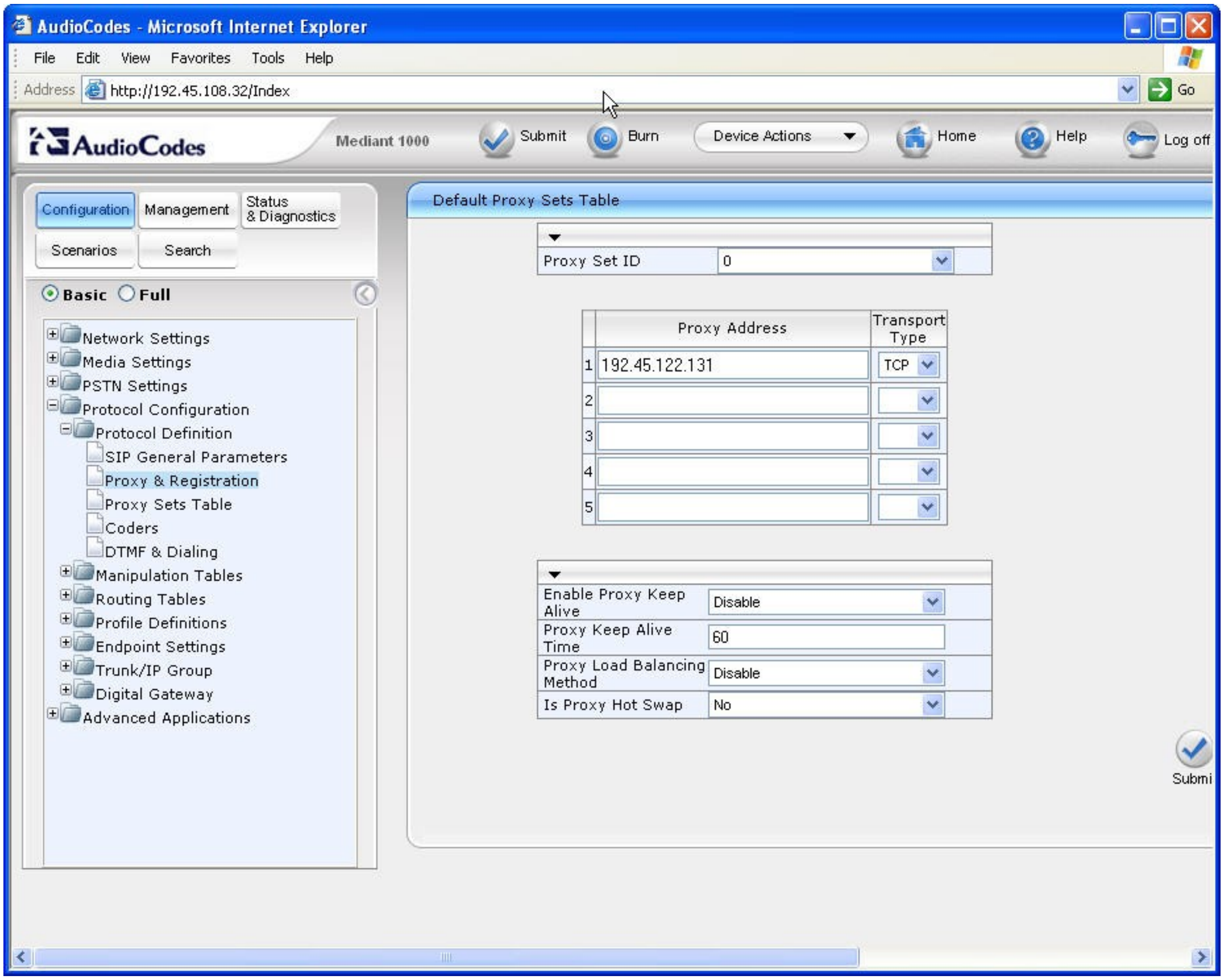
6.5. SIP Proxy and Registration

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 → Proxy & Registration**. Select ***Yes*** for the **Use Default Proxy** field. Default values may be retained for all other fields.



6.6. Proxy Sets Table

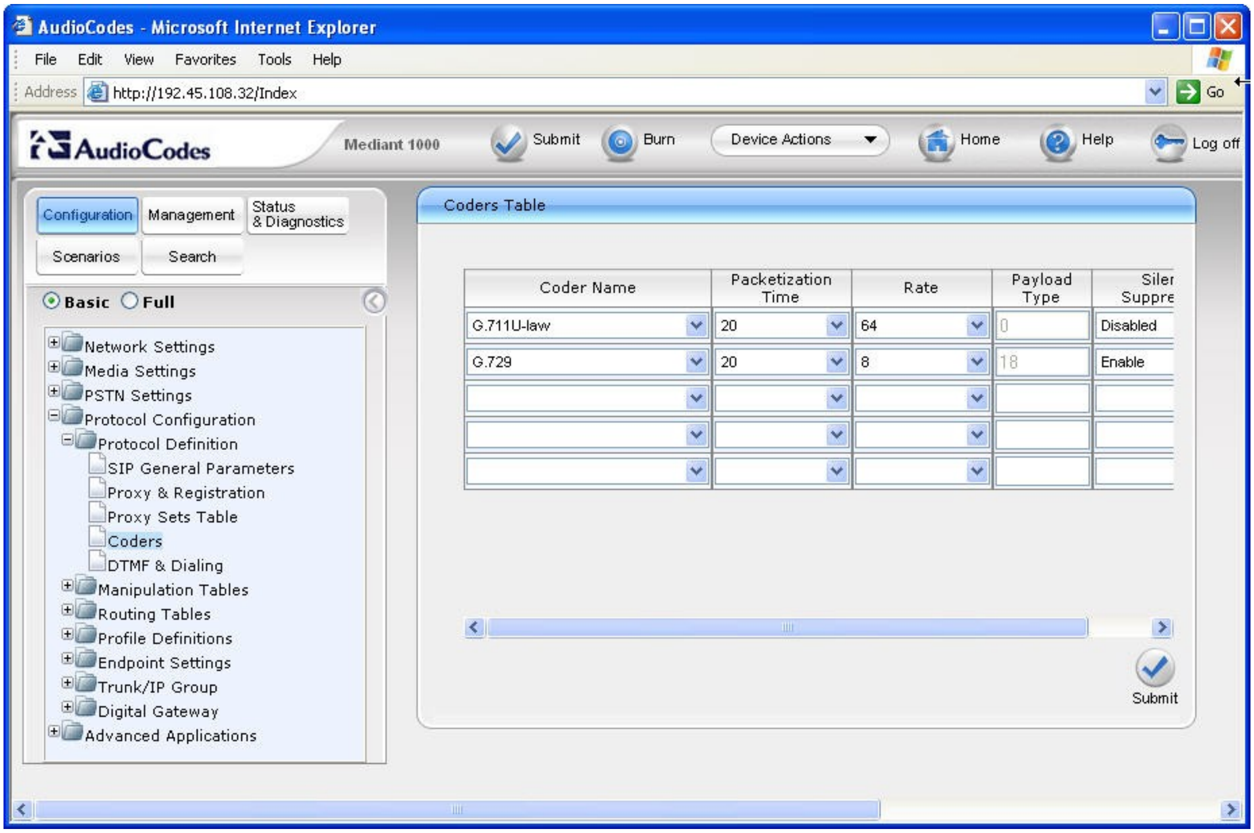
Click the right arrow icon button in the upper part of the **Proxy & Registration** page above to reach the Default Proxy Sets Table configuration page. Enter the IP address assigned to the Avaya Voice Portal MPP server for **Proxy Address**, and ***TCP*** for **Transport Type**. Default values may be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

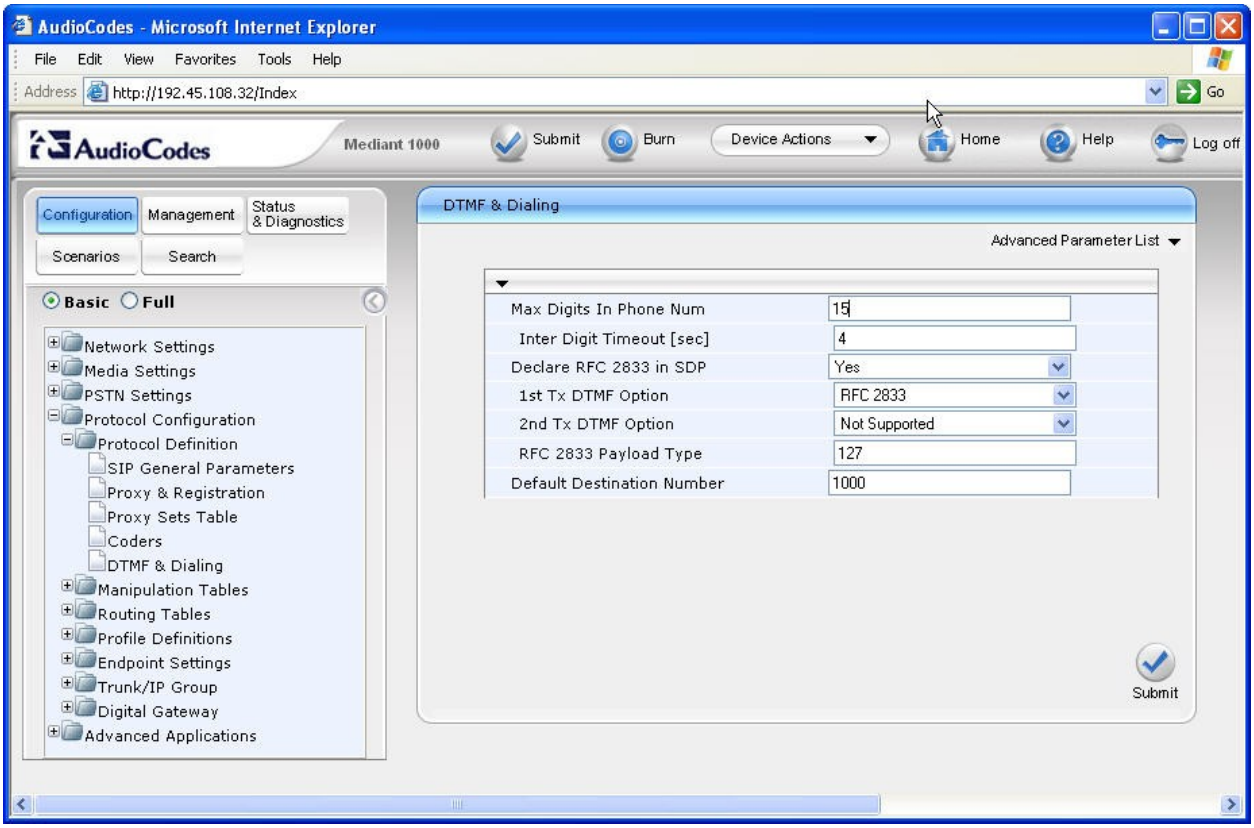


6.8. DTMF and Dialing

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**.

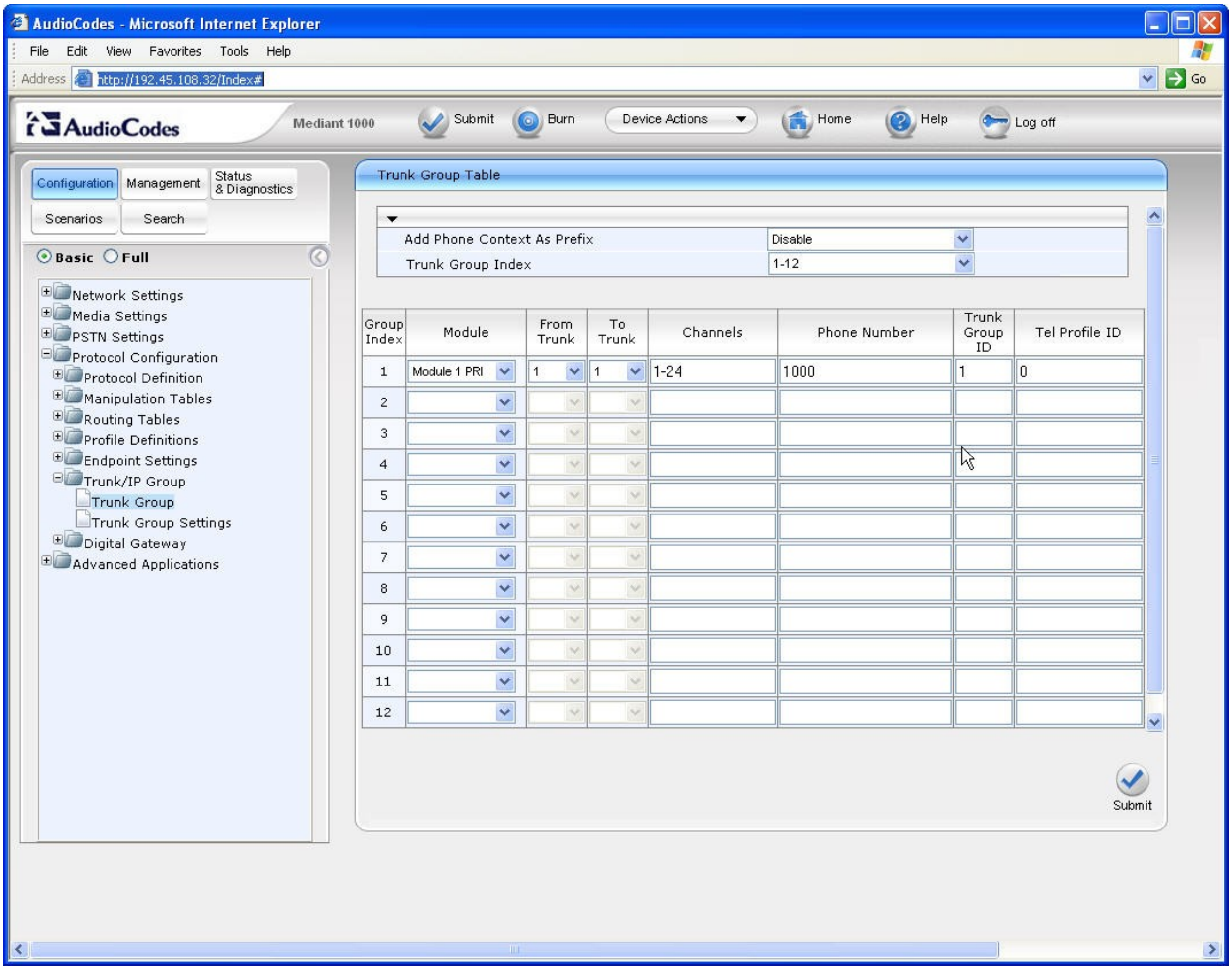
Default values may be retained for all other fields.



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.

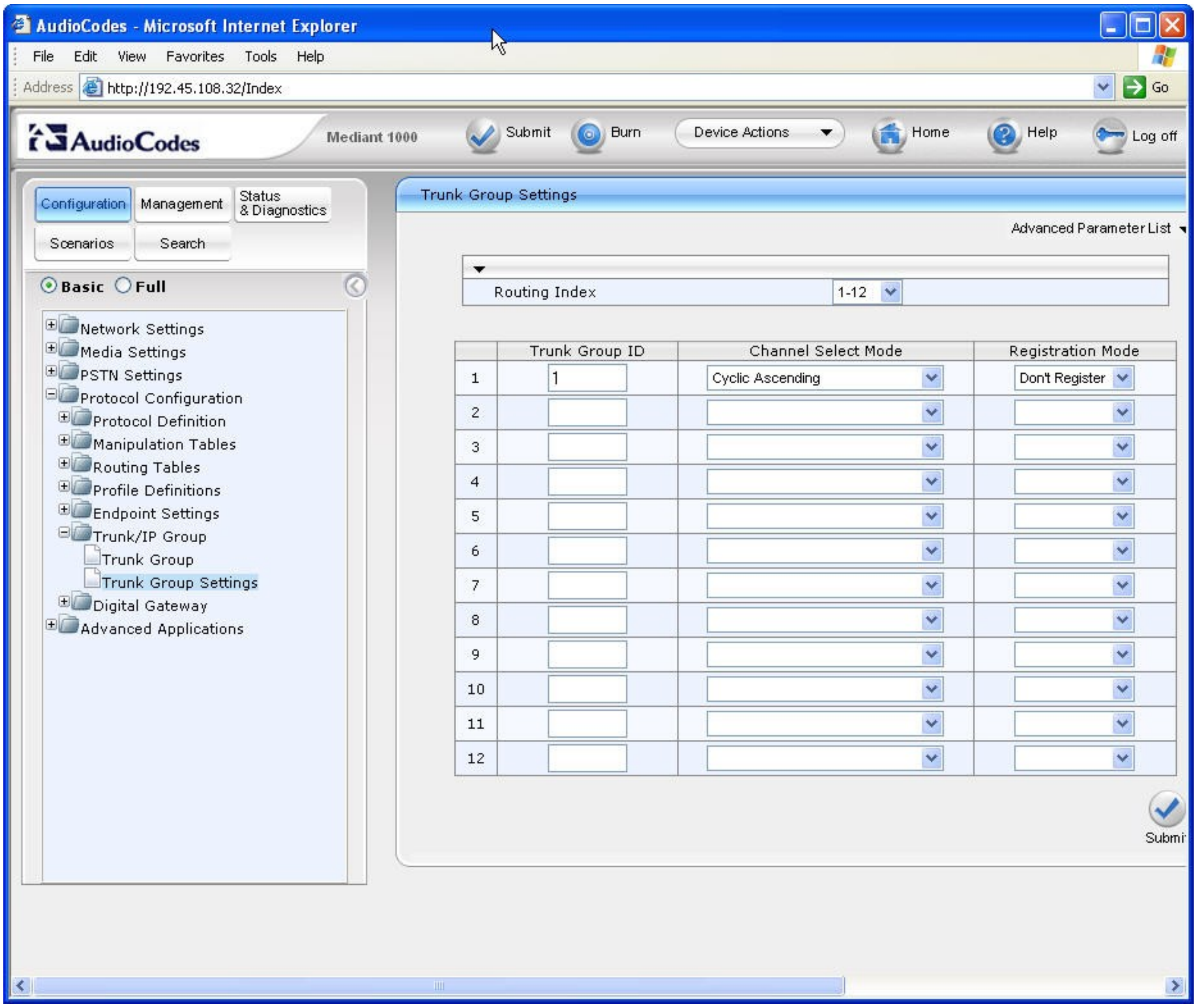
In the example below, the table entry assigns channels ***1 – 24*** of trunk ***1*** to Trunk Group ***1.*** A range of numbers arbitrary chosen to start at ***1000*** will be used for the originating calling party number if no calling party information is received from the originating PSTN trunk.



6.10. Trunk Group Settings

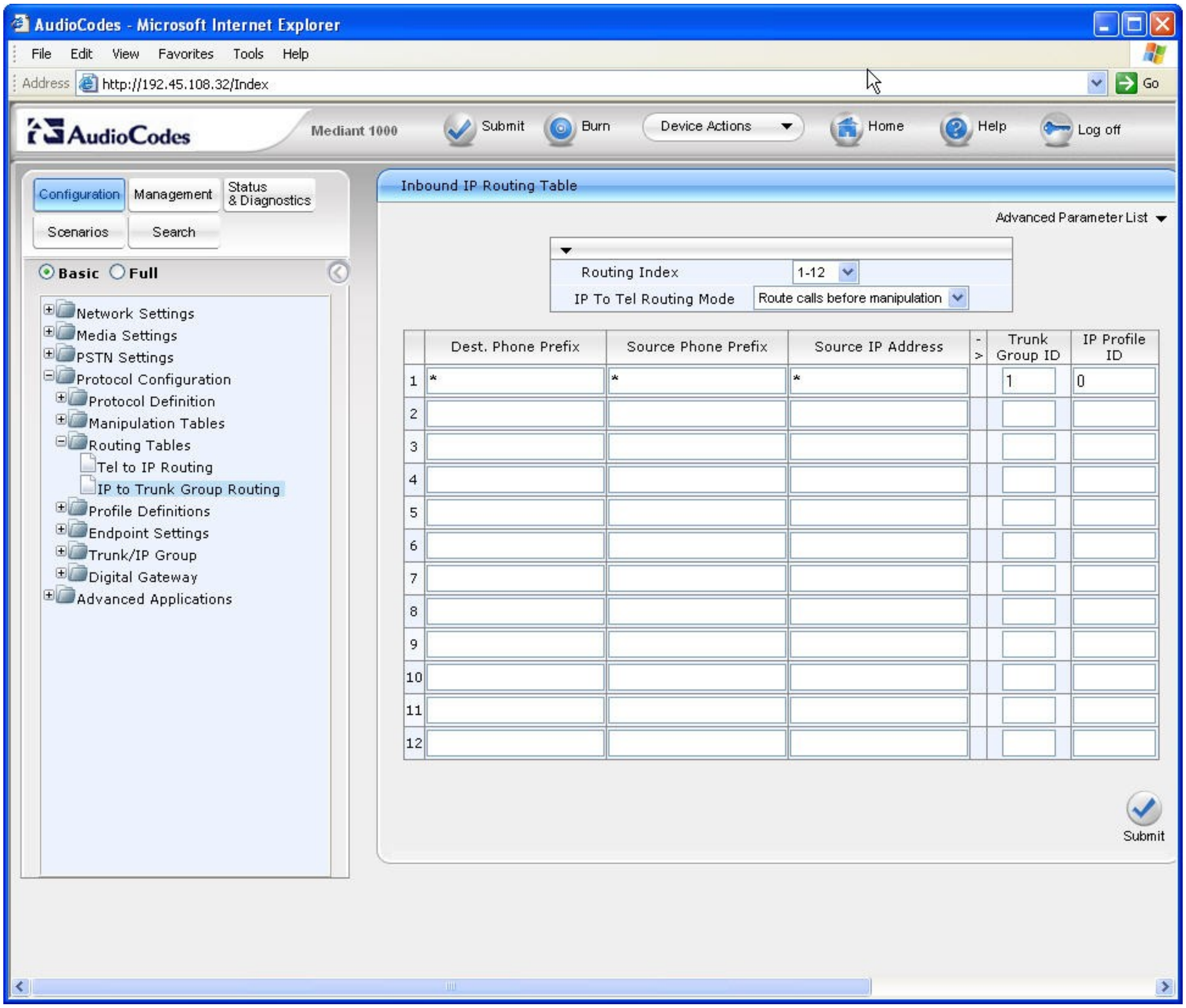
Navigate to **Protocol Configuration → Trunk/IP Group → Trunk Group Settings**. Configure the parameters as described below.

- 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

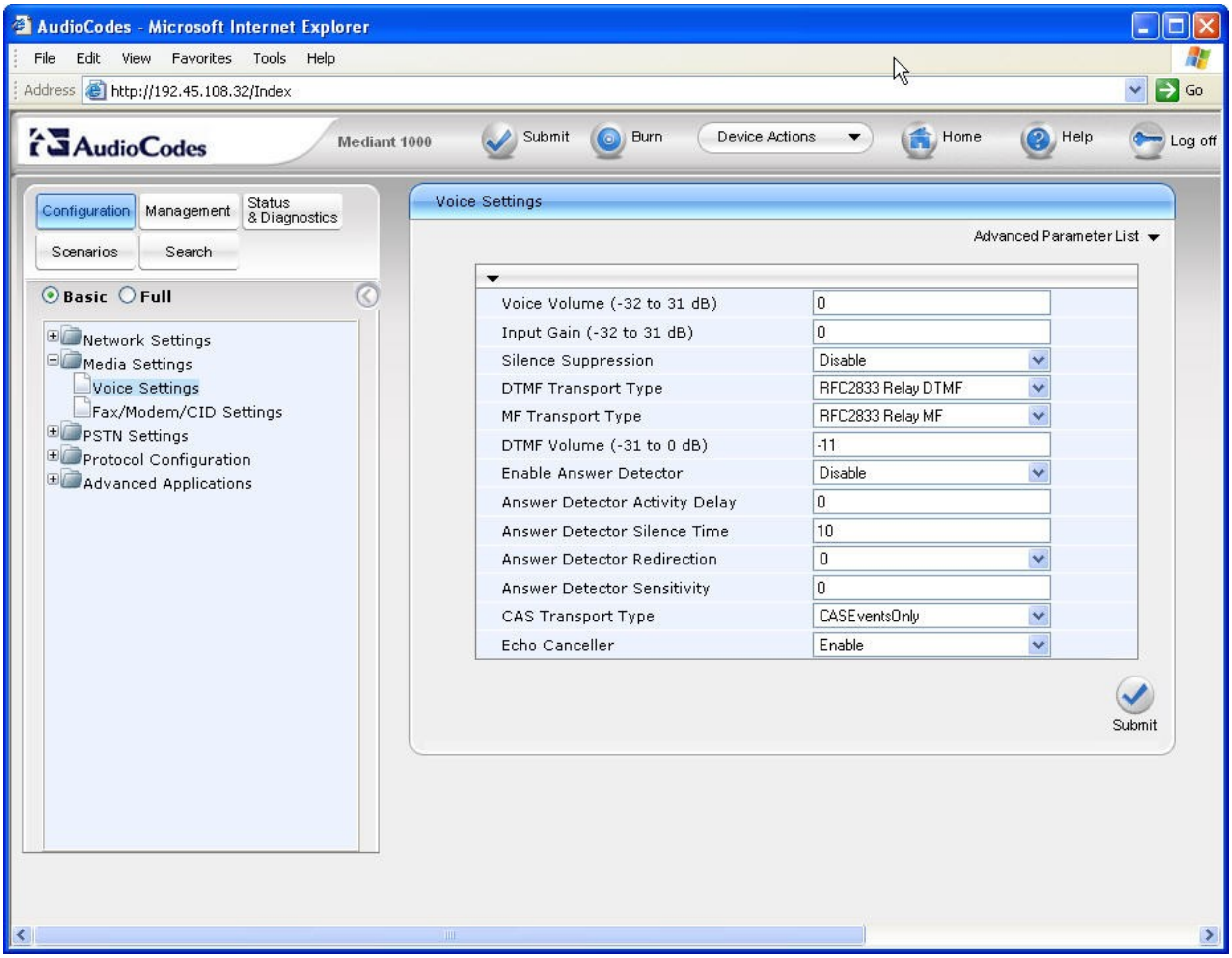
Navigate to **Protocol Configuration → Routing Tables → IP to Trunk Group Routing**. The Inbound IP Routing Table defines the mapping of IP calls to the trunk group created in **Section 6.9**. The **Dest. Phone Prefix**, **Source Phone Prefix** and **Source IP Address** columns define which calls are 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.


AudioCodes - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Address http://192.45.108.32/Index
Go
AudioCodes
Mediant 1000
Submit
Burn
Device Actions
Home
Help
Log off
Configuration
Management
Status & Diagnostics
Scenarios
Search
Basic
Full
Network Settings
Media Settings
Voice Settings
Fax/Modem/CID Settings
PSTN Settings
Protocol Configuration
Advanced Applications
Voice Settings
Advanced Parameter List
Voice Volume (-32 to 31 dB) 0
Input Gain (-32 to 31 dB) 0
Silence Suppression Disable
DTMF Transport Type RFC2833 Relay DTMF
MF Transport Type RFC2833 Relay MF
DTMF Volume (-31 to 0 dB) -11
Enable Answer Detector Disable
Answer Detector Activity Delay 0
Answer Detector Silence Time 10
Answer Detector Redirection 0
Answer Detector Sensitivity 0
CAS Transport Type CASEventsOnly
Echo Canceller Enable
Submit

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:

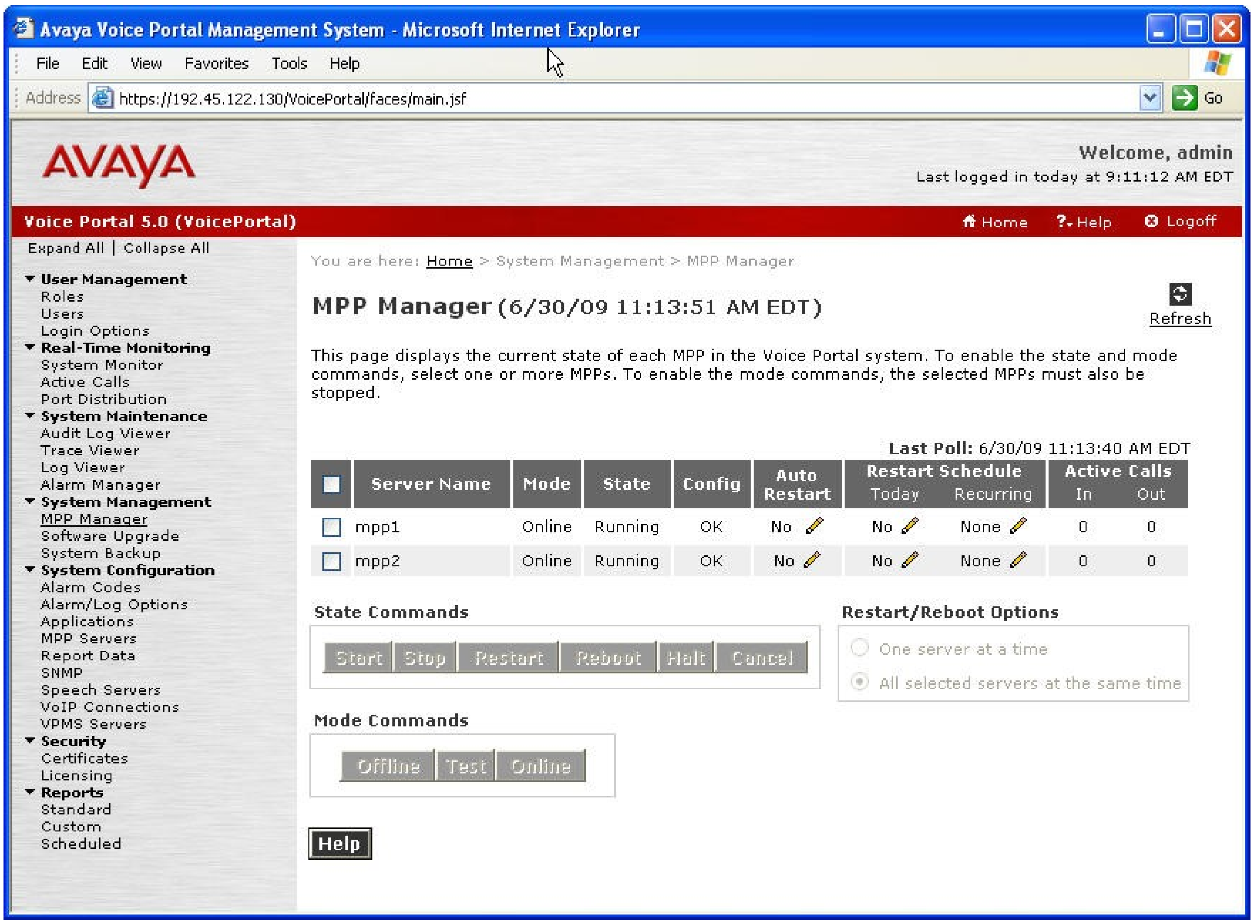
- Mediant 1000 does not forward User-to-User Information (UUI) received from Avaya Voice Portal over the ISDN interface to the far end.

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.

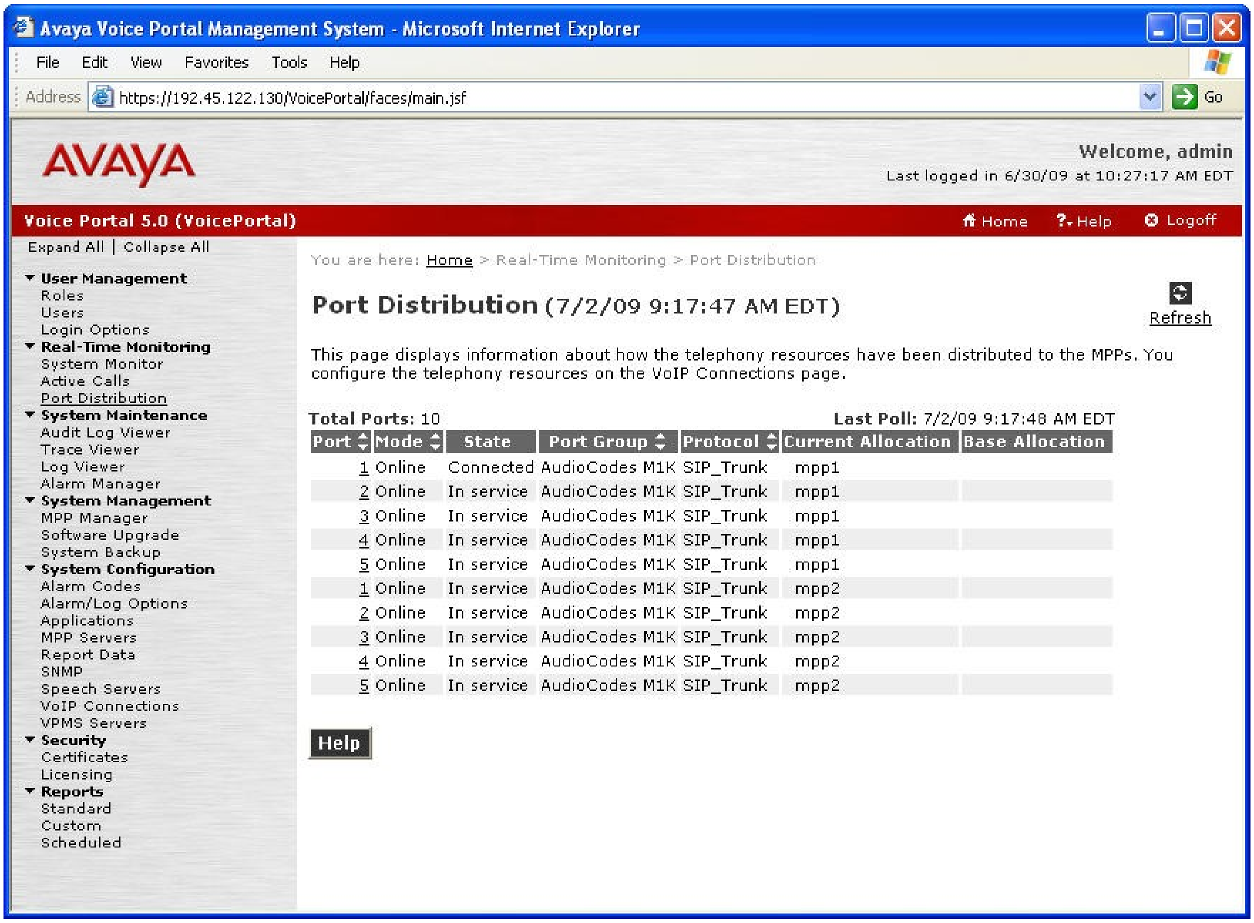
8. Verification Steps

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.

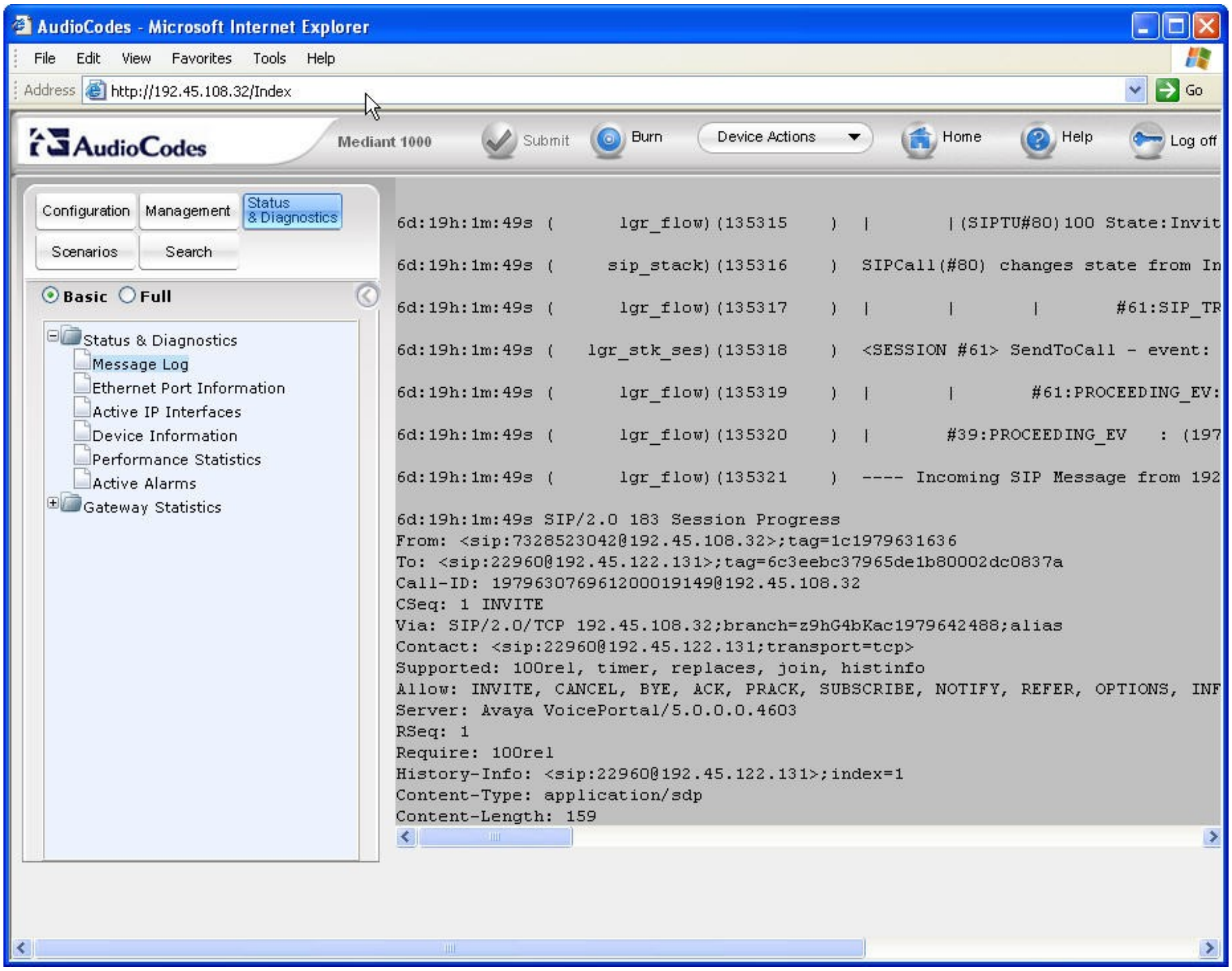
1. From VPMS (Voice Portal Management System) web interface, verify that the MPP server in use is online and running as shown below.



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



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.

1. *Administering Voice Portal,* March 2009.

2. *Avaya Voice Portal 5.0 Release Notes (Issue 2),* March 2009.

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 http://support.avaya.com.

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