Avaya Solution & Interoperability Test Lab

Application Notes for LumenVox Speech Engine with Avaya Voice Portal – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the LumenVox Speech Engine with Avaya Voice Portal. The LumenVox Speech Engine is a standards-based speech recognizer that supports multiple languages and can perform speech recognition on audio data from any audio source. It also provides speech application developers with a development and runtime platform, allowing for dynamic language, grammar, audio format, and logging capabilities to customize every step of an application.

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 configuration steps required to integrate the LumenVox Speech Engine with Avaya Voice Portal. The LumenVox Speech Engine is a standards-based speech recognizer that supports multiple languages and can perform speech recognition on audio data from any audio source. It also provides speech application developers with a development and runtime platform, allowing for dynamic language, grammar, audio format, and logging capabilities to customize every step of an application.

Figure 1 illustrates the configuration used for testing. In this configuration, Avaya Voice Portal interfaces with Avaya Communication Manager via H.323 and the LumenVox Speech Engine via Media Resource Control Protocol (MRCP). VoiceXML (VXML) scripts were run by Avaya Voice Portal and used the automatic speech recognition (ASR) engine in the LumenVox Speech Engine. Since the LumenVox Speech Engine does not support text-to-speech (TTS), an optional third-party TTS engine may be used if required by the application. A TTS engine was used during testing.

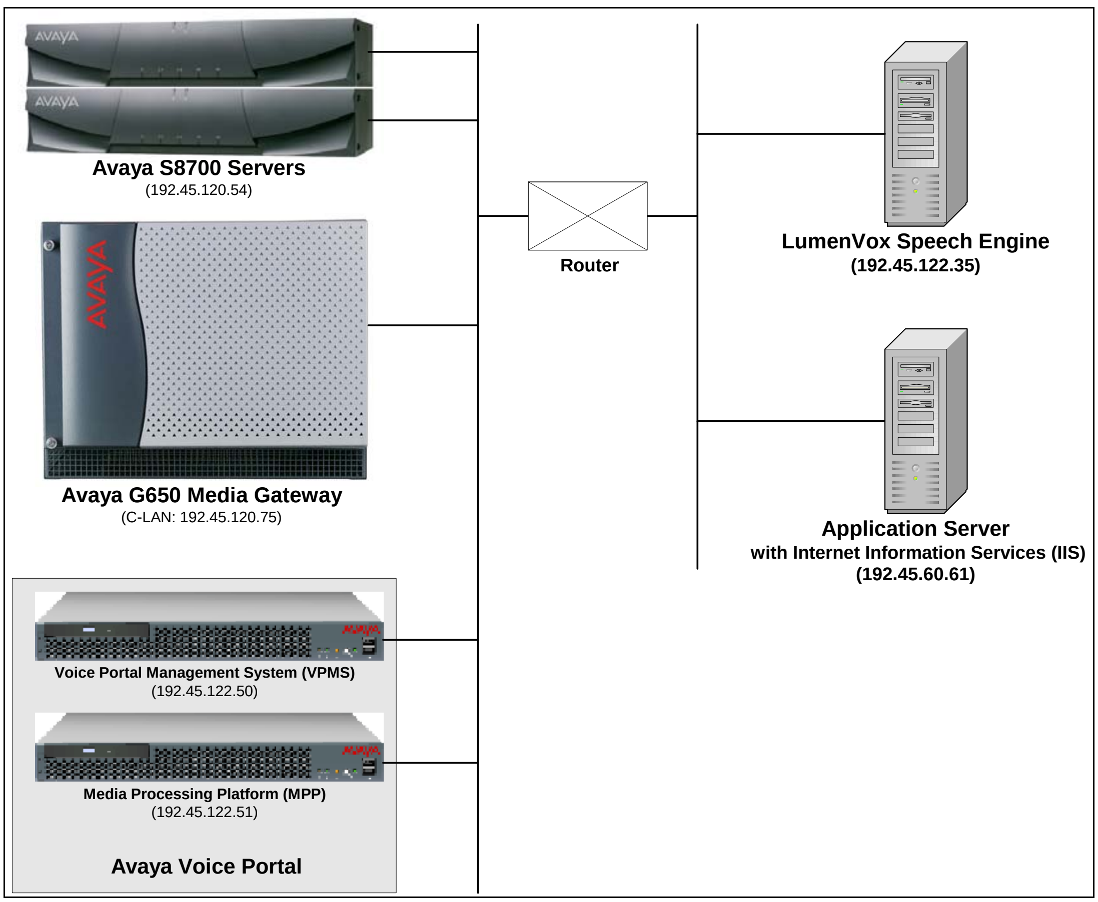

Figure 1: Configuration with Avaya Voice Portal and the LumenVox Speech Engine

1.1. Equipment and Software Validated

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

Equipment	Software
Avaya Voice Portal	4.0.0.0.2901
Avaya S8700 Servers with an Avaya G650 Media Gateway	Avaya Communication Manager 4.0 (R014x.00.1.731.2)
LumenVox Speech Engine LumenVox MRCP Server LumenVox License Server	8.0.301 8.0.301 8.0.301
Application Server – HTTP Server running Windows Server 2003	Microsoft Internet Information Services (IIS) 5.1

2. Configure Avaya Communication Manager

This section describes the configuration of H.323 stations and the IP codec set for Avaya Voice Portal. This configuration also requires a C-LAN and Media Processor board for IP communication. This configuration is outside the scope of these application notes, but the reader may refer to [1] and [2] for additional information.

From the System Access Terminal (SAT), add an H.323 station for Avaya Voice Portal. A call to this station will be routed to Avaya Voice Portal which will run a VXML script that uses the LumenVox Speech Engine. In the station form, set the **Type** to *7434ND*, provide a descriptive **Name**, set the **Security Code**, and set the **IP SoftPhone** field to *y*.

```
add station 23802                                          Page   1 of   6
                                     STATION

Extension: 23802                         Lock Messages? n             BCC: 0
     Type: 7434ND                        Security Code: XXXXX          TN: 1
     Port: S00059                      Coverage Path 1:               COR: 1
     Name: VP 192.45.122.50            Coverage Path 2:               COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 2          Personalized Ringing Pattern: 1
             Data Module? n                      Message Lamp Ext: 23802
          Display Module? y
        Display Language: english                 Coverage Module? n

          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? y

                                              IP Video Softphone? n
```

Figure 2: Station Form

In the IP codec set form associated with the IP network region of the H.323 station, configured in **Figure 2**, set the **Audio Codec** field to the appropriate value. In this configuration, *G.711MU* was used.

```
change ip-codec-set 1                                         Page   1 of   2

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU          n          2        20
 2:
 3:
 4:
```

Figure 3: IP Codec Set Form

3. Configure Avaya Voice Portal

This section covers the administration of Avaya Voice Portal. The following Avaya Voice Portal configuration steps will be covered:

- Configuring an H.323 VoIP connection
- Adding an MPP server
- Configuring the VoIP audio format (mu-law or a-law)
- Adding a speech server
- Adding applications
- Starting the 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 of the VPMS. Log in using the Administrator user role. The screen shown in **Figure 4** is displayed.

Note: All of the screens in this section are shown after the Avaya Voice Portal had been configured. The user should save the screen parameters as Avaya Voice Portal is configured.

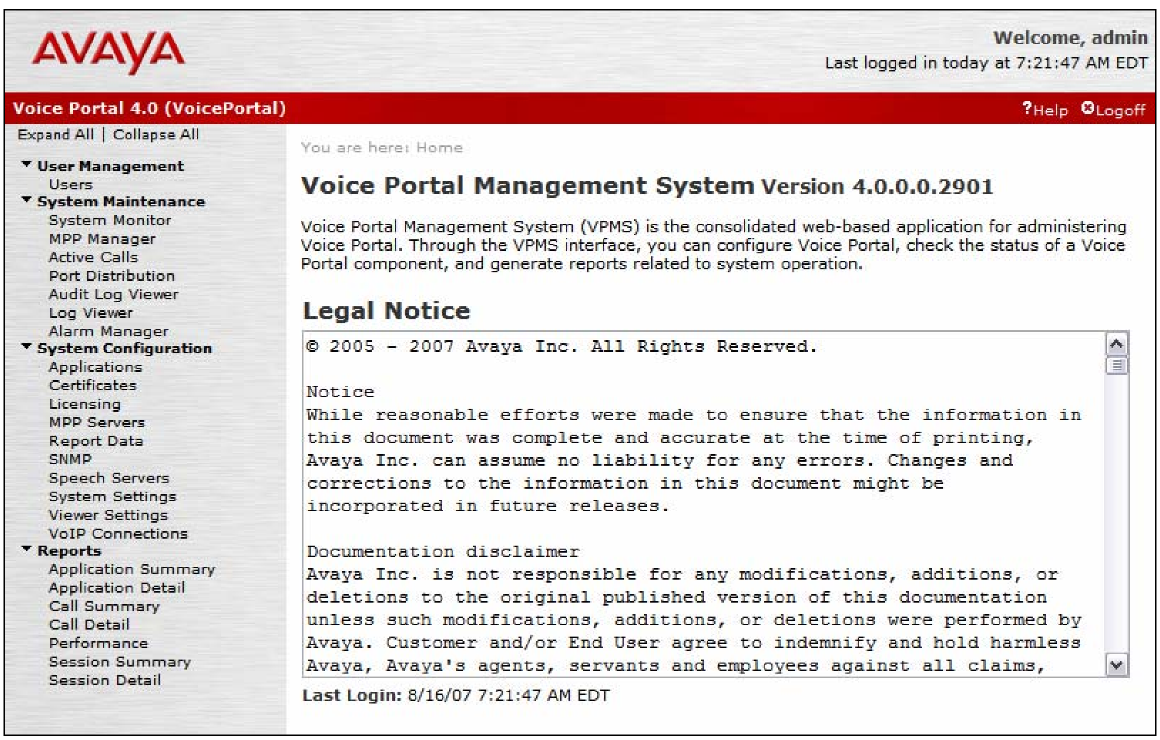

Figure 4: VPMS Main Screen

Configure the H.323 VoIP Connection. To configure an H.323 connection, navigate to the **VoIP Connections** page and then click on the **H.323** tab. In the H.323 Connection page shown in **Figure 5**, set the **Gatekeeper Address** to the IP address of the C-LAN in the Avaya G650 Media Gateway and the **Gatekeeper Port** to *1719*. Next, configure the stations for Avaya Voice Portal, which map to the 7434ND stations configured in Avaya Communication Manager. In addition, set the **Password** for the stations and set the **Station Type** to *Inbound and Outbound*. In this configuration, only station 23802 was mapped to the Avaya Voice Portal application that used the LumenVox Speech Engine.

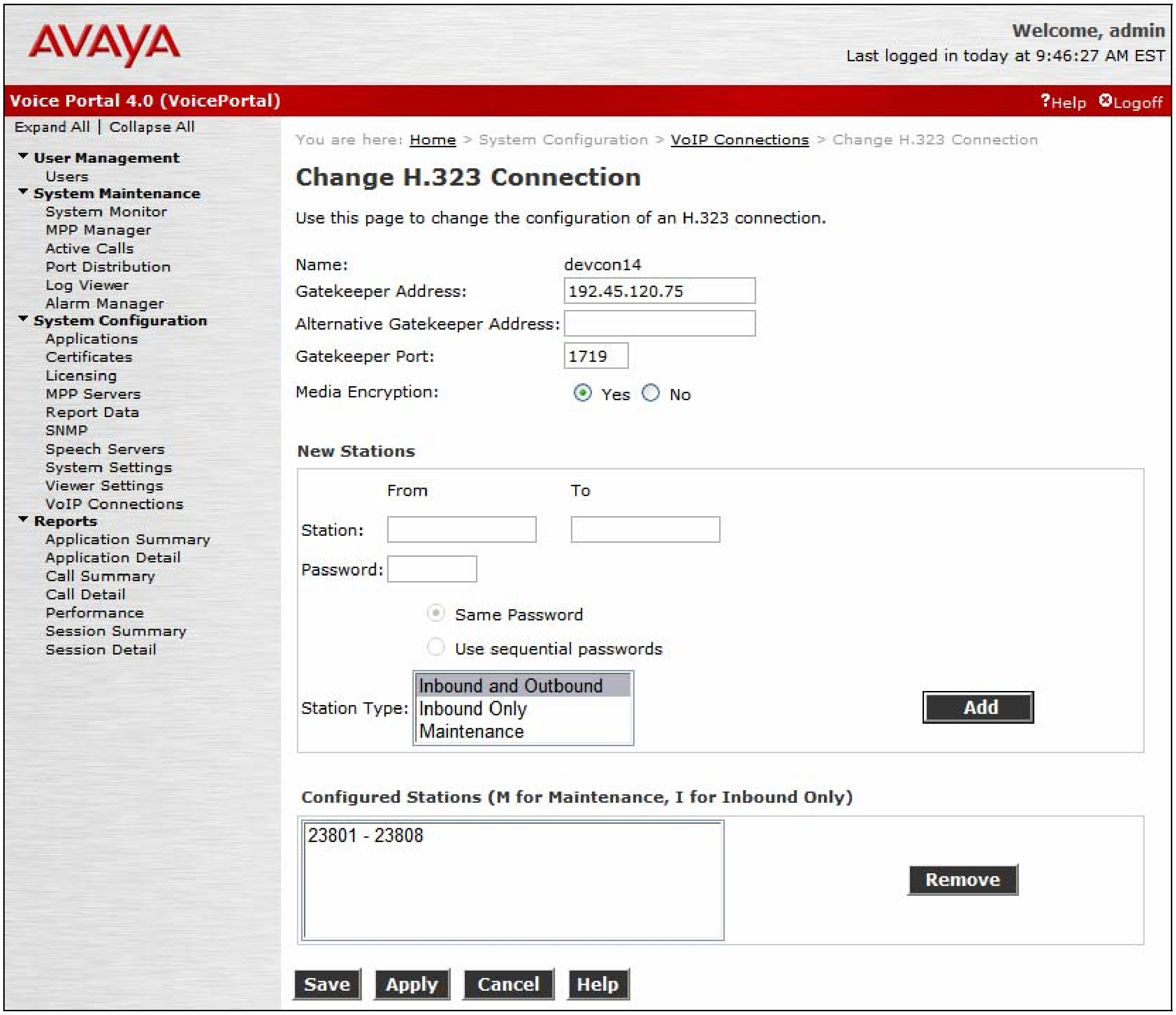

Figure 5: H.323 Connection

Add an MPP Server. Add the MPP server by navigating to the **MPP Servers** page by selecting the option from the left pane. In the MPP Server configuration page, specify a descriptive name and the **Host Address** of each MPP server. Also, specify the **Maximum Simultaneous Calls** supported by each MPP server. **Figure 6** shows the configuration for the MPP server.

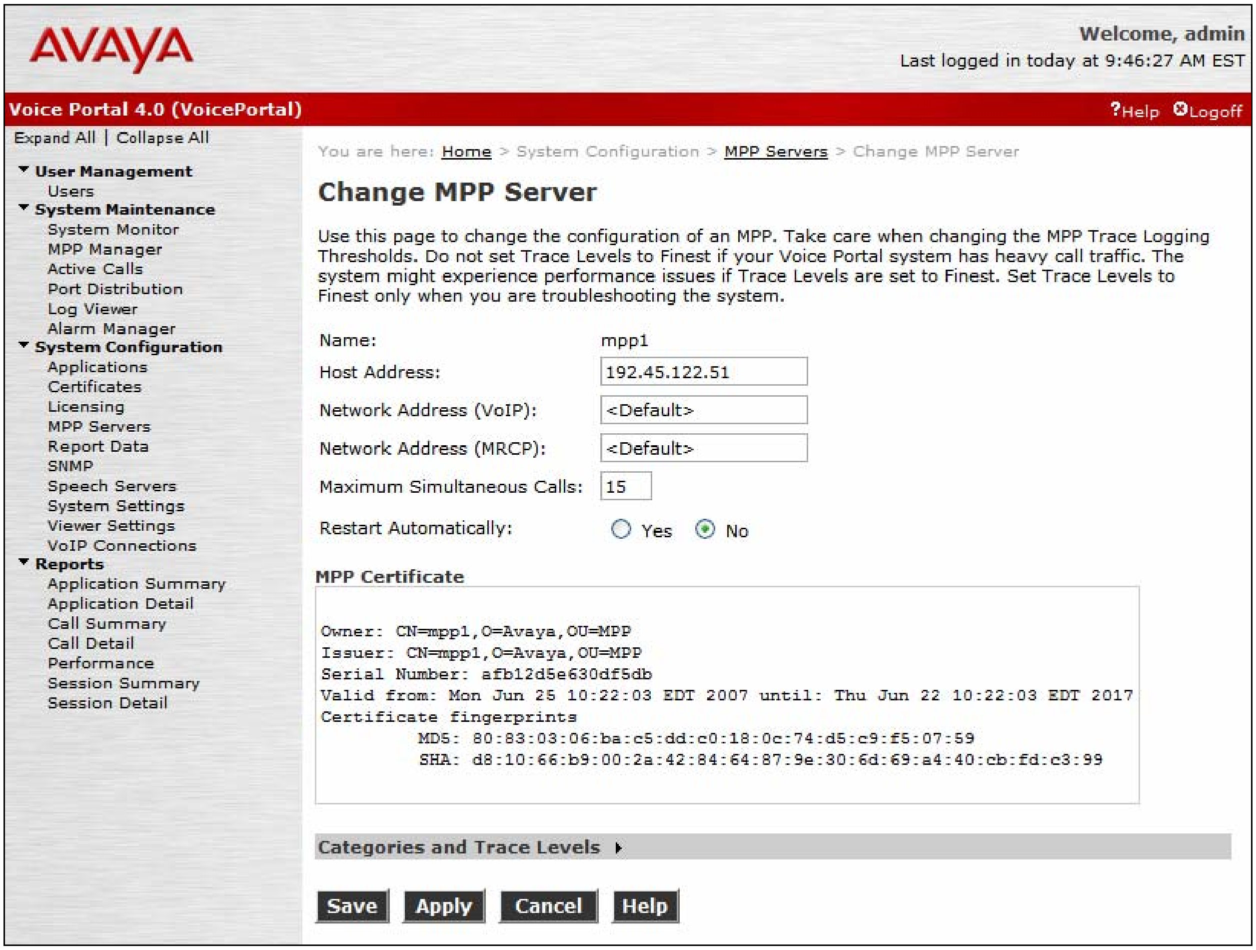

Figure 6: MPP Server

Configure the VoIP Audio Format. The **VoIP Audio Format** for the MPP server is configured in the **VoIP Settings** screen. The **MPP Native Format** field in **Figure 7** is set to *audio/basic* for mu-law.

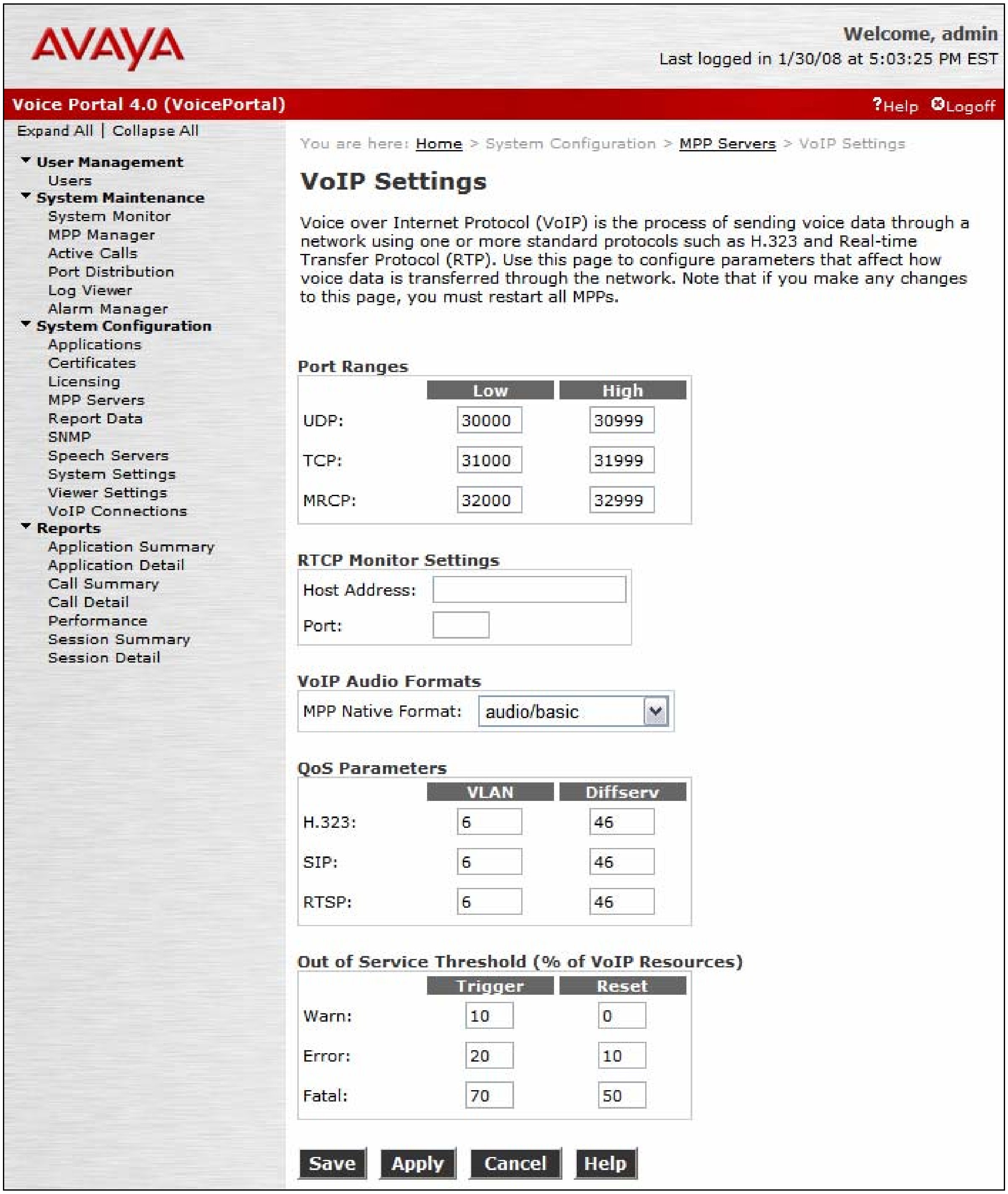

Figure 7: VoIP Settings

Add an ASR Server. To configure the ASR server, click on **Speech Servers** in the left pane, select the **ASR** tab, and then click **Add**. For the LumenVox Speech Engine, the **Engine Type** should be set to *IBM WVS*. Set the **Network Address** field to the IP address of the LumenVox Speech Engine and select the desired **Languages** to be supported. The other fields were set to their default values.

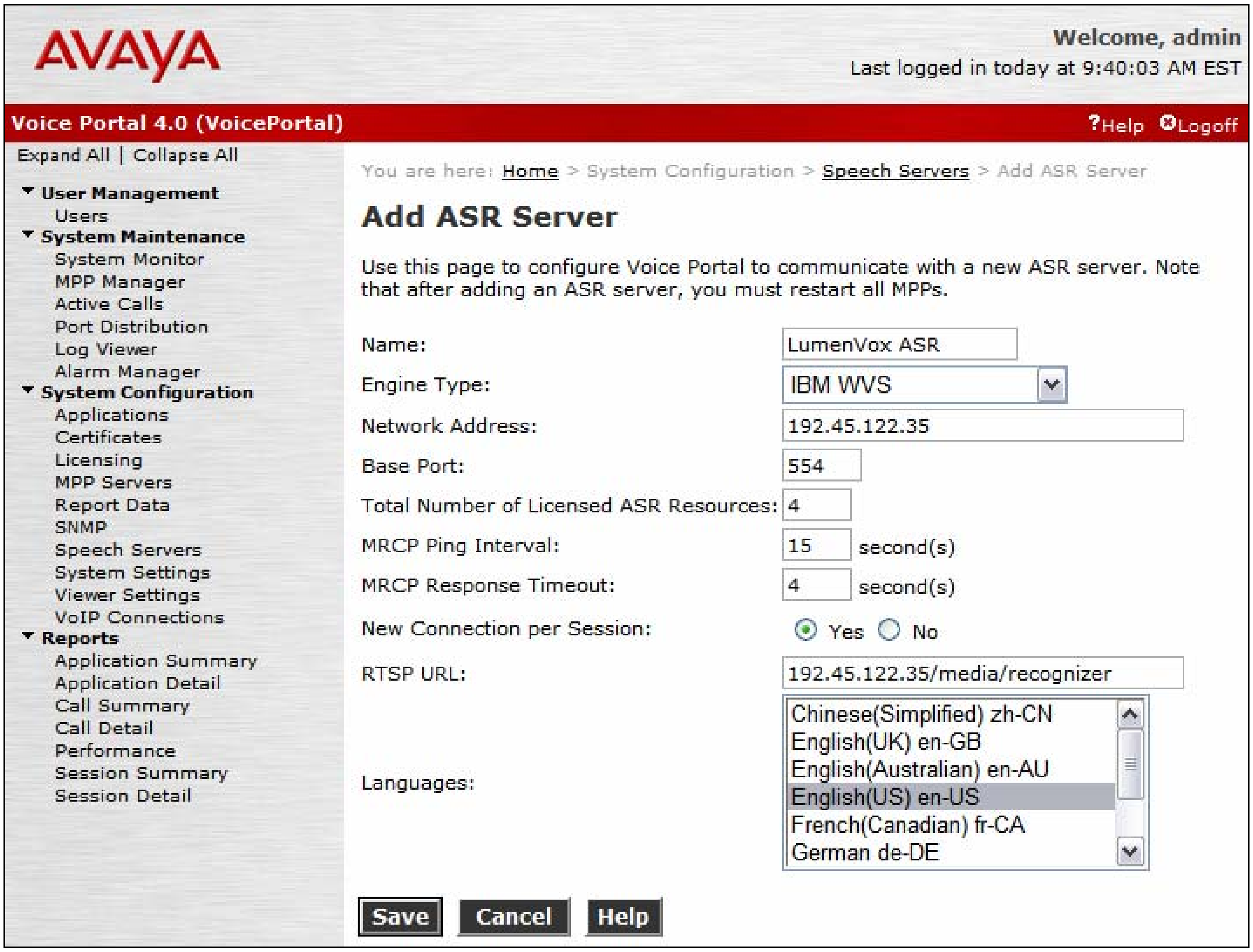

Figure 8: ASR Speech Server

Add a TTS Server. Although the LumenVox Speech Engine does not support a TTS engine, a third-party TTS server that is supported by Avaya Voice Portal may be added in the TTS tab under the **Speech Servers** option in the left pane if it is required by the Avaya Voice Portal application. For further instructions on how to add a TTS server to Avaya Voice Portal, refer to [4].

Add an Application. On the **Applications** page, add an Avaya Voice Portal application. Specify a **Name** for the application, set the **MIME Type** field to the appropriate value (e.g., VoiceXML), and set the **VoiceXML URL** field to point to a VoiceXML application hosted in 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. The **Applications** screen is shown in **Figure 9**.

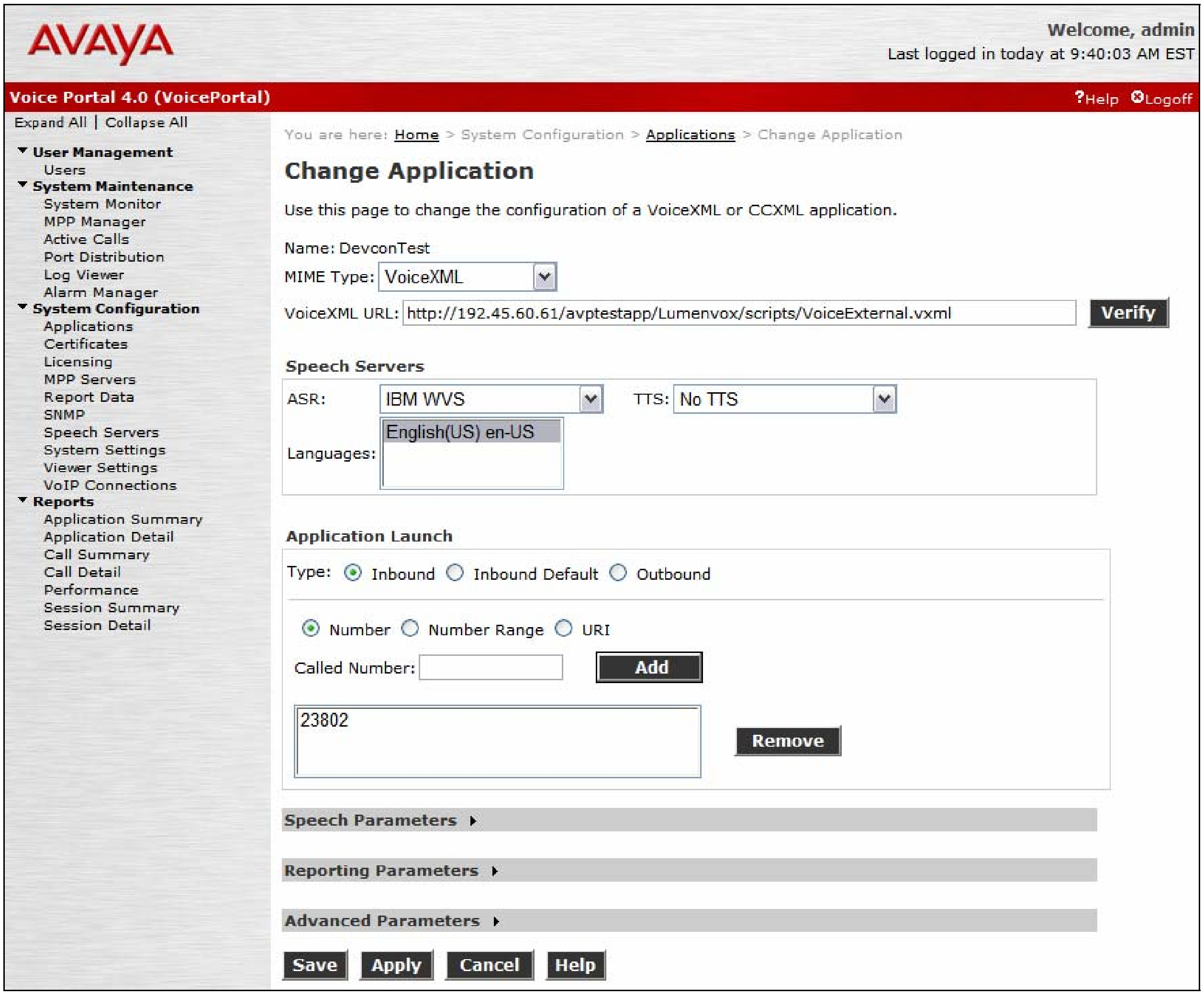

Figure 9: Applications

Start the MPP Server. Start the MPP server from the **MPP Manager** page shown in **Figure 10**. Select each MPP and then click the **Start** button. After the MPP is started, the **Mode** of the MPP should be *Online* and the **State** should be *Running*.

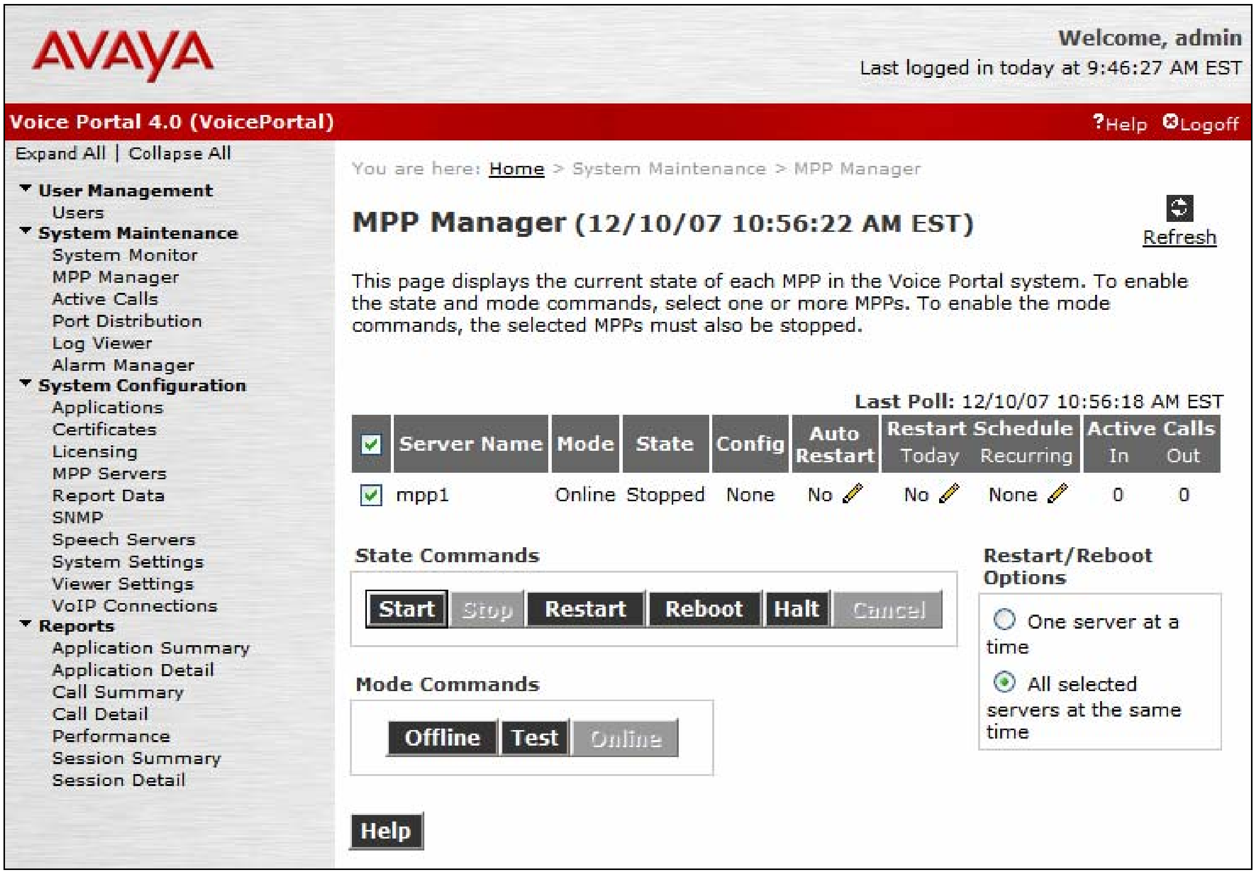

Figure 10: MPP Manager

4. Configure LumenVox Speech Engine

This section covers the configuration required for the LumenVox Speech Engine. These changes are contained in the `mrcp.config` file. The required changes are highlighted in **bold**. Refer to [5] for a complete reference for the LumenVox Speech Engine.

```
#-----------------------------------------------------------------
# this is the config file used by the Lumenvox Mrcp Server.
# the format is very simple.
#
# lines starting with '#' are comments and are ignored.
# blank lines are also ignored.
#
# valid lines have the format 'param = value'.
# spaces are stripped from beginning of the line and
# from around the equal sign.
#
# PLEASE NOTE that the parameter names are case-sensitive.
#-----------------------------------------------------------------

#-----------------------------------------------------------------
#
# custom Log file name for the mrcp server
#-----------------------------------------------------------------
mrcp_server_log            = MRCP_Log_Avaya.txt

#-----------------------------------------------------------------
# this is the only parameter that you really NEED to set.
# all the others have acceptable defaults.
# replace this number with your machine's IP address.
#-----------------------------------------------------------------
mrcp_server_ip             = 192.45.122.35

#-----------------------------------------------------------------
# this parameter sets the TCP port on which the server will listen
# for incoming RTSP requests.
#-----------------------------------------------------------------
mrcp_server_port    = 554

#-----------------------------------------------------------------
# this parameter is the lowest numbered UDP port that will be used
# for RTP and RTCP.  two sequentially numbered ports will be used
# per resource, one for RTP and the next for RTCP.
# rtpbase must be an even number.
#-----------------------------------------------------------------
mrcp_server_rtpbase = 49922

#-----------------------------------------------------------------
# the maximum number of concurrent connections allowed.
# can't be more than the number of resources. Atleast one
# resource per connection
#-----------------------------------------------------------------
mrcp_server_connmax = 100
```

```
#----------------------------------------------------------------
# the maximum number of concurrent resources.
# practically speaking, this number can not be greater than the
# number of port licenses you have for your SRE.
#----------------------------------------------------------------
mrcp server resmax  = 200

#----------------------------------------------------------------
# if you are running the MrcpServer and SRE on different machines,
# set this value to the IP address of the machine that is running
# the SRE.
#----------------------------------------------------------------
sre_ip = 192.45.122.35

#----------------------------------------------------------------
# set this value to the license type used by the speech
# recognizer. Its poissible values can be:
# Auto - picks whatever license is available
# VoxLite - picks only voxlite license
# SpeechPort - picks only full speech port license
#----------------------------------------------------------------
license type = Auto

#----------------------------------------------------------------
# this is the time in seconds since the last request received
# after which a session will automatically timeout.
#----------------------------------------------------------------
sess timeout sec = 200

#----------------------------------------------------------------
# enable logging = 1(default) or 0
#----------------------------------------------------------------
enable logging = 1

#----------------------------------------------------------------
# enable sre logging = 1 or 0 (default)
# enable or disable logging of response files in the Lang\Responses
# Directory of of the Speech Recognition Engine
#----------------------------------------------------------------
enable sre logging = 1

#----------------------------------------------------------------
#the ASR resource name string, such as  "recognizer"(default) ,
# "asr", etc
#------------------------------------------------------------------

resource_string       =  media/recognizer

#-----------------------------------------------------------------
# enable inc reco cseq = 1 or 0 (default)
# During RECOGNIZE session request, the CSeq will be increment for
# event including START-OF-SPEECH, RECOGNITION-COMPLETE if
# enable inc reco cseq sets to 1. If this value sets to 0, the CSeq
# will not be increment for those events which will be the same as
# the RECOGNIZE methos's CSeq.
#-----------------------------------------------------------------
enable inc reco cseq = 0

#------------------------------------------------------------------
# Default LumenVox Engine Specific Streaming Parameters
#----------------------------------------------------------------

dtmf payload type=96
choose model =1
enable lattice scoring =1
initial silence trimmed = 0
speech complete timeout =800
wind back time =1000
burst_thrsld =30
```

```
end of speech timeout=20000
#nbest length=4
confidence thrsld=45
sensitivity lvl=50
#speed vs accuracy=11 # not used  at this time
#dtmf term char=#
no input timeout=10000
dtmf termination timeout=50000
recognizer start timers=true
recognition timeout=60000
dtmf inter digit timeout=5000
snr sensitivity lvl=50
save waveform=false
waveform url location=file:///c:/
barge_in_timeout=150000
```

Figure 11: mrcp.config File

5. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify Avaya Voice Portal VXML applications that use the ASR engine in the LumenVox Speech Engine. This section covers the general test approach and the test results.

5.1. General Test Approach

The interoperability compliance test included feature and serviceability testing. The feature testing focused placing calls to Avaya Voice Portal that ran VoiceXML applications that use ASR engine in the LumenVox Speech Engine and the LumenVox MRCP Server. The testing verified speech and DTMF tone recognition.

The serviceability testing focused on verifying the ability of the LumenVox Speech Engine to recover from adverse conditions, such as power failures and disconnecting cables to the IP network.

5.2. Test Results

All test cases passed. Avaya Voice Portal was successful in running applications that use the ASR engine of the LumenVox Speech Engine.

6. Verification Steps

This section provides the verification steps that may be performed to verify that Avaya Voice Portal can run IVR applications that use the LumenVox Speech Engine.

1. From the VPMS web interface, verify that the MPP server is online and running in the **MPP Manager** page shown in **Figure 12**.

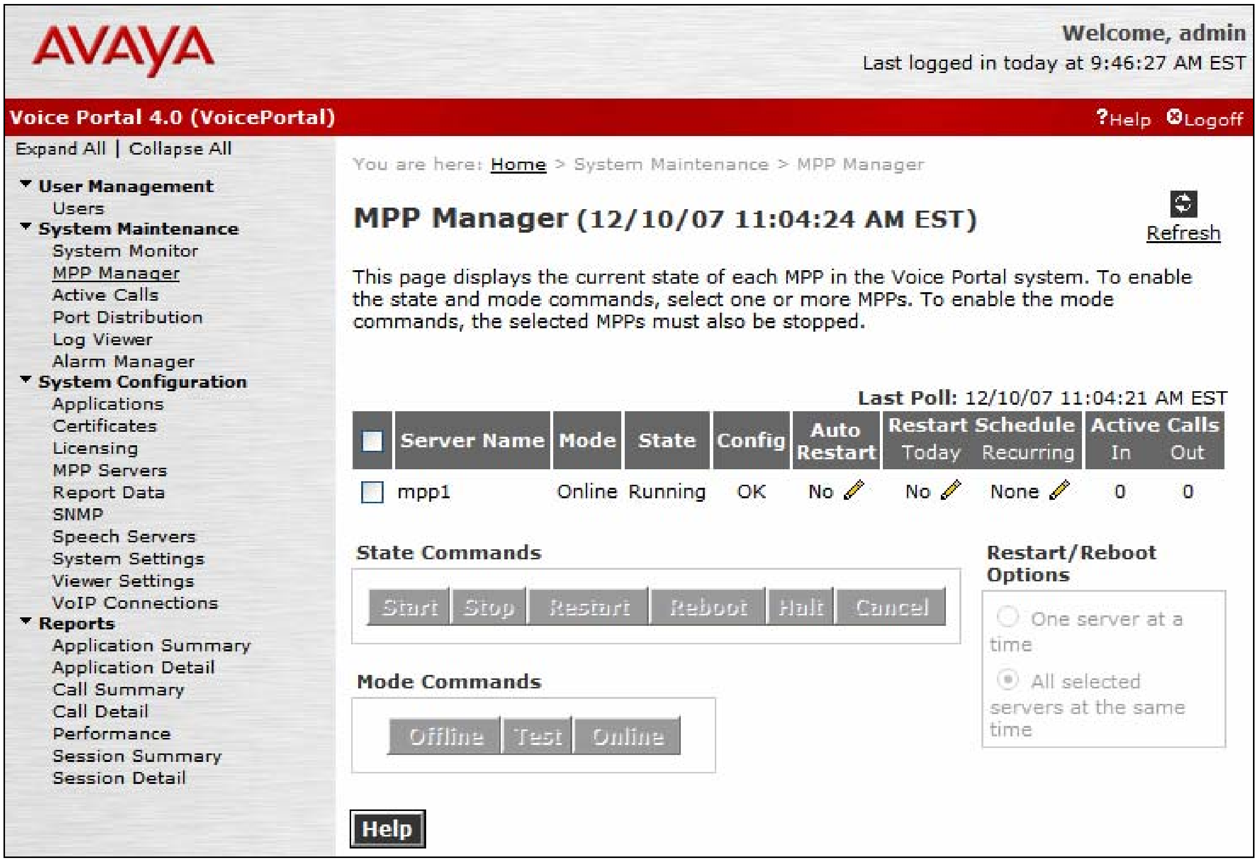

Figure 12: MPP Manager

2. From the VPMS web interface, verify that the ports on the MPP server are in service in the **Port Distribution** page shown in **Figure 13**.

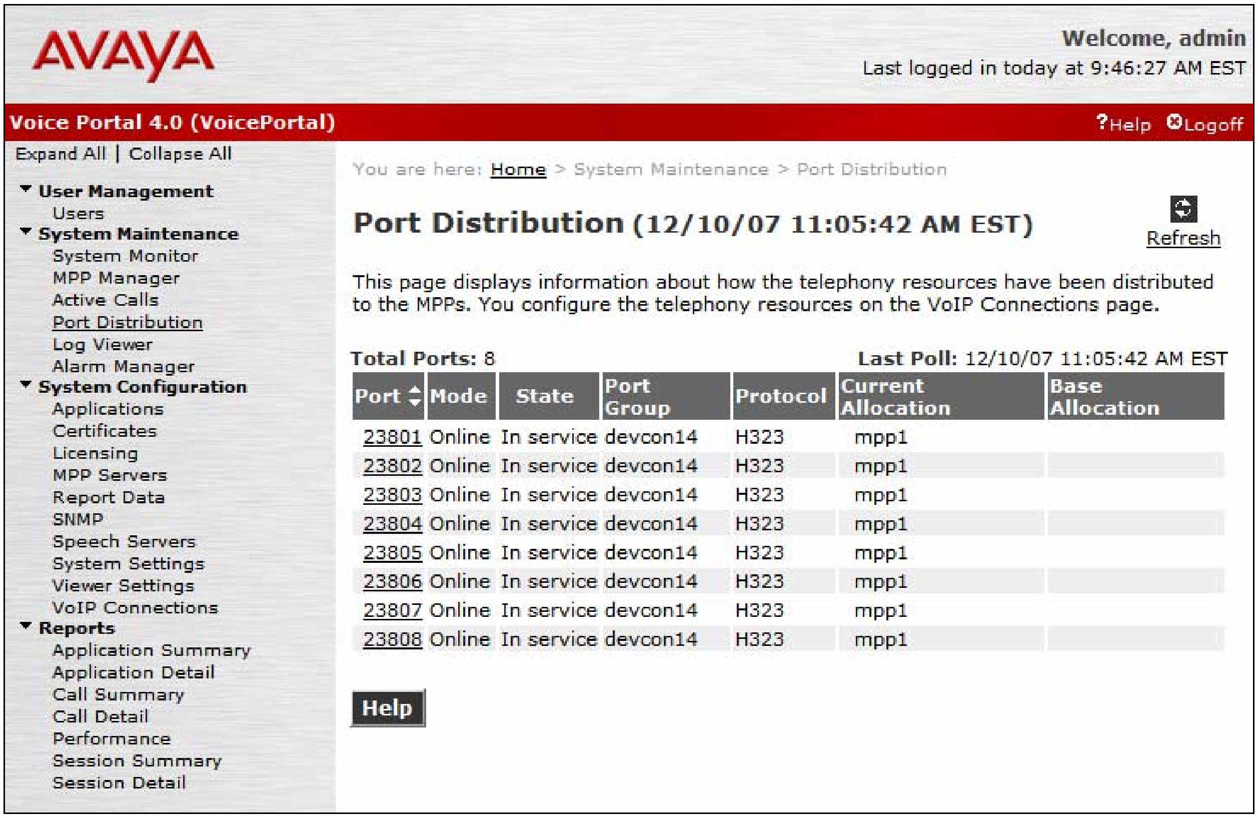

Figure 13: Port Distribution

3. Place a call to an Avaya Voice Portal extension that runs a VXML script that uses the LumenVox Speech Engine. Verify that the application answers the call and that the application is able to recognize the speech and DTMF tones input provided by the caller.

7. Support

To contact LumenVox by phone or access their website:

- **Phone:** (877) 977-0707
- **Web:** http://www.lumenvox.com

8. Conclusion

These Application Notes describe the configuration steps required to integrate the LumenVox Speech Engine with Avaya Voice Portal. All feature and serviceability test cases were completed successfully.

9. Additional References

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

[1] *Administrator Guide for Avaya Communication Manager*, Document 03-300509, Issue 3.1, February 2007, available at http://support.avaya.com.

[2] *Feature Description and Implementation for Avaya Communication Manager*, Document 555-245-205, Issue 5, February 2007, available at http://support.avaya.com.

[3] *Installing and Configuring Avaya Voice Portal 4.0*, June 2007, available at http://support.avaya.com.

[4] *Administering Avaya Voice Portal 4.0*, June 2007, available at http://support.avaya.com.

[5] *LumenVox Online Documentation* available at http://help.lumenvox.com/Robo/BIN/Robo.dll?tpc=%2Frobo%2Fprojects%2Fspeechengine%2Froot%2Fwelcome.htm&mgr=agm&project=speechengine&wnd=speechengine%7CLumenvox&agt=wsm&refer=http%3A%2F%2Fwww.lumenvox.com%2Fsupport%2F&ctxid=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.