



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for LumenVox Speech Engine with Avaya Voice Portal – Issue 0.1**

### **Abstract**

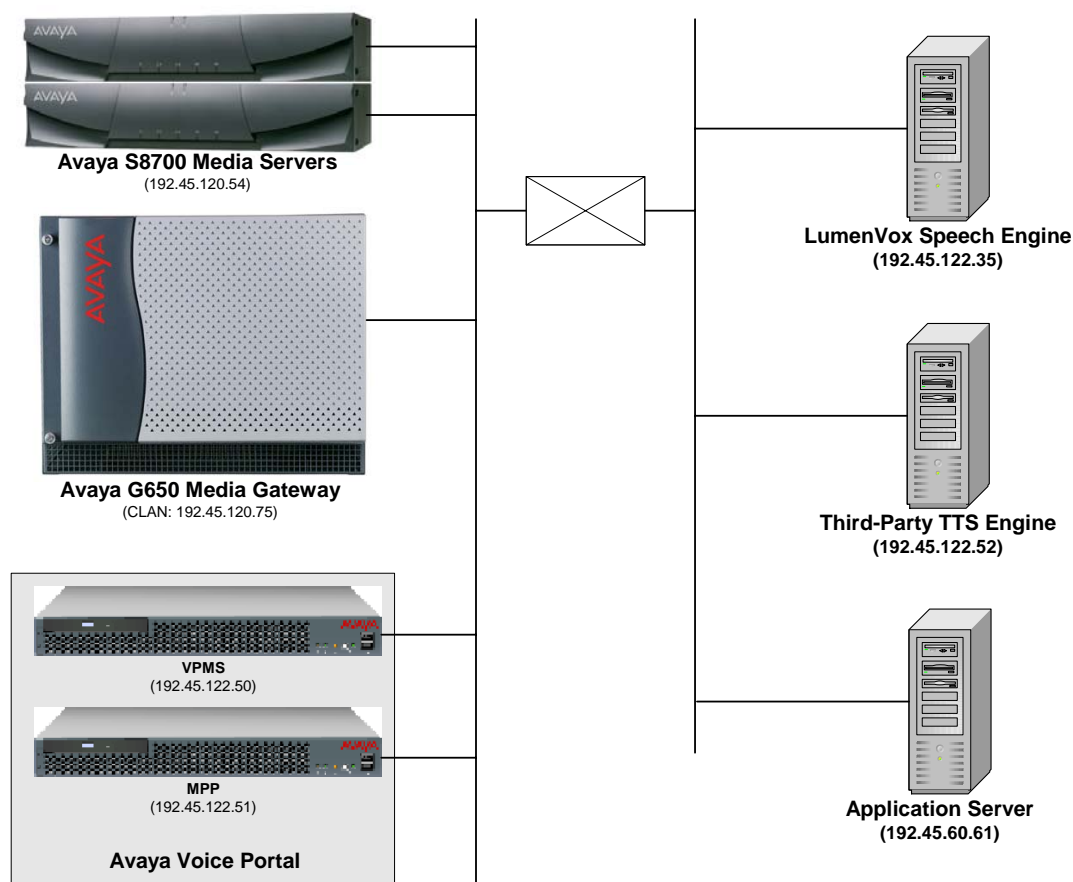
These Application Notes describe the configuration steps required to integrate the LumenVox Speech Engine with Avaya Voice Portal. The LumenVox Speech Engine is a standards-based speech recognizer that supports multiple languages and can perform speech recognition on audio data from any audio source. 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, Voice Portal interfaces with Avaya Communication Manager via H.323 and the LumenVox Speech Engine via Media Resource Control Protocol (MRCP). VXML scripts were run by Voice Portal and used the automatic speech recognition (ASR) engine in the LumenVox Speech Engine. An optional, third-party text-to-speech (TTS) engine was also 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 a G650 Media Gateway	Avaya Communication Manager 4.0 (R014x.00.0.730.5)
LumenVox Speech Engine	8.0.301
Application Server – HTTP Server running Windows Server 2003	N/A

## 2. Configure Avaya Communication Manager

This section describes the configuration of H.323 stations and the IP codec set for 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 Voice Portal. A call to this station will be routed to 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**

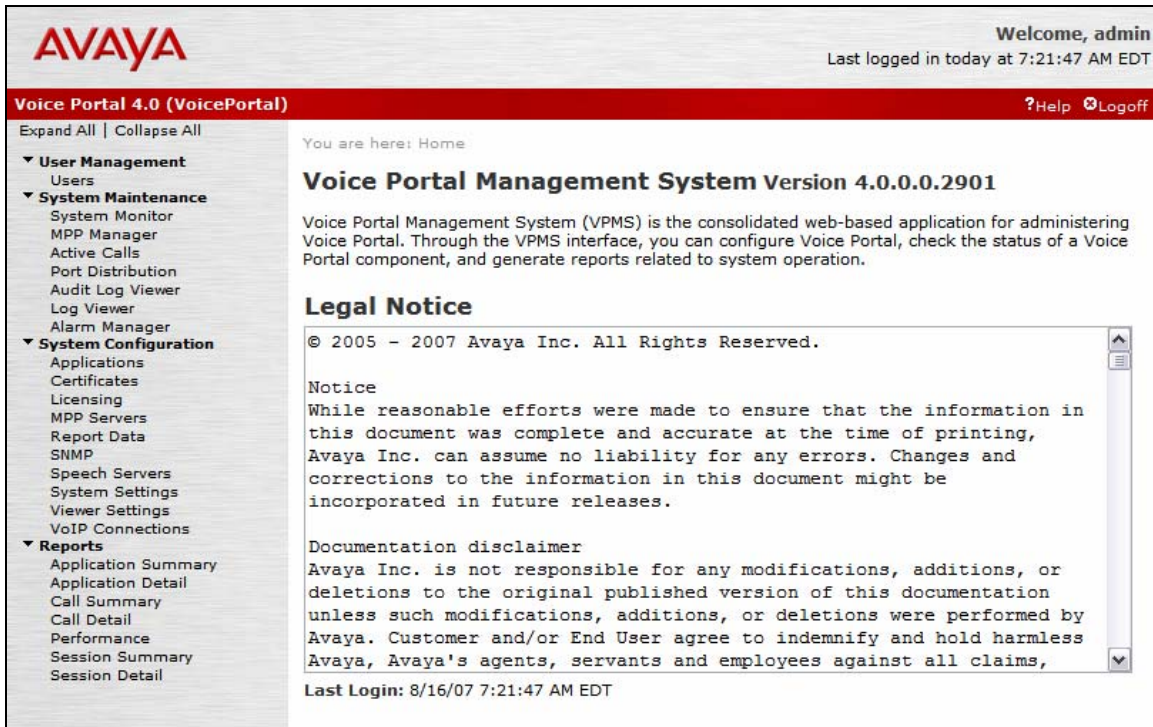
### 3. Configure Avaya Voice Portal

This section covers the administration of Avaya Voice Portal. The following 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 Voice Portal had been configured. Don't forget to save the screen parameters as you configure Avaya Voice Portal.



**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 tab shown in **Figure 5**, set the **Gatekeeper Address** to the IP address of the C-LAN in the G650 media gateway and the **Gatekeeper Port** to *1719*. Next, configure the stations for 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 mapped to the Voice Portal application that used the LumenVox Speech Engine.

**AVAYA** Welcome, admin  
Last logged in today at 9:46:27 AM EST

**Voice Portal 4.0 (VoicePortal)** ?Help Logoff

Expand All | Collapse All

You are here: [Home](#) > [System Configuration](#) > [VoIP Connections](#) > [Change H.323 Connection](#)

### Change H.323 Connection

Use this page to change the configuration of an H.323 connection.

Name: devcon14

Gatekeeper Address:

Alternative Gatekeeper Address:

Gatekeeper Port:

Media Encryption:  Yes  No

#### New Stations

From	To
Station: <input type="text"/>	<input type="text"/>
Password: <input type="text"/>	
<input checked="" type="radio"/> Same Password <input type="radio"/> Use sequential passwords	
Station Type:	<input type="text" value="Inbound and Outbound"/> <input type="text" value="Inbound Only"/> <input type="text" value="Maintenance"/>

**Add**

#### Configured Stations (M for Maintenance, I for Inbound Only)

23801 - 23808	<b>Remove</b>
---------------	---------------

**Save** **Apply** **Cancel** **Help**

**Figure 5: H.323 Connection**

**Add an MPP Server.** Add the MPP server by navigating to the **MPP Servers** screen 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 first MPP server.

The screenshot shows the Avaya Voice Portal 4.0 (VoicePortal) interface. The top right corner displays "Welcome, admin" and "Last logged in today at 9:46:27 AM EST". The left navigation pane includes sections for User Management, System Maintenance, System Configuration, and Reports. The main content area is titled "Change MPP Server" and contains the following configuration fields:

- Name: mpp1
- Host Address: 192.45.122.51
- Network Address (VoIP): <Default>
- Network Address (MRCP): <Default>
- Maximum Simultaneous Calls: 15
- Restart Automatically:  Yes  No

Below the configuration fields is an "MPP Certificate" section with the following details:

```

Owner: CN=mpp1,O=Avaya,OU=MPP
Issuer: CN=mpp1,O=Avaya,OU=MPP
Serial Number: afb12d5e630df5db
Valid from: Mon Jun 25 10:22:03 EDT 2007 until: Thu Jun 22 10:22:03 EDT 2017
Certificate fingerprints
MD5: 80:83:03:06:ba:c5:dd:c0:18:0c:74:d5:c9:f5:07:59
SHA: d8:10:66:b9:00:2a:42:84:64:87:9e:30:6d:69:a4:40:cb:fd:c3:99
    
```

At the bottom of the configuration area, there is a "Categories and Trace Levels" section and a row of buttons: Save, Apply, Cancel, and Help.

**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 alaw.

**AVAYA** Welcome, admin  
Last logged in 1/30/08 at 5:03:25 PM EST

**Voice Portal 4.0 (VoicePortal)** ? Help Logoff

Expand All | Collapse All

You are here: [Home](#) > [System Configuration](#) > [MPP Servers](#) > [VoIP Settings](#)

### VoIP Settings

Voice over Internet Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols such as H.323 and Real-time Transfer Protocol (RTP). Use this page to configure parameters that affect how voice data is transferred through the network. Note that if you make any changes to this page, you must restart all MPPs.

**Port Ranges**

	Low	High
UDP:	<input type="text" value="30000"/>	<input type="text" value="30999"/>
TCP:	<input type="text" value="31000"/>	<input type="text" value="31999"/>
MRCP:	<input type="text" value="32000"/>	<input type="text" value="32999"/>

**RTCP Monitor Settings**

Host Address:

Port:

**VoIP Audio Formats**

MPP Native Format:

**QoS Parameters**

	VLAN	Diffserv
H.323:	<input type="text" value="6"/>	<input type="text" value="46"/>
SIP:	<input type="text" value="6"/>	<input type="text" value="46"/>
RTSP:	<input type="text" value="6"/>	<input type="text" value="46"/>

**Out of Service Threshold (% of VoIP Resources)**

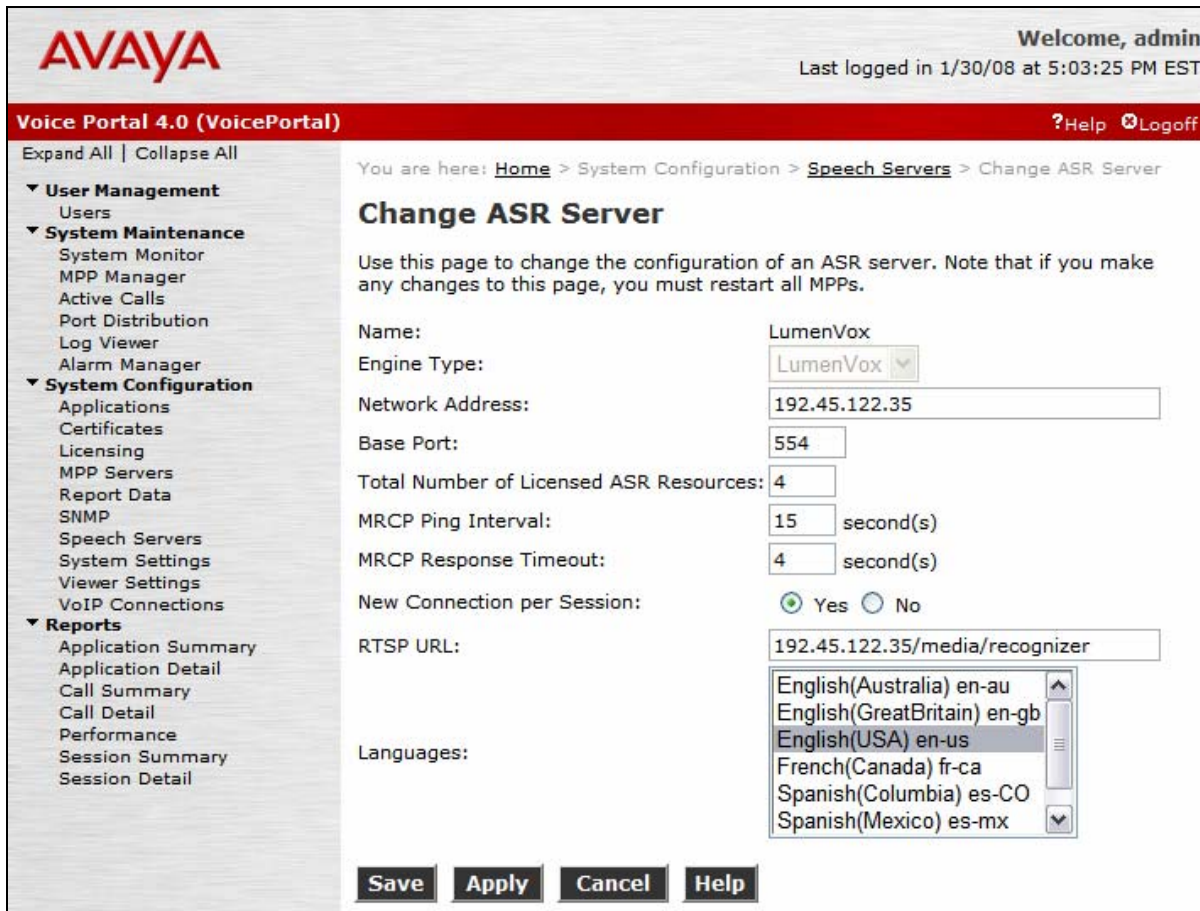
	Trigger	Reset
Warn:	<input type="text" value="10"/>	<input type="text" value="0"/>
Error:	<input type="text" value="20"/>	<input type="text" value="10"/>
Fatal:	<input type="text" value="70"/>	<input type="text" value="50"/>

**Save** **Apply** **Cancel** **Help**

**Figure 7: VoIP Settings**



**Add an ASR Server.** To configure the automatic speech recognition (ASR) server, click on **Speech Servers** in the left pane, select the **ASR** tab, and then click **Add**. **Figure 8** shows the screen after the ASR server has already been configured. 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 text-to-speech (TTS) engine, a third-party TTS server that is supported by Voice Portal may be added in the TTS tab under the **Speech Servers** option in the left pane if it is required by the Voice Portal application. For further instructions on how to add a TTS server to Voice Portal, refer to [4].

**Add an Application.** On the **Applications** page, add a 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**.

The screenshot displays the Avaya Voice Portal 4.0 interface for configuring an application. The top navigation bar includes the Avaya logo, the user name 'admin', and the login time '1/30/08 at 5:03:25 PM EST'. The main content area is titled 'Change Application' and provides instructions for modifying a VoiceXML or CCXML application. The configuration fields are as follows:

- Name:** DevconTest
- MIME Type:** VoiceXML
- VoiceXML URL:** http://192.45.60.61/avptestapp/Lumenvox/scripts/VoiceExternal.vxml
- Speech Servers:**
  - ASR:** LumenVox
  - TTS:** No TTS
  - Languages:** English(USA) en-us
- Application Launch:**
  - Type:** Inbound (selected), Inbound Default, Outbound
  - Number:** 23802 (selected), Number Range, URI

At the bottom of the form, there are buttons for 'Save', 'Apply', 'Cancel', and 'Help'. The left sidebar contains a navigation menu with categories like User Management, System Maintenance, System Configuration, and Reports.

**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*.

**AVAYA** Welcome, admin  
Last logged in today at 9:46:27 AM EST

**Voice Portal 4.0 (VoicePortal)** ?Help Logoff

Expand All | Collapse All

**▼ User Management**  
Users

**▼ System Maintenance**  
System Monitor  
MPP Manager  
Active Calls  
Port Distribution  
Log Viewer  
Alarm Manager

**▼ System Configuration**  
Applications  
Certificates  
Licensing  
MPP Servers  
Report Data  
SNMP  
Speech Servers  
System Settings  
Viewer Settings  
VoIP Connections

**▼ Reports**  
Application Summary  
Application Detail  
Call Summary  
Call Detail  
Performance  
Session Summary  
Session Detail

You are here: [Home](#) > System Maintenance > MPP Manager

### MPP Manager (12/10/07 10:56:22 AM EST) 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: 12/10/07 10:56:18 AM EST

<input checked="" type="checkbox"/>	Server Name	Mode	State	Config	Auto Restart	Restart Schedule	Active Calls		
						Today	Recurring	In	Out
<input checked="" type="checkbox"/>	mpp1	Online	Stopped	None	No	No	None	0	0

**State Commands**

**Mode Commands**

**Restart/Reboot Options**  
 One server at a time  
 All selected servers at the same time

**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 possible 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 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. 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 Voice Portal can run IVR applications that use the LumenVox Speech Engine.

1. From the VPMS web interface, verify that the MPP servers are online and running in the **System Monitor** page shown in **Figure 12**.

The screenshot shows the Avaya Voice Portal 4.0 (VoicePortal) interface. The top header includes the Avaya logo, the user name 'admin', and the last login time '9:46:27 AM EST'. The main navigation bar shows 'Voice Portal 4.0 (VoicePortal)' and 'Logoff'. The left sidebar contains a tree view with categories: User Management, System Maintenance (including System Monitor, MPP Manager, Active Calls, Port Distribution, Log Viewer, Alarm Manager), System Configuration (including Applications, Certificates, Licensing, MPP Servers, Report Data, SNMP, Speech Servers, System Settings, Viewer Settings, VoIP Connections), and Reports (including Application Summary, Application Detail, Call Summary, Call Detail, Performance, Session Summary, Session Detail). The main content area is titled 'MPP Manager (12/10/07 11:04:24 AM EST)' and includes a 'Refresh' button. Below the title is a descriptive paragraph: '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.' A table shows the status of MPP servers, with a 'Last Poll' of '12/10/07 11:04:21 AM EST'. The table has columns for 'Server Name', 'Mode', 'State', 'Config', 'Auto Restart', 'Restart Schedule' (Today, Recurring), and 'Active Calls' (In, Out). The row for 'mpp1' shows it is 'Online Running' with 'OK' config, 'No' auto restart, and 'No' restart schedule, with 0 active calls in and 0 out. Below the table are 'State Commands' (Start, Stop, Restart, Reboot, Halt, Cancel) and 'Mode Commands' (Offline, Test, Online). A 'Restart/Reboot Options' section has radio buttons for 'One server at a time' and 'All selected servers at the same time'.

Server Name	Mode	State	Config	Auto Restart	Restart Schedule		Active Calls	
					Today	Recurring	In	Out
<input type="checkbox"/> mpp1	Online	Running	OK	No	No	None	0	0

**Figure 12:** System Monitor

- 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**.

**AVAYA** Welcome, admin  
Last logged in today at 9:46:27 AM EST

**Voice Portal 4.0 (VoicePortal)** ?Help Logoff

Expand All | Collapse All

**▼ User Management**  
Users

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System Monitor  
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Active Calls  
Port Distribution  
Log Viewer  
Alarm Manager

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Applications  
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Report Data  
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Speech Servers  
System Settings  
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VoIP Connections

**▼ Reports**  
Application Summary  
Application Detail  
Call Summary  
Call Detail  
Performance  
Session Summary  
Session Detail

You are here: [Home](#) > System Maintenance > Port Distribution

### Port Distribution (12/10/07 11:05:42 AM EST) Refresh

This page displays information about how the telephony resources have been distributed to the MPPs. You configure the telephony resources on the VoIP Connections page.

**Total Ports: 8** **Last Poll: 12/10/07 11:05:42 AM EST**

Port	Mode	State	Port Group	Protocol	Current Allocation	Base Allocation
<a href="#">23801</a>	Online	In service	devcon14	H323	mpp1	
<a href="#">23802</a>	Online	In service	devcon14	H323	mpp1	
<a href="#">23803</a>	Online	In service	devcon14	H323	mpp1	
<a href="#">23804</a>	Online	In service	devcon14	H323	mpp1	
<a href="#">23805</a>	Online	In service	devcon14	H323	mpp1	
<a href="#">23806</a>	Online	In service	devcon14	H323	mpp1	
<a href="#">23807</a>	Online	In service	devcon14	H323	mpp1	
<a href="#">23808</a>	Online	In service	devcon14	H323	mpp1	

[Help](#)

**Figure 13: Port Distribution**

- Place a call to Voice Portal that runs a VXML script and 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>

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