



Avaya Solution & Interoperability Test Lab

Application Notes for LumenVox Automated Speech Recognizer, LumenVox Text-to-Speech Server and Call Progress Analysis with Avaya Aura® Experience Portal – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate LumenVox Automated Speech Recognizer, Text-to-Speech Server and Call Progress Analysis with Avaya Aura® Experience Portal.

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

The objective of compliance test was to validate interoperability of LumenVox Automated Speech Recognizer, Text-to-Speech Server and Call Progress Analysis with Avaya Aura® Experience Portal.

LumenVox provides a complete set of speech recognition and text-to-speech technologies for use in interactive voice response (IVR) applications. The product set includes the LumenVox Automatic Speech Recognizer (ASR) and Text-to-Speech (TTS) Server. Both products are used in conjunction with the LumenVox Media Server which provides an interface to Avaya Aura® Experience Portal using the Media Resource Control Protocol (MRCP). LumenVox Call Progress Analysis (CPA) solution leverages the strength of LumenVox Automated Speech Recognizer (ASR) by constantly listening for various tones, just as it would when performing speech recognition.

2. General Test Approach and Test Results

General test approach was to test various VoiceXML scripts that exercise various types of grammars in LumenVox ASR and TTS. A predefined set of VoiceXML scripts tested built-in grammars, menu grammars and SRGS grammars. Also, to test several call scenarios that would test the capabilities of LumenVox CPA.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability tests. Feature tests focused on the ability of LumenVox ASR and TTS to successfully exercise appropriate grammar and return expected results. Additionally, feature tests for CPA focused on the ability of LumenVox CPA to detect tones for Voicemail or Answering Machine and Human Voice.

Serviceability testing focused on verifying the ability of LumenVox ASR and TTS server to recover from adverse conditions, such as restart, power failures and network disconnects.

2.2. Test Results

All test cases were passed.

2.3. Support

To obtain technical support for LumenVox:

- **Web:** www.lumenvox.com/help/
- **Email:** support@lumenvox.com
- **Phone:** (858)707-7700

3. Reference Configuration

Following diagram shows the configuration used during interoperability compliance test.

Reference configuration consisted of:

- Avaya Aura® Experience Portal
- Avaya Aura® Communication Manager
- Avaya Aura® Session Manager
- Avaya Aura® System Manager
- Avaya G450 Media Gateway
- Avaya 9600 Series Deskphones
- LumenVox

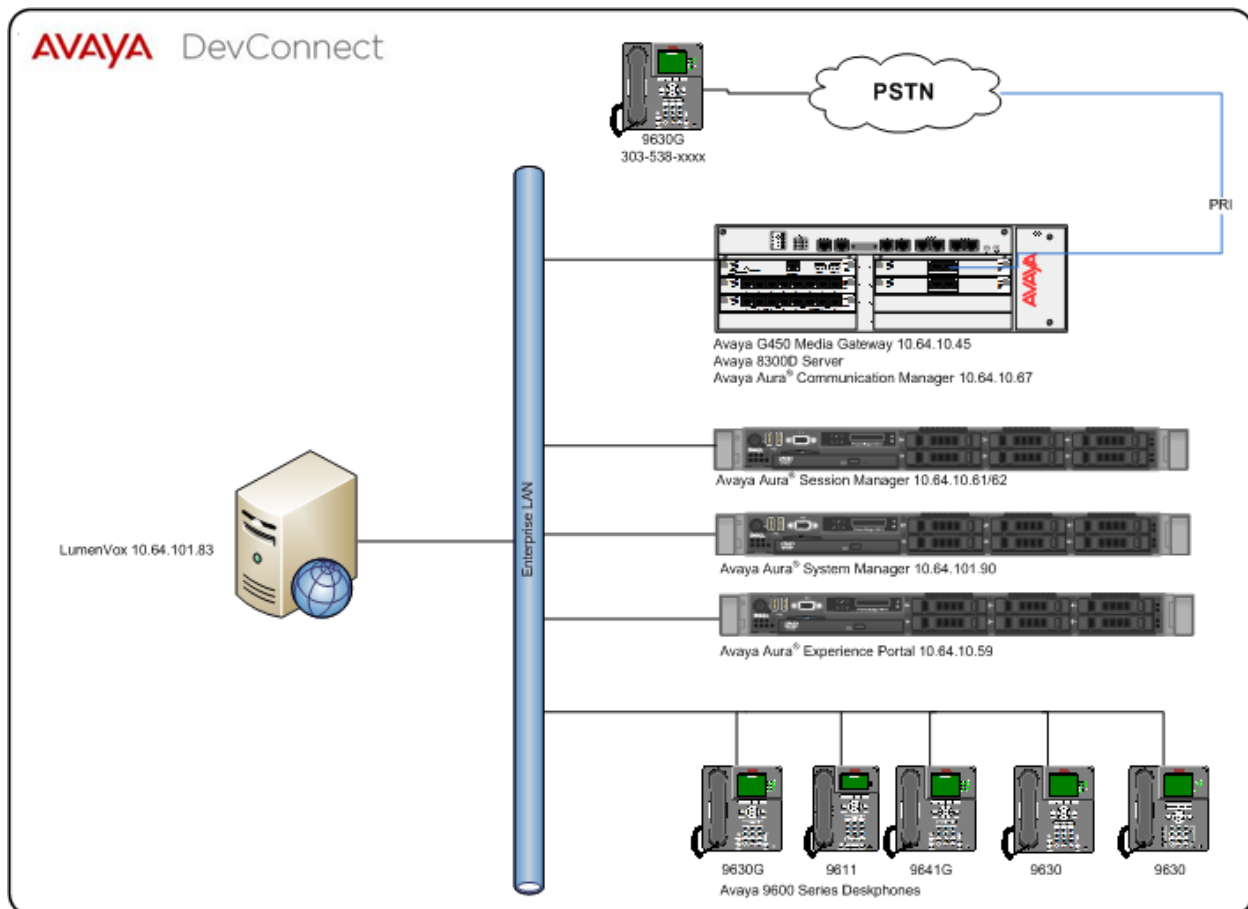


Figure 1: Reference Configuration

3.1. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Experience Portal	7.0
Avaya G450 Media Gateway	31.20.1
Avaya Aura® Communication Manager	6.3 SP6
Avaya 9600 Series IP Telephones <ul style="list-style-type: none">• 96x1 IP Telephones - H.323• 96x1 IP Telephones – SIP• 96x0 IP Telephones – H.323	6.23 6.3.1 3.22
Avaya Aura® Session Manager	6.3 SP7
Avaya Aura® System Manager	6.3 SP2
LumenVox	12.1

4. Configure Avaya Aura® Experience Portal

Avaya Aura® Experience Portal is configured via the Experience Portal Manager (EPM) web interface. To access the web interface enter `http://<ip-addr>/` as the URL in a web browser where `<ip-addr>` is the IP address of the EPM. Log in using the appropriate credentials.

AVAYA Welcome, admin
Last logged in Jun 6, 2014 at 8:27:59 AM MDT

Avaya Aura® Experience Portal 7.0 (ExperiencePortal) Home Help Logoff

Expand All | Collapse All

You are here: Home

Avaya Aura® Experience Portal Manager

Avaya Aura® Experience Portal Manager (EPM) is the consolidated web-based application for administering Experience Portal. Through the EPM interface you can configure Experience Portal, check the status of an Experience Portal component, and generate reports related to system operation.

Installed Components

Media Processing Platform
Media Processing Platform (MPP) is an Avaya media processing server. When an MPP receives a call from a PBX, it invokes a VoiceXML (or CCXML) application on an application server. It then communicates with ASR and TTS servers as necessary to process the call.

Email Service
Email Service is an Experience Portal feature which provides e-mail capabilities.

Proactive Outreach Manager
Avaya Proactive Outreach Manager (POM) provides a solution for unified, multichannel, inbound and outbound architecture, with the capability to communicate through different channels of interaction, from Short Message Service (SMS) to e-mail to the traditional voice and video.

Short Message Server
Short Message Server (SMS) is an Experience Portal feature which provides SMS capabilities.

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4.1. Add VoIP Connections

On the left pane, click on **VoIP Connections**.

4.1.1. SIP Connection

To add a SIP Connection, click on **SIP** tab on **VoIP Connections** page (not shown).

- Fill in **Name**.
- In the **Address** and **Port** boxes, fill the the IP address and Port of SM.
- In **SIP Domain**, type in the domain.
- Type in **Maximum Simultaneous Calls**.
- Rest of all values is left at **Default**.
- Click **Save** to save changes.

Add SIP Connection

Use this page to add a new SIP connection.

Name:

Enable: ☒ Yes ☐ No

Proxy Transport:

☒ Proxy Servers ☐ DNS SRV Domain

Address	Port	Priority	Weight	
<input type="text" value="10.64.10.62"/>	<input type="text" value="5060"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	Remove

[Additional Proxy Server](#)

Listener Port:

SIP Domain:

P-Asserted-Identity:

Maximum Redirection Attempts:

Consultative Transfer: ☒ INVITE with REPLACES ☐ REFER

SIP Reject Response Code: ☒ ASM (503) ☐ SES (480) ☐ Custom

SIP Timers

T1: milliseconds

T2: milliseconds

B and F: milliseconds

Call Capacity

Maximum Simultaneous Calls:

☒ All Calls can be either inbound or outbound

☐ Configure number of inbound and outbound calls allowed

Save **Cancel** **Help**

4.2. Add Speech Servers

On the left pane, click on **Speech Servers**.

4.2.1. ASR Server

To add an ASR server, click on **ASR** tab, and click **Add** (not shown).

- Type in a **Name**.
- Set **Enable** to **Yes**.
- Set **Engine Type** to **Nuance**.
- Type in the IP address of LumenVox Automated Speech Recognizer in **Network Address**.
- In **Base Port**, type in **554**.
- Type in appropriate value in **Total Number of Licensed ASR Resources**.
- Set **New Connection per Session** to **Yes**.
- Set **Languages** to **English(USA) en-US**.
- Set **RTSP URL** to **/<lumenvox_ASR_IP>/media/speechrecognizer**.
- Click **Save** to save changes.

Add ASR Server

Use this page to configure Experience Portal to communicate with a new ASR server.

Name:	<input type="text" value="LumenVox_ASR"/>
Enable:	<input checked="" type="radio"/> Yes <input type="radio"/> No
Engine Type:	<input type="text" value="Nuance"/>
Network Address:	<input type="text" value="10.64.101.83"/>
Base Port:	<input type="text" value="554"/>
Total Number of Licensed ASR Resources:	<input type="text" value="10"/>
New Connection per Session:	<input checked="" type="radio"/> Yes <input type="radio"/> No
Languages:	<div><div>Dutch(Netherlands) nl-NL</div><div>English(Australia) en-AU</div><div>English(UK) en-GB</div><div>English(India) en-IN</div><div>English(Singapore) en-SG</div><div>English(USA) en-US</div></div>
MRCP	
Ping Interval:	<input type="text" value="15"/> seconds
Response Timeout:	<input type="text" value="4"/> seconds
Protocol:	<input type="text" value="MRCP V1"/>
RTSP URL:	<input type="text" value="10.64.101.83/media/speechrecognizer"/>
<div><div>Save</div><div>Cancel</div><div>Help</div></div>	

4.2.2. TTS Server

To add a TTS server, click on **TTS** tab on **Speech Servers** page, and click **Add** (not shown).

- Type in a **Name**.
- Set **Enable** to **Yes**.
- Set **Engine Type** to **Nuance**.
- Type in the IP address of LumenVox Text-to-Speech in **Network Address**.
- In **Base Port**, type in **554**.
- Type in appropriate value in **Total Number of Licensed TTS Resources**.
- Set **New Connection per Session** to **Yes**.
- Set **Languages** to **English(USA) en-US Jennifer F**.
- Set **RTSP URL** to **/<lumenvox_ASR_IP>/media/speechsynthesizer**.
- Click **Save** to save changes.

Add TTS Server

Use this page to configure Experience Portal to communicate with a new TTS server.

Name:	<input type="text" value="LumenVox_TTS"/>
Enable:	<input checked="" type="radio"/> Yes <input type="radio"/> No
Engine Type:	<input type="text" value="Nuance"/>
Network Address:	<input type="text" value="10.64.101.83"/>
Base Port:	<input type="text" value="554"/>
Total Number of Licensed TTS Resources:	<input type="text" value="10"/>
New Connection per Session:	<input checked="" type="radio"/> Yes <input type="radio"/> No
Voices:	<div><div>English(Irish) en-IE Moira F</div><div>English(South_African) af-ZA Tessa F</div><div>English(Scottish) en-SC Fiona F</div><div>English(USA) en-US Donna F</div><div>English(USA) en-US Erica F</div><div>English(USA) en-US Jennifer F</div></div>
MRCP	
Ping Interval:	<input type="text" value="15"/> seconds
Response Timeout:	<input type="text" value="4"/> seconds
Protocol:	<input type="text" value="MRCP V1"/>
RTSP URL:	<input type="text" value="10.64.101.83/media/speechsynthesiz"/>
<div><div>Save</div><div>Cancel</div><div>Help</div></div>	

5. Configure LumenVox

Configure LumenVox as follows:

1. Edit the **media_server.conf** file
 - a. On Windows, located in **Program Files\LumenVox\Engine\config**
 - b. On Linux, located in **/etc/lumenvox**
2. Set **mrcp_server_ip** to the IP address of the server running the LumenVox Media Sever.
 - a. It should not be left as the default **127.0.0.1**

Other configurations are available that may be turned on if desired:

- Logging for debugging can be enabled by editing the **client_property.conf** file and setting **LOGGING_VERBOSITY = 3**
- Utterance files for speech tuning can be enabled by editing the **media_server.conf** file and setting
 - **enable_sre_logging = 3** for ASR utterance files

For more information on available configuration options, see the [LumenVox Knowledge Base](#) or the [LumenVox Developers Network](#).

6. Verification Steps

6.1. Avaya Aura® Experience Portal

This section provides the verification steps that may be performed to verify that Avaya Aura® Experience Portal can run LumenVox ASR and TTS servers.

1. From the EPM web interface, verify that the MPP servers are online and running. On the left pane, click on **MPP Manager**.

MPP Manager (Jun 9, 2014 4:25:18 AM MDT)

[Refresh](#)

This page displays the current state of each MPP in the Experience 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: Jun 9, 2014 4:25:16 AM MDT

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

State Commands

[Start](#)[Stop](#)[Restart](#)[Reboot](#)[Halt](#)[Cancel](#)

Restart/Reboot Options

☒ One server at a time

☐ All servers

Mode Commands

[Offline](#)[Test](#)[Online](#)[Help](#)

2. Verify that the ports on the MPP server are in service. On the left lane, click on **Port Distribution**. Select the MPP server and click **OK** (not shown).

Port Distribution Report (Jun 9, 2014 4:26:42 AM MDT)

[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.

Servers: MPPLocal

Total Ports: 10

Last Poll: Jun 9, 2014 4:26:22 AM MDT

Port	Mode	State	Port Group	Protocol	Current Allocation	Base Allocation
10	Online	In service	sm1sip	SIP_Trunk	MPPLocal	

[Help](#)

6.2. LumenVox

From a command line, run the following command:

- Linux:
 - `/usr/bin/lv_show_config -mrCP_test`
- Windows:
 - `cd "C:\Program Files\LumenVox\Engine\"`
 - `LVShowConfig.exe -mrCP_test`

This will run a series of tests to verify that ASR, TTS and the LumenVox Media Server are working correctly:

```
[interop@LumenVox bin]$ ./lv_show_config -mrCP_test

=====

Testing LumenVox Media Server

Testing MRCP v1 TTS Synthesis
The MRCP v1 TTS Synthesis completed successfully

Testing MRCP v2 TTS Synthesis (SIP/UDP)
The MRCP v2 TTS Synthesis completed successfully

Testing MRCP v2 TTS Synthesis (SIP/TCP)
The MRCP v2 TTS Synthesis completed successfully

Testing MRCP v1 ASR Recognition
The MRCP v1 ASR Recognition completed successfully

Testing MRCP v2 ASR Recognition (SIP/UDP)
The MRCP v2 ASR Recognition completed successfully

Testing MRCP v2 ASR Recognition (SIP/TCP)
The MRCP v2 ASR Recognition completed successfully

=====
```

Note that if ASR or TTS is not licensed, some or all of these tests may fail (e.g. if only TTS is licensed, it is expected that ASR tests fail).

7. Conclusion

These Application Notes describe the configuration steps required to integrate LumenVox Automated Speech Recognizer, LumenVox Text-to-Speech Server and Call Progress Analysis with Avaya Aura® Experience Portal. All feature and serviceability test cases were completed successfully.

8. Additional References

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

[1] Administering Avaya Aura® Experience Portal, Release 7.0, Issue 1, December 2013

LumenVox documentation is always available from <http://www.lumenvox.com/knowledgebase/>

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