

Avaya Solution & Interoperability Test Lab

Application Notes for LumenVox Speech Suite 19.2 with Avaya Aura® Experience Portal 8.1 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for LumenVox Speech Suite 19.2 to interoperate with Avaya Aura® Experience Portal 8.1.

In the compliance testing, LumenVox Speech Suite provided Automatic Speech Recognizer, Text-to-Speech, and Call Progress Analysis capabilities for Experience Portal applications.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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 for LumenVox Speech Suite 19.2 to interoperate with Avaya Aura® Experience Portal 8.1.

In the compliance testing, LumenVox used Media Resource Control Protocol (MRCP) version 2 to provide Automatic Speech Recognizer (ASR), Text-to-Speech (TTS), and Call Progress Analysis (CPA) capabilities for Experience Portal applications.

2. General Test Approach and Test Results

The feature test cases were performed manually with incoming calls from PSTN and Avaya H.323 and SIP endpoints to Experience Portal to invoke sample VXML applications that used LumenVox for ASR and TTS.

The CPA test cases were performed by sending audio types such as answering machine tone directly to LumenVox and verifying proper detection. This was accomplished by using a LumenVox utility via Linux shell to feed pertinent audio files and verify proper detection results.

The serviceability test cases were performed manually by disconnecting/reconnecting the Ethernet connection to LumenVox.

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For testing associated with these Application Notes, the interfaces between Avaya systems and LumenVox did not include use of any specific encryption features.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing focused on verifying the following on LumenVox:

- Communication with Experience Portal via MRCP V2.
- ASR and TTS capabilities invoked by incoming calls to sample VXML applications on Experience Portal, and CPA capabilities invoked by a LumenVox utility via the Linux shell.
- ASR capabilities including DTMF, built-in grammars, menu grammars, internal grammars, and external grammars.
- TTS capabilities including male and female voices and prosody to change rate, volume, and pitch of voices.
- CPA capabilities including live person, answering machine, fax, and special information tone detection.
- Proper responses for scenarios involving invalid input and missing input from caller.

The serviceability testing focused on verifying the ability of LumenVox to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to LumenVox.

2.2. Test Results

All test cases were executed and verified. The following is the observation on LumenVox from the compliance testing.

• The current release of LumenVox does not support G.729 and requires the G.711 codec to be configured on Experience Portal. Note that the G.711 codec variants are enabled on Experience Portal by default, and the MPP Server VoIP Settings were checked to make certain the proper G711 codec variant remains enabled as shown in **Section 5.5**.

2.3. Support

Technical support on LumenVox can be obtained through the following:

• **Phone:** (858) 707-7700

• Web: https://www.lumenvox.com/support

• Email: support@LumenVox.com

3. Reference Configuration

The configuration used for the compliance testing is shown in **Figure 1**.

The detailed administration of basic connectivity between Communication Manager, Experience Portal, System Manager, and Session Manager are not the focus of these Application Notes and will not be described.

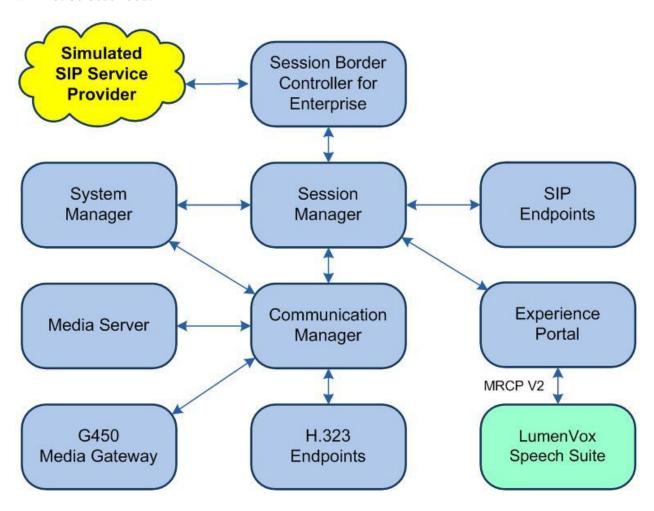


Figure 1: Compliance Testing Configuration

4. Equipment and Software Validated

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

Equipment/Software	Release/Version	
Avaya Aura® Communication Manager in Virtual Environment	8.1.3 (8.1.3.0.1.890.26685)	
Avaya G650 Media Gateway	NA	
Avaya Aura® Media Server in Virtual Environment	8.0.2.138	
Avaya Aura® Experience Portal in Virtual Environment	8.1 (8.1.0.0.0316)	
Avaya Aura® Session Manager in Virtual Environment	8.1.3 (8.1.3.0.813014)	
Avaya Aura® System Manager in Virtual Environment	8.1.3 (8.1.3.0.1012091)	
Avaya J179 IP Deskphone (H.323)	6.8502	
Avaya J169 IP Deskphone (SIP)	4.0.7.1.5	
LumenVox Speech Suite on CentOS Linux	19.2.100 7.9.2009	

5. Configure Avaya Aura® Experience Portal

This section provides the procedures for configuring Experience Portal. The procedures include the following areas:

- Administer custom properties files
- Launch Experience Portal Manager
- Administer ASR server
- Administer TTS server
- Administer MPP VoIP settings
- Restart MPP server
- Administer application

5.1. Administer Custom Properties Files

When adding an ASR server and TTS server on Experience Portal, the engine type option does not include LumenVox by default. For customers that prefer to have LumenVox appear in the engine type selection, two custom properties files can be obtained from LumenVox to apply to Experience Portal.

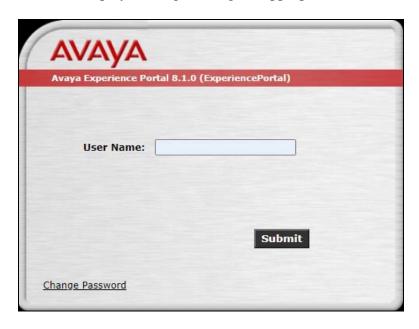
Copy the **asrCustom.properties** and **ttsCustom.properties** files provided by LumenVox to the /**tmp** directory of the Experience Portal server containing the EPM component. Access the Linux shell of this server as super user.

Navigate to the **\$CATALINA_HOME/lib/config** directory. Enter the commands below to save a copy of the original **asrCustom.properties** and **ttsCustom.properties** files, replace the two files with the version provided by LumenVox, and restart the **vpms** service as shown below.

```
[xxxx@ep-epm ~] # cd $CATALINA_HOME/lib/config
[xxxx@ep-epm config] #
[xxxx@ep-epm config] # cp asrCustom.properties asrCustom.properties.orig
[xxxx@ep-epm config] # cp ttsCustom.properties ttsCustom.properties.orig
[xxxx@ep-epm config] #
[xxxx@ep-epm config] # cp /tmp/asrCustom.properties .
[xxxx@ep-epm config] # cp /tmp/ttsCustom.properties .
[xxxx@ep-epm config] #
[xxxx@ep-epm config] # systemctl restart vpms
```

5.2. Launch Experience Portal Manager

Access the Experience Portal Manager (EPM) web interface by using the URL "https://ip-address" in an Internet browser window, where "ip-address" is the IP address of the EPM server. The screen below is displayed. Log in using the appropriate credentials.



The Avaya Experience Portal Manager screen is displayed next.



5.3. Administer ASR Server

Select System Configuration → Speech Servers from the left pane. In the Speech Servers screen (not shown), select the ASR tab followed by Add. The Add ASR Server screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

• Name: A descriptive name.

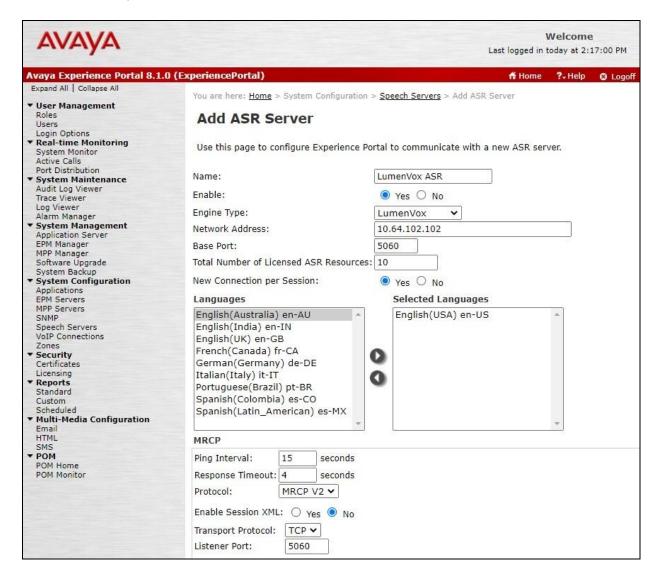
• Engine Type: "LumenVox"

• Network Address: IP address of the LumenVox server.

• **Selected Voices:** Select the desired languages.

• **Protocol:** "MRCP V2"

For **Total Number of Licensed ASR Resources**, enter the pertinent number of ASR license from LumenVox, in this case "10".



5.4. Administer TTS Server

Select System Configuration \rightarrow Speech Servers from the left pane. In the Speech Servers screen (not shown), select the TTS tab followed by Add. The Add TTS Server screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

• Name: A descriptive name.

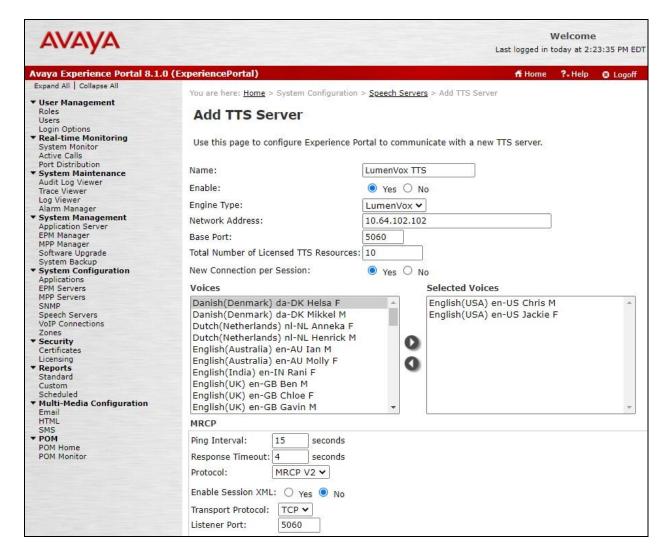
• Engine Type: "LumenVox"

• Network Address: IP address of the LumenVox server.

• **Selected Voices:** Select the desired voices.

• **Protocol:** "MRCP V2"

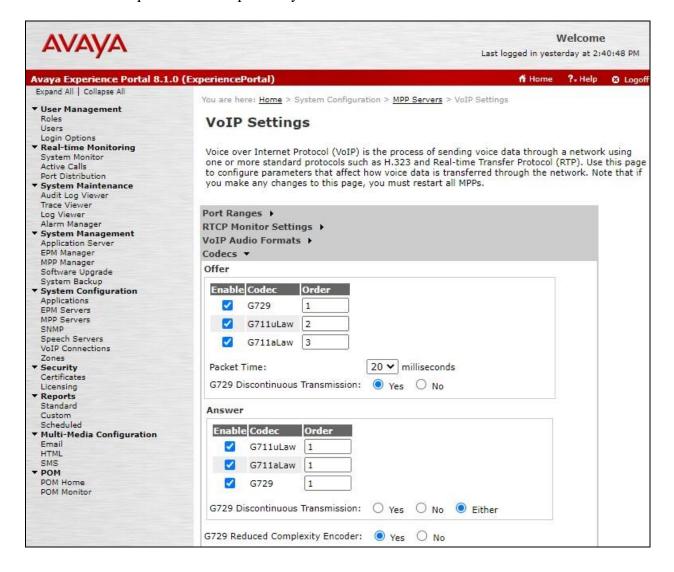
For **Total Number of Licensed TTS Resources**, enter the pertinent number of TTS license from LumenVox, in this case "10".



5.5. Administer MPP VoIP Settings

Select System Configuration → MPP Servers from the left pane. In the MPP Servers screen (not shown), select VoIP Settings.

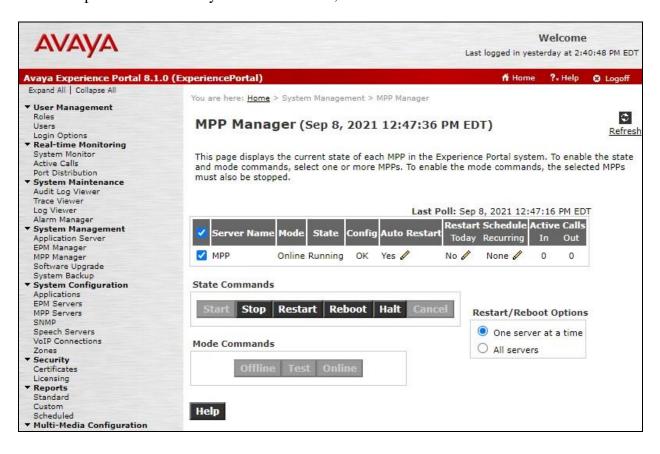
The **VoIP Settings** screen is displayed. Expand the **Codecs** sub-section. Make certain that the pertinent G711 variant is checked in both **Offer** and **Answer** as shown below. Note that the G711 codec is required for interoperability with LumenVox.



5.6. Restart MPP Server

Select **System Management** → **MPP Manager** from the left pane to display the **MPP Servers** screen.

Select the pertinent MPP entry and click **Restart**, as shown below.



5.7. Administer Application

Select **System Configuration** \rightarrow **Applications** from the left pane. In the **Applications** screen (not shown), click **Add** to add a new application for testing with LumenVox.

The **Add Application** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

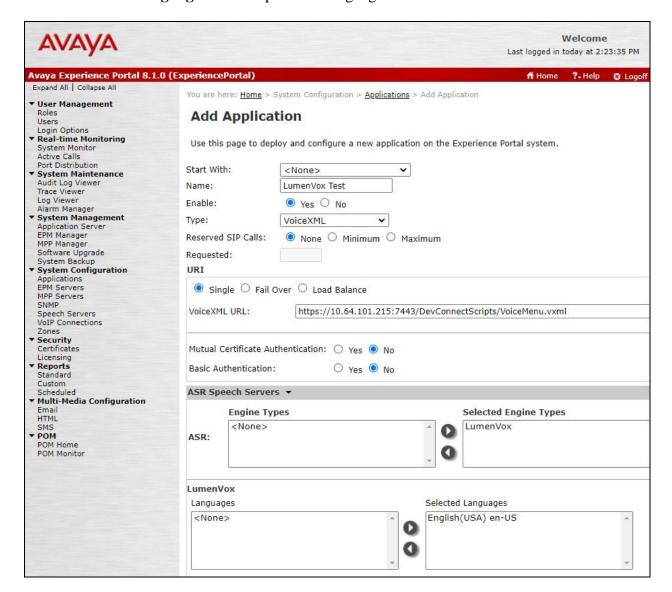
• Name: A descriptive name.

• **Type:** Select the pertinent application type, in this case "VoiceXML".

• VoiceXML URL: The URL for the pertinent VXML test application.

• **ASR Engine Types:** Select "LumenVox".

• LumenVox Languages: Select pertinent languages.

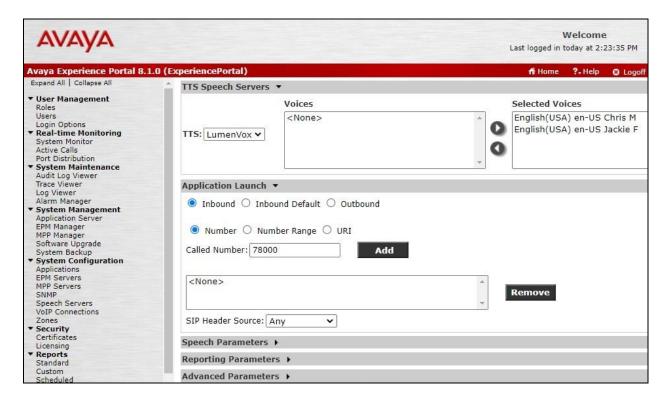


Scroll down to the **TTS Speech Servers** and **Application Launch** sub-sections. Enter the following values for the specified fields and retain the default values for the remaining fields.

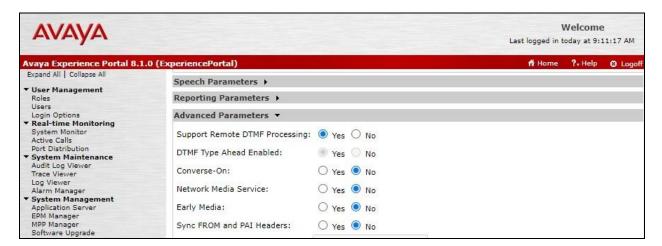
• TTS: "LumenVox"

• TTS Voices: Select the desired voices.

• Called Number: An available number to associate with the application.



Scroll down and expand the **Advanced Parameters** sub-section. Enable **Support Remote DTMF Processing** as shown below to allow LumenVox to perform DTMF processing instead of Experience Portal. Retain the default values in the remaining fields.



6. Configure LumenVox Speech Suite

This section provides the procedures for configuring LumenVox. The procedures include the following areas:

- Launch LumenVox Dashboard
- Administer Media Server

6.1. Launch LumenVox Dashboard

Access the LumenVox Dashboard web interface by using the URL "http://ip-address:8080" in an Internet browser window, where "ip-address" is the IP address of the LumenVox server. The LumenVox Dashboard screen is displayed, as shown below.

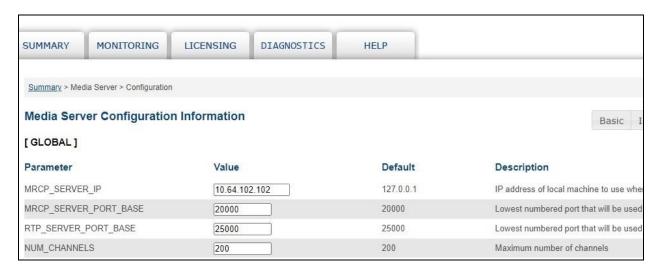
Locate the **Media Server** entry and select the associated **Configuration** button.



6.2. Administer Media Server

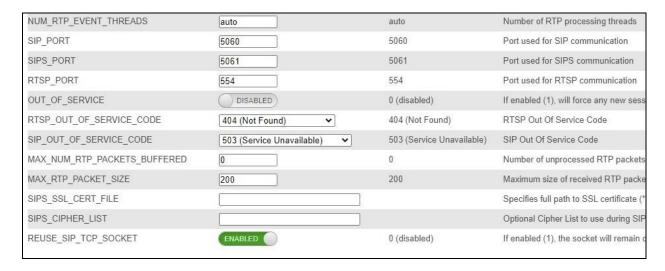
The **Media Server Configuration Information** screen is displayed. Select the **Advanced** (not shown) view toward the top right of screen.

In the **GLOBAL** sub-section, set **MRCP_SERVER_IP** to the IP address of the LumenVox server.



Scroll down to locate the **SIP_PORT** parameter and make certain that the value matches to the base port value for ASR server in **Section 5.3** and for TTS server in **Section 5.4**.

Locate the **REUSE_SIP_TCP_SOCKET** parameter at the end of this sub-section and toggle the setting to **ENABLED** as shown below. Note that this setting retains the socket connection on LumenVox at the end of ASR and TTS exchanges, which is required by Experience Portal.

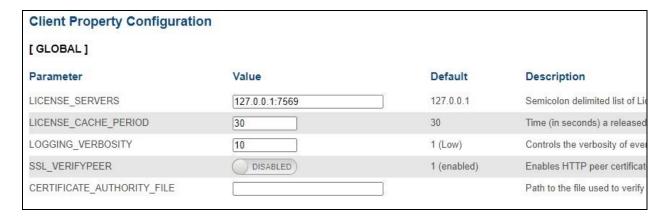


Scroll down to the **VENDORSPECIFIC** sub-section. Set **COMPATIBILITY_MODE** to "1" as shown below. Note that this setting is required for LumenVox to encode responses in the format expected by Experience Portal.

Parameter	Value	Default	Description
WIND_BACK_TIME	480	480	Milliseconds of audio prior to V
BARGE_IN_TIMEOUT	15000	15000	Default number of milliseconds
END_OF_SPEECH_TIMEOUT	20000	20000	Default number of milliseconds before timing out
VAD_STREAM_INIT_DELAY	100	100	Amount non-speech-data before
VAD_BARGEIN_THRESHOLD	50	50	Adjusts the setting of how sure bargein
COMPATIBILITY_MODE	1	0	Enables compatibility encoding
SECURE_CONTEXT	0	0	When set to 1, sensitive recogn
TTS_SECURE_CONTEXT	0	0	When set to 1, sensitive synthe

Scroll down to the **Client Property Configuration** sub-section. For customers with application scripts requiring external files to be accessed by LumenVox via secured HTTPS connection, follow LumenVox documentation to install the CA certificate supplied by the customer or disable **SSL_VERIFYPEER** if desired.

In the compliance testing, several application scripts required HTTPS access to external grammar files and the **SSL_VERIFYPEER** parameter was disabled for simplicity.

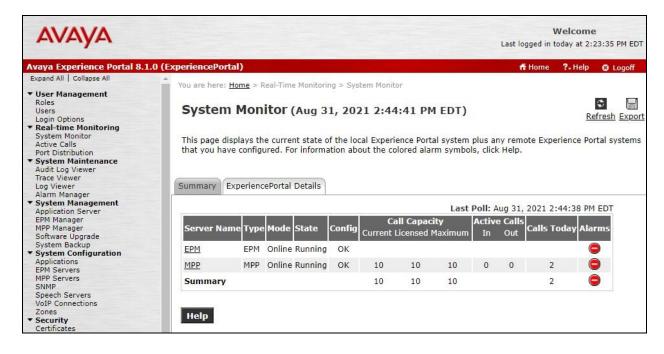


7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Experience Portal and LumenVox.

From the EPM web interface, select **Real-time Monitoring > System Monitor** from the left pane to display the **System Monitor** screen in the right pane.

Select the **ExperiencePortal Details** tab followed by the applicable **MPP** entry.

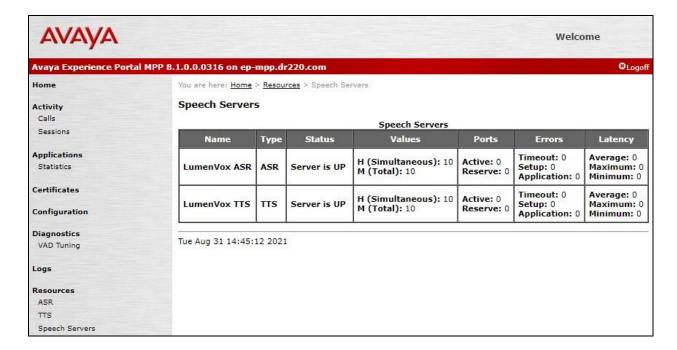


The **MPP Details** screen is displayed. Select **Service Menu** toward the bottom of the screen to launch the Media Server Service Menu.



The Avaya Experience Portal MPP screen is launched and displayed. Select Resources > Speech Servers from the left pane to display the Speech Servers screen in the right pane.

Verify that there are two entries for LumenVox with **Status** of "**Server is UP**" for both, as shown below.



8. Conclusion

These Application Notes describe the configuration steps required for LumenVox Speech Suite 19.2 to interoperate with Avaya Experience Portal 8.1. All feature and serviceability test cases were completed with an observation noted in **Section 2.2**.

9. Additional References

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

- **1.** *Administering Avaya Aura*® *Communication Manager*, Release 8.1.x, Issue 8, November 2020, available at http://support.avaya.com.
- **2.** Administering Avaya Experience Portal, Release 8.1, Issue 1, July 2021, available at http://support.avaya.com.

LumenVox documentation is available at https://www.lumenvox.com/knowledgebase.

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