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

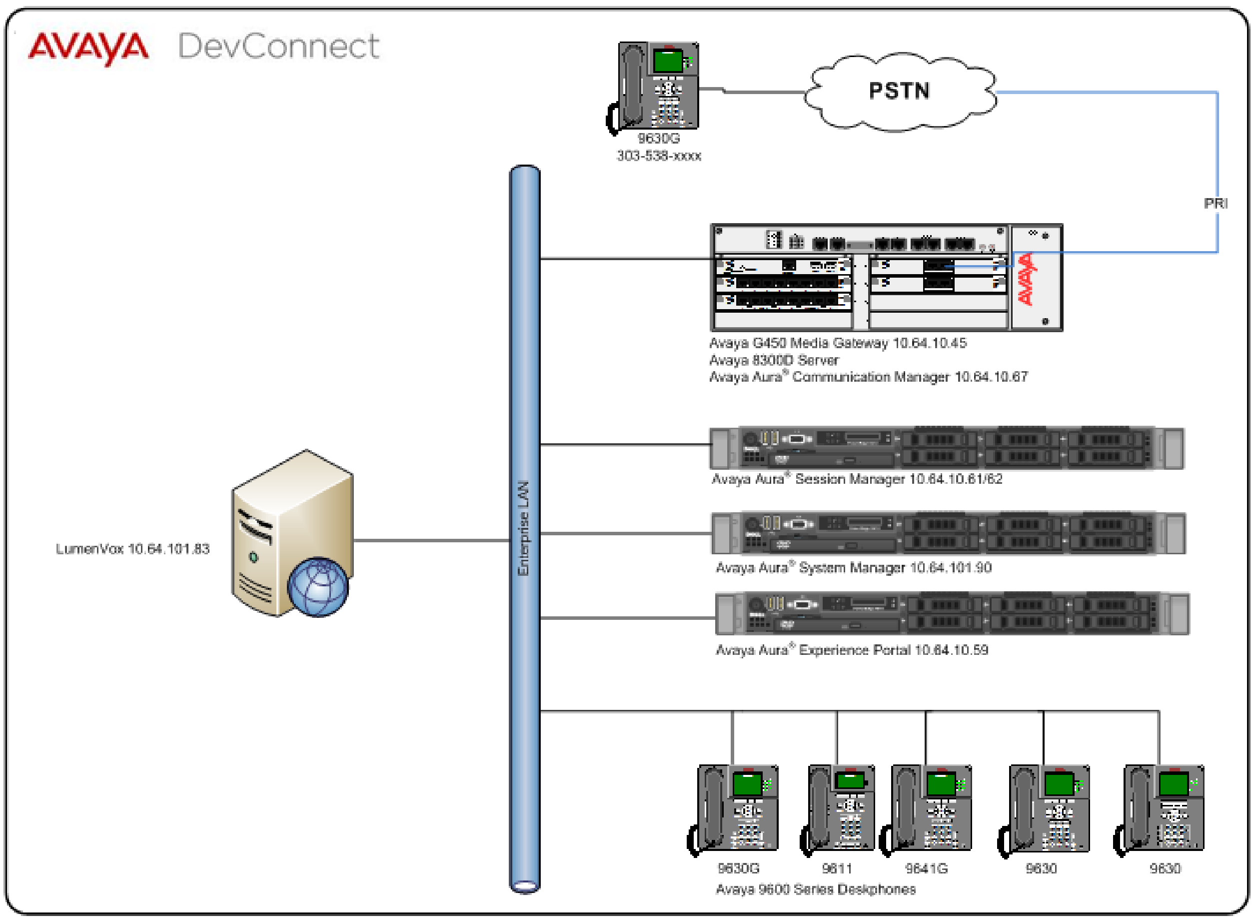


Figure 1: Reference Configuration

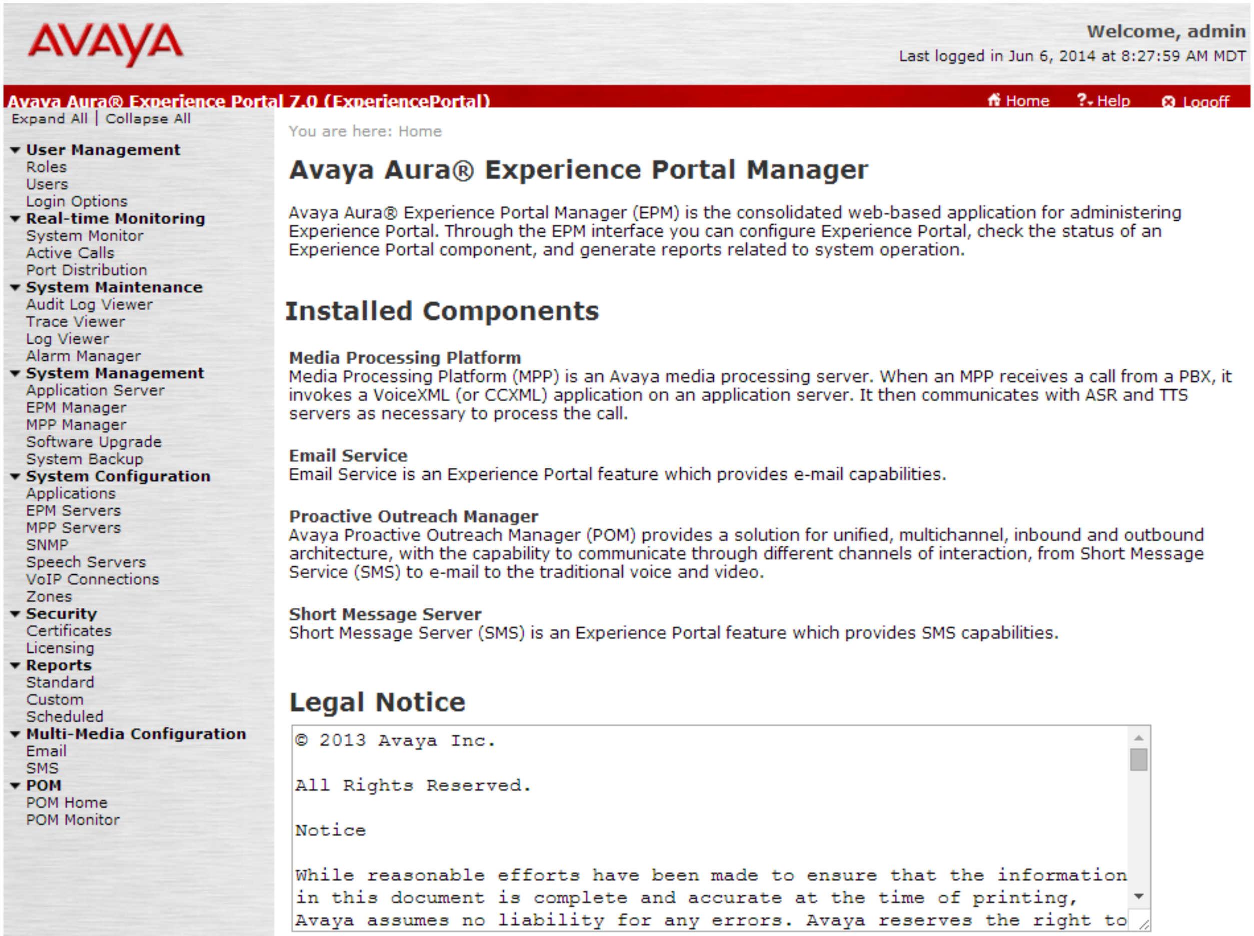
3.1. Equipment and Software Validated

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 • 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.



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.

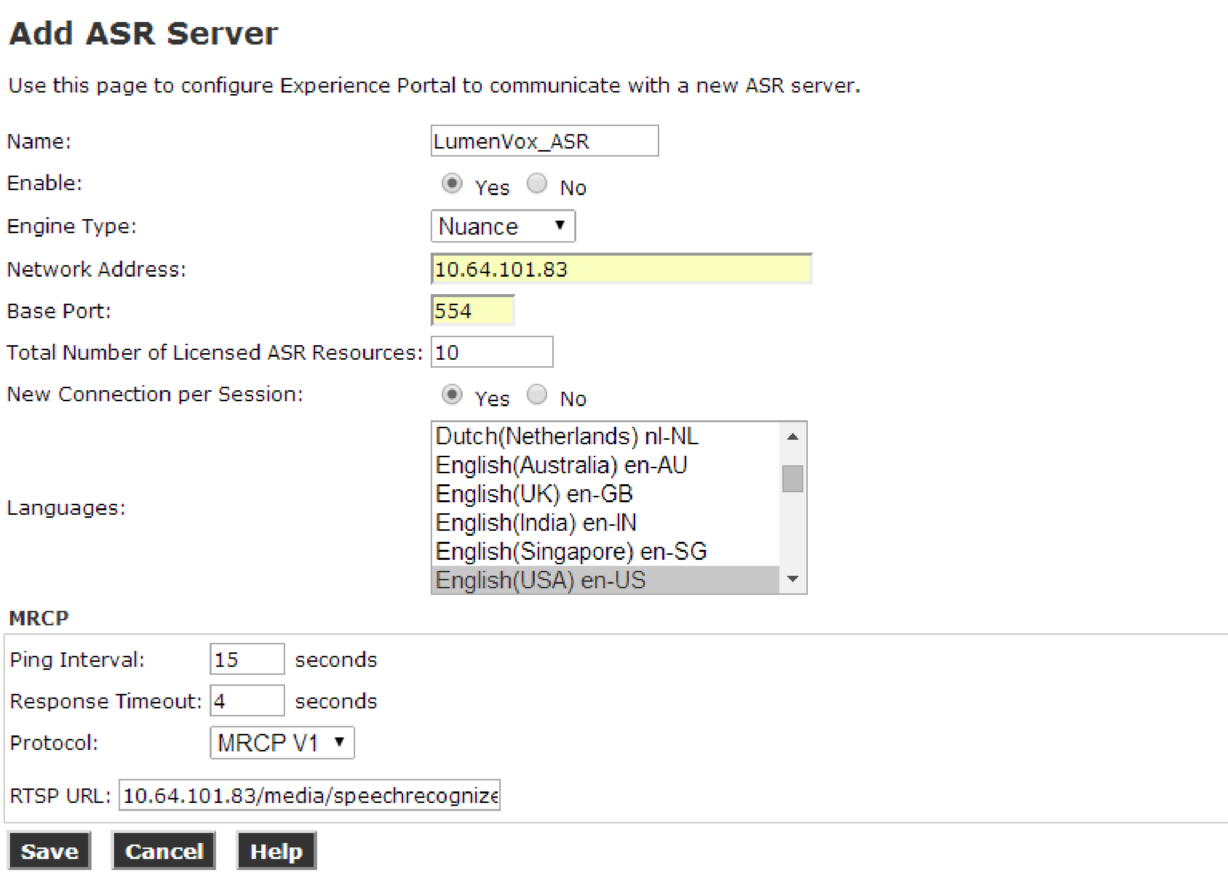
4.2. Add Speech Servers

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).

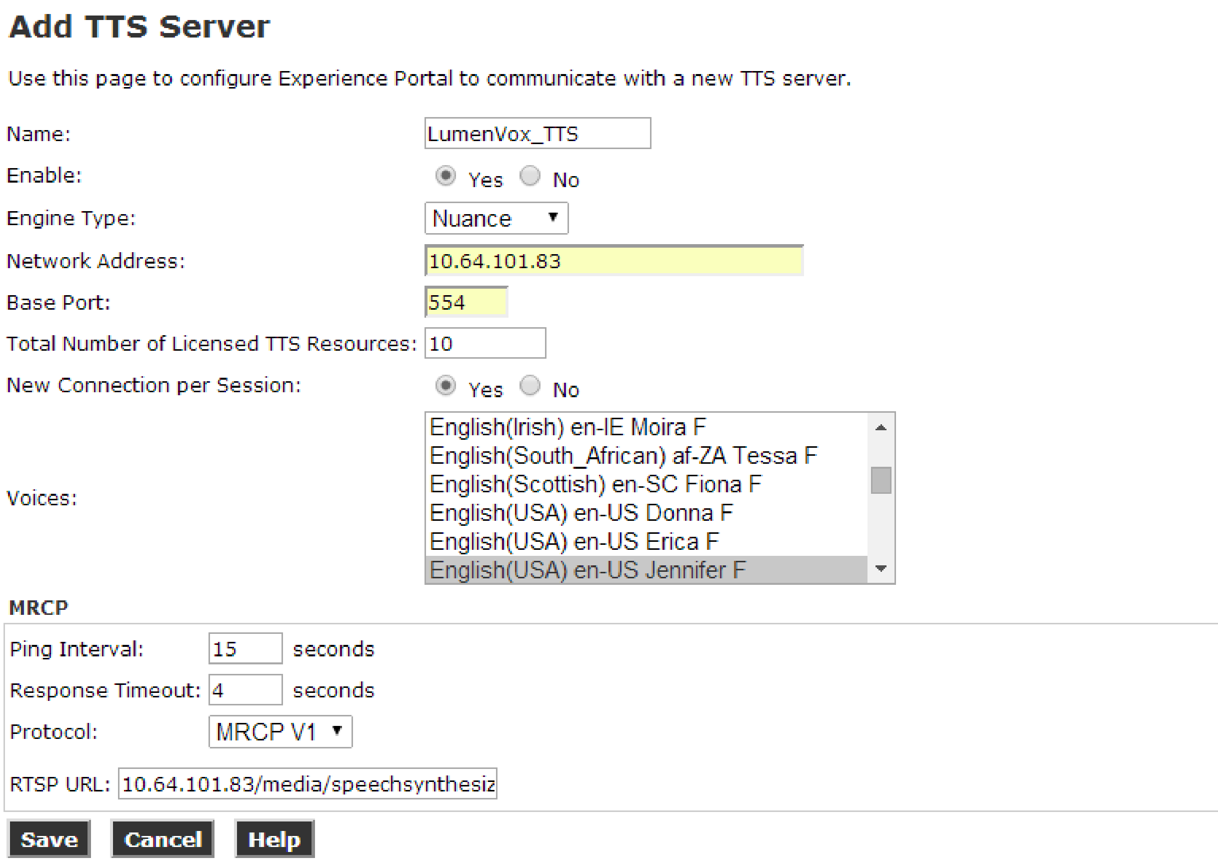
- Type in a **Name.**
- Set **Enable** to **Yes.**
- Set **Engine Type** to **Nuance.**
- Type in the IP address of LumenVox Automated Speech Recognizer in **Network Address.**
- In **Base Port**, type in **554.**
- Type in appropriate value in **Total Number of Licensed ASR Resources.**
- Set **New 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.



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.



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

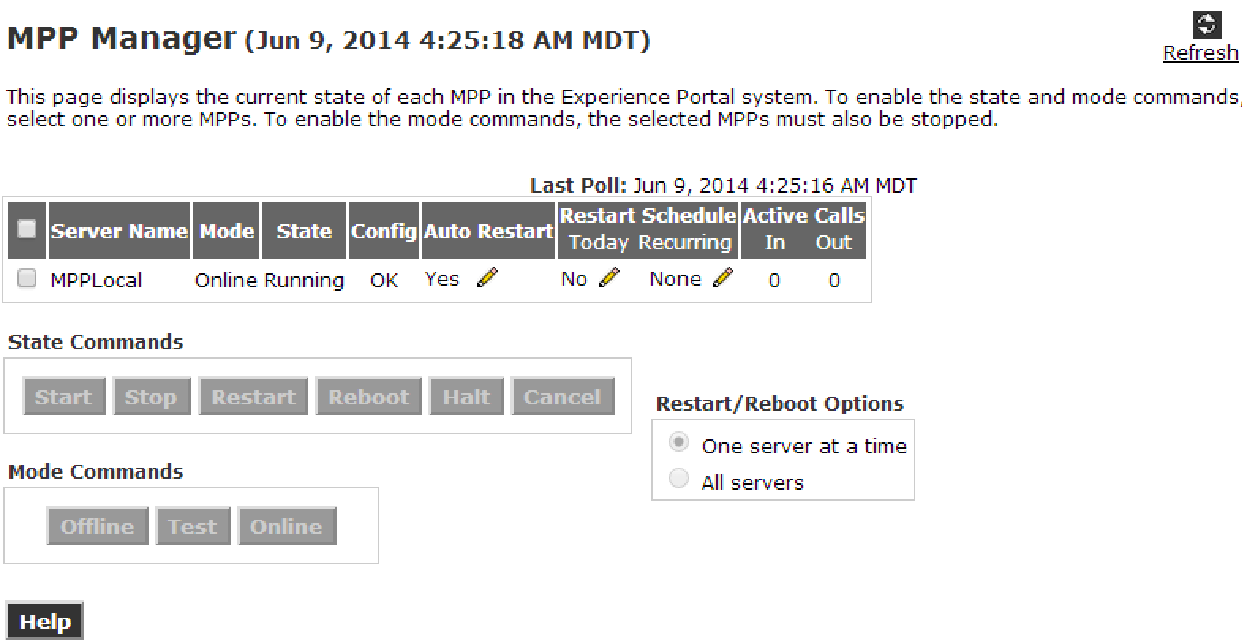
For more information on available configuration options, see the LumenVox Knowledge Base or the LumenVox Developers Network.

6. Verification Steps

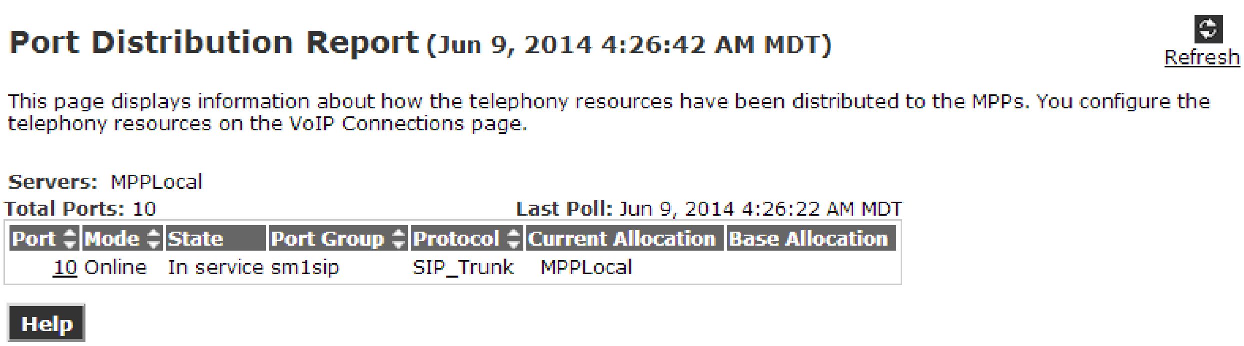
6.1. Avaya Aura® Experience Portal

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**.



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**)**.



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

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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

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```

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, December2013

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