Avaya Solution & Interoperability Test Lab

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

Abstract

These Application Notes describe the configuration steps required to integrate LumenVox Automated Speech Recognizer and LumenVox Text-to-Speech Server 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 and LumenVox Text-to-Speech Server 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).

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 Speech Recognition Grammar Specification (SRGS) grammars.

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.

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 the interoperability compliance test. Reference configuration consisted of:

- Avaya Aura® Experience Portal
- Avaya S8300D Server running Avaya Aura® Communication Manager
- Avaya G450 Media Gateway
- Avaya Aura® Session Manager
- Avaya Aura® System Manager
- Avaya IP Telephones
- Application Server
- LumenVox Automated Speech Recognizer
- LumenVox Text-to-Speech Server

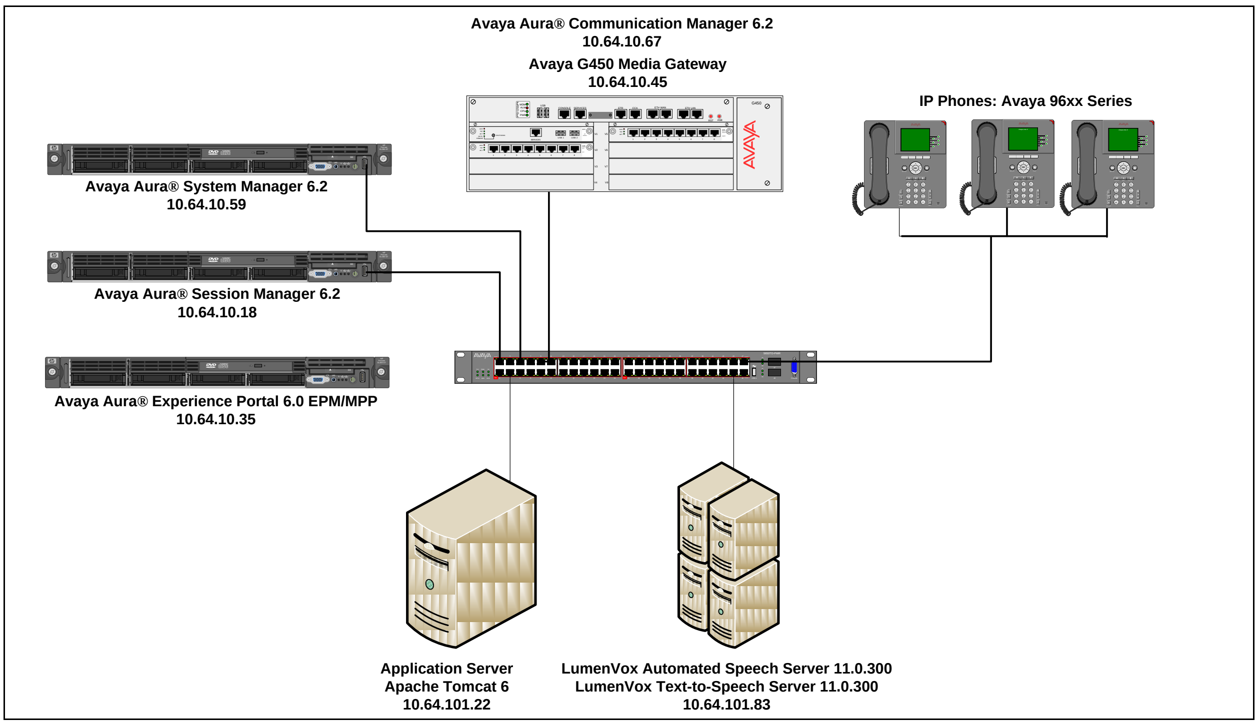


Figure 1: Reference Configuration

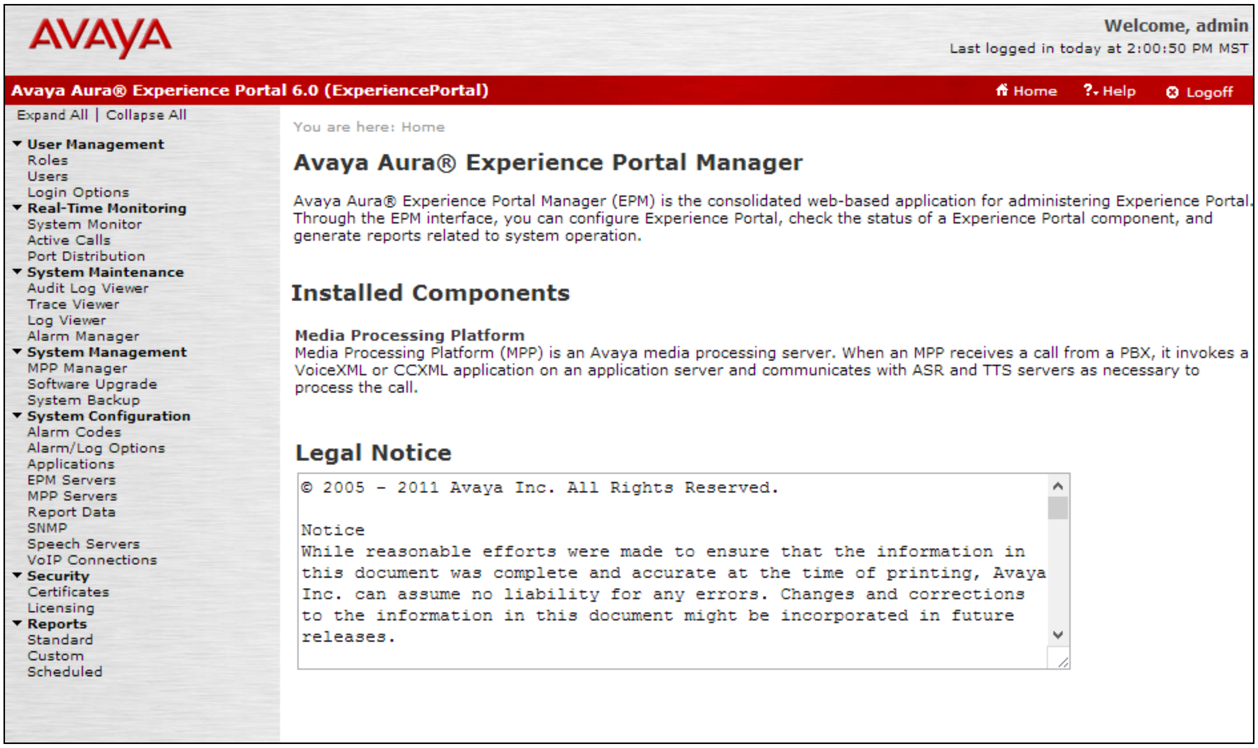
4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Experience Portal	6.0 SP1
Avaya G450 Media Gateway	31.20.1
Avaya S8300D Server running Avaya Aura® Communication Manager	6.2-02.0.823.0
Avaya 9600 Series IP Telephones	H.323 3.1 SIP 2.6.4
Avaya Aura® Session Manager	6.2.2.0.622005
Avaya Aura® System Manager	6.2.12.0
LumenVox Automated Speech Recognizer	11.0.300
LumenVox Automated Speech Recognizer	11.0.300
Tomcat Apache Web Server	6.0

5. 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 Administrator user role.



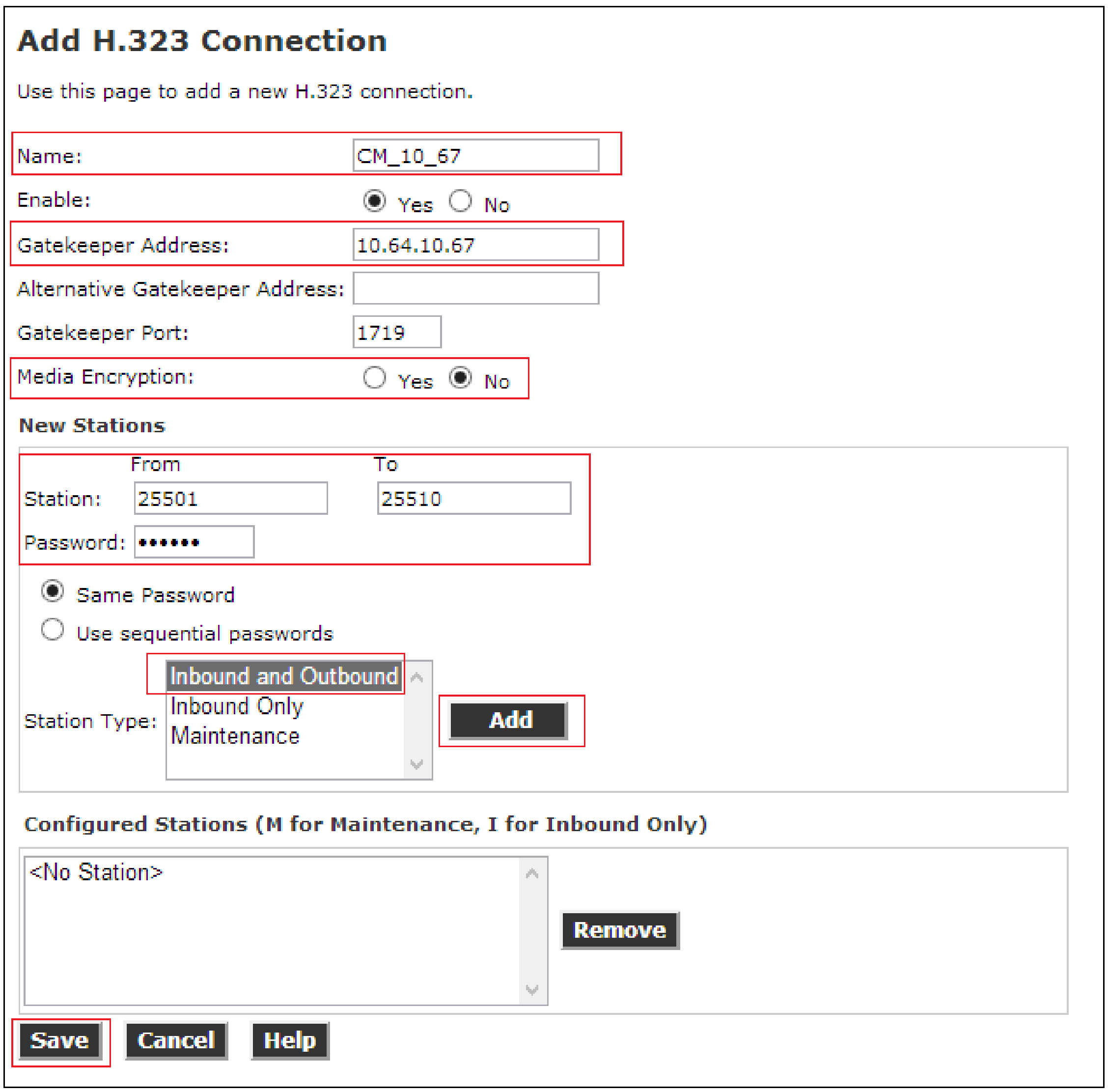
5.1. Add VoIP Connections

On the left pane, click on the **System Configuration →VoIP Connections** tab to configure VoIP connections (not shown).

5.1.1. H.323 Connection

To add a H.323 Connection, click on **H.323** tab (not shown) and click **Add** (not shown)

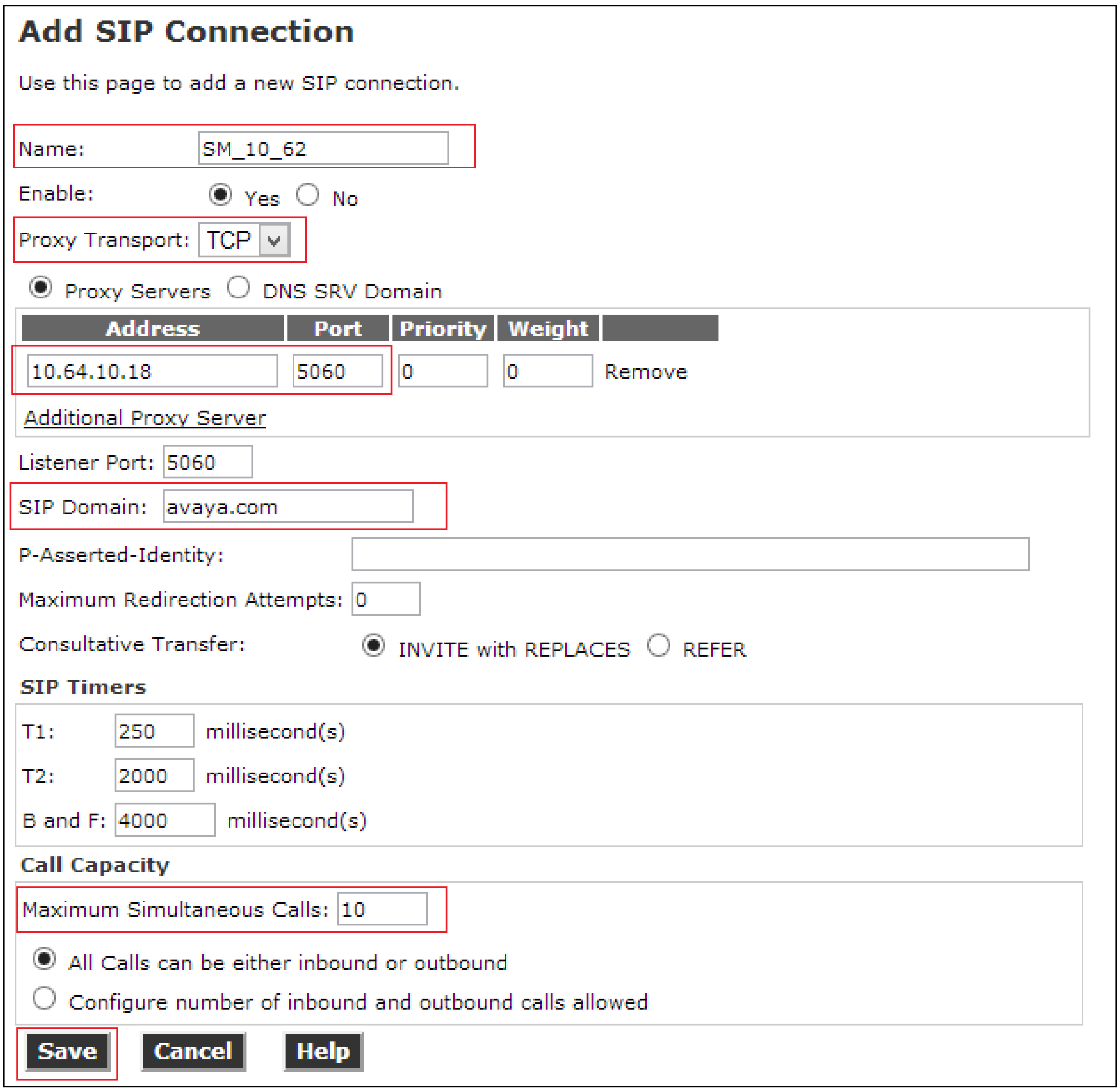
- **Name**: Enter a descriptive name
- **Gatekeeper Address**: Enter the IP address of Communication Manager.
- **Media Encryption**: Set to **No**.
- **New Stations**: Enter **Station From** and **To**, and **Password**. This information will be used from configuration performed on Communication Manager for adding stations for **Inbound and Outbound** and click **Add.**
- Retain the default values in the remaining fields Click **Save** to save changes.



5.1.2. SIP Connection

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

- **Name**: Enter a descriptive name..
- Set **Proxy Transport** to **TCP**.
- In the **Address** and **Port** boxes, enter the IP address and Port of Session Manager.
- **SIP Domain**: Enter the domain used in Session Manager.
- **Maximum Simultaneous Calls**: During the test, **10** was used for the Maximum Simulataneous Calls field.
- Retain the default values in the remaining fields Click **Save** to save changes.



5.2. Add Speech Servers

On the left pane, click on the **System Configuration →Speech Servers** tab to add Speech Server.

5.2.1. ASR Server

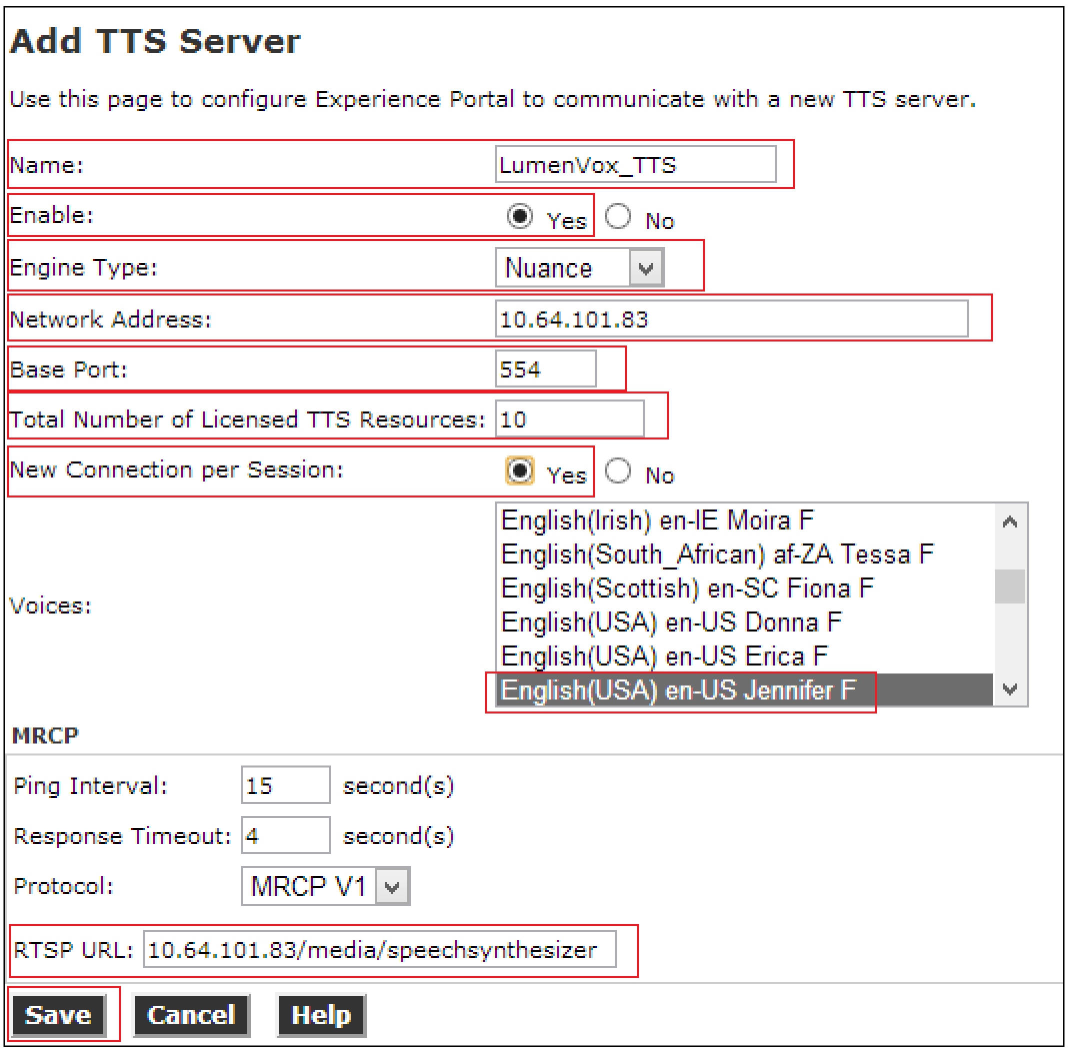
To add an **ASR** server, click on **ASR** tab (not shown), and click **Add (**not shown)**.**

- **Name**: Enter a descriptive name**.**
- **Enable**: Set to **Yes.**
- **Engine Type**: Set to **Nuance**, using the drop down menu
- **Network Address**: Enter the IP address of LumenVox Automated Speech Recognizer**.**
- **Base Port**: Enter **554.**
- **Total Number of Licensed ASR Resources**: Enter an appropriate value
- **New Connection per Session**: Select **Yes.**
- **Languages**: Select **English(USA) en-US.**
- **RTSP URL**: Enter**<LumenVox_ASR_IP address>/media/speechrecognizer**
- Click **Save** to save changes.

5.2.2. TTS Server

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

- **Name**: Enter a descriptive name.
- **Enable**: Set to **Yes.**
- **Engine Type**: Set to **Nuance** using the drop down menu.
- **Network Address**: Enter the IP address of LumenVox Text to Speech server.
- **Base Port**: Enter **554.**
- **Total Number of Licensed TTS Resources**: Enter an appropriate value.
- **New Connection per Session**: Select **Yes.**
- **Languages**: Select **English(USA) en-US Jennifer F.**
- **RTSP URL**: Enter **<LumenVox_ASR_IP address>/media/speechsynthesizer**
- Click **Save** to save changes.



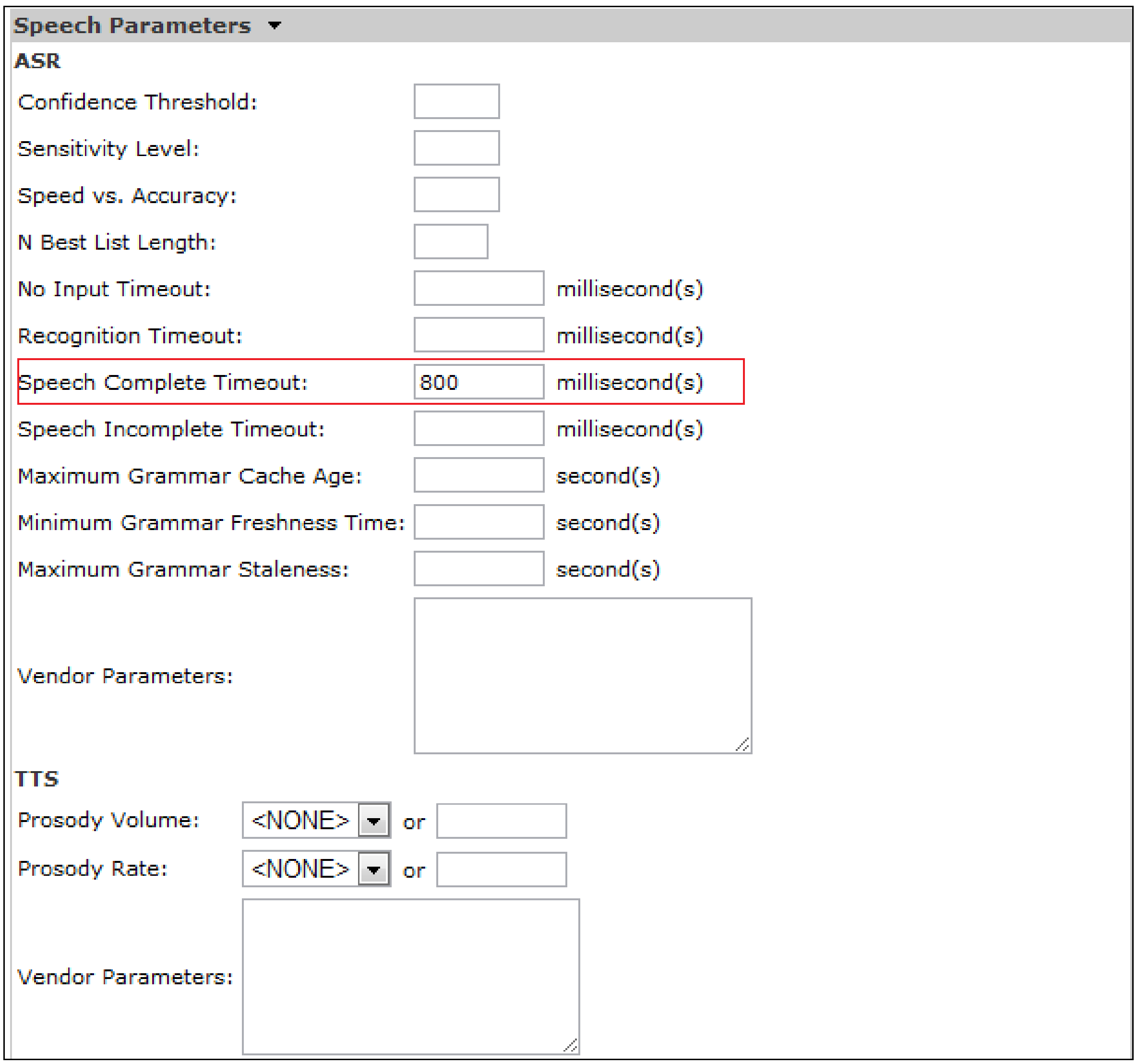
6. Configure LumenVox Automated Speech Recognizer

All configurations for LumenVox applications were performed by a LumenVox Engineer.

Log on to LumenVox server using a SSH client. The **/etc/lumenvox/media_server.conf** file needs to be modified for the following fields:

- The value of **mrcp_server_ip** must be set to the IP address of the machine that LumenVox is installed on. This must be an IP address that the Experience Portal can reach and route traffic to. Please contact LumenVox Support for questions about configuring firewalls if they will be running.
- The value of **compatibility_mode** must be changed from the default **0** to **1**

Note: When configuring an application in Experience portal to use the LumenVox ASR, set the "**Speech Complete Timeout**" parameter under **Speech Parameters** to a non-0 value:



7. Configure LumenVox Text-to-Speech Server

The LumenVox Media Server must be configured as described in "Configure LumenVox Automated Speech Recognizer." There are no special configurations for the Text-to-Speech Server.

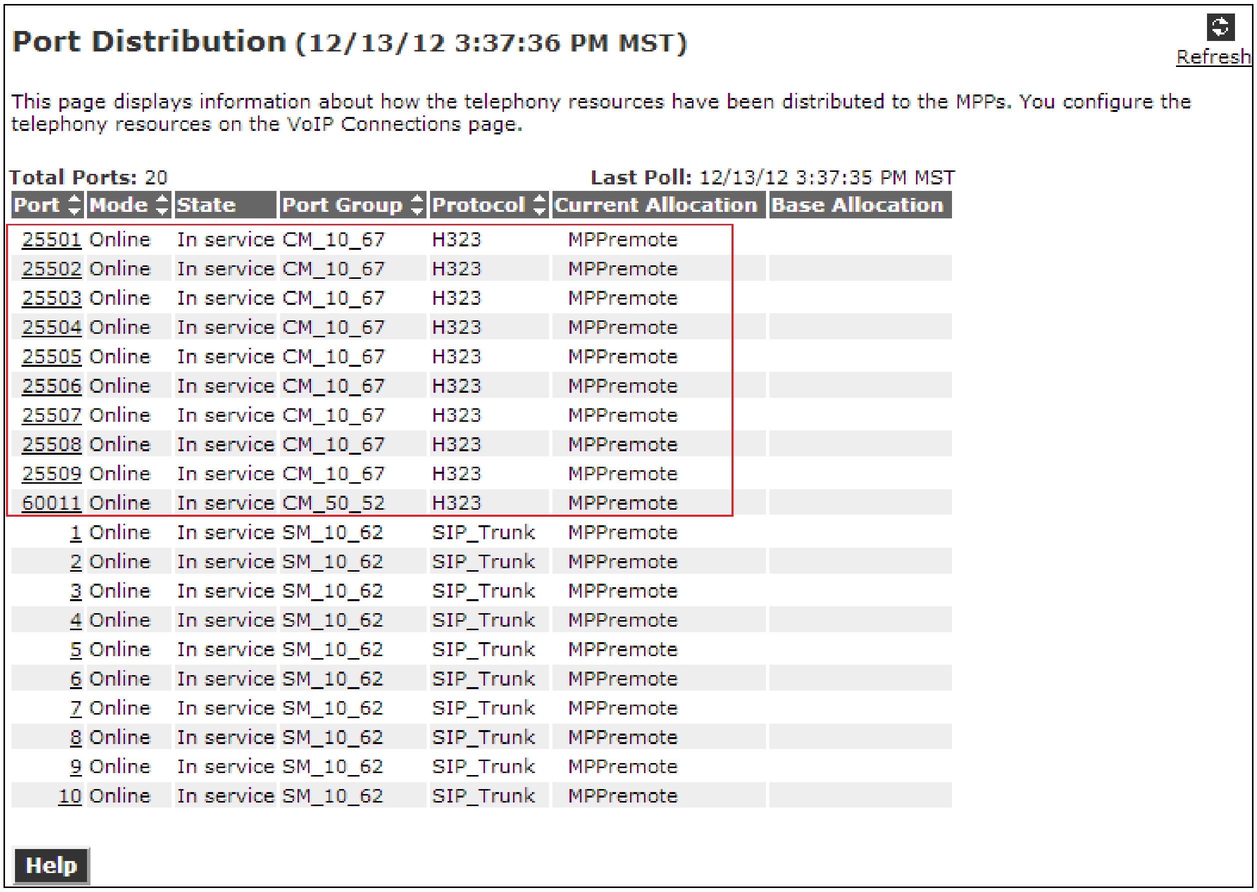
8. Verification Steps

8.1. Avaya Aura® Experience Portal

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

1. From the EPM web interface, navigate to **System Management → MPP Manager**. From the MPP Manager screen, shown below, verify that the Media Processing Platform (MPP) servers are **Online** and **Running**.

2. . On the left pane, navigate to **Real-Time Monitoring → Port Distribution**. From the Port Distribution page, verify that the ports on the MPP server are in service



Port Distribution (12/13/12 3:37:36 PM MST)

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: 20 Last Poll: 12/13/12 3:37:35 PM MST

Port	Mode	State	Port Group	Protocol	Current Allocation	Base Allocation
25501	Online	In service	CM_10_67	H323	MPPremote	
25502	Online	In service	CM_10_67	H323	MPPremote	
25503	Online	In service	CM_10_67	H323	MPPremote	
25504	Online	In service	CM_10_67	H323	MPPremote	
25505	Online	In service	CM_10_67	H323	MPPremote	
25506	Online	In service	CM_10_67	H323	MPPremote	
25507	Online	In service	CM_10_67	H323	MPPremote	
25508	Online	In service	CM_10_67	H323	MPPremote	
25509	Online	In service	CM_10_67	H323	MPPremote	
60011	Online	In service	CM_50_52	H323	MPPremote	
1	Online	In service	SM_10_62	SIP_Trunk	MPPremote	
2	Online	In service	SM_10_62	SIP_Trunk	MPPremote	
3	Online	In service	SM_10_62	SIP_Trunk	MPPremote	
4	Online	In service	SM_10_62	SIP_Trunk	MPPremote	
5	Online	In service	SM_10_62	SIP_Trunk	MPPremote	
6	Online	In service	SM_10_62	SIP_Trunk	MPPremote	
7	Online	In service	SM_10_62	SIP_Trunk	MPPremote	
8	Online	In service	SM_10_62	SIP_Trunk	MPPremote	
9	Online	In service	SM_10_62	SIP_Trunk	MPPremote	
10	Online	In service	SM_10_62	SIP_Trunk	MPPremote	

Help

8.2. LumenVox Automated Speech Recognizer

The Avaya test application (usually installed with the MPP in **/mpp/misc/avptestapp/intro.vxml**) may be used to test the ASR. Perform the following steps:

1. Configure the test application to use the LumenVox ASR.
    a. As noted in **Section 5**, ensure that the "Speech Complete Timeout" value is set to a non-0 value (800 is a default value).
2. Dial into the application.
3. At the main menu, press 1 for speech recognition test.
4. When prompted, speak "Open the window."
5. Confirm that the application understands the utterance.

Note: For optimal results, avoid use of a speakerphone when testing the ASR, as it may introduce recognition issues.

8.3. LumenVox Test-to-Speech Server

The Avaya test application (usually installed with the MPP in **/mpp/misc/avptestapp/intro.vxml**) may be used to test the TTS. Perform the following steps:

1. Configure the test application to use the LumenVox TTS.
2. Dial into the application.
3. At the main menu, press 2 for text-to-speech test.
4. Confirm TTS speaking.

9. Conclusion

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

10. 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*, April 2012

LumenVox help documentation, including detailed installation and configuration instructions, is available online at http://www.lumenvox.com/help/

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.