Avaya Solution & Interoperability Test Lab

Application Notes for LumenVox Speech Engine and LumenVox MRCPv1 Server with Avaya Interactive Response – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the LumenVox Speech Engine and LumenVox MRCPv1 Server with Avaya Interactive Response. 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. The LumenVox MRCPv1 Server handles MRCP communication from Interactive Response and passes requests to the Speech Engine for decode. As the product name implies, this is done via an MRCPv1 connection.

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 and LumenVox MRCPv1 Server with Avaya Interactive Response. 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. The LumenVox MRCPv1 Server[1] handles MRCP communication from Interactive Response and passes requests to the Speech Engine for decode. As the product name implies, this is done via an MRCPv1 connection.

Another LumenVox related solution is described in [5], *Application Notes for LumenVox Speech Engine and LumenVox Media Server with Avaya Voice Portal*.

1.1. Interoperability Compliance Testing

Interoperability compliance testing included feature and serviceability testing. The feature testing focused on placing calls to Avaya Interactive Response (IR) that ran VoiceXML and TAS applications that use the ASR engine in the LumenVox Speech Engine and the LumenVox MRCPv1 Server. Various grammar types were used by the VXML applications, including inline, built-in, menu, and external Speech Recognition Grammar Specification (SRGS) grammars. The testing verified both speech and DTMF tone recognition.

The serviceability testing focused on verifying the ability of the LumenVox Speech Engine and LumenVox MRCPv1 Server to recover from adverse conditions, such as power failures and disconnecting cables to the IP network.

1.2. Support

For technical support on the LumenVox Speech Engine and MRCPv1 Server, contact LumenVox via phone, email, or internet.

- **Phone:** (877) 977-0707
- **Email:** support@lumenvox.com
- **Web:** http://www.lumenvox.com

[1] LumenVox Speech Engine provides two options for an MRCP server: LumenVox MRCPv1 Server and LumenVox Media Server. These Application Notes cover the MRCPv1 Server.

2. Reference Configuration

Figure 1 illustrates the configuration used for testing. In this configuration, Avaya IR interfaces with Avaya Aura™ Communication Manager via H.323, and interfaces to the LumenVox Speech Engine and MRCPv1 Server. VoiceXML (VXML) scripts were run by Avaya IR and used the automatic speech recognition (ASR) engine in the LumenVox Speech Engine. Since the LumenVox Speech Engine does not support text-to-speech (TTS), an optional third-party TTS engine may be used if required by the application. A TTS engine was used during testing (not shown).

Note: Configuration of the H.323 interface between Avaya IR and Avaya Aura™ Communication Manager is outside the scope of these Application Notes. The reader should refer to the documentation in the References section for additional information. These Application Notes will focus on the speech server configuration on Avaya IR and configuration of the LumenVox Speech Engine and MRCPv1 Server.

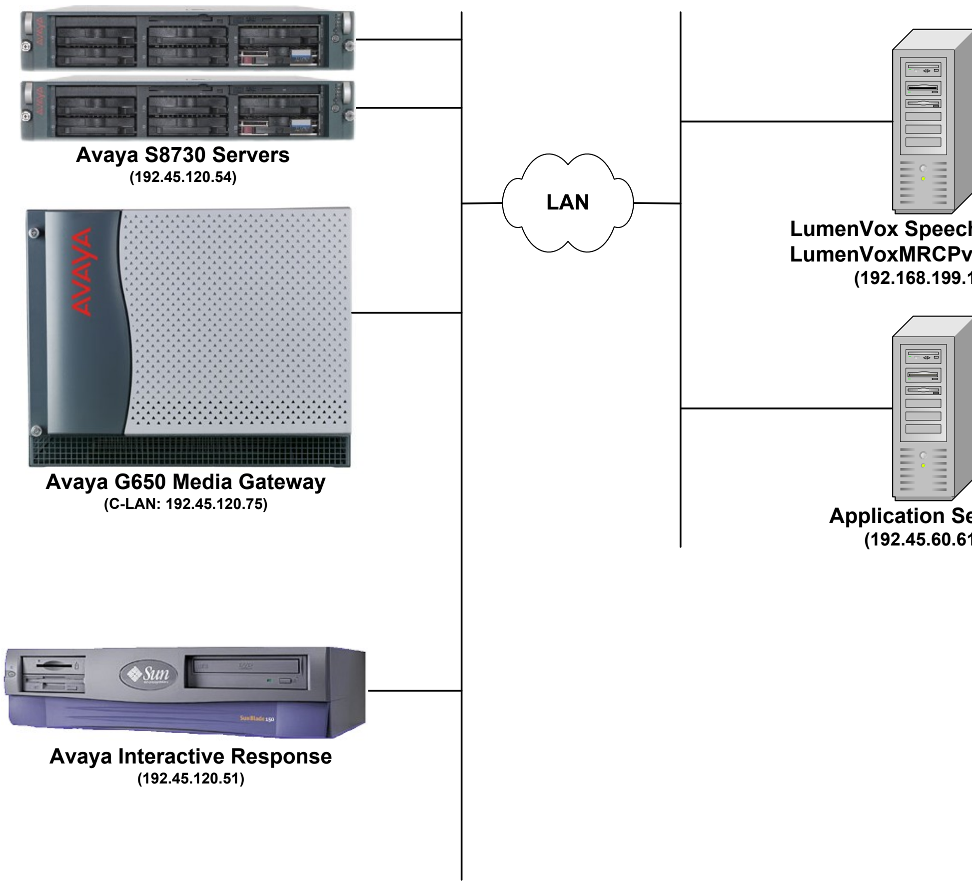

Figure 1: Configuration with Avaya IR, LumenVox Speech Engine, and LumenVox MRCPv1 Server

2.1. Equipment and Software Validated

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

Equipment	Software
Avaya Interactive Response	4.0 with Service Pack 2
Avaya S8730 Servers with an Avaya G650 Media Gateway	Avaya Aura™ Communication Manager 5.2 (R015x.02.0.947.3) with Service Pack 1 (Patch 17294)
LumenVox Speech Engine LumenVox MRCPv1 Server LumenVox License Server	9.0.601 9.0.601 9.0.601
Application Server – HTTP Server running on Windows Server 2003	Internet Information Services (IIS) 5.1

3. Configure Avaya Interactive Response

This section covers the administration of Avaya IR. The following configuration steps are covered:

- Configure the Speech Server.
- Restart the Voice System.
- Assign services (VXML or TAS application) to channels.

Refer to [4] for additional information on configuring Avaya IR.

The Avaya IR configuration is performed via a web browser. Enter the IP address of Avaya IR in the URL field of the web browser. The initial Avaya IR webpage is displayed as shown in **Figure 2**. Select the **Web Administration** link to display the log in screen (not shown), and log into Avaya IR with the appropriate credentials.

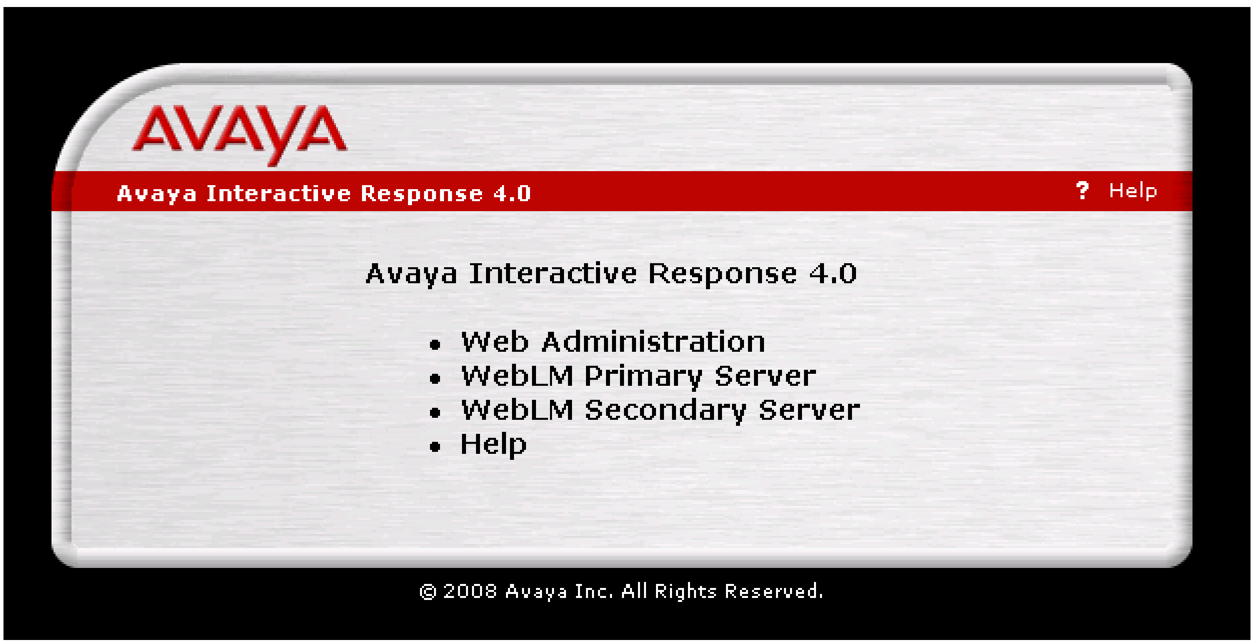

Figure 2: Initial Avaya IR Screen

After successfully logging into Avaya IR, the main Avaya IR configuration webpage is displayed as shown in **Figure 3**.

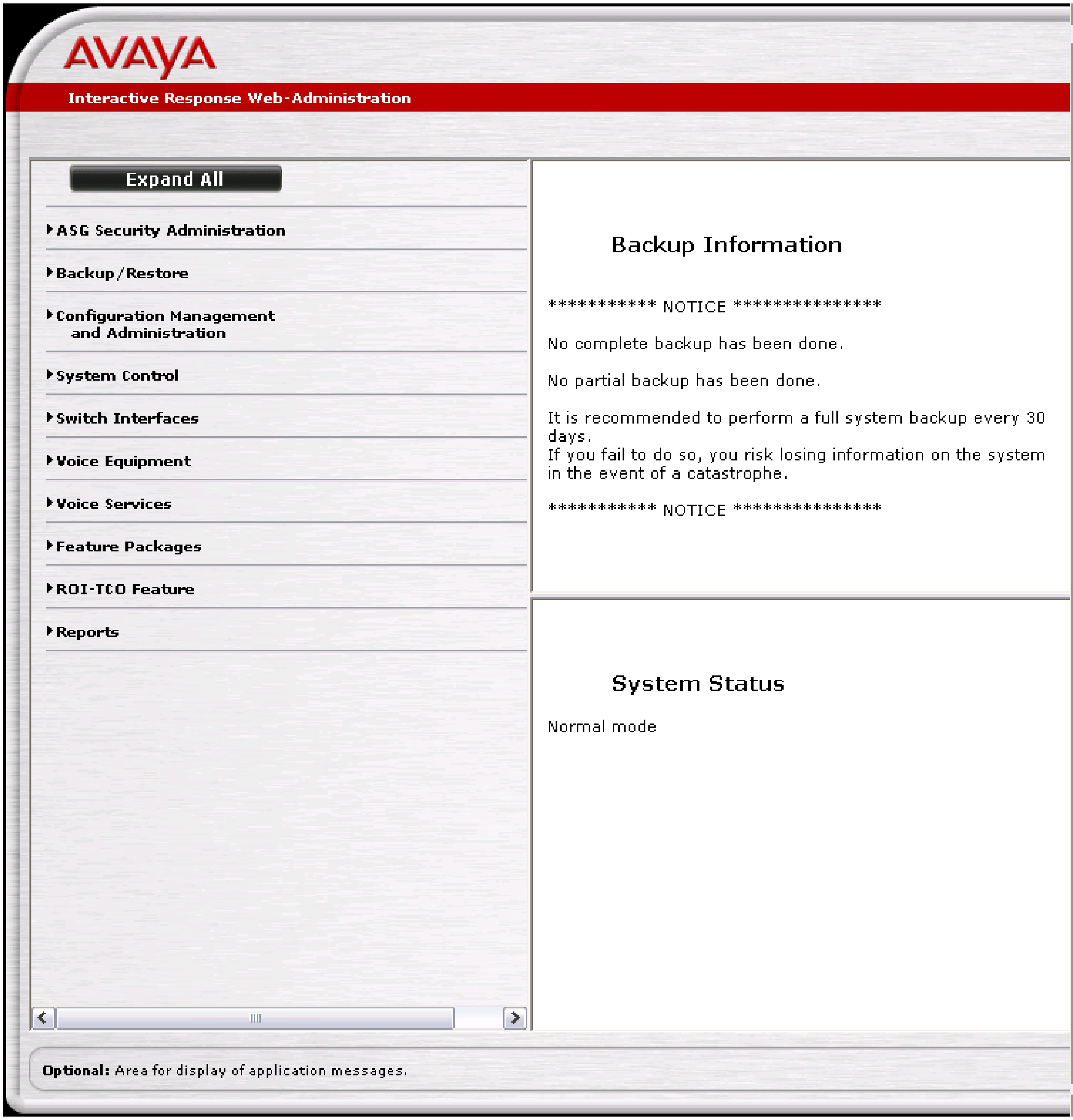

Figure 3: Main Avaya IR Webpage

Configure the speech server in Avaya IR by selecting the **Administration** option under **Speech and DPR Administration** in the left pane. The page in **Figure 4** is displayed. On this page, click the **Assign New** button. The **Assign Speech Recognition or DPR Type** page is displayed as shown in **Figure 5**.

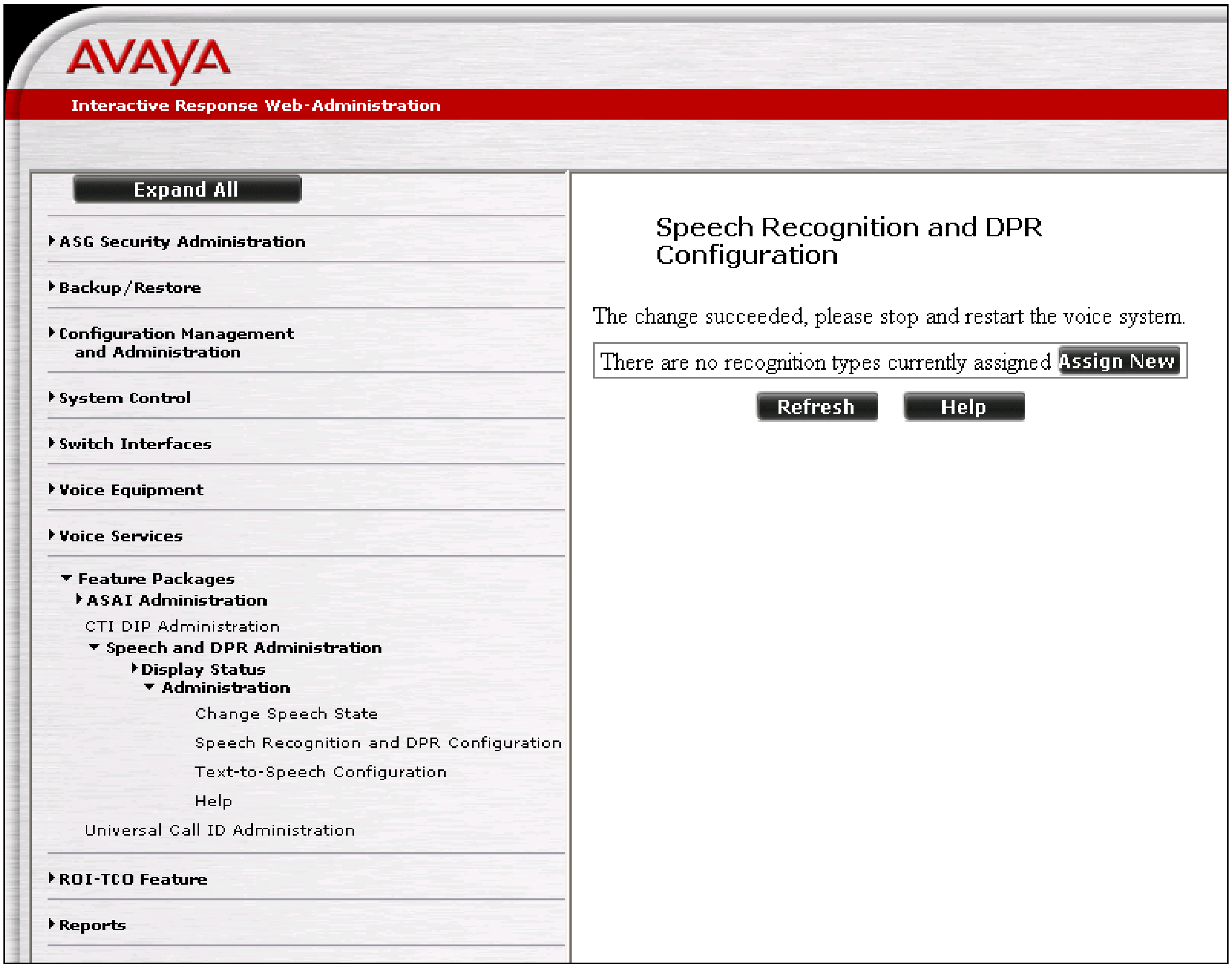

Figure 4: Speech Recognition and DPR Configuration

In the **Assign Speech Recognition or DPR Type** page, set the **Engine** field to *mrcp-ibm* as shown below. Click **Submit**.

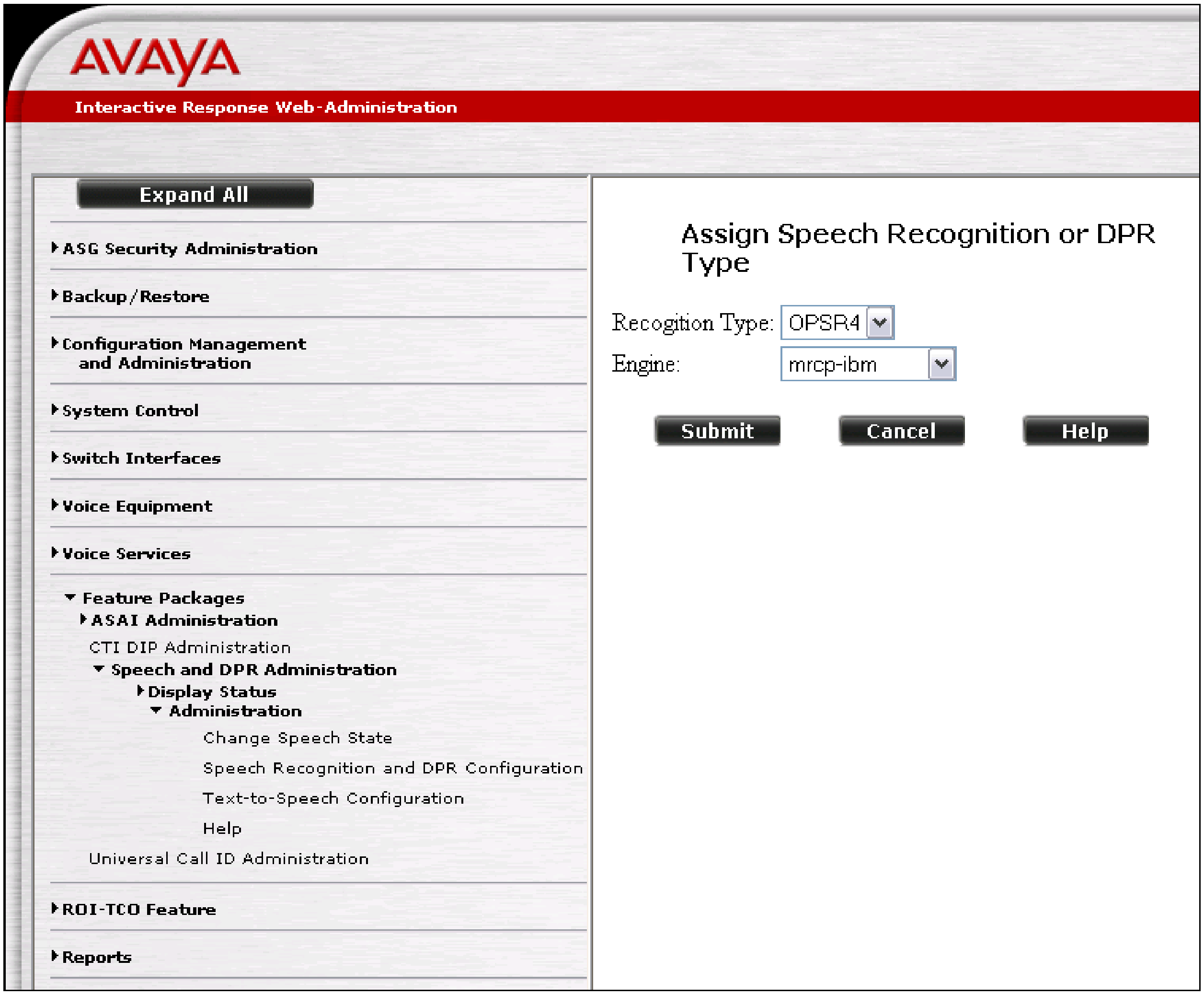

Figure 5: Assign Speech Recognition or DPR Type

On the page shown below, click the **Assign New Server** button to display the **Speech Recognition or DPR Server** page shown in **Figure 7**.

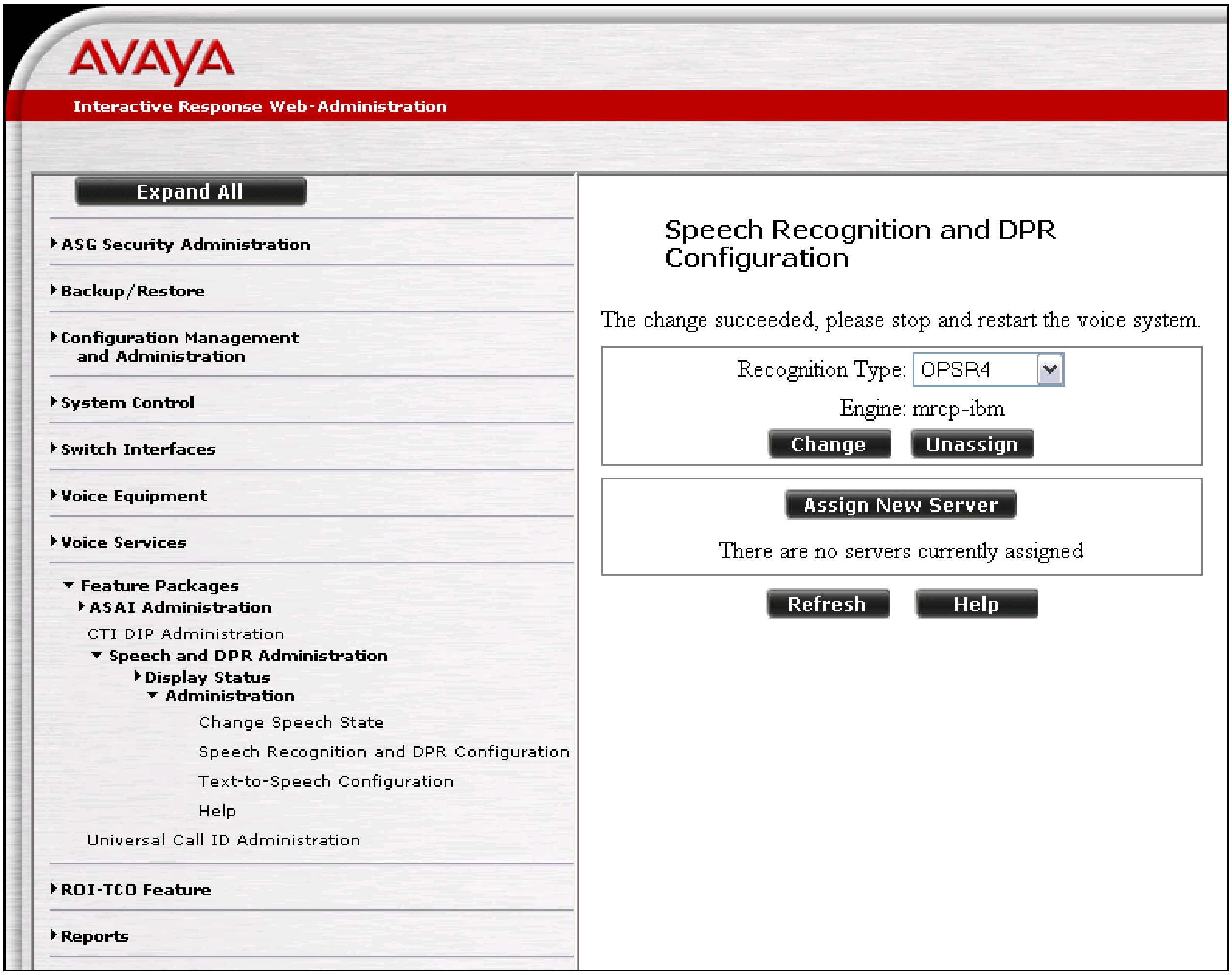

Figure 6: Speech Recognition and DPR Configuration – No Servers Assigned

In the **Assign Speech Recognition or DPR Server** page shown in **Figure 7**, set the **Server Name** and **IP Address** fields to the IP address of the LumenVox Speech Engine and LumenVox MRCPv1 Server, and set the **Ports** field to the number of ports available on the LumenVox Speech Engine according to its installed license. By default, *media/recognizer* is automatically appended to the value in the **Server Name** field. Click **Submit**.

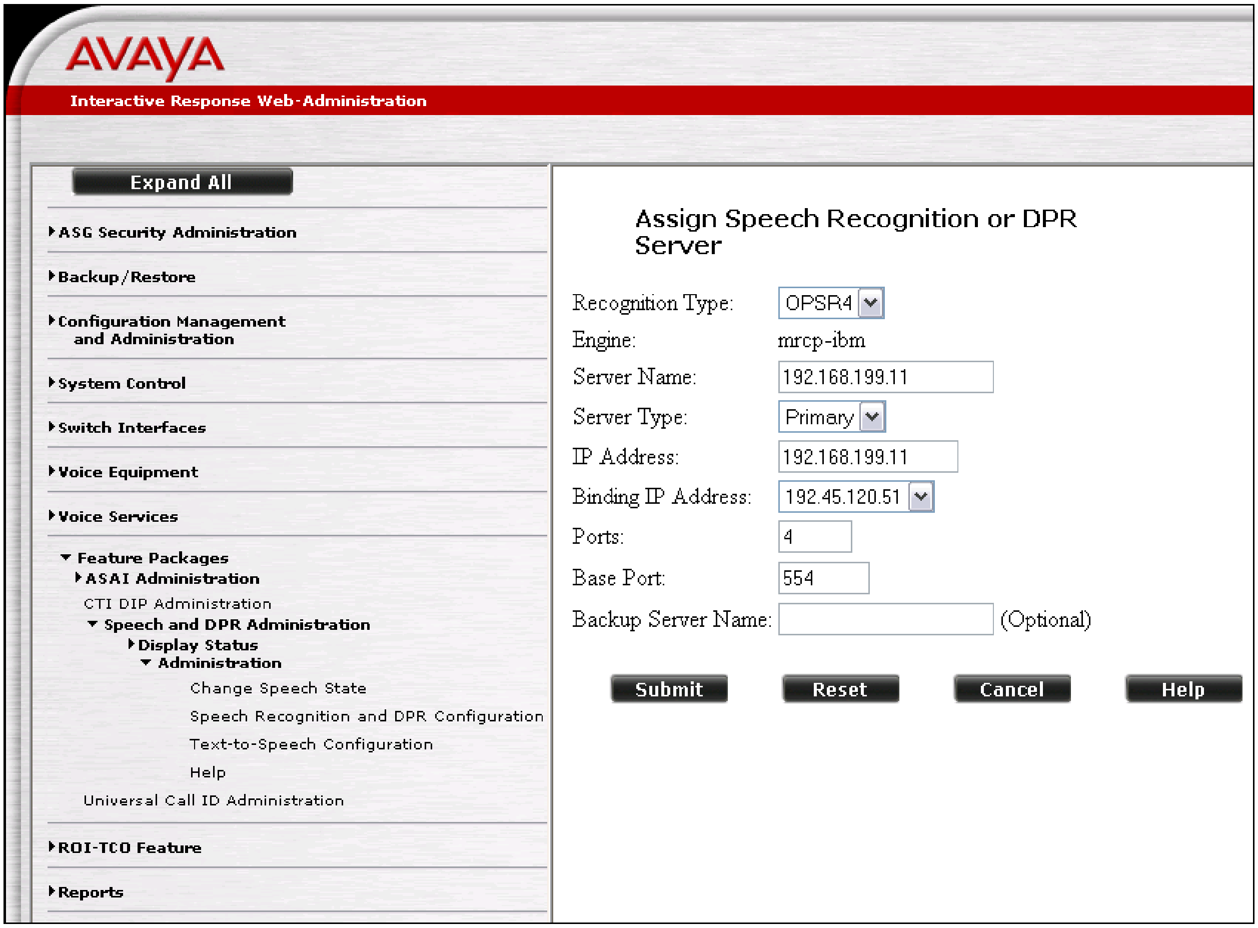

Figure 7: Assign Speech Recognition or DPR Server Parameters

Once the configuration of the speech server is complete, a configuration summary page is displayed as shown below.

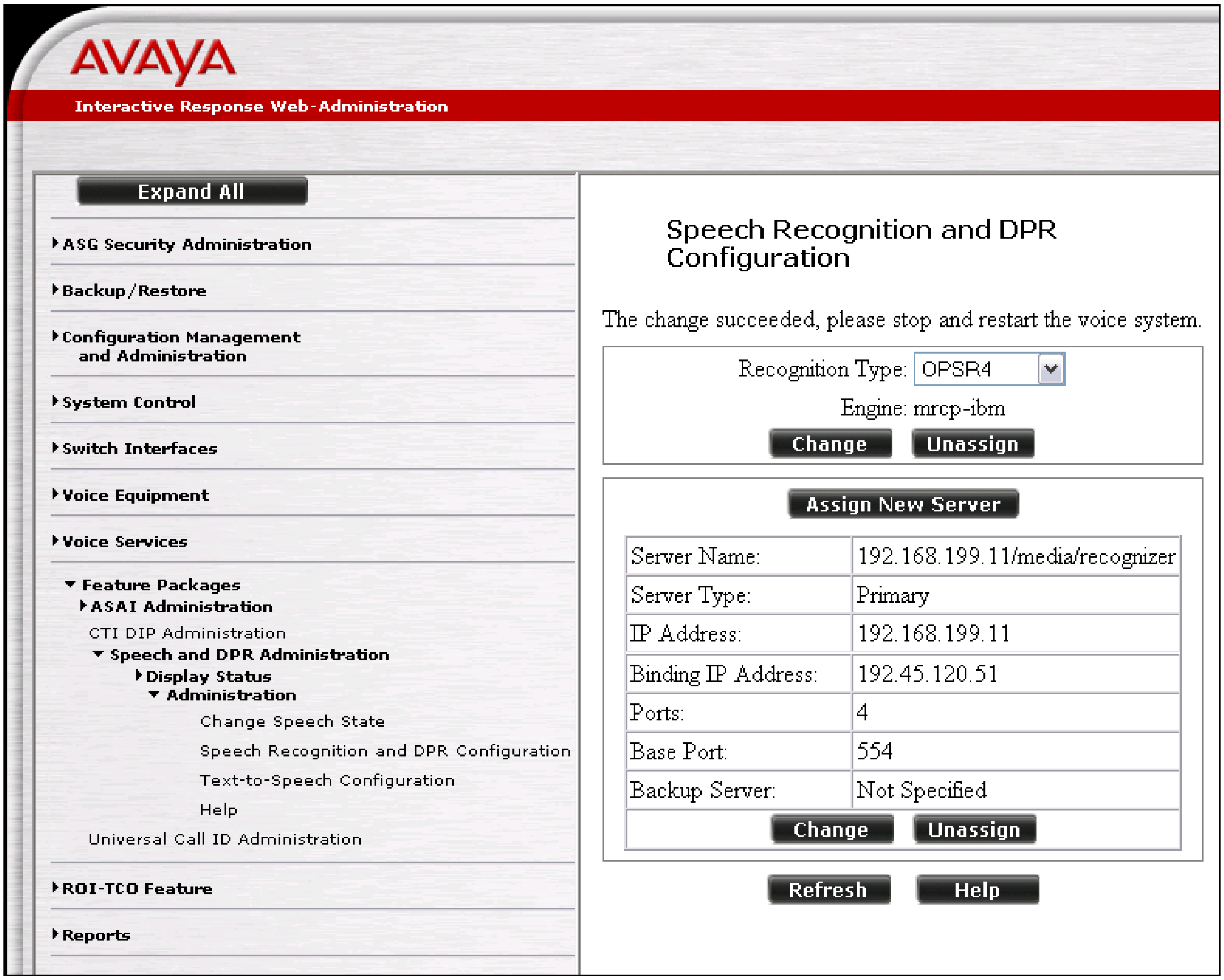

Figure 8: Speech Recognition and DPR Configuration Summary

Note: Although the LumenVox Speech Engine does not support a TTS engine, a third-party TTS server that is supported by Avaya IR may be added in the Text-to-Speech configuration page accessible under the Administration option in the left pane. For further instructions on how to add a TTS server to Avaya IR, refer to [4].

After the speech server is successfully configured, restart the Avaya IR by selecting **Stop Voice System** under **System Control** in the left pane. After the voice system has been successfully stopped, restart it by selecting **Start Voice System** in the left pane. Before proceeding, wait for the system to display a message indicating that the startup of the voice system is complete.

Now, assign a VXML application to channel 48 (i.e., the first H.323 channel in the system). This specifies which application a particular Avaya IR channel should run when it receives a call. Select **Channel Services** from the left pane under **Voice Services** to display the **Channel Services** page in **Figure 9**. Enable the checkbox for channel 48 and then click the **Assign Selected** button.

Note: It is assumed that the VoiceXML application has already been developed and is hosted on the application server.

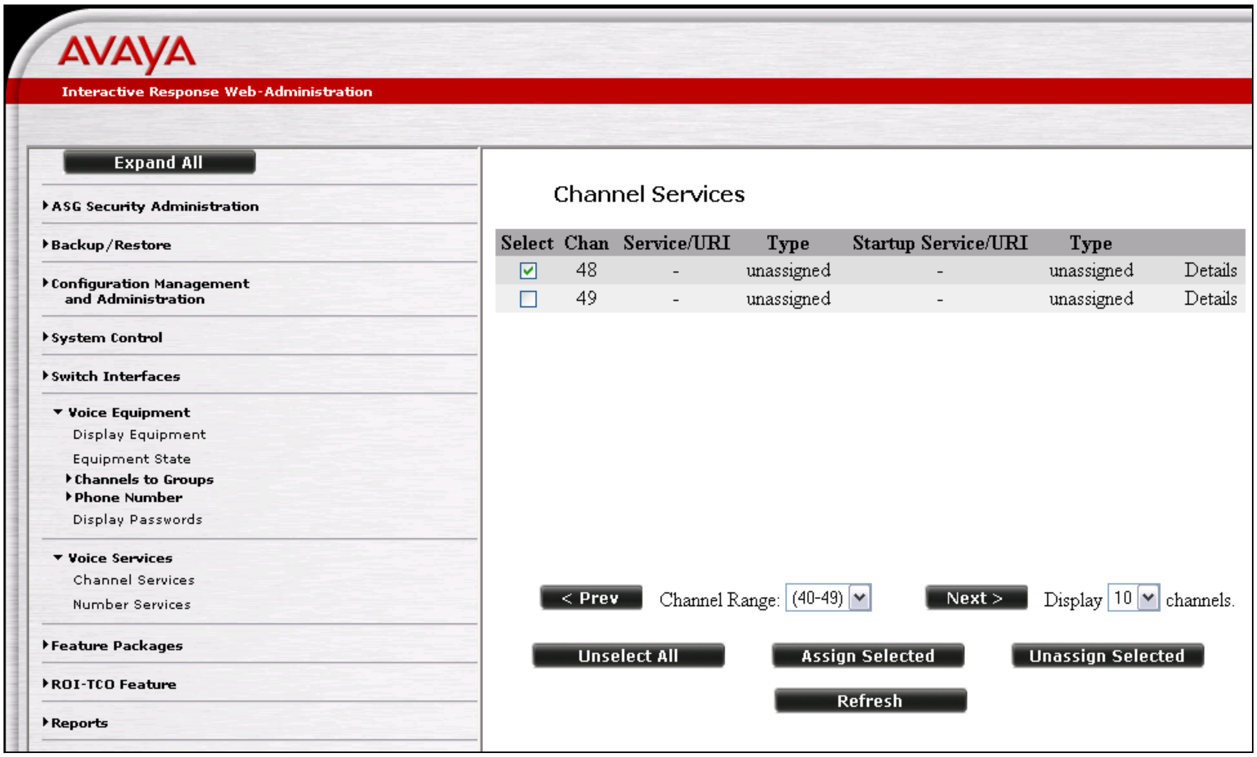

Figure 9: Channel Services

Configure the **Assign Services to Channels** page as shown in **Figure 10**. This configuration assigns a VoiceXML application named *VoiceExternal.xml* to channel 48. Set the **Assign** field to *VXML URI,* set the **Primary URI** field to *http://192.45.60.61/avptestapp/Lumenvox/scripts/VoiceExternal.vxml*, and set the **To Chan(s)** field to '48'. The VoiceXML script is hosted in an application server. Repeat this procedure for all channels that should run this application. Click **Submit**.

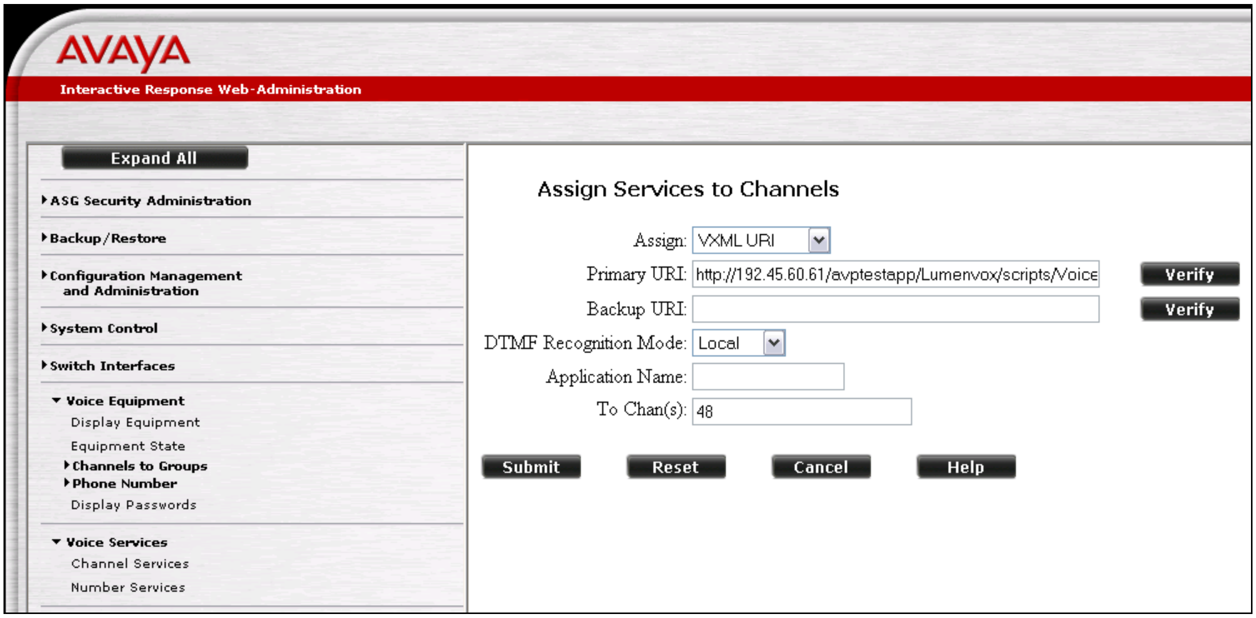

Figure 10: Assign Services to Channels – VXML Application

The following step will show how to assign a TAS script to an Avaya IR channel. Navigate to the **Channel Services** page shown in **Figure 9**, enable the checkbox for channel 49, and click the **Assign Selected** button. In the **Assign Services to Channels** page shown in **Figure 11**, set the **Assign** field to *TAS Service*, set the **Service** and **Startup Services** fields to the desired TAS application. In this example, the *avftst* (stands for Avaya transfer test) application was used. Set the **To Chan(s)** field to '49'. Click **Submit**. The TAS application was installed on Avaya IR, not the application server.

Note: It is assumed that the TAS application has already been developed and installed on Avaya IR. Refer to [4] for instructions on how to install a TAS application on Avaya IR using Avaya IVR Designer.

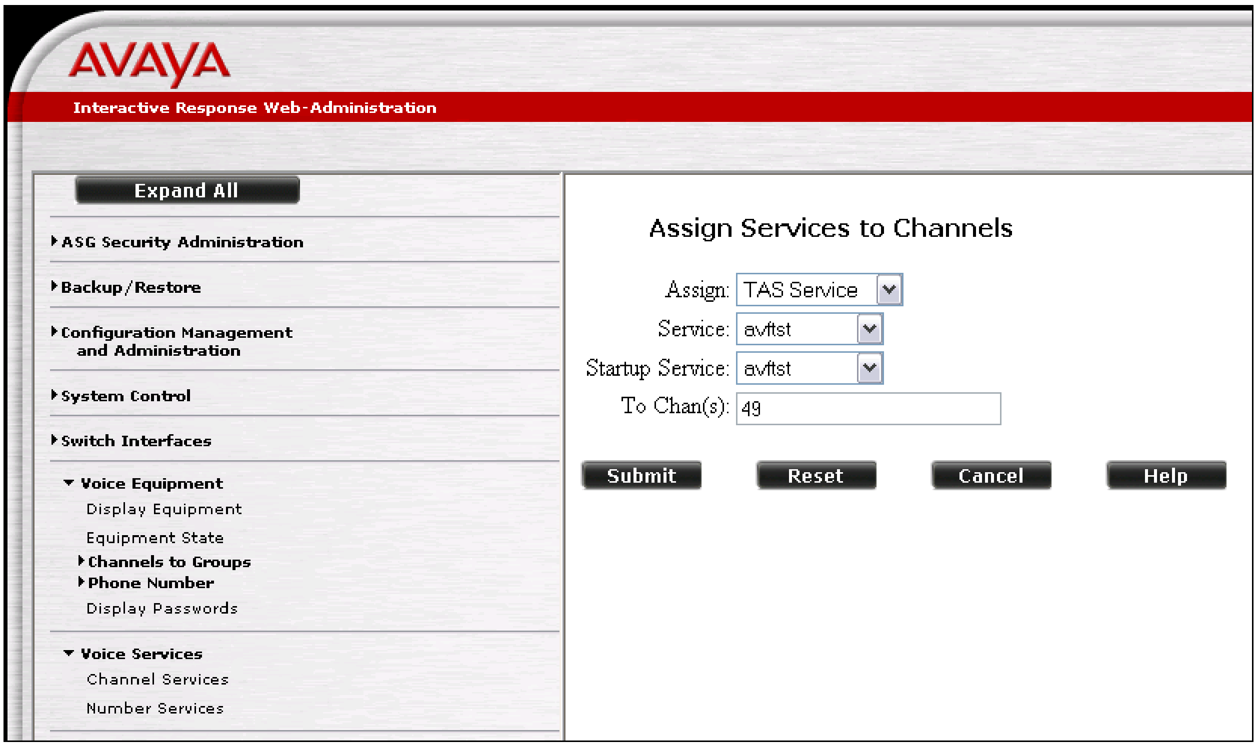

Figure 11: Assign Services to Channels – TAS Application

To view the status of the channels and the channel configuration details, select **Display Equipment** from the left pane. The page in **Figure 12** is displayed. Verify the status of the configured channels. In this configuration, channels 0-3 are in service and channels 48 and 49 have been assigned a VXML and TAS application, respectively. Channel 48 was assigned phone number 23301 and channel 49 was assigned phone number 23302.

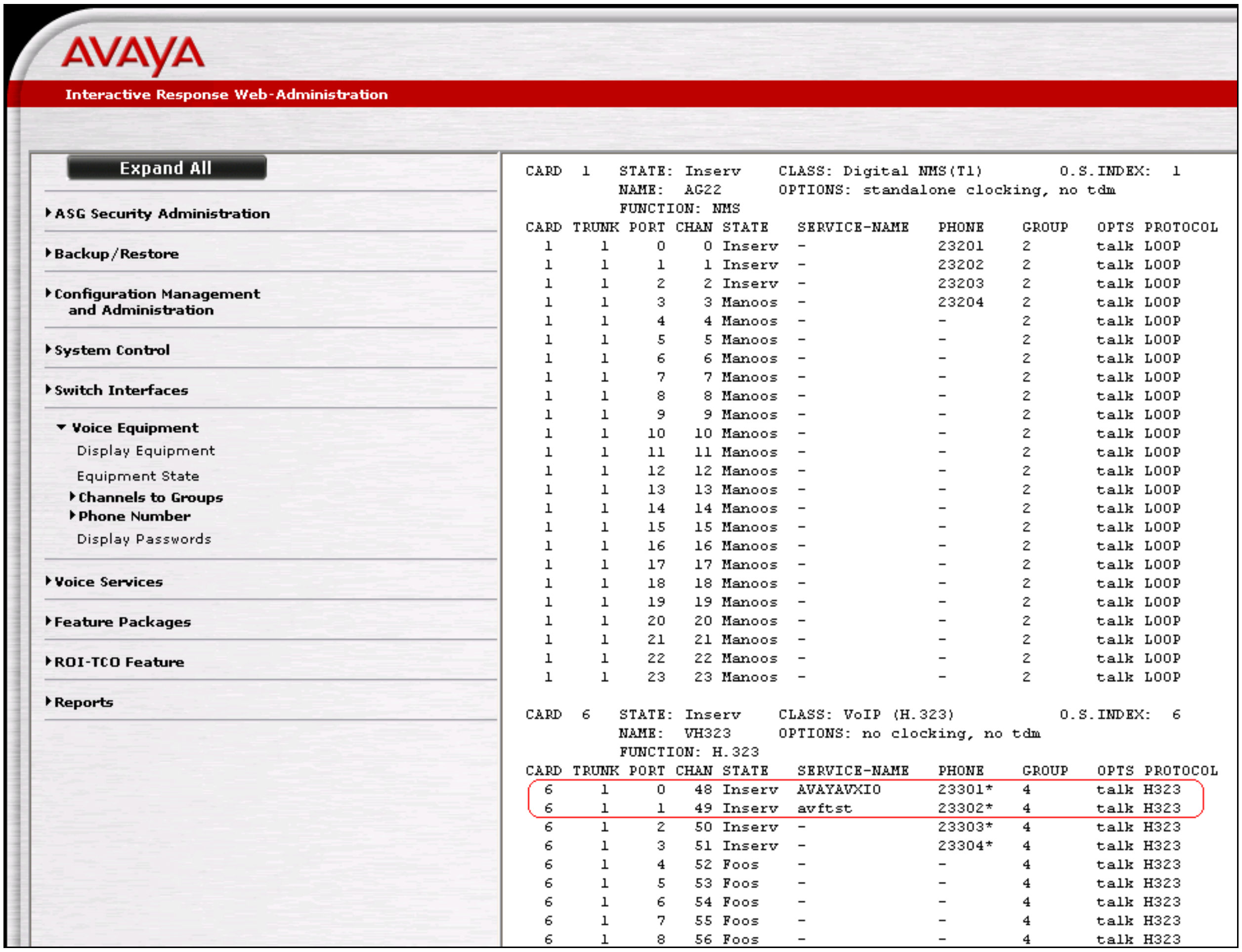

Figure 12: Display Equipment

4. Configure LumenVox Speech Engine and LumenVox MRCPv1 Server

This section covers the configuration required for the LumenVox Speech Engine and LumenVox MRCPv1 Server. This is accomplished by editing a file called **mrcp.config** that is put into **C:\Program Files\Lumenvox\MRCP Server\config** in Windows by default. The following parameters should be configured as follows:

- *rtsp_port* should match the *Base Port* field configured for the ASR server in Figure 7
- *mrcp_server_ip* should be set to the IP address of the LumenVox MRCPv1 Server
- *resource_string* should be set to *media/recognizer*, which is used by Avaya IR
- *compatibility_mode* should be set to '0' so that the LumenVox MRCPv1 Server behaves like an IBM WVS speech server

Refer to [6] 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            = MRCPv1_Log.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.  Only leave this default loop back # IP address if you install the MRCP
server on the same machine as # the platform that is hosting your application.
#-----------------------------------------------------------------
mrcp_server_ip             = 192.168.199.11

#-----------------------------------------------------------------
# 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 = 127.0.0.1

#-----------------------------------------------------------------
# set this value to the license type used by the speech # recognizer. Its poissible
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

#------------------------------------------------------------------
```

```
# rtsp_timeout_param_transmission = yes or no (default) # In the response to the SETUP
command you can select weither the # timeout parameter is transmitted back with the
session id.
#-------------------------------------------------------------------
rtsp_timeout_param_transmission = no

#--------------------------------------------------------------------
# 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=false
recognition_timeout=60000
dtmf_inter_digit_timeout=5000
snr_sensitivity_lvl=50

save_waveform=false
waveform_url_location=file:///c:/

barge_in_timeout=150000

compatibility_mode=0
```

Figure 13: mrcp.config File

5. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify Avaya IR VXML and TAS applications that use the ASR engine in the LumenVox Speech Engine. This section covers the general test approach and the test results.

Interoperability compliance testing included feature and serviceability testing. The feature testing focused on placing calls to Avaya IR that ran VoiceXML applications that use the ASR engine in the LumenVox Speech Engine and the LumenVox MRCPv1 Server. Various grammar types were used by the VXML applications, including inline, built-in, menu, and external Speech Recognition Grammar Specification (SRGS) grammars. The testing verified both speech and DTMF tone recognition.

The serviceability testing focused on verifying the ability of the LumenVox Speech Engine and LumenVox MRCPv1 Server to recover from adverse conditions, such as power failures and disconnecting cables to the IP network.

All test cases passed. Avaya IR 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 Avaya IR can run applications that use the LumenVox Speech Engine.

1. From the Avaya IR web interface, verify that the Avaya IR channels are in-service as shown in **Figure 12**.

2. From the Avaya Communication Manager SAT, verify that the H.323 channels are in-service using the **status station** **`<extension>`** as shown in **Figure 14.**

```
status station 23301                                            Page   1 of   7
                              GENERAL STATUS
     Administered Type: 7434ND             Service State: in-service/on-hook
        Connected Type: N/A           TCP Signal Status: connected
             Extension: 23301
                  Port: S00305        Parameter Download: not-applicable
           Call Parked? no                 SAC Activated? no
      Ring Cut Off Act? no
Active Coverage Option: 1            one-X Server Status: N/A

          EC500 Status: N/A        Off-PBX Service State: N/A
   Message Waiting:
   Connected Ports:


  Limit Incoming Calls? no

 User Cntrl Restr: none                        HOSPITALITY STATUS
Group Cntrl Restr: none                     Awaken at:
                                             User DND: not activated
                                            Group DND: not activated
                                          Room Status: non-guest room
```

Figure 14: Status Station

3. From the Avaya IR web interface, navigate to **Feature Packages→Speech and DRP Administration→Display Status→Speech Resource Status** in the left pane to check the status of the LumenVox Speech Engine. In the resulting page, select the resource associated with the LumenVox Speech Engine (e.g., OPSR4). The following page will be displayed. Check that the ports are INSERV.

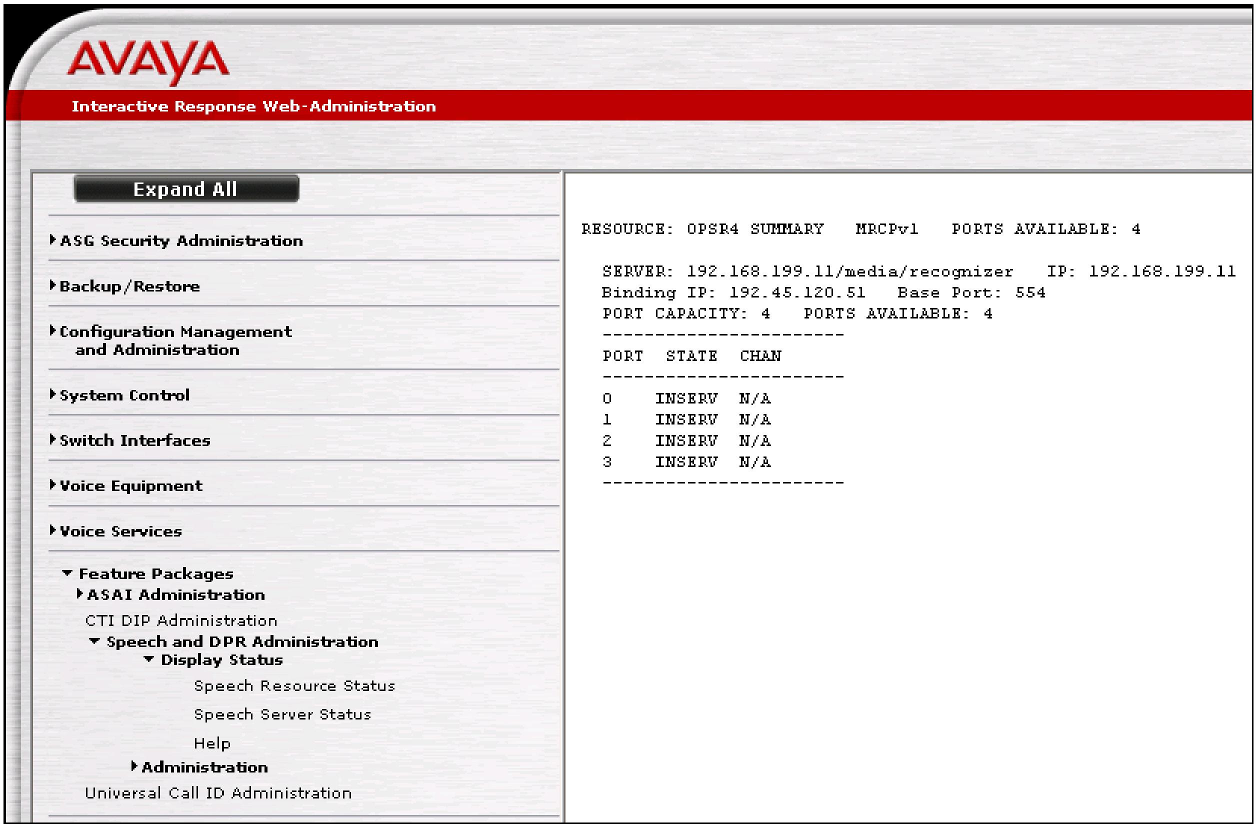

Figure 15: OPSR Summary

4. Place a call to an Avaya IR extension to run a VXML application that uses the LumenVox Speech Engine. Verify that the application answers the call and that the application is able to recognize the speech input provided by the caller.

7. Conclusion

These Application Notes describe the configuration steps required to integrate the LumenVox Speech Engine and LumenVox MRCPv1 Server with Avaya Interactive Response. VXML applications that use various grammar types were used and the speech input was recognized accurately.

8. Additional References

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

[1] *Administering Avaya Aura™ Communication Manager*, Document 03-300509, Issue 5, May 2009, available at http://support.avaya.com.

[2] *Avaya Aura™ Communication Manager Feature Description and Implementation*, Document 555-245-205, Issue 7, May 2009, available at http://support.avaya.com.

[3] *Feature Description and Implementation for Avaya Communication Manager*, Document 555-245-205, Issue 5, February 2007, available at http://support.avaya.com.

[4] *Avaya Interactive Response Release 4.0 Documentation Library*, December 2008, available at http://support.avaya.com.

[5] *Application Notes for LumenVox Speech Engine and LumenVox Media Server with Avaya Voice Portal*, Issue 1.0, available at http://support.avaya.com.

[6] *LumenVox Online Documentation* available at http://www.lumenvox.com/help/speechEngine/index.htm.

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.