Avaya Solution & Interoperability Test Lab

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

Abstract

These Application Notes describe the configuration steps required to integrate the LumenVox Speech Engine 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. It also provides speech application developers with a development and runtime platform, allowing for dynamic language, grammar, audio format, and logging capabilities to customize every step of an application.

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 with Avaya Interactive Response (IR). 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. It also provides speech application developers with a development and runtime platform, allowing for dynamic language, grammar, audio format, and logging capabilities to customize every step of an application.

Figure 1 illustrates the configuration used for testing. In this configuration, Avaya Interactive Response (IR) interfaces with Avaya Communication Manager via T1 and the LumenVox Speech Engine via Media Resource Control Protocol (MRCP). 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 text-to-speech (TTS) engine may be used if required by the application. A TTS engine was used during testing.

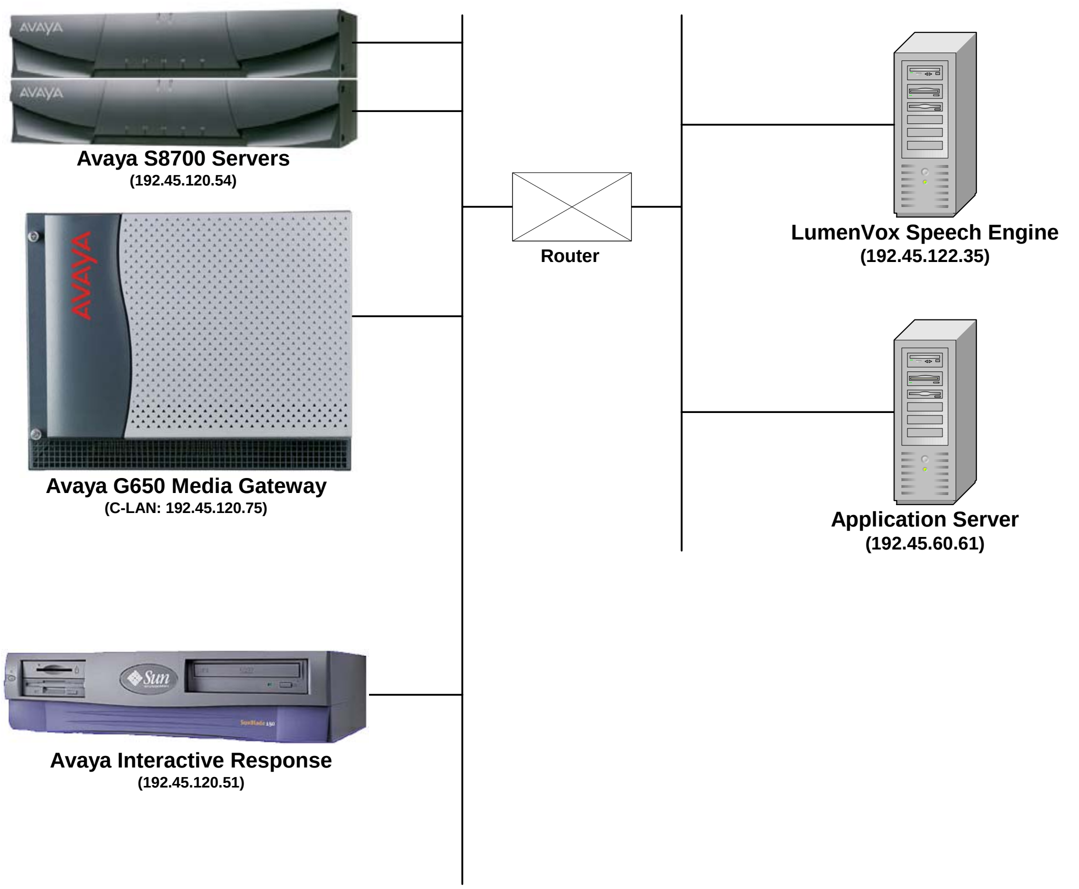


Figure 1: Configuration with Avaya IR and the LumenVox Speech Engine

1.1. Equipment and Software Validated

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

Equipment	Software
Avaya Interactive Response	3.0
Avaya S8700 Servers with an Avaya G650 Media Gateway	Avaya Communication Manager 4.0 (R014x.00.1.731.2)
LumenVox Speech Engine LumenVox MRCP Server LumenVox License Server	8.0.301 8.0.301 8.0.301
Application Server – HTTP Server running on Windows Server 2003	Internet Information Services (IIS) 5.1

2. Configure Avaya Communication Manager

This section describes the configuration of the T1/Robbed-Bit Signaling link between Avaya Communication Manager and Avaya IR and the stations that correspond to each Avaya IR port. Refer to [1] [2] [3] for additional information on configuring Avaya Communication Manager.

From the System Access Terminal (SAT), configure the DS1 board that provides T1 connectivity to Avaya IR. The **Signaling Mode** of the DS1 board is set to *robbed-bit* signaling with a **Line Coding** and **Framing Mode** of *b8zs* and *esf*, respectively.

```
add ds1 a11                                                 Page   1 of   2
                                DS1 CIRCUIT PACK

            Location: 01A11                         Name: Avaya IR
            Bit Rate: 1.544                  Line Coding: b8zs
   Line Compensation: 1                     Framing Mode: esf
      Signaling Mode: robbed-bit


Interface Companding: mulaw
           Idle Code: 11111111

      Slip Detection? n                Near-end CSU Type: other
```

Figure 2: DS1 Circuit Pack

Configure a station for each DS1 port utilized. Set the **Type** field to *DS1FD*, set the **Port** field to the desired DS1 port, and set the **Name** field to a descriptive name. In this configuration, two stations for Avaya IR were configured with extensions 23201 and 23202. Although not covered in this configuration, these stations can be members of a hunt group so that callers can dial a single number that maps to a hunt group extension. The hunt group can then route the call to an available member (i.e., DS1FD station). Agent Login IDs can also be used to have the stations or Avaya IR ports automatically logged into the hunt group (or split). Refer to [3] for addition information on adding hunt groups and agent login IDs.

```
add station 23201                                           Page   1 of   4
                                     STATION

Extension: 23201                         Lock Messages? n             BCC: 0
     Type: DS1FD                         Security Code:                TN: 1
     Port: 01A1101                     Coverage Path 1:               COR: 1
     Name: IR Port 1                   Coverage Path 2:               COS: 1
                                       Hunt-to Station:             Tests? y
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 4
    Off Premises Station? y
       R Balance Network? n

          Survivable COR: internal
   Survivable Trunk Dest? y
```

Figure 3: Station for Avaya IR Port

3. Configure Avaya Interactive Response

This section covers the configuration of Avaya IR. Avaya Communication Manager routes incoming calls to Avaya IR over a T1 interface. Each channel of the T1 interface is assigned a phone number that should match the corresponding station extension configured on Avaya Communication Manager and an Avaya IR VoiceXML or TAS script. Refer to [4] for additional information on Avaya IR.

The configuration steps required on Avaya IR are summarized below.

- Access the Avaya IR via an Internet web browser and log in.
- Stop the Voice System (i.e., Avaya IR) prior to configuring the T1 interface.
- Configure the T1 interface to the Avaya G650 Media Gateway.
- Configure the Speech Server.
- Start the Voice System.
- Assign channels to equipment groups.
- Assign phone numbers to channels.
- Assign services (TAS application) to channels.

The Avaya IR configuration was 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 4**. Select the **Web Administration** link to display the log in screen (not shown), and log into Avaya IR with the appropriate credentials.

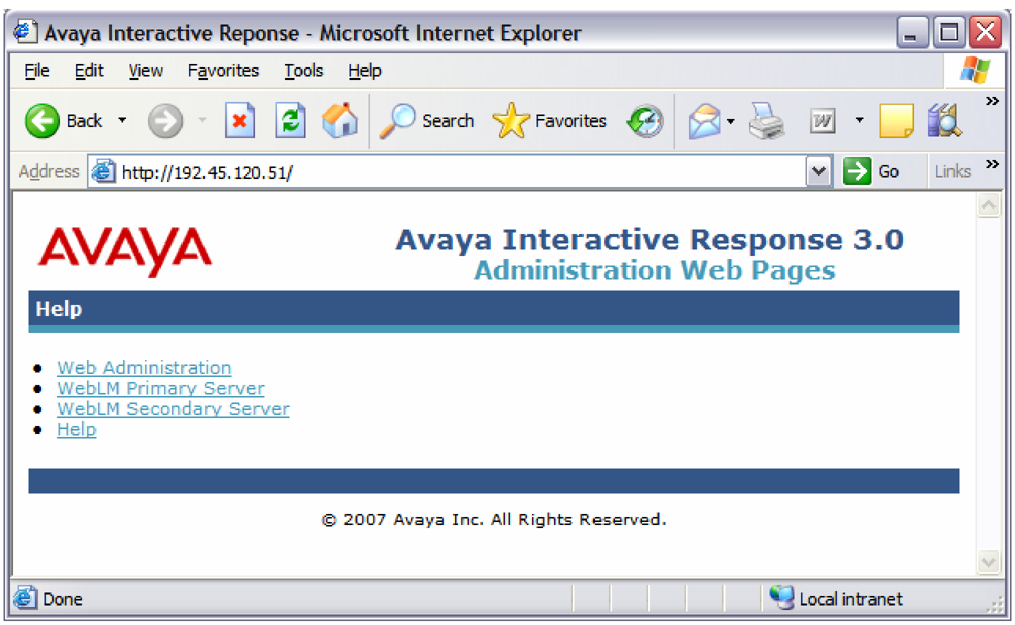


Figure 4: Initial Avaya IR Screen

After successfully logging into Avaya IR, the main Avaya IR configuration webpage is displayed as shown in **Figure 5**.

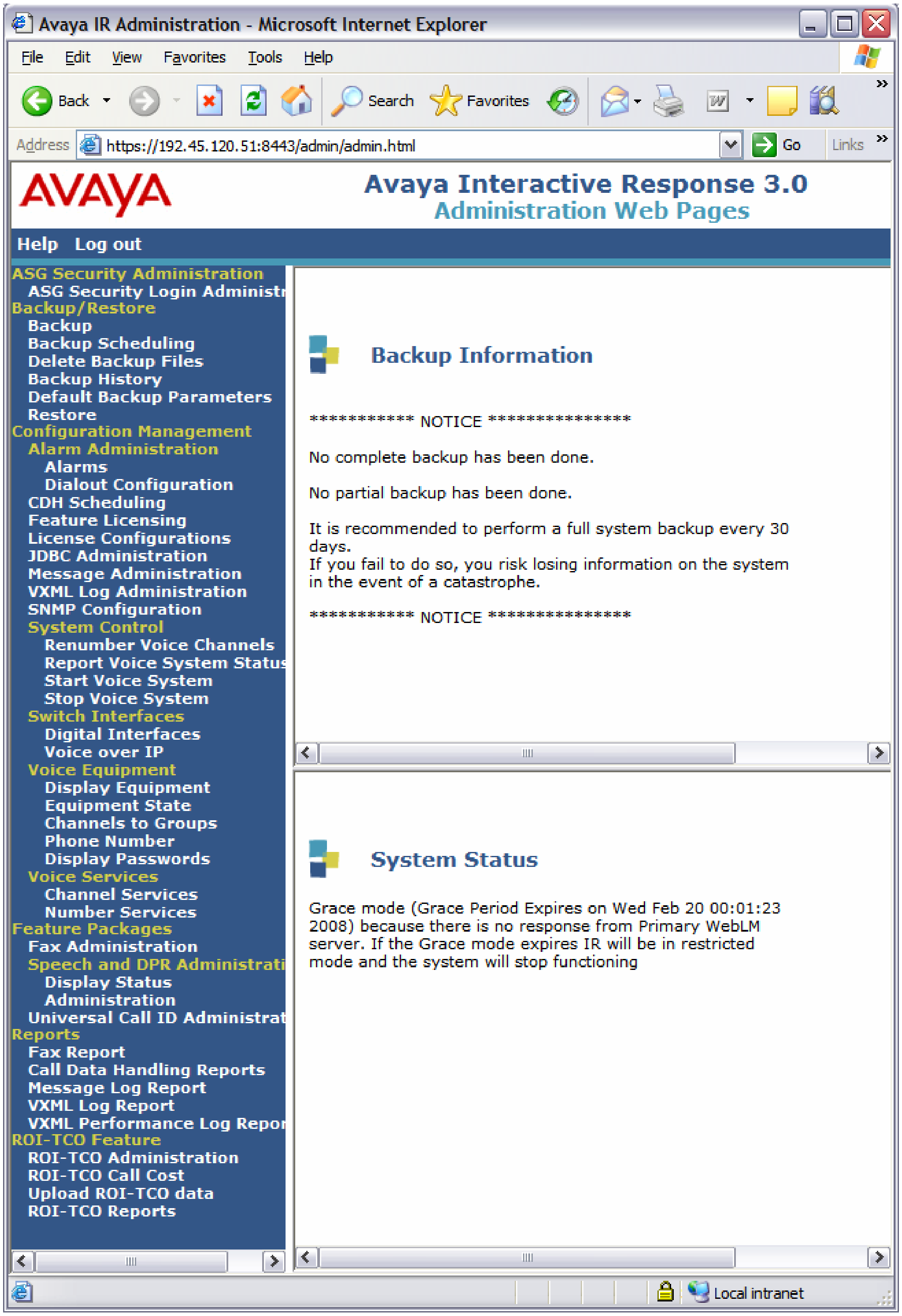


Figure 5: Main Avaya IR Webpage

Prior to configuring the T1 interface to the Avaya G650 Media Gateway, stop the Avaya IR by selecting the **Stop Voice System** link from the left pane in **Figure 5**. The **Stop Voice System** page is displayed. Click the **Submit** button and wait until the system displays a message at the bottom of the page indicating that the voice system has completely stopped.

To configure the T1 interface to the Avaya G650 Media Gateway follow these steps:

1. Under **Switch Interfaces** in the left pane, select the **Digital Interfaces** option to display the **Digital Interfaces Protocols** page.
2. Select the **Assign Card** link to display the **Assign Card** page shown in **Figure 6**. On this page, set the **Card** field to the appropriate number, set the **Card Type** field to *T1*, and set the **Trunk 1** field to *Loop Start T1*. When complete, click the **Submit** button to display the **Assign Card 1: Type T1** page shown in **Figure 7**.

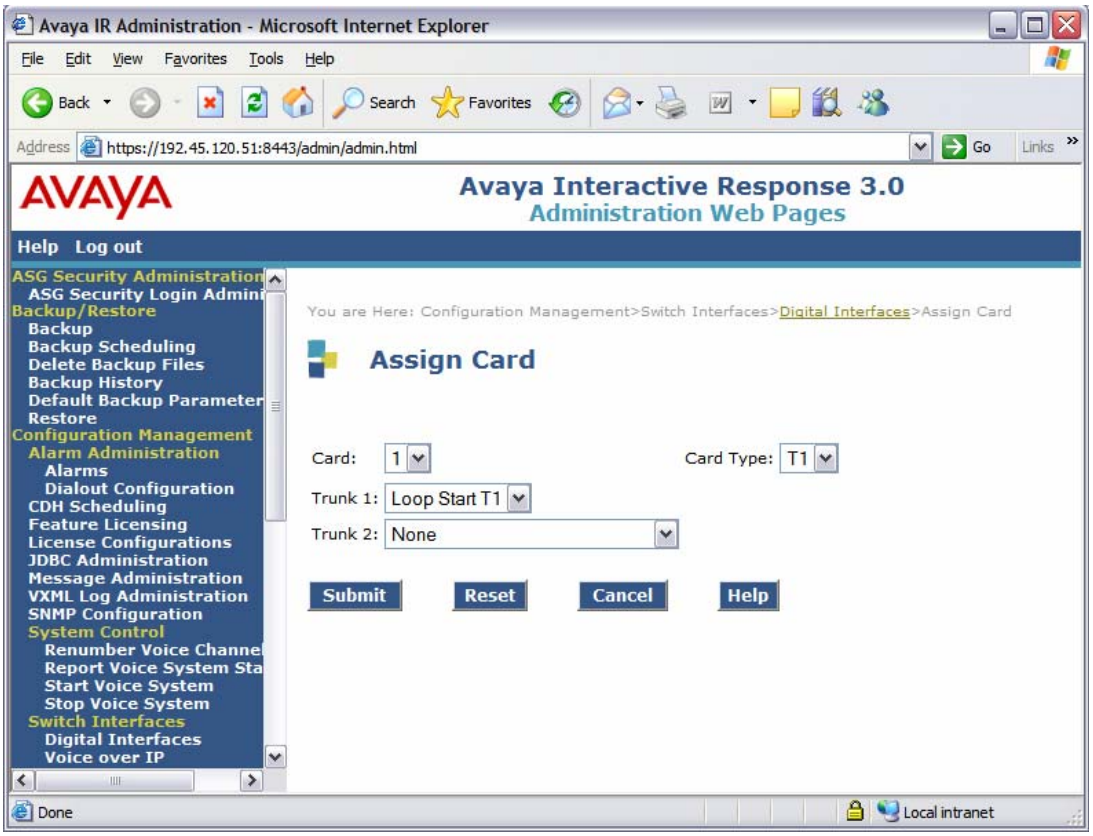


Figure 6: Assign Card

3. On the page shown in **Figure 7**, set the **Frame Type** field to *ESF* and the **Line Code** field to *B8ZS*. Accept the default for the other fields as shown in the figure. Click the **Submit** button.

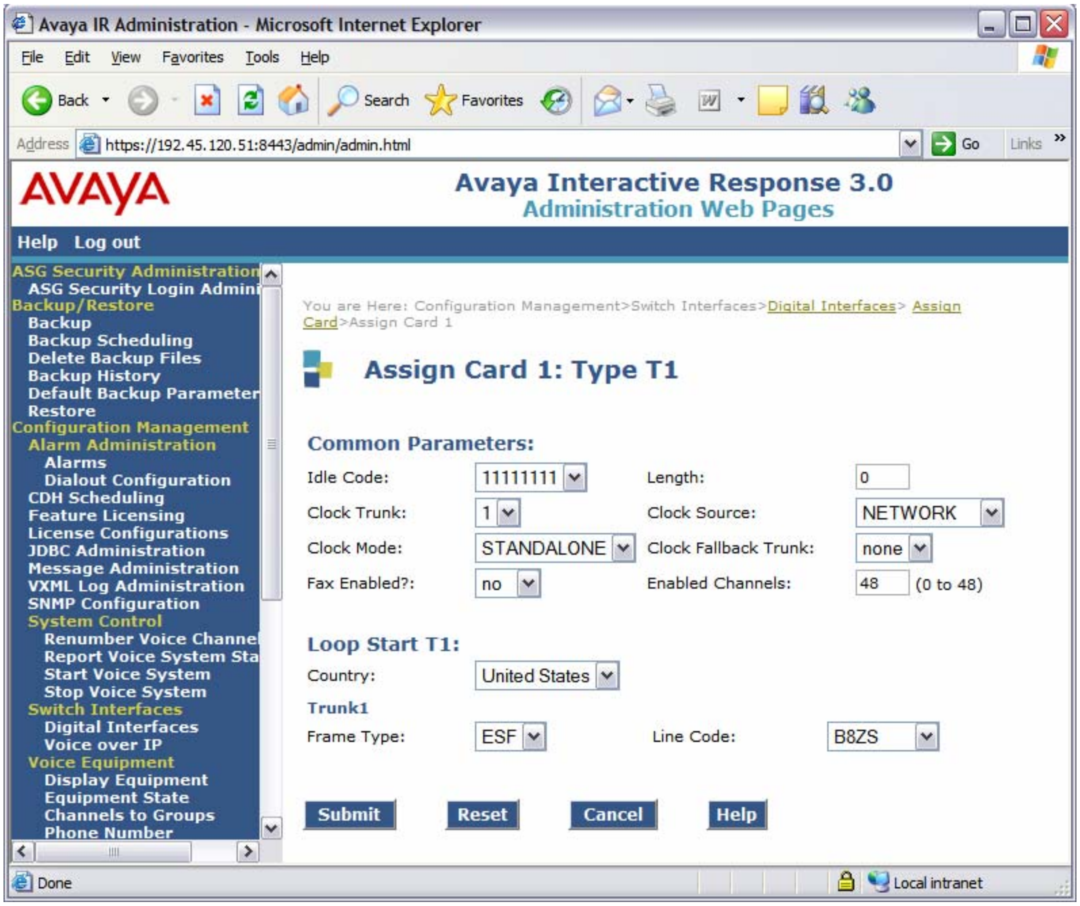


Figure 7: Assign Card Parameters

Next, 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 8** is displayed. On this page, click the **Assign New Recognition Type** button. The **Assign Speech Recognition or DPR Type** page is displayed as shown in **Figure 9**.

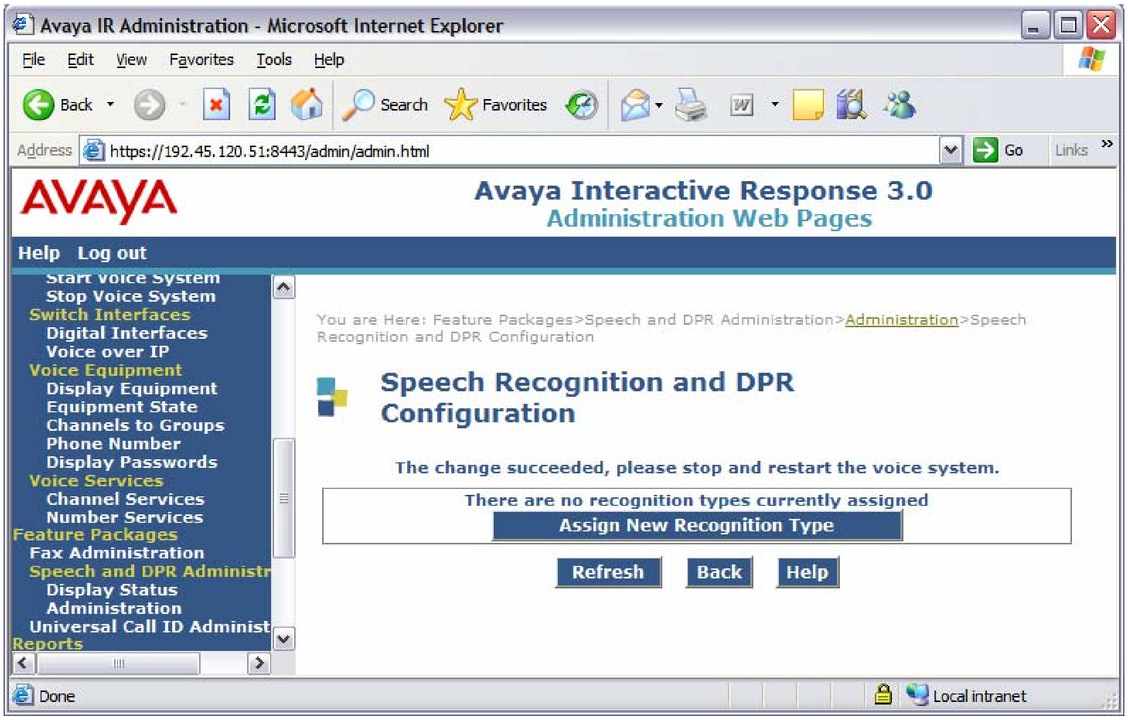


Figure 8: Speech Recognition and DPR Configuration

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

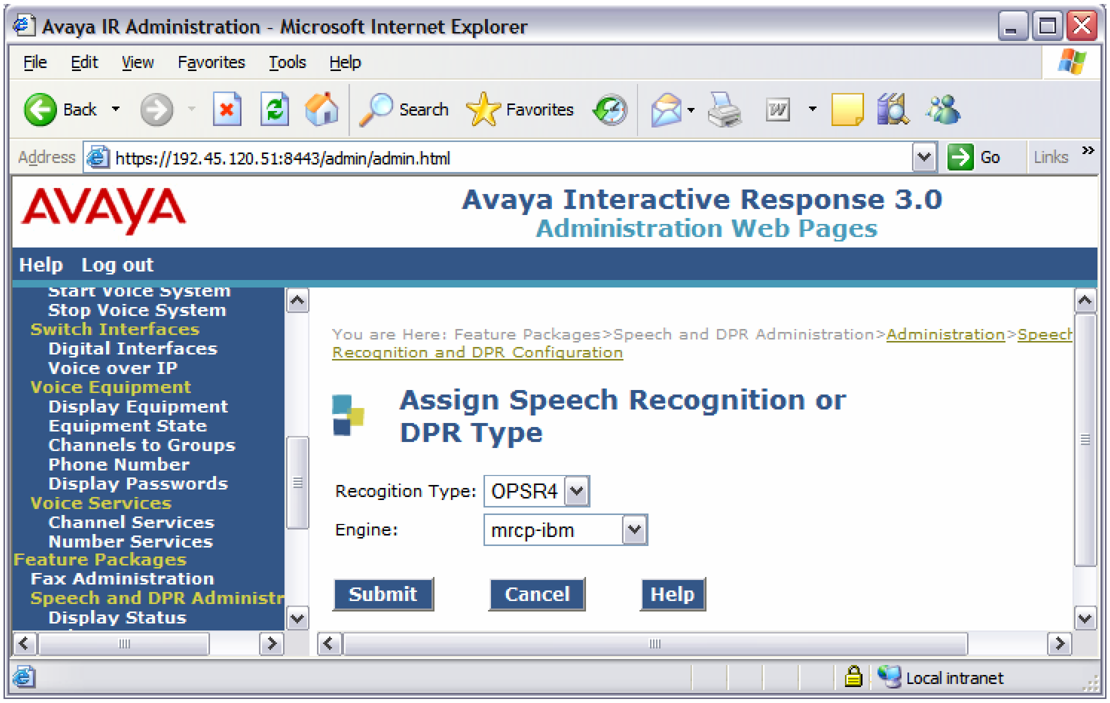


Figure 9: Assign Speech Recognition or DPR Type

On the page shown below, click the **Assign New Server** button to display the **Speech Recognition or DPR Server** page shown in **Figure 11**.

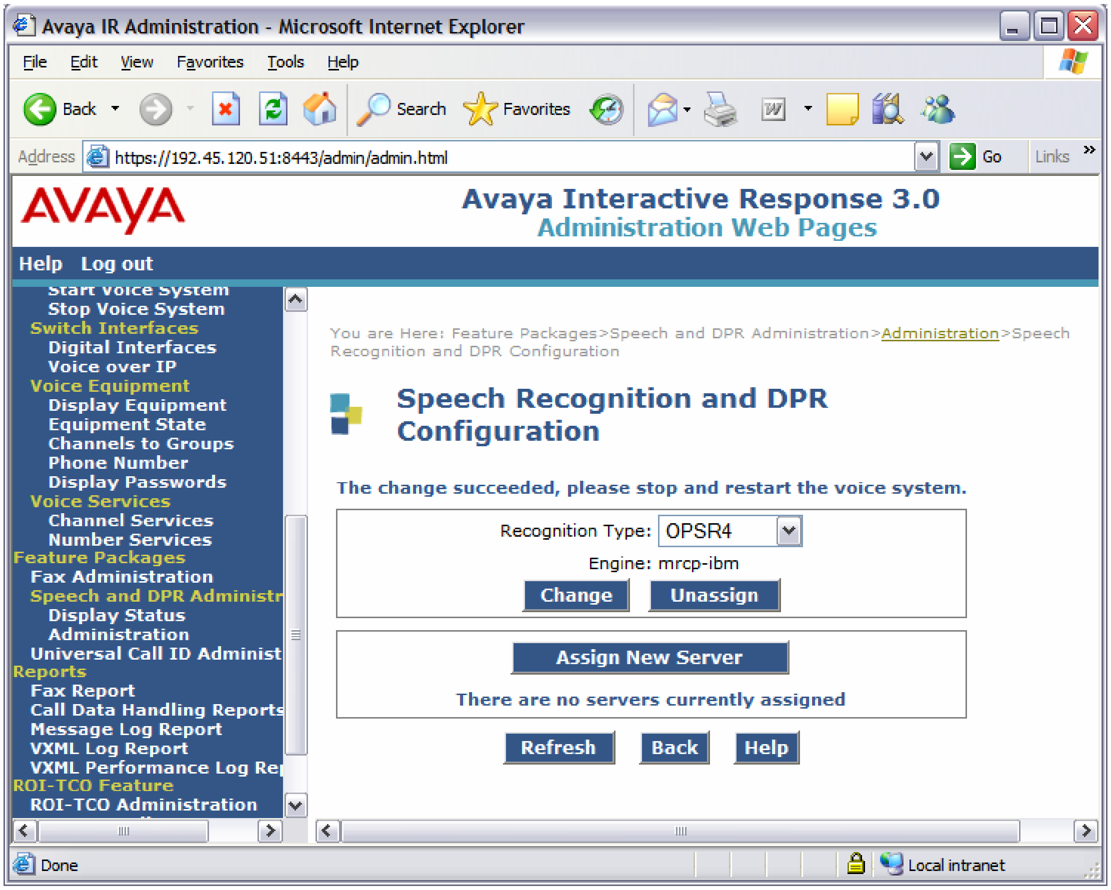


Figure 10: Speech Recognition and DPR Configuration – No Servers Assigned

In the **Assign Speech Recognition or DPR Server** page shown in **Figure 11**, set the **Server Name** and **IP Address** fields to the IP address of the LumenVox Speech Engine, and set the **Ports** field to the number of ports available on the LumenVox Speech Engine according to its installed license. Click **Submit**.

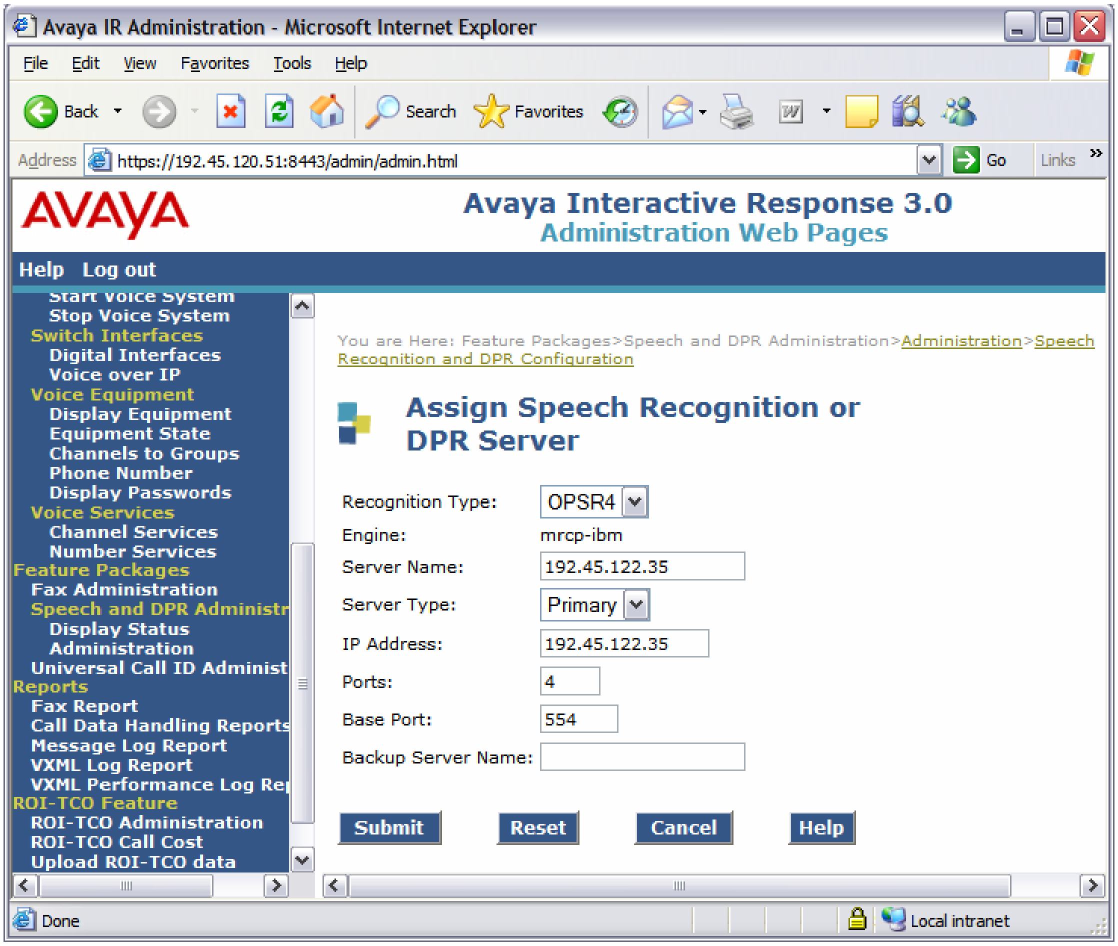


Figure 11: Assign Speech Recognition or DPR Server Parameters

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

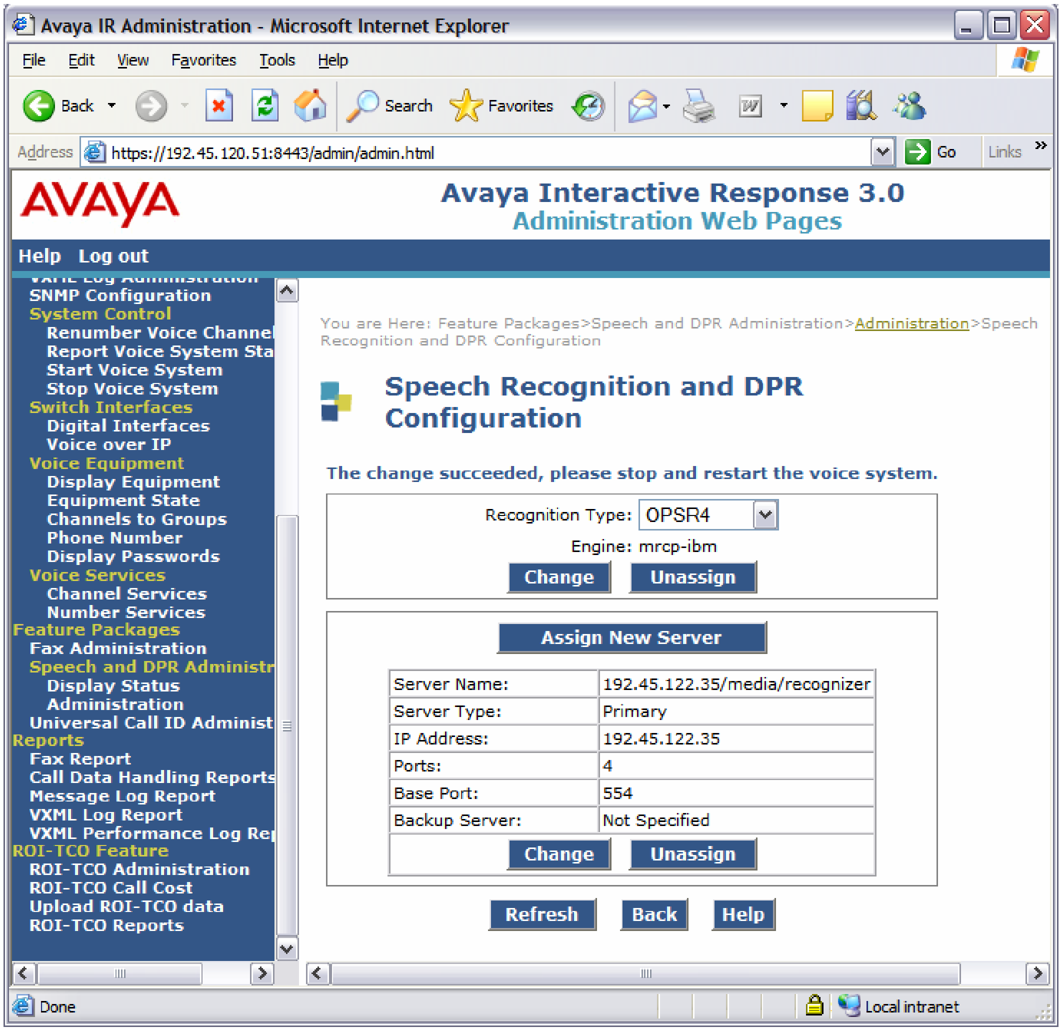


Figure 12: Speech Recognition and DPR Configuration Summary

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 T1 card and speech server are successfully configured, start the Avaya IR by selecting **Start Voice System** under **System Control** in the left pane. Before proceeding, wait for the system to display a message indicating that the startup of the voice system is complete.

Next, assign channels of the T1 card '1' to equipment group '3'. Select the **Channels to Groups** option under Voice Equipment in the left pane, and then select the **Assign** link to display the **Assign Channels to Equipment Groups** page shown in **Figure 13**. Assign group '3' to channels 0-23, which corresponds to the 24 T1 channels, and then click **Submit**. Note that not all of the channels of the T1 were used in this configuration.

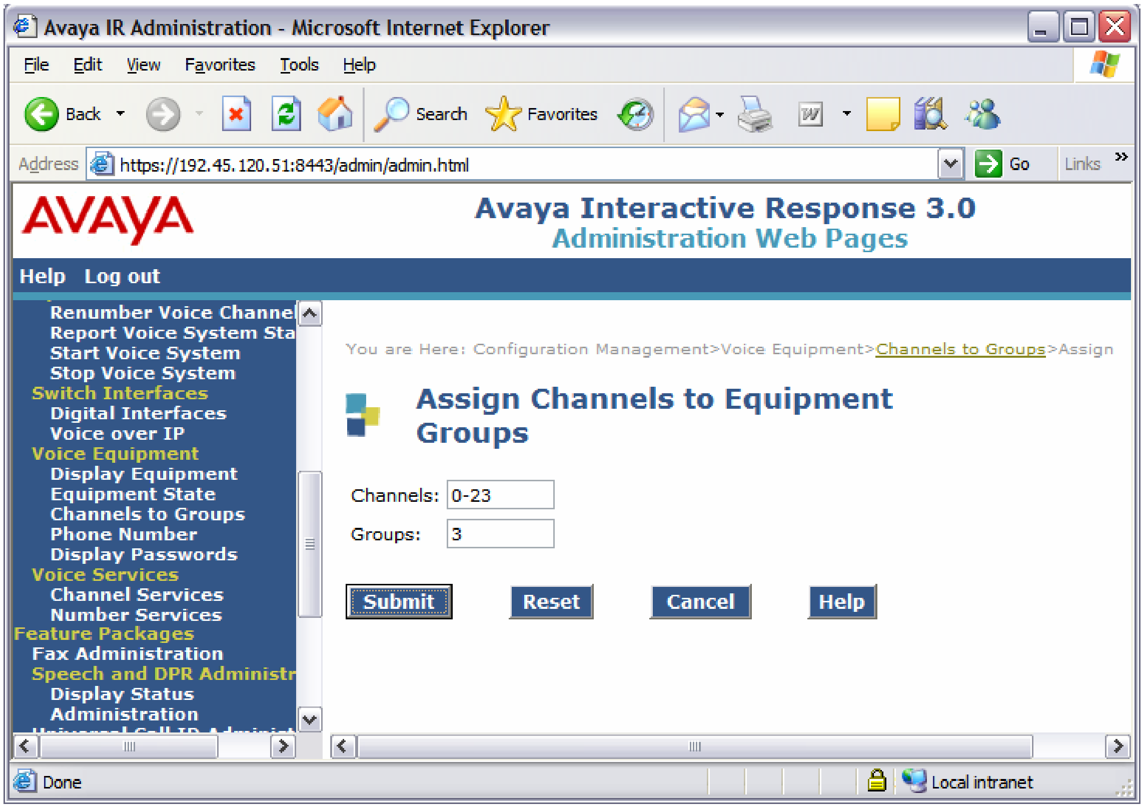


Figure 13: Assign Channels to Equipment Groups

After assigning channels to groups, assign phone numbers to channels. Select the **Phone Number** link under **Voice Equipment** in the left pane to display the **Phone Number – Channel Assignment** page and select the **Assign** link. Assign phone numbers 23201 to 23224 to channels 0 to 23, respectively, as shown in **Figure 14** and then click **Submit**. Essentially, the extensions of the DS1FD stations configured in **Figure 3** are assigned to each T1 channel. Note that not all of the channels of the T1 were used in this configuration.

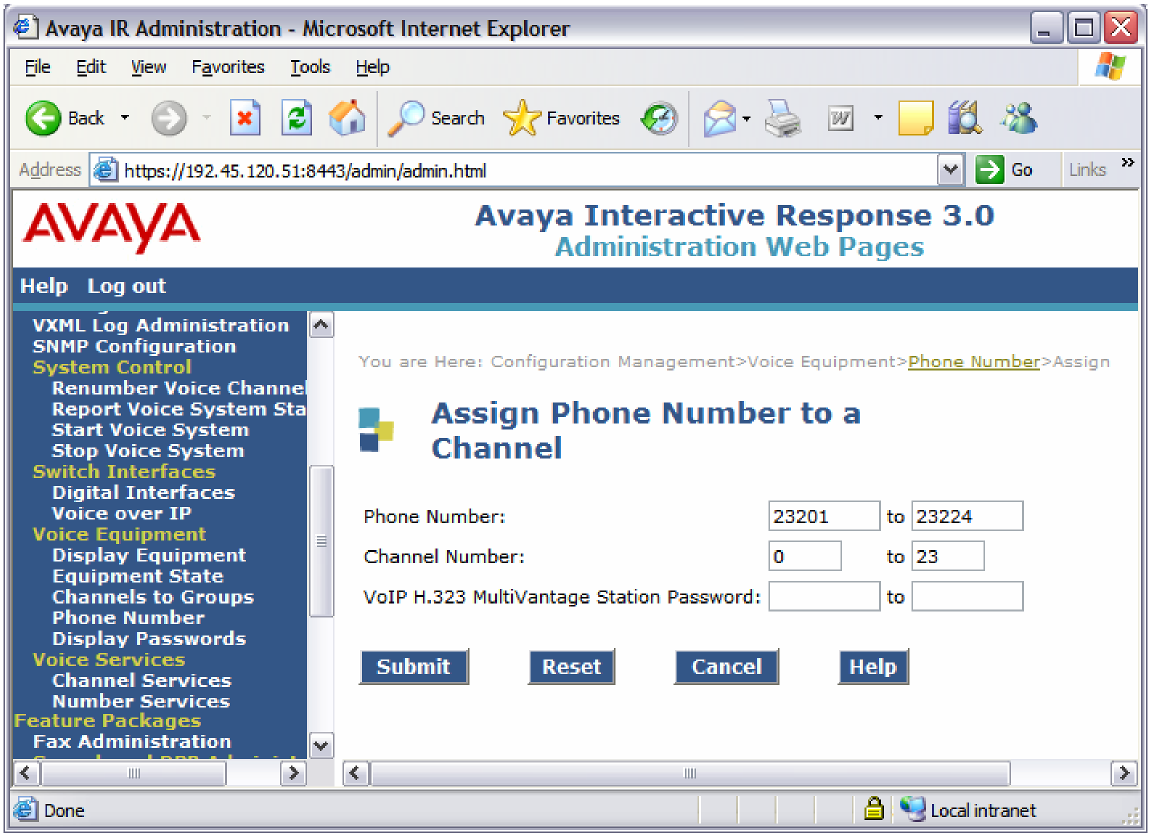


Figure 14: Assign Phone Number

Now, assign a VXML application to channel 0 (i.e., the first channel of the T1 interface). This specifies which application a particular IVR channel would run when it receives a call. Select **Channel Services** from the left pane to display the **Channel Services** page in **Figure 15**. Enable the checkbox for channel 0 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.

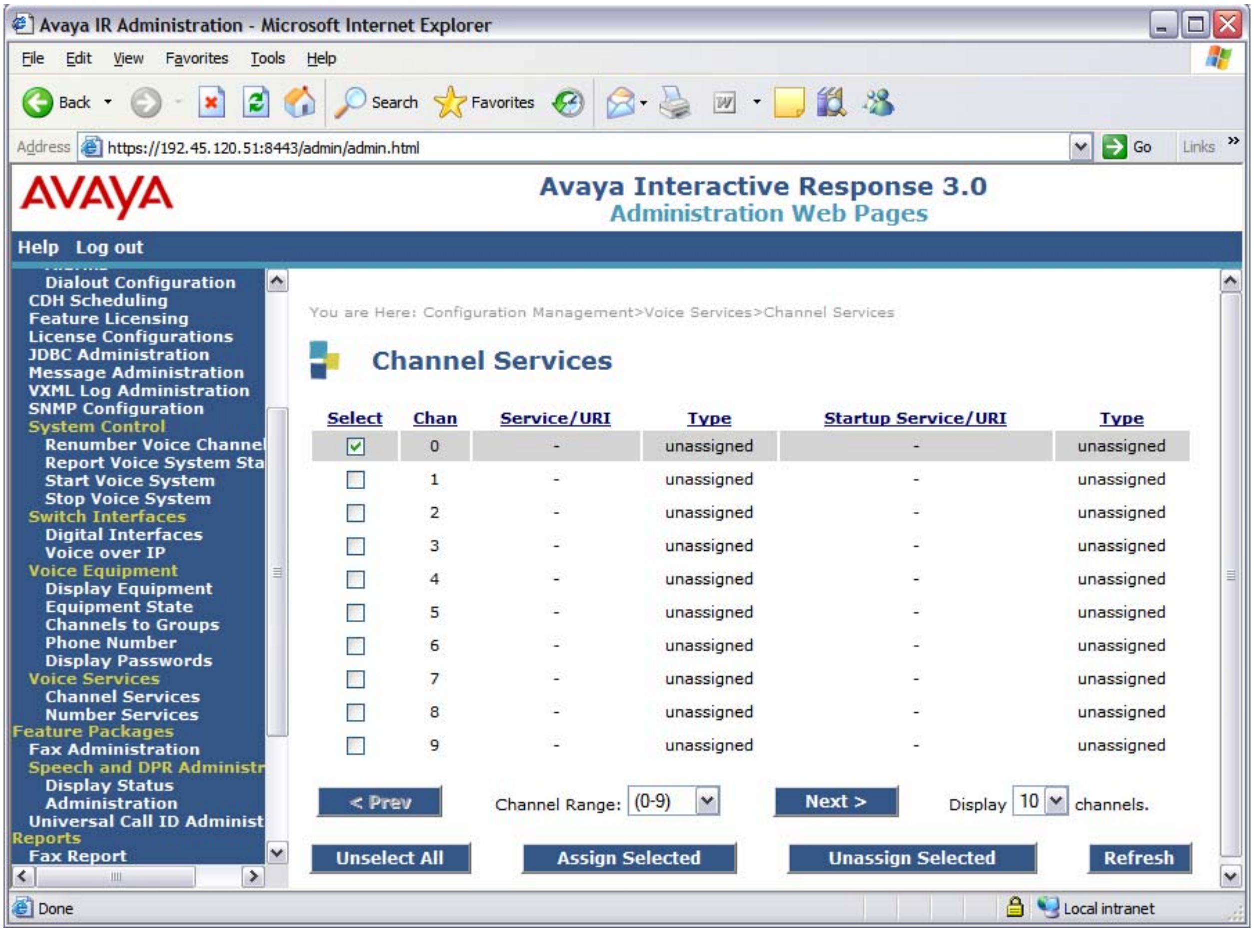


Figure 15: Channel Services

Configure the **Assign Services to Channels** page as shown in **Figure 16**. This configuration assigns a VoiceXML application named *VoiceExternal.xml* to channel 0. Set the **Assign** field to *VXML URI*, set the **URI** field to *http://192.45.60.61/avptestapp/Lumenvox/scripts/VoiceExternal.vxml*, and set the **To Chan(s)** field to '0'. The VoiceXML script is hosted in an application server. Repeat this procedure for all channels that should run this application. Note that the user may change the **To Chan(s)** field to 0-23 to assign the application to the 24 T1 channels in a single step. Click **Submit**.

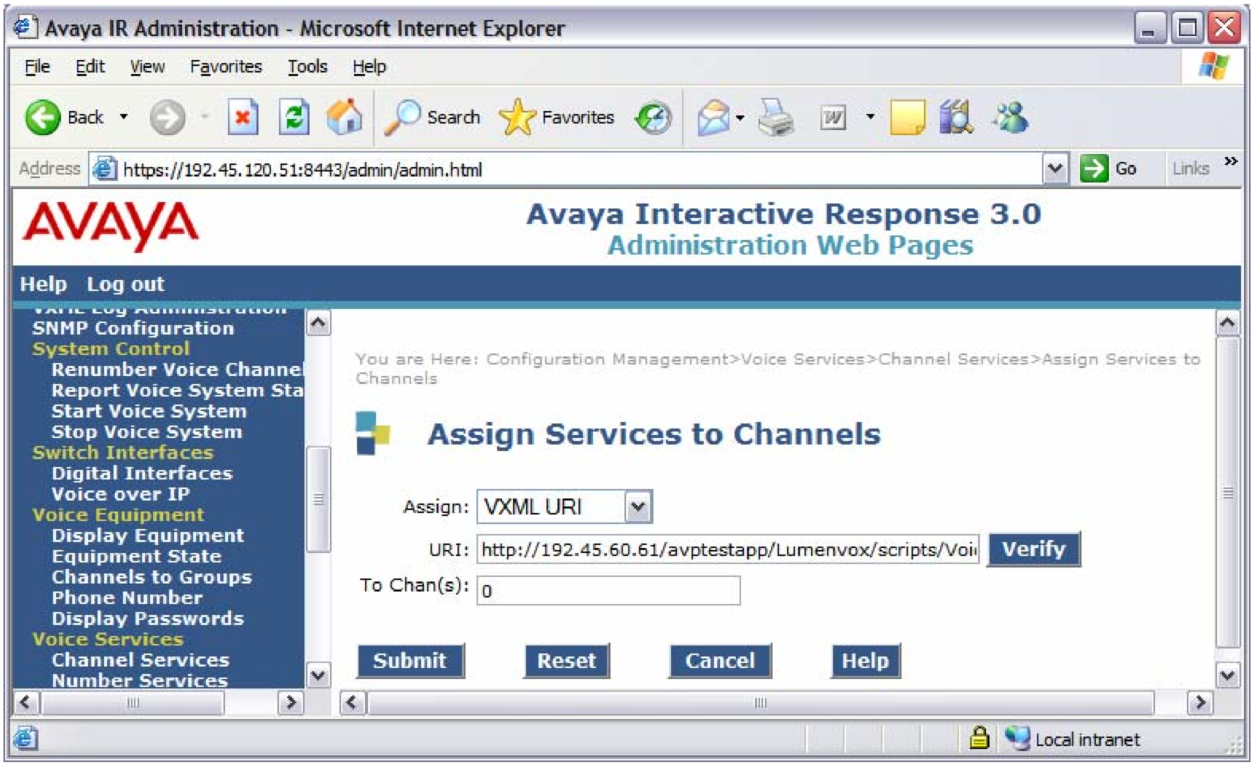


Figure 16: Assign Services to Channels – VXML Application

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 15**, enable the checkbox for channel 1, and click the **Assign Selected** button. In the **Assign Services to Channels** page shown in **Figure 17**, set the **Assign** field to *TAS Service*, set the **Service** and **Startup Services** fields to the TAS application named *avftst* (stands for Avaya transfer test), and set the **To Chan(s)** field to '1'. 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.

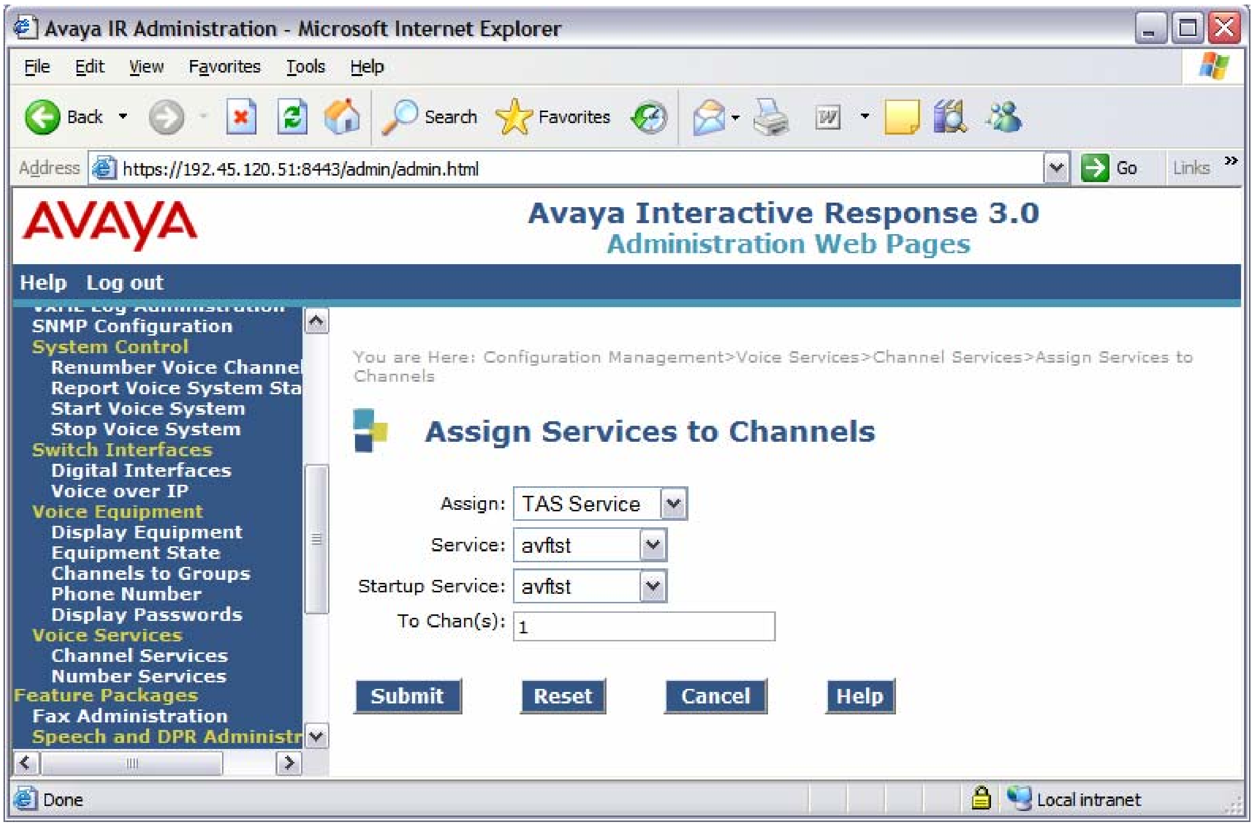
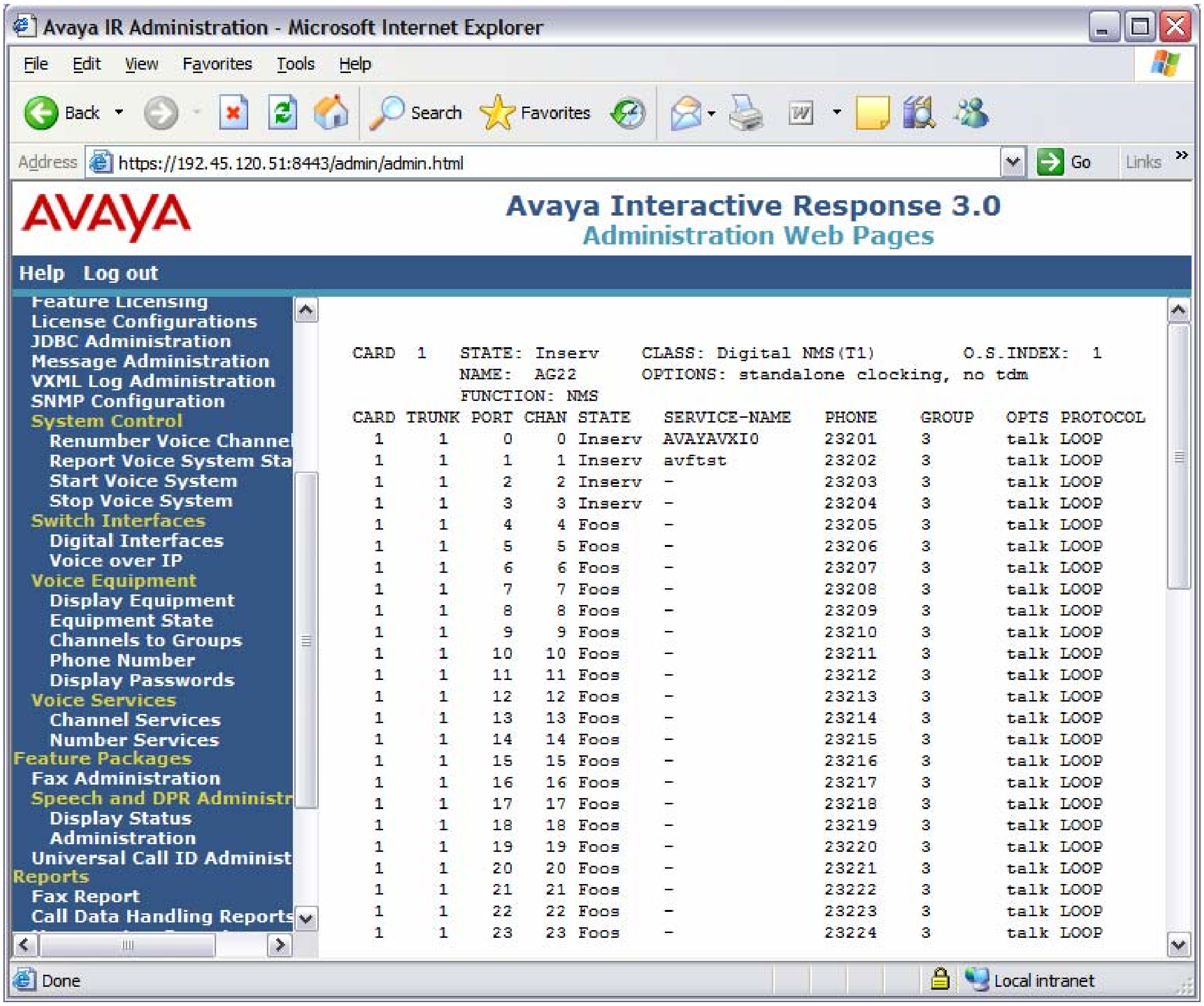


Figure 17: Assign Services to Channels – TAS Application

To view the status of the channels and the channel configuration details, select **Display Equipment** from the left pane. The page in **Figure 18** is displayed. Verify the status of the configured channels. In this configuration, channels 0-3 are in service and channels 0 and 1 have been assigned a VXML and TAS application, respectively. Channel 0 is assigned phone number 23201 and channel 1 has been assigned phone number 23202.



CARD	TRUNK	PORT	CHAN	STATE	SERVICE-NAME	PHONE	GROUP	OPTS	PROTOCOL
1	1	0	0	Inserv	AVAYAVXI0	23201	3	talk	LOOP
1	1	1	1	Inserv	avftst	23202	3	talk	LOOP
1	1	2	2	Inserv	-	23203	3	talk	LOOP
1	1	3	3	Inserv	-	23204	3	talk	LOOP
1	1	4	4	Foos	-	23205	3	talk	LOOP
1	1	5	5	Foos	-	23206	3	talk	LOOP
1	1	6	6	Foos	-	23207	3	talk	LOOP
1	1	7	7	Foos	-	23208	3	talk	LOOP
1	1	8	8	Foos	-	23209	3	talk	LOOP
1	1	9	9	Foos	-	23210	3	talk	LOOP
1	1	10	10	Foos	-	23211	3	talk	LOOP
1	1	11	11	Foos	-	23212	3	talk	LOOP
1	1	12	12	Foos	-	23213	3	talk	LOOP
1	1	13	13	Foos	-	23214	3	talk	LOOP
1	1	14	14	Foos	-	23215	3	talk	LOOP
1	1	15	15	Foos	-	23216	3	talk	LOOP
1	1	16	16	Foos	-	23217	3	talk	LOOP
1	1	17	17	Foos	-	23218	3	talk	LOOP
1	1	18	18	Foos	-	23219	3	talk	LOOP
1	1	19	19	Foos	-	23220	3	talk	LOOP
1	1	20	20	Foos	-	23221	3	talk	LOOP
1	1	21	21	Foos	-	23222	3	talk	LOOP
1	1	22	22	Foos	-	23223	3	talk	LOOP
1	1	23	23	Foos	-	23224	3	talk	LOOP

Figure 18: Display Equipment

4. Configure LumenVox Speech Engine

This section covers the configuration required for the LumenVox Speech Engine. These changes are contained in the `mrcp.config` file. The required changes are highlighted in **bold**. Refer to [5] 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            = MRCP_Log_Avaya.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.
#----------------------------------------------------------------
mrcp_server_ip             = 192.45.122.35

#----------------------------------------------------------------
# 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 = 192.45.122.35

#-----------------------------------------------------------------
# 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

#-------------------------------------------------------------------
# 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=true
recognition timeout=60000
dtmf inter digit timeout=5000
snr sensitivity lvl=50
save waveform=false
waveform url location=file:///c:/
barge_in_timeout=150000
```

Figure 19: 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.

5.1. General Test Approach

The interoperability compliance test included feature and serviceability testing. The feature testing focused on placing calls to Avaya IR to run VoiceXML applications that used the ASR engine in the LumenVox Speech Engine and the LumenVox MRCP Server. The compliance testing focused on placing calls to verify speech and DTMF tone recognition.

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

5.2. Test Results

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 IVR 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 18**.

2. From the Avaya Communication Manager SAT, verify that the T1 channels are in-service using the **status station** `<extension>` as shown in **Figure 20.**

```
status station 23201                                            Page   1 of   3
GENERAL STATUS
Administered Type: DS1FD                Service State: in-service/on-hook
Connected Type: N/A
Extension: 23201
Port: 01A1101       Parameter Download: not-applicable
Call Parked? no                 SAC Activated? no
Ring Cut Off Act? no
Active Coverage Option: 1

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 20: Status Station

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

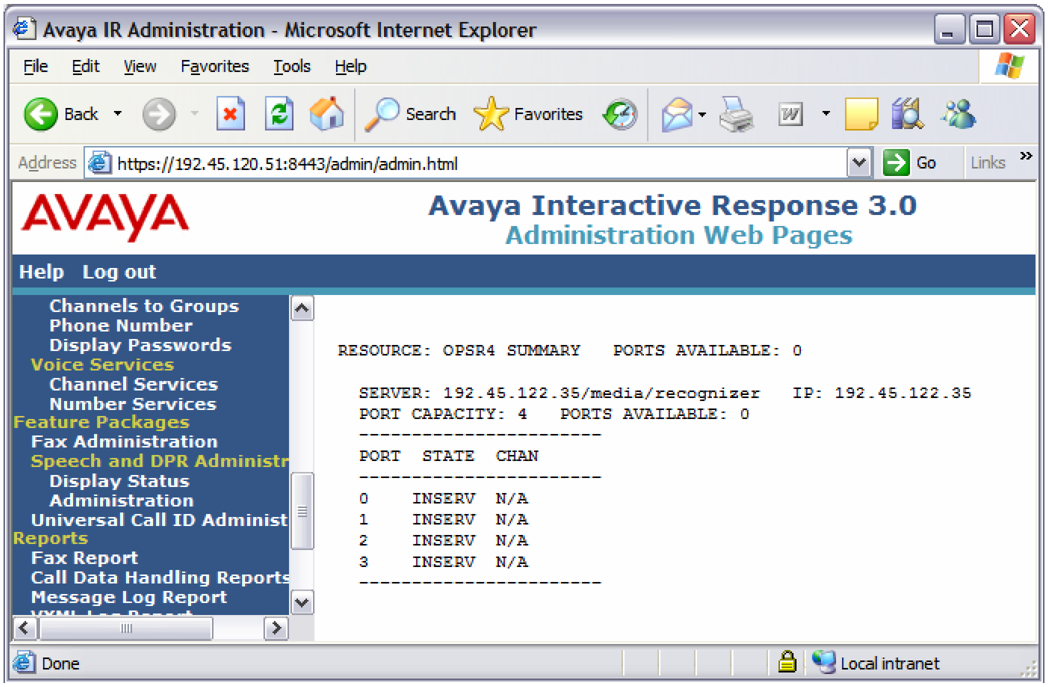


Figure 21: OPSR Summary

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

7. Support

To contact LumenVox by phone or access their website:

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

8. Conclusion

These Application Notes describe the configuration steps required to integrate the LumenVox Speech Engine with Avaya Interactive Response. All feature and serviceability test cases were completed successfully.

9. Additional References

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

[1] *Administrator Guide for Avaya Communication Manager*, Document 03-300509, Issue 3.1, February 2007, available at http://support.avaya.com.

[2] *Feature Description and Implementation for Avaya Communication Manager*, Document 555-245-205, Issue 5, February 2007, 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 (IR) Release 3.0 Documentation Library*, June 2007, available at http://support.avaya.com.

[5] *LumenVox Online Documentation* available at http://help.lumenvox.com/Robo/BIN/Robo.dll?tpc=%2Frobo%2Fprojects%2Fspeechengine%2Froot%2Fwelcome.htm&mgr=agm&project=speechengine&wnd=speechengine%7CLumenvox&agt=wsm&refer=http%3A%2F%2Fwww.lumenvox.com%2Fsupport%2F&ctxid=support

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.