



Avaya Solution & Interoperability Test Lab

Application Notes for Nuance Open Speech Attendant with Avaya Voice Portal - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Nuance OpenSpeech Attendant (OSA) to interoperate with Avaya Voice Portal. The Nuance OSA allows callers to speak the name or enter the phone number of a person, department, service, or location and be automatically transferred to the requested destination.

Information in these Application Notes has been obtained through *DeveloperConnection* compliance testing and additional technical discussions. Testing was conducted via the *DeveloperConnection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

Nuance OSA is a VXML based auto attendant solution. The Nuance OSA allows callers to speak the name or enter the phone number of a person, department, service, or location, and be automatically transferred to the requested destination. The Nuance OSA can also route callers through menu-driven options and provide frequently requested information such as operating hours, mailing address, and driving directions. However, the compliance testing focused on voice and DTMF touch tone recognition, and call routing functionality.

The Nuance OSA consisted of Nuance OpenSpeech Recognizer for speech recognition and Nuance RealSpeak to convert text into synthesized speech. A VoiceXML2.0 compliant sample speech application was used to respond to callers and to transfer calls to correct destinations. The Nuance OSA used the Apache HTTP Server and Microsoft SQL Server to manage access and storage tasks such as storing/editing phone directory, grammars and call flow.

In the compliance testing, calls were placed to Avaya Communication Manager and delivered to Avaya Voice Portal over available lines, administered as phantom IP stations with type “7434ND” on Avaya Communication Manager. The Avaya Voice Portal ran the Nuance sample speech application from the Apache HTTP Server, and used phone directory from the Microsoft SQL Server to transfer calls to correct destinations. The phone directory with transfer entries is created in the Nuance OSA. When a caller speaks the name or enters the phone number of a destination, the Nuance OSA searches the phone directory for matching transfer entry. Nuance OSA provides the destination phone number from the matching entry to Avaya Voice Portal for call transfer. The configuration used for the compliance testing is shown in **Figure 1**.

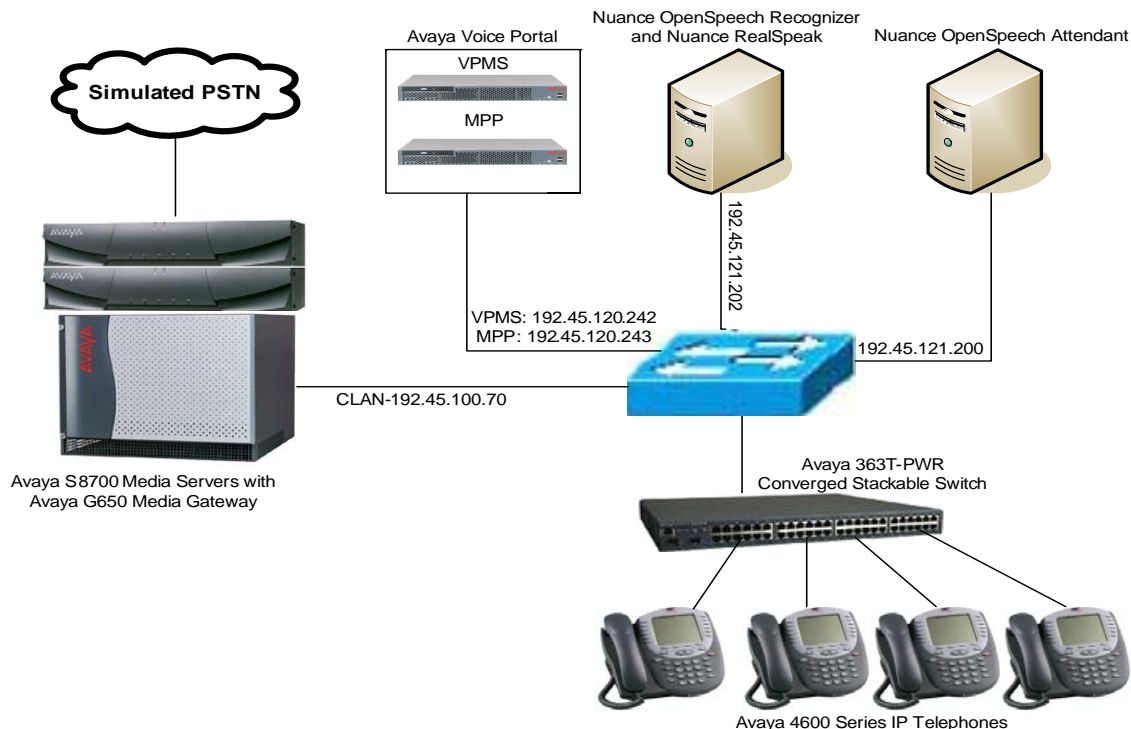


Figure 1: Compliance Test Configuration

2. Equipment and Software Validation

The following equipment and software/firmware were used for the configuration utilized in the testing.

Equipment	Software/Firmware
Avaya S870 Media Server with Avaya G650 Media Gateway	Avaya Communication Manager 3.1.2, R013x.01.2.632.1
Avaya Voice Portal <ul style="list-style-type: none"> Voice Portal Management System (VPMS) Media Processing Platform (MPP) 	3.0.1.2.2904 3.0.1.2.2904:3.0.1.3-0002
Avaya 4600 Series H.323 IP Telephones <ul style="list-style-type: none"> 4610SW 4620SW 	2.3 2.3
Nuance OSA	2.1 with Patch 210HF33
Nuance OpenSpeech Recognizer	3.0.11
Nuance RealSpeak	4.0.12

3. Configure Avaya Communication Manager

This section provides the procedure for configuring Avaya Communication Manager. The Avaya System Access Terminal (SAT) is used to issue the commands to the S8700 Media Server. The procedure includes the following areas:

- Display available license
- Administer system parameters features
- Administer IP codec set
- Administer IP network region
- Administer stations
- Administer hunt group

These Application Notes assume that the necessary configuration on Avaya Communication Manager is in place to enable calls between two H.323 IP telephones. Also, it is assumed that four H.323 IP stations with station extensions 77201, 77202, 77203, and 77204 are in place. These stations are physical IP stations, which are used to answer transferred calls from Avaya Voice Portal.

3.1. Display Available Licenses

Step	Description																														
1.	Use the “display system-parameters customer-options” command. On Page 10, verify that there are sufficient IP_API_A licenses. If not, contact an authorized Avaya account representative to obtain the license.																														
	<div><div>display system-parameters customer-options</div><div>Page 10 of 11</div><div>MAXIMUM IP REGISTRATIONS BY PRODUCT ID</div><table><tr><th>Product ID</th><th>Rel. Limit</th><th>Used</th></tr><tr><td>IP_API_A</td><td>: 100</td><td>7</td></tr><tr><td>IP_API_B</td><td>: 0</td><td>0</td></tr><tr><td>IP_API_C</td><td>: 0</td><td>0</td></tr><tr><td>IP_Agent</td><td>: 1000</td><td>0</td></tr><tr><td>IP_IR_A</td><td>: 100</td><td>0</td></tr><tr><td>IP_Phone</td><td>: 12000</td><td>8</td></tr><tr><td>IP_ROMax</td><td>: 12000</td><td>0</td></tr><tr><td>IP_Soft</td><td>: 1000</td><td>0</td></tr><tr><td>IP_eCons</td><td>: 0</td><td>0</td></tr></table></div>	Product ID	Rel. Limit	Used	IP_API_A	: 100	7	IP_API_B	: 0	0	IP_API_C	: 0	0	IP_Agent	: 1000	0	IP_IR_A	: 100	0	IP_Phone	: 12000	8	IP_ROMax	: 12000	0	IP_Soft	: 1000	0	IP_eCons	: 0	0
Product ID	Rel. Limit	Used																													
IP_API_A	: 100	7																													
IP_API_B	: 0	0																													
IP_API_C	: 0	0																													
IP_Agent	: 1000	0																													
IP_IR_A	: 100	0																													
IP_Phone	: 12000	8																													
IP_ROMax	: 12000	0																													
IP_Soft	: 1000	0																													
IP_eCons	: 0	0																													

3.2. Administer System Parameters Features

Step	Description
1.	<p>Use the “change system-parameters features” command. On Page 6, set 7434ND to “y”.</p> <pre> change system-parameters features FEATURE-RELATED SYSTEM PARAMETERS Public Network Trunks on Conference Call: 5 Auto Start? n Conference Parties with Public Network Trunks: 6 Auto Hold? n Conference Parties without Public Network Trunks: 6 Attendant Tone? y Night Service Disconnect Timer (seconds): 180 Bridging Tone? n Short Interdigit Timer (seconds): 3 Conference Tone? n Unanswered DID Call Timer (seconds): Intrusion Tone? n Line Intercept Tone Timer (seconds): 30 Mode Code Interface? n Long Hold Recall Timer (seconds): 0 Reset Shift Timer (seconds): 0 Station Call Transfer Recall Timer (seconds): 0 DID Busy Treatment: tone Allow AAR/ARS Access from DID/DIOD? y Allow ANI Restriction on AAR/ARS? n Use Trunk COR for Outgoing Trunk Disconnect? n 7405ND Numeric Terminal Display? n 7434ND? y DISTINCTIVE AUDIBLE ALERTING Internal: 1 External: 2 Priority: 3 Attendant Originated Calls: external </pre>

3.3. Administer IP Codec Set

The Avaya Voice Portal supports only two audio codecs, G711MU and G.711A, and both types of media encryptions, AES and AEA. In the compliance testing, the G.711MU audio codec with no media encryption was used for the phantom IP stations to communicate with the Avaya Voice Portal. The phantom IP stations are different than the physical IP stations. The phantom IP stations are used for voice over IP connections between Avaya Voice Portal and Avaya Communication Manager.

Step	Description
1.	<p>Use the “change ip-codec-set <n>” command, where <n> is the codec set number. On Page 1, enter the following values for the first line entry. Retain the default values for the remaining fields.</p> <ul style="list-style-type: none"> • Audio Codec: Enter audio codec “G.711MU”. • Silence Suppression: Enter “n”. • Frames Per Pkt: Enter “2” <p>Enter “none” in the first field under Media Encryption.</p> <pre> change ip-codec-set 1 Page 1 of 2 IP Codec Set Codec Set: 1 Audio Silence Frames Packet Codec Suppression Per Pkt Size(ms) 1: G.711MU n 2 20 2: 3: 4: 5: 6: 7: Media Encryption 1: none 2: 3: </pre>

3.4. Administer IP Network Region

This section describes how to configure the IP network region, where the phantom IP stations are registered.

Step	Description
1.	<p>Use the “change ip-network-region <n>” command, where <n> is a valid network region number. On Page 1, enter the audio codec set number “1” from Section 3.3, in the Codec Set field. Retain the default values in the remaining fields.</p> <pre> change ip-network-region 1 Page 1 of 19 IP NETWORK REGION Region: 1 Location: Authoritative Domain: Name: MEDIA PARAMETERS Intra-region IP-IP Direct Audio: yes Codec Set: 1 Inter-region IP-IP Direct Audio: yes UDP Port Min: 2048 IP Audio Hairpinning? y UDP Port Max: 3029 DIFFSERV/TOS PARAMETERS RTCP Reporting Enabled? y Call Control PHB Value: 34 RTCP MONITOR SERVER PARAMETERS Audio PHB Value: 46 Use Default Server Parameters? y Video PHB Value: 26 802.1P/Q PARAMETERS Call Control 802.1p Priority: 7 Audio 802.1p Priority: 6 Video 802.1p Priority: 5 AUDIO RESOURCE RESERVATION PARAMETERS H.323 IP ENDPOINTS RSVP Enabled? n H.323 Link Bounce Recovery? y Idle Traffic Interval (sec): 20 Keep-Alive Interval (sec): 5 Keep-Alive Count: 5 </pre>

3.5. Configure Stations

This section describes how to configure the phantom IP stations that are used by the Avaya Voice Portal to connect to Avaya Communication Manager. In the compliance testing, four phantom IP stations were used so Avaya Voice Portal can have four available lines to communicate with Avaya Communication Manager.

Step	Description
1.	<p>Use the “add station <n>” command, where <n> is a valid unused station extension. On Page 1, enter the following values and retain the default values for the remaining fields.</p> <ul style="list-style-type: none"> • Type: Enter station type “7434ND”. • Port: Enter “IP”. • Name: Enter a descriptive name. • Security Code: Enter a desired station security code. • Display Module: Enter “y”. • IP Softphone: Enter “y”.
	<pre> add station 77101 Page 1 of 5 STATION Extension: 77101 Lock Messages? n BCC: 0 Type: 7434ND Security Code: * TN: 1 Port: IP Coverage Path 1: COR: 1 Name: Voice Portal Station 1 Coverage Path 2: COS: 1 Hunt-to Station: STATION OPTIONS Loss Group: 2 Personalized Ringing Pattern: 1 Data Module? n Message Lamp Ext: 77101 Display Module? y Display Language: english Coverage Module? n Media Complex Ext: IP SoftPhone? y IP Video Softphone? n </pre>

Step	Description
	<p>On Page 2, set the Multimedia Mode field to “enhanced”. Retain the default values for the remaining fields.</p>
	<div> <div>add station 77101</div> <div>Page 2 of 5</div> </div> <div> <div>STATION</div> <div> <div>FEATURE OPTIONS</div> <div> <div>LWC Reception: spe</div> <div>Auto Select Any Idle Appearance? n</div> <div>LWC Activation? y</div> <div>Coverage Msg Retrieval? y</div> <div>LWC Log External Calls? n</div> <div>Auto Answer:</div> <div>none</div> <div>CDR Privacy? n</div> <div>Data Restriction? n</div> <div>Redirect Notification? y</div> <div>Idle Appearance Preference? n</div> <div>Per Button Ring Control? n</div> <div>Bridged Idle Line Preference? n</div> <div>Bridged Call Alerting? n</div> <div>Restrict Last Appearance? y</div> <div>Active Station Ringing: single</div> <div>Conf/Trans on Primary Appearance? n</div> <div>H.320 Conversion? n</div> <div>Per Station CPN - Send Calling Number?</div> <div>Service Link Mode: as-needed</div> <div>Multimedia Mode: enhanced</div> <div>MWI Served User Type:</div> <div>Display Client Redirection? n</div> <div>AUDIX Name:</div> <div>Select Last Used Appearance? n</div> <div>Coverage After Forwarding? s</div> <div>Remote Softphone Emergency Calls: as-on-local</div> <div>Direct IP-IP Audio Connections? y</div> <div>Emergency Location Ext: 77101</div> <div>Always Use? n</div> <div>IP Audio Hairpinning? y</div> </div> </div> </div>
	<p>On Page 3, set the first two fields under BUTTON ASSIGNMENTS to “call-appr”. Retain the default values for the remaining fields.</p>
	<div> <div>add station 77101</div> <div>Page 3 of 5</div> </div> <div> <div>STATION</div> <div> <div>SITE DATA</div> <div> <div>Room:</div> <div>Headset? n</div> <div>Jack:</div> <div>Speaker? n</div> <div>Cable:</div> <div>Mounting: d</div> <div>Floor:</div> <div>Cord Length: 0</div> <div>Building:</div> <div>Set Color:</div> </div> </div> <div> <div>ABBREVIATED DIALING</div> <div> <div>List1:</div> <div>List2:</div> <div>List3:</div> </div> </div> <div> <div>BUTTON ASSIGNMENTS</div> <div> <div>1: call-appr</div> <div>6:</div> <div>2: call-appr</div> <div>7:</div> <div>3:</div> <div>8:</div> <div>4:</div> <div>9:</div> </div> </div> </div>

Step	Description
	On Page 5, set the first field under DISPLAY BUTTON ASSIGNMENTS to “normal”.
	<div> <div>add station 77101</div> <div>Page 5 of 5</div> <div>STATION</div> <div>DISPLAY BUTTON ASSIGNMENTS</div> <div>1: normal</div> <div>2:</div> <div>3:</div> </div>
2.	Repeat Step 1 as necessary to add additional phantom IP stations 77102-77104. Note: In the compliance testing, the same value for the Security Code field was used for all the phantom IP stations.

3.6. Configure Hunt Group

This section describes how to configure the hunt group. A hunt group is created for the phantom IP stations, administered in Section 3.5. The hunt group extension is the pilot number that is used to call the sample speech application. In the compliance testing, two hunt groups each with two group members were used for multiple entry points in the Nuance OSA.

Step	Description
1.	<p>Use the “add hunt-group <n>” command, where <n> is an unused hunt group number. On Page 1, enter the following values and retain the default values in the remaining fields.</p> <ul style="list-style-type: none"> • Group Name: Enter a descriptive name. • Group Extension: Enter a valid group extension, for example “77100”. • ISDN/SIP Caller Display: Enter “grp-name”. This enables Avaya Communication Manager to display the name as entered in the Group Name field on the caller’s telephone. <pre> add hunt-group 77 Page 1 of 60 HUNT GROUP Group Number: 77 ACD? n Group Name: OSA VP Queue? n Group Extension: 77100 Vector? n Group Type: ucd-mia Coverage Path: TN: 1 Night Service Destination: COR: 1 MM Early Answer? n Security Code: Local Agent Preference? n ISDN/SIP Caller Display: grp-name </pre> <p>On Page 3, enter the phantom IP station extension administered in Section 3.5, in the Ext fields of the GROUP MEMBER ASSIGNMENTS.</p> <pre> add hunt-group 77 Page 3 of 60 HUNT GROUP Group Number: 77 Group Extension: 77100 Group Type: ucd-mia Member Range Allowed: 1 - 1500 Administered Members (min/max): 1 / 2 Total Administered Members: 2 GROUP MEMBER ASSIGNMENTS Ext Name (24 characters) Ext Name (24 characters) 1: 77101 OSAVP Station1 14: 2: 77102 OSAVP Station2 15: 3: 16: 4: 17: 5: 18: </pre>
2.	<p>Repeat Step 1 to add an additional hunt group with extension 77200, and administer the phantom IP stations 77103 and 77104 from Section 3.5 as members of this hunt group.</p>

4. Configure Avaya Voice Portal

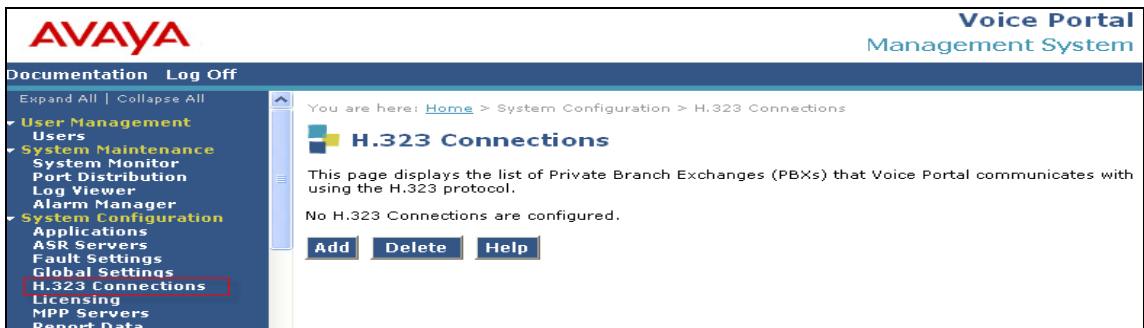
This section provides the procedure for configuring the Avaya Voice Portal. The procedure includes the following areas:

- Administer H.323 connection
- Administer MPP server
- Administer ASR (Automatic Speech Recognition) server
- Administer TTS (Text-to-Speech) server
- Administer speech application
- Administer VoIP settings

The Avaya Voice Portal consisted of a VPMS server and a MPP server. It is assumed that the VPMS and the MPP servers are installed and have appropriate licenses. The VPMS manages the MPPs and provides a web interface for administering Avaya Voice Portal. The MPP communicates with Nuance OpenSpeech Recognizer, Nuance RealSpeak and Avaya Communication Manager to provide voice response media services. The MPP used a H.323 connection to communicate with Avaya Communication Manager.

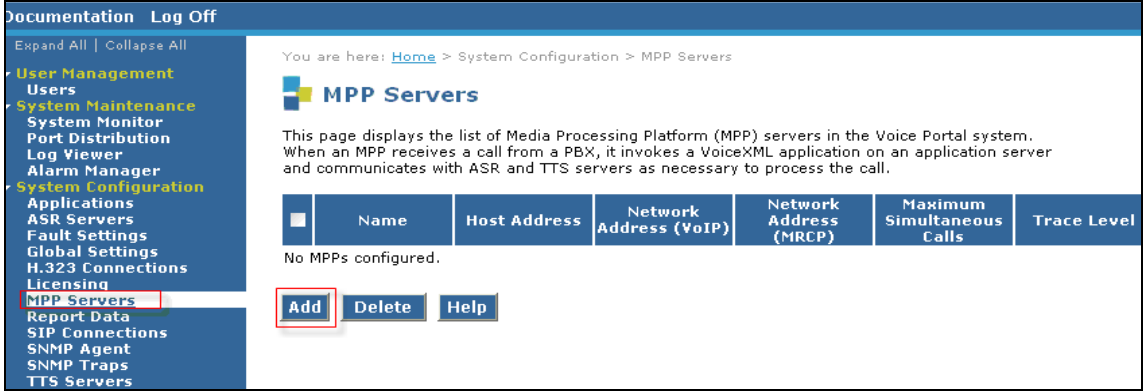
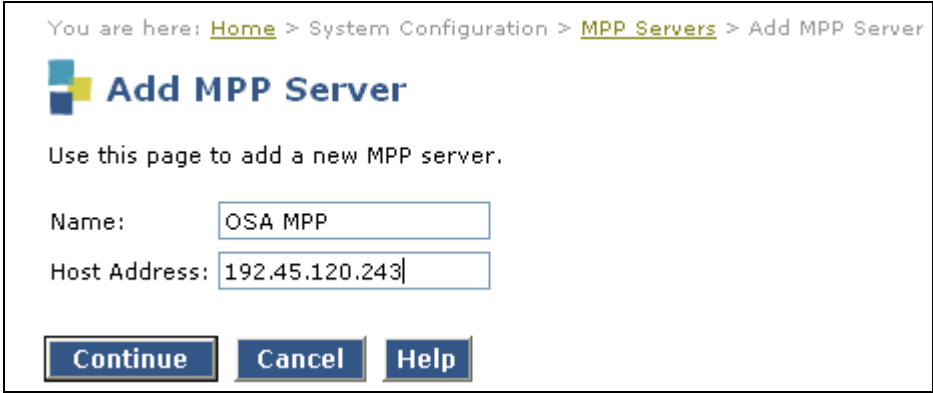
The VPMS web interface can be accessed by entering <http://<Hostname or IP address of VPMS server>:8080/VoicePortal> in the URL. For example, <http://192.45.120.242:8080/VoicePortal>. Log into VPMS with proper credentials.

4.1. Administer H.323 Link

Step	Description
1.	<p>In the left pane of the Voice Portal Management System, navigate to System Configuration → H.323 Connections. The H.323 Connections screen appears in the right pane. Click Add.</p>  <p>The screenshot shows the Avaya Voice Portal Management System web interface. The left navigation pane is expanded to 'System Configuration', and 'H.323 Connections' is selected and highlighted with a red box. The main content area displays the 'H.323 Connections' page, which includes a breadcrumb trail 'Home > System Configuration > H.323 Connections', a description of the page, and a status message 'No H.323 Connections are configured.' with 'Add', 'Delete', and 'Help' buttons.</p>

Step	Description															
2.	<p>Next, the Add H.323 Connection screen appears. Enter the following values:</p> <ul style="list-style-type: none">• Name: Enter a descriptive name, for example “Nuance OSA”.• Gatekeeper Address: Enter the IP address of the C-LAN.• Gatekeeper Port: Retain the default value.• Media Encryption: Select the “No” radio button. This setting should match with the media encryption settings on the IP Codec Set screen from Section 3.3.• From: Enter the first phantom IP station extension “77101” administered in Section 3.5.• To: Enter the last phantom IP station extension “77104” administered in Section 3.5.• Password: Enter the same value administered in the Security Code field in Section 3.5.• Use same password for all: Select the radio button.• Click Add. <p>Note: If the phantom IP station extensions are not configured in a consecutive range, then add the extensions individually.</p> <p>Scroll down to the bottom of the screen, and click Save.</p> <div><p>You are here: Home > System Configuration > H.323 Connections > Add H.323 Connection</p><h3>Add H.323 Connection</h3><p>Use this page to add a new H.323 connection.</p><p>Name: <input type="text" value="Nuance OSA"/></p><p>Gatekeeper Address: <input type="text" value="192.45.100.70"/></p><p>Gatekeeper Port: <input type="text" value="1719"/></p><p>Media Encryption: <input type="radio"/> Yes <input checked="" type="radio"/> No</p><h4>New Phone Numbers</h4><table><thead><tr><th></th><th>From</th><th>To</th></tr></thead><tbody><tr><td>Phone Number:</td><td><input type="text" value="77101"/></td><td><input type="text" value="77104"/></td></tr><tr><td>Password:</td><td colspan="2"><input type="text" value="•••••"/></td></tr><tr><td></td><td colspan="2"><input checked="" type="radio"/> Use same password for all <input type="radio"/> Use sequential passwords</td></tr><tr><td>Maintenance:</td><td colspan="2"><input type="radio"/> Yes <input checked="" type="radio"/> No</td></tr></tbody></table><p style="text-align: right;">Add</p><h4>Configured Phone Numbers (M for Maintenance)</h4><div><div><No phone number></div><div>Remove</div></div><p>Save Cancel Help</p></div>		From	To	Phone Number:	<input type="text" value="77101"/>	<input type="text" value="77104"/>	Password:	<input type="text" value="•••••"/>			<input checked="" type="radio"/> Use same password for all <input type="radio"/> Use sequential passwords		Maintenance:	<input type="radio"/> Yes <input checked="" type="radio"/> No	
	From	To														
Phone Number:	<input type="text" value="77101"/>	<input type="text" value="77104"/>														
Password:	<input type="text" value="•••••"/>															
	<input checked="" type="radio"/> Use same password for all <input type="radio"/> Use sequential passwords															
Maintenance:	<input type="radio"/> Yes <input checked="" type="radio"/> No															

4.2. Administer MPP Server

Step	Description
1.	<p>In the left pane of the Voice Portal Management System, navigate to System Configuration → MPP Servers. The MPP Servers screen appears in the right pane. Click Add.</p> 
2.	<p>The Add MPP Server screen appears. Enter the following values:</p> <ul style="list-style-type: none"> • Name: Enter a descriptive name, for example “OSA MPP”. • Host Address: Enter the IP address of MPP server. <p>After entering the values, click Continue.</p> 

Step

Description

3.

Next, the **Add MPP Server** screen appears. For the **Maximum Simultaneous Calls** field, enter a number from the range of 1 to 128. In the compliance testing, the value “4” was used because there were four phantom IP stations that were administered in Section 3.5. In this case, the Avaya Voice Portal allows maximum of four simultaneous calls to Avaya Communication Manager. Check the **Trust this certificate** checkbox and retain the default values for all the remaining fields. Scroll down to the bottom of the screen, click **Save**.

You are here: [Home](#) > System Configuration > [MPP Servers](#) > Add MPP Server

Add MPP Server

Use this page to change the configuration of an MPP. Take care when changing the MPP Trace Logging Thresholds. Do not set Trace Levels to Finest if your Voice Portal system has heavy call traffic. The system might experience performance issues if Trace Levels are set to Finest. Set Trace Levels to Finest only when you are troubleshooting the system.

Name:

OSA MPP

Host Address:

192.45.120.243

Network Address (VoIP):

192.45.120.243

Network Address (MRCP):

192.45.120.243

Maximum Simultaneous Calls:

4

MPP Certificate

The following certificate was sent by the MPP for verification. The displayed certificate should be identical to the certificate established during the installation of the target MPP. Acceptance of the certificate will allow the MPP access to privileged services on the VPMS. If the certificate does not match, ensure that the host address has been entered correctly.

Owner: CN=mpp21,O=Avaya,OU=MPP

Issuer: CN=mpp21,O=Avaya,OU=MPP

Serial Number: 0

Valid from: Tue Jan 23 10:09:07 EST 2007 until: Fri Jan 20 10:09:07 EST 2017

Certificate fingerprints

MD5: d7:c4:26:2e:a5:80:9b:97:3a:f0:1c:cc:13:09:5c:ef

SHA: bc:74:93:d8:3a:c0:ca:52:d1:8a:60:6f:9e:b9:ed:7a:f5:df:57:17

☒ Trust this certificate


Categories and Trace Levels


☒ Use Global Settings

☐ Custom

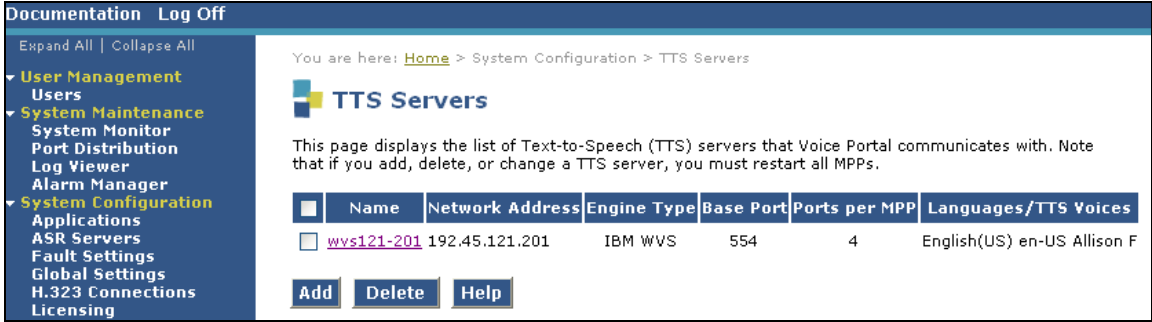
	Off	Fine	Finer	Finest
ASR	<input type="radio"/>	<input type="radio"/>	<input checked="" type="radio"/>	<input type="radio"/>
Call Data Handler	<input type="radio"/>	<input type="radio"/>	<input checked="" type="radio"/>	<input type="radio"/>
CCXML Browser	<input type="radio"/>	<input type="radio"/>	<input checked="" type="radio"/>	<input type="radio"/>
MPP System Manager	<input type="radio"/>	<input type="radio"/>	<input checked="" type="radio"/>	<input type="radio"/>
MRCP	<input type="radio"/>	<input type="radio"/>	<input checked="" type="radio"/>	<input type="radio"/>


4.3. Administer ASR Server

Step	Description														
1.	<p>In the left pane of the Voice Portal Management System, navigate to System Configuration → ASR Servers. The ASR Server screen appears in the right pane. Click Add.</p>  <p>The screenshot shows the 'ASR Servers' page in the Voice Portal Management System. The left navigation pane has 'System Configuration' expanded, and 'ASR Servers' is selected. The main content area displays the 'ASR Servers' page with a table of servers and an 'Add' button highlighted with a red box.</p> <table><tr><th></th><th>Name</th><th>Network Address</th><th>Engine Type</th><th>Base Port</th><th>Ports per MPP</th><th>Languages</th></tr><tr><td><input type="checkbox"/></td><td>wys121-201</td><td>192.45.121.201</td><td>IBM WVS</td><td>554</td><td>4</td><td>English(US) en-US</td></tr></table> <p>Buttons: Add, Delete, Help</p>		Name	Network Address	Engine Type	Base Port	Ports per MPP	Languages	<input type="checkbox"/>	wys121-201	192.45.121.201	IBM WVS	554	4	English(US) en-US
	Name	Network Address	Engine Type	Base Port	Ports per MPP	Languages									
<input type="checkbox"/>	wys121-201	192.45.121.201	IBM WVS	554	4	English(US) en-US									

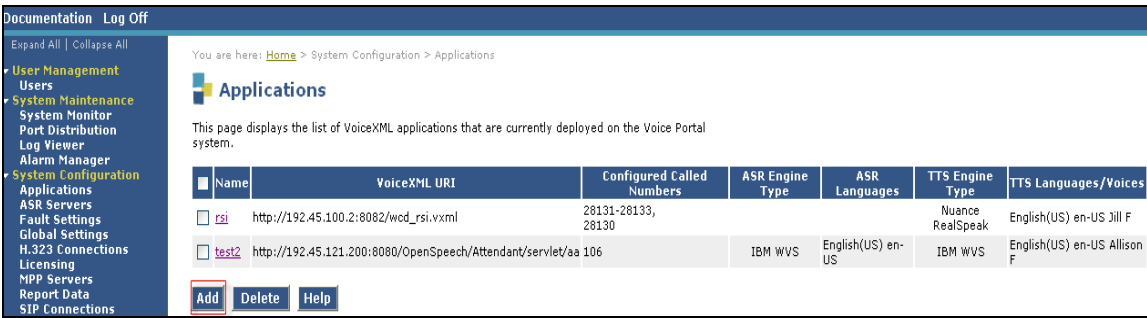
Step	Description
2.	<p>The Add ASR Server screen appears. Enter the following values and retain the default values for the remaining fields.</p> <ul style="list-style-type: none"> • Name: Enter a descriptive name, for example “OSA ASR”. • Engine Type: Select “Nuance OSR” from the drop down list. • Network Address: Enter the IP address of the Nuance OpenSpeech Recognizer server. • Ports per MPP: Enter the maximum number of simultaneous port connections the MPP can make to the Nuance OpenSpeech Recognizer. The valid numbers are 1 to 1000. <p>Click Save.</p> <div data-bbox="289 741 1433 1556"> <p>You are here: Home > System Configuration > ASR Servers > Add ASR Server</p> <h3> Add ASR Server</h3> <p>Use this page to configure Voice Portal to communicate with a new ASR server. Note that after adding an ASR server, you must restart all MPPs.</p> <p>Name: <input type="text" value="OSA ASR"/></p> <p>Engine Type: <input type="text" value="Nuance OSR"/></p> <p>Network Address: <input type="text" value="192.45.121.202"/></p> <p>Base Port: <input type="text" value="4900"/></p> <p>Ports per MPP: <input type="text" value="4"/></p> <p>MRCP Ping Interval: <input type="text" value="15"/> second(s)</p> <p>MRCP Response Timeout: <input type="text" value="4"/> second(s)</p> <p>New Connection per Session: <input type="radio"/> Yes <input checked="" type="radio"/> No</p> <p>RTSP URL: <input type="text" value="192.45.121.202/media/speechrecognize"/></p> <p>Languages: <div> <div>English(USA) en-us</div> <div>Cantonese(Hong_Kong) cn-HK</div> <div>Catalan(Spain) ca-ES</div> <div>Czech(Czech_Republic) cs-CZ</div> <div>Danish(Denmark) da-DK</div> <div>Dutch(Netherlands) nl-be</div> </div></p> <p><input type="button" value="Save"/> <input type="button" value="Cancel"/> <input type="button" value="Help"/></p> </div>

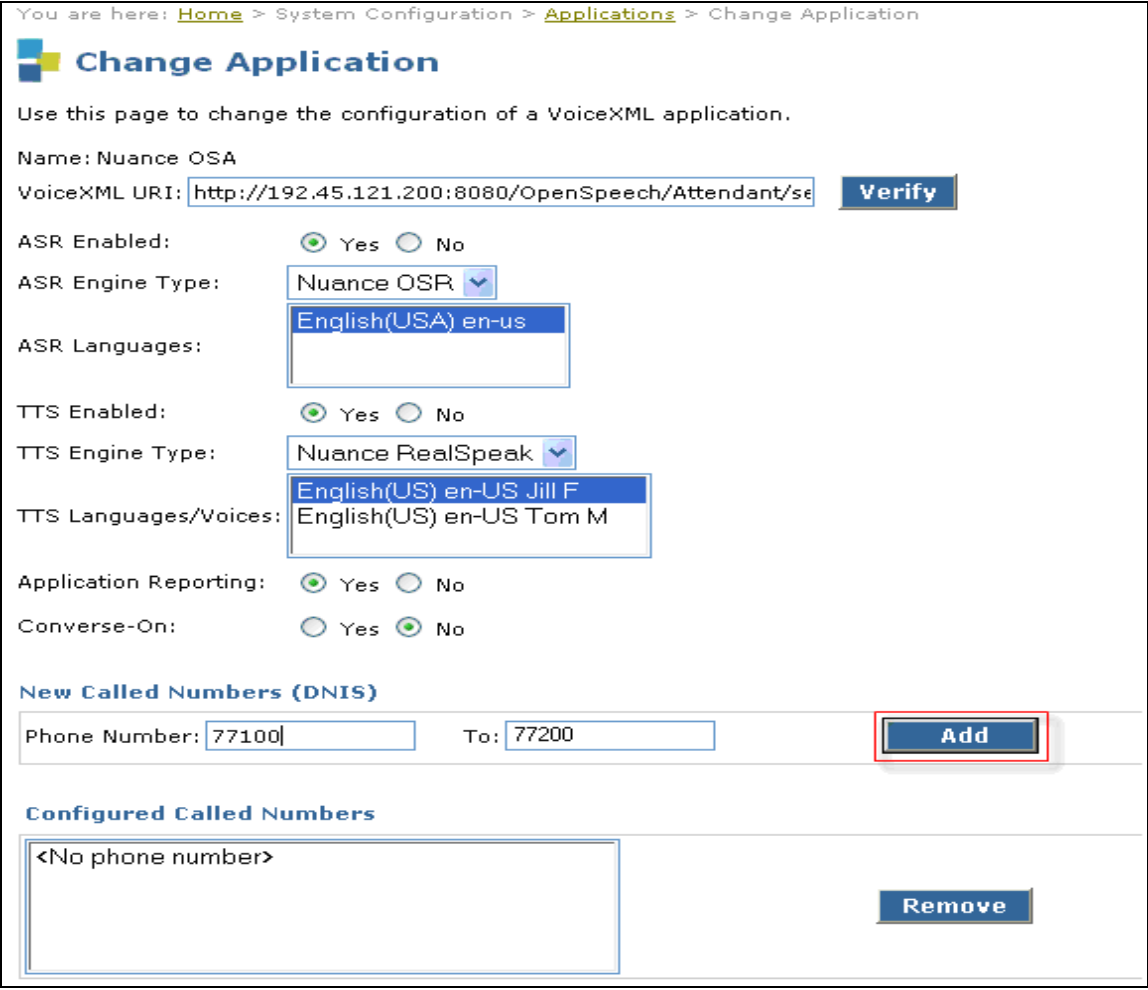
4.4. Administer TTS Server

Step	Description
1.	<p>In the left pane of the Voice Portal Management System, navigate to System Configuration → TTS Servers. The TTS Servers screen appears in the right pane. Click Add.</p> 


Step	Description
2.	<p>Next, the Add TTS Server screen appears. Enter the following values and retain the default values for the remaining fields.</p> <ul style="list-style-type: none"> • Name: Enter a descriptive name, for example “OSA TTS”. • Engine Type: Select “Nuance RealSpeak” from the drop down list. • Network Address: Enter the IP address of the Nuance RealSpeak server. • Ports per MPP: Enter the maximum number of simultaneous port connections the MPP can make to the Nuance RealSpeak. The valid numbers are 1 to 1000. • Languages/TTS Voices: Select the appropriate Languages/TTS Voices as installed on Nuance RealSpeak. <p>Click Save.</p> <div data-bbox="289 741 1433 1619"> <p>You are here: Home > System Configuration > TTS Servers > Add TTS Server</p> <h3> Add TTS Server</h3> <p>Use this page to configure Voice Portal to communicate with a new TTS server. Note that after adding a TTS server, you must restart all MPPs.</p> <p>Name: <input type="text" value="OSA TTS"/></p> <p>Engine Type: <input type="text" value="Nuance RealSpeak"/></p> <p>Network Address: <input type="text" value="192.45.121.202"/></p> <p>Base Port: <input type="text" value="4900"/></p> <p>Ports per MPP: <input type="text" value="4"/></p> <p>MRCP Ping Interval: <input type="text" value="15"/> second(s)</p> <p>MRCP Response Timeout: <input type="text" value="4"/> second(s)</p> <p>New Connection per Session: <input type="radio"/> Yes <input checked="" type="radio"/> No</p> <p>RTSP URL: <input type="text" value="192.45.121.202/media/speechsynthesiz"/></p> <p>Languages/TTS Voices: <div> <div>English(US) en-US Jennifer F</div> <div>English(US) en-US Jill F</div> <div>English(US) en-US Tom M</div> <div>Basque eu-ES Arantxa F</div> <div>Chinese(Cantonese) zh-HK Sin-Ji F</div> <div>Chinese(Mandarin) zh-CN Mei-Ling F</div> </div></p> <p><input type="button" value="Save"/> <input type="button" value="Cancel"/> <input type="button" value="Help"/></p> </div>

4.5. Administer Speech Application

Step	Description																					
1.	<p>In the left pane of Voice Portal Management System, navigate to System Configuration → Applications. The Applications screen appears in the right pane. Click Add.</p>  <p>The screenshot shows the 'Applications' screen in the Voice Portal Management System. The left sidebar contains a navigation menu with the following items: Documentation, Log Off, Expand All Collapse All, User Management, Users, System Maintenance, System Monitor, Port Distribution, Log Viewer, Alarm Manager, System Configuration (highlighted), Applications (selected), ASR Servers, Fault Settings, Global Settings, H.323 Connections, Licensing, MPP Servers, Report Data, and SIP Connections. The main content area displays the 'Applications' screen with a breadcrumb trail: You are here: Home > System Configuration > Applications. Below the breadcrumb is a heading 'Applications' and a description: 'This page displays the list of VoiceXML applications that are currently deployed on the Voice Portal system.' A table lists the applications:</p> <table><tr><th>Name</th><th>VoiceXML URI</th><th>Configured Called Numbers</th><th>ASR Engine Type</th><th>ASR Languages</th><th>TTS Engine Type</th><th>TTS Languages/Voices</th></tr><tr><td><input type="checkbox"/> rsi</td><td>http://192.45.100.2:8082/wcd_rsi.vxml</td><td>28131-28133, 28130</td><td></td><td></td><td>Nuance RealSpeak</td><td>English(US) en-US Jill F</td></tr><tr><td><input type="checkbox"/> test2</td><td>http://192.45.121.200:8080/OpenSpeech/Attendant/servlet/aa 106</td><td></td><td>IBM WVS</td><td>English(US) en-US</td><td>IBM WVS</td><td>English(US) en-US Allison F</td></tr></table> <p>At the bottom of the table are three buttons: Add (highlighted with a red box), Delete, and Help.</p>	Name	VoiceXML URI	Configured Called Numbers	ASR Engine Type	ASR Languages	TTS Engine Type	TTS Languages/Voices	<input type="checkbox"/> rsi	http://192.45.100.2:8082/wcd_rsi.vxml	28131-28133, 28130			Nuance RealSpeak	English(US) en-US Jill F	<input type="checkbox"/> test2	http://192.45.121.200:8080/OpenSpeech/Attendant/servlet/aa 106		IBM WVS	English(US) en-US	IBM WVS	English(US) en-US Allison F
Name	VoiceXML URI	Configured Called Numbers	ASR Engine Type	ASR Languages	TTS Engine Type	TTS Languages/Voices																
<input type="checkbox"/> rsi	http://192.45.100.2:8082/wcd_rsi.vxml	28131-28133, 28130			Nuance RealSpeak	English(US) en-US Jill F																
<input type="checkbox"/> test2	http://192.45.121.200:8080/OpenSpeech/Attendant/servlet/aa 106		IBM WVS	English(US) en-US	IBM WVS	English(US) en-US Allison F																

Step	Description
2.	<p>The Add Application screen appears. Enter the following values and retain the default values for the remaining fields.</p> <ul style="list-style-type: none"> • Name: Enter a descriptive name, for example “Nuance OSA”. • VoiceXML URI: Enter the URI value “http://<IP address of Application server>:8080/<path to speech application folder>”. For example, the URI for compliance testing was http://192.45.121.200:8080/OpenSpeech/Attendant/servlet/aa. • Phone Number: Enter the pilot number “77100” administered in Section 3.6. • To: Enter the pilot number “77200” administered in Section 3.6. Click Add. <p>Scroll down to the bottom of the screen, and click Save.</p> 

4.6. Administer VoIP Settings

Step	Description												
1.	<p>In the left pane of the Voice Portal Management System, navigate to System Configuration → VoIP Settings. The VoIP Settings screen appears in the right pane. Select “audio/basic” from the MPP Native Format drop down list. This setting is equivalent to the audio codec setting “G711MU” on Avaya Communication Manager, administered in Section 3.3 Step 1. Retain the default values for the remaining fields. Click Save.</p> <div><p>You are here: Home > System Configuration > VoIP Settings</p><div> VoIP Settings</div><p>Voice over Internet Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols such as H.323 and Real-time Transfer Protocol (RTP). Use this page to configure parameters that affect how voice data is transferred through the network. Note that if you make any changes to this page, you must restart all MPPs.</p><div>Port Ranges<table><tr><th></th><th>Low</th><th>High</th></tr><tr><td>UDP:</td><td><input type="text" value="30000"/></td><td><input type="text" value="30999"/></td></tr><tr><td>TCP:</td><td><input type="text" value="31000"/></td><td><input type="text" value="31999"/></td></tr><tr><td>MRCP:</td><td><input type="text" value="32000"/></td><td><input type="text" value="32999"/></td></tr></table></div><div>RTCP Monitor Settings<div><div>Host Address:</div><input type="text"/></div><div><div>Port:</div><input type="text"/></div></div><div>VoIP Audio Formats<div>MPP Native Format: <input type="text" value="audio/basic"/> ▼</div></div><div><div>Save</div><div>Cancel</div><div>Help</div></div></div>		Low	High	UDP:	<input type="text" value="30000"/>	<input type="text" value="30999"/>	TCP:	<input type="text" value="31000"/>	<input type="text" value="31999"/>	MRCP:	<input type="text" value="32000"/>	<input type="text" value="32999"/>
	Low	High											
UDP:	<input type="text" value="30000"/>	<input type="text" value="30999"/>											
TCP:	<input type="text" value="31000"/>	<input type="text" value="31999"/>											
MRCP:	<input type="text" value="32000"/>	<input type="text" value="32999"/>											

5. Configure Nuance OSA

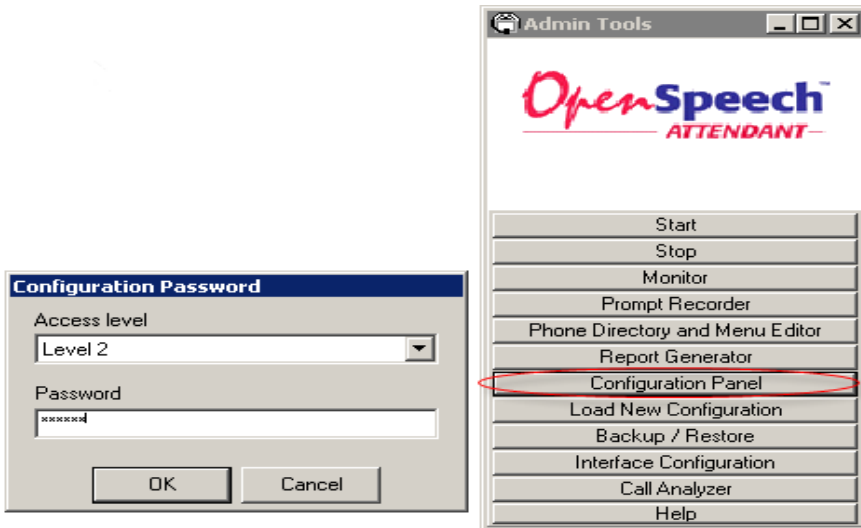
This section provides the procedures for configuring Nuance OSA. The procedure includes the following areas:

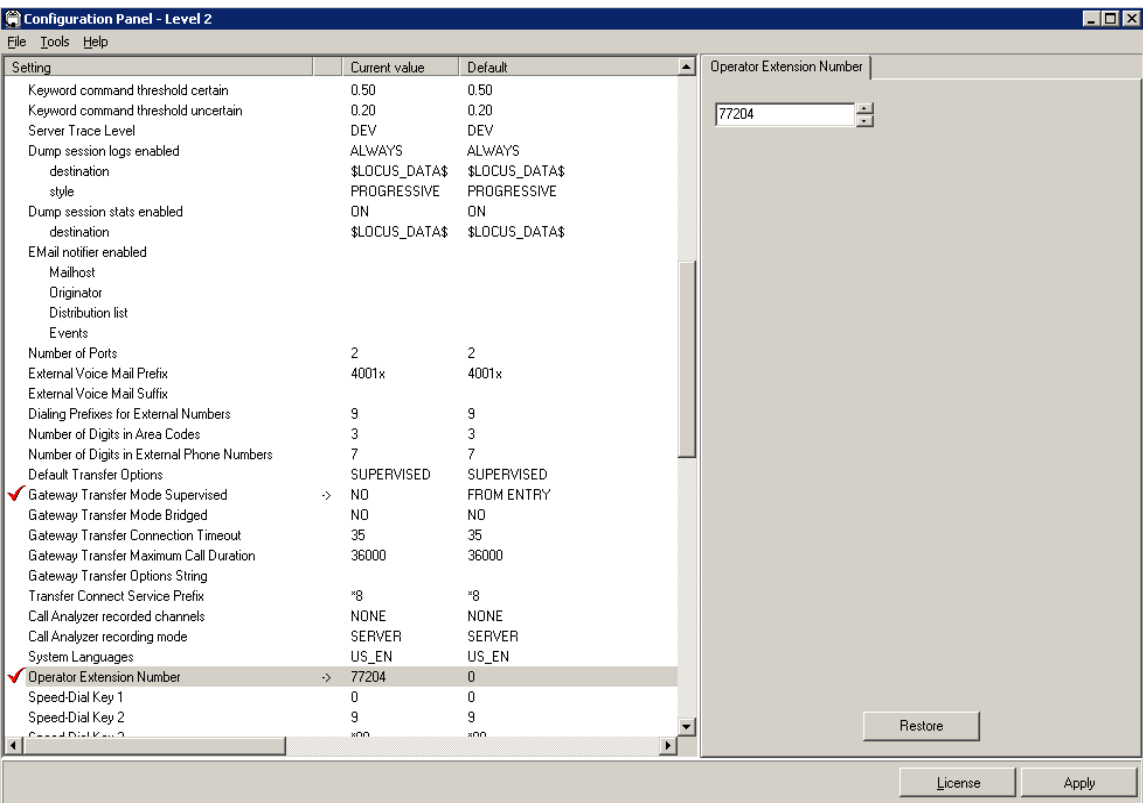
- Administer settings in configuration panel
- Administer phone directory and menu editor
- Activate changes
- Administer new entry point

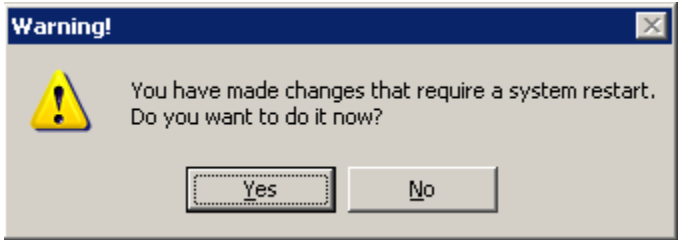

The Nuance OSA consisted of Nuance OpenSpeech Recognizer for speech recognition and Nuance RealSpeak to convert text to synthesized speech. For the compliance testing, it is assumed that the Nuance OpenSpeech Recognizer and the Nuance RealSpeak are installed and have appropriate licenses. Configuration of Nuance OSA is typically performed by Nuance technicians. The procedural steps are presented in these Application Notes for informational purposes.

5.1. Administer Settings in Configuration Panel

There are three different access levels to the Configuration Panel. The Level 2 and Level 3 access privileges are similar, which allows access to all the settings, including the gateway dependent options. The Level 1 has access only to the customer-related settings.

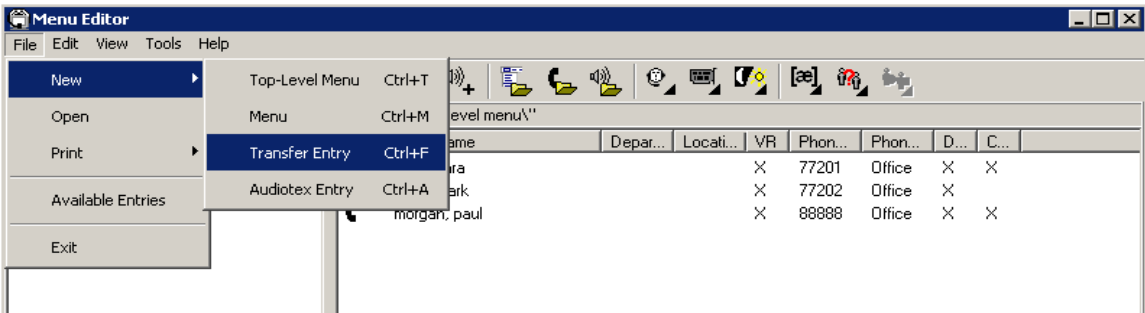
Step	Description
1.	<p>To administer the OSA application, launch Admin Tools window by navigating to All Programs → OSA → Admin Tools. Click Configuration Panel. The Configuration Password window appears. Select “Level 2” from the Access level dropdown list and enter the password provided by Nuance distributor in the Password field.</p>  <p>The screenshot shows two windows. On the left is the 'Configuration Password' dialog box with a title bar. It contains a dropdown menu for 'Access level' set to 'Level 2', a 'Password' field with masked characters, and 'OK' and 'Cancel' buttons. On the right is the 'Admin Tools' window with a title bar and the 'OpenSpeech ATTENDANT' logo. It contains a list of buttons: 'Start', 'Stop', 'Monitor', 'Prompt Recorder', 'Phone Directory and Menu Editor', 'Report Generator', 'Configuration Panel' (which is circled in red), 'Load New Configuration', 'Backup / Restore', 'Interface Configuration', 'Call Analyzer', and 'Help'.</p>

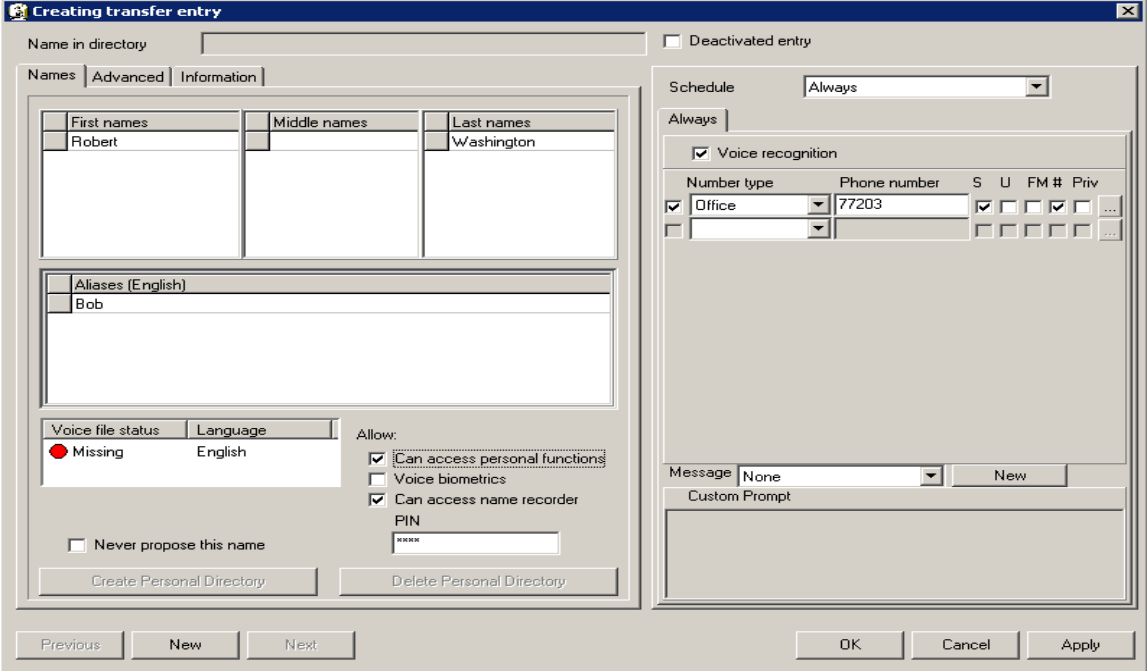
Step	Description
2.	<p>The Configuration Panel – Level 2 window appears. When a field is selected in the left pane, the corresponding field value appears in the right pane. Modify values for the following fields:</p> <ul style="list-style-type: none"> • Gateway Transfer Mode Supervised: Select “No” from the drop down list. • Operator Extension Number: Enter a valid physical IP station extension that is assigned to an operator telephone. In the compliance testing, the physical IP station extension “77204” was used as an operator extension. <p>Click Apply, and close the Configuration Panel – Level 2 window.</p>  <p>The screenshot shows the 'Configuration Panel - Level 2' window. The left pane lists various settings. The 'Gateway Transfer Mode Supervised' setting is selected, and its 'Current value' is 'NO'. The 'Operator Extension Number' setting is also selected, and its 'Current value' is '77204'. The right pane shows the 'Operator Extension Number' field with the value '77204'. The 'Apply' button is visible at the bottom right.</p>

Step	Description
3.	<p>The OSA application displays a Warning dialog box as shown below. Click Yes.</p> 
4.	<p>Click OK, on the Starting OpenSpeech Attendant dialog box.</p> 

5.2. Configure Phone Directory and Menu Editor

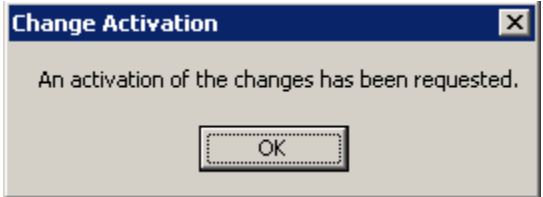
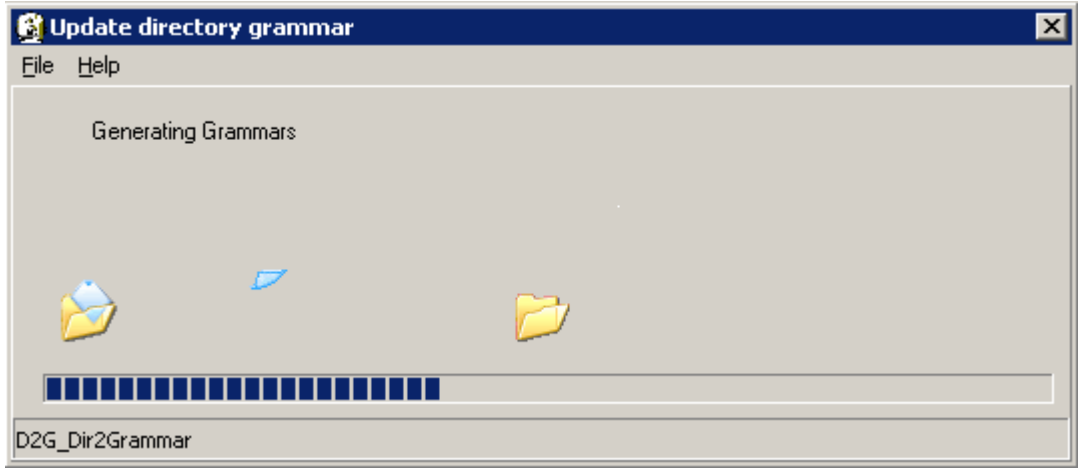
This section describes how to create transfer entries for the phone directory. The transfer entries need to be created for each employee or service to which the Nuance OSA can transfer calls. A single transfer entry can contain several different extensions or phone numbers such as office, cell phone and pager. However, in the compliance testing, the entries included only the office extensions.

Step	Description
1.	<p>In the Admin Tools window shown in Section 5.1, click Phone Directory and Menu Editor. The Menu Editor window appears. From the menu bar of the Menu Editor window, navigate to File → New → Transfer Entry.</p> 

Step	Description
2.	<p>Next, the Creating transfer entry window appears. Change the values for the following fields and retain the default values for the remaining fields.</p> <ul style="list-style-type: none"> In the left pane, click the Names tab: <ul style="list-style-type: none"> First names: Enter the first name of the employee or the name of the service. Last names: Enter the last name of the employee or the name of the service. Aliases: Enter alias in this field. This is an optional field. Allow: Check Can access personal functions and Can access name recorder check boxes. These options allow employee to change recording and PIN using the Personal Administration Mode (PAM). PIN: Enter desired PIN. This PIN is required when using PAM. In the right pane, under the Always tab: <ul style="list-style-type: none"> Number type: Select “Office” from the drop down list. Phone number: Enter a valid extension assigned to the employee’s telephone. In the compliance testing, the physical IP station extension “77203” was used as one of the employee’s telephone extension. <p>Click Apply then, Click OK.</p> 
3.	Repeat Step 1 - 2 as necessary to add additional transfer entries.

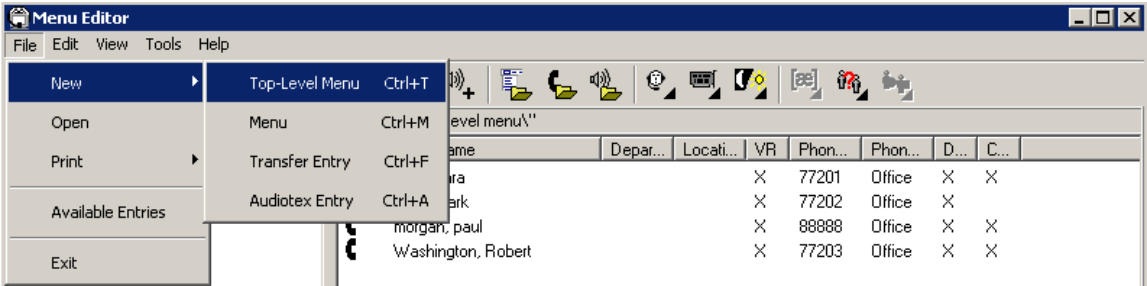
5.3. Activate Changes

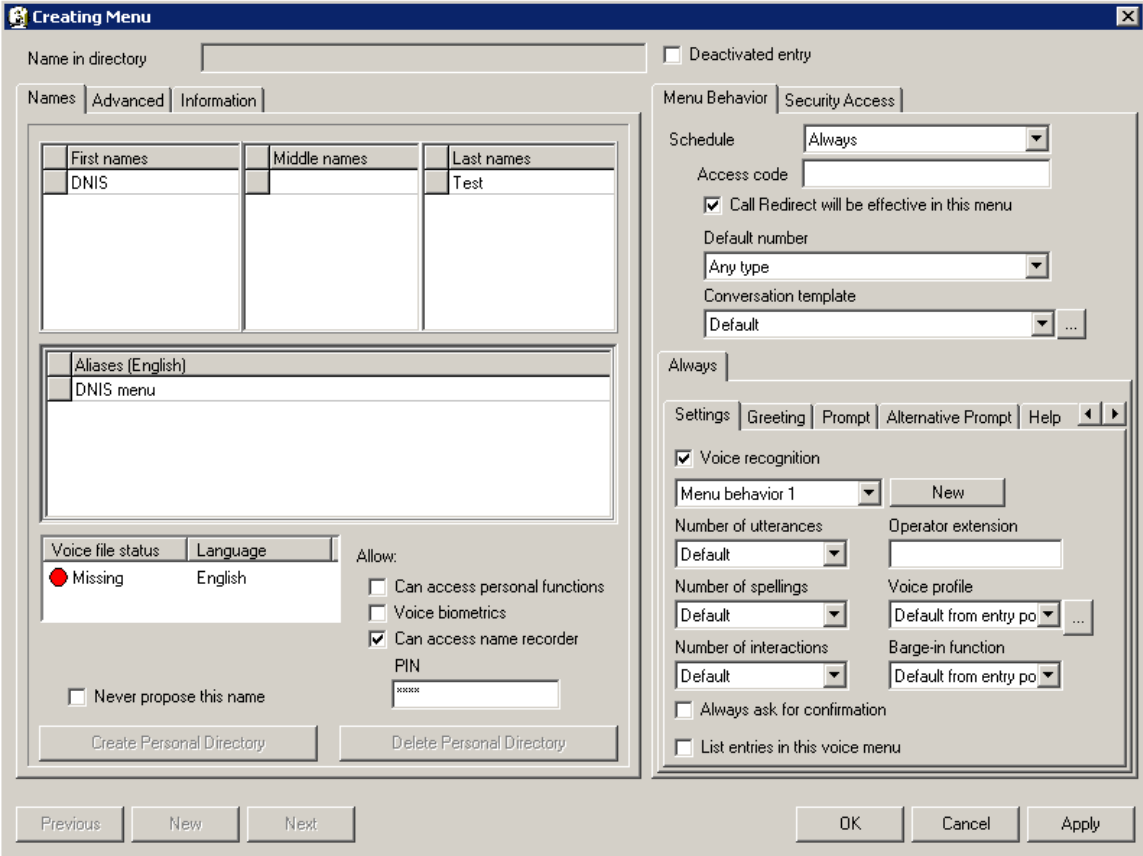
The following steps apply the changes to the phone directory. Activate the changes at the end of the Phone Directory and Menu Editor configuration. Otherwise, new or updated directory entries are not available to the callers.

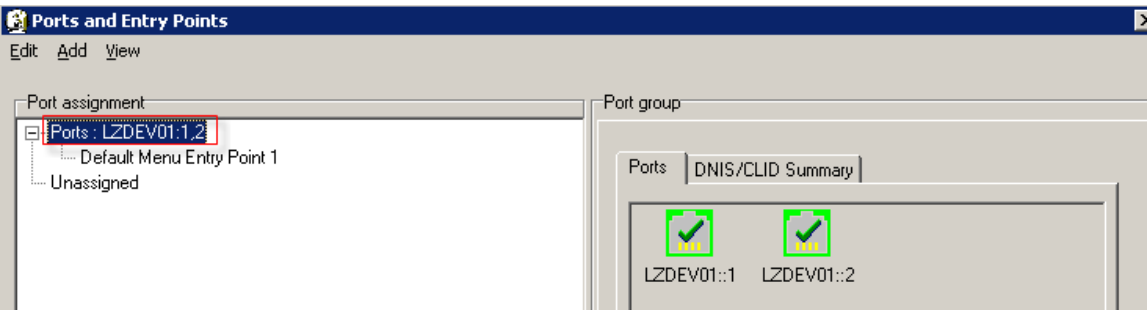
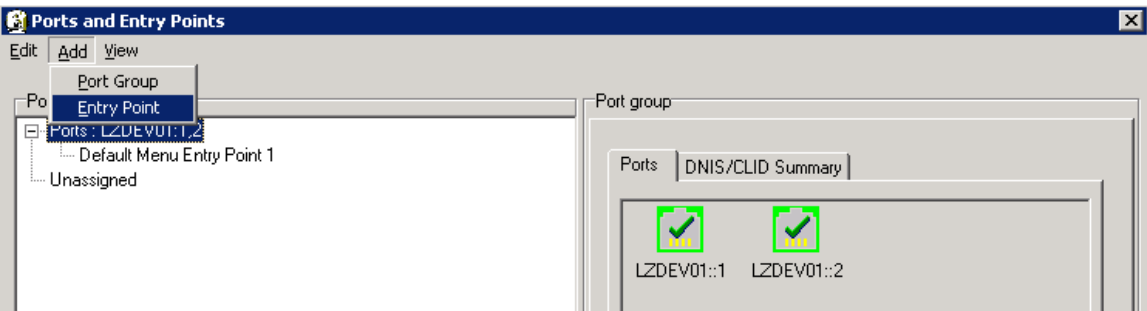
Step	Description
1.	<p>From the menu bar of the Menu Editor window shown in Section 5.2, navigate to Tools → Activate Changes. Click OK, when the Change Activation dialog appears.</p>  <p>The system generates updated grammars and displays progress in the Updated directory grammar dialog box shown below.</p> 

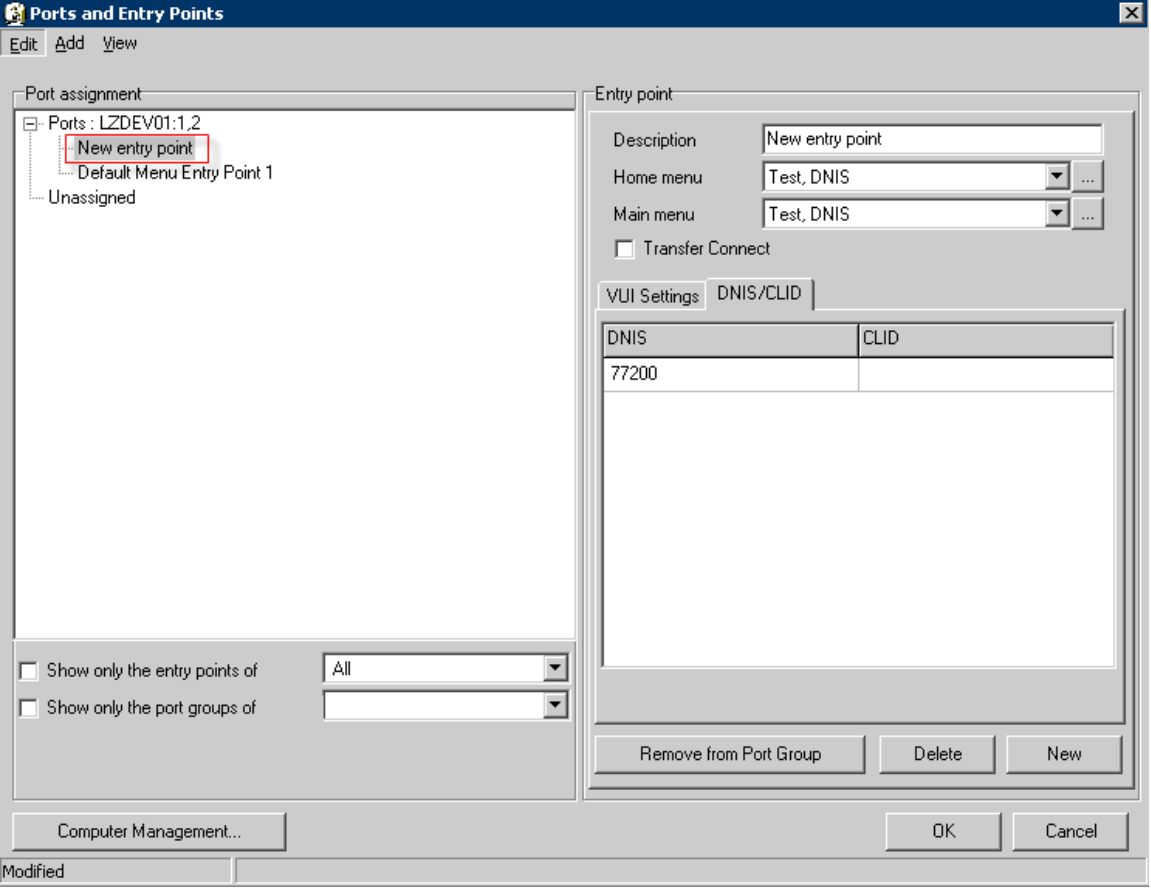
5.4. Configure New Entry Point

The Nuance OSA allows configuration of multiple entry points. The use of multiple entry point enables individual service or department to be assigned to a specific entry point. The entry point can be configured using DNIS or CLID number. When a caller calls, the Nuance OSA can route the call to the appropriate entry point based on dialed number (DNIS) or the caller's phone number (CLID). In the compliance testing, an entry point was configured using the DNIS option.

Step	Description
1.	<p>From the menu bar of the Menu Editor window, navigate to File → New → Top-Level Menu.</p> 

Step	Description
2.	<p>Next, the Creating Menu window appears. Enter the following values and retain the default values for the remaining fields.</p> <ul style="list-style-type: none"> In the left pane, click the Names tab: <ul style="list-style-type: none"> First names: Enter a descriptive first name for the entry point. Last names: Enter a descriptive last name for the entry point. Aliases: Enter a descriptive alias in this field. This is an optional field. Allow: Check the Can access name recorder check box. This option allows changing the name recording for this entry point using the PAM feature. PIN: Enter desired PIN. This PIN is required when using PAM feature. <p>Click Apply, and then click OK.</p> 

Step	Description
3.	Follow the steps in Section 5.3 to activate these changes.
4.	<p>Next, from the menu bar of the Menu Editor window, navigate to Edit → Ports and Entry Points. The Ports and Entry Points window appears. In the Port assignment, select Ports: <n>:<number of ethernet ports>, where <n> is the name of the Nuance OSA server.</p> 
5.	<p>Then, from the menu bar of the Ports and Entry Points window, navigate to Add → Entry Point.</p> 

Step	Description
6.	<p>In the Ports assignment section in the left pane, the New entry point appears. Click on the New entry point. In the Entry point section in the right pane, enter the following values and retain the default values for the remaining fields.</p> <ul style="list-style-type: none"> • Description: Enter a short description for the new entry point. • Home Menu: Select the new entry point created in Step 2 of this section from the Home menu drop down list. • Main Menu: Select the new entry point created in Step 2 of this section from the Main menu drop down list. <p>Select the DNIS/CLID tab and enter the pilot number from Section 3.6 in the DNIS field, for example “77200”. Click OK.</p> 

6. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. Feature testing focused on verifying that the Nuance OSA can successfully recognize names or entered extensions and transfer the call to the correct destination. Serviceability testing verified that Nuance OSA recovered from adverse conditions, such as rebooting Nuance OSA server and disconnecting the Ethernet cable to the Nuance OSA server.

6.1. General Test Approach

The feature and serviceability test cases were performed manually to verify proper operation. The general test approach included:

- Establishing connectivity to Avaya Voice Portal, Avaya Communication Manager, Nuance OSA, Nuance OpenSpeech Recognizer and Nuance RealSpeak.
- Verify voice and DTMF touch-tone recognition from the Nuance OSA, and call transfer from Avaya Voice Portal.
- Verify when an entry point is configured using DNIS in the Nuance OSA, the call is routed to the correct entry point based on the dialed number.
- Testing also included rainy day scenario to verify situations in the absence of transfer entry in the phone directory.

6.2. Test Results

All feature and serviceability test cases were completed successfully.

7. Verification Steps










7.1. Avaya Communication Manager

The following steps can be used to verify the communication between Avaya Communication Manager and the Avaya Voice Portal.

Step	Description																																																																																																																																																									
1.	<p>Verify that the phantom IP stations used by the Avaya Voice Portal, administered in Section 3.5, are registered to Avaya Communication Manager. Use the “list registered-ip-stations” command from the Avaya SAT to verify that the phantom IP stations are registered in Net Rgn “1” and the Gatekeeper IP Address (C-LAN) is “192.45.100.70”. Also, verify that the Station IP Address is “192.45.120.243”, which is the IP address of the MPP server.</p>																																																																																																																																																									
	<pre>list registered-ip-stations</pre>																																																																																																																																																									
	<table><thead><tr><th colspan="9">REGISTERED IP STATIONS</th></tr><tr><th>Station Ext</th><th>Set Type</th><th>Product ID</th><th>Prod Rel</th><th>Station IP Address</th><th>Net Rgn</th><th>Orig Port</th><th>Gatekeeper IP Address</th><th>TCP Skt</th></tr></thead><tbody><tr><td>22710</td><td>4610</td><td>IP_Phone</td><td>2.200</td><td>192.45.30.224</td><td>1</td><td></td><td>192.45.100.70</td><td>y</td></tr><tr><td>24141</td><td>4620</td><td>IP_Phone</td><td>2.300</td><td>192.45.100.213</td><td>1</td><td></td><td>192.45.100.70</td><td>y</td></tr><tr><td>24142</td><td>4620</td><td>IP_Phone</td><td>2.300</td><td>192.45.100.223</td><td>2</td><td></td><td>192.45.145.20</td><td>y</td></tr><tr><td>24144</td><td>4621</td><td>IP_Phone</td><td>2.300</td><td>192.45.60.31</td><td>1</td><td></td><td>192.45.100.70</td><td>y</td></tr><tr><td>24145</td><td>4610</td><td>IP_Phone</td><td>2.300</td><td>192.45.60.32</td><td>1</td><td></td><td>192.45.100.70</td><td>y</td></tr><tr><td>26613</td><td>4624</td><td>IP_Phone</td><td>1.830</td><td>192.45.30.245</td><td>1</td><td></td><td>192.45.100.70</td><td>y</td></tr><tr><td>28131</td><td>7434ND</td><td>IP_API_A</td><td>3.300</td><td>192.45.120.243</td><td>1</td><td></td><td>192.45.100.70</td><td>y</td></tr><tr><td>28132</td><td>7434ND</td><td>IP_API_A</td><td>3.300</td><td>192.45.120.243</td><td>1</td><td></td><td>192.45.100.70</td><td>y</td></tr><tr><td>28133</td><td>7434ND</td><td>IP_API_A</td><td>3.300</td><td>192.45.120.243</td><td>1</td><td></td><td>192.45.100.70</td><td>y</td></tr><tr><td>28201</td><td>7434ND</td><td>IP_API_A</td><td>1.300</td><td>192.45.120.51</td><td>1</td><td></td><td>192.45.100.70</td><td>y</td></tr><tr><td>28202</td><td>7434ND</td><td>IP_API_A</td><td>1.330</td><td>192.45.121.122</td><td>1</td><td></td><td>192.45.100.70</td><td>y</td></tr><tr><td>77101</td><td>7434ND</td><td>IP_API_A</td><td>3.300</td><td>192.45.120.243</td><td>1</td><td></td><td>192.45.100.70</td><td>y</td></tr><tr><td>77102</td><td>7434ND</td><td>IP_API_A</td><td>3.300</td><td>192.45.120.243</td><td>1</td><td></td><td>192.45.100.70</td><td>y</td></tr><tr><td>77103</td><td>7434ND</td><td>IP_API_A</td><td>3.300</td><td>192.45.120.243</td><td>1</td><td></td><td>192.45.100.70</td><td>y</td></tr><tr><td>77104</td><td>7434ND</td><td>IP_API_A</td><td>3.300</td><td>192.45.120.243</td><td>1</td><td></td><td>192.45.100.70</td><td>y</td></tr></tbody></table>	REGISTERED IP STATIONS									Station Ext	Set Type	Product ID	Prod Rel	Station IP Address	Net Rgn	Orig Port	Gatekeeper IP Address	TCP Skt	22710	4610	IP_Phone	2.200	192.45.30.224	1		192.45.100.70	y	24141	4620	IP_Phone	2.300	192.45.100.213	1		192.45.100.70	y	24142	4620	IP_Phone	2.300	192.45.100.223	2		192.45.145.20	y	24144	4621	IP_Phone	2.300	192.45.60.31	1		192.45.100.70	y	24145	4610	IP_Phone	2.300	192.45.60.32	1		192.45.100.70	y	26613	4624	IP_Phone	1.830	192.45.30.245	1		192.45.100.70	y	28131	7434ND	IP_API_A	3.300	192.45.120.243	1		192.45.100.70	y	28132	7434ND	IP_API_A	3.300	192.45.120.243	1		192.45.100.70	y	28133	7434ND	IP_API_A	3.300	192.45.120.243	1		192.45.100.70	y	28201	7434ND	IP_API_A	1.300	192.45.120.51	1		192.45.100.70	y	28202	7434ND	IP_API_A	1.330	192.45.121.122	1		192.45.100.70	y	77101	7434ND	IP_API_A	3.300	192.45.120.243	1		192.45.100.70	y	77102	7434ND	IP_API_A	3.300	192.45.120.243	1		192.45.100.70	y	77103	7434ND	IP_API_A	3.300	192.45.120.243	1		192.45.100.70	y	77104	7434ND	IP_API_A	3.300	192.45.120.243	1		192.45.100.70	y
REGISTERED IP STATIONS																																																																																																																																																										
Station Ext	Set Type	Product ID	Prod Rel	Station IP Address	Net Rgn	Orig Port	Gatekeeper IP Address	TCP Skt																																																																																																																																																		
22710	4610	IP_Phone	2.200	192.45.30.224	1		192.45.100.70	y																																																																																																																																																		
24141	4620	IP_Phone	2.300	192.45.100.213	1		192.45.100.70	y																																																																																																																																																		
24142	4620	IP_Phone	2.300	192.45.100.223	2		192.45.145.20	y																																																																																																																																																		
24144	4621	IP_Phone	2.300	192.45.60.31	1		192.45.100.70	y																																																																																																																																																		
24145	4610	IP_Phone	2.300	192.45.60.32	1		192.45.100.70	y																																																																																																																																																		
26613	4624	IP_Phone	1.830	192.45.30.245	1		192.45.100.70	y																																																																																																																																																		
28131	7434ND	IP_API_A	3.300	192.45.120.243	1		192.45.100.70	y																																																																																																																																																		
28132	7434ND	IP_API_A	3.300	192.45.120.243	1		192.45.100.70	y																																																																																																																																																		
28133	7434ND	IP_API_A	3.300	192.45.120.243	1		192.45.100.70	y																																																																																																																																																		
28201	7434ND	IP_API_A	1.300	192.45.120.51	1		192.45.100.70	y																																																																																																																																																		
28202	7434ND	IP_API_A	1.330	192.45.121.122	1		192.45.100.70	y																																																																																																																																																		
77101	7434ND	IP_API_A	3.300	192.45.120.243	1		192.45.100.70	y																																																																																																																																																		
77102	7434ND	IP_API_A	3.300	192.45.120.243	1		192.45.100.70	y																																																																																																																																																		
77103	7434ND	IP_API_A	3.300	192.45.120.243	1		192.45.100.70	y																																																																																																																																																		
77104	7434ND	IP_API_A	3.300	192.45.120.243	1		192.45.100.70	y																																																																																																																																																		

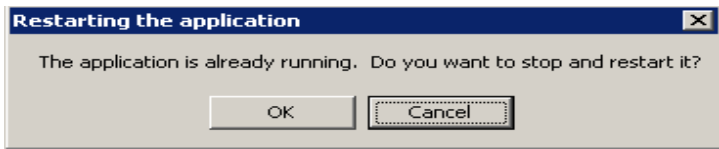
7.2. Avaya Voice Portal

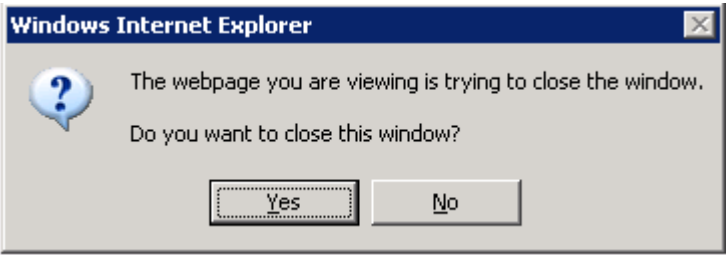
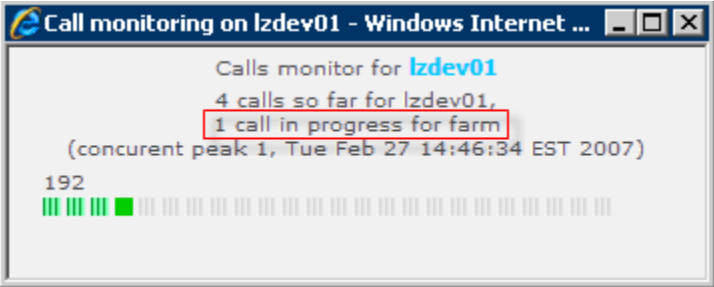
The following steps can be used to verify the communication between Avaya Voice Portal components, VPMS and MPP, and Avaya Communication Manager.

Step	Description																																				
1.	<p>From the VPMS, click on System Maintenance → System Monitor. Verify the status of the System Monitor as shown below:</p> <ul style="list-style-type: none">• Mode: “online”• State: “Running”• Config: “OK”• Restart: “No”• Call Capacity: The number of calls displayed is the same as the maximum simultaneous calls allowed on the MPP, the ASR and the TTS, administered in Section 4.2, Section 4.3, and Section 4.4 respectively.• Alarms: No alarms sign is displayed, which is indicated by the green check box. A different alarm sign is displayed, when there is a connection problem with Avaya Communication Manager. <table><tr><th>System Name</th><th>Mode</th><th>State</th><th>Config</th><th>Restart</th><th>Call Capacity</th><th>Active Calls</th><th>Calls Today</th><th>Alarms</th></tr><tr><td>VPMS</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr><tr><td>OSA MPP</td><td>Online</td><td>Running</td><td>OK</td><td>No</td><td>4 / 4 / 4</td><td>0</td><td>21</td><td></td></tr><tr><td>Overall Summary</td><td></td><td></td><td></td><td></td><td>4 / 4 / 4</td><td>0</td><td>21</td><td></td></tr></table> <div><div>Start All</div><div>Stop All</div><div>Restart All</div><div>Reboot All</div><div>Halt All</div></div> <div><div>Refresh</div><div>Help</div></div>	System Name	Mode	State	Config	Restart	Call Capacity	Active Calls	Calls Today	Alarms	VPMS									OSA MPP	Online	Running	OK	No	4 / 4 / 4	0	21		Overall Summary					4 / 4 / 4	0	21	
System Name	Mode	State	Config	Restart	Call Capacity	Active Calls	Calls Today	Alarms																													
VPMS																																					
OSA MPP	Online	Running	OK	No	4 / 4 / 4	0	21																														
Overall Summary					4 / 4 / 4	0	21																														

7.3. Nuance OSA

The following steps can be used to verify that the Nuance OSA is running and able to recognize active calls. The following steps also verify that the Nuance OpenSpeech Recognizer and the Nuance RealSpeak servers can communicate with Avaya Voice Portal and Nuance OSA.

Step	Description
1.	<p>From the Admin Tools window shown in Section 5.1, click Start. The Restarting the application dialog appears. If the Nuance OSA application is already running, click Cancel.</p> <div>A screenshot of a Windows-style dialog box titled "Restarting the application". The text inside says "The application is already running. Do you want to stop and restart it?". There are two buttons at the bottom: "OK" and "Cancel". The "Cancel" button is highlighted with a dashed border.</div>

Step	Description
2.	<p>From the Admin Tools window shown in Section 5.1, click Monitor. The OSA web page appears with a Windows Internet Explorer dialog box. Click Yes to close the window.</p>  <p>The Call monitoring window appears. Make a call to the pilot number administered in Section 3.6, and verify that the call in progress number is updated.</p> 
3.	Call the pilot number administered in Section 3.6, for example “77100”
4.	<p>Speak one of the names of a destination, for example “Robert Washington” or the alias “Bob”, which was administered in Section 5.2.</p> <p>Note: The DTMF touch-tone input can be used for this step. In this case, enter the extension associated with the transfer entry, which is “77203”.</p>
5.	Verify that the caller hears a confirmation tone.
6.	Verify that the call is transferred to the station 77203.

8. Support

Technical support on Nuance OSA can be obtained through the following:

- **Web:** www.network.nuance.com.
- **Phone:** From Montreal (514) 390-3922, or (866) 434-2564 from elsewhere.
- **Email:** SpeechAttendant.Support@nuance.com.

9. Conclusion

These Application Notes describe the configuration steps required for Nuance OSA 2.1 to interoperate with Avaya Voice Portal 3.0.1.2 using Nuance OpenSpeech Recognizer as the ASR application and Nuance RealSpeak as the TTS application with Avaya Communication Manager Release 3.1.2. All feature and serviceability test cases were completed successfully.

10. Additional References

- *Administrator Guide for Avaya Communication Manager*, Document 03-300509, Issue 2.1, May 2006, available at <http://support.avaya.com>.
- *Installing, Upgrading, and Configuring Avaya Voice Portal 3.0.1.1*, June 2006. Avaya Voice Portal product documentation is provided with the service pack image.
- *OSA Installation and Configuration Guide*, Version 2.1, January 2006. Nuance product documentations are installed with all versions of the product.
- *OSA Administration Guide*, Version 2.1, January 2006. Nuance product documentations are installed with all versions of the product.

©2007 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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 Developer*Connection* Program at devconnect@avaya.com.