



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Nuance OpenSpeech Attendant with Avaya Voice Portal – Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps required to integrate the Nuance OpenSpeech Attendant with Avaya Voice Portal and Avaya Aura™ Communication Manager. Nuance OpenSpeech Attendant allows callers to speak the name of a person, department, service, or location and be automatically transferred to the requested party without waiting to speak to an operator. In addition, the caller may dial an extension number to transfer to the requested party.

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 Nuance OpenSpeech Attendant with Avaya Voice Portal and Avaya Aura™ Communication Manager. Nuance OpenSpeech Attendant allows callers to speak the name of a person, department, service, or location and be automatically transferred to the requested party without waiting to speak to an operator. In addition, the caller may dial an extension number to transfer to the requested party.

## 1.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. Feature testing focused on Nuance OpenSpeech Attendant (OSA) successfully recognizing spoken names and extensions entered via DTMF and transfer the call to the correct destination. Blind, supervised, and bridged transfers were verified. Other features covered included barge-in / no barge-in, adding new transfer entries, recording caller utterances, and accessing Maintenance Mode and Personal Administration Mode to record name and change PIN.

Serviceability testing focused on verifying the ability of the Nuance OSA to recover from adverse conditions, such as server restarts, power failures, and disconnecting cables to the IP network.

**Note:** Nuance OSA was tested with Avaya Voice Portal using an H.323 and SIP integration. The Avaya Voice Portal SIP integration used Avaya Aura™ SIP Enablement Services 5.2.1. These Application Notes cover the configuration of Avaya Voice Portal using the H.323 integration.

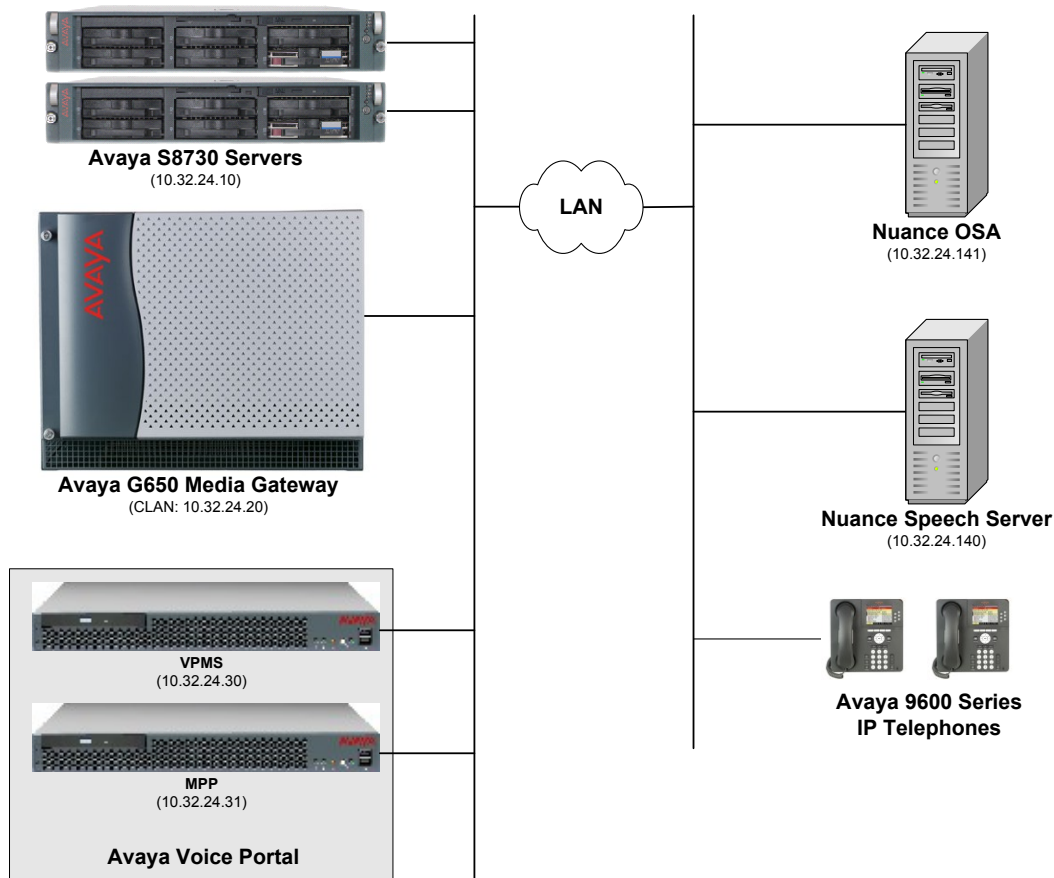
## 1.2. Support

To obtain technical support for Nuance OpenSpeech Attendant, contact Nuance via email or through their website.

- **Web:** [www.network.nuance.com](http://www.network.nuance.com)
- **Email:** [SpeechAttendant.Support@nuance.com](mailto:SpeechAttendant.Support@nuance.com)
- **Phone:** (866) 434-2564 or (514) 390-3922

## 2. Reference Configuration

**Figure 1** illustrates the configuration used to verify the Nuance OpenSpeech Attendant (OSA) solution with Avaya Voice Portal, Avaya Aura™ Communication Manager, and the Nuance Speech Server. Nuance OSA is deployed on a dedicated application server running Windows 2003 Server. Avaya Voice Portal interfaces to Avaya Aura™ Communication Manager using a VoIP H.323 interface. Avaya Voice Portal manages the interactions with speech server resources (i.e., speech recognition and text-to-speech) using VXML applications. VXML pages generated by Nuance OSA are loaded and interpreted by Avaya Voice Portal, which controls the interaction with the user. To access the Nuance OSA application, a VoIP channel on Avaya Voice Portal must be configured to invoke the VXML application when an incoming call is received on that channel.



**Figure 1: Configuration with Avaya Voice Portal and Nuance OpenSpeech Attendant**

## 2.1. Equipment and Software Validated

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

Equipment	Software
Avaya Voice Portal	5.1 (5.1.0.0.4201)
Avaya S8730 Servers with a G650 Media Gateway	Avaya Aura™ Communication Manager 5.2 (R015x.02.0.947.3) with Service Pack 3 (Patch 17579)
Avaya 9600 Series IP Telephones	3.0 (H.323)
Nuance OpenSpeech Attendant (OSA)	4.0
Nuance Speech Server <ul style="list-style-type: none"><li>• Nuance License Manager</li><li>• Nuance Speech Server</li><li>• Nuance Recognizer</li><li>• Nuance Vocalizer Network</li></ul>	11.1.0 5.1.1 9.0.12 5.0.0

### 3. Configure Avaya Aura™ Communication Manager

This section describes the configuration of H.323 stations and the IP codec set for Voice Portal. This configuration also requires a C-LAN and Media Processor board for IP communication. This configuration is outside the scope of these application notes, but the reader may refer to [1] and [2] for additional information.

**Note:** Special application SA8874 – Call Status Messages for 7434ND IP Softphone is required to support supervised transfers from Voice Portal.

From the System Access Terminal (SAT), add an H.323 station for Voice Portal. A call to this station will terminate to Voice Portal which will invoke the Nuance OSA application. In the station form, set the **Type** to *7434ND*, set the **Port** field to *IP*, provide a descriptive **Name**, set the **Security Code**, and set the **IP Softphone** field to *y*.

add station 23802		Page 1 of 6	
STATION			
Extension: 23802	Lock Messages? n	BCC: 0	
<b>Type: 7434ND</b>	<b>Security Code: XXXXX</b>	TN: 1	
<b>Port: IP</b>	Coverage Path 1:	COR: 1	
<b>Name: VP 192.45.122.50</b>	Coverage Path 2:	COS: 1	
	Hunt-to Station:		
STATION OPTIONS			
Time of Day Lock Table:			
Loss Group: 2	Personalized Ringing Pattern: 1		
Data Module? n	Message Lamp Ext: 23802		
Display Module? y	Coverage Module? n		
Display Language: english			
Survivable COR: internal	Media Complex Ext:		
Survivable Trunk Dest? y	<b>IP SoftPhone? y</b>		
	IP Video Softphone? n		

**Figure 2: Station Form**

In the IP codec set form associated with the IP network region of the H.323 station, configured in **Figure 2**, set the **Audio Codec** field to the appropriate value. In this configuration, *G.711MU* was used.

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: <b>G.711MU</b>	<b>n</b>	<b>2</b>	<b>20</b>
2:			

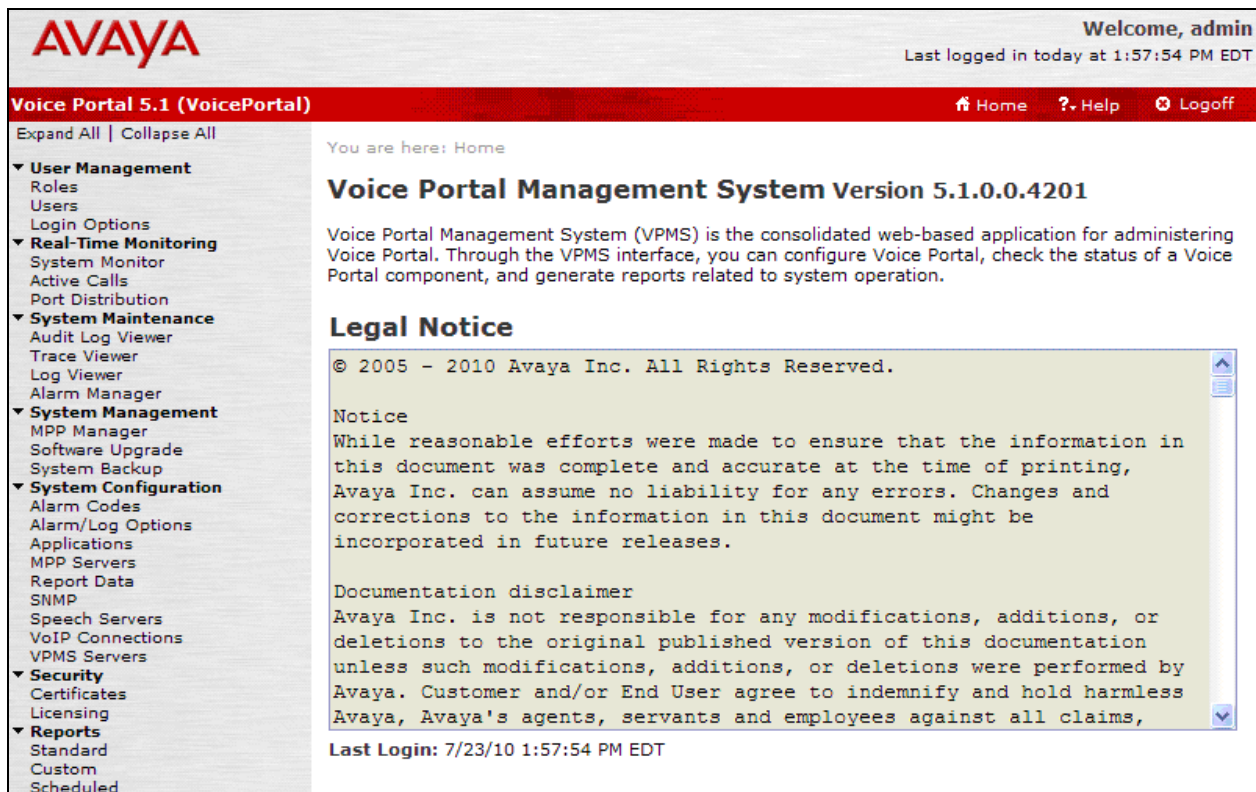
**Figure 3: IP Codec Set Form**

## 4. Configure Avaya Voice Portal

This section covers the administration of Voice Portal. The following Voice Portal configuration steps will be covered:

- Configuring an H.323 VoIP connection
- Adding an MPP server
- Configuring the VoIP audio format (mu-law or a-law)
- Adding a speech server
- Adding applications
- Starting the MPP server

Voice Portal is configured via the Voice Portal Management System (VPMS) web interface. To access the web interface, enter *http://<ip-addr>/VoicePortal* as the URL in an internet browser, where *<ip-addr>* is the IP address of the VPMS. Log in using the Administrator user role. The screen shown in **Figure 4** is displayed.



**Figure 4: VPMS Main Screen**

## 4.1. Configure the H.323 VoIP Connection

To configure an H.323 connection, navigate to the **VoIP Connections** page and then click on the **H.323** tab. Click on the **Add** button. In the **Add H.323 Connection** page, set the **Gatekeeper Address** to the IP address of the C-LAN in the G650 media gateway and the **Gatekeeper Port** to 1719. Next, configure the stations for Voice Portal, which map to the 7434ND stations configured in Communication Manager. In addition, set the **Password** for the stations and set the **Station Type** to *Inbound and Outbound*. Click the **Add** button when done. In this configuration, stations 23801 to 23808 are assigned to Voice Portal. However, only stations 23802 and 23803 are mapped to the Nuance OSA application on Voice Portal in Figure 10. Click **Save**.

The screenshot shows the Avaya Voicemail 5.1 (VoicePortal) configuration interface. The top navigation bar includes the Avaya logo, a welcome message for 'admin', and links for Home, Help, and Logoff. The left sidebar contains a tree view of configuration options, with 'System Configuration' expanded. The main content area is titled 'Add H.323 Connection' and includes a breadcrumb trail: 'Home > System Configuration > VoIP Connections > Add H.323 Connection'. The form contains the following fields and options:

- Name:** devcon13
- Enable:** ☒ Yes ☐ No
- Gatekeeper Address:** 10.32.24.20
- Alternative Gatekeeper Address:** (empty)
- Gatekeeper Port:** 1719
- Media Encryption:** ☒ Yes ☐ No

**New Stations**

From	To
Station: 23801	23808

**Password:** (masked with dots)

☒ Same Password  
☐ Use sequential passwords

**Station Type:** Inbound and Outbound (selected), Inbound Only, Maintenance

**Add** button

**Configured Stations (M for Maintenance, I for Inbound Only)**

<No Station>
--------------

**Remove** button

**Save** **Cancel** **Help** buttons

Figure 5: H.323 Connection

## 4.2. Add an MPP Server

Add the MPP server by navigating to the **MPP Servers** screen and selecting the option from the left pane. In the MPP Server configuration page, specify a descriptive **Name** and the **Host Address** of the MPP server. Also, specify the **Maximum Simultaneous Calls** supported by the MPP server. The previously configured MPP server is shown below.

**AVAYA** Welcome, admin  
Last logged in today at 1:57:54 PM EDT

**Voice Portal 5.1 (VoicePortal)** Home Help Logoff

Expand All | Collapse All

- ▼ **User Management**
  - Roles
  - Users
  - Login Options
- ▼ **Real-Time Monitoring**
  - System Monitor
  - Active Calls
  - Port Distribution
- ▼ **System Maintenance**
  - Audit Log Viewer
  - Trace Viewer
  - Log Viewer
  - Alarm Manager
- ▼ **System Management**
  - MPP Manager
  - Software Upgrade
  - System Backup
- ▼ **System Configuration**
  - Alarm Codes
  - Alarm/Log Options
  - Applications
  - MPP Servers
  - Report Data
  - SNMP
  - Speech Servers
  - VoIP Connections
  - VPMS Servers
- ▼ **Security**
  - Certificates
  - Licensing
- ▼ **Reports**
  - Standard
  - Custom
  - Scheduled

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

### Change 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: MPP

Host Address: 10.32.24.31

Network Address (VoIP): <Default>

Network Address (MRCP): <Default>

Network Address (AppSvr): <Default>

Maximum Simultaneous Calls: 20

Restart Automatically: ☐ Yes ☒ No

#### MPP Certificate

Owner: CN=mpp,O=Avaya,OU=MPP  
Issuer: CN=mpp,O=Avaya,OU=MPP  
Serial Number: 8a71206004103538  
Valid from: July 19, 2010 10:55:15 AM EDT until: July 16, 2020 10:55:15 AM EDT  
Certificate fingerprints  
MD5: a7:c5:14:7c:de:84:fc:4c:64:46:c4:2b:d0:92:5d:26  
SHA: b6:46:4f:8b:d1:a2:f2:68:af:bc:b8:5f:1e:a7:7d:72:bb:f8:2f:ea

Categories and Trace Levels ▶

**Save** **Apply** **Cancel** **Help**

Figure 6: MPP Server



### 4.3. Configure the VoIP Audio Format

The **VoIP Audio Format** for the MPP server is configured in the **VoIP Settings** screen accessible by selecting **MPP Servers** in the left pane. Click on the **VoIP Settings** button. In the **VoIP Settings** screen, set the **MPP Native Format** field to *audio/basic* for mu-law as shown in Figure 7. Click Save.

**AVAYA** Welcome, admin  
Last logged in today at 1:57:54 PM EDT

**Voice Portal 5.1 (VoicePortal)** Home ? Help Logoff

Expand All | Collapse All

**▼ User Management**  
Roles  
Users  
Login Options

**▼ Real-Time Monitoring**  
System Monitor  
Active Calls  
Port Distribution

**▼ System Maintenance**  
Audit Log Viewer  
Trace Viewer  
Log Viewer  
Alarm Manager

**▼ System Management**  
MPP Manager  
Software Upgrade  
System Backup

**▼ System Configuration**  
Alarm Codes  
Alarm/Log Options  
Applications  
MPP Servers  
Report Data  
SNMP  
Speech Servers  
VoIP Connections  
VPMS Servers

**▼ Security**  
Certificates  
Licensing

**▼ Reports**  
Standard  
Custom  
Scheduled

You are here: [Home](#) > [System Configuration](#) > [MPP Servers](#) > [VoIP Settings](#)

### VoIP Settings

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.

#### Port Ranges

	Low	High
UDP:	23000	30999
TCP:	31000	31999
MRCP:	32000	32999
H.323 Station:	35000	50000

#### RTCP Monitor Settings

Host Address:

Port:

#### VoIP Audio Formats

MPP Native Format: **audio/basic**

#### Audio Codecs

Packet Time: **20**

G729: ☒ Yes ☐ No

Reduced Complexity Encoder: ☒ Yes ☐ No

Discontinuous Transmission: ☒ Yes ☐ No

First Offered: **G729**

#### QoS Parameters

	VLAN	Diffserv
H.323:	6	46
SIP:	6	46
RTSP:	6	46

#### Out of Service Threshold (% of VoIP Resources)

	Trigger	Reset
Warn:	10	0
Error:	20	10
Fatal:	70	50

**Save** **Apply** **Cancel** **Help**

Figure 7: VoIP Settings

## 4.4. Add an ASR Server

To configure the ASR (automatic speech recognition) server, click on **Speech Servers** in the left pane, select the **ASR** tab, and then click **Add**. For a Nuance Speech Server, the **Engine Type** should be set to *Nuance*. Set the **Network Address** field to the IP address of the Nuance Speech Server and select the desired **Languages** supported by the application. The **Total Number of Licensed ASR Resources** should also be set to the appropriate value. Set the **Protocol** field to *MRCP V2*. The other fields were left at their default values. Click **Save**.

**AVAYA** Welcome, admin  
Last logged in today at 1:57:54 PM EDT

**Voice Portal 5.1 (VoicePortal)** Home Help Logoff

Expand All | Collapse All

**▼ User Management**  
Roles  
Users  
Login Options

**▼ Real-Time Monitoring**  
System Monitor  
Active Calls  
Port Distribution

**▼ System Maintenance**  
Audit Log Viewer  
Trace Viewer  
Log Viewer  
Alarm Manager

**▼ System Management**  
MPP Manager  
Software Upgrade  
System Backup

**▼ System Configuration**  
Alarm Codes  
Alarm/Log Options  
Applications  
MPP Servers  
Report Data  
SNMP  
Speech Servers  
VoIP Connections  
VPMS Servers

**▼ Security**  
Certificates  
Licensing

**▼ Reports**  
Standard  
Custom  
Scheduled

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

### Add ASR Server

Use this page to configure Voice Portal to communicate with a new ASR server.

Name:

Enable: ☒ Yes ☐ No

Engine Type:

Network Address:

Base Port:

Total Number of Licensed ASR Resources:

New Connection per Session: ☐ Yes ☒ No

Languages:

**MRCP**

Ping Interval:  second(s)

Response Timeout:  second(s)

Protocol:

Transport Protocol:

Listener Port:

**Save** **Cancel** **Help**

Figure 8: ASR Speech Server

## 4.5. Add a TTS Server

To configure the TTS (text to speech) server, click on **Speech Servers** in the left pane, select the **TTS** tab, and then click **Add**. For a Nuance Speech Server, the **Engine Type** should be set to *Nuance*. Set the **Network Address** field to the IP address of the Nuance Speech Server and select the desired **Voices** supported by the application. The **Total Number of Licensed TTS Resources** should also be set to the appropriate value. Set the **Protocol** field to *MRCP V2*. The other fields were left at their default values. Click **Save**.

The screenshot displays the Avaya Voice Portal 5.1 (VoicePortal) interface. The top header shows the Avaya logo, the user 'admin', and the login time 'Last logged in today at 1:57:54 PM EDT'. The left navigation pane lists various system management options, with 'System Configuration' expanded. The main content area is titled 'Add TTS Server' and provides instructions for configuring a new TTS server. The configuration fields are as follows:

- Name:** Nuance TTS
- Enable:** ☒ Yes ☐ No
- Engine Type:** Nuance
- Network Address:** 10.32.24.140
- Base Port:** 5060
- Total Number of Licensed TTS Resources:** 4
- New Connection per Session:** ☐ Yes ☒ No
- Voices:** A list box containing several voice options, with 'English(USA) en-US Samantha F' selected.
- MRCP:**
  - Ping Interval:** 15 second(s)
  - Response Timeout:** 4 second(s)
  - Protocol:** MRCP V2
  - Transport Protocol:** TCP
  - Listener Port:** 5060

At the bottom of the form are three buttons: 'Save', 'Cancel', and 'Help'.

Figure 9: TTS Server

## 4.6. Add an Application

On the **Applications** page, add a Voice Portal application. Specify a **Name** for the application, set the **Type** field to *VoiceXML*, and set the **VoiceXML URL** field to point to the Nuance OSA application. Next, specify the type of ASR and TTS servers to be used by the application and the called numbers that invoke the Nuance OSA application. The called number 23802 is used to access the “top-level menu” entry point and the called number 23803 is used to access the “DNIS Test” entry point. The **Applications** screen is shown in **Figure 10**. Click **Save**.

**AVAYA** Welcome, admin  
Last logged in today at 11:20:57 AM EDT

**Voice Portal 5.1 (VoicePortal)**

Expand All | Collapse All

- ▼ **User Management**
  - Roles
  - Users
  - Login Options
- ▼ **Real-Time Monitoring**
  - System Monitor
  - Active Calls
  - Port Distribution
- ▼ **System Maintenance**
  - Audit Log Viewer
  - Trace Viewer
  - Log Viewer
  - Alarm Manager
- ▼ **System Management**
  - MPP Manager
  - Software Upgrade
  - System Backup
- ▼ **System Configuration**
  - Alarm Codes
  - Alarm/Log Options
  - Applications
  - MPP Servers
  - Report Data
  - SNMP
  - Speech Servers
  - VoIP Connections
  - VPMS Servers
- ▼ **Security**
  - Certificates
  - Licensing
- ▼ **Reports**
  - Standard
  - Custom
  - Scheduled

Name:

Enable: ☒ Yes ☐ No

Type:

**URL**

☒ Single ☐ Fail Over ☐ Load Balance

VoiceXML URL:  **Verify**

Mutual Certificate Authentication: ☐ Yes ☒ No

Basic Authentication: ☐ Yes ☒ No

**Speech Servers**

ASR:  TTS:

Languages:  Voices:

**Application Launch**

☒ Inbound ☐ Inbound Default ☐ Outbound

☒ Number ☐ Number Range ☐ URI

Called Number:  **Add**

**Remove**

**Speech Parameters** ▶

**Reporting Parameters** ▶

**Advanced Parameters** ▶

**Save** **Cancel** **Help**

**Figure 10: Applications**

## 4.7. Start the MPP Server

Start the MPP server from the **MPP Manager** page shown in **Figure 11**. Select the MPP and then click the **Start** button. After the MPP is started, the **Mode** of the MPP should be *Online*, the **State** should be *Running*, and **Config** should be *OK*.

**AVAYA** Welcome, admin  
Last logged in today at 1:57:54 PM EDT

**Voice Portal 5.1 (VoicePortal)** Home ? Help Logoff

Expand All | Collapse All

**User Management**  
Roles  
Users  
Login Options

**Real-Time Monitoring**  
System Monitor  
Active Calls  
Port Distribution

**System Maintenance**  
Audit Log Viewer  
Trace Viewer  
Log Viewer  
Alarm Manager

**System Management**  
MPP Manager  
Software Upgrade  
System Backup

**System Configuration**  
Alarm Codes  
Alarm/Log Options  
Applications  
MPP Servers  
Report Data  
SNMP  
Speech Servers  
VoIP Connections  
VPMS Servers

**Security**  
Certificates  
Licensing

**Reports**  
Standard  
Custom  
Scheduled

You are here: [Home](#) > System Management > MPP Manager

**MPP Manager (7/23/10 3:13:21 PM EDT)** Refresh

This page displays the current state of each MPP in the Voice Portal system. To enable the state and mode commands, select one or more MPPs. To enable the mode commands, the selected MPPs must also be stopped.

Last Poll: 7/23/10 3:13:17 PM EDT

<input checked="" type="checkbox"/>	Server Name	Mode	State	Config	Auto Restart	Restart Schedule		Active Calls	
						Today	Recurring	In	Out
<input checked="" type="checkbox"/>	MPP	Online	Stopped	None	No	No	None	0	0

**State Commands**

Start Stop Restart Reboot Halt Cancel

**Mode Commands**

Offline Test Online

**Restart/Reboot Options**

☐ One server at a time  
☒ All selected servers at the same time

**Help**

Figure 11: MPP Manager

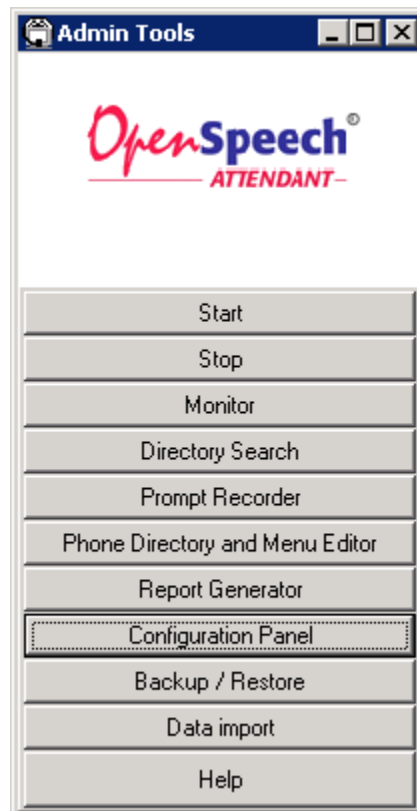
## 5. Configure Nuance OpenSpeech Attendant

This section covers the procedure for configuring Nuance OpenSpeech Attendant (OSA). The procedure includes the following areas:

- Administer settings in the Configuration Panel
- Administer transfer entries in the Phone Directory and Menu Editor
- Administer top-level menu in the Phone Directory and Menu Editor
- Administer caller utterance recording

**Note:** The Vocalizer config file, `ttsrshclient.xml`, located in `C:\Program Files\Nuance\Nuance Vocalizer for Network 5.0\config` directory of the speech server must be tuned to interoperate successfully with Nuance OSA. The tag `<ssml_validation>` must be set to *none*. The default value is *strict*.

Nuance OSA is configured through **Admin Tools** which can be started by navigating to **Start→Nuance→Admin Tools**. The initial screen is displayed as shown in **Figure 12**.

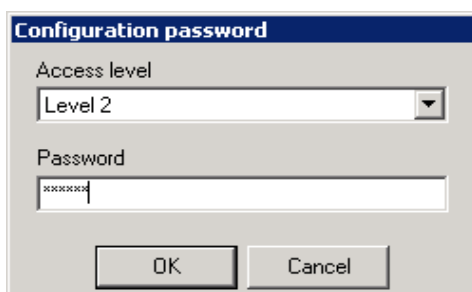


**Figure 12: Admin Tools**



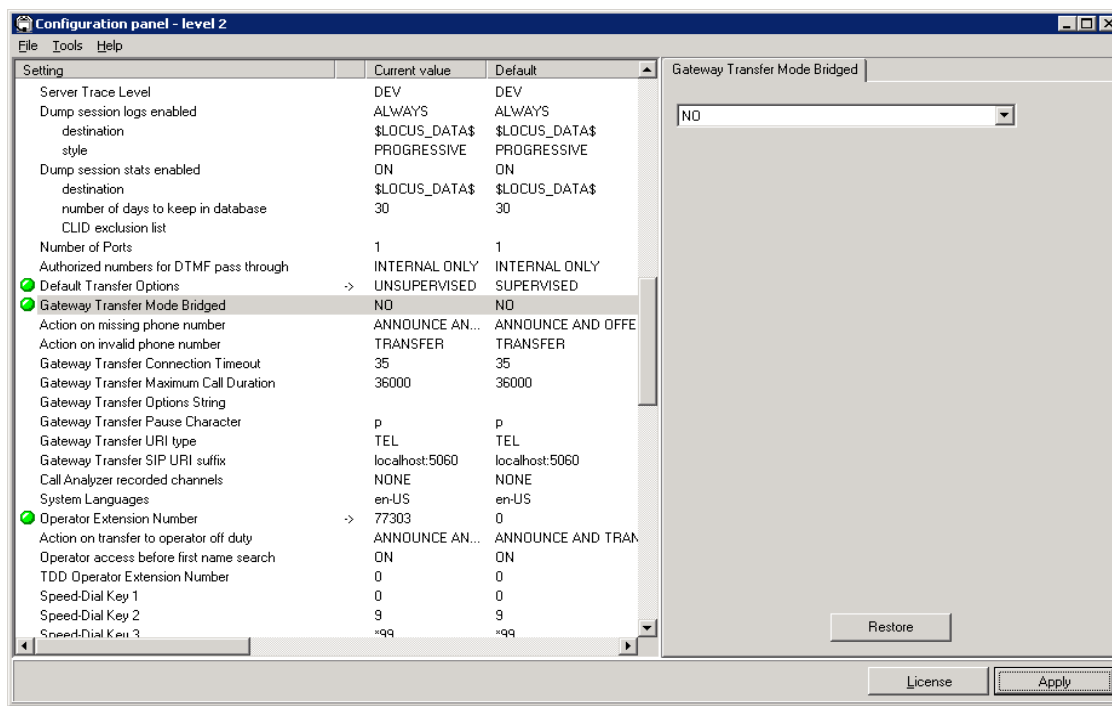
## 5.1. Administer Settings in the Configuration Panel

To open the **Configuration Panel**, click on this option in the **Admin Tools** window in **Figure 12**. The login prompt will be displayed to the user as shown in **Figure 13**. Log in with the appropriate credentials using *Level 2* access level.



**Figure 13: Configuration Panel Login Window**

The **Configuration Panel** shown in **Figure 14** is displayed. The **Configuration Panel** allows the transfer mode and operator extension number to be configured. Nuance OSA supports blind, supervised, and bridged transfers with Voice Portal. To enable OSA for blind transfers, set the **Gateway Transfer Mode Supervised** and **Gateway Transfer Mode Bridged** fields to *NO* as shown in the figure below. For supervised and bridged transfers set the corresponding fields to *YES*. The **Operator Extension Number** field should be set to a valid extension on Communication Manager. Click **Apply**.



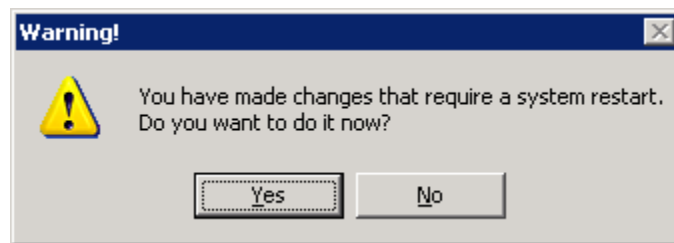
**Figure 14: Configuration Panel**



**Note:** To support supervised transfers, the special application SA8874 – Call Status Messages for 7434ND IP Softphone must be enabled on Communication Manager. When using supervised transfers, the caller won't hear any ringing during the transfer attempt.

Next, close the **Configuration Panel**. Activate the changes when prompted by the system as shown

**Figure 15.**



**Figure 15: Activate Changes in Configuration Panel**

The following window is displayed. Click **OK**.

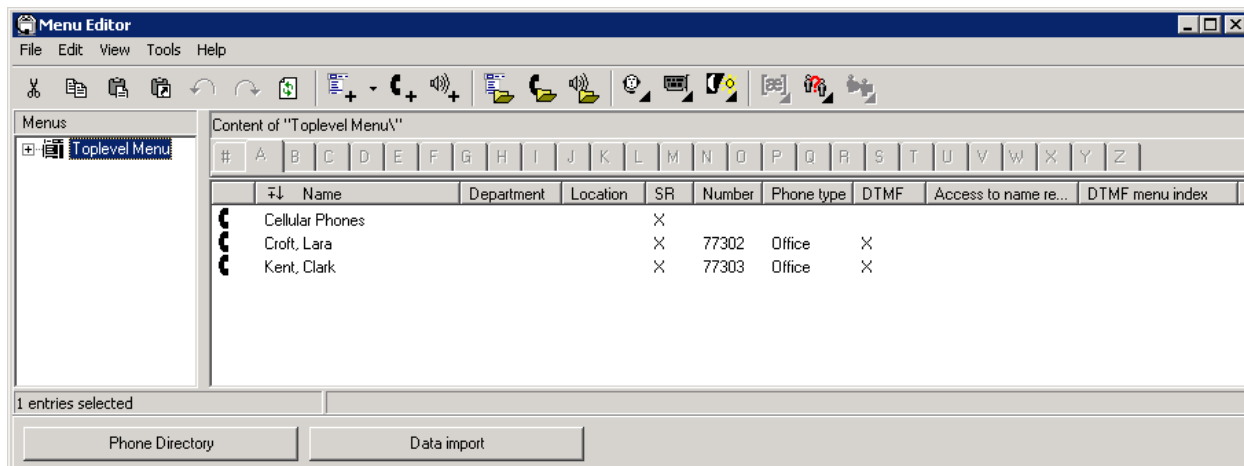


**Figure 16: Startup in Progress Window**

## 5.2. Administer Transfer Entries in Phone Directory and Menu Editor

From **Admin Tools**, click on the **Phone Directory and Menu Editor** option.

**Figure 17** is displayed.



**Figure 17: Phone Directory and Menu Editor**

Next, select **File→New→Transfer Entry** from the menu options. The **Creating transfer entry** window is displayed as shown in

**Figure 18.** Configure the **First names** and **Last names** for this entry and set the **Number** to a valid extension. Enable the **Access to personal functions** and **Access to name recorder** options and set the **PIN** field so that this user can access the Personal Administration Mode (PAM) to change their name recording. Click **OK**.

**Creating transfer entry**

Name in directory: \_\_\_\_\_

☐ Deactivated entry ☐ DTMF Menu Index

Names | Advanced | Call Accounting | Information

First names	Middle names	Last names
David		Wells

Aliases (English (US))

Allow:

☒ Access to personal functions      PIN: XXXXXX

☒ Voice biometrics      Password: \_\_\_\_\_

☒ Access to name recorder

☐ Never propose this name

Create Personal Directory      Delete Personal Directory

Schedule: Always

☒ Speech recognition

Number type	Number	S	U	FM #	Priv
<input checked="" type="checkbox"/> Office	77301	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>		<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

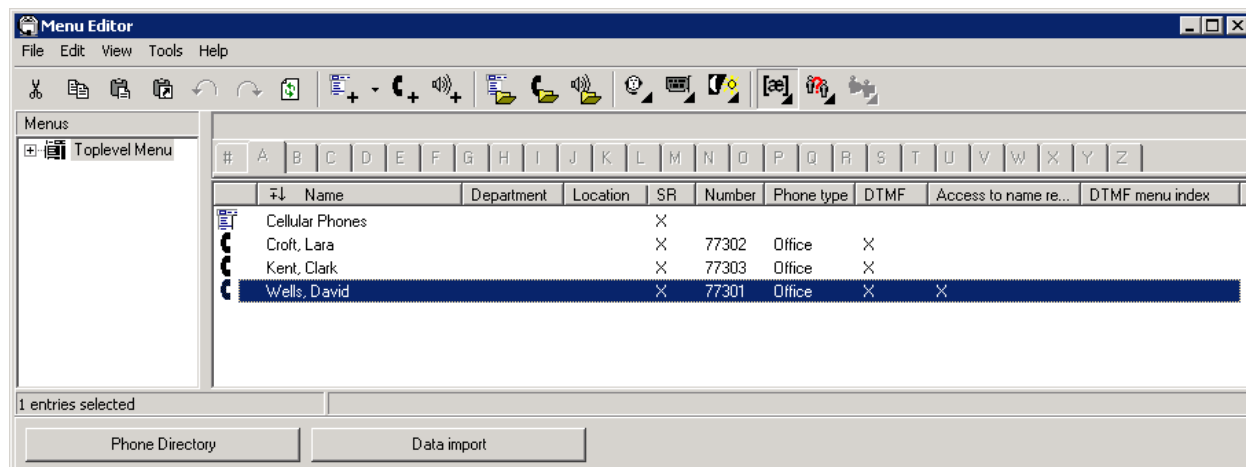
Message: None      New

Custom Prompt

Previous      New      Next      OK      Cancel      Apply

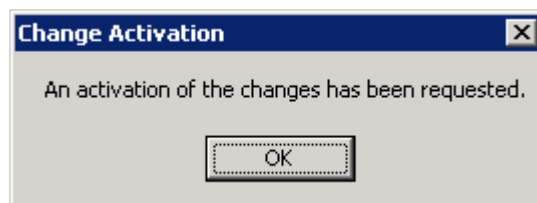
**Figure 18: Creating Transfer Entry**

The new transfer entry is now displayed in the Menu Editor window, but the entry has not been activated yet. Highlight the new transfer entry and then click on the **Activate Changes** icon



**Figure 19: Menu Editor with New Transfer Entry**

The following prompt is displayed to activate the new transfer entry so that it would be recognized by Nuance OSA.



**Figure 20: Change Activation**

### 5.3. Administer Top-Level Menu in Phone Directory and Menu Editor

Nuance OSA allows the configuration of multiple entry points. The use of multiple entry points enable individual services or departments to be assigned a different entry point, which can be configured using a DNIS or CLID number. When a call is received, Nuance OSA can route the call to the appropriate entry point based on the dialed number (DNIS) or the caller's phone number (CLID). In this example, an entry point was configured using the DNIS option.

Create a new entry point by selecting **File→New→Top-Level Menu** from the menu options. The window in

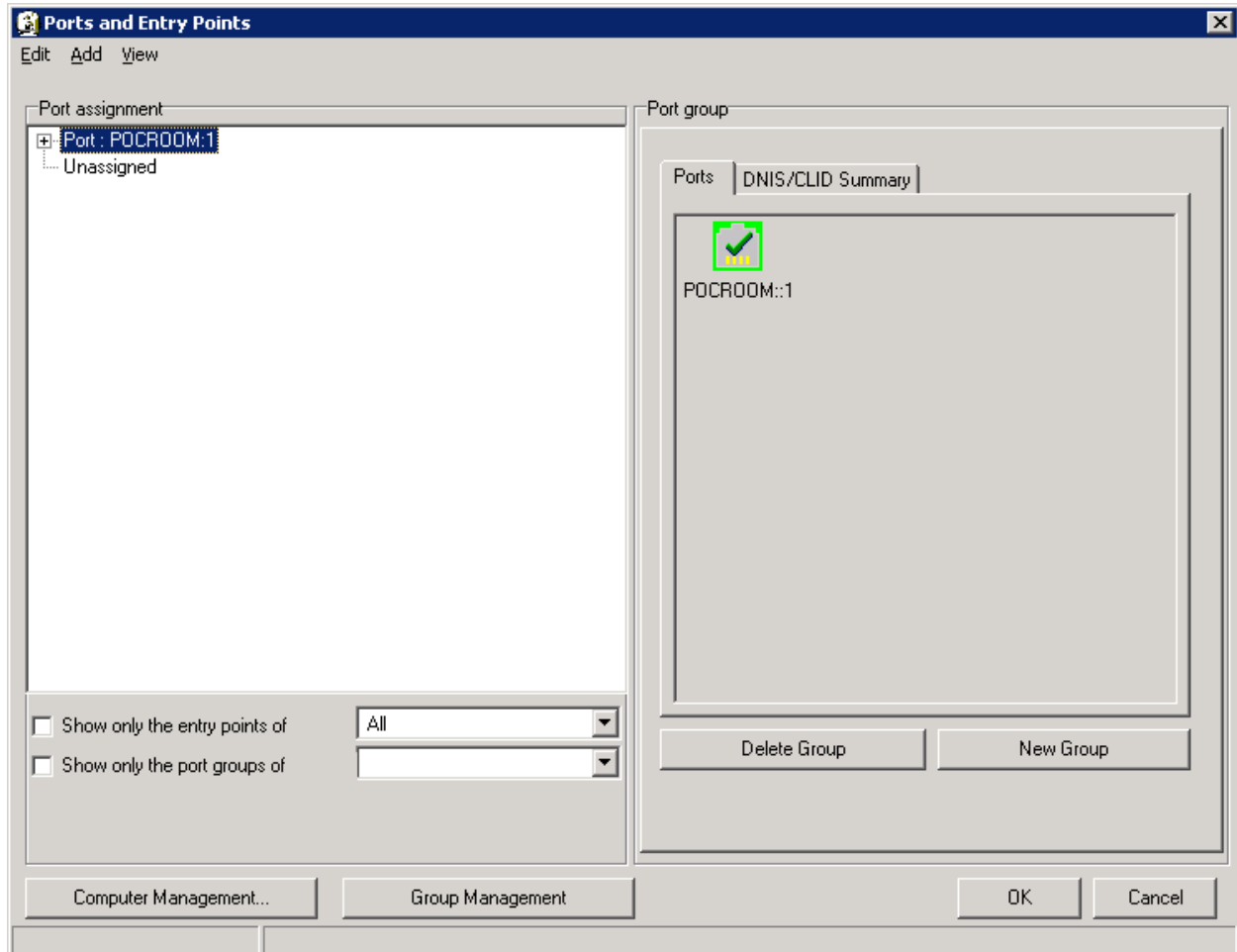
**Figure 21** is displayed. Enter a descriptive name in the **Aliases** field, enable the **Access to name recorder** option, and set the **PIN** field for use with PAM. Click **OK** to activate the changes.

The screenshot shows the 'Creating Menu' dialog box with the following details:

- Names Tab:**
  - Name in directory: [Empty field]
  - First names: [Empty field]
  - Middle names: [Empty field]
  - Last names: [Empty field]
  - Aliases (English (US)): [Empty field]
  - DNIS Test: [Empty field]
- Advanced Tab:**
  - Allow:
    - ☐ Access to personal functions
    - ☐ Voice biometrics
    - ☒ Access to name recorder
    - ☐ Never propose this name
  - PIN: [Empty field]
  - Password: [Empty field]
- Call Accounting Tab:**
  - DTMF Menu Index: [Empty field]
- Information Tab:**
  - Menu Behavior: [Always]
  - Access code: [Empty field]
  - ☒ Call redirect permitted
  - Default number: [Any type]
  - Conversation template: [Default]
  - Next step: [Operator]
  - Action trigger: [None]
- Always Section:**
  - ☒ Speech recognition
  - Menu behavior 1: [Menu behavior 1] (New button)
  - Number of utterances: [Default]
  - Operator extension: [Empty field]
  - Number of spellings: [Default]
  - Voice profile: [Default from entry po]
  - Number of interactions: [Default]
  - Barge-in function: [Default from entry po]
  - ☐ Always ask for confirmation
  - ☐ Listing of entries

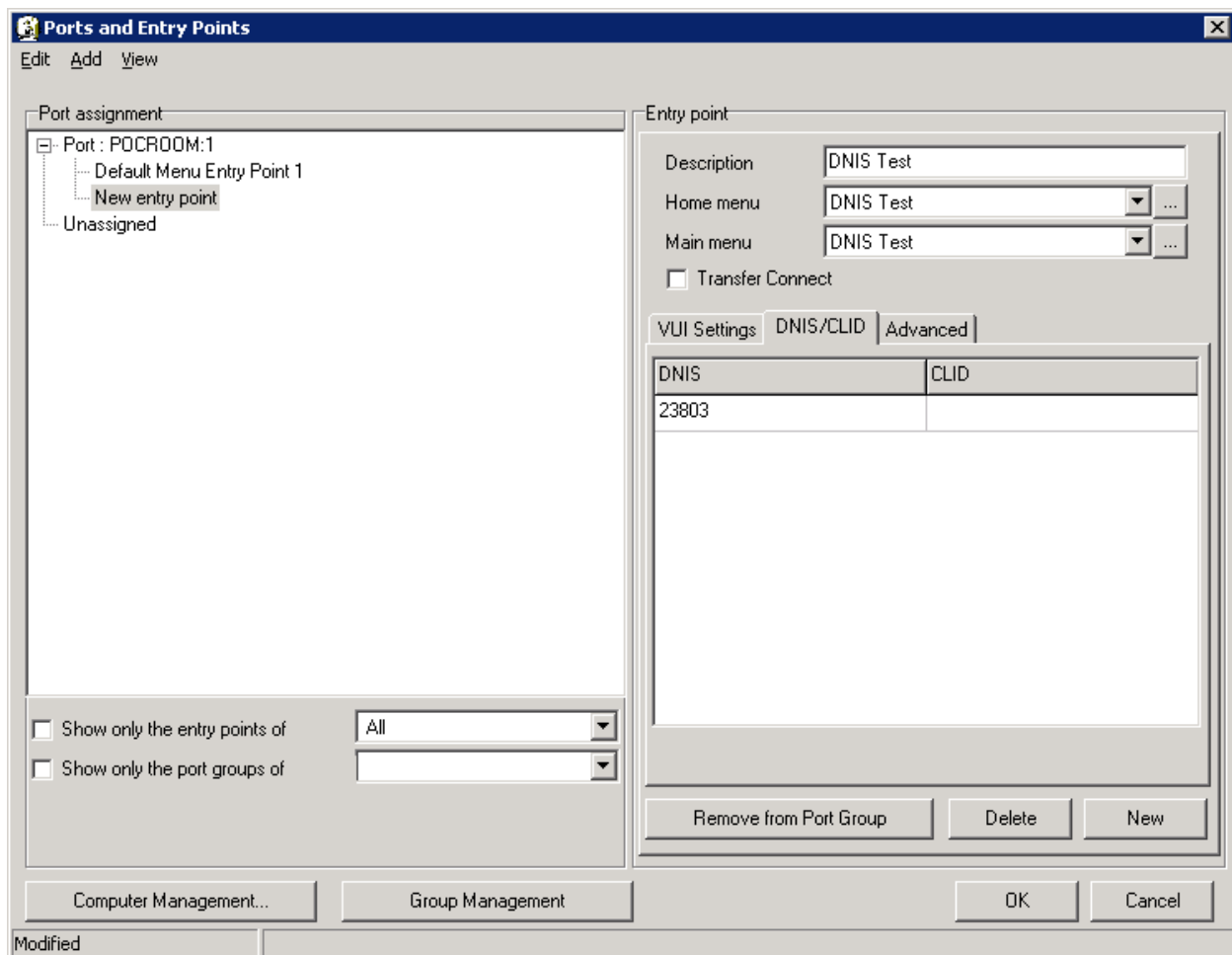
**Figure 21: Creating Menu**

From the **Menu Editor**, navigate to **Edit→Ports and Entry Points** to display the window in **Figure 22**. Select the first item under **Port assignment** in the left pane and then select **Add→Entry Point** from the menu options. **Figure 23** is displayed.



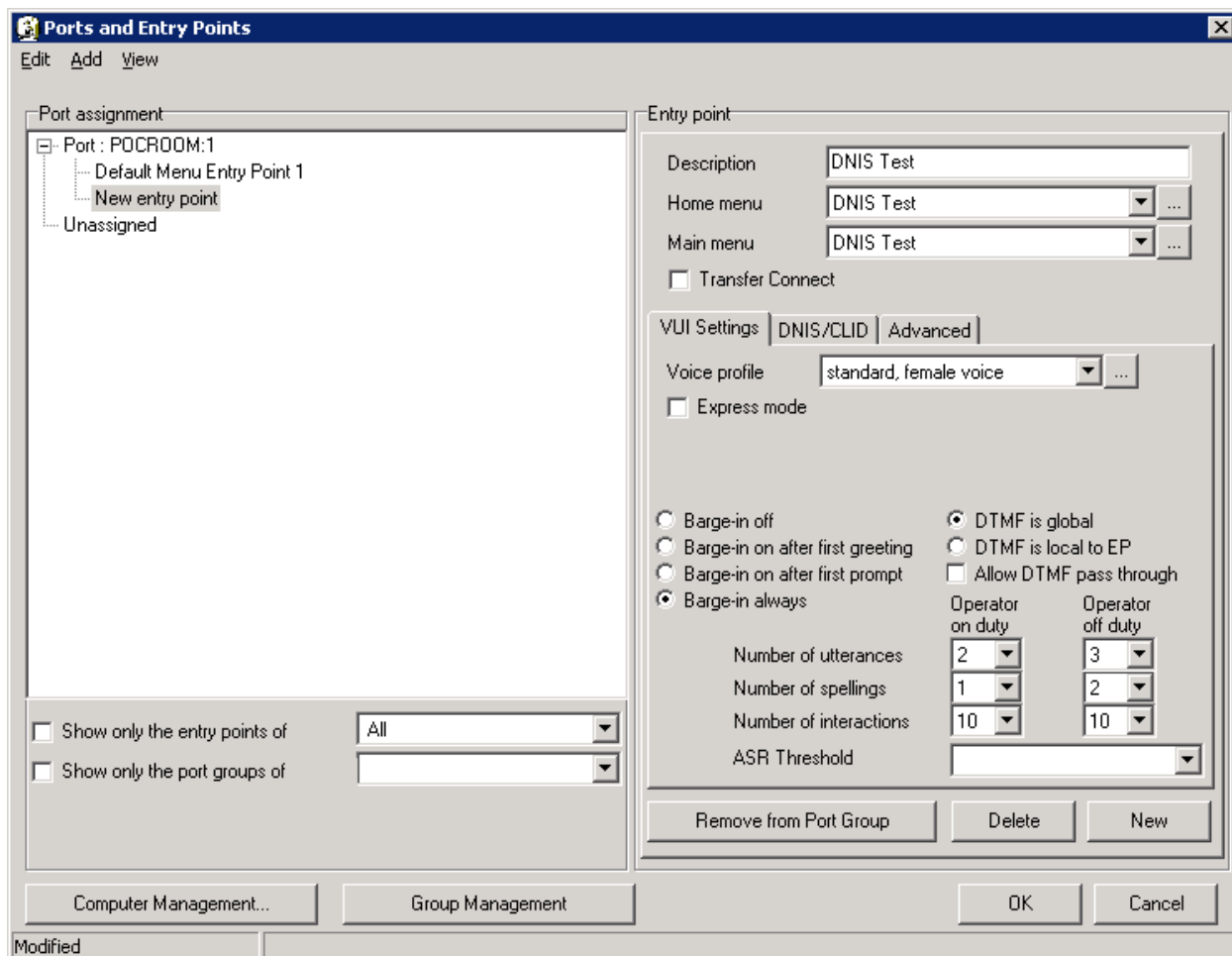
**Figure 22: Ports and Entry Points**

In the **Ports and Entry Points** window in **Figure 23**, set the **Home menu** and **Main menu** to the new entry point configured in **Figure 21**. Select the **DNIS/CLID** tab and enter an extension associated with a Voice Portal station configured on Communication Manager as shown in **Figure 2**. Click **OK**. In this example, this entry point will be used when Nuance OSA receives a DNIS of 23803. The other extension, 23802, is used to direct the user to a different top-level menu.



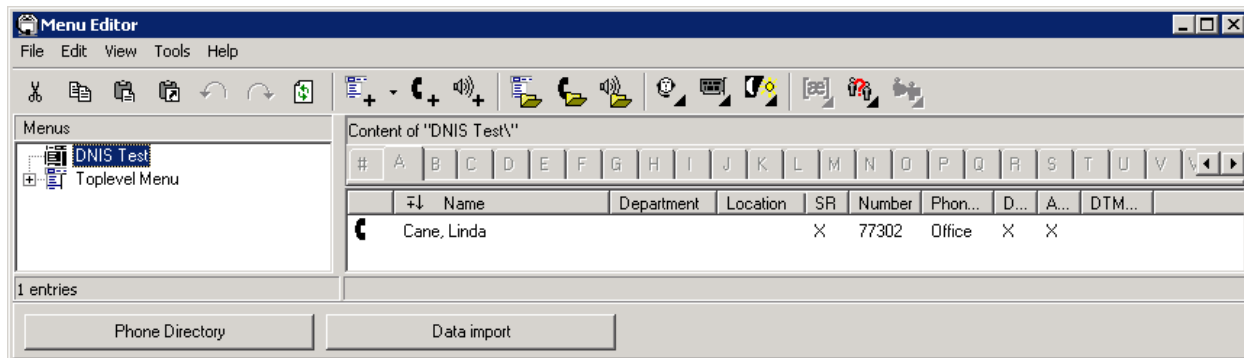
**Figure 23: New Entry Point (DNIS/CLID Tab)**

In the **VUI Settings** tab, set the **Voice Profile** as shown below. Click **OK**.



**Figure 24: Ports and Entry Points (VUI Settings Tab)**

A transfer entry was also added under the **DNIS Test** top-level menu using the procedure described in **Section 5.2**.



**Figure 25: Transfer Entry for DNIS Test Top-Level Menu**



## 5.4. Administer Caller Utterance Recordings

The **Monitor** tool available from the **Admin Tools** window allows a system administrator to listen to caller utterances. To enable the recording of caller utterances, click on **Configuration Panel** in **Admin Tools** and log in using Level 3 access. Set the **Call Analyzer recorded channels** parameter to *ALL* and set the **Dump output documents enabled** parameter to *ON*. Apply the changes as described above. Caller utterances can then be retrieved from the **Monitor** tool.

From **Admin Tools**, click on **Monitor** and log in with the appropriate credentials. Click on **Call Logs** in the left pane to display a list of calls. Select the call for which you would like to listen to the caller utterance. The caller utterance will be displayed as shown below and may be listened to by clicking on the **play / stop** button.

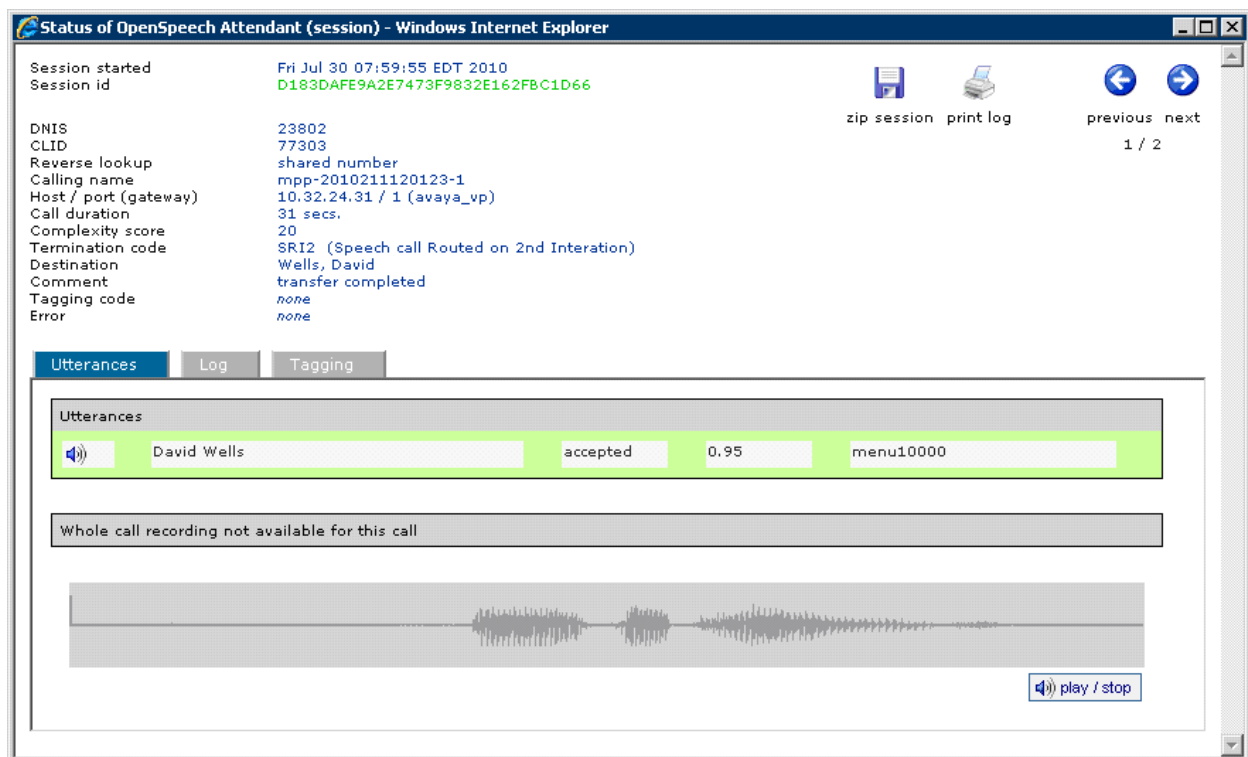


Figure 26: Caller Utterance

## 6. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. Feature testing focused on Nuance OSA successfully recognizing spoken names and extensions entered via DTMF, and then transferring the call to the correct destination. Blind, supervised, and bridged transfers were verified. Other features covered included barge-in / no barge-in, adding new transfer entries, recording caller utterances, and accessing Maintenance Mode and Personal Administration Mode to record name and change PIN.

Serviceability testing focused on verifying the ability of the Nuance OSA to recover from adverse conditions, such as server restarts, power failures, and disconnecting cables to the IP network.

All test cases passed. One observation made during testing was that during a supervised transfer, ringback is not heard by the caller. For blind and bridged transfers, ringback is heard by the caller.

## 7. Verification Steps

This section provides the verification steps that may be performed to verify that Voice Portal can access the Nuance OSA.

1. From the VPMS web interface, verify that the MPP server is online and running in the **MPP Manager** page as shown in
2. **Figure 27.**

**AVAYA** Welcome, admin  
Last logged in today at 1:57:54 PM EDT

**Voice Portal 5.1 (VoicePortal)** Home Help Logoff

Expand All | Collapse All

- ▼ **User Management**
  - Roles
  - Users
  - Login Options
- ▼ **Real-Time Monitoring**
  - System Monitor
  - Active Calls
  - Port Distribution
- ▼ **System Maintenance**
  - Audit Log Viewer
  - Trace Viewer
  - Log Viewer
  - Alarm Manager
- ▼ **System Management**
  - MPP Manager**
  - Software Upgrade
  - System Backup
- ▼ **System Configuration**
  - Alarm Codes
  - Alarm/Log Options
  - Applications
  - MPP Servers
  - Report Data
  - SNMP
  - Speech Servers
  - VoIP Connections
  - VPMS Servers
- ▼ **Security**
  - Certificates
  - Licensing
- ▼ **Reports**
  - Standard
  - Custom
  - Scheduled

You are here: [Home](#) > System Management > MPP Manager

**MPP Manager (7/23/10 3:28:23 PM EDT)** Refresh

This page displays the current state of each MPP in the Voice Portal system. To enable the state and mode commands, select one or more MPPs. To enable the mode commands, the selected MPPs must also be stopped.

Last Poll: 7/23/10 3:28:12 PM EDT

	Server Name	Mode	State	Config	Auto Restart	Restart Schedule	Active Calls		
						Today	Recurring	In	Out
<input type="checkbox"/>	MPP	Online	Running	OK	No	No	None	0	0

**State Commands**

Start Stop Restart Reboot Halt Cancel

**Mode Commands**

Offline Test Online

**Restart/Reboot Options**

☐ One server at a time

☒ All selected servers at the same time

Help

## **Figure 27: MPP Manager**

3. From the VPMS web interface, verify that the ports on the MPP server are in-service from the **Port Distribution** page shown in
4. **Figure 28.**

**AVAYA** Welcome, admin  
Last logged in today at 1:57:54 PM EDT

**Voice Portal 5.1 (VoicePortal)** Home ? Help Logoff

Expand All | Collapse All

- ▼ **User Management**
  - Roles
  - Users
  - Login Options
- ▼ **Real-Time Monitoring**
  - System Monitor
  - Active Calls
  - Port Distribution
- ▼ **System Maintenance**
  - Audit Log Viewer
  - Trace Viewer
  - Log Viewer
  - Alarm Manager
- ▼ **System Management**
  - MPP Manager
  - Software Upgrade
  - System Backup
- ▼ **System Configuration**
  - Alarm Codes
  - Alarm/Log Options
  - Applications
  - MPP Servers
  - Report Data
  - SNMP
  - Speech Servers
  - VoIP Connections
  - VPMS Servers
- ▼ **Security**
  - Certificates
  - Licensing
- ▼ **Reports**
  - Standard
  - Custom
  - Scheduled

You are here: [Home](#) > Real-Time Monitoring > Port Distribution

### Port Distribution (7/23/10 3:29:02 PM EDT)

[Refresh](#)

This page displays information about how the telephony resources have been distributed to the MPPs. You configure the telephony resources on the VoIP Connections page.

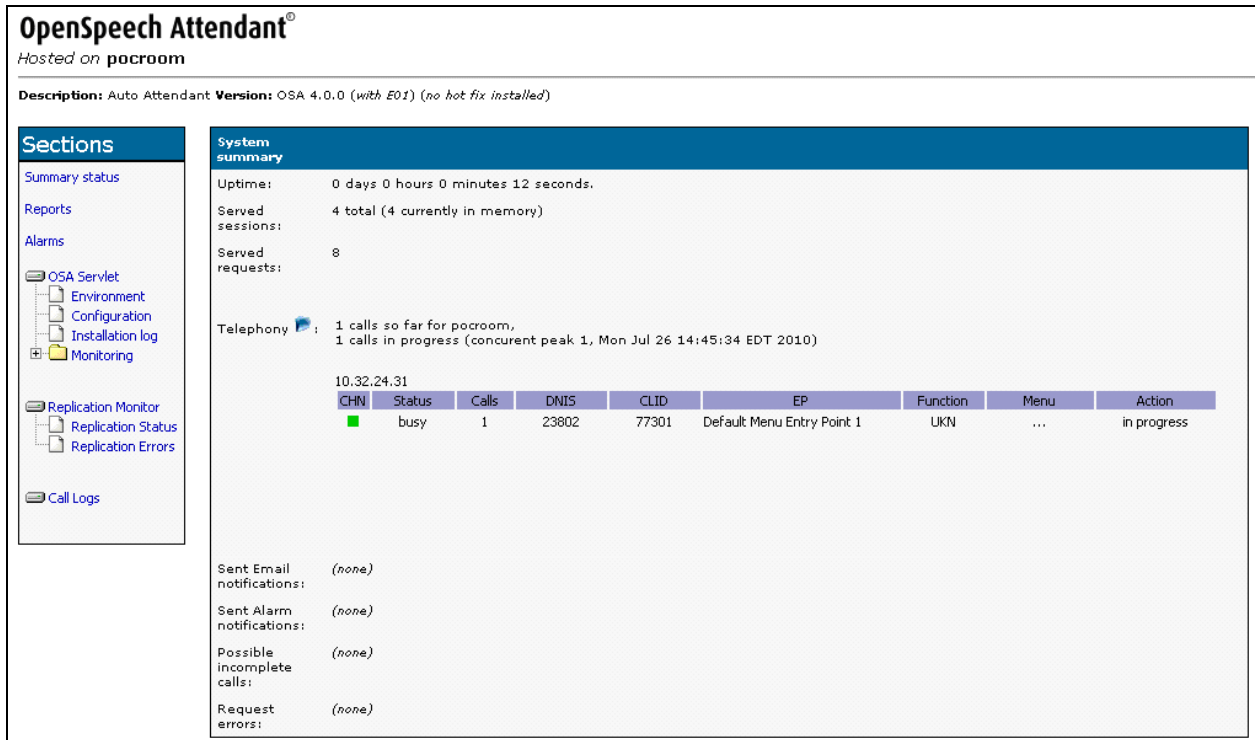
Total Ports: 8 Last Poll: 7/23/10 3:29:03 PM EDT

Port	Mode	State	Port Group	Protocol	Current Allocation	Base Allocation
23801	Online	In service	devcon13	H323	MPP	
23802	Online	In service	devcon13	H323	MPP	
23803	Online	In service	devcon13	H323	MPP	
23804	Online	In service	devcon13	H323	MPP	
23805	Online	In service	devcon13	H323	MPP	
23806	Online	In service	devcon13	H323	MPP	
23807	Online	In service	devcon13	H323	MPP	
23808	Online	In service	devcon13	H323	MPP	

[Help](#)

**Figure 28: Port Distribution**

5. Place a call to Voice Portal that invokes the Nuance OSA application. From the Nuance OSA server, open the OSA Monitor from **Admin Tools**. Verify that it detects an active call as shown in
6. **Figure 29.**



**Figure 29: OSA Monitor**

7. Verify that the Nuance OSA greeting is heard and OSA transfers the call to the proper destination specified in a spoken name or extension entered via DTMF.

8. From the OSA Monitor, click on **Call logs** in the left pane. Verify that the call log shown in
9. **Figure 30** is displayed with the correct call information and status.

**OpenSpeech Attendant®**  
Hosted on pocroom

Description: Auto Attendant Version: OSA 4.0.0 (with E01) (no hot fix installed)

**Sections**

- Summary status
- Reports
- Alarms
- OSA Servlet
  - Environment
  - Configuration
  - Installation log
  - Monitoring
- Replication Monitor
  - Replication Status
  - Replication Errors
- Call Logs**

**CALL LOGS**

Select period:  
Date from: 07/26/2010  
Date to: 07/26/2010

Filter by:  
DNIS: Starts With  
CLID: Starts With

Filter by call termination codes:  
None  
ADNT - Announce number but Do Not Transfer (CS)  
B - Busy (INF)  
BO - Busy on Operator (INF)  
CC - Cancelled by Caller (INF)

Filter by call complexity: <=   
Filter by call duration: <= seconds

all types last 50 calls Submit

Logs (from database), showing first 50 out of 177 sessions ... too many sessions.

Call start	DNIS	CLID	Call complexity	Call duration	Error	Termination code	Destination	System comment	Tagging
26/07/2010 14:47:00	23810	77301	0	24		HAFNF	Toplevel Menu	caller hangup	
26/07/2010 14:46:19	23810	77301	0	9		HG	Toplevel Menu	caller hangup	
26/07/2010 14:45:33	23802	77301	0	24		HAFNF	Toplevel Menu	caller hangup	
26/07/2010 14:43:26	75200	77301	0	17		HAFNF	Toplevel Menu	caller hangup	
26/07/2010 14:42:19	75200	76301	0	24		HAFNF	Toplevel Menu	caller hangup	
26/07/2010 14:41:32	...	...	0	16		HG	Toplevel Menu	caller hangup	
26/07/2010 14:06:55	75200	77303	0	19		HG	DNIS Test	caller hangup	
26/07/2010 14:04:39	75200	77303	0	13		HG	DNIS Test	caller hangup	

**Figure 30: Call Log**

## 8. Conclusion

These Application Notes describe the configuration steps required to integrate Nuance OpenSpeech Attendant with Avaya Voice Portal. All feature and serviceability test cases were completed successfully.

## 9. Additional References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Administering Avaya Aura<sup>TM</sup> Communication Manager*, Release 5.2, May 2009, Issue 5.0, Document Number 03-300509.
- [2] *Avaya Aura<sup>TM</sup> Communication Manager Feature Description and Implementation*, Release 5.2, May 2009, Issue 7, Document 555-245-205.
- [3] *Administering Voice Portal*, June 2010.

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