



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

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# **Application Notes for Configuring Nuance Speech Attendant with Avaya Aura® Session Manager R6.3 and Avaya Aura® Communication Manager R6.2 – Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps required to integrate the Nuance Speech Attendant with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP integration. Nuance Speech 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 and be transferred 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 Speech Attendant (SA) with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP integration. Nuance Speech 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 be transferred to the requested party.

## 2. General Test Approach and Test Results

The interoperability compliance testing included feature and serviceability test cases.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

### 2.1. Interoperability Compliance Testing

Feature testing focused on Nuance SA successfully recognizing spoken names and extensions entered via DTMF, and then transferring the call to the correct destination. Blind and supervised transfers were verified. Other features covered included: DNIS and CLID handling, 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 SA to recover from adverse conditions, such as server restarts, power failures, and disconnecting cables to the IP network.

### 2.2. Test Results

All test cases passed with the following observation:

- Nuance SA does not support shuffling (i.e., direct IP-IP media) and should be disabled either in the IP Network Region or SIP signaling group as shown in **Section 5.2**.

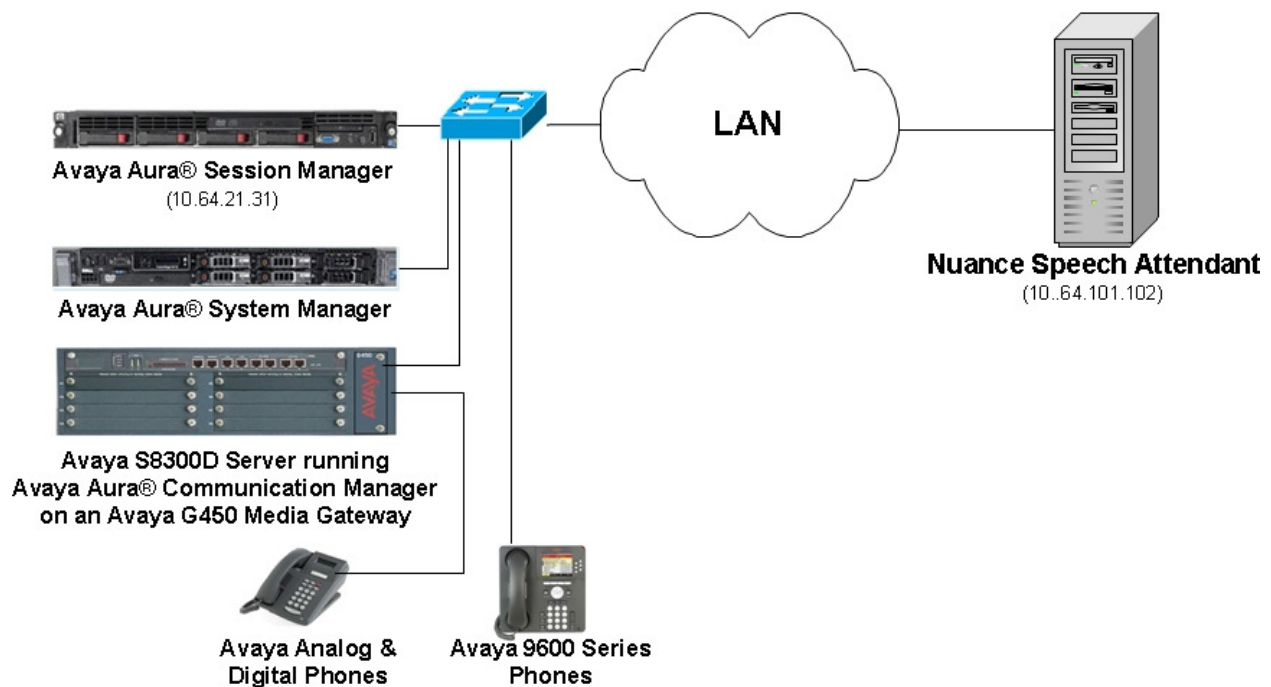
### 2.3. Support

To obtain technical support for Nuance Speech Attendant, contact Nuance via their website, email, or phone number.

- **Web:** [www.network.nuance.com](http://www.network.nuance.com)
- **Email:** [SpeechAttendant.Support@nuance.com](mailto:SpeechAttendant.Support@nuance.com)
- **Phone:** +1 (866) 434-2564 or +1 (514) 390-3922

### 3. Reference Configuration

**Figure 1** illustrates the setup used to verify the Nuance Speech Attendant (SA) solution with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Nuance SA is deployed on a dedicated server running Windows 2008 R2 Enterprise server. Session Manager interfaces to Communication Manager using a SIP trunk, and Nuance SA interfaces to Session Manager via SIP. To access the Nuance SA application, a call is simply routed from Communication Manager to the SA server through the Session Manager. Multiple SIP ports were configured on the Nuance SA server. Avaya Aura® System Manager was used to configure Session Manager.



**Figure 1: Configuration with Nuance Speech Attendant in an Avaya SIP Network.**

## 4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Avaya S8300D Server with an Avaya G450 Media Gateway	6.2 Service Pack 5
Avaya Aura® Session Manager running on a HP Proliant DL360 G7	6.3 with SP1
Avaya Aura® System Manager running on a Dell™ PowerEdge™ R610 Server	6.3 with SP1
Avaya one-X® 9600 Series IP Telephones <ul style="list-style-type: none"><li>• 96x0 (H.323)</li><li>• 96x0 (SIP)</li><li>• 96x1 (H.323)</li><li>• 96x1 (SIP)</li></ul>	Avaya one-X® Deskphone Edition 3.1.5 Avaya one-X® Deskphone Edition 2.6.9 Avaya one-X® Deskphone Edition 6.2.2 Avaya one-X® Deskphone Edition 6.2.1
Avaya 6210 Analog Phone	-
Avaya 2420 Digital Phone	-
Nuance Speech Attendant	11.1 Hotfix 44

## 5. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager. The procedures include the following areas:

- Verify SIP Trunk Capacity
- Configure a SIP trunk between Communication Manager and Session Manager
- Configure AAR Call Routing to Nuance SA

Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials. In the Communication Manager SAT screens shown throughout this document, the SAT command used to access each screen is displayed in the upper left-hand corner of the screen.

### 5.1. Verify SIP Trunk Capacity

Using the SAT, verify that SIP trunks are enabled in the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On **Page 2** of the **system-parameters customer-options** form, verify that the number of SIP trunks supported by the system is sufficient.

display <b>system-parameters customer-options</b>	Page 2 of 11
OPTIONAL FEATURES	
IP PORT CAPACITIES	USED
Maximum Administered H.323 Trunks: 12000	68
Maximum Concurrently Registered IP Stations: 18000	11
Maximum Administered Remote Office Trunks: 12000	0
Maximum Concurrently Registered Remote Office Stations: 18000	0
Maximum Concurrently Registered IP eCons: 414	0
Max Concur Registered Unauthenticated H.323 Stations: 100	0
Maximum Video Capable Stations: 41000	4
Maximum Video Capable IP Softphones: 18000	2
<b>Maximum Administered SIP Trunks: 24000</b>	<b>70</b>
Maximum Administered Ad-hoc Video Conferencing Ports: 24000	0
Maximum Number of DS1 Boards with Echo Cancellation: 522	0
Maximum TN2501 VAL Boards: 128	0
Maximum Media Gateway VAL Sources: 250	1
Maximum TN2602 Boards with 80 VoIP Channels: 128	0
Maximum TN2602 Boards with 320 VoIP Channels: 128	0
Maximum Number of Expanded Meet-me Conference Ports: 300	0
(NOTE: You must logoff & login to effect the permission changes.)	

## 5.2. Configure SIP Trunk

In the **IP Node Names** form, assign a host name and an IP address for the Session Manager SIP interface (e.g. SM\_21\_21 and 10.64.21.31). The host name will be used throughout the other configuration screens of Communication Manager.

change node-names ip		Page 1 of 2
<b>IP NODE NAMES</b>		
Name	IP Address	
<b>SM_21_31</b>	<b>10.64.21.31</b>	
default	0.0.0.0	
msgserver	10.64.21.41	

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager in **Section 6.1**. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G450 Media Gateway. However, Nuance SA does not support shuffling so it should be disabled either in the IP Network Region or in the SIP signaling group configured below. The **IP Network Region** form also specifies the **Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling group.

change ip-network-region 1		Page 1 of 20
<b>IP NETWORK REGION</b>		
Region: 1		
Location:	<b>Authoritative Domain: avaya.com</b>	
Name:		
MEDIA PARAMETERS	<b>Intra-region IP-IP Direct Audio: no</b>	
<b>Codec Set: 1</b>	<b>Inter-region IP-IP Direct Audio: no</b>	
UDP Port Min: 2048	IP Audio Hairpinning? n	
UDP Port Max: 3329		
DIFFSERV/TOS PARAMETERS		
Call Control PHB Value: 46		
Audio PHB Value: 46		
Video PHB Value: 26		
802.1P/Q PARAMETERS		
Call Control 802.1p Priority: 6		
Audio 802.1p Priority: 6		
Video 802.1p Priority: 5	<b>AUDIO RESOURCE RESERVATION PARAMETERS</b>	
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		

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to Nuance SA. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set *1* was specified in **IP Network Region** form *1* shown above. The default settings of the **IP Codec Set** form are shown below. Nuance SA supports G.711mu-law.

change ip-codec-set 1

Page 1 of 2

IP Codec Set

Codec Set: 1

Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1: <b>G.711MU</b>	<b>n</b>	<b>2</b>	<b>20</b>
2:			

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tls*.
- Specify the processor (*procr*) board and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was disabled on this form since Nuance SA does not support shuffling.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.  
Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

add signaling-group 1		Page 1 of 2															
<b>SIGNALING GROUP</b>																	
<table style="width: 100%; border: none;"> <tr> <td style="width: 33%;">Group Number: 1</td> <td style="width: 33%;">Group Type: sip</td> <td style="width: 33%;"></td> </tr> <tr> <td>IMS Enabled? n</td> <td>Transport Method: tls</td> <td></td> </tr> <tr> <td>Q-SIP? n</td> <td></td> <td></td> </tr> <tr> <td>IP Video? y</td> <td>Priority Video? n</td> <td>Enforce SIPS URI for SRTP? y</td> </tr> <tr> <td>Peer Detection Enabled? y</td> <td>Peer Server: SM</td> <td></td> </tr> </table>			Group Number: 1	Group Type: sip		IMS Enabled? n	Transport Method: tls		Q-SIP? n			IP Video? y	Priority Video? n	Enforce SIPS URI for SRTP? y	Peer Detection Enabled? y	Peer Server: SM	
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<table style="width: 100%; border: none;"> <tr> <td style="width: 50%;">Near-end Node Name: procr</td> <td style="width: 50%;">Far-end Node Name: SM_21_31</td> </tr> <tr> <td>Near-end Listen Port: 5061</td> <td>Far-end Listen Port: 5061</td> </tr> <tr> <td></td> <td>Far-end Network Region: 1</td> </tr> </table>			Near-end Node Name: procr	Far-end Node Name: SM_21_31	Near-end Listen Port: 5061	Far-end Listen Port: 5061		Far-end Network Region: 1									
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<table style="width: 100%; border: none;"> <tr> <td style="width: 50%;">Far-end Domain: avaya.com</td> <td style="width: 50%;">Bypass If IP Threshold Exceeded? n</td> </tr> <tr> <td>Incoming Dialog Loopbacks: eliminate</td> <td>RFC 3389 Comfort Noise? n</td> </tr> <tr> <td>DTMF over IP: rtp-payload</td> <td>Direct IP-IP Audio Connections? n</td> </tr> <tr> <td>Session Establishment Timer(min): 3</td> <td>IP Audio Hairpinning? n</td> </tr> <tr> <td>Enable Layer 3 Test? y</td> <td>Initial IP-IP Direct Media? n</td> </tr> <tr> <td>H.323 Station Outgoing Direct Media? n</td> <td>Alternate Route Timer(sec): 6</td> </tr> </table>			Far-end Domain: avaya.com	Bypass If IP Threshold Exceeded? n	Incoming Dialog Loopbacks: eliminate	RFC 3389 Comfort Noise? n	DTMF over IP: rtp-payload	Direct IP-IP Audio Connections? n	Session Establishment Timer(min): 3	IP Audio Hairpinning? n	Enable Layer 3 Test? y	Initial IP-IP Direct Media? n	H.323 Station Outgoing Direct Media? n	Alternate Route Timer(sec): 6			
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Configure the **Trunk Group** form as shown below. This trunk group is used for calls to SIP endpoints. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

add trunk-group 1		Page 1 of 21	
<b>TRUNK GROUP</b>			
Group Number: 1	<b>Group Type: sip</b>	CDR Reports: y	
<b>Group Name: SM_21_31</b>	COR: 1	TN: 1	TAC: 101
Direction: two-way	Outgoing Display? n		
Dial Access? n	Night Service:		
Queue Length: 0			
<b>Service Type: tie</b>	Auth Code? n		
	Member Assignment Method: auto		
	<b>Signaling Group: 1</b>		
	<b>Number of Members: 12</b>		

On **Page 3** of the trunk group form, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number sent to the far-end.

add trunk-group 1		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
<b>Numbering Format: private</b>			
UUI Treatment: service-provider			
Replace Restricted Numbers? n			
Replace Unavailable Numbers? n			
Modify Tandem Calling Number: no			
Show ANSWERED BY on Display? y			

Configure the **Numbering – Private Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with ‘5’, whose calls are routed over any trunk group, have the extension sent to the Nuance SA for proper CLID handling.

change private-numbering 0		Page 1 of 2	
<b>NUMBERING – PRIVATE FORMAT</b>			
Ext	Ext	Trk	Private
Len	Code	Grp(s)	Prefix
5	5		
		Total	
		Len	
		5	Total Administered: 2
			Maximum Entries: 540

In the **AAR Digit Analysis Table**, specify a **Dialed String** that would match the Nuance SA extension (70000) and specify the **Route Pattern** that will be used to route the call.

In the **Route Pattern** form, specify the SIP trunk group for **Grp No**, and set the **Numbering Format** for the route preference to *lev0-pvt*.

10 of 29  
NuanceSA11CMSM

## 6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain.
- Logical/physical Locations that can be occupied by SIP Entities.
- SIP Entities corresponding to Session Manager and Communication Manager.
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities.
- Routing Policies, which control call routing between the SIP Entities.
- Dial Patterns, which govern to which SIP Entity a call is routed.

Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL “https://<ip-address>”, where <ip-address> is the IP address of System Manager. Log in with the appropriate credentials (not shown).

### 6.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **Domains** on the left and clicking the **New** button on the right (not shown). The following screen will then be shown. Fill in the following:

- **Name:** The authoritative domain name (e.g., *avaya.com*).
- **Notes:** Descriptive text (optional).

Click **Commit**.

Since the sample configuration does not deal with any other domains, no additional domains need to be added.

The screenshot shows the Avaya Aura System Manager 6.3 web interface. The top header includes the Avaya logo, the title 'Avaya Aura® System Manager 6.3', and a user status bar indicating 'Last Logged on at March 28, 2013 1:48 PM' with links for 'Help', 'About', 'Change Password', and 'Log off admin'. The navigation sidebar on the left lists various configuration categories: Routing, Domains (selected), Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Domain Management' and contains a table with one entry: 'avaya.com' under the 'Name' column, 'sip' under the 'Type' column, and an empty 'Notes' column. There are 'Commit' and 'Cancel' buttons at the top right and bottom right of the table area.

Name	Type	Notes
* avaya.com	sip	

## 6.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, select **Locations** on the left and then click the **New** button on the right (not shown). The following screen will then be shown. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

Under *Location Pattern*:

- **IP Address Pattern:** A pattern used to logically identify the location.
- **Notes:** Descriptive text (optional).

The screen below shows addition of a location which includes the Communication Manager, Session Manager, and Speech Attendant. Click **Commit** to save the Location definition.

**AVAYA** Avaya Aura® System Manager 6.3

Last Logged on at March 29, 2013 11:04 AM  
Help | About | Change Password | Log off admin

Routing x Home

Home / Elements / Routing / Locations

Location Details Commit Cancel

Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth. Note: If this setting is disabled, you should return to this form to review settings for multimedia bandwidth.  
See Session Manager -> Session Manager Administration -> Global Settings

**General**

\* Name: .21 and .101 Subnet

Notes:

**Overall Managed Bandwidth**

Managed Bandwidth Units: Kbit/sec

Total Bandwidth:

**Per-Call Bandwidth Parameters**

\* Default Audio Bandwidth: 80 Kbit/sec

**Alarm Threshold**

Audio Alarm Threshold: 80 %

\* Latency before Audio Alarm Trigger: 5 Minutes

**Location Pattern**

Add Remove

2 Items Refresh Filter: Enable

IP Address Pattern	Notes
* 10.64.21.*	
* 10.64.101.*	

Select : All, None

Commit Cancel

## 6.3. Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager, Communication Manager, and Nuance Speech Attendant.

### 6.3.1. Avaya Aura® Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right (not shown). The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

**Note:** The **Ports** section of this screen (not shown) is assumed to have been previously configured.

The screenshot displays the Avaya Aura System Manager 6.3 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.3", and a user status area indicating "Last Logged on at March 28, 2013 1:48 PM" with links for "Help | About | Change Password | Log off admin". The main interface features a left-hand menu with options: Routing, Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled "Home / Elements / Routing / SIP Entities" and contains the "SIP Entity Details" form. The form has a "General" tab and includes fields for: Name (SM\_21\_31), FQDN or IP Address (10.64.21.31), Type (Session Manager), Notes, Location (.21 and .101 Subnet), Outbound Proxy, Time Zone (America/Denver), and Credential name. At the bottom, there is a "SIP Link Monitoring" section with a dropdown menu set to "Use Session Manager Configuration". Buttons for "Commit" and "Cancel" are located at the top right of the form area.

### 6.3.2. Avaya Aura® Communication Manager

A SIP Entity must be added for the Communication Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right (not shown). The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., procr board) on the telephony system.
- **Type:** Select *CM*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

**AVAYA** Avaya Aura® System Manager 6.3

Last Logged on at March 28, 2013 1:48 PM  
Help | About | Change Password | Log off admin

Routing \* Home

Home / Elements / Routing / SIP Entities

SIP Entity Details

Commit Cancel

**General**

\* Name: CM\_21\_41

\* FQDN or IP Address: 10.64.21.41

Type: CM

Notes: Evolution Server - 8300D

Adaptation:

Location: .21 and .101 Subnet

Time Zone: America/Denver

Override Port & Transport with DNS SRV: ☐

\* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: both

**SIP Link Monitoring**

SIP Link Monitoring: Use Session Manager Configuration

### 6.3.3. Nuance Speech Attendant

A SIP Entity must be added for Nuance SA. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right (not shown). The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** Nuance SA IP address.
- **Type:** Select *SIP Trunk*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

Defaults may be used for the remaining fields. Click **Commit** to save each SIP Entity definition.



Avaya Aura® System Manager 6.3

Last Logged on at March 28, 2013 1:48 PM  
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

## 6.4. Add Entity Link

The SIP trunk from Session Manager to Communication Manager and Nuance SA are described by Entity Links. To add an Entity Link, select **Entity Links** on the left and click on the **New** button on the right (not shown). Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name.
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of Communication Manager or Nuance SA.
- **Port:** Port number on which the other system receives SIP requests.
- **Trusted:** Select *Trusted*.

The following screens display the two Entity Links. The first entity link is for Session Manager and Communication Manager. The second entity link is for Session Manager and Nuance SA.



Avaya Aura® System Manager 6.3

Last Logged on at March 28, 2013 1:48 PM  
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service	Notes
* CM_21_41	* SM_21_31	TLS	* 5061	* CM_21_41	* 5061	Trusted	<input type="checkbox"/>	



Avaya Aura® System Manager 6.3

Last Logged on at March 28, 2013 1:48 PM  
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service	Notes
* SA Link	* SM_21_31	UDP	* 5060	* SpeechAttendant	* 5060	Trusted	<input type="checkbox"/>	



## 6.5. Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities. Two routing policies were added – one for Communication Manager and one for Nuance SA. To add a routing policy, select **Routing Policies** on the left and click on the **New** button on the right (not shown). The following screen is displayed. Fill in the following:

Under *General*:

Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:

Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screen shows the Routing Policy for Communication Manager.

The screenshot displays the Avaya Aura System Manager 6.3 web interface. The left sidebar shows the navigation menu with 'Routing Policies' selected. The main content area is titled 'Home / Elements / Routing / Routing Policies'. It contains a 'Routing Policy Details' section with a 'General' tab. The 'General' tab has fields for 'Name' (CM\_21\_41), 'Disabled' (unchecked), 'Retries' (0), and 'Notes'. Below this is the 'SIP Entity as Destination' section with a 'Select' button and a table showing the selected entity: CM\_21\_41, 10.64.21.41, CM, Evolution Server - 8300D. The 'Time of Day' section has 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. It shows a table with one item: Ranking 0, Name 24/7, with checkboxes for Mon, Tue, Wed, Thu, Fri, Sat, Sun, all checked. The start time is 00:00 and the end time is 23:59. The filter is set to 'Enable'.

AVAYA Avaya Aura® System Manager 6.3

Last Logged on at March 28, 2013 1:48 PM  
Help | About | Change Password | Log off admin

Routing \* Home

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit Cancel

General

\* Name: CM\_21\_41

Disabled: ☐

\* Retries: 0

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CM_21_41	10.64.21.41	CM	Evolution Server - 8300D

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

The following screen shows the Routing Policy for Nuance SA. Note in the example below, this routing policy was given a **Ranking** of 1. The ranking will be discussed in the next section.



[Routing](#) \* [Home](#)

[Home](#) / [Elements](#) / [Routing](#) / [Routing Policies](#)

[Help ?](#)

### Routing Policy Details

#### General

\* Name:

Disabled: ☐

\* Retries:

Notes:

#### SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
SpeechAttendant	10.64.101.102	SIP Trunk	

#### Time of Day

1 Item [Refresh](#)
Filter: [Enable](#)

<input type="checkbox"/>	Ranking <a href="#">1</a> ▲	Name <a href="#">2</a> ▲	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	<input type="text" value="1"/>	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : [All](#), [None](#)

## 6.6. Add Dial Patterns

Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration, 5-digit extensions beginning with “5” reside on Communication Manager, extension “70000” is the Nuance SA number. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button on the right (not shown). Fill in the following:

Under *General*:


- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain of dial pattern.

Under *Originating Locations and Routing Policies*:

Click **Add**, and then select the appropriate location and routing policy from the list.

Default values can be used for the remaining fields. The following screen shows the dial pattern definitions for local extensions on Communication Manager. Click **Commit** to save this dial pattern.

The following screen shows the dial pattern definition for Communication Manager extensions beginning with “5”.

Avaya Aura® System Manager 6.3

Last Logged on at March 28, 2013 1:48 PM  
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / Dial Patterns

Dial Pattern Details

CommitCancel

Help ?

General

\* Pattern: 5

\* Min: 5

\* Max: 5

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: avaya.com

Notes:

Originating Locations and Routing Policies

AddRemove

1 Item Refresh

Filter: Enable

<input type="checkbox"/>	Originating Location Name 1	Originating Location Notes	Routing Policy Name	Rank 2	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any originating location	CM_21_41		<input type="checkbox"/>	CM_21_41	

Select : All, None

The following screen shows the dial pattern definition for the Nuance SA number (70000). As mentioned in **Section 6.5**, the *ToSpeechAttendant* routing policy was assigned a **Rank** of 1. If desired, additional routing policies may be created with the desired ranks (the lower the rank number, the higher the priority). In the example below, under **Originating Locations and Routing Policies**, a second routing policy (AAM\_21\_72) is assigned to this dial pattern. The AAM\_21\_72 routing policy was assigned a rank of 2 (not shown). When a call to extension 70000 is dialed, Session Manager will first use the routing policy with the lowest rank and route the call to Nuance SA. In **Section 7.1**, Nuance SA was configured to send a 503 SIP message when all the SA ports are busy. As a result, when Session Manager receives a 503 SIP message response, Session Manager will then attempt to reroute the call using the next lowest ranked routing policy.



## Avaya Aura® System Manager 6.3

Last Logged on at April 12, 2013 1:44 PM  
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Routing](#) \* [Home](#)

[Home](#) / [Elements](#) / [Routing](#) / [Dial Patterns](#)

- ▼ Routing
- Domains
- Locations
- Adaptations
- SIP Entities
- Entity Links
- Time Ranges
- Routing Policies
- Dial Patterns**
- Regular Expressions
- Defaults

[Commit](#) [Cancel](#)

### Dial Pattern Details

**General**

\* **Pattern:**

\* **Min:**

\* **Max:**

**Emergency Call:** ☐

**Emergency Priority:**

**Emergency Type:**

**SIP Domain:**

**Notes:**

### Originating Locations and Routing Policies

[Add](#) [Remove](#)
2 Items [Refresh](#)
Filter: [Enable](#)

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any originating location	ToSpeechAttendant		<input type="checkbox"/>	SpeechAttendant	
<input type="checkbox"/>	-ALL-	Any originating location	AAM_21_72		<input type="checkbox"/>	AAM_21_72	Avaya Aura Messaging

Select : All, None

### Denied Originating Locations

[Add](#) [Remove](#)
0 Items [Refresh](#)
Filter: [Enable](#)

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

[Commit](#) [Cancel](#)

## 7. Configure Nuance Speech Attendant

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

- Configure SIP interface in the Configuration Panel.
- Configure the number of SIP ports supported by Nuance SA.

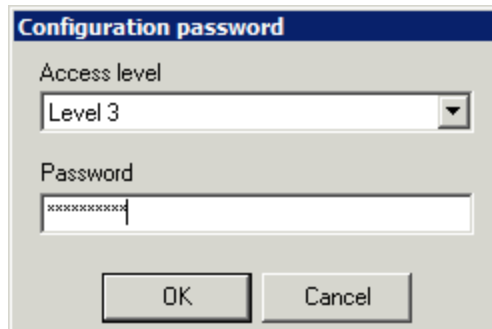
**Note:** Configuration of transfer entries and menus in Nuance SA is outside the scope of these Application Notes and will not be covered.

Nuance SA is configured through **Admin Tools** which can be started by navigating to **Start→Programs→Speech Attendant→Admin Tools** (not shown). The initial screen is displayed below.



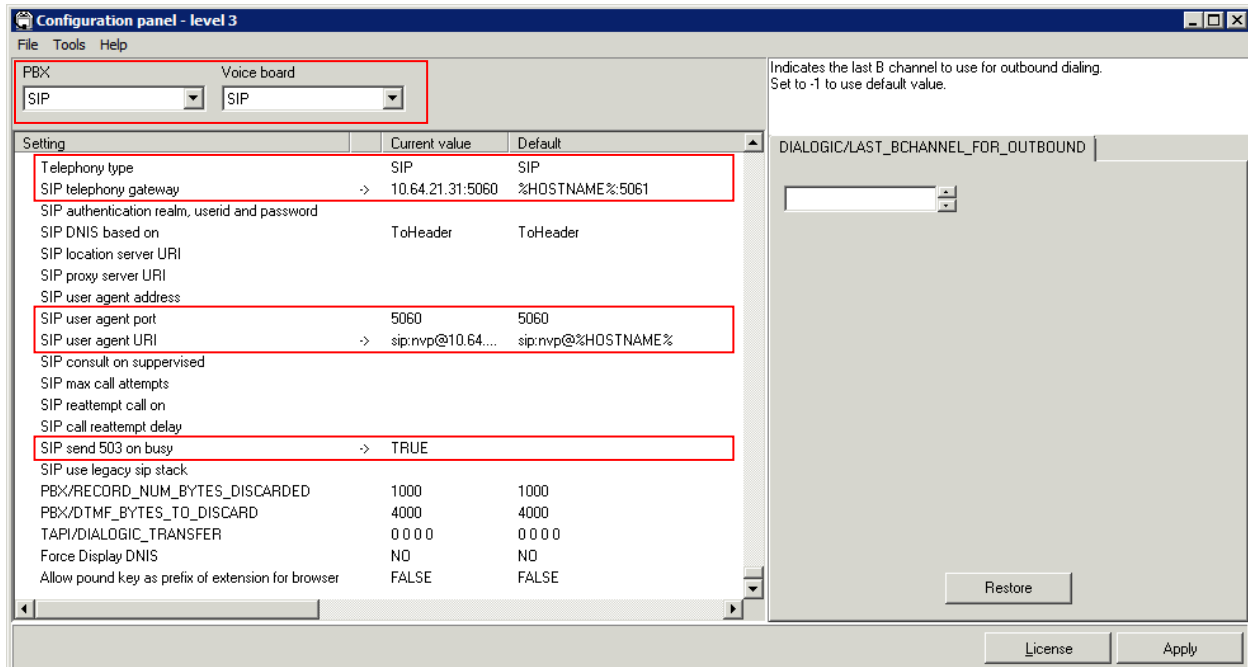
## 7.1. Configure SIP Interface in Configuration Panel

To open the **Configuration Panel**, click on this option in the **Admin Tools** window shown above. The login prompt will be displayed to the user as shown below. Log in with the appropriate credentials using *Level 3* access level.



A dialog box titled "Configuration password" with a blue header. It contains two input fields: "Access level" with a dropdown menu showing "Level 3", and "Password" with a text box containing "XXXXXXXX". At the bottom are "OK" and "Cancel" buttons.

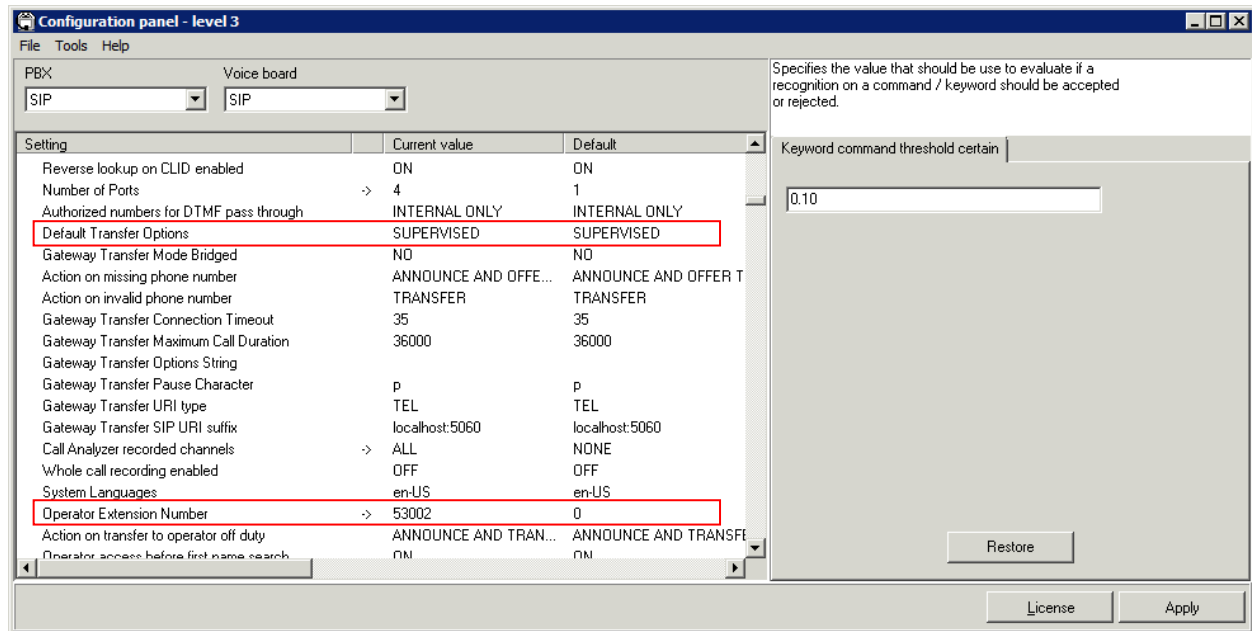
The **Configuration Panel** shown below is displayed. The **Configuration Panel** allows the SIP interface, transfer mode, and operator extension number to be configured. SA supports blind and supervised transfers. To configure the SIP interface, set the **PBX** and **Voice Board** fields to *SIP*. Next, set the **Telephony type** field to *SIP* and specify the Session Manager IP address and port in the **SIP telephony gateway** field. Configure the SIP port that SA listens on in the **SIP user agent port** field and specify the **SIP user agent URI** for SA. Set **SIP send 503 on busy** to *TRUE*.



A screenshot of the "Configuration panel - level 3" window. The window has a menu bar (File, Tools, Help) and a title bar. The main area is divided into several sections. At the top, there are two dropdown menus: "PBX" and "Voice board", both set to "SIP". Below these is a table with columns "Setting", "Current value", and "Default". The table lists various settings, with several rows highlighted by red boxes. To the right of the table is a section titled "DIALOGIC/LAST\_BCHANNEL\_FOR\_OUTBOUND" with a text box and a "Restore" button. At the bottom of the window are "License" and "Apply" buttons.

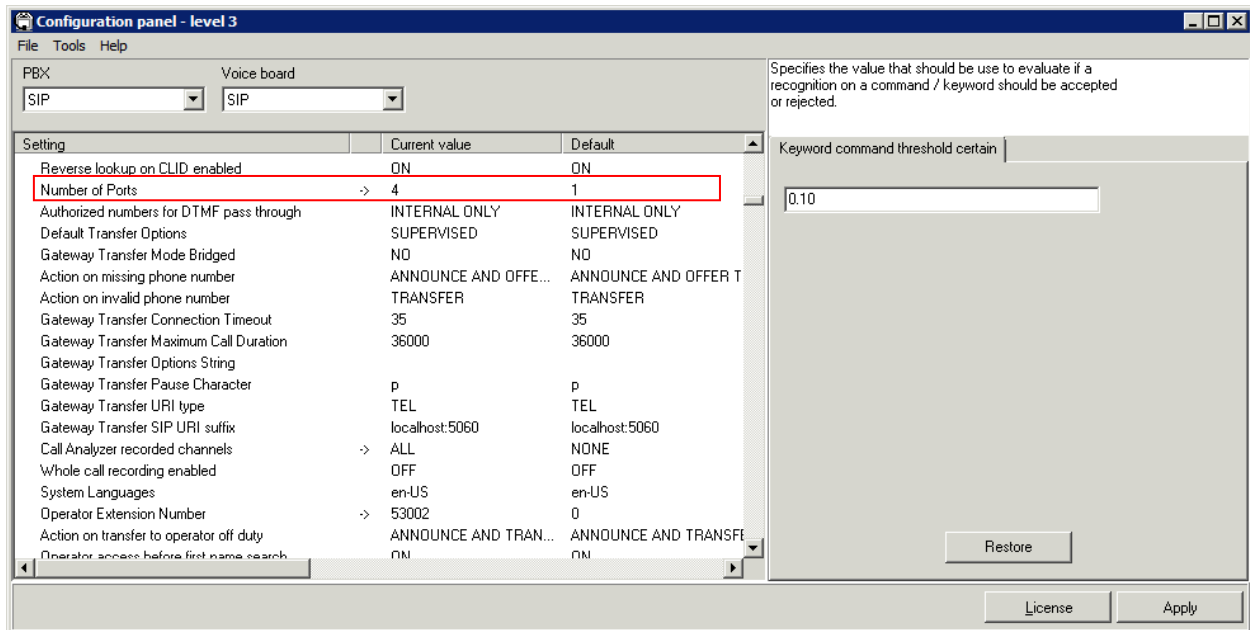
Setting	Current value	Default
Telephony type	SIP	SIP
SIP telephony gateway	10.64.21.31:5060	%HOSTNAME%:5061
SIP authentication realm, userid and password	ToHeader	ToHeader
SIP DNIS based on	ToHeader	ToHeader
SIP location server URI		
SIP proxy server URI		
SIP user agent address		
SIP user agent port	5060	5060
SIP user agent URI	sip:nvp@10.64...	sip:nvp@%HOSTNAME%
SIP consult on supervised		
SIP max call attempts		
SIP reattempt call on		
SIP call reattempt delay		
SIP send 503 on busy	TRUE	
SIP use legacy sip stack		
PBX/RECORD_NUM_BYTES_DISCARDED	1000	1000
PBX/DTMF_BYTES_TO_DISCARD	4000	4000
TAPI/DIALOGIC_TRANSFER	0 0 0 0	0 0 0 0
Force Display DNIS	NO	NO
Allow pound key as prefix of extension for browser	FALSE	FALSE

In the **Configuration Panel**, specify the transfer mode (supervised or unsupervised) in the **Default Transfer Options** field, and specify the **Operator Extension Number**, which should be set to a valid extension on Communication Manager.



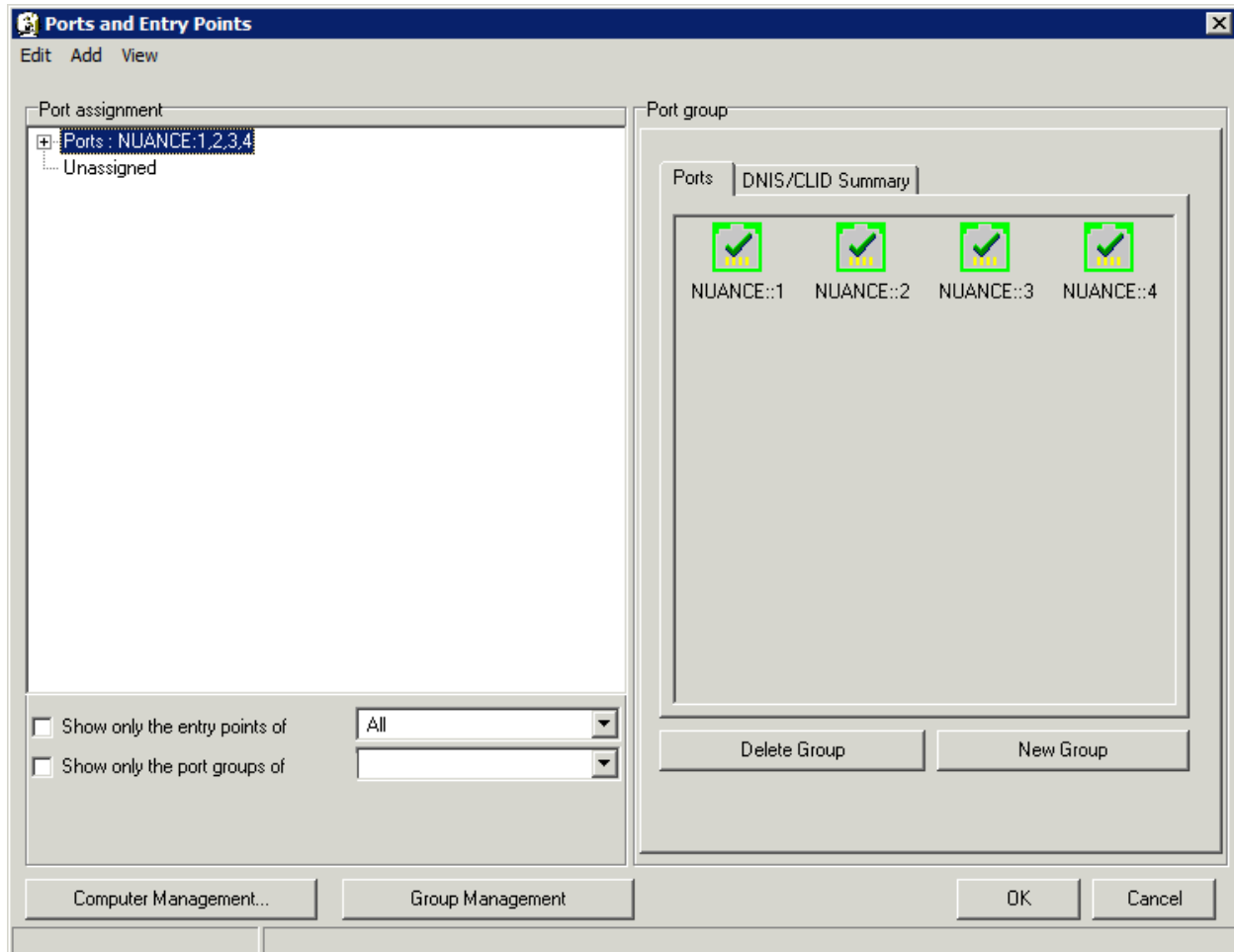
## 7.2. Configure Number of Ports Supported

In the **Configuration Panel**, set the **Number of Ports** field to the desired value. In this example, 4 ports were configured. Click **Apply** and then close the **Configuration Panel**. Allow the SA application to be restarted when prompted.

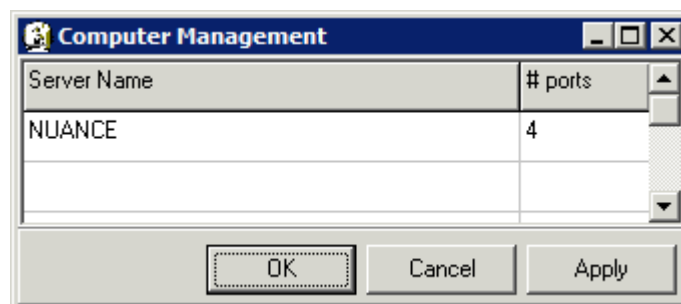




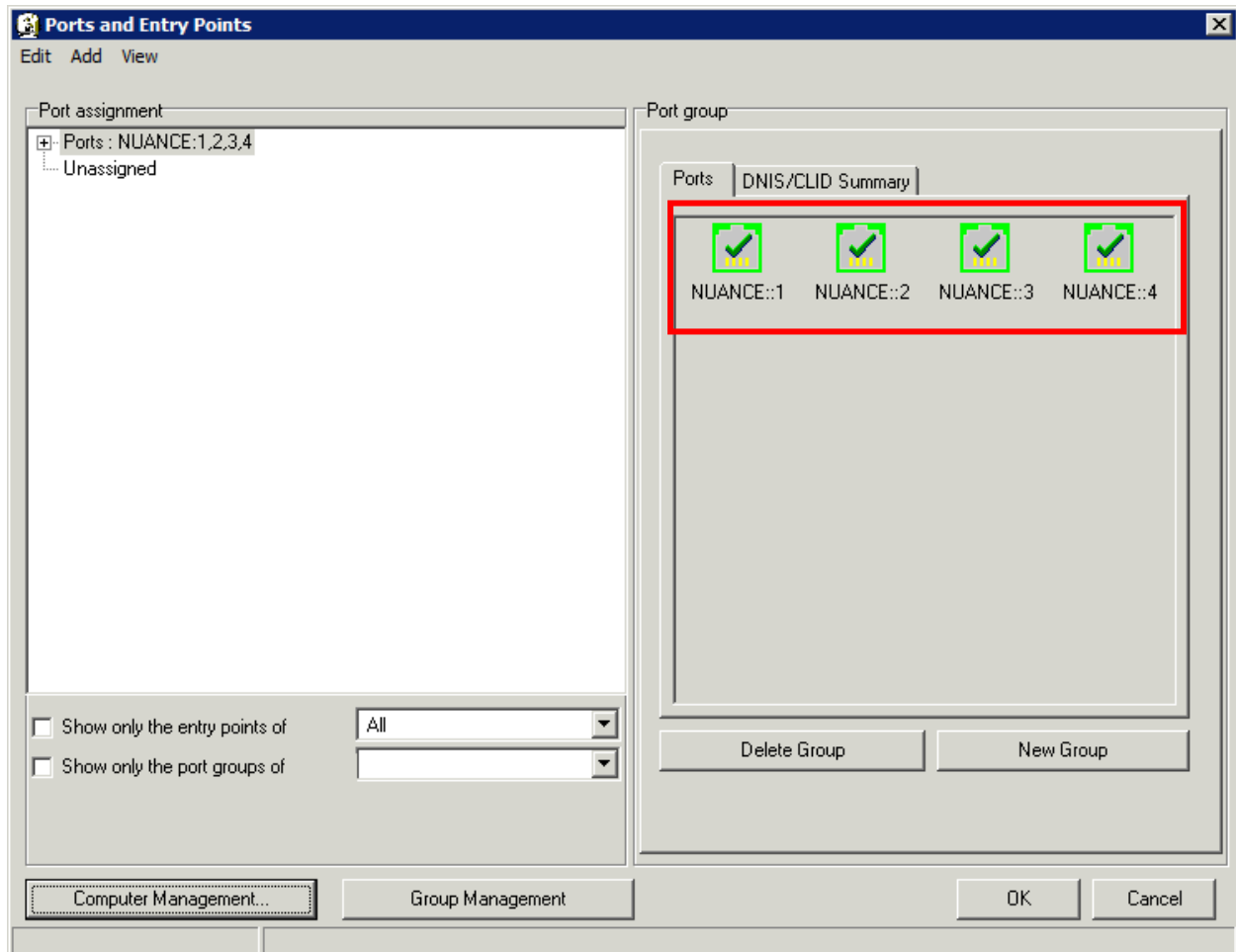
From **Admin Tools**, click on the **Phone Directory and Menu Editor** option and login with the appropriate credentials. In the **Menu Editor** (not shown), navigate to **Edit→Ports and Entry Points** to display the window below. Once the **Ports and Entry Points** window is opened, click on the **Computer Management** button at the bottom of the screen.



In the **Computer Management** window, set the # ports column to the desired value. In this example, 4 ports were configured. Click **OK**.



Under the Port group section of the **Ports and Entry Points** window, a list of ports in the system will be displayed. Make sure they are all green (enabled). If the ports are grey (disabled), click on each icon to enable them. Click **OK**.



Next, close the **Configuration Panel**. Allow the system to restart when prompted.

## 8. Verification Steps

This section provides the verification steps that may be performed to verify that Nuance SA is operating properly with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP integration.

1. Place a call to Nuance SA. From the Nuance SA server, open the SA **Monitor** from **Admin Tools** (refer to **Section 7** for accessing **Admin Tools**) and login with the appropriate credentials. Verify that SA detects an active call as shown below. For example, the first entry indicates that the call is *in progress*.

### SpeechAttendant®

Hosted on Nuance

**Description:** Auto Attendant **Version:** SA 11.0.0 (with E01) (latest hot fix installed 1100HF33, 1100HF34, 1100HF35, 1100HF36, 1100HF37, 1100HF38, 1100HF39, 1100HF40, 1100HF41, 1100HF42, 1100HF43, 1100HF44)

**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: 02/27/2013  
Date to: 02/27/2013

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: <= seconds  
all types  
last 50 calls  
Submit

Logs (from database), 5 sessions

Call start	DNIS	CLID	Call complexity	Call duration	Error	Termination code	Destination	System comment	Tagging
27/02/2013 11:07:36	70000	53003	0	0		...		in progress	
27/02/2013 11:06:22	70000	53003	5	11		SRI1	Kent, Clark	transfer completed	
27/02/2013 11:05:58	70000	53004	0	21		HAFNF	Toplevel Menu	caller hangup	
27/02/2013 11:04:42	70000	53004	5	12		SRI1	Croft, Lara	transfer completed	
27/02/2013 11:04:03	70000	53004	5	13		ORV	Operator	transfer completed	

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

## 9. Conclusion

These Application Notes describe the configuration steps required to integrate Nuance Speech Attendant with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP integration. All feature and serviceability test cases were completed successfully. Refer to **Section 2.2** for test results and any observations.

## 10. Additional References

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

- [1] *Administering Avaya Aura® Communication Manager*, December 2012, Release 6.2, Issue 7.0, Document Number 03-300509.
- [2] *Administering Avaya Aura® Session Manager*, December 2012, Release 6.3, Document.

Nuance product documentation is available at <https://network.nuance.com/portal/server.pt>.

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