



Avaya Solution & Interoperability Test Lab

Application Notes for Nuance Speech Attendant with Avaya Aura® Session Manager and Avaya Aura® Communication Manager – 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 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 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 transfer to the requested party.

1.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. Feature testing focused on that Nuance SA can successfully recognize spoken names and extensions entered via DTMF and transfer the call to the correct destination. Blind and supervised transfers were verified. Other features covered included: DNIS and CLID handling, adding new transfer entries, recording caller utterances, and accessing Maintenance Mode and Personal Administration Mode to record name and change PIN. In addition, failover support was also tested, which verified the ability to re-route the call to Nuance SA to an alternate destination if it was busy or unavailable.

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.

1.2. Support

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

- **Web:** www.network.nuance.com
- **Email:** 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 Speech Attendant (SA) solution with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Nuance SA is deployed on a dedicated server running Windows 2003 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.

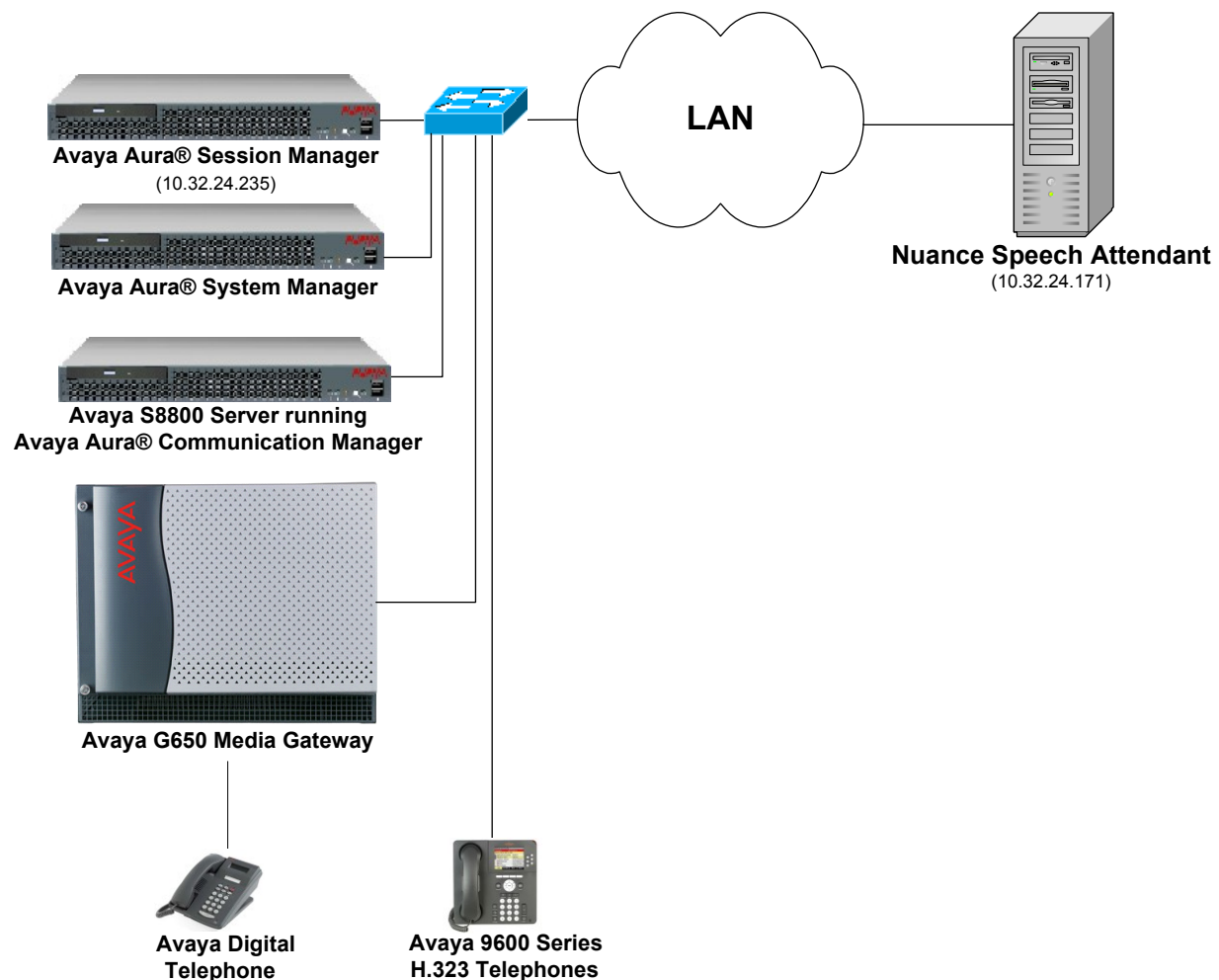


Figure 1: Configuration with Nuance Speech Attendant in an Avaya SIP Network

2.1. Equipment and Software Validated

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

Equipment	Software
Avaya S8800 Server with a G650 Media Gateway	Avaya Aura® Communication Manager 6.0 with Service Pack 1
Avaya Aura® Session Manager	6.0 (6.0.0.0.600020)
Avaya Aura® System Manager	6.0 (6.0.0.0.556-3.0.6.1)
Avaya 9600 Series IP Telephones	3.110b (H.323)
Nuance Speech Attendant (SA)	11.0 with Hotfix 5

3. Configure Avaya Aura® Communication Manager

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

- Verify Communication Manager license
- 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.

3.1. 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 system-parameters customer-options		Page 2 of 11
OPTIONAL FEATURES		
IP PORT CAPACITIES	USED	
Maximum Administered H.323 Trunks:	12000	40
Maximum Concurrently Registered IP Stations:	18000	17
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:	18000	0
Maximum Video Capable IP Softphones:	18000	0
Maximum Administered SIP Trunks:	24000	30
Maximum Administered Ad-hoc Video Conferencing Ports:	24000	0
Maximum Number of DS1 Boards with Echo Cancellation:	522	0
Maximum TN2501 VAL Boards:	128	1
Maximum Media Gateway VAL Sources:	250	0
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.)		

3.2. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for the C-LAN board in the G650 Media Gateway, and the Session Manager SIP interface. The host names will be used throughout the other configuration screens of Communication Manager.

change node-names ip	Page 1 of 2
IP NODE NAMES	
Name	IP Address
Gateway001	10.32.24.1
ModMsg	192.50.10.45
clancrm	10.32.24.20
default	0.0.0.0
devcon-asm	10.32.24.235
medprocrm	10.32.24.21
procr	10.32.24.10
procr6	::
(8 of 8 administered node-names were displayed)	
Use 'list node-names' command to see all the administered node-names	
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name	

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. 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 G650 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 **IP 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
IP NETWORK REGION	
Region: 1	
Location: 1	Authoritative Domain: avaya.com
Name:	
MEDIA PARAMETERS	Intra-region IP-IP Direct Audio: no
Codec Set: 1	Inter-region IP-IP Direct Audio: no
UDP Port Min: 2048	IP Audio Hairpinning? y
UDP Port Max: 3029	
DIFFSERV/TOS PARAMETERS	
Call Control PHB Value: 34	
Audio PHB Value: 46	
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	

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 '1' shown above. The default settings of the **IP Codec Set** form are shown below. Nuance SA supports G.711.

change ip-codec-set 1

Page1 of 2

IP Codec Set

Codec Set: 1

	Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1:	G.711MU	n	2	20
2:				
3:				
4:				
5:				
6:				
7:				

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 *tcp*.
- Specify the C-LAN 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 TCP port value of *5060* 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 50		Page 1 of 1
SIGNALING GROUP		
<div style="display: flex; justify-content: space-between;"> <div> Group Number: 50 IMS Enabled? n Q-SIP? n IP Video? n Peer Detection Enabled? y </div> <div> Group Type: sip Transport Method: tcp Peer Server: SM </div> <div> SIP Enabled LSP? n Enforce SIPS URI for SRTP? y </div> </div>		
<div style="display: flex; justify-content: space-between; margin-top: 20px;"> <div> Near-end Node Name: clancrm Near-end Listen Port: 5060 </div> <div> Far-end Node Name: devcon-asm Far-end Listen Port: 5060 Far-end Network Region: 1 </div> </div>		
Far-end Domain: avaya.com		
<div style="display: flex; justify-content: space-between;"> <div> Incoming Dialog Loopbacks: eliminate DTMF over IP: rtp-payload Session Establishment Timer(min): 3 Enable Layer 3 Test? n H.323 Station Outgoing Direct Media? n </div> <div> Bypass If IP Threshold Exceeded? n RFC 3389 Comfort Noise? n Direct IP-IP Audio Connections? n IP Audio Hairpinning? n Initial IP-IP Direct Media? n Alternate Route Timer(sec): 6 </div> </div>		

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 50		Page 1 of 21	
TRUNK GROUP			
Group Number: 50	Group Type: sip	CDR Reports: y	
Group Name: To devcon-asm	COR: 1	TN: 1	TAC: 1050
Direction: two-way	Outgoing Display? n		
Dial Access? n	Night Service:		
Queue Length: 0			
Service Type: tie	Auth Code? n		
		Member Assignment Method: auto	
		Signaling Group: 50	
		Number of Members: 10	

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 50		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Numbering Format: private			
		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 **Private Numbering 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 '7' whose calls are routed over any trunk group, including SIP trunk group "50", have the extension sent to the Nuance SA for proper CLID handling.

change private-numbering 0		Page 1 of 2	
NUMBERING - PRIVATE FORMAT			
Ext	Ext	Trk	Private
Len	Code	Grp (s)	Prefix
5	7		
		Total	
		Len	
		5	Total Administered: 1
		Maximum Entries: 540	

3.3. Configure AAR Call Routing to Nuance Speech Attendant

In the **AAR Digit Analysis Table**, specify a **Dialed String** that would match the Nuance SA extension (22005) and specify the **Route Pattern** that will be used to route the call.

change aar analysis 7							Page 1 of 2
AAR DIGIT ANALYSIS TABLE							
Location: all							Percent Full: 1
	Dialed String	Total Min Max	Route Pattern	Call Type	Node Num	ANI Req'd	
	22005	5 5	50	aar		n	

In the *Route Pattern* form, specify the SIP trunk group and set the **Numbering Format** for the route preference to *unk-unk*. This would prevent a + sign to be prepended to the dialed digits in conjunction with using private numbering format in the SIP trunk group.

change route-pattern 50														Page	1 of	3						
Pattern Number: 50														Pattern Name: To devcon-asm								
SCCAN? n														Secure SIP? n								
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted							DCS/	IXC							
No			Mrk	Lmt	List	Del	Digits							QSIG								
							Dgts							Intw								
1:	50	0											n	user								
2:											n	user										
3:											n	user										
4:											n	user										
5:											n	user										
6:											n	user										
BCC VALUE														TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	No.	Numbering	LAR
0		1	2	M	4	W	Request								Dgts	Format						
														Subaddress								
1:	y	y	y	y	y	n	n	rest						unk-unk	none							
2:	y	y	y	y	y	n	n	rest							none							
3:	y	y	y	y	y	n	n	rest							none							
4:	y	y	y	y	y	n	n	rest							none							
5:	y	y	y	y	y	n	n	rest							none							
6:	y	y	y	y	y	n	n	rest							none							

4. 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
- Session Manager, corresponding to the Session Manager Server to be managed by System Manager

Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL “https://<ip-address>/SMGR”, where <ip-address> is the IP address of System Manager. Log in with the appropriate credentials.

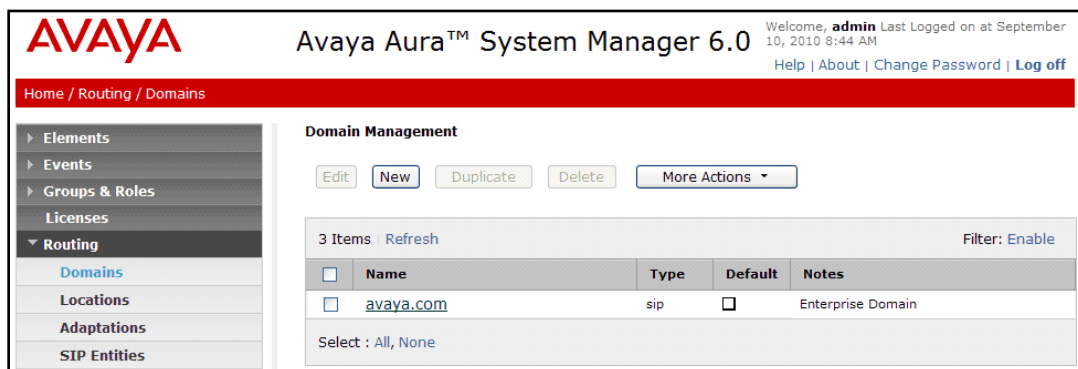
4.1. Specify SIP Domain

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. 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.



4.2. 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 click on the **New** button (not shown) on the right. 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 the *BR-DevConnect* location, which includes the Communication Manager and Session Manager. Click **Commit** to save the Location definition.

AVAYA Avaya Aura™ System Manager 6.0 Welcome, **admin** Last Logged on at September 10, 2010 8:44 AM
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Routing / Locations / Location Details

Location Details Commit Cancel

General

* **Name:**

Notes:

Managed Bandwidth: Kbit/sec

* **Average Bandwidth per Call:** Kbit/sec

Location Pattern

Add Remove

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

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.32.24.*	<input type="text"/>

Select : All, None

* **Input Required** Commit Cancel

Help

[Help for Locations Details fields](#)

[Help for Committing configuration changes](#)

4.3. Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager, the C-LAN in the G650 Media Gateway, and Nuance Speech Attendant

4.3.1. 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 (not shown) on the right. 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 one of the locations defined previously.
- **Time Zone:** Time zone for this location.

AVAYA Avaya Aura™ System Manager 6.0 Welcome, **admin** Last Logged on at September 10, 2010 8:44 AM
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Routing / SIP Entities / SIP Entity Details

SIP Entity Details Commit Cancel

General

* **Name:** devcon-asm

* **FQDN or IP Address:** 10.32.24.235

Type: Session Manager

Notes:

Location: BR-DevConnect

Outbound Proxy:

Time Zone: America/New_York

Credential name:

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

4.3.2. 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. 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., C-LAN board) on the telephony system.
- **Type:** Select *CM*.
- **Location:** Select one of the locations 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 6.0 Welcome, **admin** Last Logged on at August 31, 2010 1:45 PM

[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Routing / SIP Entities / SIP Entity Details

SIP Entity Details Commit Cancel

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Adaptation:

Location:

Time Zone:

Override Port & Transport with DNS SRV: ☐

* SIP Timer B/F (in seconds):

Credential name:

Call Detail Recording:

SIP Link Monitoring

SIP Link Monitoring:

Entity Links

0 Items [Refresh](#) Filter: Enable

	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted
--	--------------	----------	------	--------------	------	---------

* Input Required Commit Cancel

4.3.3. 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 (not shown) on the right. 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 *Other*.
- **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 Avaya Aura™ System Manager 6.0 Welcome, **admin** Last Logged on at January 14, 2011 9:29 AM
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Routing / SIP Entities / SIP Entity Details

SIP Entity Details Commit Cancel

General

* **Name:**

* **FQDN or IP Address:**

Type:

Notes:

Adaptation:

Location:

Time Zone:

Override Port & Transport with DNS SRV: ☐

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

Credential name:

Call Detail Recording:

SIP Link Monitoring

SIP Link Monitoring:

4.4. 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. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *devcon13 Link* or *SA Link*).
- **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:** Check this box. *Note: If this box is not checked, calls from the associated SIP Entity specified in Section 4.4 will be denied.*

The following screen displays two Entity Links after being configured per the instructions above. The first entity link is for Session Manager and Communication Manager and the second entity link is for Session Manager and Nuance SA.

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at January 14, 2011 9:29 AM

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Home / Routing / Entity Links

Entity Links

[Edit](#) [New](#) [Duplicate](#) [Delete](#) [More Actions](#)

13 Items [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
<input type="checkbox"/>	devcon13 Link	devcon-asm	TCP	5060	devcon13	5060	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	SA Link	devcon-asm	UDP	5060	SpeechAttendant	5060	<input checked="" type="checkbox"/>	

Select : All, None

4.5. 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 (not shown) on the right. 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.

Under *Dial Patterns*:

Enter the dialed digits that would be routed to the specified SIP entity.

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 6.0 interface. The left sidebar shows a navigation menu with 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and includes a 'Commit' button. The 'General' section has a 'Name' field set to 'To devcon13', a 'Disabled' checkbox, and a 'Notes' field. The 'SIP Entity as Destination' section has a 'Select' button. Below this is a table with one row: 'devcon13' with FQDN or IP Address '10.32.24.20' and Type 'CM'. The 'Time of Day' section has 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. It shows a table with one row for '24/7' with checkboxes for all days of the week and a time range of '00:00' to '23:59'. The 'Dial Patterns' section has 'Add' and 'Remove' buttons. It shows a table with one row for pattern '77' with Min '5', Max '5', and Originating Location 'BR-DevConnect'. A 'Help' section is visible at the bottom left.

AVAYA Avaya Aura™ System Manager 6.0 Welcome, **admin** Last Logged on at January 14, 2011 9:29 AM [Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Routing / Routing Policies / Routing Policy Details

Routing Policy Details [Commit](#) [Cancel](#)

General

* Name:

Disabled: ☐

Notes:

SIP Entity as Destination

[Select](#)

Name	FQDN or IP Address	Type	Notes
devcon13	10.32.24.20	CM	

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

Dial Patterns

[Add](#) [Remove](#)

1 Item Refresh Filter: Enable

Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
77	5	5	<input type="checkbox"/>	avaya.com	BR-DevConnect	Avaya CM

Select : All, None

Help

- Help for Routing Policy Details fields
- Help for SIP Entity List
- Help for Time Range List
- Help for Pattern List
- Help for Regular Expressions List
- Help for Committing configuration changes

The following screen shows the Routing Policy for Nuance SA.

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at January 14, 2011 9:29 AM
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Routing / Routing Policies / Routing Policy Details

Elements
Events
Groups & Roles
Licenses
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Security
System Manager Data
Users

Help

Help for Routing Policy Details fields
Help for SIP Entity List
Help for Time Range List
Help for Pattern List
Help for Regular Expressions List
Help for Committing configuration changes

Routing Policy Details

CommitCancel

General

* Name: ToSpeechAttendant

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
SpeechAttendant	10.32.24.171	Other	

Time of Day

AddRemoveView Gaps/Overlaps

1 Item RefreshFilter: Enable

<input type="checkbox"/>	Ranking 1	Name 2	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	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

Dial Patterns

AddRemove

1 Item RefreshFilter: Enable

<input type="checkbox"/>	Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
<input type="checkbox"/>	22	5	5	<input type="checkbox"/>	avaya.com	BR-DevConnect	Nuance Speech Attendant

Select : All, None

4.6. 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 “77” reside on Communication Manager, extension “22005” is the Nuance SA number. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. 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.
- **Notes** Comment on purpose 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 “77”.

AVAYA Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at January 14, 2011 9:29 AM

Help | About | Change Password | Log off

Home / Routing / Dial Patterns / Dial Pattern Details

Dial Pattern Details [Commit] [Cancel]

General

* Pattern: 77

* Min: 5

* Max: 5

Emergency Call: ☐

SIP Domain: avaya.com

Notes: Avaya CM

Originating Locations and Routing Policies

[Add] [Remove]

1 Item Refresh

<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"/>	BR-DevConnect	Juan's Subnet(s)	To devcon13	0	<input type="checkbox"/>	devcon13	

Select : All, None

Denied Originating Locations

[Add] [Remove]

0 Items Refresh

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

* Input Required

[Commit] [Cancel]

The following screen shows the dial pattern definition for the Nuance SA number (22005).

AVAYA

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at January 14, 2011 9:29 AM

[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Routing / Dial Patterns / Dial Pattern Details

Elements

Events

Groups & Roles

Licenses

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Security

System Manager Data

Users

Help

Help for Dial Pattern Details fields

Help for Location and Routing Policy Lists

Help for Denied Location fields

Help for Committing configuration changes

Dial Pattern Details

Commit

Cancel

General

* Pattern: 22

* Min: 5

* Max: 5

Emergency Call: ☐

SIP Domain: avaya.com

Notes: Nuance Speech Attendant

Originating Locations and Routing Policies

Add

Remove

1 Item Refresh

<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"/>	BR-DevConnect	Juan's Subnet(s)	ToSpeechAttendant	0	<input type="checkbox"/>	SpeechAttendant	

Select : All, None

Denied Originating Locations

Add

Remove

0 Items Refresh

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

* Input Required

Commit

Cancel

4.7. Add Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add** (not shown), and fill in the fields as described below and shown in the following screen:

Under *Identity*:

- **SIP Entity Name:** Select the name of the SIP Entity added for Session Manager
- **Description:** Descriptive comment (optional)
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager
- **Default Gateway:** Enter the IP address of the default gateway for Session Manager

Use default values for the remaining fields. Click **Save** to add this Session Manager.

AVAYA Avaya Aura™ System Manager 6.0 Welcome, **admin** Last Logged on at September 10, 2010 8:44 AM
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Elements / Session Manager / Session Manager Administration / Edit Session Manager

Edit Session Manager Commit Cancel

General | Security Module | NIC Bonding | Monitoring | CDR | Personal Profile Manager (PPM) - Connection Settings | Event Server | Expand All | Collapse All

General

SIP Entity Name: devcon-asm

Description:

*Management Access Point Host Name/IP: 10.32.24.233

*Direct Routing to Endpoints:

Security Module

SIP Entity IP Address: 10.32.24.235

*Network Mask: 255.255.255.0

*Default Gateway: 10.32.24.1

*Call Control PHB: 46

*QOS Priority: 6

*Speed & Duplex:

VLAN ID:

5. 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

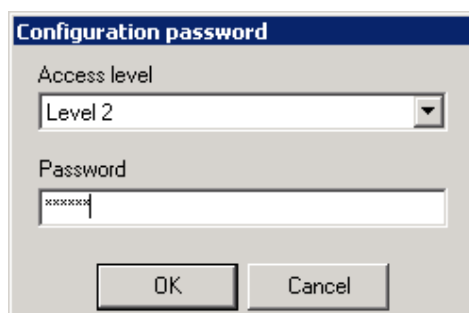
Note: Configuration of transfer entries and menus in Nuance SA is outside the scope of these Application Notes and will not be covered.

Nuance SA is configured through **Admin Tools** which can be started by navigating to **Start→Programs→Speech Attendant→Admin Tools**. The initial screen is displayed below.



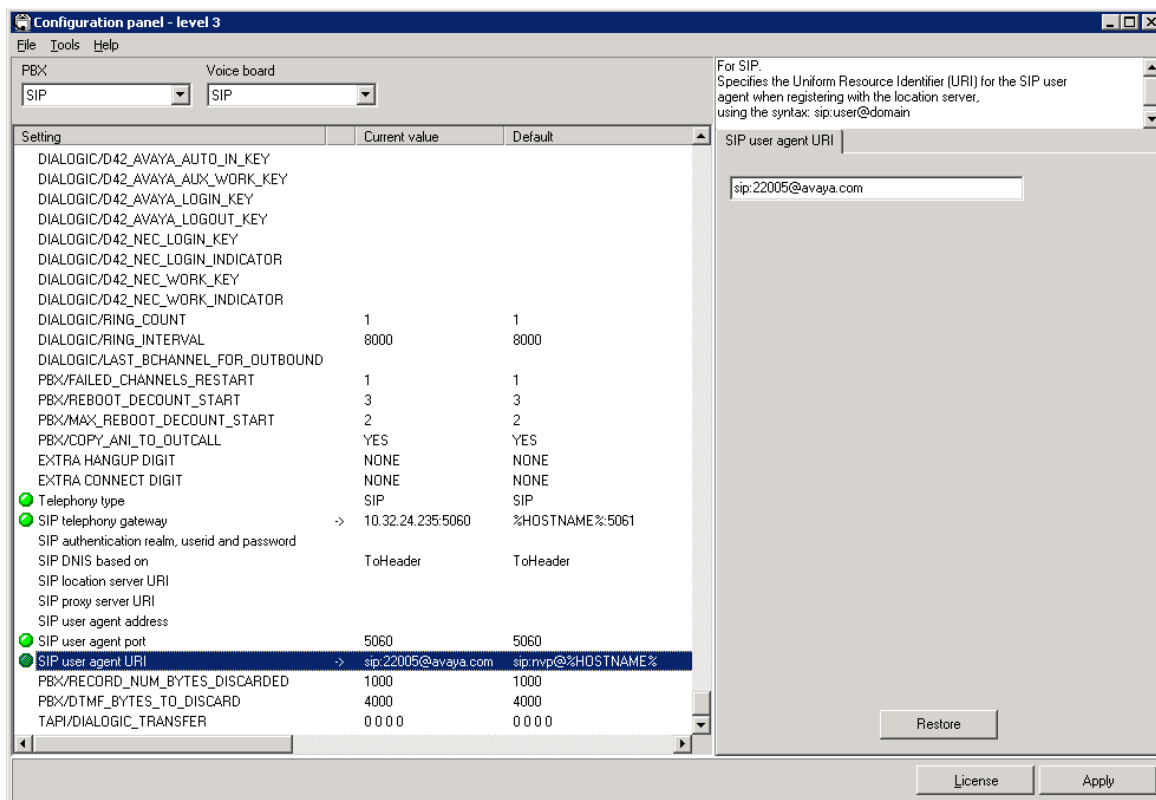
5.1. Configure SIP Interface in Configuration Panel

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 2* access level.



A small dialog box titled "Configuration password". It contains two input fields: "Access level" with a dropdown menu showing "Level 2", and "Password" with a text box containing "xxxxxx". 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. Lastly, configure the SIP port that SA listens on in the **SIP user agent port** field and specify the SIP URI for SA, which contains its extension and domain name (e.g., *sip:22005@avaya.com*).

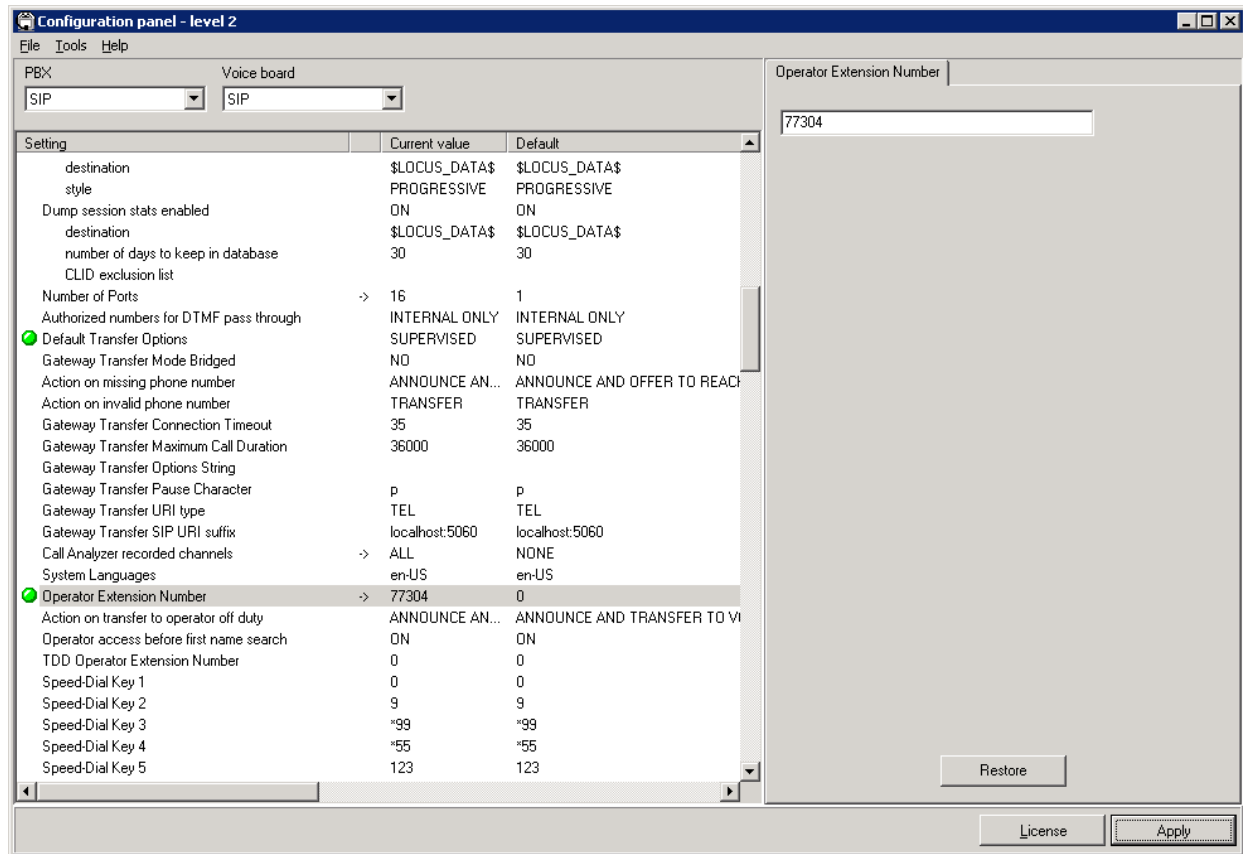


The "Configuration panel - level 3" window. It has a menu bar (File, Tools, Help) and two dropdowns at the top: "PBX" (set to SIP) and "Voice board" (set to SIP). Below these is a table with columns "Setting", "Current value", and "Default".

Setting	Current value	Default
DIALOGIC/D42_AVAYA_AUTO_IN_KEY		
DIALOGIC/D42_AVAYA_AUX_WORK_KEY		
DIALOGIC/D42_AVAYA_LOGIN_KEY		
DIALOGIC/D42_AVAYA_LOGOUT_KEY		
DIALOGIC/D42_NEC_LOGIN_KEY		
DIALOGIC/D42_NEC_LOGIN_INDICATOR		
DIALOGIC/D42_NEC_WORK_KEY		
DIALOGIC/D42_NEC_WORK_INDICATOR		
DIALOGIC/RING_COUNT	1	1
DIALOGIC/RING_INTERVAL	8000	8000
DIALOGIC/LAST_BCHANNEL_FOR_OUTBOUND		
PBX/FAILED_CHANNELS_RESTART	1	1
PBX/REBOOT_DECOUNT_START	3	3
PBX/MAX_REBOOT_DECOUNT_START	2	2
PBX/COPY_ANI_TO_OUTCALL	YES	YES
EXTRA HANGUP DIGIT	NONE	NONE
EXTRA CONNECT DIGIT	NONE	NONE
Telephony type	SIP	SIP
SIP telephony gateway	-> 10.32.24.235:5060	%HOSTNAME%:5061
SIP authentication realm, userid and password		
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:22005@avaya.com	sip:nvp:%HOSTNAME%
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

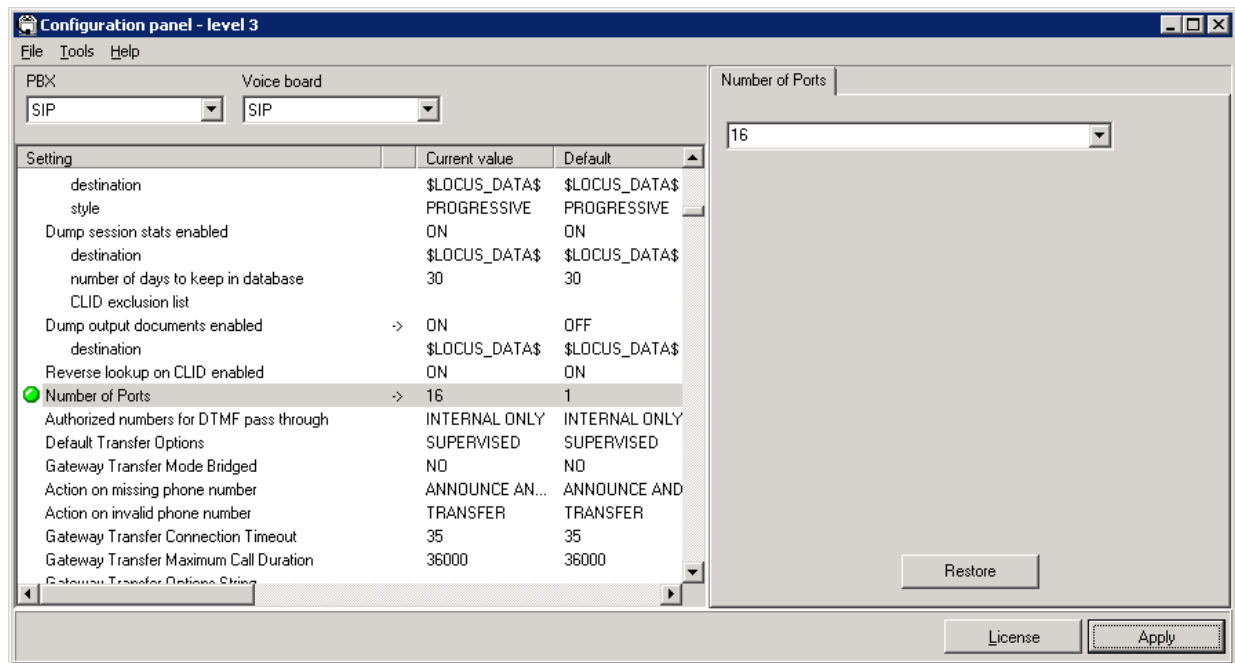
On the right side, there is a text box for "SIP user agent URI" containing "sip:22005@avaya.com". Below the table is a "Restore" button. At the bottom are "License" and "Apply" buttons.

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.

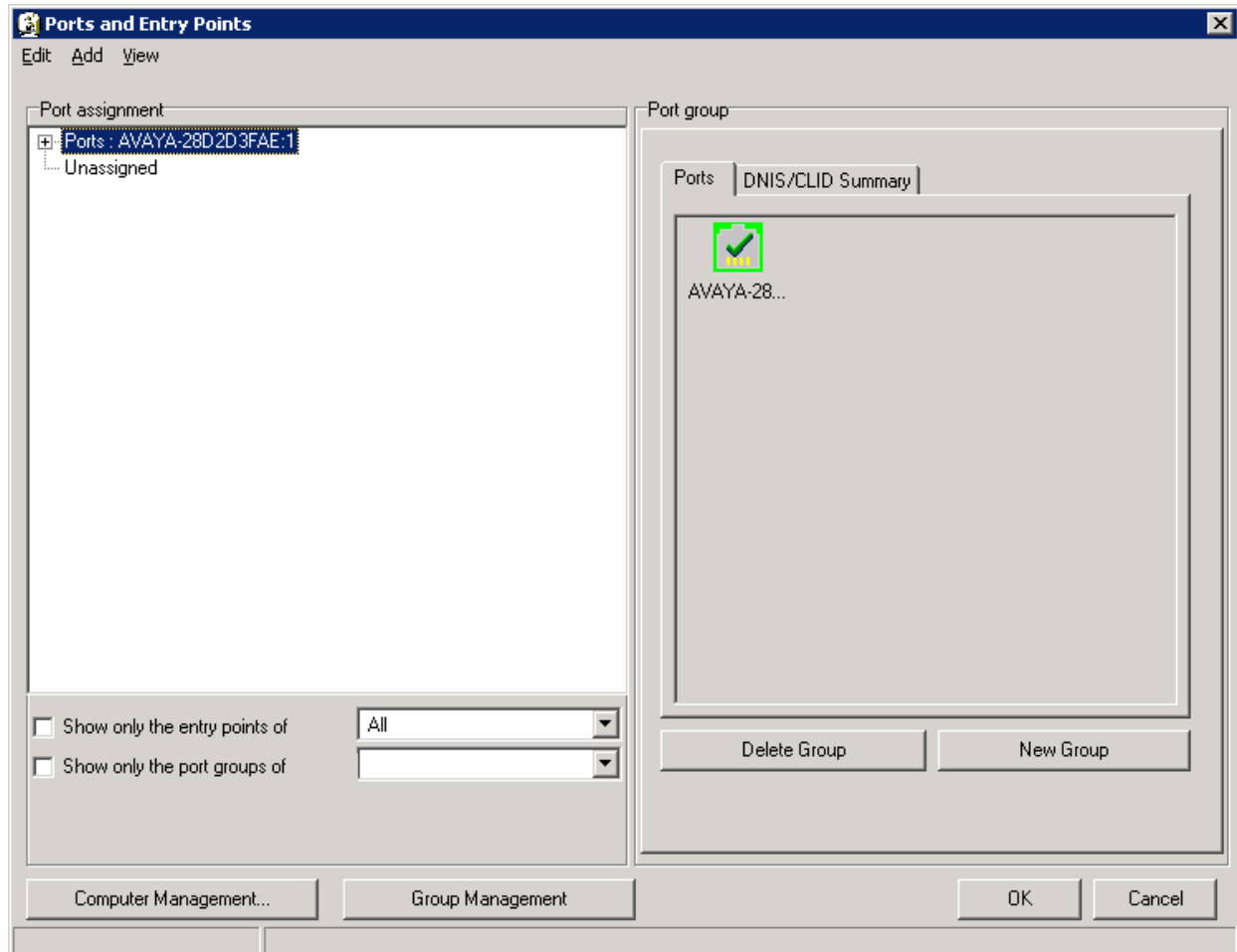


5.2. Configure Number of Ports Supported

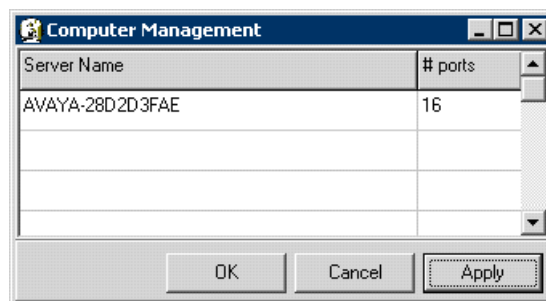
In the **Configuration Panel**, set the **Number of Ports** field to the desired value. In this example, 16 ports were configured. Click **Apply** and then close the **Configuration Panel**. Allow the SA application to be restarted when prompted.



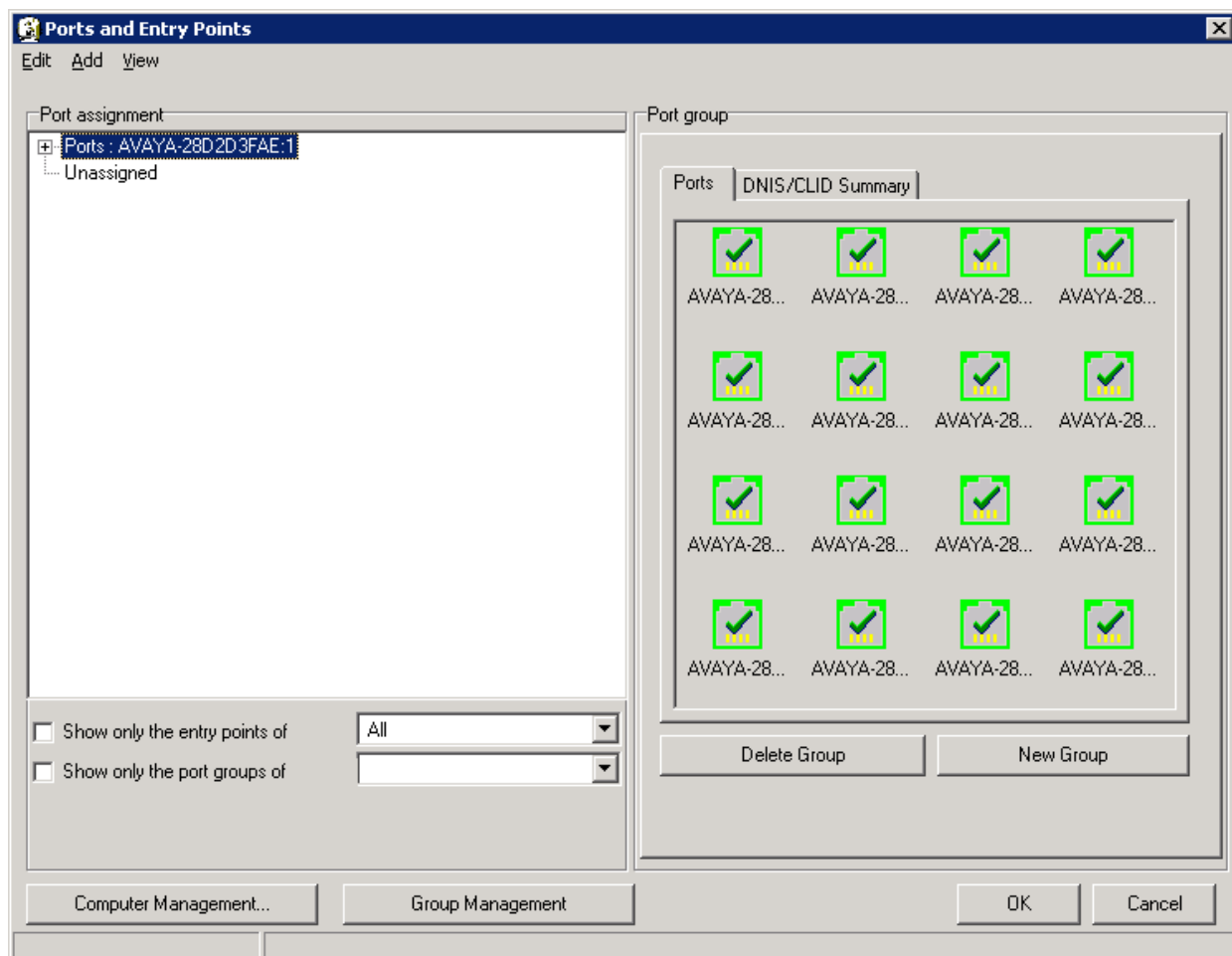
From **Admin Tools**, click on the **Phone Directory and Menu Editor** option and login with the appropriate credentials. In the **Menu Editor** (not shown), navigate to **Edit→Ports and Entry Points** to display the window below. Once the **Ports and Entry Points** window is opened, click on the **Computer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



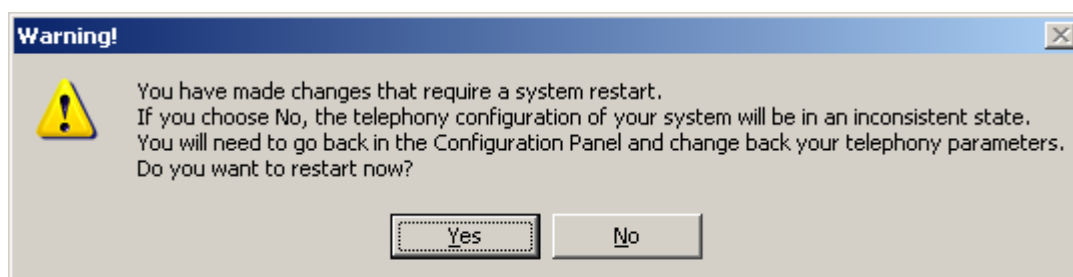
In the **Computer Management** window, set the # ports column to the desired value. In this example, 16 ports were configured. Click **OK**.



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 as shown below. Click **OK**.



Next, close the **Configuration Panel**. Allow the system to restart when prompted by clicking **Yes** in the dialog box below.



6. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability 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 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.

All test cases passed with the following observations noted:

- If all SIP ports on Nuance SA are busy, the call is not re-routed by Session Manager to another destination when an alternate route is specified in its Route Policies and Dial Patterns. Session Manager does not re-route calls based on a busy condition (i.e., a “486 Busy Here” message is received). However, if a “503 Service Not Available” message is returned to Session Manager, it will re-route the call to an alternate destination, if configured properly. In a future SA hotfix, Nuance will provide the ability to return a 503 message to Session Manager when it is busy. If SA is down or not running, Session Manager will re-route the call.
- 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.

7. 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** 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*.

The screenshot displays the SpeechAttendant® SA Monitor interface. The top header shows the product name and version: "SpeechAttendant® Hosted on avaya-28d2d3fae". Below this, a description states: "Description: Auto Attendant Version: SA 11.0.0 (with E03) (latest hot fix installed 1100HF01, 1100HF02, 1100HF03, 1100HF04, 1100HF05)".

The interface is divided into two main sections. On the left is a "Sections" sidebar with links for Summary status, Reports, Alarms, OSA Servlet, Environment, Configuration, Installation log, Monitoring, Replication Monitor, Replication Status, Replication Errors, and Call Logs. The "Call Logs" link is selected.

The main area is titled "CALL LOGS" and contains several filter options: "Select period:" with date pickers for "Date from" (11/30/2010) and "Date to" (11/30/2010); "Filter by:" with dropdowns for "DNIS" and "CLID" (both set to "Starts With"); "Filter by call termination codes:" with a dropdown menu showing options like "None", "ADNT - Announce number but Do Not Transfer (CS)", "B - Busy (INF)", "BO - Busy on Operator (INF)", and "CC - Cancelled by Caller (INF)"; "Filter by call complexity:" with a dropdown set to "all types"; and "Filter by call duration:" with a dropdown set to "last 50 calls" and a "Submit" button.

Below the filters, a table titled "Logs (from database), 33 sessions" displays the following data:

Call start	DNIS	CLID	Call complexity	Call duration	Error	Termination code	Destination	System comment	Tagging
30/11/2010 11:59:26	22006	77304	0	0		...		In progress	
30/11/2010 11:59:13	22005	77304	0	8		HG	Toplevel Menu	caller hangup	
30/11/2010 11:58:47	22005	77301	25	19		SRI3	Kent, Clark	transfer completed	
30/11/2010 11:58:26	22005	77303	5	15		SRI1	Croft, Lara	transfer completed	
30/11/2010 11:57:42	22005	77302	5	34		SRI1	Cane, Linda	transfer completed	
30/11/2010 11:40:22	22005	77303	0	9		HG	Toplevel Menu	caller hangup	

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.

8. 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.

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® Communication Manager*, June 2010, Release 6.0, Issue 6.0, Document Number 03-300509.
- [2] *Administering Avaya Aura® Session Manager*, August 2010, Issue 3, Release 6.0, Document Number 03-603324.

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