



Avaya Solution & Interoperability Test Lab

Application Notes for Nuance SpeechAttendant 12.1 with Avaya Aura® Session Manager 7.0 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Nuance SpeechAttendant 12.1 to interoperate with Avaya Aura® Session Manager 7.0 and Avaya Aura® Communication Manager 7.0 using SIP trunks. Nuance SpeechAttendant automates call routing by asking callers to speak the name or dial the extension of a destination.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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 for Nuance SpeechAttendant 12.1 to interoperate with Avaya Aura® Session Manager 7.0 and Avaya Aura® Communication Manager 7.0 using SIP trunks. Nuance SpeechAttendant automates call routing by asking callers to speak the name or dial the extension of a destination.

In the compliance testing, calls from internal and external callers were routed over SIP trunks to Nuance SpeechAttendant. Nuance SpeechAttendant played different greeting announcements based on ANI and/or DNIS, used speech recognition and/or DTMF digits to determine the route destination, and used SIP REFER to transfer calls to destinations on Avaya Aura® Communication Manager or on the PSTN.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were placed manually from users on the PSTN and on Communication Manager to SpeechAttendant. Speech and DTMF input were used from the callers for requesting transfer to internal user and group destinations on Communication Manager, and to external destinations on the PSTN.

The serviceability test cases were performed manually by disconnecting and reconnecting the Ethernet connection to SpeechAttendant.

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

The interoperability compliance test included feature and serviceability testing.

The feature testing included G.711MU, session refresh, ANI, DNIS, speech recognition, DTMF, speaking ahead (barge-in), dialing ahead, call forwarding, find me, voicemail, invalid number, blind transfer, supervised transfer, outgoing call screening, and simultaneous calls.

The serviceability testing focused on verifying the ability of SpeechAttendant to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to SpeechAttendant.

2.2. Test Results

All test cases were executed, and the following were observations on SpeechAttendant:

- The application only supports the G.711MU codec, and does not support codec negotiation and media shuffling.
- When an internal user destination has call forwarding activated for both internal and external calls, all transferred calls from SpeechAttendant will follow the forwarding destination for external calls.
- The difference between SpeechAttendant's implementation of blind versus supervised transfer resides in when the application drops from the call after sending REFER with transfer-to destination as Refer To.

2.3. Support

Technical support on SpeechAttendant can be obtained through the following:

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

3. Reference Configuration

As shown in **Figure 1**, SIP trunks were used between Session Manager and SpeechAttendant, and the applicable domain name was “dr220.com”.

A five digit Uniform Dial Plan (UDP) was used to facilitate routing with SpeechAttendant. Unique extension ranges were assigned to users on Communication Manager (6xxxx), and to SpeechAttendant (52xxx).

The configuration of Session Manager is performed via the web interface of System Manager. The detailed administration of basic connectivity between Communication Manager, System Manager, and Session Manager is not the focus of these Application Notes and will not be described.

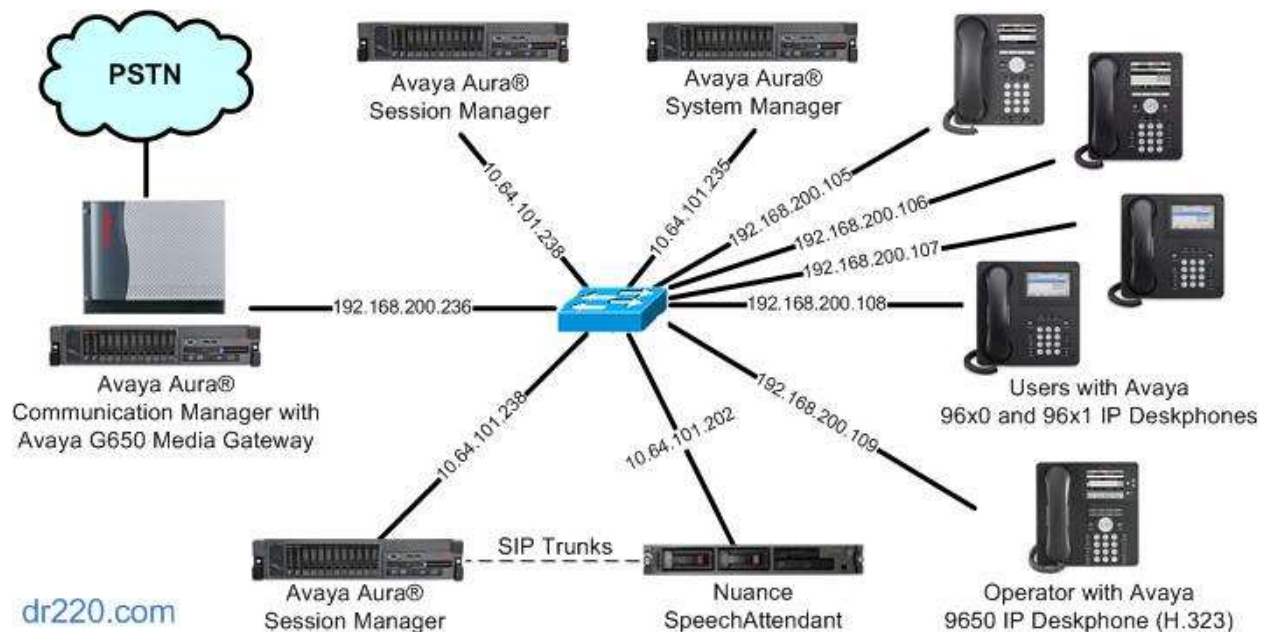


Figure 1: Compliance Testing Configuration

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager in Virtual Environment	7.0 SP1 (7.0.0.1.0.441.22477)
Avaya G650 Media Gateway	NA
Avaya Aura® Media Server in Virtual Environment	7.7.0.236
Avaya Aura® Session Manager in Virtual Environment	7.0 (7.0.0.0.0.700007)
Avaya Aura® System Manager in Virtual Environment	7.0 (7.0.0.0.0.4036)
Avaya 9608 IP Deskphone (H.323)	6.6029
Avaya 9620C & 9650 IP Deskphones (H.323)	3.250A
Avaya 9621G & 9641G IP Deskphones (SIP)	7.0.0.39
Nuance SpeechAttendant on Microsoft Windows Server 2012	12.1.0 HotFix 1210HF01, 1210HF02 R2 Standard

5. Configure Avaya Aura® Communication Manager

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

- Verify license
- Administer system parameters features
- Administer SIP trunk group
- Administer SIP signaling group
- Administer SIP trunk group members
- Administer IP network region
- Administer IP codec set
- Administer route pattern
- Administer private numbering
- Administer uniform dial plan
- Administer AAR analysis

In the compliance testing, a separate set of codec set, network region, trunk group, and signaling group were used for integration with SpeechAttendant.

5.1. Verify License

Log into the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the “display system-parameters customer-options” command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

display system-parameters customer-options		Page 2 of 12
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	12000	10
Maximum Concurrently Registered IP Stations:	1800	1
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	1
Maximum Video Capable IP Softphones:	24000	20
Maximum Administered SIP Trunks:	24000	0
Maximum Administered Ad-hoc Video Conferencing Ports:	24000	0
Maximum Number of DS1 Boards with Echo Cancellation:	522	0

5.2. Administer System Parameters Features

Use the “change system-parameters features” command to allow for trunk-to-trunk transfers.

For ease of interoperability testing, the **Trunk-to-Trunk Transfer** field was set to “all” to enable all trunk-to-trunk transfers on a system wide basis. Note that this feature poses significant security risk, and must be used with caution. For alternatives, the trunk-to-trunk feature can be implemented on the Class Of Restriction or Class Of Service levels. Refer to reference [1] for more details.

```
change system-parameters features                               Page 1 of 19
      FEATURE-RELATED SYSTEM PARAMETERS
      Self Station Display Enabled? n
      Trunk-to-Trunk Transfer: all
      Automatic Callback with Called Party Queuing? n
      Automatic Callback - No Answer Timeout Interval (rings): 3
      Call Park Timeout Interval (minutes): 10
      Off-Premises Tone Detect Timeout Interval (seconds): 20
      AAR/ARS Dial Tone Required? y

      Music (or Silence) on Transferred Trunk Calls? Call-wait
      DID/Tie/ISDN/SIP Intercept Treatment: attendant
      Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
      Automatic Circuit Assurance (ACA) Enabled? n
```

5.3. Administer SIP Trunk Group

Use the “add trunk-group n” command, where “n” is an available trunk group number, in this case “52”. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** “sip”
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** “tie”

add trunk-group 52		Page 1 of 21	
TRUNK GROUP			
Group Number: 52	Group Type: sip	CDR Reports: y	
Group Name: Nuance	COR: 1	TN: 1	TAC: 1052
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n	Auth Code? n		
Queue Length: 0	Member Assignment Method: auto		
Service Type: tie	Signaling Group:		
	Number of Members: 0		

Navigate to **Page 3**, and enter “private” for **Numbering Format**.

add trunk-group 52		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			
Hold/Unhold Notifications? y			
Modify Tandem Calling Number: no			

5.4. Administer SIP Signaling Group

Use the “add signaling-group n” command, where “n” is an available signaling group number, in this case “52”. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** “sip”
- **Transport Method:** “tcp”
- **Near-end Node Name:** An existing C-LAN node name or “procr”.
- **Far-end Node Name:** The existing node name for Session Manager.
- **Near-end Listen Port:** An available port for integration with Nuance.
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**.
- **Far-end Network Region:** An existing network region to use with Nuance.
- **Far-end Domain:** The applicable domain name for the network.
- **Direct IP-IP Audio Connections:** “n”

add signaling-group 52		Page 1 of 2
SIGNALING GROUP		
Group Number: 52	Group Type: sip	
IMS Enabled? n	Transport Method: tcp	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? y	
Peer Detection Enabled? y	Peer Server: Others	
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? n		
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? y		
Near-end Node Name: procr	Far-end Node Name: sm7-sig	
Near-end Listen Port: 5052	Far-end Listen Port: 5052	
	Far-end Network Region: 2	
Far-end Domain: dr220.com		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? n	
Enable Layer 3 Test? y	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? n	
	Alternate Route Timer(sec): 6	

5.5. Administer SIP Trunk Group Members

Use the “change trunk-group n” command, where “n” is the trunk group number from **Section 5.3**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Signaling Group:** The signaling group number from **Section 5.4**.
- **Number of Members:** The desired number of members, in this case “10”.

change trunk-group 52		Page 1 of 21	
TRUNK GROUP			
Group Number: 52	Group Type: sip	CDR Reports: y	
Group Name: Nuance	COR: 1	TN: 1	TAC: 1052
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0			
Service Type: tie	Auth Code? n		
Member Assignment Method: auto			
Signaling Group: 52			
Number of Members: 10			

5.6. Administer IP Network Region

Use the “change ip-network-region n” command, where “n” is the existing far-end network region number used by the SIP signaling group from **Section 5.4**.

For **Authoritative Domain**, enter the applicable domain for the network. Enter a descriptive **Name**. Enter “no” for **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio**, as shown below. For **Codec Set**, enter an available codec set number for integration with SpeechAttendant.

change ip-network-region 2		Page 1 of 20	
IP NETWORK REGION			
Region: 2			
Location:		Authoritative Domain: dr220.com	
Name: Nuance		Stub Network Region: n	
MEDIA PARAMETERS		Intra-region IP-IP Direct Audio: no	
Codec Set: 2		Inter-region IP-IP Direct Audio: no	
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			

Navigate to **Page 4**, and specify this codec set to be used for calls with network regions used by Avaya endpoints and by the trunk to the PSTN. In the compliance testing, network region “1” was used by the Avaya endpoints and by the trunk to the PSTN.

change ip-network-region 2										Page 4 of 20			
Source Region: 2										Inter Network Region Connection Management			
										I		M	
										G	A	t	
dst	codec	direct	WAN-BW-limits		Video		Intervening			Dyn	A	G	c
rgn	set	WAN	Units	Total	Norm	Prio	Shr	Regions	CAC	R	L	e	
1	2												
2	2												
3		all											
4													
5													
6													
7													
8													

Use the “change ip-codec-set n” command, where “n” is the codec set number from **Section 5.6**. Update the audio codec types in the **Audio Codec** fields as necessary. Note that SpeechAttendant only supports the G.711 codec variant. The codec shown below was used in the compliance testing.

5.8. Administer Route Pattern

- **Pattern Name:** A descriptive name.
- **Grp No:** The SIP trunk group number from **Section 5.3**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.

12 of 36
Nuance-SM7

5.9. Administer Private Numbering

Use the “change private-numbering 0” command, to define the calling party number to send to SpeechAttendant. Add an entry for the trunk group defined in **Section 5.3**. In the example shown below, all calls originating from a 5-digit extension beginning with 6 and routed to trunk group 52 will result in a 5-digit calling number. The calling party number will be in the SIP “From” header.

change private-numbering 0					Page 1 of 2
NUMBERING - PRIVATE FORMAT					
Ext	Ext	Trk	Private	Total	
Len	Code	Grp(s)	Prefix	Len	
5	6	52		5	Total Administered: 1
					Maximum Entries: 540

5.10. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 52xxx to SpeechAttendant. Note that other routing methods may be used. Use the “change uniform-dialplan 0” command, and add an entry to specify the use of AAR for routing of digits 52xxx, as shown below.

change uniform-dialplan 0					Page 1 of 2
UNIFORM DIAL PLAN TABLE					
					Percent Full: 0
Matching			Insert	Node	
Pattern	Len	Del	Digits	Net Conv Num	
52	5	0		aar n	

5.11. Administer AAR Analysis

Use the “change aar analysis 0” command, and add an entry to specify how to route calls to 52xxx. In the example shown below, calls with digits 52xxx will be routed as an AAR call using route pattern “52” from **Section 5.8**.

change aar analysis 0							Page 1 of 2
AAR DIGIT ANALYSIS TABLE							
Location: all						Percent Full: 2	
	Dialed	Total		Route	Call	Node	ANI
	String	Min	Max	Pattern	Type	Num	Reqd
52		5	5	52	unku		n

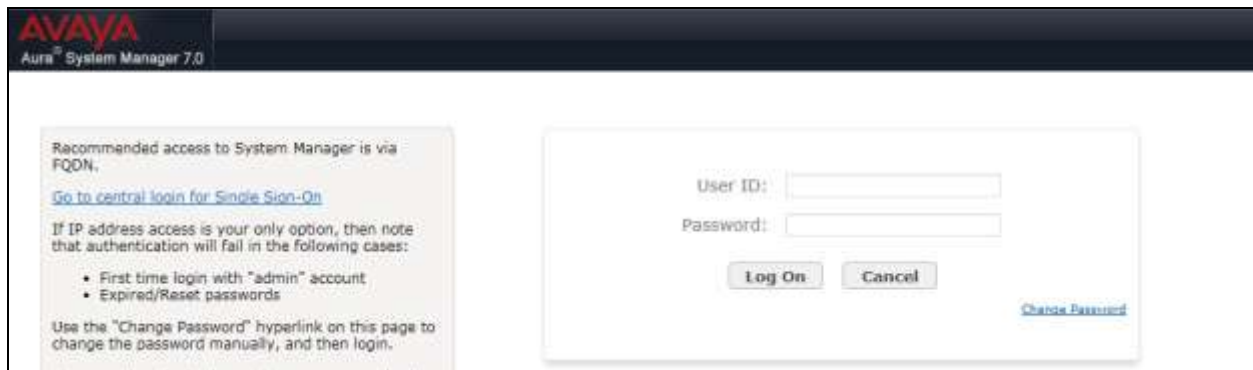
6. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer locations
- Administer SIP entities
- Administer routing policies
- Administer dial patterns

6.1. Launch System Manager

Access the System Manager web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of System Manager. Log in using the appropriate credentials.



6.2. Administer Locations

In the subsequent screen (not shown), select **Elements → Routing** to display the **Introduction to Network Routing Policy** screen below. Select **Routing → Locations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new location for SpeechAttendant.



The **Location Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name** and optional **Notes**. Retain the default values in the remaining fields.

Scroll down to the **Location Pattern** sub-section, click **Add** and enter the IP address of SpeechAttendant in **IP Address Pattern**, as shown below. Retain the default values in the remaining fields.

6.3. Administer SIP Entities

Add two new SIP entities, one for SpeechAttendant and one for the new SIP trunks with Communication Manager.

6.3.1. SIP Entity for SpeechAttendant

Select **Routing** → **SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for SpeechAttendant.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of the SpeechAttendant server.
- **Type:** “SIP Trunk”
- **Notes:** Any desired notes.
- **Location:** Select the SpeechAttendant location name from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left navigation pane is expanded, showing the 'Routing' menu with 'SIP Entities' selected. The main content area displays the 'SIP Entity Details' form. The form has a 'General' tab and several sections: 'General' with fields for Name, FQDN or IP Address, Type, Notes, Adaptation, Location, Time Zone, SIP Timer B/F (in seconds), Credential name, Securable, and Call Detail Recording; 'Loop Detection' with fields for Loop Detection Mode, Loop Count Threshold, and Loop Detection Interval (in msec); and 'SIP Link Monitoring' with a field for SIP Link Monitoring. The form includes 'Commit' and 'Cancel' buttons at the top right. The top of the interface shows the Avaya logo, 'Aura System Manager 7.0', and a 'Log off' button.

Scroll down to the **Entity Links** sub-section, and click **Add** to add an entity link. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case “DR-SM7”.
- **Protocol:** “UDP”
- **Port:** “5060”
- **SIP Entity 2:** The SpeechAttendant entity name from this section.
- **Port:** “5060”
- **Connection Policy:** “trusted”

Note that SpeechAttendant can support UDP, TCP, and TLS, and the compliance testing used the UDP protocol.

Entity Links

Override Port & Transport with DNS SRV: ☐

Add Remove

1 Item Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	* SM7-SA	DR-SM7	UDP	* 5060	Nuance-SA	* 5060	trusted	<input type="checkbox"/>

Select : All, None

SIP Responses to an OPTIONS Request

Add Remove

0 Items Filter: Enable

<input type="checkbox"/>	Response Code & Reason Phrase	Mark Entity Up/Down	Notes
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Commit Cancel

6.3.2. SIP Entity for Communication Manager

Select **Routing** → **SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for Communication Manager. Note that this SIP entity is used for integration with SpeechAttendant.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of an existing CLAN or the processor interface.
- **Type:** “CM”
- **Notes:** Any desired notes.
- **Adaptation:** Select the applicable adaptation for Communication Manager.
- **Location:** Select the applicable location for Communication Manager.
- **Time Zone:** Select the applicable time zone.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left navigation pane has 'Routing' expanded, and 'SIP Entities' is selected. The main content area is titled 'SIP Entity Details' and contains the following fields:

- Name:** DR-CM7-5052
- FQDN or IP Address:** 10.64.101.236
- Type:** CM
- Notes:** CM7 Port 5052 for Nuance
- Adaptation:** DR-CM7-Adaptation
- Location:** DR-Loc
- Time Zone:** America/New_York
- SIP Timer B/F (in seconds):** 4
- Credential name:** (empty field)
- Securable:** ☐
- Call Detail Recording:** none
- Loop Detection Mode:** On
- Loop Count Threshold:** 5
- Loop Detection Interval (in msec):** 200
- SIP Link Monitoring:** Use Session Manager Configuration

Scroll down to the **Entity Links** sub-section, and click **Add** to add an entity link. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case “DR-SM7”.
- **Protocol:** The signaling group transport method from **Section 5.4**.
- **Port:** The signaling group listen port number from **Section 5.4**.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** The signaling group listen port number from **Section 5.4**.
- **Connection Policy:** “trusted”

Entity Links
Override Port & Transport with DNS ☐
SRV: ☐

Add Remove

1 Item Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	*DR-CM7-5052	DR-SM7	TCP	*5052	DR-CM7-5052	*5052	trusted	<input type="checkbox"/>

Select: All, None

SIP Responses to an OPTIONS Request

Add Remove

0 Items Filter: Enable

<input type="checkbox"/>	Response Code & Reason Phrase	Mark Entity Up/Down	Notes
--------------------------	-------------------------------	---------------------	-------

Commit Cancel

6.4. Administer Routing Policies

Add two new routing policies, one for SpeechAttendant and one for the new SIP trunks with Communication Manager.

6.4.1. Routing Policy for SpeechAttendant

Select **Routing** → **Routing Policies** from the left pane, and click **New** in the subsequent screen (not shown) to add a new routing policy for SpeechAttendant.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the SpeechAttendant entity name from **Section 6.3.1**. The screen below shows the result of the selection.

AVAYA
Aura® System Manager 7.0

Last Logged on at December 7, 2015 1:46 PM
Log off

Home Routing

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

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit Cancel Help ?

General

* Name: To-SA

Disabled: ☐

* Retries: 0

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Nuance-SA	10.64.101.202	SIP Trunk	

6.4.2. Routing Policy for Communication Manager

Select **Routing** → **Routing Policies** from the left pane, and click **New** in the subsequent screen (not shown) to add a new routing policy for Communication Manager.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Communication Manager entity name from **Section 6.3.2**. The screen below shows the result of the selection.

AVAYA
Aura® System Manager 7.0

Last logged on at December 7, 2013 1:48 PM
Log off

Home Routing

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit Cancel Help ?

General

* Name: To-CM7-5052

Disabled: ☐

* Retries: 0

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
DR-CM7-5052	10.64.101.236	CM	CM7 Port 5052 for Nuance

6.5. Administer Dial Patterns

Add a new dial pattern for SpeechAttendant, and update existing dial patterns for Communication Manager.

6.5.1. Dial Pattern for SpeechAttendant

Select **Routing** → **Dial Patterns** from the left pane, and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach SpeechAttendant. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match, in this case “52”.
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** The signaling group domain name from **Section 5.4**.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching SpeechAttendant. In the compliance testing, the entry allowed for call originations from all Communication Manager endpoints in locations “DR-Loc” and “NJ-Loc”. The SpeechAttendant routing policy from **Section 6.4.1** was selected as shown below.

AVAYA
Aura® System Manager 7.0

Last Logged on at December 10, 2015 4:22 PM

Home / Routing

Home / Elements / Routing / Dial Patterns

Dial Pattern Details

Commit Cancel Help ?

General

* Pattern: 52

* Min: 5

* Max: 5

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: -ALL- ▼

Notes:

Originating Locations and Routing Policies

Add Remove

2 Items 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"/>	DR-Loc	TLT DR Network	To-SA	0	<input type="checkbox"/>	Nuance-SA	
<input type="checkbox"/>	NJ-Loc	TLT NJ Network	To-SA	0	<input type="checkbox"/>	Nuance-SA	

Select : All, None

6.5.2. Dial Pattern for Communication Manager

Select **Routing** → **Dial Patterns** from the left pane, and click on the first existing dial pattern for Communication Manager in the subsequent screen, in this case dial pattern “6” (not shown). The **Dial Pattern Details** screen is displayed.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new policy as necessary for calls from SpeechAttendant. In the compliance testing, the new policy allowed for call origination from the SpeechAttendant location from **Section 6.2**, and the Communication Manager routing policy from **Section 6.4.2** was selected as shown below. Retain the default values in the remaining fields.

Follow the procedures in this section to make similar changes to the applicable Communication Manager dial pattern to reach the PSTN. In the compliance testing, SpeechAttendant will add the prefix “9” for outbound calls to the PSTN, and therefore the existing dial pattern for “9” was also changed (not shown below).

AVAYA
Aura® System Manager 7.0

Last Logged on: at December 7, 2015 1:46 PM

Home Routing

Home / Elements / Routing / Dial Patterns

Dial Pattern Details

Commit Cancel

General

* Pattern: 6

* Min: 5

* Max: 5

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: -ALL-

Notes: To CM7

Originating Locations and Routing Policies

Add Remove

3 Items Filter: Enable

	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	DR-Loc	TLT DR Network	To-CM7	0	<input type="checkbox"/>	DR-CM7	
<input type="checkbox"/>	NJ-Loc	TLT NJ Network	To-CM7	0	<input type="checkbox"/>	DR-CM7	
<input checked="" type="checkbox"/>	SA-Loc	Nuance SpeechAttendant	To-CM7-5052	0	<input type="checkbox"/>	DR-CM7-5052	

Select : All, None

7. Configure Nuance SpeechAttendant

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

- Launch Admin Tools
- Administer configuration panel
- Administer phone directory and menu editor
- Administer dialing properties
- Administer ports and entry points

The configuration of SpeechAttendant is typically performed by Nuance Professional Services. The procedural steps are presented in these Application Notes for informational purposes.

7.1. Administer Configuration Panel

From the SpeechAttendant server, double-click the **Admin Tools** icon shown below, which was automatically created as part of installation.



7.2. Administer Configuration Panel

The **Admin Tools** screen is displayed. Select **Configuration Panel**.



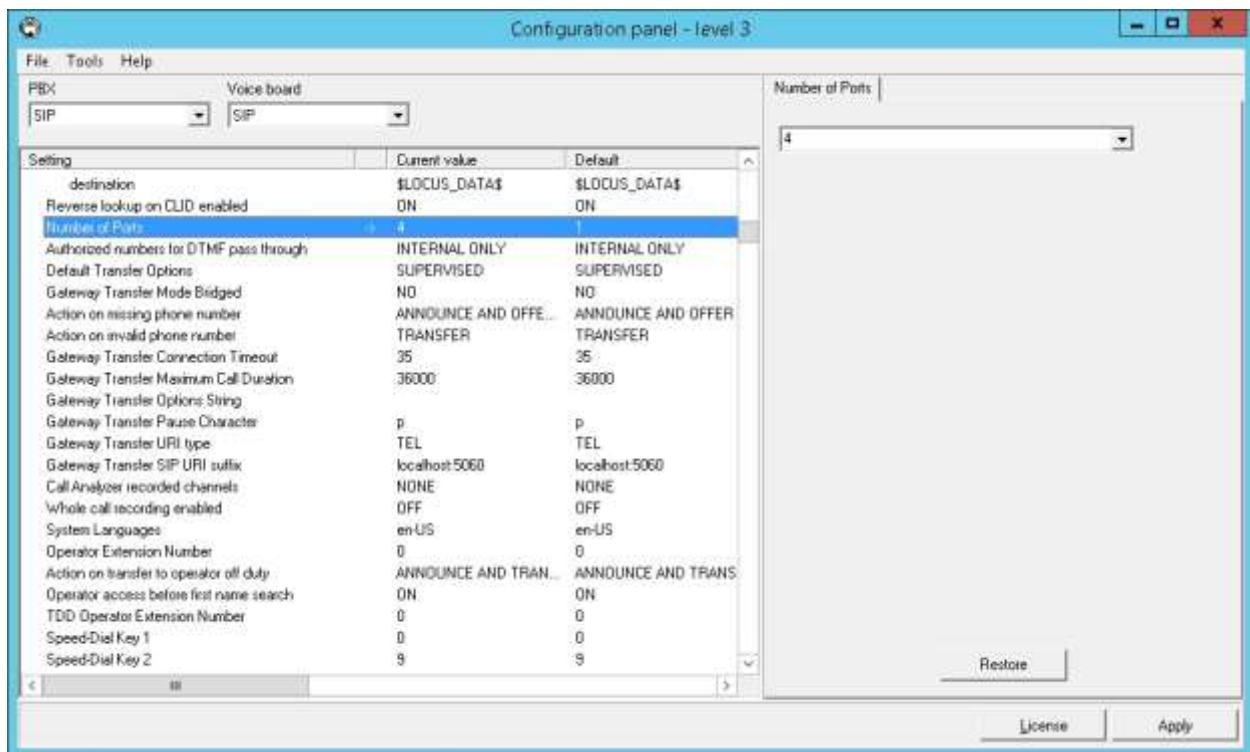
The **Configuration password** screen is displayed. Select “Level 3” and enter the appropriate credential.



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

The **Configuration panel – level 3** screen is displayed next. In the upper left pane, set **PBX** and **Voice board** to “SIP”, as shown below.

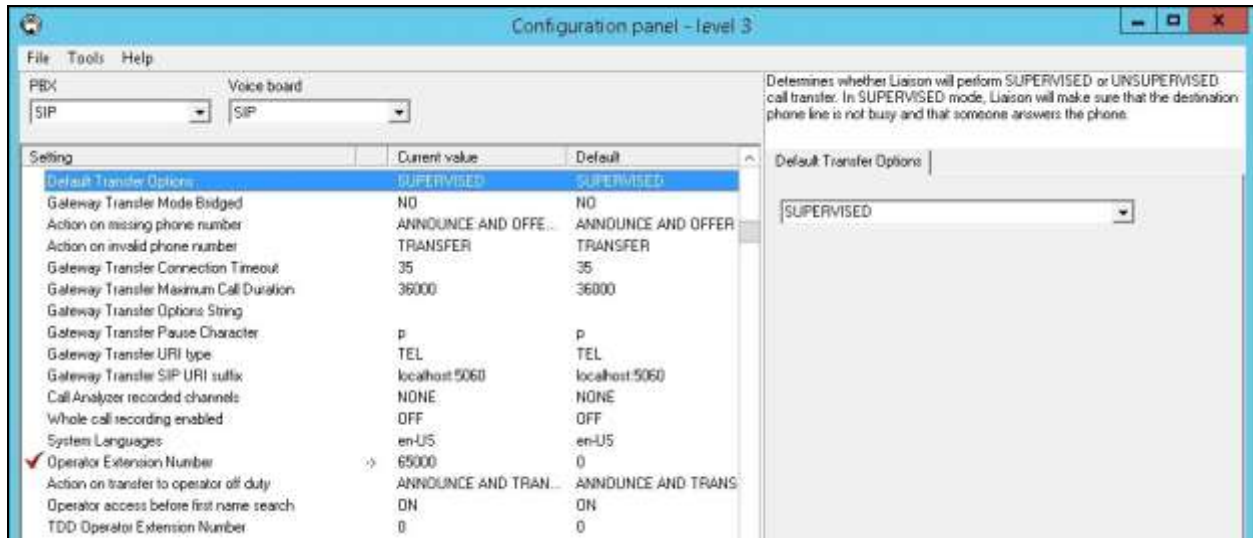
Scroll the screen in the left pane as necessary, and update the **Number of Ports** parameter to the number allowed for by the SpeechAttendant license, in this case “4”.



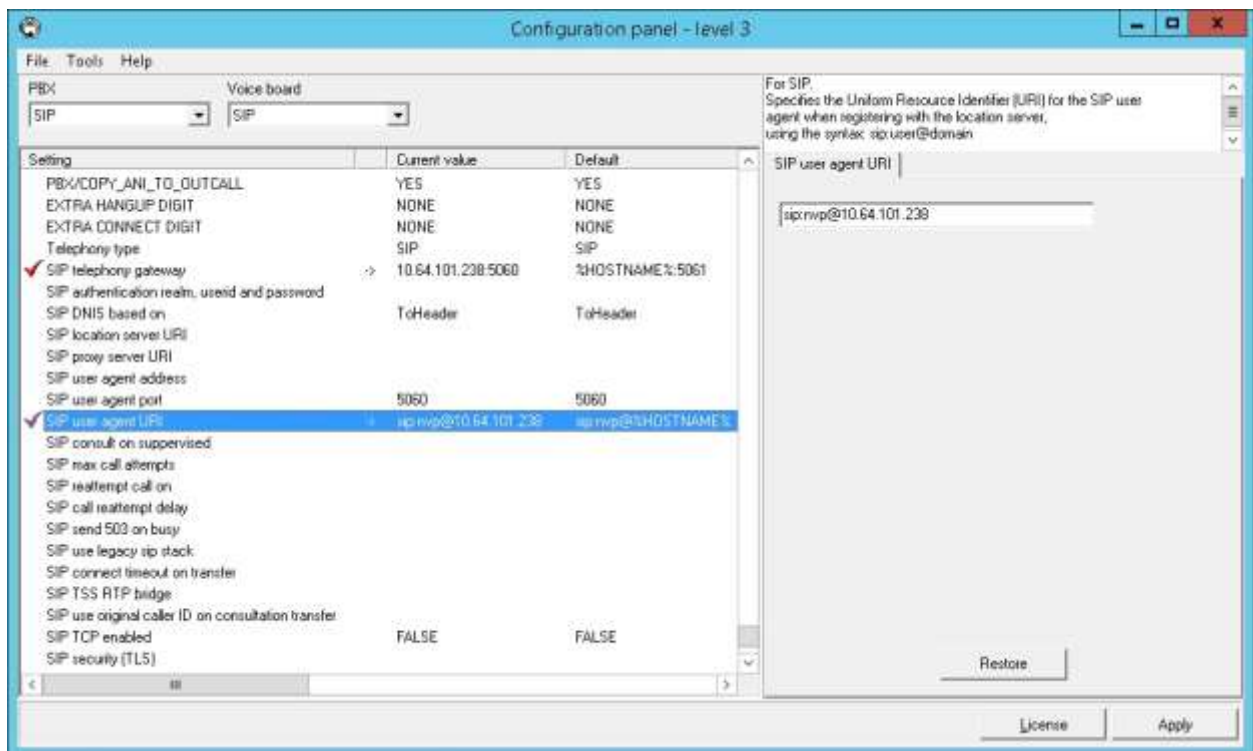
A screenshot of the "Configuration panel - level 3" window. The window has a menu bar (File, Tools, Help) and a title bar. It is divided into several sections. On the left, there are two dropdown menus: "PBX" and "Voice board", both set to "SIP". Below these is a table with three columns: "Setting", "Current value", and "Default". The "Number of Ports" setting is highlighted in blue, showing a current value of "4" and a default of "1". To the right of the table is a large text area with the number "4" entered. At the bottom right of the window are "License" and "Apply" buttons, and a "Restore" button is located above the "License" button.

Setting	Current value	Default
destination	\$LOCUS_DATA\$	\$LOCUS_DATA\$
Reverse lookup on CLID enabled	ON	ON
Number of Ports	4	1
Authorized numbers for DTMF pass through	INTERNAL ONLY	INTERNAL ONLY
Default Transfer Options	SUPERVISED	SUPERVISED
Gateway Transfer Mode Bridged	NO	NO
Action on missing phone number	ANNOUNCE AND OFFER	ANNOUNCE AND OFFER
Action on invalid phone number	TRANSFER	TRANSFER
Gateway Transfer Connection Timeout	35	35
Gateway Transfer Maximum Call Duration	36000	36000
Gateway Transfer Options String		
Gateway Transfer Pause Character	p	p
Gateway Transfer URI type	TEL	TEL
Gateway Transfer SIP URI suffix	localhost:5060	localhost:5060
Call Analyzer recorded channels	NONE	NONE
Whole call recording enabled	OFF	OFF
System Languages	en-US	en-US
Operator Extension Number	0	0
Action on transfer to operator off duty	ANNOUNCE AND TRANSFER	ANNOUNCE AND TRANSFER
Operator access before first name search	ON	ON
TDID Operator Extension Number	0	0
Speed-Dial Key 1	0	0
Speed-Dial Key 2	9	9

Scroll the screen in the left pane as necessary, and update the **Operator Extension Number** parameter with the extension of an endpoint on Communication Manager for use as the operator. SpeechAttendant will automatically transfer a caller to the operator when all attempts to understand the caller requests have failed. Callers can also ask for the operator directly.



Scroll the screen in the left pane as necessary, to locate the **SIP telephony gateway** and **SIP user agent URI** parameters. Update the two parameters with the IP address of the Session Manager signaling interface and the port number from **Section 6.3.1**, as shown below.



7.3. Administer Phone Directory and Menu Editor

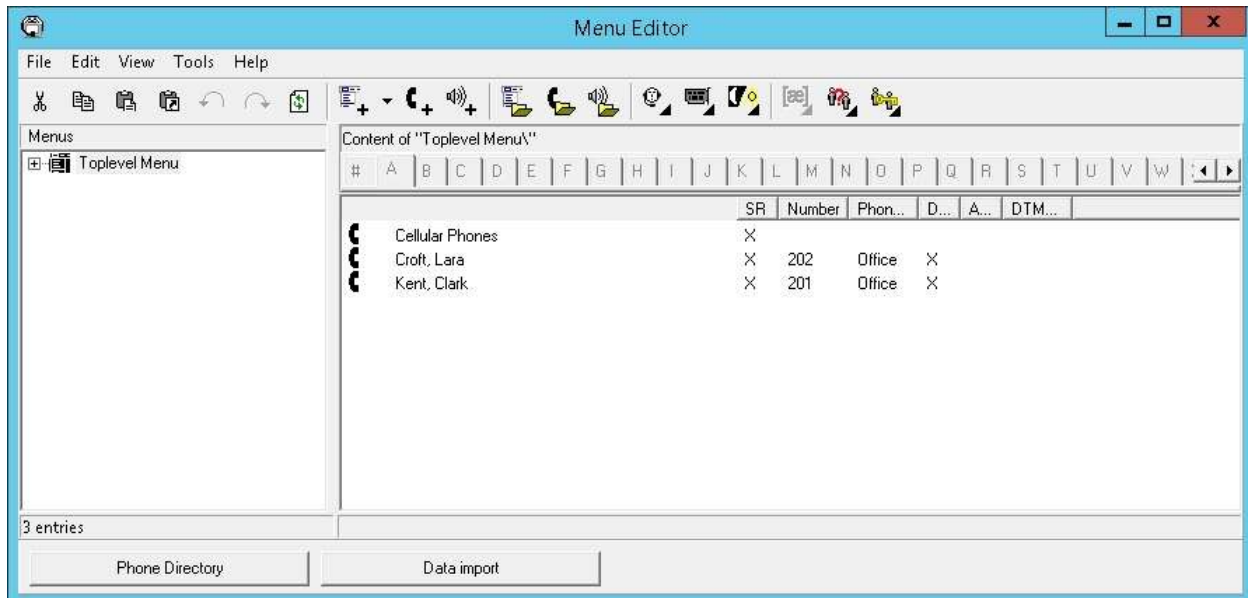
From the **Admin Tools** screen, select **Phone Directory and Menu Editor**, as shown below.



The **Phone Directory and Menu Editor** screen below is displayed next. Log in using the appropriate credentials.

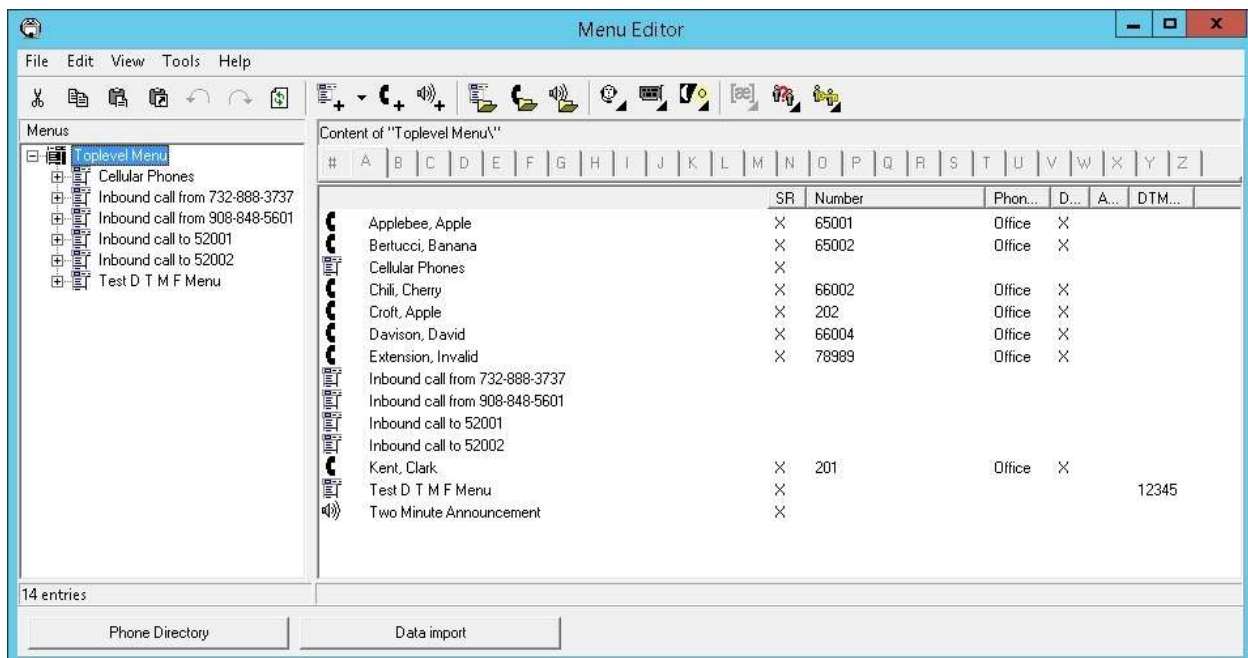


The **Menu Editor** screen below is displayed, with default directory entries in the right pane.



Follow reference [3] to create additional entry points in the left pane and additional directory entries in the right pane pertinent to customer needs. The screenshot below shows the entry points and directory entries used in the compliance testing. A mixture of blind and supervised transfer methods were configured for the directory entries.

Select **Tools → Dialing Properties** from the top menu.



7.4. Administer Dialing Properties

The **Default dialing properties** screen is displayed. Follow reference [3] to update and add dialing properties entries as necessary for routing of calls pertinent to the customer network.

In the compliance testing, the last entry in the screenshot below was added for routing of calls to internal destinations consisting of 5-digit extensions.

	Mask	Routing number	Announce number	DTMF input	Number type	Transfer type	Comment
1	+1-??-??-??-??-??	???	???	???	INTERNAL	FROM ENTRY	Default internal
2	??-??-??	9-??-??-??	??-??-??	??-??-??	LOCAL	FROM ENTRY	Default local
3	+1-??-??-??-??	9-1-??-??-??	??-??-??		LONG DISTANCE	FROM ENTRY	Default long distance
4	+??-??-??-??-??	9-011-??-??-??-??-??	??-??-??-??-??		INTERNATIONAL	FROM ENTRY	Default international
5	+??-??-??-??-??	9-011-??-??-??-??-??	??-??-??-??-??		INTERNATIONAL	FROM ENTRY	Default international
6	0	0			INTERNAL	UNSUPERVISED	Default operator
7	????	????	????	????	INTERNAL	FROM ENTRY	5 digit internal
8							
9							
10							
11							
12							
13							
14							
15							
16							
17							

Move up Move down

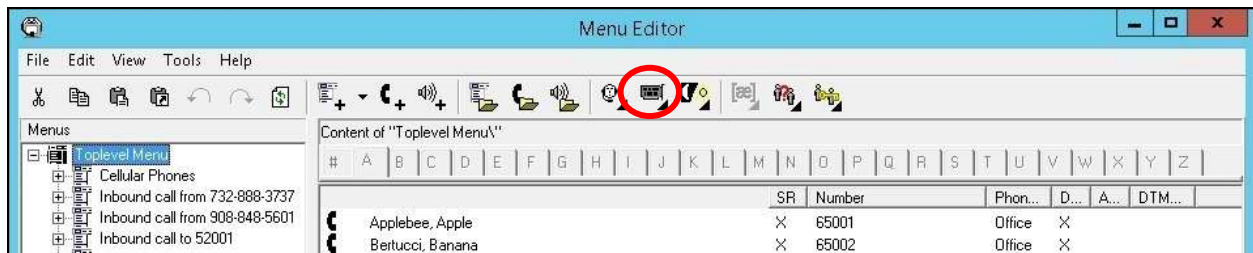
Number in directory Test

Rule matched Number dialed
Number announced

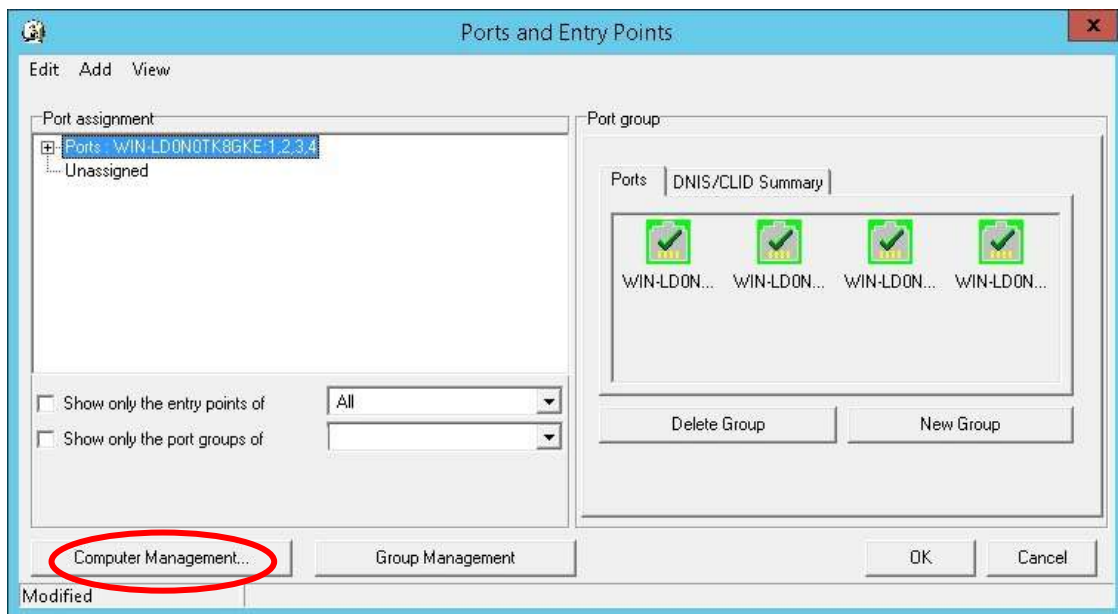
OK Cancel Apply

7.5. Administer Ports and Entry Points

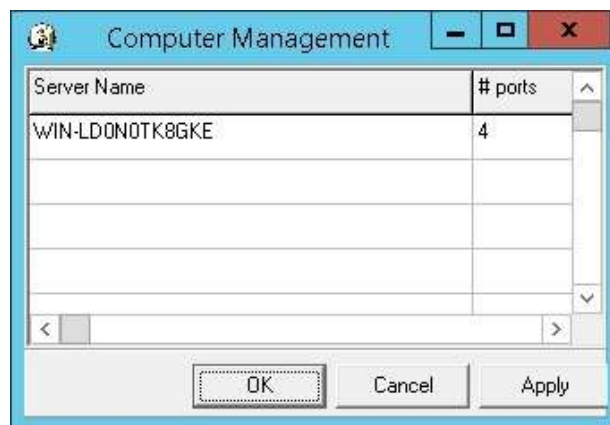
From the **Menu Editor** screen, click the **Ports and entry points** icon shown below.



The **Ports and Entry Points** screen below is displayed. Select **Computer Management**.



The **Computer Management** screen is displayed. Set the **# ports** value to the value allowed for by the license, in this case “4”.



8. Verification Steps

This section provides tests that can be performed to verify proper configuration of Communication Manager, Session Manager, and SpeechAttendant.

8.1. Verify Avaya Aura® Communication Manager

From the SAT interface, verify the status of the SIP trunk groups by using the “status trunk n” command, where “n” is the trunk group number administered in **Section 5.3**. Verify that all trunks are in the “in-service/idle” state as shown below.

```
status trunk 52
```

TRUNK GROUP STATUS			
Member	Port	Service State	Mtce Connected Ports Busy
0052/001	T00146	in-service/idle	no
0052/002	T00147	in-service/idle	no
0052/003	T00148	in-service/idle	no
0052/004	T00149	in-service/idle	no
0052/005	T00150	in-service/idle	no
0052/006	T00151	in-service/idle	no
0052/007	T00152	in-service/idle	no
0052/008	T00153	in-service/idle	no
0052/009	T00154	in-service/idle	no
0052/010	T00155	in-service/idle	no

Verify the status of the SIP signaling groups by using the “status signaling-group n” command, where “n” is the signaling group number administered in **Section 5.4**. Verify that the **Group State** is “in-service”, as shown below.

```
status signaling-group 52
```

STATUS SIGNALING GROUP	
Group ID:	52
Group Type:	sip
Group State:	in-service

8.2. Verify Avaya Aura® Session Manager

From the System Manager home page (not shown), select **Elements** → **Session Manager** to display the **Session Manager Dashboard** screen (not shown).

Select **Session Manager** → **System Status** → **SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Click the SpeechAttendant entity name from **Section 6.3.1**.

AVAYA
Aura® System Manager 7.0

Last Logged on at December 8, 2015 3:29 PM
Log off

Home / Elements / Session Manager / System Status / SIP Entity Monitoring

SIP Entity Link Monitoring Status Summary

This page provides a summary of Session Manager SIP entity link monitoring status.

SIP Entities Status for All Monitoring Session Manager Instances

Run Monitor

1 Items Refresh Filter: Enable

Session Manager	Type	Monitored Entities					Total
		Down	Partially Up	Up	Not Monitored	Deny	
<input type="checkbox"/> DR-SM2	Core	5	0	5	0	0	10

Select: All, None

All Monitored SIP Entities

Run Monitor

10 Items Refresh Filter: Enable

SIP Entity Name
<input type="checkbox"/> DR-CM2
<input type="checkbox"/> IPO1-IP500V2
<input type="checkbox"/> IPO2-IPOSE
<input checked="" type="checkbox"/> Nuance-SA

The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that the **Conn Status** and **Link Status** are “UP”, as shown below.

AVAYA
Aura System Manager 7.0

Last Logged on at December 8, 2015 3:29 PM

Home / Elements / Session Manager / System Status / SIP Entity Monitoring

SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

All Entity Links to SIP Entity: Nuance-SA

Status Details for the selected Session Manager:

Summary View

1 Items Refresh Filter: Enable

Session Manager	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
DR-SMZ	10.64.101.202	5060	UDP	FALSE	UP	200 OK	UP

8.3. Verify Nuance SpeechAttendant

From a PC, launch an Internet browser window and access the SpeechAttendant web-based status interface by using the URL “http://<ip-address>/OpenSpeech/Attendant/servlet/aa?action=status”, where “ip-address” is the IP address of the SpeechAttendant server.

The **Speech Attendant Login** screen is displayed. Log in using the appropriate credentials.

Speech Attendant
Login

Enter your user name and password.

User name:

Password:

Login Forgot your password ?

The screen below is displayed next. Verify that the **Status** for all channels are “Idle”, as shown below.

SpeechAttendant®
Hosted on **WIN-LD0N0TK8GKE**


Description: Auto Attendant Version: SA 12.1.0 (with E01) (latest hot fix installed 1210HFD1, 1210HFD2)

Sections





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

System summary

Uptime: 0 days 0 hours 0 minutes 41 seconds.
Served sessions: 1 total (1 currently in memory)
Served requests: 3

Telephony  no call so far for WIN-LD0N0TK8GKE.
no call in progress

win-lD0n0tk8gke

CHN	Status	Calls	DNIS	CLID	EP	Function	Menu	Action
	idle		---	---	---	---	---	---
	idle		---	---	---	---	---	---
	idle		---	---	---	---	---	---
	idle		---	---	---	---	---	---

Establish an incoming trunk call from PSTN with SpeechAttendant. Verify that the calling party hears the appropriate greeting, and that the status screen reflects the active call with pertinent call information, as shown below.

SpeechAttendant®
Hosted on **WIN-LD0N0TK8GKE**


Description: Auto Attendant Version: SA 12.1.0 (with E01) (latest hot fix installed 1210HFD1, 1210HFD2)

Sections





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

System summary

Uptime: 0 days 0 hours 0 minutes 41 seconds.
Served sessions: 1 total (1 currently in memory)
Served requests: 5

Telephony  1 calls so far for WIN-LD0N0TK8GKE.
1 calls in progress (concurrent peak 1, Fri Dec 11 17:55:10 PST 2015)

win-lD0n0tk8gke

CHN	Status	Calls	DNIS	CLID	EP	Function	Menu	Action
	busy	1	52000	9088485601	Call from 908-848-5601	AA	Inbound call from 908-848-5601	In progress
	idle		---	---	---	---	---	---
	idle		---	---	---	---	---	---
	idle		---	---	---	---	---	---

9. Conclusion

These Application Notes describe the configuration steps required for Nuance SpeechAttendant 12 to successfully interoperate with Avaya Aura® Session Manager 7.0 and Avaya Aura® Communication Manager 7.0 using SIP trunks. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

10. Additional References

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

1. *Administering Avaya Aura® Communication Manager*, Release 7.0, Issue 1, August 2015, available at <http://support.avaya.com>.
2. *Administering Avaya Aura® Session Manager*, Release 7.0, Issue 1, August 2015, available at <http://support.avaya.com>.
3. *Nuance SpeechAttendant Nuance OpenSpeech Attendant Administration Guide*, April 2014, available at <https://network.nuance.com/portal/server.pt>.

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