

Avaya Solution & Interoperability Test Lab

### Application Notes for Nuance SpeechAttendant 12.2 with Avaya IP Office Server Edition 10.0 – Issue 1.0

### Abstract

These Application Notes describe the configuration steps required for Nuance SpeechAttendant 12.2 to interoperate with Avaya IP Office Server Edition 10.0 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 12.2 to interoperate with Avaya IP Office Server Edition 10.0 using SIP trunks. SpeechAttendant automates call routing by asking callers to speak the name or dial the extension of a destination.

The IP Office Server Edition configuration consisted of two IP Office systems, a primary Linux server at the Main site and an expansion IP500V2 at the Remote site that were connected via Small Community Network (SCN) trunks.

In the compliance testing, calls from PSTN and internal callers were routed over SIP trunks to SpeechAttendant. 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 INVITE and SIP REFER to perform supervised transfer of calls to destinations on the primary IP Office system at the Main site, to destinations on the expansion IP Office system at the Remote site, and to destinations on the PSTN.

The SIP trunks connection from SpeechAttendant can be with either the primary Linux server or the expansion IP500V2 IP Office system. The configuration shown in these Application Notes used the primary Linux server IP Office system for SIP trunks connectivity.

## 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 primary and expansion IP Office systems to SpeechAttendant. Speech and DTMF input were used from the callers for requesting transfer to internal user and group destinations on the two IP Office systems, 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, supervised transfer, speaking ahead (barge-in), dialing ahead, no answer, do not disturb, busy, call forwarding, follow me, voicemail, mobile twinning, hot desking, invalid number, supervised transfer, call pickup, call screening, and simultaneous calls.

The feature testing call flows included calls with resources on the primary IP Office system, calls with resources on the expansion IP Office system, as well as calls with resources between the two IP Office systems.

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.711 codec, and does not support codec negotiation and media shuffling.
- The default RTP packet size of 30ms from SpeechAttendant can cause audio degradation with H.323 users on IP Office, and the workaround is to configure SpeechAttendant to use 20ms.
- The SpeechAttendant implementation of unsupervised transfer, which involves a REFER to the transfer destination without an INVITE, is not supported by IP Office. Therefore, all transfers from SpeechAttendant are required to use the supervised method.

### 2.3. Support

Technical support on SpeechAttendant can be obtained through the following:

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

# 3. Reference Configuration

The IP Office Server Edition configuration used in the compliance testing consisted of a primary Linux server at the Main site, and an expansion IP500V2 at the Remote site, with SCN trunks connectivity between the two systems. Each IP Office system has connectivity to the PSTN, for testing of cross systems PSTN scenarios. As shown in **Figure 1**, SIP trunks were used between the primary IP Office at the Main site and SpeechAttendant.

The detailed administration of IP Office resources is not the focus of these Application Notes and will not be described. As shown in **Figure 1** below, one SpeechAttendant server was deployed with SIP trunks connectivity to the primary IP Office system.

A five digit dial plan was used to facilitate routing with SpeechAttendant. Unique extension ranges were assigned to users on the primary IP Office system (210xx), to users on the expansion IP Office system (220xx), and to SpeechAttendant (2155x).



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
Main Site	
Avaya IP Office Server Edition (Primary) in Virtual Environment	10.0.0.1.0
Avaya 9608 & 9641G IP Deskphone (H.323)	6.6302
Avaya 1120E IP Deskphone (SIP)	4.4.23.0
Nuance SpeechAttendant on Microsoft Windows Server 2012	12.2 R2 Standard
Remote Site	
Avaya IP Office on IP500 V2 (Expansion)	10.0.0.1.0
Avaya 9620C IP Deskphones (H.323)	3.270B
Avaya 1120E IP Deskphone (SIP)	4.4.23.0
Avaya 9508 Digital Deskphone	NA

Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with IP Office Server Edition in all configurations.

# 5. Configure Avaya IP Office

This section provides the procedures for configuring the IP Office systems. The procedures include the following area:

- Verify license
- Administer system
- Administer line
- Administer incoming call route
- Administer short code

### 5.1. Verify License

From a PC running the IP Office Manager application, select **Start**  $\rightarrow$  **Programs**  $\rightarrow$  **IP Office**  $\rightarrow$  **Manager** to launch the application. Select the proper primary IP Office system, and log in using the appropriate credentials.

The Avaya IP Office Manager for Server Edition IPO2-IPOSE screen is displayed, where IPO2-IPOSE is the name of the primary IP Office system.

From the configuration tree in the left pane, select **License** under the IP Office system that will be used for SIP trunks connection with SpeechAttendant, in this case "IPO2-IPOSE", and a list of licenses is displayed in the right pane. Verify that there is a license for **SIP Trunk Channels** and that the **Status** is "Valid", as shown below.

e Edit View Tools Help						
02-IPOSE 🔹 License		-	2 16 - 18	🖳 🔜 🔺 🖌 🔙		
Configuration				er - 🖻	$\times$	<
IPO2-IPOSE	License Remote Server					
世一句 System (1) 田一行( Line (2)	Avaya Softphone Licence	1000	Valid	Never	PLDS Nodal	
E- Control Unit (8)	Basic User	1000	Obsolete	Never	PLDS Nodal	
🗄 🛷 Extension (9)	CTI Link Pro	2	Valid	Never	PLDS Nodal	
🕀 🚹 User (9)	Devlink3 External Recorder	1	Valid	Never	PLDS Nodal	
Group (10)     Short Code (57)	IP500 Universal PRI (Additional cl	na 100	Obsolete	Never	PLDS Nodal	
Service (0)	IPSec Tunnelling	1	Obsolete	Never	PLDS Nodal	
Incoming Call Route	Office Worker	1000	Valid	Never	PLDS Nodal	
- A Directory (0)	Power User	1000	Valid	Never	PLDS Nodal	
	Receptionist	10	Valid	Never	PLDS Nodal	
IP Route (1)	Server Edition R10	150	Valid	Never	PLDS Nodal	=
Account Lode (U)	SIP Trunk Channels	256	Valid	Never	PLDS Nodal	
H- License (22)	SM Trunk Channels	128	Valid	Never	PLDS Nodal	
🕀 🍸 ARS (2)	UMS Web Services	1000	Valid	Never	PLDS Nodal	
🕀 🚋 Location (2)	VMPro Recordings Administrator	s 1	Valid	Never	PLDS Nodal	
	VMPro TTS Professional	40	Valid	Never	PLDS Nodal	

#### 5.2. Administer System

From the configuration tree in the left pane, select **System** under the IP Office system used for SIP trunks connection with SpeechAttendant, to display the system screen in the right pane.

Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later to configure SpeechAttendant. Note that IP Office can support SIP trunks on the LAN1 and/or LAN2 interfaces, and the compliance testing used the LAN1 interface.



Select the **VoIP** sub-tab. Make certain that **SIP Trunks Enable** is checked, as shown below. Retain the default values in the remaining fields.

📶 Avaya IP Office Manager for Serve	er Edition IPO2-IPOSE [10.0.0.1.0 buil	d 53]			
File Edit View Tools He	lp				
IPO2-IPOSE • System	✓ IPO2-IPOSE	-   2 🖻 ·	· 🗐 💽 🖬 🔔	1	
Configuration	E	IP02-IP0SE		<b>M</b>	- 🖻   🗙   🖌   <   >
PO2-IPOSE	System LAN1 LAN2 DNS	Voicemail Telephony	Directory Services Sys	tem Events SMTP	SMDR VoIP Vo + +
	LAN Settings VoIP Network	Topology			
<ul> <li>              ← (*) Line (2)          </li> <li>             ← Control Unit (8)         </li> <li>             ← Extension (9)         </li> <li>             ↓ User (9)         </li> <li>             ∰ Group (10)         </li> </ul>	<ul> <li>H.323 Gatekeeper Enable</li> <li>Auto-create Extension</li> <li>H.323 Signaling over TLS</li> </ul>	Auto-create Us	er 🔳	] H.323 Remote Exten	sion Enable
Short Code (57)  Service (0)  Good Service (0)  Good Service (0)  Good Service (0)  Good Service (0)  Servic	<ul> <li>SIP Trunks Enable</li> <li>SIP Registrar Enable</li> <li>Auto-create Extension/User</li> </ul>			E	SIP Remote Extension
Account Code (0) 	SIP Domain Name	dr220.com		19	
ie → ★ ARS (2) ie → ↓ Location (2) → ☆ Authorization Co	SIP Registrar FQDN	UDP	UDP Port 5060	Rem	ote UDP Port 5060
	Layer 4 Protocol	📝 ТСР	TCP Port 5060	Rem	ote TCP Port 5060
Enter (4)		TLS	TLS Port 5061	Rem Rem	ote TLS Port 5061 👻

TLT; Reviewed: SPOC 4/20/2017

Solution & Interoperability Test Lab Application Notes ©2017 Avaya Inc. All Rights Reserved. 7 of 24 Nuance-IPOSE10 Select the **Telephony** tab, followed by the **Telephony** sub-tab in the right pane. Uncheck **Inhibit Off-Switch Forward/Transfer**, if transfer from SpeechAttendant to PSTN destinations is desired. In the compliance testing, this parameter was disabled.

ver Edition IPO2-IPOSE [10.0.0.1.0 build	53]			
•IP • IPO2-IPOSE	- 2 - 1 - 2 - 1	. 🗸 🕢		
12	IP02-IP0SE*		🗠 - 🗐   🗙	(  🗸   <
System LAN1 LAN2 DNS	Voicemail Telephony Directory Services Sys	stem Events SMTP	SMDR VoIP	VoIP Sec 4
Telephony Park & Page Tones &	& Music Ring Tones SM Call Log TUI			
Dial Delay Time (sec)	4	Compandi	ing Law	
Dial Delay Count	0	Switch		Line
Default No Answer Time (sec)	15	U-Law	t.	🔘 U-Law
Hold Timeout (sec)	120			
Park Timeout (sec)	300	🔘 A-Law	ф -	🔘 A-Law
Ring Delay (sec)	5	DSS Stat	us	
Call Priority Promotion Time (sec)	Disabled	III Auto Ho	14	3
Default Currency	▼ d2U	Dial By N	Jame	
Default Name Priority	Favor Trunk 🔹	Show Ac	rount Code	
Media Connection Preservation	Enabled 👻	🗐 Inhibit O	)ff-Switch Forwar	/d/Transfer
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### 5.3. Administer Line

From the configuration tree in the left pane, right-click on **Line** under the IP Office system used for SIP trunks connection with SpeechAttendant, and select **New**  $\rightarrow$  **SIP Line** from the pop-up list to add a new SIP line.

Select the **Transport** tab. For **ITSP Proxy Address**, enter the IP address of the SpeechAttendant server. Retain the defaults in the remaining fields. Note that SpeechAttendant can support UDP and TLS, and the compliance testing used the UDP protocol.

🕻 Avaya IP Office Manager for Se	rver Edition IPO2-IPOSE (10.0.0.1.0	) build 53]		
File Edit View Tools H IPO2-IPOSE - Line	Help ➡ 8	• 2 6 - 9 • 9 5	1 🗸 🖂 🕢	
Configuration	12	SIP Line - Line 1*		i - □   ×   <   :
Location(2)	SIP Line Transport SIP URI Vo	oIP SIP Credentials SIP Advanced Engineering		
G → G 2 H G 2 H → G 2 H	ITSP Proxy Address 10.64.1 Network Configuration Layer 4 Protocol Use Network Topology Info	UDP Send Port None Listen Port	5060 5060	A V A V
Short Code (57)     Service (0)     Def Incoming Call Rov     Control of the Profile (0)     Def Incoming Profile (0)	Explicit DNS Server(s) Calls Route via Registrar 🖉	0 · 0 · 0 · 0 0 0 · 0 · 0	0	
Account Code (0)	Separate Registrar			

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Solution & Interoperability Test Lab Application Notes ©2017 Avaya Inc. All Rights Reserved. 8 of 24 Nuance-IPOSE10 Select the **SIP URI** tab, and click **Add** to display the **New URI** sub-section. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- Identity: "Auto"
- Incoming Group: An available incoming group number.
- Outgoing Group: An available outgoing group number.
- Max Sessions: The maximum number of simultaneous calls.

🗶 Avaya IP Office Manager for Ser	ver Edition IPO2-IPOSE [10	.0.0.1.0 build 53]	
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Configuration	12	SIP Line - Line 1*	📸 - 🖻   X   🗸   <   >
Location(2) IPO2-IPOSE System (1) IPO2-IPOSE Control Unit (8) Extension (9) Extension (9) Service (0) Firewall Product (1) Firewall Profile (0) Firewall Profile (1) Firewall	SIP Line Transport SIP URI Groups New URI Local URI Contact Display Name Identity Identity Header Forwarding And Tw Originator Number Send Caller ID Diversion Header Registration Incoming Group Outgoing Group Max Sessions	URI       VolP       SIP Credentials       SIP Advanced       Engineering         Local URI       Contact       Display       Identity       Header       Originator Number         Auto       Auto       Auto       Identity       Header       Identity       Identity         Auto       Identity       Identity       Identity       Identity       Identity       Identity         Auto       Identity       Identity       Identity       Identity       Identity       Identity         Auto       Identity       Identity       Identity       Identity       Identity       Identity         None       Identity       Identity       Identity       Identity       Identity       Identity         I       Identity       Identity       Identity       Identity       Identity       Identity         Identity       Identity       Ident	Send Caller ID Diversion Add Remove Edit OK Cancel

Select the **VoIP** tab. For **Codec Selection**, select "Custom" from the drop-down list. Retain the applicable G.711 codec variant in the **Selected** column, in this case "G.711 ULAW 64K".

Check <b>Re-invite Supported</b> .	Retain the default values in t	he remaining fields.
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📶 Avaya IP Office Manager for Sen	Server Edition IPO2-IPOSE [10.0.0.1.0 build 53]				
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IPO2-IPOSE    Line	- 8	- (İ.	26-1	🔺 🔛 📰 🧘 🛹 🔤 🖉	7
Configuration	12	SIP I	Line - Line	1*	iii - 1 ×   <   >
E tocation(2)	SIP Line Transport SIP L	JRI VoIP SIP Credentials SI	P Advanced En	igineering	
=					🔲 Local Hold Music
IPO2-IPOSE					👿 Re-invite Supported
E-~ Control Unit (8)	Codec Selection	Custom			Codec Lockdown
Ertension (9)		Unused		Selected	Allow Direct Media Path
🕀 🔐 Oser (9)		G.711 ALAW 64K	>>>	G.711 ULAW 64K	E Force direct media with
🕀 🥦 Short Code (57)		G.729(a) 8K CS-ACELP			
Service (0)			Ŷ		PRACK/100rel Supported
Directory (0)					
Time Profile (0)			<<<		
Account Code (0)					
License (22)					
🕀 🌆 User Rights (11)			>>>		
				<u></u>	
	Eav Transport Support	None			-
E- A Location (2)					
- 3: Basking Ric	DTMF Support	RFC2833/RFC4733			•
Authorization Cc	Media Security	Disabled		•	
÷ System (1)		9-		254	
⊕ - 行了 Line (4) 🗧					

### 5.4. Administer Incoming Call Route

From the configuration tree in the left pane, right-click on **Incoming Call Route** under the IP Office system used for SIP trunks connection with SpeechAttendant, and select **New** from the pop-up list to add a new route for incoming calls from SpeechAttendant.

For **Line Group Id**, select the incoming group number from **Section 5.3**, in this case "1". Retain the default value in the remaining fields.

🐮 Avaya IP Office Manager for Server I	Edition IPO2-IPOSE [10.0.0.1.0 bu	ild 53]	
File Edit View Tools Help	- 0 202520000		
Confirmation			
Configuration	2	U	
Location(2)     PO2-IPOSE     System (1)     IPO2-IPOSE     f Line (3)         1         2         8     Control Unit (8)     Extension (9)     User (9)     Group (10)     Service (0)     Incorring Call Route     8 3035300000     Time Profile (0)     Time Profile (0)     I PR oute (1)     Account Code (0)     License (22)     User Rights (11)     ARS (2)     S0: Main     S1: Main     S1: Main	Standard Voice Recording Bearer Capability Line Group ID Incoming Number Incoming Sub Address Incoming CLI Locale Priority Tag Hold Music Source Ring Tone Override	Any Voice   I   I   I   I   I   System Source   None	

Select the **Destinations** tab. For **Destination**, enter "." to match any dialed number from SpeechAttendant.

🕐 Avaya IP Office Manager for Server I	Edition IPO2-IPOSE [	10.0.0.1.0 build 53]			
File Edit View Tools Help IPO2-IPOSE - Incoming C	all Route 🝷 8	30353)00000	•	I	
Configuration	<b>I</b> ₽		0*		📸 - 🖻   🗙   🗸   <   >
IPO2-IPOSE	Standard Voice F	lecording Destination	15		
trane (3)	TimeProf	le	Destination	Fallback Ext	ension
🕀 \prec Control Unit (8)	I Unit (8) Default Value			•	•
ter (9) ter (9)					
🖶 🎲 Group (10) 🕀 🕬 Short Code (57)					
Service (0)					
- A Directory (0)					

### 5.5. Administer Short Code

From the configuration tree in the left pane, right-click on **Short Code** under the IP Office used for SIP trunks connection with SpeechAttendant, and select **New** from the pop-up list to add a new short code for outgoing calls to SpeechAttendant. In the compliance testing, all calls to 2155x are routed over the SIP trunks to SpeechAttendant.

For **Code**, enter the appropriate value, in this case "2155x". For **Telephone Number**, enter "." to match the dialed number.

For **Line Group ID**, enter the outgoing group number from **Section 5.3**. Retain the default values in the remaining fields.

🗂 Avaya IP Office Manager for Serve	er Edition IPO2-IPOSE [10.0.0.1	.0 build 53]	
File Edit View Tools He	lp		
IPO2-IPOSE	de 🔫 2155x	- 🔍 🖻 - 🖬 💽 🖬 📣 🛹 🕾 🛍	9
Configuration	12	<short code:0="">: Dial*</short>	
	Short Code Code Feature	2155x Dial 🔹	
Control Unit (8)	Telephone Number Line Group ID	1	
<ul> <li>⑦ 2 User (9)</li> <li>⑦ 30 (700µ (10))</li> <li>⑦ 9 Short Code (58)</li> <li>□ 30 Service (0)</li> <li>□ 10 Incoming Call Rout</li> </ul>	Locale Force Account Code Force Authorization Code		

Repeat this section to add similar short code for the expansion IP Office system, which is named **IPO2-IP500V2** in this case. For **Line Group ID**, select the applicable outgoing group ID for the SCN trunk that connects to the primary IP Office system, in this case "999999" as shown below.

File Edit View Tools H	elp		
IPO2-IP500V2 • Short Co	de 🔫 *02	- i 🚨 🗁 - 🔜 i 🛋 🔜 🖌 🛹 🛶 🐼	
Configuration	32	<short code:0="">: Dial*</short>	🚽 - 🖻   🗙   🖌   <
Directory(0)	Short Code		
Account Code(0)	Code	2155×	
⊕−¶s User Rights(11) ⊕−∰ Location(2)	Feature	Dial 🔹	
i⊞	Telephone Number		
in ····································	Line Group ID	999999 👻	
E Control Unit (4)	Locale	<b>_</b>	
🖶 🐗 Extension (31) 🕀 🧃 User (10)	Force Account Code		
Group (2)	Force Authorization Code		

### 6. 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 NSServer.cfg
- Restart SpeechAttendant

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

### 6.1. Launch Admin Tools

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



### 6.2. Administer Configuration Panel

The Admin Tools screen is displayed. Select Configuration Panel.

🛱 Admin T 🗕 🗖 🗙
SpeechAttendant <sup>°</sup>
Start
Stop
Monitor
Directory Search
Prompt Recorder
Phone Directory and Menu Editor
Report Generator
Contiguration Panel
Backup / Restore
Data import
Help

Solution & Interoperability Test Lab Application Notes ©2017 Avaya Inc. All Rights Reserved. The **Configuration password** screen is displayed. Select "Level 2" and enter the appropriate credential.

Level 2
Password

The **Configuration panel – level 2** screen is displayed next. In the upper left pane, set **PBX** to "IPOFFICE-SIP" and **Voice board** to "SIP", as shown below.

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 primary IP Office LAN1 IP address and UDP port number from **Section 5.2**. Note that any name can be used as part of **SIP user agent URI**, and in the compliance testing the name "sa1" was used.

Scroll the screen in the left pane as necessary, to locate the **Force Display DNIS** parameter. Set the value to "YES", as shown below.

٩		Cont	figuration panel - leve	el 2		-	. C	3	×
File Tools Help									
PBX Voice board					SIP telephony gateway				
IPOFFICE-SIP	-	-							
J		_			10.64.101.234:5060				
Setting		Current value	Default	^	1				
Prompt Volume		0	0						
Agent action when port idle									
Telephony type		SIP	SIP						
SIP telephony gateway		10.64.101.234.5060	%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:sa1@10.64.101.234	sip:nvp@%HOSTNAME%						
SIP consult on suppervised		TRUE	TRUE						
SIP max call attempts		40	40						
SIP reattempt call on		503	503						
SIP call reattempt delay		500	500						
SIP send 503 on busy		FALSE	FALSE						
SIP connect timeout on transfer SIP TSS RTP bridge		60	60						
SIP use original caller ID on consultation transfer		TRUE	TRUE						
SIP TCP enabled		FALSE	FALSE						
SIP security (TLS)									
SIP TLS Port									
SIP TLS certificate path									
SIP TLS domain name									
✓ Force Display DNIS	->	YES	NO						
Allow pound key as prefix of extension for browser SIP aai based on		FALSE	FALSE	≡					
SIP aai header on transfer				~	Bestore				
< 111				>	Trestore				

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### 6.3. Administer Phone Directory and Menu Editor

From the Admin Tools screen, select Phone Directory and Menu Editor.

🛱 Admin T 💻 🗖 🗙
SpeechAttendant <sup>®</sup>
Start
Stop
Monitor
Directory Search
Prompt Recorder
Enone Directory and Menu Editor
Report Generator
Configuration Panel
Backup / Restore
Data import
Help

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

Phone I	Directory and Menu Editor
Ø	SpeechAttendant
Administrator	
Password	
ок	Cancel

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

٢	Menu Editor
File Edit View Tools Help	
x 🖻 🛱 🕏 🕂 🕞 🔂	Ĩ <sub>+</sub> → ( <sub>+</sub> 4) <sub>+</sub> Ĩ <u>,</u> ( <sub>2</sub> 4) <sub>2</sub> ( <sup>2</sup> , III ( <sup>3</sup> ) ( <sup>2</sup> )
Menus	Content of "Toplevel Menu\"
⊞ d <b>iğî</b> Toplevel Menu	# A B C D E F G H I J K L M N O P Q R S T U V W X Y Z
	↓ Name Depar Locati SR Number Phon D A DTM
	🛱 Cellular Phones X
	Croft, Lara X 202 Office X
	Kent, Clark X 201 Office X

Follow reference [2] to create additional entry points in the left pane and additional directory entries in the right pane pertinent to customer needs. The screen below shows the entry points and directory entries used in the compliance testing. Note that operator destinations were configured as part of the entry points, and that all directory entries were configured with the supervised transfer method (not shown below), as required in this integration.

Select **Tools**  $\rightarrow$  **Dialing Properties** from the top menu.

٩	Menu Editor
File Edit View Tools Help	
x 🖻 🛱 🖻 🔶 🕞 😒	<sup>™</sup> <sub>+</sub> - ( <sub>+</sub> 4) <sub>+</sub> <sup>™</sup> <sub>2</sub> ( <sub>2</sub> 4) <sub>2</sub> <sup>™</sup> <sub>2</sub> <sup>™</sup> <sub>2</sub> <sup>™</sup> <sub>2</sub> <sup>™</sup> <sub>3</sub> <sup>™</sup> <sub>3</sub> <sup>™</sup> <sub>3</sub>
Menus	Content of "Toplevel Menuk"
⊡∰ Toplevel Menu ⊕≣ Cellular Phones	# A B C D E F G H I J K L M N O P Q R S T U V W X Y Z
- 🖺 Inbound call from 732-888-3737	∓↓         Name         Depar         Locati         SR         Number         Phon         D         A         DTM
	🛱 Cellular Phones X
Inbound call to 21551	Croft, Lara X 202 Office X
「「Inbound call to 21552」 「開発 Task D. T. M. F. Manua	Expansion Digital X 22021 Office X
Inst D I M F Menu	Expansion Group X 22991 Office X
	Expansion H 3 2 3 X 22031 Office X
	Expansion SIP X 22041 Office X
	Inbound call from 732-888-3737 X
	Inbound call from 908-953-2222 X
	Inbound call to 21551 X
	Inbound call to 21552 X
	Kent, Clark X 201 Office X
	Primary Group X 21991 Office X
	Primary H 3 2 3 X 21031 Office X
	C Primary SIP X 21041 Office X
	I Test DT M F Menu X
	Three Minutes Announcement X
3 entries	
Phone Directory	Data import

### 6.4. Administer Dialing Properties

The **Default dialing properties** screen is displayed. Follow reference [2] to update and add dialing properties entries as necessary for routing of calls pertinent to the customer network.

An entry needs to exist for routing of calls to the operator extension specified as part of entry points. In the compliance testing, the only operator extension used was "21035", and the first entry in the screen below was created for the operator with **Transfer type** of "SUPERVISED", as shown below.

The second entry in the screen below was added for routing of calls to internal destinations consisting of 5-digit extensions. In the compliance testing, all directory entries associated with internal destinations were configured in **Section 6.3** with supervised as transfer type, therefore the **Transfer type** below was left at the default value of "FROM ENTRY".

In the compliance testing, all other entries below were left at their default settings. Rearrange the order of the entries as necessary to reflect the desired matching order for the network.

21	1025	nouting number	Announce number	DTMF input	Number type	Transfer type	Comment
??		21035			INTERNAL	SUPERVISED	Operator transfer
	7777	?????	22225	?????	INTERNAL	FROM ENTRY	5 digital internal
+1	1-???-???-????x????	????	????	????	INTERNAL	FROM ENTRY	Default internal
??	77-777-7777	9-???-???-????	???-???-????		LOCAL	FROM ENTRY	Default local
+1	1-???-???-????	9-1-???-?????????	777-777-7777		LONG DISTANCE	FROM ENTRY	Default long distance
+?	??-??-??-??-??	9-011-??-??-??-??-??-??	77-77-77-77-77-77		INTERNATIONAL	FROM ENTRY	Default international
+?	777-77-77-77-77	9-011-???-??-??-??-??	777-77-77-77-77		INTERNATIONAL	FROM ENTRY	Default international
L.							
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#### 6.5. Administer NSServer.cfg

From the SpeechAttendant server, navigate to the SpeechAttendant **config** directory, in this case **C:\NuanceVoice\Nuance\VoicePlatform\ Speech Server\config**, and edit the **NSServer.cfg** file shown below.

🎉 l 💽 🗓 = 1	C:\NuanceVoice\Nuance\\	/oice Platform\Speech Se	erver\config	-	. 🗆 🗙
File Home Share	View				~ O
🕞 💿 🔻 🕇 🚺 « Nu	anceVoice 🕨 Nuance 🕨 Voice Platform	▶ Speech Server ▶ config	V 🖒 Se	arch config	Q
☆ Favorites	Name	Date modified	Туре	Size	^
Desktop	NSSserver.cfg	3/10/2017 12:17 PM	CFG File	57 KB	
🐊 Downloads	NSSserver-NVP.tl	11/9/2016 7:34 AM	TL File	57 KB	
3 Recent places ≡	NVP-httpd.conf	3/9/2017 11:33 AM	Text Document	2 KB	
	NVP-httpd.conf.tl	10/26/2016 6:08 PM	TL File	2 KB	
🌉 This PC	osrrecorderErrors.xml	3/22/2016 1:51 PM	XML File	2 KB	
🖵 A on TLT-PC20-E	osrspeechrecogErrors.xml	3/22/2016 1:52 PM	XML File	4 KB	=
💬 C on TLT-PC20-E	prsspeechrecogErrors.xml	3/22/2016 1:52 PM	XML File	5 KB	
🖵 D on TLT-PC20-E	rsspeechsynthErrors.xml	6/22/2012 9:51 AM	XML File	4 KB	
📔 Desktop	SBcacheErrors.xml	10/2/2012 3:52 PM	XML File	4 KB	
Documents	SBinetErrors.xml	10/2/2012 3:52 PM	XML File	10 KB	
\rm Downloads	serverErrors.xml	12/9/2015 1:59 PM	XML File	6 KB	
🚍 E on TLT-PC20-D 🗸	tones.all.off	1/8/2013 2:04 PM	OFF File	5 KB	V
23 items 1 item selected	56.3 KB				

Scroll down to locate the **server.mrcp2.rsspeechsynth.rtpPacketSamples** parameter, and update the value to "160" for use of 20ms for RTP packet size.



### 6.6. Restart SpeechAttendant

From the **Admin Tools** screen, select **Start** to restart the application for all changes to take effect.

SpeechAttendant Stop Monitor Directory Search Prompt Recorder Phone Directory and Menu Editor Report Generator Configuration Panel Backup / Restore Data import	Admin T
SrarrSrarr	SpeechAttendant <sup>®</sup>
Stop Monitor Directory Search Prompt Recorder Phone Directory and Menu Editor Report Generator Configuration Panel Backup / Restore Data import	Sran
Monitor Directory Search Prompt Recorder Phone Directory and Menu Editor Report Generator Configuration Panel Backup / Restore Data import	Stop
Directory Search Prompt Recorder Phone Directory and Menu Editor Report Generator Configuration Panel Backup / Restore Data import	Monitor
Prompt Recorder Phone Directory and Menu Editor Report Generator Configuration Panel Backup / Restore Data import	Directory Search
Phone Directory and Menu Editor Report Generator Configuration Panel Backup / Restore Data import	Prompt Recorder
Report Generator Configuration Panel Backup / Restore Data import	Phone Directory and Menu Editor
Configuration Panel Backup / Restore Data import	Report Generator
Backup / Restore	Configuration Panel
Data import	Backup / Restore
11-1-	Data import
Heip	Help

# 7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of IP Office and SpeechAttendant.

### 7.1. Verify Avaya IP Office

From the Avaya IP Office Manager for Server Edition IPO2-IPOSE screen shown in Section 5.1, select File  $\rightarrow$  Advanced  $\rightarrow$  System Status to launch the System Status application, and log in using the appropriate credentials.

The Avaya IP Office System Status – IPO2-IPOSE screen is displayed. Expand Trunks in the left pane and select the SIP line from Section 5.3, in this case "1".

Verify that the **SIP Trunk Summary** screen shows all channels with **Current State** of "Idle", as shown below.



### 7.2. 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.

|--|

speech Attendant
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.

SpeechAttendar	ו <b>ל</b> © סדא8gke									
Description: Auto Attendant	Version: SA 12.2	2.0 (with E	501) (no ho	t fix installed	d)					
Sections	System summary									
Summary status	Uptime:	1 days 2 hours 8 minutes 7 seconds.								
Reports	Served sessions:	11 total (1 currently in memory)								
Alarms  OSA Servlet  Configuration  Installation log	Served requests: Telephony 🍺 :	153 11 calls no call	s so far for in progress	WIN-LDONO7 (concurent )	DT 2017)					
		win-ld0n0tk8gke								
Replication Monitor		CHN	Status	Calls	DNIS	CLID	EP	Function	Menu	Action
Replication Status		III	idle	11						
Replication Errors			idle							
		III	idle							
Call Logs		Ш	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.



## 8. Conclusion

These Application Notes describe the configuration steps required for Nuance SpeechAttendant 12.2 to successfully interoperate with Avaya IP Office Server Edition 10.0 using SIP trunks. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

## 9. Additional References

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

- **1.** *Administering Avaya IP Office*<sup>™</sup> *Platform with Manager*, Release 10.0, September 2016, available at <u>http://support.avaya.com</u>.
- **2.** *Nuance SpeechAttendant Nuance OpenSpeech Attendant Administration Guide*, April 2014, available at <u>https://network.nuance.com/portal/server.pt</u>.

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