



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

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# **Application Notes for Nuance SpeechAttendant 12.1 with Avaya IP Office Server Edition 9.1 – Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps required for Nuance SpeechAttendant 12.1 to interoperate with Avaya IP Office Server Edition 9.1 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 IP Office Server Edition 9.1 using SIP trunks. Nuance SpeechAttendant automates call routing by asking callers to speak the name or dial the extension of a destination.

The Avaya IP Office Server Edition configuration consisted of two Avaya 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 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 INVITE and SIP REFER to perform supervised transfer of calls to destinations on the primary IP Office system at the Main site, destinations on the expansion IP Office system at the Remote site, and destinations on the PSTN.

The SIP trunks connection from Nuance SpeechAttendant can be with either the primary Linux server or the expansion IP500V2 IP Office systems. 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, resiliency, and simultaneous calls.

The feature testing call flows included calls with IP Office resources on the primary linux system, calls with IP Office resources on the expansion IP500V2 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.711MU 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.
- When a user with a digital telephone on the expansion IP Office system dialed SpeechAttendant for directory transfer to an internal destination with Follow Me or Forwarding set to a user on the primary IP Office system or on the PSTN, then the redirected destination cannot hear the audio from the digital caller. This is not expected to be a common scenario, and has been reported to the IP Office product team for investigation.

## 2.3. Support

Technical support on SpeechAttendant can be obtained through the following:

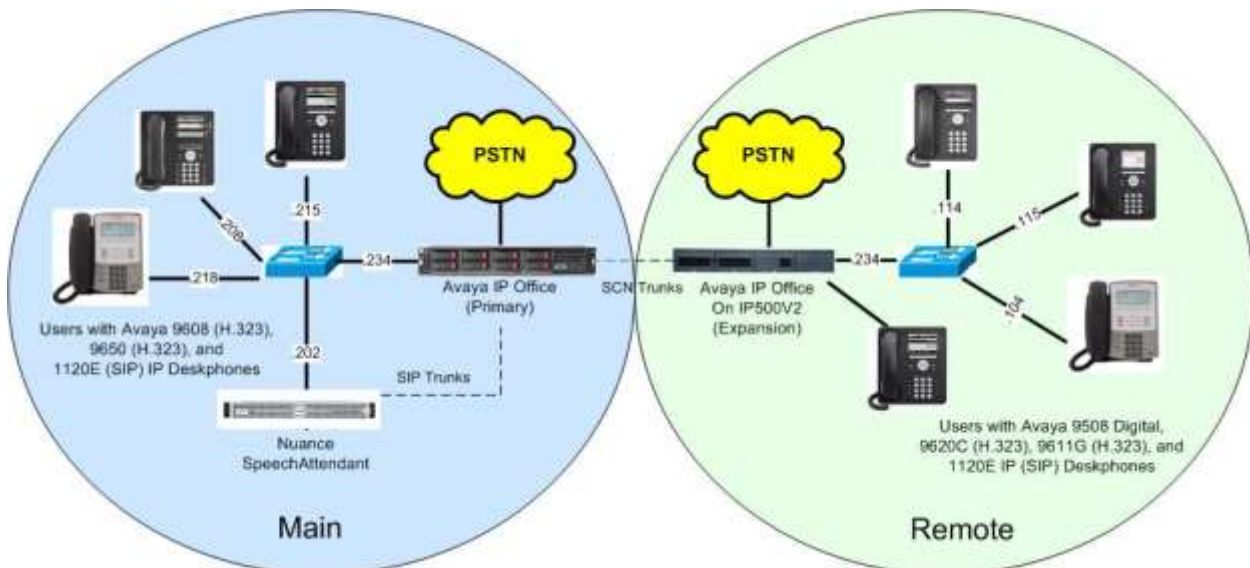
- **Phone:** (866) 434-2564 or (514) 390-3922
- **Email:** [SpeechAttendant.Support@nuance.com](mailto:SpeechAttendant.Support@nuance.com)
- **Web :** [www.network.nuance.com](http://www.network.nuance.com)

### 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 cross systems PSTN scenarios. As shown in **Figure 1**, SIP trunks were used between 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 connection 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
<b>Main Site</b>	
Avaya IP Office Server Edition (Primary) in Virtual Environment	9.1.400.137
Avaya 9650 IP Deskphone (H.323)	3.250A
Avaya 9608 IP Deskphone (H.323)	6.6029
Avaya 1120E IP Deskphone (SIP)	4.4.18.0
Nuance SpeechAttendant on Microsoft Windows Server 2012	12.1.0 HotFix 1210HF01, 1210HF02 R2 Standard
<b>Remote Site</b>	
Avaya IP Office on IP500 V2 (Expansion)	9.1.400.137
Avaya 9620C IP Deskphones (H.323)	3.250A
Avaya 9611G IP Deskphone (H.323)	6.6029
Avaya 1120E IP Deskphone (SIP)	4.4.18.0
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:

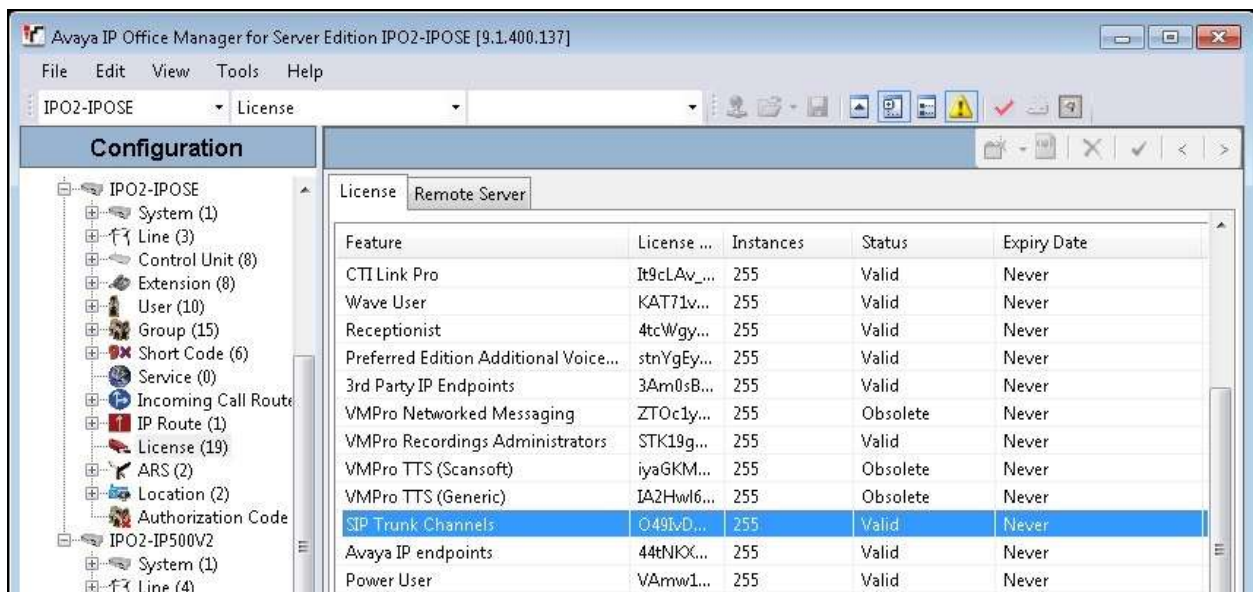
- Verify license
- Administer system
- Administer line
- Administer incoming call route
- Administer short code

### 5.1. Verify License

From a PC running the IP Office Manager application, select **Start → Programs → IP Office → 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.

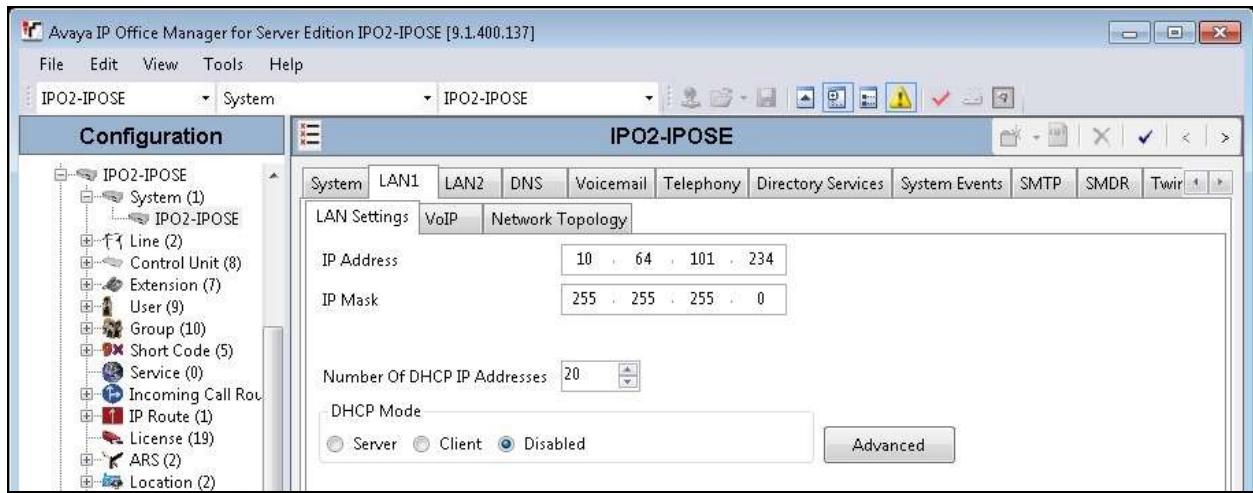


Feature	License ...	Instances	Status	Expiry Date
CTI Link Pro	It9cLAv...	255	Valid	Never
Wave User	KAT71v...	255	Valid	Never
Receptionist	4tcWgy...	255	Valid	Never
Preferred Edition Additional Voice...	stnYgEy...	255	Valid	Never
3rd Party IP Endpoints	3Am0sB...	255	Valid	Never
VMPPro Networked Messaging	ZTOc1y...	255	Obsolete	Never
VMPPro Recordings Administrators	STK19g...	255	Valid	Never
VMPPro TTS (Scansoft)	iyaGKM...	255	Obsolete	Never
VMPPro TTS (Generic)	IA2Hwl6...	255	Obsolete	Never
<b>SIP Trunk Channels</b>	<b>O49lvD...</b>	<b>255</b>	<b>Valid</b>	<b>Never</b>
Avaya IP endpoints	44tNKOC...	255	Valid	Never
Power User	VAmw1...	255	Valid	Never

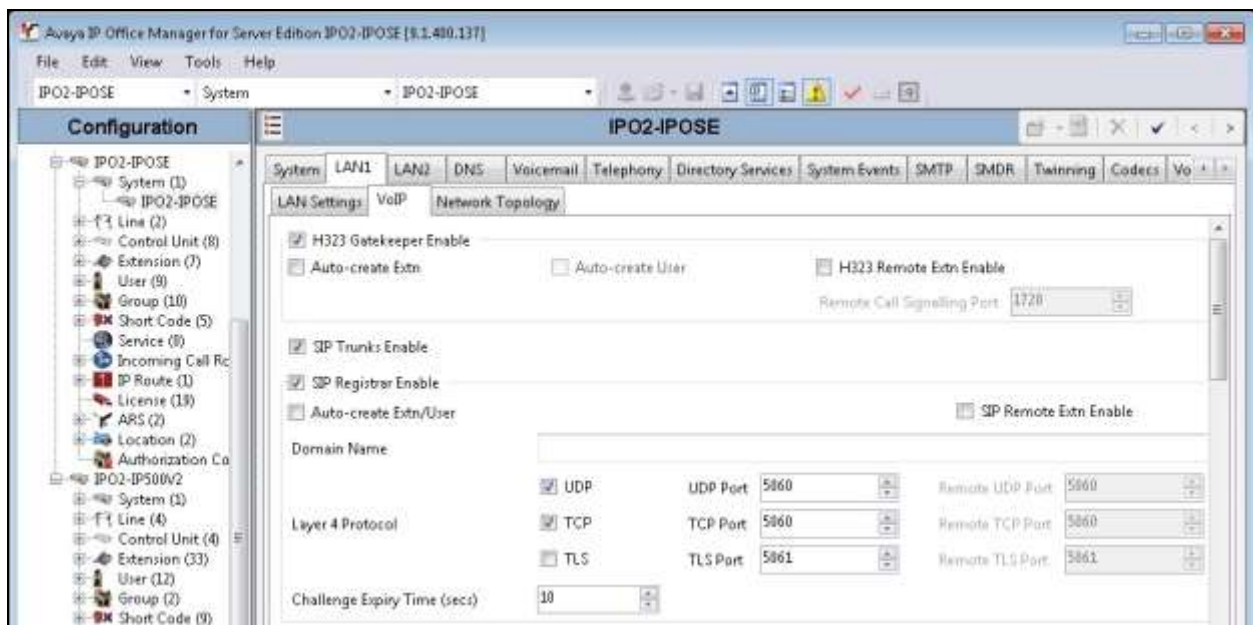
## 5.2. Administer System

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.



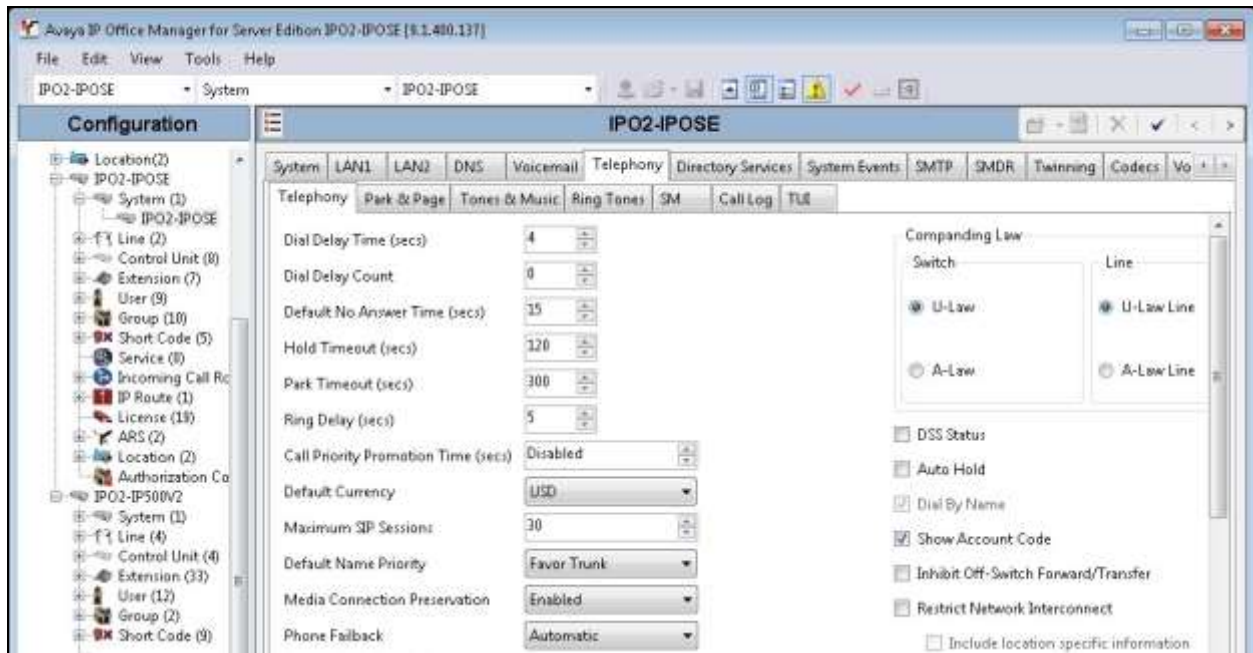
Select the **VoIP** sub-tab. Make certain that **SIP Trunks Enable** is checked, as shown below. Retain the default values in the remaining fields.





Select the **Telephony** tab, followed by the **Telephony** sub-tab in the right pane. Uncheck **Inhibit Off-Switch Forward/Transfer** to allow call forwarding and transfer with SpeechAttendant over SIP trunks.

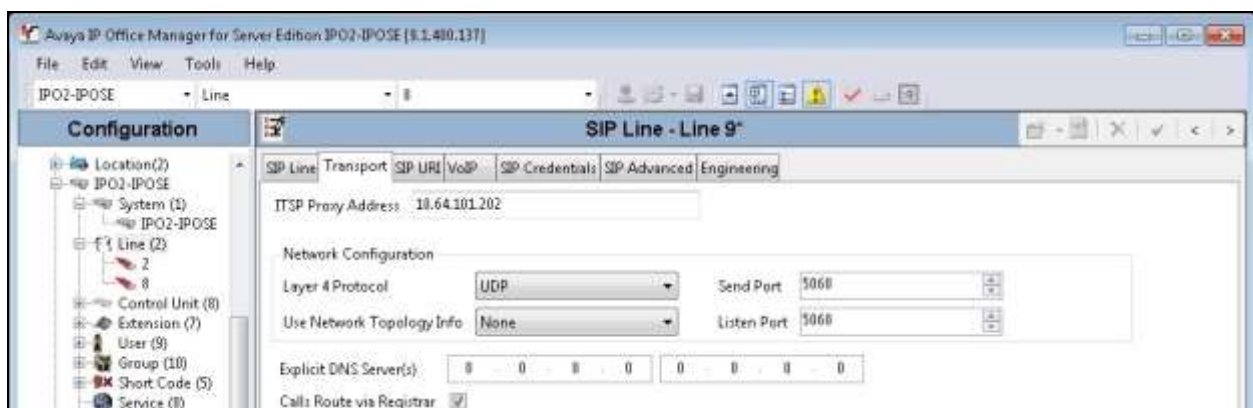
For **Maximum SIP Sessions**, set to the desired total number of SIP telephone and trunk calls that can occur at the same time.



### 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 → SIP Line** from the pop-up list to add a new SIP line.

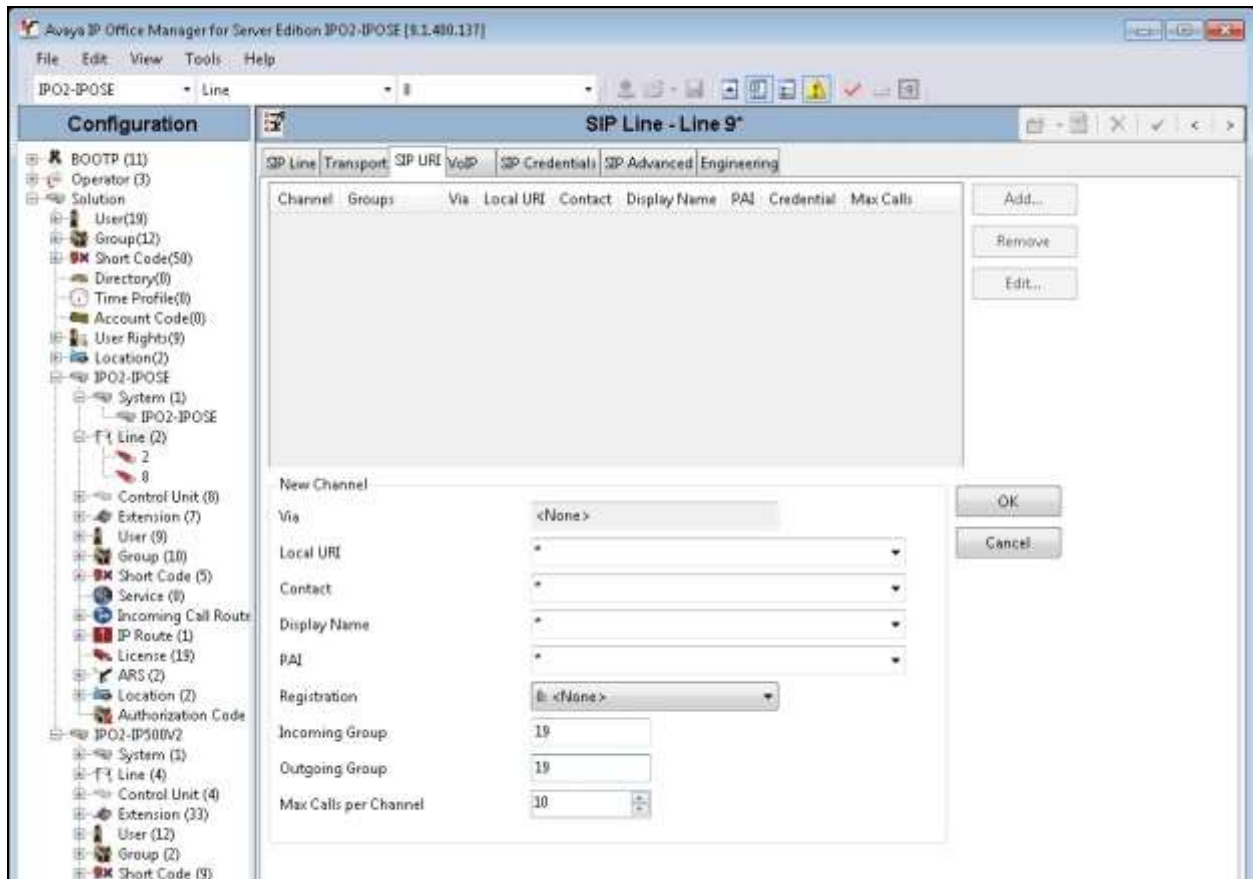
Select the **Transport** tab. For **ITSP Proxy Address**, enter the IP address of the SpeechAttendant server. Retain the defaults in the remaining fields. Note that SpeechAttendant can support UDP, TCP, and TLS, and the compliance testing used the UDP protocol.





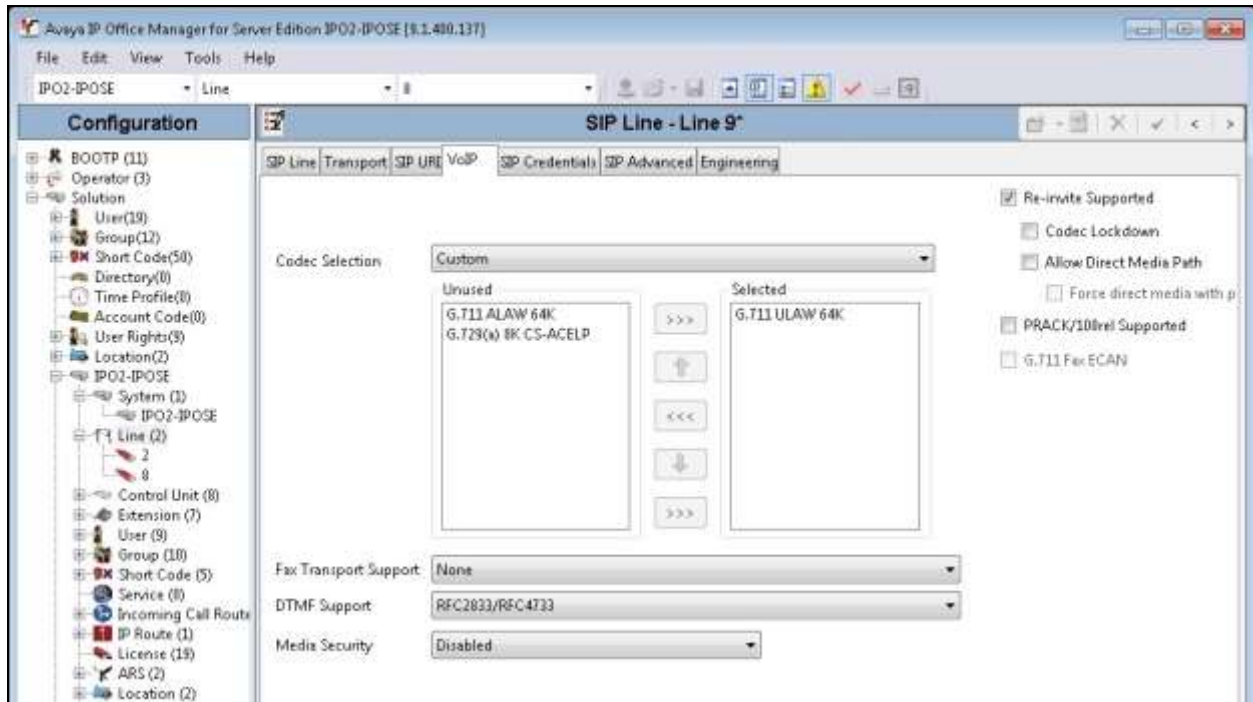
Select the **SIP URI** tab, and click **Add** to display the **New Channel** sub-section. Enter the wildcard character “\*” for **Local URI**, **Contact**, **Display Name**, and **PAI**.

For **Incoming Group** and **Outgoing Group**, enter available group numbers. Set **Max Calls per Channel** to support the applicable maximum number of simultaneous calls. Retain the default values in the remaining fields.



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”.

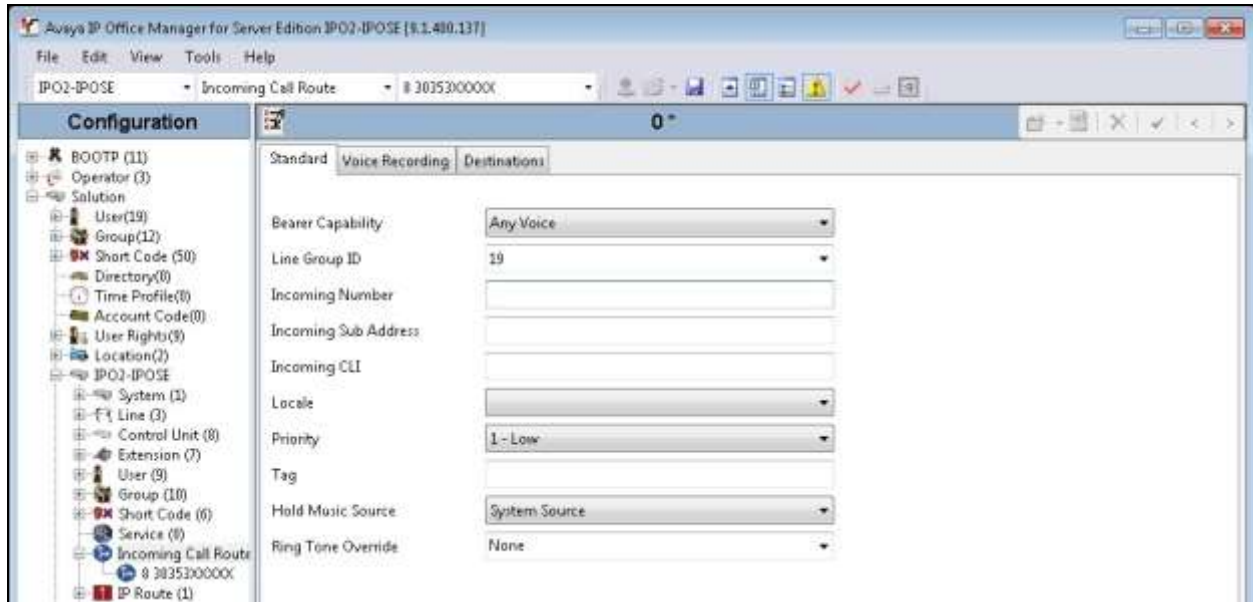
Check **Re-invite Supported**. Retain the default values in the remaining fields.



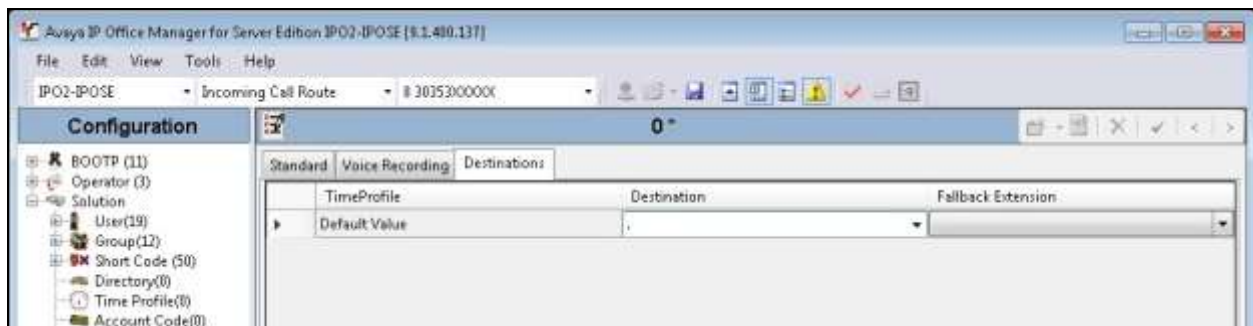
## 5.4. Administer Incoming Call Route

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 “19”. Retain the default value in the remaining fields.



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

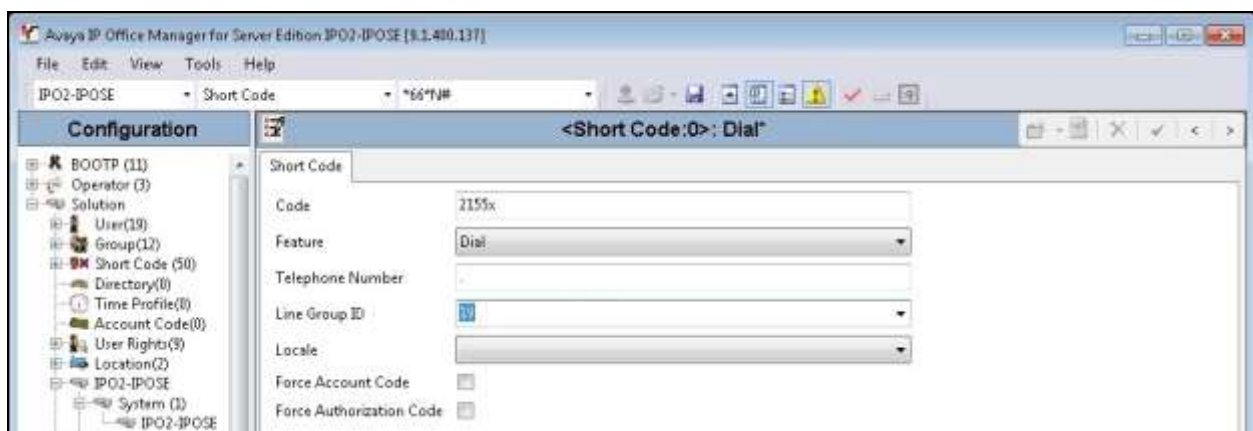


## 5.5. Administer Short Code

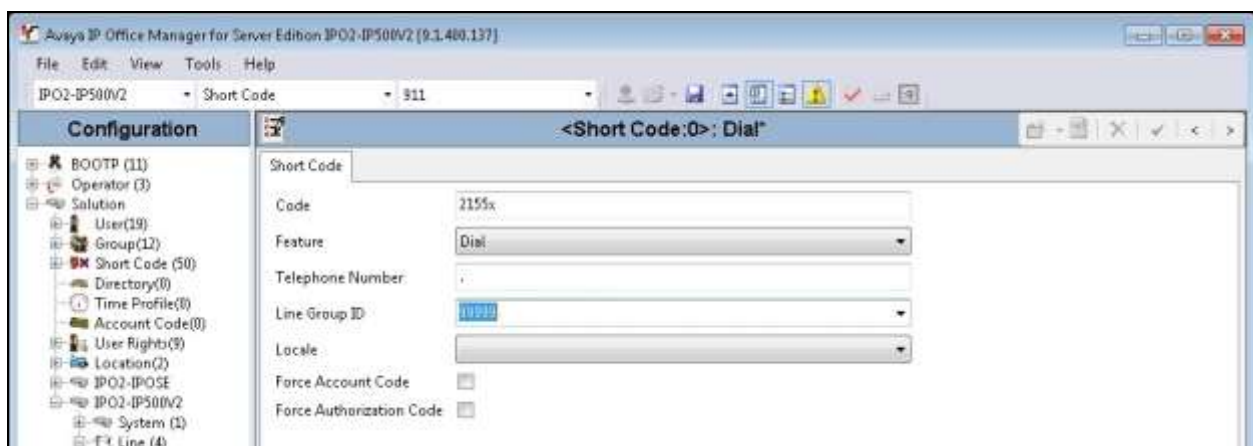
From the configuration tree in the left pane, right-click on **Short Code** under the IP Office used for SIP trunks connection with SpeechAttendant, and select **New** from the pop-up list to add a new short code for outgoing calls to SpeechAttendant. In the compliance testing, all calls to 2155x are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



Repeat this section to add similar short code for the other IP Office system, in this case the expansion **IPO2-IP500V2** system. For **Line Group ID**, select the applicable outgoing group ID of the SCN trunk with the primary IP Office system, as shown below.



## 6. Configure Nuance SpeechAttendant

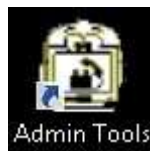
This section provides the procedures for configuring SpeechAttendant. The procedures include the following areas:

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

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**.



The **Configuration password** screen is displayed. Select “Level 2” and enter the appropriate credential.

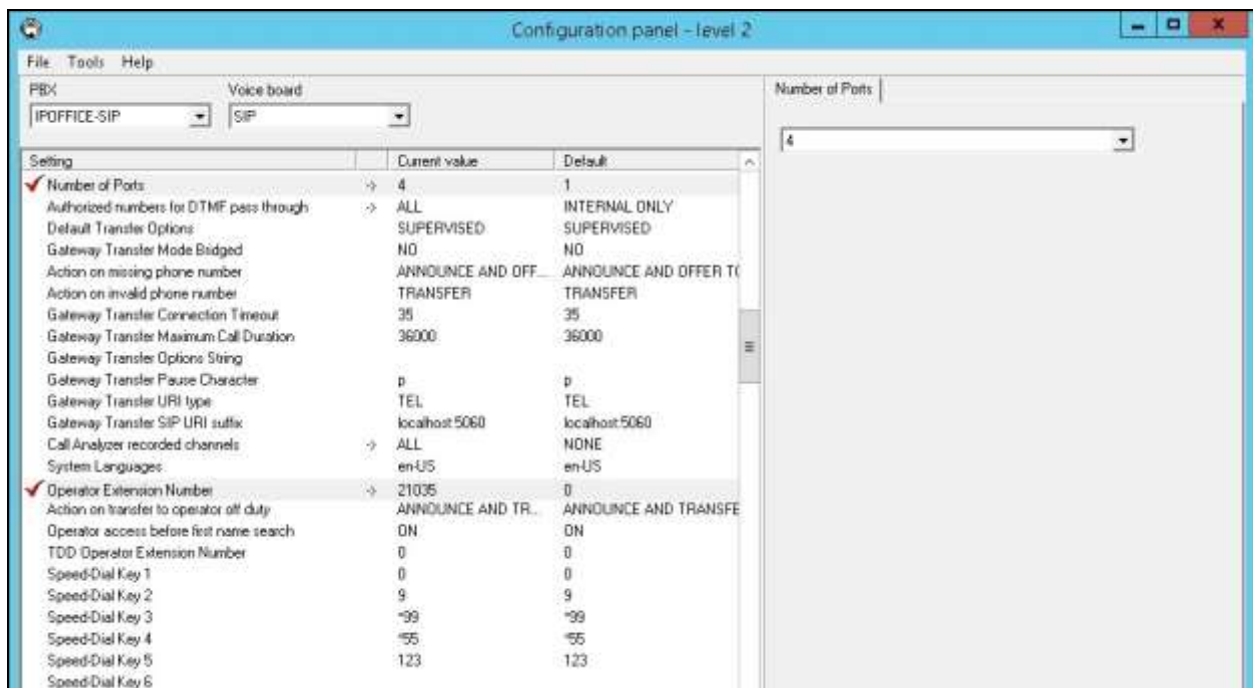


A dialog box titled "Configuration password" with a light blue border. It contains two input fields: "Access level" with a dropdown menu showing "Level 2", and "Password" with an empty text box. At the bottom are "OK" and "Cancel" buttons.

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, and update the **Number of Ports** parameter to the number allowed for by the SpeechAttendant license, in this case “4”.

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

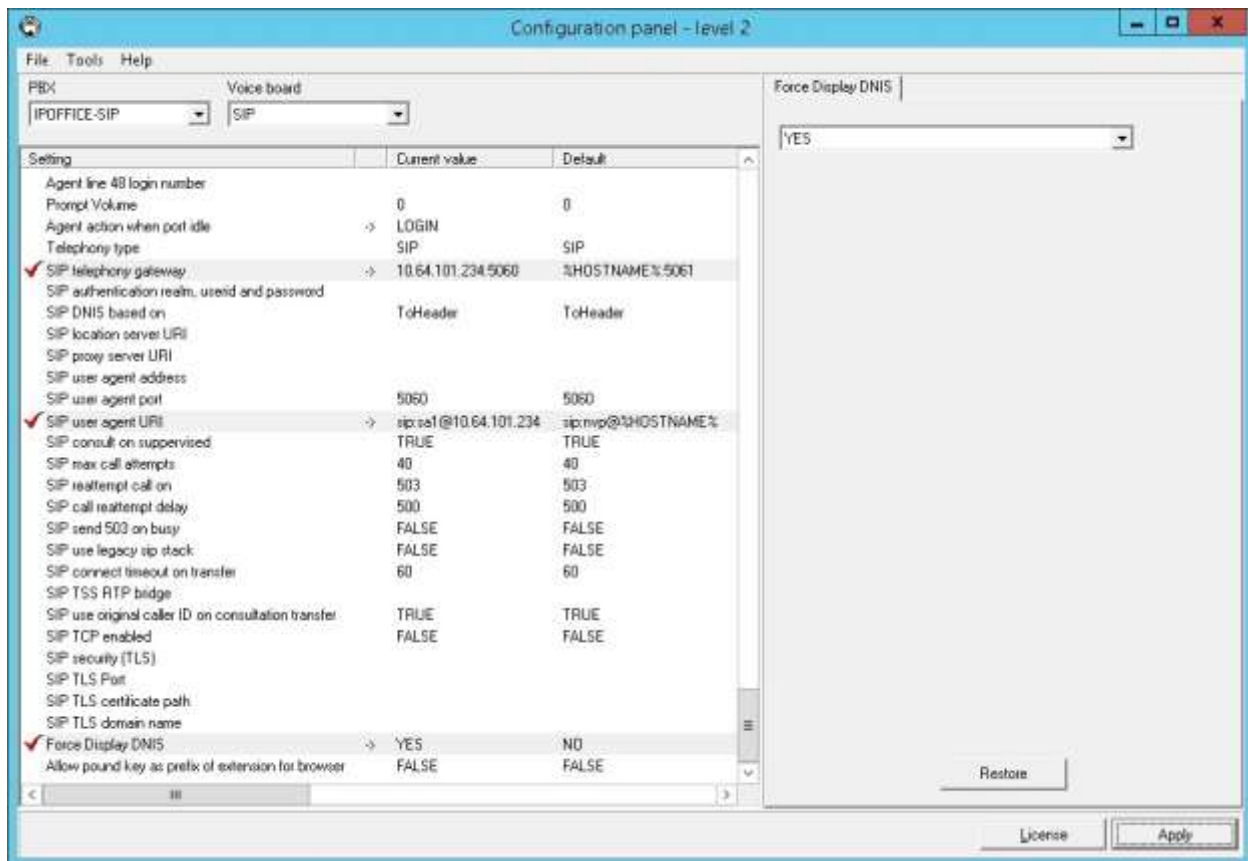


A screenshot of the "Configuration panel - level 2" window. The window has a menu bar (File, Tools, Help) and a title bar. The main area is divided into two panes. The left pane shows a list of settings with columns for "Setting", "Current value", and "Default". The right pane shows a "Number of Ports" dropdown menu set to "4".

Setting	Current value	Default
✓ Number of Ports	4	1
Authorized numbers for DTMF pass through	ALL	INTERNAL ONLY
Default Transfer Options	SUPERVISED	SUPERVISED
Gateway Transfer Mode Bridged	NO	NO
Action on missing phone number	ANNOUNCE AND OFF...	ANNOUNCE AND OFFER TO
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	ALL	NONE
System Languages	en-US	en-US
✓ Operator Extension Number	21035	0
Action on transfer to operator off duty	ANNOUNCE AND TR...	ANNOUNCE AND TRANSFE
Operator access before first name search	ON	ON
TDD Operator Extension Number	0	0
Speed-Dial Key 1	0	0
Speed-Dial Key 2	9	9
Speed-Dial Key 3	*99	*99
Speed-Dial Key 4	*55	*55
Speed-Dial Key 5	123	123
Speed-Dial Key 6		

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**, as shown below. Note that any name can be used as part of **SIP user agent URI**, and in the compliance testing the “sa1” name was used.

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





### 6.3. Administer Phone Directory and Menu Editor

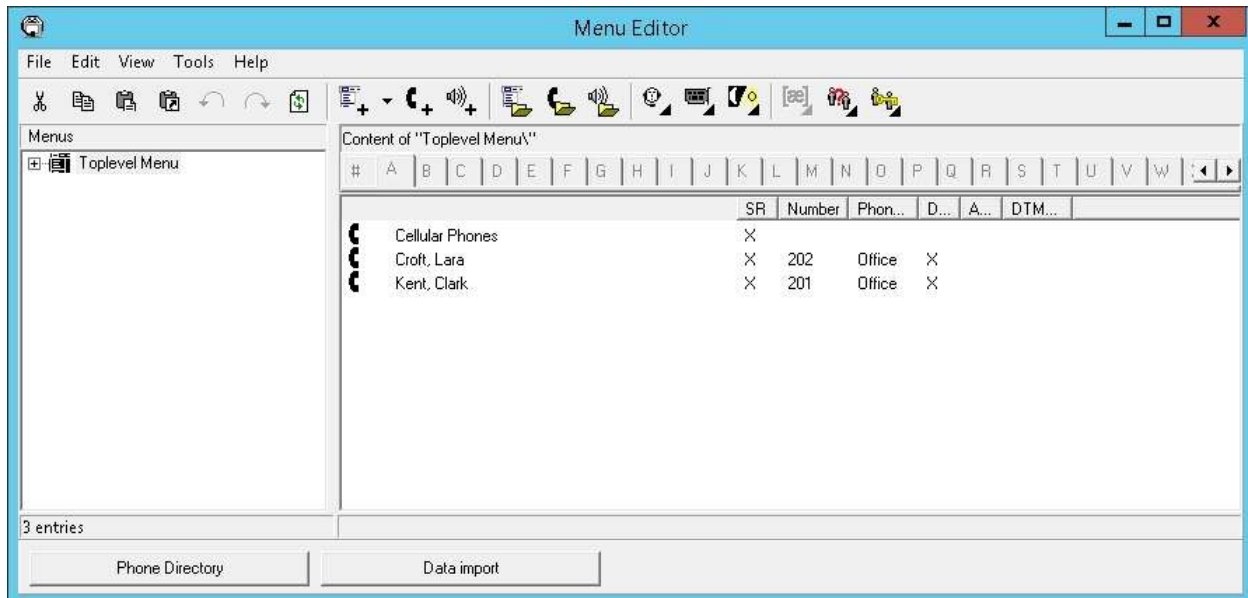
From the **Admin Tools** screen, select **Phone Directory and Menu Editor**.



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

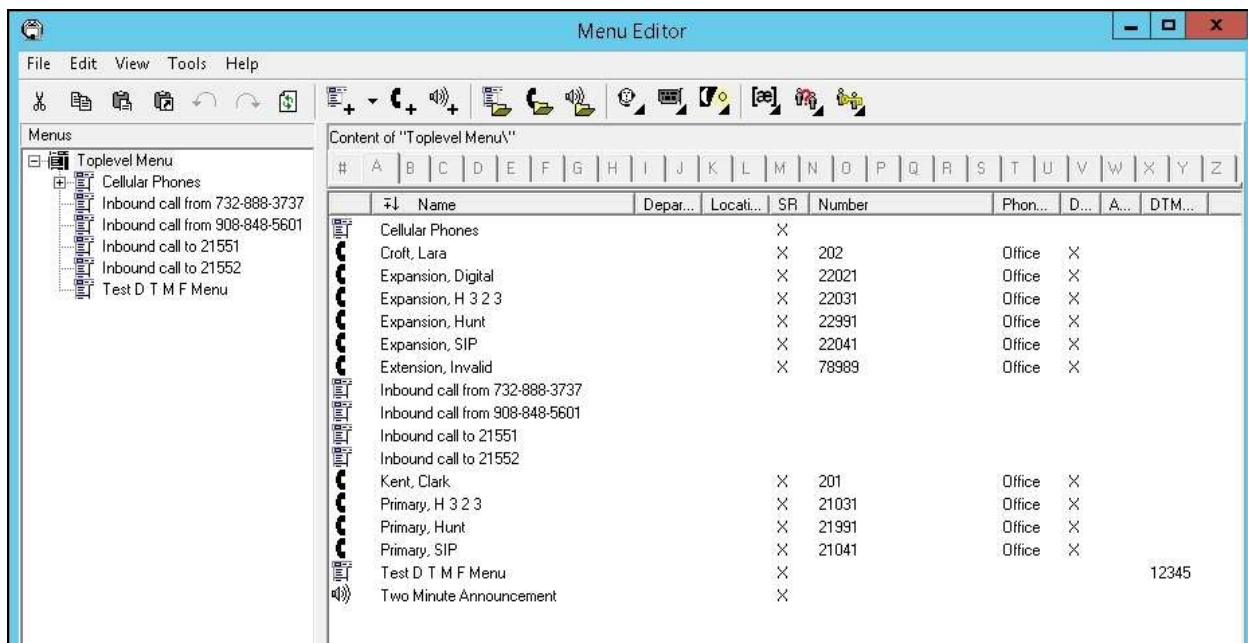


The **Menu Editor** screen below is displayed, with default directory entries in the right pane.



Follow reference [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 the supervised transfer method is required for all directory entries.

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



## 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.

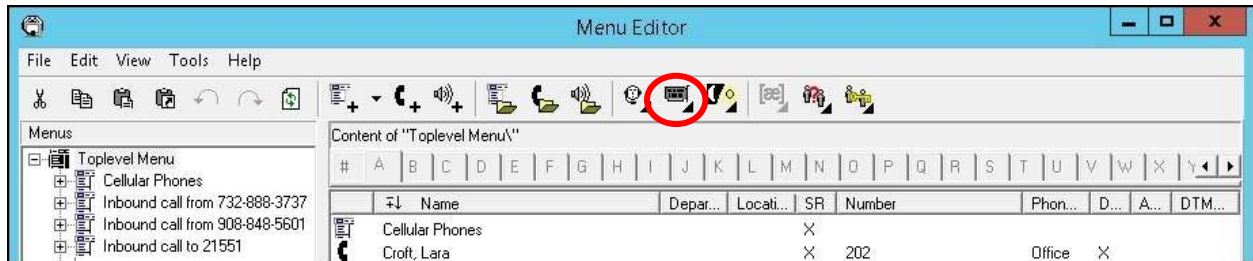
In the compliance testing, the last entry in the screen below was added for routing of calls to internal destinations consisting of 5-digit extensions. In addition, an entry needs to exist for routing of calls to the operator, in this case to “21035”. Note that all entries are required to have **Transfer type** to be set to “SUPERVISED”.

The screenshot shows a window titled "Default dialing properties" with two tabs: "Dialing rules" and "Parameters". The "Dialing rules" tab is active, displaying a table with 7 columns: Mask, Routing number, Announce number, DTMF input, Number type, Transfer type, and Comment. The table contains 17 rows, with the first 7 rows populated and the remaining 10 rows empty. The first row is highlighted in blue. Below the table are buttons for "Move up" and "Move down". At the bottom, there are input fields for "Number in directory", "Rule matched", "Number dialed", and "Number announced", along with "OK", "Cancel", and "Apply" buttons.

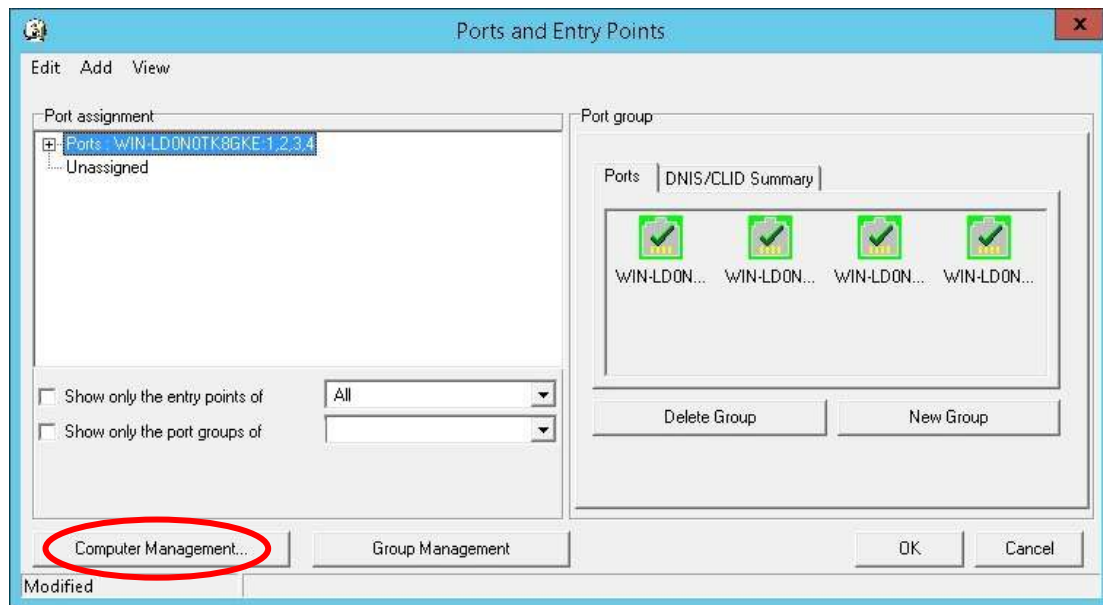
	Mask	Routing number	Announce number	DTMF input	Number type	Transfer type	Comment
1	+1-??-??-??-??-??	???	???	???	INTERNAL	SUPERVISED	Default internal
2	??-??-??	9-??-??-??	??-??-??	??-??-??	LOCAL	SUPERVISED	Default local
3	+1-??-??-??-??	9-1-??-??-??	??-??-??		LONG DISTANCE	SUPERVISED	Default long distance
4	+??-??-??-??-??	9-011-??-??-??-??-??	??-??-??-??-??		INTERNATIONAL	SUPERVISED	Default international
5	21035	21035			INTERNAL	SUPERVISED	Operator Transfer
6	+??-??-??-??-??	9-011-??-??-??-??-??	??-??-??-??-??		INTERNATIONAL	SUPERVISED	Default international
7	?????	?????	?????	?????	INTERNAL	SUPERVISED	5 digit internal
8							
9							
10							
11							
12							
13							
14							
15							
16							
17							

## 6.5. Administer Ports and Entry Points

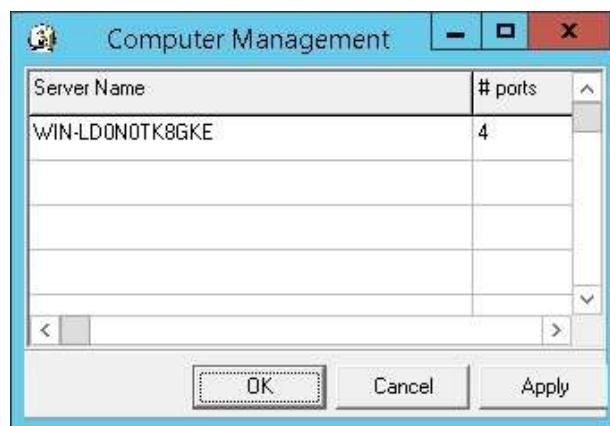
From the **Menu Editor** screen, click the **Ports and entry points** icon shown below.



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

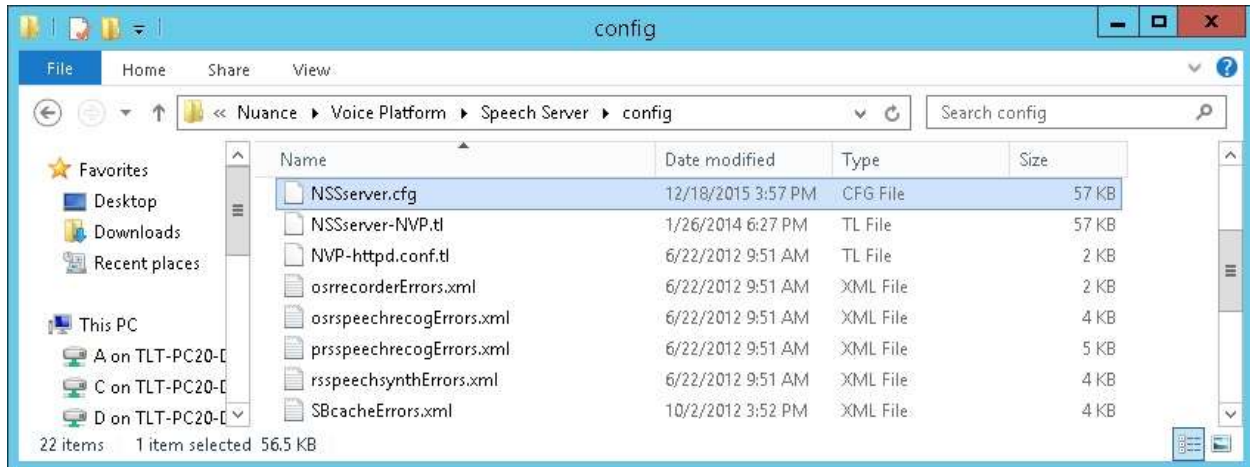


The **Computer Management** screen is displayed. Set the **# ports** value to the value allowed for by the license, in this case “4”.

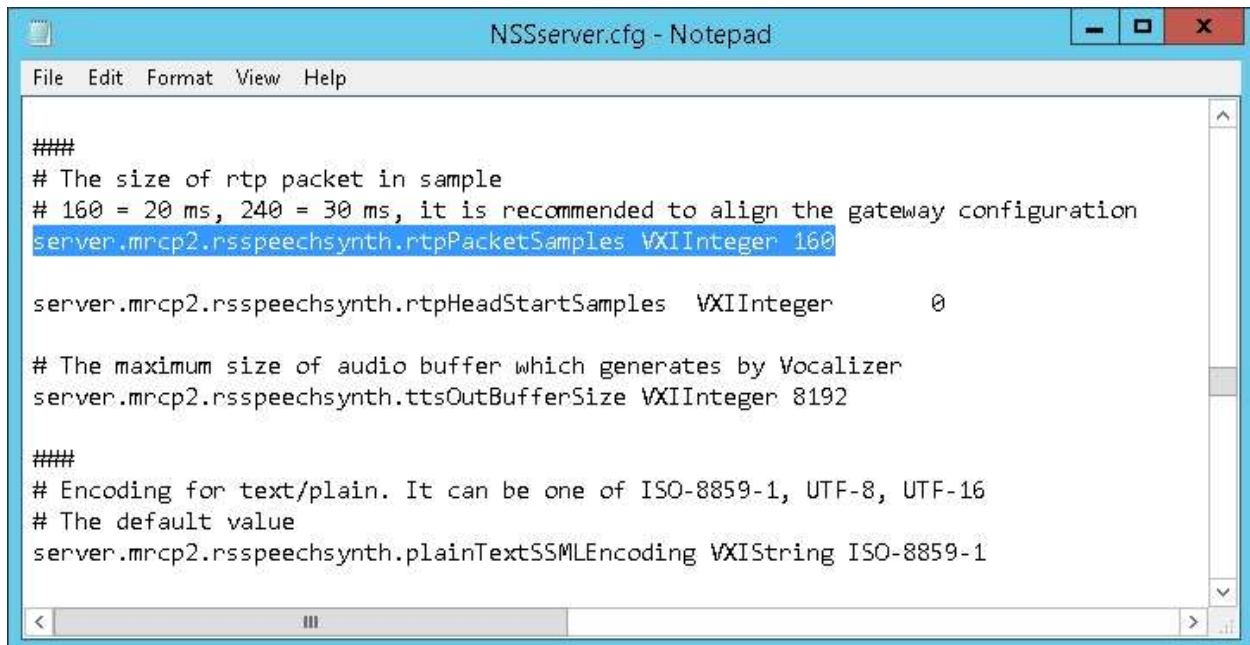


## 6.6. Administer NSServer.cfg

From the SpeechAttendant server, navigate to the **C:\Program Files\Nuance\VoicePlatform\Speech Server\config** directory to edit the **NSServer.cfg** file shown below.



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



## 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 → Advanced → 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 “9”.

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

Avaya IP Office System Status - IPO2-IPOSE (18.64.181.234) - IP Office Linux PC 9.1.4.0 build 131

### IP Office System Status

Help Snapshot LogOff Exit About

- System
- Alarms (2)
- Extensions (3)
- Trunks (3)
  - Line2
  - Line3
  - Line4
- Active Calls
- Resources
- Voicemail
- IP Networking
- Locations

#### SIP Trunk Summary

Line Service State: In Service  
Peer Domain Name: sip://10.64.101.202  
Resolved Address: 10.64.101.202  
Line Number: 9  
Number of Administered Channels: 10  
Number of Channels in Use: 0  
Administered Compression: G711 Mu  
Enable Faststart: Off  
Silence Suppression: Off  
Media Stream: RTP  
Layer 4 Protocol: UDP  
SIP Trunk Channel Licenses: Unlimited  
SIP Trunk Channel Licenses in Use: 0  
SIP Device Features: REFER (Incoming and Outgoing)

Channel Number	URI	Call G... Ref	Current State	Time in State	Remote Media A...	Co...	Conne...	Caller ID or Dial...	Other Party on...	Direction of Call	Round Trip D...	Receive Jitter	Receive Packet...	Transmit Jitter	Transmit Packet...
1			Idle	00:03:48											
2			Idle	00:03:48											
3			Idle	00:03:48											
4			Idle	00:03:48											
5			Idle	00:03:48											
6			Idle	00:03:48											
7			Idle	00:03:48											
8			Idle	00:03:48											
9			Idle	00:03:48											
10			Idle	00:03:48											

Trace Trace All Pause Ping Call Details Graceful Shutdown Force Out of Service Print... Save As...

9:32:12 AM Online



## 7.2. Verify Nuance SpeechAttendant

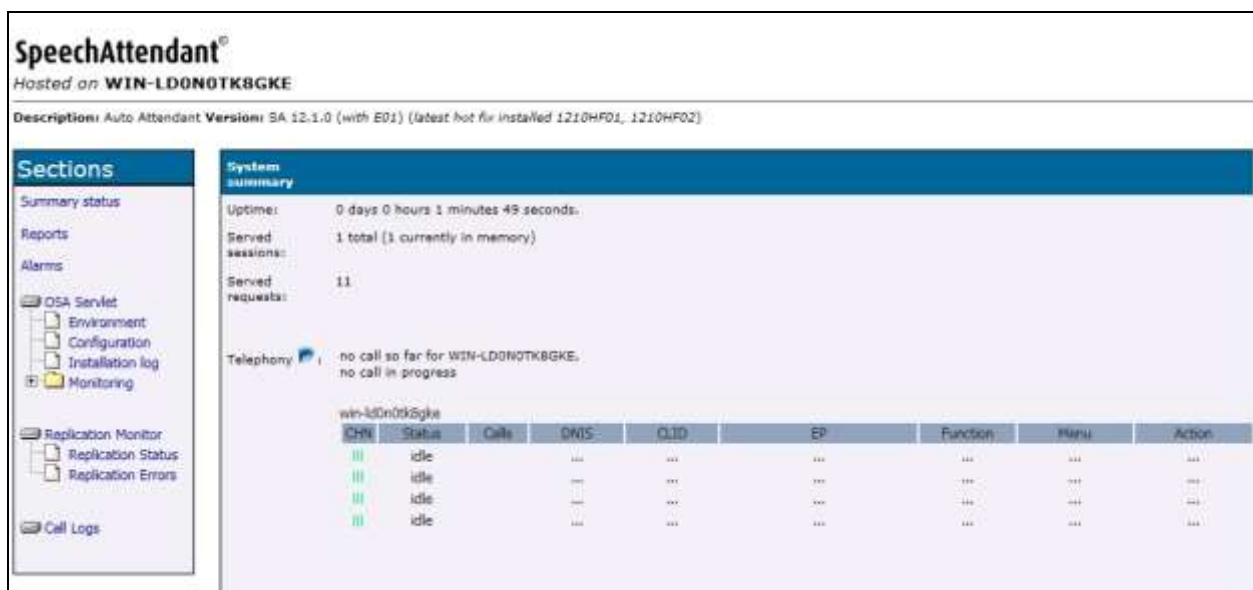
From a PC, launch an Internet browser window and access the SpeechAttendant web-based status interface by using the URL “http://<ip-address>/OpenSpeech/Attendant/servlet/aa?action=status”, where “ip-address” is the IP address of the SpeechAttendant server.

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



The image shows the Speech Attendant Login screen. It has a light blue background with a stylized wave pattern. The title "Speech Attendant" is centered at the top, with "Login" below it. A central box with a green border contains the login form. The box has a blue header that says "Enter your user name and password." Below this are two input fields: "User name:" and "Password:". At the bottom of the box are a "Login" button and a link that says "Forgot your password ?".

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



The image shows the SpeechAttendant System Summary screen. The top header says "SpeechAttendant®" and "Hosted on WIN-LDONOTKBGKE". Below this is a description: "Description: Auto Attendant Version: SA 12.1.0 (with E01) (latest hot fix installed 1210HF01, 1210HF02)". On the left is a "Sections" menu with options: Summary status, Reports, Alarms, OSA Servlet (Environment, Configuration, Installation log, Monitoring), Replication Monitor (Replication Status, Replication Errors), and Call Logs. The main area is titled "System summary" and contains the following information:

- Uptime: 0 days 0 hours 1 minutes 49 seconds
- Served sessions: 1 total (1 currently in memory)
- Served requests: 11
- Telephony: no call so far for WIN-LDONOTKBGKE, no call in progress

Below this is a table for "win-lدونتکبگک" showing channel status:

CHN	Status	Calls	DNIS	CLD	EP	Function	Menu	Action
III	idle		xxx	xxx	xxx	xxx	xxx	xxx
III	idle		xxx	xxx	xxx	xxx	xxx	xxx
III	idle		xxx	xxx	xxx	xxx	xxx	xxx
III	idle		xxx	xxx	xxx	xxx	xxx	xxx



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 601) (latest hot fix installed 1210HP01, 1210HP02)

Sections

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

System summary

Uptime: 0 days 0 hours 1 minutes 49 seconds.  
 Served sessions: 1 total (1 currently in memory)  
 Served requests: 11  
 Telephony: 1 calls so far for WIN-LD0N0TK8GKE.  
           1 calls in progress (concurrent peak 1, Mon Dec 28 08:16:12 PST 2015)

win-LD0N0TK8GKE

CHN	Status	Calls	DNIS	CLID	EP	Function	Menu	Action
	busy	1	21550	9088485601	Call from 908-848-5601	AA	Inbound call from 908-848-5601	In progress
	idle		---	---	---	---	---	---
	idle		---	---	---	---	---	---
	idle		---	---	---	---	---	---

## 8. Conclusion

These Application Notes describe the configuration steps required for Nuance SpeechAttendant 12 to successfully interoperate with Avaya IP Office Server Edition 9.1 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™ Platform with Manager*, Release 9.1.0, Issue 10.03, February 2015, available at <http://support.avaya.com>.
2. *Nuance SpeechAttendant Nuance OpenSpeech Attendant Administration Guide*, April 2014, available at <https://network.nuance.com/portal/server.pt>.

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