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:** SpeechAttendant.Support@nuance.com
- **Web :** 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 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).

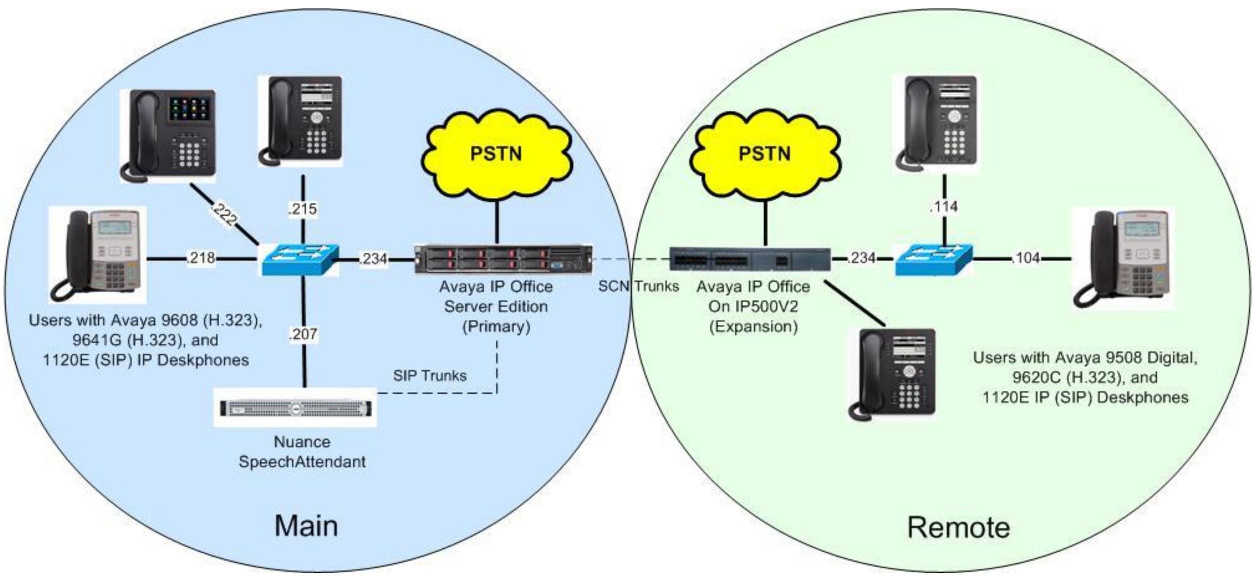


Figure 1: Compliance Testing Configuration

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:

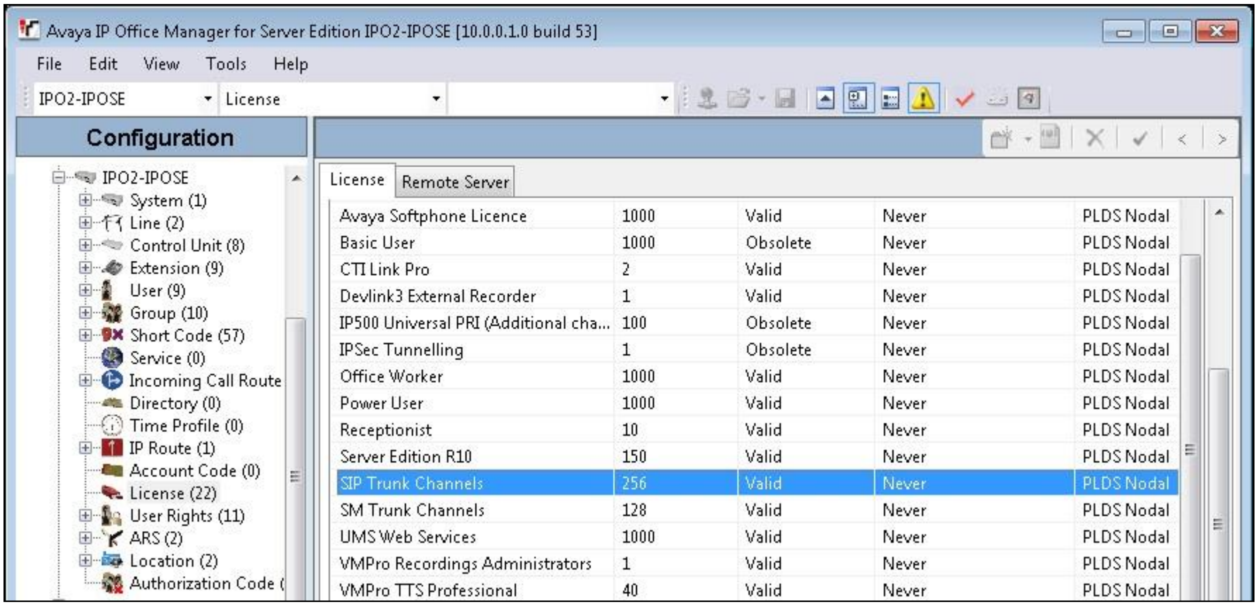
- Verify license
- Administer system
- Administer line
- Administer incoming call route
- Administer short code

5.1. Verify License

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

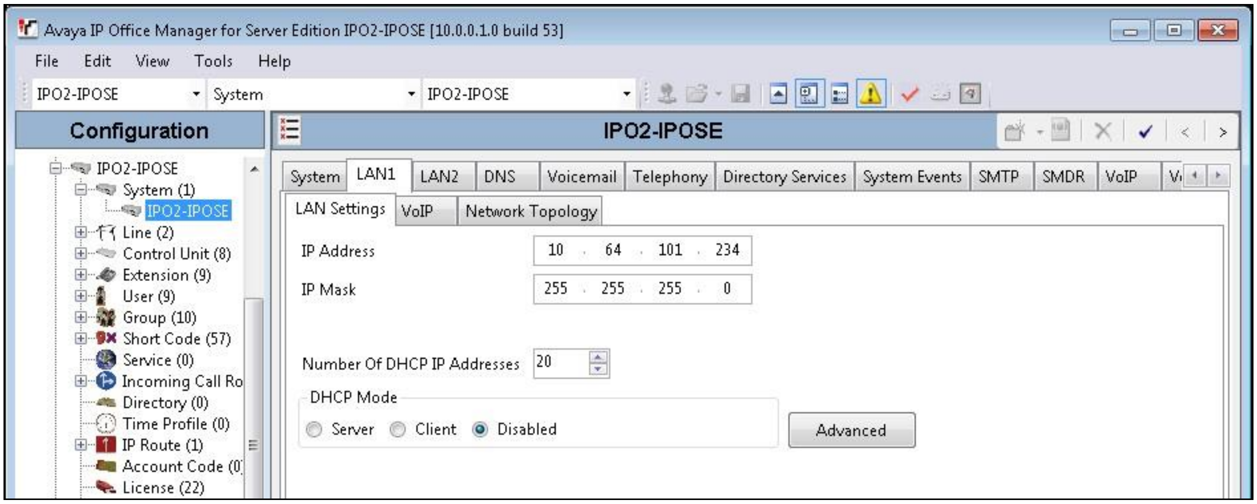
From the configuration tree in the left pane, select **License** under the IP Office system that will be used for SIP trunks connection with SpeechAttendant, in this case "IPO2-IPOSE", and a list of licenses is displayed in the right pane. Verify that there is a license for **SIP Trunk Channels** and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



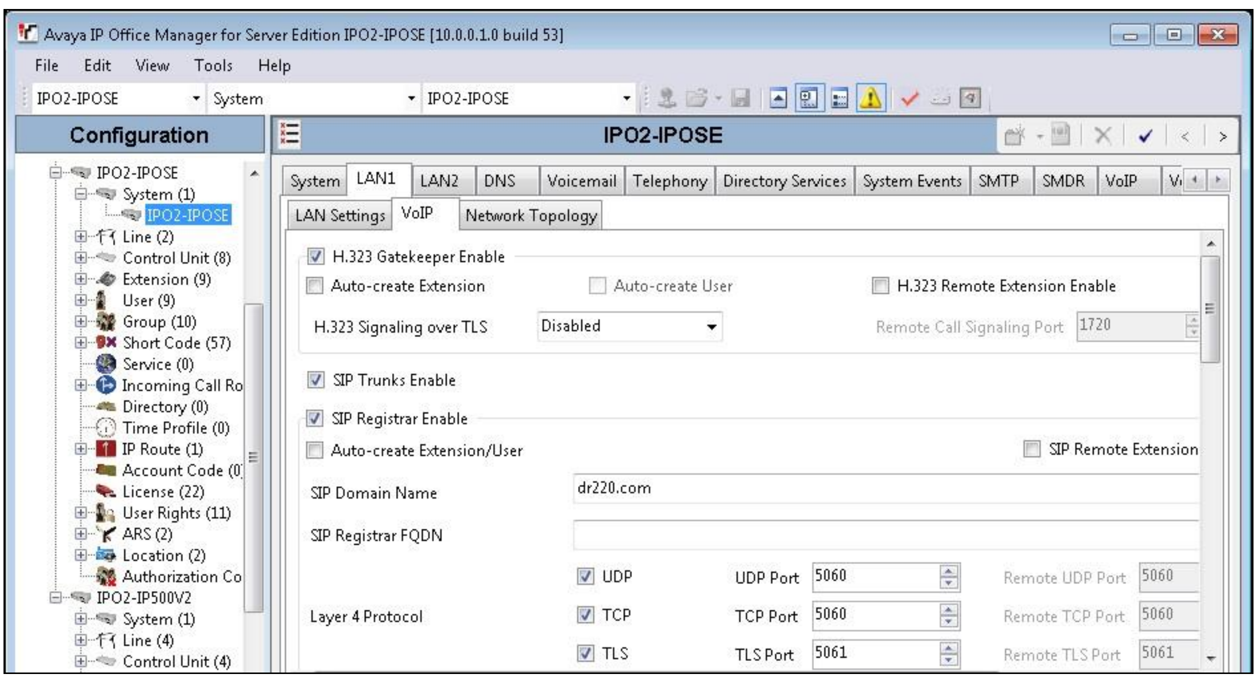
5.2. Administer System

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.

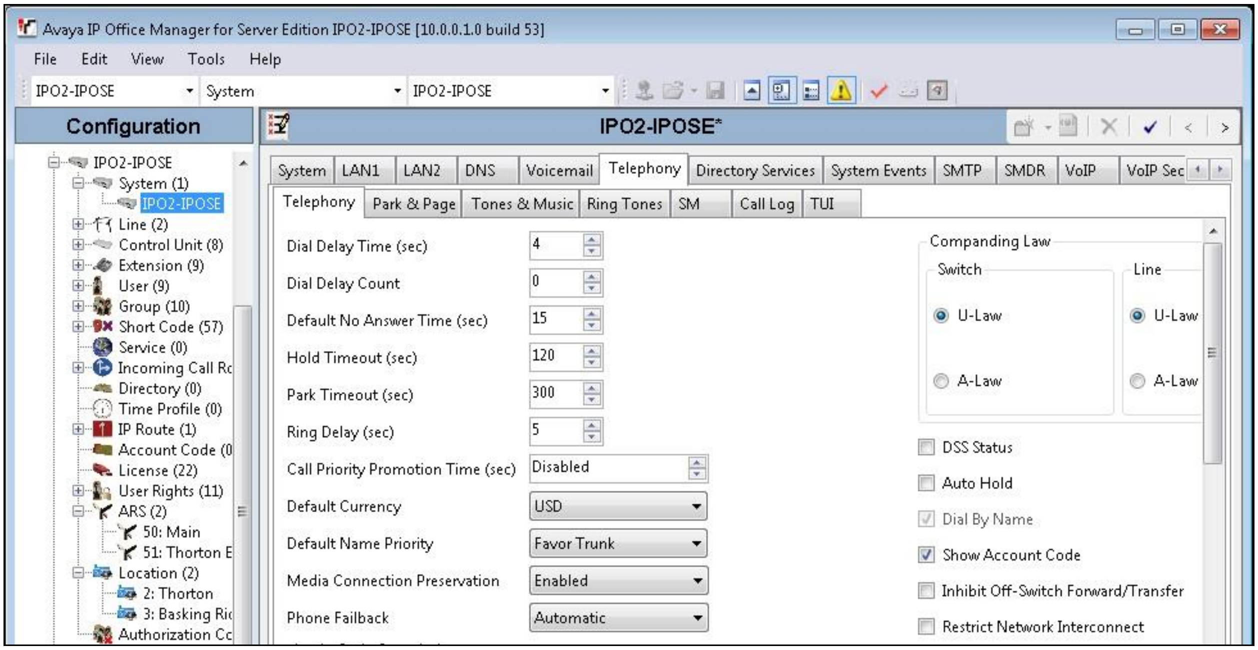
Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later to configure SpeechAttendant. Note that IP Office can support SIP trunks on the LAN1 and/or LAN2 interfaces, and the compliance testing used the LAN1 interface.



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



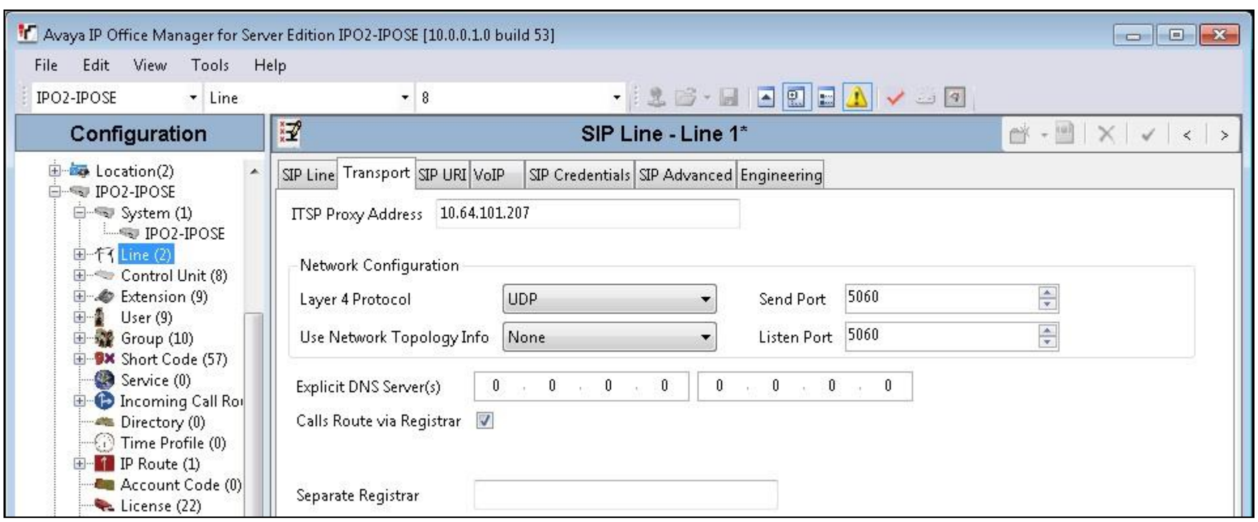
Select the **Telephony** tab, followed by the **Telephony** sub-tab in the right pane. Uncheck **Inhibit Off-Switch Forward/Transfer**, if transfer from SpeechAttendant to PSTN destinations is desired. In the compliance testing, this parameter was disabled.



5.3. Administer Line

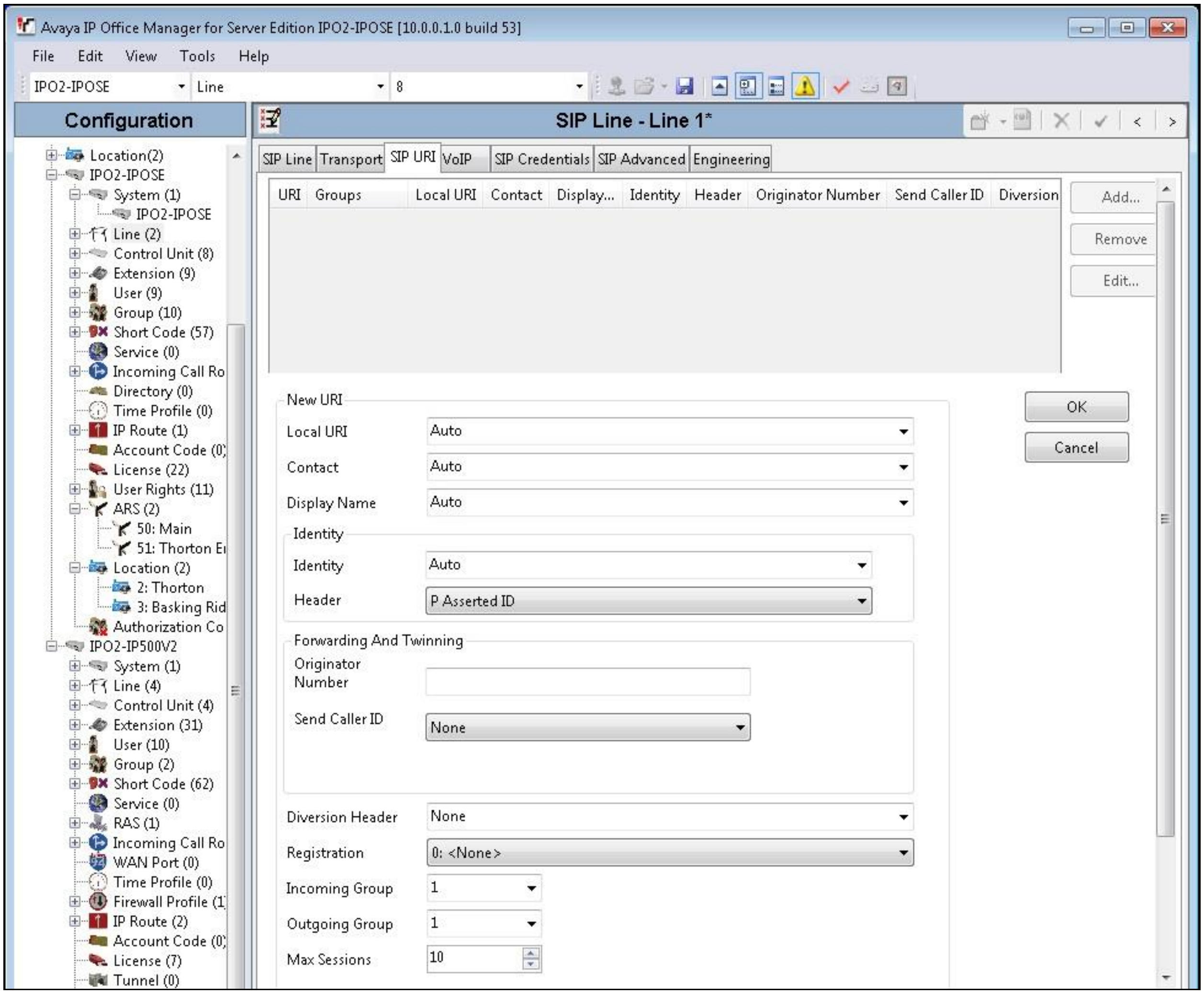
From the configuration tree in the left pane, right-click on **Line** under the IP Office system used for SIP trunks connection with SpeechAttendant, and select **New → SIP Line** from the pop-up list to add a new SIP line.

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 and TLS, and the compliance testing used the UDP protocol.



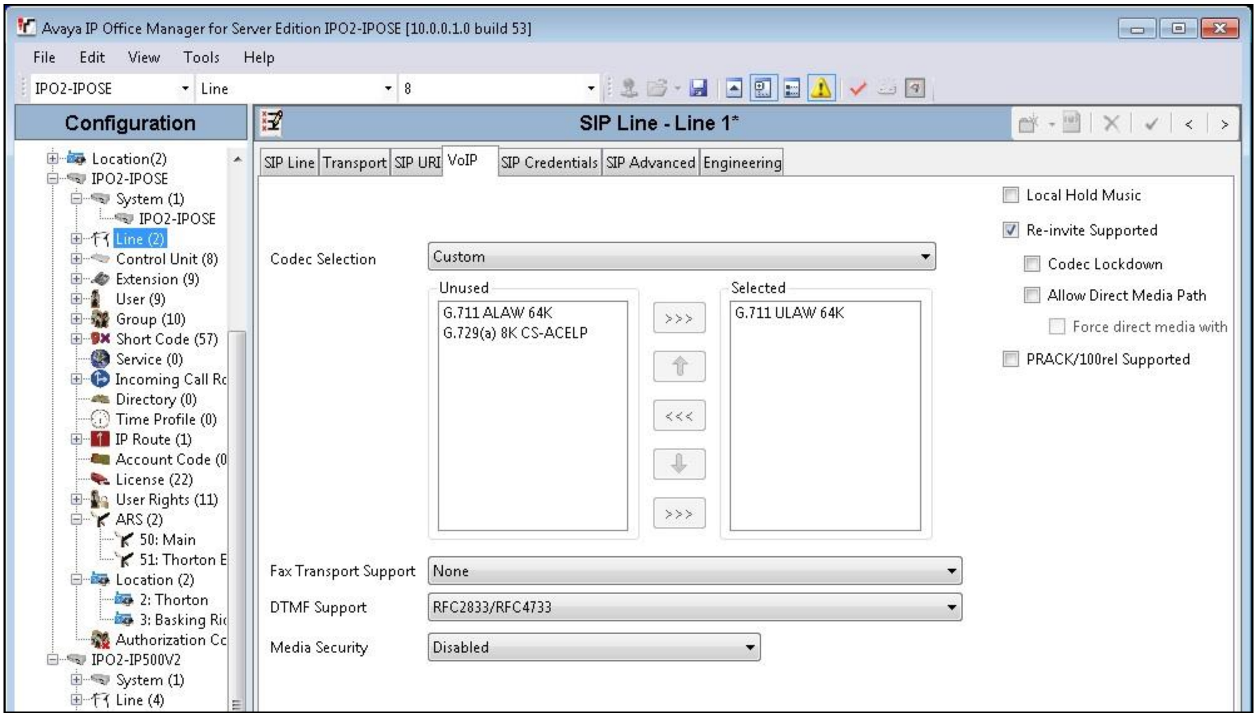
Select the **SIP URI** tab, and click **Add** to display the **New URI** sub-section. Enter the following values for the specified fields, and retain the default values for the remaining fields.

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



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".

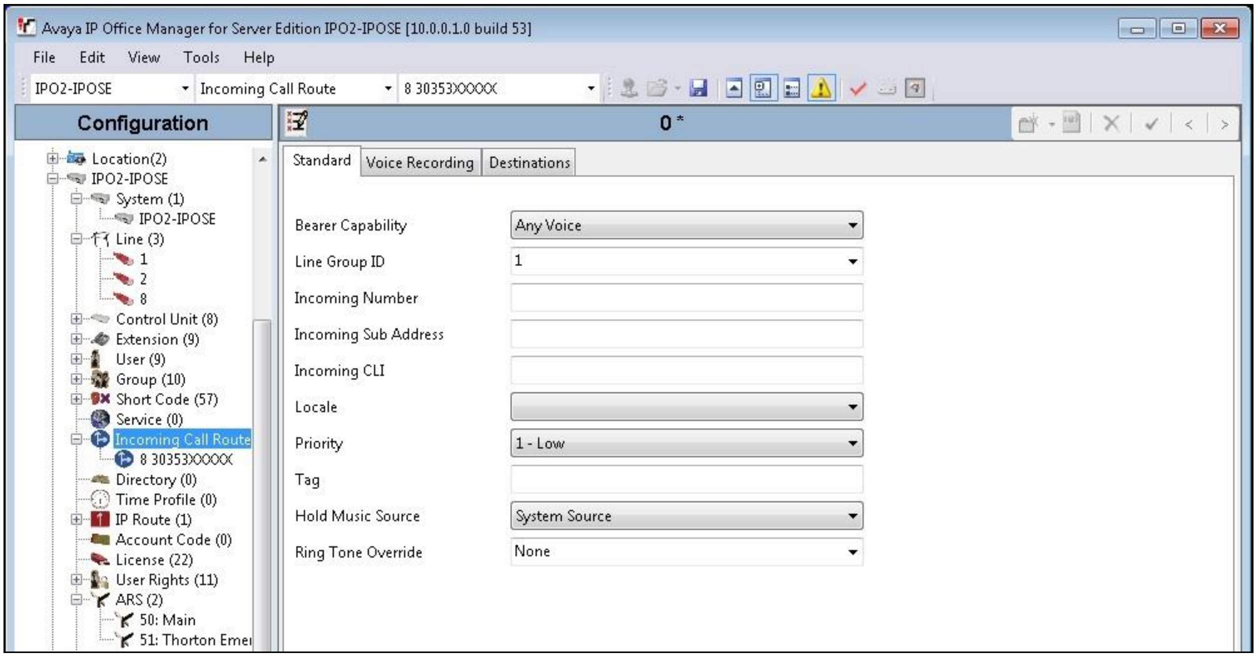
Check **Re-invite Supported**. Retain the default values in the remaining fields.



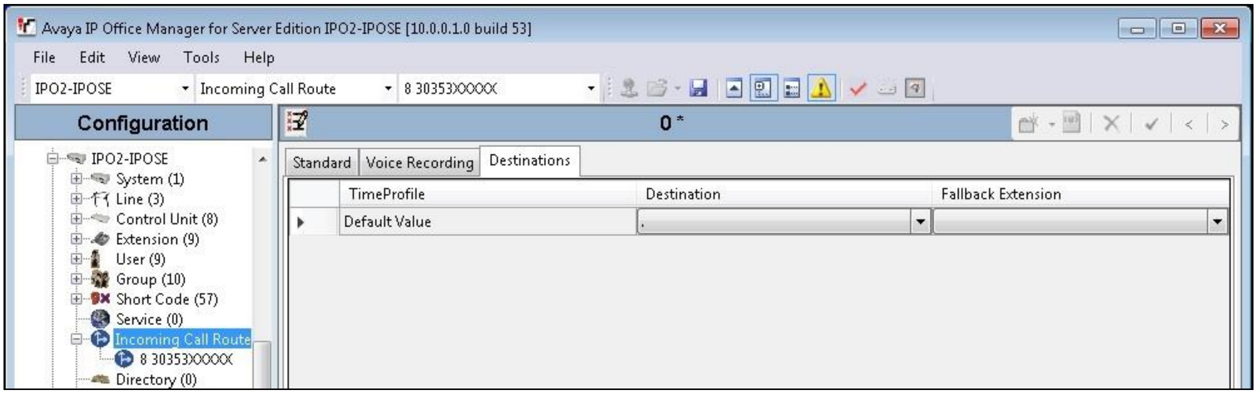
5.4. Administer Incoming Call Route

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 "1". Retain the default value in the remaining fields.



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

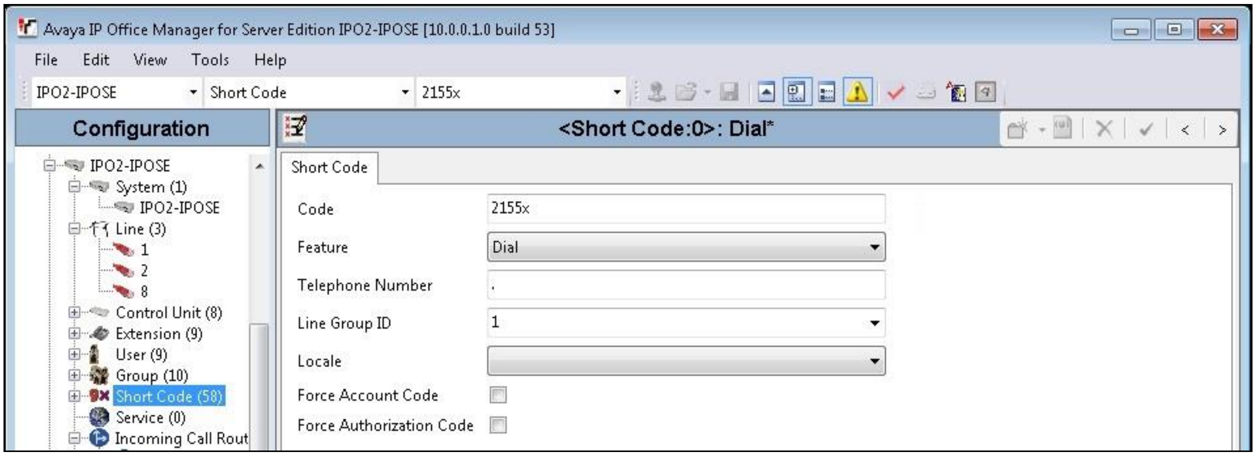


5.5. Administer Short Code

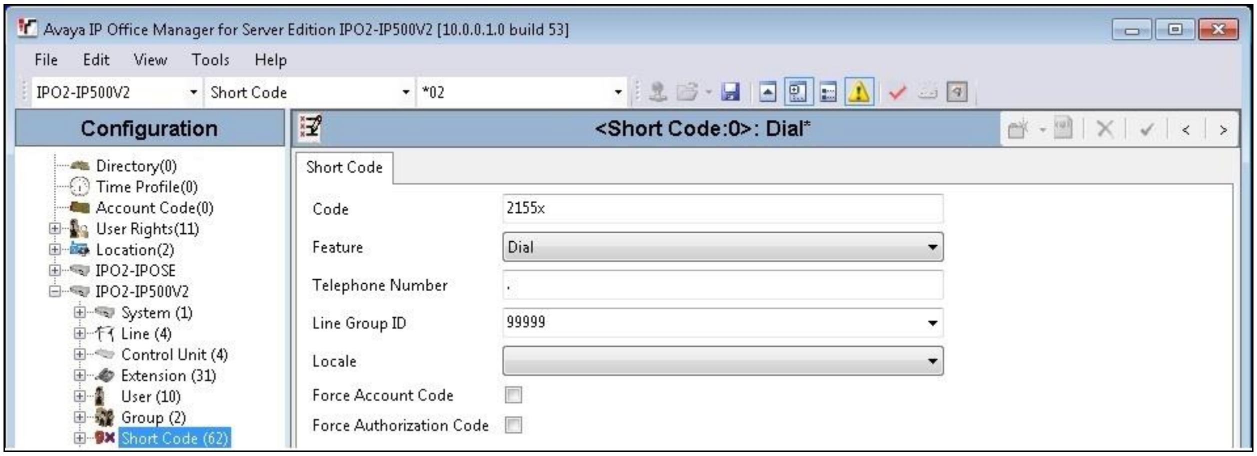
From the configuration tree in the left pane, right-click on **Short Code** under the IP Office used for SIP trunks connection with SpeechAttendant, and select **New** from the pop-up list to add a new short code for outgoing calls to SpeechAttendant. In the compliance testing, all calls to 2155x are 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 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 "99999" 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 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

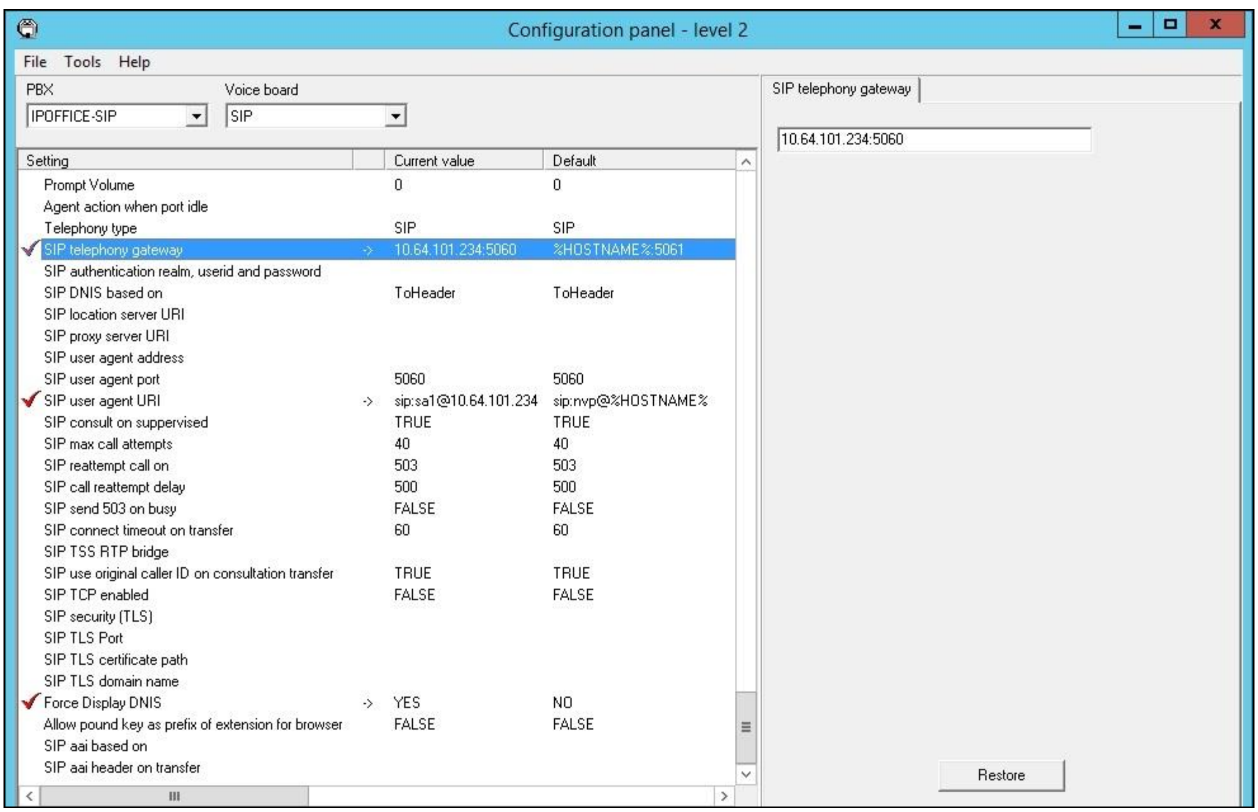
The **Admin Tools** screen is displayed. Select **Configuration Panel**.

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

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.

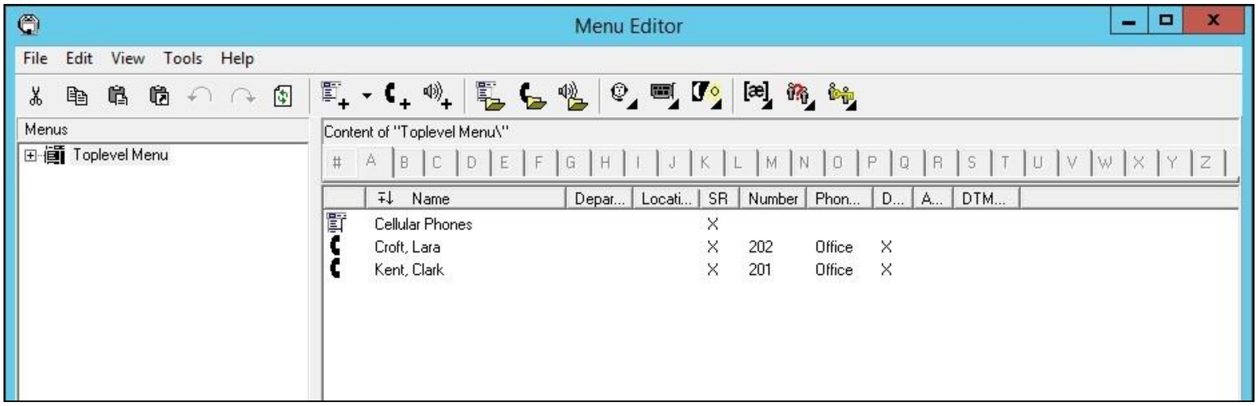


6.3. Administer Phone Directory and Menu Editor

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

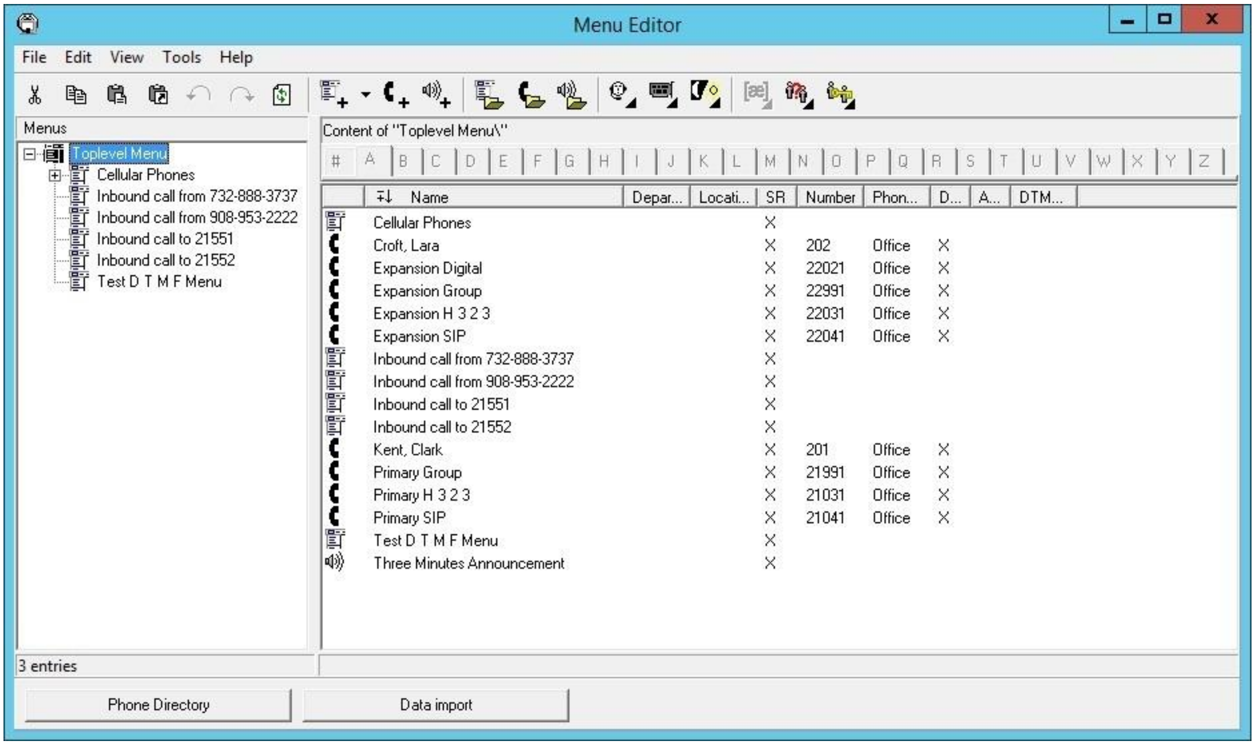
The **Phone Directory and Menu Editor** screen below is displayed next. Log in using the appropriate credentials.

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 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 → Dialing Properties** from the top menu.



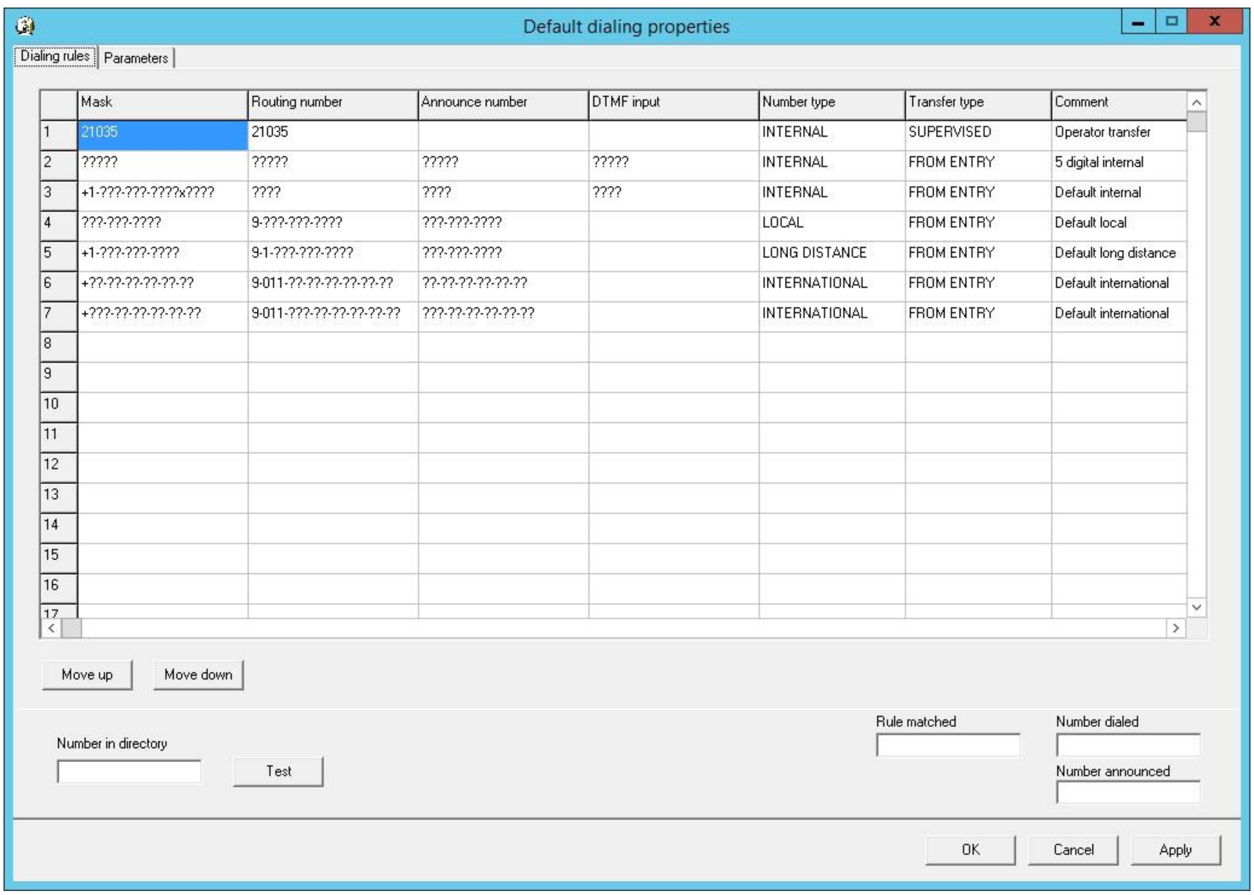
6.4. Administer Dialing Properties

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.

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.

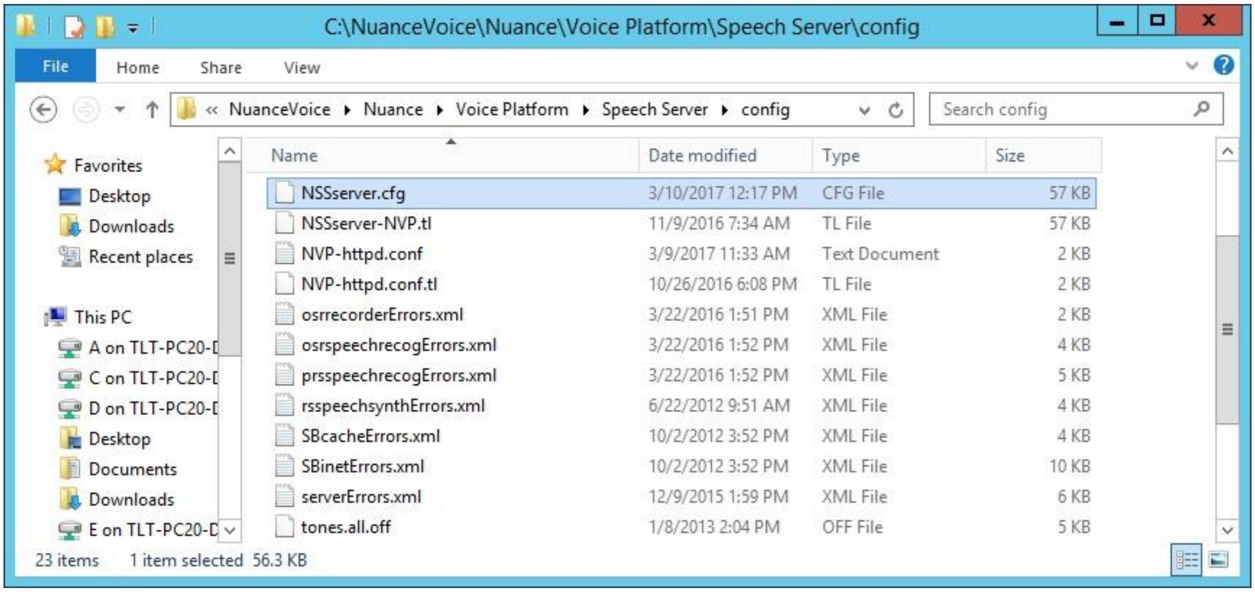
The second entry in the screen below was added for routing of calls to internal destinations consisting of 5-digit extensions. In the compliance testing, all directory entries associated with internal destinations were configured in **Section 6.3** with supervised as transfer type, therefore the **Transfer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

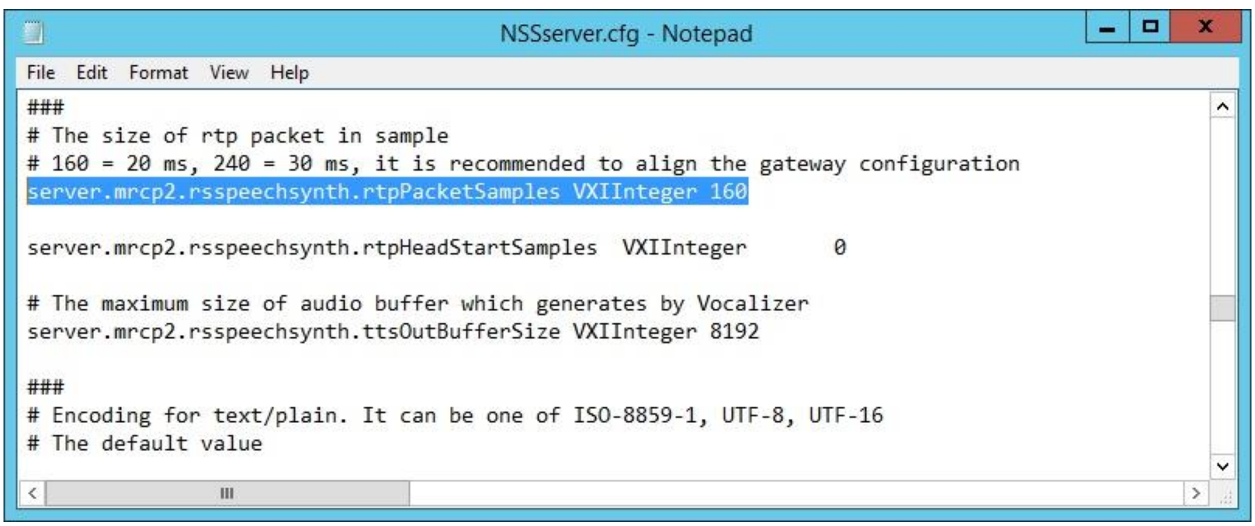


6.5. Administer NSServer.cfg

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.



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.

7. Verification Steps

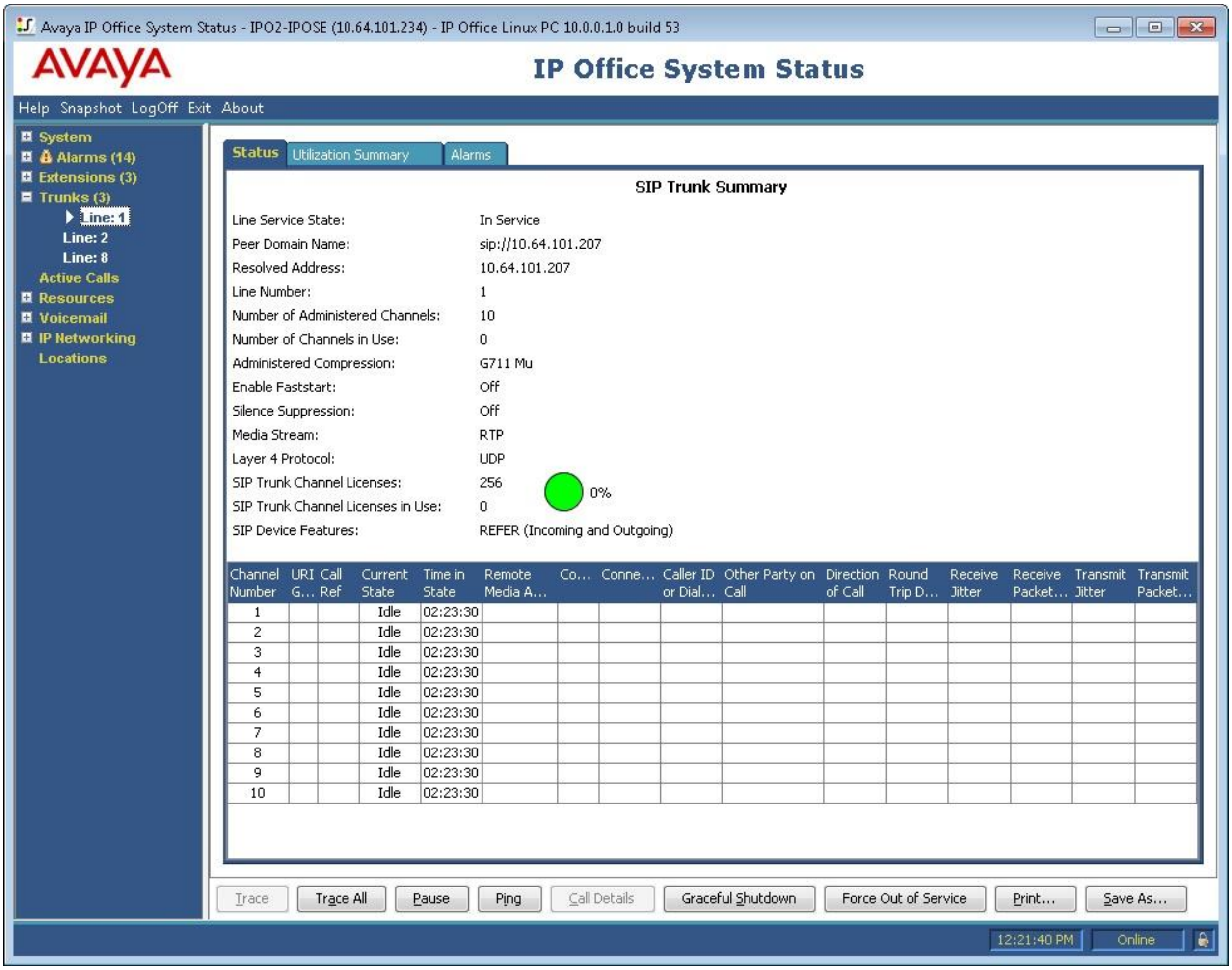
This section provides the tests that can be performed to verify proper configuration of IP Office and SpeechAttendant.

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 "1".

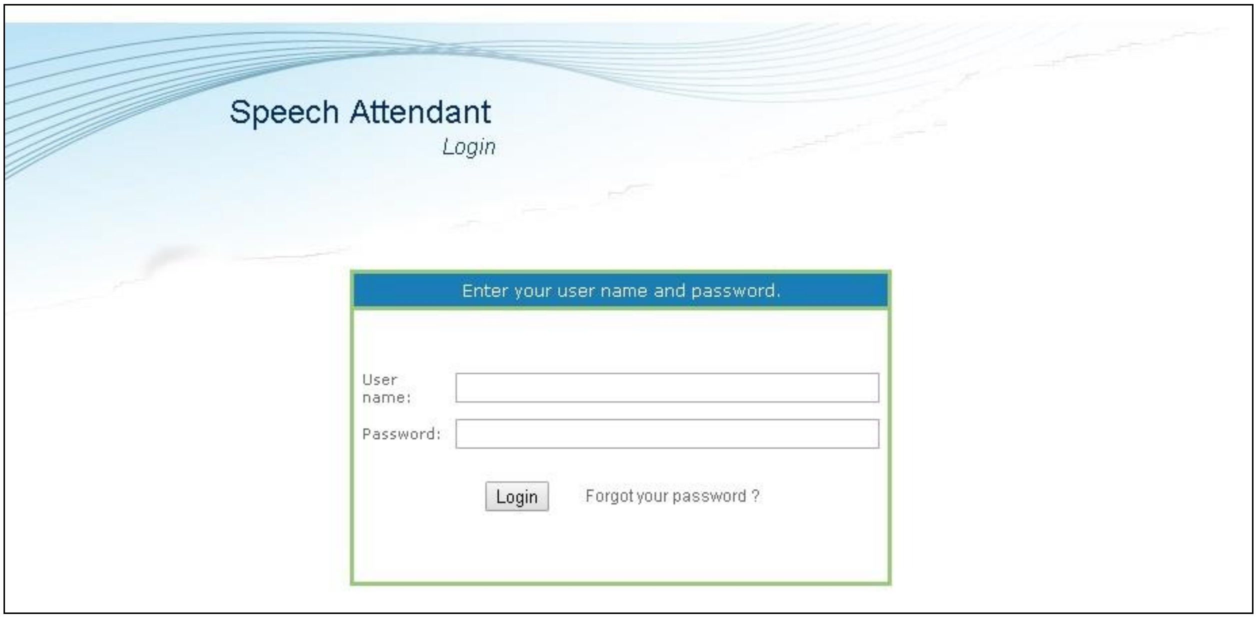
Verify that the **SIP Trunk Summary** screen shows all channels with **Current State** of "Idle", as shown below.



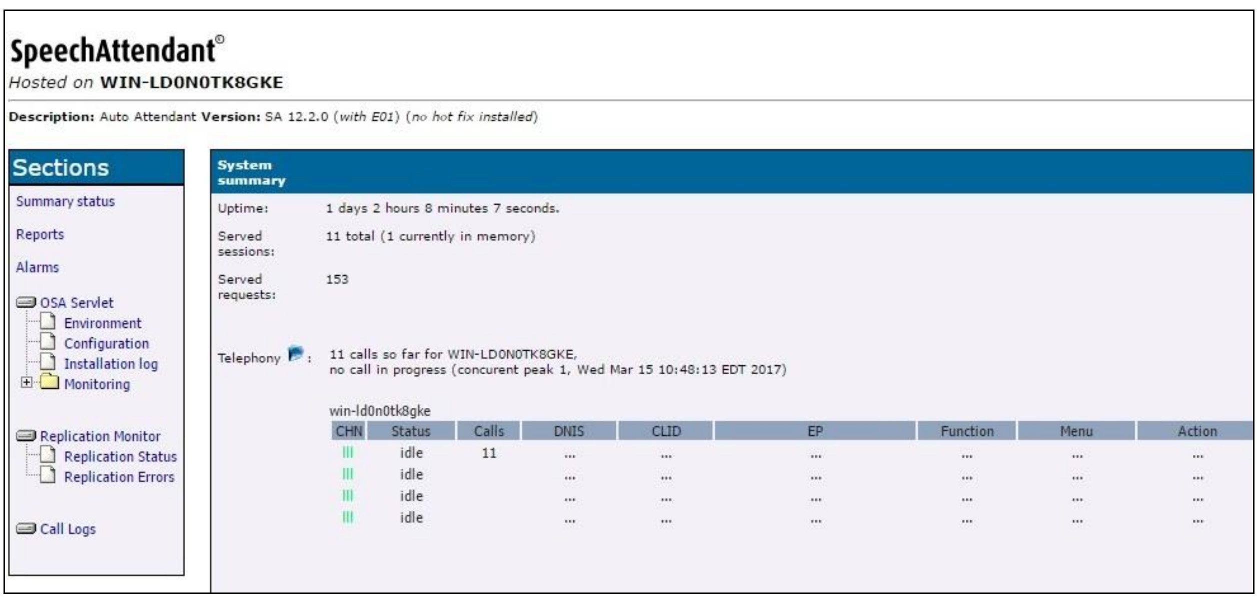
7.2. Verify Nuance SpeechAttendant

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.

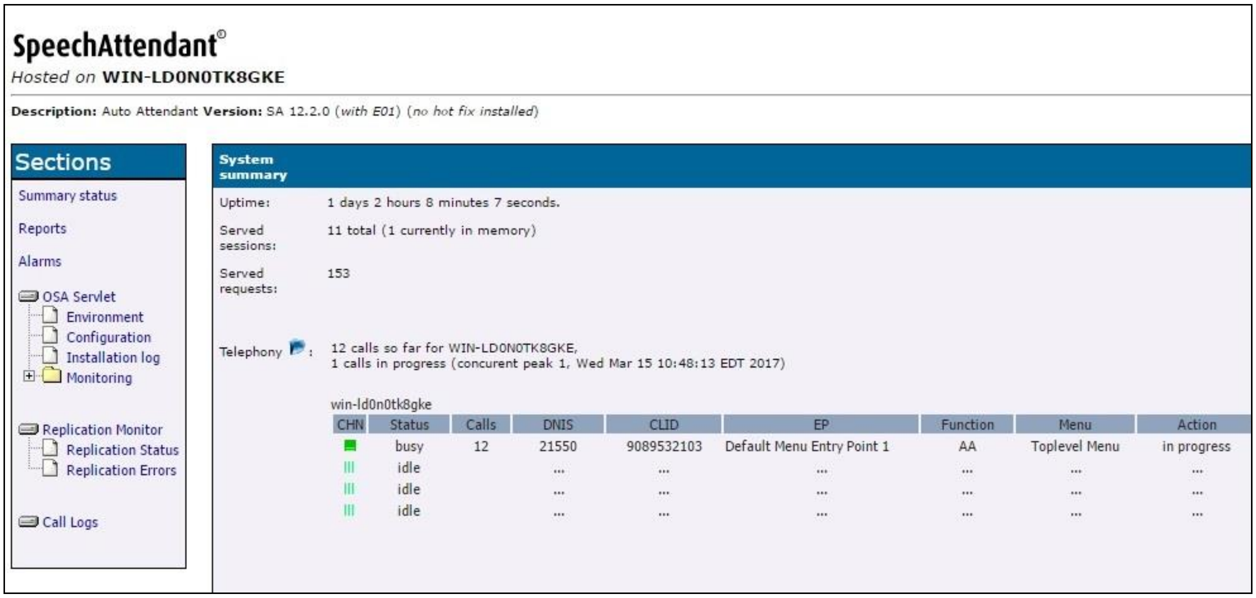
The **Speech Attendant Login** screen is displayed. Log in using the appropriate credentials.



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



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™ Platform with Manager*, Release 10.0, September 2016, 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.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.