**Avaya Solution & Interoperability Test Lab**

# Application Notes for Configuring Nuance Speech Attendant with Avaya IP Office 8.1 – Issue 1.0

## Abstract

These Application Notes describe the configuration steps required to integrate the Nuance Speech Attendant with Avaya IP Office using a SIP integration. Nuance Speech Attendant allows callers to speak the name of a person, department, service, or location and be automatically transferred to the requested party without waiting to speak to an operator. In addition, the caller may dial an extension number and be transferred to the requested party.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

# 1. Introduction

These Application Notes describe the configuration steps required to integrate the Nuance Speech Attendant (SA) with Avaya IP Office using a SIP integration. Nuance Speech Attendant allows callers to speak the name of a person, department, service, or location and be automatically transferred to the requested party without waiting to speak to an operator. In addition, the caller may dial an extension number to be transferred to the requested party.

# 2. General Test Approach and Test Results

The interoperability compliance testing included feature and serviceability test cases.

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

Feature testing focused on Nuance SA successfully recognizing spoken names and extensions entered via DTMF, and then transferring the call to the correct destination. Blind and supervised transfers were verified. Other features covered included: DNIS and CLID handling, barge-in / no barge-in, adding new transfer entries, recording caller utterances, and accessing Maintenance Mode and Personal Administration Mode to record name and change PIN.

Serviceability testing focused on verifying the ability of the Nuance SA to recover from adverse conditions, such as server restarts, power failures, and disconnecting cables to the IP network.

## 2.2. Test Results

All test cases passed with the following observation:

- Nuance SA does not support shuffling.
- Avaya IP Office does not support a SIP REFER for a blind transfer. Only consultation/supervised transfers are supported with this solution.

## 2.3. Support

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

- **Web:** www.network.nuance.com
- **Email:** SpeechAttendant.Support@nuance.com
- **Phone:** +1 (866) 434-2564 or +1 (514) 390-3922

# 3. Reference Configuration

**Figure 1** illustrates the setup used to verify the Nuance Speech Attendant (SA) solution with Avaya IP Office. Nuance SA is deployed on a dedicated server running Windows 2008 R2 Enterprise server. Nuance SA interfaces to Avaya IP Office via SIP. To access the Nuance SA application, a call is simply routed from IP Office to the SA server over a SIP trunk. Multiple SIP ports were configured on the Nuance SA server.

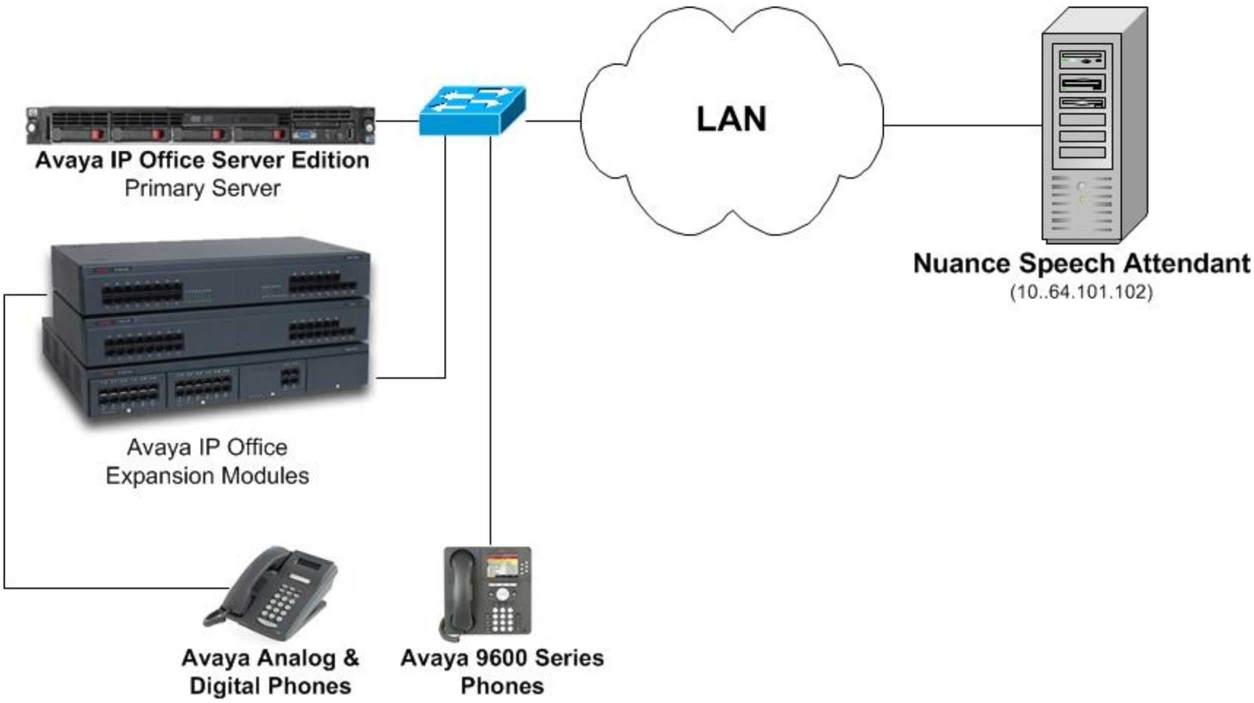



**Figure 1: Avaya IP Office with Nuance Speech Attendant**

## 4. Equipment and Software Validated

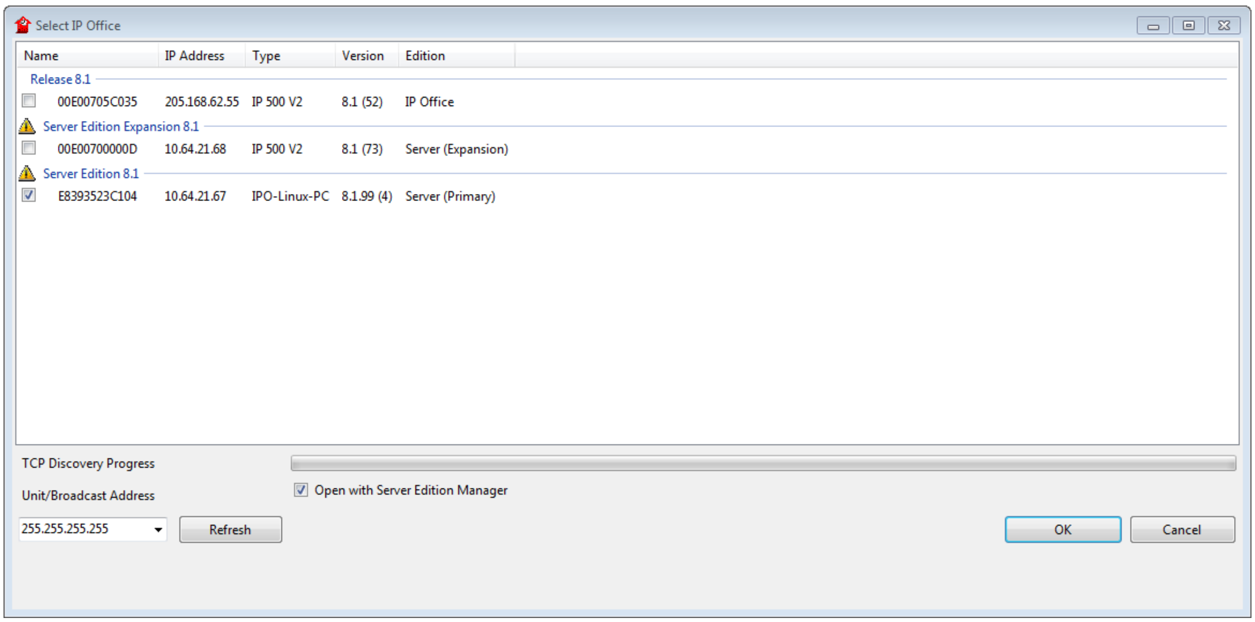
The following equipment and software were used for the sample configuration:

| Equipment/Software | Release/Version |
|---|---|
| Avaya IP Office Server Edition | 8.1 |
| Avaya one-X® 9600 Series IP Telephones<br>• 96x0 (H.323)<br>• 96x1 (H.323) | Avaya one-X® Deskphone Edition 3.1.5<br>Avaya one-X® Deskphone Edition 6.2.2 |
| Avaya Analog and Digital Phones | - |
| Nuance Speech Attendant | 11.1 Hotfix 50 |

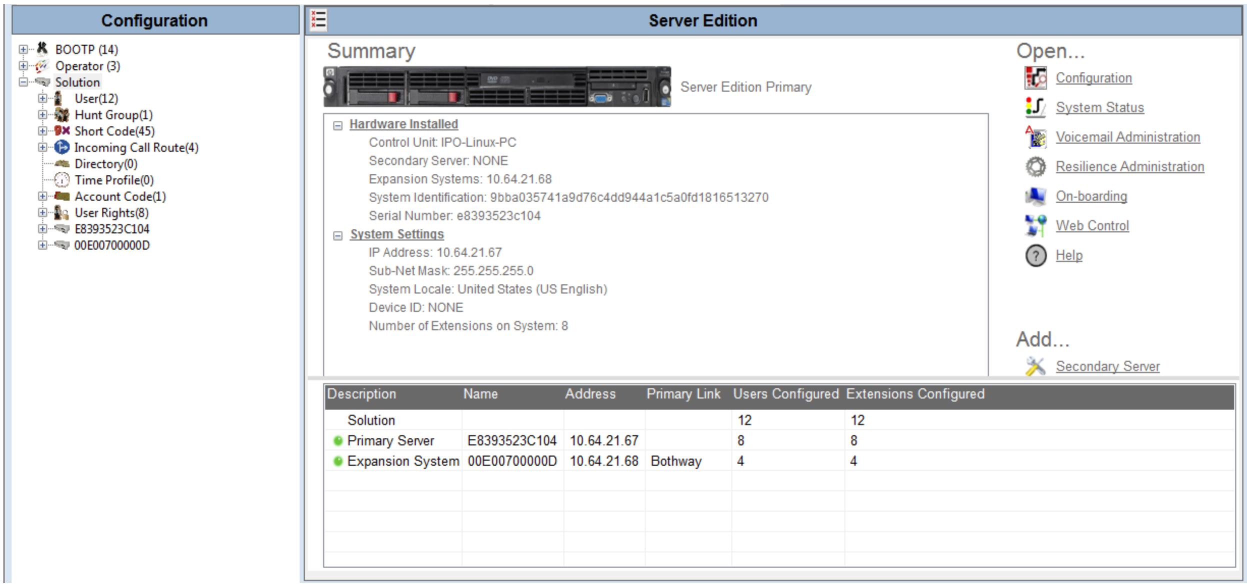
# 5. Configure Avaya IP Office Server Edition

This section describes the Avaya IP Office Server Edition configuration necessary to support connectivity to Nuance Speech Attendant. It is assumed that the initial installation and provisioning of the Server Edition Primary Server and Expansion System has been previously completed and therefore is not covered in these Application Notes. For information on these installation tasks refer to reference [**1**] in the **Additional References** section.

The solution is configured through the Avaya IP Office Server Edition Manager PC application. From the PC running the IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office Server Edition system, making sure the box for **Open with Server Edition Manager** is checked. Log in using appropriate credentials.



The Solution View screen will appear, similar to the one shown below. This screen includes the system inventory of the servers and links for administration and configuration tasks.




In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center and the Details pane on the right side. These panes will be referenced throughout the rest of this section.

Note that the Navigation pane includes solution settings, under the Solution menu, which apply to all the systems in the Server Edition solution, and individual system settings, each grouped under the Primary Server and the Expansion System menus.

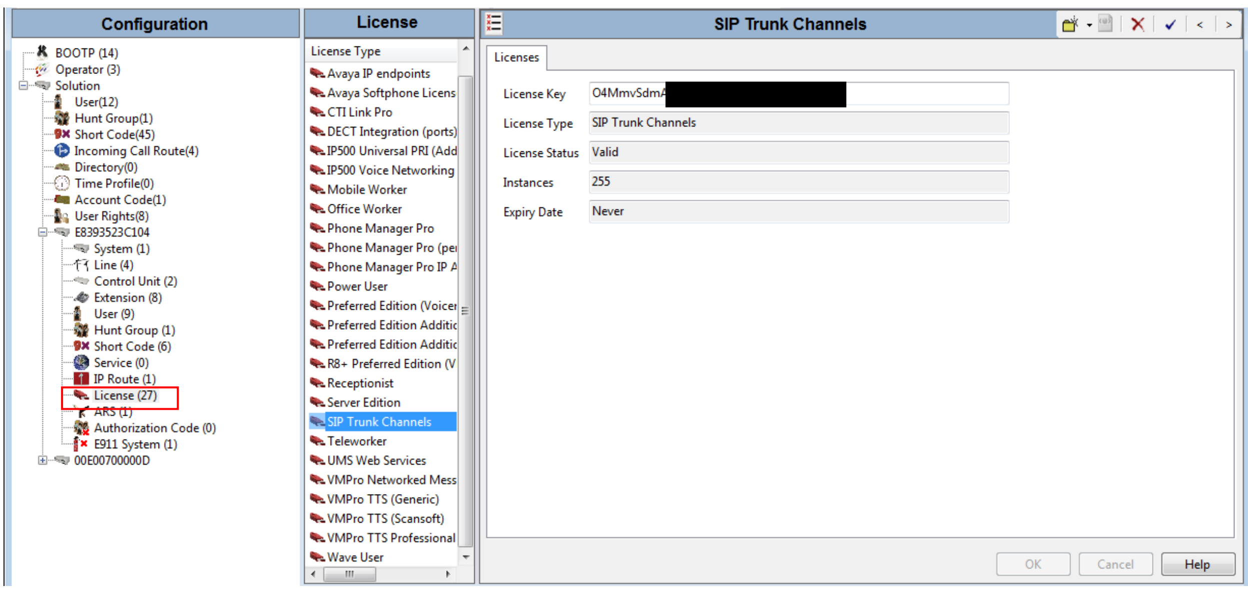
Standard feature configurations that are not directly related to the interfacing with the service provider are assumed to be already in place, and they are not part of these Application Notes.

## 5.1. Licensing

The configuration and features described in these Application Notes require the IP Office Server Edition system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

Licenses for an IP Office Server Edition solution are based on a combination of centralized licensing done through the IP Office Server Edition Primary Server, and server specific licenses that are entered into the configuration of the system requiring the feature. SIP Trunk Channels are centralized licenses, and they are entered into the configuration of the Primary Server. Note that when centralized licenses are used to enable features on other systems, such as SIP trunk channels, the Primary Server allocates those licenses to the other systems only after it has met its own license needs.

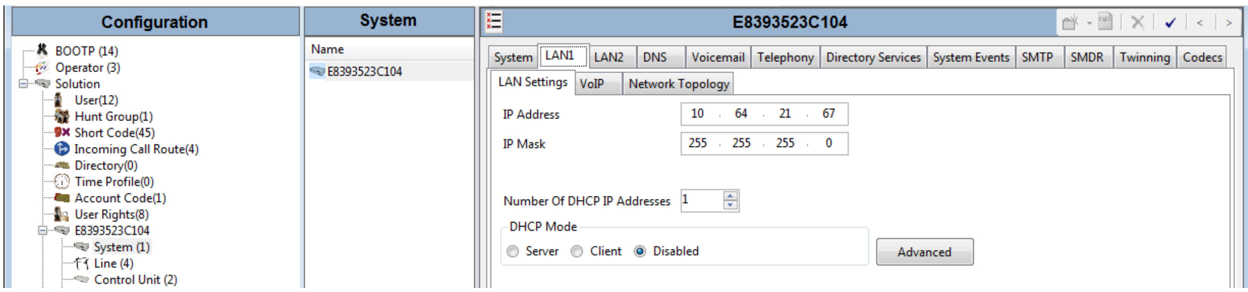
To verify that there is a SIP Trunk Channels license with sufficient capacity, select **License** under the Primary server on the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm that there is a valid license with sufficient "Instances" (trunk channels) in the Details pane. Note that the actual License Key in the screen below was edited for security purposes.



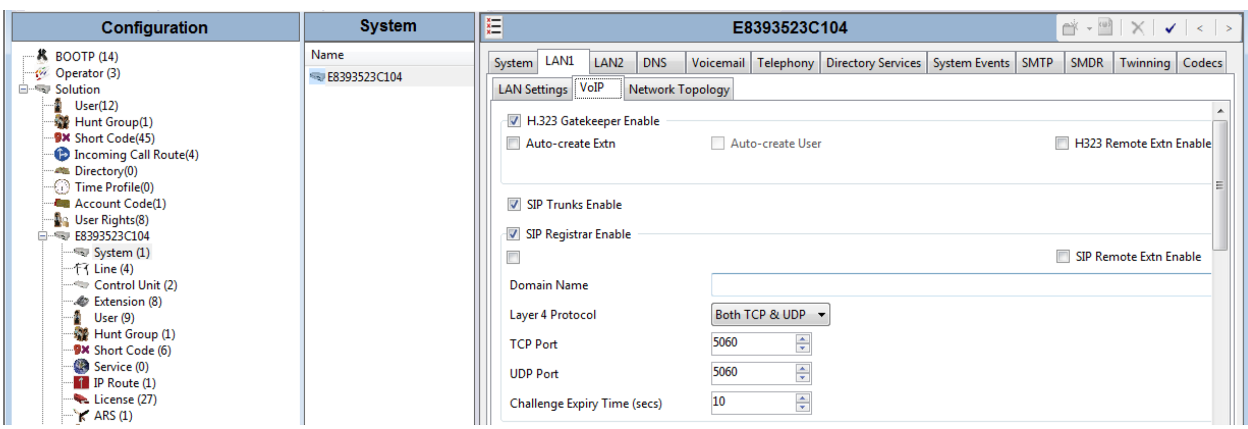
## 5.2. LAN1 Settings

In the sample configuration, LAN1 is used to connect both the Primary Server and the Expansion System to the enterprise network.

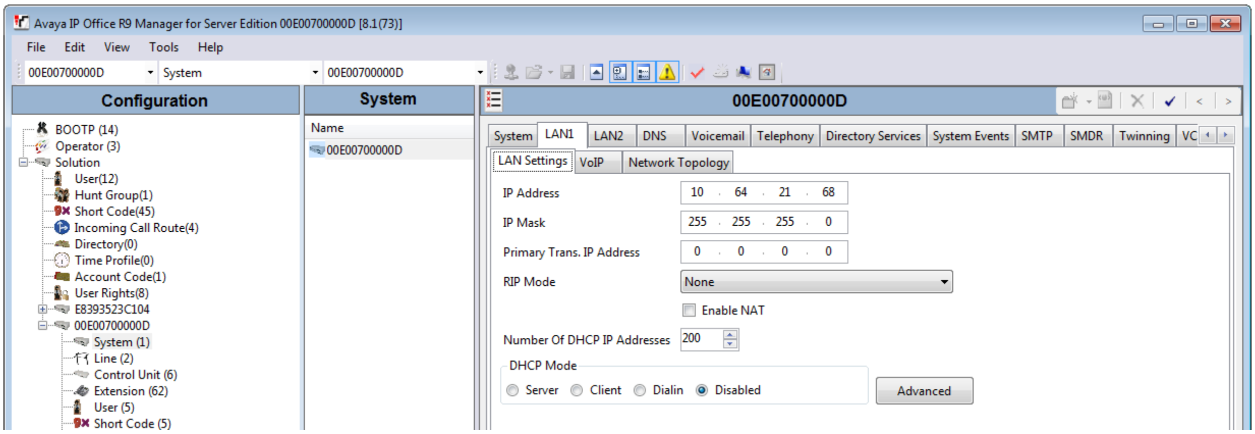
To configure the LAN1 settings on the Primary Server, complete the following steps. Navigate to **Primary** (in this case, **E83935223C104**) **→ System (1)** in the Navigation pane and then to the **LAN1 → LAN Settings** tab in the Details pane. The **IP Address** and **IP Mask** fields should be populated with the values assigned during the Primary Server initial installation process. Verify the configuration or modify the values if needed. While DHCP was disabled during the compliance test, this parameter should be set according to customer requirements.




On the **VoIP** tab in the Details pane, the **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol. The **SIP Trunks Enabled** box is checked to support SIP trunking. The **RTP Port Number Range** (not shown) can be customized to a specific range of listening ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range.



To configure the LAN1 settings for the Expansion System, navigate to **Expansion** (in this case, **00E00700000D**) → **System (1)** on the Navigation pane and then navigate to the **LAN1** → **LAN Settings** tab in the Details pane. The **IP Address** and **IP Mask** fields should be populated with the values assigned during the Expansion System initial installation process. Verify the configuration or modify the values if needed. While DHCP was disabled during the compliance test, this parameter should be set according to customer requirements. Other settings were left at their default values.

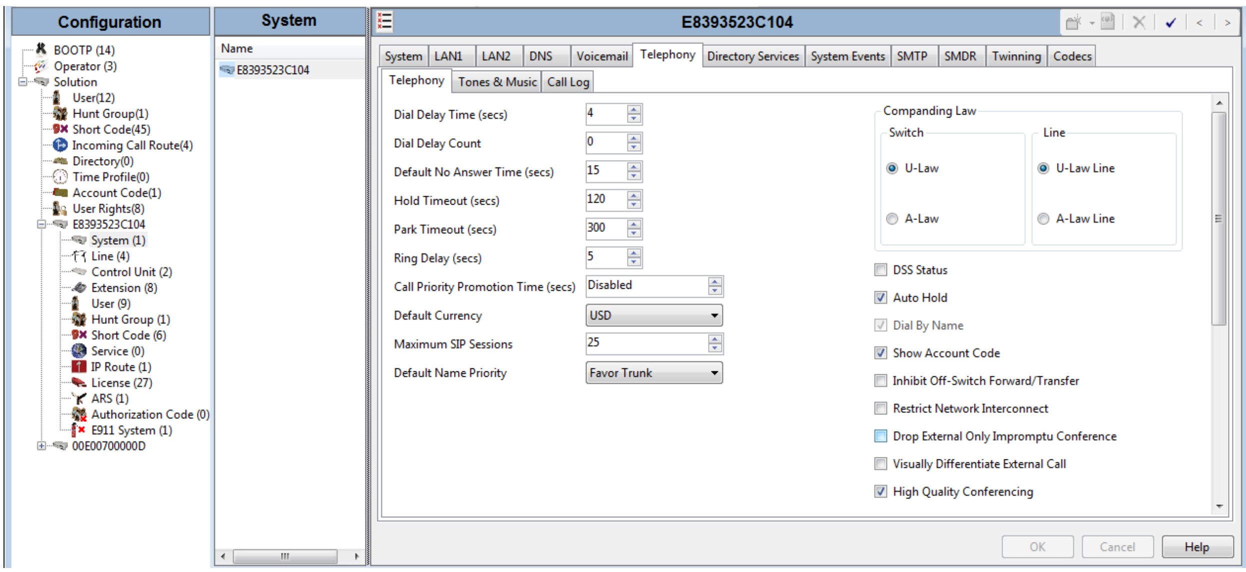



The remaining parameters in the **VoIP** tab for LAN1 in the Expansion System can be configured using the same values previously described for the LAN1 settings in the Primary Server. Use the configuration steps and screens for these tabs previously shown in this section to complete the configuration of the LAN1 settings in the Expansion System.

## 5.3. System Telephony Settings

Navigate to **System(1)** under the Primary server on the Navigation pane and then to **Telephony → Telephony** tab in the Details Pane to configure the Telephony settings for the Primary Server. Choose the **Companding Law** typical for the enterprise location. ***U-Law*** was used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfers to the PSTN.

The **Maximum SIP Sessions** field appears only in Server Edition systems. This value determines the number of SIP Trunk Channel licenses reserved for concurrent sessions on SIP trunks provided by this server. These licenses are reserved from the pool of SIP Trunk Channel licenses shown on **Section 5.1**. In the compliance test, **25** sessions were reserved on the Primary Server. Defaults were used for all other settings.




Navigate to **Expansion → System(1)** and repeat the steps above to configure the Telephony settings for the Expansion System. Since the SIP trunk will be terminated on the Primary Server, it was not necessary to enter a value in the **Maximum SIP Sessions** field in this case, and the default value of **0** was used (not shown).

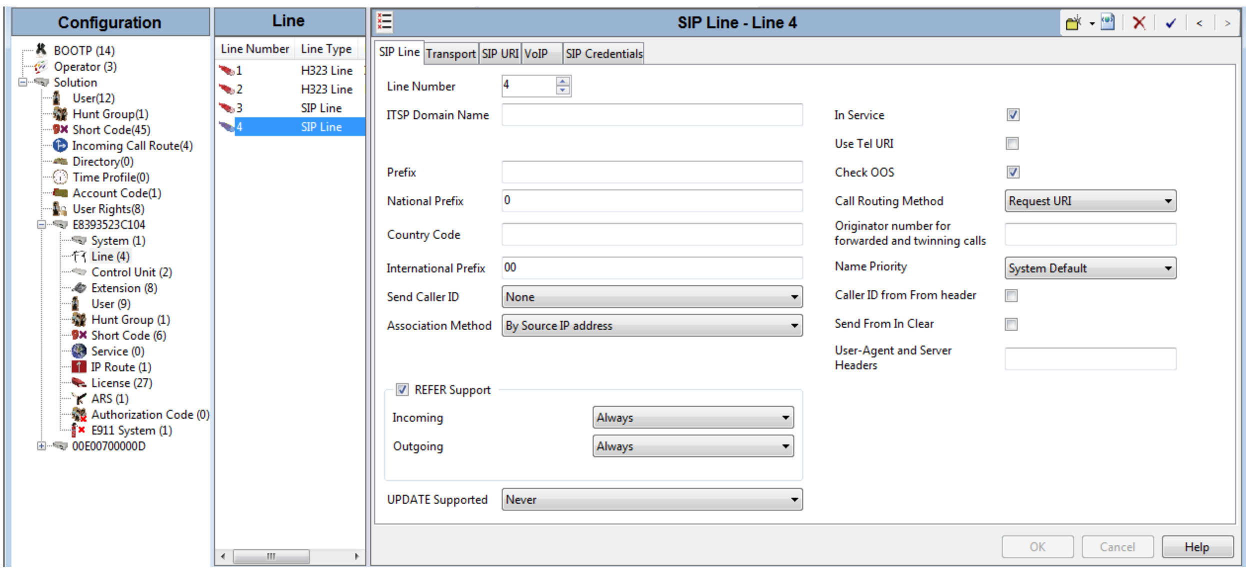
## 5.4. Administer SIP Line

A SIP line is created to establish the SIP connection between the Server Edition Primary Server and the Nuance Speech Attendant server. This line will carry all inbound and outbound traffic between the IP Office and Nuance Speech Attendant. To create the SIP line, navigate to **Primary → Line** in the Navigation pane. Right-click and select **New→ SIP Line** (not shown).

### 5.4.1. SIP Line Tab

On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

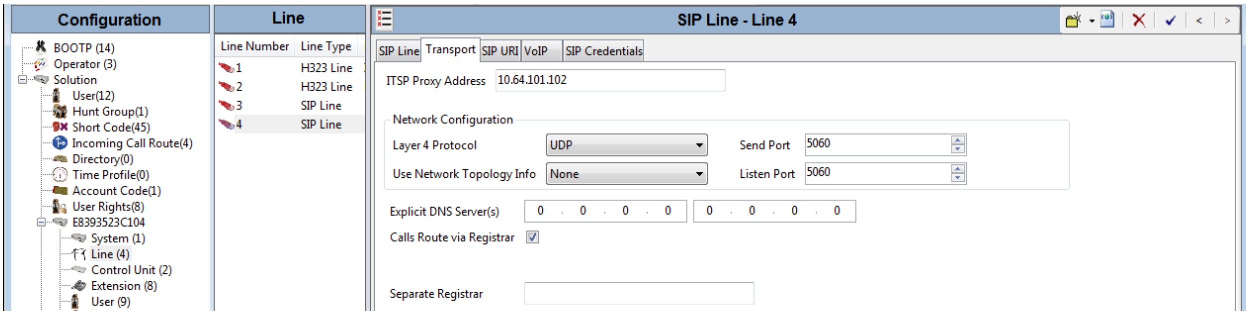
- Leave the ITSP Domain **Name** field blank. With this setting, the IP address of LAN1 in the Primary Server is automatically used in the domain part of the SIP URIs sent to Nuance Speech Attendant.
- Check the **In Service** box.
- Check the **Check OOS** box. With this option selected, the SIP OPTIONS method will be used to periodically check the SIP Line.
- Default values may be used for all other parameters.



## 5.4.2. Transport Tab

Select the **Transport** tab and set the following:

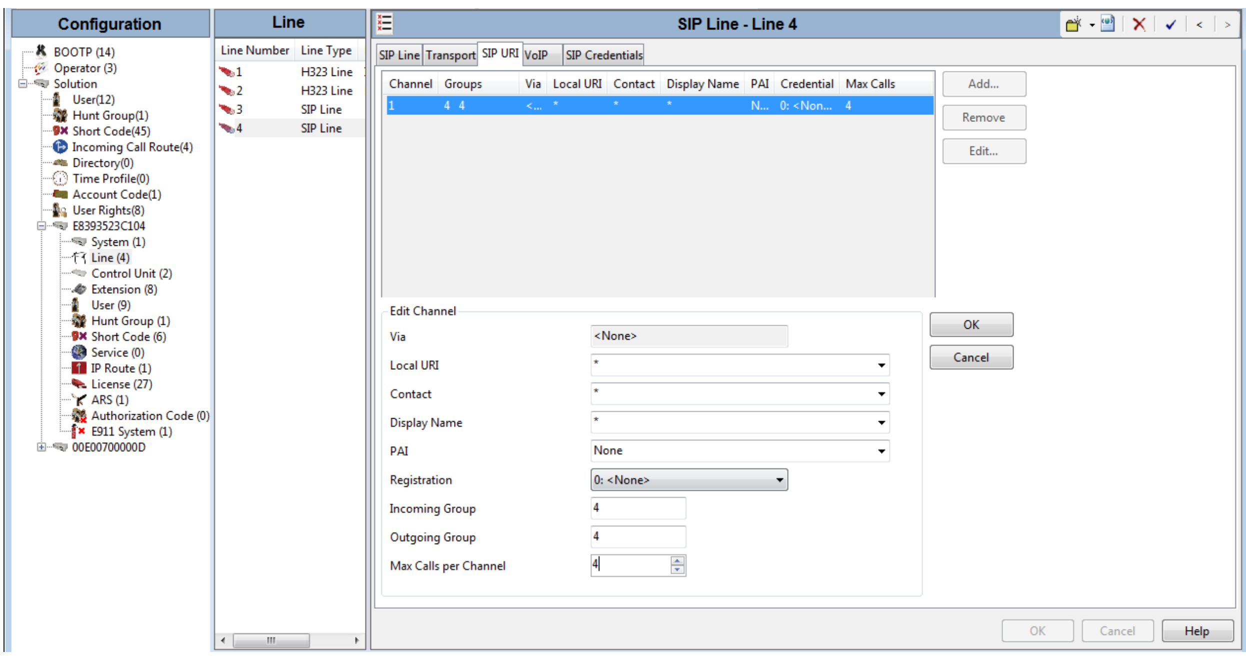
- Set the **ITSP Proxy Address** to the IP address of Nuance Speech Attendant.
- Set the **Layer 4 Protocol** to ***UDP***.
- Set the **Send Port** to ***5060***.
- Default values may be used for all other parameters.



### 5.4.3. SIP URI Tab

A SIP URI entry needs to be created to match each incoming number that Avaya IP Office will accept on this line. Select the **SIP URI** tab and click the **Add** button. The **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the example screen below, a previously configured entry is edited. For the compliance test, a single SIP URI entry was created that matched any number assigned to an Avaya IP Office user. The entry was created with the parameters shown below:

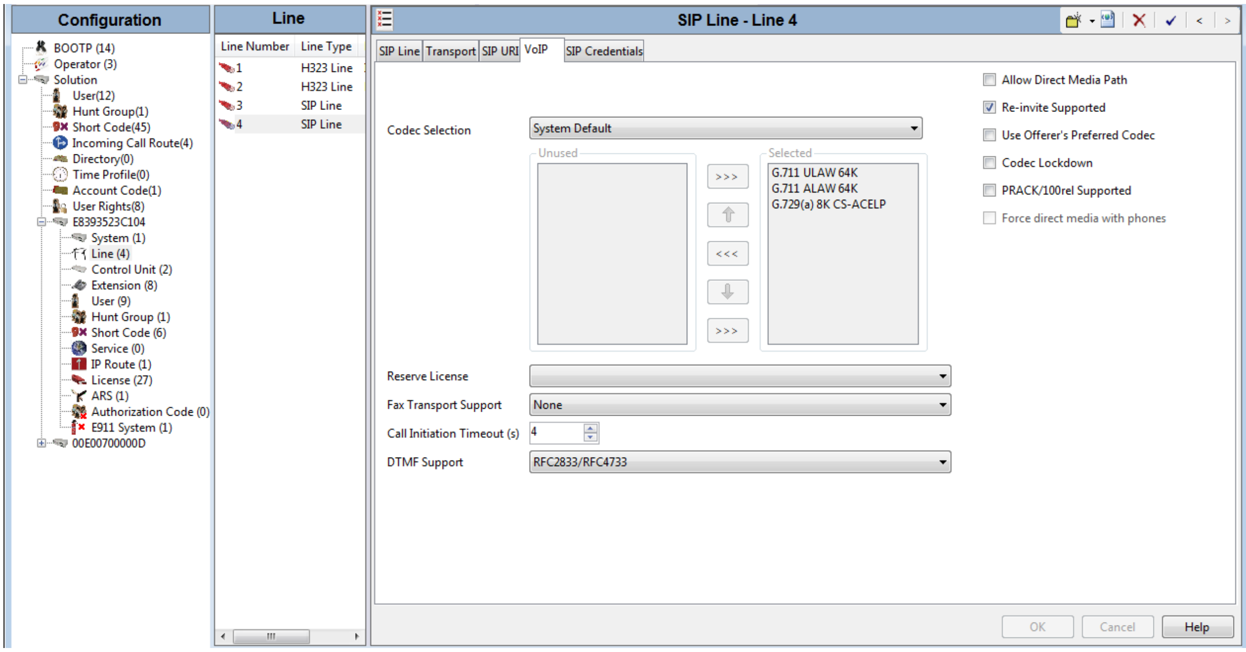
- Set **Local URI**, **Contact**, **Display Name** and **PAI** to *****.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group ***4*** was defined that only contains this line (line 4).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.



### 5.4.4. VoIP Tab

Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- In the sample configuration, the **Codec Selection** was configured using the ***System Default*** option.
- Check the **Re-invite Supported** box. This is necessary to allow the use of re-invites used for codec and direct media path re-negotiation.
- Default values may be used for all other parameters.



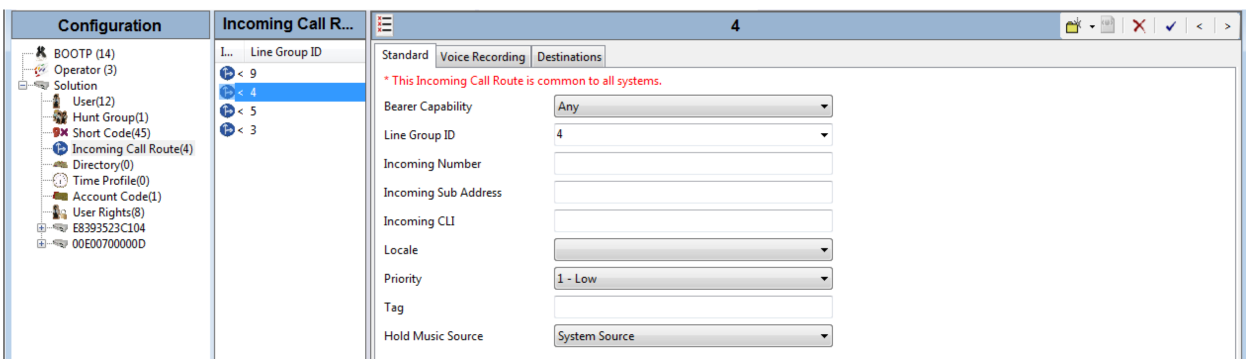
## 5.5. Incoming Call Route

An incoming call route maps inbound numbers on a specific line to internal extensions, hunt groups, short codes, etc, within the IP Office Server Edition solution. Note that in Server Edition systems, Incoming Call Routes are solution settings, shared by all the systems in the solution.

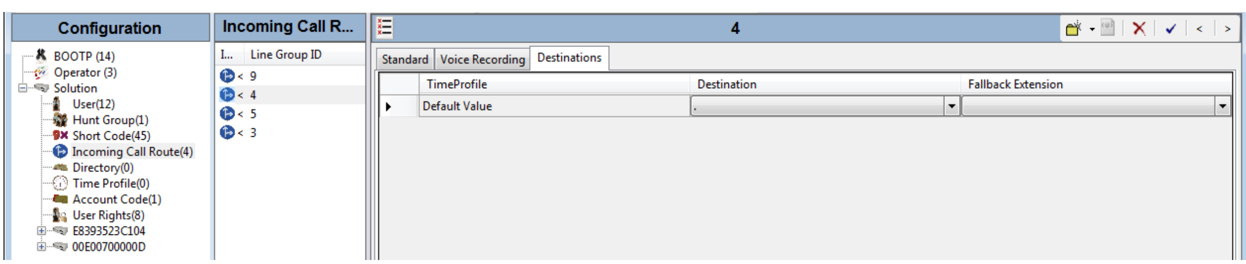
In a scenario like the one used for the compliance test, only one incoming route is needed, which allows any incoming number arriving on the SIP trunk to reach any predefined extension in the IP Office Primary Server or Expansion System.

On the left Navigation Pane, navigate to **Solution**. Right-click on **Incoming Call Route** and select **New** (not shown)**.** On the Details Pane, under the **Standard** tab, set the parameters as show bellow:

- Set **Bearer Capacity** to ***Any***.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.4.3**.
- Leave the **Incoming Number** field blank.
- Default values may be used for all other parameters.




- Under the **Destinations** tab, enter "**.**" For the **Default Value Destination**. This setting will allow the call to be routed to any destination.

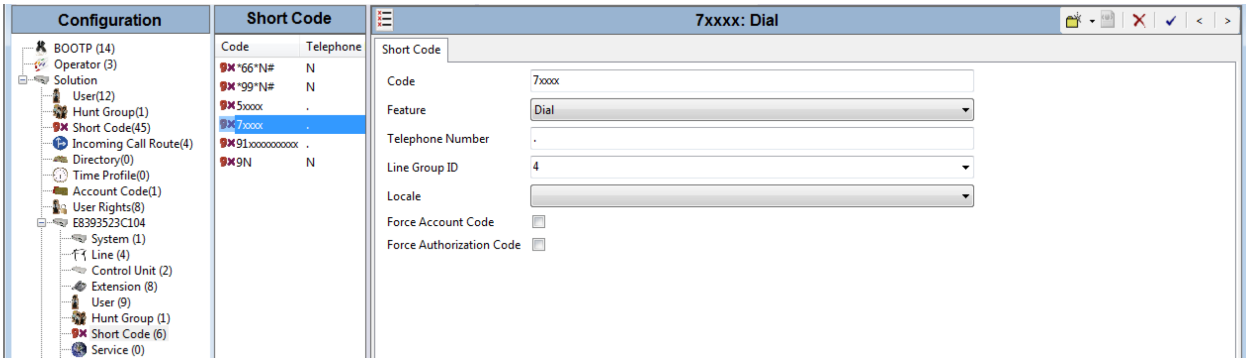


## 5.6. Outbound Call Routing

For outbound call routing, system short codes were used for compliance testing.

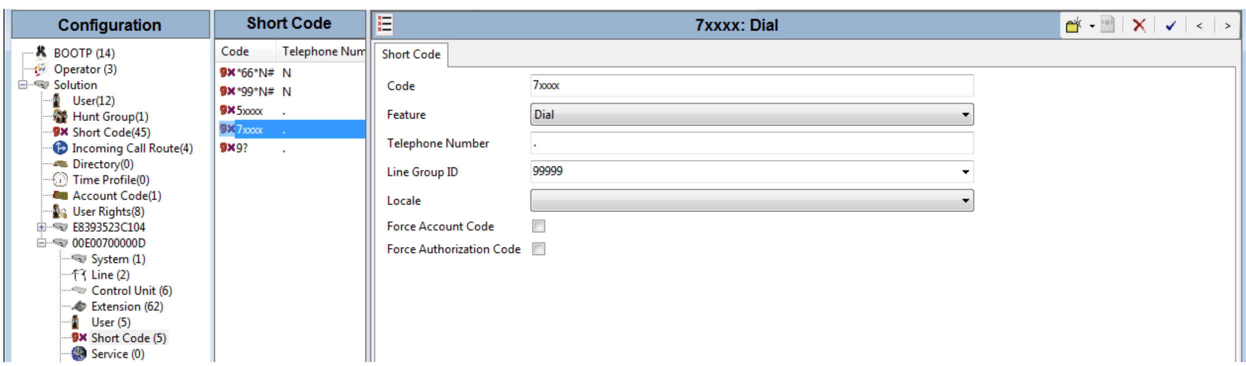
### 5.6.1. Short Codes in the Primary Server

On the left Navigation pane, select right-click on **Primary → Short Code** and select **New** (not shown)**.** The screen below shows the short code added ***7xxxx***, used in the Primary Server to route the digits ***70000*** through ***79999*** to **Line Group ID *4*** to Nuance Speech Attendant.



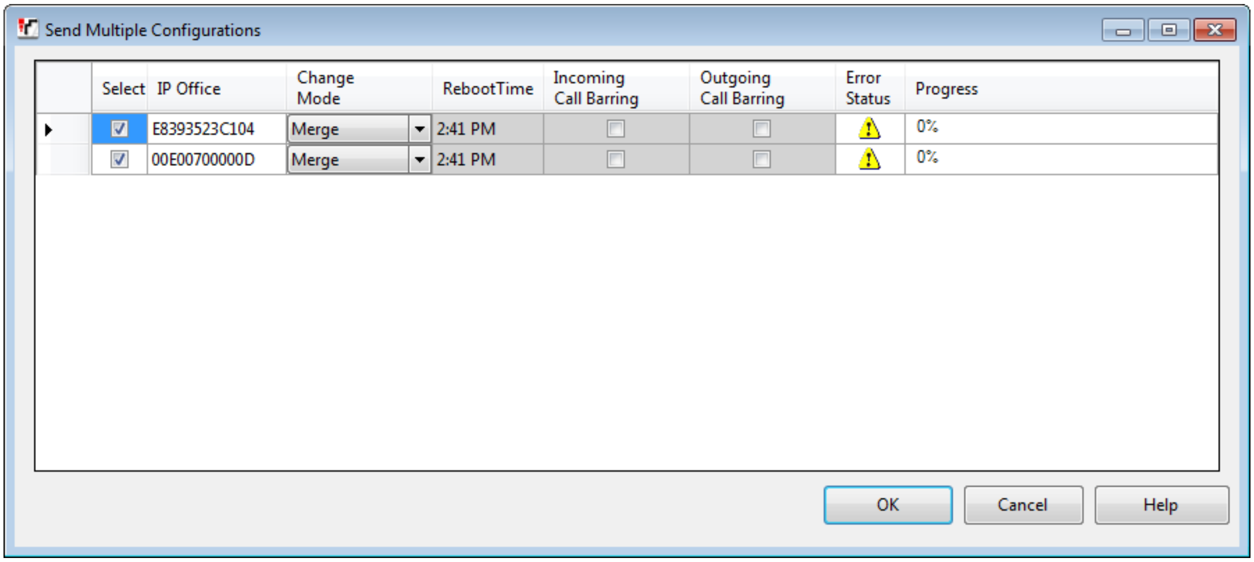
### 5.6.2. Short Codes in the Expansion System

On the left Navigation pane, select right-click on **Expansion → Short Code** and select **New** (not shown)**.** The screen below shows the short code added ***7xxxx***, used in the Expansion Server to route the digits ***70000*** through ***79999*** to **Line Group ID *9999***. Line Group ID 9999 routes calls to the Primary Server. The Primary Server would then route the call to Nuance Speech Attendant as mentioned in the previous section.



## 5.7. Save Configuration

Navigate to **File → Save Configuration** in the menu bar at the top left of the screen (not shown) to save the configuration performed in the preceding sections. A screen like the one shown below is displayed, showing details for those systems where the system configuration has been changed and needs to be sent back to the system. **Reboot** or **Merge** is shown for each system under the **Change Mode** column, based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a service disruption. Click **OK** to save the configuration.



# 6. Configure Nuance Speech Attendant

This section covers the procedure for configuring Nuance Speech Attendant (SA). The procedure includes the following areas:

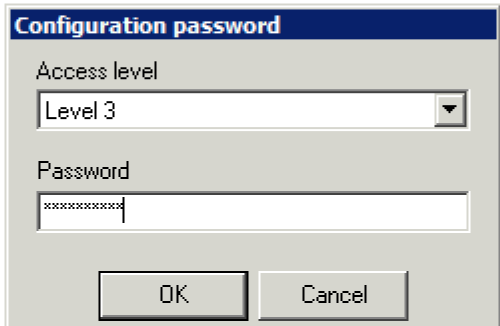
- Configure SIP interface in the Configuration Panel.
- Configure the number of SIP ports supported by Nuance SA.

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

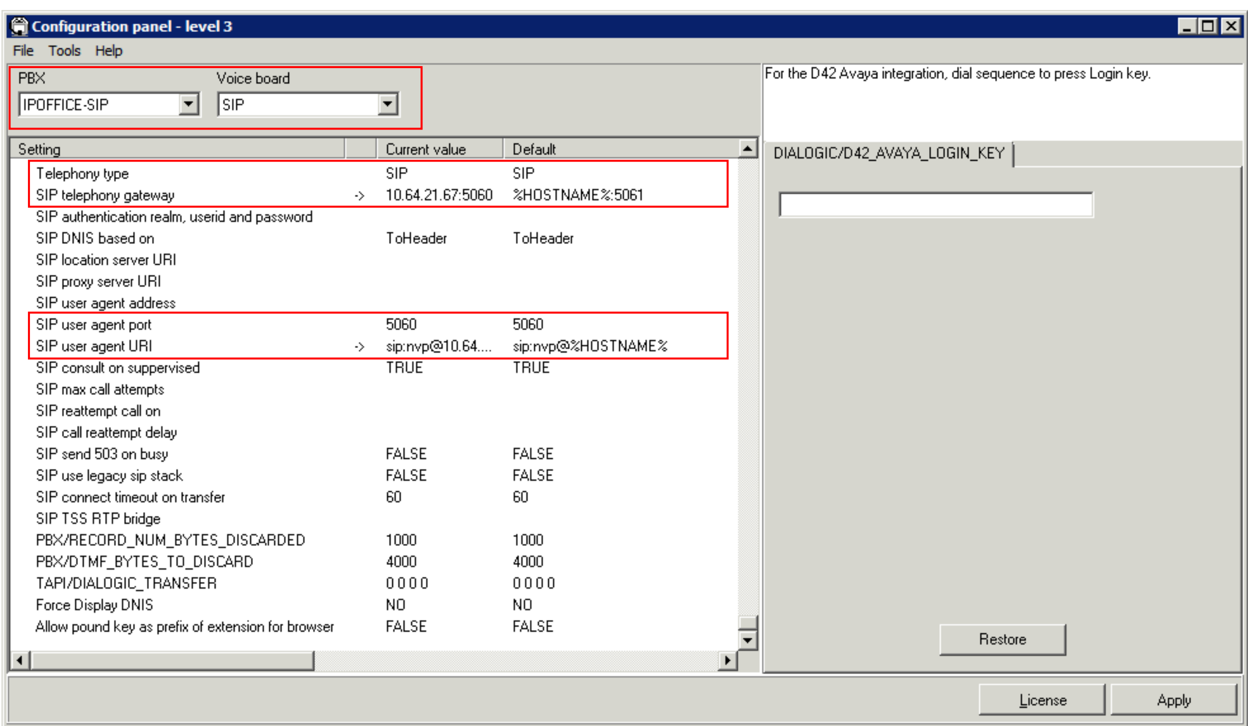
Nuance SA is configured through **Admin Tools** which can be started by navigating to **Start→Programs→Speech Attendant→Admin Tools** (not shown). The initial screen is displayed below.

## 6.1. Configure SIP Interface in Configuration Panel

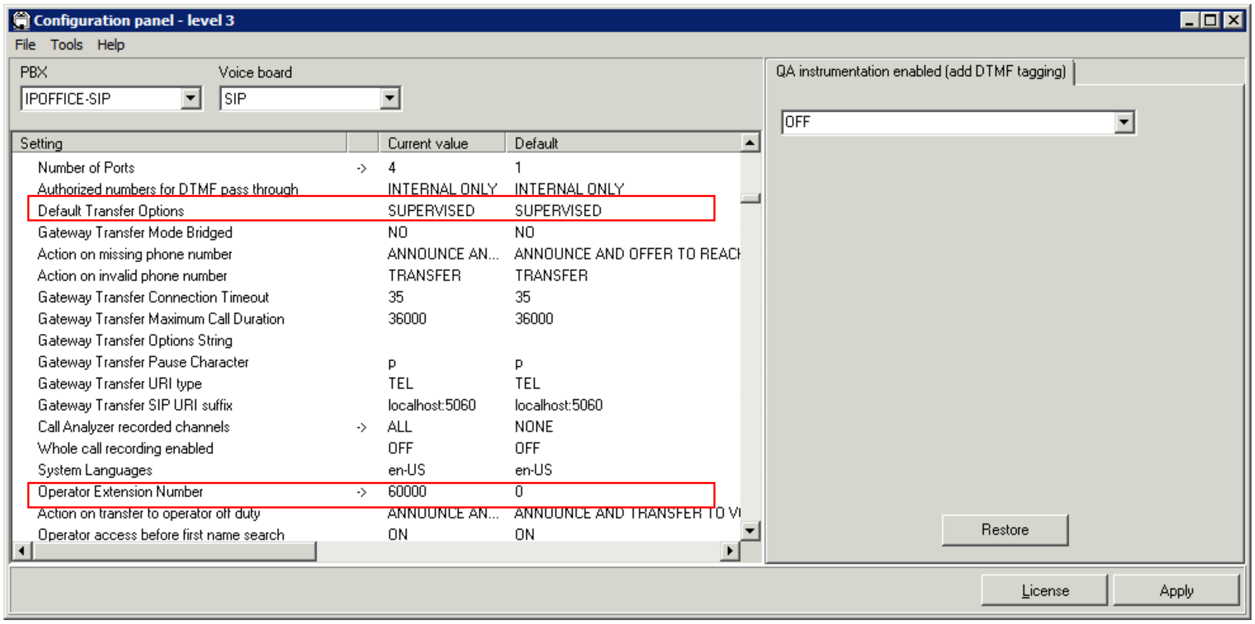
To open the **Configuration Panel**, click on this option in the **Admin Tools** window shown above. The login prompt will be displayed to the user as shown below. Log in with the appropriate credentials using *Level 3* access level.




The **Configuration Panel** shown below is displayed. The **Configuration Panel** allows the SIP interface, transfer mode, and operator extension number to be configured. In general, SA supports both blind and supervised transfers; however, Avaya IP Office does not support a SIP REFER for blind transfers. Therefore, only supervised transfers work and are supported with this solution. To configure the SIP interface, set **PBX** to *IPOFFICE-SIP* and **Voice Board** to *SIP*. Next, set the **Telephony type** field to *SIP* and specify the IP Office primary server IP address and port in the **SIP telephony gateway** field. Configure the SIP port that SA listens on in the **SIP user agent port** field and specify the **SIP user agent URI** for SA.

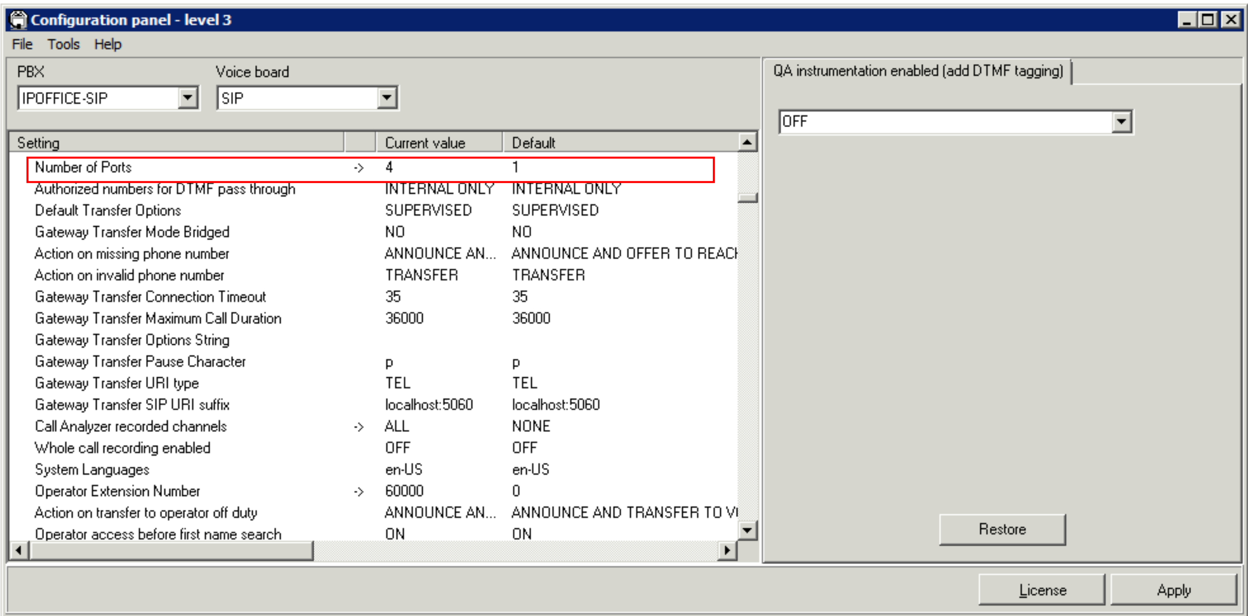


In the **Configuration Panel**, specify the transfer mode (*SUPERVISED*) in the **Default Transfer Options** field, and specify the **Operator Extension Number**, which should be set to a valid extension on IP Office.

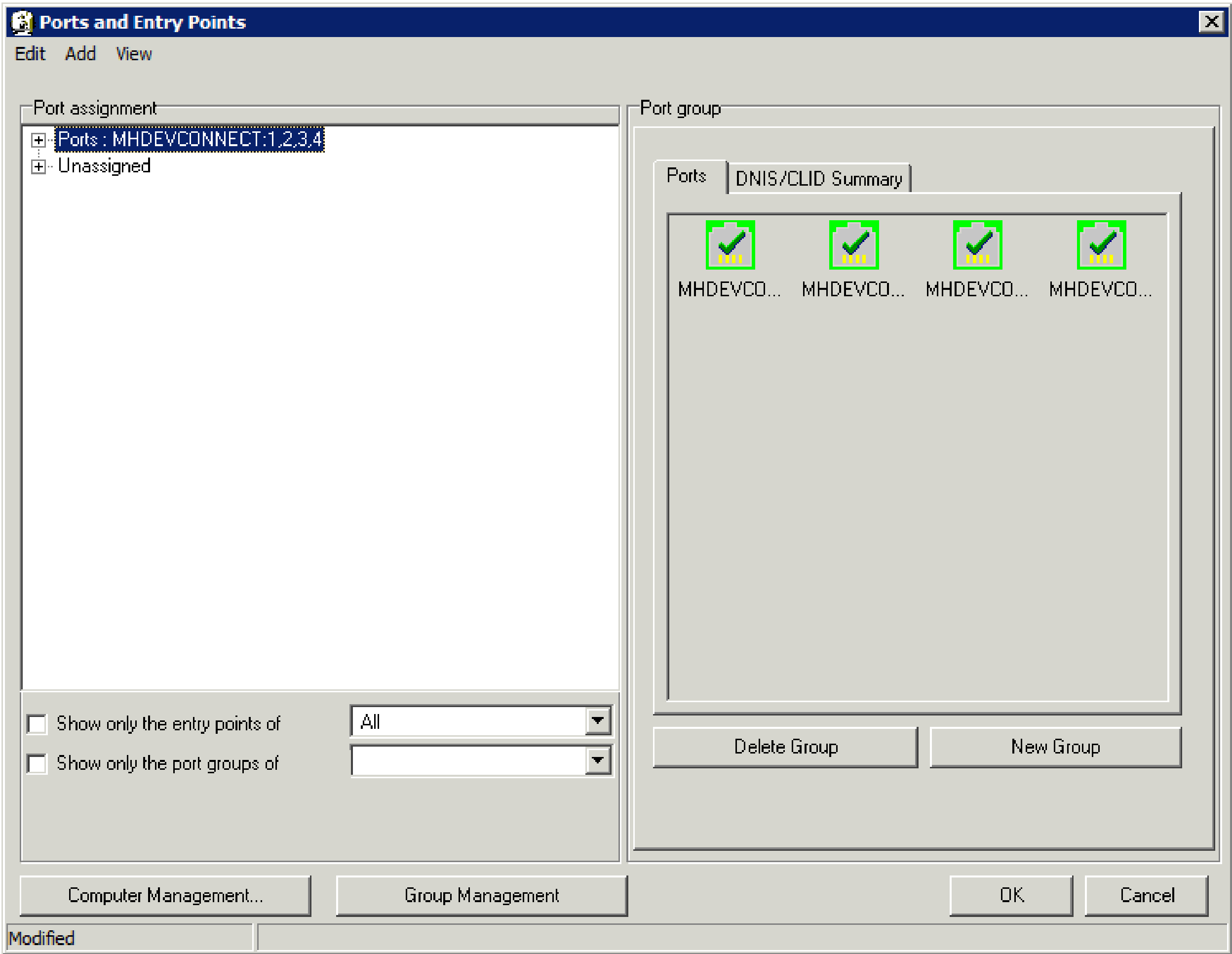


## 6.2. Configure Number of Ports Supported

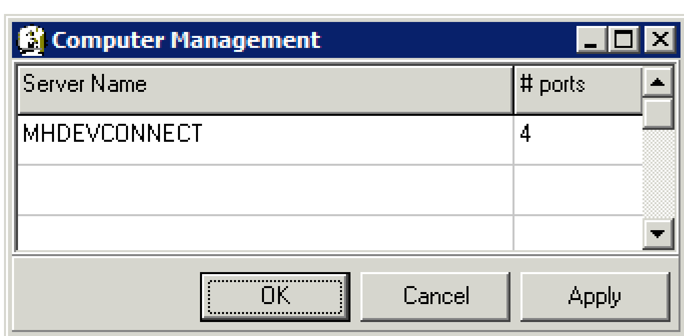
In the **Configuration Panel**, set the **Number of Ports** field to the desired value. In this example, 4 ports were configured. Click **Apply** and then close the **Configuration Panel**. Allow the SA application to be restarted when prompted.



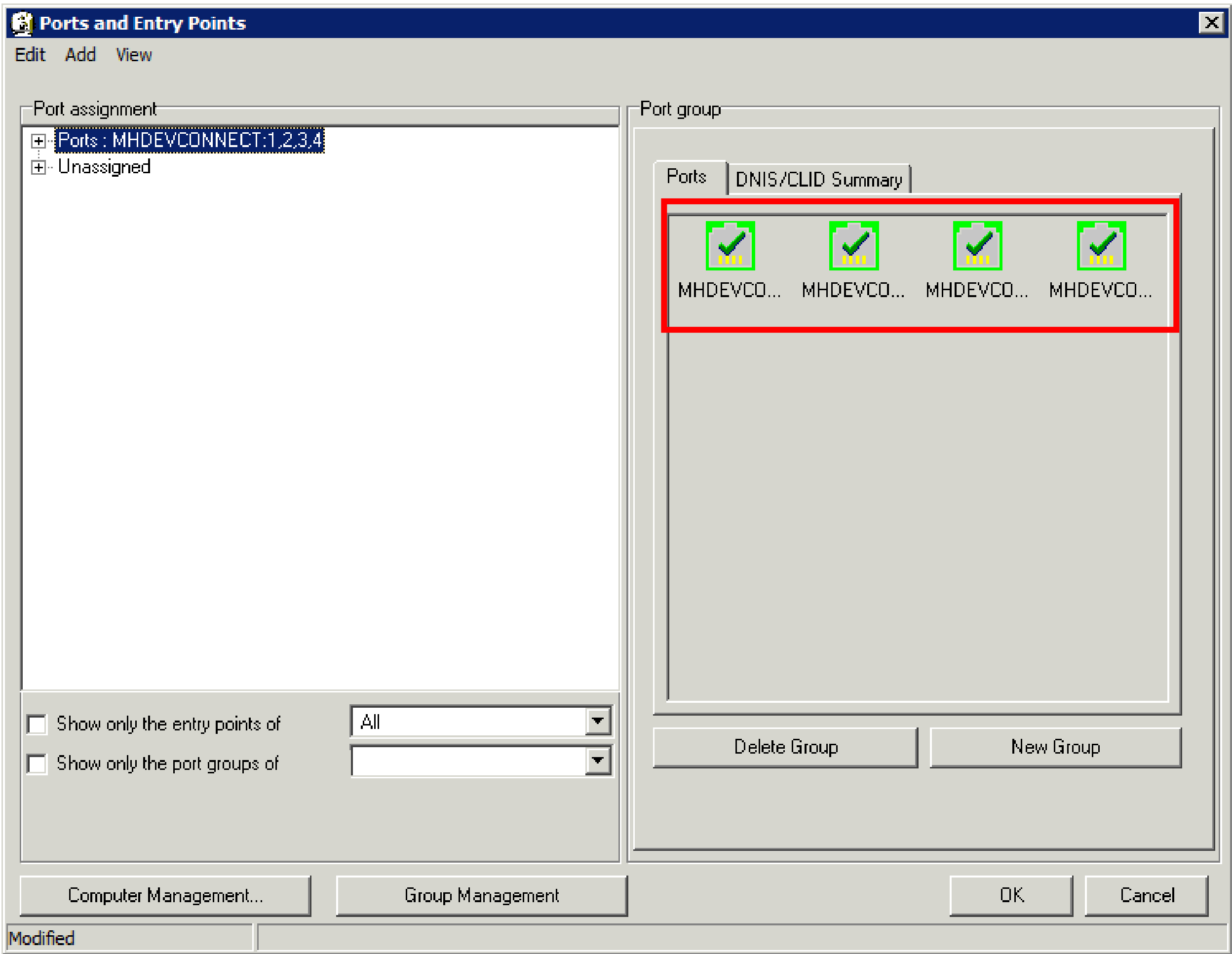
From **Admin Tools**, click on the **Phone Directory and Menu Editor** option and login with the appropriate credentials. In the **Menu Editor** (not shown), navigate to **Edit→Ports and Entry Points** to display the window below. Once the **Ports and Entry Points** window is opened, click on the **Computer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




In the **Computer Management** window, set the # ports column to the desired value. In this example, 4 ports were configured. Click **OK**.



Under the Port group section of the **Ports and Entry Points** window, a list of ports in the system will be displayed. Make sure they are all green (enabled). If the ports are grey (disabled), click on each icon to enable them. Click **OK**.




Next, close the **Configuration Panel**. Allow the system to restart when prompted.

# 7. Verification Steps

This section provides the verification steps that may be performed to verify that Nuance SA is operating properly with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP integration.

1. Place a call to Nuance SA. From the Nuance SA server, open the SA **Monitor** from **Admin Tools** (refer to **Section 6** for accessing **Admin Tools**) and login with the appropriate credentials. Verify that SA detects an active call as shown below.

2. Verify that the Nuance SA greeting is heard and SA transfers the call to the proper destination specified in a spoken name or extension entered via DTMF.

# 8. Conclusion

These Application Notes describe the configuration steps required to integrate Nuance Speech Attendant with Avaya IP Office using a SIP integration. All feature and serviceability test cases were completed successfully. Refer to **Section 2.2** for test results and any observations.

# 9. Additional References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at http://support.avaya.com.


[1] *Deploying IP Office Server Edition Solution IP Office 8.1*, Document 15-604134, December 2012

[2] *IP Office Server Edition Reference Configuration IP Office 8.1*, Document 15-604135, December 2012

[3] *IP Office R8.1 FP1, Manager 10.1*, Document Number 15-601011, April 2013

[4] *Avaya IP Office Knowledgebase*, http://marketingtools.avaya.com/knowledgebase


Nuance product documentation is 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.