



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

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# **Application Notes for DuVoice 6.0 with Avaya IP Office Server Edition 10.1 – Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps required for the DuVoice 6.0 to interoperate with Avaya IP Office Server Edition 10.1. DuVoice is a hospitality messaging system.

In the compliance testing, DuVoice used the SIP User and Configuration Web Service interfaces from Avaya IP Office to provide automated attendant, voicemail, wake-up call, do not disturb, user name and profile template change, and room maid status features.

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 the DuVoice 6.0 to interoperate with Avaya IP Office Server Edition 10.1. DuVoice is a hospitality messaging system.

In the compliance testing, DuVoice used the SIP User and Configuration Web Service interfaces from IP Office to provide automated attendant, voicemail, wake-up call, do not disturb, user name and profile template change, and room maid status features.

The IP Office Server Edition configuration consisted of two IP Office systems, a primary Linux server and an expansion IP500V2 that were connected via Small Community Network (SCN) trunks. The SIP connection between DuVoice and IP Office can be with either the primary or the expansion IP Office system. The configuration shown in these Application Notes used the primary IP Office system for SIP connectivity with DuVoice.

For customer configurations that include a Property Management System (PMS), DuVoice supports the PMS interface for initiation of hospitality features, which is outside the scope of the compliance test. For customer configurations that do not include the PMS, such as in the case of the compliance test configuration, the DuVoice Hospitality Tester tool can be used for initiation of hospitality features. In the compliance testing, the Hospitality Tester tool was running on the DuVoice server and used for initiation of hospitality requests such as check-in, check-out, room move, wakeup call, and do not disturb.

The Configuration Web Service interface was used by DuVoice to provide user name and profile template changes. DuVoice established Configuration Web Service connections with both IP Office systems, for update of user name and profile template on both systems. Two common user rights templates were configured on IP Office for use for check-in, check-out, and room moves.

The SIP User interface was used by DuVoice to provide all remaining hospitality features. In the compliance testing, four virtual SIP users were configured as members of a hospitality group on the primary IP Office system, and were registered by DuVoice with the primary IP Office system. Incoming calls to the hospitality group were delivered over an available virtual SIP user to DuVoice. DuVoice used the SIP packets to determine the type of call and hence the service to provide, such as automated attendant for incoming calls, voicemail coverage for redirected call, voicemail retrieval by subscribers, setting of room maid status from guest telephones, and scheduling of wakeup calls from staff telephones. All SIP communications on DuVoice were supported using the Dialogic Host Media Processing SIP stack.

For the auto attendant feature, incoming calls from the PSTN to the hospitality group were routed via available virtual SIP users to DuVoice. DuVoice played the appropriate greeting announcement and used the inputted DTMF digits along with the SIP REFER method to perform unsupervised transfer of calls to appropriate user destinations on IP Office.

For support of the room maid status feature, a common short code was created on IP Office. The short code was dialed manually from the guest room telephones, with the call routed via an available virtual SIP user to DuVoice, for setting of appropriate maid status for the guest room.

In the compliance testing, subscribers of DuVoice voicemail consisted of all front desk, staff, and guest users on both IP Office systems. The Call Forwarding feature from IP Office was used to redirect calls to DuVoice via virtual SIP users. Three common short codes were configured on IP Office, two were dialed by virtual SIP users for activation and deactivation of the Call Forwarding Unconditional feature for guest users, and one was used for setting the user's Forward Number on IP Office. Upon the guest manually making a request to the front desk or to staff for do not disturb, the Hospitality Tester tool was used in the compliance testing to initiate the Call Forwarding Unconditional feature activation/deactivation on behalf of guests.

Voicemail messages were recorded and saved on DuVoice. Two common short codes were configured on IP Office, and dialed by virtual SIP users for activation and deactivation of subscriber's Message Waiting Indicator (MWI).

For the wakeup call feature, wakeup call requests were made manually to the front desk or staff. The front desk or staff can use his/her telephone to call the hospitality group to schedule a wakeup call on behalf of guests, or use the Hospitality Tool to schedule the wakeup calls. The virtual SIP users were used by DuVoice to initiate wakeup calls to guests.

## **2. General Test Approach and Test Results**

The feature test cases were performed manually. Calls were made from the PSTN and from local users to the hospitality group, for various hospitality features such as auto attendant, voicemail retrieval, etc. The Hospitality Tester was used to manually initiate check-in, check-out, room moves, activate/deactivate do not disturb, and for monitoring of room maid status.

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

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the Configuration Web Service interface between IP Office and DuVoice was encrypted by default with TLS, and the SIP users interface was UDP so did not include use of any specific encryption features as requested by DuVoice.

## **2.1. Interoperability Compliance Testing**

The interoperability compliance test included feature and serviceability testing.

The feature testing focused on verifying the following on DuVoice:

- Registration of virtual SIP users.
- Automated attendant navigation for incoming trunk calls, such as transfer to guests, front desk, and staff.
- Voicemail recording and retrieval, with proper MWI activation/deactivation for users with analog, digital, H.323, and SIP telephone types.
- Scheduling and delivering of wake-up call requests, including retried attempts and escalation to operator.
- Handling of maid codes as indication of room maid status for hospitality group calls from guest telephones.
- Use of virtual SIP users and short codes to set Forwarding Unconditional feature on IP Office for support of do not disturb requests from guests.
- Use of Configuration Web Services to update guest user name and user rights template associated with check-in, check-out, and move requests.

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

## 2.2. Test Results

All test cases were executed and verified. The following were observations on DuVoice from the compliance testing.

- By design, PSTN caller ID was not made available to voicemail subscribers as part of message retrieval.
- Depending on the customer network, greeting and announcement playback from DuVoice to users with analog and digital telephones can be chopped off at times, such that the users may not hear the beginning portion of the playback. DuVoice has a system wide delay variable that can be lengthened to help with this situation, and the default value for the variable is set to one second.
- Depending on the customer network, additional pauses may need to be added for the room maid status short code on IP Office, in order for room maid status setting from guest rooms with analog and digital telephones to be processed successfully by DuVoice. Refer to the room maid status short code configuration on **Section 5.8.4**.
- In the rare case that the trunk connection with the PSTN is SIP trunks via the expansion IP Office system, there can be a no audio connection problems when the PSTN caller calls the hospitality group configured on the primary IP Office system to request transfer to a user on the expansion IP Office system that does not answer with call covered to DuVoice. The no audio issue for this situation is under investigation by the IP Office team. Note that this issue does not exist when the trunk connection with the PSTN is SIP trunks via the primary IP Office system, or when the PSTN trunk connection is PRI via the expansion IP Office system.

## 2.3. Support

Technical support on DuVoice can be obtained through the following:

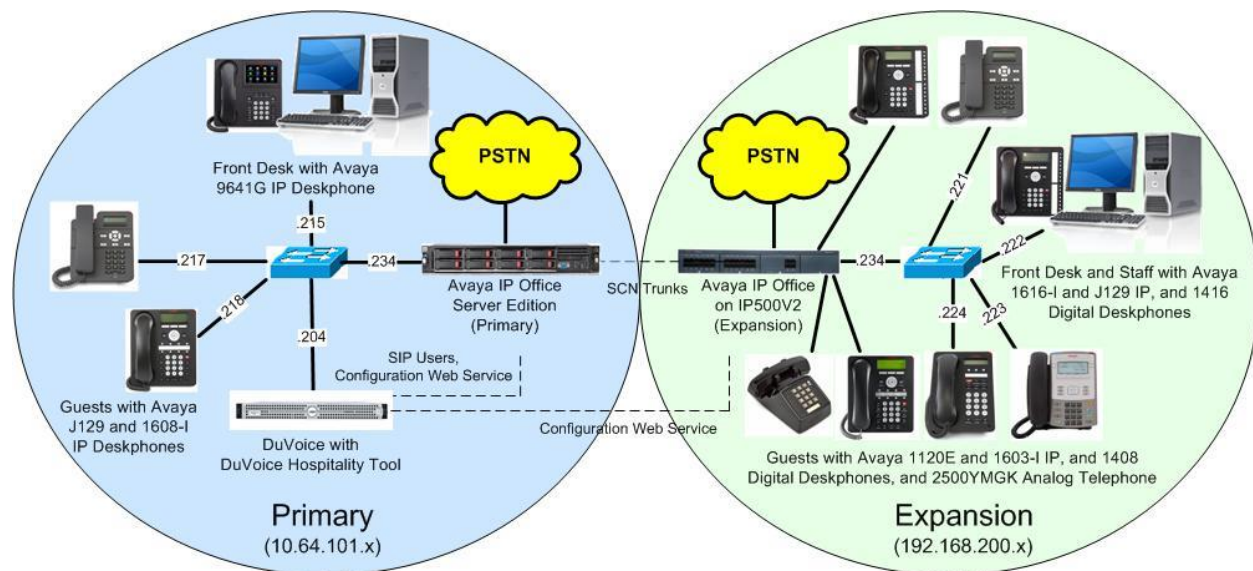
- **Phone:** (425) 250-2393
- **Email:** [support@duvoice.com](mailto:support@duvoice.com)

### 3. Reference Configuration

The configuration used for the compliance testing is shown in **Figure 1**. The DuVoice server used in the testing included the Dialogic Host Media Processing Software for support of the SIP protocol.

The IP Office resources used in the compliance testing are shown in the table below.

Users	Extensions
<b>Primary</b>	
Front Desk	21030
Guests	21031, 21041
<b>Expansion</b>	
Front Desk and Staff	22020, 22030, 22040
Guests	22001, 22021, 22031, 22041



**Figure 1: Compliance Testing Configuration**

## 4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office Server Edition (Primary) in Virtual Environment	10.1.0.0.0
Avaya IP Office on IP500 V2	10.1.0.0.0
Avaya 1120E IP Deskphone (SIP)	4.4.23.0
Avaya J129 IP Deskphone (SIP)	1.1.0.0.15
Avaya 1603-I, 1608-I, 1616-I IP Deskphones (H.323)	1.3110
Avaya 9641G IP Deskphone (H.323)	6.6506
Avaya 1408 and 1416 Digital Deskphones	46.0
2500YMGK Analog Phone	NA
DuVoice on Microsoft Windows 10 Pro	6.0.73
• Dialogic Host Media Processing Software	3.0.375
• Hospitality Tester	NA
• Avaya Configuration Web Service SDK	10.1

*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 IP Office. The procedures include the following areas:

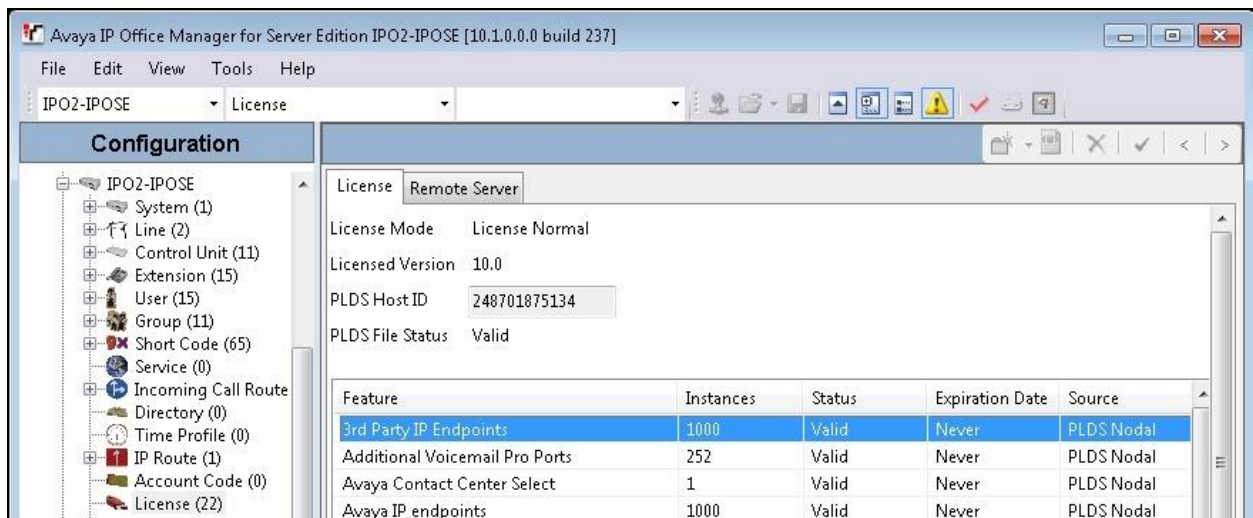
- Verify license
- Obtain LAN IP address
- Administer SIP Registrar
- Administer SIP extensions
- Administer SIP users
- Administer hospitality group
- Administer incoming call route
- Administer short codes
- Administer voicemail users
- Administer user rights
- Administer security service

### 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 users connections with DuVoice, in this case “IPO2-IPOSE”, a list of licenses is displayed in the right pane. Verify that there are sufficient licenses for **3<sup>rd</sup> Party IP Endpoints**, as shown below.

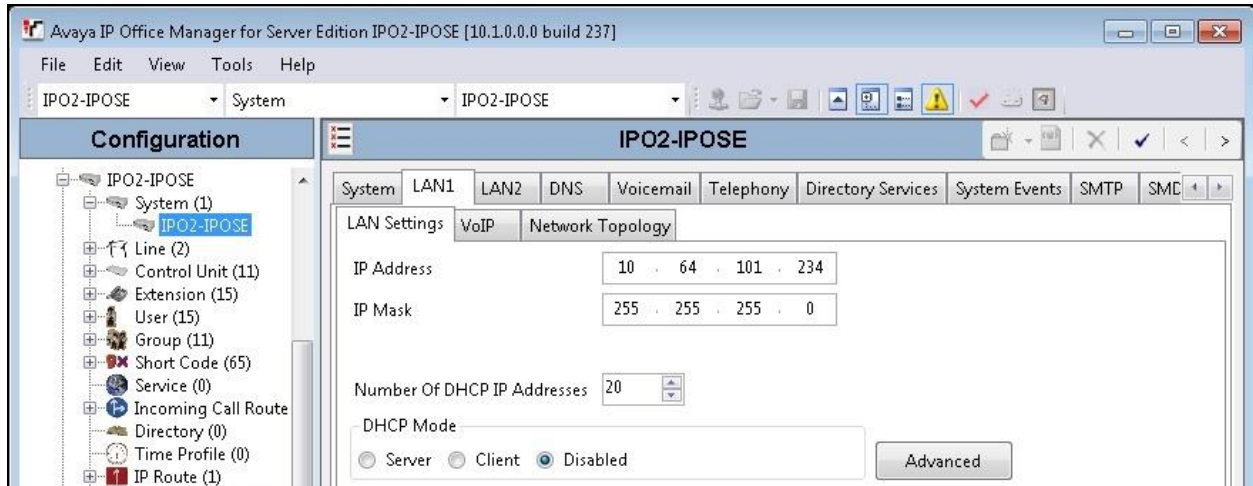




## 5.2. Obtain LAN IP Address

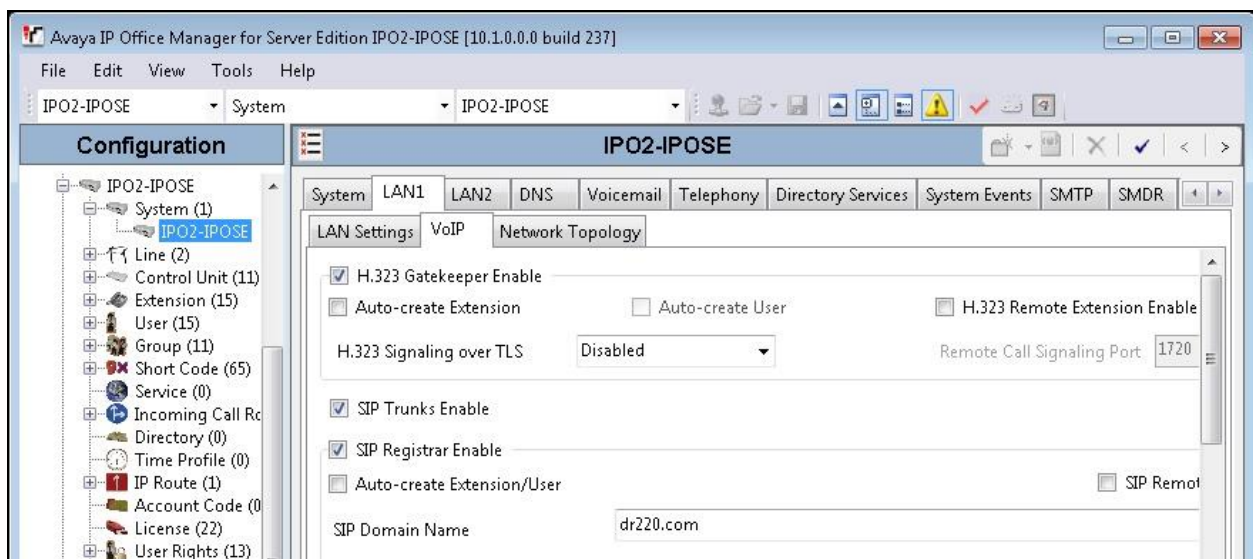
From the configuration tree in the left pane, select **System** under the IP Office system that will be used for SIP user connections with DuVoice, in this case “IPO2-IPOSE”. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane.

Make a note of the **IP Address**, which will be used later to configure DuVoice. Note that IP Office can support SIP on the LAN1 and/or LAN2 interfaces, and the compliance testing used the LAN1 interface.



## 5.3. Administer SIP Registrar

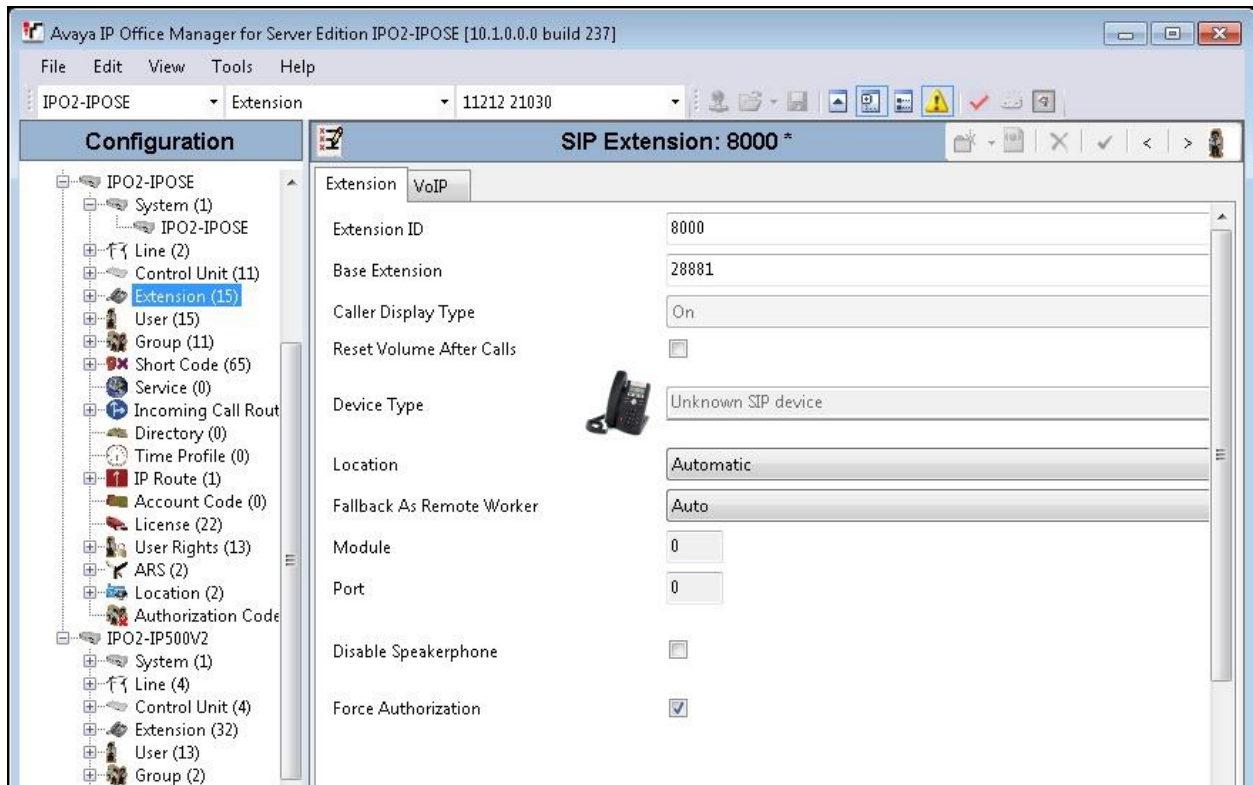
Select the **VoIP** sub-tab. Make certain that **SIP Registrar Enable** is checked, as shown below. Make a note of the **SIP Domain Name** field value, which will be used later to configure DuVoice.



## 5.4. Administer SIP Extensions

From the configuration tree in the left pane, right-click on **Extension** under the IP Office system that will be used for SIP user connections with DuVoice, in this case “IPO2-IPOSE”, and select **New → SIP Extension** from the pop-up list to add a new SIP extension.

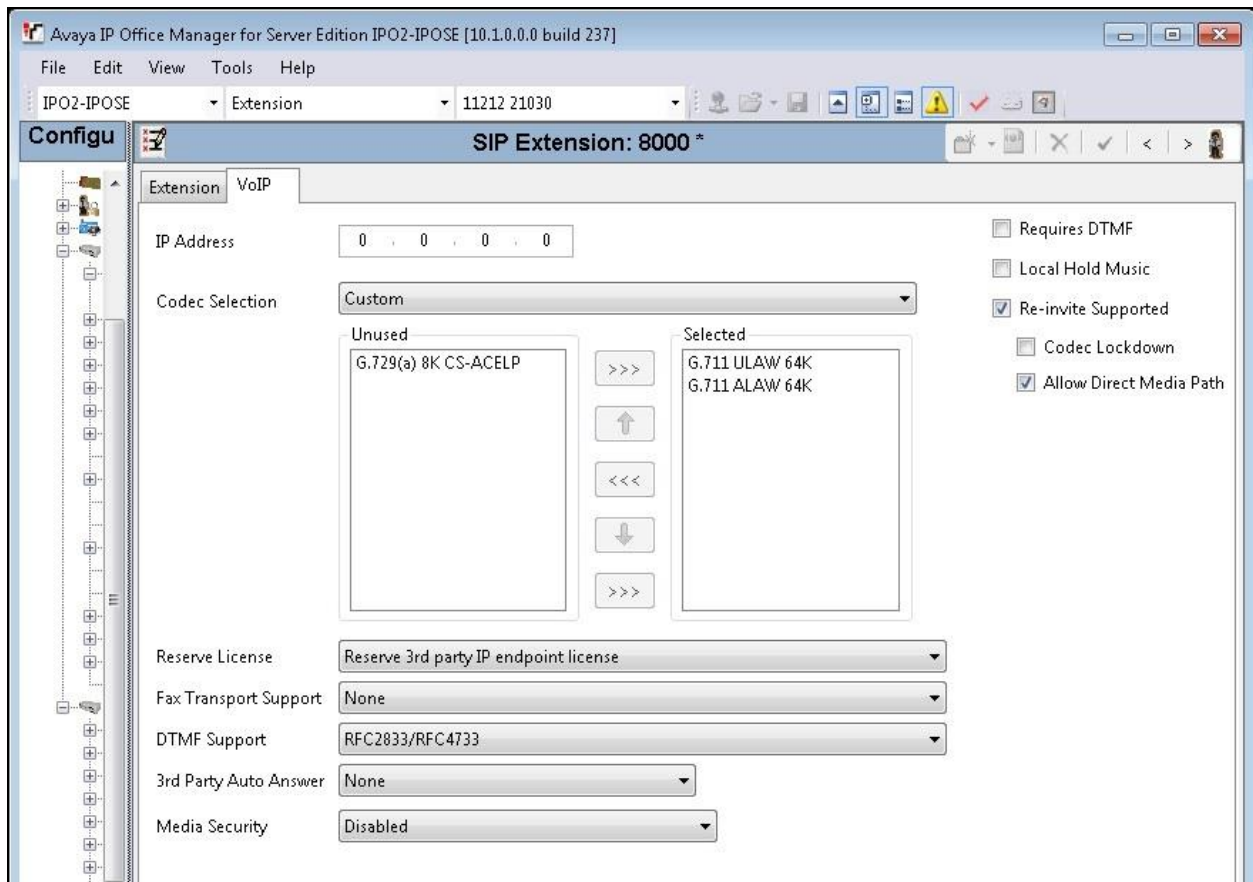
Enter the desired digits for **Base Extension**, and retain the default values in the remaining fields.



Select the **VoIP** tab. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Codec Selection:** “Custom”
- **Selected:** Retain only the G.711 codec variants.
- **Reserve License:** “Reserve 3rd party IP endpoint license”
- **Media Security:** “Disabled”

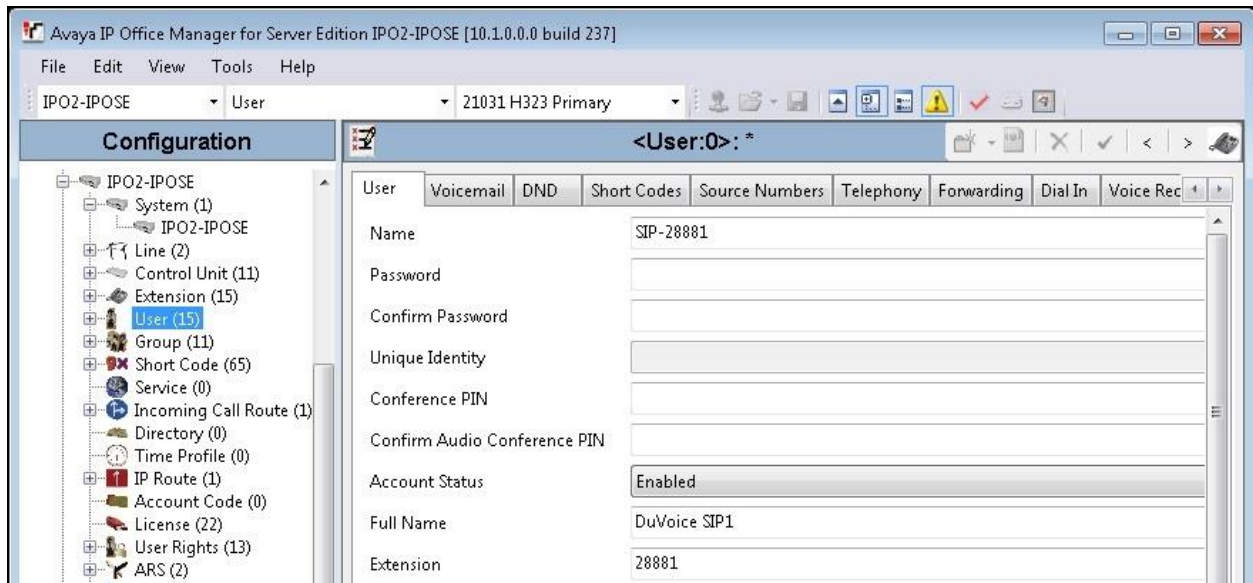
Repeat this section to add the desired number of SIP extensions. In the compliance testing, four SIP extensions with base extensions of 28881-28884 were created.



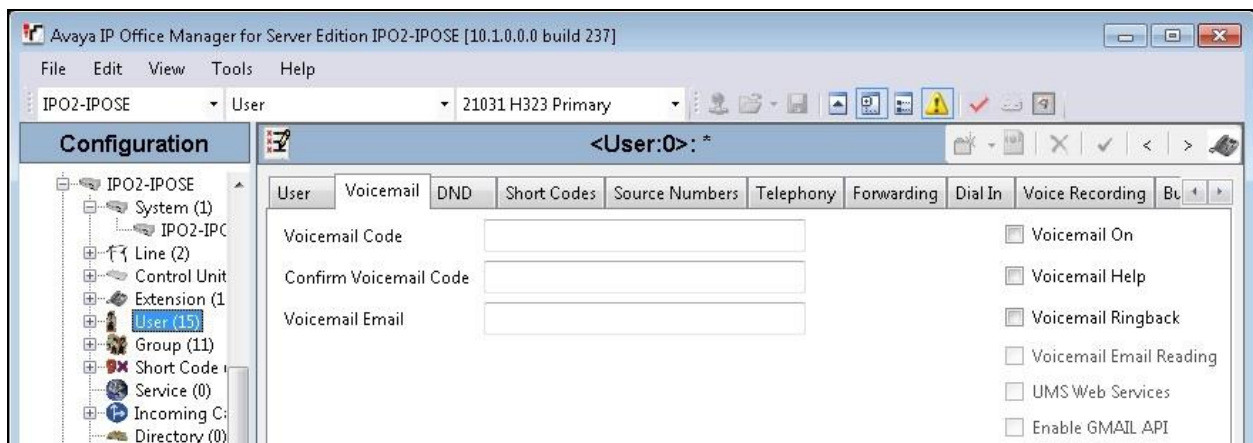
## 5.5. Administer SIP Users

From the configuration tree in the left pane, right-click on **User** under the IP Office system that will be used for SIP users connections with DuVoice, in this case “IPO2-IPOSE”, and select **New** from the pop-up list to add a new user.

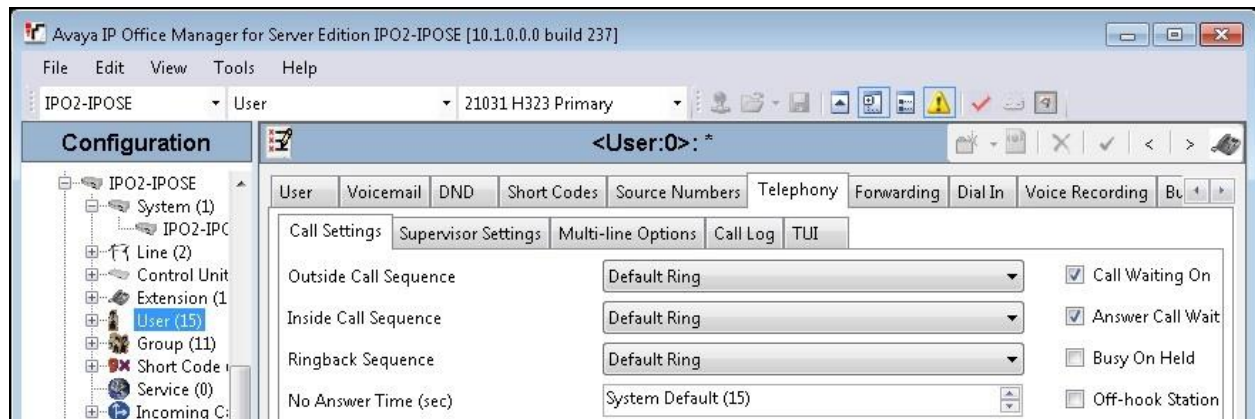
Enter desired values for **Name** and **Full Name**. For **Extension**, enter the first SIP base extension from **Section 5.4**. Retain the default values in the remaining fields.



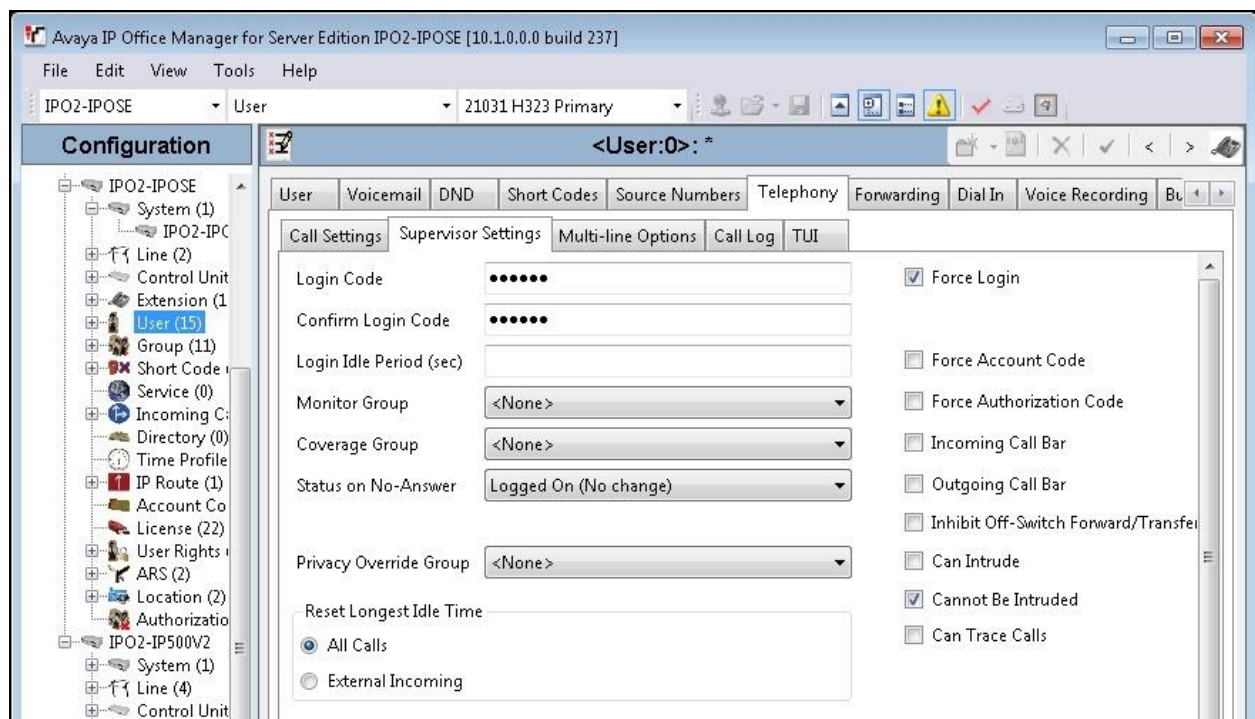
Select the **Voicemail** tab, and uncheck **Voicemail On**, as shown below.



Select the **Telephony** tab, followed by the **Call Settings** sub-tab. Check **Call Waiting On**, as shown below. Retain the default values in the remaining fields.



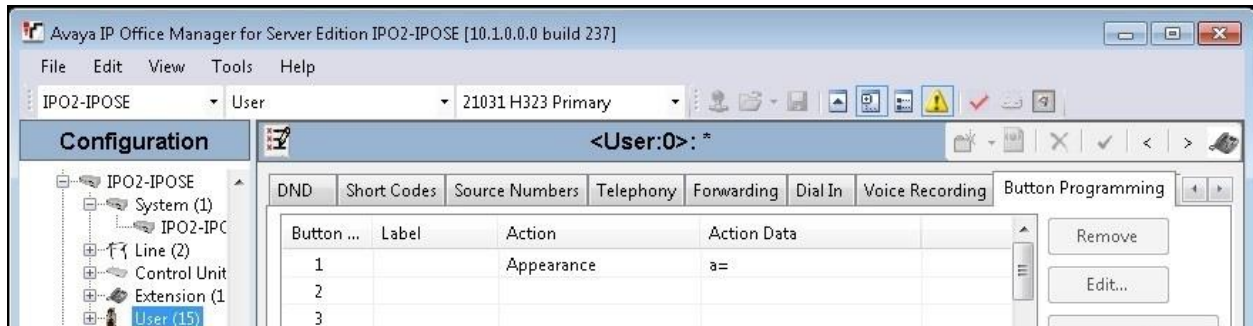
Select the **Supervisor Settings** sub-tab. Enter desired password for **Login Code** and **Confirm Login Code**, and check **Force Login**. Retain the default values in the remaining fields.





Select the **Button Programming** tab. Retain only the first **Appearance** button and remove all others as shown below.

Repeat this section to add a new user for each remaining SIP extensions from **Section 5.4**. In the compliance testing, four users with names of “SIP-28881” to “SIP-28884” were created.

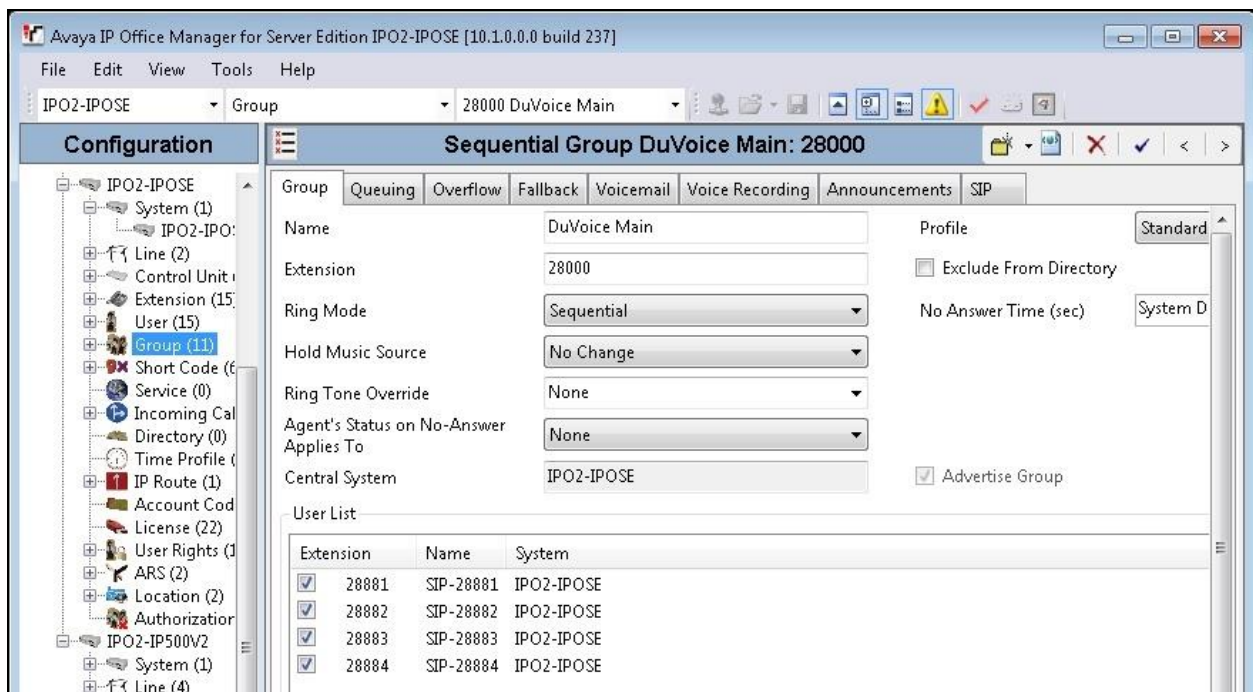


## 5.6. Administer Hospitality Group

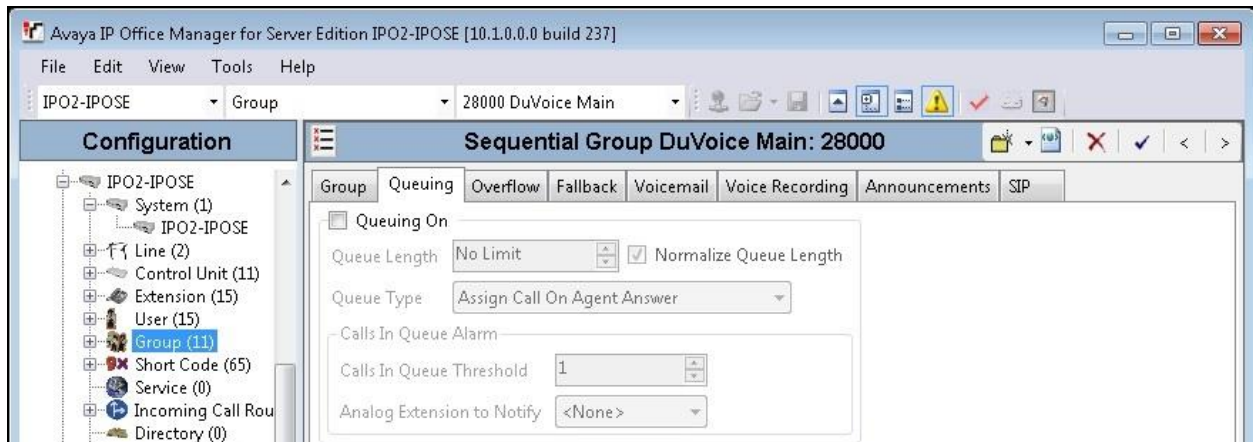
From the configuration tree in the left pane, right-click on **Group** under the IP Office system that will be used for SIP users connections with DuVoice, in this case “IPO2-IPOSE”, and select **New** from the pop-up list to add a new group. This group will be used to deliver calls to DuVoice for hospitality features.

Enter desired values for **Name** and **Extension**. For **Ring Mode**, select “Sequential” from the drop-down list. Retain the default values in the remaining fields.

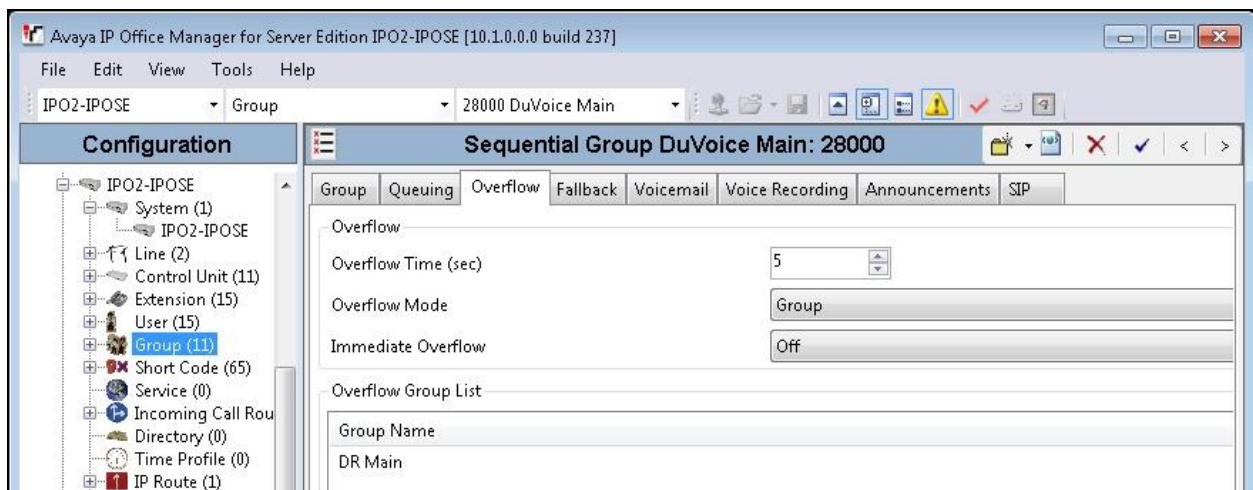
In the **User List** sub-section, add all SIP users from **Section 5.5** as members, as shown below.



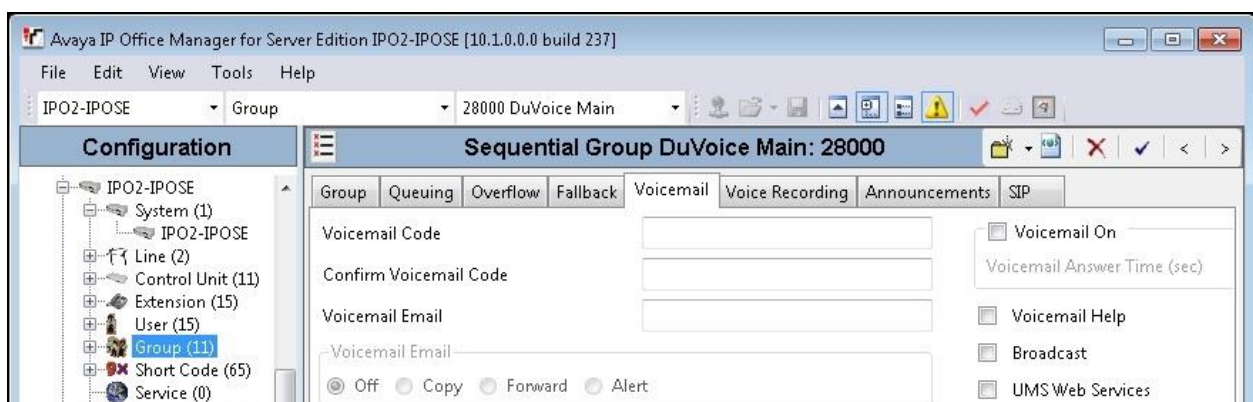
Select the **Queuing** tab, and uncheck **Queuing On**, as shown below.



Select the **Overflow** tab. In the **Overflow Group List** sub-section, add a desired group for coverage of non-answered calls for the hospitality group. In the compliance testing, “DR Main” is a pre-existing group consisting of front desk and staff users on both IP Office systems.



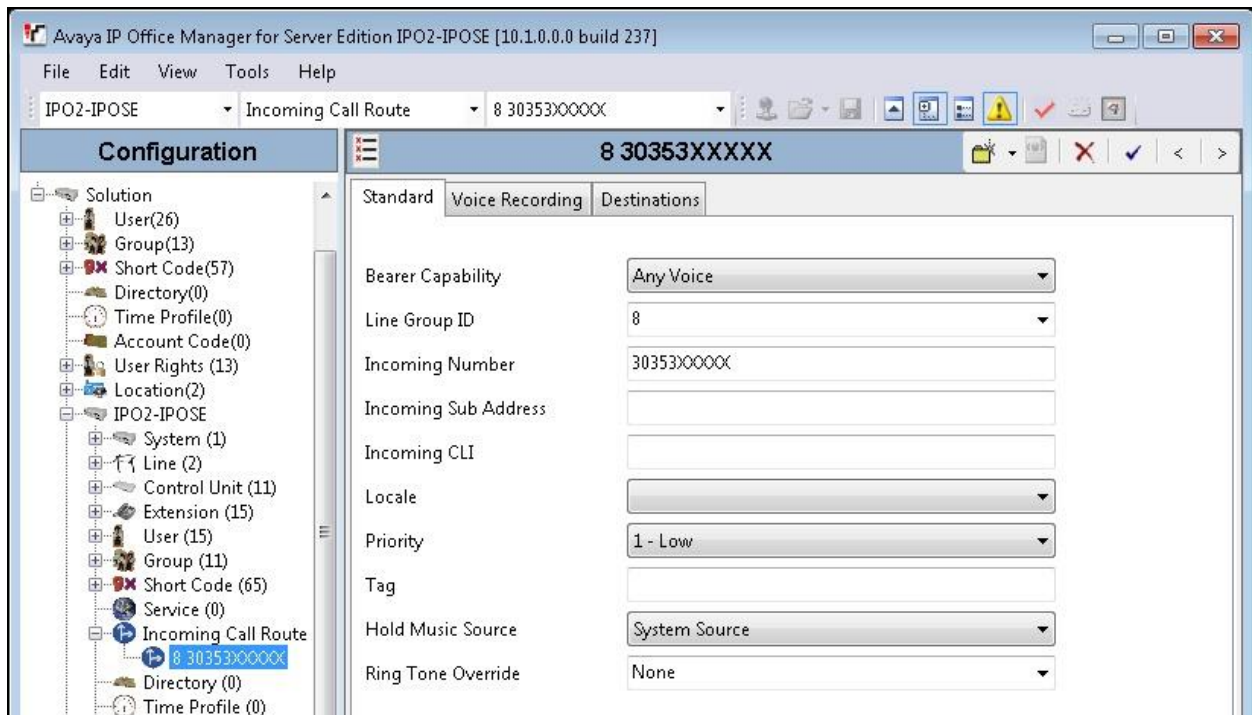
Select the **Voicemail** tab, and uncheck **Voicemail On**, as shown below.



## 5.7. Administer Incoming Call Route

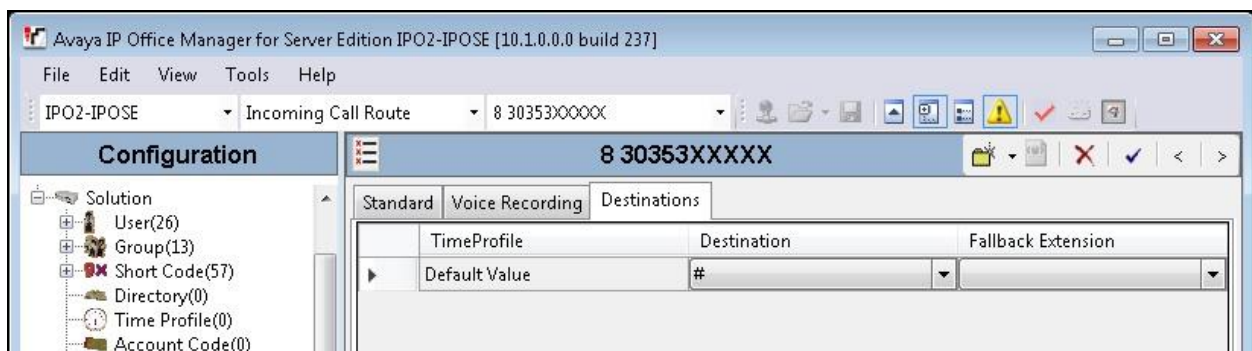
If necessary, create an incoming call route to route incoming calls to the hospitality group. In the compliance testing, the existing incoming call route on the primary IP Office system for connection to the PSTN can route to any five digit extensions on both IP Office systems.

As shown in the screen below, the **Incoming Number** for the line is “30353xxxxx”, which uses five single digit wildcards “x” allowing the last five digits to be any number.



In the **Destinations** tab, the use of “#” in the **Destination** field enables the routing to be based on the “xxxxx” from the **Incoming Number** field from above. Therefore, incoming calls to “3035328000” will be routed to the hospitality group configured in **Section 5.6**.

If desired, the hospitality group can be selected from the **Destination** drop-down to route all incoming trunk calls to DuVoice.



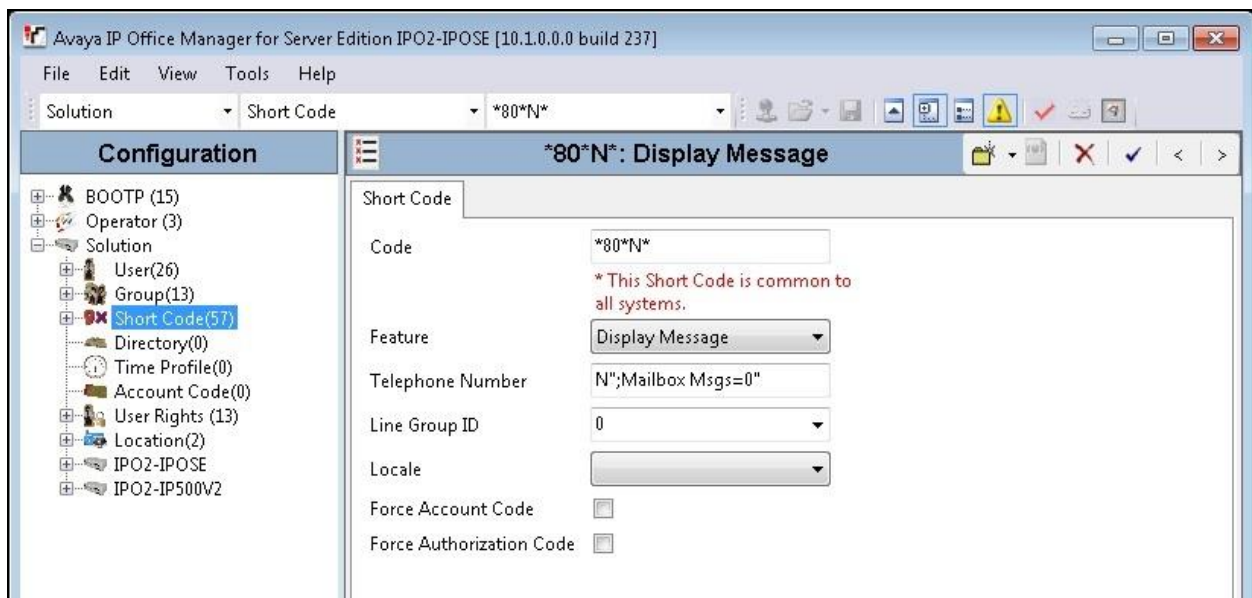
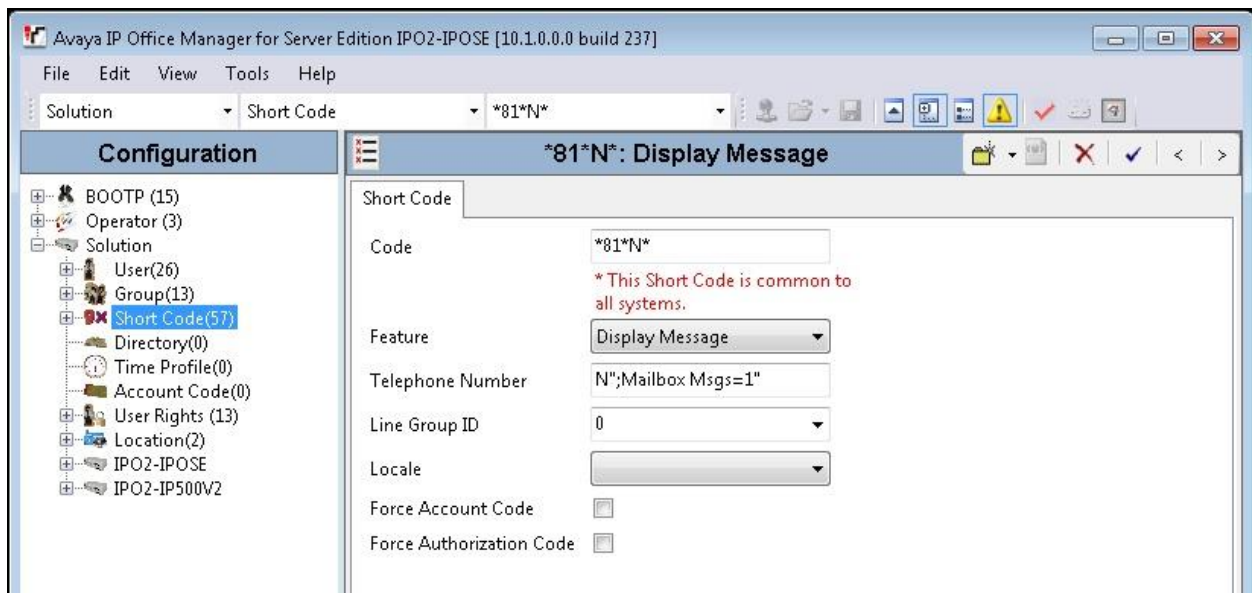


## 5.8. Administer Short Codes

From the configuration tree in the left pane, under **Solution** → **Short Code**, create a set of common short codes for support of MWI, Call Forwarding Unconditional, operator, and room maid status.

### 5.8.1. MWI

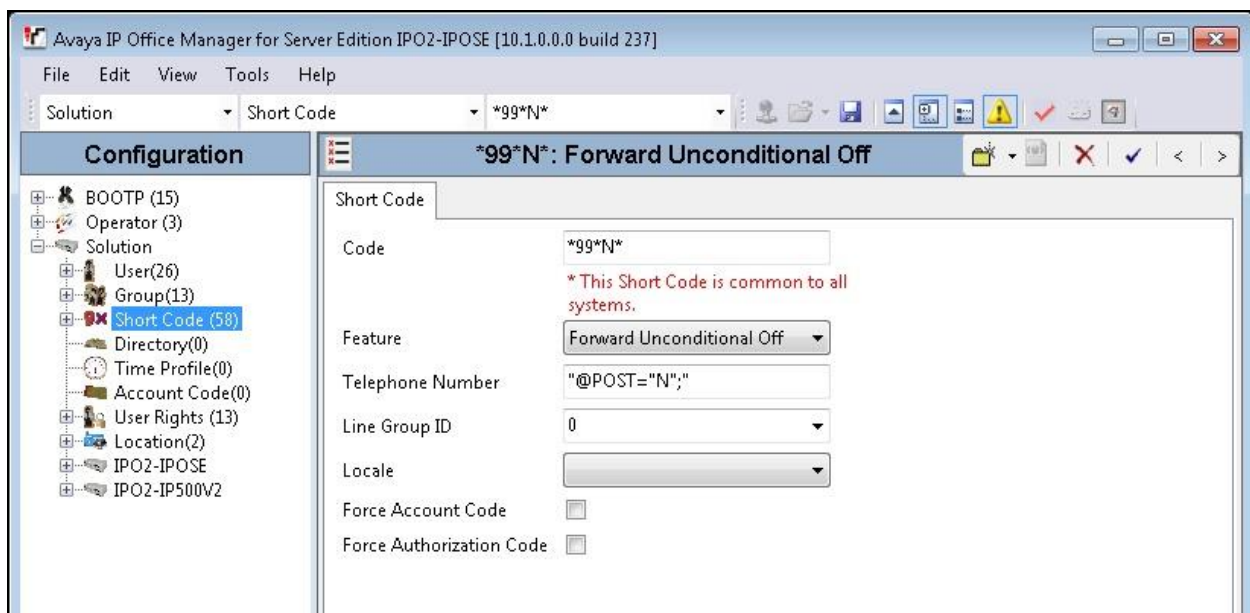
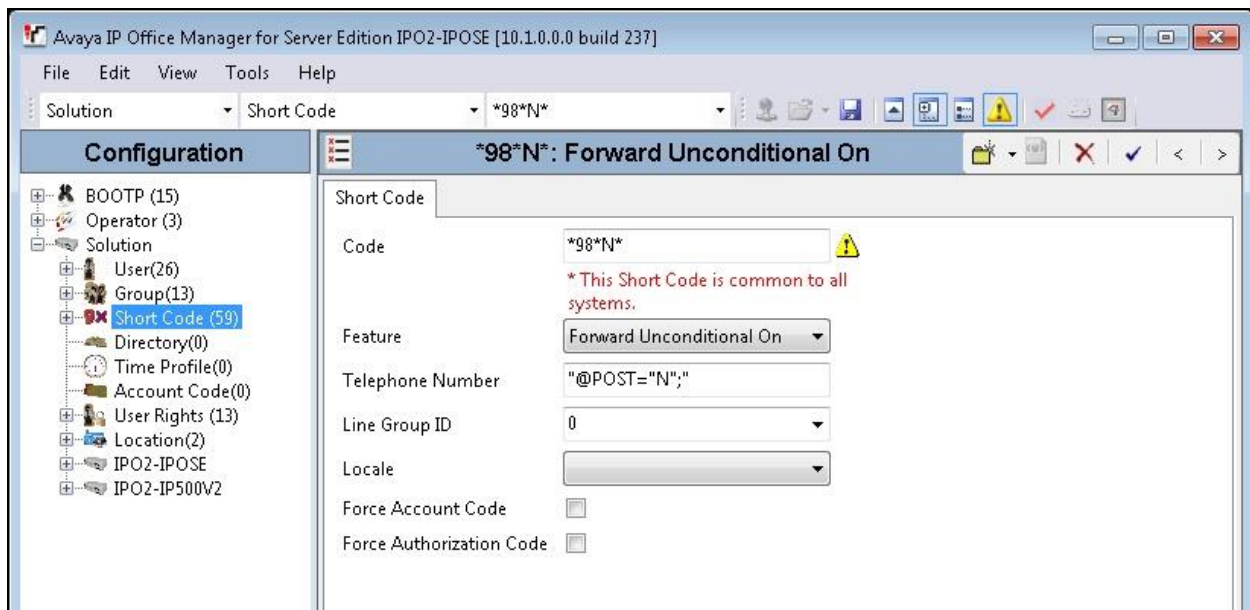
Create two common short codes for activation and deactivation of MWI, as shown below. These two short codes are dialed by DuVoice via the virtual SIP users. Note that the default DuVoice configuration expects the MWI short codes to use the exact code values shown below. Should different values be used, then the new values need to be configured in the DuVoice setup wizard in **Section 6.1**.



## 5.8.2. Call Forwarding

Create two common short codes for activation and deactivation of Call Forwarding Unconditional, and one common short code for use as Forward Number.

The two Call Forwarding Unconditional short codes are dialed by DuVoice via the virtual SIP users, when a guest manually places a do not disturb request/cancel to the front desk or staff. Note that DuVoice requires the two short codes to use the same exact values shown below.



Create one common short code for use as Forward Number for all users that use DuVoice for voicemail. For **Telephone Number**, “28000” is the hospitality group extension from **Section 5.6**. Note that DuVoice requires this short code to contain the exact format shown below.

The screenshot shows the Avaya IP Office Manager interface. The left sidebar displays a tree view with the following items: BOOTP (15), Operator (3), Solution, User(26), Group(13), Short Code (59) (highlighted), Directory(0), Time Profile(0), Account Code(0), User Rights (13), Location(2), IPO2-IPOSE, and IPO2-IP500V2. The main window is titled '\*67;: Dial Extension'. The 'Short Code' field is set to '\*67;'. A red note states: '\* This Short Code is common to all systems.' The 'Feature' is set to 'Dial Extension'. The 'Telephone Number' is set to '28000|>>E'. The 'Line Group ID' is set to '0'. The 'Locale' is set to an empty dropdown. The 'Force Account Code' and 'Force Authorization Code' checkboxes are both unchecked.

### 5.8.3. Operator

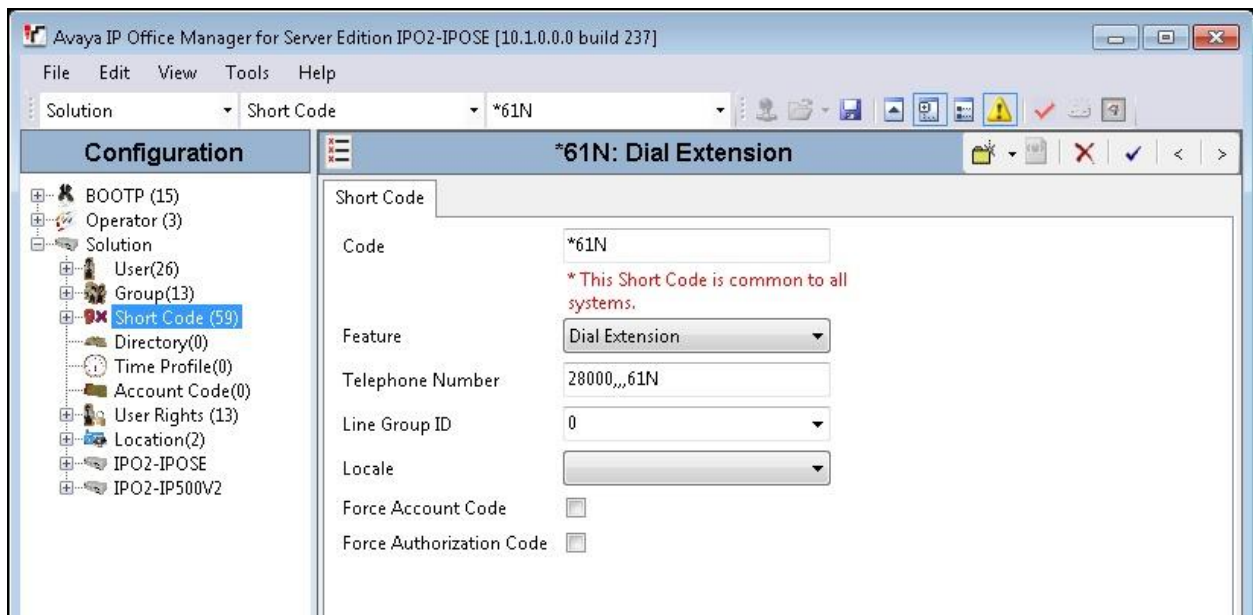
DuVoice recommends a common short code be created for reaching the operator. This is used by DuVoice for coverage of failed wakeup calls and for supporting the operator option for calls to the hospitality group. In the compliance testing, **Telephone Number** was set to the extension of a pre-existing group consisting of front desk and staff users on both IP Office systems. Note that the code value can vary, and “0” is used in this case.

The screenshot shows the Avaya IP Office Manager interface. The left sidebar displays a tree view with the following items: BOOTP (15), Operator (3), Solution, User(26), Group(13), Short Code (59) (highlighted), Directory(0), Time Profile(0), Account Code(0), User Rights (13), Location(2), IPO2-IPOSE, and IPO2-IP500V2. The main window is titled '0: Dial Extension'. The 'Short Code' field is set to '0'. A red note states: '\* This Short Code is common to all systems.' The 'Feature' is set to 'Dial Extension'. The 'Telephone Number' is set to '21991'. The 'Line Group ID' is set to '0'. The 'Locale' is set to an empty dropdown. The 'Force Account Code' and 'Force Authorization Code' checkboxes are both unchecked.

#### 5.8.4. Room Maid Status

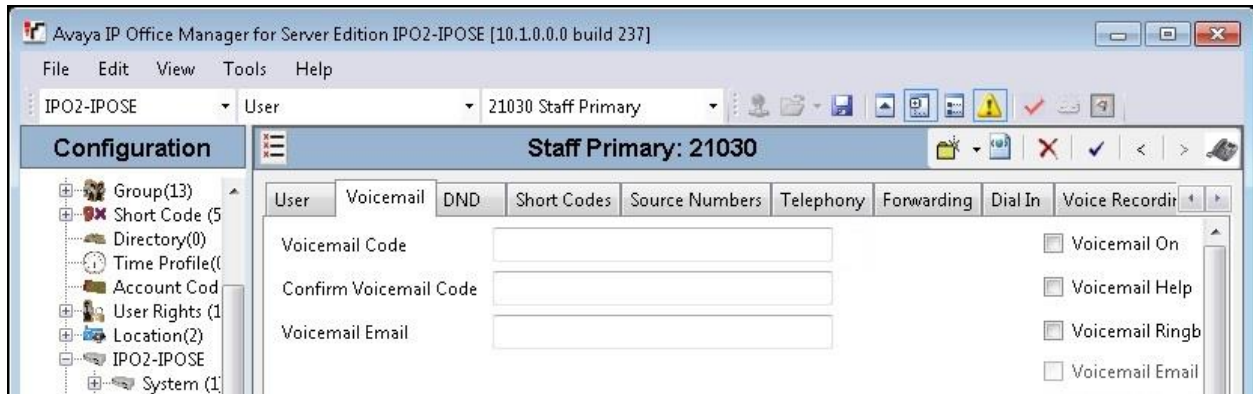
Create a common short code for room maid status. This short code is dialed by maids from the guest rooms, to reach DuVoice for update of room maid status.

For **Telephone Number**, enter “28000,,,61N”, where “28000” is the hospitality group extension from **Section 5.6**. Note that the number of commas in the telephone number may vary. Each comma represents a pause, and three commas were needed in the compliance testing for maid status reporting from all guest rooms on both IP Office systems to be recognized. Note that DuVoice requires this short code to use the exact format shown below, with only variation in the hospitality group extension and the number of commas for pauses.



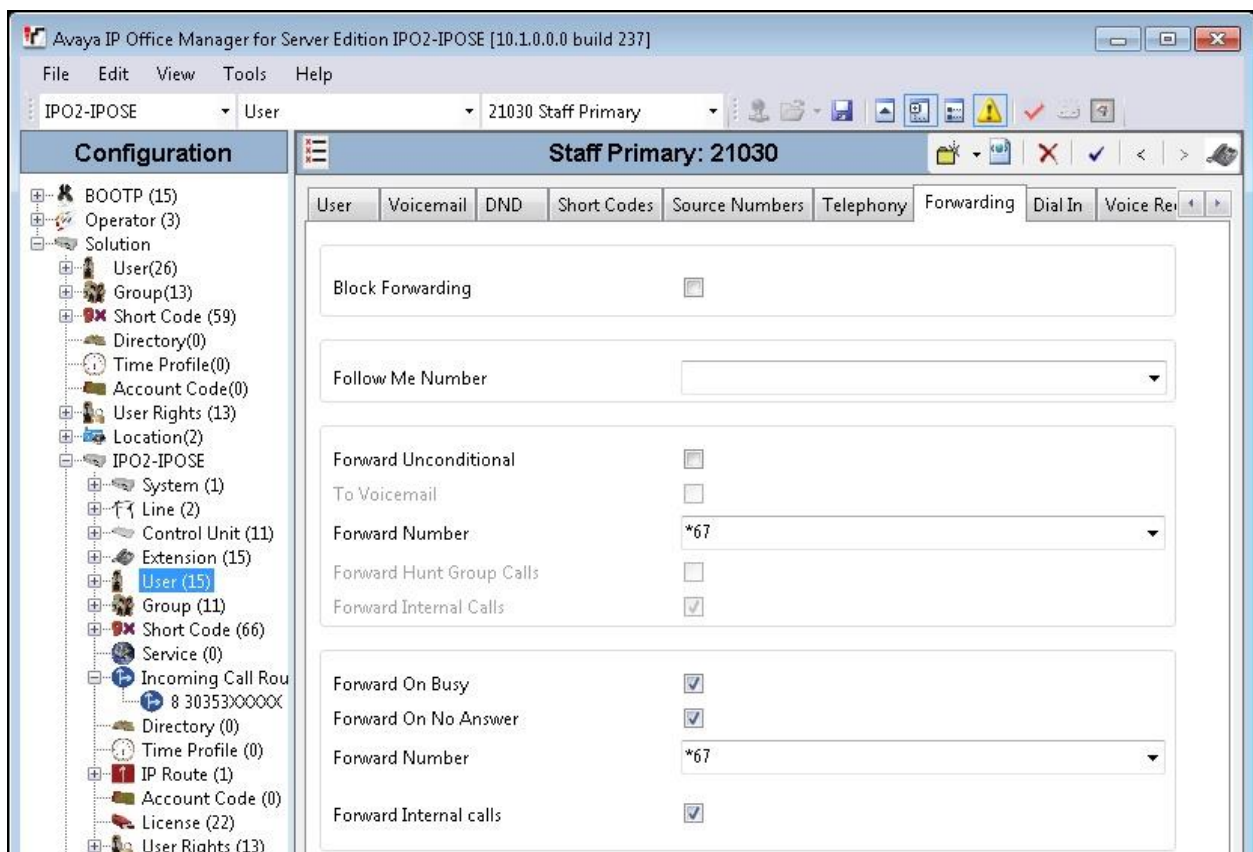
## 5.9. Administer Voicemail Users

From the configuration tree in the left pane, select the first user under the primary IP Office system from **Section 3** that will be using DuVoice for voicemail, in this case “21030”. Select the **Voicemail** tab. Uncheck **Voicemail On**, as shown below.



Select the **Forwarding** tab. Set the **Forward Number** to the Forward Number short code from **Section 5.8.2**. Check all forwarding parameters in the bottom sub-section, as shown below.

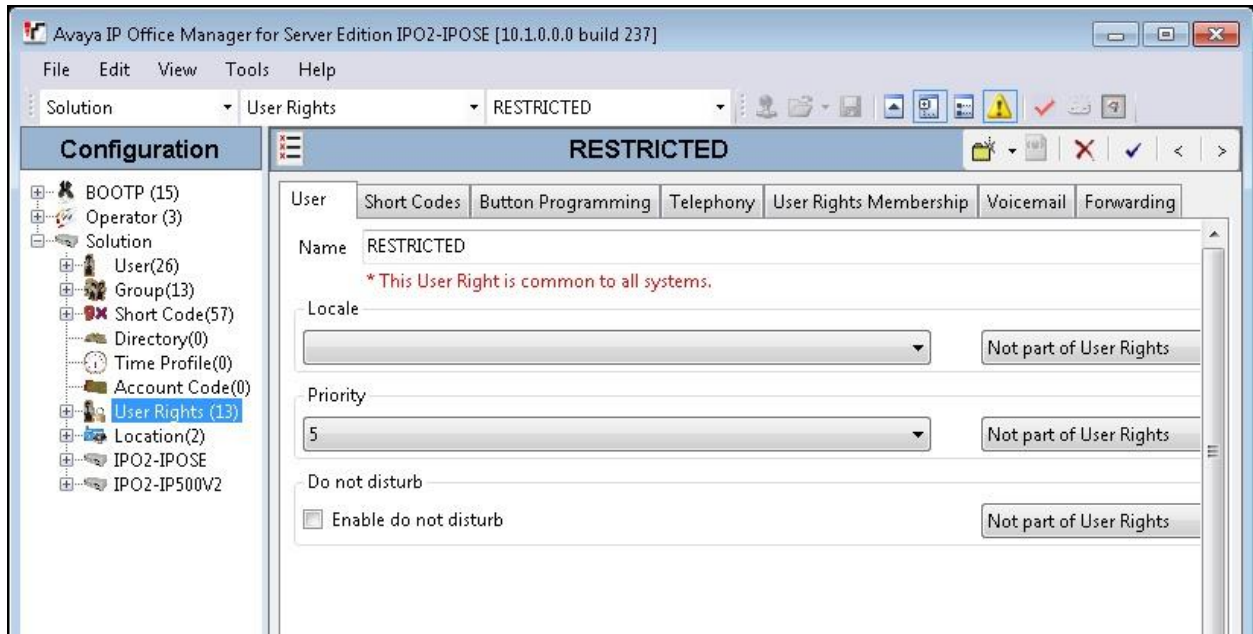
Repeat this section for all users on both IP Office systems that will use DuVoice for voicemail.



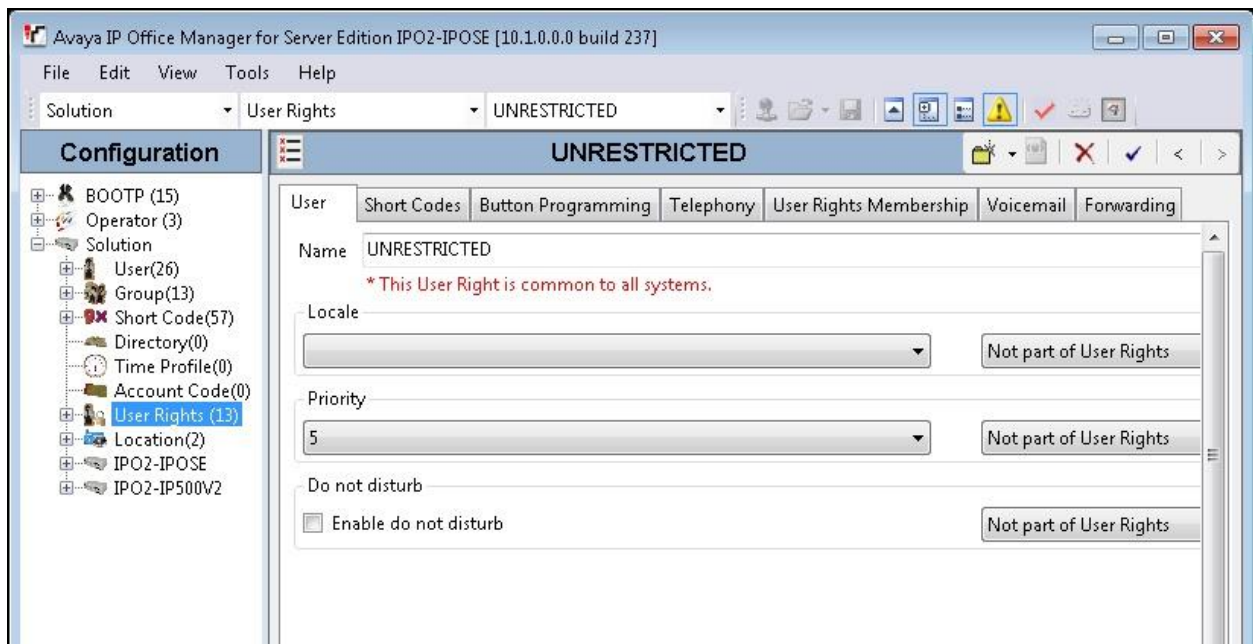


## 5.10. Administer User Rights

From the configuration tree in the left pane, right-click on **Solution** → **User Rights** and select **New** to create a new common user rights template. For **Name**, enter “UNRESTRICTED” as user rights for use by guest rooms in the check-in state. Configure the remaining parameters as desired.



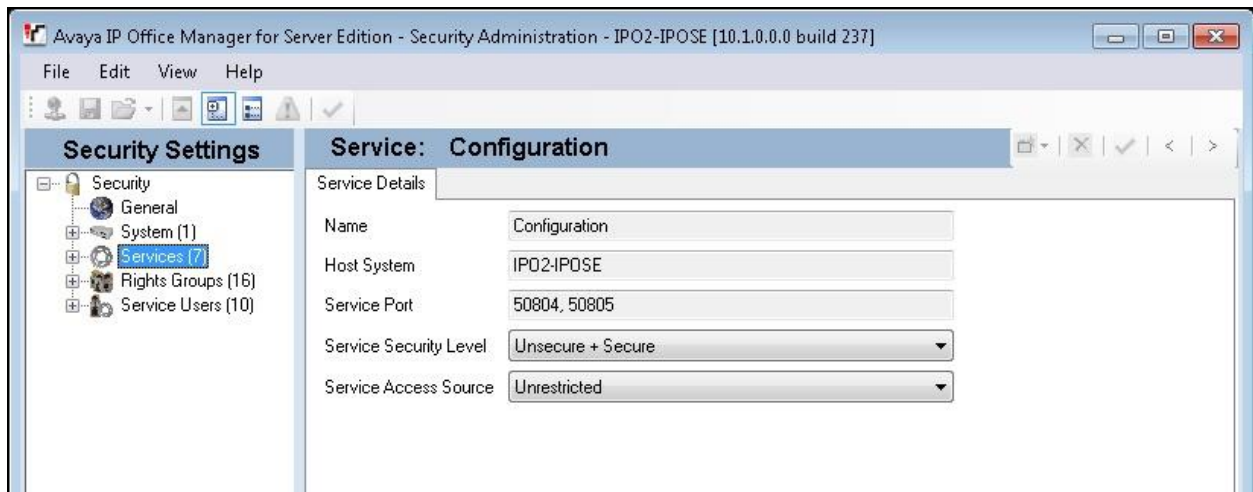
Repeat this section to create another common user rights template for guest rooms in the check-out state. In the compliance testing, two user rights templates with names of “UNRESTRICTED” and “RESTRICTED” were created.



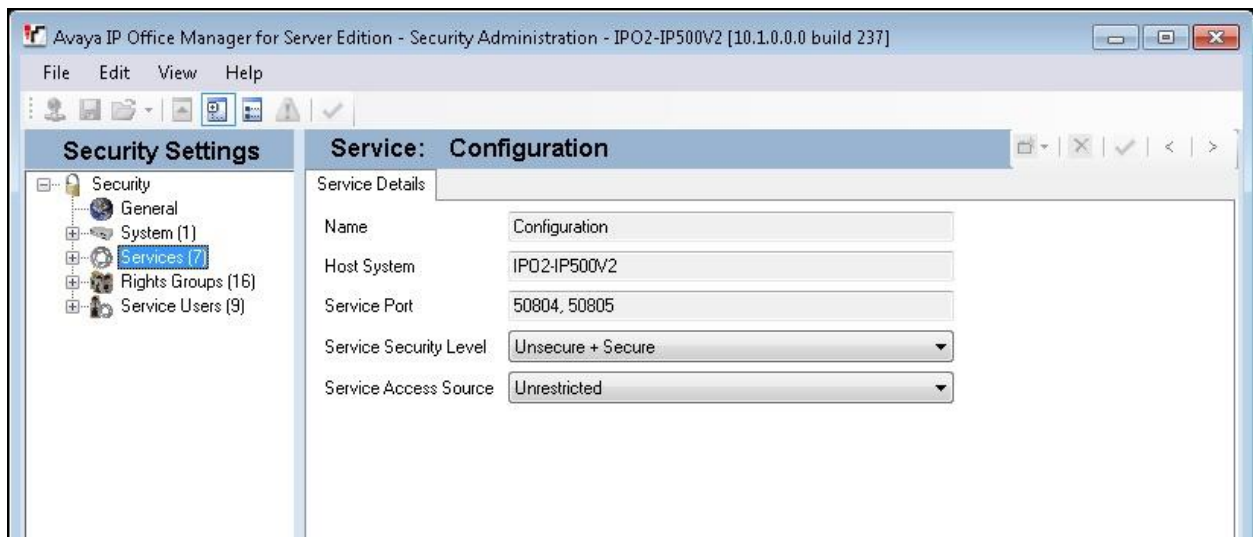
## 5.11. Administer Security Service

From the configuration tree in the left pane, select the primary IP Office system, in this case “IPO2-IPOSE” (not shown), followed by **File → Advanced → Security Settings** from the top menu.

The **Avaya IP Office Manager for Server Edition – Security Administration** screen is displayed. Select **Security → Services** in the left pane to display the **Service: Configuration** screen in the right pane. For **Service Security Level**, select “Unsecure + Secure” as shown below. The additional “Secure” level is needed for the Configuration Web Service interface.



Repeat the procedures in this section to set the security level for the expansion IP Office system.



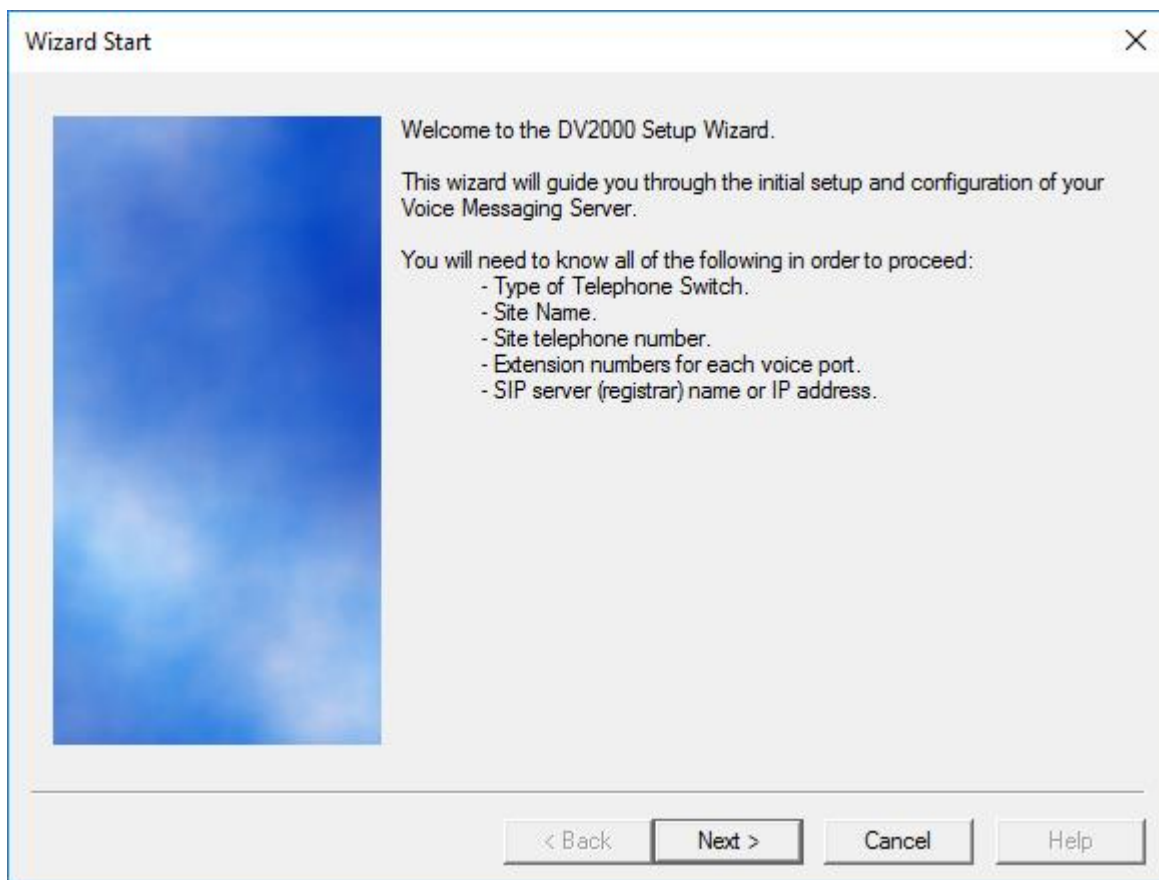
## 6. Configure DuVoice

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

- Administer setup wizard
- Administer profile configuration
- Administer SIP configuration
- Administer hospitality configuration
- Administer mailboxes

### 6.1. Administer Setup Wizard

From the DuVoice server, select **Start → DV2000 → System Configuration**. The **Wizard Start** screen is displayed upon initial access.





The **Site Information** screen is displayed next. Enter descriptive values for the required fields.



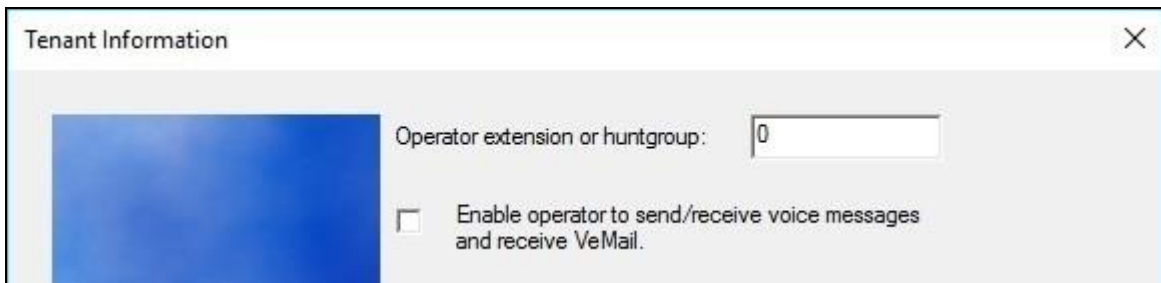
Site Information

Site Name is required. All other entries should be filled

Site Name:

Site Telephone	908-953-2103
Dealer Name	Avaya
Dealer Telephone	908-953-2103
Address	350 Mount Kemble Ave
Address	
State	NJ
Zip	07960
City	Morristown
Country	USA

The **Tenant Information** screen is displayed next. Enter the operator short code value from **Section 5.8.3**.

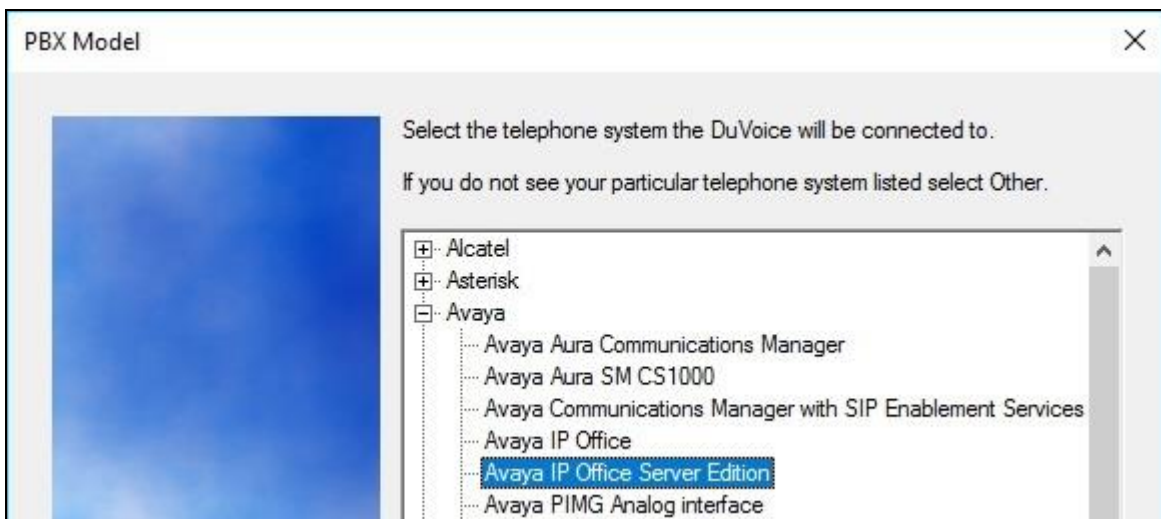


Tenant Information

Operator extension or huntgroup:

☐ Enable operator to send/receive voice messages and receive VeMail.

The **PBX Model** screen is displayed next. Expand and select **Avaya → Avaya IP Office Server Edition**, as shown below.



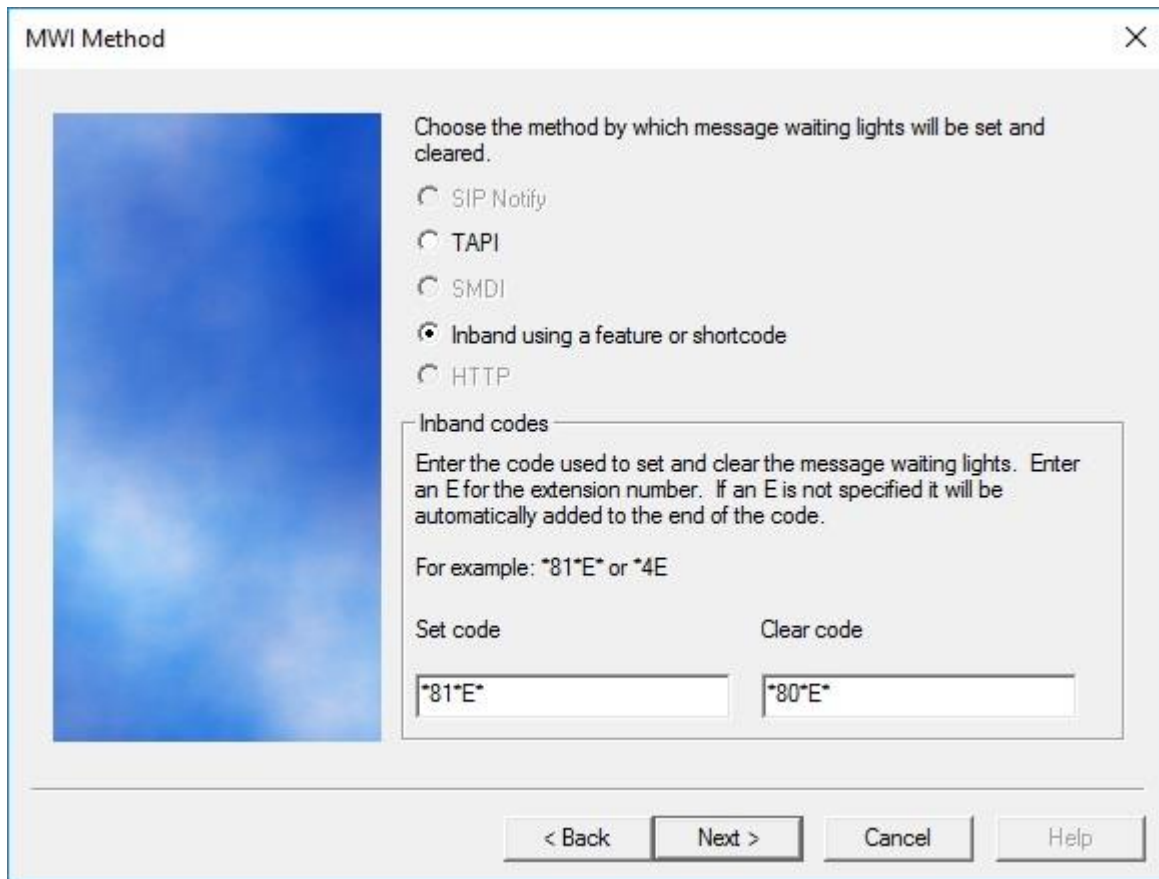
PBX Model

Select the telephone system the DuVoice will be connected to.

If you do not see your particular telephone system listed select Other.

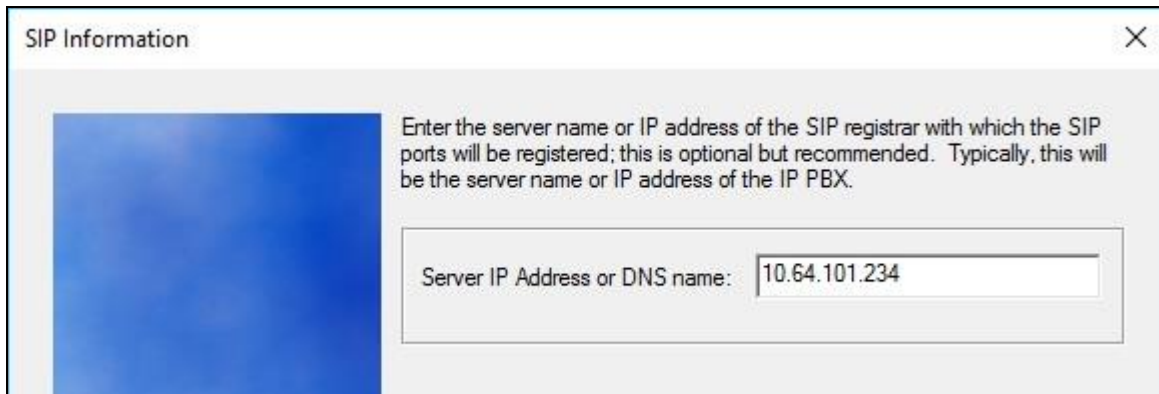
- [-] Alcatel
- [-] Asterisk
- [+] Avaya
  - Avaya Aura Communications Manager
  - Avaya Aura SM CS1000
  - Avaya Communications Manager with SIP Enablement Services
  - Avaya IP Office
  - Avaya IP Office Server Edition**
  - Avaya PIMG Analog interface

The **MWI Method** screen is displayed. Select **Inband using a feature or shortcode**, and retain the default values in the remaining fields.



The MWI Method window is titled "MWI Method" and contains a close button (X) in the top right corner. On the left is a blue decorative image. The main text reads: "Choose the method by which message waiting lights will be set and cleared." Below this are five radio button options: "SIP Notify", "TAPI", "SMDI", "Inband using a feature or shortcode" (which is selected), and "HTTP". Below the radio buttons is a section titled "Inband codes" with the instruction: "Enter the code used to set and clear the message waiting lights. Enter an E for the extension number. If an E is not specified it will be automatically added to the end of the code. For example: \*81\*E\* or \*4E". This section contains two input fields: "Set code" with the value "\*81\*E\*" and "Clear code" with the value "\*80\*E\*". At the bottom are four buttons: "< Back", "Next >", "Cancel", and "Help".

The **SIP Information** screen is displayed next. For **Server IP Address or DNS name**, enter the IP address or DNS name of the primary IP Office system.



The SIP Information window is titled "SIP Information" and contains a close button (X) in the top right corner. On the left is a blue decorative image. The main text reads: "Enter the server name or IP address of the SIP registrar with which the SIP ports will be registered; this is optional but recommended. Typically, this will be the server name or IP address of the IP PBX." Below this is a single input field labeled "Server IP Address or DNS name:" containing the value "10.64.101.234".

The **Voice Ports** screen is displayed. For **Voicemail Huntgroup**, enter the hospitality group extension from **Section 5.6**.

When the configured SIP extensions from **Section 5.4** are sequential, check **Auto increment extension numbers based on line 1**, and enter the first SIP extension as shown below. Otherwise, enter the relevant SIP extensions next to each port number.

In the compliance testing, four DuVoice ports were configured to correspond to the four SIP extensions created on IP Office.

Click **Next**, followed by **Finish** in the subsequent screen (not shown) to complete the wizard.

Voice Ports

This system will be configured for 4 voice mail ports. If you know the extension of each port enter it in the space provided by clicking the ports extension field below. Entering the extension numbers is required for some integrations and will help with resolving integration issues.

If You do not know the extensions leave them blank, they can be entered later in System Configuration.

Voicemail Huntgroup: 28000

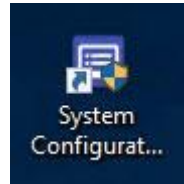
☒ Auto increment extension numbers based on line 1.

Number	Extension
Port 1	28881
Port 2	
Port 3	
Port 4	

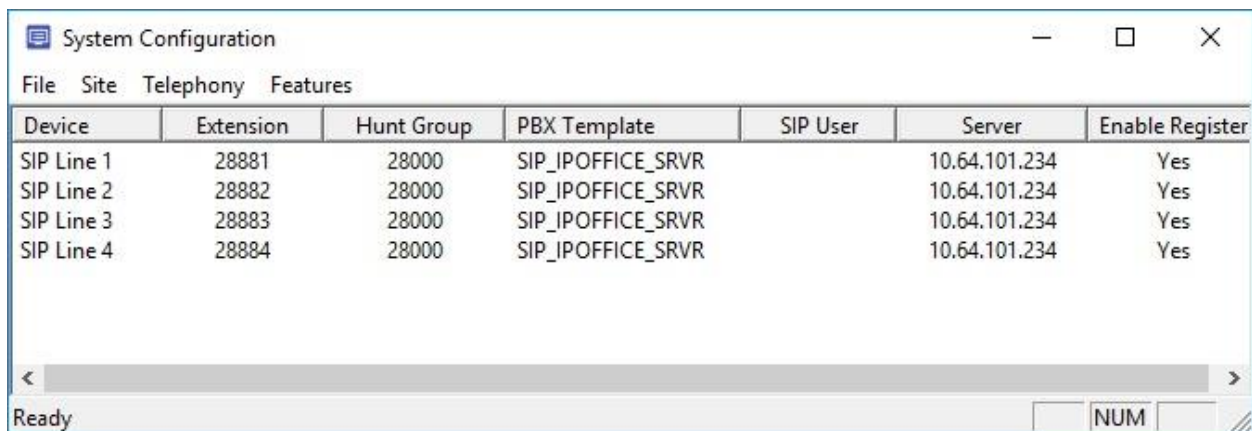
< Back Next > Cancel Help

## 6.2. Administer Profile Configuration

From the DuVoice server, double-click on the **System Configuration** icon shown below, which was created as part of server installation.



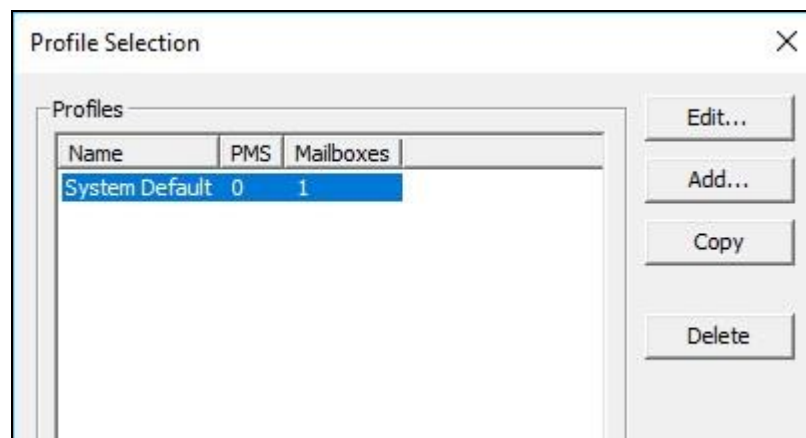
The **System Configuration** screen below is displayed. Select **Site** → **Profiles** from the top menu.

A screenshot of the "System Configuration" application window. It has a menu bar with "File", "Site", "Telephony", and "Features". Below the menu bar is a table with the following data:

Device	Extension	Hunt Group	PBX Template	SIP User	Server	Enable Register
SIP Line 1	28881	28000	SIP_IPOFFICE_SRVR		10.64.101.234	Yes
SIP Line 2	28882	28000	SIP_IPOFFICE_SRVR		10.64.101.234	Yes
SIP Line 3	28883	28000	SIP_IPOFFICE_SRVR		10.64.101.234	Yes
SIP Line 4	28884	28000	SIP_IPOFFICE_SRVR		10.64.101.234	Yes

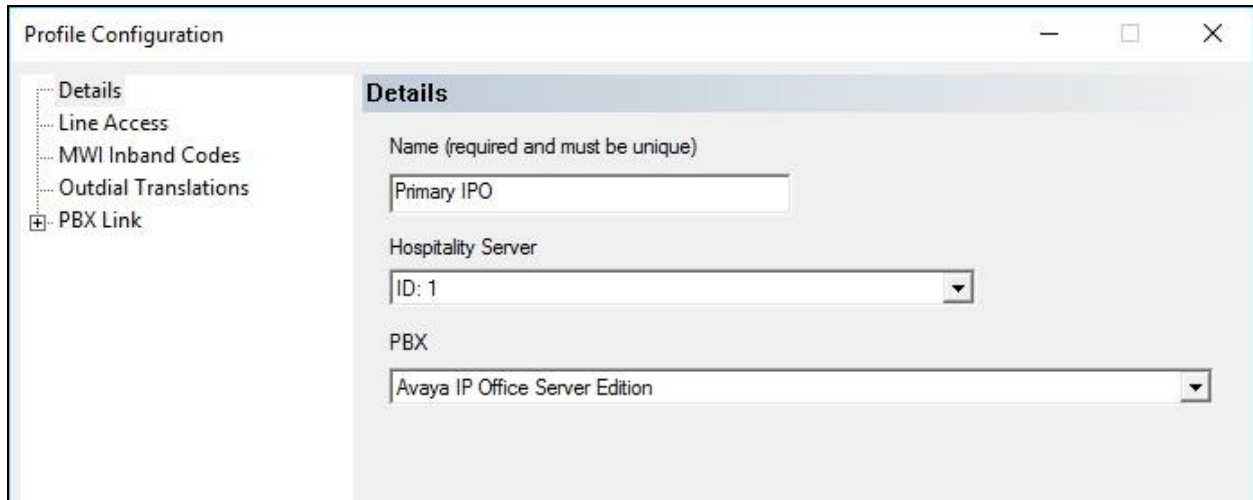
The window also has a status bar at the bottom that says "Ready" and a "NUM" button.

The **Profile Selection** screen is displayed next. Select the default entry and click **Edit**.



The **Profile Configuration** screen below is displayed. Select **Details** in the left pane. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive profile name for the primary IP Office system.
- **Hospitality Server:** The applicable pre-configured hospitality server.
- **PBX:** “Avaya IP Office Server Edition”

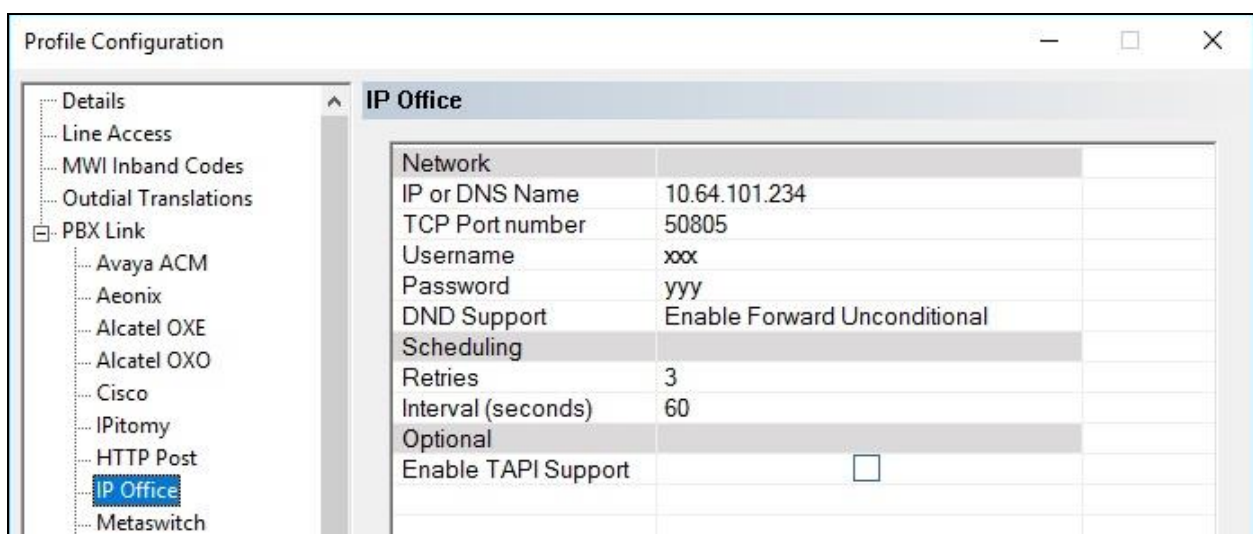


The screenshot shows the 'Profile Configuration' window with the 'Details' tab selected in the left pane. The main area displays the following fields:

- Name (required and must be unique):** A text box containing 'Primary IPO'.
- Hospitality Server:** A dropdown menu showing 'ID: 1'.
- PBX:** A dropdown menu showing 'Avaya IP Office Server Edition'.

Expand **PBX Link** in the left pane, and select **IP Office**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **IP or DNS Name:** IP address of the primary IP Office system.
- **Username:** The default administrator credentials for the primary IP Office system.
- **Password:** The default administrator credentials for the primary IP Office system.
- **DND Support:** “Enable Forward Unconditional”



The screenshot shows the 'Profile Configuration' window with the 'PBX Link' expanded in the left pane and 'IP Office' selected. The main area displays the following fields:

Network	
IP or DNS Name	10.64.101.234
TCP Port number	50805
Username	xxx
Password	yyy
DND Support	Enable Forward Unconditional
Scheduling	
Retries	3
Interval (seconds)	60
Optional	
Enable TAPI Support	<input type="checkbox"/>

Repeat this section to add a new profile for the expansion IP Office system, as shown below.

The screenshot shows the 'Profile Configuration' window with the 'Details' tab selected. The left sidebar lists configuration categories: Details, Line Access, MWI Inband Codes, Outdial Translations, and PBX Link. The main area contains the following fields:

- Name (required and must be unique):** Expansion IPO
- Hospitality Server:** ID: 1
- PBX:** Avaya IP Office Server Edition

The screenshot shows the 'Profile Configuration' window with the 'IP Office' tab selected. The left sidebar lists configuration categories: Details, Line Access, MWI Inband Codes, Outdial Translations, and PBX Link. Under 'PBX Link', several options are listed, with 'IP Office' highlighted. The main area displays the following configuration details:

Network	
IP or DNS Name	192.168.200.234
TCP Port number	50805
Username	xxx
Password	yyy
DND Support	Enable Forward Unconditional
Scheduling	
Retries	3
Interval (seconds)	60
Optional	
Enable TAPI Support	<input type="checkbox"/>



### 6.3. Administer SIP Configuration

The **System Configuration** screen below is displayed again. Select **Telephony → SIP Integration** from the top menu.

System Configuration						
File Site Telephony Features						
Device	Extension	Hunt Group	PBX Template	SIP User	Server	Enable Register
SIP Line 1	28881	28000	SIP_IPOFFICE_SRVR		10.64.101.234	Yes
SIP Line 2	28882	28000	SIP_IPOFFICE_SRVR		10.64.101.234	Yes
SIP Line 3	28883	28000	SIP_IPOFFICE_SRVR		10.64.101.234	Yes
SIP Line 4	28884	28000	SIP_IPOFFICE_SRVR		10.64.101.234	Yes

The **SIP Configuration** screen is displayed. Select **PBX Settings** from the left pane. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **PBX IP or DNS:** IP address of the primary IP Office system.
- **Domain Name:** The IP Office domain name from **Section 5.3**.
- **Realm:** “ipoffice”
- **Register using:** Select **Domain Name**.

PBX Settings

Lines

MWI / DMG Routing

PBX Settings

Local address

10 . 64 . 101 . 205

PBX IP or DNS

10.64.101.234

Port

5060

Domain Name

dr220.com

Register expire time

160

seconds

Realm

ipoffice

Optional Backup Servers

☐ Default to primary on restart.

Order	PBX IP or DNS Name
1	
2	
3	
4	
5	

Transport Protocol

☒ UDP

☐ TCP

Register using

☐ PBX address

☐ Local IP address

☒ Domain Name

Select **Lines** from the left pane. For **Password**, enter the corresponding SIP user login code from **Section 5.5** for each SIP line entry, as shown below.

SIP Configuration

PBX Settings  
Lines  
MWI / DMG Routing

Lines

Line	<input checked="" type="checkbox"/>	Register	Extension	Account	Password	User Ag
1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	28881	28881	123456	DuVoice
2	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	28882	28882	234567	DuVoice
3	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	28883	28883	345678	DuVoice
4	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	28884	28884	456789	DuVoice

< >

Auto Number Extension

Auto Number Account

Match Passwords

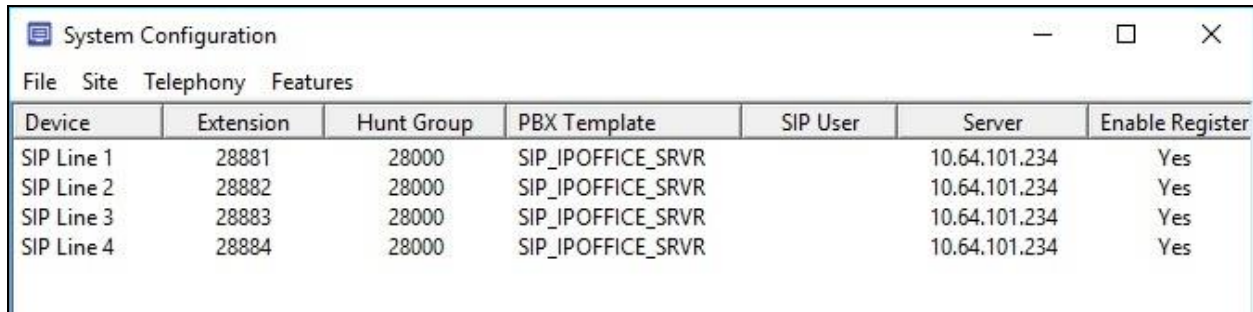
OK

Cancel



## 6.4. Administer Hospitality Configuration

The **System Configuration** screen below is displayed again. Select **Features** → **Hospitality** from the top menu.

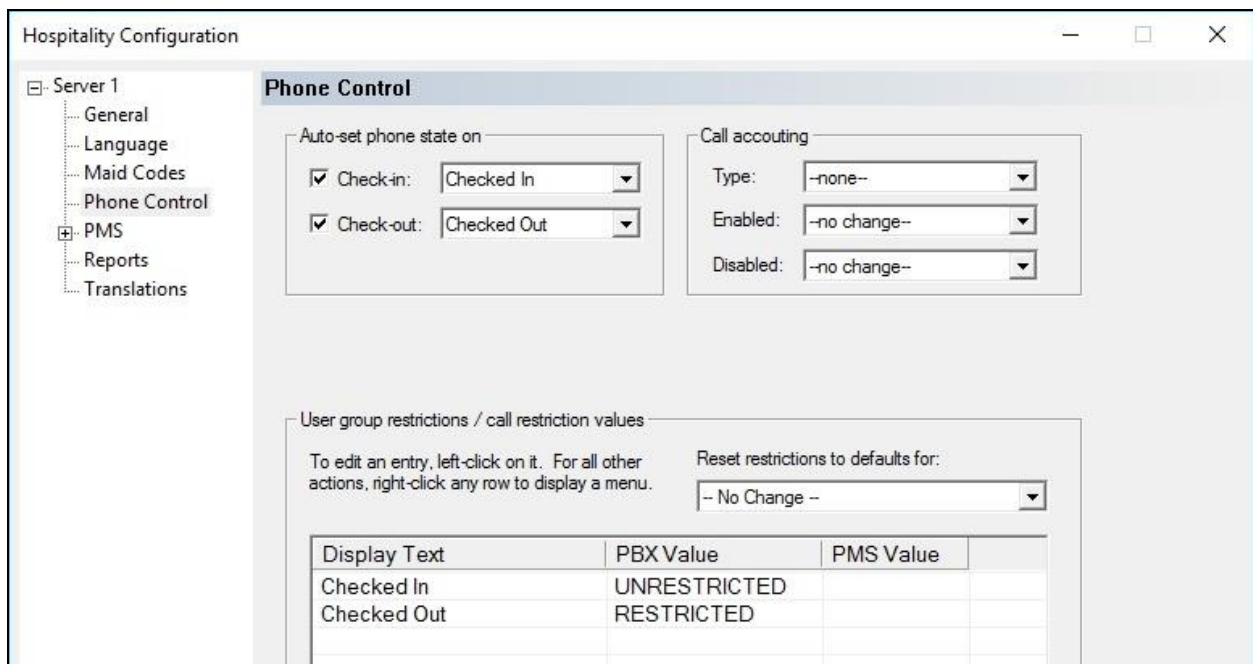


Device	Extension	Hunt Group	PBX Template	SIP User	Server	Enable Register
SIP Line 1	28881	28000	SIP_IPOFFICE_SRVR		10.64.101.234	Yes
SIP Line 2	28882	28000	SIP_IPOFFICE_SRVR		10.64.101.234	Yes
SIP Line 3	28883	28000	SIP_IPOFFICE_SRVR		10.64.101.234	Yes
SIP Line 4	28884	28000	SIP_IPOFFICE_SRVR		10.64.101.234	Yes

The **Hospitality Configuration** screen is displayed next. Select **Server 1** → **Phone Control** from the left pane.

In the **User group restrictions / call restriction values** sub-section in the right pane, create two entries for check-in and check-out. Use descriptive values for **Display Text**, and use the user rights template values from **Section 5.10** for **PBX Value**.

After creating the above entries, in the **Auto-set phone state on** sub-section in the right pane, check and select the relevant values for **Check-in** and **Check-out**, as shown below.



**Auto-set phone state on**

☒ Check-in: Checked In

☒ Check-out: Checked Out

**Call accounting**

Type: -none--

Enabled: -no change--

Disabled: -no change--

**User group restrictions / call restriction values**

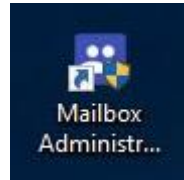
To edit an entry, left-click on it. For all other actions, right-click any row to display a menu.

Reset restrictions to defaults for: -- No Change --

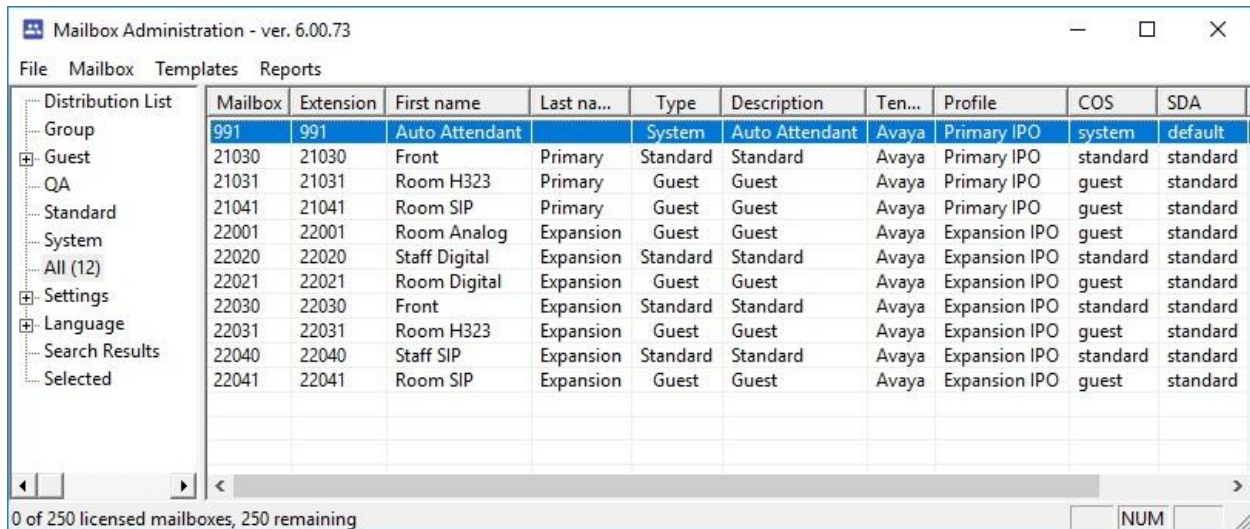
Display Text	PBX Value	PMS Value
Checked In	UNRESTRICTED	
Checked Out	RESTRICTED	

## 6.5. Administer Mailboxes

From the DuVoice server, double-click on the **Mailbox Administration** icon shown below, which was created as part of server installation.



The **Mailbox Administration** screen is displayed next. Follow reference [2] to create a mailbox for each user on IP Office that will be using DuVoice for voicemail. In the compliance testing, an entry was created for each IP Office user from **Section 3**.

A screenshot of the "Mailbox Administration - ver. 6.00.73" application window. The window has a menu bar with "File", "Mailbox", "Templates", and "Reports". On the left is a "Distribution List" tree with nodes: "Group", "Guest", "QA", "Standard", "System", "All (12)", "Settings", "Language", "Search Results", and "Selected". The main area is a table with columns: "Mailbox", "Extension", "First name", "Last na...", "Type", "Description", "Ten...", "Profile", "COS", and "SDA". The table contains 11 rows of data. At the bottom, it says "0 of 250 licensed mailboxes, 250 remaining" and has a "NUM" button.

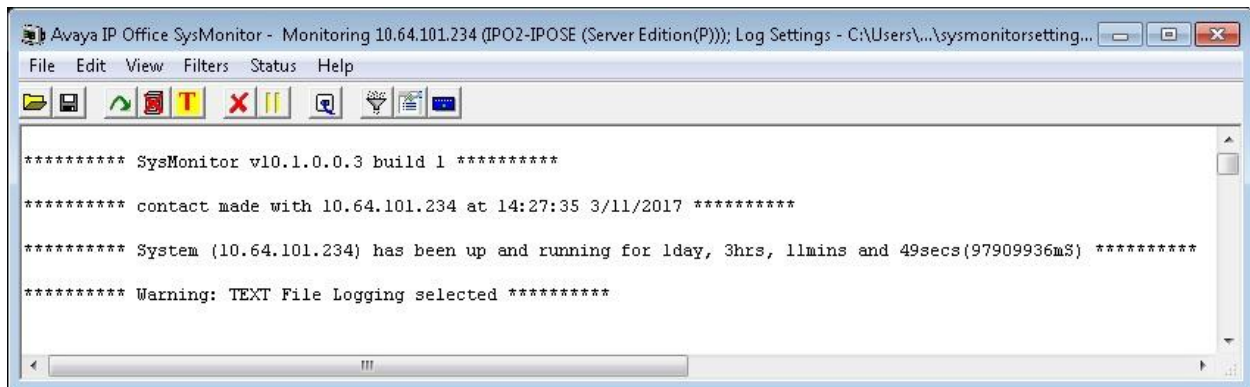
Mailbox	Extension	First name	Last na...	Type	Description	Ten...	Profile	COS	SDA
991	991	Auto Attendant		System	Auto Attendant	Avaya	Primary IPO	system	default
21030	21030	Front	Primary	Standard	Standard	Avaya	Primary IPO	standard	standard
21031	21031	Room H323	Primary	Guest	Guest	Avaya	Primary IPO	guest	standard
21041	21041	Room SIP	Primary	Guest	Guest	Avaya	Primary IPO	guest	standard
22001	22001	Room Analog	Expansion	Guest	Guest	Avaya	Expansion IPO	guest	standard
22020	22020	Staff Digital	Expansion	Standard	Standard	Avaya	Expansion IPO	standard	standard
22021	22021	Room Digital	Expansion	Guest	Guest	Avaya	Expansion IPO	guest	standard
22030	22030	Front	Expansion	Standard	Standard	Avaya	Expansion IPO	standard	standard
22031	22031	Room H323	Expansion	Guest	Guest	Avaya	Expansion IPO	guest	standard
22040	22040	Staff SIP	Expansion	Standard	Standard	Avaya	Expansion IPO	standard	standard
22041	22041	Room SIP	Expansion	Guest	Guest	Avaya	Expansion IPO	guest	standard

## 7. Verification Steps

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

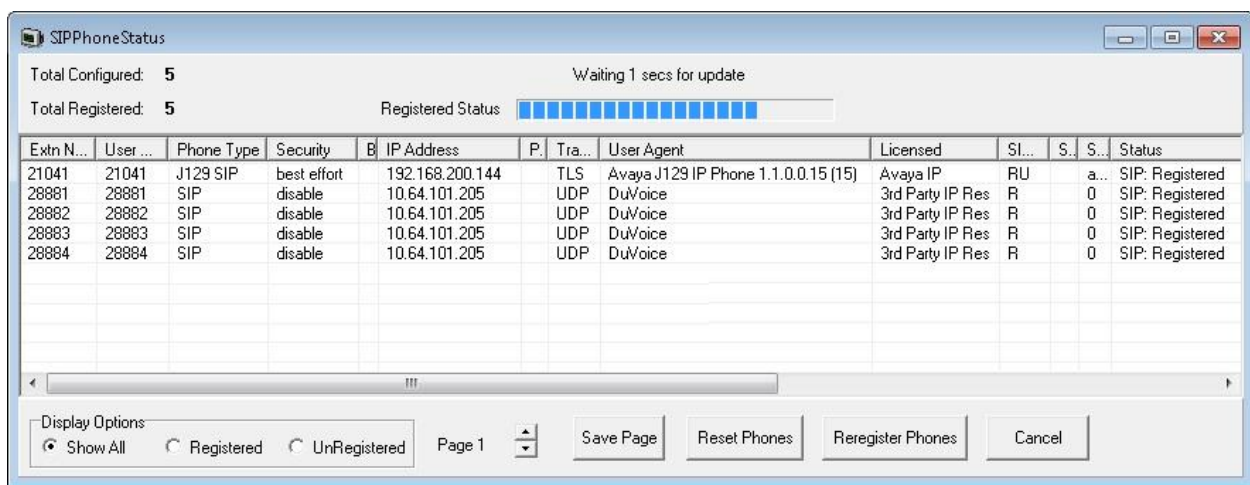
### 7.1. Verify SIP User Integration

From a PC running the IP Office Monitor application, select **Start → All Programs → IP Office → Monitor** to launch the application, and connect to the primary IP Office system. The **Avaya IP Office SysMonitor** screen is displayed. Select **Status → SIP Phone Status** from the top menu.



The **SIPPhoneStatus** screen is displayed. Verify that there is an entry for each SIP extension from **Section 5.4**, that the **User Agent** is “DuVoice”, and that the **Status** is “SIP: Registered”, as shown below.

Place an incoming call from the PSTN to the hospitality group. Verify that the calling party hears the greeting announcement from DuVoice. Enter the extension of a guest user on the primary IP Office system, and verify that the call is transferred to the guest user. Repeat with a call transfer to a guest user on the expansion IP Office system.



## 7.2. Verify Configuration Web Service Integration

From the DuVoice server, select **Start → DV2000 Testing → Hospitality Tester** to launch the tool. The **Hospitality Tester** screen is displayed.

For **Rooms**, select a desired guest extension on the primary IP Office system, in this case “21031”. Enter the desired **First name** and **Last name**, and click **Check in**. Verify that the pertinent guest user name and user rights template are updated properly on the primary IP Office system.

Repeat with a check-in request for a guest user on the expansion IP Office system.

The screenshot shows the 'Hospitality Tester' web application. At the top, there's a 'Rooms' section with a dropdown menu showing '21031' and a 'Reload' button. To the right are 'Refresh' and 'Update All' buttons. Below this is the 'Current Settings' section, which includes checkboxes for 'Checked in/out', 'VIP', and 'DND'. It also has input fields for 'Extension' (21031), 'Profile' (Primary IPO), 'Guest ID', 'Maid status' (0:Dirty), 'Language', 'Text count' (0), and 'Phone COS'. The bottom section contains fields for 'First name' (Room H323), 'Last name' (Primary), 'Title', and a 'VIP' checkbox. Below these are dropdowns for 'Language' (Default), 'Maid status', and 'Phone COS'. At the bottom, there are several action buttons: 'Update', 'Check in', 'Add text count', 'Set DND', 'Wakeup call', 'Check out', 'Subtract text count', 'Clear DND', and 'Move guest'.

## 8. Conclusion

These Application Notes describe the configuration steps required for DuVoice 6.0 to successfully interoperate with Avaya IP Office Server Edition 10.1. 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 10.1, June 2017, available at <http://support.avaya.com>.
2. *DV2000 System Reference Guide*, Version 6.0.61, available at <http://support.duvoice.com/vs6/manual/home>.

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