Avaya Solution & Interoperability Test Lab

Applications Notes for Avaya IP Office 6.1 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office with the AT&T IP Toll Free service. Avaya IP Office solution was tested with AT&T IP Toll Free service supports using **MIS/PNT** transport.

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks for business customers.

AT&T is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program.

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1. Introduction

These Application Notes describe the steps for configuring Avaya IP Office Preferred Edition with the AT&T IP Toll Free service. Avaya IP Office solution was tested with AT&T IP Toll Free service supports using **MIS/PNT** transport.

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks for business customers.

2. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise with IP Office, Avaya phones and fax machines (Ventafax application).
- A laboratory version of the AT&T IP Toll Free service, to which the simulated enterprise was connected via MIS/PNT transport.

The main test objectives were to verify the following features and functionality:

- Inbound AT&T IP Toll Free service calls to IP Office hunt groups/telephones.
- Call and two-way talk path establishment between PSTN and IP Office phones via the AT&T Toll Free service.
- Basic supplementary telephony features such as hold, resume, transfer, and conference.
- G.729 and G.711 codecs.
- T.38 fax calls between IP Office and the AT&T IP Toll Free service/PSTN G3 and SG3 fax endpoints.
- DTMF tone transmission using RFC 2833 between IP Office and the AT&T IP Toll Free service/PSTN automated access systems.
- Inbound AT&T IP Toll Free service calls to IP Office that are directly routed to stations, and unanswered, can be covered to Voicemail Pro.
- Long duration calls.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound and outbound call flows (see **Section 3.2** for examples) between IP Office and the AT&T IP Toll Free service.

The compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the AT&T network. Calls were made from the PSTN across the AT&T network (see **Section 3.2** for sample call flows). The following features were tested as part of this effort:

- SIP trunking
- T.38 Fax
- Passing of DTMF events and their recognition by navigating automated voice menus
- PBX and AT&T IP Toll Free service features such as hold, resume, conference and transfer
- Legacy Transfer Connect
- Alternate Destination Routing

2.2. Known Limitations/Test Results

1. G.711 faxing is not supported between IP Office and the AT&T IP Toll Free service. T.38 faxing is supported, as is Group 3 and Super Group 3 fax.
2. AT&T IP Transfer connect option of the AT&T IP Toll Free service was not verified with Avaya IP Office 6.1 and hence not supported.
3. Shuffling is not supported for SIP trunks in Avaya IP Office 6.1.
4. G.726 codec is not supported Avaya IP Office 6.1.

The test objectives stated in **Section 2** with limitations as noted in this section were verified.

2.3. Support

AT&T customers may obtain support for the AT&T IP Toll Free service by calling (888) 325-5555.

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. In the United States, (866)GO-AVAYA (866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers (provided on http://support.avaya.com) to directly access specific support and consultation services based upon their Avaya support agreements.

3. Reference Configuration

The reference configuration used in these Application Notes is shown in the figure below and consists of several components:

- IP Office provides the voice communications services for a particular enterprise site. In the reference configuration, IP Office runs on an IP 500. This solution is extensible to other Avaya IP Office hardware too.
- Avaya "desk" phones are represented with Avaya 1616, 4625 and 9630 IP Telephones running H.323 software, Avaya 1416 digital phone, Avaya 6211 Analog Telephone, Avaya SIP Phones (1140E and 1230) and PC based IP Office Softphone.
- Voicemail Pro provides the voice messaging capabilities in the reference configuration and its provisioning is beyond the scope of this document.
- Inbound calls from PSTN were sent from AT&T IP Toll Free service to IP Office. IP Office terminated the call to the appropriate agent/phone or fax extension. Signaling is between IP Office public interface and the AT&T Border Element.
- Enterprise sites may have additional or alternate routes to PSTN using analog or digital TDM trunks. However these trunks were not used in this reference configuration.

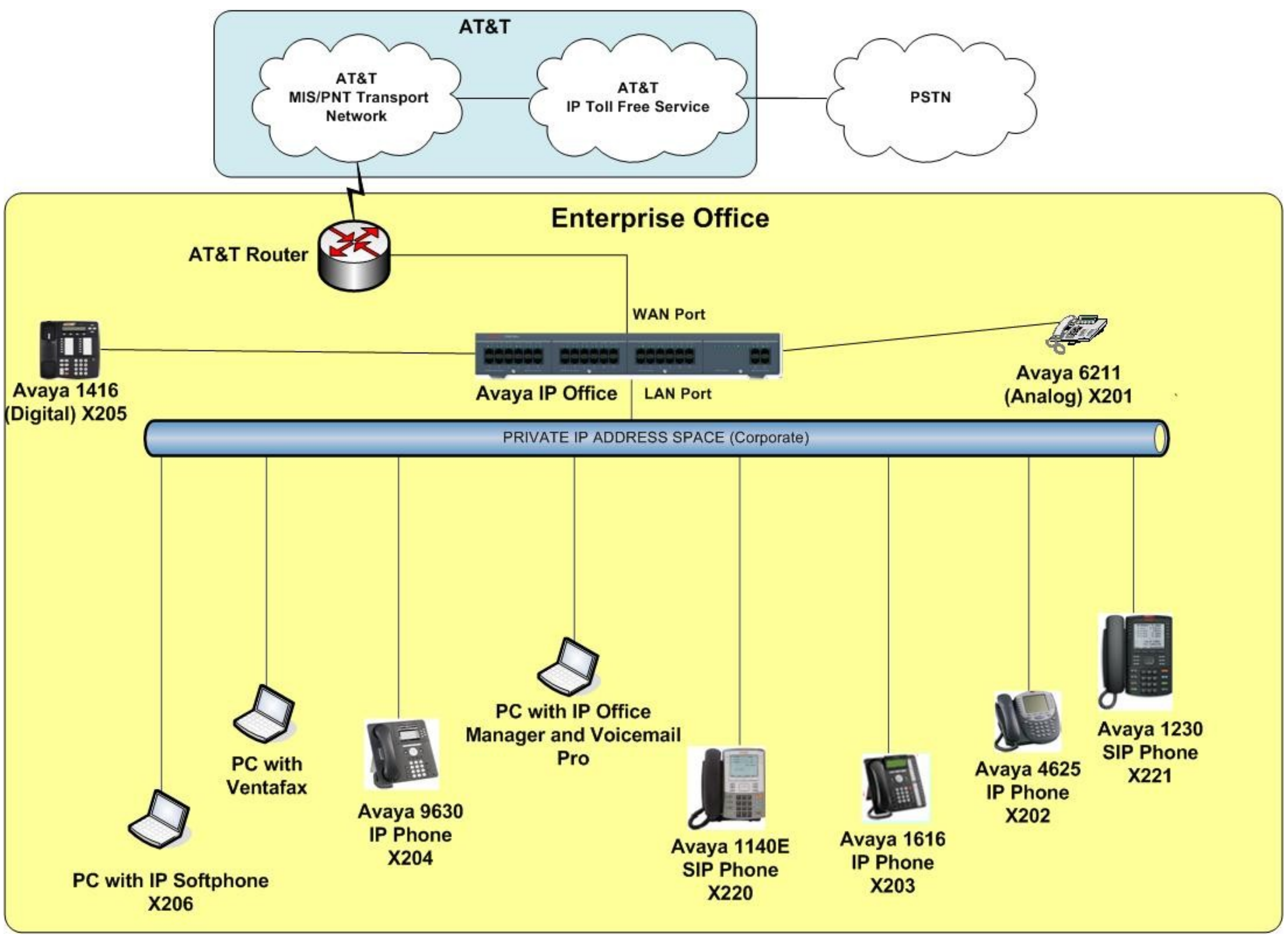


Figure 1: Reference configuration

3.1. Illustrative Configuration Information

The specific values listed in the table below and in subsequent sections are used in the reference configuration described in these Application Notes, and are **for illustrative purposes only**. Customers must obtain and use the specific values for their own specific configurations.

Note - The AT&T IP Toll Free service border element IP addresses shown in this document are examples. AT&T Customer Care will provide the actual IP addresses as part of the AT&T IP Toll Free service provisioning process.

Component	Illustrative Value in these Application Notes
Avaya IP Office	
Public IP Address	192.168.62.56
Private IP Address	10.80.130.56
Avaya IP Office Extensions	201 = Analog 202,203,204=H323 205=Digital 206=Softphone 220,221= SIP phones
AT&T IP Toll Free Service	
Border Element IP Address	135.242.225.200
Digits passed in SIP-URI Request	000001017 to 000001021

Table 1: Illustrative Values Used in these Application Notes

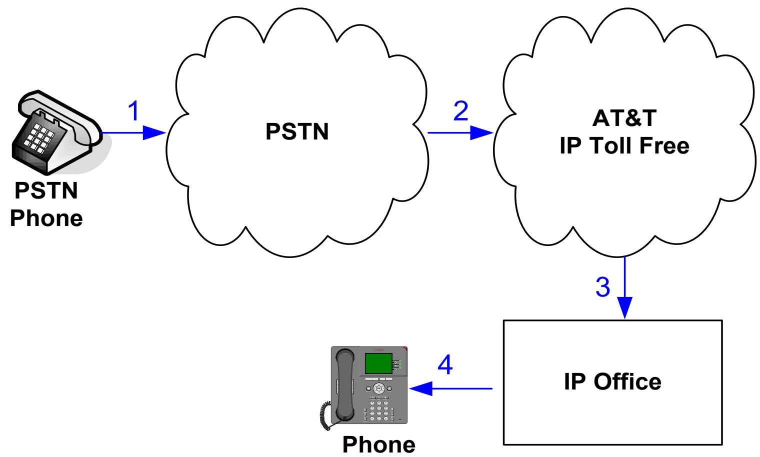
3.2. Call Flows

To understand how inbound and outbound AT&T IP Toll Free service calls are handled by IP Office, two basic call flows are described in this section.

3.2.1. Inbound

The first call scenario illustrated in the figure below is an inbound AT&T IP Toll Free service call that arrive on IP Office, which in turn routes the call to a hunt group, phone or a fax.

1. A PSTN phone originates a call to an AT&T IP Toll Free service number.
2. The PSTN routes the call to the AT&T IP Toll Free service network.
3. The AT&T IP Toll Free service routes the call to IP Office.
4. Depending on the called number, IP Office routes the call to
   - A hunt group, which in turn, routes the call to an agent
   - Directly to an agent or a phone/fax extension.

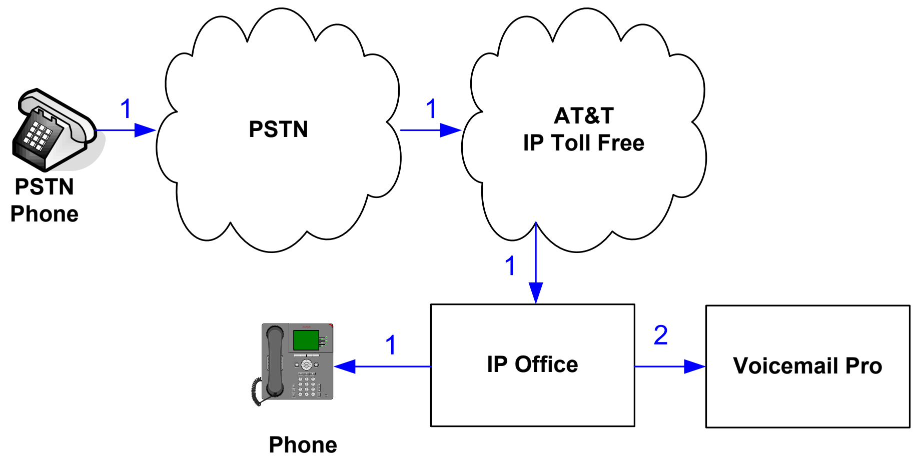


Inbound - AT&T IP Toll Free

3.2.2. Coverage to Voicemail

The call scenario illustrated in the figure below is an inbound call that is covered to voicemail. In this scenario, the voicemail system is Voicemail Pro software installed on a PC.

1. Same as the first call scenario in **Section 3.2.1**.
2. The IP Office phone does not answer the call, and the call covers to the phone's voicemail. IP Office forwards the call to Voicemail Pro.



Coverage to Voicemail

4. Equipment and Software Validated

The following equipment and software was used for the reference configuration described in these Application Notes.

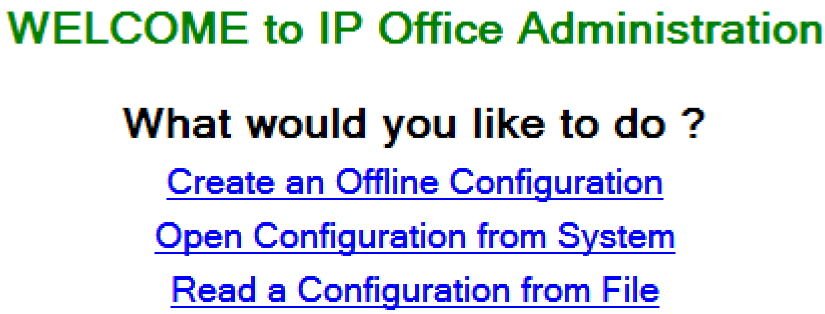
Note – Although Avaya IP Office Preferred Edition was used during this testing; Avaya IP Office Essential and Advanced Editions are also supported.

Component	Version
Avaya IP Office 500	Release 6.1 (6.1.5) (Preferred Edition)
Avaya IP Office Manager	Release 8.1 (8.1.5) (Preferred Edition)
Avaya IP Office Voicemail Pro	Release 6.1.16
Avaya IP Office Voicemail Pro Client	Version 6.1 (16)
Avaya 1616IP-Series Telephones (H.323)	Release 1.3
Avaya 9630 IP Telephone	Avaya one-X® Deskphone Edition H.323 Version S3.11
Avaya 4625SW IP Telephone	a25d01a2_9_1.bin
Avaya IP Office Softphone	Release 3.1.2.17 59616
Avaya 1416 Digital Telephone	-
Avaya 6211 Analog phone	-
Avaya 1140E SIP Telephone	04.00.04.00 (SIP1140)
Avaya 1230 SIP Telephone	04.00.04.00 (SIP1230)
Fax device	Ventafax Home Version 6.2
AT&T IP Toll Free Service using MIS/PNT transport service connections.	VNI 18

Table 2: Equipment and Software Versions

5. Avaya IP Office

IP Office is configured via the IP Office Manager program. For more information on IP Office Manager, consult reference [IPO-MGR]. From the IP Office Manager PC, select **Start** → **Programs** → **IP Office** → **Manager** to launch the Manager application. A screen that includes the following in the center may be displayed:



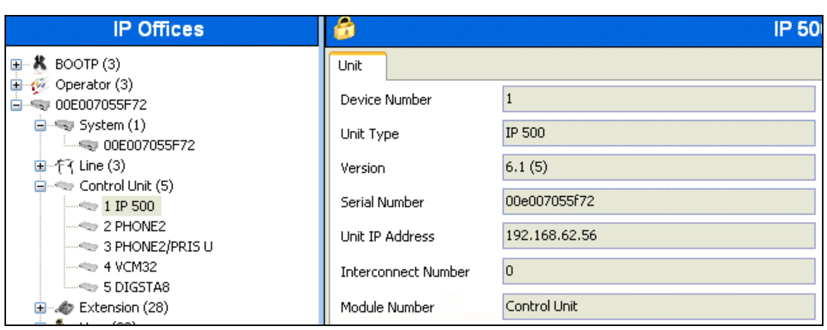
Open the IP Office configuration, either by reading the configuration from the IP Office server, or from file. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, and the Details pane on the right side.

5.1. Physical, Network, and Security Configuration

This section describes attributes of the reference configuration, but is not meant to be prescriptive. Consult reference [IPO-INSTALL] for more information on the topics in this section.

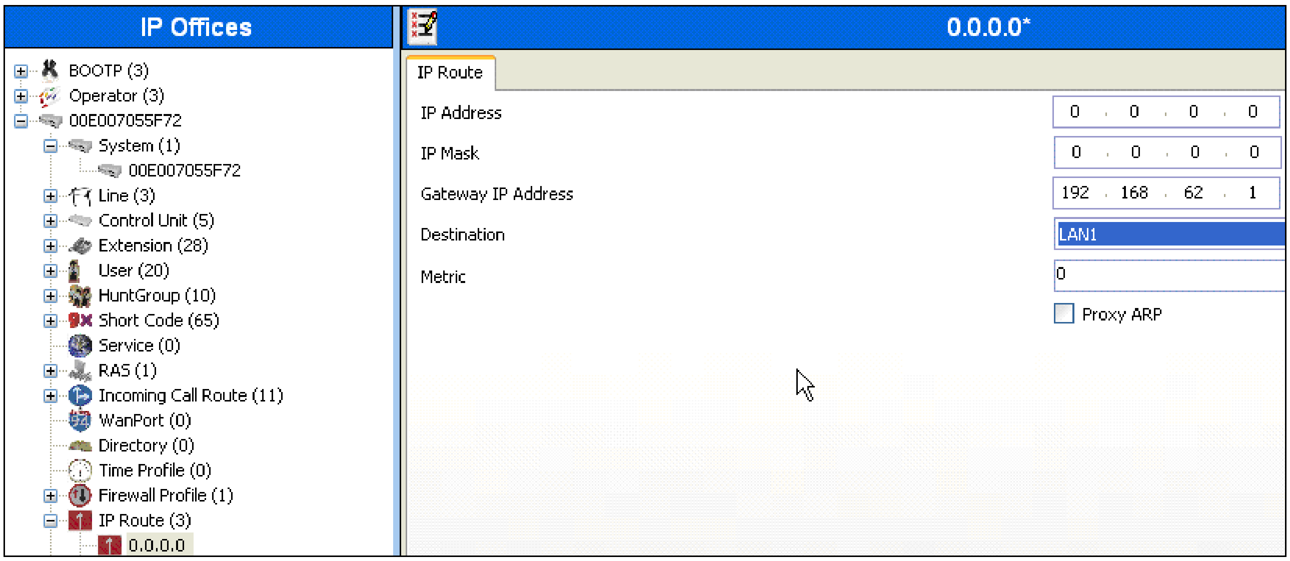
In the reference configuration, the IP Office 500 contains a Phone2 (analog), Phone2/PRIS U (Combo module), VCM32, and a DIGSTA8 (Digital) modules. The VCM32 is a Voice Compression Module supporting VoIP codecs. The Analog module (Phone2) was used in this reference configuration to support analog telephones or fax machines. The Digital module (DIGSTA8) was used to support digital phone extensions.

The following screen shows the modules in the IP Office used in the sample configuration. To access such a screen, select **Control Unit** in the Navigation pane. The modules appear in the Details pane. In the screen below, **IP 500** is selected in the Navigation pane, revealing additional information about the IP 500 in the Details pane.

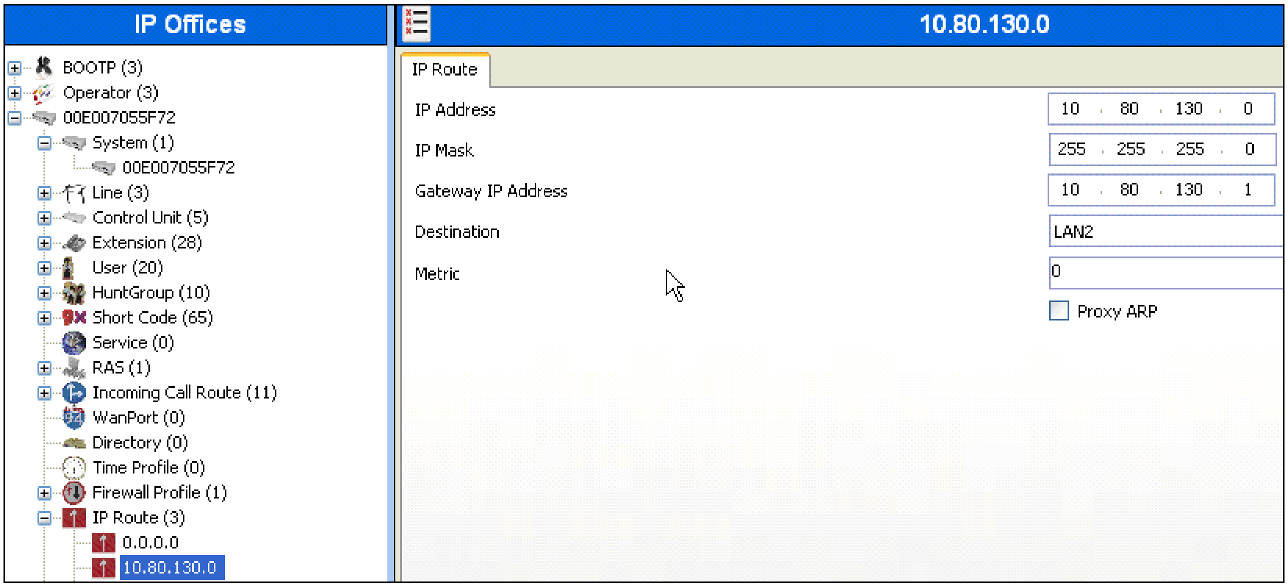


In the sample configuration, the IP Office LAN1 port (labeled as WAN in Figure 1) is physically connected to the public network at the IP Office customer site. The default gateway for this network is 192.168.62.1.

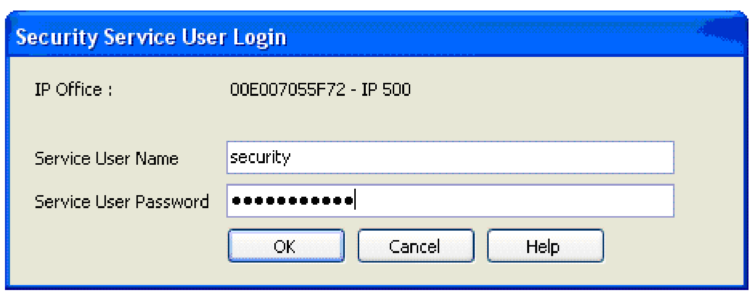
1. To add an IP Route in IP Office, right-click **IP Route** from the Navigation pane, and select **New.** To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the Details pane with the relevant default route using **Destination** LAN1 (Refer **Section 5.3.2**).



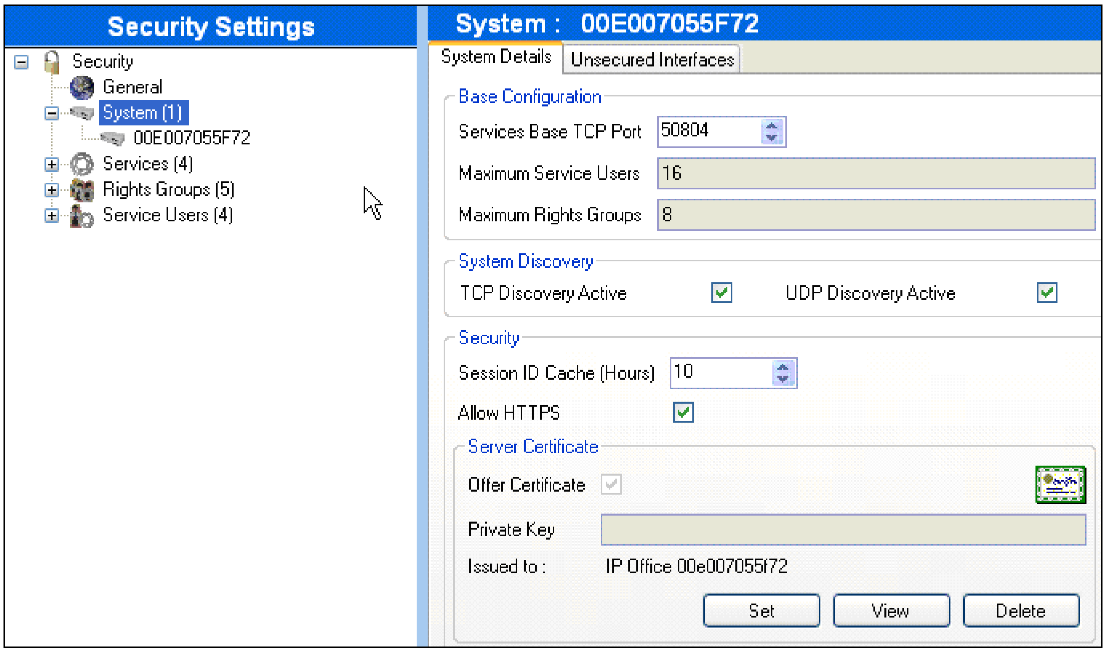
2. Another route was added for the enterprise side (LAN2) as shown in the screen below. All the IP devices were part of this 10.80.130.x network in this reference configuration.



3. For use of Avaya IP Office Softphone, https was enabled in this reference configuration. To check whether https is enabled, navigate to **File → Advanced → Security Settings** and login in with proper credentials in the screen below.



4. After logging in, select **System** from the Navigation pane and the appropriate IP Office system. In the Details pane, select the **System Details** tab. Verify that **Allow HTTPS** is checked. If not, check the box, click **OK**, and heed the on-screen prompts and warnings. Note that this action may be service disrupting.

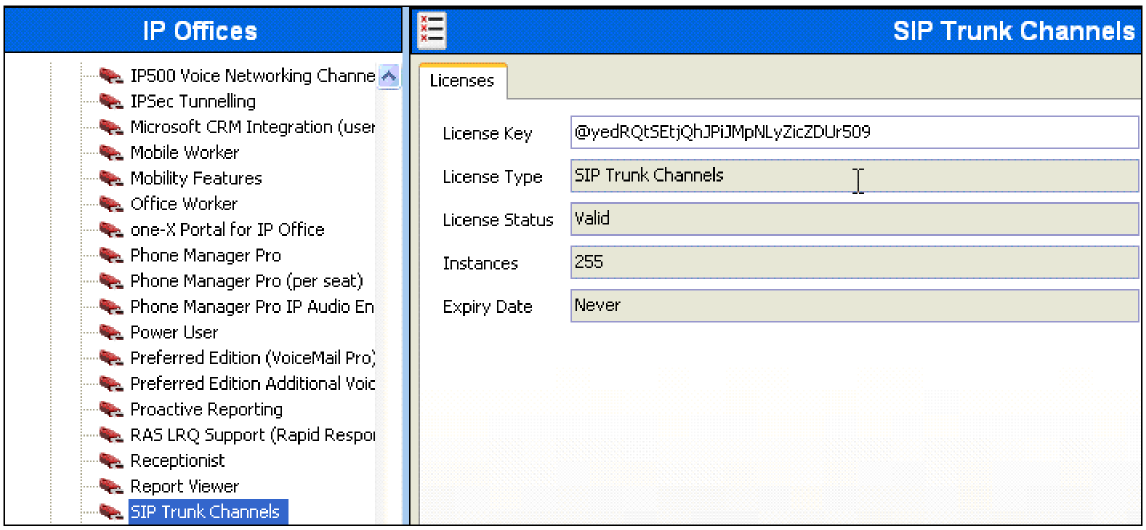


5. When complete, select **File → Configuration** to return to configuration activities.

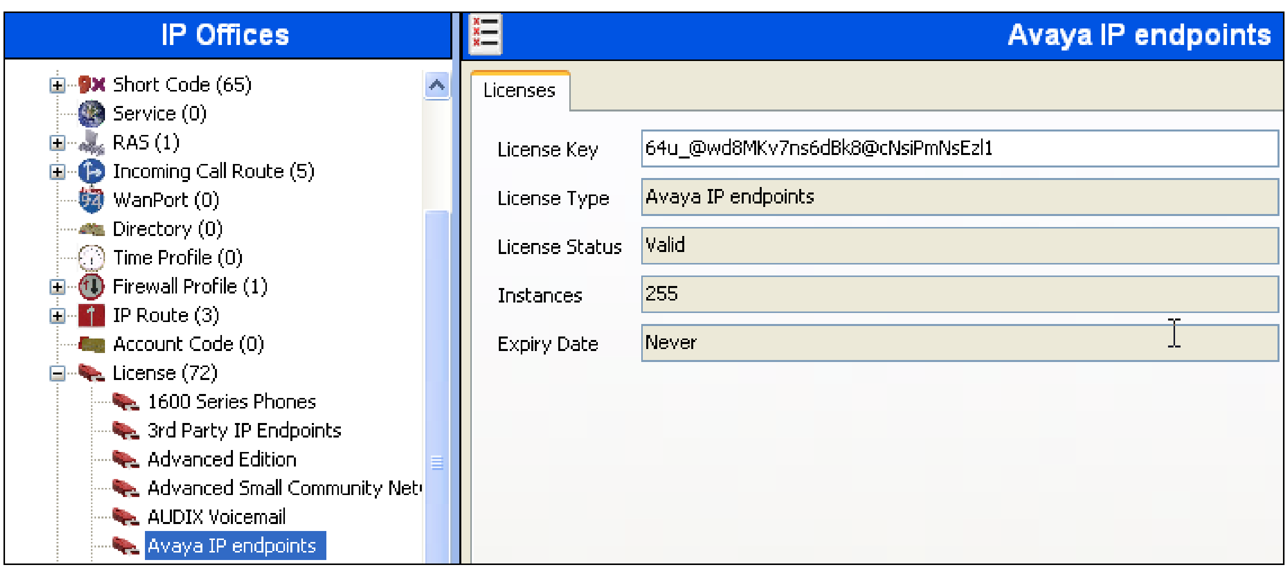
5.2. Licensing

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

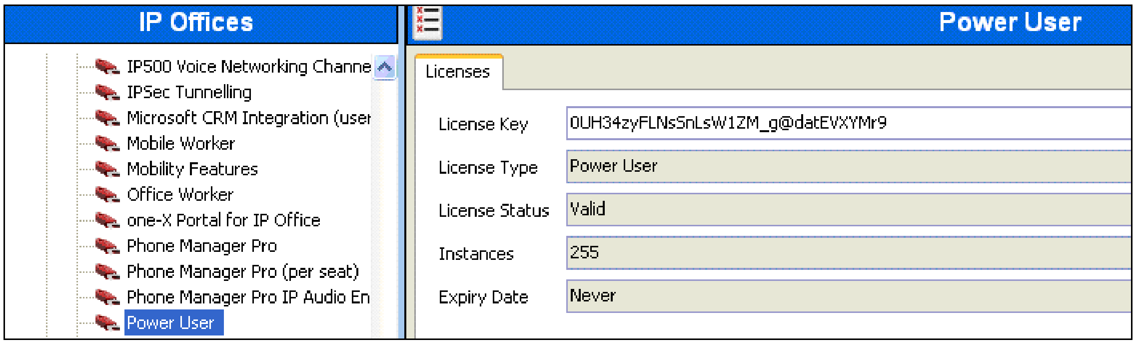
1. To verify that there is a SIP Trunk Channels License with sufficient capacity, click **License** in the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm a valid license with sufficient "Instances" (trunk channels) in the Details pane.



2. If Avaya IP Telephones will be used, verify the Avaya IP endpoints license. Click **License** in the Navigation pane and **Avaya IP endpoints** in the Group pane. Confirm a valid license with sufficient "Instances" in the Details pane.



3. The following screen shows the availability of a valid license for **Power User** features. In the sample configuration, the user with extension **203** will be configured as a **Power User** and will be capable of using the IP Office Softphone.

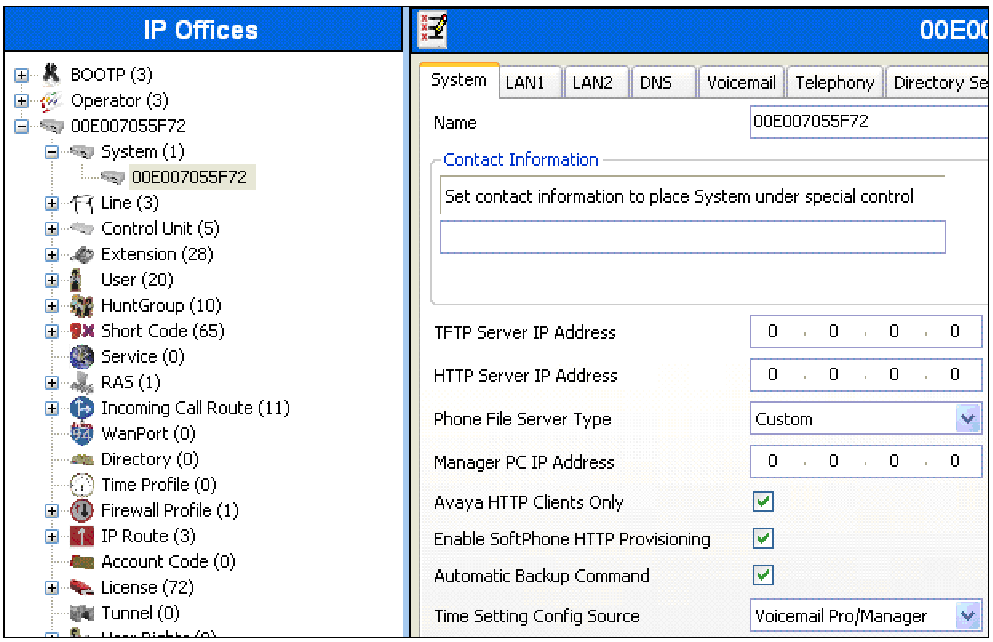


5.3. System Settings

This section illustrates the configuration of system settings. Select **System** in the Navigation pane to configure these settings. The configuration in following sections is for reference purposes only.

5.3.1. System Tab

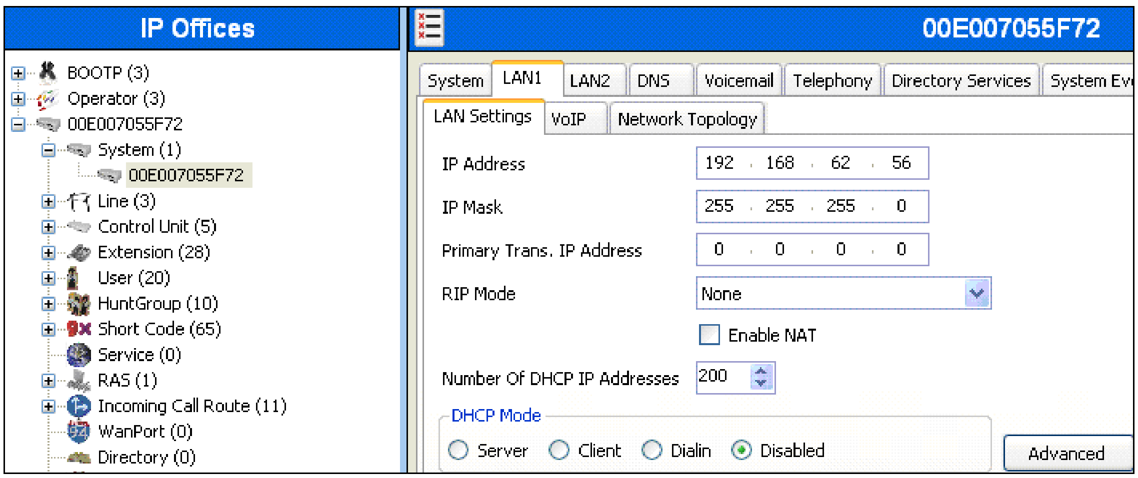
With the proper system name selected in the Group pane, select the **System** tab in the Details pane. The following screen shows a portion of the **System** tab. The **Name** field can be used for a descriptive name of the system. In this case, the MAC address is used as the name. The **Avaya HTTP Clients Only** and **Enable SoftPhone HTTP Provisioning** boxes are checked to facilitate IP Office Softphone usage.



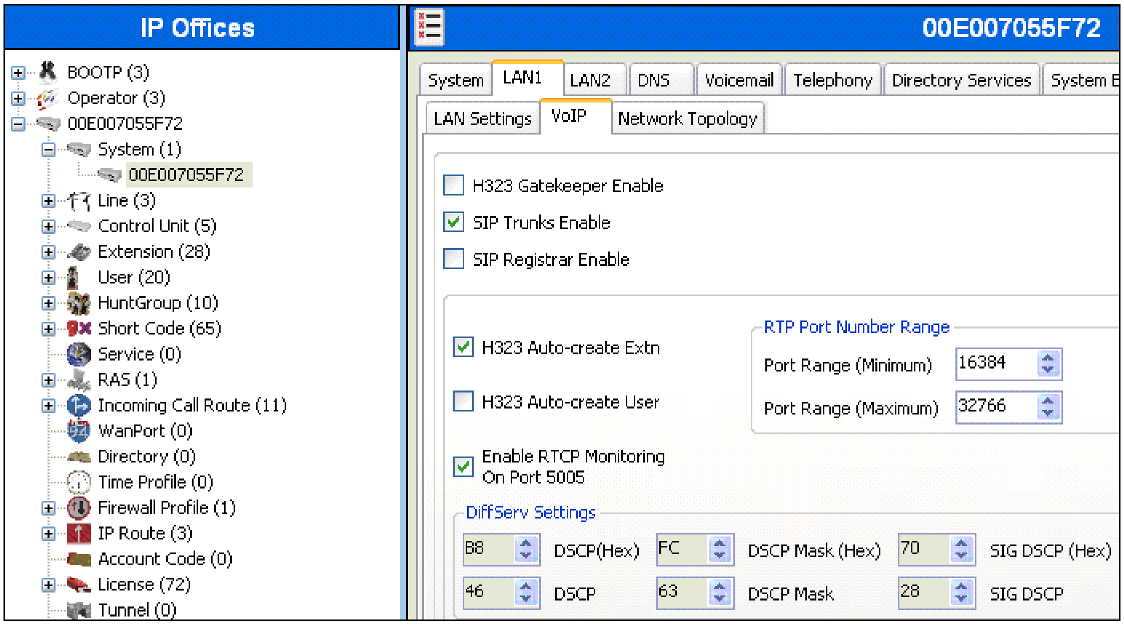
5.3.2. LAN Settings

In the sample configuration, LAN1 was used to connect the IP Office to AT&T Network and LAN2 was used to connect to the enterprise network.

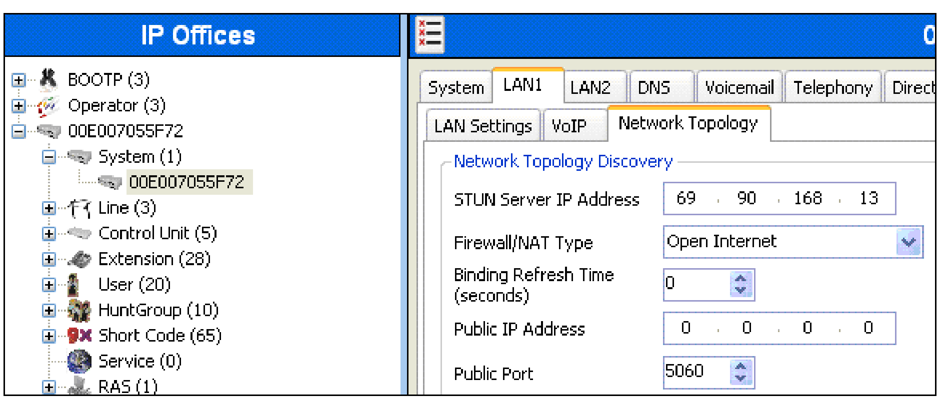
1. Select the **LAN1** tab followed by the **LAN Settings** tab and configure as follows:
   - **IP Address** – Set to **192.168.62.56** which is the IP address of IP Office known to AT&T network
   - **IP Mask** – Set to a valid value e.g. 255.255.255.0
   - **Primary Trans**. **IP Address** – Set to **0.0.0.0**
   - **DHCP Mode** – Select the **Disabled** radio button



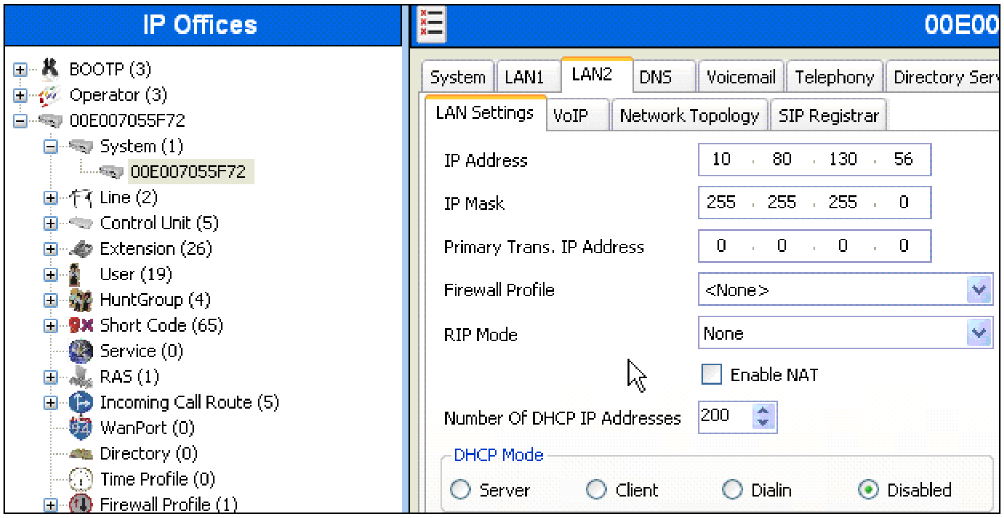
2. Select the **VoIP** tab as shown in the following screen and configure as follows:
   - **SIP Trunks Enable** – Check this box to enable the configuration of SIP trunks
   - **RTP Port Range (Minimum)** – Set to **16384** (As required by AT&T)
   - **RTP Port Range (Maximum)** – Set to **32766** (As required by AT&T). Although AT&T requires the maximum value to be **32767**, IP Office requires an even number to be entered in this field



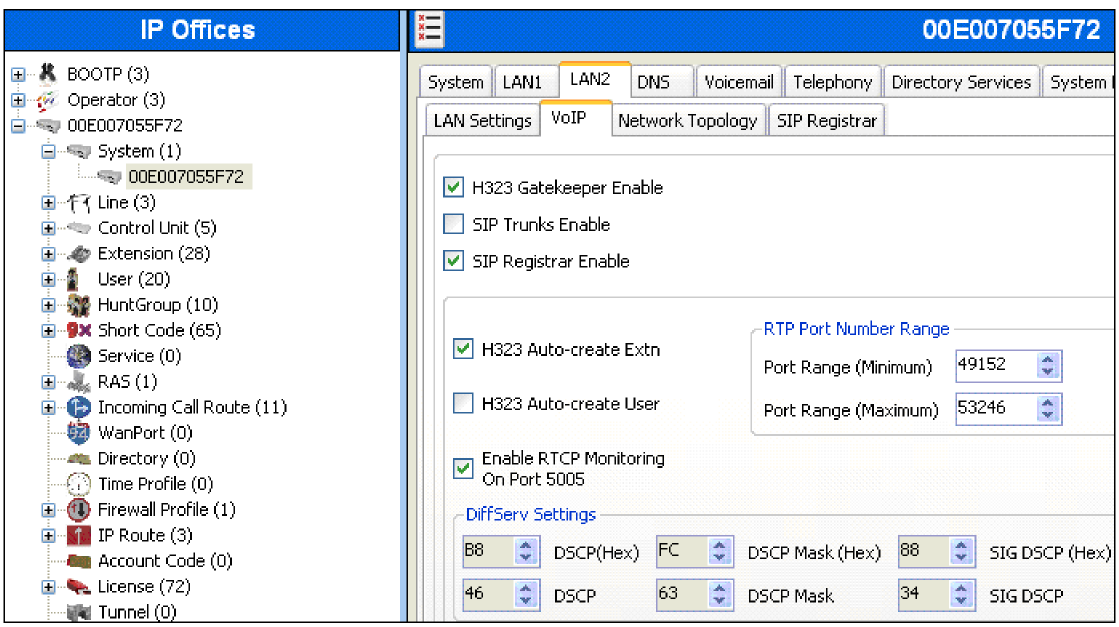
3. Select the **Network Topology** tab as shown in the following screen and configure as follows:
   - **Firewall/NAT Type** - Select **Open Internet** from the drop-down list. With this configuration, STUN will not be used but make sure to leave **STUN Server IP Address** to its default value.
   - **Public IP Address** – Set to 0.0.0.0 (default)
   - **Public Port** – Set to **5060** (default)



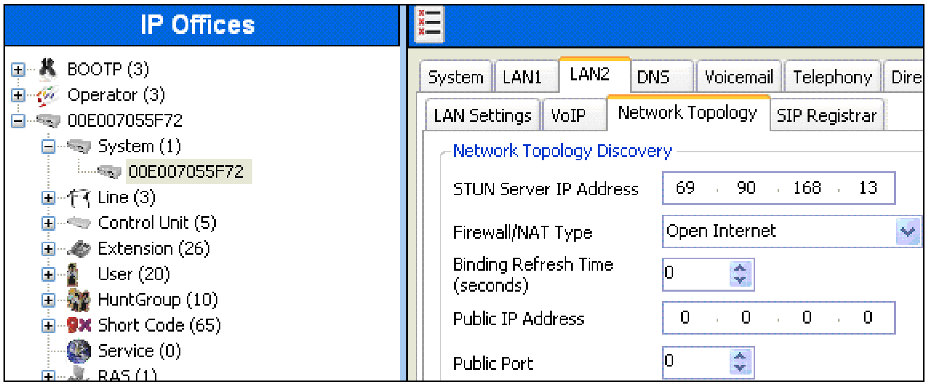
4. Select the **LAN2** tab followed by the **LAN Settings** tab and set **IP Address** of the IP Office on the enterprise side to **10.80.130.56** and **IP Mask** to **255.255.255.0**.



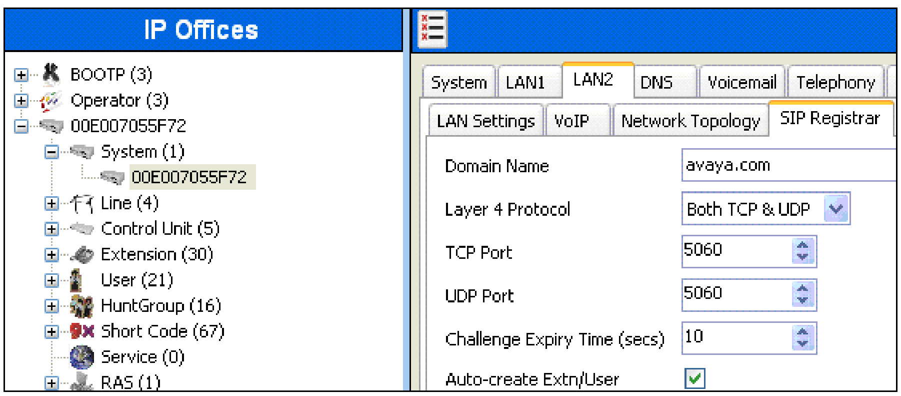
5. Select the **VoIP** tab as shown in the following screen and configure as follows:
   - **H323 Gatekeeper Enable** – Check this box to allow the use of Avaya IP Phones
   - **SIP Registrar Enable** – Check this box to allow SIP phones and IP Office Softphone usage



6. The Network Topology screen is set the same as it was set for LAN1.



7. Set the **Domain Name** field to **avaya.com** and leave all the other fields to their default values. This domain name is used to register the SIP telephones. Also, make sure that the **Layer 4 Protocol** field is set to **Both TCP & UDP** as Avaya IP Softphone uses UDP and the SIP phones require TCP.

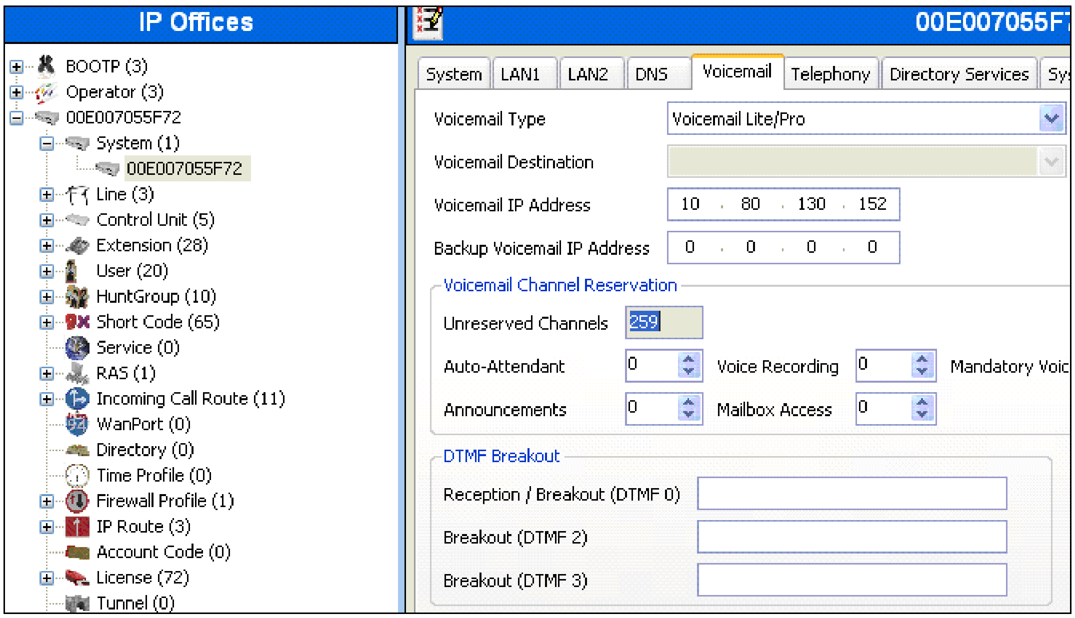


8. Click **OK** [not shown] to commit.

5.3.3. Voicemail

The settings presented here simply illustrate this reference configuration and are not intended to be prescriptive. Select **Voicemail** tab and configure as follows:

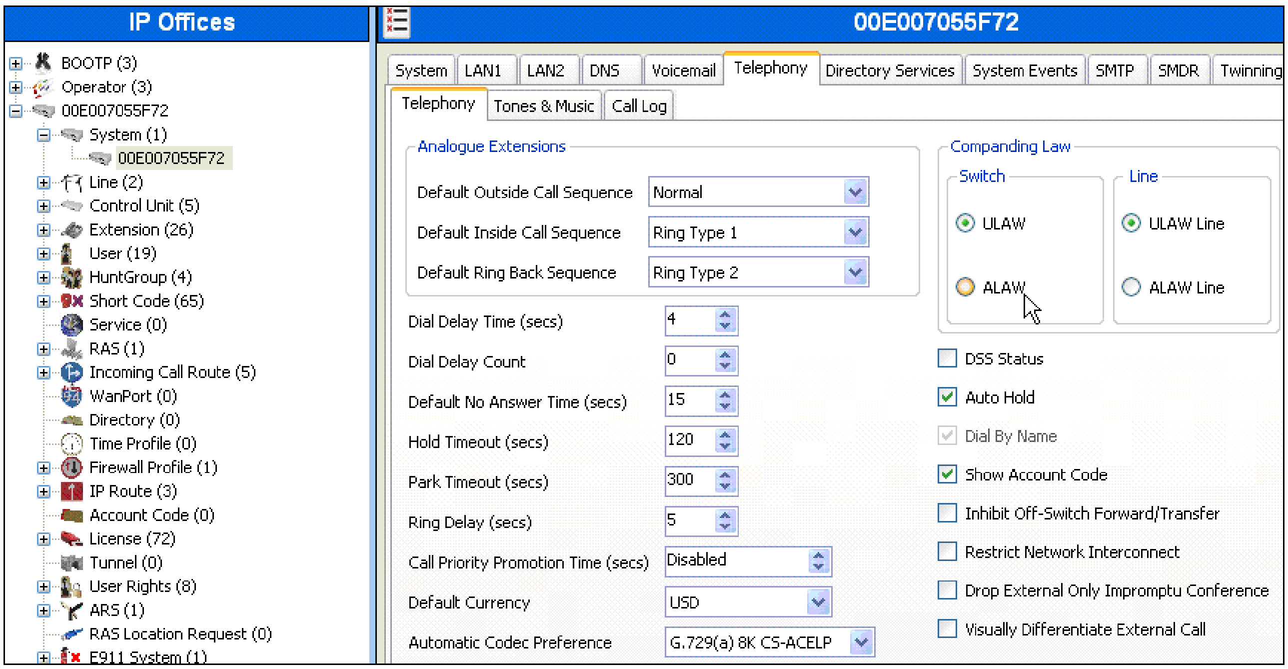
- **Voicemail Type** – Set to **Voicemail Lite/Pro** from the drop-down list
- **Voicemail IP Address** – Set to **10.80.130.152**, the IP Address of the PC running the Voicemail Pro software.



5.3.4. System Telephony Configuration

The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. Select the **Telephony** tab and **Telephony** sub-tab as shown and configure as follows:

- **Companding Law** – Check the **ULAW** box for **Switch** field and **ULAW Line** box for the **Line** field.
- **Inhibit Off-Switch Forward/Transfer** – Uncheck this box so that call forwarding and call transfer to PSTN destinations via the AT&T Toll Free service can be tested.
- **Automatic Codec Preference** – Use the default value of **G.729(a) 8K CS-ACELP**.



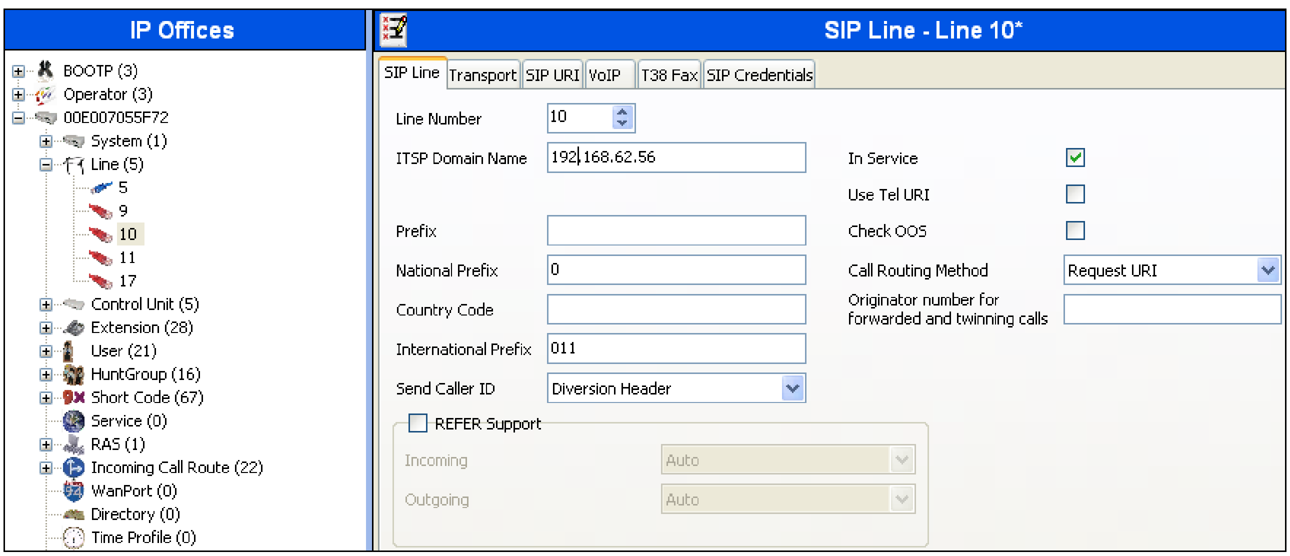
5.4. SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 6.1. To add a new SIP Line, right click on **Line** in the Navigation pane, and select **New → SIP Line**. A new Line Number will be assigned automatically. The settings presented here simply illustrate this reference configuration and are not intended to be prescriptive.

5.4.1. SIP Line - SIP Line Tab

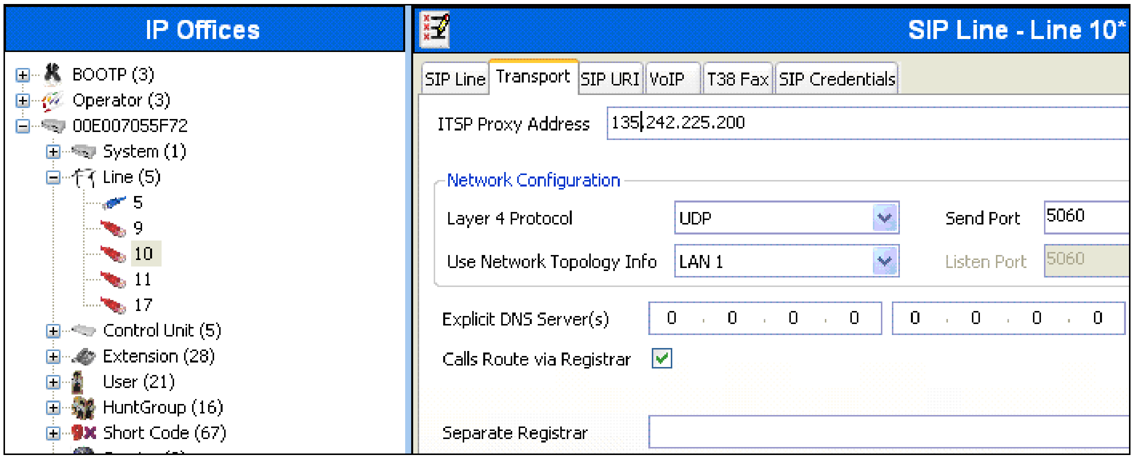
Select **SIP Line** tab as shown below for Line Number 10 used for AT&T and configure as follows:

- **ITSP Domain Name** – Set to the IP Office LAN1 address (192.168.62.56) configured in **Section 5.3.2** so that IP Office uses this IP address in the host portion of SIP headers such as the From header and Diversion header.
- **In Service** – Default is checked
- **Check OOS** – Uncheck this box. If this box is checked, it enables IP Office to use the SIP OPTIONS method to periodically check the SIP Line and if no response is received, the SIP line is taken out of service. See **Section 5.9** for additional information related to configuring the periodicity of SIP OPTIONS.
- **Call Routing Method** – Set to **Request URI** (default)



5.4.2. SIP Line - Transport Tab

Select the **Transport** tab and set the **ITSP Proxy Address** to the AT&T Border Element IP Address. The **Use Network Topology Info** parameter is set to **LAN 1** configured in **Section 5.3.2**. Default values are used for the other fields.

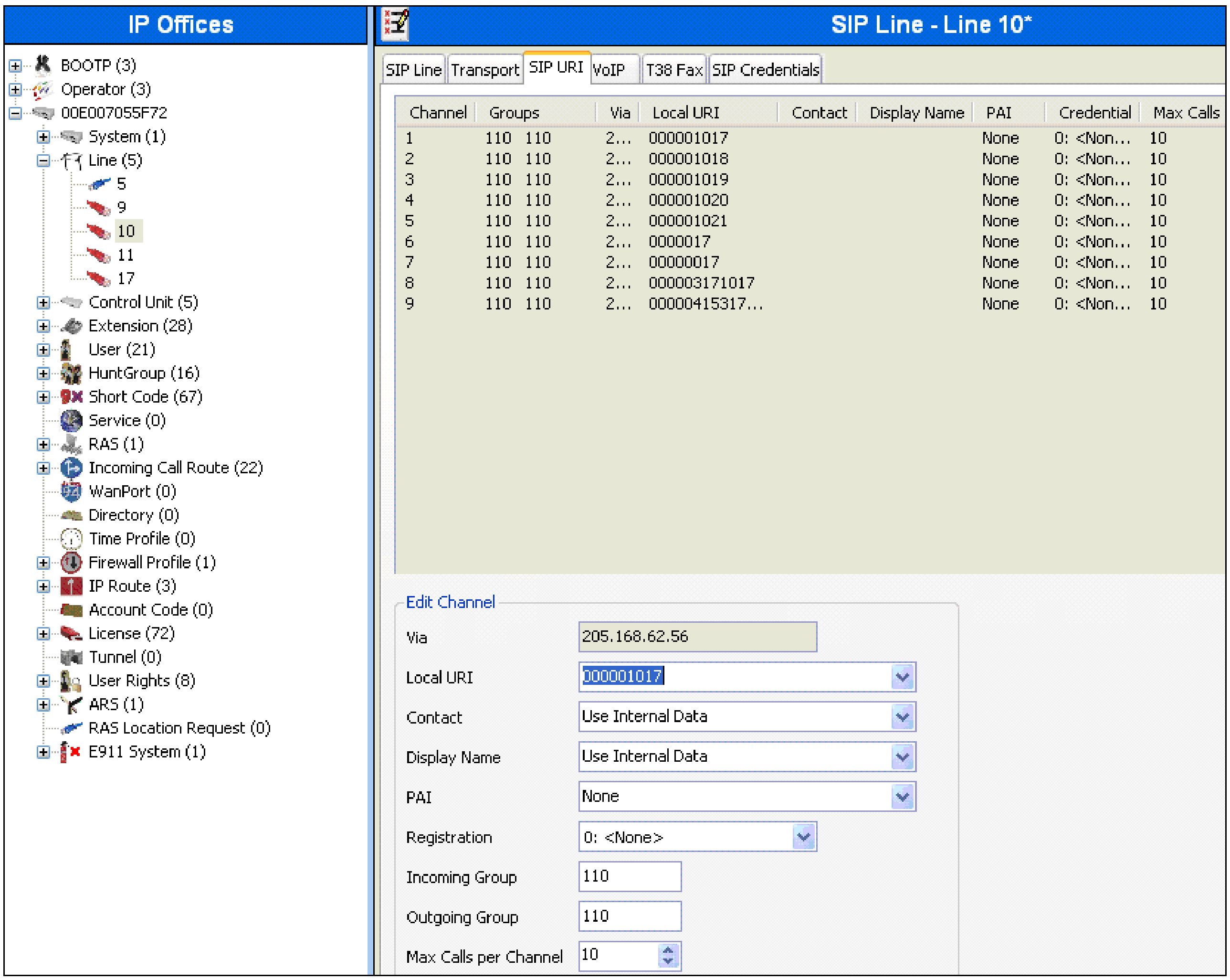


5.4.3. SIP Line - SIP URI Tab

Select the **SIP URI** tab and click the **Add…** button to add a new SIP URI. Configure the **New Channel** section displayed as follows:

- **Local URI** – Set to the DNIS sent by AT&T IP Toll Free service in the SIP URI. In this example it is set to 000001017 which is one of the DNIS mentioned in **Table 1**.
- **Registration** - Set to **0: <None>**
- **Incoming Group** and **Outgoing Group** – Set to **110**

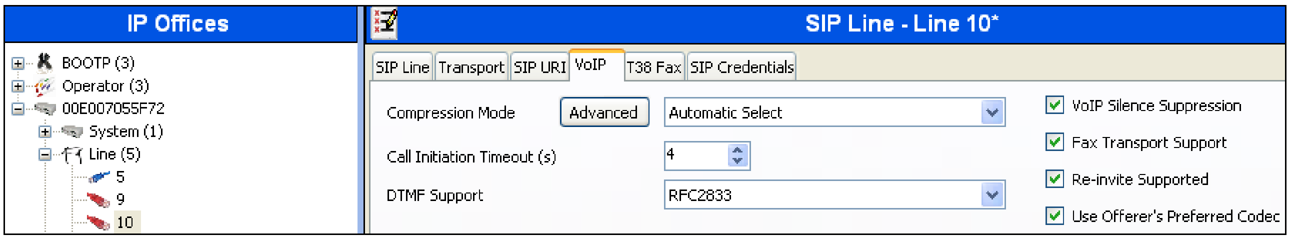
Repeat above steps for other DNIS provided by AT&T.



5.4.4. SIP Line - VoIP Tab

Select the **VoIP** tab and configure as follows:

- **Compression Mode** – Set to **Automatic Select** from the drop-down list
- **DTMF Support** - Set to the default value **RFC2833**.
- **Re-invite Supported** – Check to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk
- **Use Offerer's Preferred Codec** – Check this box so that the top codec offered to IP Office is used if IP Office supports that codec
- **Fax Transport Support** – Check to allow T38 fax support.



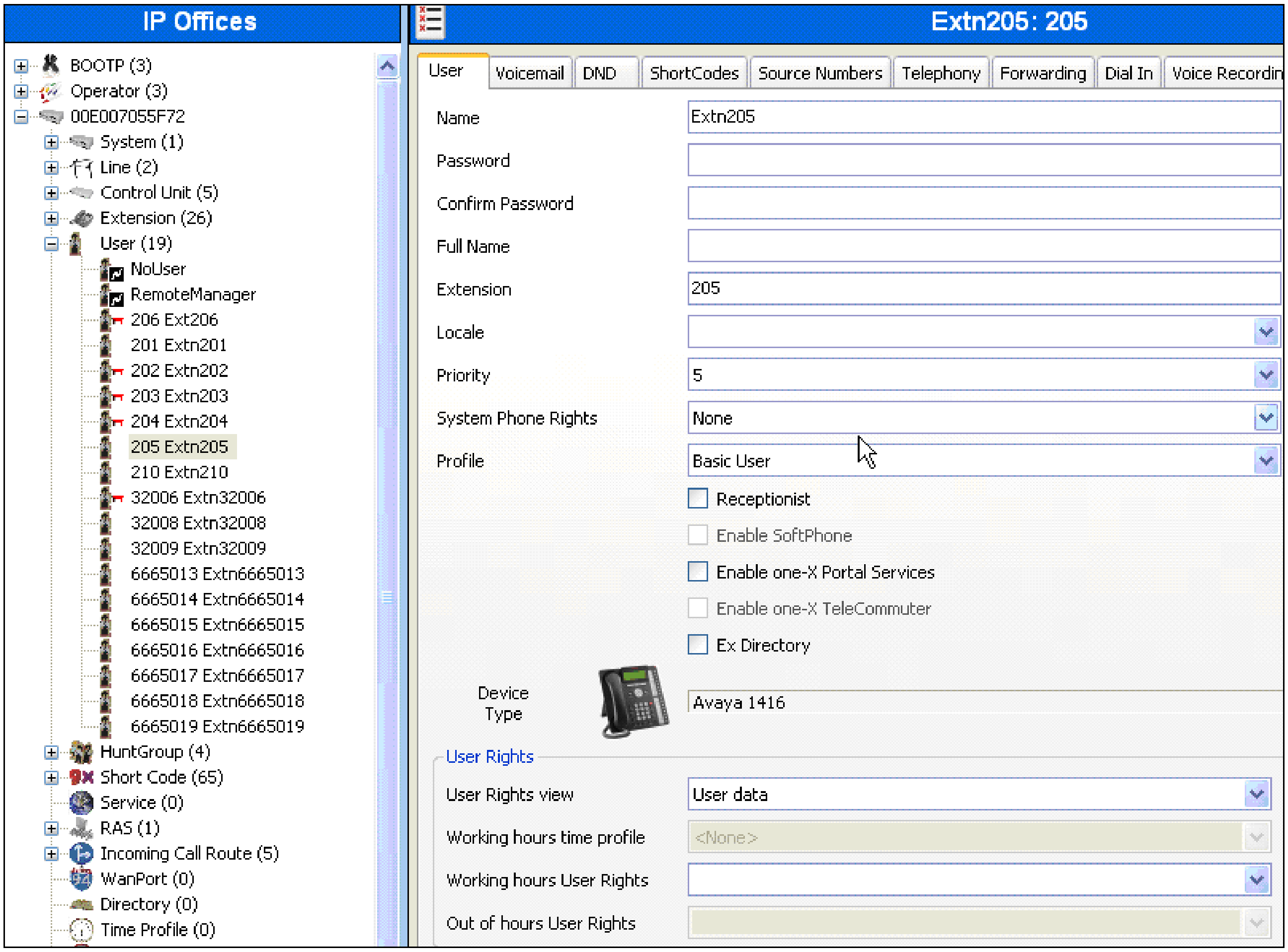
Since default values were used for T38 fax and AT&T IP Toll Free does not require registration, the **T38 Fax** and **SIP Credentials** tabs need not be visited. Click **OK** (not shown) to commit the SIP Line configuration.

5.5. Users, Extensions, and Hunt Groups

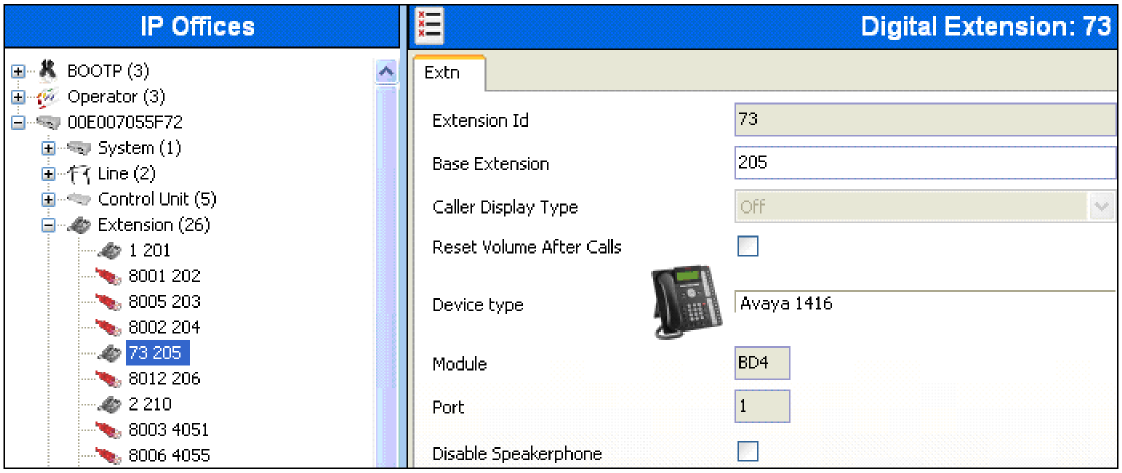
In this section, examples of IP Office Users, Extensions, and Hunt Groups will be illustrated. In the interests of brevity, not all users and extensions used will be presented, since the configuration can be easily extrapolated to other users. To add a User, right click on **User** in the Navigation pane, and select **New.** To edit an existing User, select **User** in the Navigation pane, and select the appropriate user to be configured in the Group pane.

5.5.1. Digital Telephone User 205

The following screen shows the **User** tab for User 205. This user corresponds to a digital phone.

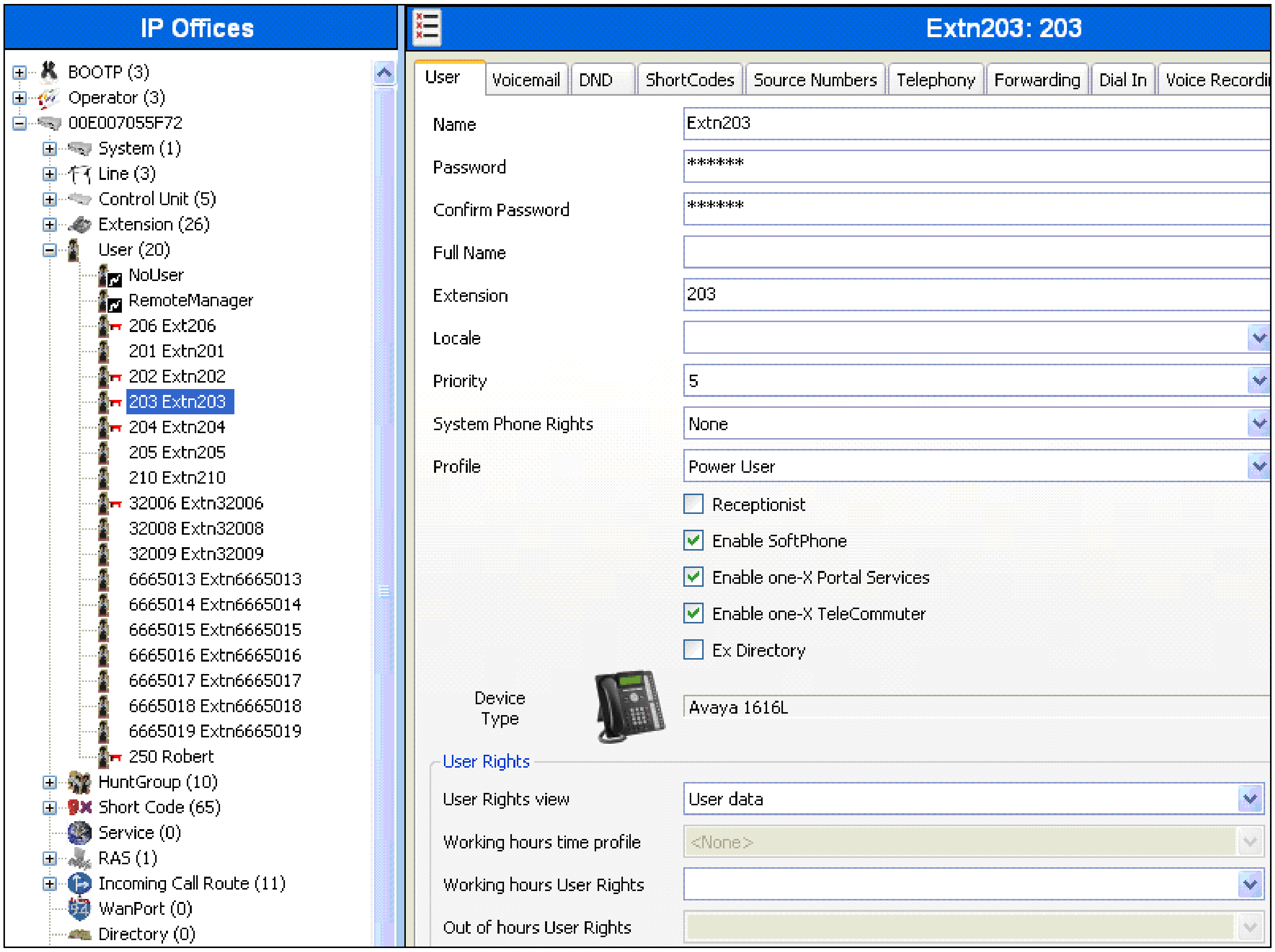


The following screen shows the Extension information for this user. To view, select **Extension** and the appropriate extension from the Navigation pane.

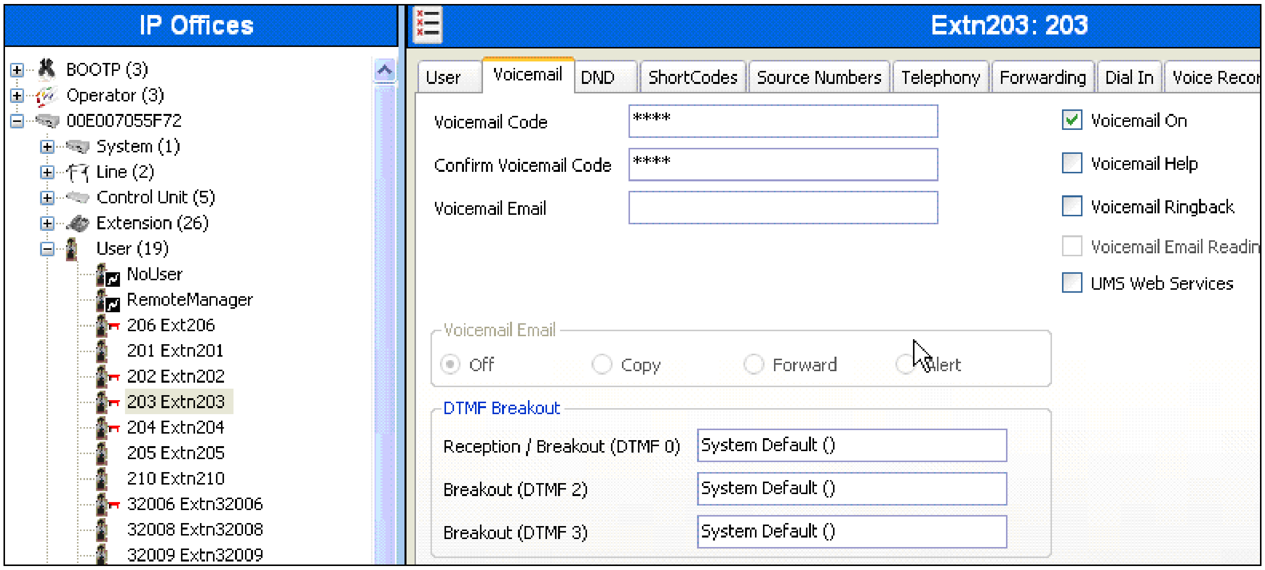


5.5.2. IP Telephone User 203

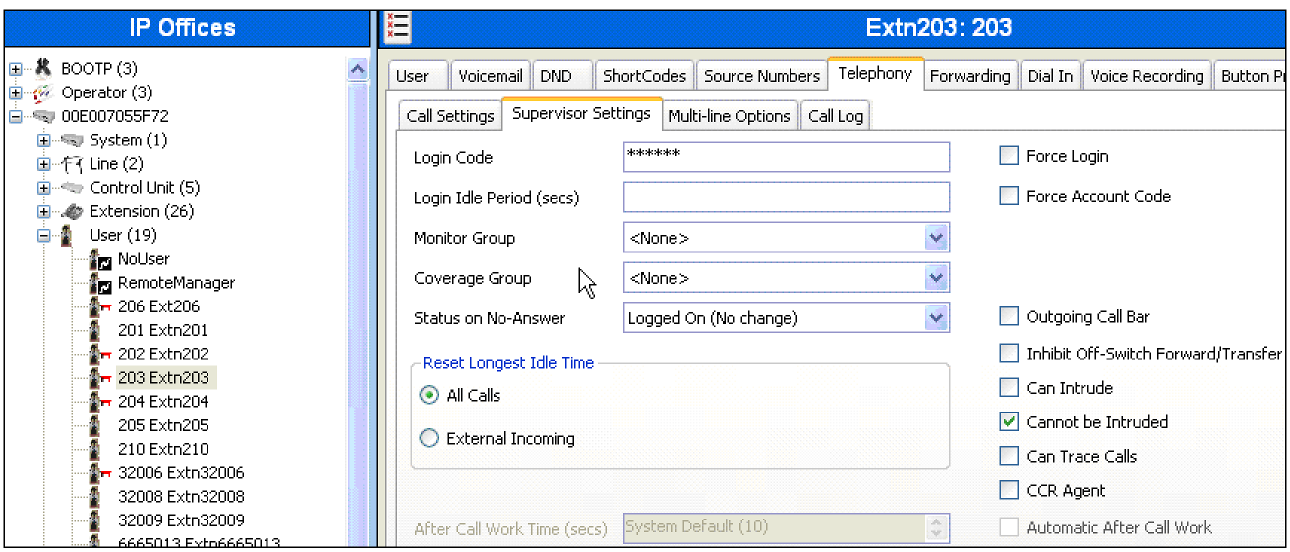
The following screen shows the **User** tab for User 203. This user corresponds to an Avaya 1616 IP Telephone that is configured as power user with IP Office Softphone features enabled as shown below.



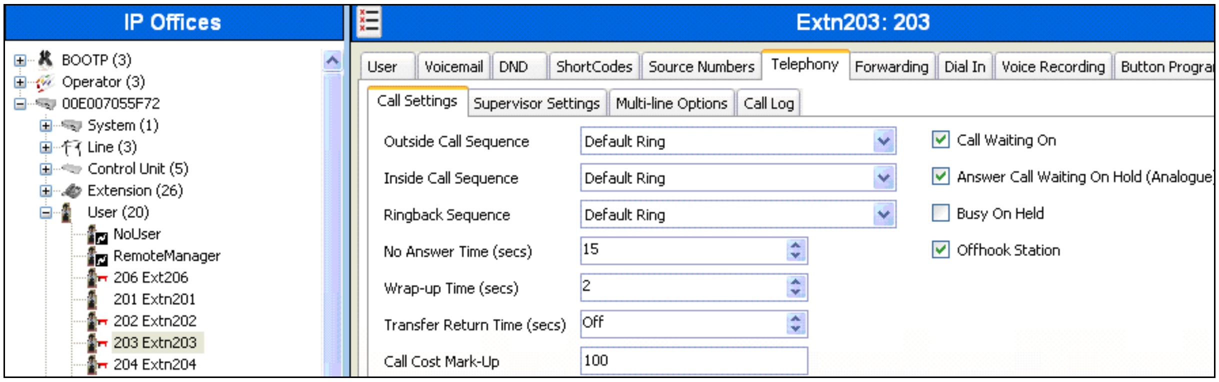
The following screen shows the **Voicemail** tab for this user. The **Voicemail On** box is checked, and a voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters.



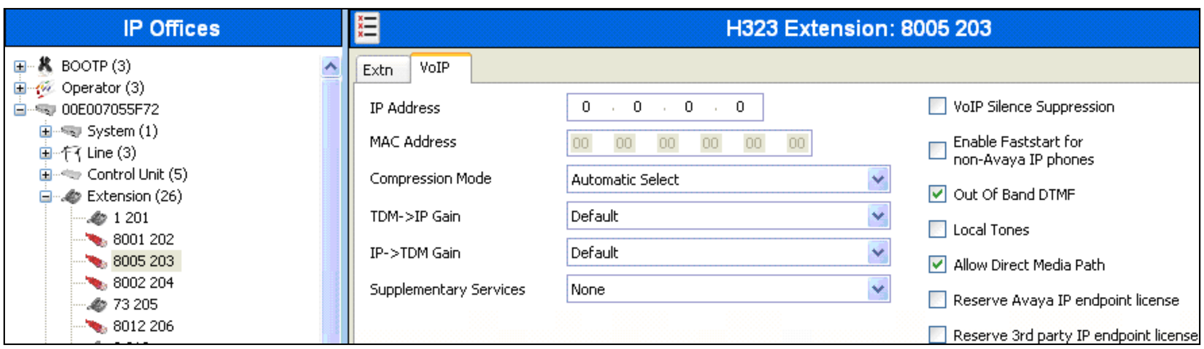
Select the **Supervisor Settings** tab as shown below. To allow hot desking, enter a **Login Code**.



Select the **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow an IP Office Softphone logged in as this extension to have multiple call appearances (e.g., necessary for call transfer).

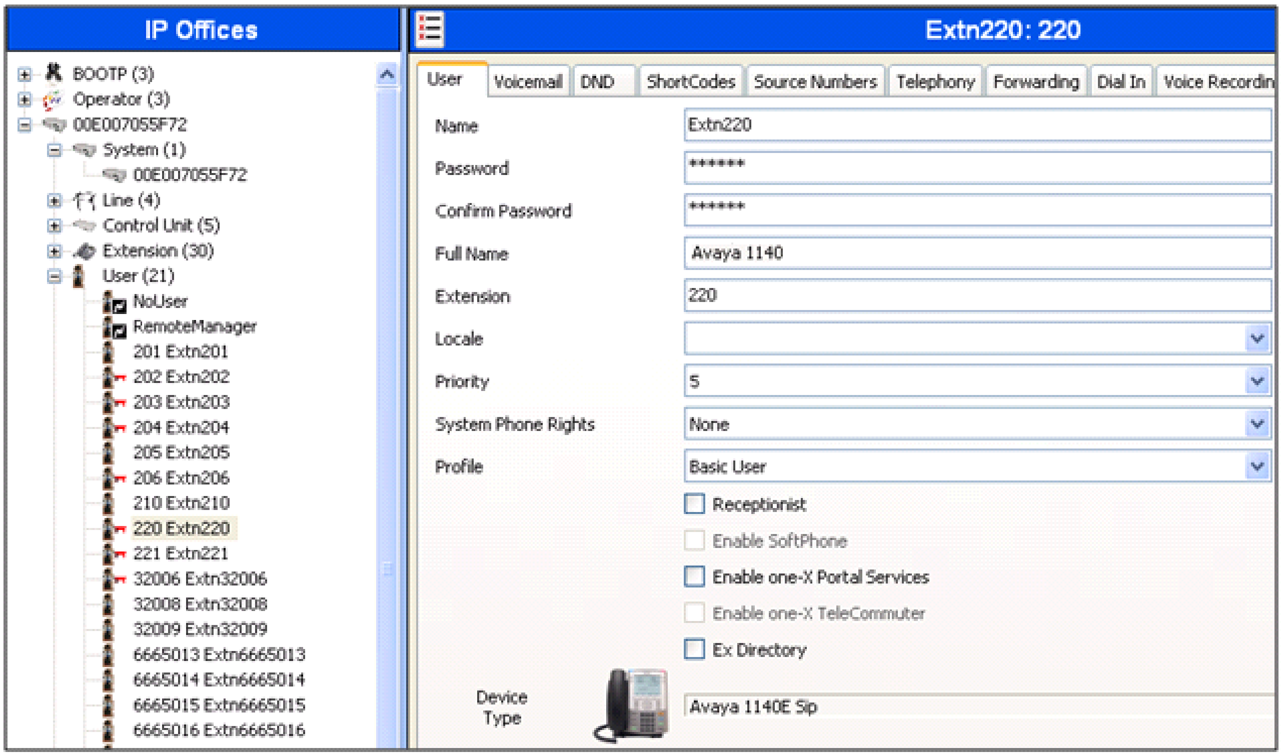


The following screen shows the Extension information for this user, simply to illustrate the **VoIP** tab available for an IP Telephone.

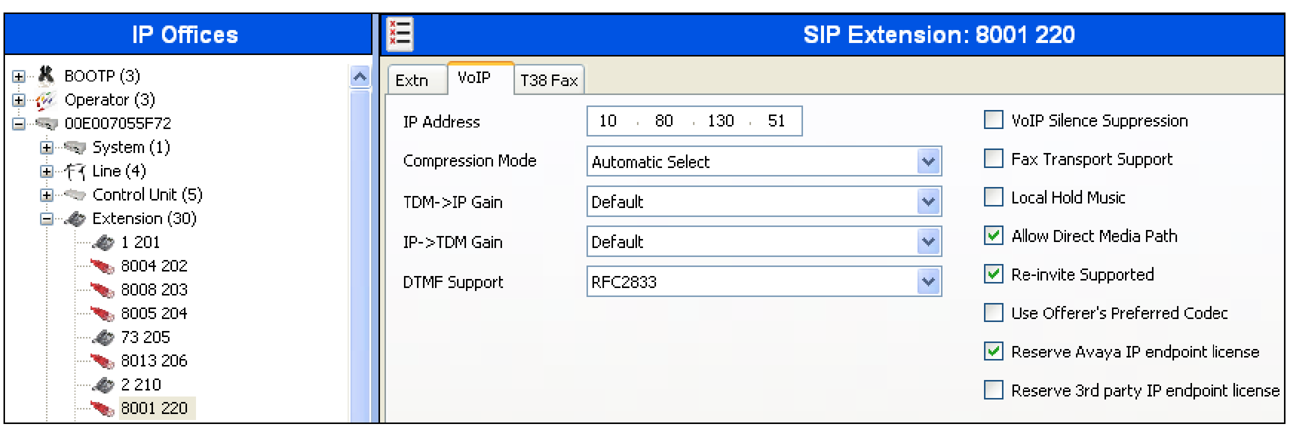


5.5.3. SIP Telephone User 220

The following screen shows the **User** tab for User 220. This user corresponds to an Avaya 1140E SIP Telephone and configured as below:



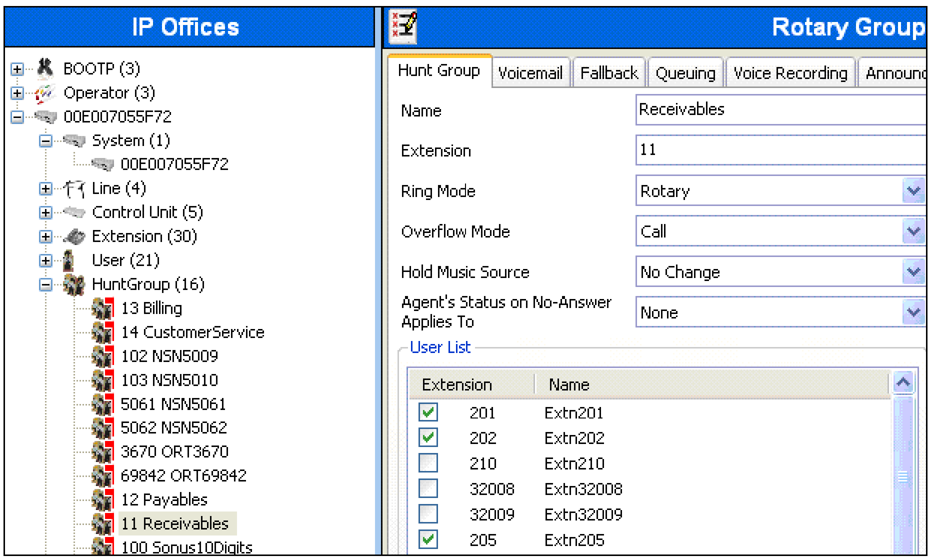
The following screen shows the Extension information for this user. Note that for a SIP telephone, the IP Address configured for the phone needs to be specified. In this example, 10.80.130.51 was assigned to the Avaya 1140E telephone. All other screens are configured the same way as in **Section 5.5.2**.



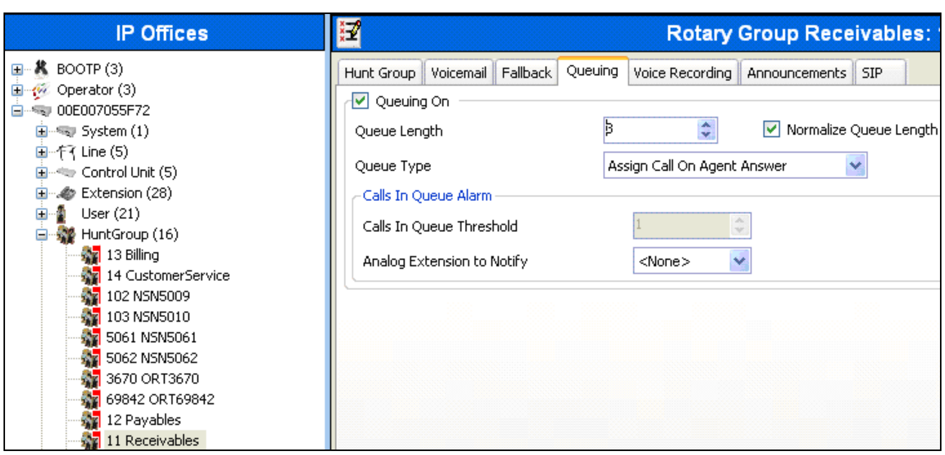
5.5.4. Hunt Groups

Hunt groups were used in this reference configuration to route the incoming calls on a SIP Trunk from AT&T Toll Free service to the an agent with the right skill set. To configure a new hunt group, right-click **HuntGroup** from the Navigation pane, and select **New**. To view or edit an existing hunt group, select **HuntGroup** and choose the appropriate hunt group from the Navigation pane.

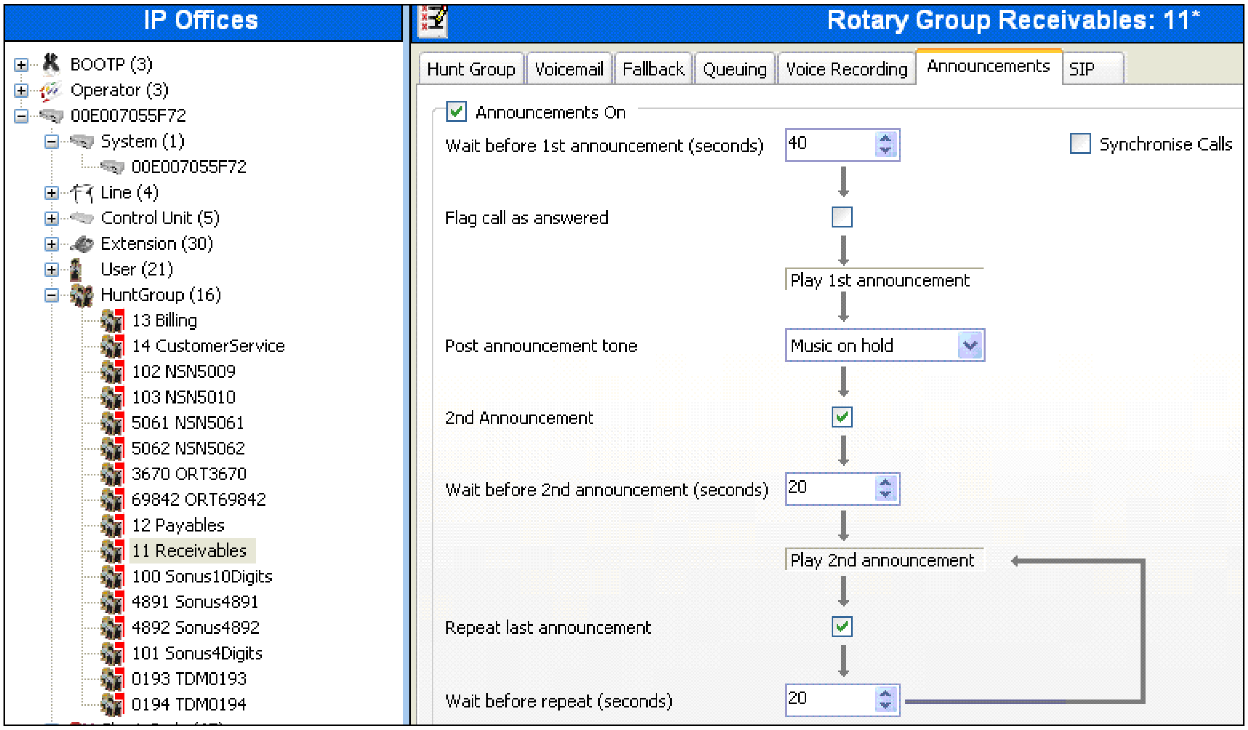
The following screen shows the **Hunt Group** tab for hunt group 11. The group name is set to **Receivables**. Several extensions/agents are part of this hunt group. Since the **Ring Mode** field is set to **Rotary**, this will enable the telephones to ring in a round robin fashion. Click the **Edit** button to change the **User List**. Once a user is part of a hunt group **User List**, it can be enabled/disabled by checking/unchecking the box in the Extension field.



Under the **Queuing** tab, check the **Queuing On** box and set the **Queue Length** field to any desirable value. Use the default values for all the other fields.



Under the **Announcements** tab and check the **Announcements On** box. The wait time can be set to any desirable value. These announcements are played if an agent for a particular skill is unavailable.



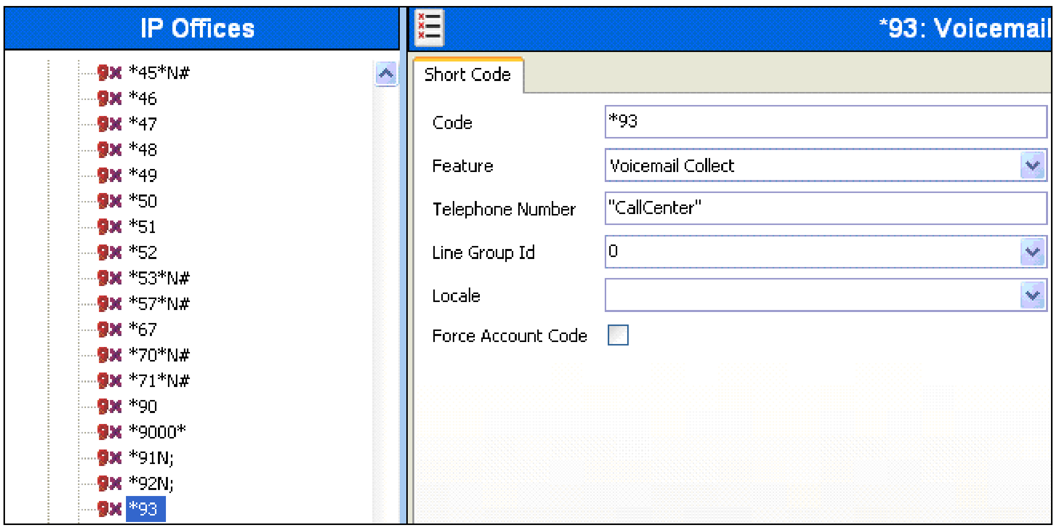
Additional hunt groups Billing, Payables and Customer Service were created in this reference configuration to exercise the Call Center functionality within IP Office.

5.6. Short Codes

In this section, various examples of IP Office short codes will be illustrated. To add a short code, right click on **Short Code** in the Navigation pane, and select **New.** To edit an existing short code, click **Short Code** and select the code to be edited in the Navigation pane.

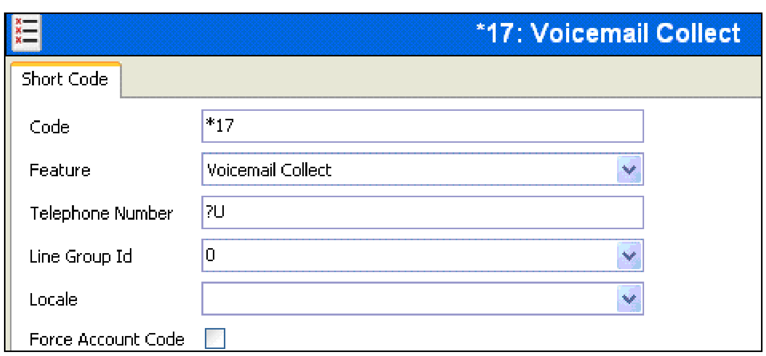
5.6.1. Call Center Codes

Call Center functionality is configured on Voicemail Pro. **Section 5.8** lists some of the configuration steps to provide this functionality. In order to access this functionality, short codes can be used. In this reference configuration, **CallCenter** was configured on Voicemail Pro. The following screen shows the short codes set for this functionality.



5.6.2. Voicemail Retrieval Code

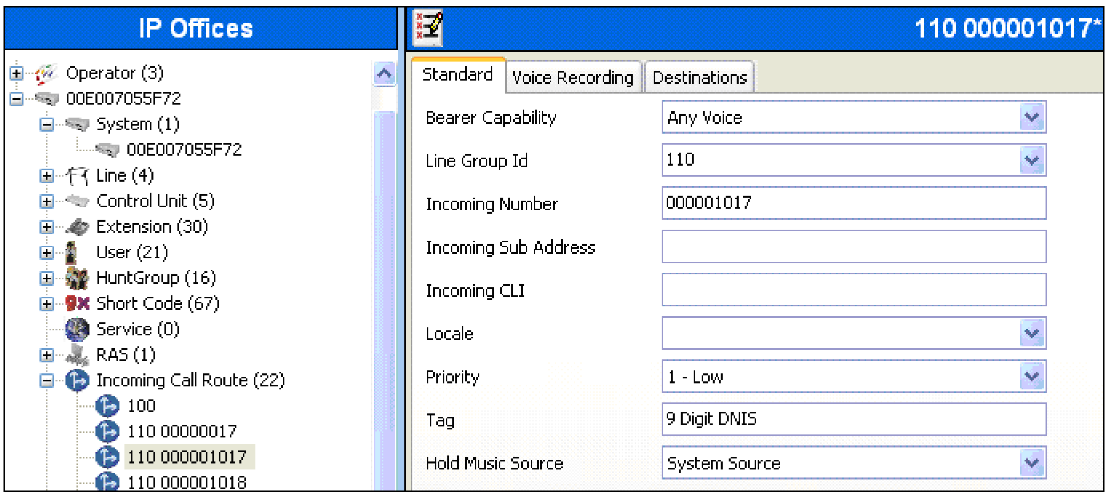
To retrieve voicemails left in individual mailboxes, this code was configured in this reference configuration. When a user enters, ***17**, they can retrieve the messages in their mailbox. Additionally, this short code can be also in the Incoming Call Route configured in **Section 5.7**.



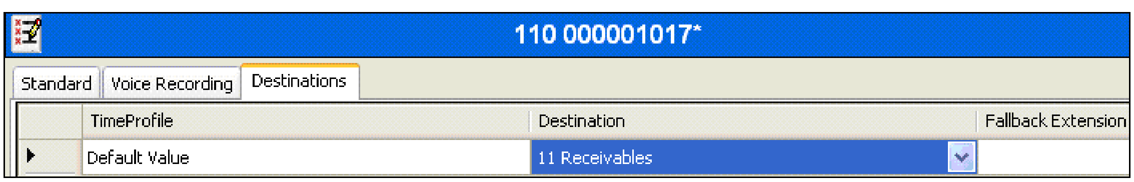
5.7. Incoming Call Routes

In this section, IP Office Incoming Call Routes are illustrated. Each Incoming Call Route will map a specific AT&T IP Toll Free DNIS to a destination user, group, or function on IP Office. To add an incoming call route, right click on **Incoming Call Route** in the Navigation pane, and select **New.** To edit an existing incoming call route, select **Incoming Call Route** and the appropriate route in the Navigation pane.

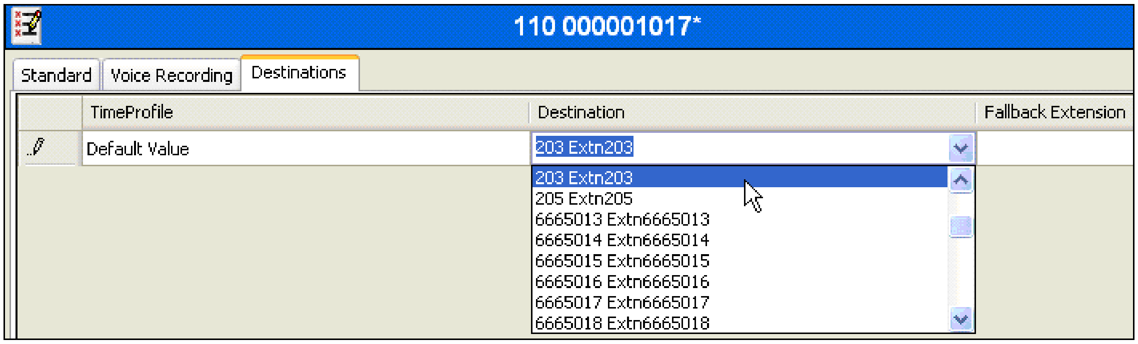
The screen shown below matches the AT&T IP Toll Free DID **000001017** in the **Incoming Number** field on the **Line Group Id** (**110**). The **Line Group Id** matches the **Incoming Group** field and **Incoming Number** matches the **Local URI** field configured in the **SIP URI** tab for the SIP Line to AT&T IP Toll Free service in **Section 5.4**.



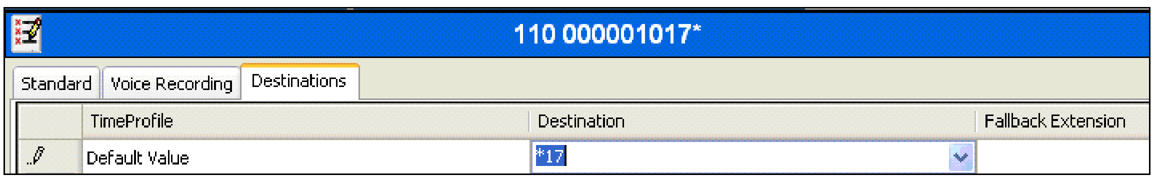
Select the **Destinations** tab and a value can be either selected from the drop-down list or manually entered. In the screen shown below, the hunt group configured in **Section 5.5.4** was selected.

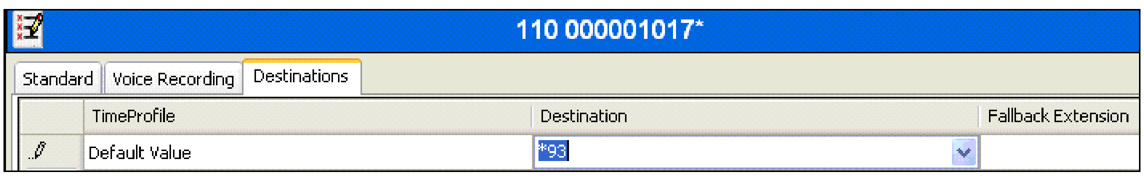


Similarly, in the screen below, an extension configured in **Section 5.5.2** was selection.

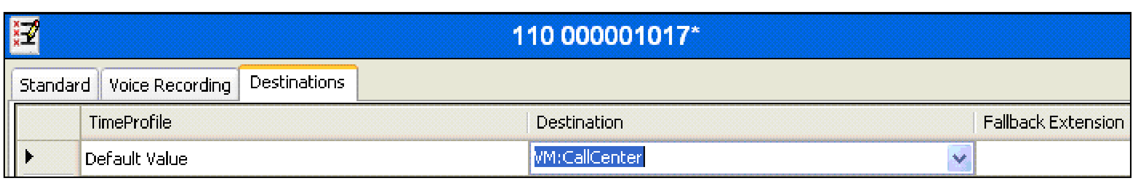


The following screens displays how a short code can be manually assigned in the **Destination** field to route the call for voicemail retrieval and access Call Center functionality.





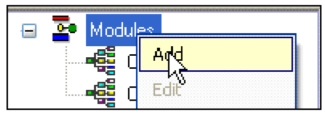
The following screen displays another mechanism to access the Call Center functionality without using the short code. The Call Center functionality is configured in Voicemail Pro as detailed in **Section 5.8**.



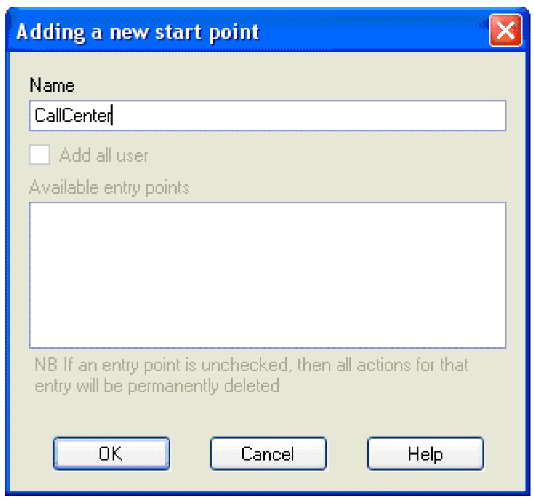
5.8. Call Center Provisioning in Voicemail Pro

The call center functionality was configured in Voicemail Pro. Following steps highlight the configuration of this functionality. For further information, consult [IPO-VMPRO].

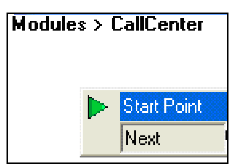
1. Navigate to **Start→Voicemail Pro Client** and right click on modules and select **Add** to add a new module.



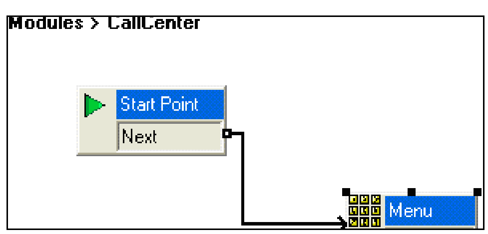
2. In the screen below, enter **CallCenter** in the **Name** field and click OK.



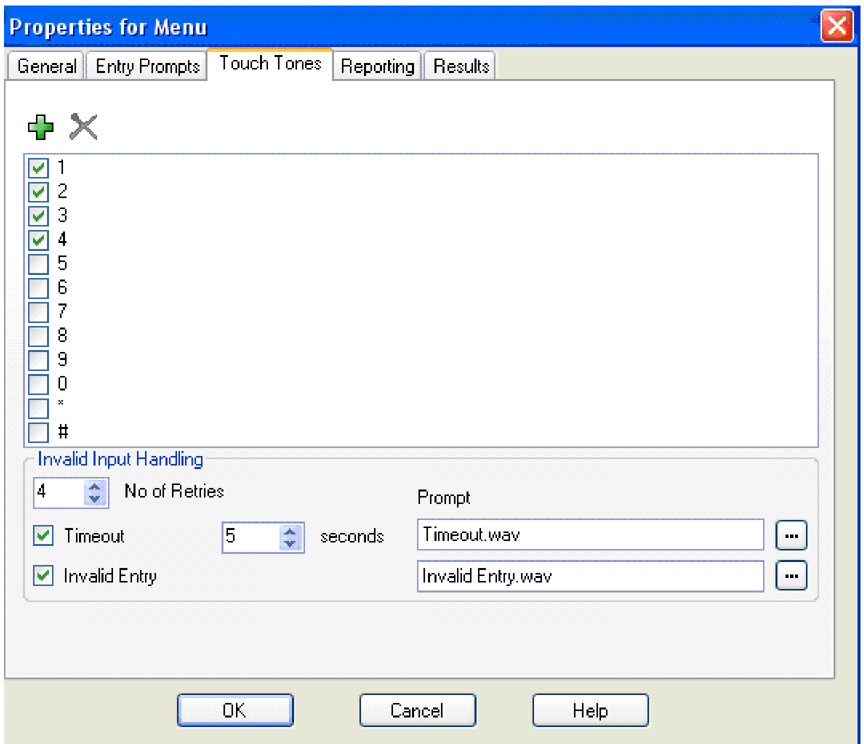
3. Following screen is displayed indicating the starting point for the Call Center functionality.



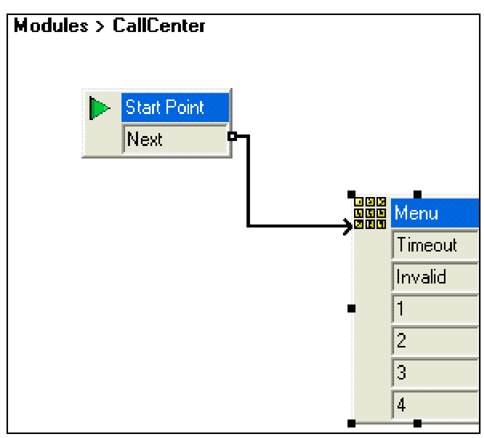
4. Under the **Actions** tab, select **Basic Actions** [not shown]. Select **Menu** and place it on the right side of the pane and then connect the **Start Point** to **Menu** as shown below:



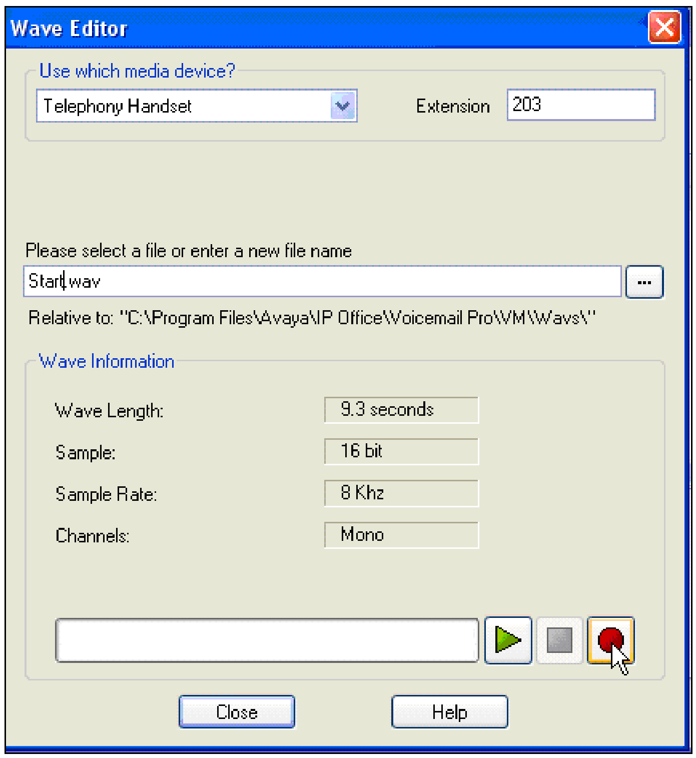
5. Right click on **Menu** and select **Touch Tones** tab. Check the appropriate boxes. In this reference configuration, 1, 2, 3, 4, Timeout and Invalid boxes were selected. This allows caller to enter any of the digits from 1 to 4 to go to the appropriate agent. Digit **1** was used for **Receivables**, Digit **2** was used for **Payables**, Digit **3** was used for **Billing** and Digit **4** was used for **Customer Service** Hunt Groups/Skills in this reference configuration. Digits have to be entered within a certain time and within the specified range otherwise an error recording may be played. Enter any valid number in the **No. of Retries** field. This field dictates the number of retries allowed to the caller for entering a digit.



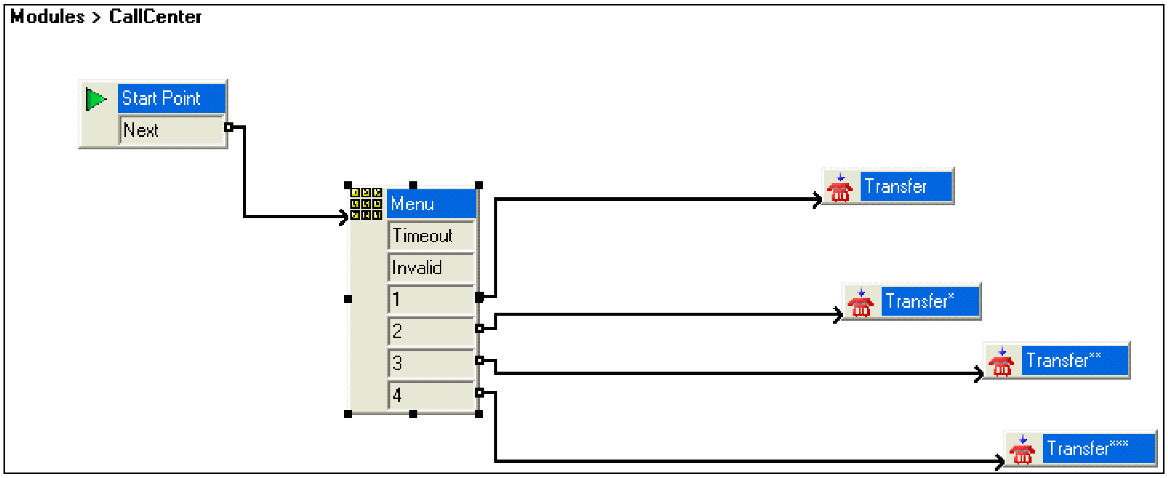
Click **OK** and following screen is displayed:



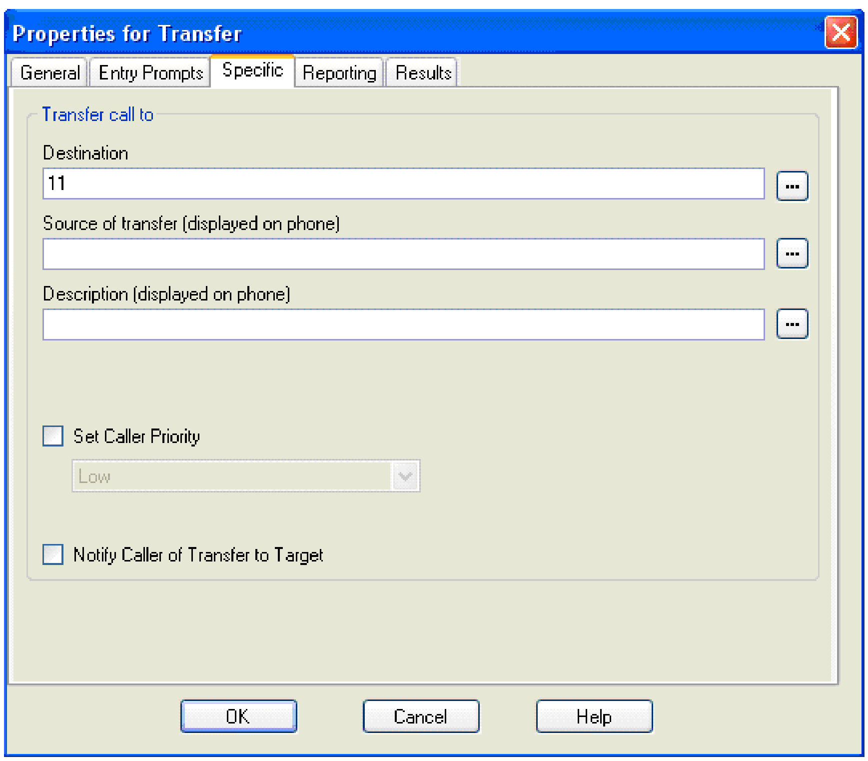
6. Right click on the **Menu** action and select the **Entry Prompts** tab and click on the ✚ sign [not shown]. On the following screen, enter the **Extension** where the recording is done and the filename for the recording in the **Please select a file or enter a new file name** field and press the Red record button as shown. In this reference configuration the phone at extension **203** rings and Voicemail Pro prompts the user to record an announcement which is played back to the caller when the call comes into IP Office.



7. Under the **Actions** button, select **Transfer** [not shown]. Repeat this step for additional actions. In this reference configuration four **Transfer** actions were created for each of the selections in **Step 5** and connected to them.



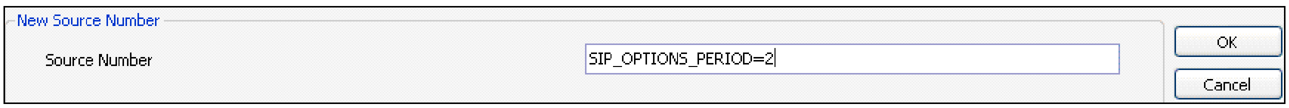
Right click on the **Transfer** action and select the Specifics tab. In the **Destination** field enter the hunt group/skill number created in **Section 5.5.4** and click **OK**. This will enable the call to be routed to the appropriate skill. Repeat this step for all the **Transfer** actions.



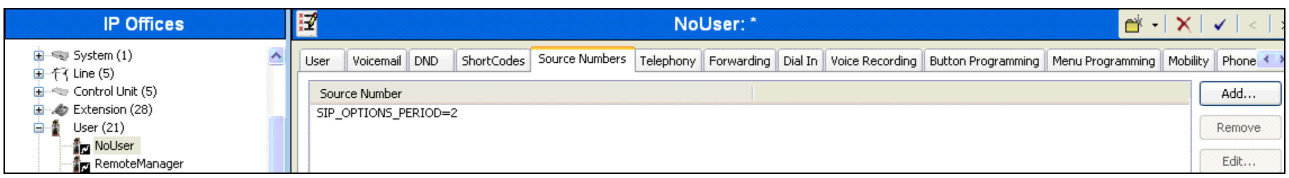
5.9. SIP OPTIONS Frequency

From the Navigation pane, select **User** and then select the user named **NoUser**. In the **NoUser** Details pane, select the tab **Source Numbers**. Press the **Add…** button to the right of the list of any previously configured Source Numbers. In the **Source Number** field shown below, type **SIP_USE_PAI_FOR_PRIVACY**. Click **OK**.

Similarly SIP OPTIONS frequency was configured by adding another source number by entering **SIP_OPTIONS_PERIOD=2** in the **Source Number** field. This will set the frequency of the SIP OPTIONS message sent by IP Office to 2 minutes.

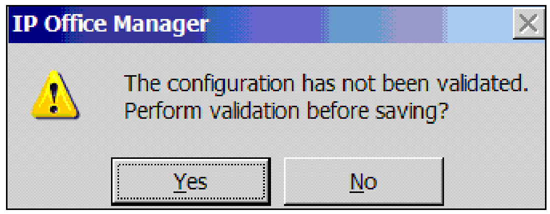


The following screen displays the Source Numbers configured in this reference configuration.

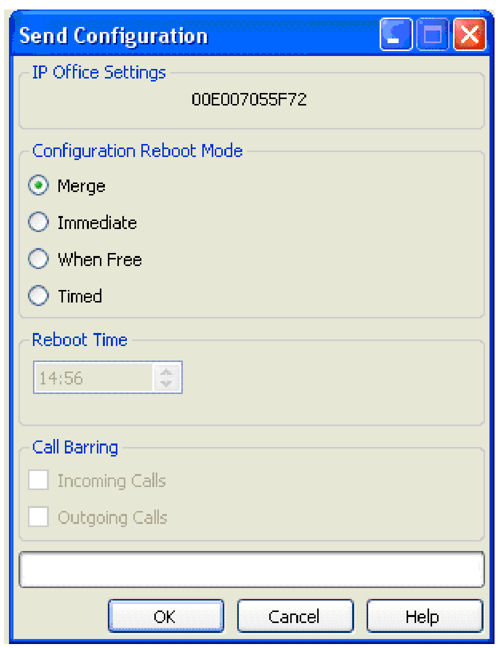


5.10. Saving Configuration Changes to IP Office

When desired, send the configuration changes made in IP Office Manager to the IP Office server, to cause the changes to take effect. Click the "disk" icon that is the third icon from the left (i.e., common "save" icon with mouse-over help "Save Configuration File"). Click **Yes** to validate the configuration, if prompted.



Once the configuration is validated, a screen similar to the following will appear, with either "Merge" or "Immediate" selected, based on the nature of the configuration changes made since the last save. Note that clicking OK may cause a service disruption. Click **OK** if desired.



6. Verification Steps

The following steps may be used to verify the configuration:

6.1. General

- Place an inbound call, answer the call, and verify that two-way talk path exists. Verify that the call remains stable for several minutes and disconnect properly.
- Place an inbound call to an agent or phone, but do not answer the call. Verify that the call covers to Voicemail Pro and messages can be retrieved using the appropriate short codes.
- Use the **System Status** application to verify the status of trunks, extensions and call progress.
- Use the **Monitor** application to monitor the activity on IP Office.

7. Conclusion

As illustrated in these Application Notes, Avaya IP Office can be configured to interoperate successfully with the AT&T IP Toll Free service. This solution provides users of Avaya IP Office the ability to support inbound toll free calls over an AT&T IP Toll Free SIP trunk service connection via MIS/PNT transport.

The reference configuration shown in these Application Notes is representative of a basic enterprise customer configuration and is intended to provide **configuration guidance** to supplement other Avaya product documentation. It is based upon formal interoperability compliance testing as part of the Avaya DevConnect Service Provider program.

8. References

This section references documentation relevant to these Application Notes. In general, Avaya product documentation is available at http://support.avaya.com

[IPO-INSTALL] IP Office 6.1 Installation Manual, Issue 22g, November 17 2010
Document Number 15-601042
https://support.avaya.com/css/P8/documents/100119958

[IPO-MGR] IP Office Release 6.1 Manager 8.1, Issue 25i, November 23, 2010
Document Number 15-601011
https://support.avaya.com/css/P8/documents/100119917

[IPO-SYSSTAT] IP Office Release 6.0 System Status Application, Issue 05a, February 12, 2010
Document Number 15-601758
http://support.avaya.com/css/P8/documents/100073300

[IPO-VMPRO] IP Office Release 6.1 Voicemail Pro Installation and Maintenance, Issue 23c, November 5, 2010
Document Number 15-601063
https://support.avaya.com/css/P8/documents/100119901

[IPO-MON] IP Office System Monitor, Issue 02b, November 28, 2008
Document Number 15-601019
http://support.avaya.com/css/P8/documents/100073350

Additional IP Office documentation can be found at:
http://marketingtools.avaya.com/knowledgebase/

AT&T IP Toll Free Service Descriptions:
[1] *AT&T IP Toll Free*
http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-toll-free-enterprise/

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.