Avaya Solution & Interoperability Test Lab

Applications Notes for Avaya IP Office 7.0 with AT&T IP Flexible Reach Business in a Box (SM) SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office with the AT&T IP Flexible Reach with Business in a Box services. AT&T IP Flexible Reach service with Business in a Box (SM) only supports **MIS/PNT** transport connection.

The AT&T IP Flexible Reach with Business in a Box service is one of several SIP-based Voice over IP (VoIP) services offered to enterprises for a variety of voice communications needs. The AT&T IP Flexible Reach service allows enterprises in the U.S.A. to place outbound local and long distance calls, receive inbound Direct Inward Dialing (DID) calls from the PSTN, and place calls between an enterprise's sites.

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 (IP Office) with the AT&T IP Flexible Reach with Business in a Box service. AT&T IP Flexible Reach service with Business in a Box (SM) only supports MIS/PNT transport connection.

The AT&T IP Flexible Reach Business in a Box service, referred to AT&T IP Flexible Reach from hereon, is one of several SIP-based Voice over IP (VoIP) services offered to enterprises for a variety of voice communications needs. The AT&T IP Flexible Reach service allows enterprises in the U.S.A. to place outbound local and long distance calls, receive inbound Direct Inward Dialing (DID) calls from the PSTN, and place calls between an enterprise's sites.

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 Flex Reach 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 and Outbound calls between AT&T IP Flex Reach service and IP Office endpoints.
- Call and two-way talk path establishment between PSTN and IP Office phones via the AT&T Flex Reach service.
- Basic supplementary telephony features such as hold, resume, transfer, and conference.
- G.729 and G.711 codecs.
- T.38 and G.711 fax calls from AT&T IP Flex Reach service/PSTN to Avaya IP Office G3 and SG3 fax endpoints.
- DTMF tone transmission using RFC 2833 between IP Office and the AT&T IP Flex Reach service/PSTN automated access systems.
- Inbound AT&T IP Flex Reach 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 Flexible Reach service.

This 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 to and 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 and G.711 Fax
- Passing of DTMF events and their recognition by navigating automated menus
- PBX features such as hold, resume, conference and transfer
- Call redirection with Diversion Header

2.2. Known Limitations

1. G.711 faxing is not supported between IP Office and the AT&T IP Flexible Reach service. T.38 faxing is supported for both Group 3 and Super Group 3 fax.
2. **Emergency 911/E911 Services Limitations and Restrictions** - Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer configuration guide will properly operate with AT&T IP Flexible Reach to complete 911/E911 calls; therefore, it is Customer's responsibility to ensure proper operation with its equipment/software vendor.

   While AT&T IP Flexible Reach services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when that E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found at http://new.serviceguide.att.com. Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the Customer's location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Guide in detail to understand the limitations and restrictions.
3. Shuffling is not supported for SIP trunks in Avaya IPO 7.0
4. **G**.722 codec is not supported by Avaya IP Office 7.0.
5. Avaya IP Office Softphone should be configured in high bandwidth mode otherwise after resuming the call back from hold for an inbound call, audio is lost from Avaya IP Office to PSTN.

2.3. Support

AT&T customers may obtain support for the AT&T IP Flexible Reach service by calling (877) 288-8362.

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 V2. This solution is extensible to other Avaya IP Office hardware too.
- Avaya "desk" phones are represented with Avaya 1616I, 4625 and 9630 IP Telephones running H.323 software, Avaya Digital Phones (1416, T7100 and 7316E), 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.
- Outbound calls are originated from a phone or fax provisioned on IP Office. Signaling is between IP Office and the AT&T SIP Trunk IP Address.
- 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.

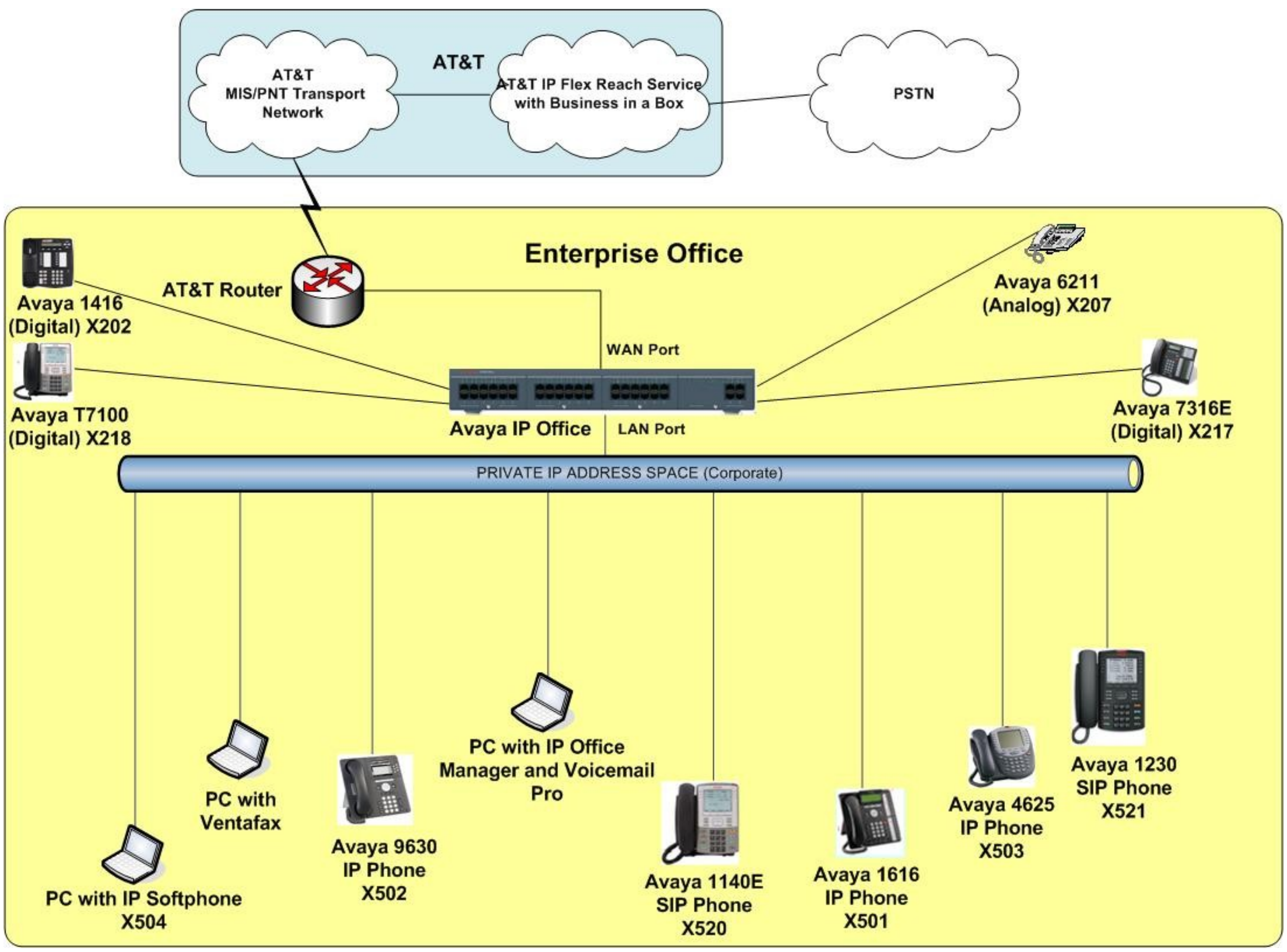


Figure 1: Reference configuration

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 Flexible Reach service SIP Trunk IP address shown in this document is an example. AT&T Customer Care will provide the actual IP addresses as part of the AT&T IP Flexible Reach 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	207 = Analog 501,502,503=H323 202,217,218=Digital 504=Softphone 520,521= SIP phones
AT&T IP Flexible Reach Service	
SIP Trunk IP Address	135.242.225.210

Table 1: Illustrative Values Used in these Application Notes

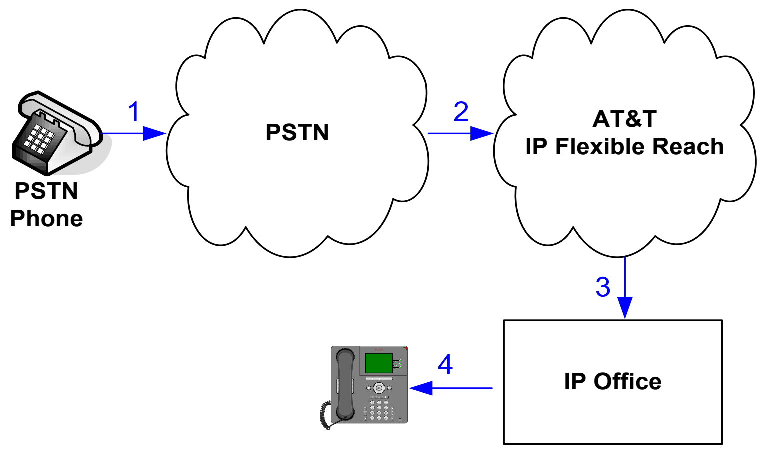
3.2. Call Flows

To understand how inbound and outbound AT&T IP Flexible Reach service calls are handled by IP Office, four 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 Flexible Reach 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 Flexible Reach service number.
2. The PSTN routes the call to the AT&T IP Flexible Reach service network.
3. The AT&T IP Flexible Reach service routes the call to IP Office.
4. IP Office applies any necessary digit manipulations based upon the DID and routes the call to a hunt group, phone or a fax.

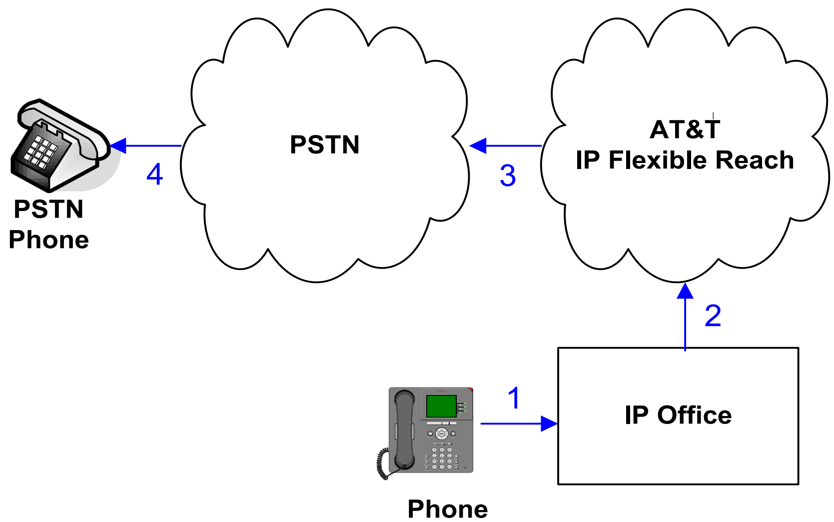


Inbound - AT&T IP Flexible Reach

3.2.2. Outbound

The second call scenario illustrated in the figure below is an outbound call initiated on IP Office for delivery to AT&T IP Flexible Reach service.

1. IP Office phone or fax originates a call to an AT&T IP Flexible Reach service number for delivery to PSTN.
2. IP Office applies any necessary origination treatment (verifying permissions, determining the proper route, selecting the outgoing trunk, etc.) and routes the call to AT&T IP Flexible Reach service.
3. The AT&T IP Flexible Reach service delivers the call to PSTN.
4. PSTN delivers the call to a phone or fax.



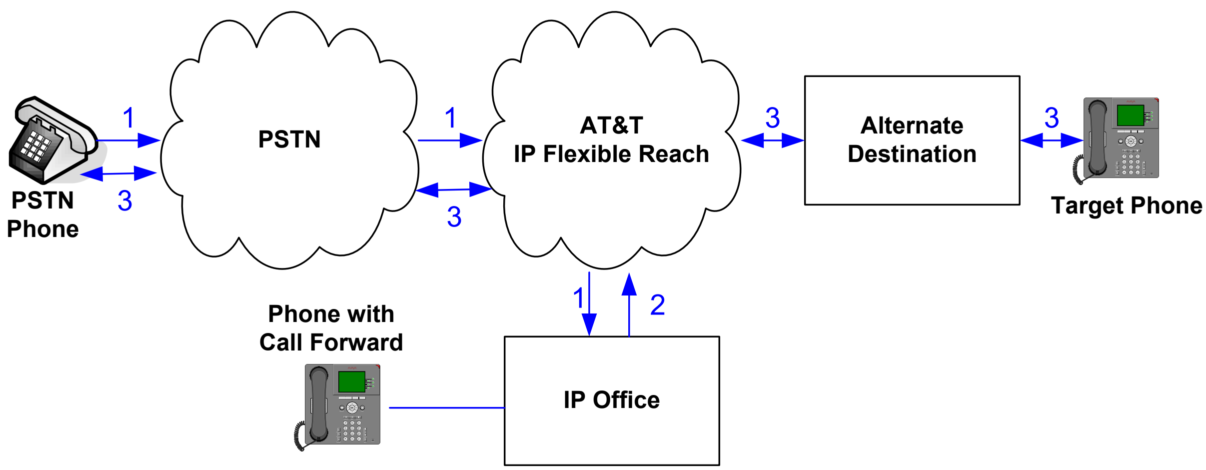
Outbound - AT&T IP Flexible Reach Service

3.2.3. Call Forward Re-direction

The third call scenario illustrated in the figure below is an inbound AT&T IP Flexible Reach service call that arrives IP Office and it routes the call to a destination station, however the station has set Call Forwarding to an alternate destination. Without answering the call, IP Office immediately redirects the call back to the AT&T IP Flexible Reach service for routing to the alternate destination.

Note –AT&T requires the diversion header when a call is redirected to ATT IP Flexible Reach service telephone number. (see **Section 5.4.1**).

1. Same as the first call scenario in **Section 3.2.1**.
2. Because the IP Office phone has set Call Forward to another AT&T IP Flexible Reach service number, IP Office initiates a new call back out to the AT&T IP Flexible Reach service network.
3. The AT&T IP Flexible Reach service places a call to the alternate destination and upon answer, IP Office connects the calling party (PSTN Phone) to the target party (Target Phone).

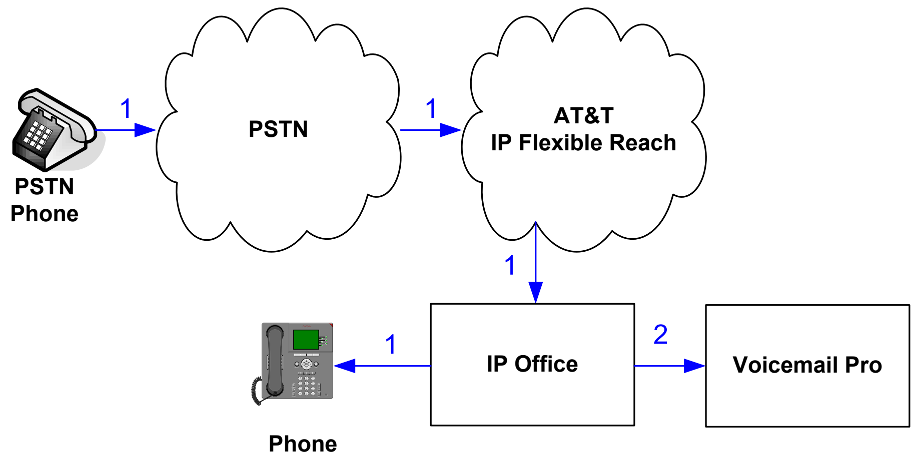


Re-directed (e.g. Call Forward) - AT&T IP Flexible Reach Call

3.2.4. 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.

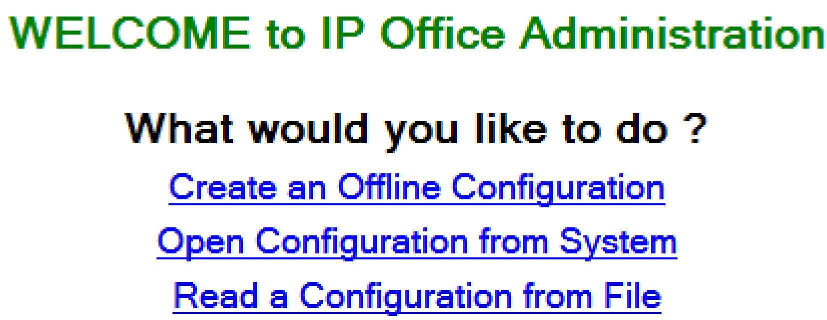
Component	Version
Avaya IP Office 500 V2	Release 7.0 (5) (Preferred Edition)
Avaya IP Office Manager	Release 9.0 (5) (Preferred Edition)
Avaya IP Office Voicemail Pro	Release 7.0 (17)
Avaya IP Office Voicemail Pro Client	Version 7.0 (17)
Avaya 1616IP-Series Telephones (H.323)	Release 1.3
Avaya 9630 IP 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 T7100 Digital Phone	-
Avaya 7316E Digital Phone	-
Avaya 6211 Analog phone	-
Avaya 1140E SIP Telephone	04.00.13.00 (SIP1140)
Avaya 1230 SIP Telephone	04.00.13.00 (SIP1230)
Fax device	Ventafax Home Version 6.2
AT&T IP Flexible Reach Service using MIS/PNT transport service connections.	VNI 18

Table 2: Equipment and Software Versions

5. Avaya IP Office

This section describes attributes of the reference configuration, but is not meant to be prescriptive. The configuration steps described here are only for the fields where a value was changed. For all the other fields default values are used. Additionally, the screen shots referenced in these section may not be complete at times. Consult reference [IPO-INSTALL] for more information on the topics in this section.

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:

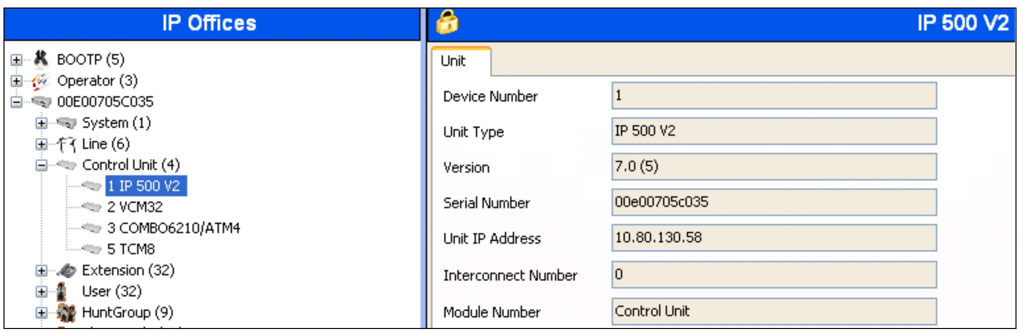


Open the IP Office configuration, either by reading the configuration from the IP Office server, or from file. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left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

5.1. Physical, Network, and Security Configuration

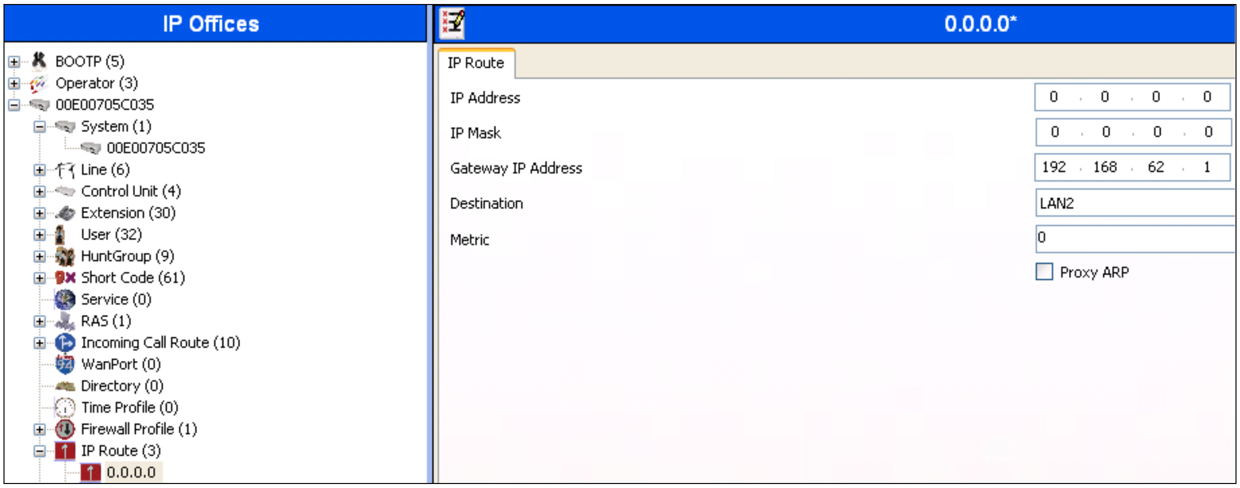
In the reference configuration, the IP Office 500 V2 contains a VCM32 module, COMBO6210/ATM4 module, and a TCM8 module. The VCM32 is a Voice Compression Module supporting VoIP codecs. The COMBO6210/ATM4 was used in this reference configuration to support digital and analog telephones or fax machines. The TCM8 module was used to support heritage Avaya/Nortel 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 V2** is selected in the Navigation pane, revealing additional information about the IP 500 V2 in the Details pane.

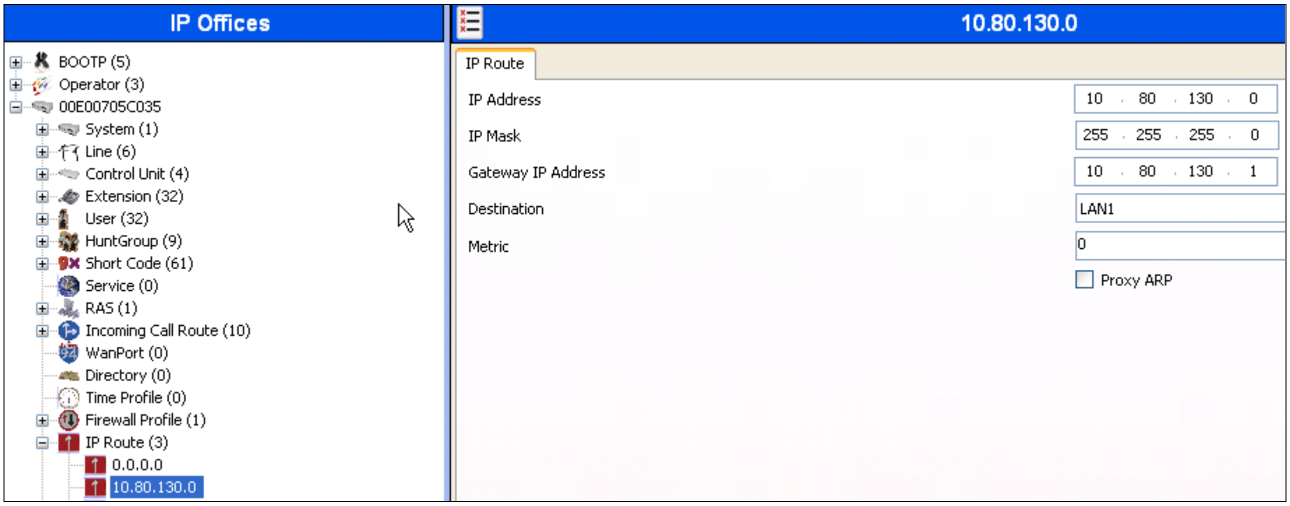


In this reference configuration, the IP Office **LAN2** port (labeled as WAN port in Figure 1) is physically connected to the public network at the IP Office customer site. The default gateway for this network is **192.168.62.1**.

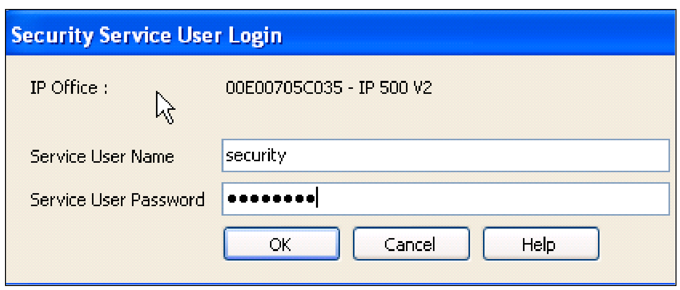
1. To add an IP Route in IP Office, right-click **IP Route** from the Navigation pane, and select **New** [not shown]. To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the Details pane with the relevant default route using **LAN2** as configured in **Destination** field (Refer **Section 5.3.2**).


IP Offices
BOOTP (5)
Operator (3)
00E00705C035
System (1)
00E00705C035
Line (6)
Control Unit (4)
Extension (30)
User (32)
HuntGroup (9)
Short Code (61)
Service (0)
RAS (1)
Incoming Call Route (10)
WanPort (0)
Directory (0)
Time Profile (0)
Firewall Profile (1)
IP Route (3)
0.0.0.0
0.0.0.0*
IP Route
IP Address
0 . 0 . 0 . 0
IP Mask
0 . 0 . 0 . 0
Gateway IP Address
192 . 168 . 62 . 1
Destination
LAN2
Metric
0
Proxy ARP

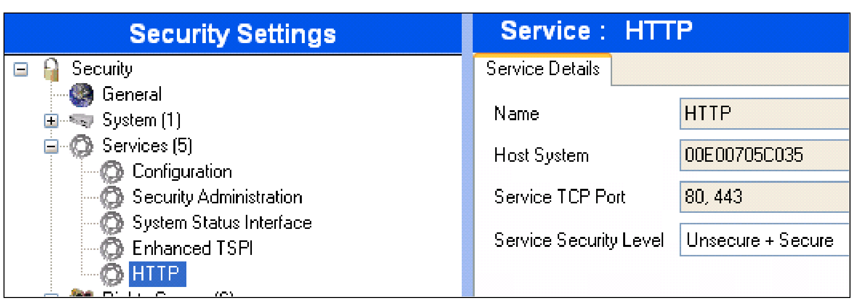
2. Another route for **10.80.130.0** subnet was added for the enterprise side **LAN1** port (labeled as LAN port in Figure 1) as shown in the screen below. All the IP devices were part of this 10.80.130.x network in this reference configuration.



3. For use of Avaya IP Office Softphone, navigate to **File → Advanced → Security Settings** and login with proper credentials in the screen shown below.



4. After logging in, navigate to **Services → HTTP** and verify that **Service Security Level** field is set to **Unsecure + Secure**. Note that this action may be service disrupting.

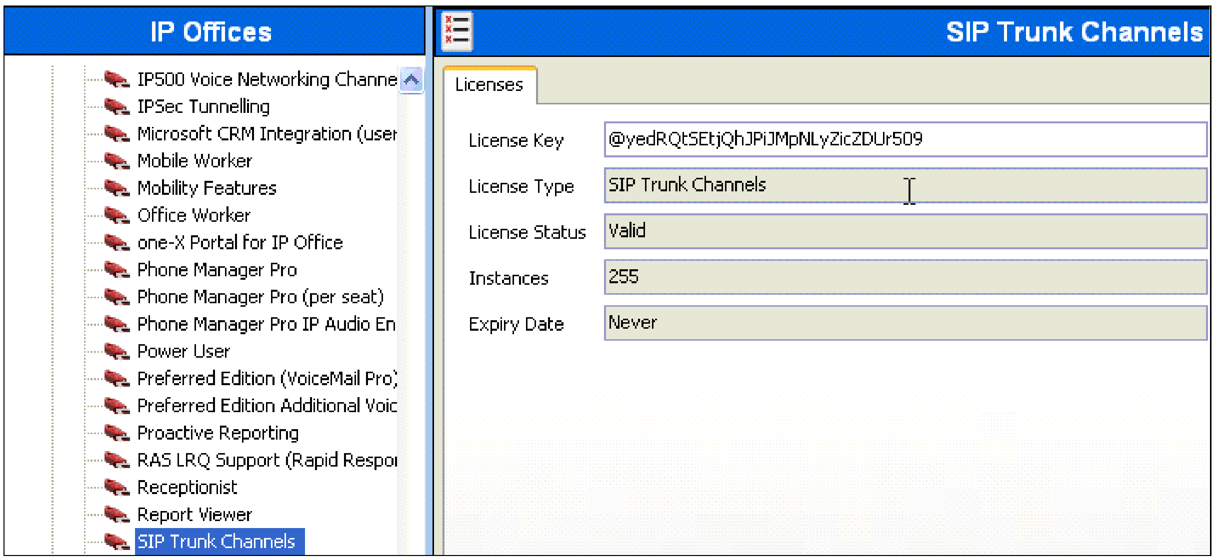


5. When complete, select **File → Configuration** to return to configuration activities.

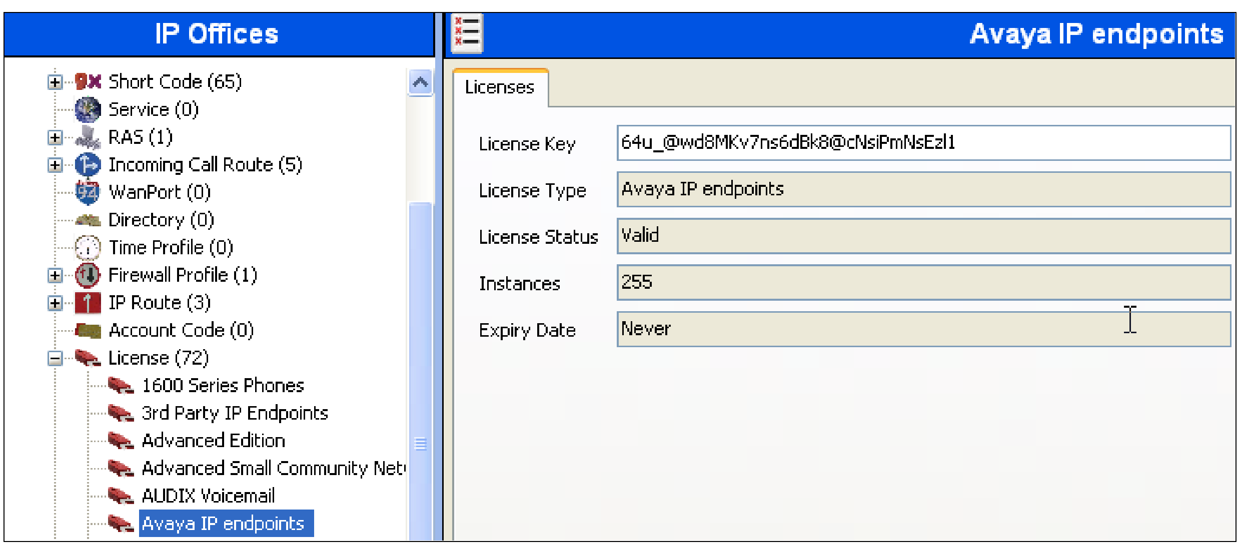
5.2. Licensing

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.

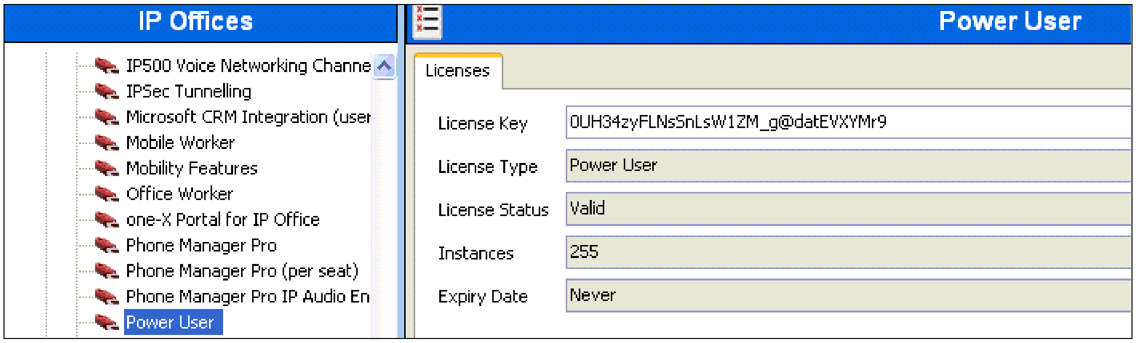
1. To verify that SIP Trunk Channels has sufficient capacity, navigate to **License → SIP Trunk Channels** in the Navigation pane and confirm a valid license with sufficient **Instances** (trunk channels) exist in the Details pane.



2. To verify Avaya IP endpoints with sufficient capacity, navigate to **License → Avaya IP endpoints** in the Navigation pane and confirm a valid license with sufficient **Instances** exist in the Details pane.



3. The following screen shows the availability of a valid license for **Power User** features. In this reference configuration, the user with extension **501** (**Section 5.5.2**) is configured as a **Power User** and is capable of using the IP Office Softphone too.

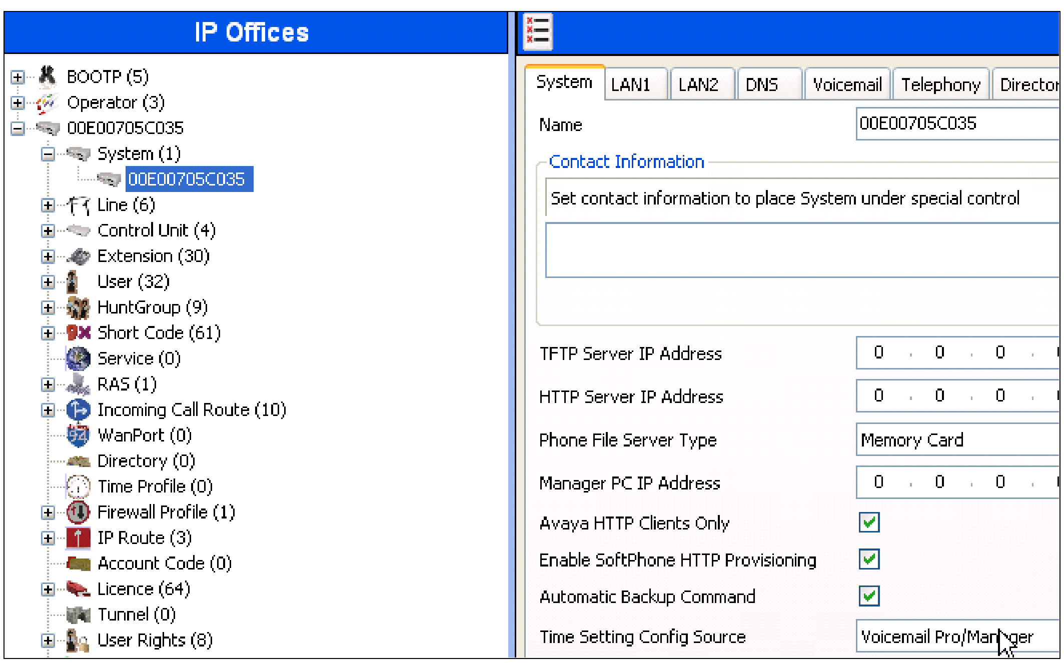


5.3. System Settings

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

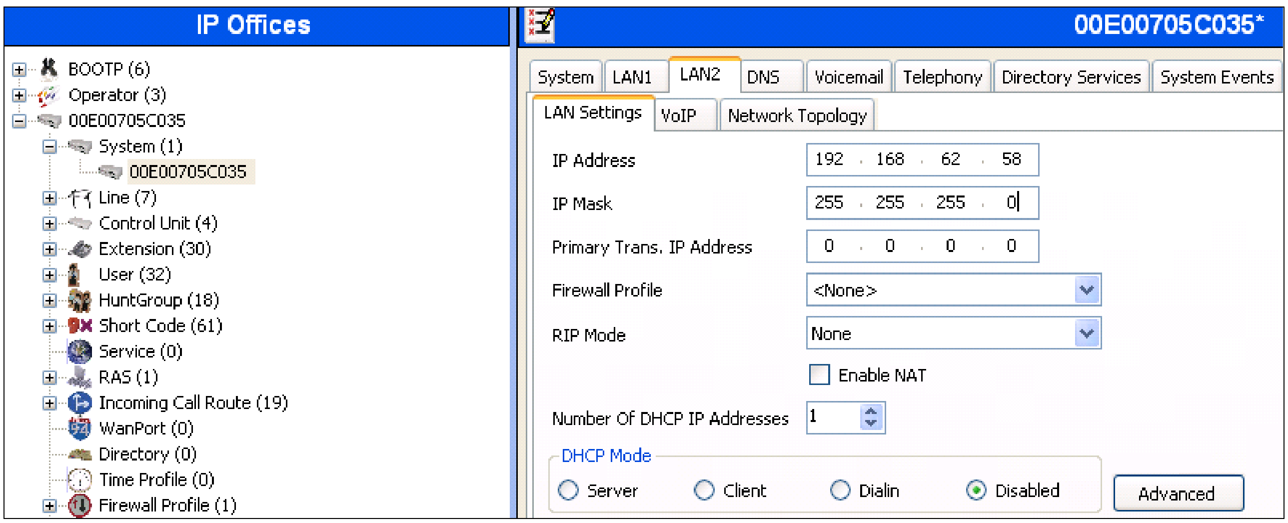
With the proper system name selected in the Group pane, select the **System** tab in the Details pane. The following screen shows a portion of the **System** tab. The **Name** field is used for a descriptive name of the system. In this case, the MAC address is used as the name. The **Avaya HTTP Clients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



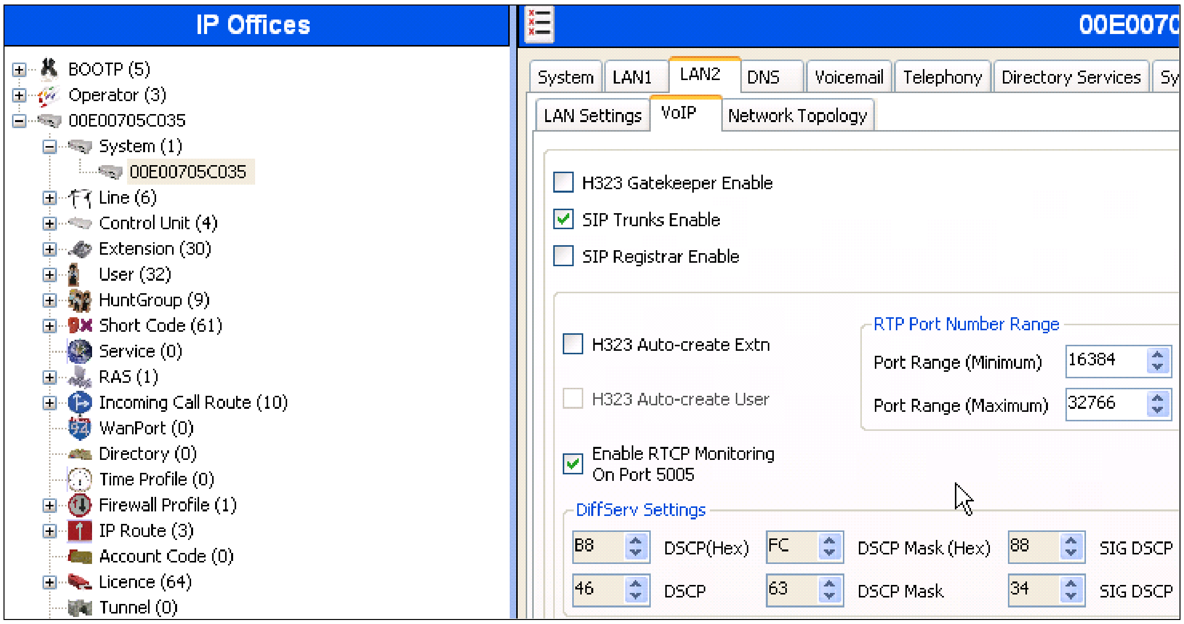
5.3.2. LAN Settings

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

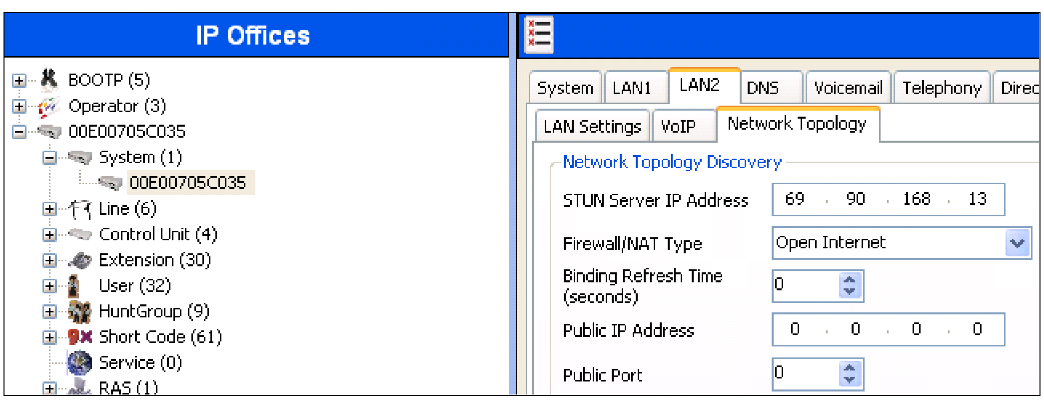
1. Select the **LAN2** tab followed by the **LAN Settings** tab and configure as follows:
   - **IP Address** – Set to **192.168.62.58** which is the IP address of IP Office known to AT&T network
   - **IP Mask** – Set to a valid value e.g **255.255.255.0**
   - **Primary Trans**. **IP Address** – Set to **0.0.0.0**
   - **DHCP Mode** – Select the **Disabled** radio button



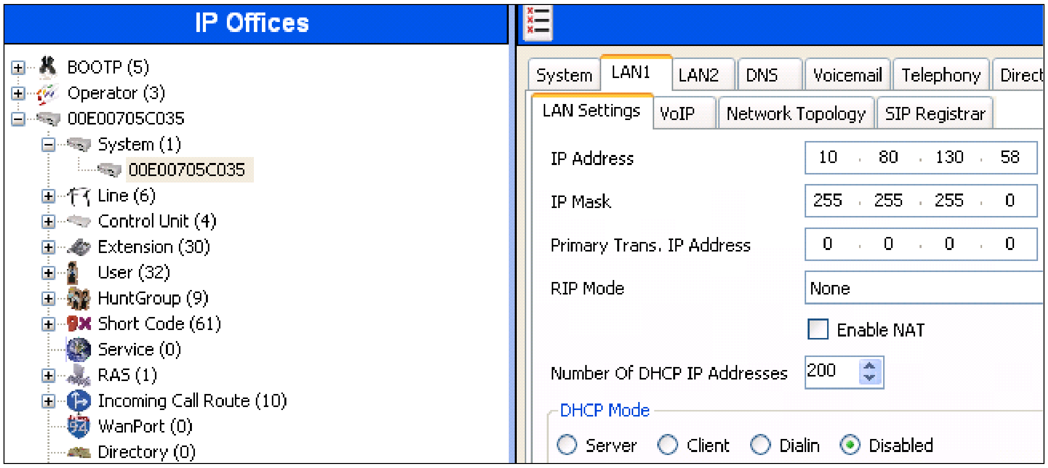
2. Select the **VoIP** tab as shown in the following screen and configure as follows:
   - **SIP Trunks Enable** – Check this box to enable the configuration of SIP trunks
   - **RTP Port Range (Minimum)** – Set to **16384** (As required by AT&T)
   - **RTP Port Range (Maximum)** – Set to **32766** (As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



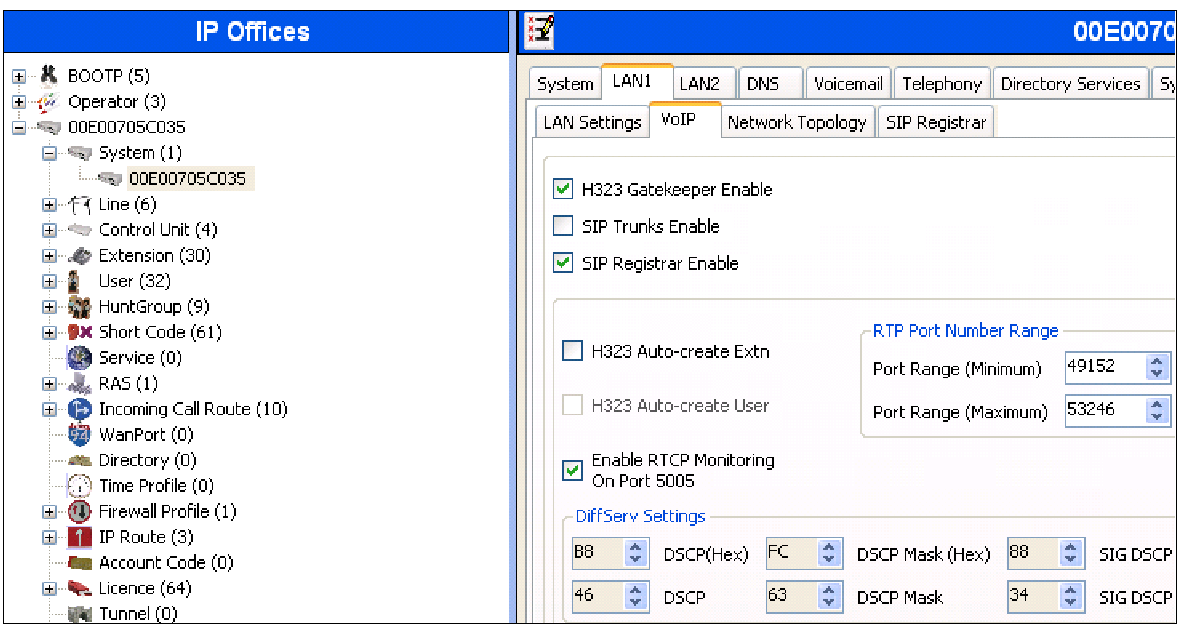
3. Select the **Network Topology** tab as shown in the following screen and set **Firewall/NAT Type** field to **Open Internet**. With this configuration, STUN will not be used but make sure to leave **STUN Server IP Address** to its default value.



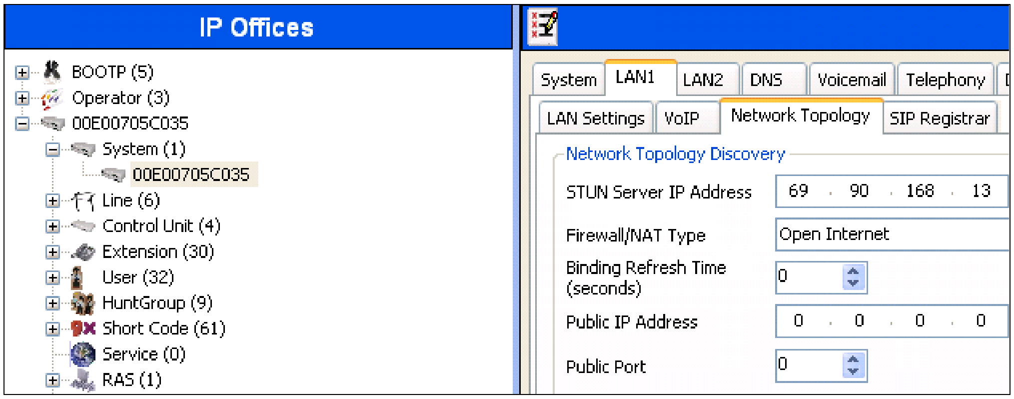
4. Select the **LAN1** tab followed by the **LAN Settings** tab and set **IP Address** of the IP Office on the enterprise side to **10.80.130.58** and **IP Mask** to **255.255.255.0**. Check the **DHCP Mode – Disabled** box.



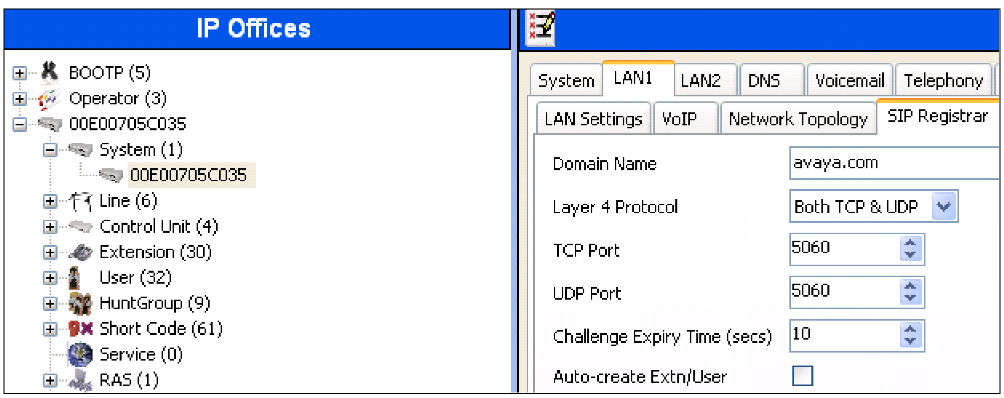
5. Select the **VoIP** tab as shown in the following screen and configure as follows:
- **H323 Gatekeeper Enable** – Check this box to allow the use of Avaya IP Phones
- **SIP Registrar Enable** – Check this box to allow SIP phones and IP Office Softphone usage



6. The Network Topology screen is set the same as it was set in **Step 3**.



7. Select the SIP Registrar tab and 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.

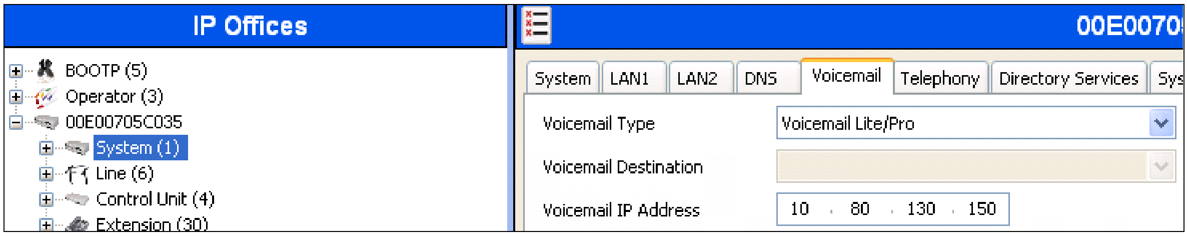


8. Click **OK** [not shown] to commit.

5.3.3. Voicemail

Select **Voicemail** tab and configure as follows:

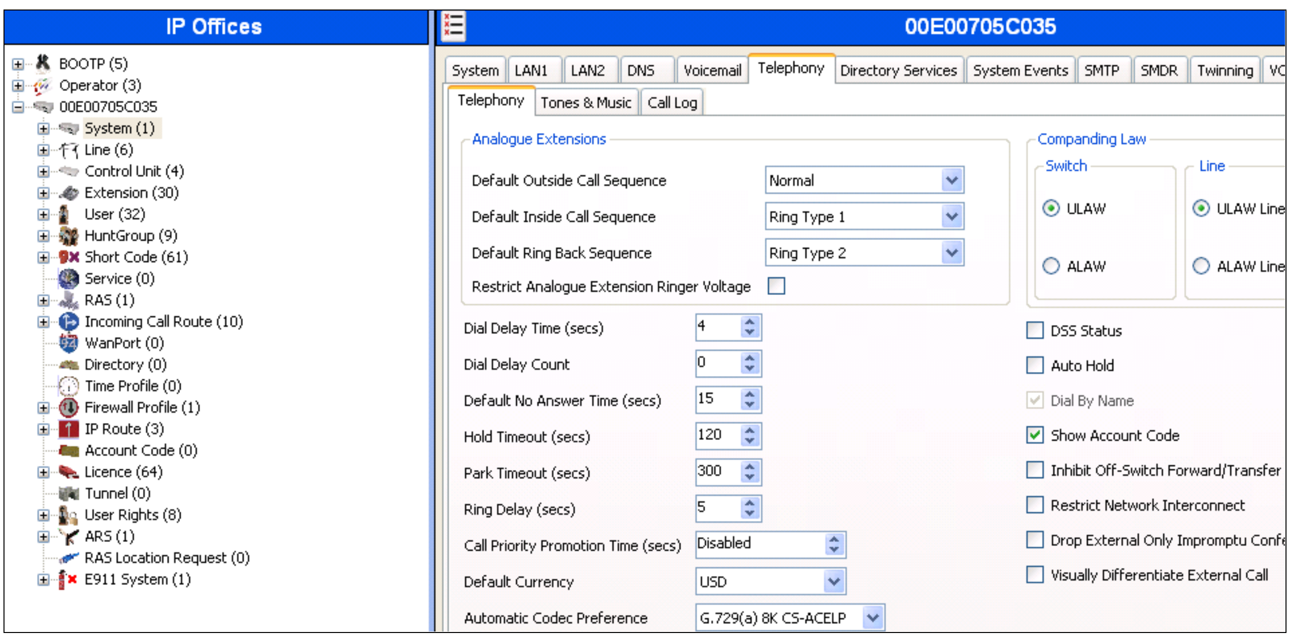
- **Voicemail Type** – Set to **Voicemail Lite/Pro** from the drop-down list
- **Voicemail IP Address** – Set to **10.80.130.150**, the IP Address of the PC running the Voicemail Pro software.



5.3.4. System Telephony Configuration

Select the **Telephony** tab and in **Telephony** sub-tab 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 Flex Reach service can be tested.
- **Automatic Codec Preference** – Use the default value of **G.729(a) 8K CS-ACELP**.



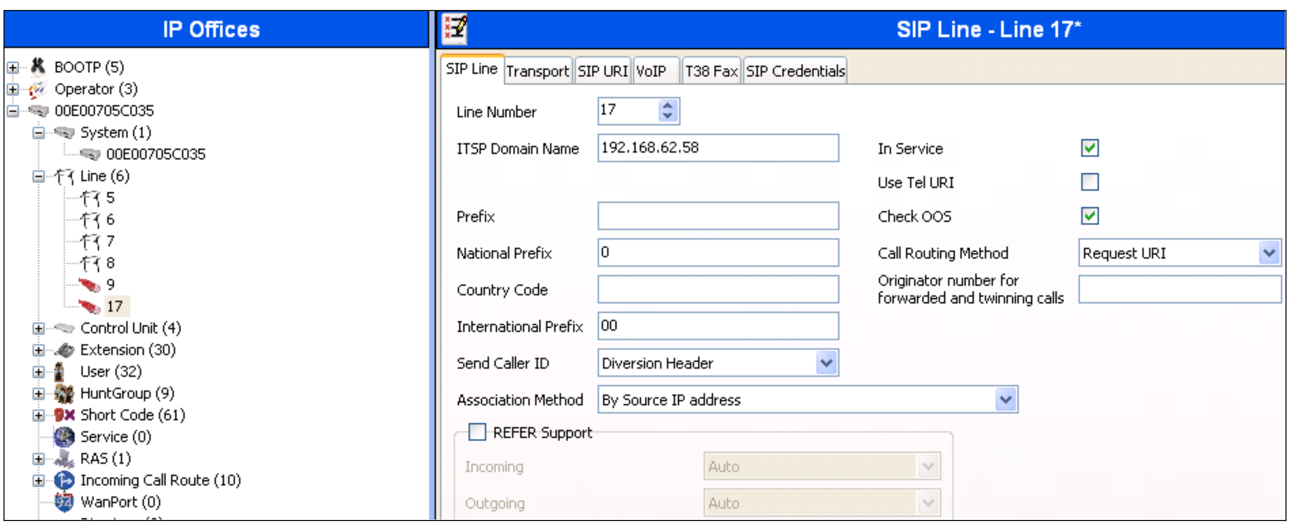
5.4. SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 7.0. To add a new SIP Line, right click on **Line** in the Navigation pane, and select **New → SIP Line** [not shown]. A new Line Number is assigned automatically..

5.4.1. SIP Line - SIP Line Tab

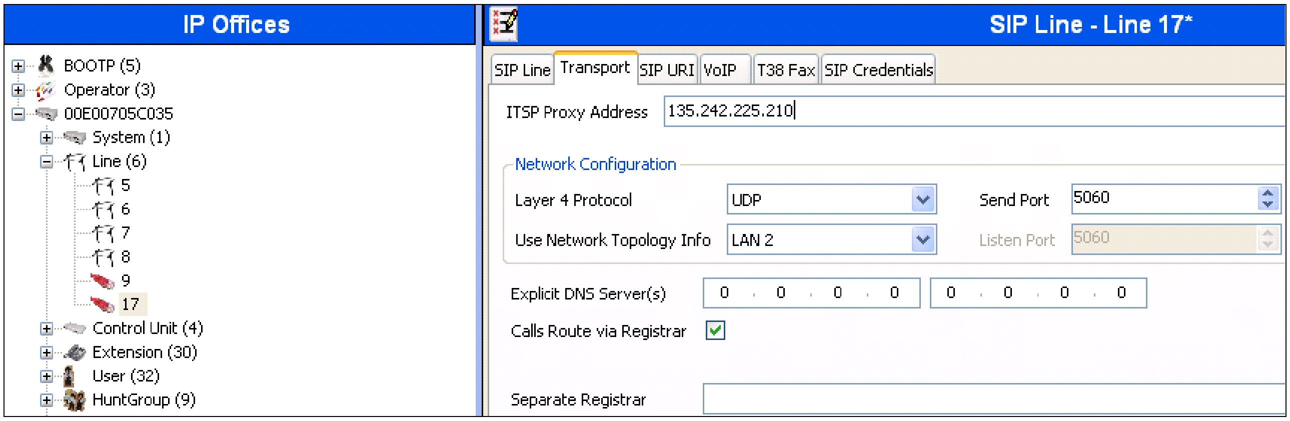
Select **SIP Line** tab as shown below for Line Number 17 used for AT&T and configure as follows:

- **ITSP Domain Name** – Set to the IP Office LAN1 address (**192.168.62.58**) configured in **Section 5.3.2**, **Step 1** 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** – 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.
- **Send Caller ID** - Select **Diversion Header** from the drop-down list which will ensure that in case alternate destination is N11, NPA-555-1212, or 8xx number, the call can be properly redirected by AT&T Flexible Reach service by inspecting the SIP Diversion header.
- **Refer Support** – Uncheck this box.
- **Call Routing Method** – Set to **Request URI** (default).



5.4.2. SIP Line - Transport Tab

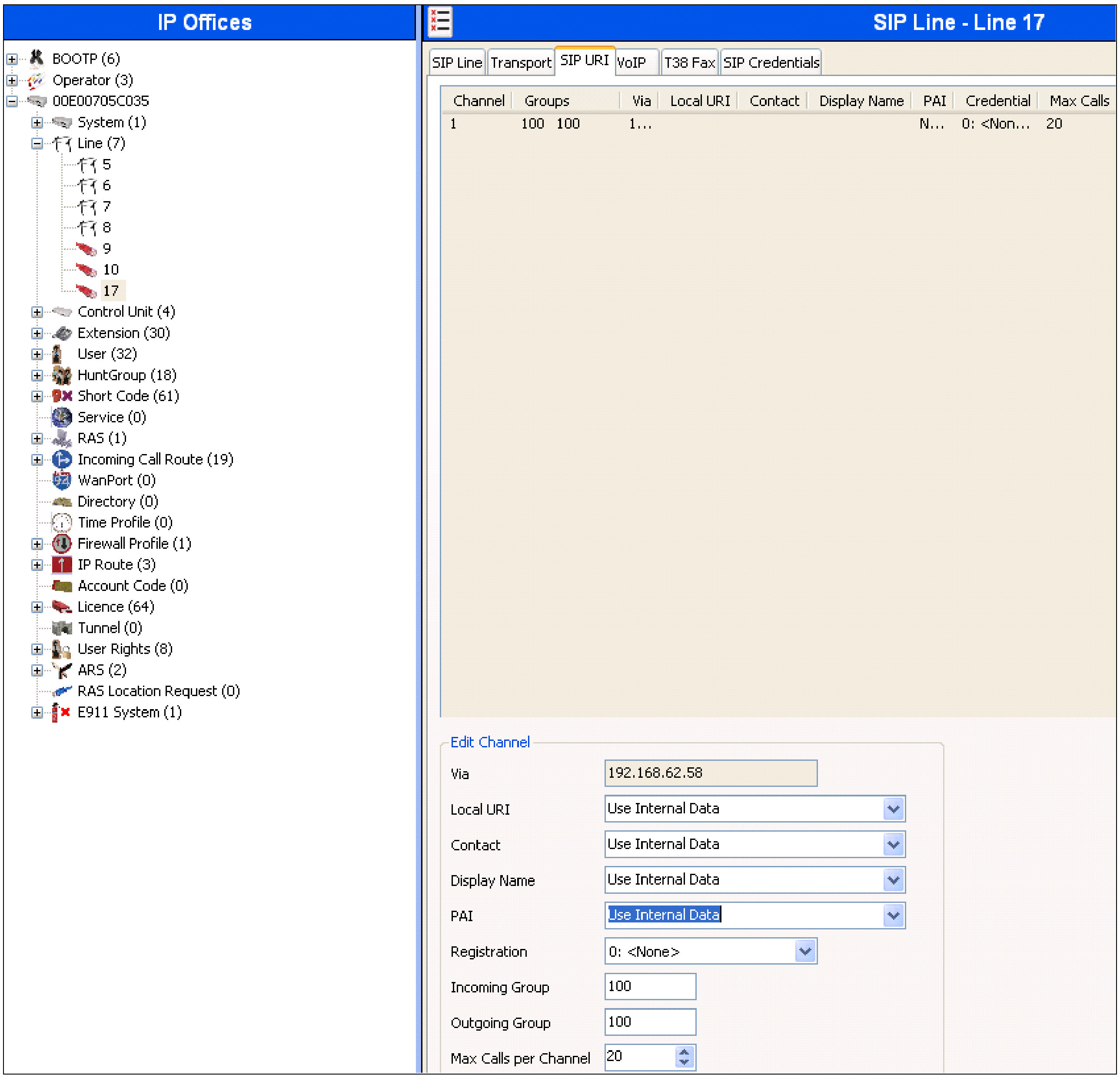
Select the **Transport** tab and set the **ITSP Proxy Address** to the AT&T SIP Trunk IP Address. The **Use Network Topology Info** parameter is set to **LAN 2** which is configured in **Section 5.3.2**. Default values are used for the other fields.



5.4.3. SIP Line - SIP URI Tab

Select the **SIP URI** tab and click the **Add…** button [not shown] to add a new SIP URI. Configure the **New Channel** section displayed as follows:

- **Local URI, Contact, Display Name** and **PAI** – Set all these fields to **Use Internal Data**
- **Registration** - Set to **0: <None>**
- **Incoming Group** and **Outgoing Group** – Set to **100**

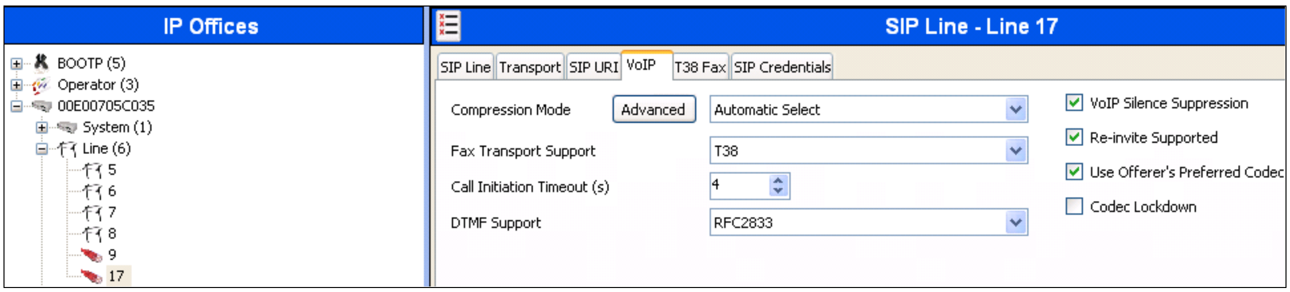


In this reference configuration, the single SIP URI shown above was sufficient to allow incoming calls for AT&T DID numbers destined for specific IP Office users via IP Office hunt groups.

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.
- **Fax Transport Support** – Select **T38** from the drop-down list.
- **DTMF Support** - Set to the default value **RFC2833**.
- **VOIP Silence Suppression** – This box is checked as AT&T Flexible Reach service requires G729b as a preferred codec.
- **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.



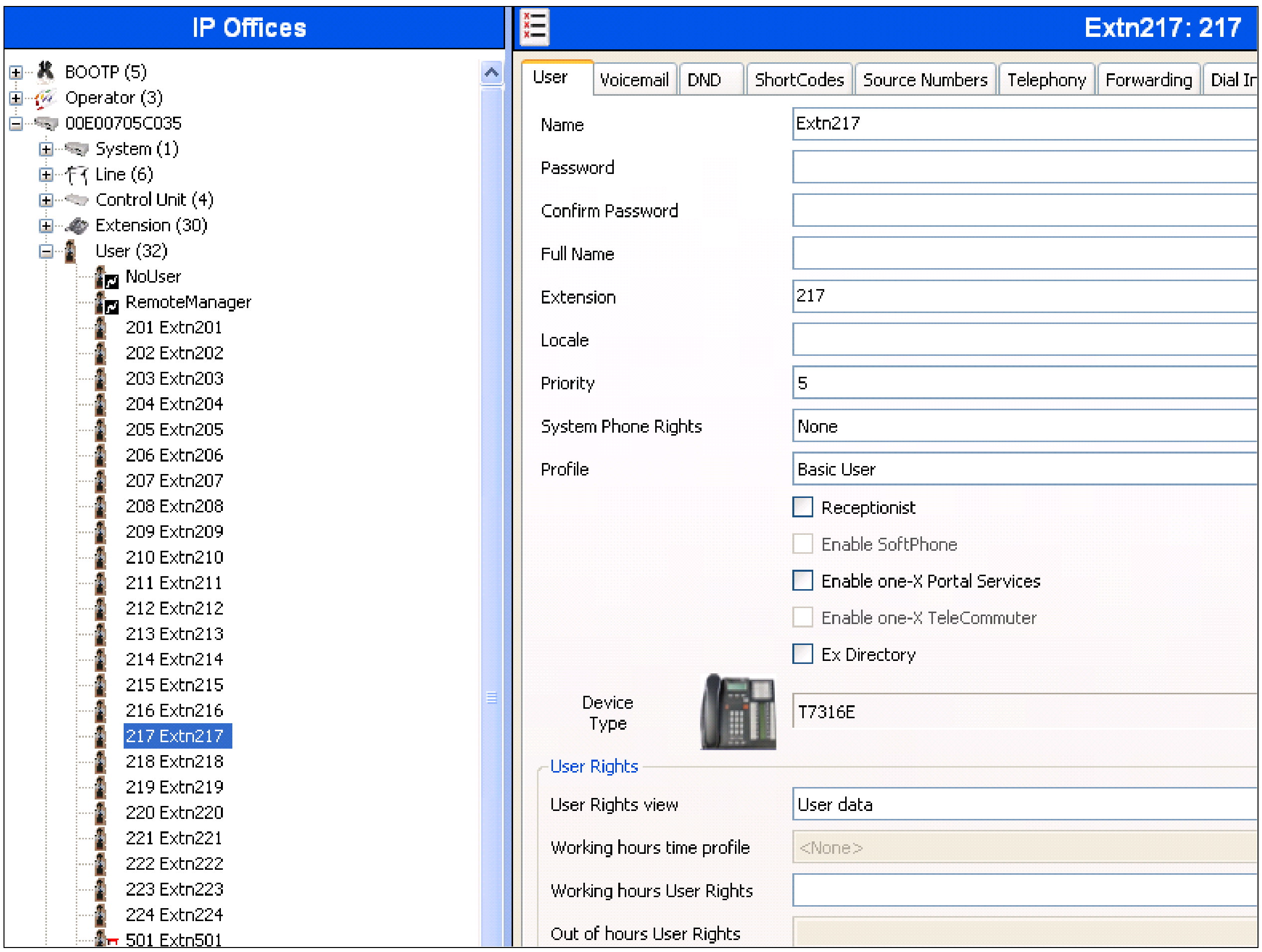
Since default values were used for T38 fax and AT&T IP Flexible Reach does not require registration, the **T38 Fax** and **SIP Credentials** tabs need not be visited. Click **OK** (not shown) to commit the SIP Line configuration.

5.5. Users, Extensions, and Hunt Groups

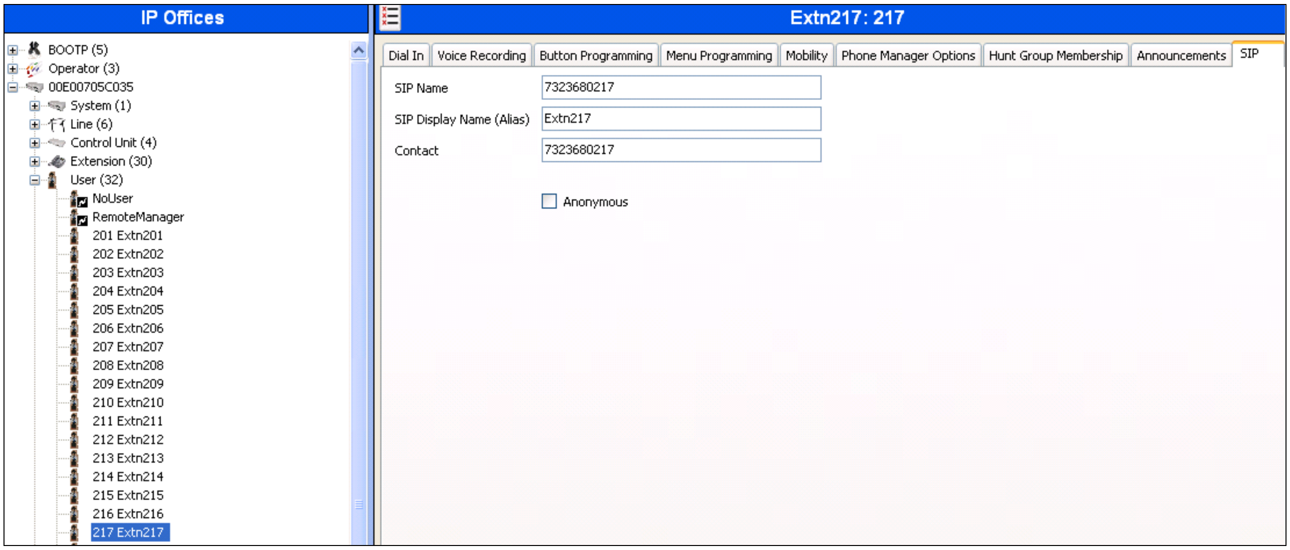
In this section, examples of IP Office Users, Extensions, and Hunt Groups will be illustrated. In the interests of brevity, not all users and extensions used will be presented, since the configuration can be easily extrapolated to other users. To add a User, right click on **User** in the Navigation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

5.5.1. Digital Telephone User 217

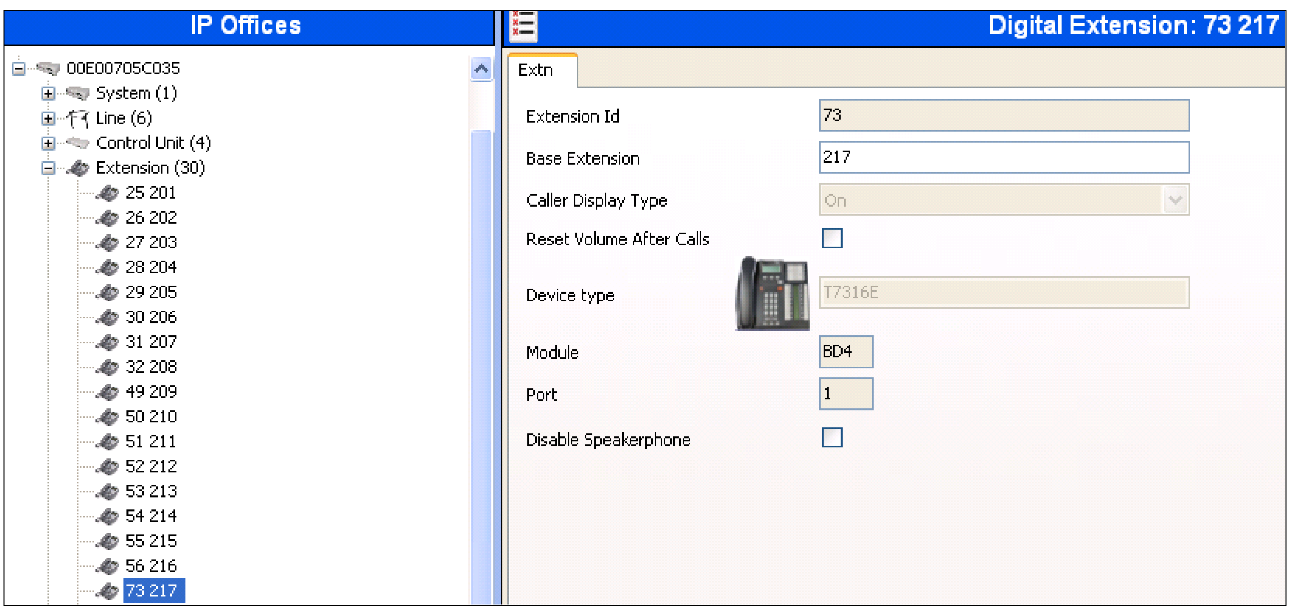
The following screen shows the **User** tab for User **217**. This user corresponds to a digital phone.



The following screen shows the **SIP** tab for User **217**. In this sample configuration, the **SIP Name** and **Contact** parameters are the user part of the SIP URI in the From header for outgoing SIP trunk calls only. The **SIP Display Name (Alias)** parameter is configured with any descriptive name. If all outgoing calls involving this user should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network. See **Section 5.6.1** for an alternate method of using a short code (rather than static user provisioning) to place an anonymous call.

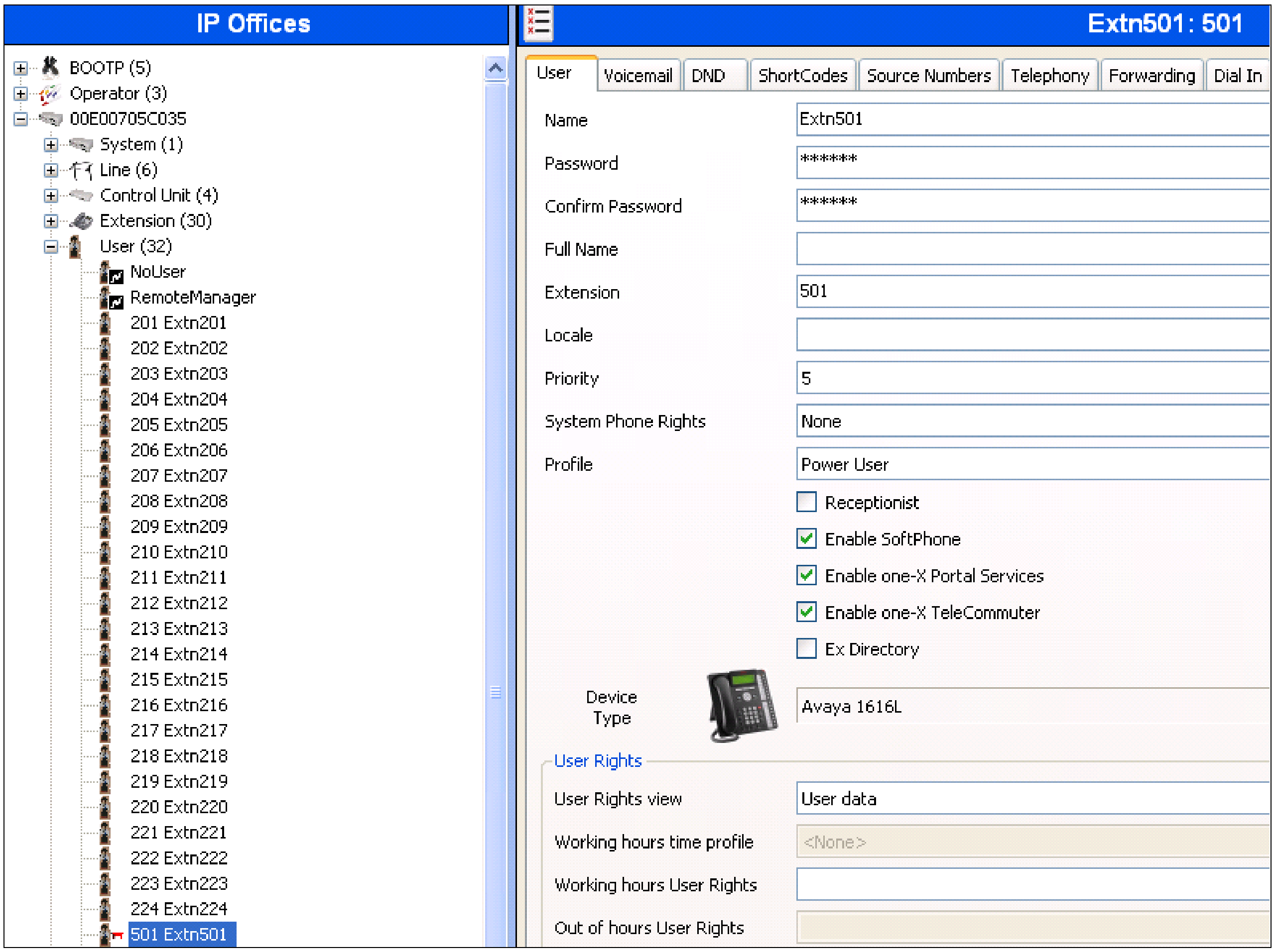


The following screen shows the Extension information for this user. To view, select **Extension** and the appropriate extension in the Navigation pane.

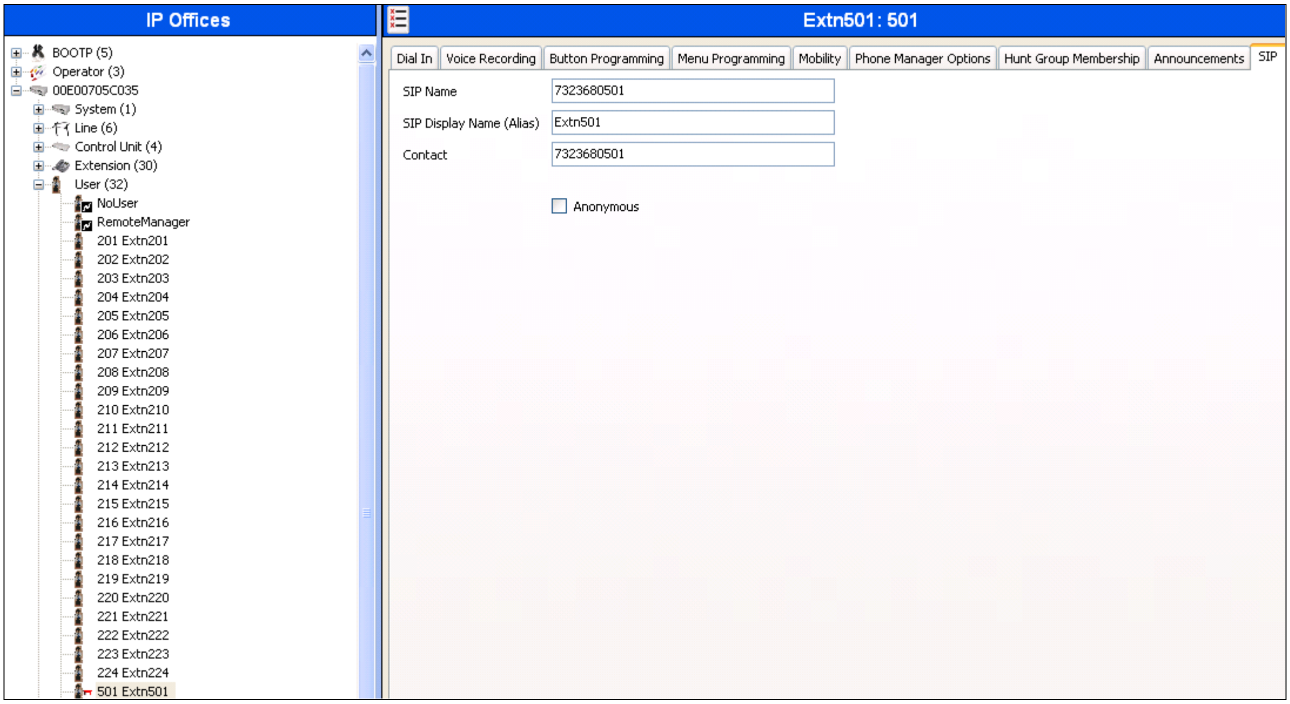


5.5.2. IP Telephone User 501

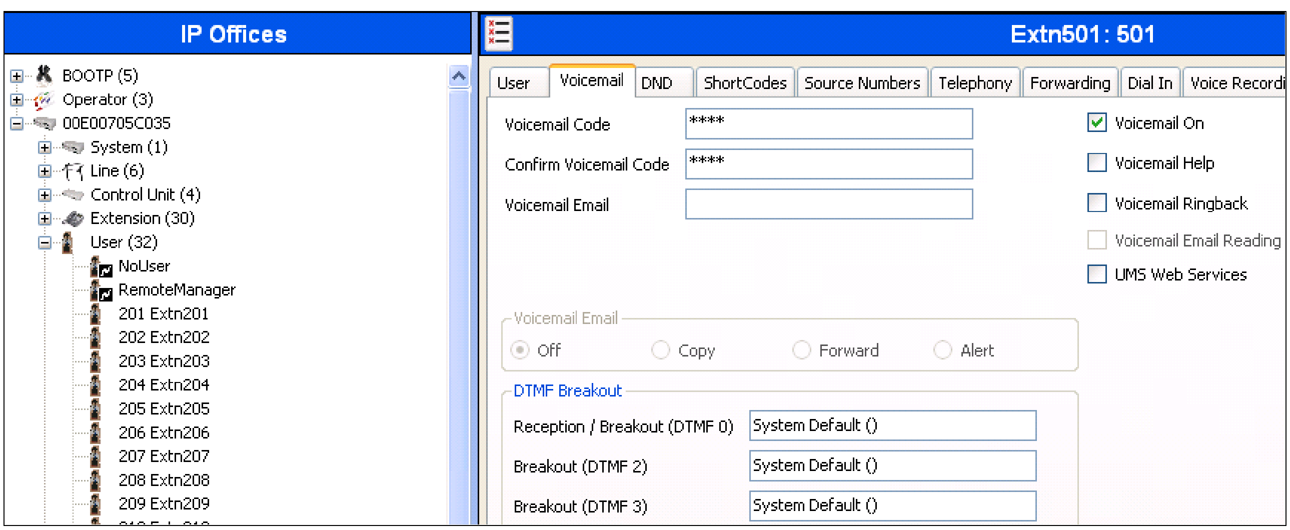
The following screen shows the **User** tab for User **501**. This user corresponds to an Avaya 1616 IP Telephone that is configured as power user with IP Office Softphone features enabled as shown below.



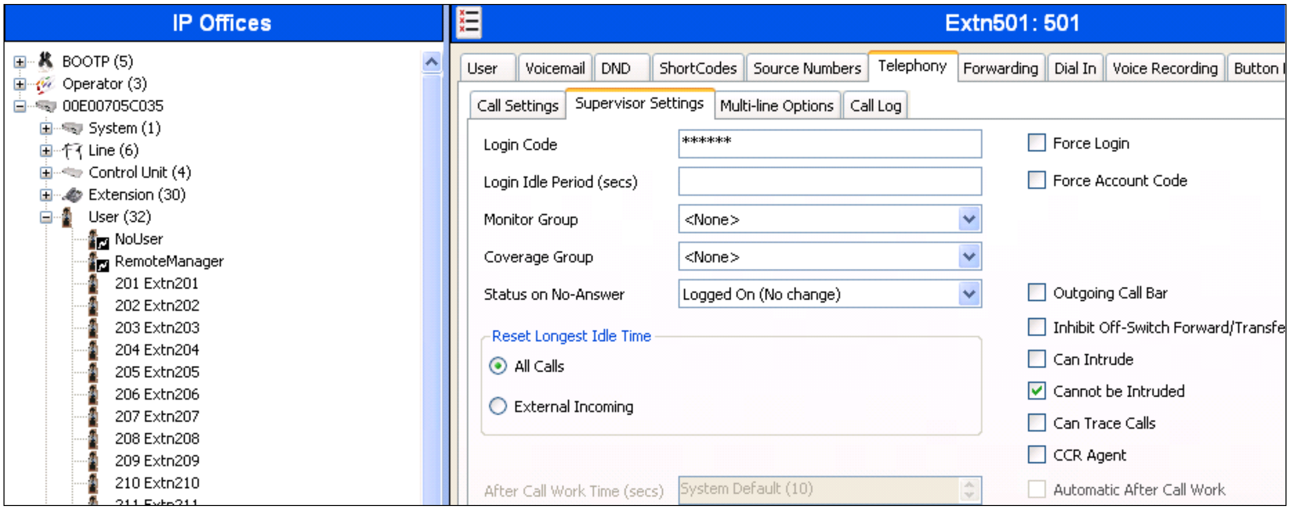
Like the user with extension **217**, the **SIP** tab for the user with extension **501** is configured with a **SIP Name, SIP Display Name (Alias)** and **Contact**.



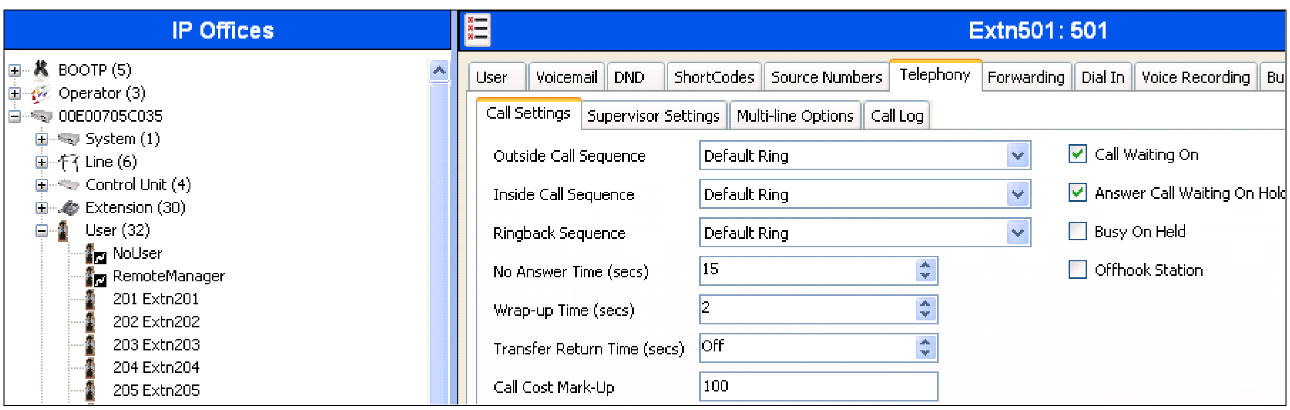
The following screen shows the **Voicemail** tab for this user. The **Voicemail On** box is checked, and a voicemail password can be configured in the **Voicemail Code** and **Confirm Voicemail Code** fields.



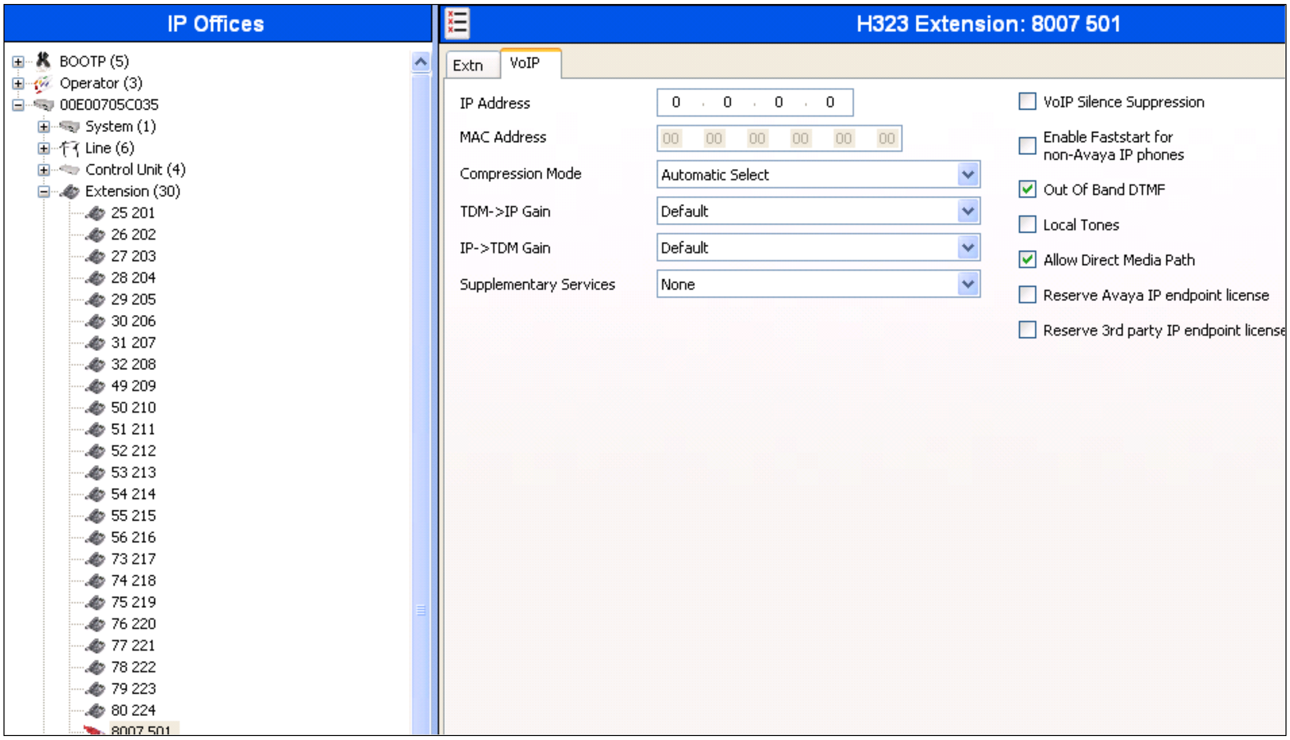
Select the **Telephony** tab and **Supervisor Settings** sub-tab as shown below. To allow hot desking, enter a **Login Code**.



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 (necessary for call transfer).



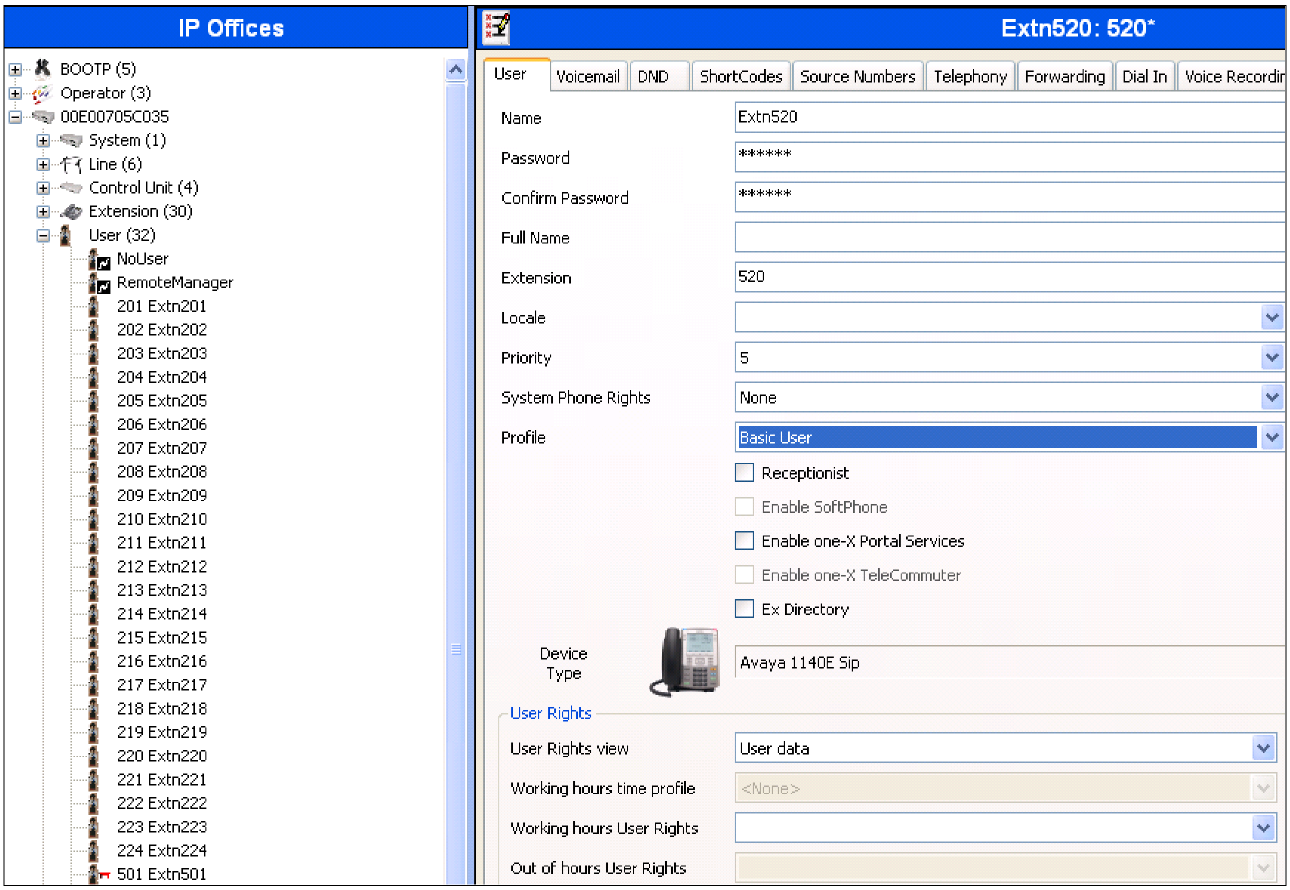
The following screen shows the Extension information for this user, simply to illustrate the **VoIP** tab available for an IP Telephone.



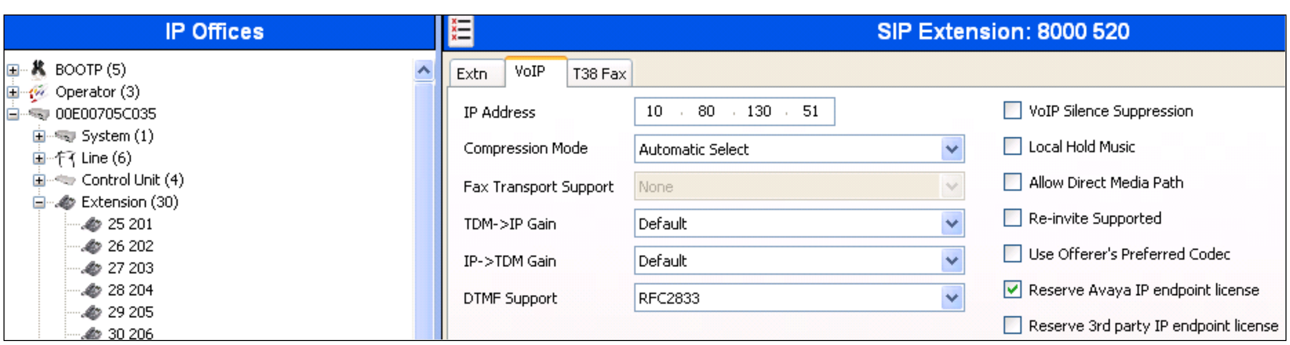
Solution & Interoperability Test Lab Application Notes

5.5.3. SIP Telephone User 520

The following screen shows the **User** tab for User **520**. This user corresponds to an Avaya 1140E SIP Telephone.



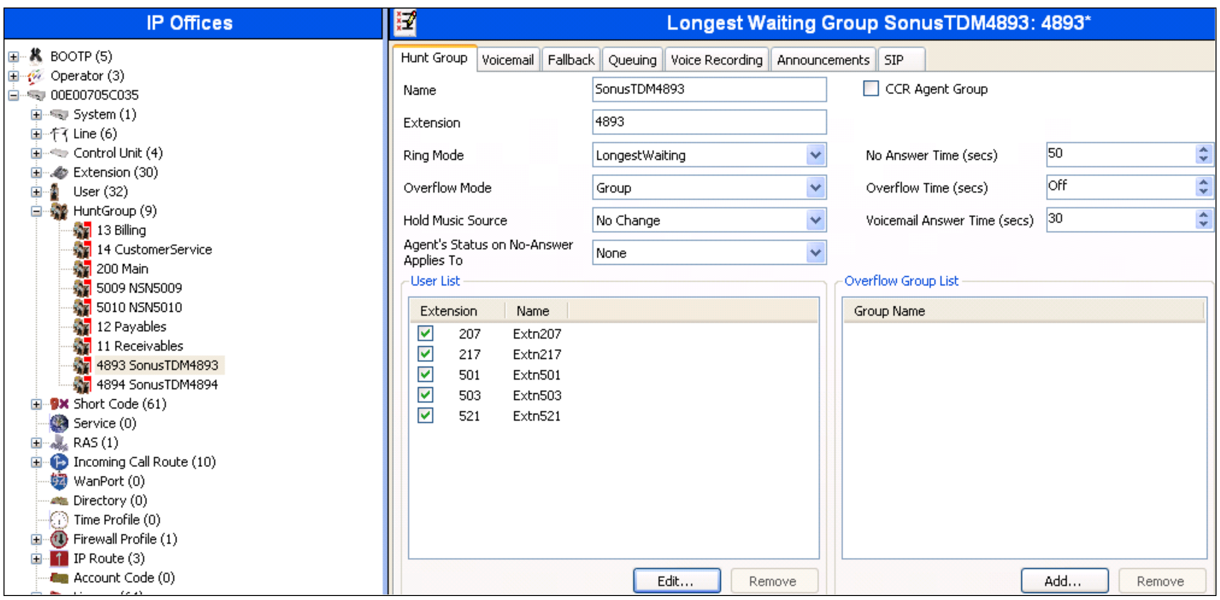
The following screen shows the Extension information for this user. Note that for a SIP telephone, the IP Address configured for the phone needs to be specified. In this example, **10.80.130.51** was assigned to the Avaya 1140E telephone. All other screens are configured the same way as in **Section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



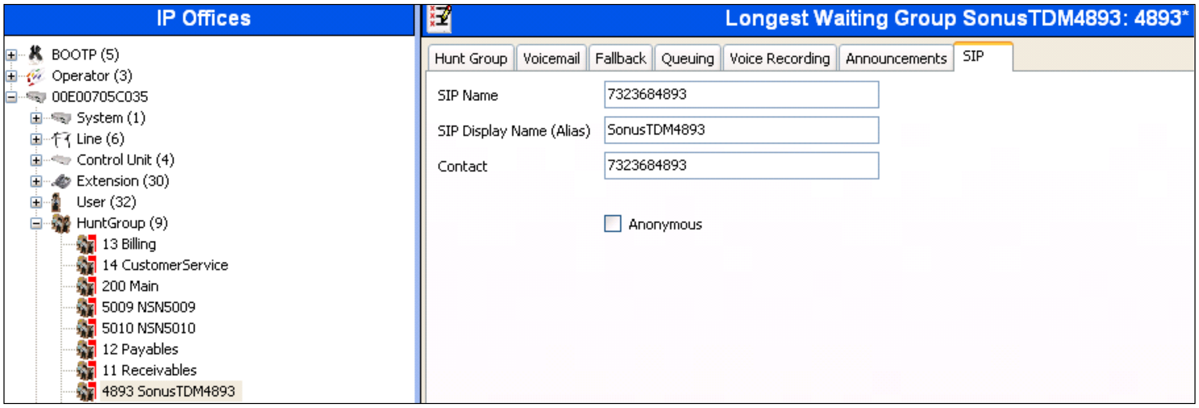
5.5.4. Hunt Groups

Hunt groups were used in this reference configuration to make sure all different endpoints could be exercised for incoming calls on a SIP Trunk from AT&T Flexible Reach service. 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 4893. This hunt group was configured to contain all of the endpoints used in this reference configuration. Since the **Ring Mode** field is set to **LongestWaiting**, this will enable the telephones to ring in a round robin fashion with a priority given to the longest waiting member of the hunt group. 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.



The following screen shows the **SIP** tab for hunt group **4893**. The **SIP Name** and **Contact** are configured with AT&T DID **7323684893**. Refer to **Section 5.7** where an incoming call route is mapped to this hunt group for the calls to be delivered to the members of this group.


IP Offices
BOOTP (5)
Operator (3)
00E00705C035
System (1)
Line (6)
Control Unit (4)
Extension (30)
User (32)
HuntGroup (9)
13 Billing
14 CustomerService
200 Main
5009 NSN5009
5010 NSN5010
12 Payables
11 Receivables
4893 SonusTDM4893
Longest Waiting Group SonusTDM4893: 4893*
Hunt Group
Voicemail
Fallback
Queuing
Voice Recording
Announcements
SIP
SIP Name
7323684893
SIP Display Name (Alias)
SonusTDM4893
Contact
7323684893
Anonymous

5.6. Short Codes

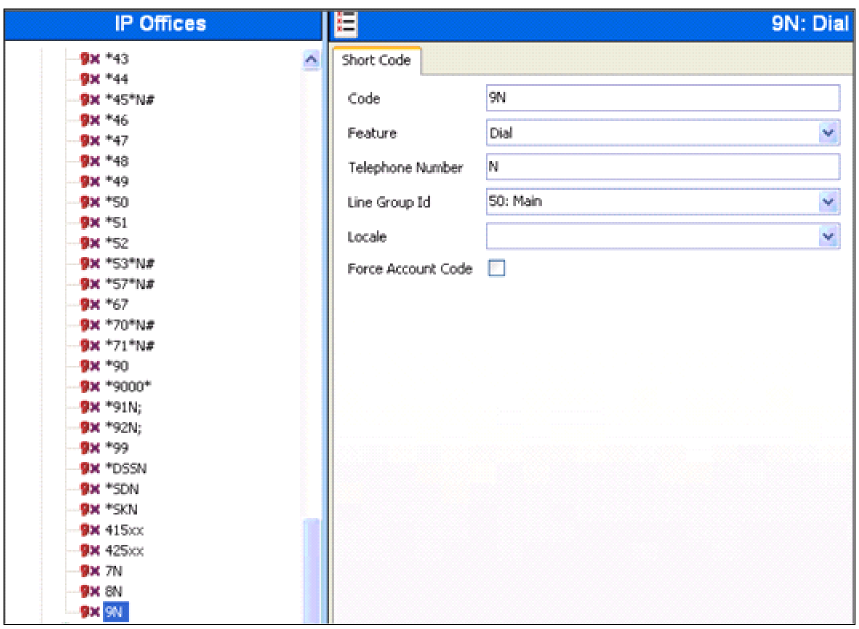
In this section, various examples of IP Office short codes will be illustrated. To add a short code, right click on **Short Code** in the Navigation pane, and select **New.** To edit an existing short code, click **Short Code** and select the code to be edited in the Navigation pane.

5.6.1. ARS Code

In this reference configuration, the Automatic Route Selection (ARS) feature was used for outgoing calls. A short code was configured as follows:

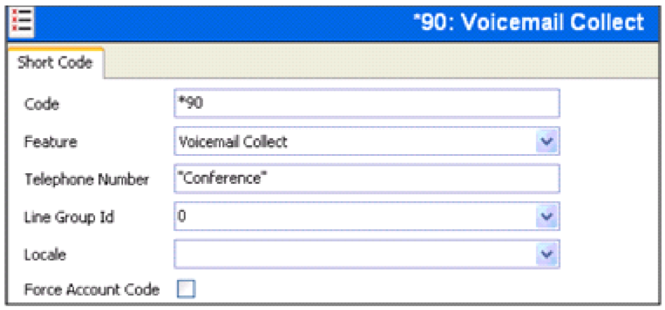
- **Code** – Set to **9N** for access to ARS
- **Feature** – Select **Dial** from the drop-down list
- **Telephone Number** – Set to **N**. If this value is set to **WN**, then the calls sent via this route will be sent as Anonymous.
- **Line Group Id** – Set to **50:Main** configured in **Section 5.8**.

So, when an IP Office user dials **9-1-303-538-1760** IP Office identifies it as an ARS call and refers to the line group setup in this short code. Refer to **Section 5.8** for how the call is handled after it get to the ARS group.



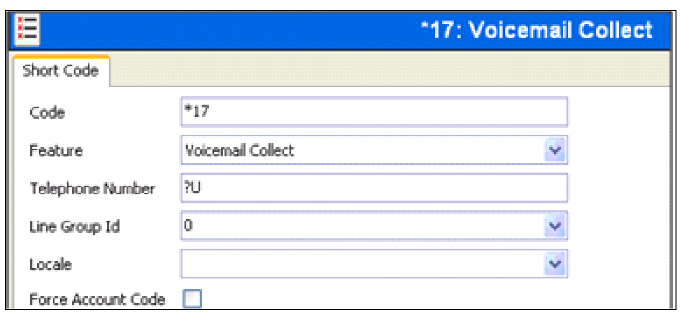
5.6.2. Meet-me Conference and Auto-attendant Codes

Features like Meet-me Conference and Auto-attendant are configured on Voicemail Pro and are beyond the scope of this document. In order to access those features, short codes can be used. In this reference configuration, for meet-me conference and auto-attendant **Conference** was configured on Voicemail Pro. The following screens show the short code set for these features. **Conference** module configured in Voicemail Pro handles both the conferencing and auto attendant features and is beyond the scope of this document.



5.6.3. Voicemail Retrieval Code

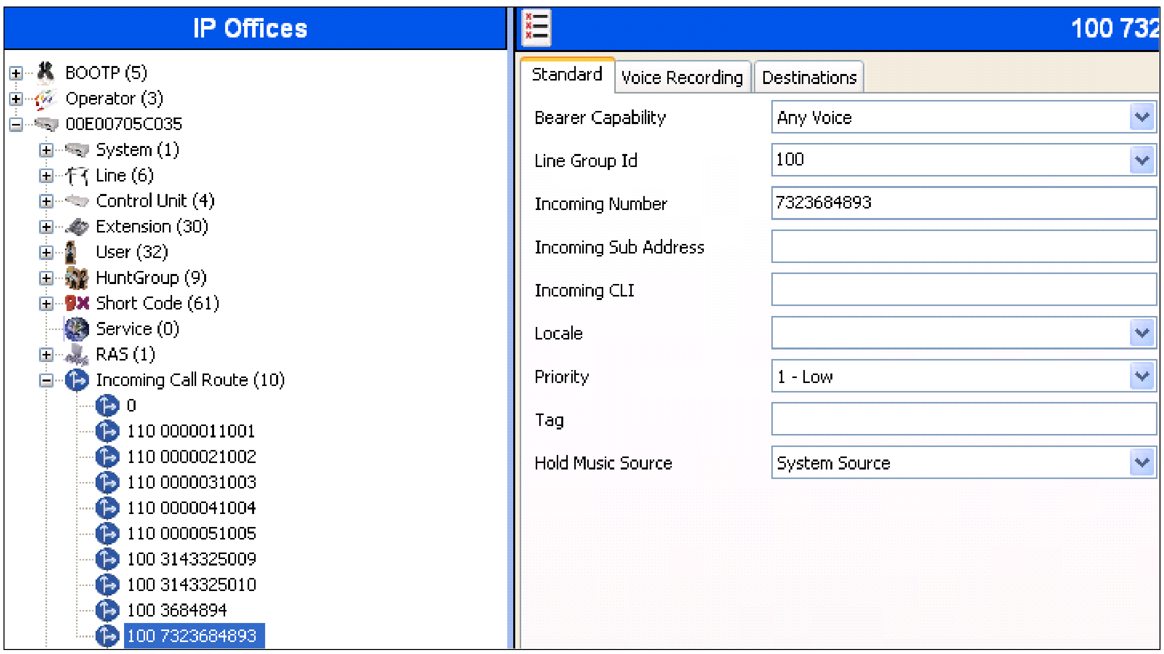
To retrieve voicemails left in individual mailboxes, this code was configured in this reference configuration. When a user enters, ***17**, they can retrieve the messages in their mailbox. Additionally, this short code can be also in the Incoming Call Route configured in **Section 5.7**.



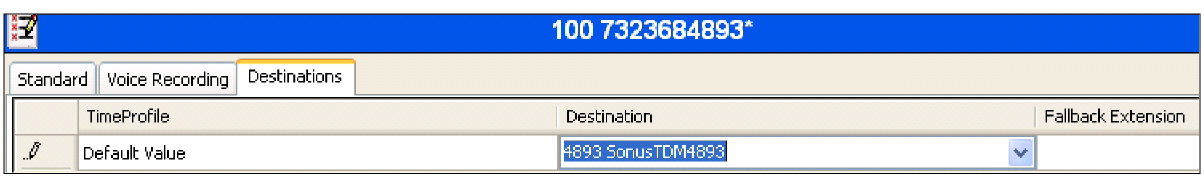
5.7. Incoming Call Routes

In this section, IP Office Incoming Call Routes are illustrated. Each Incoming Call Route will map a specific AT&T IP Flexible Reach DID number 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** [not shown]. 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 Flexible Reach DID **7323684893** in the **Incoming Number** field on the **Line Group Id** (**100**). The **Line Group Id** matches the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to AT&T IP Flexible Reach service in **Section 5.4**.

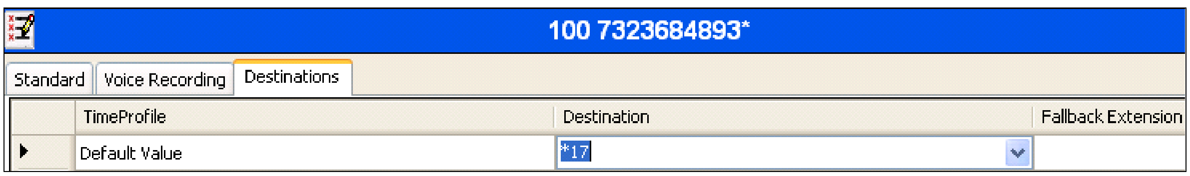


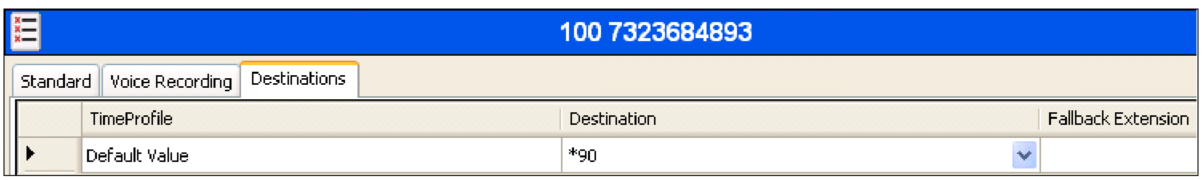
Select the **Destinations** tab and a value can be either selected from the drop-down list or manually entered. In the screen shown below, the hunt group configured in **Section 5.5.4** was selected.



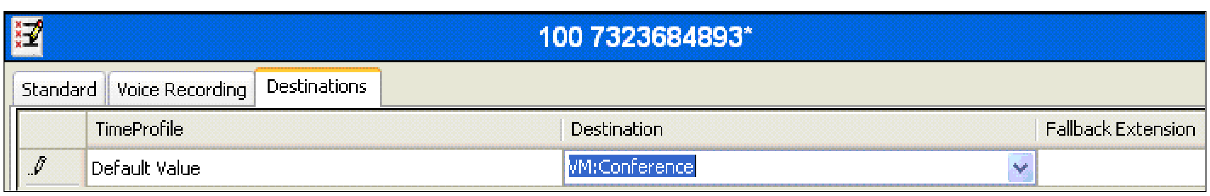
Similarly, in the screen below, any of the extensions configured in **Section 5.5** can be used.

The following screen displays how a short code can be manually assigned in the **Destination** field to route the call for voicemail retrieval. Similarly, a short code for Meet-me conference and Auto-attendant features can be used in the **Destination** field to exercise those features.





The following screen displays another mechanism to access the Meet-me conference and Auto-attendant feature without using the short code. The Conference feature was configure in Voicemail Pro and is beyond the scope of this document.

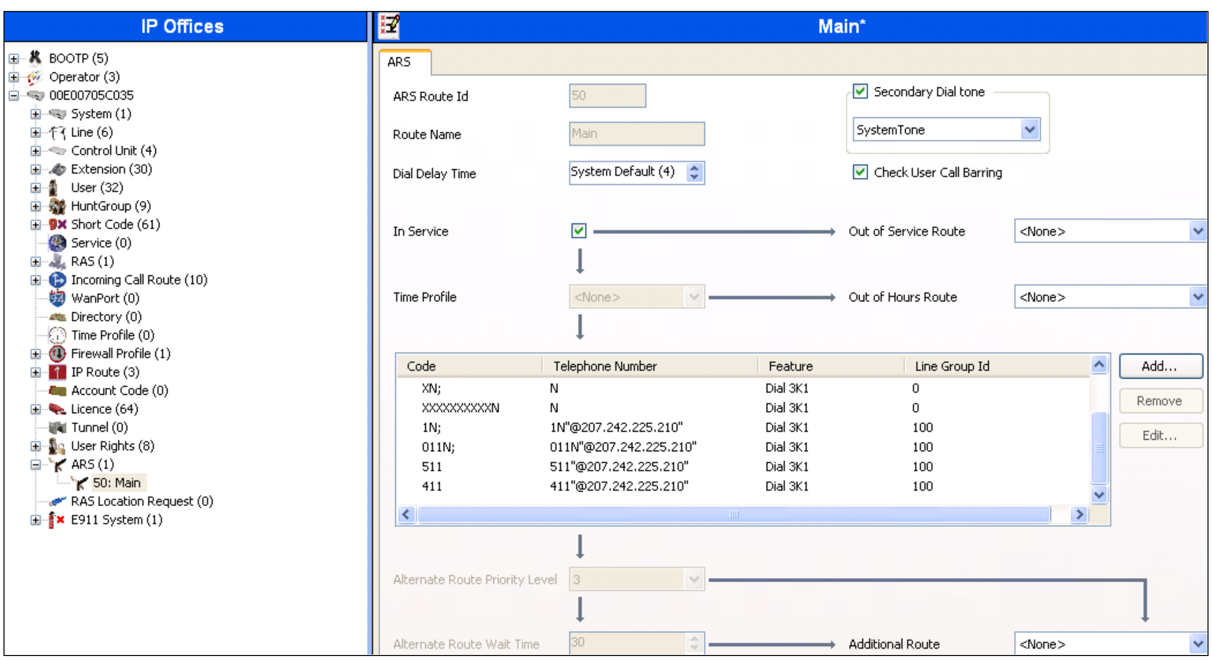


5.8. ARS and Alternate Routing

While detailed coverage of ARS is beyond the scope of these Application Notes, this section includes basic ARS screen illustrations and considerations. In this reference configuration, Automatic Route Selection (ARS) was used rather than the simple short code approach. With ARS, secondary dial tone can be provided after the access code, time-based routing criteria can be introduced, and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not available. Although not shown in this section, ARS also facilitates more specific dialed telephone number matching, enabling immediate routing and alternate treatment for different types of numbers following the access code. For example, if all 1+10 digit calls following an access code should use the SIP Line preferentially, but other local or service numbers following the access code should prefer a different outgoing line group, ARS can be used to distinguish the call behaviors.

To add a new ARS route, right-click **ARS** in the Navigation pane, and select **New** [not shown]. To view or edit an existing ARS route, select **ARS** in the Navigation pane, and select the appropriate route name in the Group pane.

The following screen shows an example ARS configuration for the route named "Main". The **In Service** parameter refers to the ARS form itself, not the Line Groups that may be referenced in the form. If the **In Service** box is un-checked, calls are routed to the ARS route name specified in the **Out of Service Route** parameter. IP Office short codes may also be defined to allow an ARS route to be disabled or enabled from a telephone. The configurable provisioning of an Out of Service Route, and the means to manually activate the Out of Service Route can be helpful for scheduled maintenance or other known service-affecting events for the primary route.



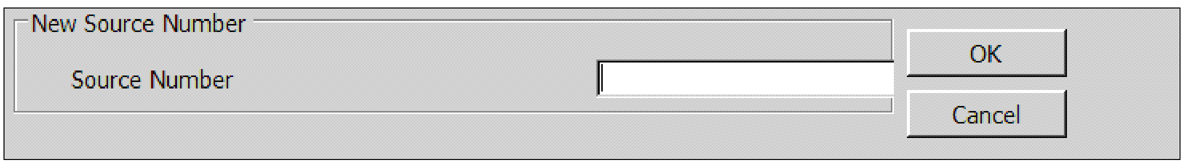
Assuming the primary route is in-service, the number passed from the short code used to access ARS (e.g., 9N in **Section 5.6**) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed **9-1-303-538-1760**, the call would be directed to Line Group **100**, configured in **Section 5.4.3**. If Line Group 100 cannot be used, the call can automatically route to the route name configured in the **Additional Route** parameter in the lower right of the screen. Since alternate routing can be considered a privilege not available to all callers, IP Office can control access to the alternate route by comparing the calling user's priority to the value in the **Alternate Route Priority Level** field.

5.9. Privacy / Anonymous Calls and SIP OPTIONS Frequency

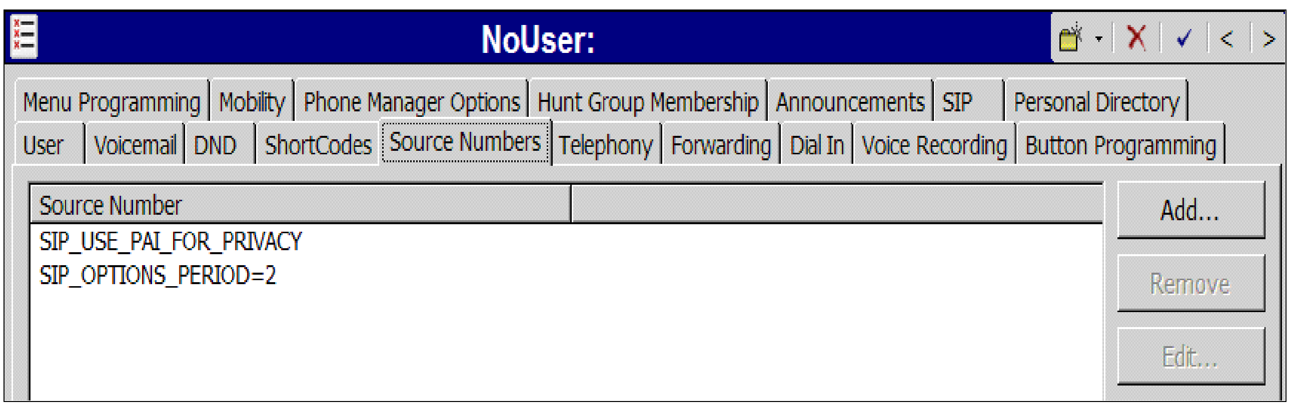
To configure IP Office to include the caller's DID number in the P-Asserted-Identity SIP header, required by AT&T Flexible Reach service to admit an otherwise anonymous caller to the network, the following procedure may be used.

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.

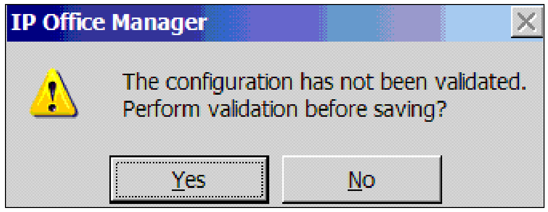


The following screen displays the Source Numbers configured in this reference configuration.

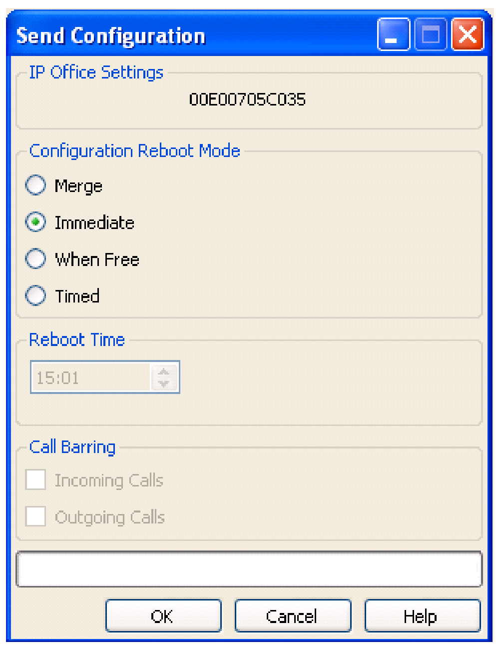


5.10. Saving Configuration Changes to IP Office

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 and outbound calls exercising the Sonus network, TDM gateway and NSN network.
- 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 outbound call, answer the call at PSTN phone and verify that two-way 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 Flexible Reach service. This solution provides users of Avaya IP Office the ability to support inbound and outbound calls over an AT&T IP Flexible Reach SIP trunk service connection via MIS/PNT transport. Additionally the ability of Avaya IP Office to provide SIP Diversion Header to the AT&T IP Flexible Reach service for certain out bound call scenarios was demonstrated.

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 7.0 Installation Manual, Issue 23k, May 22, 2011
Document Number 15-601042
https://support.avaya.com/css/P8/documents/100129376

[IPO-MGR] IP Office Release 7.0 Manager 9.0 Issue 26h, May 22, 2011
Document Number 15-601011
https://support.avaya.com/css/P8/documents/100129398

[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 7.0 Voicemail Pro Administration, Issue 26a, May 1, 2011
Document Number 15-601063
https://support.avaya.com/css/P8/documents/100129332

[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 Flexible Reach Service Descriptions:

[1] *AT&T IP Flexible Reach*
http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-flexible-reach-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.