Avaya Solution & Interoperability Test Lab

Application Notes for Avaya IP Office Release 8.1 with AT&T Business in a Box (BIB) with AT&T IP Flexible Reach - Enhanced Features – Issue 1.1

Abstract

These Application Notes describe the steps for configuring Avaya IP Office R8.1with the AT&T Business in a Box with AT&T IP Flexible Reach - Enhanced Features. AT&T Business in a Box with AT&T IP Flexible Reach - Enhanced Features supports AVPN or MIS/PNT transport connection.

The AT&T Business in a Box with AT&T IP Flexible Reach - Enhanced Features 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 R8.1 with the AT&T Business in a Box (BIB) with AT&T IP Flexible Reach - Enhanced Features.

In the reference configuration, the Avaya IP Office solution consists of an Avaya IP Office 500 V2 Release 8.1, Avaya Voicemail Pro, Avaya IP Office Softphone (SIP), and Avaya H.323, SIP, and Analog desk phones.

The AT&T AT&T Business in a Box with IP Flexible Reach - Enhanced Features 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 AT&T Business in a Box with IP Flexible Reach - Enhanced Features 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.

Note - The AT&T IP Flexible Reach - Enhanced Features service utilizes AVPN or MIS/PNT transport services, with or without BIB access. In addition, AVPN transport supports compressed RTP (cRTP), while MIS/PNT transport does not.

However, while the BIB access also supports AVPN and MIS/PNT transport, cRTP is not supported.

The AT&T IP Flexible Reach - Enhanced Features service will be referred to as IPFR-EF in the remainder of this document.

2. General Test Approach and Test Results

The test environment (see **Figure 1**) consisted of:

- A simulated enterprise with Avaya IP Office, Avaya IP Office telephones and fax machines (Ventafax application).
- AT&T Business in a Box with IPFR-EF service, to which the simulated enterprise was connected via AVPN transport. However, these Application Notes also certify MIS/PNT transport.
- The configurations described in these Application Notes certify Avaya IP Office and IPFR-EF, with and without BIB access.

The main test objectives were to verify the features and functionality described in **Section 2.1**.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound and outbound call flows (see **Section 3.2** for examples) between Avaya IP Office and AT&T Business in a Box with IPFR-EF via AVPN transport. 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. The following features were tested as part of this effort:

- Incoming and outgoing calls between PSTN, routed by the AT&T Business in a Box with IPFR-EF service, and Avaya IP Office. These calls are via the Avaya IP Office SIP Line and may be generated/answered by Avaya SIP telephones/Softphones, H.323 telephones, Analog telephones, Analog fax machines or via Hunt Groups. Coverage to Avaya IP Office Voicemail Pro, and Voicemail Pro auto-attendant applications, may also be used.
- Inbound / Outbound fax using T38 or G.711.
- Proper disconnect when the caller abandoned a call before answer for both inbound and outbound calls, and when the Avaya IP Office party or the PSTN party terminated an active call.
- Proper busy tone heard when an Avaya IP Office user called a busy PSTN user, or a PSTN user called a busy Avaya IP Office user (i.e., if no redirection was configured for user busy conditions).
- Various outbound PSTN call types were tested including long distance, international, and toll-free.
- Requests for privacy (i.e., caller anonymity) for Avaya IP Office outbound calls to the PSTN, as well as privacy requests for inbound calls from the PSTN to Avaya IP Office users.
- SIP OPTIONS monitoring of the health of the SIP trunk. In the reference configuration Avaya IP Office sent OPTIONS to the AT&T Business in a Box with IPFR-EF service Border Element and AT&T responded with *405 Method Not Allowed* (which is the expected response). That response is sufficient for Avaya IP Office to consider the connection up. The AT&T Business in a Box with IPFR-EF service Border Element does ***not*** send OPTIONS to Avaya IP Office
- Incoming and outgoing calls using the G.729(a) and G.711 ULAW codecs.
- Call redirection with Diversion Header.
- Long duration calls.
- DTMF transmission (RFC 2833) for successful voice mail navigation, including navigation of a simple auto-attendant application configured on Avaya IP Office Voicemail Pro.
  - See **Section 2.2**, **Item 1** for a DTMF issue regarding Avaya IP Office Softphone.
- Telephony features such as call waiting, hold, transfer, and conference.
- Mobile twinning to a mobile phone when the associated Avaya IP Office extension is called, as well as Mobility features such as Mobile Callback and Mobile Call Control.
- AT&T IPFR-EF service features such as:
  - Simultaneous Ring
  - Sequential Ring

- Call Forward – Always
- Call Forward – Busy
- Call Forward – Ring No Answer

2.2. Test Results/Known Limitations

Interoperability testing of the reference configuration was completed with successful results; however the following observations were noted:

- The AT&T IP Flexible Reach - Enhanced Features service "blind transfer" feature (utilizing Refer for network call redirection) is not supported. While IP Office R8.1 does support Refer, it does not support this type of call redirection.
    - Refer must be disabled on IP Office R8.1 (see **Section 5.4**).

- Avaya IP Office Softphone did not generate RFC2833 DTMF when attempting to navigate menus with an Auto Attendant system utilizing SIP 183 Session Progress signaling. Some Auto Attendant systems send 183 Session Progress in response to an inbound call, and do not send 200 OK until a menu selection has been made. In these cases, Avaya IP Office Softphone did not generate DTMF SIP Telephone Events when menu selection was attempted (even though appropriate SIP Telephone Event SDP signaling was sent in the initial Avaya IP Office Invite and received in the network 183).
    - An MR was opened with Avaya IP Office support.
    - This issue does not occur with Avaya IP Office desk phones.
    - It should be noted that if the called Auto Attendant system answers at the onset with 200 OK (prior to a menu selection), then the Avaya IP Office Softphone does generate DTMF Telephone Events during menu selection.

- The Avaya IP Office fax feature "T.38 Fallback" (to G.711) is not supported in the reference configuration.

- Avaya IP Office only supports a packet size of 20 msecs, and does not specify a PTIME value in the SIP SDP (in either requests or responses). Network responses include MAXPTIME=20, and network requests include MAXPTIME=30. Although no issues were found during testing, the AT&T IPFR-EF service recommends a value of 30ms when AVPN transport is used.

- A no audio condition was found in the AT&T test network environment when an inbound PSTN call was forwarded back to PSTN or twinned to a mobile phone. This problem was apparently a result of media deadlock condition where the security aspects of the AT&T test network environment expected Avaya IP Office endpoint to send RTP packets first, even though there was no Avaya IP Office originated media involved (since the original inbound PSTN call had been re-directed to a second PSTN destination). A work around for this issue was accomplished by enabling RTP Keepalives in Avaya IP Office to send spurious RTP packets toward the AT&T test network at the start of the call, thus breaking the media deadlock. Although this issue may not occur in production AT&T Business in a Box with IPFR-EF service connections, customers experiencing loss of audio for similar call flows over secure connections should follow the steps for enabling RTP Keepalives in Avaya IP Office as shown in **Section 5.3.3**.

- Avaya IP Office uses a fixed RFC2833 Telephone Event type of 101 in SIP requests. The network responses comply with a value of 101. However network SIP requests specify Telephone Event type 100 and Avaya IP Office complies with a value of 100. No issues were found during testing as a result of this behavior.

- 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<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.

  *   NOTE: N11 (including 911) calls are not supported unless AT&T IP Flexible Reach Local Service is ordered!

2.3. Support

2.3.1. Avaya

For technical support on the Avaya products described in these Application Notes visit http://support.avaya.com.

2.3.2. AT&T

AT&T customers may obtain support for the AT&T Business in a Box with IPFR-EF service by calling (877) 288-8362 or by visiting: http://www.business.att.com/enterprise/Service/voice-services/voip/business-in-a-box/

3. Reference Configuration

Note – Documents used to provision the reference configuration are listed in **Section 9**. Specific references to these documents are indicated in the following sections by the notation **reference [x]**, where x is the document reference number.

The reference configuration used in these Application Notes is shown in **Figure 1** below and consists of the following components:

- Avaya IP Office provides the voice communications services for a particular enterprise site. In the reference configuration, Avaya IP Office runs on an IP 500 V2. This solution is extensible to other Avaya IP Office hardware as well.
- Avaya "desk" telephones are represented with an Avaya 1608 H.323 set, an Avaya 6211 Analog set, an Avaya 1120E SIP set, and PC based Avaya IP Office SIP Softphone (in Default Mode). Fax endpoints are represented by PCs running Ventafax software connected by modem to an Avaya IP Office analog port.
- Avaya IP Office Voicemail Pro (running on a Windows 2003 server) provided the voice messaging capabilities in the reference configuration.
- Outbound and Inbound calls utilize telephone or fax User/Extensions provisioned on Avaya IP Office. Signaling is sent between Avaya IP Office and the AT&T Business in a Box via the IPFR-EF Border Element.
- The AT&T IPFR-EF service requires the following network settings between Avaya IP Office and the IPFR-EF Border Element:
    - UDP transport using port 5060
    - RTP port ranges 16384-32767
- AT&T provided the inbound and outbound access numbers (DID and DNIS) used in the reference configuration.

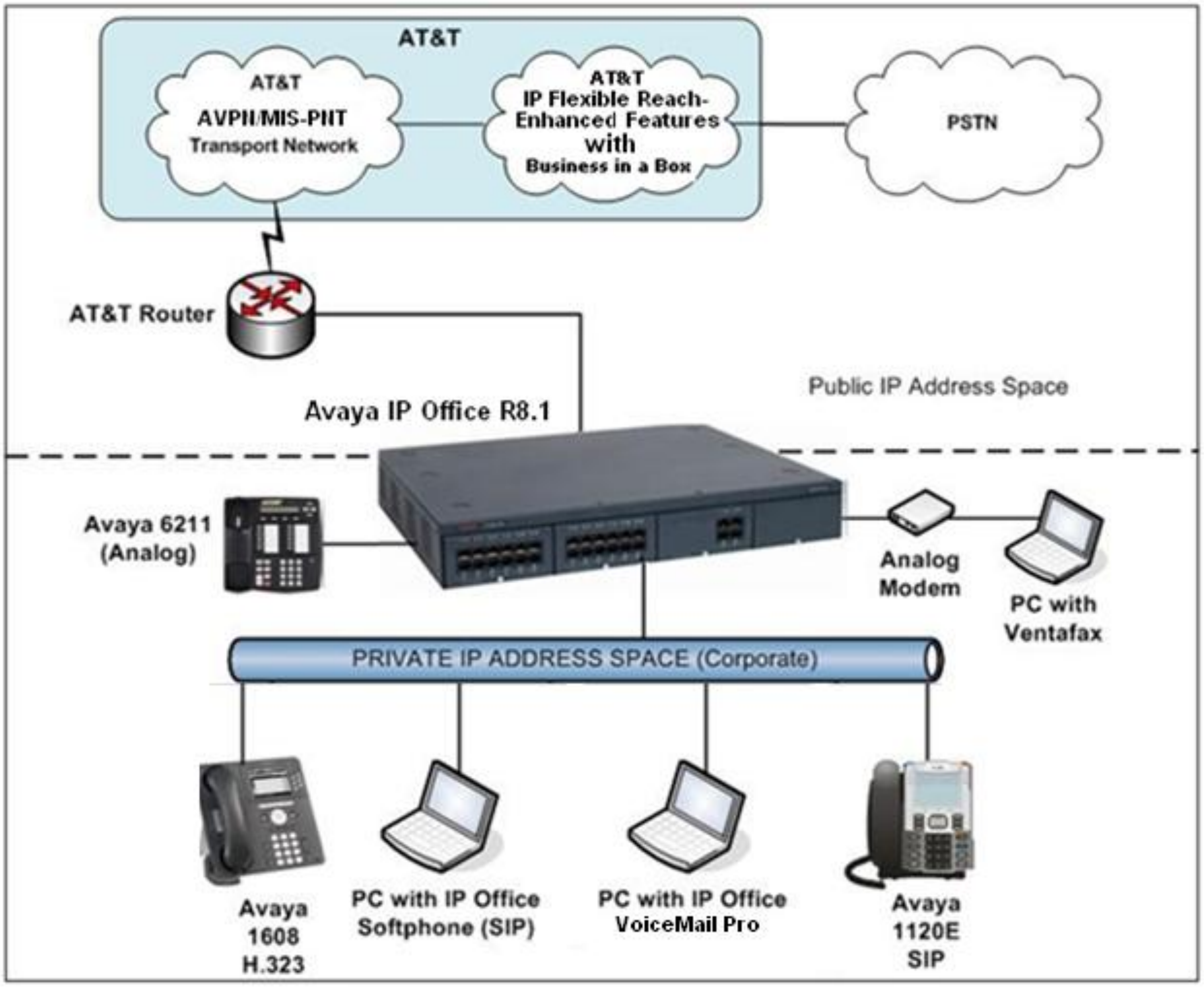


Figure 1: Reference Configuration

3.1. Illustrative Configuration Information

The specific values listed in **Table 1** 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 Business in a Box with IPFR-EF IP addressing shown in this document is an example. AT&T Customer Care will provide the actual IP addressing as part of the Business in a Box with IPFR-EF provisioning process.

Component	Illustrative Value in these Application Notes
Avaya IP Office 500 V2	
Public IP Address (LAN2 interface, labeled “WAN” on the chassis, see **Section 5.1**).	192.168.64.130
Private IP Address (LAN1 interface, labeled “LAN” on the chassis, see **Section 5.1**)	192.168.42.1
Avaya IP Office Voicemail Pro	
Windows 2003 server	172.16.6.51
AT&T Business in a Box with IPFR-EF **Service**	
Border Element IP Address	135.25.29.74
AT&T Access router interface	192.168.64.254

Table 1: Illustrative Values Used in these Application Notes

3.2. Call Flows

To understand how inbound and outbound AT&T Business in a Box with IPFR-EF service calls are handled by Avaya 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 Business in a Box with IPFR-EF service call that arrives on Avaya IP Office, which in turn routes the call to a hunt group, phone or a fax endpoint.

1. A PSTN phone originates a call to an IPFR-EF service number.
2. The PSTN routes the call to the AT&T Business in a Box with IPFR-EF service network.
3. The AT&T Business in a Box with IPFR-EF service routes the call to Avaya IP Office.
4. Avaya IP Office applies any necessary digit manipulations based upon the DID and routes the call to a hunt group, phone or a fax endpoint.

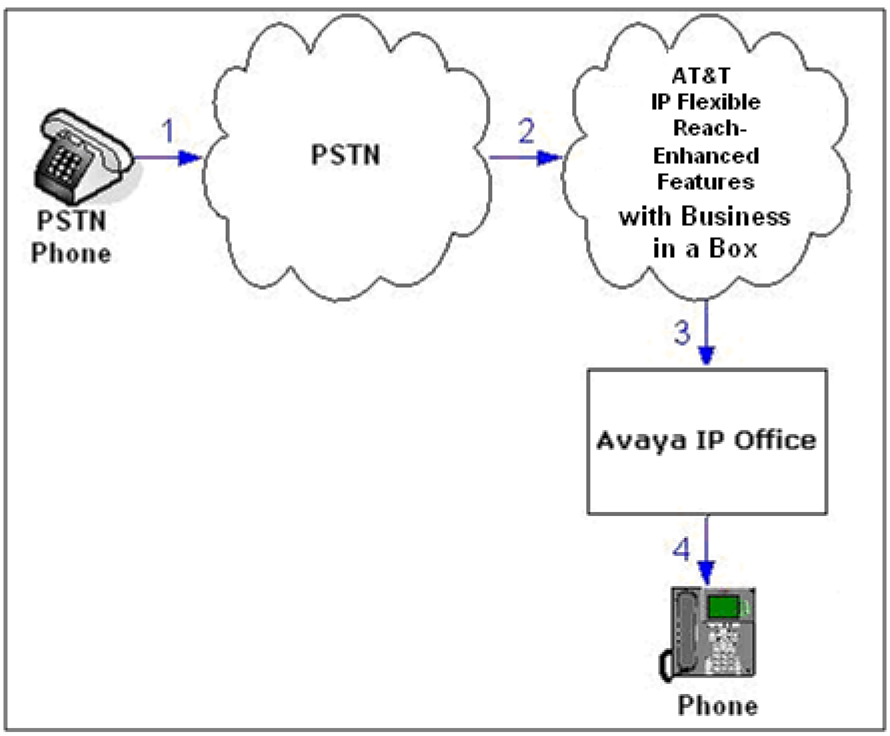


Figure 2: Inbound AT&T Business in a Box with IPFR-EF Call

3.2.2. Outbound

The second call scenario illustrated in the figure below is an outbound call initiated on Avaya IP Office for delivery to AT&T Business in a Box with IPFR-EF service.

1. An Avaya IP Office phone or fax endpoint originates a call to an AT&T Business in a Box with IP IPFR-EF service number for delivery to PSTN.

2. Avaya 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 Business in a Box with IPFR-EF service.
3. The AT&T Business in a Box with IPFR-EF service delivers the call to PSTN.
4. PSTN delivers the call to a phone or fax endpoint.

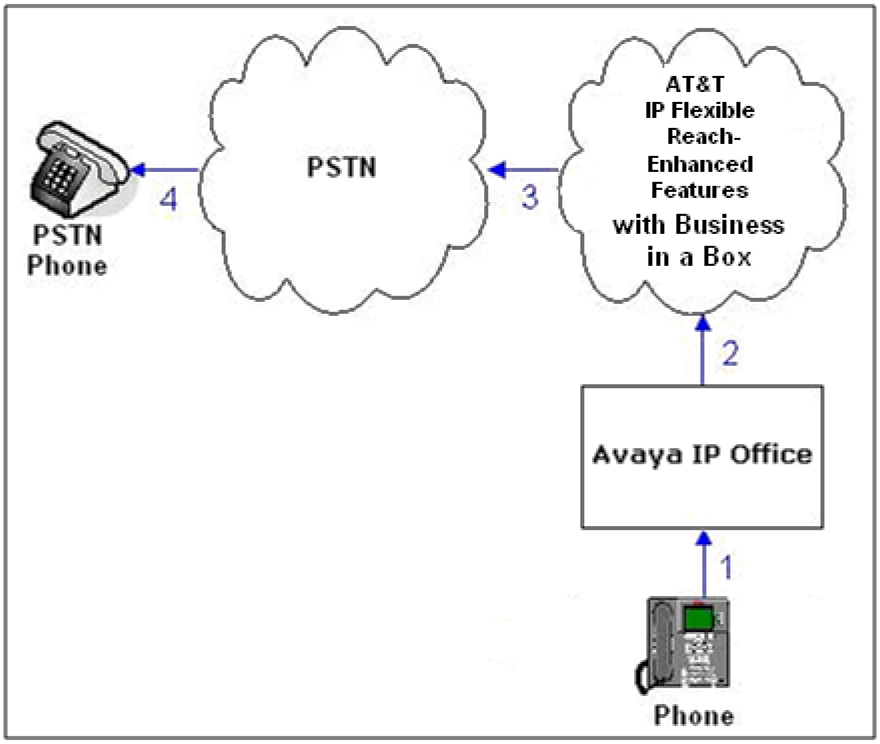


Figure 3: Outbound AT&T IP Business in a Box with IPFR-EF Call

3.2.3. Call Forward Re-direction[1]

The third call scenario illustrated in the figure below is an inbound AT&T Business in a Box with IPFR-EF service call destined for an Avaya IP Office station that has set Call Forwarding to an alternate destination. Without answering the call, Avaya IP Office immediately redirects the call back to the AT&T Business in a Box with IPFR-EF service for routing to the alternate destination.

Note – AT&T requires the Diversion header be used when a call is redirected to AT&T Business in a Box with IPFR-EF service (See **Section 5.4**).

1. Same as the first call scenario in **Section 3.2.1**.
2. Because the Avaya IP Office phone has set Call Forward to another AT&T Business in a Box with IPFR-EF service number, Avaya IP Office initiates a new call back out to the AT&T I Business in a Box with IPFR-EF service network. This ne SIP Invite will contain a Diversion Header.

[1] Note that when the redirection is completed, RTP will still flow through Avaya IP Office

3. The AT&T Business in a Box with IPFR-EF service places a call to the alternate destination and upon answer, Avaya IP Office connects the calling party (PSTN Phone) to the target party (Target Phone).

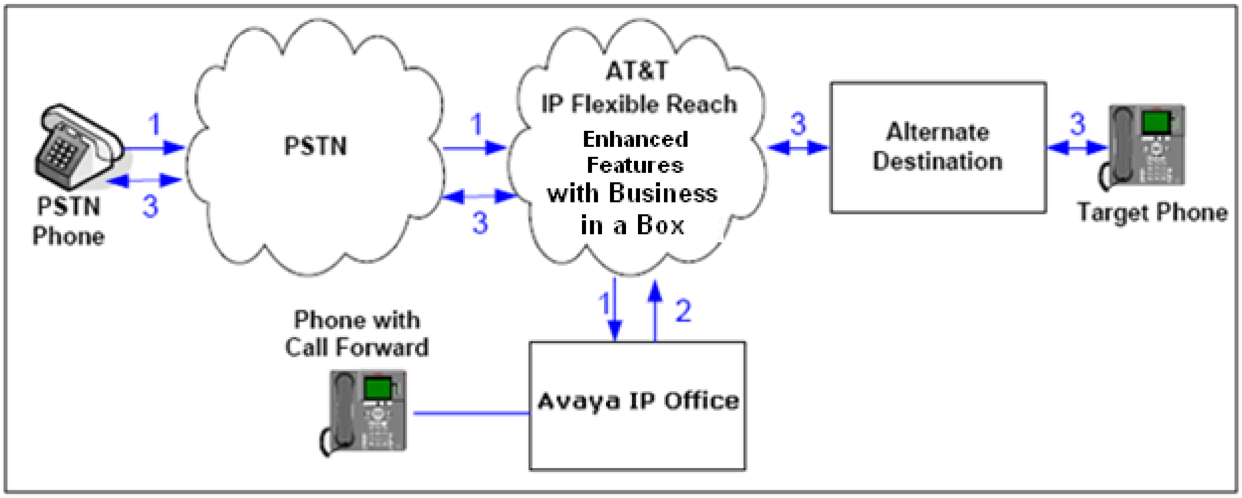


Figure 4: Re-directed (e.g. Call Forward) AT&T Business in a Box with IPFR-EF Call

3.2.4. Coverage to Voicemail

The call scenario illustrated in the figure below is an inbound call that is covered to Voicemail. In the reference configuration, the Voicemail system used is Avaya IP Office Voicemail Pro, running on a Windows 2003 server.

1. Same as the first call scenario in **Section 3.2.1**.
2. The Avaya IP Office phone does not answer the call, and the call covers to the external application Avaya IP Office Voicemail Pro.

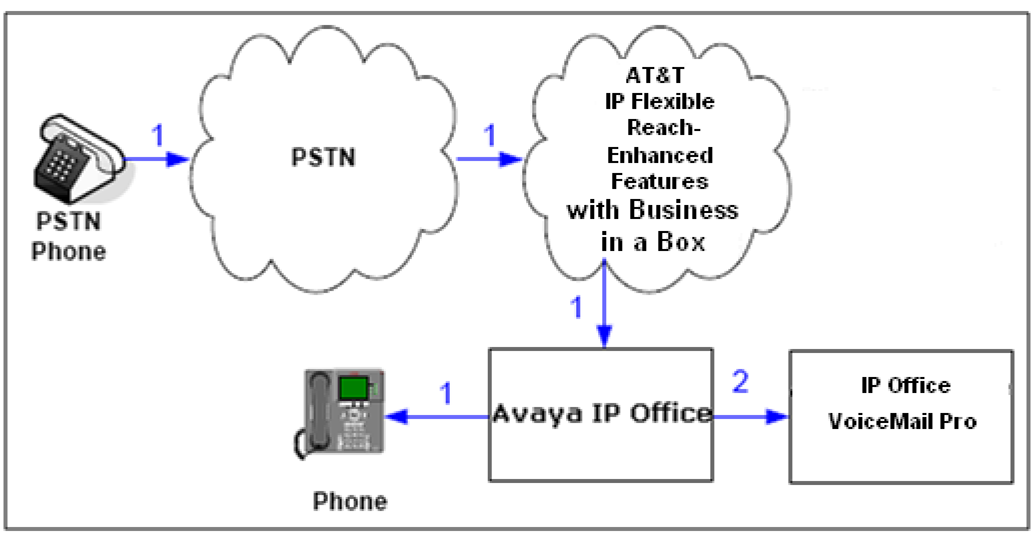


Figure 5: Coverage to Voicemail (Voicemail Pro)

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office 500 V2	R8.1 (43)
Avaya IP Office Manager	10.1 (43)
Avaya 1608 (H.323) Telephone	Ha1608ua1_3100.bin
Avaya 1120E (SIP) Telephone	04.03.12.00
Avaya IP Office Softphone (SIP)	3.2.3.20
Avaya 6211 Analog Telephone	-
Fax device	Ventafax 6.3
AT&T Business in a Box with IPFR-EF Service via MIS/PNT transport service connections.	VNI 23

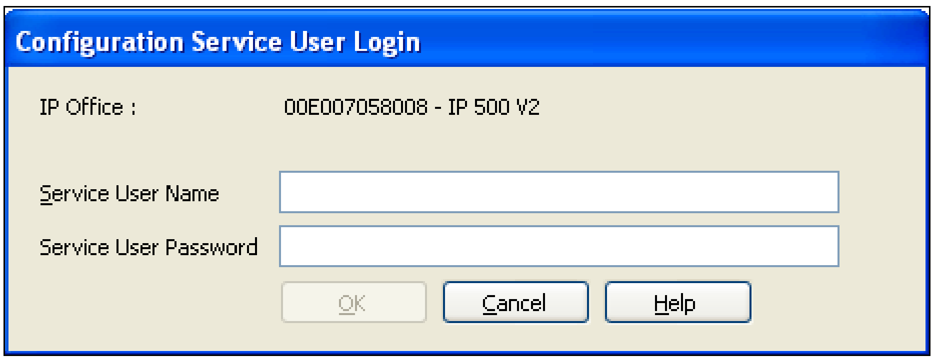
Table 2: Equipment and Software Versions

Note - Testing was performed with IP Office 500 V2 R8.1, but it also applies to IP Office Server Edition R8.1 (single site configuration only).

5. Avaya IP Office Configuration

Note - 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 sections may not be complete at times. For more information on installing Avaya IP Office consult reference **[1]**.

Avaya IP Office is configured via the Avaya IP Office Manager program. For more information on Avaya IP Office Manager, consult reference **[2]**. From the Avaya IP Office Manager PC, select **Start** → **Programs** → **Avaya IP Office** → **Manager** to launch the Manager application. Enter the appropriate credentials.

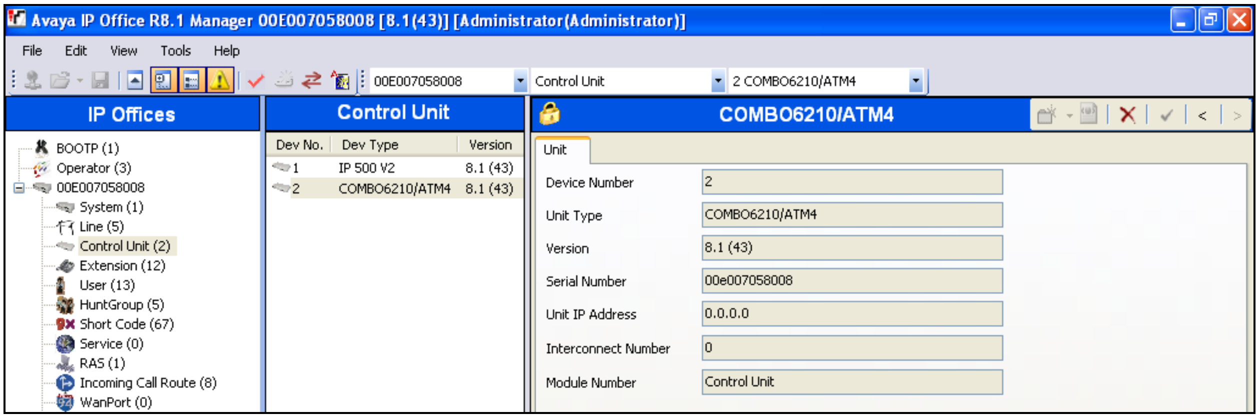


5.1. Physical, Network, and Security Configuration

This section describes attributes of the reference configuration, but is not meant to be prescriptive.

In the reference configuration the Avaya IP Office 500 V2 platform contained a COMBO6210/ATM4 module. The COMBO6210/ATM4 is used to add a combination of ports to an IP500 V2 control unit and is not supported by IP500 control units. The module supports 10 voice compression channels. Codec support is G.711mu, G729a, G.723 with 64ms echo cancellation and G.722 (supported by Avaya IP Office Release 8.0 and higher). It will support 6 Digital Station ports for digital stations in slots 1-6 (except 3800, 4100, 4400, 7400, M and T-Series), 2 Analog Extension ports in slots 7-8, and 4 Analog Trunk ports in slots 9-12.

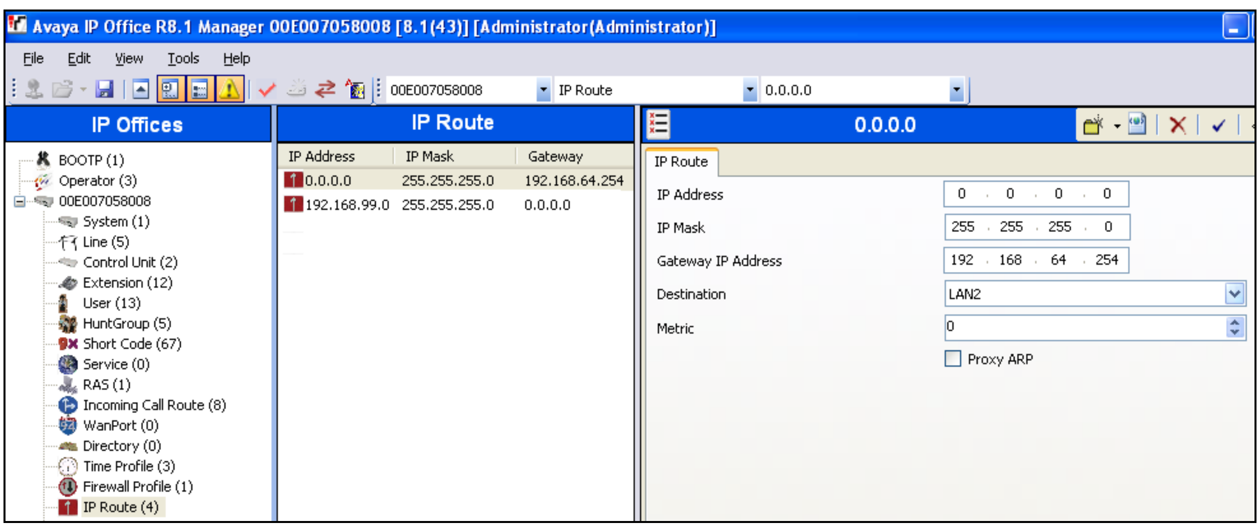
The following screen shows the modules in the Avaya IP Office used in the reference configuration. In the screen below, **IP 500 V2** is selected in the Group pane, revealing additional information about the IP 500 V2 in the Details pane.



The Avaya IP Office 500 V2 has two Ethernet ports on the back of the chassis, labeled **WAN** and **LAN**. In the reference configuration, the WAN port is connected to the AT&T network and the LAN port is connected to a local Ethernet switch. The Avaya H.323 and SIP telephones as well as the Administration/Softphone PC used in the reference configuration are also connected to this Ethernet switch.

In the Avaya IP Office Manager, the WAN port is identified as **LAN2** and the LAN port is identified as **LAN1**. Provisioning for these interfaces is described in **Section 5.3.2** and **5.3.3**.

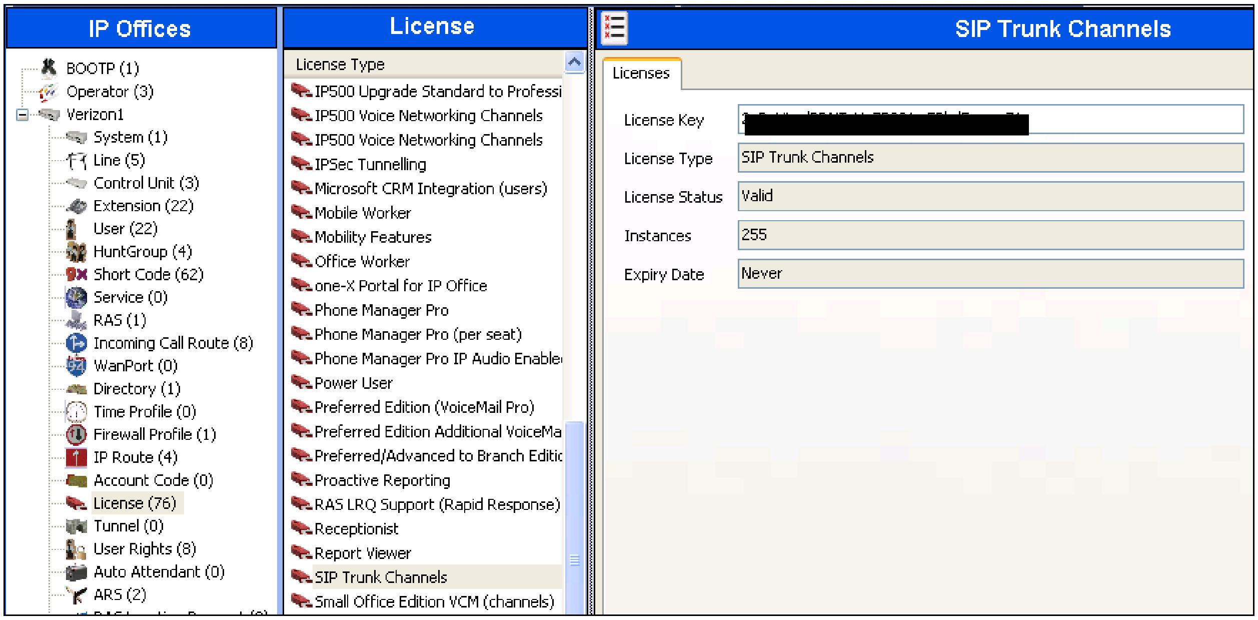
In order for the Avaya IP Office system to be able to route data to/from the AT&T network, a default route must be added specifying the reference configuration AT&T gateway (192.168.64.254). To add an IP Route in Avaya IP Office, right-click **IP Route** from the Navigation pane, and select **New.** 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 relevant default route using **Destination → LAN2**.



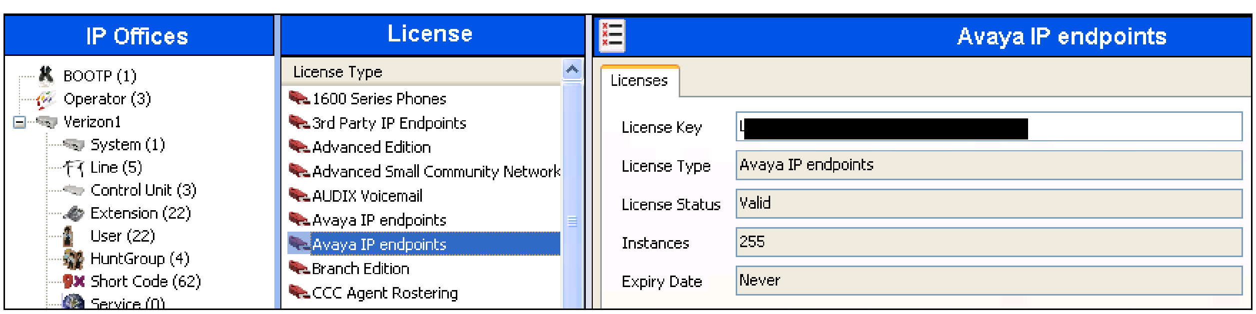
5.2. Licensing

The configuration and features described in these Application Notes require the Avaya 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.

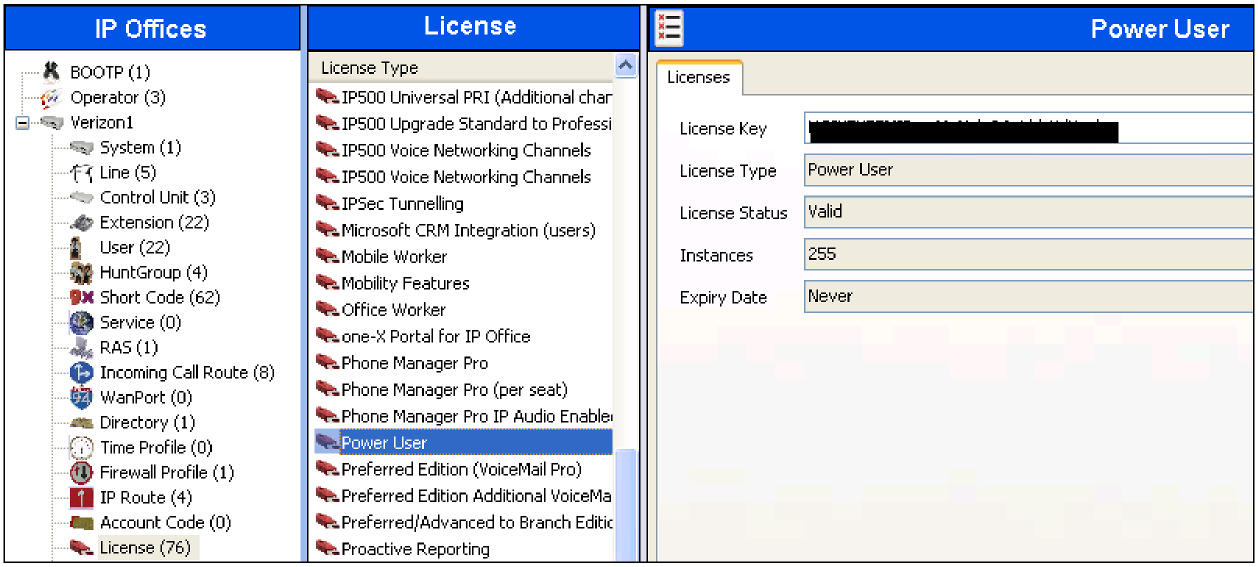
To verify that there is a SIP Trunk Channels License with sufficient capacity, click **License** in the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm a valid license with sufficient "Instances" (trunk channels) in the Details pane.



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. Note that in some cases duplicate license entries may be listed (e.g, Avaya IP endpoints below). One will display a key sequence in the License Key field while the other will display "Virtual".



The following screen shows the availability of a valid license for **Power User** features. In the reference configuration, the user with extension 500 will be configured as a "Power User".

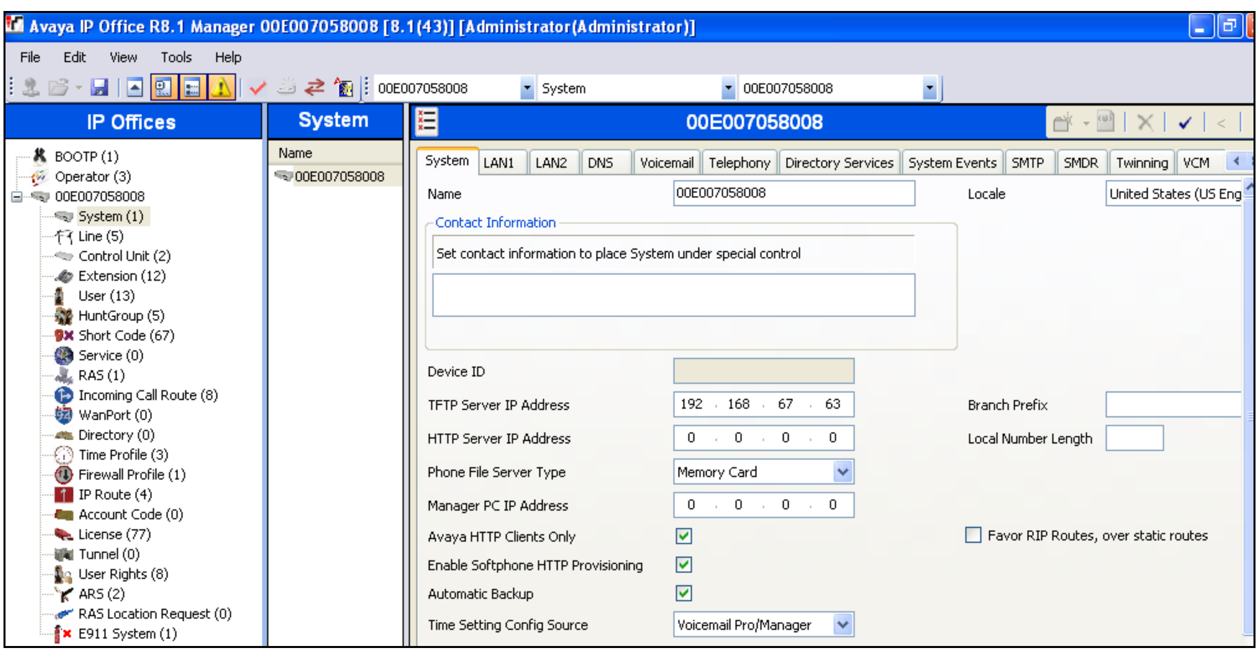


5.3. System Settings

This section illustrates the configuration of system settings. Select **System** in the Navigation pane to configure these settings.

5.3.1. System Tab

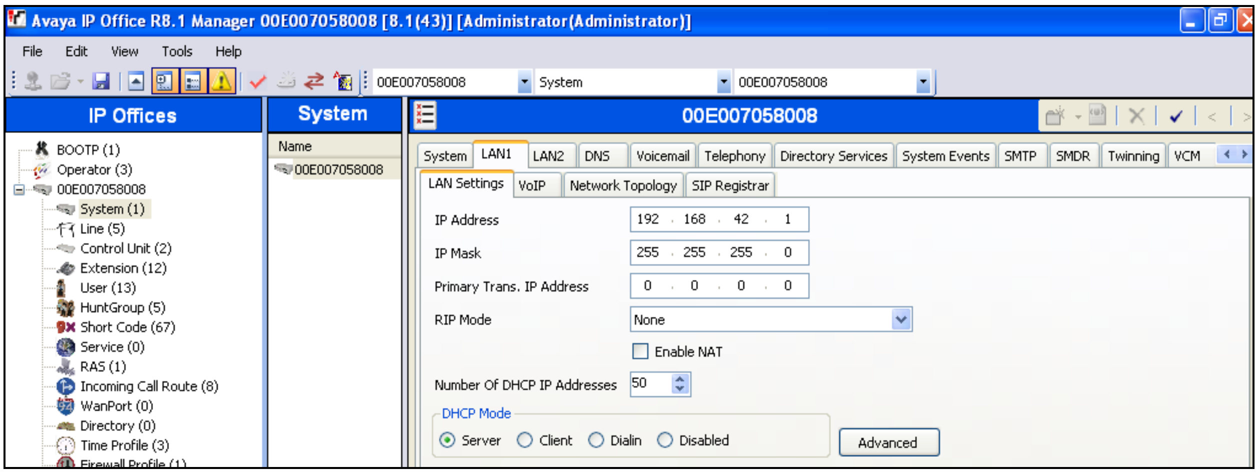
With the proper system name selected in the Group pane, select the **System** tab in the Details pane. The following screen shows a portion of the **System** tab. The **Name** field can be used for a descriptive name of the system. In this case, the default system serial number is used as the name. The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



5.3.2. LAN 1 Settings

In the reference configuration, LAN1 was used to connect the Avaya IP Office to the enterprise network. To view or configure the **IP Address** of LAN1, select the **LAN1** tab followed by the **LAN Settings** tab, and enter the following:

- **IP Address:** As shown in **Figure 1**, the IP Address of the Avaya IP Office in the reference configuration is **192.168.42.1**.
- **DHCP Mode** is also set to **Server** so that IP phones will get an IP Address from the Avaya IP Office Server. Other parameters on this screen may be set according to customer requirements.
- Click the **OK** button (not shown).

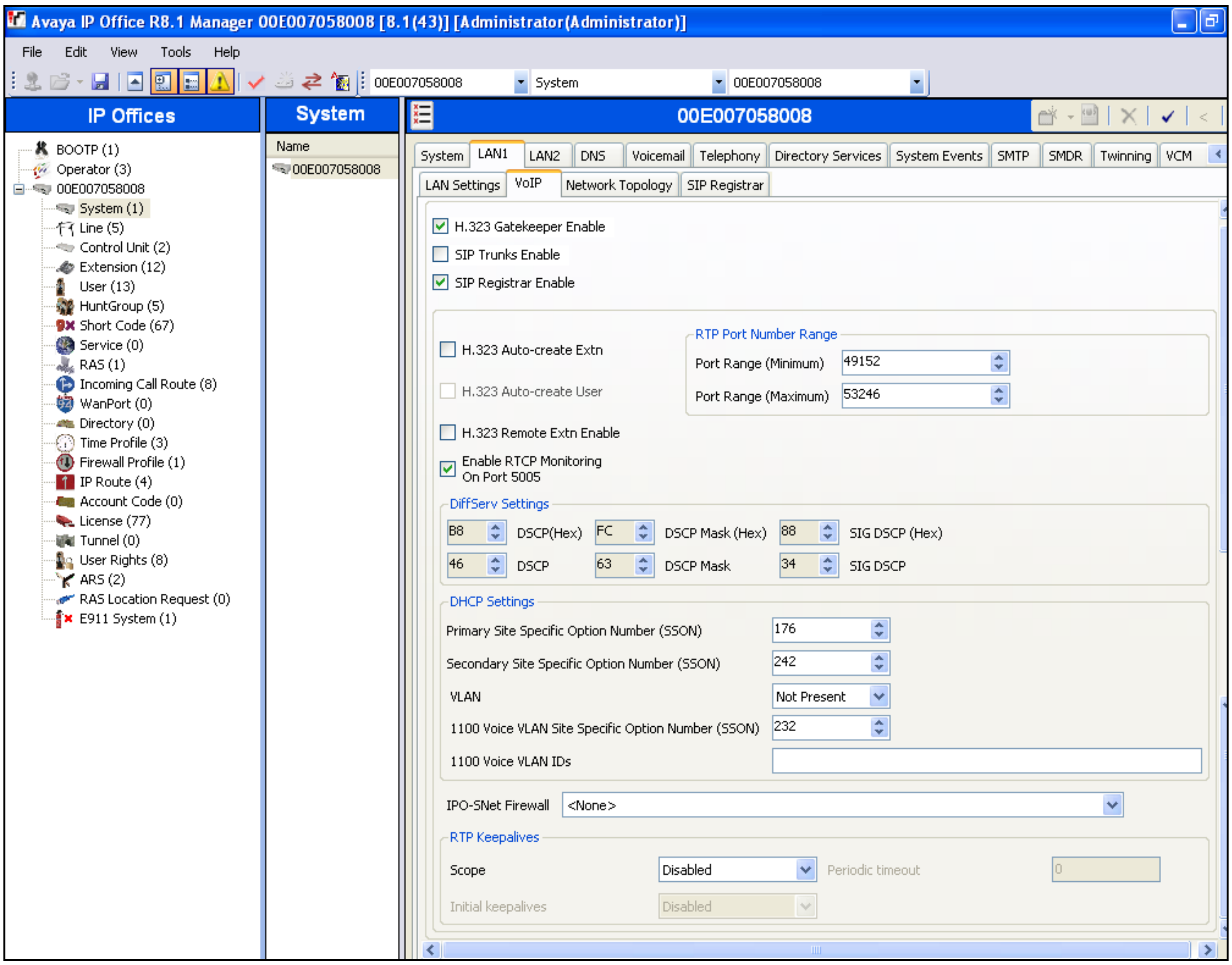


Select the **LAN1 →VoIP** tab as shown in the following screen. The following settings were used in the reference configuration:

- The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 1600-Series Telephones used in the reference configuration.
- The **SIP Registrar Enable** box is checked to allow Avaya 11xx (SIP) and Avaya IP Office Softphone (SIP) usage.
- **RTP Port Number Range:** For each VoIP call, a receive port for incoming Real Time Protocol (RTP) traffic is selected from a defined range of possible ports, using the even numbers in that range. The Real Time Control Protocol (RTCP) traffic for the same call uses the RTP port number plus 1, that is the odd numbers. For control units and Avaya H.323 IP phones, the default port range used is 49152 to 53246. On some installations, it may be a requirement to change or restrict the port range used. It is recommended that only port numbers between 49152 and 65535 are used.
  - **Port Range (Minimum):** Default = **49152**. Range = 1024 to 64510. This sets the lower limit for the RTP port numbers used by the system.

- Port Range (Maximum): Default = 53246. Range = 2048 to 65534. This sets the upper limit for the RTP port numbers used by the system. The gap between the minimum and the maximum must be at least 1024.

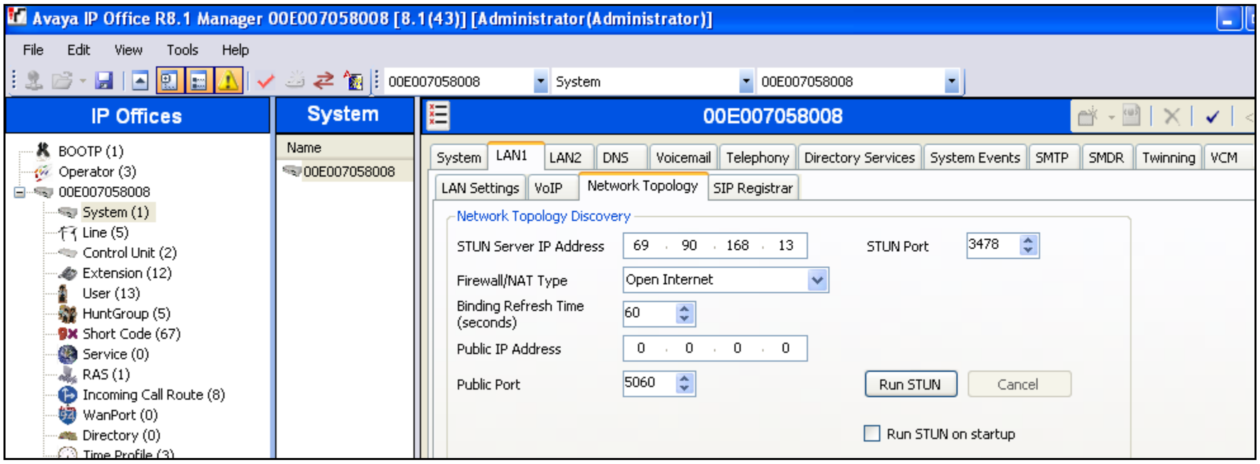
- **DiffServ Settings** (optional)**:** If desired, Avaya IP Office can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Service policies. The default values were used in the reference configuration.
- Note that on this interface, **RTP Keepalives/Scope** is set to **Disabled** (default).
- Other parameters on this screen may be set according to customer requirements.
- Click the **OK** button (not shown).



Select the **LAN1 → Network Topology** tab as shown in the following screen, and enter the following:

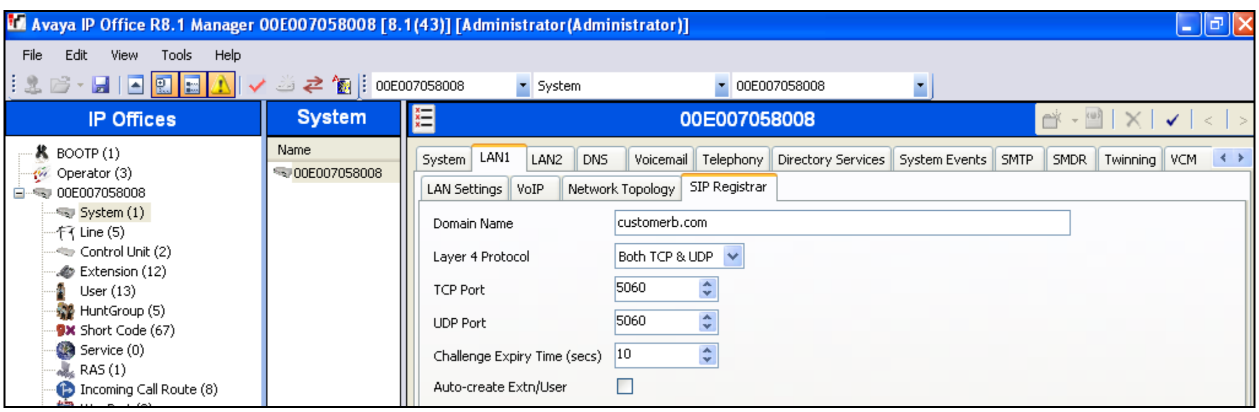
- **Public IP Address**: The **0.0.0.0** default value is used. This means Avaya IP Office will use the LAN1 IP address specified on the LAN1 **LAN Settings** tab described above (192.168.42.1).
- **Public Port** to **5060**.

- **Firewall/NAT Type** is set to **Open Internet**. With this configuration, STUN will not be used.
- Click the **OK** button (not shown).



Note: The **Firewall/NAT Type** parameter may need to be different, depending on the type of firewall or Network Address Translation device used at the customer premise.

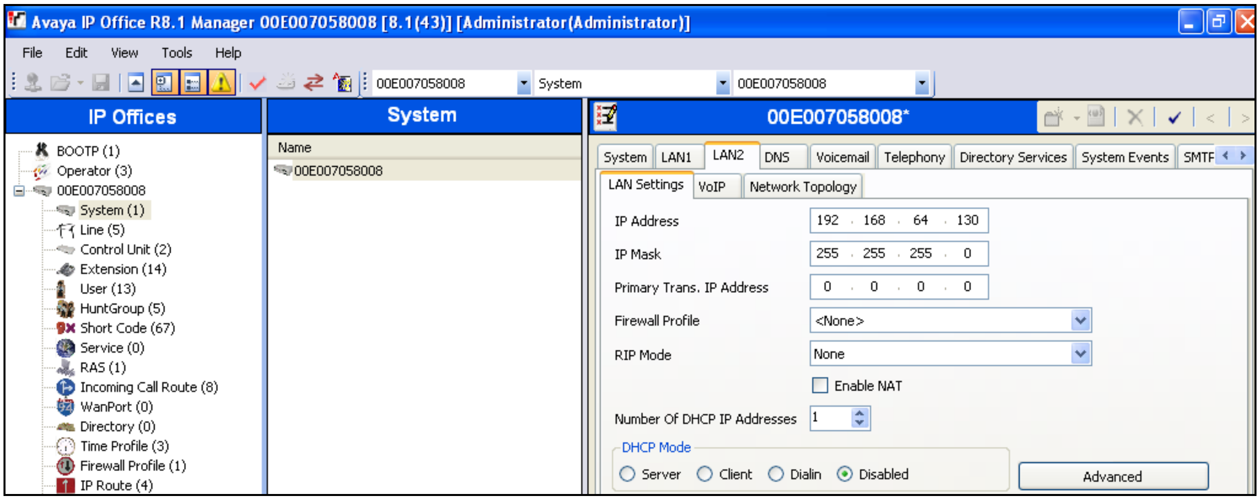
If SIP endpoints are used, select the **SIP Registrar** tab. The following screen shows the settings used in the reference configuration. Note that if the **Domain Name** field is left blank (default) the LAN IP address is used for registration.



5.3.3. LAN 2 Settings

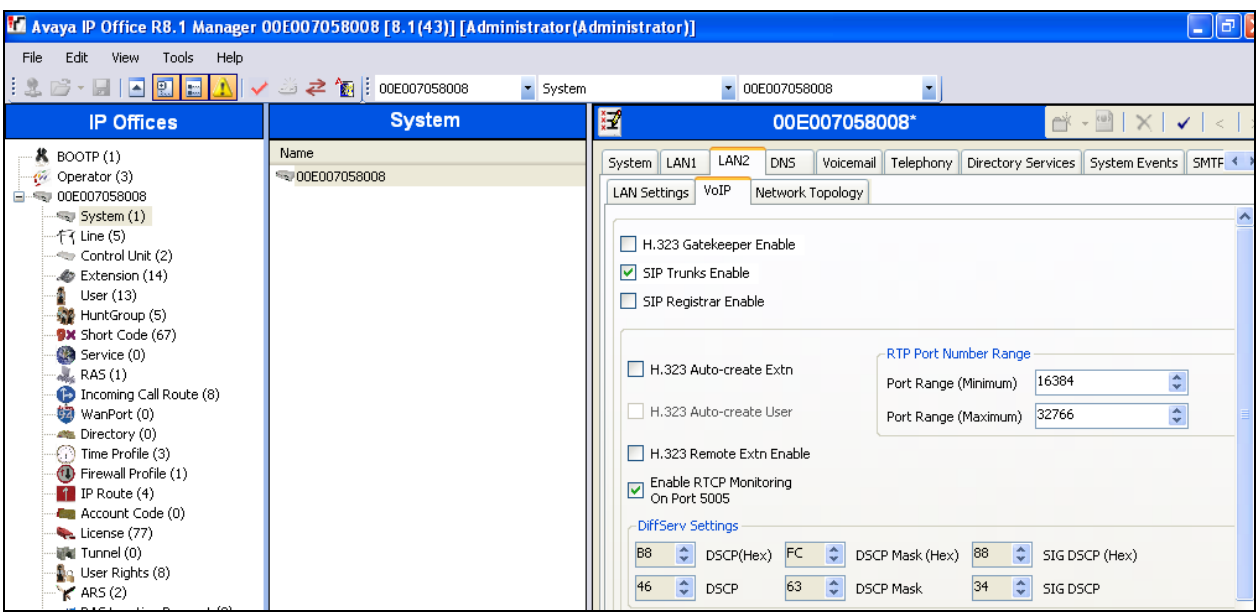
In the reference configuration, LAN2 was used to connect the Avaya IP Office to the AT&T network. To view or configure the **IP Address** of LAN2, select the **LAN2** tab followed by the **LAN Settings** tab, and enter the following:

- **IP Address:** This is the IP address of the Avaya IP Office, known to AT&T. **192.168.64.130** was used in the reference configuration.
- **DHCP Mode** is set to **Disabled** since DHCP is unnecessary towards AT&T.
- Other parameters on this screen may be set according to customer requirements.
- Click the **OK** button.

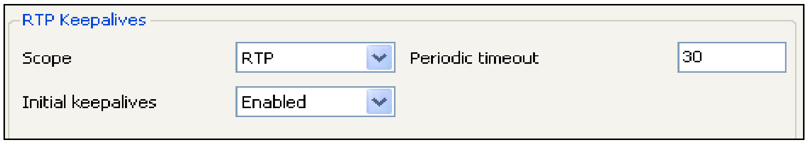


Select the **LAN2 → VoIP** tab as shown in the following screen and verify/enter the following:

- **H323 Gatekeeper Enable** and **SIP Registrar Enable:** These boxes are unchecked since IP telephones will not be registering on this link.
- **SIP Trunks Enable:** This box must be checked to enable the configuration of SIP trunks to AT&T.
- **RTP Port Number Range:** The AT&T IPFR-EF service requires that the RTP use the port range 16384 to 32767.
    - **16384** is entered in the **Port Range (Minimum)** field.
    - **32766** is entered in the **Port Range (Maximum)** field, as this field requires even numbers. See **Section 5.3.2** for more information on the RTP settings.



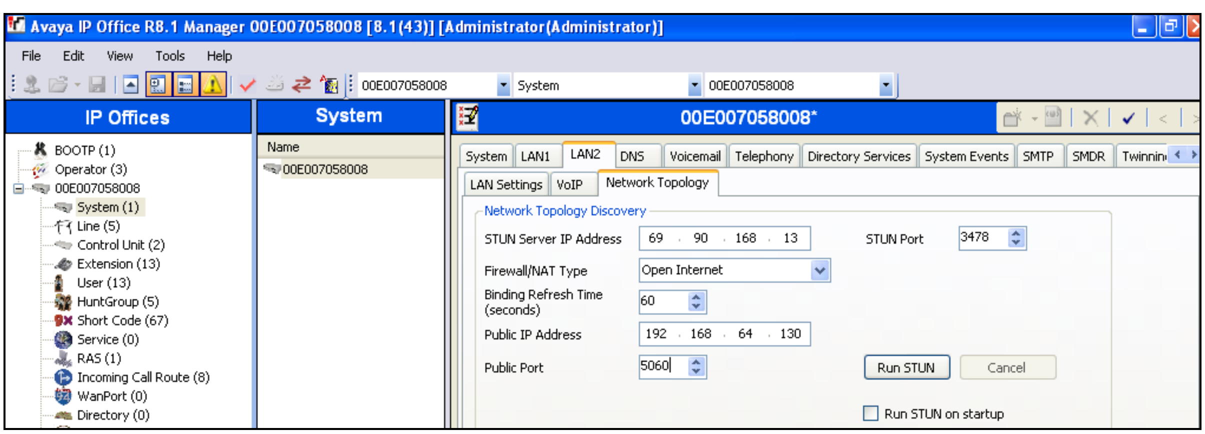
- As described in **Section 2.2 Item 4**, a no audio condition was observed in the AT&T test network environment when an inbound PSTN call was forwarded back to PSTN or twinned to a mobile phone. If a similar situation is observed, a workaround for this issue is to enable **RTP Keepalives** on the **LAN2 → VoIP** tab. Scrolling down to the bottom of the form, enter the following:
  - **Scope**: Select **RTP**
  - **Periodic Timeout**: Enter **30**
  - **Initial keepalives**: Select **Enabled**



- Let all other values default.
- Click the **OK** button.

Select the **LAN2 → Network Topology** tab as shown in the following screen, and enter the following:

- **Public IP Address**: Enter the Avaya IP Office LAN2 IP address **192.168.64.130** defined in the LAN2 **LAN Settings** tab above.
- **Firewall/NAT Type** is set to **Open Internet**. With this configuration, STUN will not be used.
- **Binding Refresh Time** can be configured to vary SIP OPTIONS timing. A value of **60** seconds was used in the reference configuration (see **Section 5.11** for more information on SIP Options timing between Avaya IP Office and AT&T).
  Note – In the reference configuration Avaya IP Office sent OPTIONS to the AT&T Business in a Box with IPFR-EF service Border Element and AT&T responded with *405 Method Not Allowed* (which is the expected response). That response is sufficient for Avaya IP Office to consider the connection up. The AT&T Business in a Box with IPFR-EF service Border Element does ***not*** send OPTIONS to Avaya IP Office.
- **Public Port** to **5060**.
- Click the **OK** button (not shown).



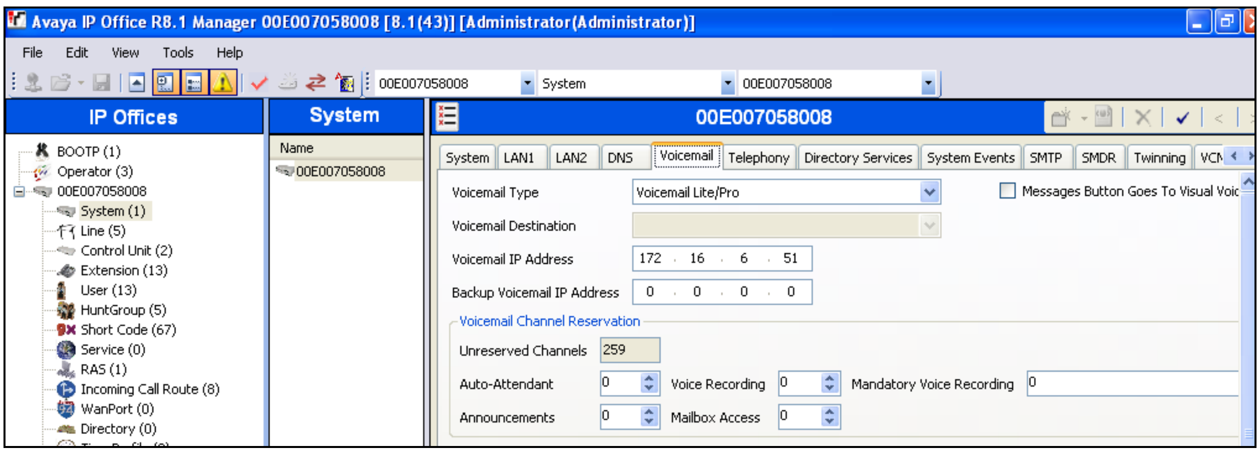
Note that since **SIP Registrar Enable** was unchecked on the LAN2 **VOIP** tab, the SIP Registrar Tab is not present for LAN2.

5.3.4. Voicemail

As described in **Sections 1** and **2**, Avaya IP Office Voicemail Pro was used in the reference configuration, running on a Windows 2003 Server. The installation and provisioning of Avaya IP Office Voicemail Pro is beyond the scope of this document. See reference **[4]** & **[5]** for more information on installing and provisioning Avaya IP Office Voicemail Pro.

To view or change Avaya IP Office Voicemail settings, select the **Voicemail** tab as shown in the following screen. The settings presented here simply illustrate the reference configuration and are not intended to be prescriptive.

- Set **Voicemail Type: Voicemail Lite/Pro**.
- Set **Voicemail IP Address:** to the IP address of the platform running Voicemail Pro.
- Other parameters on this screen may be set according to customer requirements.
- Click the **OK** button.



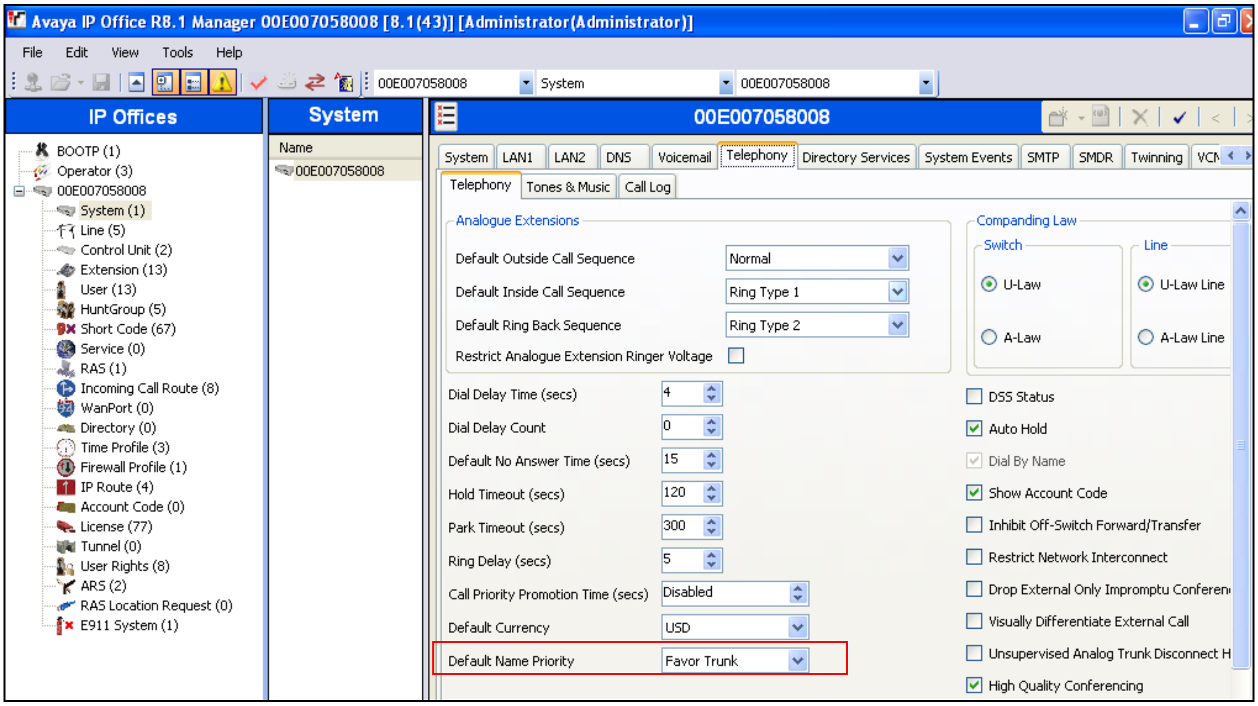
5.3.5. System Telephony Configuration

To view or change telephony settings, select the **Telephony** tab and **Telephony** sub-tab as shown in the following screen. The settings presented here simply illustrate the reference configuration and are not intended to be prescriptive.

In the reference configuration, the **Inhibit Off-Switch Forward/Transfer** box is unchecked so that call forwarding and call transfer to PSTN destinations via the AT&T Business in a Box with IPFR-EF service can be tested.

The **Companding Law** parameters are set to **U-LAW** as is typical in North America. Other parameters on this screen may be set according to customer requirements.

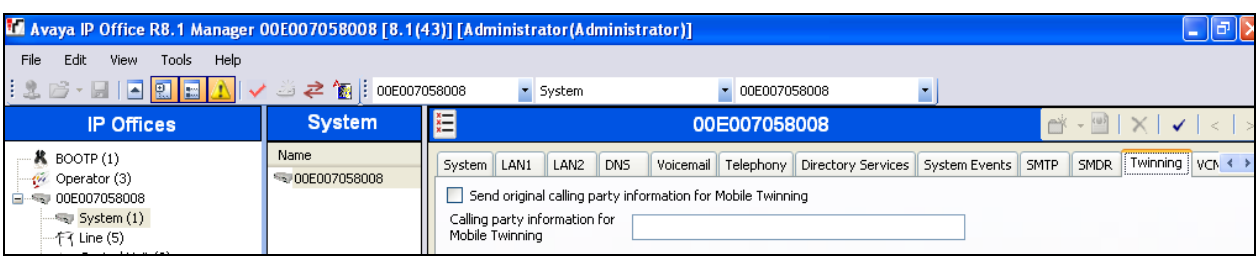
OPTIONAL: The **Default Name Priority** can be relevant to SIP Trunking. The option to **Favor Trunk** or **Favor Directory** can be set system-wide using the screen below, or set uniquely for each line. **Favor Trunk** was used in the reference configuration. With the option set to **Favor Directory**, Avaya IP Office will prefer to display names found in a personal or system directory over those arriving from the far-end, if there is a directory match to the caller ID. This capability is also defined in the **SIP Line** tab in **Section 5.4**. A user's personal directory example is shown in **Section 5.5.2**.



5.3.6. System Twinning Configuration

To view or change Twinning settings, select the **Twinning** tab as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked, and the **Calling party information for Mobile Twinning** is left blank, in the reference configuration.

With this configuration, and related configuration of Diversion Header on the SIP Line (**Section 5.4**), the true identity of a PSTN caller can be presented to the twinning destination (e.g., a user's mobile phone) when a call is twinned out via the AT&T Business in a Box with IPFR-EF service.

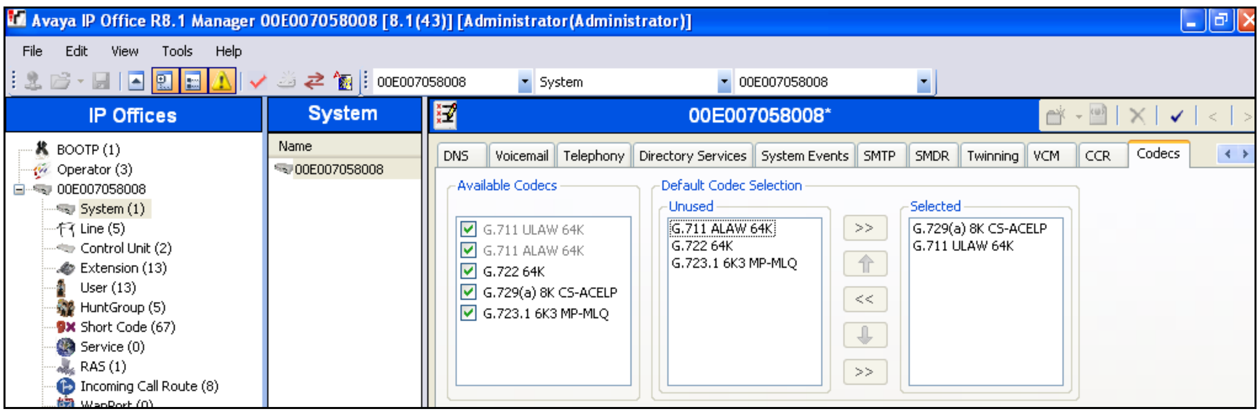


5.3.7. System Codecs Configuration

Navigate to the **System → Codecs** tab. On the left, observe the list of **Available Codecs**. By selecting codecs in this column, they will appear in the **Default Codec Selection → Unused** column. Codecs may be selected from the **Unused** list and moved to the **Selected** column by use of the >> button, thereby making the selected codecs available in other screens where codec configuration may be performed (e.g., Sip Lines and Extensions).

The up and down arrow buttons are used to order the selected codecs. By default, all IP (SIP and H.323) lines (included the SIP Line) and extensions will assume the system default **Selected** codec list, unless configured otherwise for the specific line or extension. When completed, click on **OK** (not shown).

Note - In the reference configuration the System and Extension (see **Section 5.5***)* codec lists specify G.711mu and G.729A (in that order), and the SIP Line (see **Section 5.4.3)** offers G.729A and G.711mu (in that order). In this manner local Avaya IP Office calls will offer G.711mu first, and SIP trunk calls will offer G.729A first.



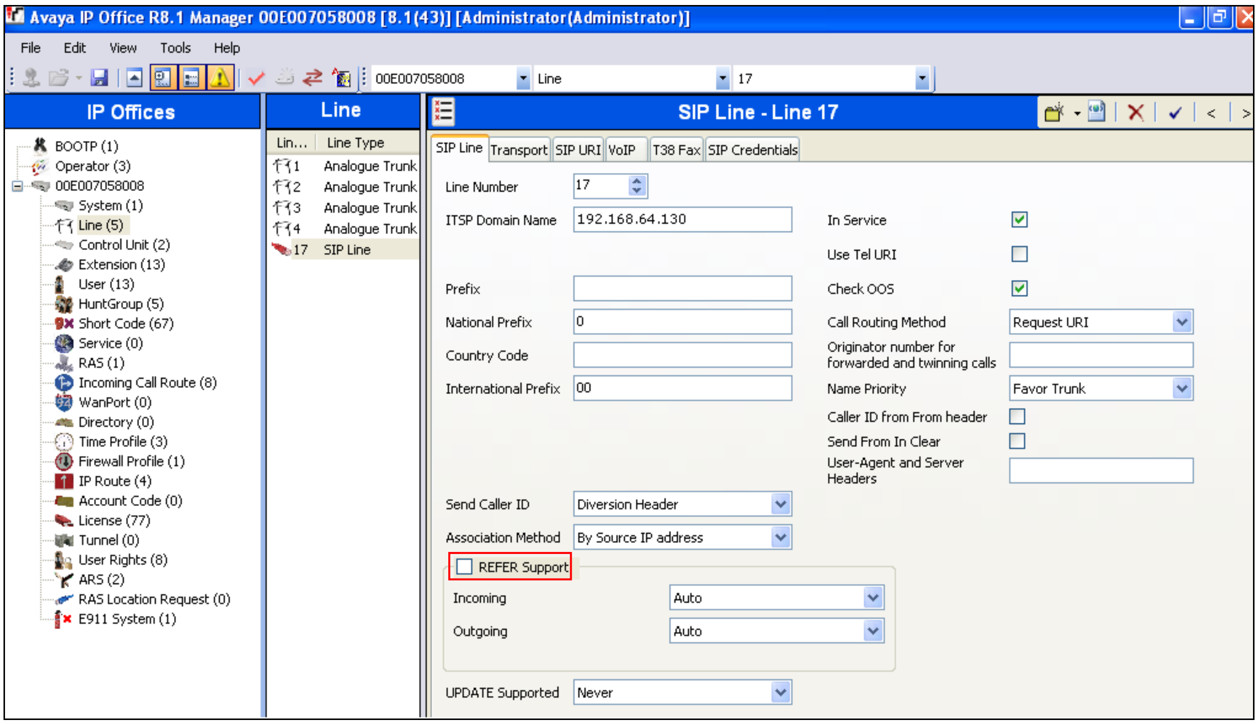
5.4. SIP Line

The **SIP Line** tab in the Details pane is shown below for **Line Number 17**, used for the AT&T SIP Trunk. Note, if no SIP Line exists, right click on the **Line** item in the **Navigation** pane and select **New → SIP Line**. SIP Line 17 will be the first SIP Line number created. The SIP Line form is completed as follows:

- **ITSP Domain Name:** Set to the IP Office LAN2 address as defined in **Section 5.3.3** (**192.168.64.130**).
- **In Service** and **Check OOS:** These boxes are checked (default).
  - Note that the Out Of Service (OOS) option is used in conjunction with SIP OPTIONS (see **Section 5.11**).
- **Call Routing Method:** Specify matching for **Incoming Call Routes** (**Section 5.7**). Matched values based on the **Request URI**, or **To Header** contents, may be selected. In the reference configuration, the default **Request URI** setting was used.
- **Country Code:** Use the default <blank>.

- **Send Caller ID:** Set to **Diversion Header**. This is required by the AT&T Business in a Box with IPFR-EF service for call redirection scenarios (e.g., Call Forward, Mobile Twinning).
- **REFER Support:** Verify that this open is *not* selected (default). See **Section 2.2, item 1**.
- Use the default values for the other fields.
- Click **OK** (not shown).

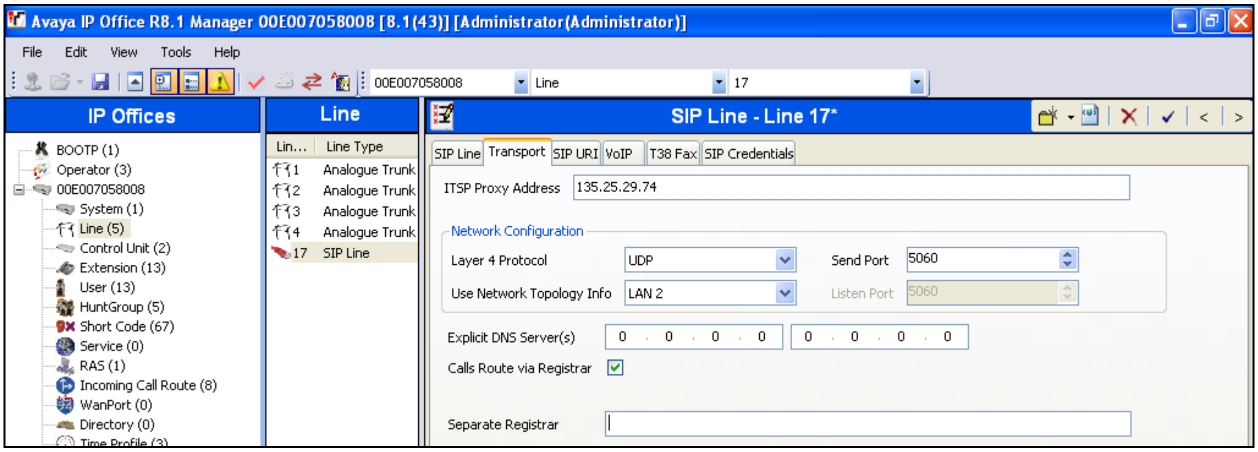
Optional: As described in **Section 5.3.5**, the **Name Priority** parameter may retain the default **Favor Trunk** setting, or can be configured to **Favor Directory**. The default **Favor Trunk** setting was used in the reference configuration.



5.4.1. SIP Line - Transport Tab

Select the **SIP Line → Transport** tab and configure the following:

- **ITSP Proxy Address:** Set to the AT&T Business in a Box with IPFR-EF Border Element IP address (e.g., **135.25.29.74**).
- **Network Configuration → Layer 4 Protocol**. Set to **UDP**.
- **Network Configuration → Send Port:** Set to **5060** (default).
- **Network Configuration → Use Network Topology Info:** Set to **LAN 2**. This associates the SIP Line with the parameters in the **System → LAN2 → Network Topology** tab.
- **Calls Route via Registrar:** Enabled (default).
- **Click OK** (not shown).



5.4.2. SIP Line - SIP URI Tab

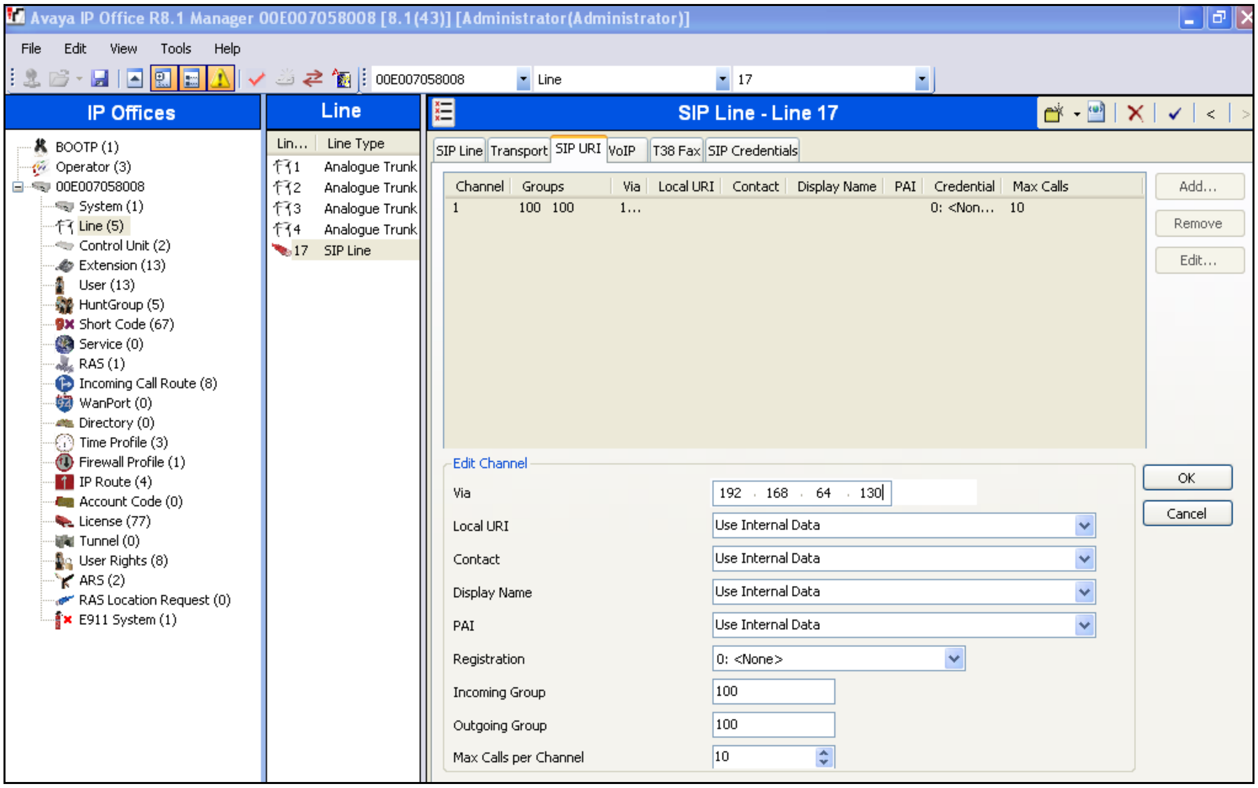
Select the **SIP Line → SIP URI** tab. On this form a list of the DNIS digits delivered by AT&T is created. To add a new SIP URI, click the **Add…** button. In the bottom of the screen, a **New Channel** area will be opened. Entries may be specified in two ways:

1. A "wild-card" entry that will use the contents of SIP headers containing "calling" information.

> **Note** - When this method is used, the inbound AT&T DNIS digits must be specified for an Avaya IP Office User or Hunt Group on its corresponding **SIP** tab (see **Section 5.5**). Otherwise the call may be denied.

In this method the following information is specified:

- The **VIA** field will automatically be populated with the IP address defined in the **ITSP Domain Name** field defined in **Section 5.4**.
- **Local URI**, **Contact**, **Display Name**, **and PAI**: Set to **Use Internal Data**.
- **Registration:** Set to the default **0: <None>**.
- **Incoming Group:** Set here to **100**. This value references the **Incoming Call Routes** in **Section 5.7**.
- **Outgoing Group:** Set to **100**. This will be used for routing outbound calls to AT&T via the **Short Codes** (**Section 5.6**) or **ARS** configuration (**Section 5.8**).
- **Max Calls per Channel:** In the reference configuration this was set to **10**. This sets the maximum number of simultaneous calls that can use the URI before Avaya IP Office returns busy to any further calls.
- Click **OK** to save the information.

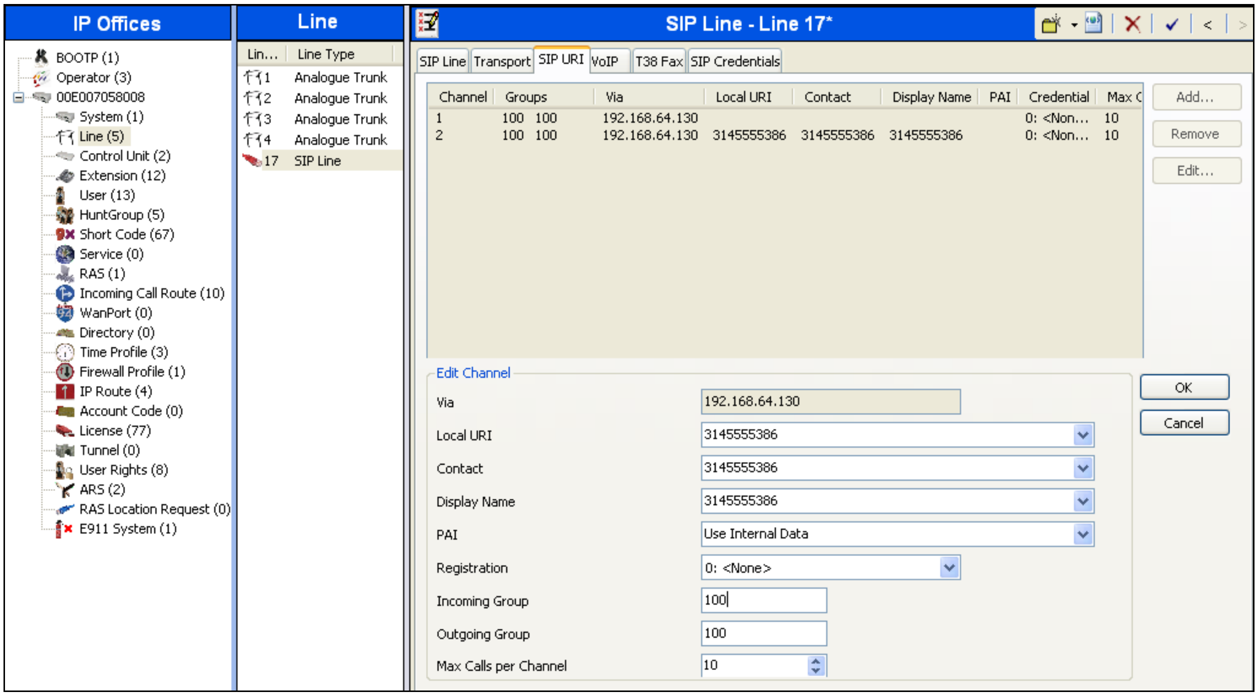


2. A specific entry that will match inbound DNIS digits from AT&T.

> **Note** – This method must be used for Avaya IP Office call destinations other than Users or Hunt Groups or the calls will be denied.

In this method the following information is specified:

- **Local URI**, **Contact**, and **Display Name**: Set to an AT&T DNIS number (e.g, **3145555386).**
- **PAI**: Set to **Use Internal Data**. When set, Avaya IP Office will use PAI for privacy signaling (see **Section 5.6.3**).
- **Registration:** Set to the default **0: <None>**.
- **Incoming Group:** Set here to **100**. This value references the **Incoming Call Routes** in **Section 5.7**.
- **Outgoing Group:** Set to **100**. This will be used for routing outbound calls to AT&T via the **Short Codes** (**Section 5.6**) or **ARS** configuration (**Section 5.8**).
- **Max Calls per Channel:** In the reference configuration to this was set to **10**. This sets the maximum number of simultaneous calls that can use the URI before Avaya IP Office returns busy to any further calls.
- Click **OK** to save the information.



- To edit an existing entry, click an entry in the list (e.g., **Channel 2 →3145555386**), and click the **Edit…**button. In the bottom of the screen, the **Edit Channel** area will be opened.
- When all SIP URI entries have been added/edited, click **OK** at the bottom of the screen (not shown).

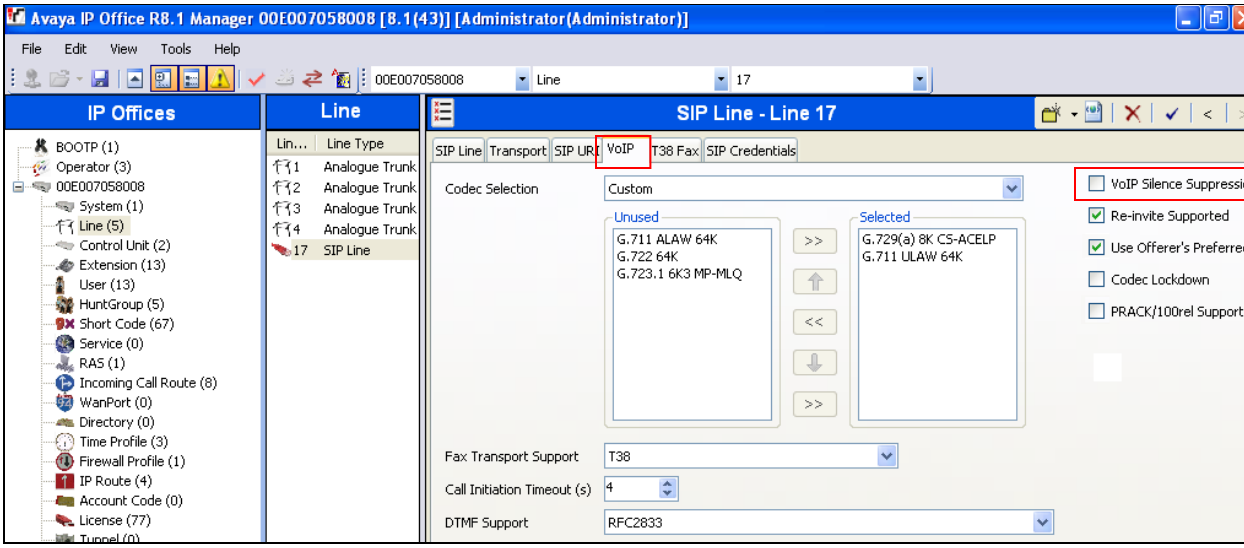
5.4.3. SIP Line - VoIP Tab

Select the **VoIP** tab. The **Codec Selection** drop-down box → **System Default** will list all available codecs, in the reference configuration, **Custom** was selected and **G729(a) 8K CS-ACELP,** and **G.711 ULAW 64K** were specified. This will cause Avaya IP Office to include these codecs in the Session Description Protocol (SDP) offer, and in the order specified. Note that G.729A is set as the preferred codec on the connection to the AT&T Business in a Box with IPFR-EF network.

- T.38 fax was used in the reference configuration. Set the **Fax Transport Support** drop-down menu to **T38**. Note that the **T.38 Fallback** option is *not* supported in the reference configuration (see **Section 2.2**). Note that Error Correction Mode (ECM) is enabled by default on the T.38 Fax tab (**Section 5.4.4**). ECM is supported by the AT&T Business in a Box with IPFR-EF service. G.711 fax also worked in the reference configuration; however T.38 is the preferred method.
- The **DTMF Support** parameter can remain set to the default value **RFC2833**.
- The **Re-invite Supported** parameter can be checked 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.
- Click **OK** (not shown).

Since the AT&T Business in a Box with IPFR-EF service does not require registration, the **SIP Credentials** tab need not be visited.

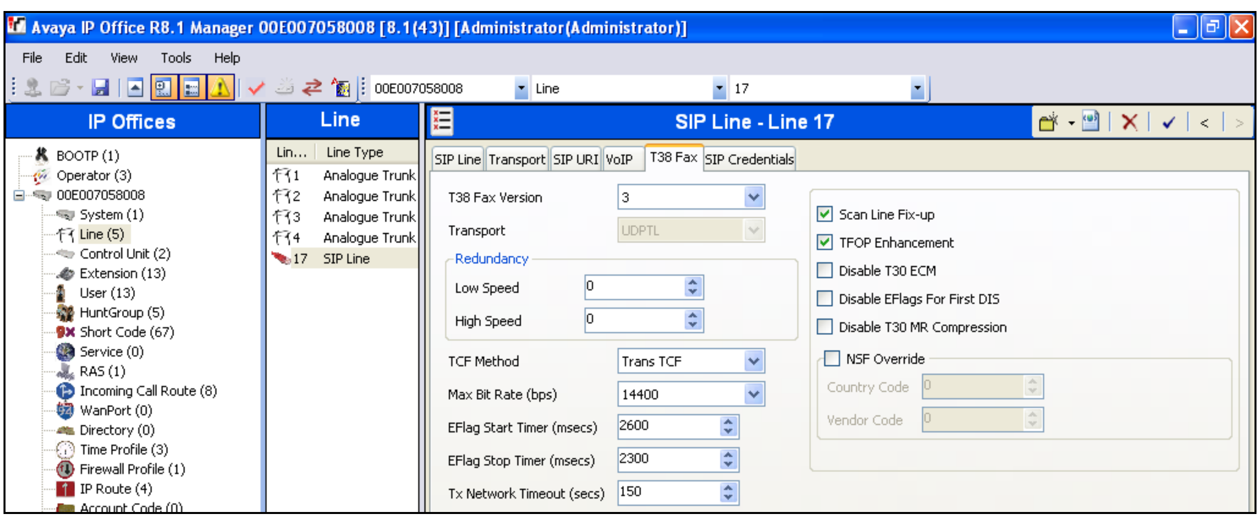
Note that by default the VoIP Silence Suppression box is not checked. This disables the use of the G.729B codec. If silence suppression is desired, check this box, and enable the **VoIP Silence Suppression** option on the Extension form **VoIP** tab (see **Section 5.5.2**).



5.4.4. SIP Line - T38 Fax

Note - The settings on this tab are only accessible if **Re-invite Supported** and a **Fax Transport Support** option are selected on the **VoIP** tab (**Section 5.4.3**).

Note - All default values were used in the reference configuration. Therefore the **Use Default Values** box is checked (not shown). If different settings are needed, uncheck this box to unlock the form.

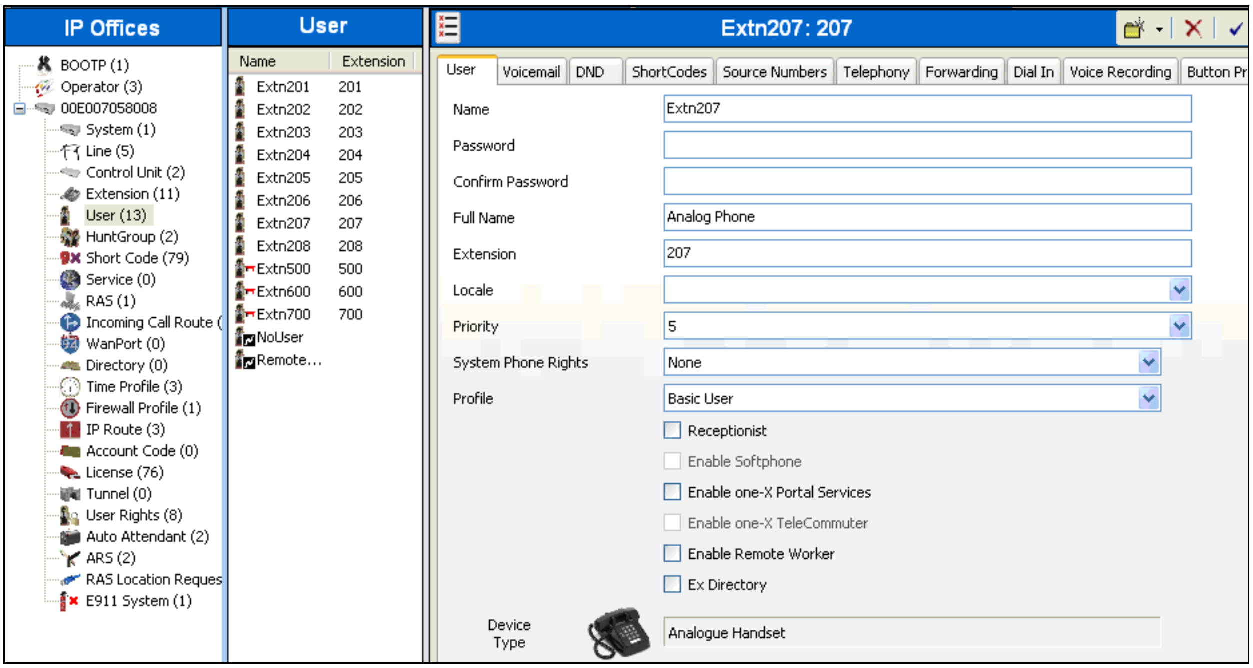


5.5. Users, Extensions, and Hunt Groups

In this section, examples of Avaya IP Office Users, Extensions, and Hunt Groups will be illustrated. 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. Analog User Extn207

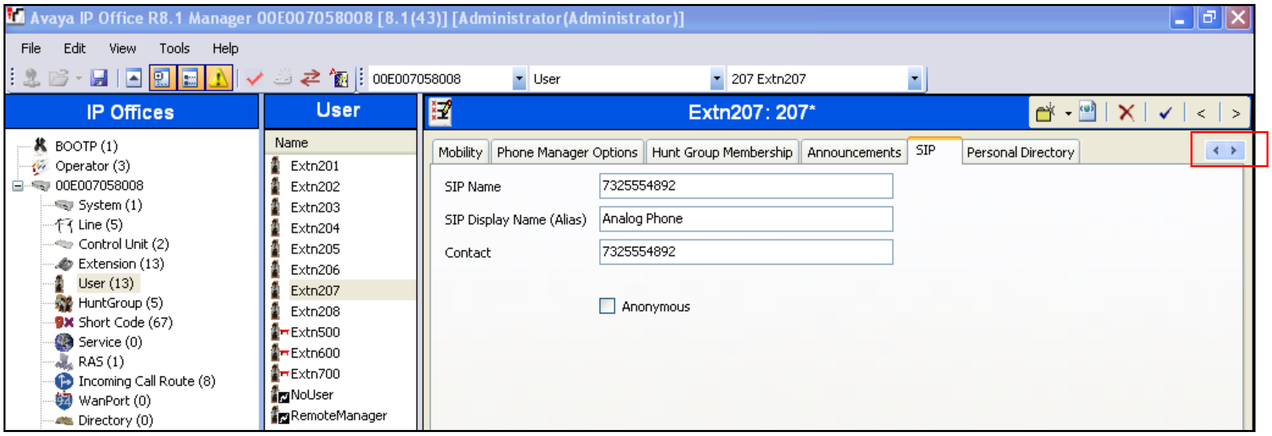
The following screen shows the **User** tab for User **Extn207**. This user corresponds to the Avaya Analog 6211 set.



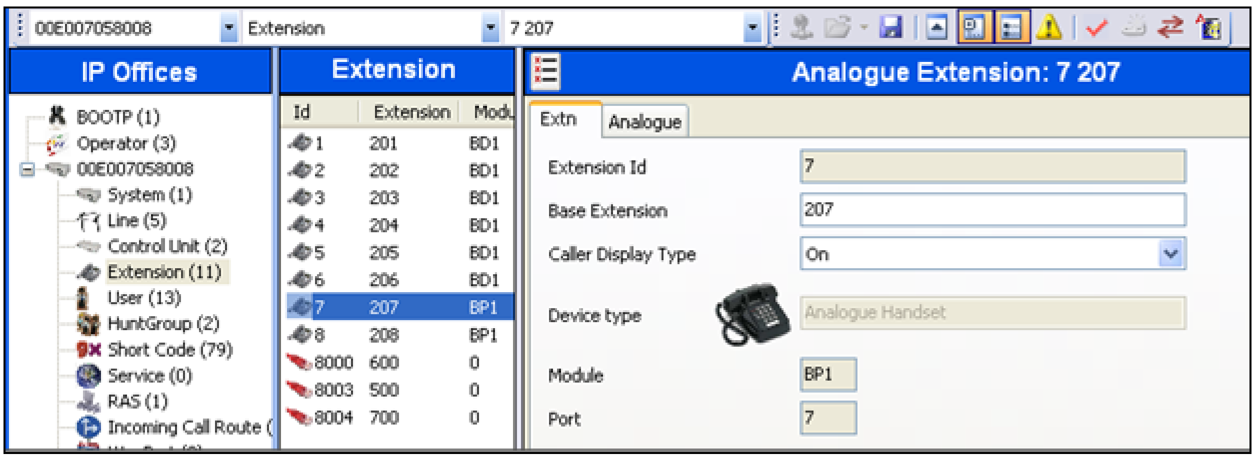
The following screen shows the **SIP** tab for User **Extn207** (use the arrow buttons in the upper right corner to navigate to the SIP tab). The **SIP Name** and **Contact** parameters are configured with the associated AT&T DNIS number of the user, (e.g., **7325554892**). These parameters configure the user part of the SIP URI in the From header for outgoing SIP trunk calls, and allow matching of the SIP URI for incoming calls, without having to enter this number as an explicit SIP URI for the SIP Line (see **Section 5.4.2**).

The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name.

If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network (see **Section 5.10**). See **Section 5.6** for a method of using a Short Code (rather than static user provisioning) to place an anonymous call.



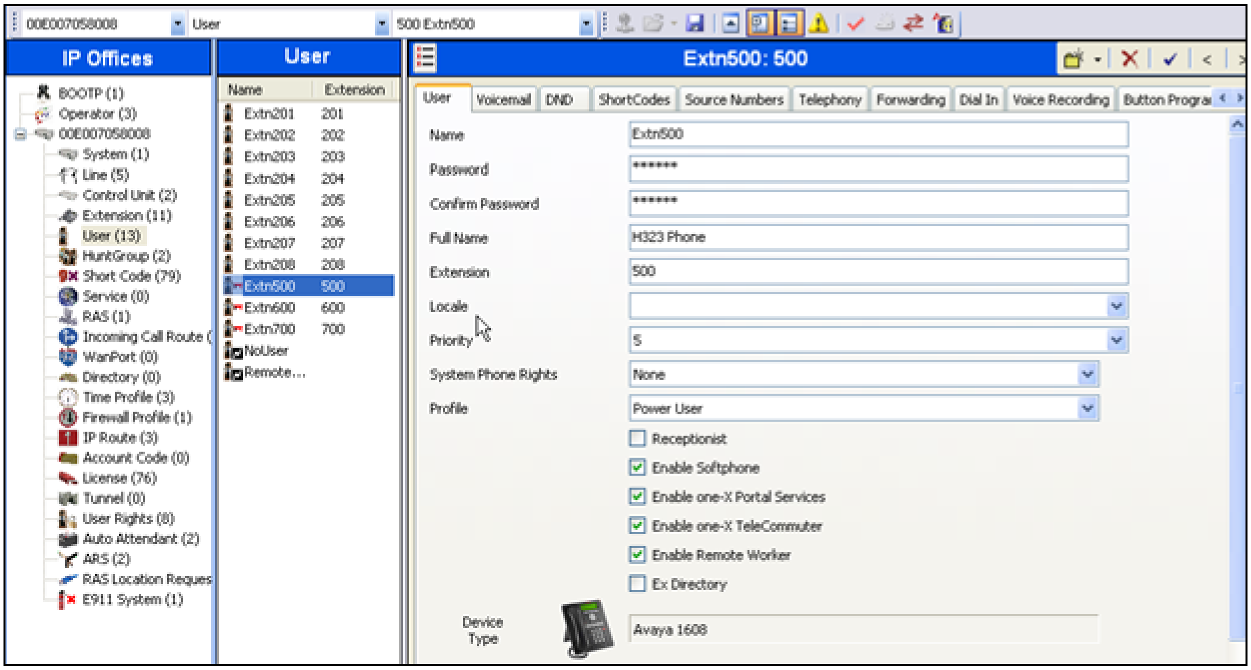
The following screen shows the Extension information for this user. To view, select **Extension** from the Navigation pane, and the appropriate extension from the Group pane (e.g., **207**).



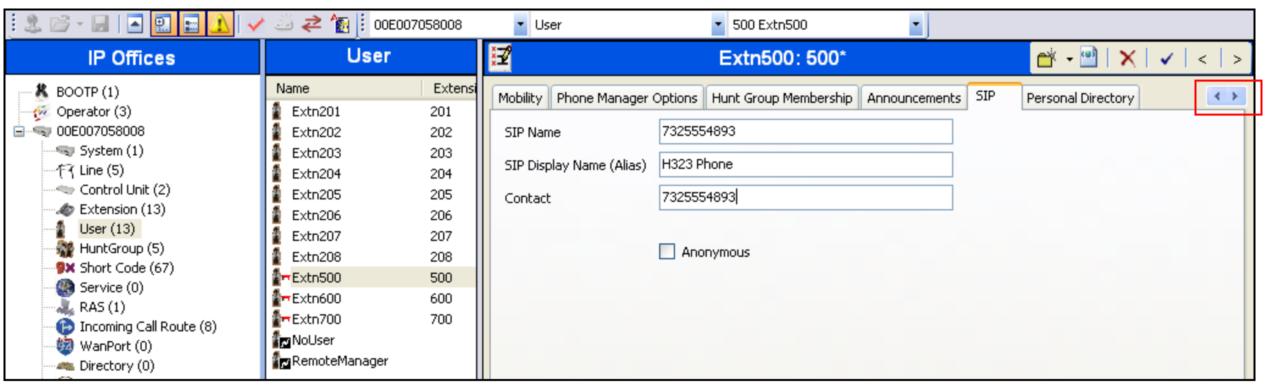
5.5.2. IP Phone User Extn500

To create a new extension, right click on **Extension** from the Navigation pane and select **New** and **H.323**. Alternatively edit an existing extension by selecting an extension in the Group pane. The following screen shows a 1608 IP Telephone provisioned in the **User** tab for User **Extn500**. In the reference configuration, this user will be granted "Power User" features.

- **Password**: This password is used by user applications such as SoftConsole, Phone Manager and TAPI. It is also used for users with Dial In access. Note that this is ***not*** the user's phone log in code (see the information on the **Telephony** tab → **Supervisor Settings** below), or their Voicemail mailbox password (see information on the **Voicemail** tab below).
- The **Profile** parameter is set to **Power User**.
- The **Enable Softphone** box is checked, along with other advanced capabilities.



Like the Analog Extn207 user, the **SIP** tab (use the arrow buttons in the upper right corner to navigate to the SIP tab) for User Extn500 is configured with a **SIP Name** and **Contact** specifying the user's associated AT&T DNIS number (e.g., **7325554893**).

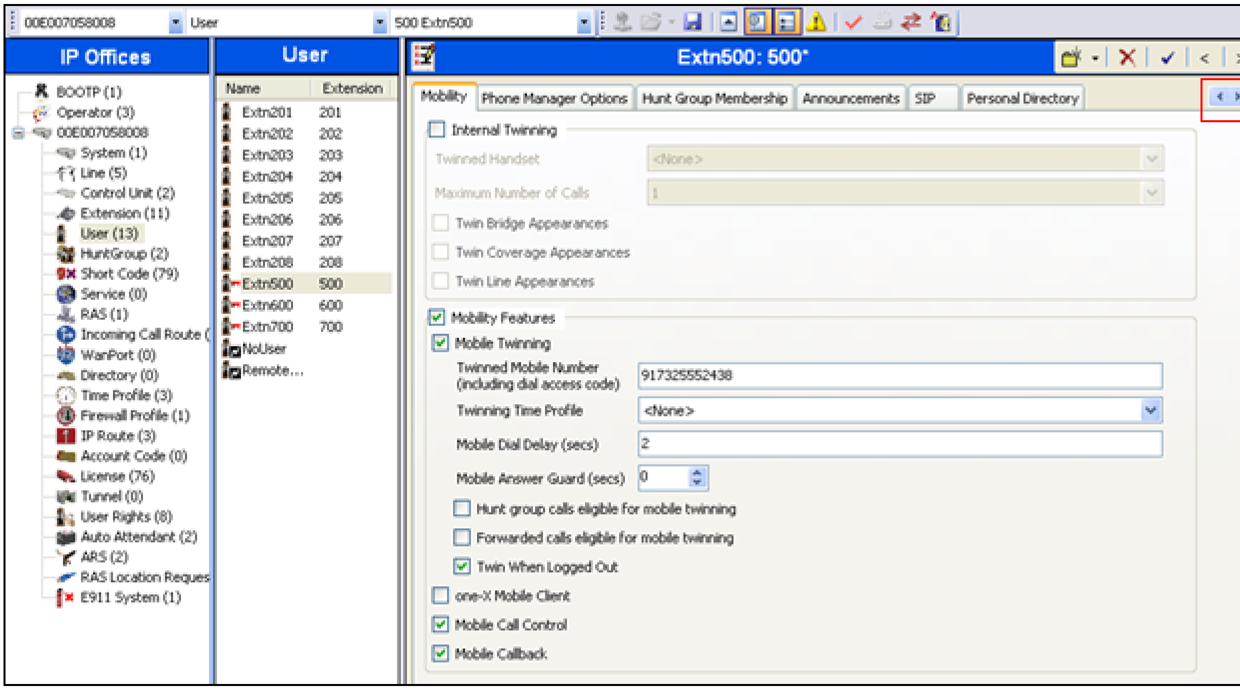


User **Extn500** will use the Mobile Twinning feature. The following screen shows the **Mobility** tab for User Extn500 (use the arrow buttons in the upper right corner to navigate to the Mobility tab).

The **Mobility Features**, **Mobile Twinning**, **Mobile Call Control**, **and Mobile Callback** boxes are checked**.**

The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case **917325552438** (note that the outbound call access code **9** is specified, as described for the Short Code 9N; in **Section 5.6.1**).

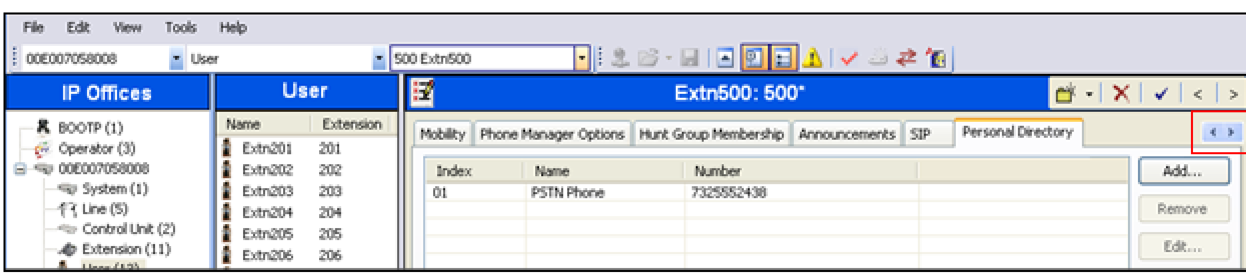
The **Mobile Call Control** and **Mobile Callback** features are accessed via Short Codes (as shown in **Section 5.6**) and Incoming Call Routes (as shown in **Section 5.7**).



Avaya IP Office offers a feature where users can define names in a Personal Directory, and display these names, based on the inbound calling number.

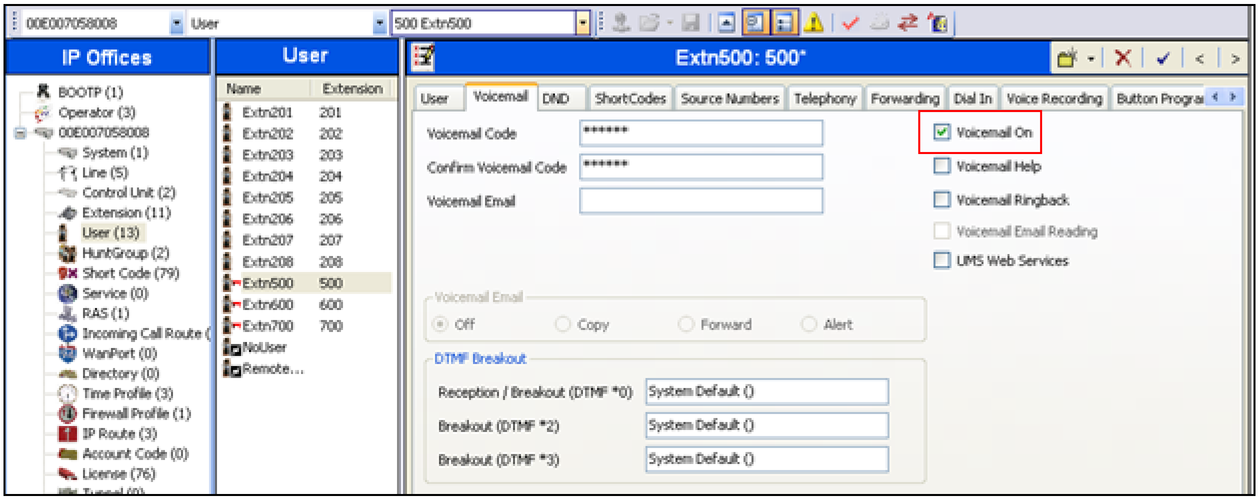
The following screen shows the **Personal Directory** tab for User **Extn500**. With the configuration shown below, if Extn500 receives an inbound AT&T call from the telephone number **7325552438**, the phone will display the name "PSTN Phone" (along with the number), even if AT&T provided a different name in the SIP INVITE message sent to Avaya IP Office.

Note – In the reference configuration, the SIP Line is configured with **Name Priority → Favor Trunk** (see **Section 5.4**). This setting needs to be changed to **Name Priority → Favor Directory**, to enable this feature.

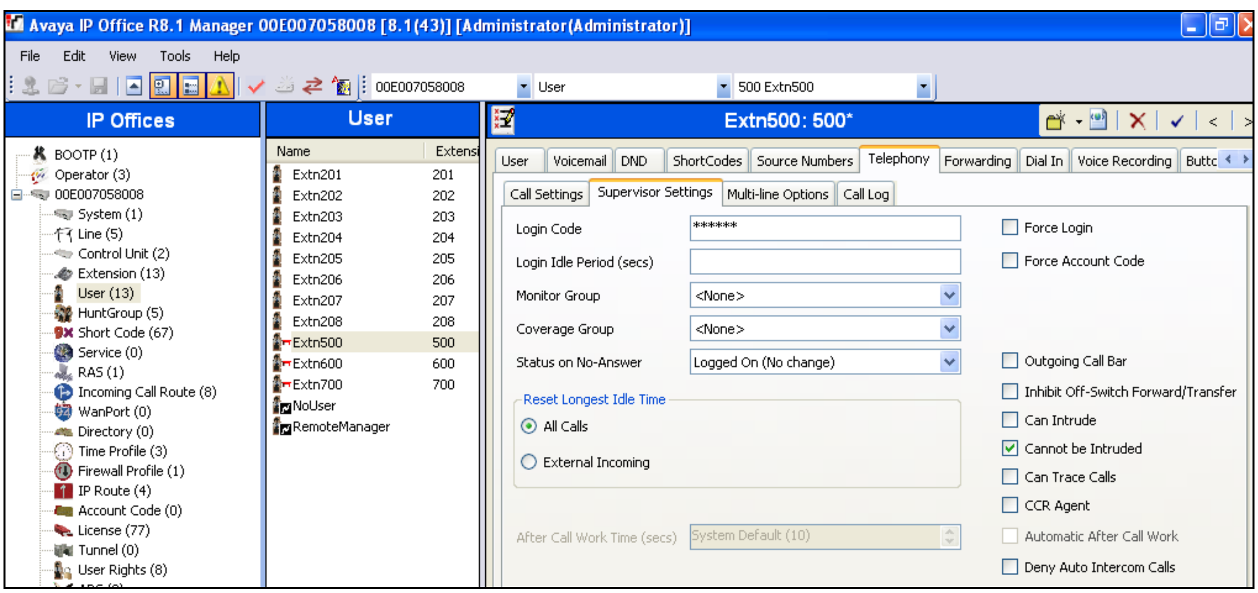


The following screen shows the **Voicemail** tab for User Extn500. The **Voicemail On** box is checked, and a Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters.

Voice mail navigation and retrieval were performed locally and from PSTN telephones, to test DTMF using RFC 2833, and to test assignment of an AT&T DNIS number to the "Voicemail Collect" feature (e.g., via the ***17** Short Code shown in **Section 5.6**). Note that the second configuration option described in the **SIP Line → SIP URI** tab (**Section 5.4.2**) is required for this type of inbound call to work.



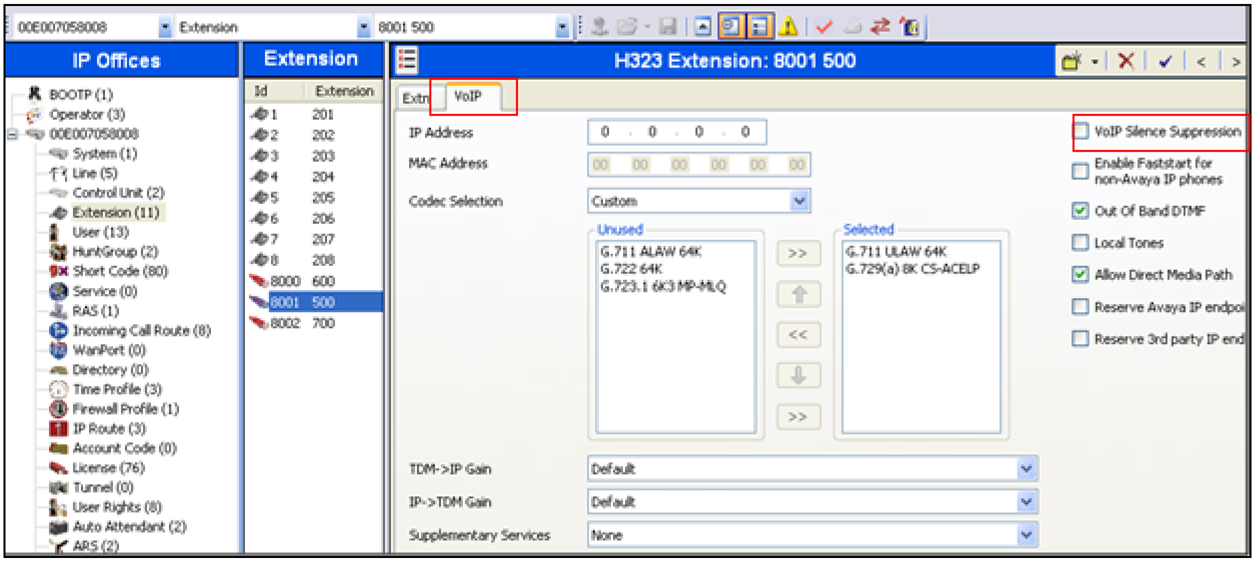
Select the **Telephony → Supervisor Settings** tab as shown below. The **Login Code** will be used by the telephone user as the phone login password.



The following screen shows the Extension information for this user, to illustrate the **VoIP** tab available for an IP Telephone. To view, select **Extension** from the Navigation pane, and the appropriate extension (**500**) from the Group pane. Select **VoIP** in the Details pane.

- Use the **IP Address** field default value (**0.0.0.0**).
- Note that the same codec list as shown in **Section 5.3.7** is used.
- Use defaults for the remaining fields.

Note that by default the VoIP Silence Suppression box is not checked (the same applies to provisioned SIP phones as shown in **Section 5.5.3**). This disables the use of the G.729B codec. If silence suppression is desired, check this box, and enable the Silence Suppression option on the SIP Line form VoIP tab (see **Section 5.4.3**).



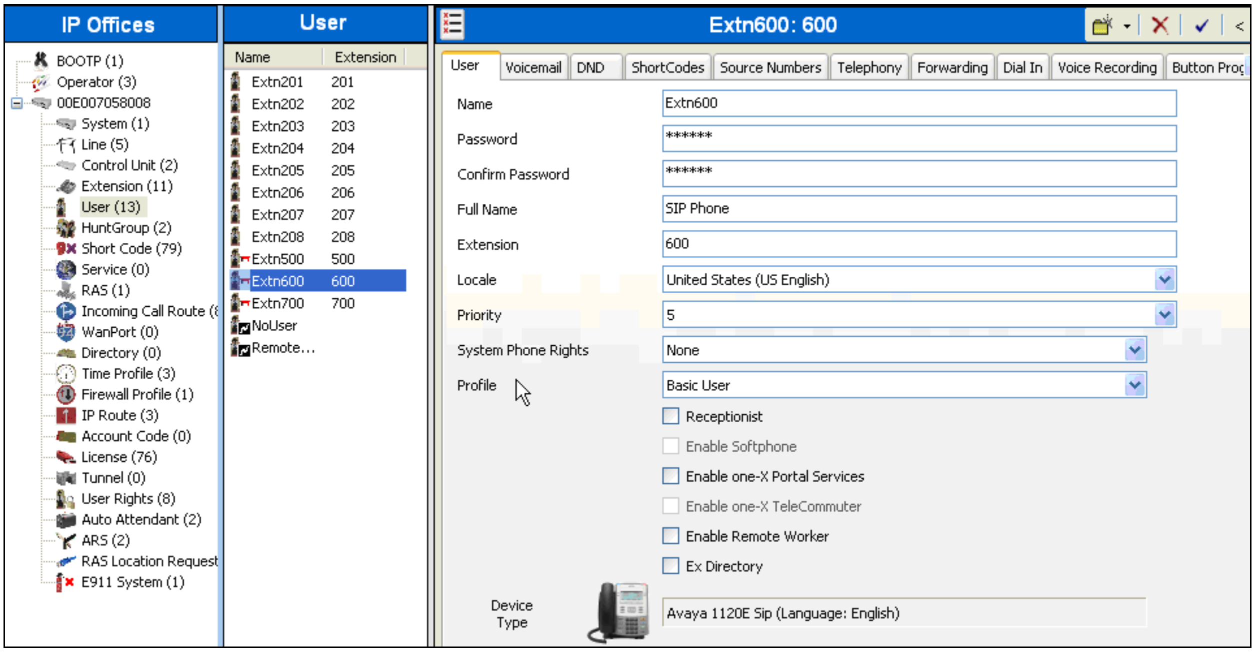
5.5.3. SIP Telephone Users (Avaya 1120E and Avaya IP Office SoftPhone)

In the reference configuration, an Avaya 1120E SIP telephone and Avaya IP Office SoftPhone were provisioned as SIP users. To create a new extension, right click on **Extension** from the Navigation pane and select **New** and **SIP**. Alternatively edit an existing extension by selecting an extension in the Group pane.

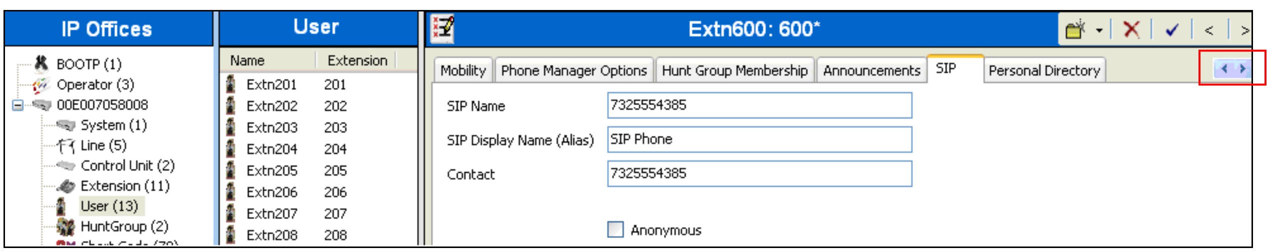
5.5.3.1 SIP Avaya 1120E

The following screen shows an1120E Telephone provisioned in the **User** tab for User **Extn600**.

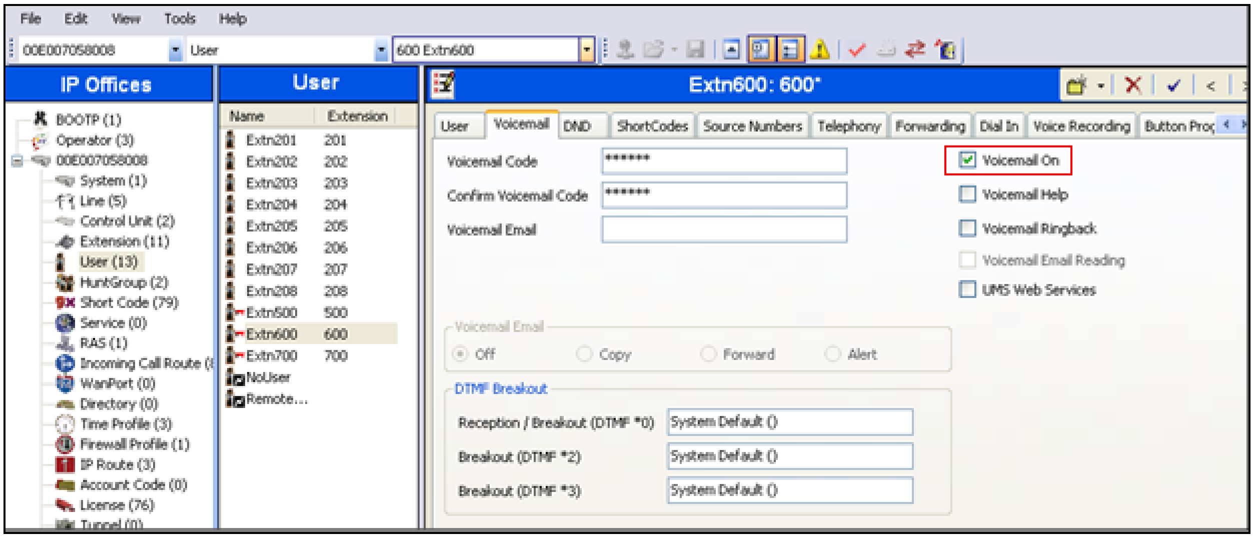
- **Password**: This password is used by user applications such as SoftConsole, Phone Manager and TAPI. It is also used for users with Dial In access. Note that this is ***not*** the user's phone log in code (see the information on the **Telephony** tab → **Supervisor Settings** below), or their Voicemail mailbox password (see the **Voicemail** tab below).
- In the reference configuration, the **Profile** parameter is set to **Basic User** (default). User Extn600 does not have the Mobile feature capabilities in the reference configuration.



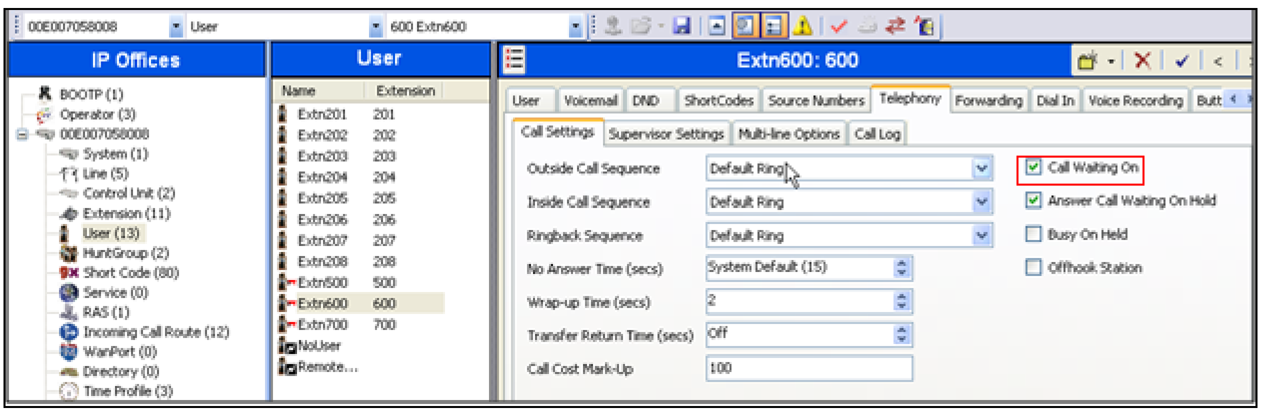
Like the H.323 Extn500 user, the **SIP** tab (use the arrow buttons in the upper right corner to navigate to the SIP tab) for User Extn600 is configured with a **SIP Name** and **Contact** specifying the user's associated AT&T DNIS number (e.g., **7325554385**). Optionally a user can be set to use privacy for all calls by selecting the **Anonymous** option.



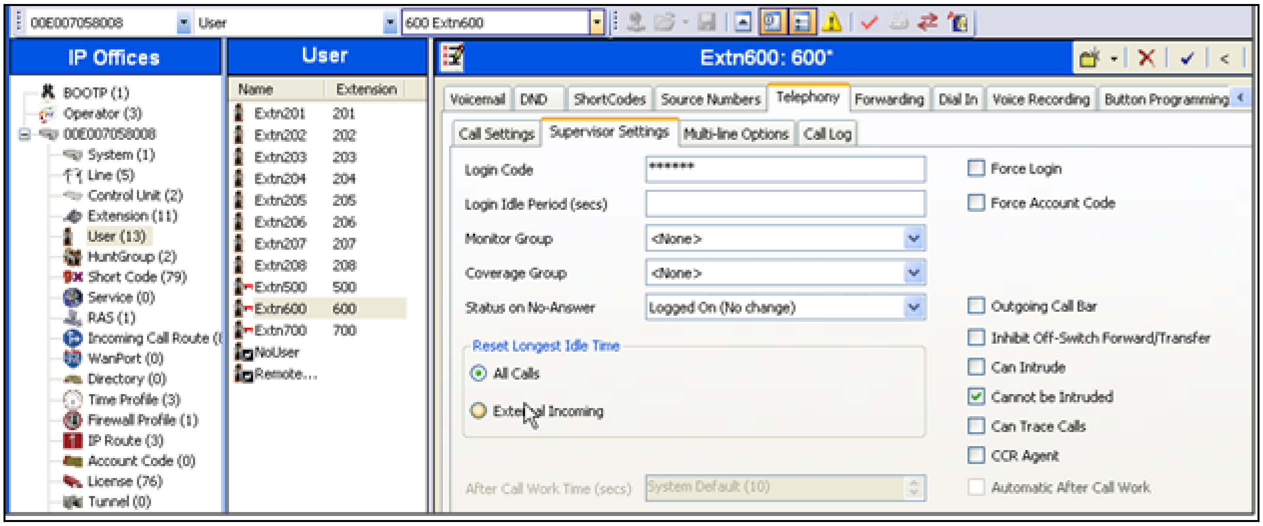
Like the H.323 Extn500 user, Extn600 also utilized the external Voicemail Pro server. The **Voicemail On** box is checked, and a Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters.



Select the **Telephony→ Call Settings** tab as shown below. Check the **Call Waiting On** box to allow multiple call appearances and transfer operations.

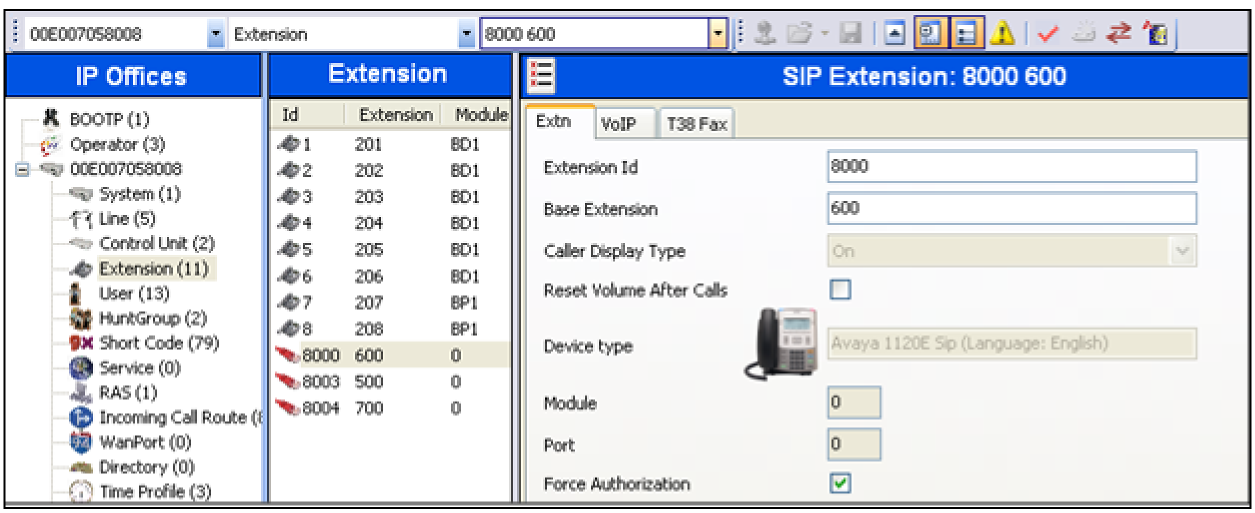


Select the **Telephony→ Supervisor Settings** tab as shown below. The **Login Code** will be used by the telephone user as the phone login password.



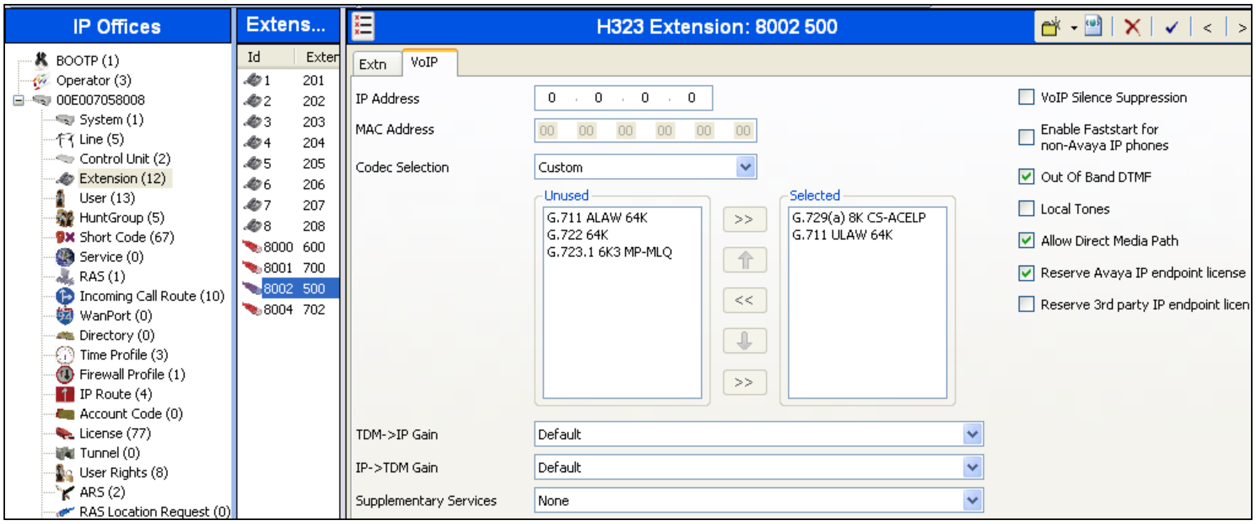
A new SIP extension may be added by right-clicking on **Extension** in the Navigation pane and selecting **New** and **SIP Extension**. Alternatively, an existing SIP extension may be selected in the group pane. The following screen shows the **Extn** tab for the extension corresponding to an Avaya 1120E.

- The **Extension ID** is automatically assigned by Avaya IP Office (e.g., **8000**). The **Base Extension** field is manually populated with the desired extension (e.g., **600**).
- Ensure the **Force Authorization** box is checked.



The following screen shows the **VoIP** tab for extension 600.

- The **IP Address** default value is used (**0.0.0.0**).
- Check the **Reserve Avaya IP endpoint license** box.
- In the reference configuration the **Codec Selection** parameter is set to **Custom**, and the same codec list used in **Section 5.3.7** is used.
- Other fields may retain default values.



5.5.3.2 SIP Avaya IP Office Softphone

NOTE – During testing, a DTMF issue was discovered with the Avaya IP Office Softphone (see **Section 2.2**, **Item 1**).

Repeat the steps shown in **Section 5.5.3.1** with the following settings.

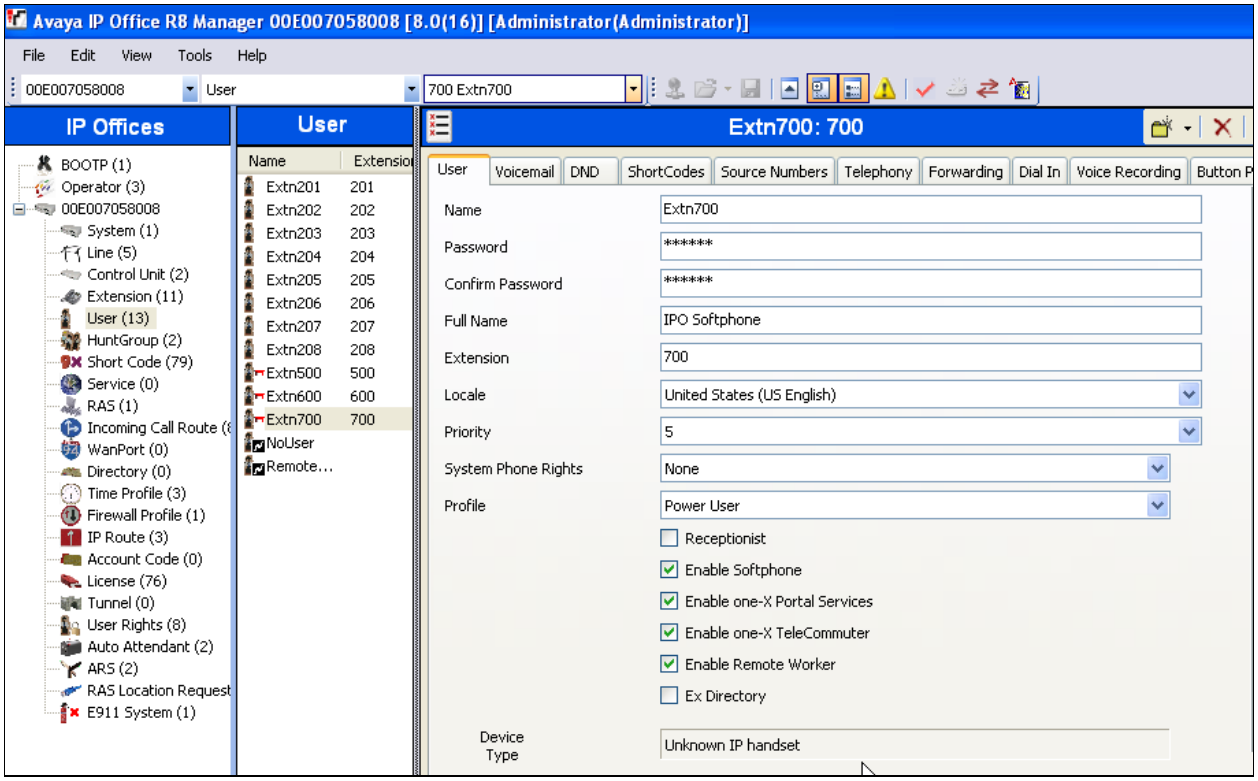
- **Defining a User**
  - **User** tab
    - The Avaya IP Office SoftPhone was provisioned as User **Extn700**.
    - The **Profile** parameter is set to **Power User** (see **Section 5.5.2**).
    - The **Enable Softphone** box is checked, the other enabled advanced capabilities shown, are optional.

  - **SIP** tab
    - **SIP Name** and **Contact** specifying the user's associated AT&T DNIS number (e.g., **7325554386**).

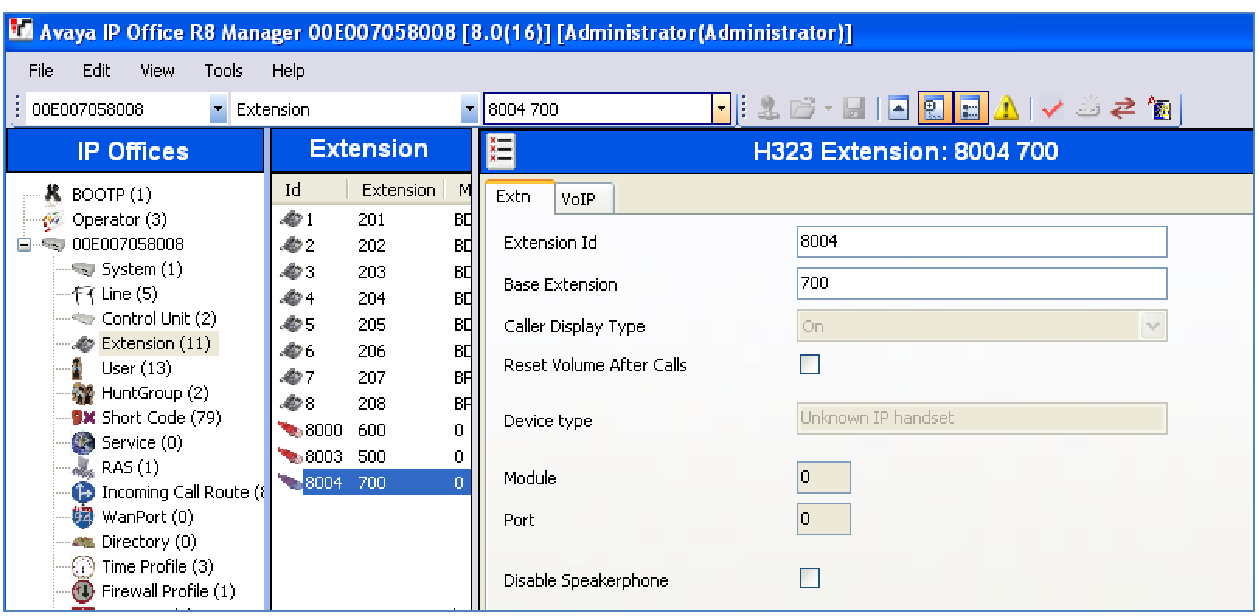
  - **Voicemail** tab
    - User Extn700 also utilized the embedded Voicemail. The **Voicemail On** box is checked, and a Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters.

  - **Telephony→ Call Settings** tab.
    - Check the **Call Waiting On** box to allow multiple call appearances and transfer operations.

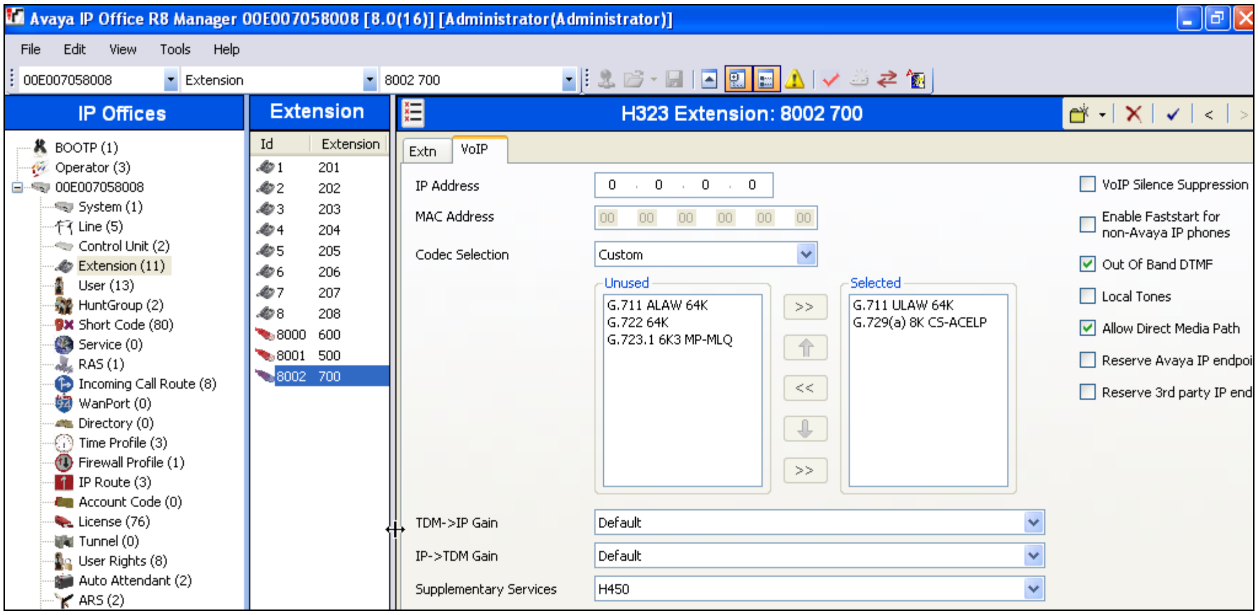
  - **Telephony→ Supervisor Settings** tab
    - The **Login Code** is the softphone login password.



- **Defining an Extension**
  - **SIP Extension** tab
    - The **Extension ID** is automatically assigned by Avaya IP Office (e.g., **8004**). The **Base Extension** field is manually populated with the desired extension (e.g., **700**).



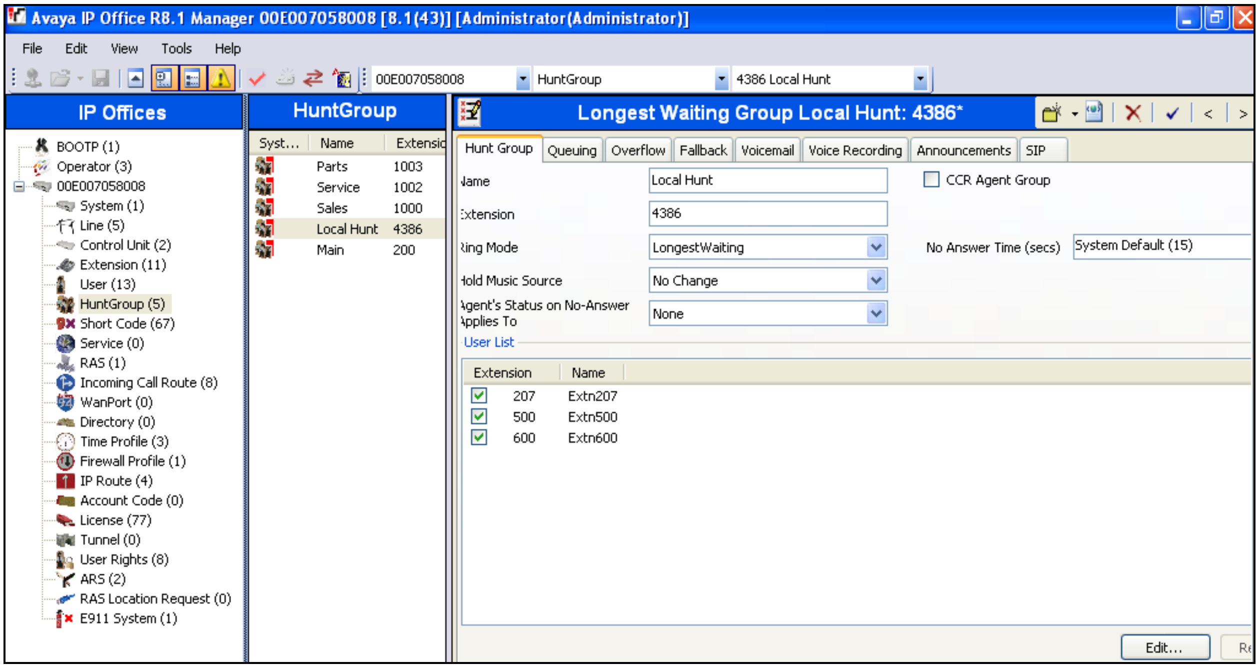
- **SIP Extension → VoIP** tab for extension 700.
    - The **IP Address** default value is used (**0.0.0.0**).
    - In the reference configuration the **Codec Selection** parameter is set to **Custom**, and the same codec list used in **Section 5.3.7** is used.
    - Other fields may retain default values.



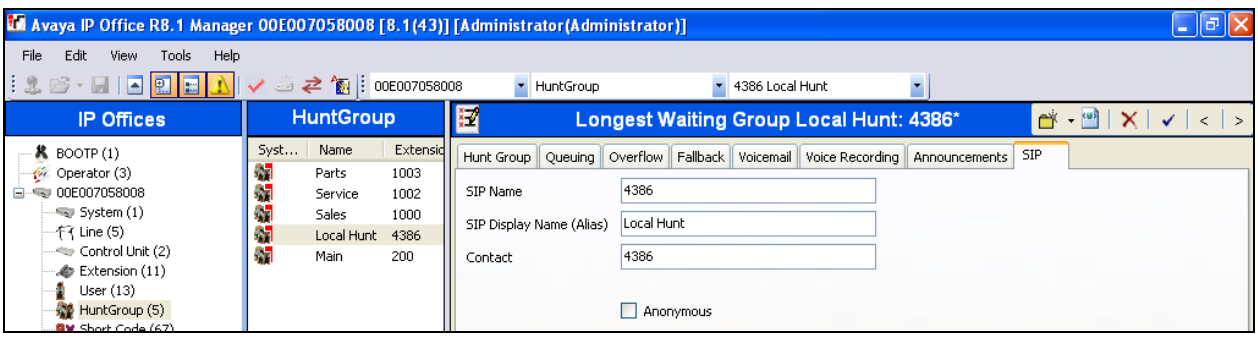
5.5.4. Hunt Groups

Users may also receive incoming calls as members of a hunt group. 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** from the Navigation pane, and the appropriate hunt group from the Group pane.

The following screen shows the **Hunt Group** tab for hunt group **4386**. This hunt group was configured to contain the Analog telephone (Extn207), the H.323 telephone (Extn500), and the SIP telephone Extn600 (1120E). In the reference configuration, these telephones extensions are rung based on idle time, due to the **Ring Mode** setting **LongestWaiting**. Click the **Edit** button to change the **User List** included in the Hunt Group from the list of available users.



The following screen shows the **SIP** tab for hunt group **4386**. The **SIP Name** and **Contact** are configured with the AT&T DNIS number **4386** (note that for testing purposes in the reference configuration, AT&T delivered this four digit DNIS number). In **Section 5.7**, an **Incoming Call Route** will map 4386 to this hunt group.



5.6. Short Codes

Avaya IP Office provides predefined Short Codes, however new Short Codes may be defined to match number strings to an action. 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** in the Navigation pane, and the Short Code to be configured in the Group pane.

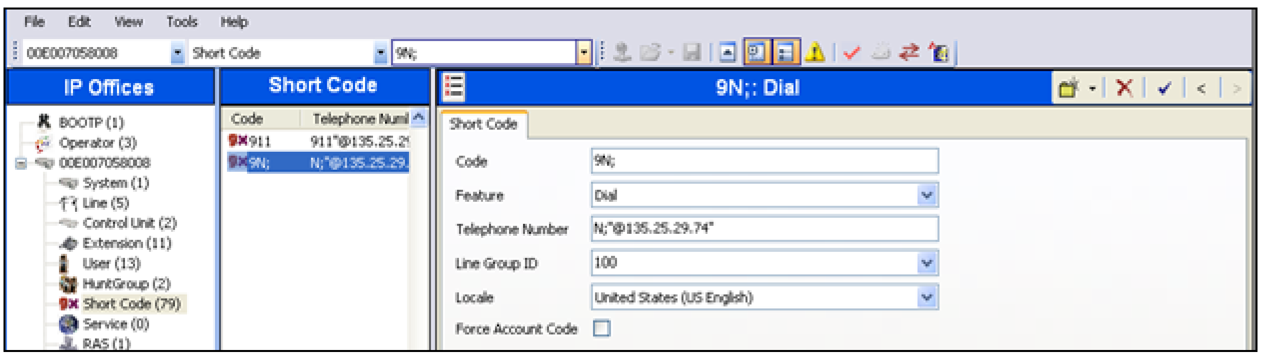
5.6.1. Short Code Direct Dialing (no ARS)

In the screen shown below, the Short Code **9N;** is illustrated (note the semicolon at the end of the string). This Short Code will allow an Avaya IP Office user to dial the digit 9 followed by any telephone number, symbolized by the letter **N**, to reach the SIP Line to AT&T. The variable

N can be any number such as a 10-digit number, a 1+10 digit number, a toll free number, directory assistance (e.g., 411), etc.

This Short Code does ***not*** use the ARS table, and this Short Code does not provide for alternate routing or an awareness of end of user dialing. When a users dial 9 plus the number, Avaya IP Office must wait for an end of dialing timeout before sending the SIP INVITE.

- The **Code** parameter is set to **9N;**
- The **Feature** parameter is set to **Dial**
- The **Telephone Number** parameter is set to **N"@135.25.29.74"** with the text string beginning with @ in quotes (address 135.25.29.74 is the IP address of the AT&T Business in a Box with IPFR-EF border element used in the test environment)**.** This field is used to construct the Request URI and To Header in the outgoing SIP INVITE message.
- The **Line Group ID** parameter is set to **100**, matching the number of the **Outgoing Group** configured on the **SIP URI** tab of **SIP Line 17** (see **Section 5.4.2**).
- Click the **OK** button (not shown).



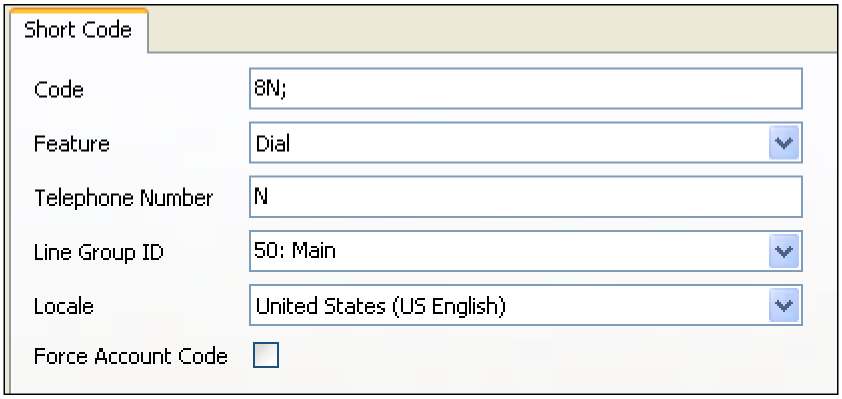
5.6.2. Short Code Dialing via Automatic Route Selection (ARS)

In the reference configuration only a single connection existed to the AT&T Business in a Box with IPFR-EF service. Therefore alternate routing via the Avaya IP Office ARS capability was not used. However, for completeness, defining an ARS Short Code is described below.

In the screen shown below, the Short Code **8N;** is illustrated (note the semicolon at the end of the string). This simple Short Code will allow an Avaya IP Office user to dial the digit 8 followed by any telephone number, symbolized by the letter **N**, to reach the SIP Line to AT&T. However, Avaya IP Office will first consult the ARS table defined in **Section 5.8**. The variable **N** can be any number such as a 10-digit number, a 1+10 digit number, a toll free number, directory assistance (e.g., 411), etc.

- The **Code** parameter is set to **8N;**
- The **Feature** parameter is set to **Dial**
- The **Telephone Number** parameter is set to **N**

- The **Line Group ID** parameter is set to **50:Main** (default value provided by Avaya IP Office).
- Click the **OK** button (not shown).

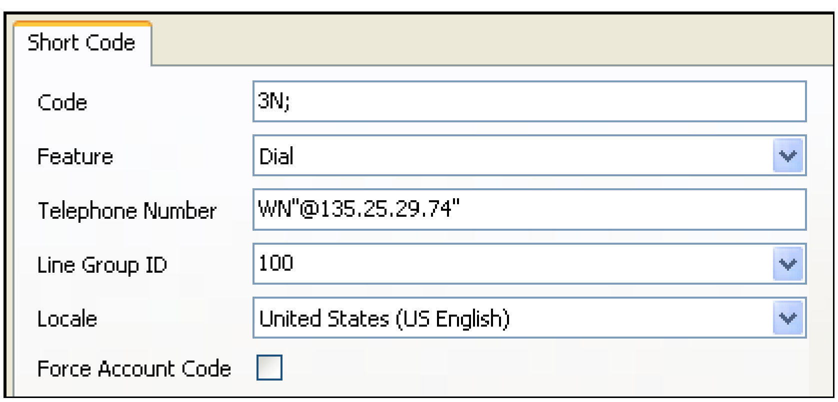


5.6.3. Privacy Dialing

Optionally, add or edit a Short Code that can be used to access the SIP Line anonymously. In the screen shown below, the Short Code **3N;** is illustrated. This code is similar to the **9N;** code. The **Telephone Number** field begins with the letter **W,** which means "withhold the outgoing calling line identification".

Note - When a user dials 3 plus the number, Avaya IP Office will include the user's telephone number in the **P-Asserted-Identity** (PAI) header along with the **Privacy: Id** header. The **From** and **Contact** headers will contain **Anonymous**.

- The **Code** parameter is set to **3N;**
- The **Feature** parameter is set to **Dial**
- The **Telephone Number** parameter is set to **WN"@135.25.29.74"**
- The **Line Group ID** parameter is set to **100**, matching the number of the **Outgoing Group** configured on the **SIP URI** tab of **SIP Line 17** (see **Section 5.4.2**).
- Click the **OK** button (not shown).

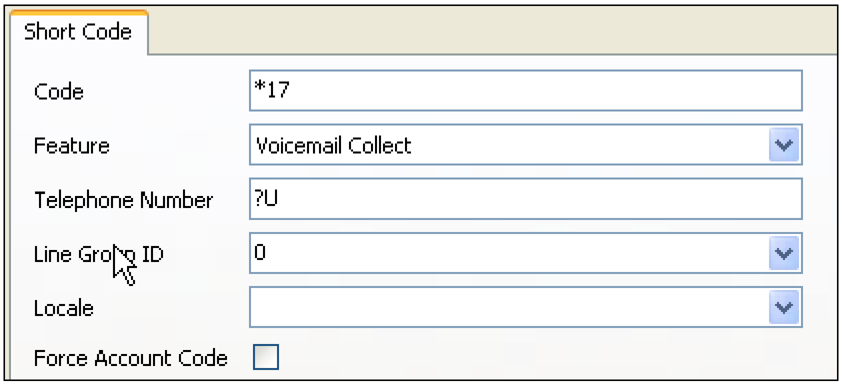


5.6.4. Feature Dialing

Optionally, add or edit a Short Code that can be used to access Avaya IP Office features rather than a SIP Line.

5.6.4.1 Voicemail Access

In this case, the **Code *17** is defined for **Feature → Voicemail Collect**. This Short Code will be used as one means to allow an AT&T DNIS number to be programmed to route directly to voice messaging, (via inclusion of this Short Code as the destination of an **Incoming Call Route** in **Section 5.7**).

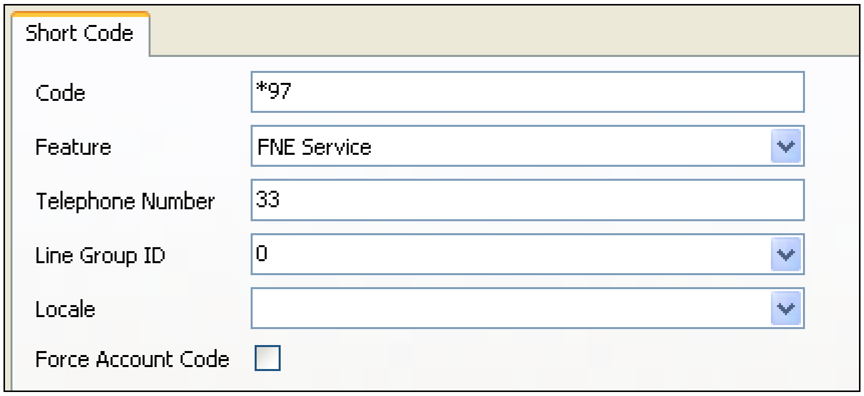


5.6.4.2 Feature Name Extension (FNE) Access

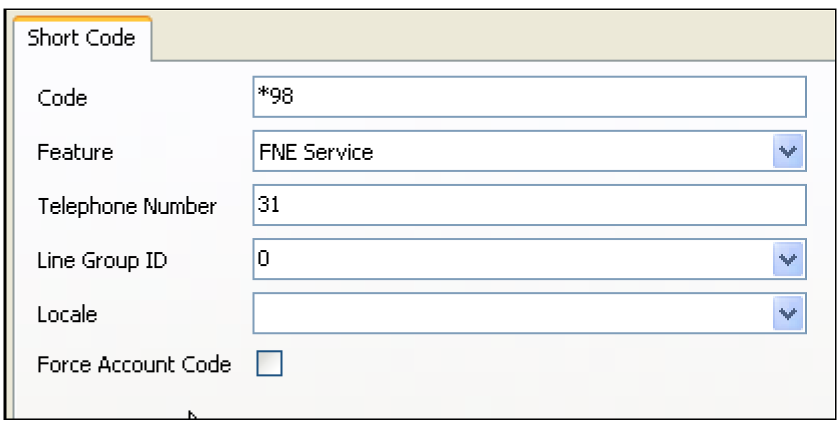
Two Avaya IP Office Mobility features, **Mobile Call Control** and **Mobile Callback**, are shown as examples of this function.

The following screen illustrates the **Code *97** which is defined for **Feature → FNE Service** and **Telephone Number → 33** for the Avaya IP Office Mobility feature **Mobile Callback** (note that 33 is predefined in Avaya IP Office for this feature). Code, ***97** was used as the destination of an Incoming Call Route for an AT&T DNIS number. This enabled DID calls from a configured twinning destination to be dialed, and then hung up by the caller while hearing ring back. Avaya IP Office will then call the caller back using the number defined in the **User → Mobility → Twinned Mobile Number** field (see **Section 5.5.2**), via AT&T Business in a Box with IPFR-EF service.

Note that for this feature to work, the inbound calling number information must match the number provisioned in the associated **User → Mobility → Twinned Mobile Number**. For example, in **Section 5.5.2** the number **917325552438** was defined for the H.323 set. Therefore, the inbound calling number must be **7325552438.**



The following screen illustrates another Mobility Short Code. In this case, the **Code *98** is defined for **Feature → FNE Service** and **Telephone Number → 31** for the Avaya IP Office Mobility feature **Mobile Call Control** (note that 31 is predefined in Avaya IP Office for this feature). Code ***98** was used as the destination of an Incoming Call Route for an AT&T DNIS number. This enabled DID access to Mobile Call Control from configured twinning destinations, allowing the mobile user to make calls as if the calls were made from the user's Avaya IP Office extension in the office.

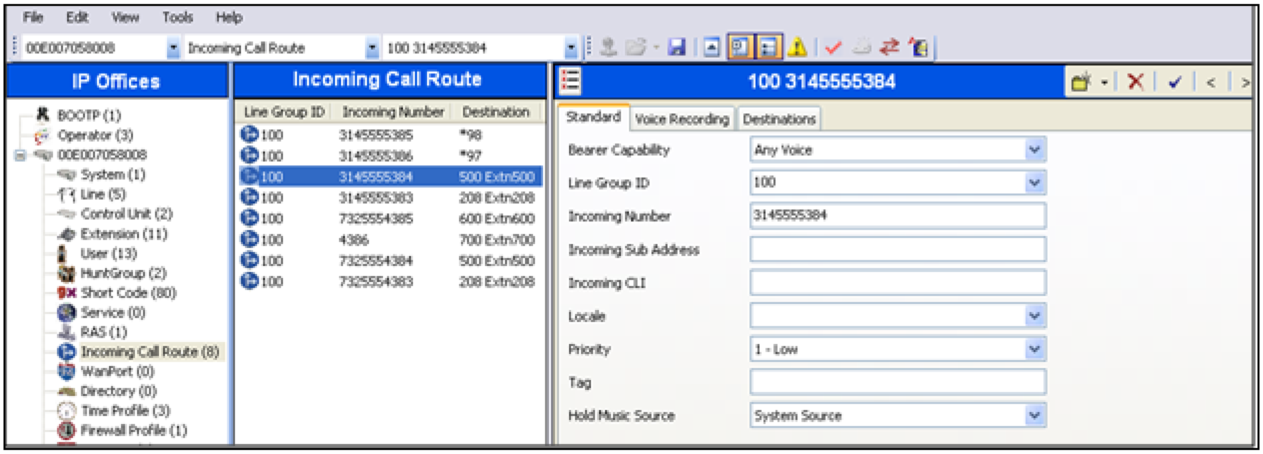


5.7. Incoming Call Routes

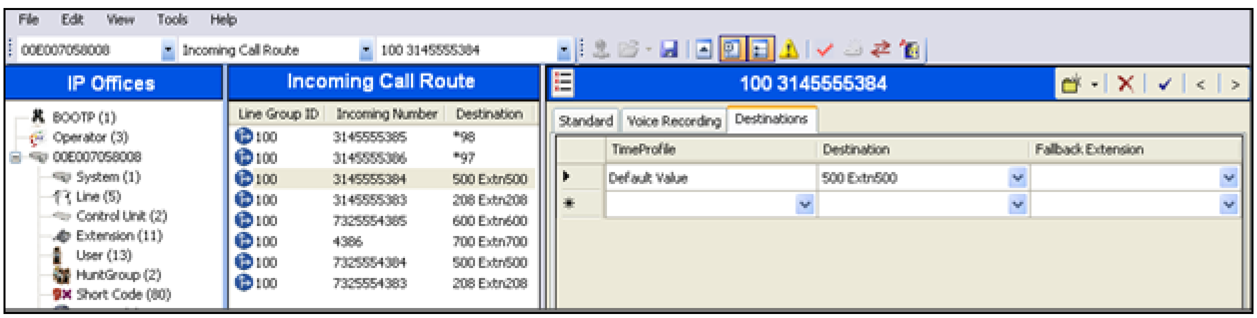
Each Incoming Call Route will map a specific AT&T DNIS number to a destination user, hunt group, or function on Avaya IP Office. To add an incoming call route, right click on **Incoming Call Route** in the Navigation pane, and select **New.** To edit an existing incoming call route, select **Incoming Call Route** in the Navigation pane, and the appropriate incoming call route to be configured in the Group pane.

In the screen shown below, the incoming call route for **Incoming Number → 3145555384** is illustrated.

The **Line Group ID** is set to **100**, matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to AT&T in **Section 5.4.2**.



Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when AT&T delivers DNIS digits 3145555384. In the reference configuration DNIS digits 3145555384 is associated with Avaya IP Office User **Extn500** (the 1608 H.323 telephone).



Repeat the process to route all AT&T DNIS numbers to their associated Avaya IP Office destinations.

Note that the **Destination** drop down menu may not contain all desired destinations (e.g., Short Codes for feature access as described in **Section 5.6**). In these cases the desired destination may be manually typed into the **Destination** field.

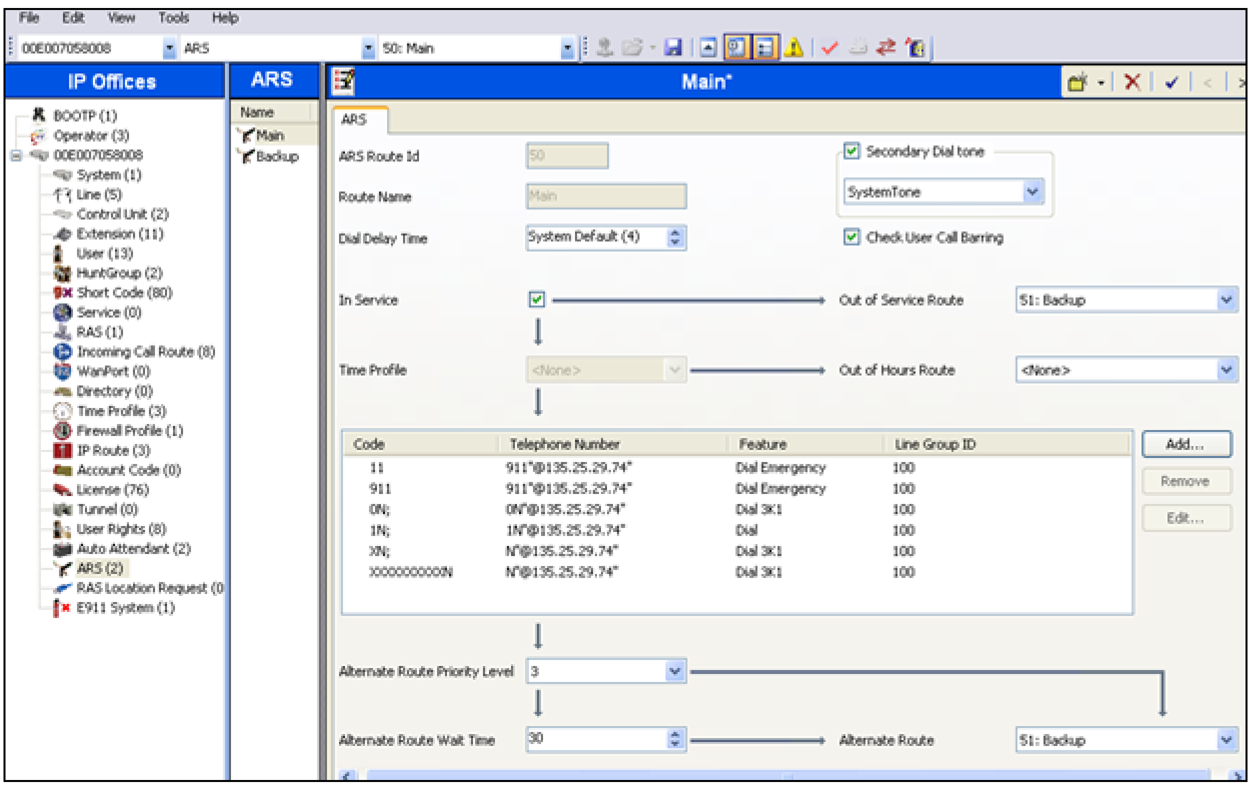
5.8. Automatic Route Selection (ARS) and Alternate Routing

In the reference configuration only a single connection existed to the AT&T Business in a Box with IPFR-EF service. Therefore, alternate routing via the Avaya IP Office ARS capability was not used (the direct dial 9N; Short Code described in **Section 5.6.1** was used instead). While detailed coverage of ARS is beyond the scope of these Application Notes, this section includes basic ARS screen illustrations and considerations, including alternate routing should the SIP Line be out of service or temporarily not responding.

Section 5.6.2 shows an example of an ARS Short Code used to access the ARS table. 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**. 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 **Main** (ARS Route ID 50). 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. Avaya 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.

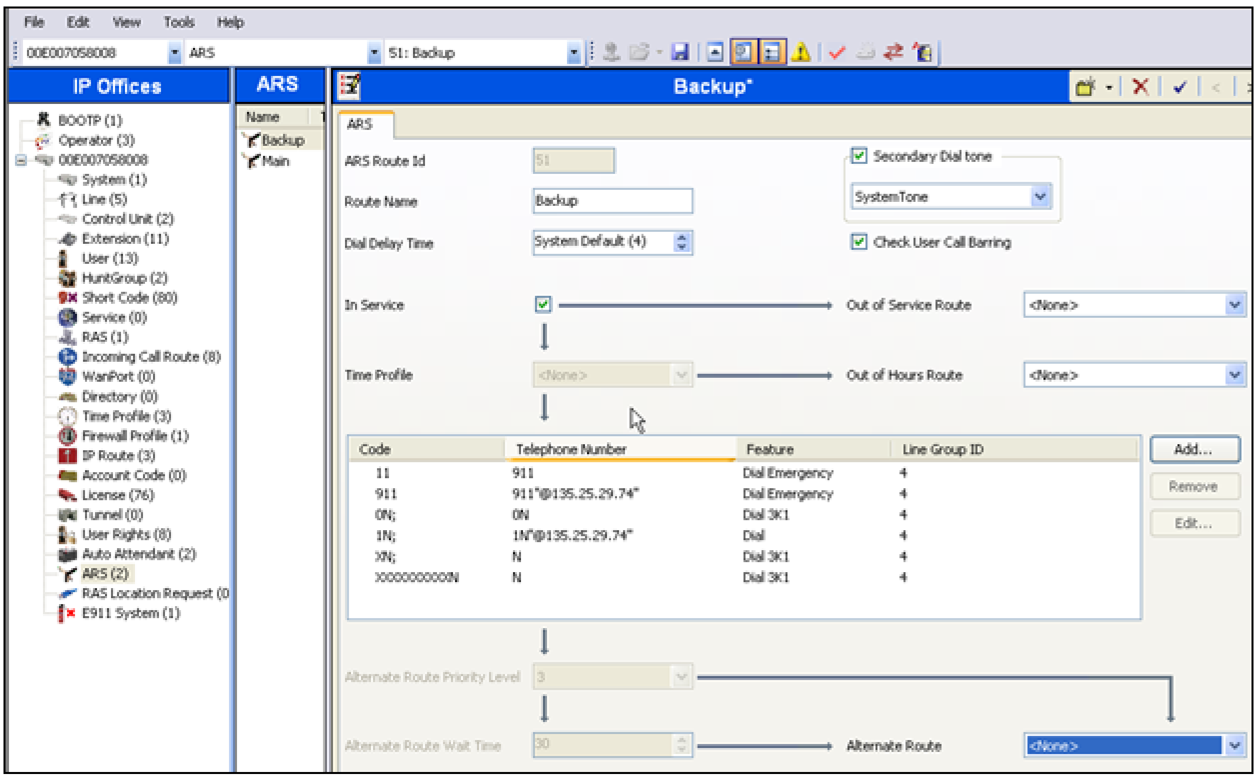


Assuming the primary route is in-service, the number passed from the Short Code used to access ARS (e.g., 8N; in **Section 5.6.2**) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed 8-1-732-555-2438, the call would be directed to Line Group 100. If Line Group 100 cannot be used, the call can automatically route to the route name configured in the **Alternate Route** parameter in the lower right of the screen (**Backup**).

Since alternate routing can be considered a privilege not available to all callers, Avaya 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.

Note – No alternate network access was available in the reference configuration. The example below is included for informational purposes.

The following screen shows an example ARS configuration for the route named **Backup** (ARS **Route ID 51**). If the user dialed 8-1-732-555-2438, and the call could not be routed via the primary route **Main** described above, the call will be delivered to this **Backup** route. Per the configuration shown below, the call would be delivered to Line Group **4** (e.g., an additional SIP Line used for backup purposes, not shown). The configuration of the **Code**, **Telephone Number**, **Feature**, and **Line Group ID** for an ARS route is similar to the configuration already shown for Short Codes in **Section 5.6**.



If a primary route experiences a network outage such that no response is received to an outbound INVITE, Avaya IP Office routes the call via the alternate route. The user receives an audible tone when the re-routing occurs and may briefly see "Waiting for Line" on the display. The redirection will occur if a call is made right after a failure of the primary route, so Avaya IP Office had not yet marked the SIP Line out of service as a result of no response to SIP OPTIONS (see the note in **Section 5.3.3**). Alternatively calls can be delivered via the alternate route when the primary route was manually marked out-of-service, or known to be out-of-service due to prior failure of SIP OPTIONS.

5.9. Auto Attendant (via Avaya IP Office Voicemail Pro)

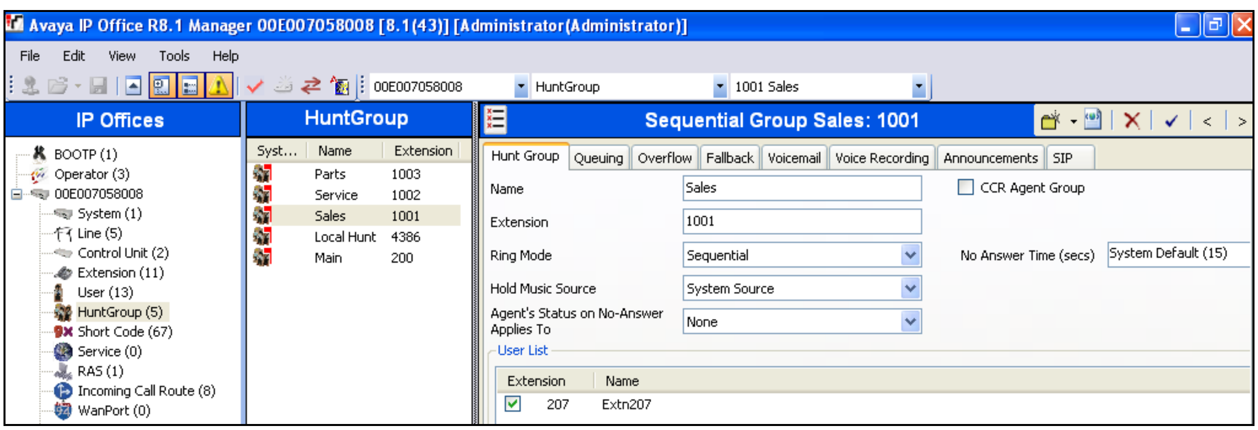
In the reference configuration, Avaya IP Office Voicemail Pro (running on a Windows 2003 server), is used for Voicemail processing. The Avaya IP Office Voicemail Pro provides for basic Auto Attendant functionality.

Note - While Avaya IP Office Voicemail Pro provisioning and programming is beyond the scope of this document, a sample Auto Attendant basic configuration is shown below.

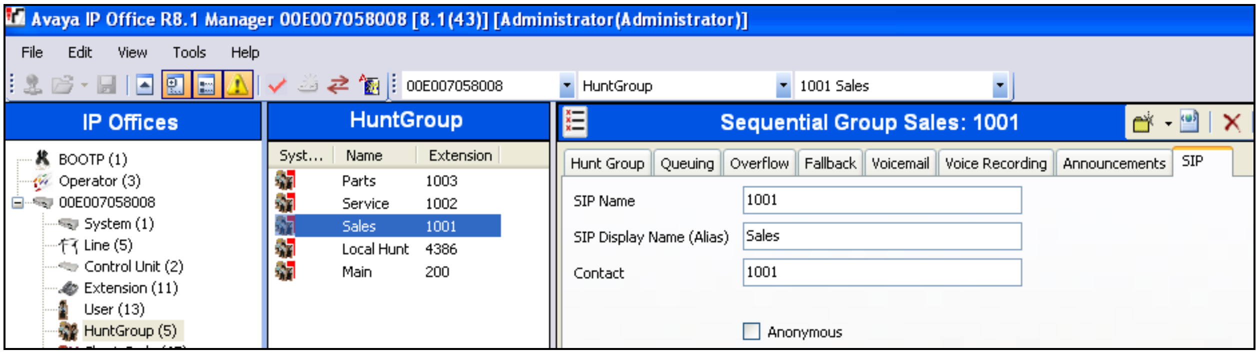
In the reference configuration an Auto Attendant was provisioned to prompt callers to select a numeric option (1, 2, or 3), that would forward the call to an associated Avaya IP Office extension (207, 500, and 600). This is accomplished via the following steps:

Step 1 - Via Avaya IP Office Manager interface, follow the steps in **Section 5.5.4**, and create hunt groups for "Sales", "Service", and "Parts".

- In the **Hunt Group** tab, enter:
    i. **Name** = **Sales**
    ii. **Extension** = **1001**
    iii. **Ring Mode** = **Sequential**
    iv. **User List** = Extension **207**
    v. Let all other values default.



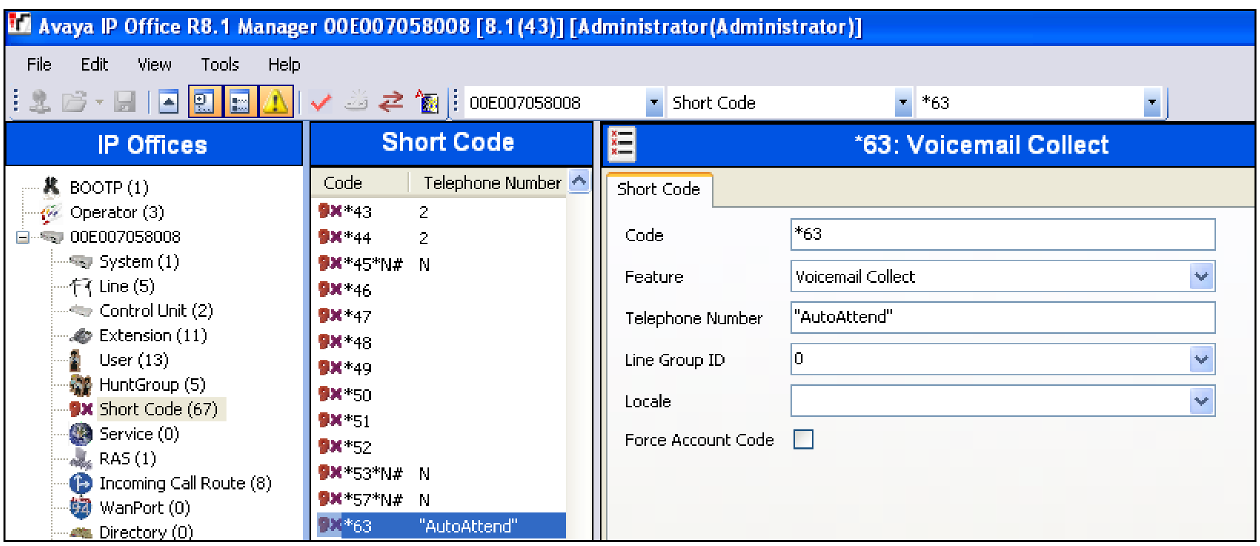
- In the SIP tab, enter:
    i. **SIP Name** = **1001**
    ii. **SIP Display Name** = **Sales**
    iii. **Contact** = **1001**



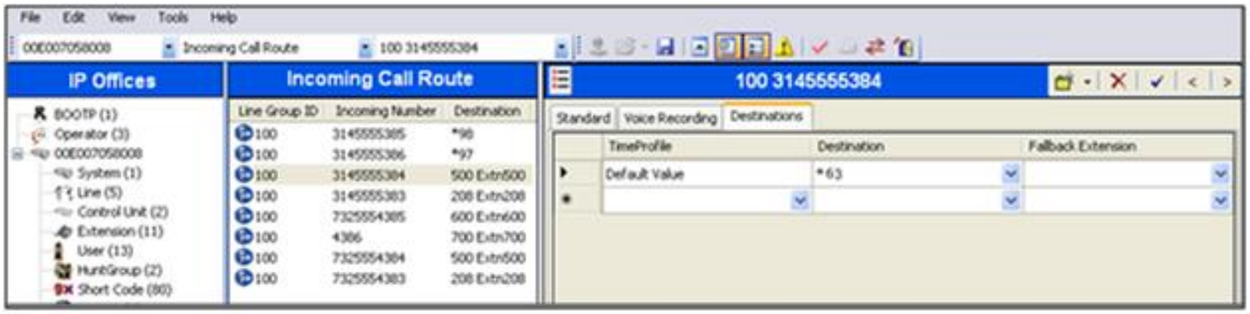
- Repeat the above steps to create the **Service** hunt group (group extension **1002** and associated with phone extension **500**) and the **Parts** hunt group (group extension **1003** and associated with phone extension **600**).

Step 2 – Following the steps shown in **Section 5.6**, create a Short Code for the Auto Attendant.

i. **Code** = Enter a Short Code value (e.g., ***63**)
ii. **Feature** = Select **Voicemail Collect** from the menu
iii. **Telephone Number** = Enter **"AutoAttend"** (the quotes must be entered, and this text value must match the name of the Voicemail Pro Module created below).
iv. **Line Group** = **0**

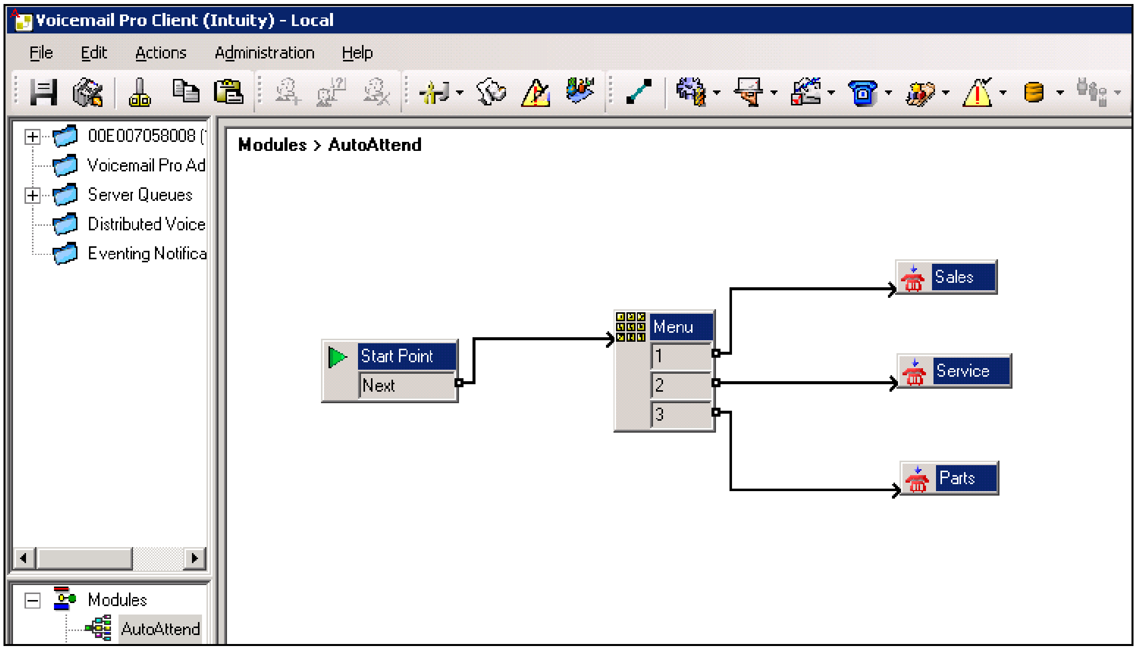


Step 3 - Following the steps shown in **Section 5.7**, create an Incoming Call Route that maps an AT&T DID to Short Code *63 (e.g., the AT&T DID 3145555304).



Step 4 - Via the Voicemail Pro GUI interface:

- Create **Start Point** (**Modules → Add**), **Menu**, and **Transfer** objects.
    - i. **Start Point** object properties (let all other fields default)
        1. **General** tab, **Token Name** = **Start Point**
    - ii. **Menu** object properties (let all other fields default)
        1. **General** tab, **Token Name** = **Menu**
        2. **Entry Prompt** tab, Create an **Entry Prompt** that will tell the caller what digits to press to reach Sales, Service, and Parts (e.g., **attendant.wav**). To modify an existing recording, double click on the .wav file and rerecord. If no .wav files exist, double click on the to open the .wav editor.
        3. **Touch Tone** tab, Select **1, 2,** and **3** as the possible entry digits.
    - iii. **Transfer** Object "**Sales**" properties
        1. **General** tab, **Token Name** = **Sales**
        2. **Specific** tab, **Destination → Mailbox → Sales**
    - iv. **Transfer** Object **"Service**" properties
        1. **General** tab, **Token Name** = **Service**
        2. **Specific** tab, **Destination → Mailbox → Service**
    - v. **Transfer** Object "**Parts**" properties
        1. **General** tab, **Token Name** = **Parts**
        2. **Specific** tab, **Destination → Mailbox → Parts**
- Select the Connector icon and:
    - i. Drag a connecting flow line from the **Start Point** box to the **Menu** box (see screen shot below).
    - ii. Drag connecting flow lines from each of the **Menu** options to their associated **Transfer** boxes (see screen shot below).



Step 5 - From the top menu select **File → Save & Make Live**, or select the  icon.

When the associated AT&T Business in a Box with IPFR-EF number is called from PSTN (e.g., 3145555304 from **Step 3**), the caller will be prompted to enter 1, 2, or 3 to access Sales, Service, or Parts. The associated Avaya IP Office extension (e.g., 207, 500, or 600) will then ring.

5.10. Privacy / Anonymous Calls

As described in **Section 5.6**, an Avaya IP Office user whose calling line identification is not typically withheld from the network, can request privacy in the reference configuration by dialing the Short Code **3N;** to access the SIP Line. The Avaya 1600-Series IP Telephones can also request privacy for a specific call, without dialing a unique Short Code, using **Features → Call Settings → Withhold Number**, on the phone itself. Specific users may be configured to always withhold calling line identification by checking the **Anonymous** field in the **SIP** tab for the user (see **Section 5.5.3**).

5.11. SIP Options Frequency

In the reference configuration, Avaya IP Office periodically checks the health of the SIP Line by sending a SIP OPTIONS message. In **Section 5.4**, the SIP Line to AT&T is shown with the **Check OOS** box checked. If there is no response, Avaya IP Office can mark the trunk out of service. Although ARS (as shown in **Section 5.8**) can include alternate routes to complete calls even if the far-end is not responding, Avaya IP Office must wait for the outbound INVITE to timeout before route advance. Once the SIP OPTIONS maintenance recognizes that the SIP Line is out-of-service, new calls will no longer be delayed before route advance. Also, once the problem with the SIP Line is resolved, the SIP OPTIONS maintenance will automatically bring the link back to the in-service state. In addition, for secure networks, the periodic sending of

OPTIONS by Avaya IP Office *may* serve to keep network Firewall “pinholes” open preventing the blockage of inbound traffic to Avaya IP Office.

In the reference configuration, Avaya IP Office sourced SIP OPTIONS every 60 seconds, (the value configured in the **Binding Refresh Time** provisioned in **Section 5.3.3**).

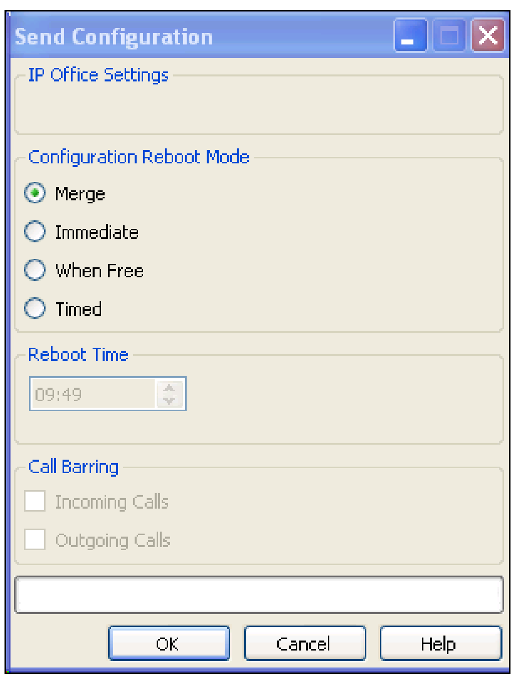
Note – In the reference configuration Avaya IP Office sent OPTIONS to the AT&T Business in a Box with IPFR-EF service Border Element and AT&T responded with *405 Method Not Allowed* (which is the expected response). That response is sufficient for Avaya IP Office to consider the connection up. The AT&T Business in a Box with IPFR-EF service Border Element does ***not*** send OPTIONS to Avaya IP Office.

5.12. Saving Configuration Changes to Avaya IP Office

The provisioning changes made in Avaya IP Office Manager must be applied to the Avaya IP Office server in order for the changes to take effect. At the top of the Avaya IP Office Manager page click **File → Save Configuration** (if that option is grayed out, no changes are pending).

A screen similar to the following will appear, with either **Merge** or **Immediate** automatically selected, based on the nature of the configuration changes. The **Merge** option will save the configuration change with no impact to the current system operation. The **Immediate** option will save the configuration and cause the Avaya IP Office server to reboot.

Click **OK** to execute the save.



The active configuration may be saved to a file at any time by selecting **File → Save Configuration As**.

6. AT&T Business in a Box with IPFR-EF Configuration

AT&T provides the Business in a Box with IPFR-EF service border element IP address, the access DID numbers, and the associated DNIS digits used in the reference configuration. In addition the AT&T EIPFR features are also assigned by AT&T. AT&T required that the Avaya IP Office LAN2 IP address be provided to the Business in a Box with IPFR-EF service, as part of the provisioning process.

7. Verification Steps

The following procedures may be used to verify the Avaya IP Office R8 with AT&T Business in a Box with IPFR-EF service configuration.

7.1. General

The following scenarios may be executed to verify Avaya IP Office functionality with the AT&T Business in a Box with IPFR-EF service:

- Incoming calls from the PSTN routed by the AT&T Business in a Box with IPFR-EF service to the Avaya IP Office location. These incoming PSTN calls arrive via the SIP Line and may be answered by Avaya SIP telephones/Softphones, H.323 telephones, Analog telephones, Analog fax machines or via Hunt Groups. Coverage to Avaya IP Office Voicemail Pro, and Voicemail Pro auto-attendant applications, may also be used. The display of caller ID on display-equipped Avaya IP Office telephones can be verified.
- Outgoing calls from the Avaya IP Office location to the PSTN routed via the SIP Line to the AT&T Business in a Box with IPFR-EF service. These outgoing PSTN calls can be originated from Avaya SIP telephones/Softphones, H.323 telephones, or Analog endpoints. The display of caller ID on display-equipped PSTN telephones may be verified.
- Inbound / Outbound fax using T38 or G.711.
- Proper disconnect when the caller abandoned a call before answer for both inbound and outbound calls, and when the Avaya IP Office party or the PSTN party terminated an active call.
- Proper busy tone heard when an Avaya IP Office user called a busy PSTN user, or a PSTN user called a busy Avaya IP Office user (i.e., if no redirection was configured for user busy conditions).
- Various outbound PSTN call types were tested including long distance, international, and toll-free.
- Requests for privacy (i.e., caller anonymity) for Avaya IP Office outbound calls to the PSTN.
- Privacy requests for inbound calls from the PSTN to Avaya IP Office users.
- SIP OPTIONS monitoring of the health of the SIP trunk. In the reference configuration Avaya IP Office sent OPTIONS to the AT&T Business in a Box with IPFR-EF service Border Element and AT&T responded with *405 Method Not Allowed* (which is the expected response). That response is sufficient for Avaya IP Office to consider the connection up. The AT&T Business in a Box with IPFR-EF service Border Element does ***not*** send OPTIONS to Avaya IP Office

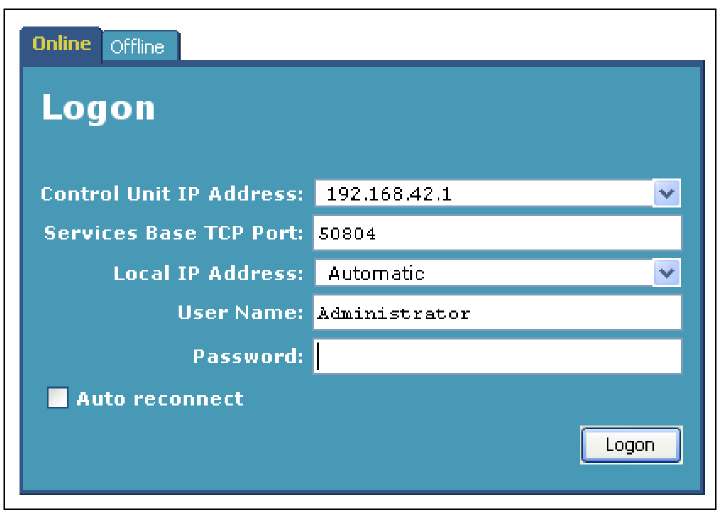
- Incoming and outgoing calls using the G.729(a) and G.711 ULAW codecs.
- DTMF transmission (RFC 2833) for successful voice mail navigation using G.729a and G.711MU for incoming and outgoing calls, including navigation of a simple auto-attendant application configured on Avaya IP Office Voicemail Pro.
    - See **Section 2.2**, **Item 1** for a DTMF issue regarding Avaya IP Office Softphone.
- Telephony features such as call waiting, hold, transfer, and conference.
- Mobile twinning to a mobile phone when the associated Avaya IP Office extension is called, as well as Mobility features such as Mobile Callback and Mobile Call Control may also be verified.
- AT&T IPFR-EF service features such as:
    - Simultaneous Ring
    - Sequential Ring
    - Call Forward – Always
    - Call Forward – Busy
    - Call Forward – Ring No Answer

7.2. Analysis/Troubleshooting

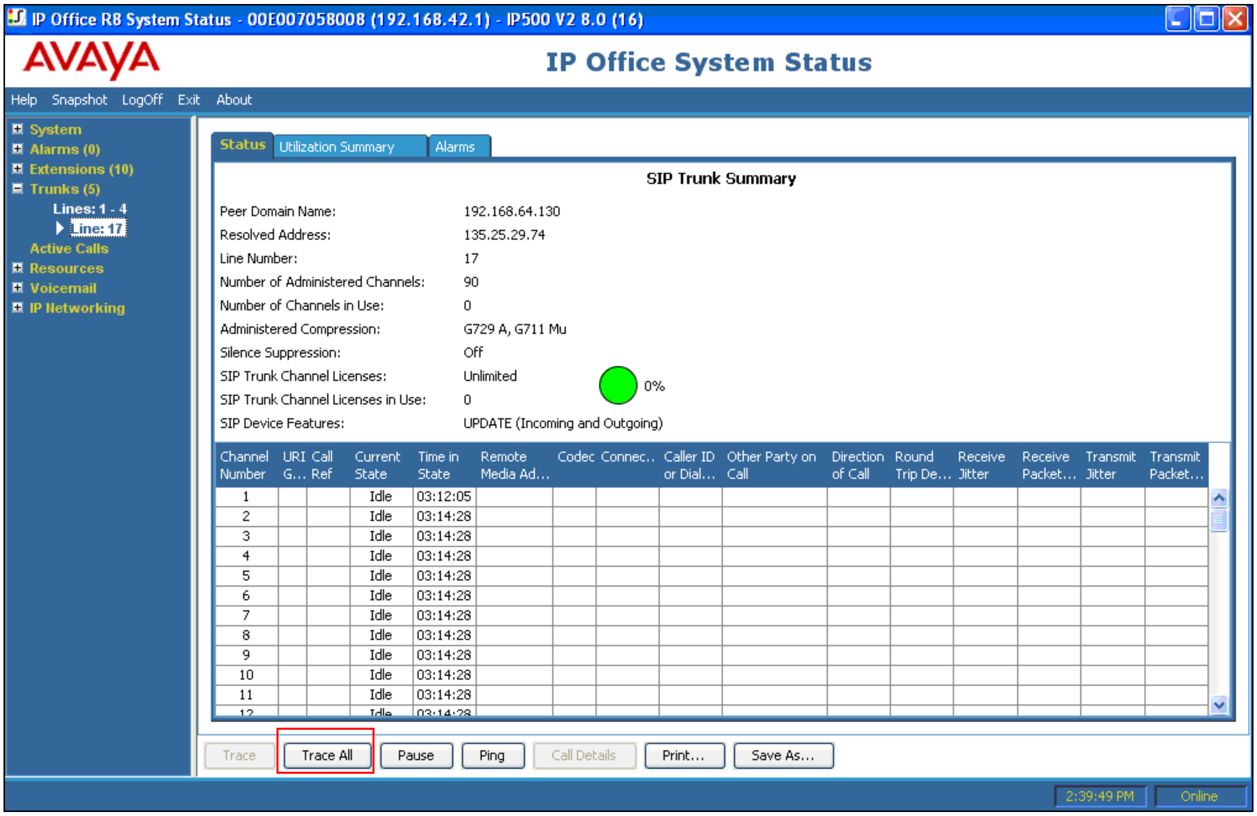
The following items may be used to analyze/troubleshoot Avaya IP Office operations.

7.2.1. System Status Application

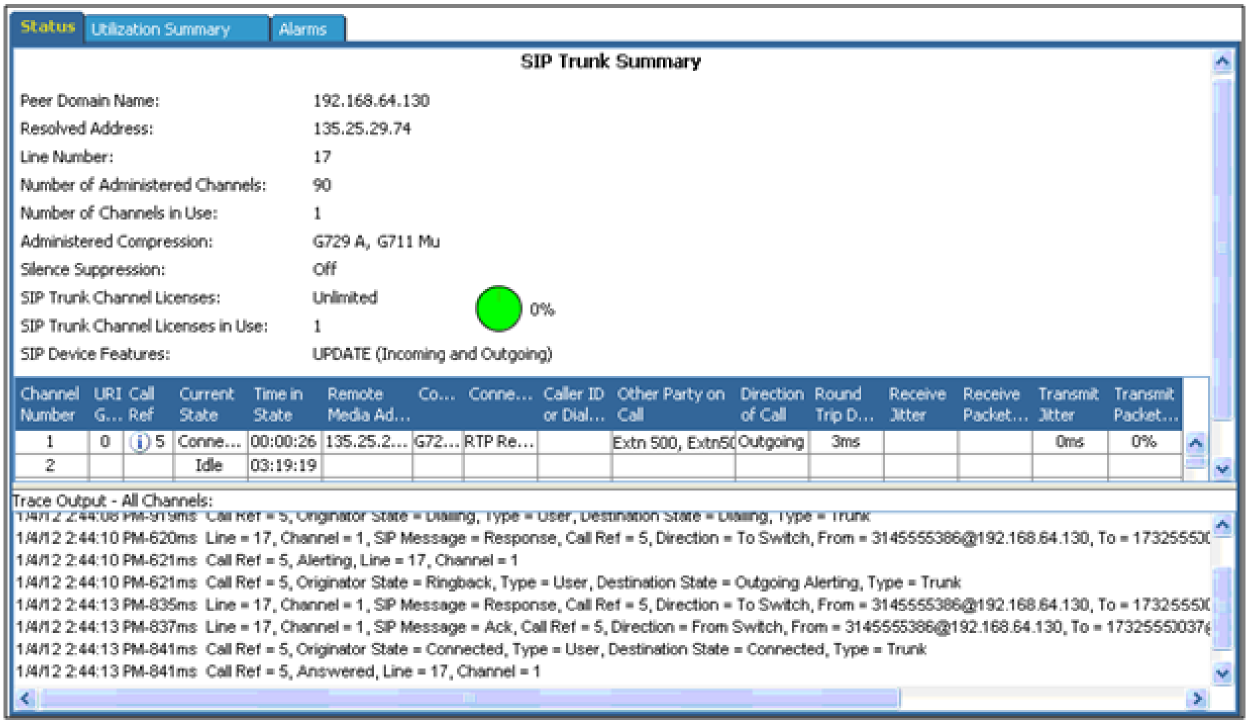
The System Status application can be used to monitor or troubleshoot Avaya IP Office. The System Status application can typically be accessed from **Start → Programs → Avaya IP Office → System Status**. The following screen shows an example **Logon** screen. Enter the Avaya IP Office IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.



After logging in, select **Trunks → Line: 17** from the left navigation menu. (SIP Line 17 is configured in **Section 5.4** to communicate with the AT&T Business in a Box with IPFR-EF service). A screen such as the one shown below is displayed. In the lower left, the **Trace All** button may be pressed to display tracing information as calls are made using this SIP Line.

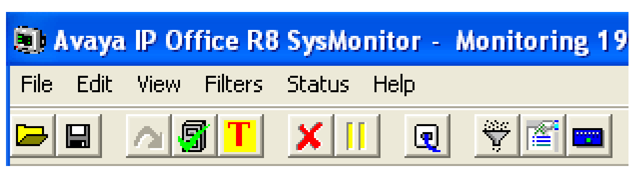


The following screen shows an example outbound call where Avaya IP Office H.323 phone Extn500 called the PSTN.



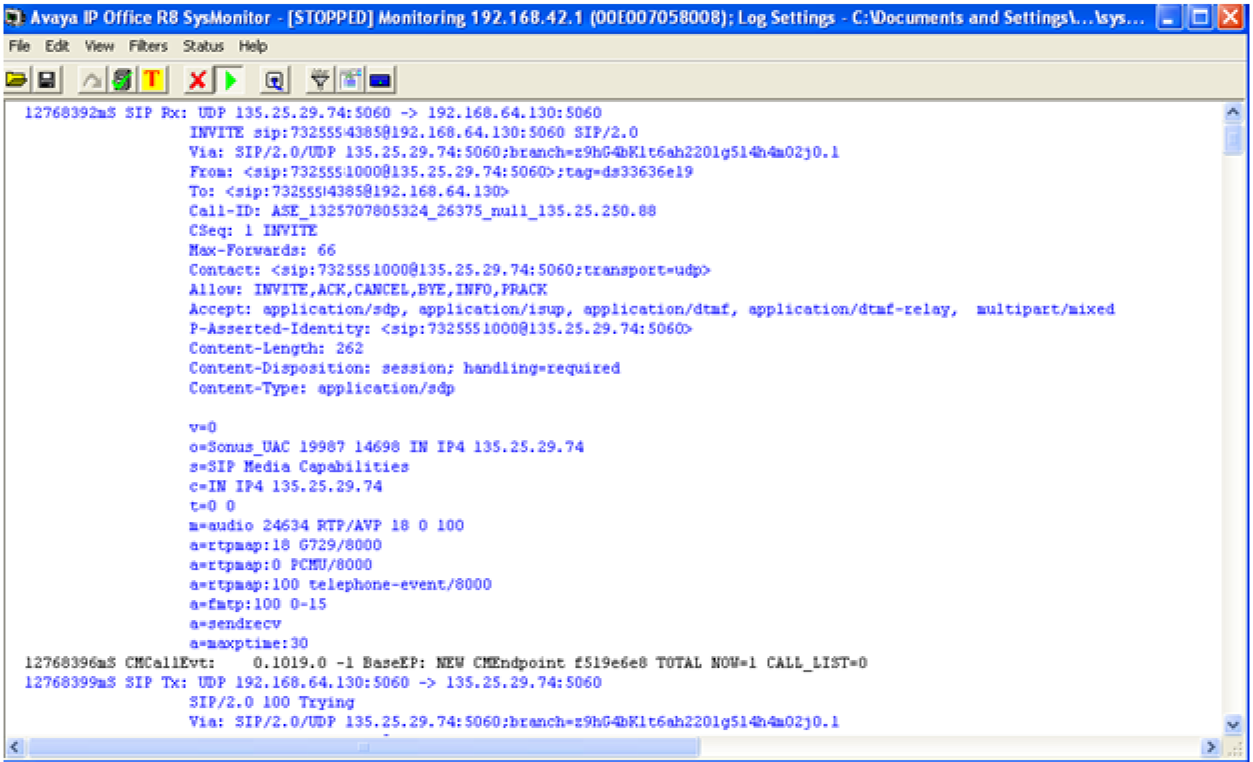
7.2.2. System Monitor Application

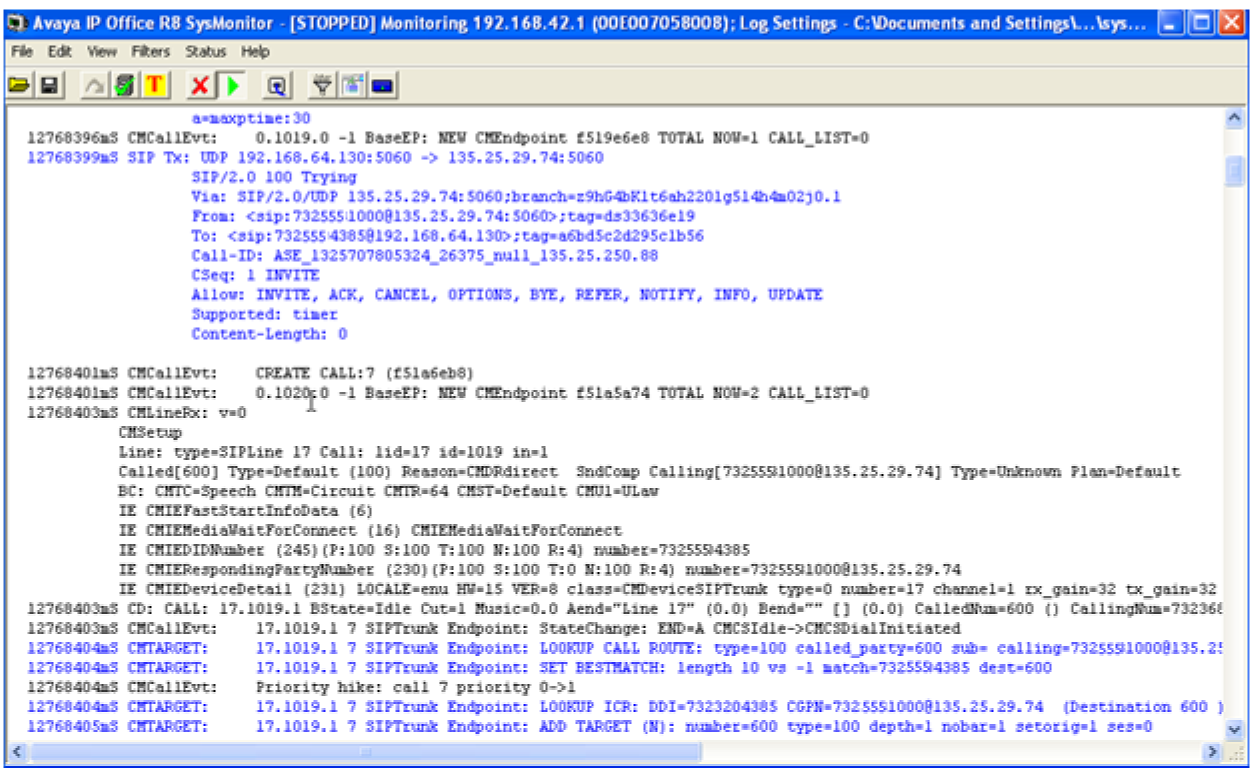
The System Monitor application can also be used to monitor or troubleshoot Avaya IP Office functionality (see reference **[3]**. The System Monitor application can typically be accessed from **Start → Programs → Avaya IP Office → Monitor**.



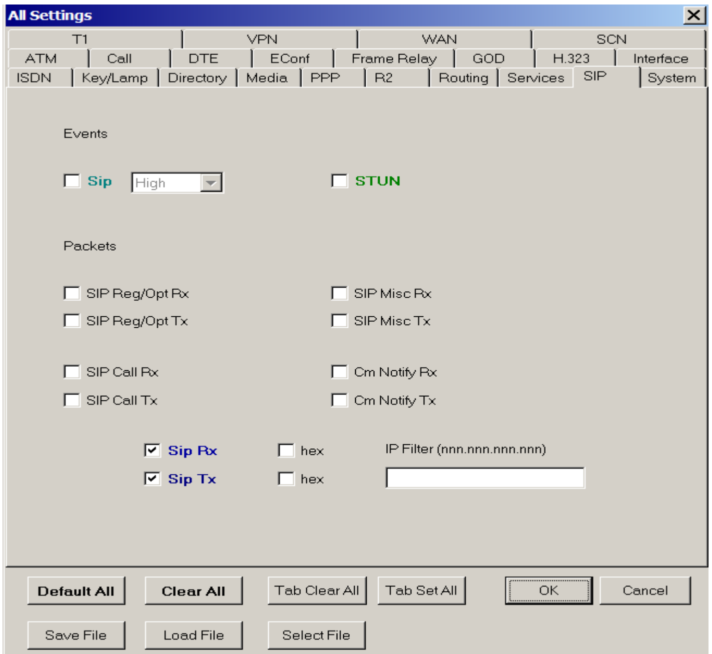
The Monitor will be active at startup. To pause the Monitor press the Pause  button. The pause button will be replaced with the Start  button. Press this button to resume the monitoring. To clear the Monitor display, press the Clear  button.

Below are samples of a monitored inbound call to Avaya IP Office SIP telephone Extn600. The Monitor will display SIP protocol (first image) as well as internal Avaya IP Office processing (second image).


Avaya IP Office R8 SysMonitor - [STOPPED] Monitoring 192.168.42.1 (00E007058008); Log Settings - C:\Documents and Settings\...\sys...
File Edit View Filters Status Help
12768392mS SIP Rx: UDP 135.25.29.74:5060 -> 192.168.64.130:5060
INVITE sip:7325554385@192.168.64.130:5060 SIP/2.0
Via: SIP/2.0/UDP 135.25.29.74:5060;branch=z9hG4bK1t6ah2201g514h4m02j0.1
From: <sip:7325551000@135.25.29.74:5060>;tag=ds33636e19
To: <sip:7325554385@192.168.64.130>
Call-ID: ASE_1325707805324_26375_null_135.25.250.88
CSeq: 1 INVITE
Max-Forwards: 66
Contact: <sip:7325551000@135.25.29.74:5060;transport=udp>
Allow: INVITE,ACK,CANCEL,BYE,INFO,PRACK
Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay, multipart/mixed
P-Asserted-Identity: <sip:7325551000@135.25.29.74:5060>
Content-Length: 262
Content-Disposition: session; handling=required
Content-Type: application/sdp
v=0
o=Sonus_UAC 19987 14698 IN IP4 135.25.29.74
s=SIP Media Capabilities
c=IN IP4 135.25.29.74
t=0 0
m=audio 24634 RTP/AVP 18 0 100
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:100 telephone-event/8000
a=fmtp:100 0-15
a=sendrecv
a=maxptime:30
12768396mS CMCallEvt: 0.1019.0 -1 BaseEP: NEW CMEndpoint f519e6e8 TOTAL NOW=1 CALL_LIST=0
12768399mS SIP Tx: UDP 192.168.64.130:5060 -> 135.25.29.74:5060
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 135.25.29.74:5060;branch=z9hG4bK1t6ah2201g514h4m02j0.1


Avaya IP Office R8 SysMonitor - [STOPPED] Monitoring 192.168.42.1 (00E007058008); Log Settings - C:\Documents and Settings\...\sys...
File Edit View Filters Status Help
a=maxptime:30
12768396mS CMCallEvt: 0.1019.0 -1 BaseEP: NEW CMEndpoint f519e6e8 TOTAL NOW=1 CALL_LIST=0
12768399mS SIP Tx: UDP 192.168.64.130:5060 -> 135.25.29.74:5060
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 135.25.29.74:5060;branch=z9hG4bK1t6ah2201g514h4m02j0.1
From: <sip:7325551000@135.25.29.74:5060>;tag=ds33636e19
To: <sip:7325554385@192.168.64.130>;tag=a6bd5c2d295c1b56
Call-ID: ASE_1325707805324_26375_null_135.25.250.88
CSeq: 1 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
Supported: timer
Content-Length: 0
12768401mS CMCallEvt: CREATE CALL:7 (f51a6eb8)
12768401mS CMCallEvt: 0.1020.0 -1 BaseEP: NEW CMEndpoint f51a5a74 TOTAL NOW=2 CALL_LIST=0
12768403mS CMLineRx: v=0
CMSetup
Line: type=SIPLine 17 Call: lid=17 id=1019 in=1
Called[600] Type=Default (100) Reason=CMDRdirect SndComp Calling[7325551000@135.25.29.74] Type=Unknown Plan=Default
BC: CMTC=Speech CMTM=Circuit CMTR=64 CMST=Default CMUl=ULaw
IE CMIEFastStartInfoData (6)
IE CMIEMediaWaitForConnect (16) CMIEMediaWaitForConnect
IE CMIEDIDNumber (245)(P:100 S:100 T:100 N:100 R:4) number=7325554385
IE CMIERespondingPartyNumber (230)(P:100 S:100 T:0 N:100 R:4) number=7325551000@135.25.29.74
IE CMIEDeviceDetail (231) LOCALE=enu HW=15 VER=8 class=CMDeviceSIPTrunk type=0 number=17 channel=1 rx_gain=32 tx_gain=32
12768403mS CD: CALL: 17.1019.1 BState=Idle Cut=1 Music=0.0 Aend="Line 17" (0.0) Bend="" [] (0.0) CalledNum=600 () CallingNum=73236
12768403mS CMCallEvt: 17.1019.1 7 SIPTrunk Endpoint: StateChange: END=A CMCSIdle->CMCSDialInitiated
12768404mS CMTARGET: 17.1019.1 7 SIPTrunk Endpoint: LOOKUP CALL ROUTE: type=100 called_party=600 sub= calling=7325551000@135.2
12768404mS CMTARGET: 17.1019.1 7 SIPTrunk Endpoint: SET BESTMATCH: length 10 vs -1 match=7325554385 dest=600
12768404mS CMCallEvt: Priority hike: call 7 priority 0->1
12768404mS CMTARGET: 17.1019.1 7 SIPTrunk Endpoint: LOOKUP ICR: DDI=7323204385 CGPN=7325551000@135.25.29.74 (Destination 600)
12768405mS CMTARGET: 17.1019.1 7 SIPTrunk Endpoint: ADD TARGET (N): number=600 type=100 depth=1 nobar=1 setorig=1 ses=0

The Monitor application allows the monitored information to be customized. To customize, select the Options button  that is third from the right in the screen above, or select **Filters → Trace Options**. The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked. All SIP messages will appear in the trace with the color blue. To customize the color, right-click on **SIP Rx** or **SIP Tx** and select the desired color.



8. Conclusion

As illustrated in these Application Notes, Avaya IP Office Release 8.1 can be configured to interoperate successfully with AT&T Business in a Box and AT&T IP Flexible Reach - Enhanced Features, within the limitations described in **Section 2.2**. This solution provides users of Avaya IP Office R8.1 the ability to support inbound and outbound calls utilizing an AT&T Business in a Box with IPFR-EF SIP trunk service connection, via AVPN or MIS/PNT transport, using the service features listed in **Section 2**. 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.

9. References

Avaya:
In general, Avaya product documentation is available at http://support.avaya.com

[1] Avaya IP Office 8.1 Installation,15-601042 Issue 26i – (23 August 2012)

[2] Avaya IP Office R8.1 Manager, 10.115-601011 Issue 29o – (03 August 2012)

[3] Avaya IP Office System Monitor, Document Number 15-601019

[4] Avaya IP Office Voicemail Pro15-601063 Issue 20b - (11 July 2008)

[5] Avaya IP Office Voicemail Pro Example Exercises, Issue 4c (5th May 2004)

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

AT&T Business in a Box with IPFR-EF Service:

[7] http://www.business.att.com/enterprise/Service/voice-services/voip/business-in-a-box/

10. Appendix: SIP Line Template

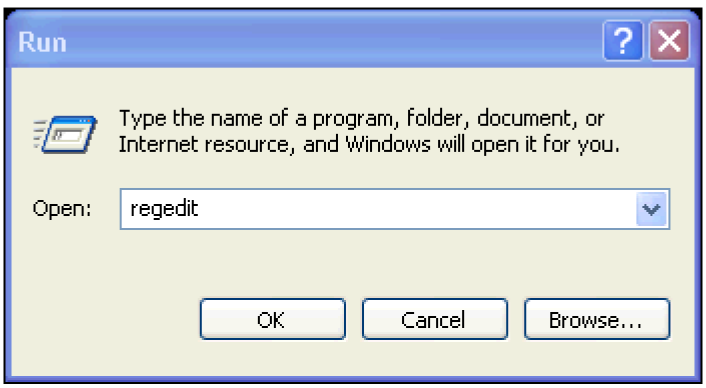
Avaya IP Office Release 8.1 supports a SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

Not all of the configuration information is included in the SIP Line Template, therefore, it is critical that the SIP Line configuration be verified/updated after a template has been imported, and additional configuration be supplemented using the settings provided in this Application Notes.

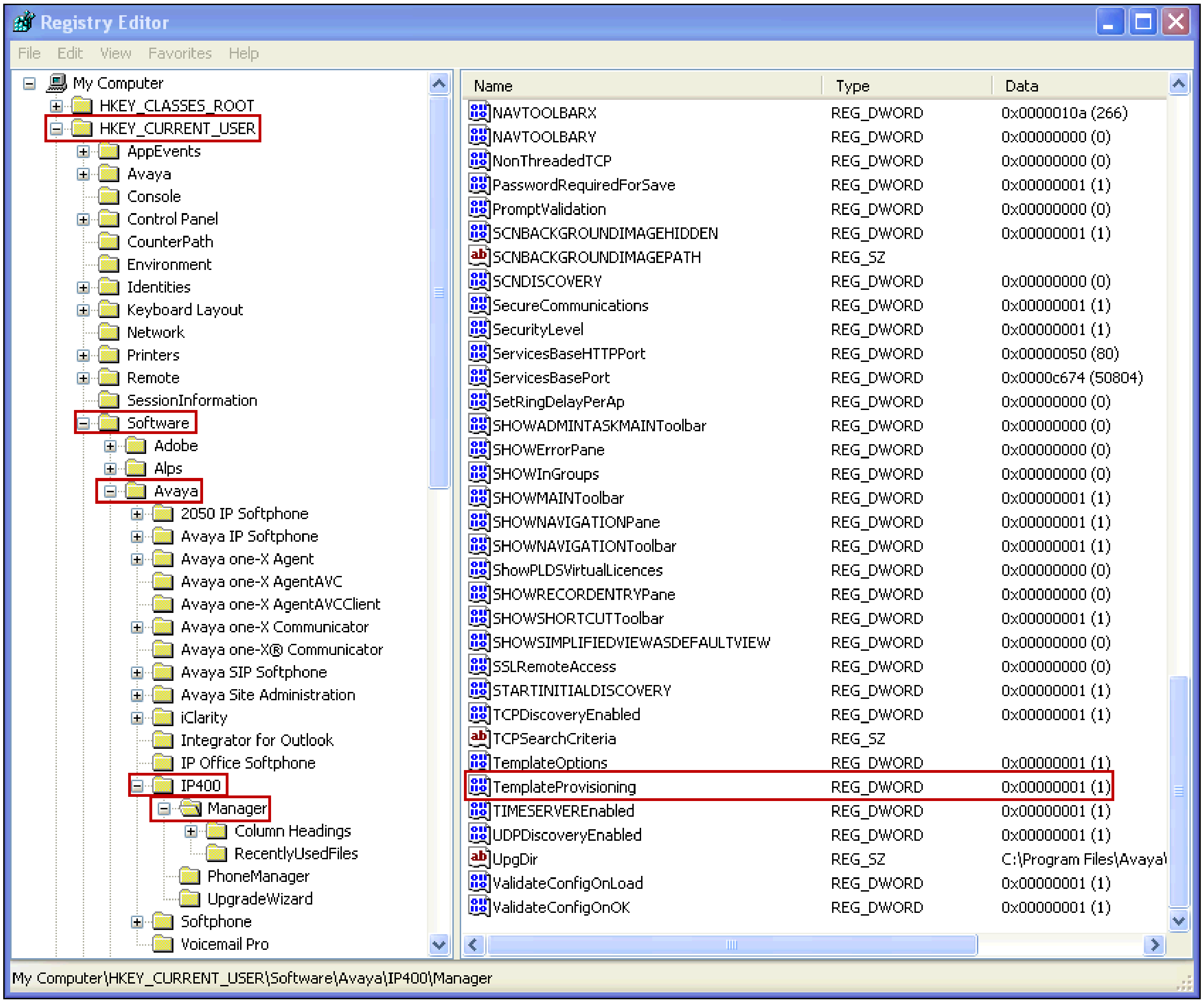
To create a SIP Line Template from the configuration described in these Application Notes, configure the parameters as described below.

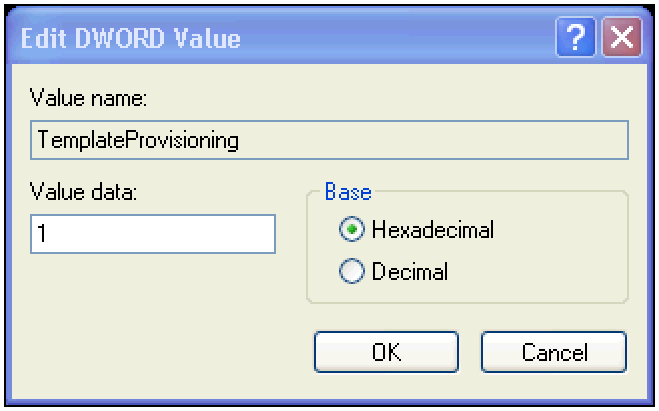
Create a new registry entry called **TemplateProvisioning** and set the **Value data** to **1**, as follows:

Select **Start**, and then **Run**. Type **regedit** as shown below



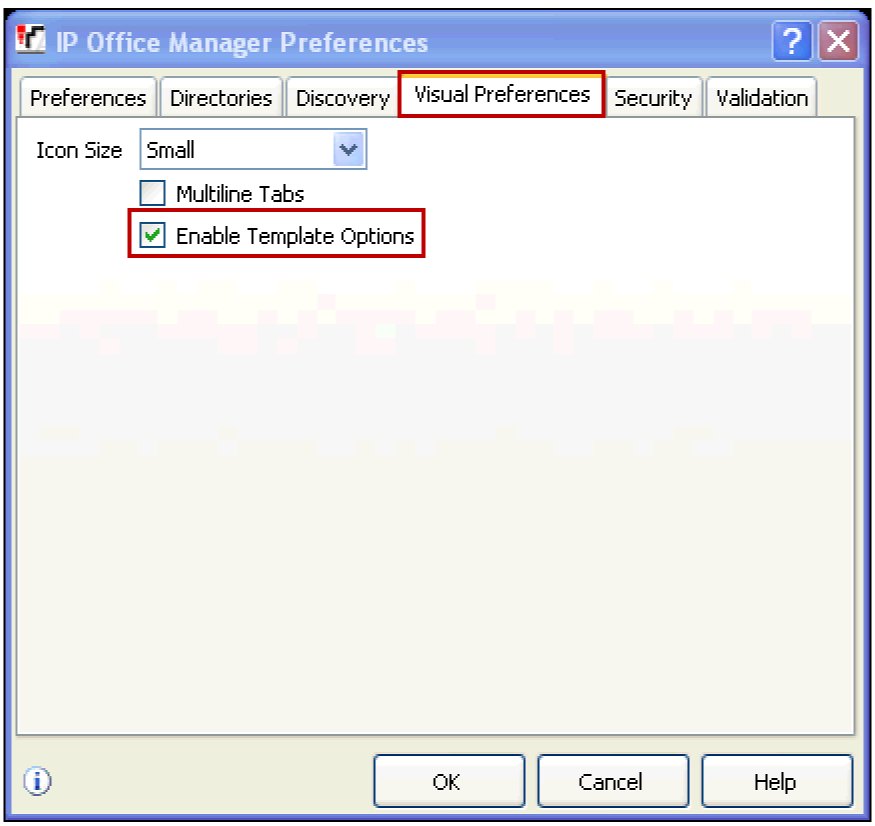
Under **HKEY_CURRENT_USER, Software, Avaya, IP400**, right click on **Manager**, then select **New, DWORD value,** then rename the newly created entry to: **TemplateProvisioning.** Right click on the newly created entry and select **Modify,** change the value under **Value Data** from "**0**" to "**1**".





Reboot the computer.

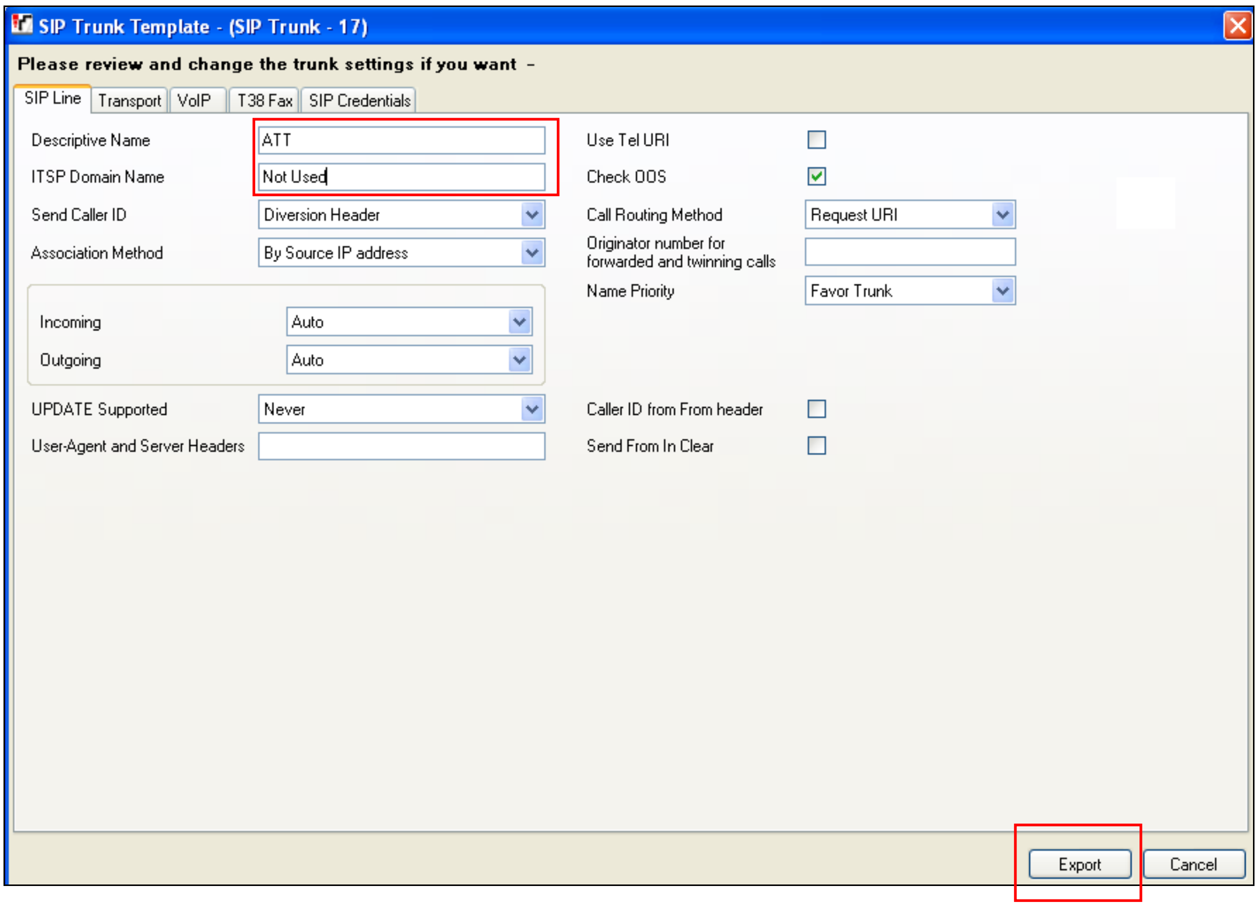
When the computer comes back up, enable the template by opening **IP Office Manager**, select **File**, and then **Preferences**. On the **Visual Preferences** tab, check the **Enable Template Options** box, and click **OK**.



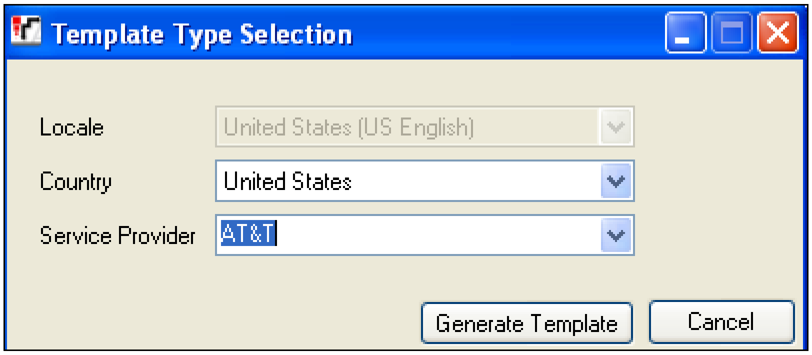
To create a SIP Line Template from the configuration, on the left Navigation Pane, right click on the Sip Line (**17**), and select **Generate SIP Trunk Template** (not shown)

Enter a descriptive name; **ATT** was used in the sample template. Note that for ITSP Domain Name **Not Used** was used (AT&T uses IP addresses instead of Domain names), an entry is required here or the template will not run. This entry (**Not Used**) should be removed after importing the configuration into a new Avaya IP Office installation.

To generate the template click on **Export**.



On the next screen, **Template Type Selection**, select the **Country,** enter the name for the **Service Provider**, and click **Generate Template**.



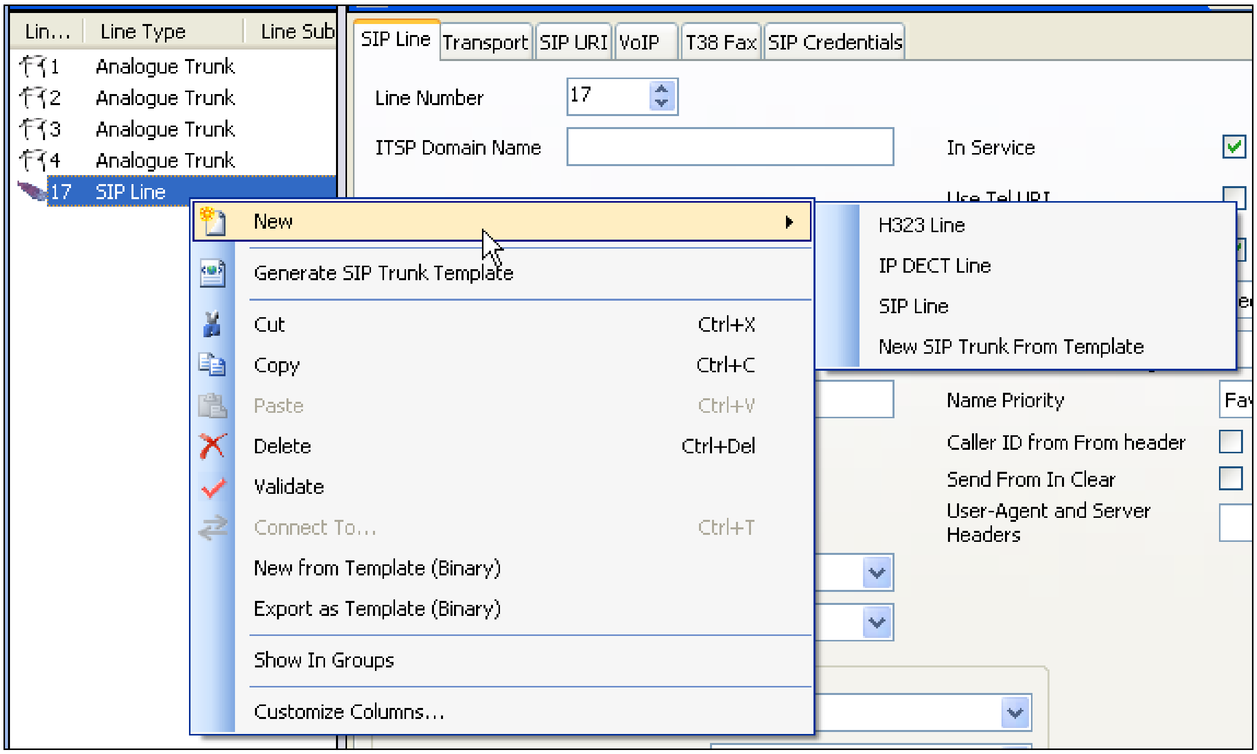
By default the template file is generated to the path **\\Program Files\Avaya\IP Office\Manager\Templates**.

The following is an example of the exported SIP Line Template file, **US_AT&T_SIPTrunk.xml**:

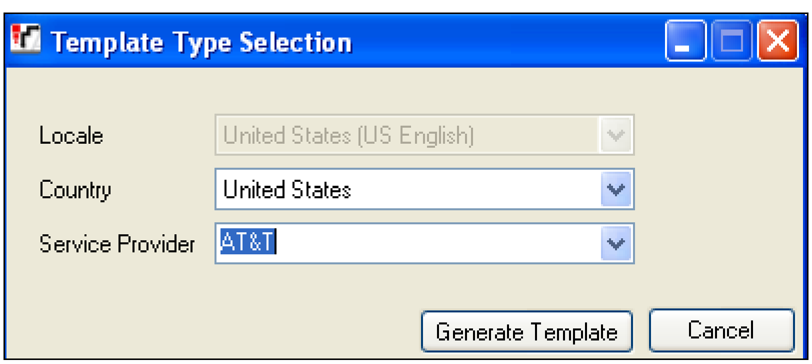

```
<?xml version="1.0" encoding="utf-8" ?>
- <Template xmlns="urn:SIPTrunk-schema">
<TemplateType>SIPTrunk</TemplateType>
<Version>20121130</Version>
<SystemLocale>enu</SystemLocale>
<DescriptiveName>ATT</DescriptiveName>
<ITSPDomainName>Not Used</ITSPDomainName>
<SendCallerID>CallerIDDIV</SendCallerID>
<ReferSupport>true</ReferSupport>
<ReferSupportIncoming>2</ReferSupportIncoming>
<ReferSupportOutgoing>2</ReferSupportOutgoing>
<RegistrationRequired>false</RegistrationRequired>
<UseTelURI>false</UseTelURI>
<CheckOOS>true</CheckOOS>
<CallRoutingMethod>1</CallRoutingMethod>
<OriginatorNumber />
<AssociationMethod>SourceIP</AssociationMethod>
<LineNamePriority>FavourTrunk</LineNamePriority>
<UpdateSupport>UpdateNever</UpdateSupport>
<UserAgentServerHeader />
<CallerIDfromFromheader>false</CallerIDfromFromheader>
<PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
<ITSPProxy>12.40.234.99</ITSPProxy>
<LayerFourProtocol>SipUDP</LayerFourProtocol>
<SendPort>5060</SendPort>
<ListenPort>5060</ListenPort>
<DNSServerOne>0.0.0.0</DNSServerOne>
```

<DNSServerTwo>**0.0.0.0**</DNSServerTwo>
<CallsRouteViaRegistrar>**true**</CallsRouteViaRegistrar>
<SeparateRegistrar />
<CompressionMode>**AUTOSELECT**</CompressionMode>
<UseAdvVoiceCodecPrefs>**true**</UseAdvVoiceCodecPrefs>
<AdvCodecPref>**G.711 ULAW 64K,G.729(a) 8K CS-ACELP**</AdvCodecPref>
<CallInitiationTimeout>**4**</CallInitiationTimeout>
<DTMFSupport>**DTMF_SUPPORT_RFC2833**</DTMFSupport>
<VoipSilenceSupression>**false**</VoipSilenceSupression>
<ReinviteSupported>**true**</ReinviteSupported>
<FaxTransportSupport>**FOIP_T38**</FaxTransportSupport>
<UseOffererPrefferedCodec>**false**</UseOffererPrefferedCodec>
<CodecLockdown>**false**</CodecLockdown>
<Rel100Supported>**true**</Rel100Supported>
<T38FaxVersion>**3**</T38FaxVersion>
<Transport>**UDPTL**</Transport>
<LowSpeed>**0**</LowSpeed>
<HighSpeed>**0**</HighSpeed>
<TCFMethod>**Trans_TCF**</TCFMethod>
<MaxBitRate>**FaxRate_14400**</MaxBitRate>
<EflagStartTimer>**2600**</EflagStartTimer>
<EflagStopTimer>**2300**</EflagStopTimer>
<UseDefaultValues>**false**</UseDefaultValues>
<ScanLineFixup>**true**</ScanLineFixup>
<TFOPEnhancement>**true**</TFOPEnhancement>
<DisableT30ECM>**false**</DisableT30ECM>
<DisableEflagsForFirstDIS>**false**</DisableEflagsForFirstDIS>
<DisableT30MRCompression>**false**</DisableT30MRCompression>
<NSFOverride>**false**</NSFOverride>
</Template>

Next, import the template into the new Avaya IP Office system by creating a new SIP Line as shown in the screenshot below. In the Navigation Pane on the left, right-click on **Line** then navigate to **New, New SIP Trunk from Template**:



On the next screen, **Template Type Selection**, verify that the information in the **Country** and **Service Provider** fields is correct. If more than one template is present, use the drop-down menus to select the required template. Click **Create new SIP Trunk** to finish the process.



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 Developer*Connection* Program at devconnect@avaya.com.