Avaya Solution & Interoperability Test Lab

Application Notes for Avaya IP Office Release 9.1, Avaya Session Border Controller for Enterprise 6.3, with AT&T IP Toll Free Service– Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office R9.1 and the Avaya Session Border Controller for Enterprise 6.3, with the AT&T IP Toll Free service using AVPN or MIS/PNT transport connections.

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

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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

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1. Introduction

These Application Notes describe the steps for configuring Avaya IP Office R9.1 and the Avaya Session Border Controller for Enterprise 6.3, with the AT&T IP Toll Free service using **AVPN** or **MIS/PNT** transport connections.

Avaya IP Office is a versatile communications solution that combines the reliability and ease of a traditional telephony system with the applications and advantages of an IP telephony solution. This converged communications solution can help businesses reduce costs, increase productivity, and improve customer service.

The Avaya Session Border Controller for Enterprise is the point of connection between Avaya IP Office and the AT&T IP Toll Free service, and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability.

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution providing toll-free services over SIP trunks for business customers. The AT&T Toll Free service utilizes AVPN[1] or MIS/PNT[2] transport services.

Note – The Avaya Session Border Controller for Enterprise will be referred to as the *Avaya SBCE* in the remainder of this document. The AT&T IP Toll Free service will be referred to as *IPTF* in the remainder of this document.

Note – The solution described in these application notes also applies to the AT&T Business in a Box service.

2. General Test Approach and Test Results

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.

The interoperability compliance testing focused on verifying inbound and outbound call flows between IPTF and the Customer Premises Equipment (CPE) containing the Avaya IP Office 9.1 and the Avaya SBCE 6.3 (see **Section 3.2** for call flow examples).

The test environment described in these Application Notes consisted of:

[1] AVPN uses compressed RTP (cRTP).
[2] MIS/PNT does not support cRTP.

- A simulated enterprise with Avaya IP Office 9.1, Avaya SBCE 6.3, Avaya SIP (1120E), H.323 (1608), and Analog telephones, as well as a fax machine emulator (Ventafax).
- Laboratory versions of the IPTF service, to which the simulated enterprise was connected via AVPN/MIS transport.

2.1. Interoperability Compliance Testing

The compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the IPTF network. Calls were made from the PSTN across the IPTF test network, to the CPE.

The interoperability compliance testing focused on verifying inbound call flows (see **Section 3.2**) between Avaya IP Office, Avaya SBCE, and the IPTF service.

The compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the AT&T network.

The following SIP trunking VoIP features were tested with the IPTF service:

- Incoming calls from PSTN, routed by the IPTF service, to the Avaya SBCE 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, were also used.
- Inbound fax using T38 or G.711, and G3 or SG3 endpoints.
- Proper disconnect when the caller abandoned a call before answer, 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).
- SIP OPTIONS monitoring of the health of the SIP trunk. In the reference configuration Avaya IP Office sent OPTIONS to the IPTF 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.
- Incoming calls using the G.729A and G.711 ULAW codecs.
- 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, as well as IPTF DTMF generated features.
- Telephony features such as call waiting, hold, transfer, and conference.
- AT&T IP Toll Free features such as Legacy Transfer Connect and Alternate Destination Routing were also tested.

2.2. Test Results

The test objectives stated in **Section 2.1**, with limitations as noted below, were verified.

1. **Avaya IP Office only supports a packet size (ptime) of 20 msecs, and therefore does not specify a ptime value in the SIP SDP (in either requests or responses) –**.
    - Although no issues were found during testing, AT&T recommends that for maximum customer bandwidth utilization, a ptime value of 30 should be specified.

2. **Avaya IP Office does not support T.38 fax and the Direct Media feature simultaneously** – Avaya IP Office supports a Direct Media feature whereby Avaya IP Office IP endpoints can send/receive media directly with the Avaya SBCE, rather than having the media routed via the Avaya IP Office V2 500 platform.
    - As T.38 fax is the preferred fax transport method, and all media must pass through the Avaya SBCE going to/from AT&T, Direct Media was disabled in the reference configuration (see **Sections 5.4.6** and **5.5.2**, **step 2**).
    - Alternatively, Direct Media can be enabled if G.711 fax is used. However, as mentioned above, any resulting Direct Media will only take place within the CPE (phone to phone, or phone to Avaya SBCE).

3. **Inbound T.38 or G.711 fax calls fail when the sender and receiver are both Super G3 (SG3) fax devices** – During testing it was found that when the sender and receiver both used SG3 fax devices, and an inbound fax call was placed to Avaya IP Office using either T.38 or G.711, approximately 80% of the fax calls failed to connect.
    - It was found that during SG3/SG3 inbound fax calls, Avaya IP Office took between 15 and 20 seconds to establish the fax connection (non SG3/SG3 calls took half this time).
    - An MR was opened with Avaya IP Office support.
    - **UPDATE** – This issue was resolved in Avaya IP Office 9.1, Service Pack 1.

4. **Avaya IP Office issues SIP Invites with incorrect Host field contents in the From Header** – If the SIP Line *ITSP Domain Name* field (see **Section 5.4.3**), is populated per the system Help file (e.g., domain or IP address of the Service Provider), then Avaya IP Office will populate outbound Invite From headers with this value. Instead, the From header should be populated with the IP address of the Avaya IP Office SIP Trunk interface (LAN 2 in the reference configuration).
    - An MR was opened with Avaya IP Office support.
    - A workaround is to populate the *ITSP Domain Name* field with the LAN 2 IP address.
        - Note that this workaround will cause Avaya IP Office to send OPTIONS messages to AT&T with the LAN 2 IP address in the R-URI and To headers. However as the OPTIONS messages are used for trunk "keep-alive" purposes only, this is not an issue.

5. **The Avaya SBCE may issue a Remote-Address header even though the option to do so is disabled** - During testing it was found that the Avaya SBCE could include a Remote-Address header to Invites or 200OKs, even though the option to do so is disabled by default.
    - No issues were caused by the inclusion of this header, however the Avaya SBCE was provisioned to remove this header to reduce overall packet size (see **Section 6.5.3**).

6. **Use of Network Address Translation (NAT) on the customer interface of the AT&T CE router when SIP Multipart headers are used.** The IPTF service may send Multipart SIP headers. Previously, use of NAT on the AT&T CE (Cisco) router would corrupt the Multipart header contents, including the SDP. The AT&T solution was not to use NAT on the CE router.
    - Recent testing with Cisco IOS *c2900-universalk9-mz.SSA-**eng-sp-153-3.M1**.bin,* **in addition to specifying the router command *ip nat service allow-multipart***, showed that NAT can be used on the CE router without corrupting the Multipart headers (Mobility and NSS). Note that Cisco IOS *c2900-universalk9-mz.SPA.**154-3.M1**.bin,* **in addition to specifying the router command *ip nat service allow-multipart***, was tested successfully as well.

2.3. Support

AT&T customers may obtain support for the AT&T IP Toll Free service by calling (800) 325-5555.

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

3. Reference Configuration

Note – Documents used to provision the test environment are listed in **Section 9**. References to these documents are indicated by the notation **[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 platform.
- The Avaya SBCE provides SIP Session Border Controller (SBC) functionality, including address translation and SIP header manipulation between the IPTF service and the CPE. In the reference configuration, the Avaya SBCE runs on a Portwell CAD-0208 platform.

- Avaya "desk" telephones are represented with an Avaya 1608 H.323 set, an Avaya 6211 Analog set, an Avaya 1120E SIP set, as well as Avaya Communicator 2.0 (SIP). Fax endpoints are represented by PCs running Ventafax emulation 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. This solution is extensible to the Avaya IP Office embedded voice mail as well.
- In the reference configuration, both the Avaya IP Office (interface "LAN 1"), and the Avaya SBCE (interface "A1") are connected to the private CPE network. The Avaya SBCE interface "B1" is connected to the AT&T network.
- UDP transport via port 5060, was used between the Avaya IP Office and the Avaya SBCE, as well as between the Avaya SBCE and AT&T.
- The AT&T IPTF service requires RTP port ranges 16384-32767.
- AT&T provided the inbound and outbound access numbers (DID and DNIS) used in the reference configuration. Note that the IPTF service may deliver various DNIS digit lengths in the SIP Invite R-URI depending on the circuit order provisioning. In the reference configuration, the IPTF service delivered 21 or 10 DNIS digits.

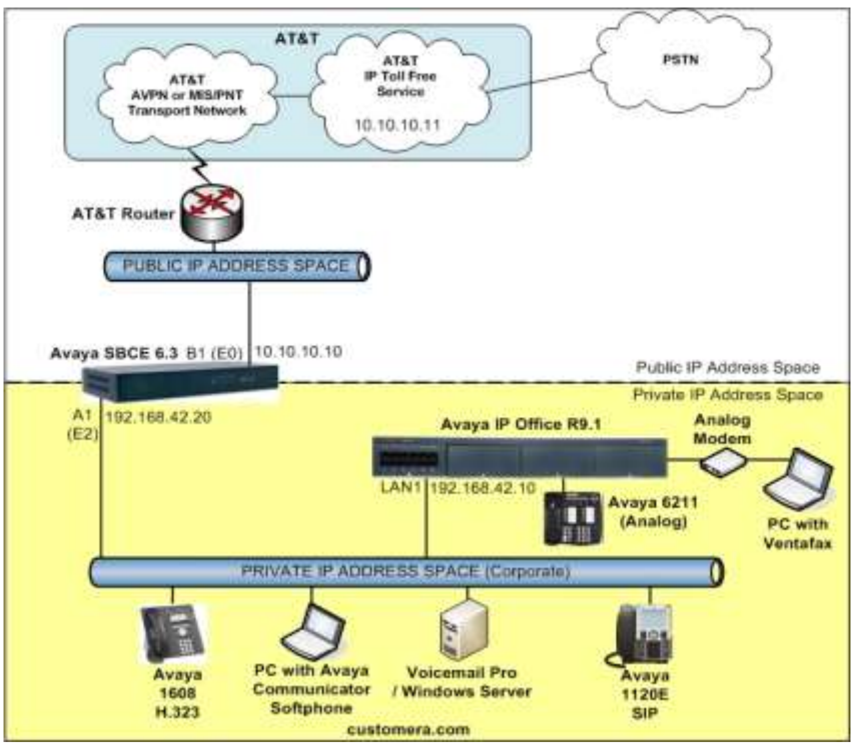


Figure 1: Reference Configuration

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 values based on their own specific configurations.

Note – The Avaya SBCE “B1” interface communicates with AT&T Border Elements (BEs) located in the AT&T IPTF network. For security reasons, the IP addresses of the AT&T BEs are not included in this document. However as placeholders in the following configuration sections, the IP addresses **10.10.10.10** (Avaya SBCE “B1”), and **10.10.10.11/10.10.10.12** (AT&T BE IP addresses), are specified. In addition, AT&T DID/DNIS numbers shown in this document are examples as well. AT&T Customer Care will provide the actual Border Element IP addresses and DID/DNIS numbers as part of the IPTF provisioning process.

Component	Illustrative Value in these Application Notes
Avaya IP Office	
Private network “LAN 1” interface.	192.168.42.10
Avaya SBCE	
Private network “A1” interface.	192.168.42.20
Public network “B1” interface.	10.10.10.10
AT&T IPTF Service	
Border Element IP Address	10.10.10.11 & 10.10.10.12

Table 1: Illustrative Values Used in these Application Notes

3.2. Call Flows

To understand inbound AT&T IPTF service calls, two basic call flows are described in this section.

3.2.1. Basic Inbound Call

The first call scenario illustrated in the figure below is an inbound AT&T IPTF 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 IPTF service number.
2. The PSTN routes the call to the AT&T IPTF service network.
3. The AT&T IPTF service routes the call to the Avaya SBCE.
4. The Avaya SBCE performs SIP Network Address Translation (NAT) and any specified SIP header modifications, and routes the call to Avaya IP Office.
5. 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.

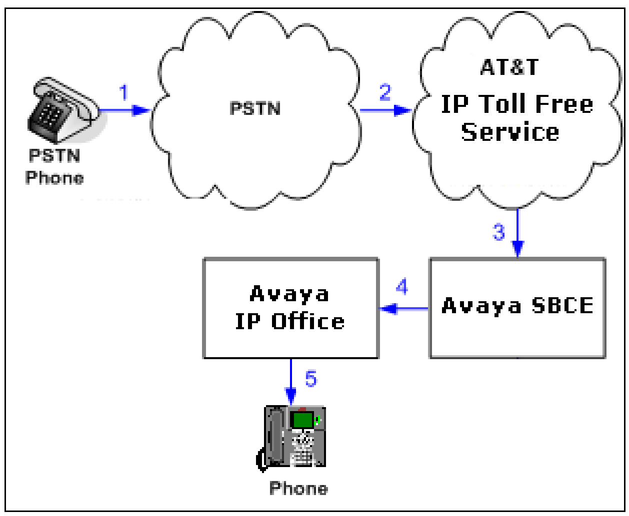


Figure 2: Inbound AT&T IPTF Call

3.2.2. 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.

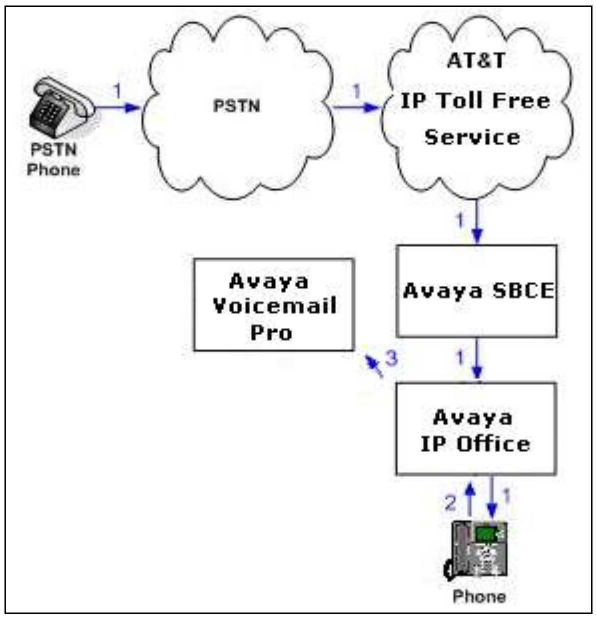


Figure 3: Coverage to Voicemail (Voicemail Pro)

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	R9.1 (437) and Service Pack 1 (9.1.1.0 build 10) (see Section 2.2, Item 3)
Avaya 1608 (H.323) Telephone	Ha1608ua1_350B.bin
Avaya 1120E (SIP) Telephone	04.04.10.00
Avaya Communicator for Windows	2.0.3.30
Avaya 6211 Analog Telephone	-
Avaya SBCE	6.3.000-19-4338
Fax device	Ventafax 6.3

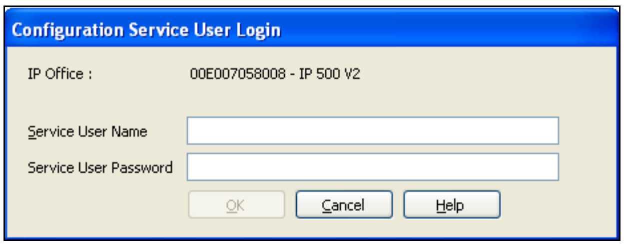
Table 2: Equipment and Software Versions

Note - Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with all configurations of IP Office Server Edition without T.38 Fax Service.

5. Avaya IP Office Configuration

Note - This section describes attributes of the reference configuration, but is not meant to be prescriptive. In the following sections, only the parameters that are highlighted in **bold** text are applicable to the reference configuration. Other parameter values may or may not match based on local configurations. Many forms contain multiple tabs. Only those tabs with provisioning related to the reference configuration are discussed. Any other tab/form should be considered default values. Additionally, the screen shots referenced in these sections may not be the complete form.

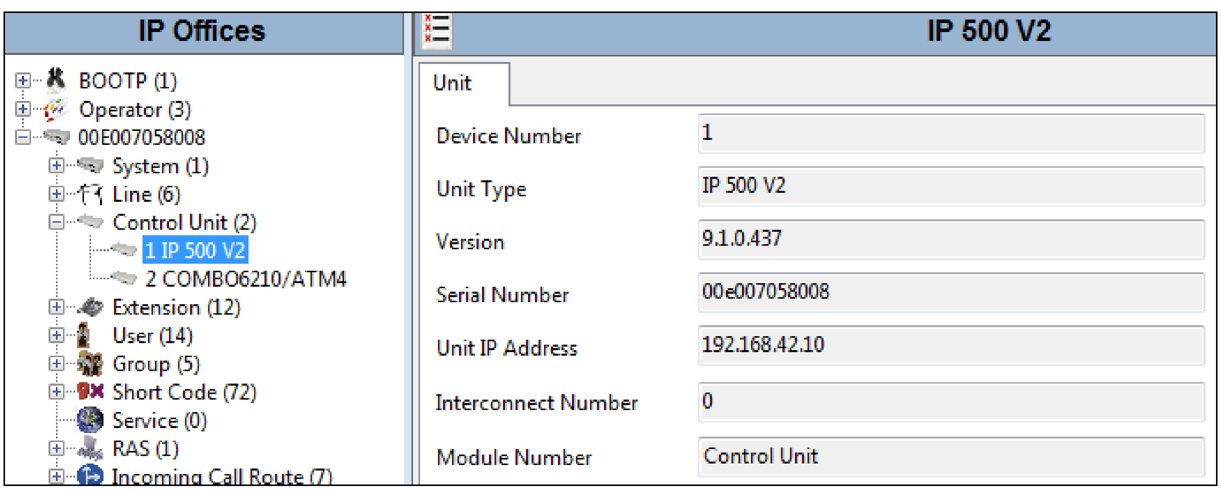
Avaya IP Office is configured via the Avaya IP Office Manager program. For more information on provisioning Avaya IP Office Manager, consult reference **[1]**. From the Avaya IP Office Manager PC, select **Start →Programs →Avaya IP Office →Manager** to launch the Manager application. Enter the appropriate credentials.



5.1. Platform Information

Note - In the following sections, the left hand Navigation pane will be used to select Avaya IP Office provisioning options.

This section describes attributes of the reference configuration. The following screen shows the Avaya IP Office module configuration used in the reference configuration. In the screen below, the **IP 500 V2** platform is displayed along with the COMBO6210/ATM4 modules.



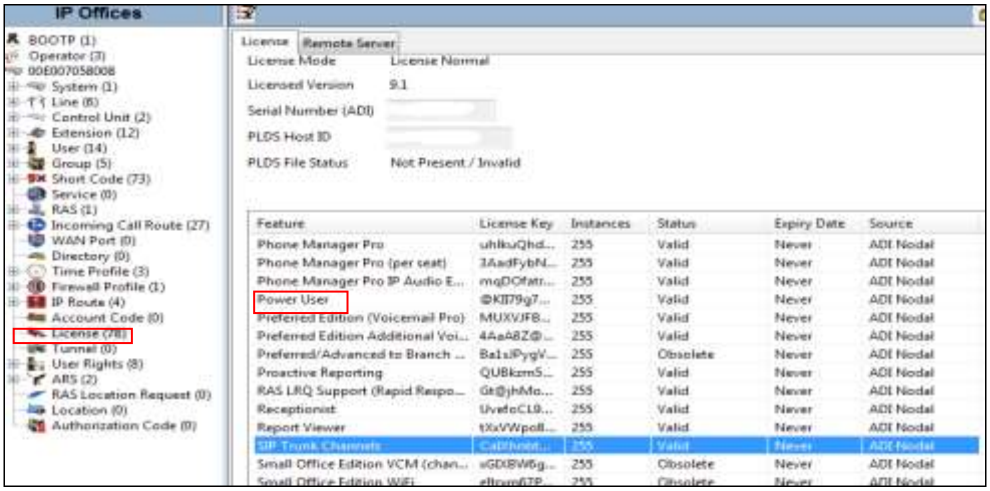
The Avaya IP Office 500 V2 has two Ethernet ports on the back of the chassis, labeled **WAN** and **LAN**. In the reference configuration, the LAN port (LAN1) is connected to the private CPE network. The Avaya SBCE, as well as the Avaya H.323 and SIP telephones, and the Avaya IP Office management/Softphone PC, are also connected to the private CPE network. The WAN port (LAN2) is not used in the reference configuration. Provisioning for the LAN1 interface is described in **Section 5.3.2**.

A default route must be added to the Avaya IP Office configuration. In the reference configuration this is **192.168.42.1**. To add an IP Route in Avaya IP Office, right-click **IP Route** from the left hand Navigation pane, and select **New** (not shown). To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the relevant default route using **Destination → LAN1**.

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 verify that **SIP Trunk Channels** has sufficient "Instances" (trunk channels). If any of those endpoints are to be defined as a **Power User**, then that must be licensed as well.



5.3. System Settings

This section illustrates the configuration of system settings. Select **System** in the left hand 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 section of the **System** tab. The **Name** field can be used for a descriptive name of the system.

5.3.2. LAN 1 Tab

In the reference configuration, LAN1 was used to connect the Avaya IP Office to the CPE network, and ultimately to the Avaya SBCE (see **Section 3**).

5.3.2.1 LAN 1 – LAN Settings Tab

To view or configure the LAN 1 IP address, select the **LAN 1 → LAN Settings** tab, and enter the following:

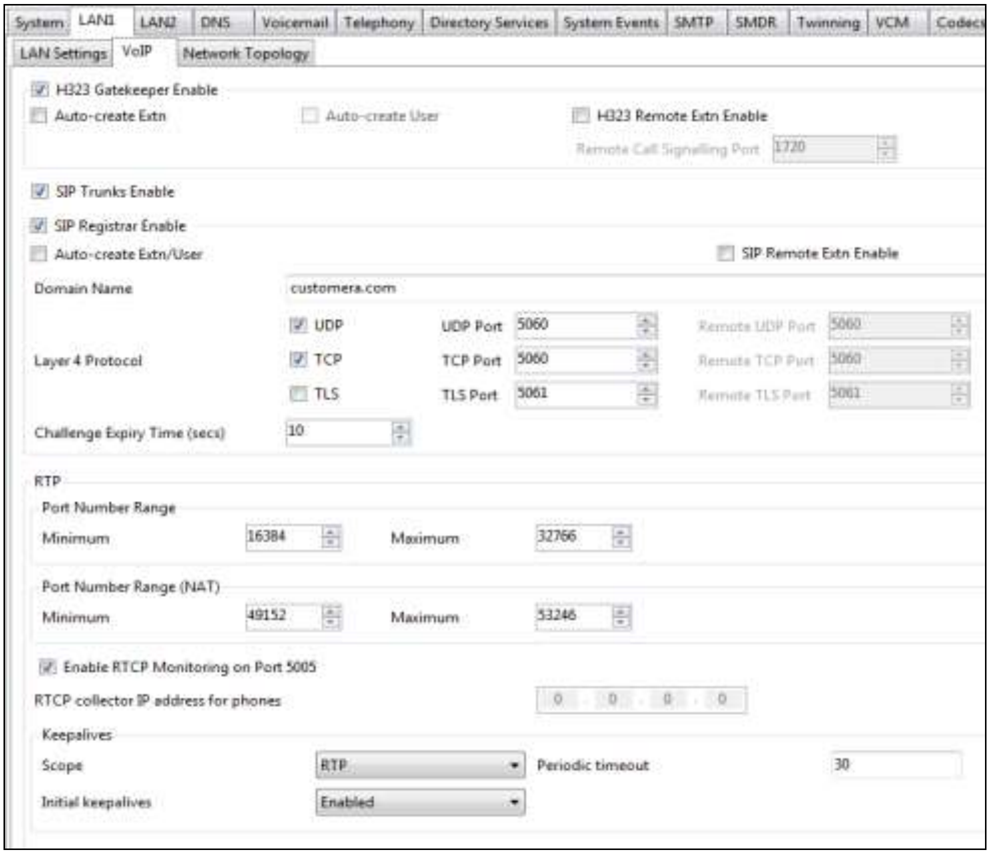
- **IP Address:** Set to **192.168.42.10** as specified in the reference configuration.

- **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).

5.3.2.2 LAN 1 - VoIP Tab

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

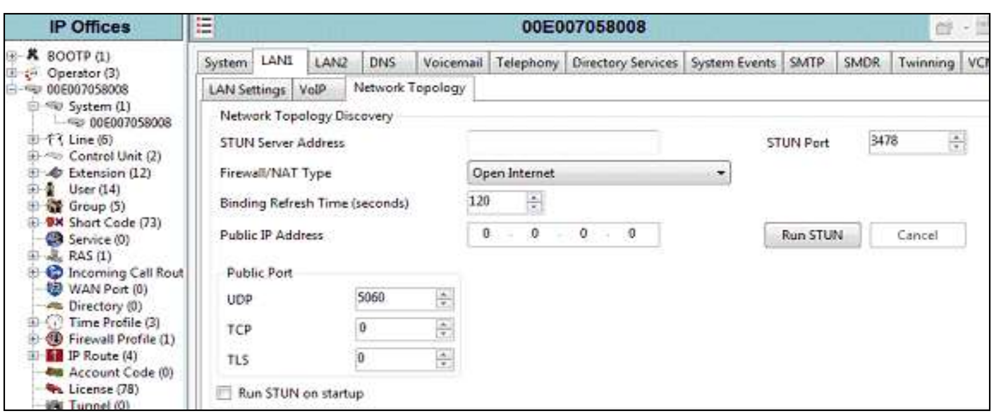
- The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 1600-Series Telephones used in the reference configuration.
- Select the **SIP Trunks Enabled** option.
- The **SIP Registrar Enable** box is checked to allow Avaya 11xx (SIP) and Avaya Communicator for Windows (SIP) usage.
- The **Domain Name** used in the reference configuration is **customera.com**.
- In the **Layer 4 Protocol** section, select **UDP/5060** and **TCP/5060**.
- **RTP Port Number Range:** The AT&T IPTF service requires that the RTP use the port range 16384 to 32767.
  - **16384** entered in the **Port Range (Minimum)** field.
  - **32766** entered in the **Port Range (Maximum)** field, as this field requires even numbers.
- OPTIONAL: To prevent possible issues with network firewalls closing idle RTP channels, **RTP Keepalives** may be enabled. Scrolling down to the bottom of the form, enter the following:
  - **Scope**: Select **RTP**
  - **Periodic Timeout**: Enter **30**
  - **Initial keepalives**: Select **Enabled**
- Other parameters on this screen are set to the defaults.
- Click the **OK** button (not shown).



5.3.2.3 LAN 1 - Network Topology Tab

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.
- **Public Port:** Enter **UDP/5060.**
- **Firewall/NAT Type** is set to **Open Internet** in the reference configuration.
- Other parameters on this screen are set to the defaults.
- Click the **OK** button (not shown).



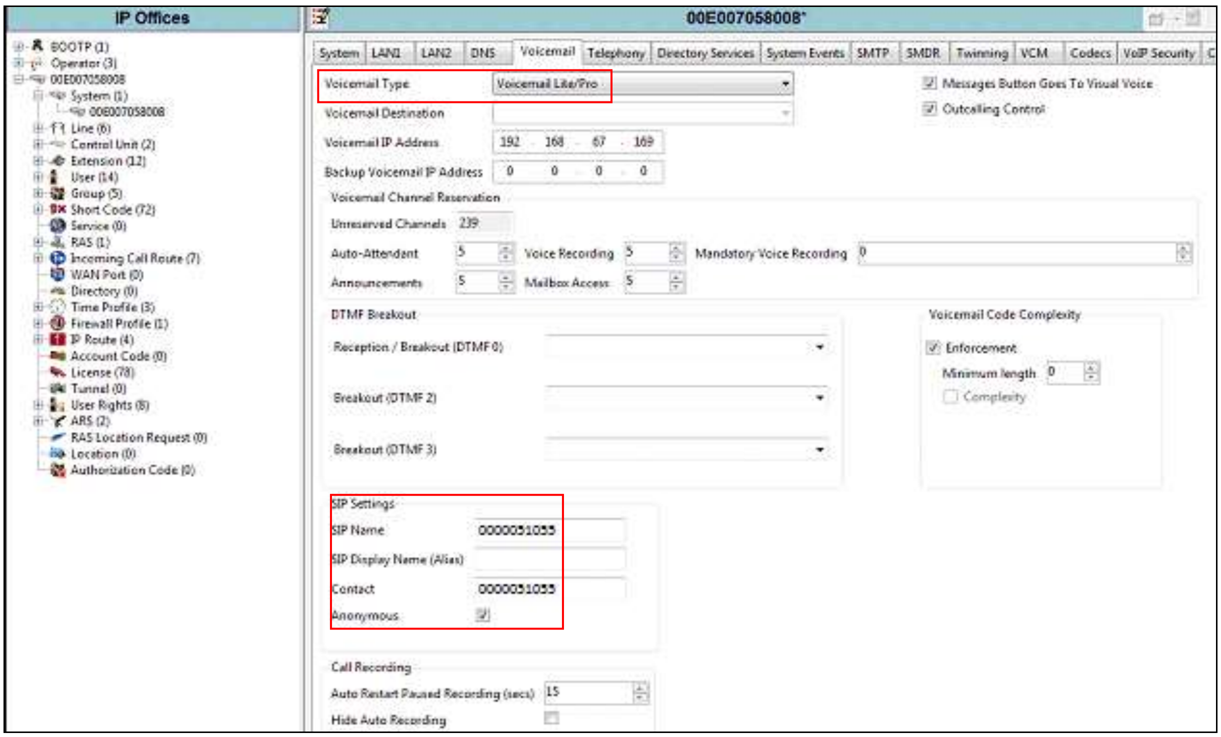
5.3.3. LAN 2 Tab

The LAN 2 interface is not used in the reference configuration.

5.3.4. Voicemail Tab

As described in **Section 3**, Avaya Voicemail Pro is used in the reference configuration.

- Set **Voicemail Type** to **Voicemail Lite/Pro**.
- In the **SIP Settings** section, set the **SIP Name** and **Contact** fields to the AT&T DNIS digits used to call directly to Voicemail for message retrieval (e.g., **0000051055**). Note that the **Anonymous** box is checked by default, so no entry is needed in the **SIP Display Name (Alias)** field.
- Other parameters on this screen are default. Click the **OK** button (not shown).

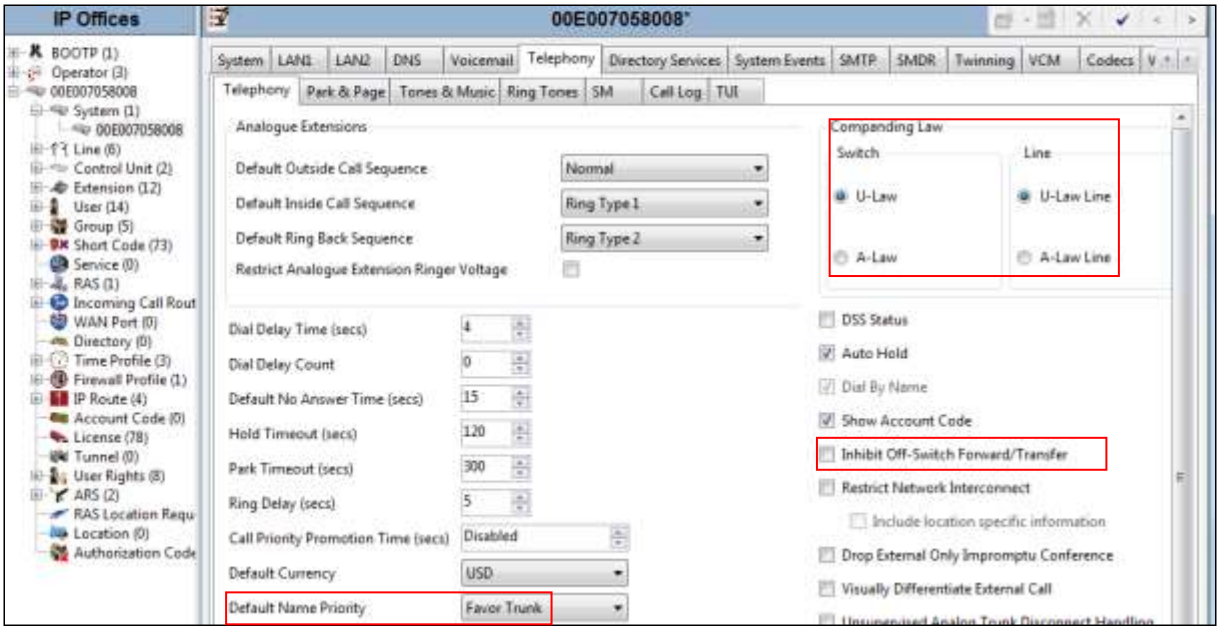


5.3.5. Telephony Tab

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

- *Uncheck* the **Inhibit Off-Switch Forward/Transfer** box. This is so that call forwarding and call transfer to PSTN destinations via the AT&T IPTF service can be tested.
- Set the **Companding Law** parameters are set to **U-LAW** as is typical in North America.

- In the reference configuration, **Default Name Priority** is set to **Favor Trunk**. 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 5.4.3**. A user's personal directory example is shown in **Section 5.5.2**.
- Default values are used in the other fields.
- Click the **OK** button (not shown).



5.3.6. 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 and extensions will assume the system default **Selected** codec list, unless configured otherwise for the specific SIP Line or extension (see the note below).

- Populate the **Selected** column with **G.711 ULAW 64K** as the first codec and **G.729(a) 8K CS-ACELP** as the second codec.
- In the **RFC2833 Default Payload** setting field, specify **100**, which is the recommended value for AT&T interoperability.
- Click the **OK** button (not shown).

Note - In the reference configuration, the Extension codec lists (see **Section 5.5.2**) also specify *G.711ULAW* and *G.729(a)* (in that order), and the SIP Line (see **Section 5.4.6)** offers *G.729(a)* and *G.711ULAW* (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. However, see **Sections 5.4.6** and **5.5.2** for methods of enabling Silence Suppression if needed.

5.4. SIP Line

The following sections describe the configuration of a SIP Line. The SIP Line terminates the CPE end of the IP Office SIP trunk to the Avaya SBCE.

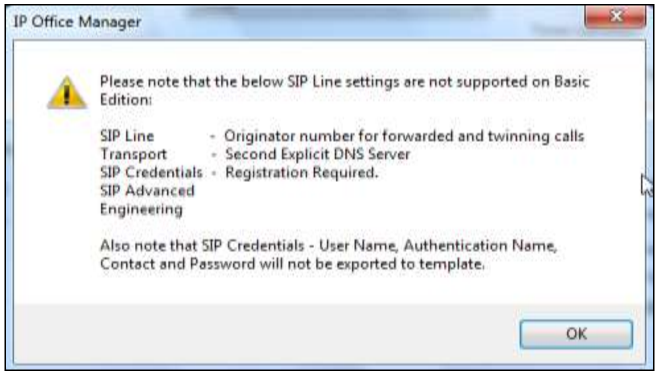
The recommended method for creating/configuring a SIP Line is to use the template associated with the provisioning described in these Application Notes. The template is an .xml file that can be used by Avaya IP Office Manager to create a new SIP Line for SIP trunking with the AT&T IPTF service. Follow the steps in **Section 5.4.2** to create a SIP Trunk from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses
- SIP Credentials (if applicable)
- SIP URI entries
- Setting of the **Use Network Topology Info** field on the Transport tab.

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary, after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration as shown in **Sections 5.4.3 – 5.4.8**.

In addition, the following SIP Line settings are not supported on Basic Edition:



Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.3 – 5.4.8**.

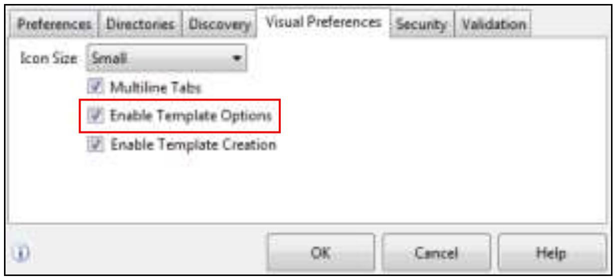
5.4.1. Importing a SIP Line Template

Note – DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500V2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer's environment.

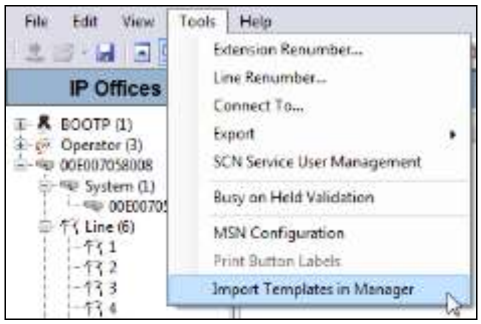
1. Copy a previously created template file to a location (e.g., *\temp*) on the same computer where IP Office Manager is installed. By default, the template file name will have the format **AF_<*user supplied text*>_SIPTrunk.xml**, where the **<*user supplied text*>** portion is entered during template file creation.

Note – If necessary, the **<*user supplied text*>** portion of the template file name may be modified, however the **AF_<*user supplied text*>_SIPTrunk.xml** format of the file name must be maintained. For example, an original template file **AF_*TEST* _SIPTrunk.xml** could be changed to **AF_*Test1*_SIPTrunk.xml**. The template file name is selected in **Section 5.4.2** to create a new SIP Line.

2. Verify that Template Options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the **Visual Preferences** tab. Check the box next to **Enable Template Options**. Click **OK**.

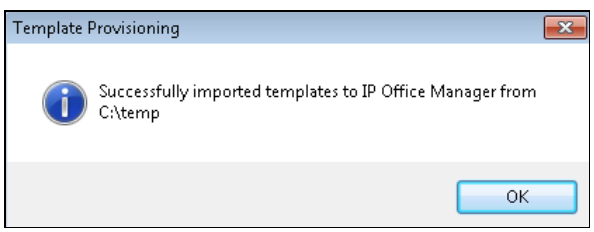


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**.

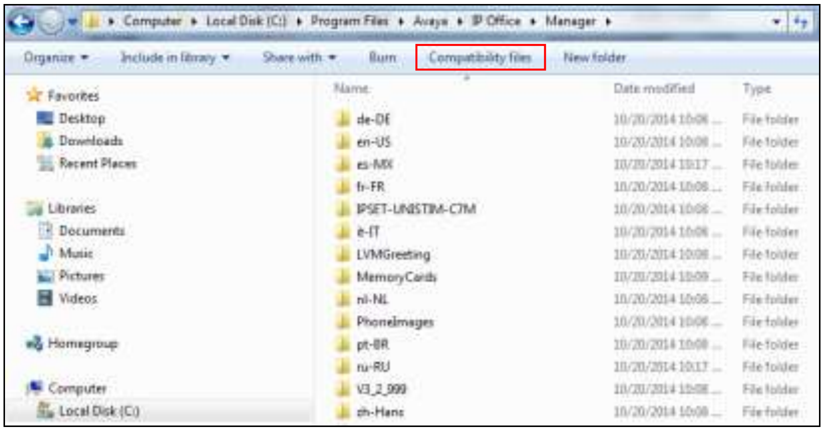


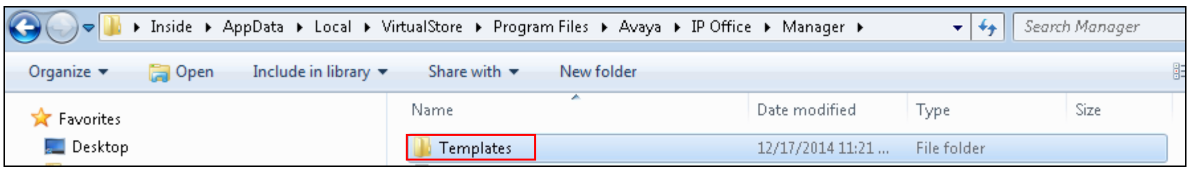
4. A folder browser will open (not shown). Select the directory used in **step 1** to store the template(s) (e.g., *\temp*). In the reference configuration, template files **AF_no_SBCE_ SIPTrunk.xml** and **AF_with_SBCE_ SIPTrunk.xml** were imported. The template files are automatically copied into the IP Office default template location, **C:\Program Files\Avaya\IP Office\Manager\Templates**.

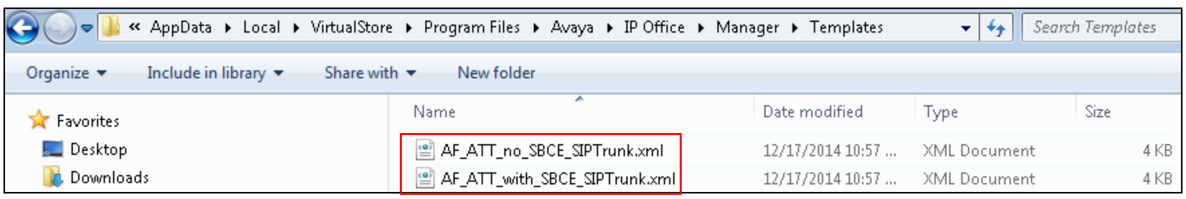
5. After the import is complete, a final import status pop-up window will open stating success or failure. Click **OK**.



Note –Windows 7 (and later) locks the Avaya IP Office 9.1 **\Templates** directory, and it cannot be viewed. To enable browsing of the **\Templates** directory, open Windows Explorer, navigate to **C:\Program Files\Avaya\IP Office\Manager** (or C:*Program Files (x86)*\Avaya\IP Office\Manager), and then click on the **Compatibility files** option shown below. The **\Templates** directory and its contents can then be viewed.

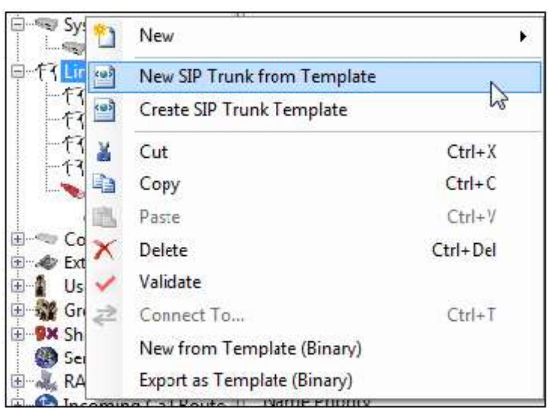






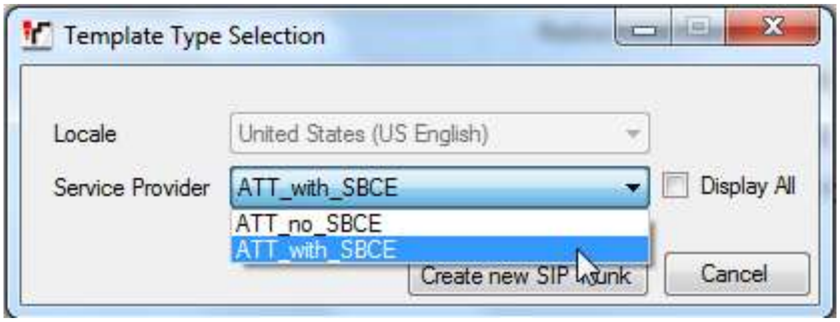
5.4.2. Creating a SIP Trunk from an XML Template

1. To create the SIP Trunk from a template, right-click on **Line** in the Navigation Pane, and select **New SIP Trunk from Template**.

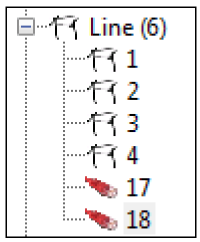


2. In the subsequent **Template Type Selection** pop-up window, from the **Service Provider** pull-down menu, select template **AF_with_SBCE**. Click **Create new SIP Trunk**.

Note – By default, the drop down menu will display only the ***<user supplied text>*** part of the template file name (see **Section 5.4.1**). If you check the **Display All** box, then the full .XML template file name is displayed.



The newly created SIP Line will appear in the Navigation pane (e.g., SIP Line **18**).

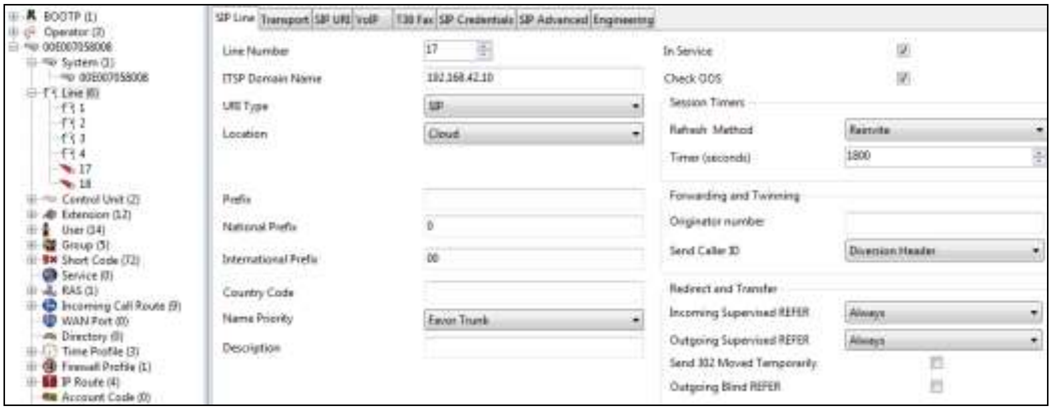


5.4.3. SIP Line – SIP Line tab

The **SIP Line** tab is shown below for **Line Number 17**, used for the SIP Trunk to the Avaya SBCE, and ultimately AT&T. Note, if no SIP Line exists, right click on the **Line** item in the **Navigation** pane and select **New → SIP Line** (not shown). In the reference configuration, SIP Line 17 was created. The SIP Line form is completed as follows:

- **ITSP Domain Name**: Set to the IP address of the Avaya IP Office LAN1 interface (e.g., **192.168.42.10**).
- **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 5.10**).
- **Refresh Method:** Set to **ReInvite**, as AT&T does not support UPDATE.
- **Send Caller ID**: Set to **Diversion Header**.
- **Incoming Supervised Refer**: Set this field to **Always**.
- **Outgoing Supervised Refer**: Set this field to **Always**.
- **Send 302 Moved Temporarily:** Verify this field is unchecked (default).
- **Outgoing Blind Refer**: Verify this field is unchecked (default).
- Use the default values for the other fields.
- Click **OK** (not shown).

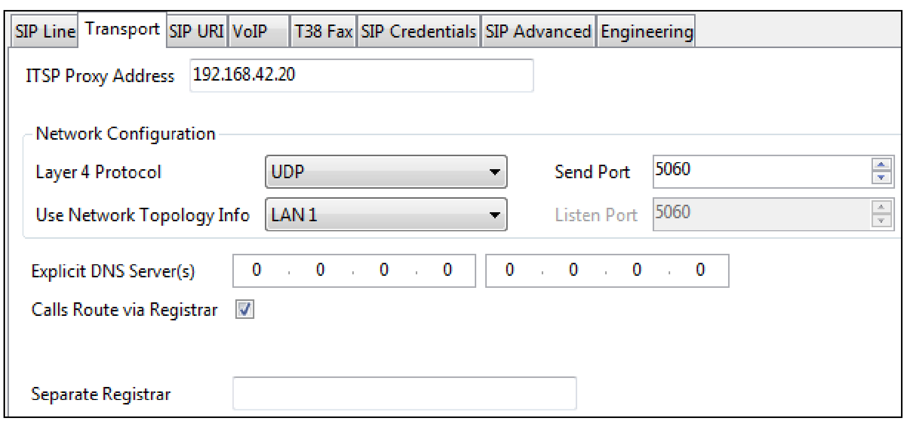
As described in **Section 5.3.5**, the **Name Priority** parameter may retain the default **Favor Trunk** setting, or can be configured to **Favor Directory**. As shown below, the default **Favor Trunk** setting was used in the reference configuration.



5.4.4. SIP Line - Transport tab

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

- **ITSP Proxy Address:** Set to the Avaya SBCE "A1" interface IP address (e.g., **192.168.42.20**).
- **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 1**.
- Verify **Calls Route via Registrar:** Enabled (default).
- **Click OK** (not shown).



5.4.5. SIP Line - SIP URI tab

A SIP URI entry needs to be created to match each number that Avaya IP Office and the service provider will accept on this line. Select the **SIP Line → SIP URI** tab. On this form a list of the DNIS digits delivered by AT&T is created.

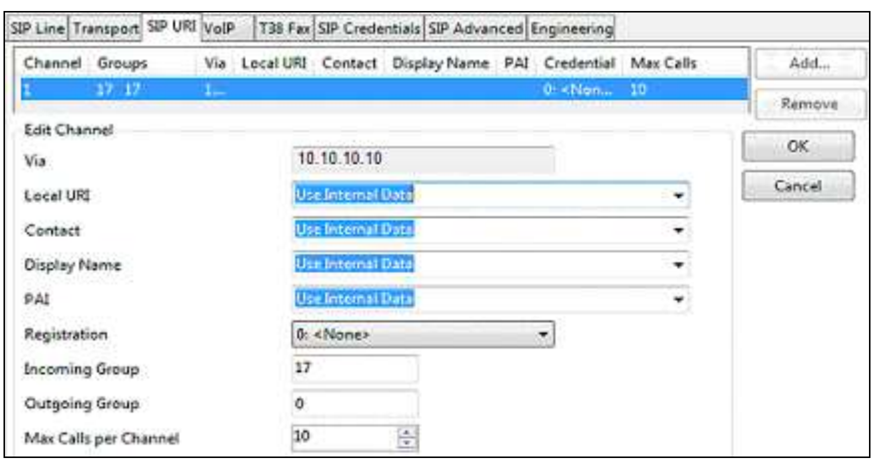
Note – In the reference configuration the AT&T IPTF service mostly delivered ten DNIS digits in the R-URI (some test calls delivered twenty-one digits). The entries below match on these DNIS digits, *not* the dialed DID number.

To add a new SIP URI, click the **Add…** button. At the bottom of the screen, a **New Channel** area will be opened. Two types of entries are used:

1. **Type 1**: A "global" entry that will use the contents of SIP headers containing "called party info" information. This type of entry is used for inbound calls to Avaya IP Office Users, Hunt Groups, or Voicemail access where the matching AT&T DNIS digits are specified on their corresponding **SIP Settings** (**Section 5.3.4**) or **SIP** tabs (see **Section 5.5**). Otherwise, the call will be denied. In this method the following information is specified:
   - The **Via** field will automatically be populated with the IP address of the LAN 2 interface with which the SIP trunk is associated (see **Section 5.3.3**).
   - **Local URI**, **Contact**, **Display Name**, **and PAI** fields: Set these fields to **Use Internal Data**.
   - Verify **Registration:** Set to the default **0: <None>**.
   - **Incoming Group:** Set to **17** (SIP Line 17). This value references the table created with **Incoming Call Routes** in **Section 5.7**.

   Note – As the IPTF service is inbound only, default **Outbound Group 0** is specified.

   - **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**.

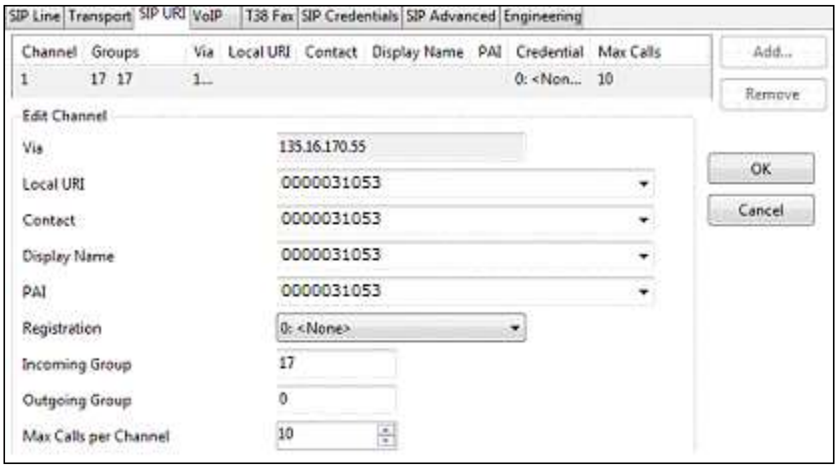


2. **Type 2**: This is an explicit entry matching inbound DNIS digits from AT&T. This method must be used for Avaya IP Office call destinations that cannot specify matching DNIS digits from AT&T. These call destinations may be Short Codes (e.g., Auto Attendant and Meet-Me conference), or other inbound destinations that do not have a SIP tab. For this method the following information is specified:

- **Local URI**, **Contact**, PAI, and **Display Name**: Set to an AT&T DNIS number (e.g., **00000031053).**
- Verify **Registration:** Set to the default **0: <None>**.
- **Incoming Group:** Set here to **17** (SIP Line 17). This value references the **Incoming Call Routes** in **Section 5.7**.

Note – As the IPTF service is inbound only, default **Outbound Group 0** is specified.

- **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.
- Repeat these steps as required, and click **OK** to save the information.



- To edit an existing entry, click an entry in the list and click the **Edit** button.
- When all SIP URI entries have been added/edited, click **OK** at the bottom of the screen (not shown).

5.4.6. SIP Line - VoIP tab

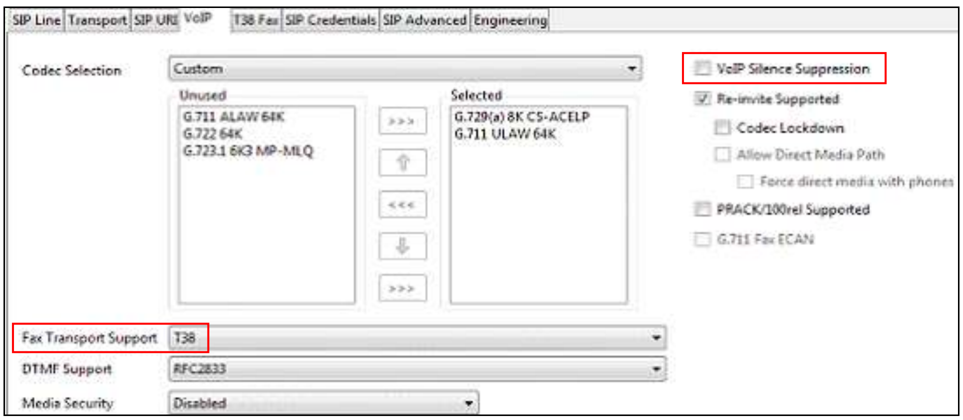
Select the **SIP Line → VoIP** tab and enter the following:

- 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 causes Avaya IP Office to include these codecs in the Session Description Protocol (SDP) offer, and in the order specified. Note that in the reference configuration G.729A is set as the preferred codec on the SIP trunk to the AT&T IPTF network (see the note below regarding IPTF and Silence Supression).
- T.38 fax was used in the reference configuration. Set the **Fax Transport Support** drop-down menu to **T.38**. Note that Error Correction Mode (ECM) is enabled by default on the **T.38 Fax** tab (**Section 5.4.7**). ECM is supported by the AT&T IPTF service. G.711 fax also worked in the reference configuration (T.38 option disabled); however T.38 is the preferred method.

> **Note** – With T.38 specified, the Avaya IP Office Direct Media feature cannot be selected. However if G.711 fax is selected, Direct Media may be used (see **Section 2.2**). Also, see **Section 2.2** relating to an issue with Super G3 (SG3) fax.

- 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.
- The **DTMF Support** parameter can remain set to the default value **RFC2833**.
- Click **OK** (not shown).

> **Note** - By default the VoIP Silence Suppression box is not checked, disabling the use of the G.729B codec. The AT&T IPTF service specifies G.729A by default. Therefore Silence Suppression (G.729B) was left disabled in the reference configuration. However G.729B is supported. If silence suppression is desired, check this box, and enable the **VoIP Silence Suppression** option on the **Extension** form **VoIP** tab for the various IP endpoints (**Section 5.5.2**).

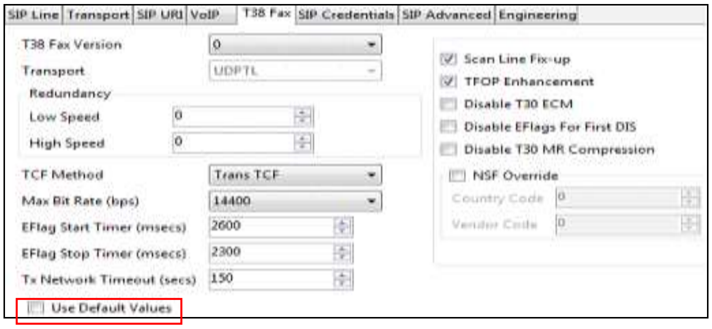


5.4.7. SIP Line - T38 Fax Tab

> **Note** - The settings on this tab are only accessible if **Re-invite Supported** and a **Fax Transport Support** option (**T38**) are selected on the **VoIP** tab (**Section 5.4.6**).

Select the **SIP Line → T.38 Fax** tab and enter the following:

- Unselect the **Use Default Values** option.
- Set the **T38 Fax Version** option to **0** (zero). This matches the version AT&T uses.
- Verify that **Disable T30 ECM** is *not* checked,
- Default values are used for the remaining fields. Select **Ok** (not shown).



5.4.8. SIP Line – SIP Advanced Tab

By default, Avaya IP Office will use the PPI (P-Preferred-Identity) header for signaling user information when privacy is invoked. However, AT&T utilizes the PAI (P-Asserted-Identity) header for privacy. Therefore Avaya IP Office is configured to use the PAI header to pass the calling party information for authentication and billing when privacy is used (see **Sections 5.4.5** and **5.9**).

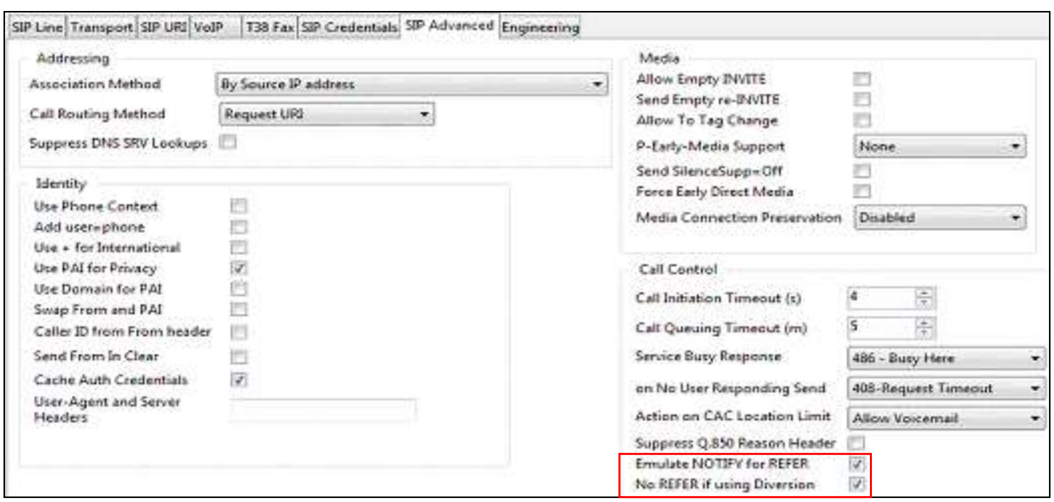
Select the **SIP Line → SIP Advanced** tab and enter the following:

- Select **Emulate NOTIFY for Refer**.

Note – The AT&T IPTF service does not support NOTIFY. Some Avaya endpoints (e.g., Avaya Communicator for Windows) require receipt of a NOTIFY when Refer based call transfers are performed. This option will send a NOTIFY to these endpoints.

- Select the **Use PAI for Privacy** option, and click **Ok** (not shown).

Note – By default, Avaya IP Office sends Refer in addition to Diversion header, for call forward scenarios. However AT&T only requires Diversion header. Therefore in the reference configuration the **No Refer if using Diversion** was selected.



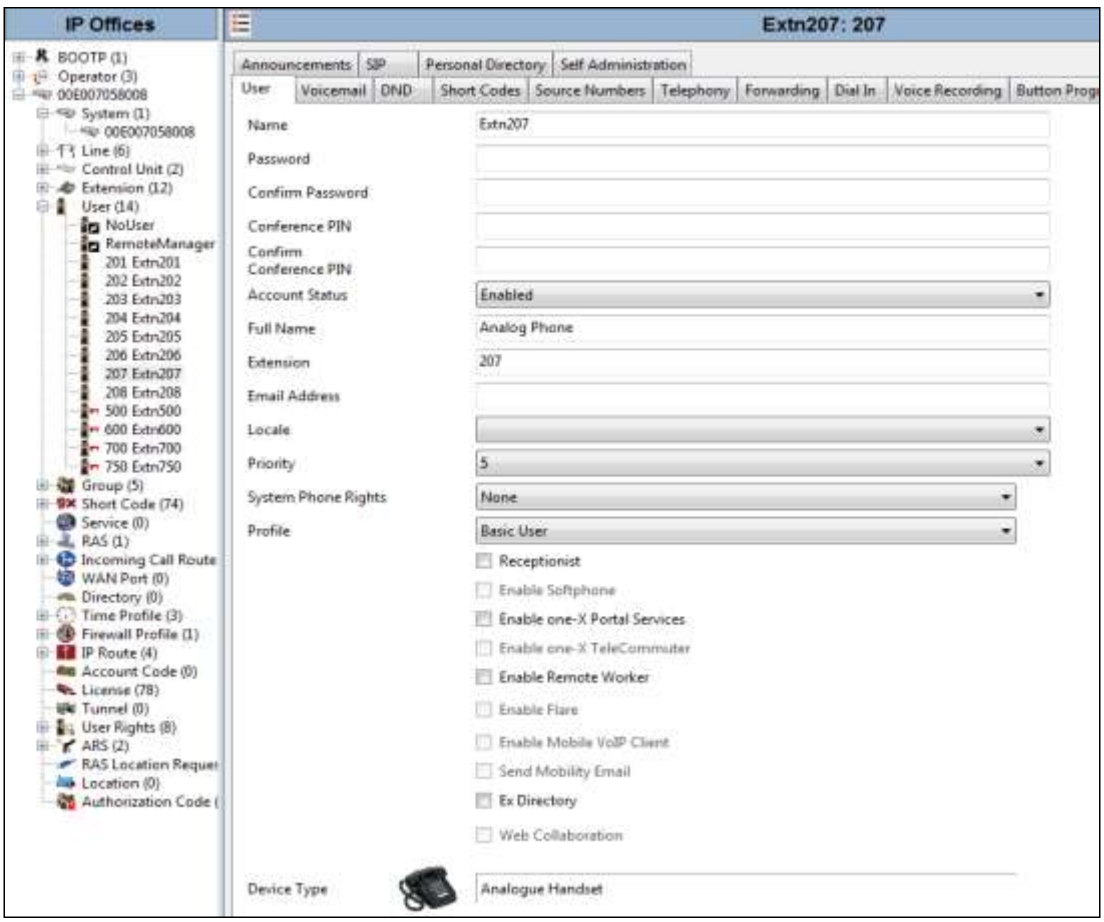
5.5. Users, Extensions, and Hunt Groups

In this section, examples of Avaya IP Office Users, Extensions, and Hunt Groups are illustrated. Note that the following examples do not discuss all available options, and the screen shots may not display all available parameters. Parameters/options note discussed, should assume to be default.

5.5.1. Analog User 207

The following screen shows the **User** tab for analog phone User **207**. This user corresponds to the Avaya Analog 6211 set.

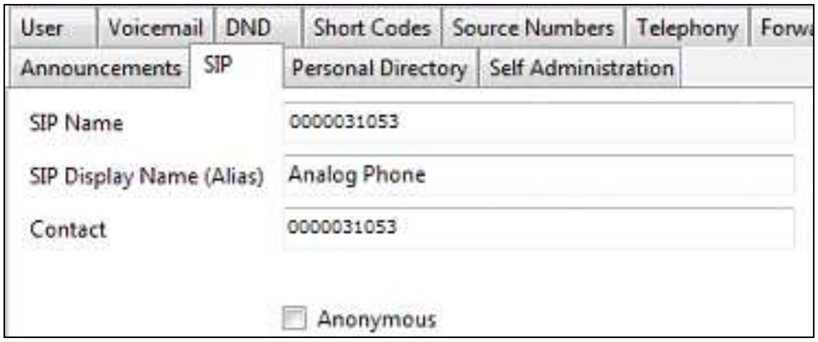
1. To add a User, right click on **User** in the Navigation pane, and select **New** (not shown). To edit an existing User, select **User** in the Navigation pane, and select the appropriate user to be configured.



The following screen shows the **SIP** tab for User **207**.

- The **SIP Name** and **Contact** parameters are configured with the associated AT&T DNIS number of the user, (e.g., **00000031053**). These parameters 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 5.4.5**).

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



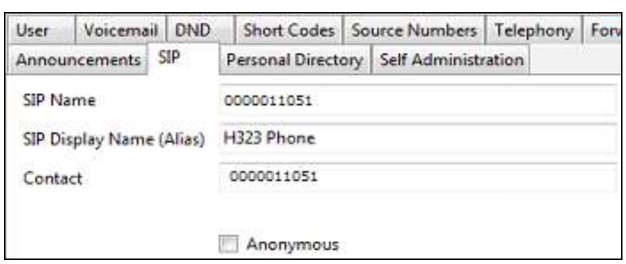
2. Analog (or digital) phone extension ports are either integral to the control unit or added by the installation of an analog or digital phone expansion module. Analog (or digital) extension records are automatically created for each physical extension port within the system. These ports cannot be added or deleted manually. For Server Edition, non-IP extensions are only supported on Expansion System (V2) units. Based on the hardware configuration used in the reference configuration, analog ports 207 and 208 are automatically defined by the system.
   - To edit an existing analog extension, select the appropriate extension to be configured (e.g., **207**).

   - Select the Analogue tab and verify that **Standard Telephone** is selected, and click the **OK** button (not shown).

5.5.2. IP Phone User 500

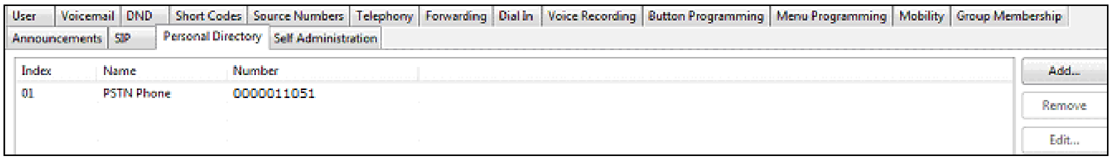
1. Following the steps shown in **Section 5.5.1**, create a 1608 H.323 IP phone user (e.g., **500**). Note that this user will be granted "Power User" features.
   - **Password**: This password is used by user applications such as SoftConsole, Phone Manager and TAPI, or users with Dial In access. Note that this is ***not*** the user's phone log in code (see the information on the **Telephony → Supervisor Settings** tab below), or their Voicemail mailbox password (see information on the **Voicemail** tab below).
   - The **Profile** parameter is set to **Power User**. This gives this user access to additional Avaya P Office features. See **[1]** for more information.

Like the analog user 207, the **SIP** tab for 500 is configured with a **SIP Name** and **Contact** specifying the user's associated AT&T number (e.g., **0000011051**).

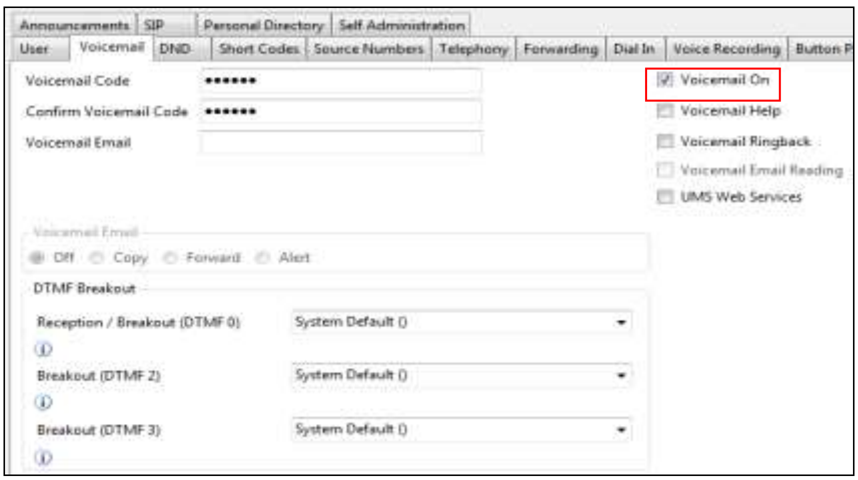


Avaya IP Office offers a feature where users can define names in a Personal Directory, and display these names, based on the inbound calling number. The following screen shows the **Personal Directory** tab for User **500**. With the configuration shown below, if user 500 receives an inbound AT&T call from the telephone number **0000011051**, 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.

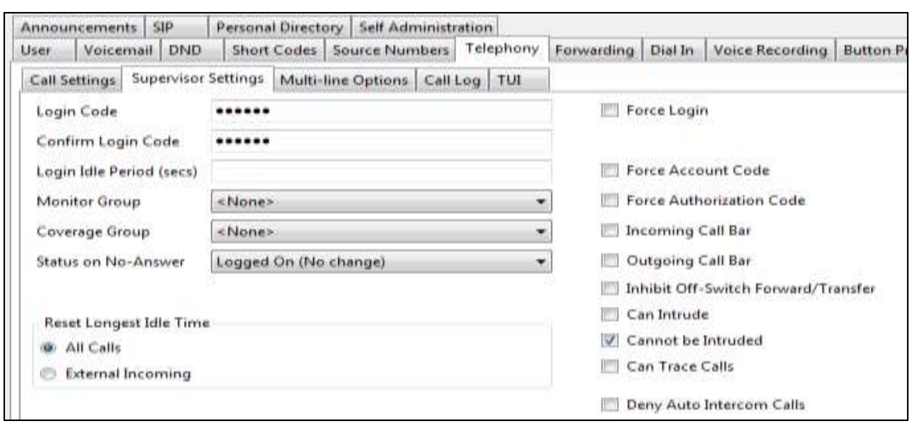
Note – In the reference configuration, the SIP Line is configured with **Name Priority → Favor Trunk** (see **Section 5.4.3**). To enable the Personal Directory feature, this setting needs to be changed to **Name Priority → Favor Directory**.



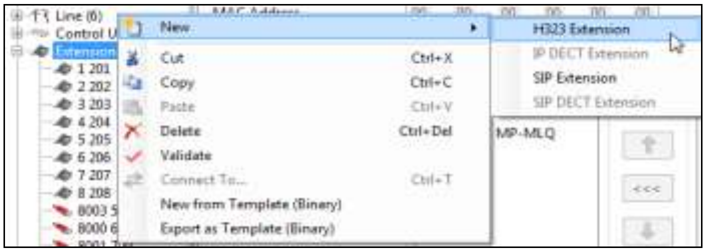
The following screen shows the **Voicemail** tab for user 500. The **Voicemail On** box is checked and a Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters.



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



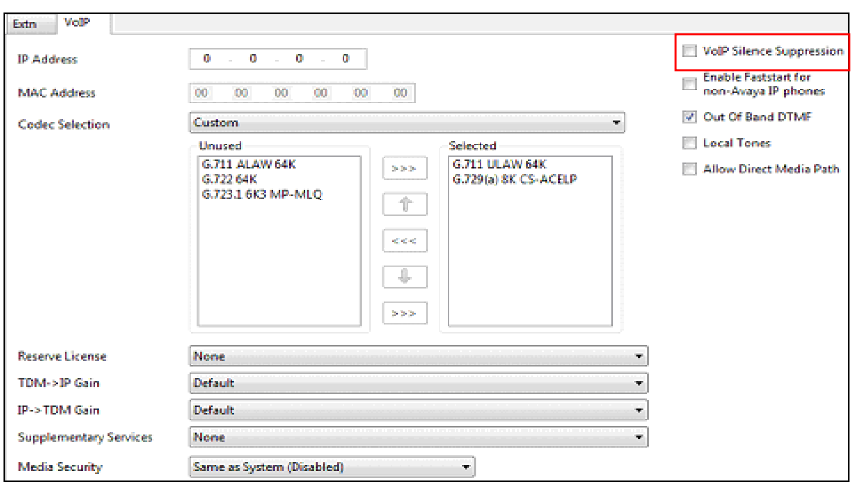
2. To create an associated extension, right click on **Extension** in the Navigation Pane, and select **New → H323 Extension**.



On the **Extn** tab, enter the **Base Extension** (e.g., **500**). Note that the **Extension ID** field will auto populate.

Select the **VoIP** tab and provision the following:

- Keep the **IP Address** field as the default value (**0.0.0.0**).
- Populate the **Selected** column with **G.711 ULAW 64K** as the first codec and **G.729(a) 8K CS-ACELP** as the second codec, (see **Section 5.3.7**).
- Verify **VoIP Silence Suppression** is not selected (see **Section 5.3.7**).
- Click the **OK** button (not shown).



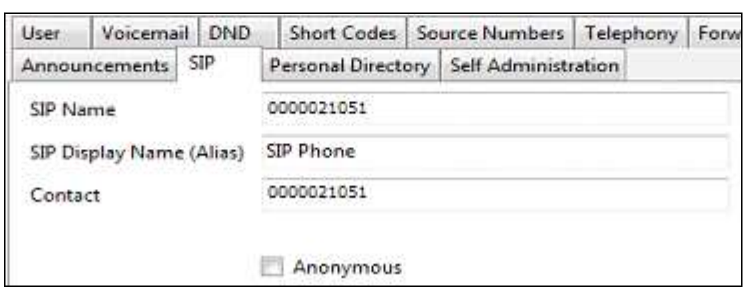
5.5.3. SIP Telephone Users (Avaya 1120E and Avaya Communicator)

In the reference configuration, an Avaya 1120E SIP telephone and Avaya Communicator softphone were provisioned as SIP users.

5.5.3.1 SIP Avaya 1120E

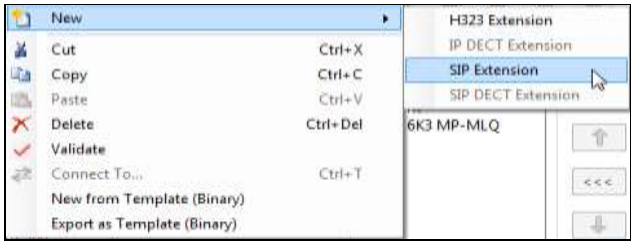
1. The following screen shows an 1120E Telephone provisioned in the **User** tab for User **600**. The provisioning of this user is the same as for the H.323 station in **Section 5.5.2**. Note that this station is set as a Basic User.

Like the H.323 500 user, the **SIP** tab for 600 is configured with a **SIP Name** and **Contact** specifying the user's associated AT&T number (e.g., **0000021051**).

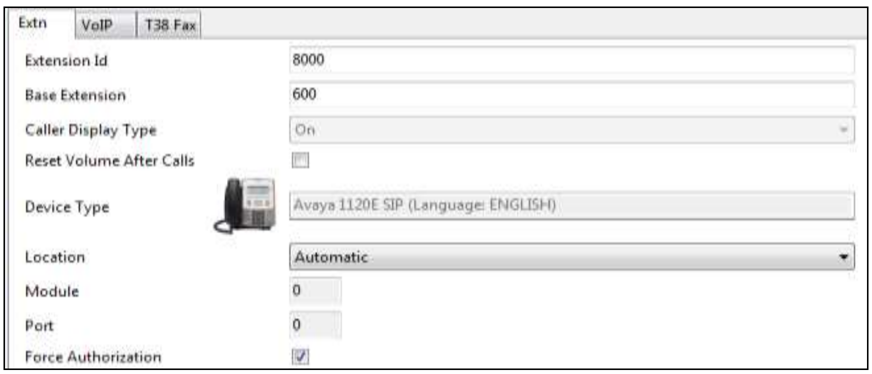


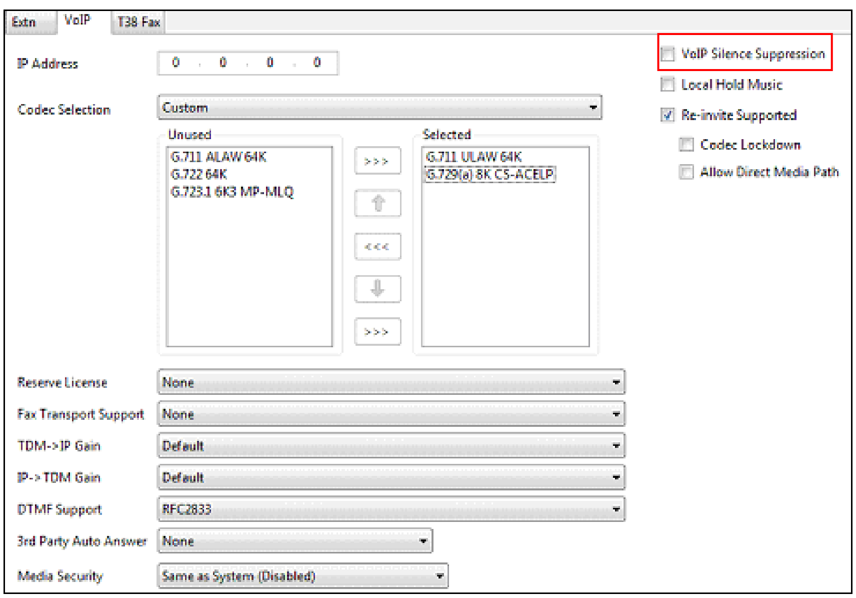
Voicemail and a **Login Code** are also defined following the examples shown for the H.323 User 500 in **Section 5.5.2**.

2. Following the steps shown in **Section 5.5.2** for the H.323 phone, create a corresponding **SIP Extension** for the 1120E SIP telephone (e.g., **600**).



The following screens show the **Extn** and **VoIP** tabs for the Avaya 1120E extension 600 (the **T.38 Fax** tab is not used). Note that the **Extension ID** on the **Extn** tab is auto populated by the system.

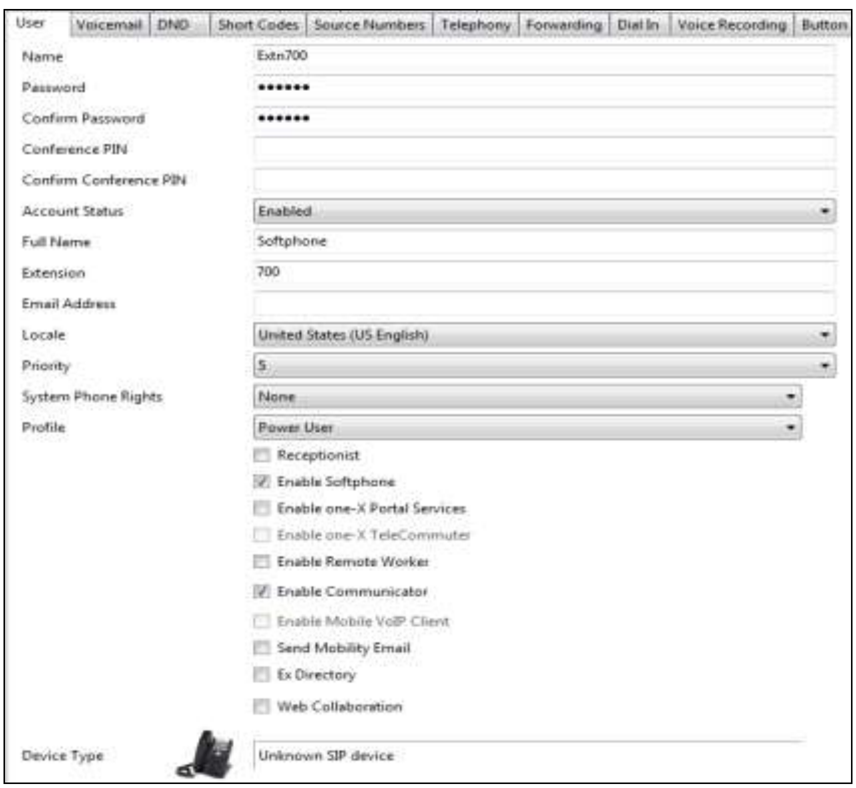


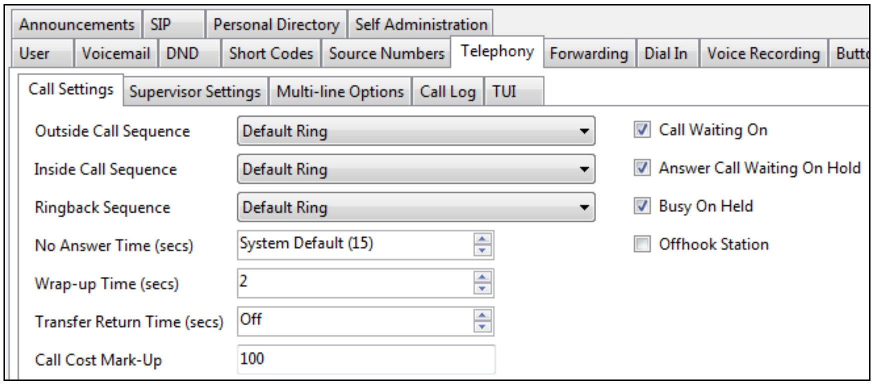


5.5.3.2 SIP Avaya Communicator Softphone

Repeat the steps shown in **Section 5.5.3.1** to create user **700** with the following settings.

1. Defining a **User**
   - **User** tab (shown below).
     - **Extension** = **700**
     - The **Profile** parameter is set to **Power User**. This gives this user access to additional Avaya P Office features. See **[1]** for more information.
     - The **Enable Softphone** box is checked (Power User required).
     - The **Enable Communicator** box is checked (Power User required).
   - **SIP** tab (not shown).
     - **SIP Name** and **Contact** specifying the user's associated AT&T DNIS number (e.g., **0000041051**).
   - **Voicemail** tab (not shown).
     - The **Voicemail On** box is checked.
   - **Telephony→ Call Settings** tab (shown below).
     - In the reference configuration, the **Call Waiting On** box, to allow multiple call appearances and transfer operations, was enabled. However depending on the desired call behavior, this setting may be mutually exclusive with the default **Busy On Held** setting. Combinations of these options should be attempted to achieve the desired effect.
   - **Telephony→ Supervisor Settings** tab (not shown)
     - The **Login Code** is specified.





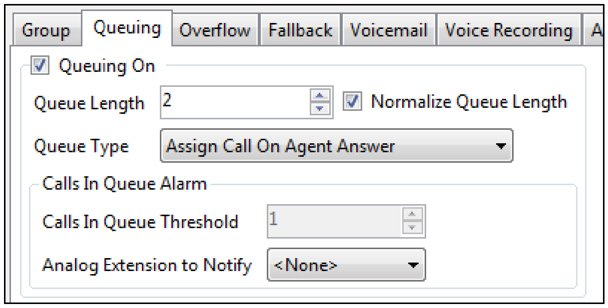
2. Define an **Extension** for the Avaya Communicator Softphone (e.g., **700**).
    - The **Extn** and **VoIP** tabs (not shown), are similar to those shown for extension 600 in **Section 5.5.3.1**.

5.5.4. Hunt Groups

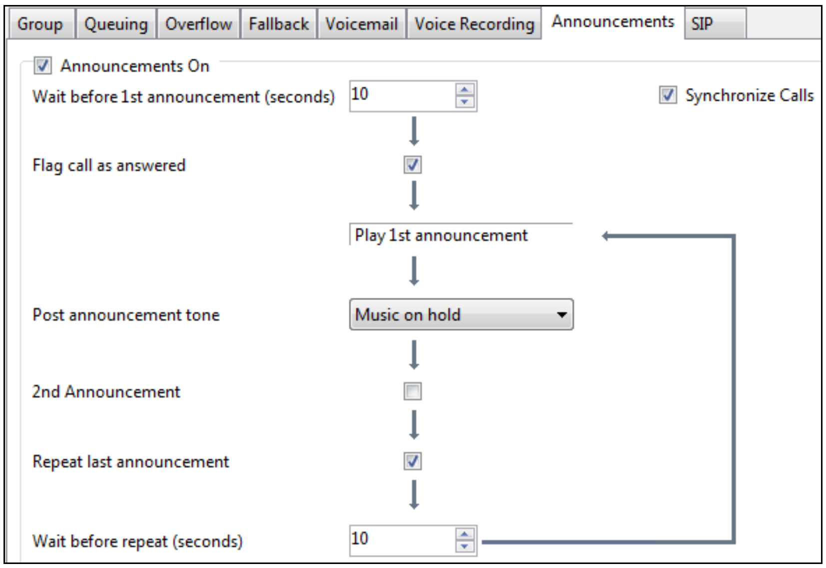
Users may also receive incoming calls as members of a hunt group. To configure a new hunt group, right-click **HuntGroup** from the Navigation pane and select **New**. To view or edit an existing hunt group, select **HuntGroup** from the Navigation pane, and the appropriate hunt group from the Group pane.

1. The following screen shows the **Hunt Group** tab for hunt group **Sales**. This hunt group was configured to contain the Analog telephone (Extn207), and the SIP Softphone (Extn700). 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 select/deselect from the **User List** included in the Hunt Group from the list of available users.

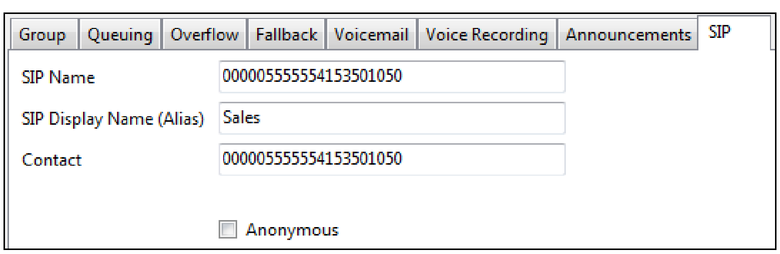
2. Under the **Queuing** tab, check the **Queuing On** box and set the **Queue Length** field to any desirable value. Use the default values for all the other fields.



3. Under the **Announcements** tab, check the **Announcements On** box. The wait time can be set to any desirable value. Make sure that the **Synchronize Calls** box is checked. These announcements are played if an agent for a particular skill is unavailable.



4. The following screen shows the **SIP** tab for hunt group **Sales**. The **SIP Name** and **Contact** are configured with the AT&T DNIS number **0000055555554153501050.** The Anonymous box may be checked is desired.



5. Click on **OK** (not shown).

In the reference configuration, these steps were used to create additional Hunt Groups "Service" (1002) and "Parts" (1003).

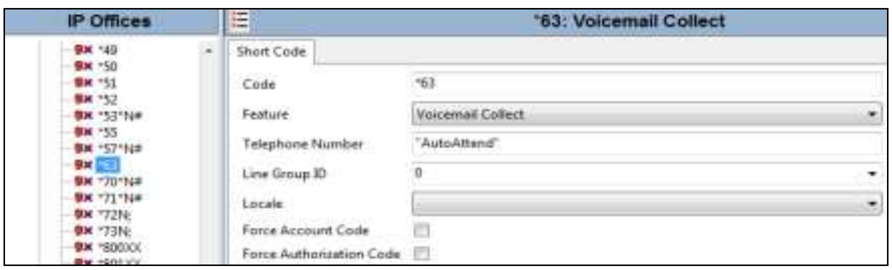
5.6. Short Codes

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. These Short Codes can be dialed directly by local IP Office users, or then can be defined as destinations in the **Incoming Call Route** table for remote access (**Section 5.7**).

5.6.1. Access Voicemail Pro Scripts

In the reference configuration, Call Center functionality is emulated via Voicemail Pro scripts (see **Sections 5.7.3** and **5.8**). Short Codes may be used to access these scripts. In the example below the following is defined to access an Auto-Attendant script:

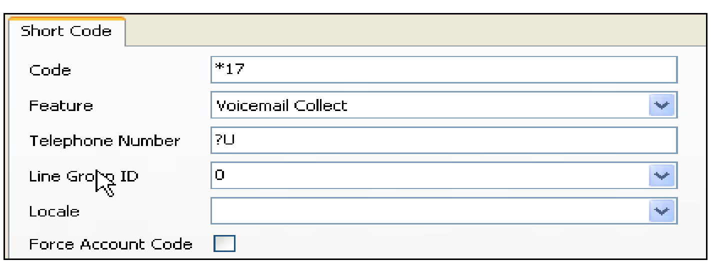
- **Code**: Enter the desired Short Code (e.g., ***63**).
- **Feature**: Select **Voicemail Collect** from the drop down menu.
- **Telephone Number**: Enter the name of the associated Voicemail Pro script (e.g., **"AutoAttend"**). Note that the script name must be in quotes, and match exactly with the name defined in Voicemail Pro.
- **Line Group ID**: Select **0** (default).
- Select **OK** (not shown).



5.6.2. Voicemail Access

In the reference configuration, Short Code ***17** is defined to access Voicemail Pro mailboxes.

- **Code**: Enter the desired Short Code (e.g., ***17**).
- **Feature**: Select **Voicemail Collect** from the drop down menu.
- **Telephone Number**: Enter **?U**.
- **Line Group**: Enter **0**.
- Click **OK** (not shown).



5.7. Incoming Call Routes

Note – The digits defined and matched in the Incoming Call Route table, are the DNIS digits specified in the AT&T Request-URI, not the DID digits dialed by the caller.

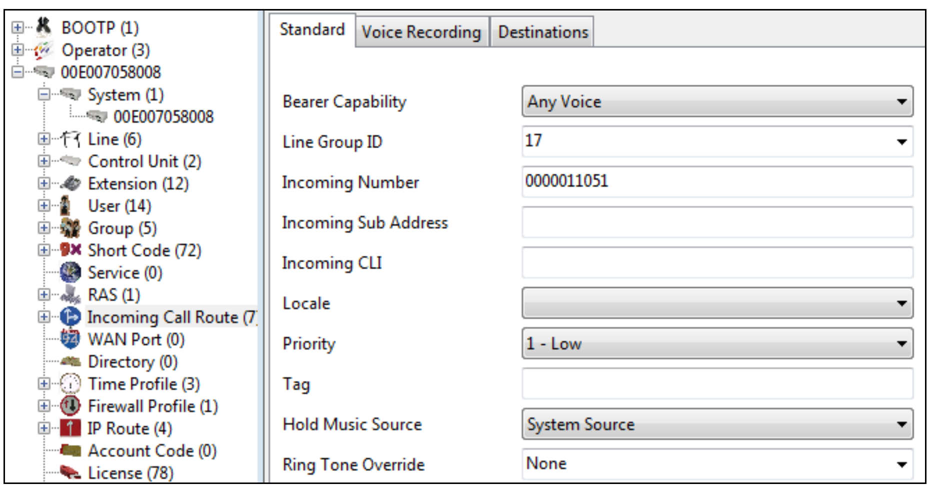
The Incoming Call Route table will map specific AT&T DNIS numbers to an IP Office User, Hunt Group, or Short Code, as well as to Voicemail Pro scripts.

To add an incoming call route, right click on **Incoming Call Route** in the Navigation pane, and select **New** (not shown). To edit an existing incoming call route, select an **Incoming Call Route** in the Navigation pane, and the associated call route information is displayed in the Group pane.

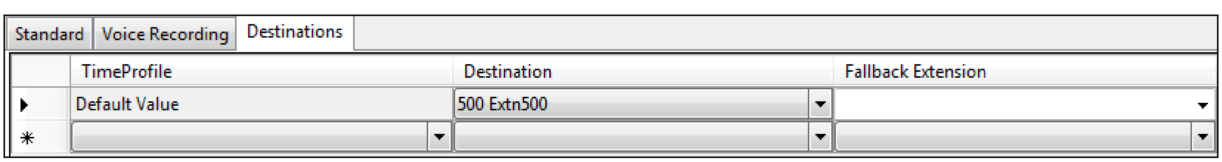
5.7.1. Calls to IP Office Stations and Hunt Groups

In the example below, the incoming number 0000011051 is directed to H.323 phone 500.

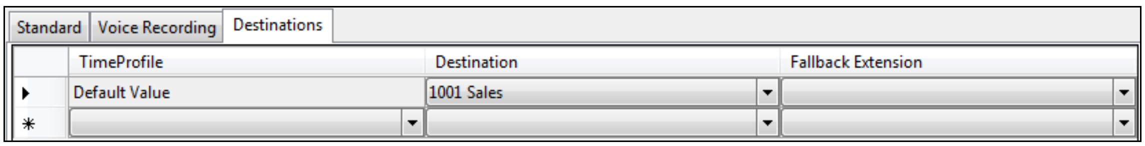
1. On the **Standard** tab enter the following:
   - **Line Group ID**: Enter the SIP Line defined in **Section 5.4** (e.g., **17**).
   - **Incoming Number**: Enter the associated DNIS digits sent by AT&T (e.g., **0000011051**).
   - Use default values for the remaining fields and click **OK** (not shown).



2. On the **Destinations** tab enter the following:
   - In the **Destinations** column, select extension **500** from the drop down menu.
   - Use default values for the remaining fields and click **OK** (not shown).



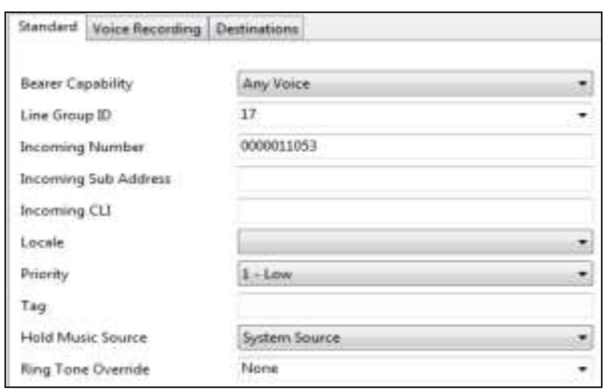
Below is an example of a call for **0000011052** being directed to Hunt Group **1001** (Sales).

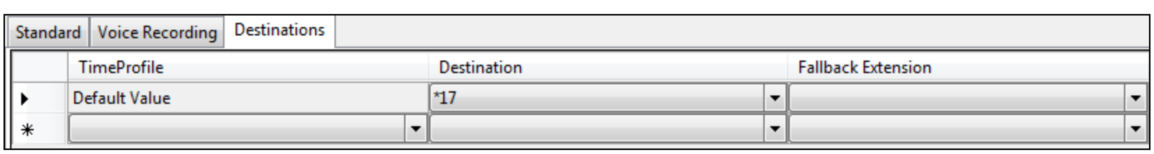


5.7.2. Calls to IP Office Short Codes

In the example below, incoming number 0000011053, is directed to Short Code *17 (access to Voicemail Pro mailboxes).

1. On the **Standard** tab repeat the steps in **Section 5.7.1**, with the following changes:
   - **Incoming Number**: Enter the associated DNIS digits sent by AT&T (e.g., **00000110513**).
2. On the **Destinations** tab enter the following:
   - In the **Destinations** column, and select Short Code ***17** from the drop down menu (note if the Short Code does not appear in the list, enter the value manually).
   - Use default values for the remaining fields and click **OK** (not shown).

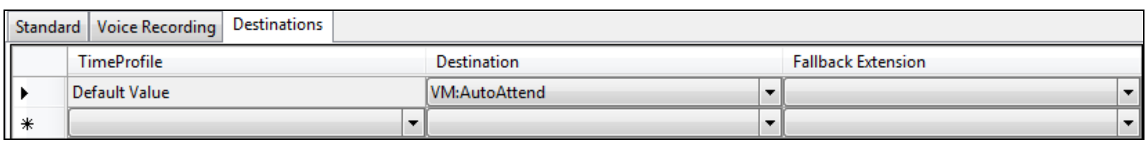




5.7.3. Calls to Voicemail Pro Scripts

As described in **Sections 5.6.1** and **5.8**, Voicemail Pro scripts are defined with specific names. These script names are specified as destinations in the Incoming Call Route table.
In the example below, incoming number **0000011054** is directed to eth Voicemail Pro Auto-Attendant script **AutoAttend**.

1. On the **Standard** tab repeat the steps in **Section 5.7.1**, with the following changes:
   - **Incoming Number**: Enter the associated DNIS digits sent by AT&T (e.g., **00000110514**).
2. On the **Destinations** tab enter the following:
   - In the **Destinations** column, enter the string **VM:AutoAttend** to the drop down menu.
   - Use default values for the remaining fields and click **OK** (not shown).



5.8. Call Center Provisioning in Avaya Voicemail Pro

Note - While Avaya Voicemail Pro provisioning and programming is beyond the scope of this document, a sample Auto-Attendant script is described below.

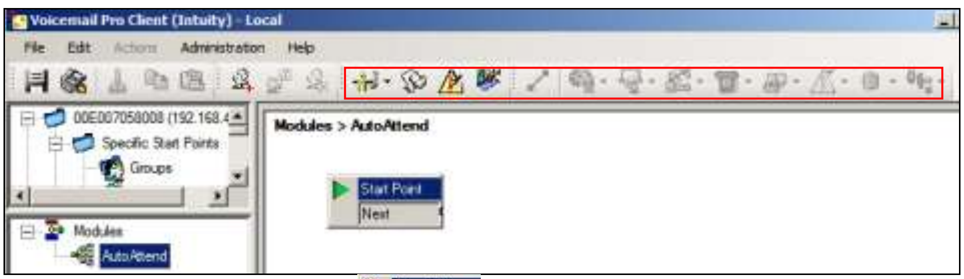
In the reference configuration, Avaya Voicemail Pro (running on a Windows 2003 server), is used for Voicemail processing, as well as for simulating basic Call Center functionality.

The Auto-Atendant function 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 Hunt Group (**Sales**, **Service**, and **Parts**). This is accomplished via the following steps:

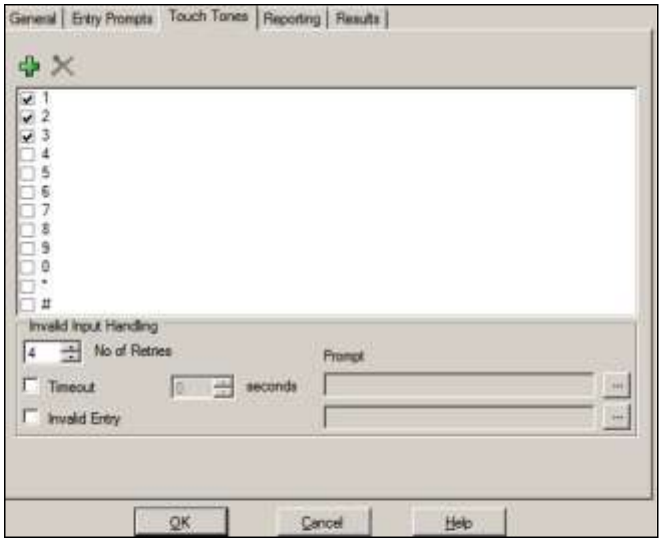
1. Hunt Groups **Sales**, **Service**, and **Parts** are created in IP Office (**Section 5.5.4)**.
2. Short Code ***63** is created in IP Office for Call Center access (**Section 5.6.1)**.
3. Incoming Call Route for DNIS digits **0000011054** is defined for access to the Auto-Attendant script (**Section 5.7.3**).
4. Via the Voicemail Pro GUI interface on the Voicemail Pro/Windows server:

- Open the **Voicemail Pro Client** application (not shown).
- Create a **Start Point** by right clicking on **Modules** and selecting **Add.**

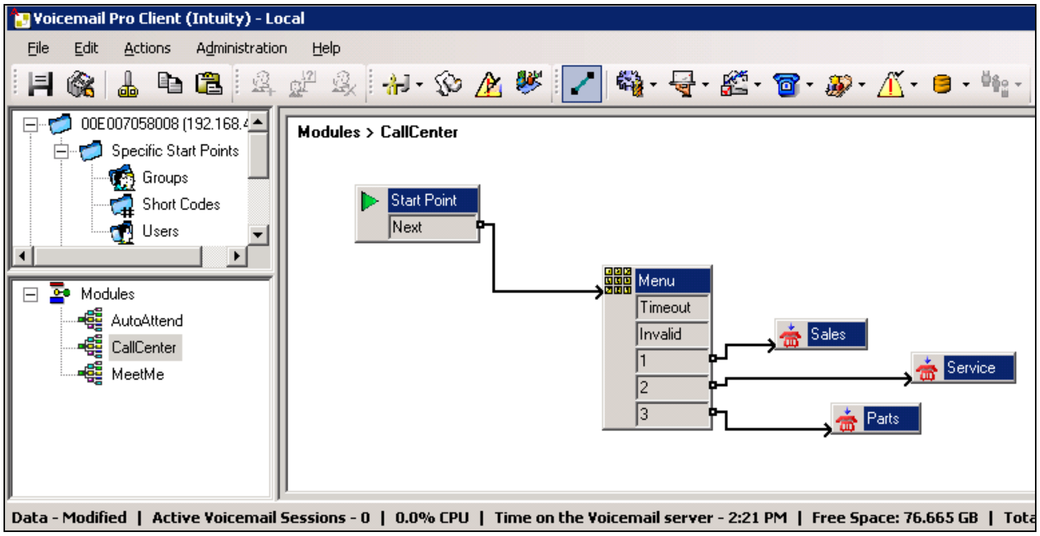
- Enter a name (e.g., **AutoAttend**) and click on **Ok** (not shown). The new script "AutoAttend" will appear under Modules, and a Start Point icon will appear in the work area. Note that most of the script options are grayed out (highlighted below).



- Click on the **Start Point** icon to activate the script options at the top of the screen. From the options, select the **Basic Actions** icon, select the **Menu** icon, and click on the work area to place the **Menu** icon.
    - i. Double click the **Start Point** icon.
        1. On the **General** tab → **Token Name**, enter **Start Point** and click **Ok** (not shown).
    - ii. Double click the **Menu** icon.
        1. On the **General** tab → **Token Name**, enter **Menu** (not shown)..
        2. On the **Entry Prompts** tab, select or 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 iconto open the .wav editor.
        3. On the **Touch Tone** tab:
            - a. Select **1, 2,** and **3** as the possible entry digits.
            - b. Select **4** for **No of Retries**.
        4. Click on **Ok.**



- Click on the Telephony Actions icon, select the Transfer icon, and click on the work area to place the **Transfer** icon in the work area. This will be used for "Sales". Select and place two more Transfer Icons (these will be used for "Service" and "Parts").
    i. Double click on the first **Transfer** icon ("**Sales**")
        1. On the **General** tab → **Token Name**, enter **Sales** (not shown).
        2. On the **Specific** tab → **Destination** → **Mailbox** → **Sales** → **Ok** (not shown).
    ii. Double click on the second **Transfer** icon **("Service**").
        1. On the **General** tab → **Token Name** = **Service** (not shown).
        2. On the **Specific** tab → **Destination** → **Mailbox** → **Service** → **Ok** (not shown).
    iii. Double Click on the third **Transfer** icon ("**Parts**").
        1. On the **General** tab, **Token Name** = **Parts** (not shown).
        2. On the **Specific** tab, **Destination** → **Mailbox** → **Parts** → **Ok** (not shown).
- From the options bar, 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).



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

When the associated AT&T DNIS number is received (e.g., **0000011054**), IP Office will send the call to Voicemail Pro. 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.

5.9. SIP Options

In the reference configuration, IP Office was configured to periodically check the status of the SIP Line by sending a SIP OPTIONS message to AT&T. This function is triggered by the setting in the **Binding Refresh Time** parameter. In the reference configuration, the Binding Refresh Time is set to 120 seconds (see **Section 5.3.3.3**). See **Section 2.2, Item 6** regarding OPTIONS header contents in the reference configuration.

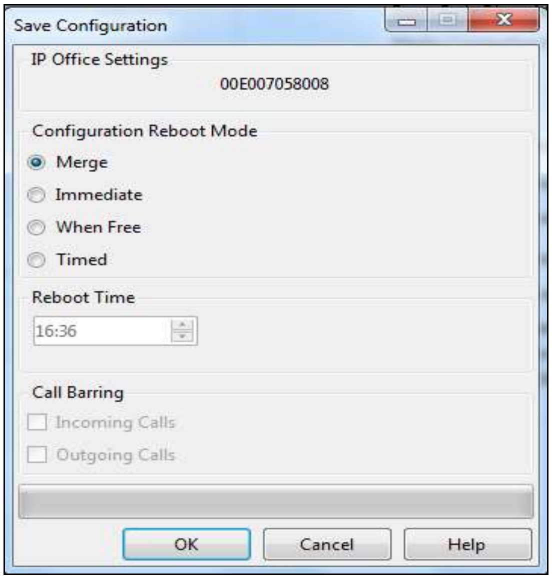
5.10. Saving Configuration Changes to Avaya IP Office

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. As noted in the previous sections, any changes made to an IP Office provisioning tab must be accepted by clicking **OK** on the associated screen. However these changes will not take effect until they are written to the IP Office configuration.

At the top of the Avaya IP Office Manager GUI, click **File → Save Configuration** (note that if that option is grayed out, no changes are pending).

A screen similar to the one below 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. Configure Avaya Session Border Controller for Enterprise

Note - Only the Avaya SBCE provisioning required for the reference configuration is described in these Application Notes.

6.1. Initial Installation/Provisioning

Note - The installation and initial provisioning of the Avaya SBCE is beyond the scope of this document. Refer to reference **[1]** and **[3]** for additional information.

IMPORTANT! – During the Avaya SBCE installation, the Management interface, (labeled "M1"), of the Avaya SBCE must be provisioned on a different subnet than either of the Avaya SBCE private and public network interfaces (e.g., A1 and B1). If this is not the case, contact your Avaya representative to have this resolved.

The Avaya SBCE installation typically defines public and private networks. As described in **Section 3**, the reference configuration defines the Avaya SBCE private interface A1 (IP address 192.168.42.20) on the same CPE network as the Avaya IP Office LAN1 interface (IP address 192.168.42.10). The connection to AT&T used the Avaya SBCE public interface B1 (IP address 10.10.10.10).

6.2. Log into the Avaya SBCE

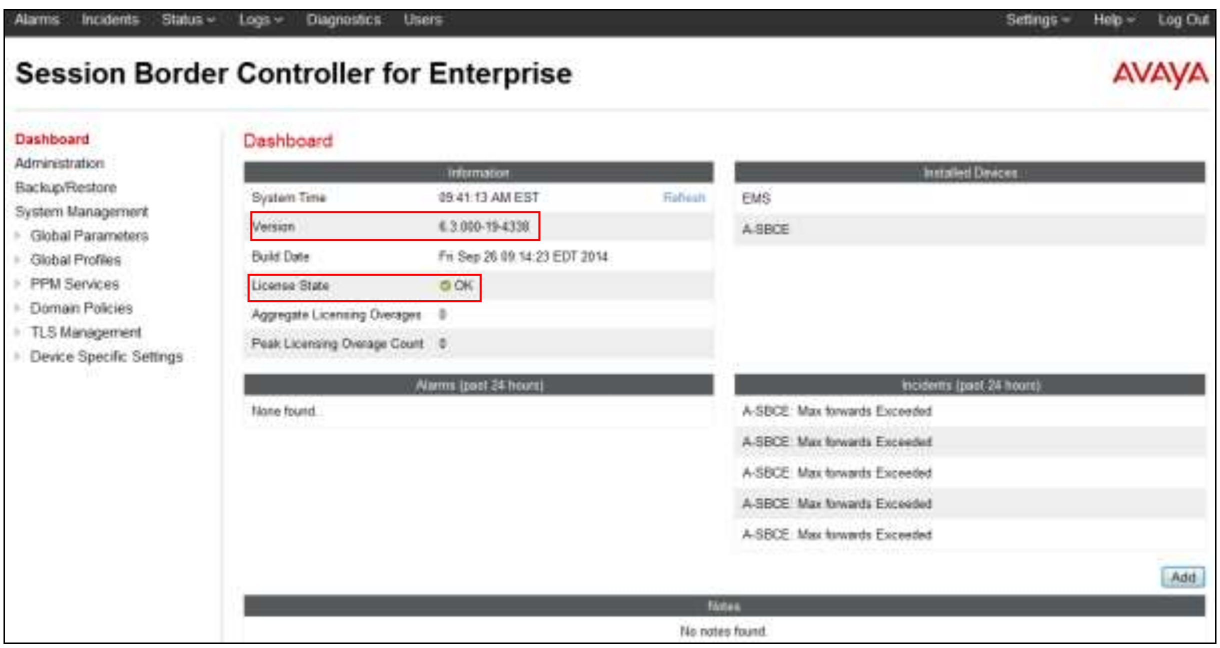
The follow provisioning is performed via the Avaya SBCE GUI interface, using the "M1" management LAN connection on the chassis.

1. Access the web interface by typing "**https://x.x.x.x**" (where x.x.x.x is the management IP address of the Avaya SBCE).
2. Enter the **Username** and click on **Continue**.

3. Enter the password and click on **Log In**.

4. The main menu window will open. Note that the installed software version is displayed. Verify that the **License State** is **OK**. The SBCE will only operate for a short time without a valid license. Contact your Avaya representative to obtain a license.

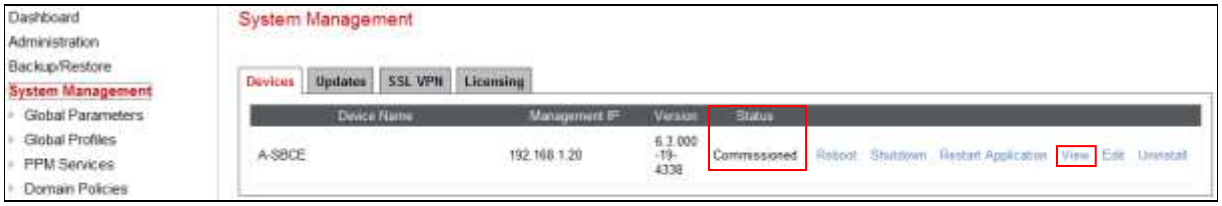
Note – The provisioning described in the following sections use the menu options listed in the left hand column.



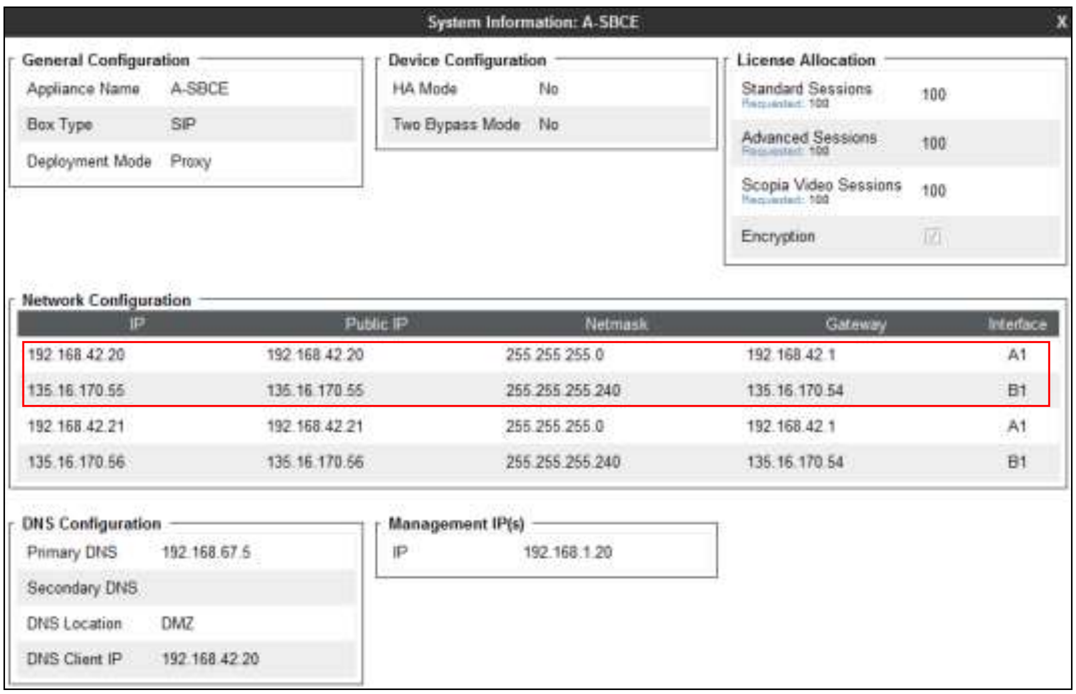
Note – The Avaya SBCE supports a Remote Worker configuration whereby Avaya IP Office endpoints residing on the public side of the Avaya SBCE, can securely register/operate with Avaya IP Office in the private CPE. While Remote Worker functionality was tested in the reference configuration, Remote Worker provisioning is beyond the scope of this document. See **[1]** for more information on Remote Worker in an Avaya IP Office environment.

6.3. System Management – Status

1. Select **System Management** and verify that the **Status** column says **Commissioned**. If not, contact your Avaya representative.



2. Click on **View** (shown above) to display the **System Information** screen. Note that the first two A1 and B1 interfaces listed are those referenced in this document for SIP trunking between Avaya IP Office and AT&T. The second two A1 and B1 Interfaces were use for Remote Worker and are not discussed.



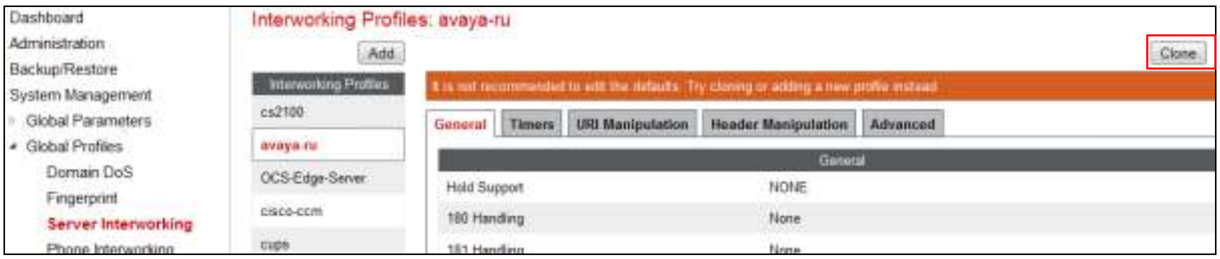
6.4. Global Profiles

Global Profiles allow for configuration of parameters across the Avaya SBCE appliances.

6.4.1. Server Interworking – to Avaya IP Office

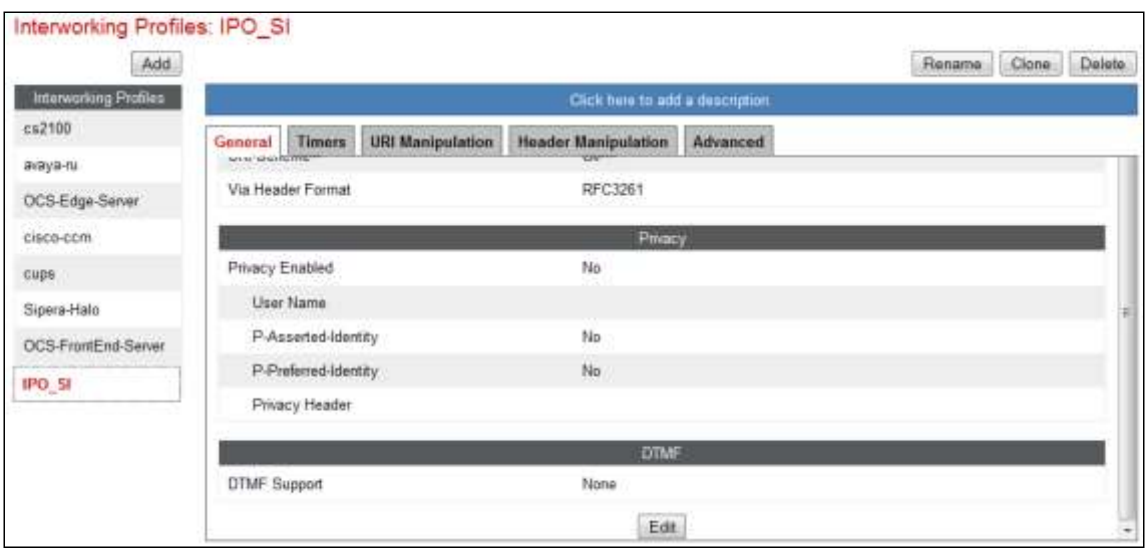
Server Interworking allows users to configure and manage various SIP call server-specific capabilities such as call hold and T.38 faxing. This section defines the connection to Avaya IP Office.

1. Select **Global Profiles → Server Interworking** from the left-hand menu.
2. Select the pre-defined **avaya-ru** profile and click the **Clone** button.

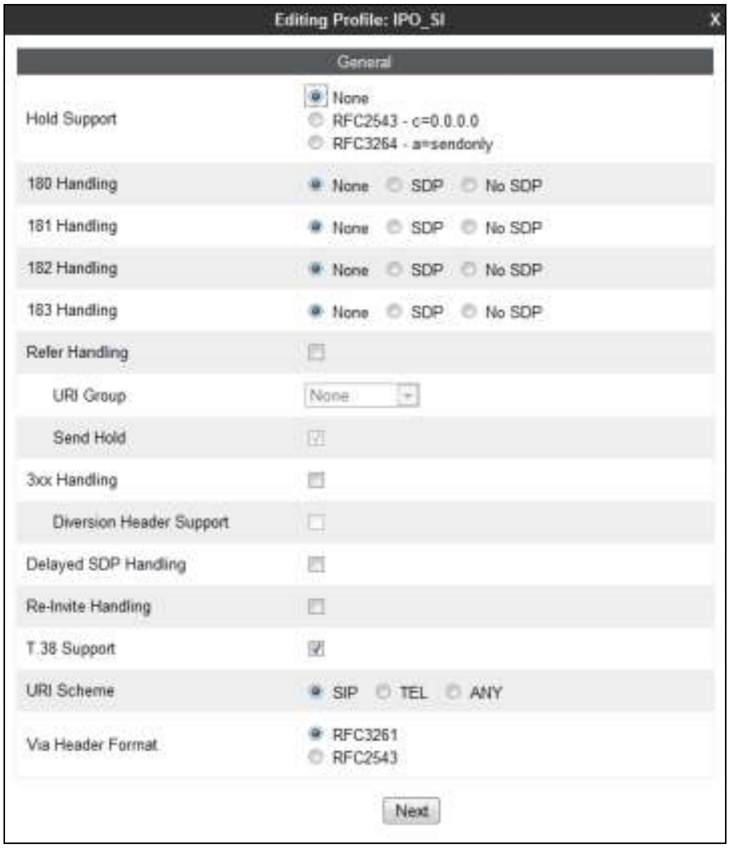


3. Enter profile name: (e.g., **IPO_SI**), and click **Finish**.

4. The new IPO_SI profile will be listed. Select it, scroll to the bottom of the Profile screen, and click on **Edit**.

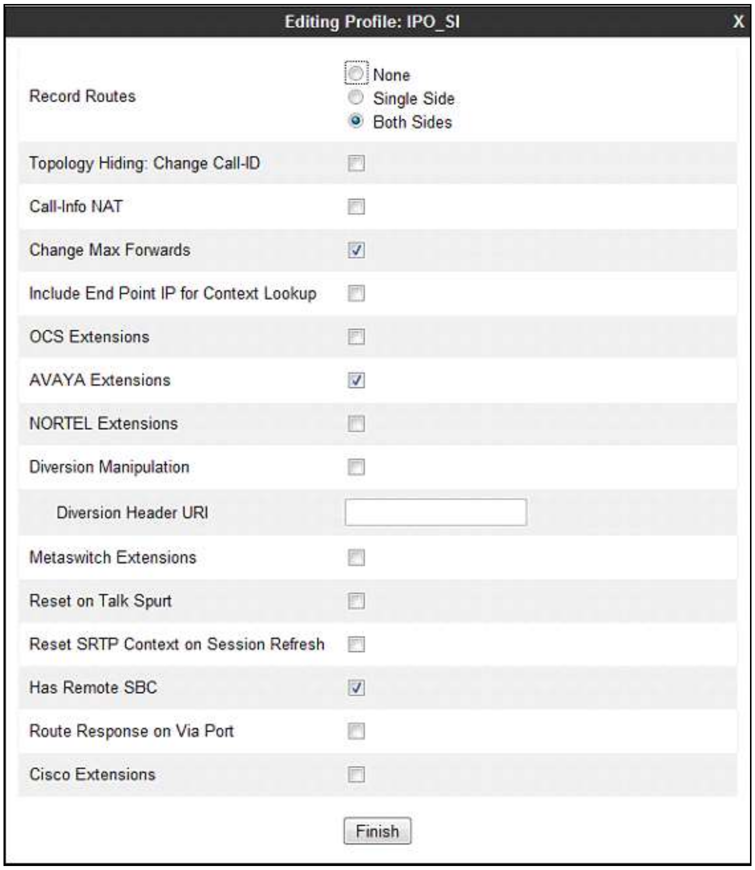


5. The **General** screen will open.
    a. Check **T38 Support**.
    b. All other options can be left with default values.
    c. Click **Next.**



6. On the **Privacy/DTMF** window, select **Finish** to accept default values.

7. Returning to the **General** screen, select the **Advanced** tab, accept the default values, and click **Finish**.



6.4.2. Server Interworking – to AT&T

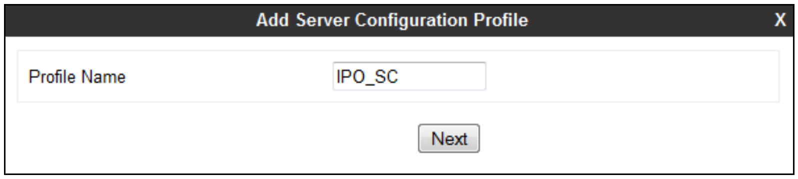
Repeat the steps shown in **Section 6.4.1** to add an Interworking Profile for the connection to AT&T via the public network, with the following changes:

1. Select **Add Profile** (not shown) and enter a profile name**:** (e.g., **ATT_SI**) and click **Next** (not shown).
2. The **General** screen will open (not shown):
    a. Check **T38 Support**
    b. All other options can be left as default.
    c. Click **Next**
3. The **Privacy/DTMF**, **SIP Timers/Transport Timers**, and **Advanced** screens will open (not shown), accept default values for all the screens by clicking **Next**, then clicking on **Finish** when completed.

6.4.3. Server Configuration – Avaya IP Office

This section defines the Server Configuration for the Avaya SBCE connection to Avaya IP Office.

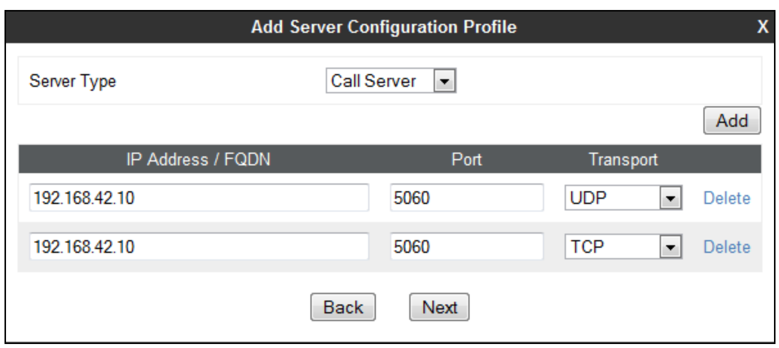
1. Select **Global Profiles → Server Configuration** from the left-hand menu.
2. Select **Add Profile** and the **Profile Name** window will open. Enter a Profile Name (e.g., **IPO_SC**) and click **Next**.



3. The **Add Server Configuration Profile** window will open.
    a. Select **Server Type**: **Call Server**
    b. **IP Address**: **192.168.42.10** (Avaya IP Office LAN1 IP Address)
    c. **Supported Transports**: Check **UDP**
    d. **UDP Port**: **5060**

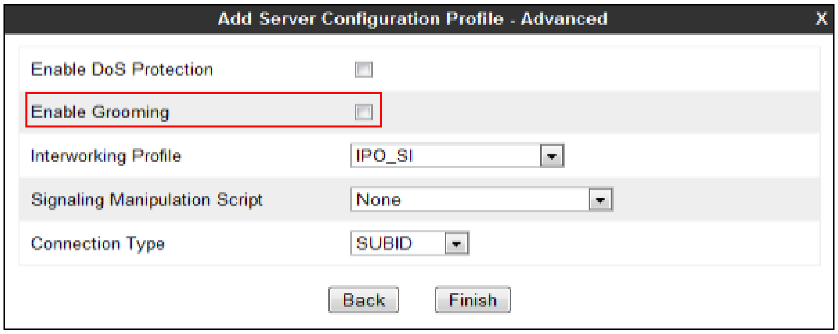
    Note – UDP is the recommended protocol to use on the connection between the Avaya SBCE and Avaya IP Office for SIP Trunking (see **Section 5.4.4** and **6.6.4**). However TCP may be used if desired and is also shown below.

    e. Select **Next**



4. The **Authentication** and **Heartbeat** windows will open (not shown).
    a. Select **Next** to accept default values.
5. The **Advanced** window will open.
    a. Select **IPO_SI** (created in **Section 6.4.1**), for **Interworking Profile**.
    b. In the **Signaling Manipulation Script** field select **none.**
    c. Select **Finish**.

Note – If TCP transport is specified in **Step 3**, then the **Enable Grooming** option should be enabled .



6.4.4. Server Configuration – AT&T

Repeat the steps in **Section 6.4.3**, with the following changes, to create a Server Configuration for the Avaya SBCE connection to AT&T.

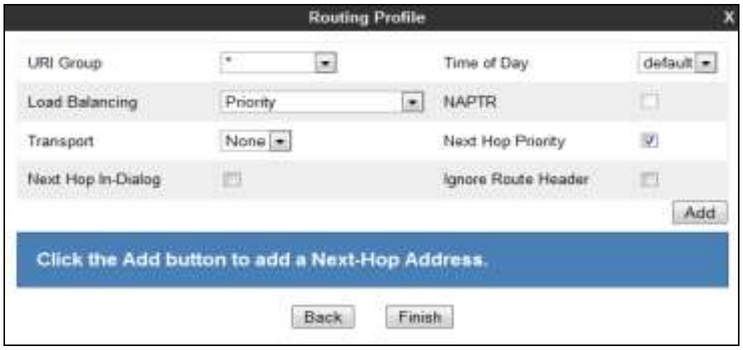
1. Select **Add Profile** and enter a Profile Name (e.g., **ATT_SC**) and select **Next**.
2. On the **General** window (not shown), enter the following.
    a. Select Server Type**: Trunk Server**
    b. **IP Address: 10.10.10.11** (AT&T Border Element IP address)
    c. **Supported Transports**: Check **UDP**
    d. **UDP Port: 5060**
    e. Select **Next**.
3. On the **Advanced** window (not shown), enter the following.
    d. Select **ATT_SI** (created in **Section 6.4.2**), for **Interworking Profile**.
    a. Select **Finish**.

6.4.5. Routing – to Avaya IP Office

This provisioning defines the Routing Profile for the connection to Avaya IP Office.

1. Select **Global Profiles → Routing** from the left-hand menu, and select **Add** (not shown).
2. Enter a **Profile Name**: (e.g., **IPO_RP)** and click **Next**.

3. The Routing Profile window will open. Using the default values shown, click on **Add.**

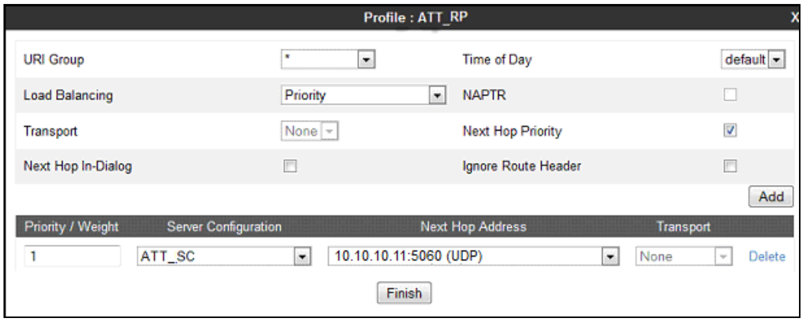


4. The Next-Hop Address window will open. Populate the following fields:
   a. **Priority/Weight** = **1**
   b. **Server Configuration** = **IPO_SC** (**from Section 6.4.3**).
   c. **Next Hop Address** = Select **192.168.42.10:5060 (UDP)** from the drop down menu (Avaya IP Office LAN1 IP address).
   d. Click on **Finish.**

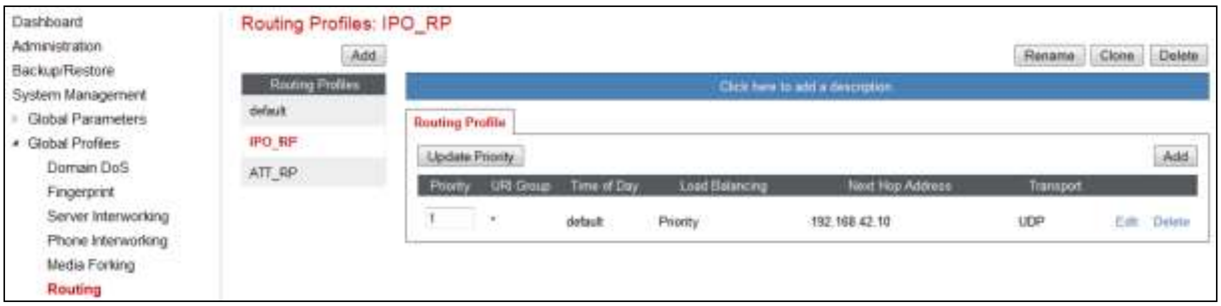
6.4.6. Routing – to AT&T

Repeat the steps in **Section 6.4.5**, with the following changes, to add a Routing Profile for the Avaya SBCE connection to AT&T.

1. On the **Global Profiles → Routing window (not shown),** enter a Profile Name: (e.g., **ATT_RP**).
2. On the Next-Hop Address window (not shown), populate the following fields:
   a. **Priority/Weight** = **1**
   b. **Server Configuration** = **ATT_SC** (**from Section 6.4.4**).
   c. **Next Hop Address** = Select **10.10.10.11:5060 (UDP)** from the drop down menu (Primary AT&T Border Element IP address).
   d. Click on **Finish.**



The following screen shows the completed Routing Profile.



6.4.7. Topology Hiding – Avaya IP Office

The **Topology Hiding** screen allows users to manage how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the security of the network. It hides the topology of the enterprise network from external networks.

1. Select **Global Profiles → Topology Hiding** from the left-hand side menu.
2. Select the **Add** button, enter Profile Name: (e.g., **Avaya_TH**), and click **Next**.

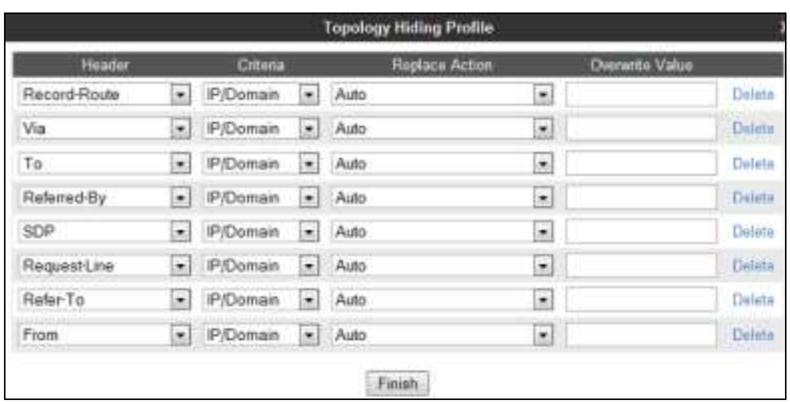
3. The **Topology Hiding Profile** window will open. Click on the **Add Header** button repeatedly until no new headers are added to the list, and the **Add Header** button is no longer displayed.

4. Populate the fields as shown below, and click **Finish**. Note that **customera.com** is the domain used by Avaya IP Office (see Section **5.3.2.2**).

6.4.8. Topology Hiding – AT&T

Repeat the steps in **Section 6.4.7,** with the following changes, to create a Topology Hiding Profile for the Avaya SBCE connection to AT&T.

1. Enter a Profile Name: (e.g., **ATT_TH**).
2. Use the default values for all fields and click **Finish**.



The following screen shows the completed **Topology Hiding Profile** form.

6.4.9. Signaling Manipulation

Signaling Manipulations are SigMa scripts the Avaya SBCE can use to manipulate SIP headers/messages**.** However, no Signaling Manipulations were used in the reference configuration.

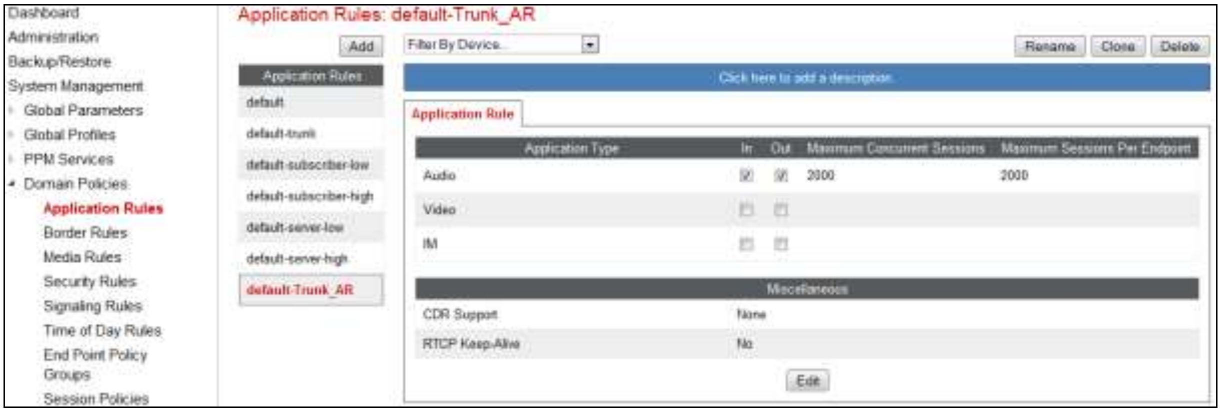
Note – The use of Signaling Manipulation scripts demands higher processing requirements for the Avaya SBCE. Therefore, the use of Signaling Rules (**Section 6.5.3**) is the preferred method for header/message manipulation. Signaling Manipulations should only be used in cases where the use of Signaling Rules does not meet the desired result. Refer to **[5]** for information on the Avaya SBCE scripting language.

6.5. Domain Policies

The Domain Policies feature allows users to configure, apply, and manage various rule sets (policies) to control unified communications based upon various criteria of communication sessions originating from or terminating in the enterprise.

6.5.1. Application Rules

1. Select **Domain Policies →Application Rules** from the left-hand side menu (not shown).
2. Select the **default-trunk** rule (not shown).
3. Select the **Clone** button (not shown), and the **Clone Rule** window will open.
    a. In the **Clone Name** field enter **default-Trunk_AR.**
    b. Click **Finish** (not shown). The completed **Application Rule** is shown below.

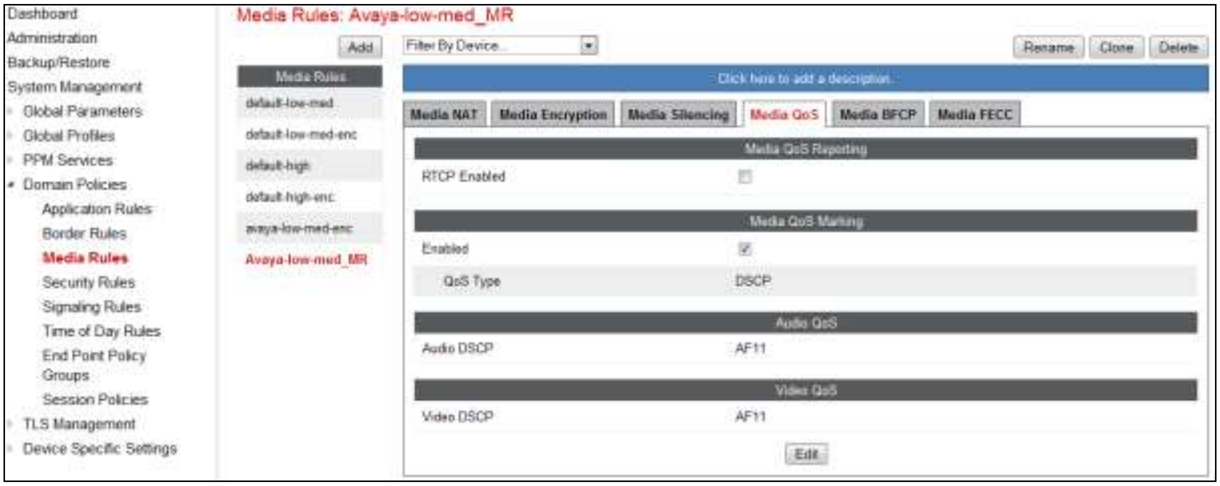


6.5.2. Media Rules

Media Rules are used to define QOS parameters. The Media Rule described below will be applied to both directions, and therefore, only one rule is needed.

1. Select **Domain Policies → Media Rules** from the left-hand side menu (not shown).
2. From the Media Rules menu, select the **default-low-med** rule.
3. Select **Clone** button (not shown), and the **Clone Rule** window will open.
   a. In the **Clone Name** field enter **Avaya-low-med_MR**
   b. Click **Finish.** The newly created rule will be displayed.
4. Highlight the **Avaya-low-med_MR** rule just created (not shown):
   a. Select the **Media QOS** tab (not shown).
   b. Click the **Edit** button and the **Media QOS** window will open.
   c. Check the **Media QOS Marking** field is **Enabled.**
   d. Select the **DSCP** box.
   e. **Audio**: Select **AF11** from the drop-down.
   f. **Video**: Select **AF11** from the drop-down.
5. Click **Finish.**

The completed **Media Rule** screen is shown below.



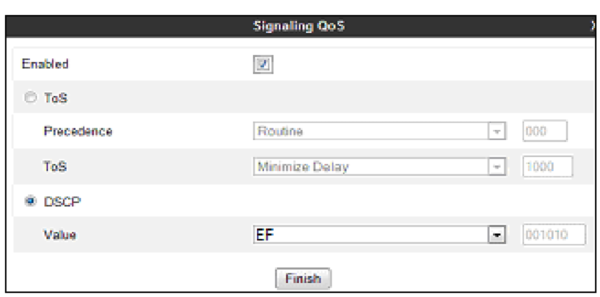
6.5.3. Signaling Rules

In the reference configuration, Signaling Rules are used to filter various SIP headers and set QOS parameters.

6.5.3.1 Avaya – Signaling Rule – QOS Tab

The Signaling Rule for Avaya IP Office, only modifies the Signaling QOS tab.

1. Select **Domain Policies → Signaling Rules** from the left-hand side menu (not shown).
2. The Signaling Rules window will open (not shown). From the Signaling Rules menu, select the **default** rule.
3. Select the **Clone** button and the **Clone Rule** window will open (not shown).
   a. In the **Rule Name** field enter **IPO_SR.**
   b. Click **Finish.** The newly created rule will be displayed.
4. Highlight the **IPO_SR** rule and enter the following:
   a. Select the **Signaling QOS** tab.
   b. Click the **Edit** button and the **Signaling QOS** window will open.
   c. Verify that **Signaling QOS** is selected.
   d. Select **DCSP**.
   e. Select **Value** = **EF**.
5. Click **Finish.**

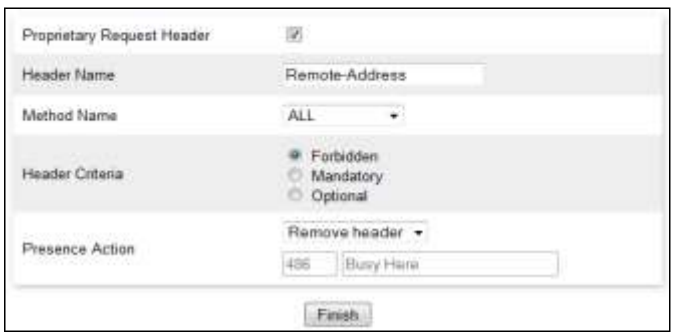


6.5.3.2 AT&T – Signaling Rules

The Signaling Rules for AT&T modify the Request Headers, Response Headers, and the Signaling QOS tabs. Repeat the steps in **Section 6.5.3.1** with the following changes:

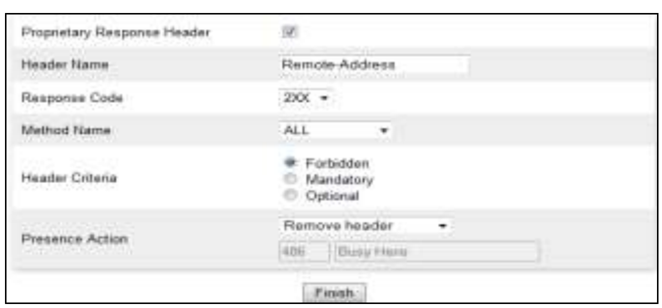
1. Select the **Clone** button and the **Clone Rule** window will open (not shown).
   - In the **Rule Name** field enter**: ATT_SR**, then click **Finish.**
2. Highlight the **ATT_SR** rule and select the **Request Headers** tab. Click the **Add Out Header Control** button.

3. Populate the **Add Header Control** form as shown below, and click **Finish**.



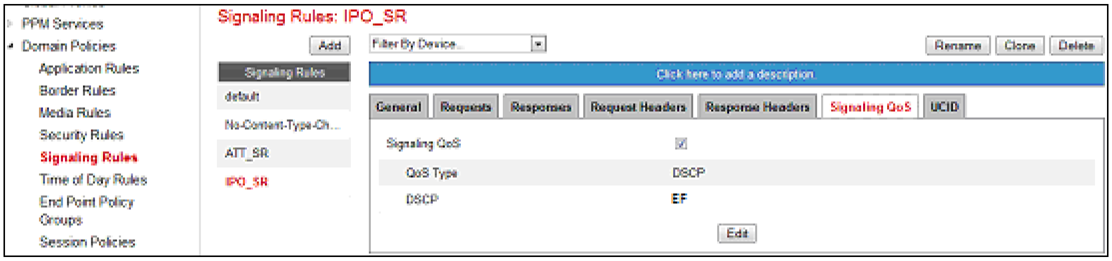
4. Highlight the **ATT_SR** rule and select the **Response Headers** tab. Click the **Add Out Header Control** button.

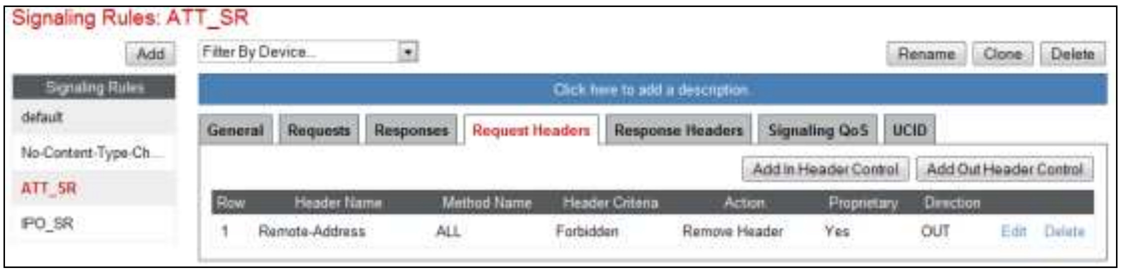
5. Populate the **Add Header Control** form as shown below, and click **Finish**.



6. Repeat steps **4** and **5** from **Section 6.5.3.1** to modify the **Signaling QOS** tab.

The completed Signaling Rule forms are shown below.

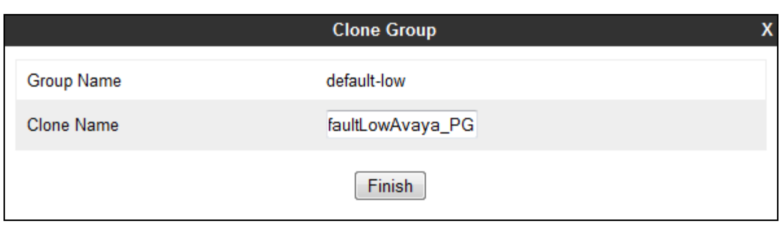




6.5.4. Endpoint Policy Groups – Avaya IP Office

Endpoint Policy Groups associate the other parameters defined under the Domain Policies section.

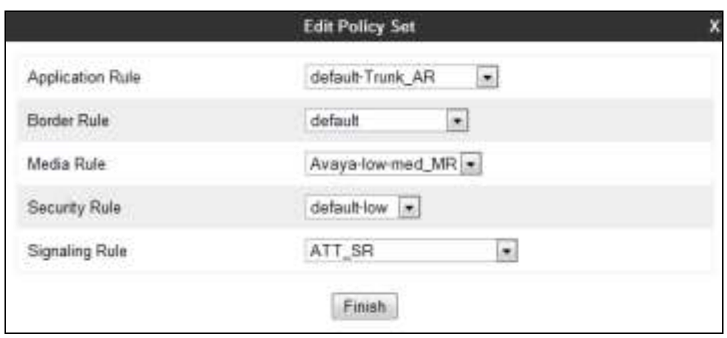
1. Select **Domain Policies → End Point Policy Groups** from the left-hand side menu.
2. Select the existing **default-low** policy and click the **Clone** button (not shown).
3. On the Clone Group window, enter a name for the group (e.g., **defaultLowAvaya_PG**), and click **Finish**.



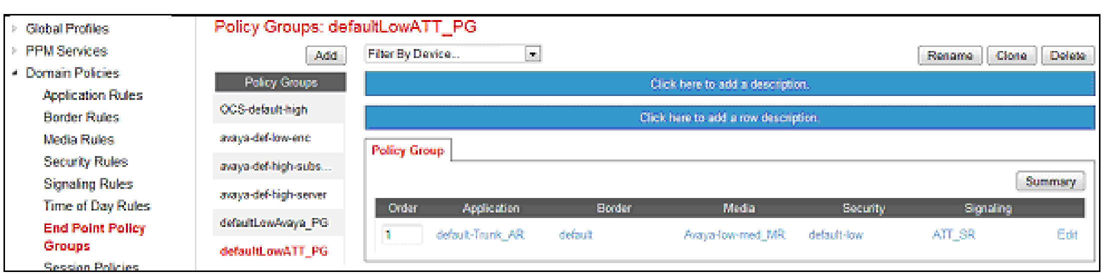
4. Select the **defaultLowAvaya_PG** policy, and click the **Edit** button (not shown). Populate the fields as follows:
    a) **Application Rule**: **default-Trunk_AR** (created in **Section 6.5.1**)
    b) **Border Rule**: **default**
    c) **Media Rule**: **Avaya-low-med_MR** (created in **Section 6.5.2**)
    d) **Security Rule**: **default-low**
    e) **Signaling Rule**: **IPO_SR** (created in **Section 6.5.3**)
5. Select **Finish.**

6.5.5. Endpoint Policy Groups – AT&T

1. Repeat the steps in **Section 6.5.4** with the following changes:
    a. **Group Name**: **defaultLowATT_PG**
    b. **Signaling Rule**: **ATT_SR** (created in **Section 6.5.3)**
2. Select **Finish**.



The completed End Point Policy Group form is shown below.



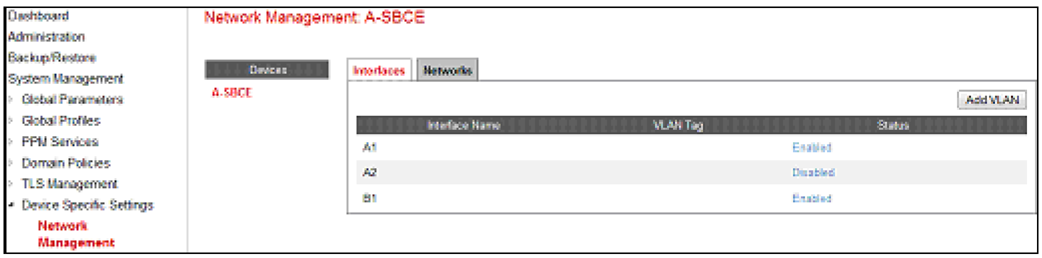
6.6. Device Specific Settings

The **Device Specific Settings** feature for SIP allows users to view system information, and manage various device-specific network parameters such as port ranges.

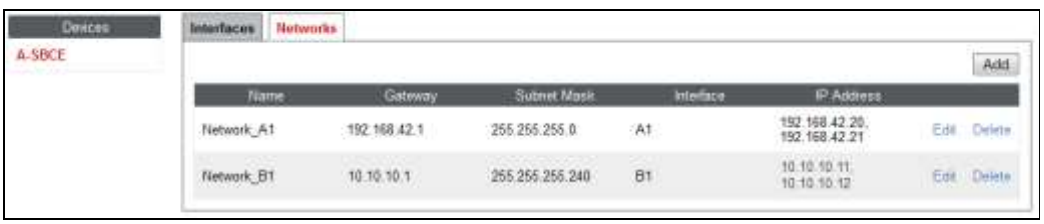
6.6.1. Network Management

1. Select **Device Specific Settings → Network Management** from the left-hand side menu.
2. The **Interfaces** tab shows the state of the physical interfaces, and allows the physical interfaces to be enabled/disabled by clicking on the **Status** value.

Note – The Avaya SBCE Portwell CAD-0208 platform supports the A1, A2, and B1 interfaces only.



3. The **Networks** tab shows the IP configuration of the interfaces.



6.6.2. Advanced Options

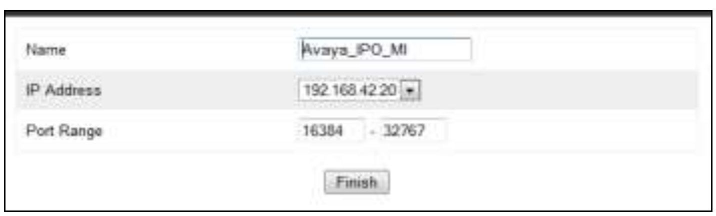
In **Section 6.5.3**, the media UDP port ranges required by AT&T are set (**16384 – 32767**). By default, part of this range is already allocated by the Avaya SBCE for internal use (22000 - 31000). The following steps reallocate the port ranges used by the Avaya SBCE so the range required by AT&T can be used.

1. Select **Device Specific Settings → Advanced Options** from the menu on the left-hand side.
2. Select the **Port Ranges** tab.
3. In the **Config Proxy Internal Signaling Port Range** row, change the range to **50001 – 51000**.
4. Scroll to the bottom of the window and select **Save**.

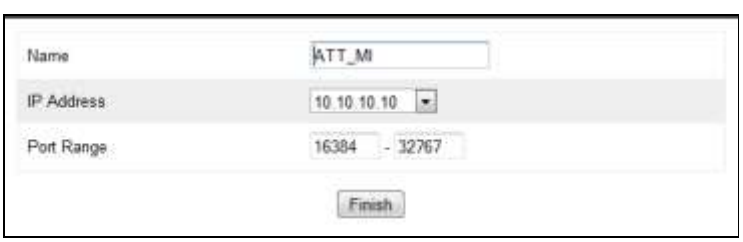
6.6.3. Media Interfaces

The AT&T IPFR-EF service specifies that customers use RTP ports in the range of **16384** – **32767**. Both inside and outside ports have been changed to this range, but only the outside is required by the AT&T IPFR-EF service.

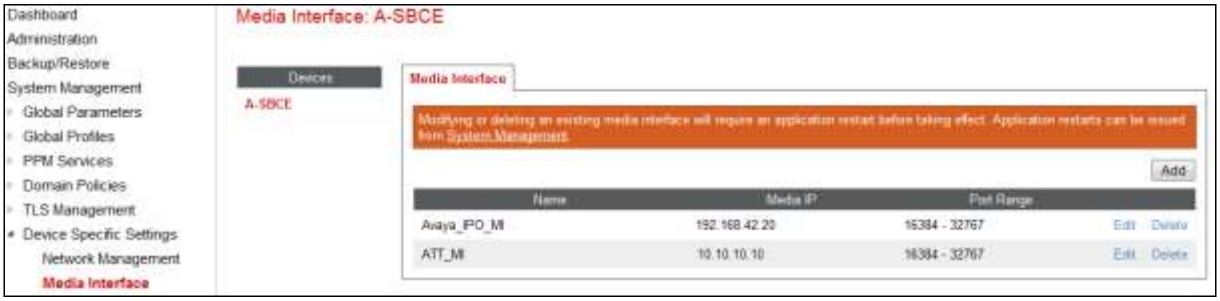
1. Select **Device Specific Settings → Media Interface** from the left-hand menu (not shown).
2. Select **Add** (not shown). The **Add Media Interface** window will open. Enter the following:
   a) **Name**: **Avaya_IPO_MI**
   b) **IP Address**: **192.168.42.20** (Avaya SBCE A1 address to Avaya IP Office)
   c) **Port Range**: **16384 - 32767**
3. Click **Finish.**



4. Select **Add** (not shown). The **Add Media Interface** window will open. Enter the following:
   a) **Name**: **ATT_MI**
   b) **IP Address**: **10.10.10.10** (Avaya SBCE B1 address toward AT&T)
   c) **Port Range**: **16384 - 32767**
5. Click **Finish**.



The completed **Media Interface** screen is shown below.

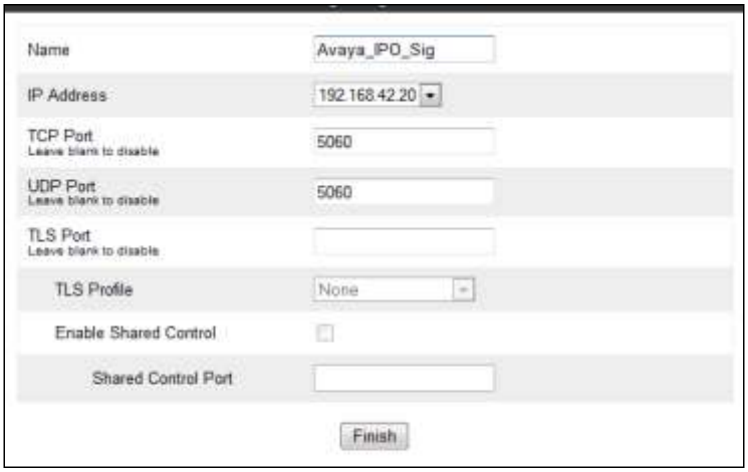


6.6.4. Signaling Interface

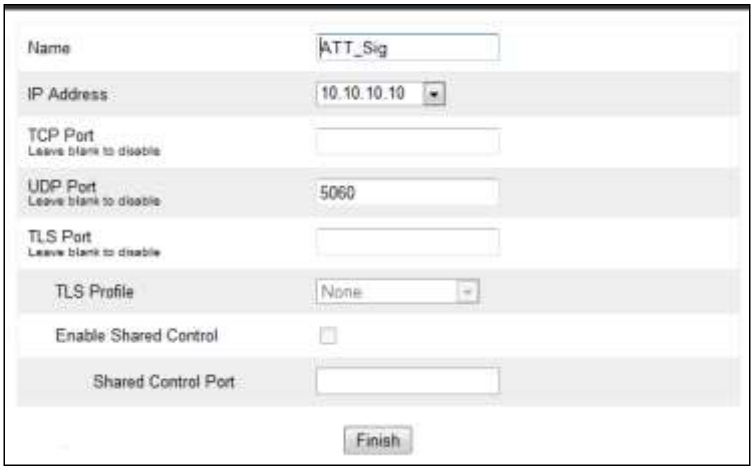
1. Select **Device Specific Settings → Signaling Interface** from the left-hand menu (not shown).
2. Select **Add** (not shown). The **Add Signaling Interface** window will open. Enter the following:
   a) **Name**: **Avaya_IPO_Sig**
   b) **IP Address**: **192.168.42.20** (Avaya SBCE A1 address to Avaya IP Office)
   c) **UDP Port**: **5060**

> **Note** – UDP is the recommended protocol to use on the connection between the Avaya SBCE and Avaya IP Office. However TCP may be used if desired (see **Sections 5.4.4** and **6.4.3**).

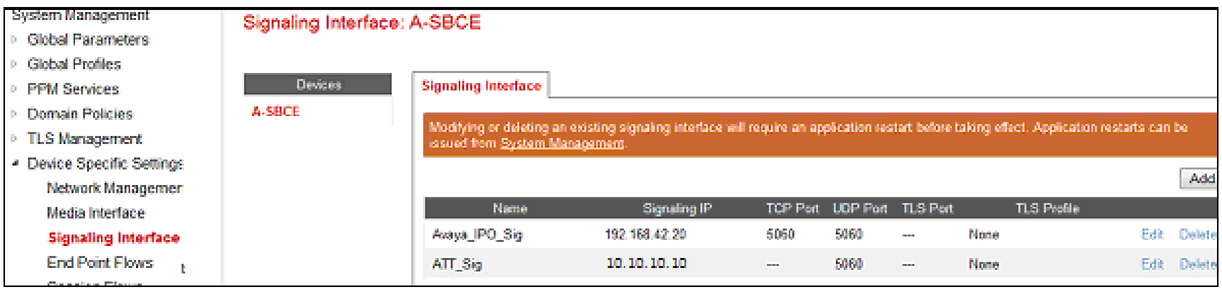
3. Click **Finish**.



4. Select **Add** again, and enter the following:
   a) **Name**: **ATT_Sig**
   d) **IP Address**: **10.10.10.10** (Avaya SBCE B1 address toward AT&T)
   b) **UDP Port**: **5060**
5. Click **Finish** (not shown).



The completed **Signaling Interface** screen is shown below.

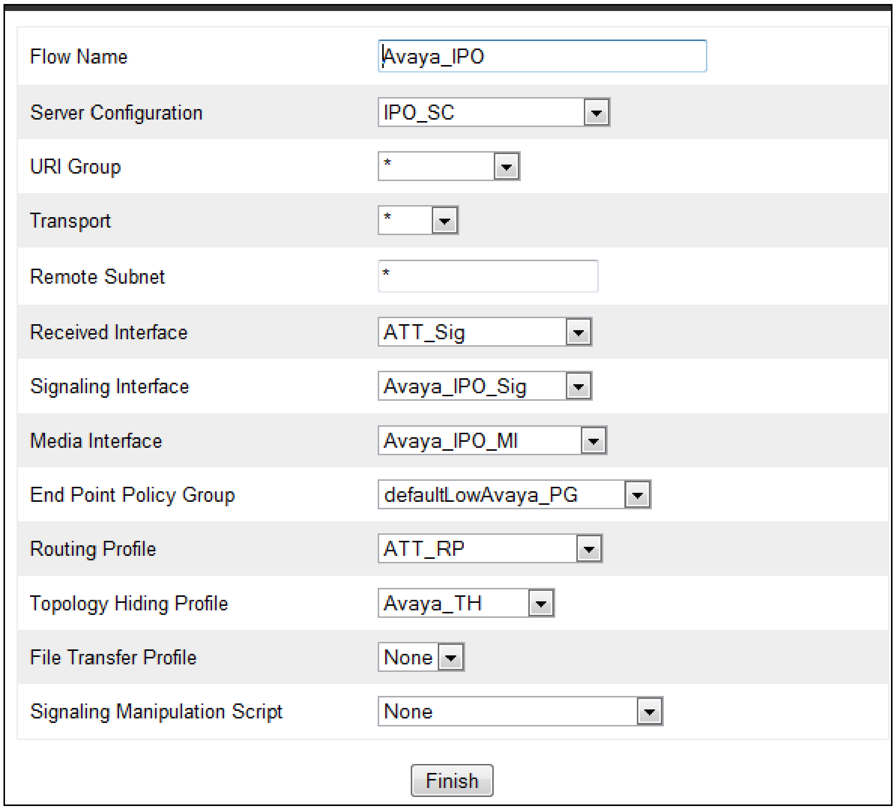


6.6.5. Endpoint Flows – Avaya IP Office

Endpoint flows determine the path to be followed by the packets passing through the Avaya SBCE.

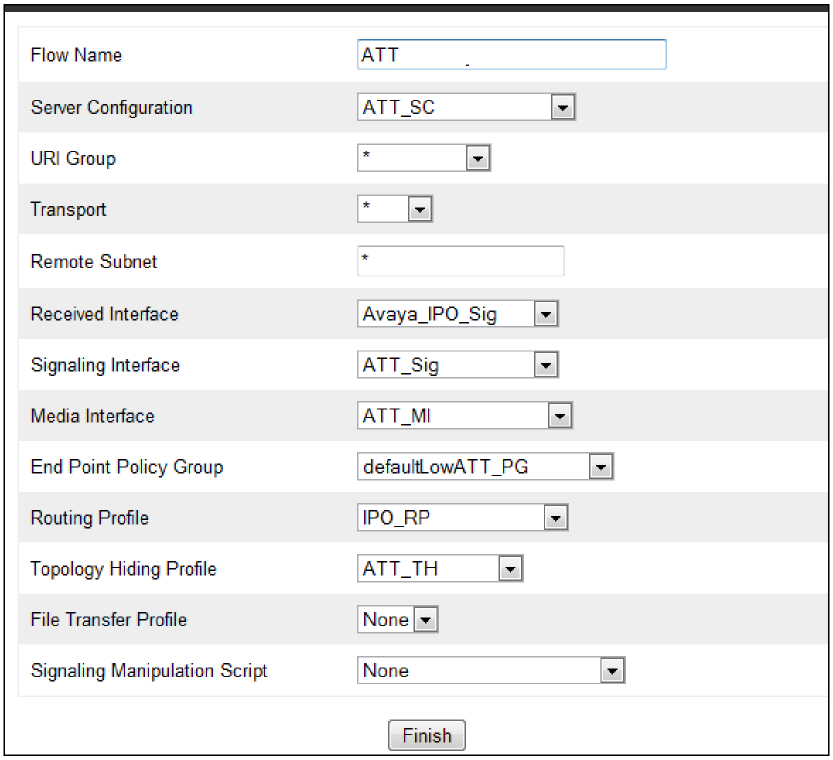
1. Select **Device Specific Settings → Endpoint Flows** from the left-hand menu (not shown).
2. Select the **Server Flows** tab (not shown).
3. Select **Add**, (not shown) and enter the following:
    a) **Name**: **Avaya_IPO**
    b) **Server Configuration**: **IPO_SC** (**Section 6.4.3**
    c) **URI Group**: *****
    d) **Transport**: *****
    e) **Remote Subnet**: *****
    f) **Received Interface**: **ATT_Sig**
    g) **Signaling Interface**: **Avaya_IPO_Sig** (**Section 6.6.4**)
    h) **Media Interface**: **Avaya_IPO_MI** (**Section 6.6.3**)
    i) **End Point Policy Group**: **defaultLowAvaya_PG** (**Section 6.5.4**)
    j) **Routing Profile**: **ATT_RP** (**Section 6.4.6**)
    k) **Topology Hiding Profile**: **Avaya_TH** (**Section 6.4.7**)
    l) **File Transfer Profile**: **None**
    m) **Signaling Manipulation Script: None**

4. Click **Finish**.

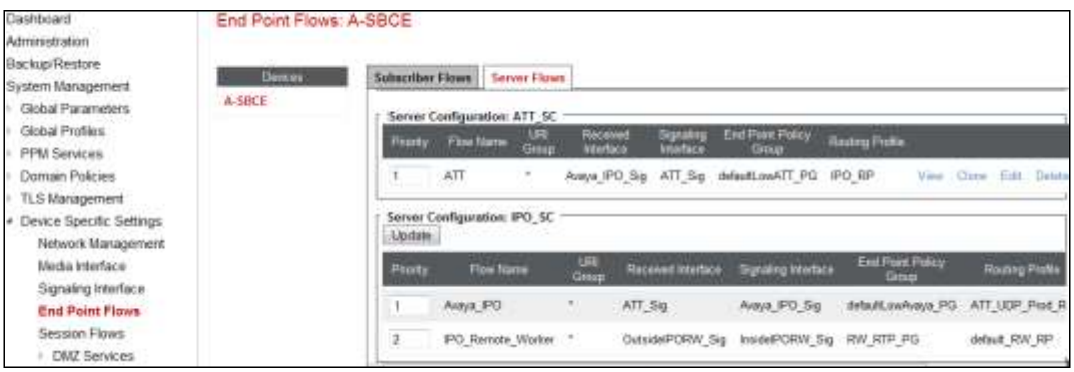


6.6.6. Endpoint Flows – AT&T

1. Repeat steps **1** through **4** from **Section 6.6.5**, with the following changes:
   a) **Name**: **ATT**
   b) **Server Configuration**: **ATT_SC** (**Section 6.4.4**).
   c) **URI Group**: *
   d) **Transport**: *
   e) **Remote Subnet**: *
   f) **Received Interface**: **Avaya_IPO_Sig** (**Section 6.6.4**).
   g) **Signaling Interface**: **ATT_Sig (Section 6.6.4**).
   h) **Media Interface**: **ATT_MI** (**Section 6.6.3**).
   i) **End Point Policy Group**: **defaultLowATT_PG** (**Section 6.5.5**).
   j) **Routing Profile**: **IPO_RP (Section 6.4.5**).
   k) **Topology Hiding Profile**: **ATT_TH** (**Section 6.4.8**).
   l) **File Transfer Profile**: **None**
   m) **Signaling Manipulation Script: None**
2. Click **Finish**.



The completed **End Point Flows** screen is shown below.



7. AT&T IP Toll Free Service Configuration

AT&T provides the IPTF service border element IP address, the access DID numbers, and the associated DNIS digits used in the reference configuration. In addition the AT&T IPTF features, and their associated access numbers, are also assigned by AT&T. AT&T requires that the Avaya SBCE public (B1) IP address be provided to the IPTF service, as part of the provisioning process.

8. Verification Steps

The following procedures may be used to verify the Avaya IP Office R9.1, Avaya SBCE 6.3, with the AT&T IP Toll Free service, configuration.

8.1. AT&T IP Toll Free Service

The following scenarios may be executed to verify Avaya IP Office R9.1 functionality with the AT&T IPTF service:

- Place inbound calls, answer the calls, and verify that two-way talk path exists. Verify that the calls remain stable for several minutes and disconnects properly.
- Incoming calls using the G.729A and G.711 ULAW codecs.
- Verify basic call functions such as hold, transfer, and conference.
- Place an inbound call to a telephone, but do not answer the call. Verify that the call covers to voicemail (e.g., Avaya Voicemail Pro). Retrieve the message either locally or from PSTN.
- Using the appropriate IPTF access numbers and codes, verify the "Legacy Transfer Connect" DTMF initiated features.
- Inbound fax using T38 or G.711.
- SIP OPTIONS monitoring of the health of the SIP trunk.

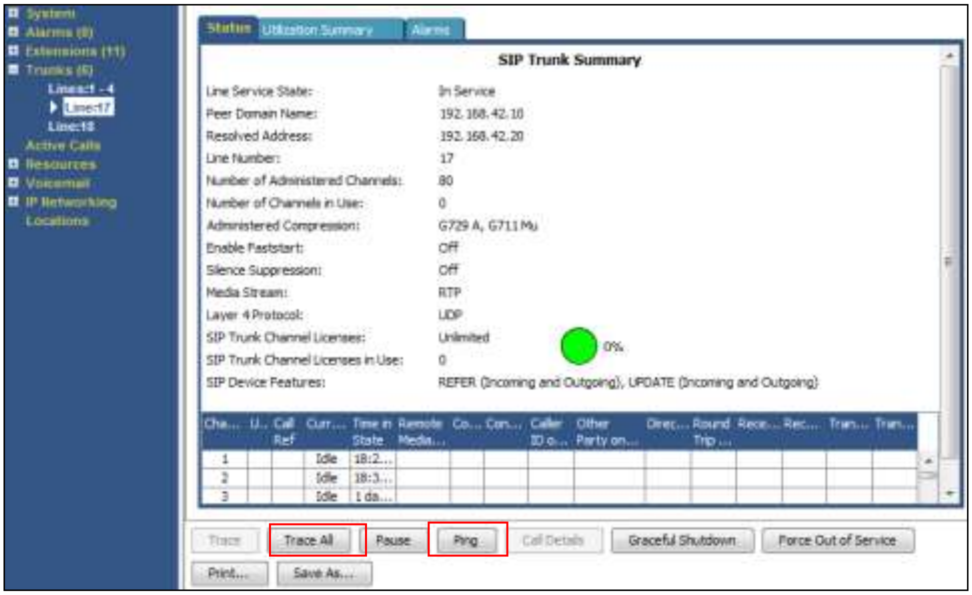
8.2. Avaya IP Office 9.1

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

8.2.1. 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**.

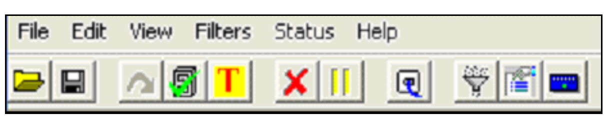
After logging in, select **Trunks → Line: 17** from the left navigation menu. (SIP Line 17 is configured in **Section 5.4**). A screen such as the one shown below is displayed. In the lower left, the **Trace All** button may be pressed to display tracing information as calls are made using this SIP Line. The **Ping** button can be used to ping the other end of the SIP trunk (e.g., the Avaya SBCE).

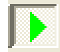


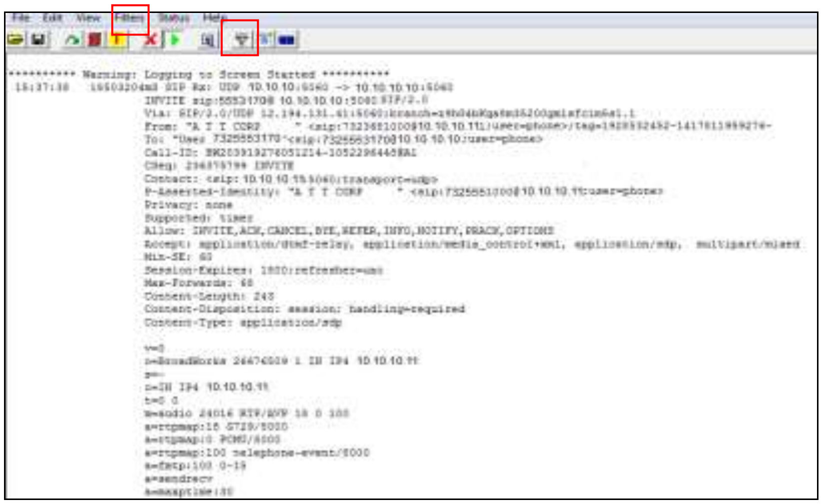
By navigating to **IP Networking → IP Routes**, and clicking on **Ping**, an IP Office **Source Interface**, and any **Destination IP Address**, may be specified for a ping buy clicking **OK**.

8.2.2. System Monitor Application

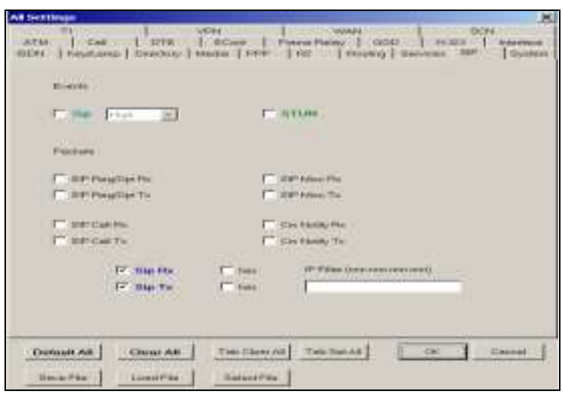
The System Monitor application can also be used to monitor or troubleshoot Avaya IP Office functionality (see reference **[1]**. The System Monitor application can typically be accessed from **Start → Programs → Avaya IP Office → Monitor**.



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 is a sample of a monitored inbound call to Avaya IP Office SIP telephone 500.



The displayed data may be customized. Select the **Options** button , or select **Filters** → **Trace Options**. The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, only the **SIP Rx** and **SIP Tx** boxes are selected.



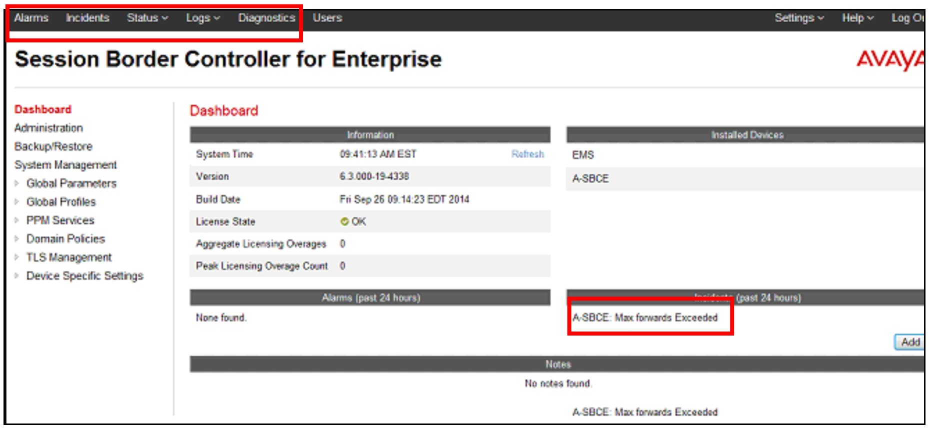
8.3. Avaya Session Border Controller for Enterprise 6.3

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

8.3.1. System Status

Various system conditions monitored by the Avaya SBCE may be displayed as follows.

Step 1 – Log into the Avaya SBCE as shown in **Section 6**. Across the top of the display are options to display **Alarms**, **Incidents**, **Logs**, and **Diagnostics**. In addition, the most recent Incidents are listed in the lower right of the screen.



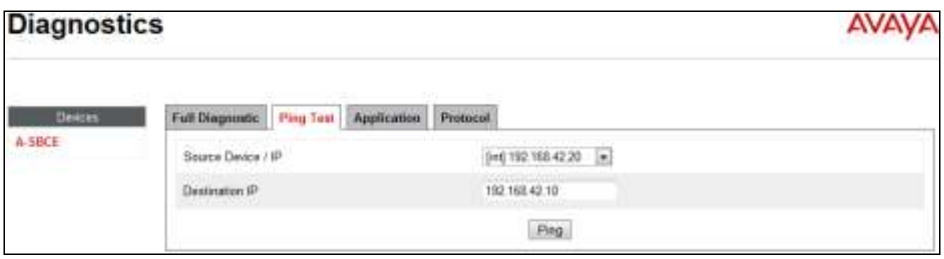
8.3.2. Ping Test

The Avaya SBCE can very network connectivity by issuing a ping.

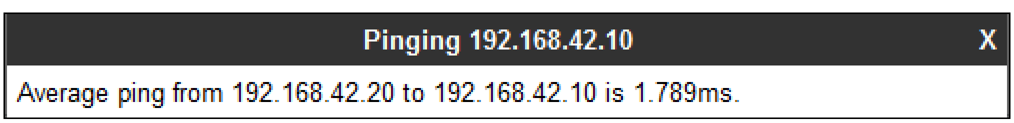
Step 1 - Select **Diagnostics** from the selections shown above. The diagnostics window will open.

Step 2 – Select the **Ping** tab.

- **Source Device / IP**: From the drop-down menu, select the Avaya SBCE interface to issue the ping.
- **Destination IP**: Enter the IP address to be pinged (e.g., Avaya IP Office LAN1).
- Click on the **Ping** button.



When the ping attempts are completed, the results are displayed:



8.3.3. Protocol Traces

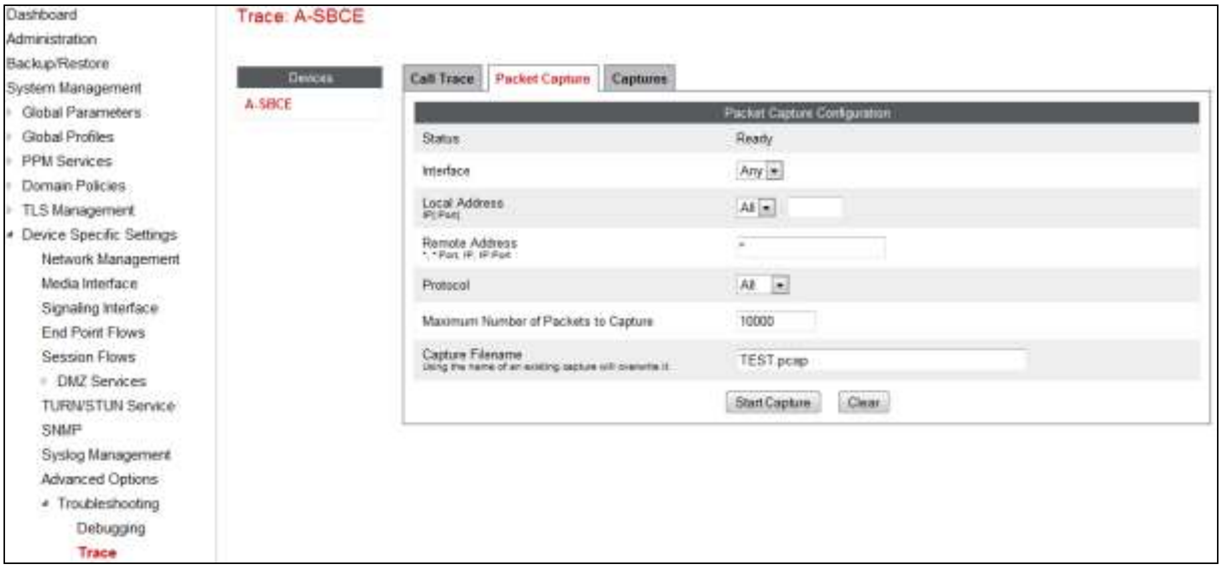
The Avaya SBCE can take internal traces of specified interfaces.

Step 1 - Navigate to **Device Specific Settings → Troubleshooting → Trace**

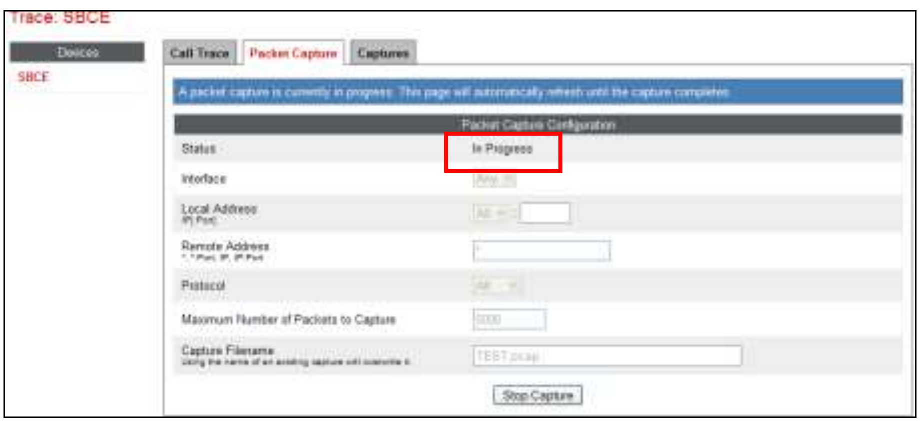
Step 2 - Select the **Packet Capture** tab and enter the following:

- **Interface**: Select the desired interface where the trace will be run from the drop down menu. Selecting **Any** will result in a trace showing activity on both the A1 (inside) and B1 (outside) interfaces.

- **Local Address**: Select **All**.
- **Remote Address**: Select *****.
- **Protocol**: Select **All**.
- **Maximum Number of Packets to Capture**: Specify the number of packets to capture (e.g., **10000**). Note that the number specified should be a best guess based on the duration of the test.
- **Capture Filename**: Enter a name for the trace file. Note that the trace will be saved in Wireshark *.pcap* format.
- Click **Start Capture** to begin the trace.



The capture process will initialize and then display the following status window. Note that the **Status** will change to **In Progress** when the trace begins, and the screen will begin to refresh.

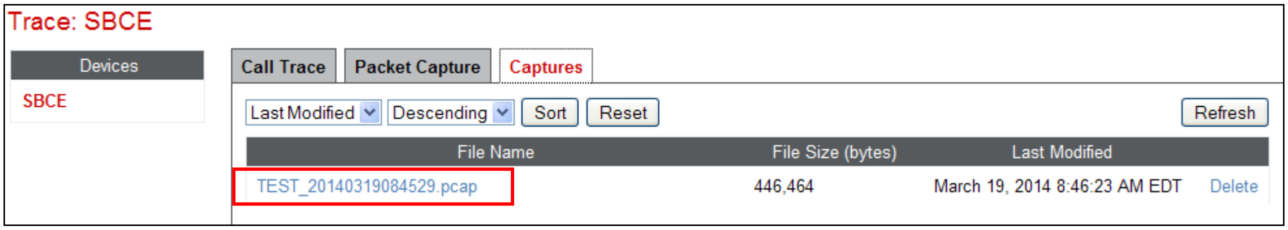


Step 3 – Run the test.

Step 4 – At the conclusion of the test. Select the **Stop Capture** button shown above.

Step 5 - Click on the **Captures** tab and the packet capture is listed as a *.pcap* file with the date and time added to filename specified in **Step 2**.

Step 6 - Click on the **File Name** link to download the file and use Wireshark to open the trace.



9. Conclusion

As illustrated in these Application Notes, Avaya IP Office R9.1 and the Avaya Session Border Controller for Enterprise 6.3 can be configured to interoperate successfully with the AT&T IP Toll Free service using **AVPN** or **MIS/PNT** transport connections, utilizing service features listed in **Section 2.1,** and within the limitations described in **Section 2.2**.

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.

10. References

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

[1] Administering Avaya IP Office™ Platform with Manager, Release 9.1.0, 10.0.2, January 2015

[2] Administering Avaya IP Office™ Platform Voicemail Pro, 15-601063 Issue 10d - (25 March 2015)

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

[4] Deploying Avaya Session Border Controller for Enterprise, Release 6.3, Issue 4, October 2014

[5] Administering Avaya Session Border Controller for Enterprise, Release 6.3, Issue 4, October 2014

AT&T IPTF Service:

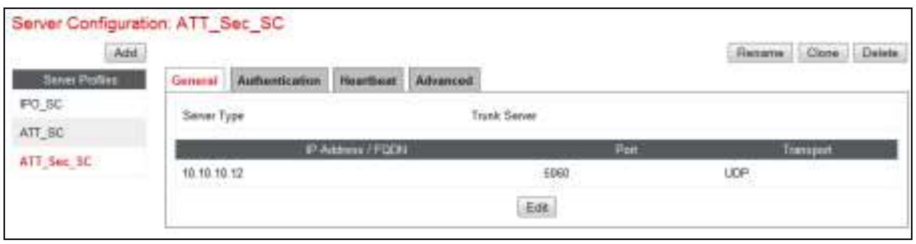
[6] AT&T IP Toll Free Service description - http://www.business.att.com/enterprise/Service/voice-services/contact-center-solutions/ip-toll-free/

11. Addendum 1 – Multiple AT&T Border Elements

AT&T may provide two network border elements for redundancy purposes. The Avaya SBCE can be provisioned to support this redundant configuration.

Given two AT&T border elements **10.10.10.11** (Primary) and **10.10.10.12** (Secondary), the Avaya SBCE is provisioned as follows to include the backup trunk connection.
Step 1 – Create a secondary AT&T Server Configuration.

1. Select **Global Profiles → Server Configuration** from the left-hand menu.
2. Select **Add Profile**
   - **Name: ATT_Sec_SC**
3. On the **Add Server Configuration Profile – General** tab:
   - Select **Server Type: Trunk Server**
   - **IP Address: 10.10.10.12** (Address for a secondary location)
   - **Supported Transports**: Check **UDP**
   - **UDP Port: 5060**
   - Select **Next** (not shown)



4. On the **Authentication** tab:
   - Select **Next** to accept defaults (not shown).
5. On the **Heartbeat** tab:
   - Check **Enable Heartbeat**
   - **Method: OPTIONS**
   - **Frequency:** Set an appropriate interval (e.g., **60** seconds)
   - **From URI: secondary@customera.com**
   - **To URI: secondary@customera.com**
   - Select **Next**.

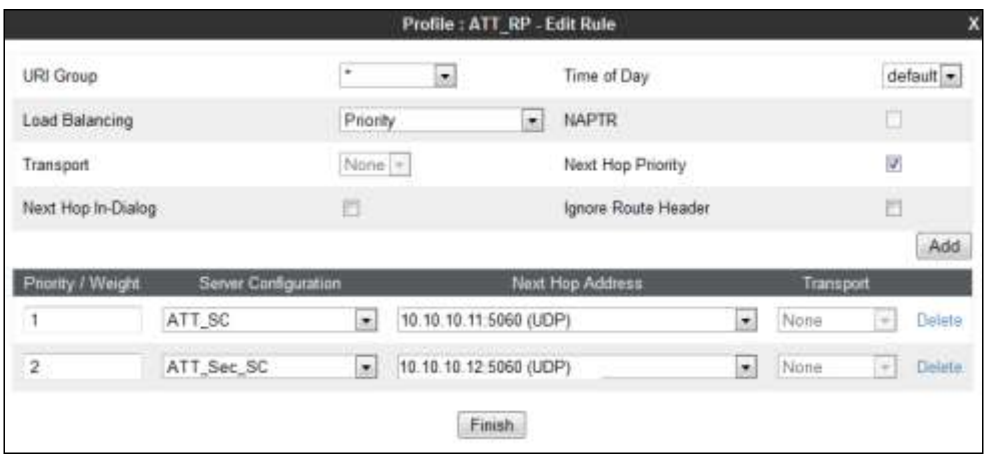
6. On the **Advanced** Tab, click **Finish** to accept defaults (not shown).

Step 2 – Add Heartbeat to the AT&T Primary Server Configuration.

1. Select the **Server Configuration** created in **Section 6.4.4** (e.g., **ATT_SC**)
2. Select the **Heartbeat Tab**
3. Select **Edit**
4. Repeat **Step 5**, but with information for the Primary Trunk as shown below.

Step 3 - Add Secondary IP Address to Routing.

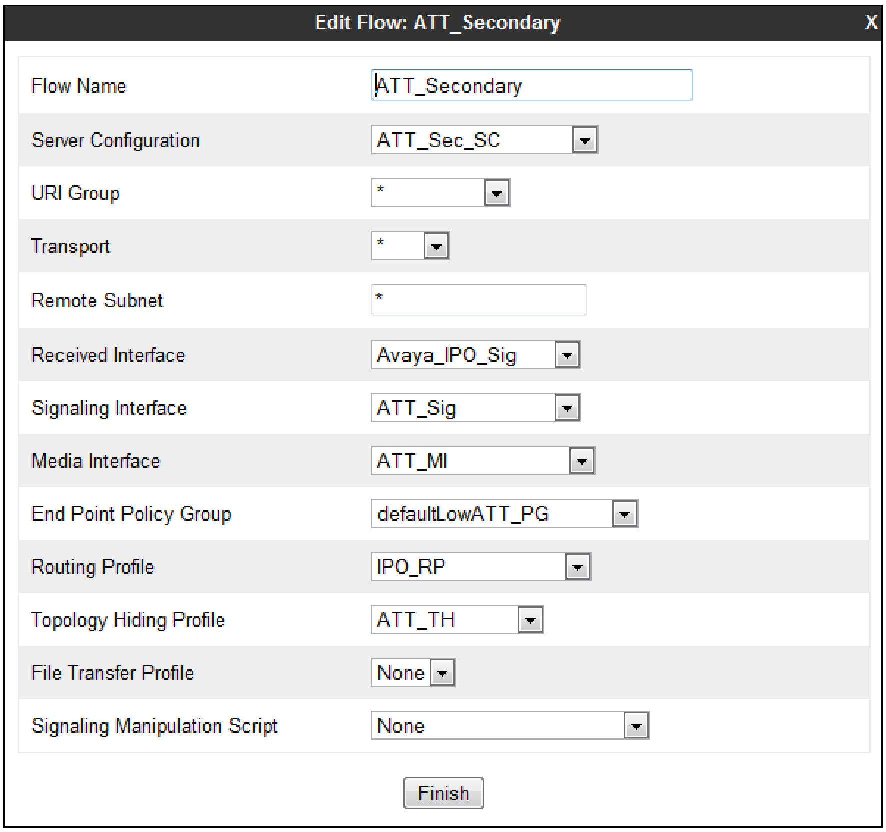
1. Select **Global Profiles → Routing** from the left-hand menu.
2. Select the **Routing Profile** created in **Section 6.4.6** (e.g., **ATT_RP**)
3. Click **Edit** (not shown), and enter the following:
   a) **Priority/Weight** = **2**
   b) **Server Configuration** = **ATT_Sec_SC**
   c) **Next Hop Address** = Select **10.10.10.12:5060 (UDP)** from the drop down menu.
   d) Click on **Finish.**



3. Configure End Point Flow for the AT&T Secondary Border Element.
   1. Select **Device Specific Settings** from the menu on the left-hand side.
   2. Select **Endpoint Flows.**
   3. Select the **Server Flows** Tab.
   4. Select **Add Flow.**
      a) **Name: ATT_Secondary**
      b) **Server Configuration: ATT_Sec_SC**

c) **URI Group: ***
d) **Transport: ***
e) **Remote Subnet: ***
f) **Received Interface: Avaya_IPO_Sig**
g) **Signaling Interface: ATT_Sig**
h) **Media Interface: ATT_MI**
i) **End Point Policy Group**: **defaultLowATT_PG**
j) **Routing Profile: IPO_RP**
k) **Topology Hiding Profile: ATT_TH**
l) **File Transfer Profile: None**

5. Click **Finish.**



When completed the Avaya SBCE will issue OPTIONS messages to the primary (10.10.10.11) and secondary (10.10.10.12) border elements. If the SBCE fails to get a response to the OPTIONS sent to 10.10.10.11, the SBCE will direct outbound calls to 10.10.10.12.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect program at devconnect@avaya.com.