Avaya Solution & Interoperability Test Lab

Application Notes for Avaya IP Office Release 8.1 and Avaya Session Border Controller for Enterprise 6.2, with AT&T IP Toll Free Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office R8.1 and the Avaya Session Border Controller for Enterprise 6.2, with the AT&T IP Toll Free service using AVPN or MIS/PNT transport connections.

The Avaya Session Border Controller for Enterprise is the point of connection between Avaya 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.

Avaya IP Office 8.1 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 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.

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

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1. Introduction

These Application Notes describe the steps for configuring Avaya IP Office R8.1 (Avaya IP Office) and the Avaya Session Border Controller for Enterprise 6.2, (Avaya SBCE), 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.

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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 AT&T IP Toll Free service will be referred to as IPTF in the remainder of this document.

2. General Test Approach and Test Results

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

- A simulated enterprise with Avaya IP Office, Avaya IP Office telephones and fax machines (Ventafax application), and the Avaya SBCE.
- Laboratory versions of the IPTF service, to which the simulated enterprise was connected via AVPN/MIS transport.

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

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

2.1. Interoperability Compliance Testing

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

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

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

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

- Incoming calls from PSTN, routed by the IPTF service, via the Avaya SBCE, to Avaya IP Office. These calls are via the Avaya IP Office SIP Line and may be generated/answered by Avaya SIP telephones/Softphones, H.323 telephones, Analog telephones, Analog fax machines or via Hunt Groups. Coverage to Avaya IP Office Voicemail Pro, and Voicemail Pro auto-attendant applications, may also be used.
- Inbound 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.729 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

Interoperability testing of the sample configuration and features described in **Section 2.1** were completed successfully. The following observations were noted during testing:

2.2.1. Known Limitations

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

2. **G.726 codec support** - G.726 codec is not supported by IP Office.

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

4. **Avaya IP Office uses fixed RFC2833 Telephone Event type 101 in SIP requests –** Avaya IP Office uses a fixed RFC2833 Telephone Event type of 101, and the AT&T network responses do comply with a value of 101. However, AT&T network SIP requests specify Telephone Event type 100, and Avaya IP Office complies with a value of 100.
    - No issues were found during testing as a result of this behavior.

5. **IP Trunk shuffling is not supported with Avaya IP Office 8.1 using the 500v2 platform**.
    - IP Trunk shuffling *is* supported with Avaya IP Office 8.1 using the server platform.

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 reference configuration are listed in **Section 10**. Specific references to these documents are indicated in the following sections by the notation **reference [x]**, where x is the document reference number.

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

- Avaya IP Office provides the voice communications services for a particular enterprise site. In the reference configuration, Avaya IP Office runs on an IP 500 V2 platform. This solution is extensible to Avaya IP Office platforms as well (see **Item 5** in **Section 2.2.1**).
- Avaya "desk" telephones are represented with an Avaya 1608 H.323 set, an Avaya 6211 Analog set, an Avaya 1120E SIP set, and PC based Avaya IP Office SIP Softphone (in Default Mode). Fax endpoints are represented by PCs running Ventafax software connected by modem to an Avaya IP Office analog port.

- Avaya IP Office Voicemail Pro (running on a Windows 2003 server) provided the voice messaging capabilities in the reference configuration. This solution is extensible to the Avaya IP Office embedded voice mail as well.
- Avaya Session Border Controller for Enterprise running on a CAD-0208 platform. This solution is extensible to other Avaya Session Border Controller for Enterprise platforms as well.
- UDP/5060 is the recommended transport protocol/port to use on the IP Office LAN1 connection. However TCP/5060 may be used if necessary.
- Inbound calls utilize telephone or fax User/Extensions provisioned on Avaya IP Office. Signaling is sent between Avaya IP Office and the IPTF service via the Avaya SBCE.
- The AT&T IPTF service requires the following SIP trunk network settings to the IPTF Border Element:
    - UDP transport using port 5060
    - RTP port ranges 16384-32767
- The AT&T IP Toll Free service provided the inbound access numbers (DID and DNIS) used in the reference configuration.

The Avaya IP Office 500V2 platform and the Avaya SBCE CAD-0208 platform used in the reference configuration deployed using the following configuration (referred to as an IP Office **"one-wire"** configuration).

- Avaya IP Office LAN1 interface (labeled "LAN") connected to the CPE private network.
- Avaya SBCE A1 interface connected to the CPE private network.
- Avaya SBCE B1 interface connected to the AT&T IP Toll Free service network router.

Note – In the reference configuration, the IP Office LAN2 interface, and the Avaya SBCE E3 interface (CAD-0208 platform), are both connected to a separate "management" subnet.

Note – The IP Office "one-wire" configuration described in this document is the preferred configuration for the IP Office/Avaya SBCE solution. However an IP Office "two-wire" configuration is also supported.

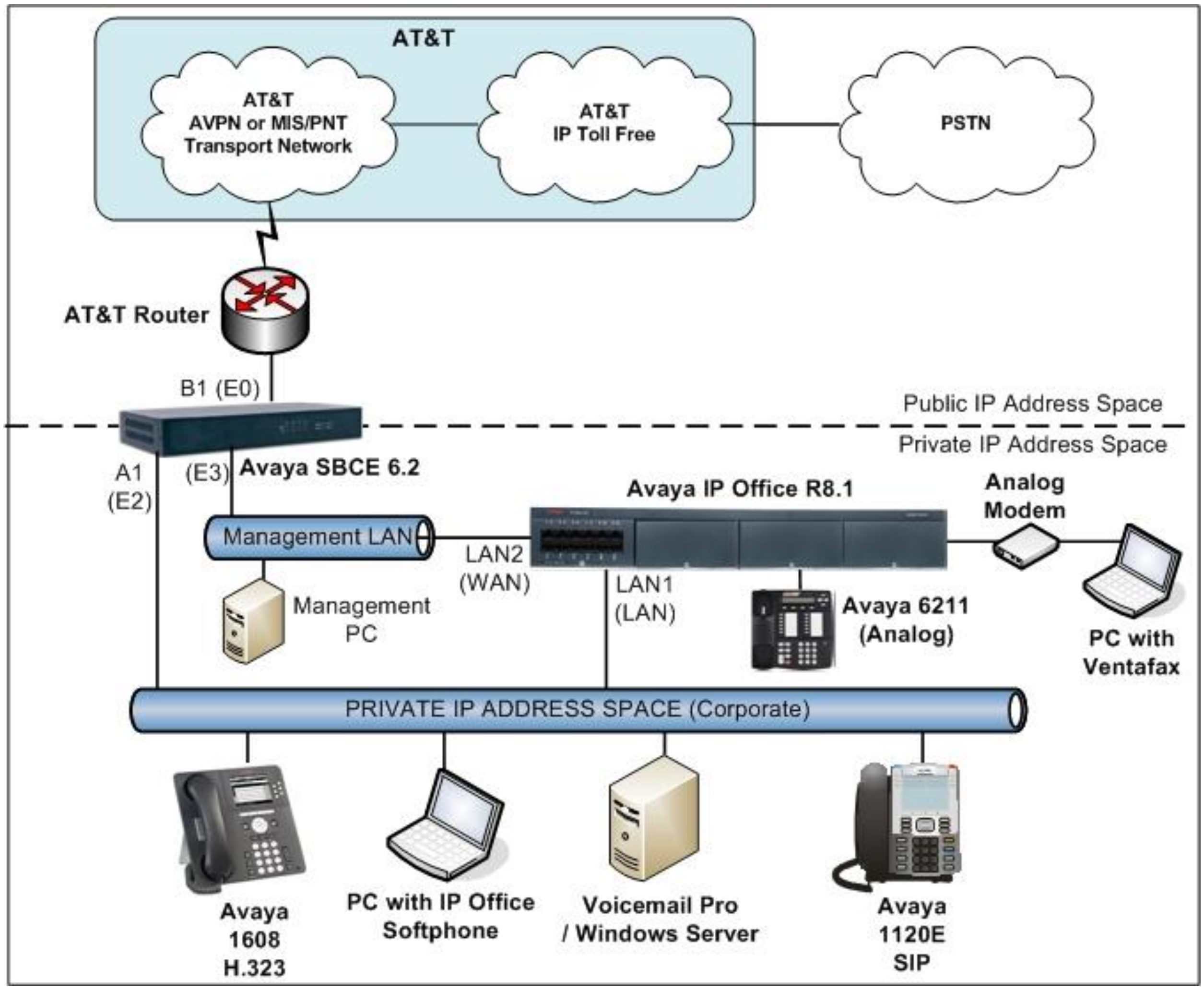


Figure 1: Reference Configuration

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 AT&T IPTF IP addressing shown in this document is an example. AT&T Customer Care will provide the actual IP addressing as part of the IPTF provisioning process.

Component	Illustrative Value in these Application Notes
Avaya IP Office 500 V2 Platform	
Private IP Address (LAN1 interface, labeled "LAN" on the chassis, see **Section 5.1**)	192.168.42.1
LAN2 interface, (labeled "WAN" on the chassis), for management access.	192.168.1.22
Avaya SBCE CAD-0208 Platform	
Private IP Address (A1 interface).	192.168.42.20
Public IP Address (B1 interface).	192.168.64.130
Management IP address (interface labeled "E3").	192.168.1.20
AT&T IPTF Service	
Border Element IP Address	135.25.29.74
AT&T Access router interface	192.168.64.254

Table 1: Illustrative Values Used in these Application Notes

3.2. Call Flows

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

3.2.1. Inbound

The first call scenario illustrated in the figure below is an inbound AT&T 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.

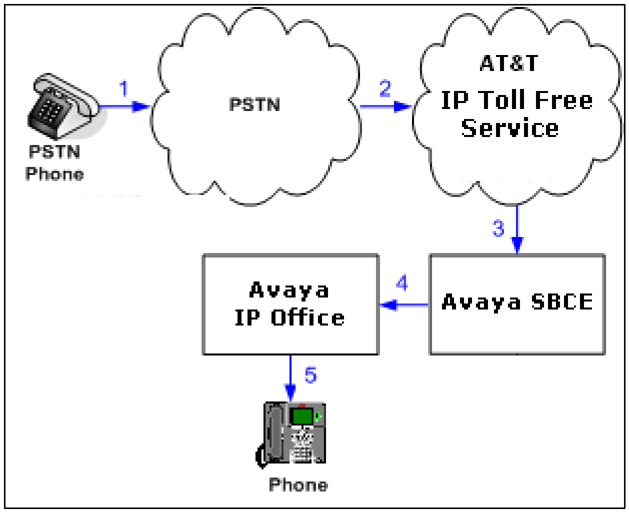


Figure 2: Inbound AT&T IPTF Call

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.

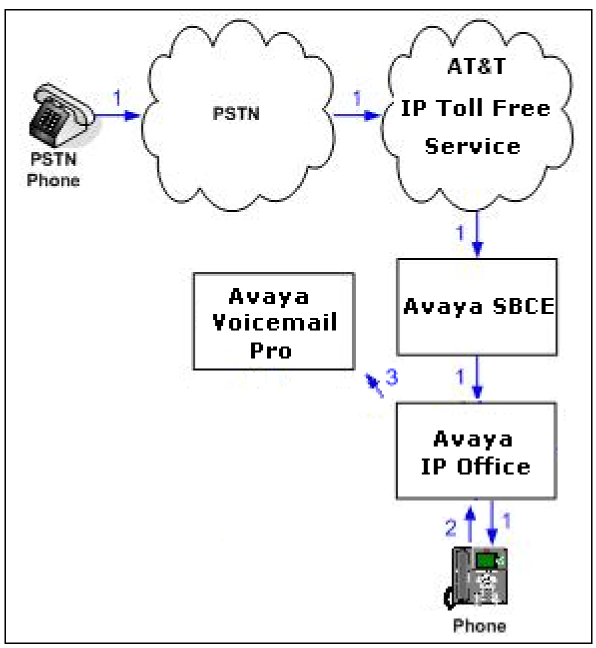


Figure 3: Coverage to Voicemail (Voicemail Pro)

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 500 V2	R8.1 (63)
Avaya IP Office Manager	10.1 (63)
Avaya 1608 (H.323) Telephone	Ha1608ua1_3200.bin
Avaya 1120E (SIP) Telephone	04.03.12.00
Avaya IP Office Softphone (SIP)	3.2.3.20 (64770)
Avaya 6211 Analog Telephone	-
Avaya Session Border Controller for Enterprise	6.2.0 Q30
Fax device	Ventafax 6.3
AT&T IPTF Service via MIS/PNT transport service connections.	VNI 26

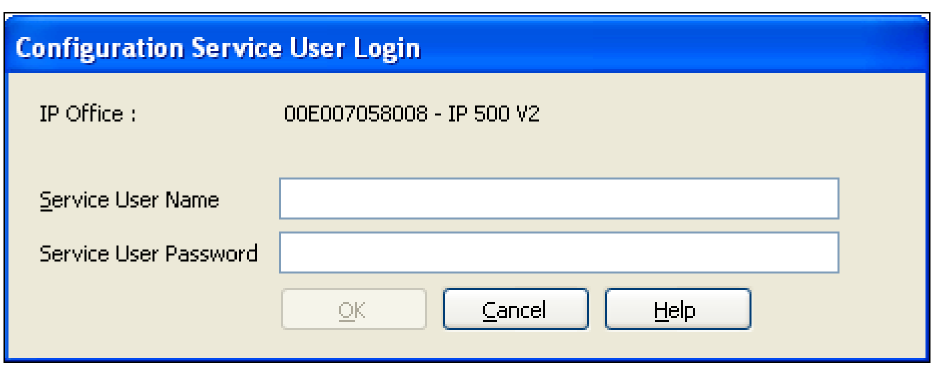
Table 2: Equipment and Software Versions

Note - Testing was performed with IP Office 500 R8.1, but it also applies to IP Office Server Edition R8.1. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R8.1 to support analog or digital endpoints or trunks.

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 this document. Other parameter values may or may not match based on local configurations. Additionally, the screen shots referenced in these sections may not be complete at times. For more information on installing Avaya IP Office consult reference **[1]**.

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

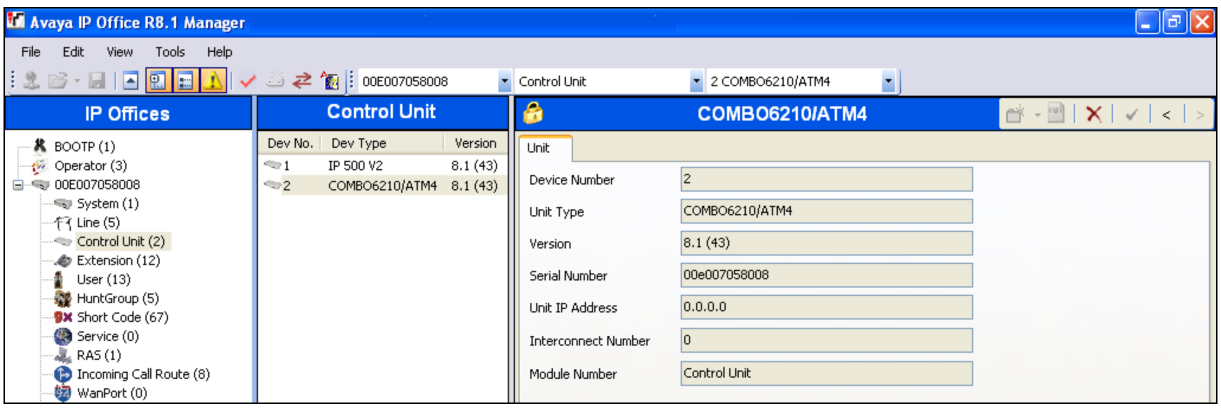


5.1. Physical, Network, and Security Configuration

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

In the reference configuration the Avaya IP Office 500 V2 platform contained a COMBO6210/ATM4 module. The COMBO6210/ATM4 is used to add a combination of ports to an IP500 V2 control unit. The module supports 10 voice compression channels. Codec support is G.711mu, G729a, G.723 with 64ms echo cancellation and G.722 (supported by Avaya IP Office Release 8.0 and higher). The module also supports 6 Digital Station ports for digital stations in slots 1-6 (except 3800, 4100, 4400, 7400, M and T-Series), 2 Analog Extension ports in slots 7-8, and 4 Analog Trunk ports in slots 9-12.

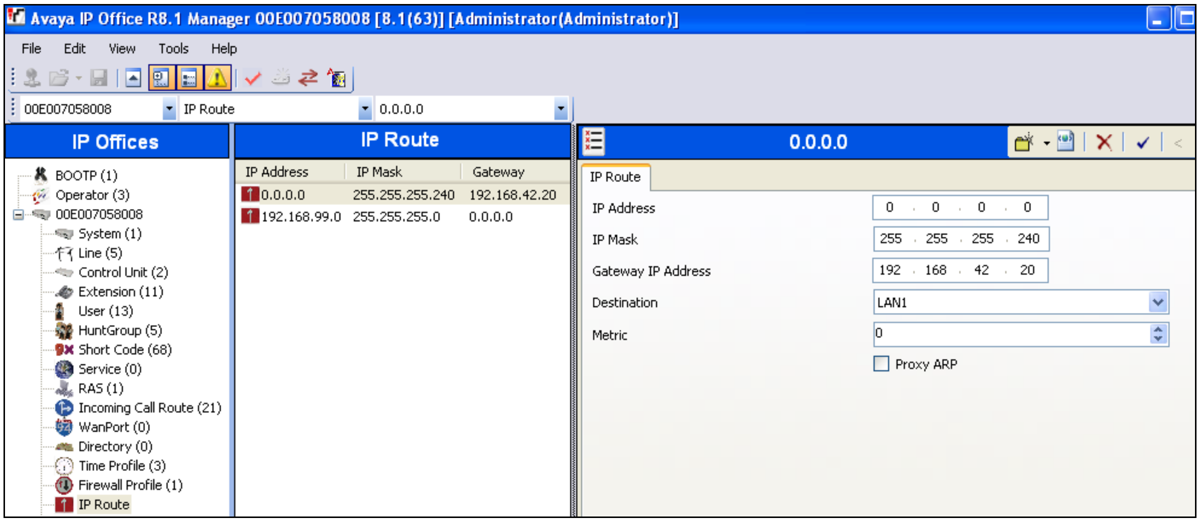
The following screen shows the Avaya IP Office module configuration used in the reference configuration. In the screen below, **IP 500 V2** is selected in the Group pane, revealing additional information about the IP 500 V2 in the Details pane.



The Avaya IP Office 500 V2 has two Ethernet ports on the back of the chassis, labeled **LAN** (LAN1) and **WAN** (LAN2). In the reference configuration, the LAN port (LAN1) is connected to the private CPE network, as described in **Section 3**. The Avaya SBCE, as well as H.323 telephones, SIP telephones, and the management/Softphone PC used in the reference configuration, are also connected to the private CPE network. The WAN port (LAN2) is connected to the CPE management LAN subnet.

Provisioning for these interfaces is described in **Section 5.3.2** and **5.3.3**.

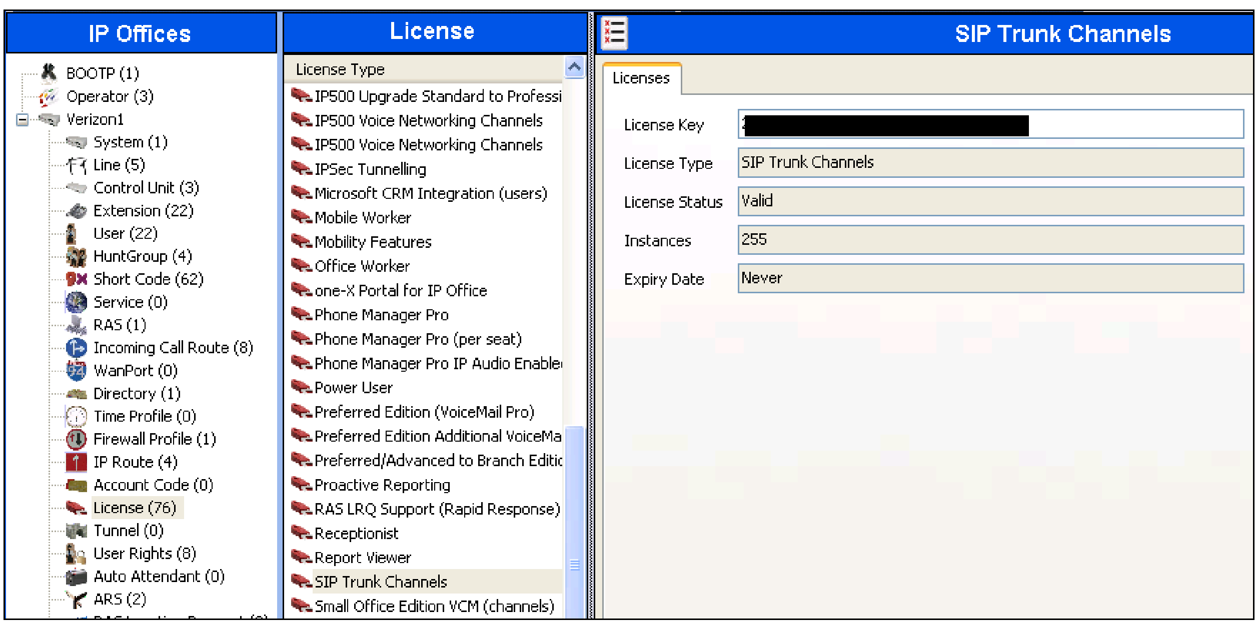
In order for the Avaya IP Office system to be able to route data to/from the AT&T network, a default route must be added specifying the Avaya SBCE “A1” interface (e.g., **192.168.42.20**). To add an IP Route in Avaya IP Office, right-click **IP Route** from the Navigation pane, and select **New.** To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the relevant default route using **Destination → LAN1**.



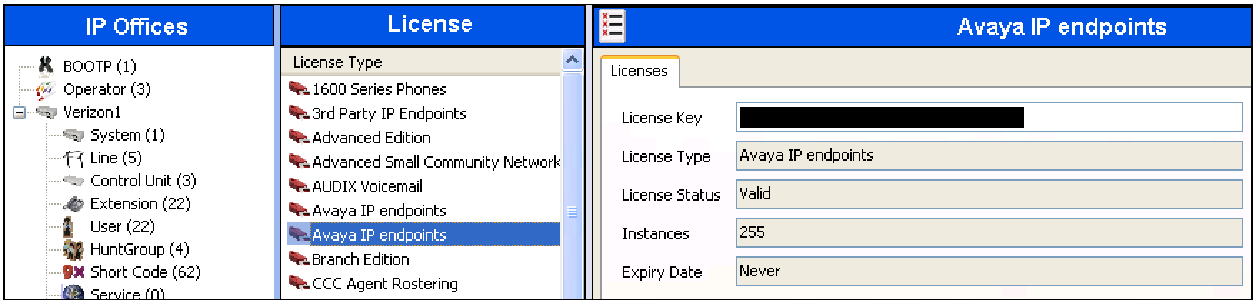
5.2. Licensing

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.

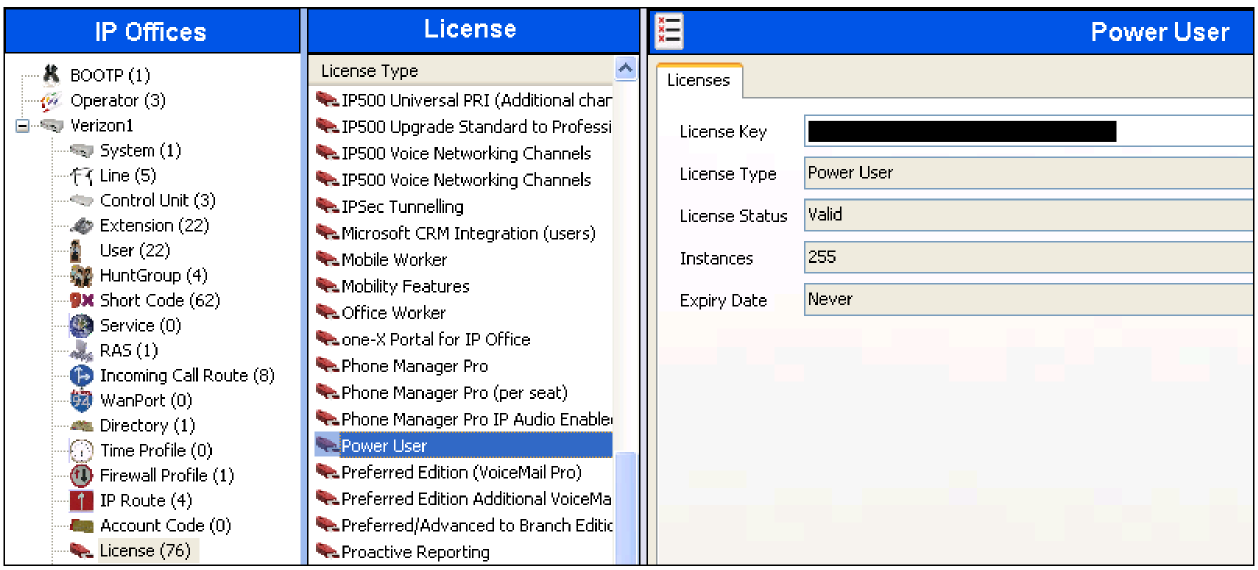
To verify that there is a SIP Trunk Channels License with sufficient capacity, click **License** in the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm a valid license with sufficient "Instances" (trunk channels) in the Details pane.



If Avaya IP Telephones will be used, verify the Avaya IP endpoints license. Click **License** in the Navigation pane and **Avaya IP endpoints** in the Group pane. Confirm a valid license with sufficient "Instances" in the Details pane. Note that in some cases duplicate license entries may be listed (e.g, Avaya IP endpoints below). One will display a key sequence in the License Key field while the other will display “Virtual”.



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

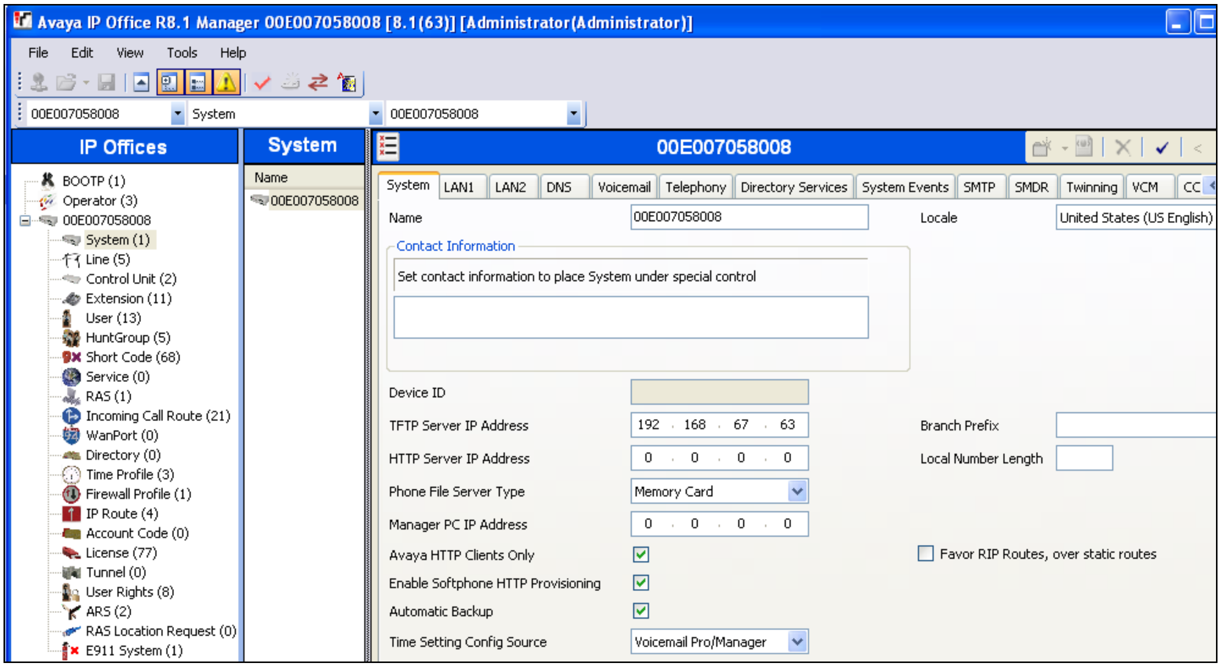


5.3. System Settings

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

5.3.1. System Tab

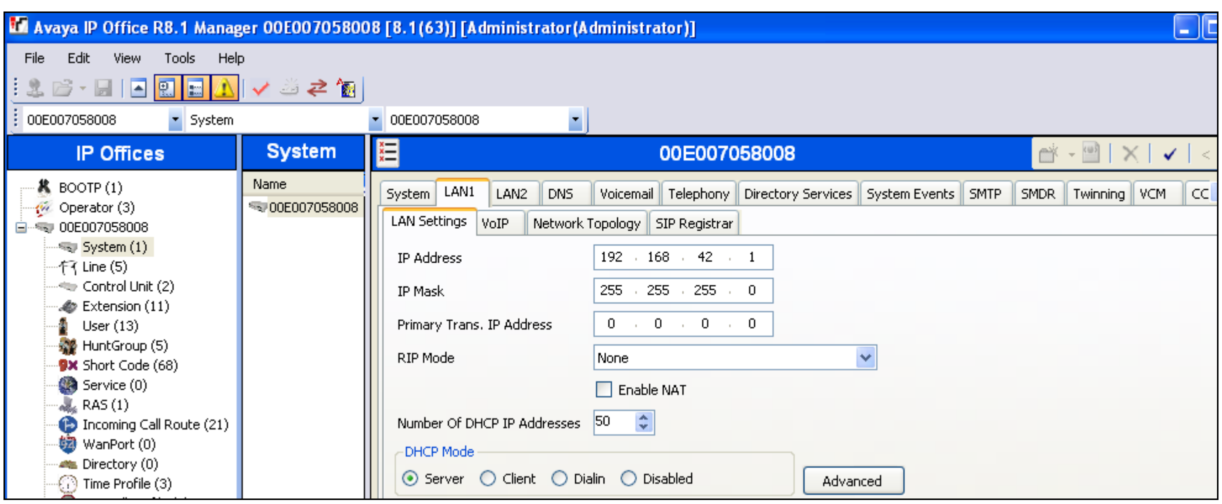
With the proper system name selected in the Group pane, select the **System** tab in the Details pane. The following screen shows a portion of the **System** tab. The **Name** field can be used for a descriptive name of the system. In this case, the default system serial number is used as the name. The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



5.3.2. LAN 1 Settings

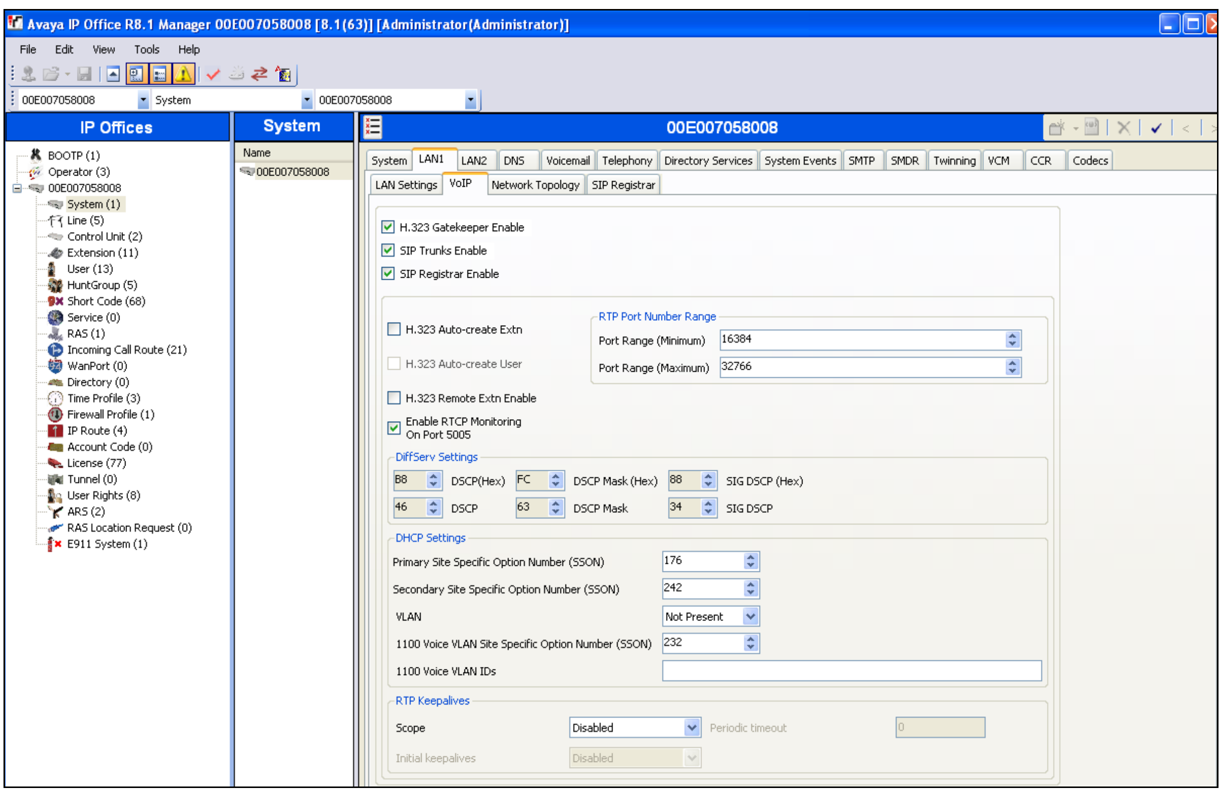
In the reference configuration, LAN1 was used to connect the Avaya IP Office to the CPE network (see the note in **Section 3**). To view or configure the IP address, select the **LAN1** tab followed by the **LAN Settings** tab, and enter the following:

- **IP Address:** Set to **192.168.42.1** as used 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).



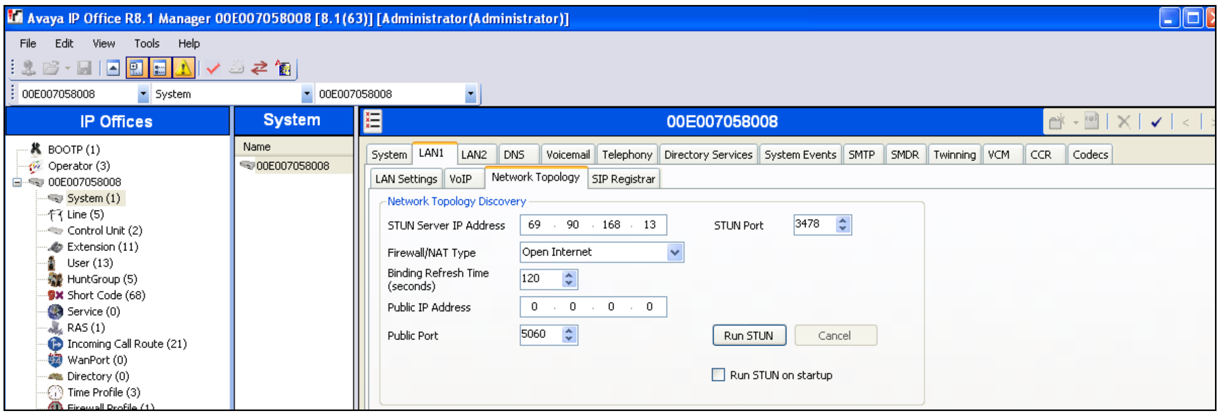
Select the **LAN1 →VoIP** tab as shown in the following screen. The following settings were used in the reference configuration:

- The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 1600-Series Telephones used in the reference configuration.
- The **SIP Trunks Enabled** box is checked to support SIP trunking.
- The **SIP Registrar Enable** box is checked to allow Avaya 11xx (SIP) and Avaya IP Office Softphone (SIP) usage.
- **RTP Port Number Range:** The AT&T IPTF service requires that the RTP use the port range 16384 to 32767.
  - **16384** is entered in the **Port Range (Minimum)** field.
  - **32766** is entered in the **Port Range (Maximum)** field, as this field requires even numbers. See **Section 6.5.3** for more information on the RTP settings.
- **DiffServ Settings** (optional)**:** If desired, Avaya IP Office can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Service policies. The default values were used in the reference configuration.
- Note that on this interface, **RTP Keepalives/Scope** is set to **Disabled** (default).
- Other parameters on this screen may be set according to customer requirements.
- Click the **OK** button (not shown).



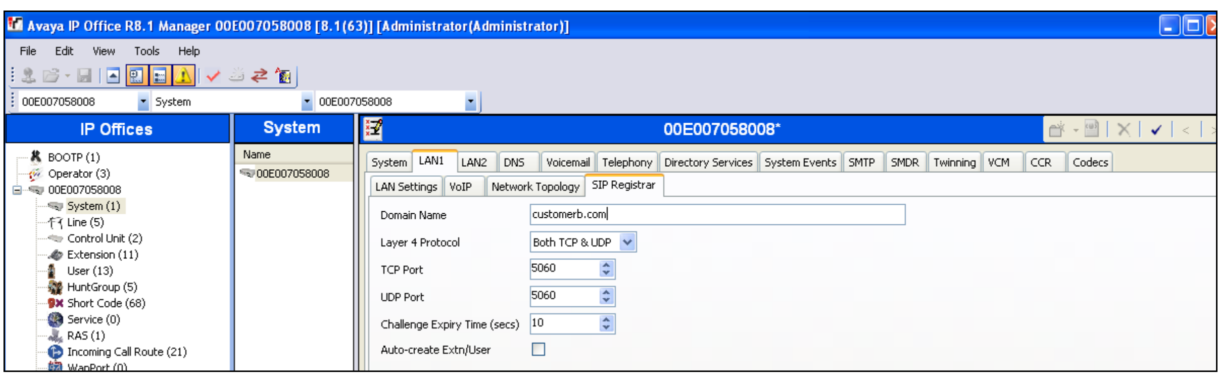
Select the **LAN1 → Network Topology** tab as shown in the following screen, and enter the following:

- **Public IP Address**: The **0.0.0.0** default value is used. This means Avaya IP Office will use the LAN1 IP address specified on the LAN1 **LAN Settings** tab described above (192.168.42.1).
- **Public Port** to **5060**.
- **Firewall/NAT Type** is set to **Open Internet**. With this configuration, STUN will not be used.
- **Binding Refresh Time** is set to **120** (used for OPTIONS interval, see **Section 5.9**).
- Click the **OK** button (not shown).



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

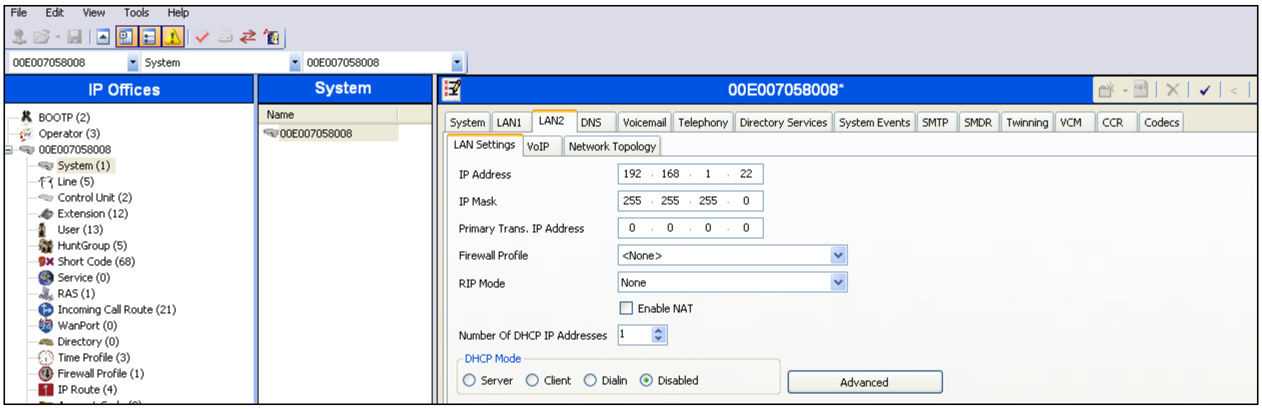
If SIP endpoints are used, select the **SIP Registrar** tab. The following screen shows the settings used in the reference configuration. Note that the **Domain Name** field is set to **customerb.com**, (the CPE domain), otherwise the LAN1 IP address is used for registration.



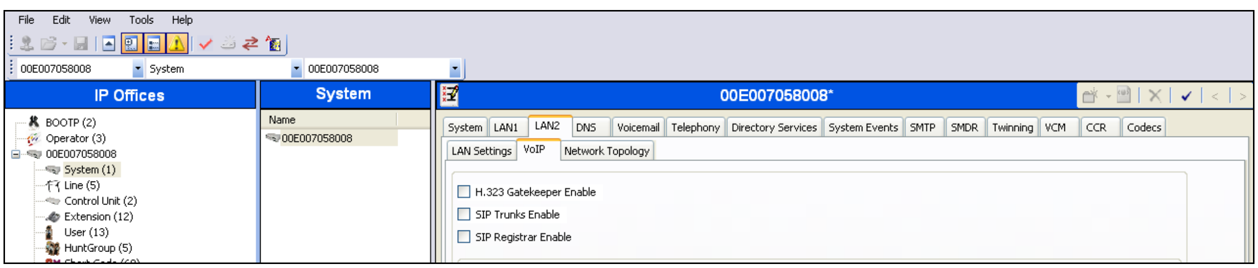
5.3.3. LAN 2 Settings

As described in **Section 3**, the LAN2 interface was not used for VoIP traffic in the "one-wire" configuration implemented in the reference configuration. However the LAN2 interface was used to manage the IP Office platform and was connected to a separate management subnet.

- **IP Address:** In the reference configuration the IP Office management address is **192.168.1.22**.
- Other parameters on this screen were set to defaults.



- On the **VoIP** tab verify that the H.323 Gatekeeper, SIP Trunks, and SIP Registrar boxes are *not* checked.



- The **Network Topology** tab uses default values.
- Click the **OK** button (not shown).

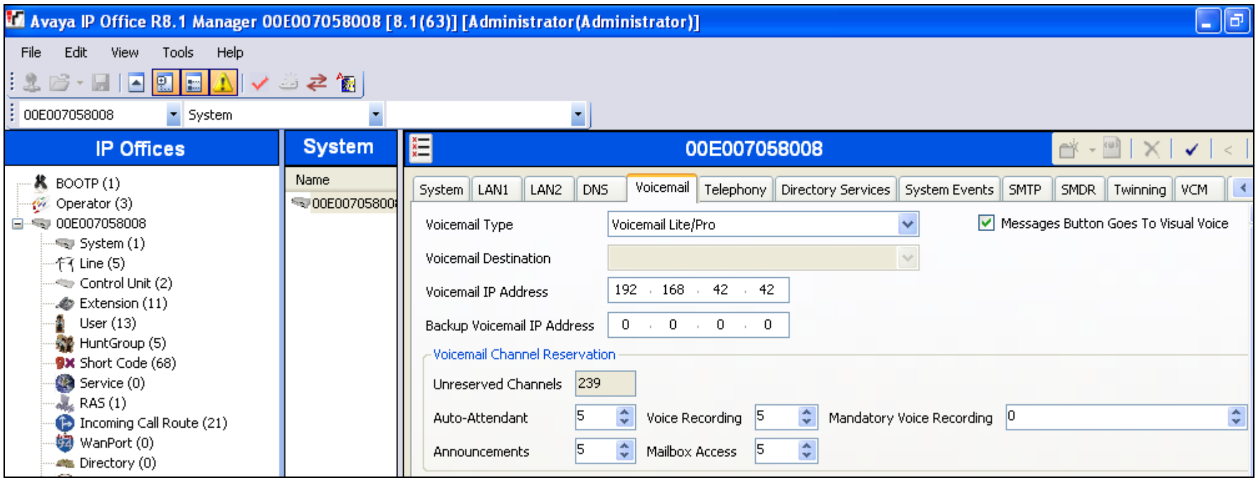
5.3.4. Voicemail

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

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

- Set **Voicemail Type: Voicemail Lite/Pro**.

- Set **Voicemail IP Address:** to the IP address of the platform running Voicemail Pro.
- Other parameters on this screen may be set according to customer requirements.
- Click the **OK** button (not shown).



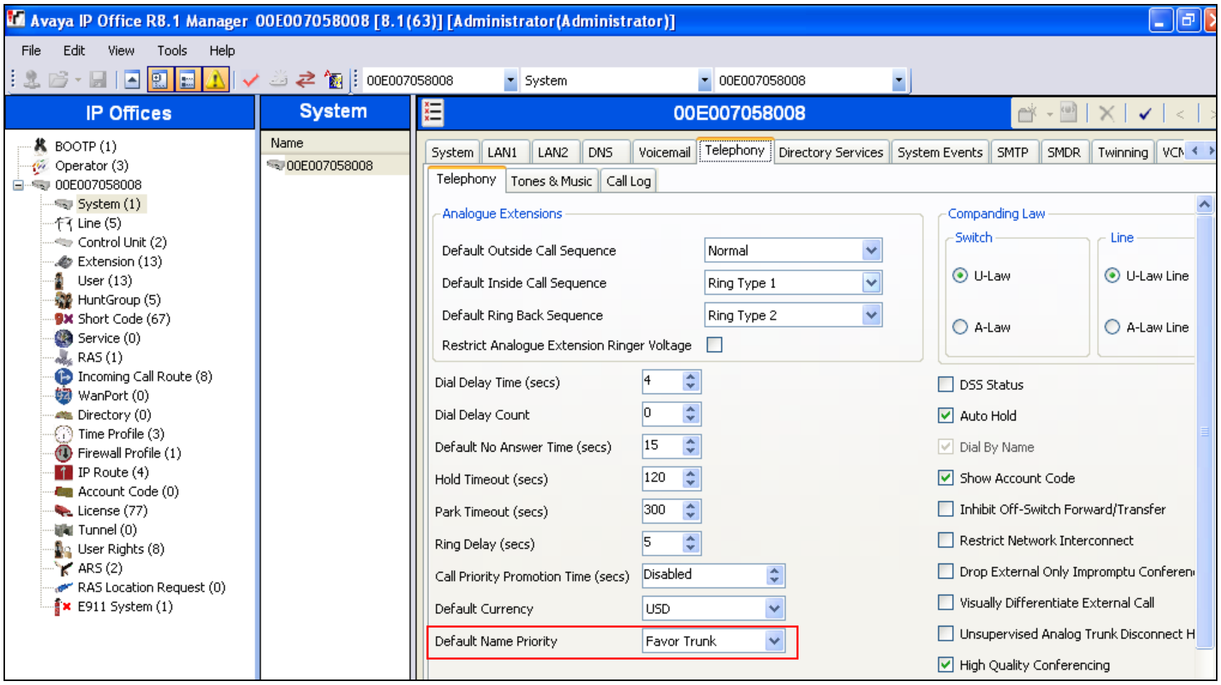
5.3.5. System Telephony Configuration

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

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

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

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

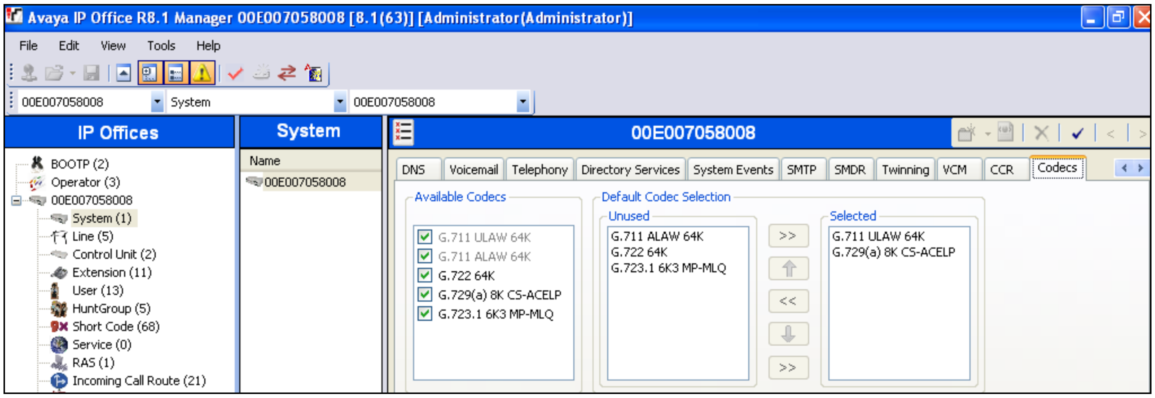


5.3.6. System Codecs Configuration

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

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

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

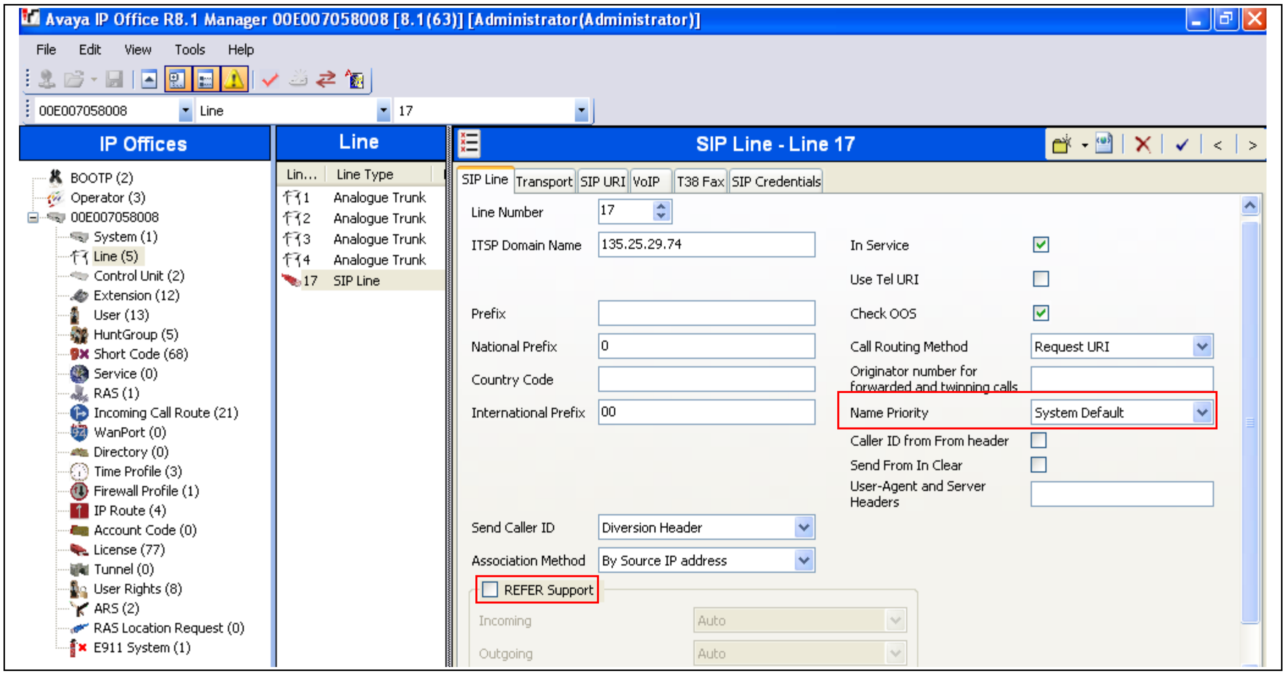


5.4. SIP Line

The **SIP Line** tab in the Details pane is shown below for **Line Number 17**, used for the 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**. SIP Line 17 will be the first SIP Line number created. The SIP Line form is completed as follows:

- **ITSP Domain Name:** Set to the AT&T border element IP address supplied by AT&T (e.g., **135.25.29.74**).
- **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 5.9**).
- **Call Routing Method:** Matched values based on the **Request URI**, or **To Header** contents, may be selected. In the reference configuration, the default **Request URI** setting was used.
- **Country Code:** Use the default <blank>.
- **Send Caller ID:** Set to **Diversion Header**.
- **REFER Support:** Verify that this open is *not* selected (default).
- Use the default values for the other fields.
- Click **OK**.

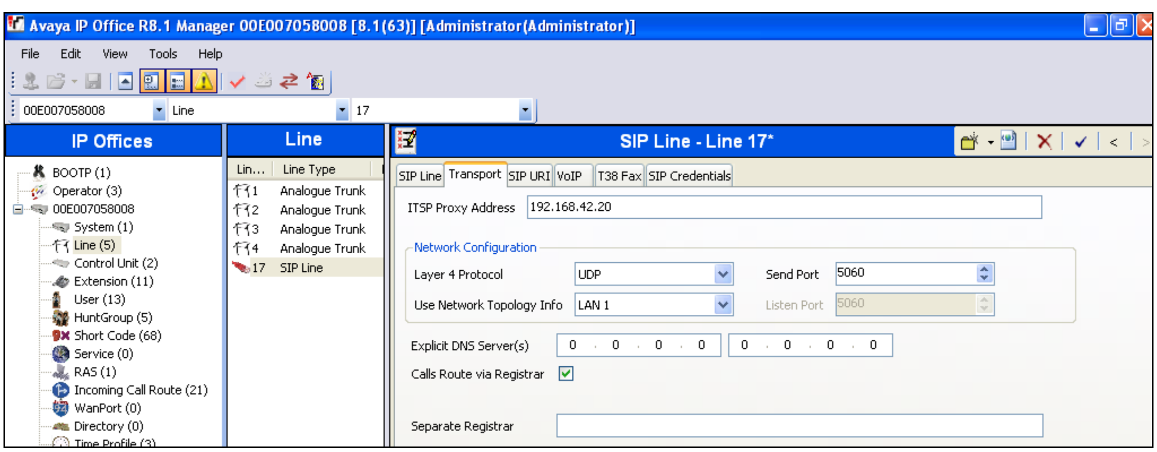
Optional: As described in **Section 5.3.5**, the **Name Priority** parameter may retain the default **System Default** setting, or can be specifically configured to **Favor Trunk** or **Favor Directory**. The default **System Default** setting was used in the reference configuration (see **Section 5.3.5** for the System setting of **Favor Trunk**).



5.4.1. 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 **192.168.42.20** (see **Section 3**).
- **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**.
- **Calls Route via Registrar:** Enabled (default).
- **Click OK** (not shown).



5.4.2. SIP Line - SIP URI Tab

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

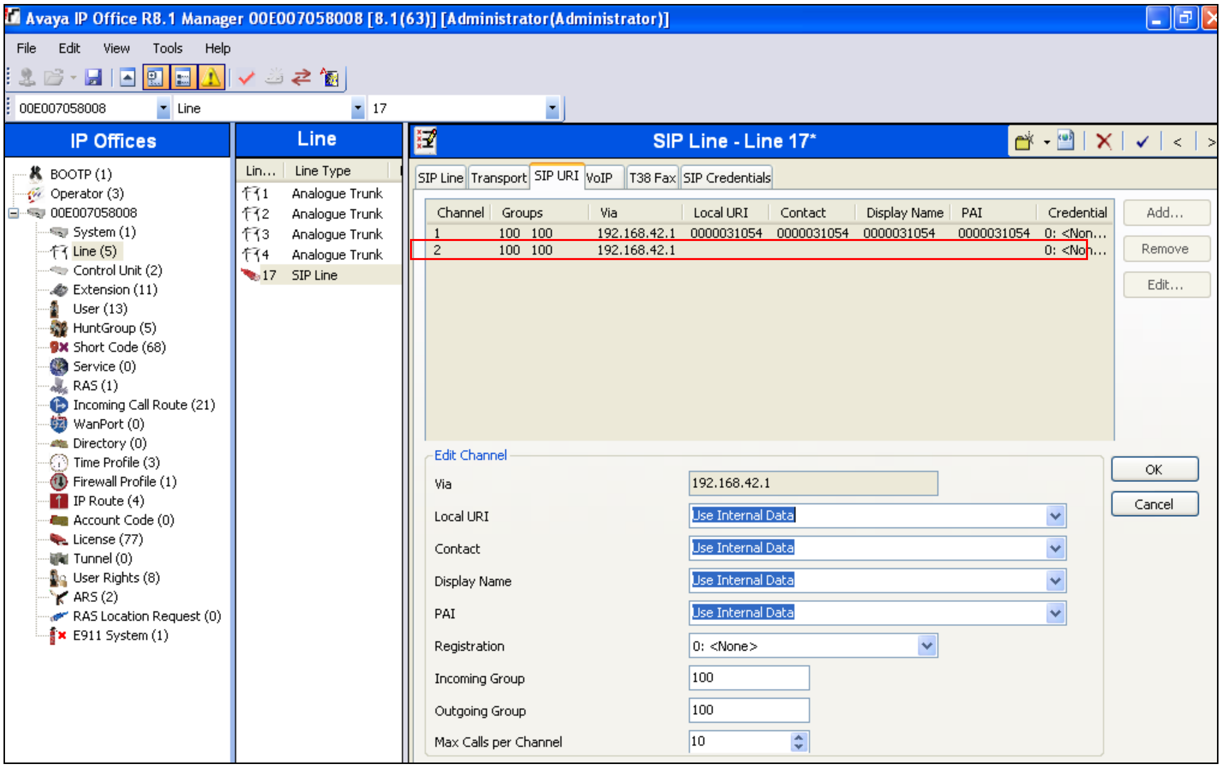
1. A "wild-card" entry that will use the contents of SIP headers containing "calling" information.

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

In this method the following information is specified:

- The **VIA** field will automatically be populated with the IP address of the system LAN interface (LAN 1) with which the SIP trunk is associated (see **Sections 5.3.2** and **5.4.1)**.
- **Local URI**, **Contact**, **Display Name**, **and PAI**: Set to **Use Internal Data**.

  > **Note –** This PAI setting directs Avaya IP Office to send the PAI (P-Asserted-Identity) header instead of the default PPI (P-Preferred-Identity) header when appropriate (e.g., privacy calls). The PAI header will be populated from the data set in the **SIP** tab of the call initiating **User** as shown in **Section 5.5.**
- **Registration:** Set to the default **0: <None>**.
- **Incoming Group:** Set here to **100**. This value references the **Incoming Call Routes** in **Section 5.7**.
- **Outgoing Group:** Set to **100**. Note that although the AT&T IP Toll Free service does not support outbound calls, this value was set to match the Incoming Group.
- **Max Calls per Channel:** In the reference configuration this was set to **10**. This sets the maximum number of simultaneous calls that can use the URI before Avaya IP Office returns busy to any further calls.
- Click **OK** to save the information.

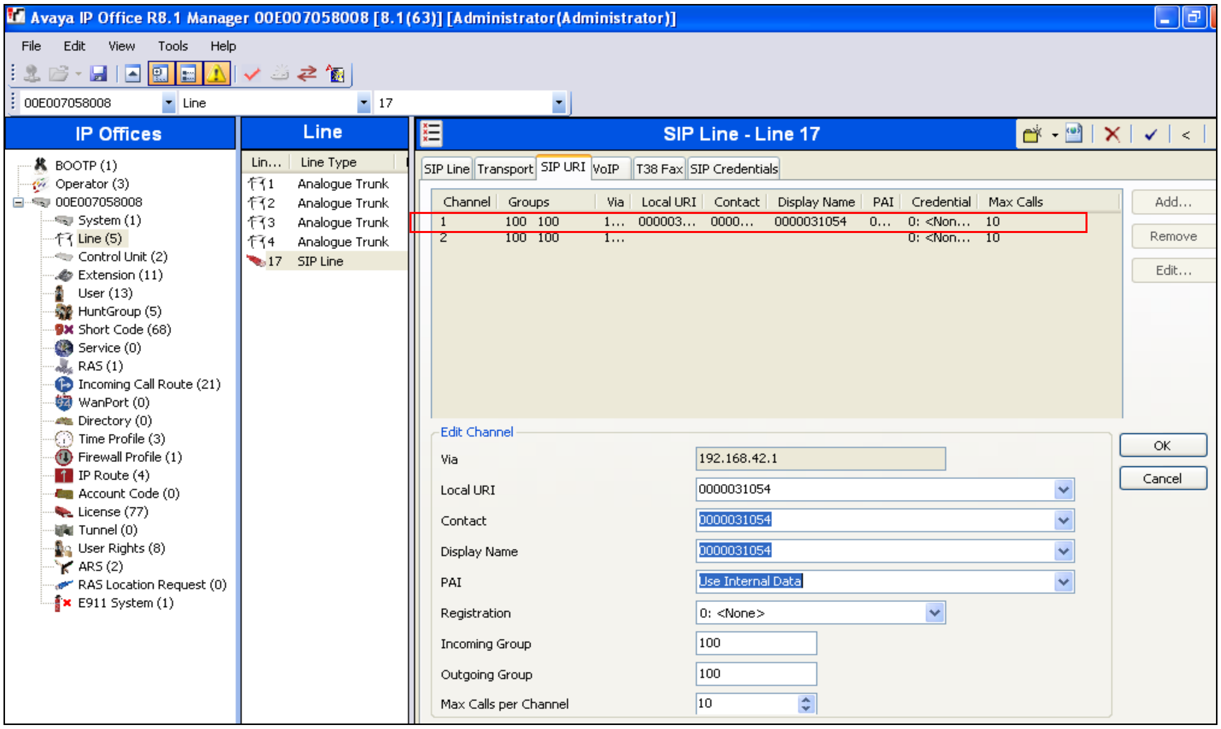


2. A specific entry that will match inbound DNIS digits from AT&T.

> **Note** – This method must be used for Avaya IP Office call destinations other than Users or Hunt Groups, (e.g., Auto Attendant or direct calls to Voicemail Pro), or the calls will be denied.

In this method the following information is specified:

- **Local URI**, **Contact**, and **Display Name**: Set to an AT&T DNIS number (e.g., **0000041053** to Voicemail Pro)**.**
- **PAI**: Set to **Use Internal Data**.
- **Registration:** Set to the default **0: <None>**.
- **Incoming Group:** Set here to **100**. This value references the **Incoming Call Routes** in **Section 5.7**.
- **Outgoing Group:** Set to **100**. Note that although the AT&T IP Toll Free service does not support outbound calls, this value was set to match the Incoming Group.
- **Max Calls per Channel:** In the reference configuration to this was set to **10**. This sets the maximum number of simultaneous calls that can use the URI before Avaya IP Office returns busy to any further calls.
- Click **OK** to save the information.



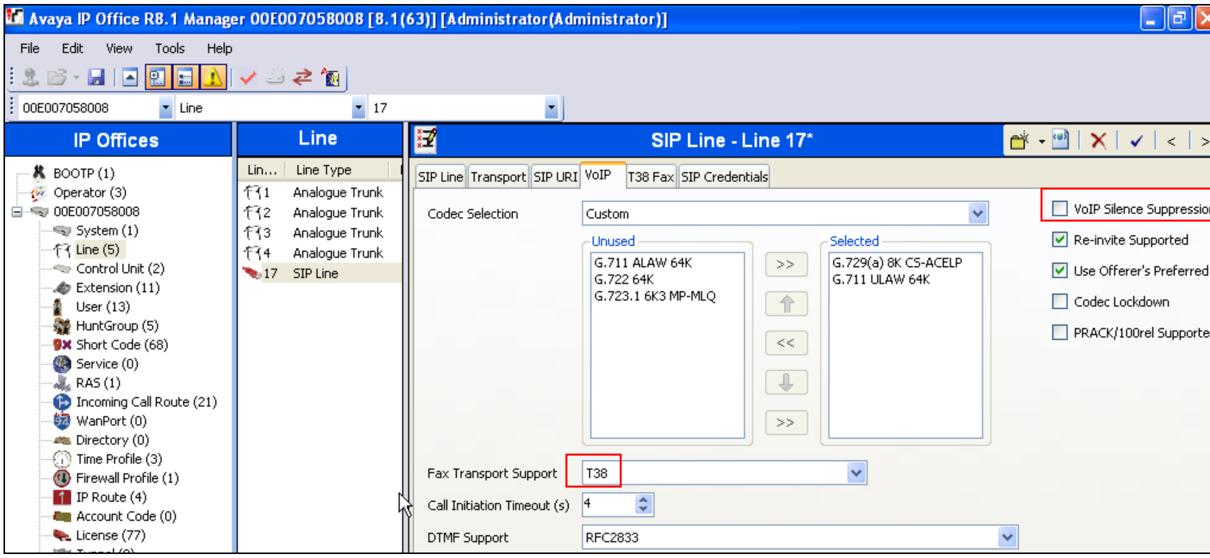
- To edit an existing entry, click an entry in the list and click the **Edit…**button.
- When all SIP URI entries have been added/edited, click **OK** at the bottom of the screen (not shown).

5.4.3. SIP Line - VoIP Tab

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

- T.38 fax was used in the reference configuration. Set the **Fax Transport Support** drop-down menu to **T38**. Note that the **T.38 Fallback** option is *not* supported in the reference configuration (see **Section 2.2.1**). Note that Error Correction Mode (ECM) is enabled by default on the T.38 Fax tab (**Section 5.4.4**). ECM is supported by the AT&T IPTF service. G.711 fax also worked in the reference configuration (T.38 option disabled); however T.38 is the preferred method.
- The **DTMF Support** parameter can remain set to the default value **RFC2833**.
- The **Re-invite Supported** parameter can be checked to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk.
- Click **OK** (not shown).

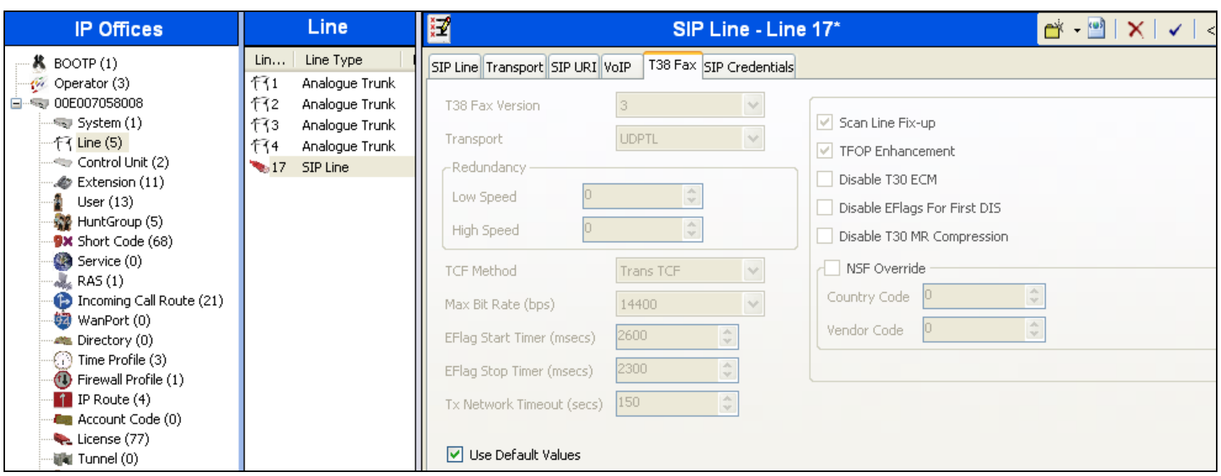
Note - By default the VoIP Silence Suppression box is not checked. This disables the use of the G.729B codec. If silence suppression is desired, check this box, and enable the **VoIP Silence Suppression** option on the Extension form **VoIP** tab (see **Section 5.5.2**).



5.4.4. SIP Line - T38 Fax

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

Note - All default values were used in the reference configuration. Therefore the **Use Default Values** box is checked. If different settings are needed, uncheck this box to unlock the form.



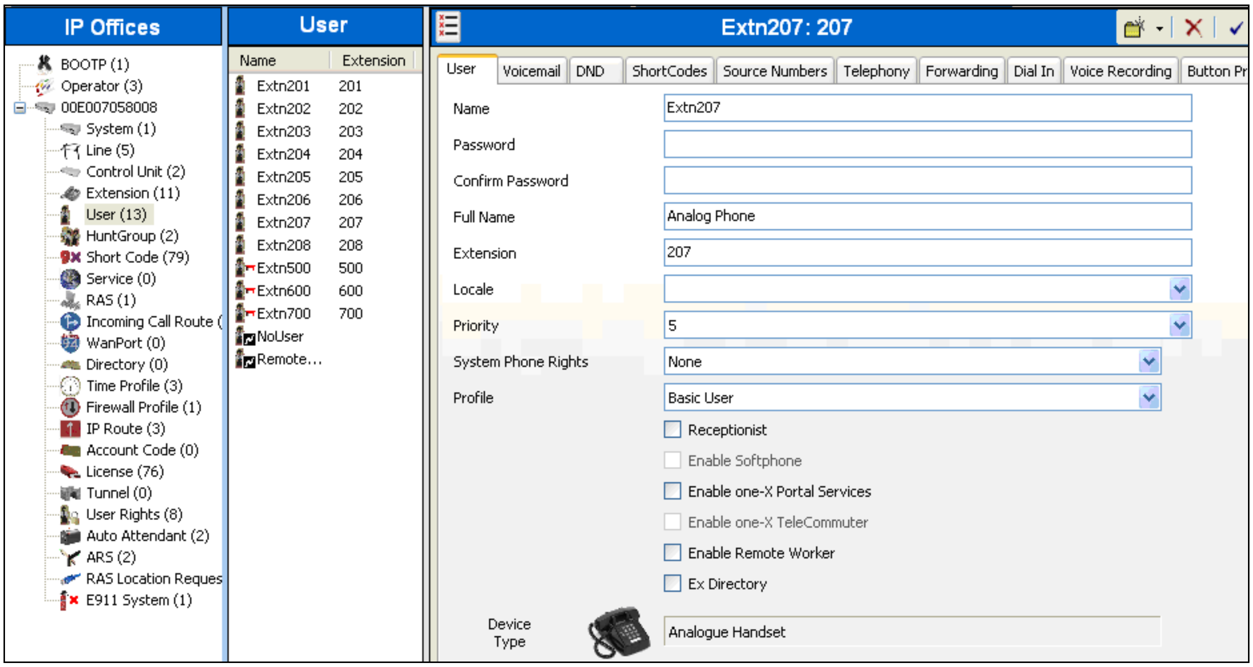
Note - Since the AT&T IPTF service does not require registration, the **SIP Credentials** tab need not be visited.

5.5. Users, Extensions, and Hunt Groups

In this section, examples of Avaya IP Office Users, Extensions, and Hunt Groups will be illustrated. To add a User, right click on **User** in the Navigation pane, and select **New.** To edit an existing User, select **User** in the Navigation pane, and select the appropriate user to be configured in the Group pane.

5.5.1. Analog User Extn207

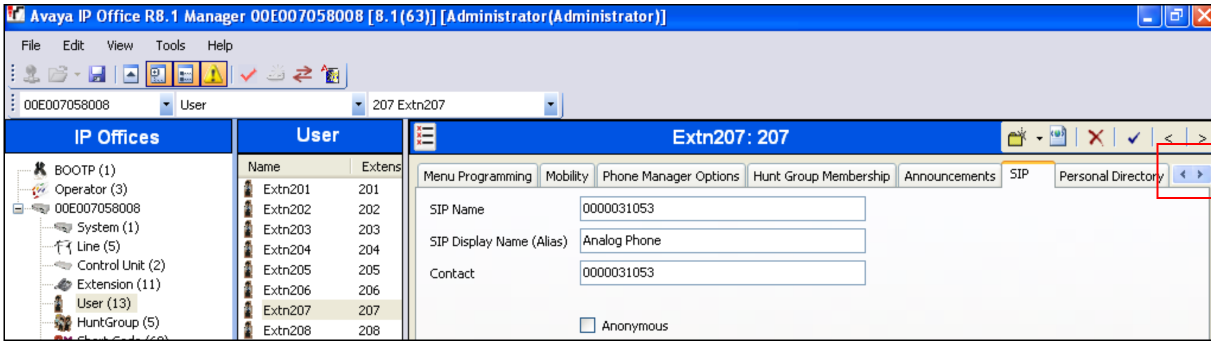
The following screen shows the **User** tab for User **Extn207**. This user corresponds to the Avaya Analog 6211 set.



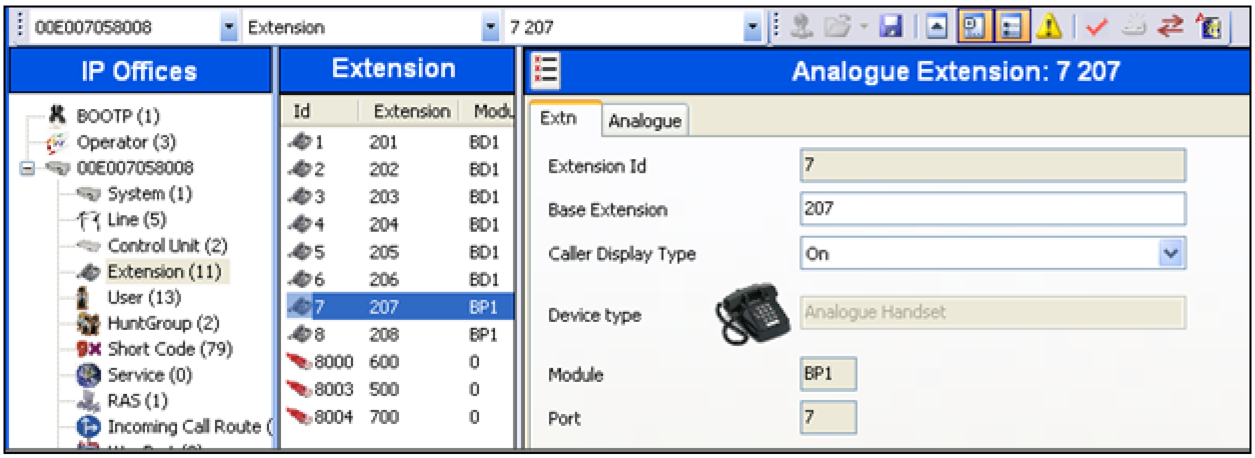
The following screen shows the **SIP** tab for User **Extn207** (use the arrow buttons in the upper right corner to navigate to the SIP tab). The **SIP Name** and **Contact** parameters are configured with the associated AT&T DNIS number of the user, (e.g., **0000031053**). These parameters configure the user part of the SIP URI in the From header for outgoing SIP headers, and allow matching of the SIP URI for incoming calls, without having to enter this number as an explicit SIP URI for the SIP Line (see **Section 5.4.2**).

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

If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network.



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

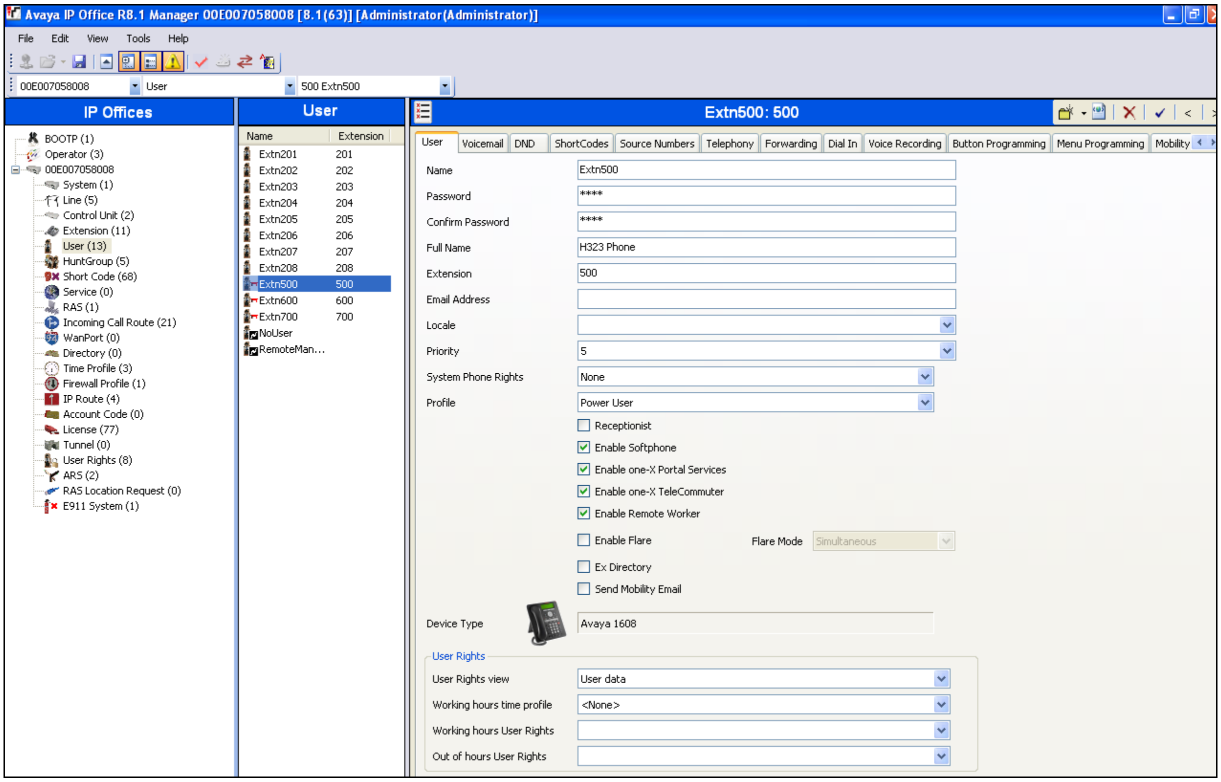


5.5.2. IP Phone User Extn500

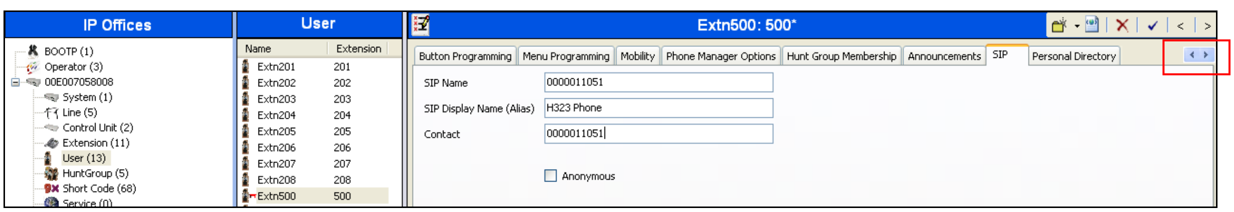
To create a new extension, right click on **Extension** from the Navigation pane and select **New** and **H.323**. Alternatively edit an existing extension by selecting an extension in the Group pane.

Step 1 - The following screen shows a 1608 IP Telephone provisioned in the **User** tab for User **Extn500**. In the reference configuration, this user will be granted "Power User" features.

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

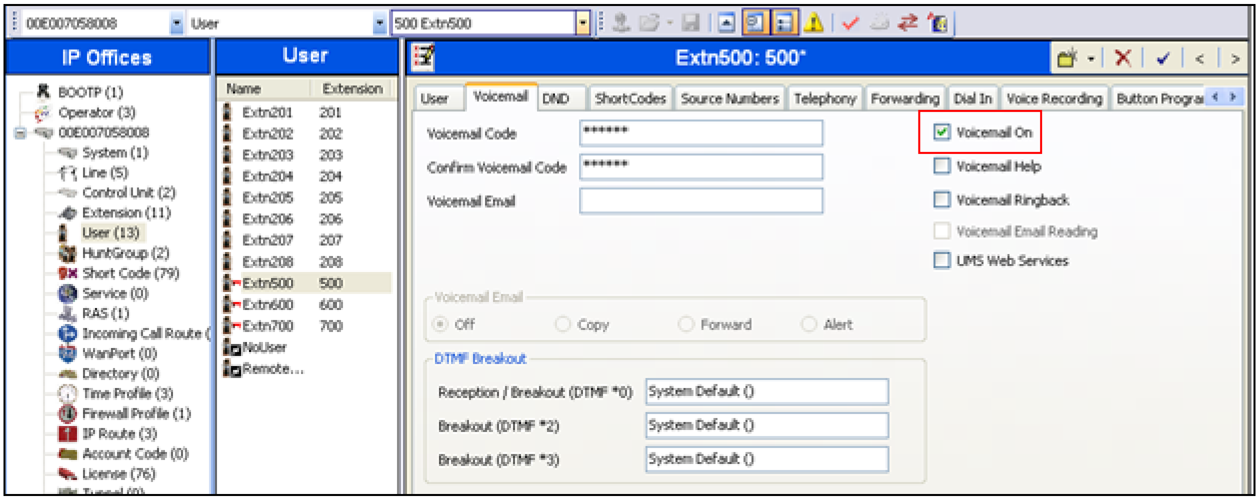


Step 2 - Like the Analog Extn207 user, the **SIP** tab (use the arrow buttons in the upper right corner to navigate to the SIP tab) for User Extn500 is configured with a **SIP Name** and **Contact** specifying the user's associated AT&T DNIS number (e.g., **0000011051**).

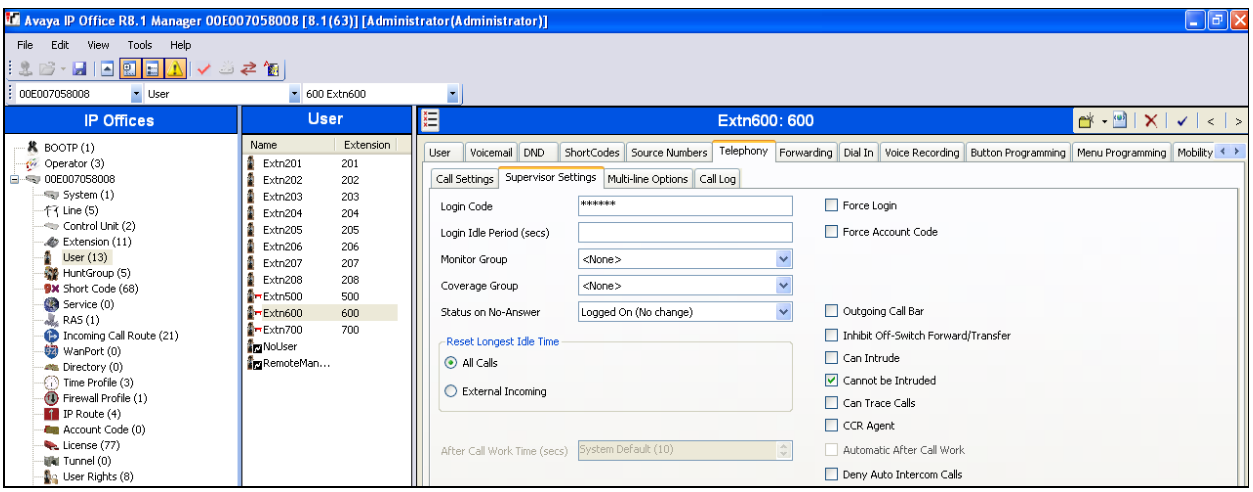


Step 3 - The following screen shows the **Voicemail** tab for User Extn500. The **Voicemail On** box is checked, and a Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters.

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



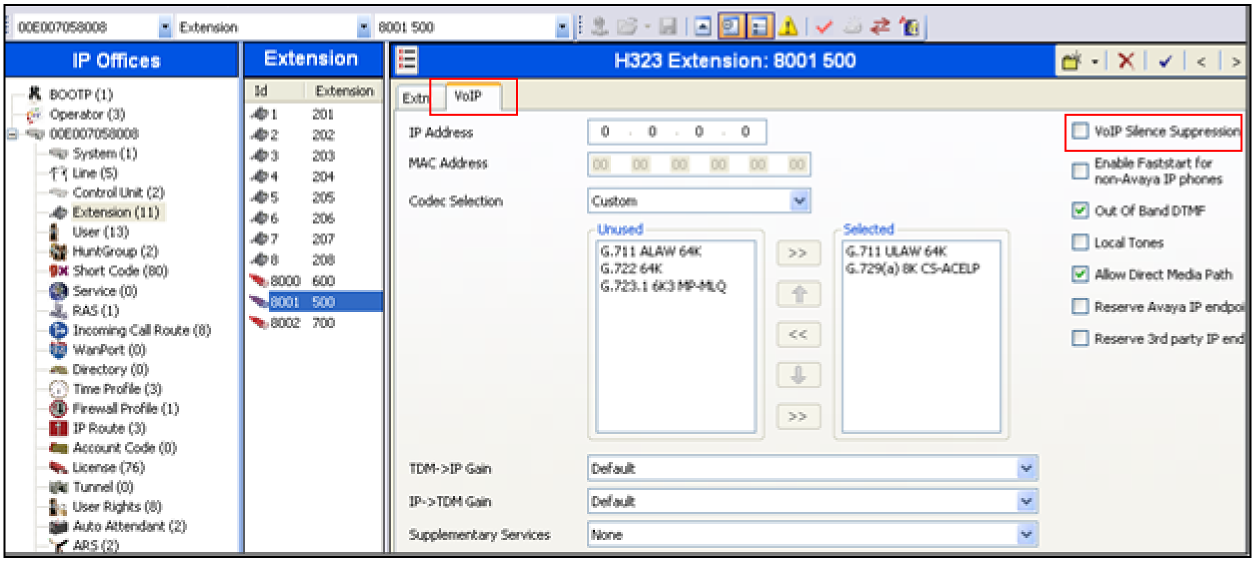
Step 4 - Select the **Telephony → Supervisor Settings** tab as shown below. The **Login Code** will be used by the telephone user as the phone login password.



Step 5 - The following screen shows the Extension information for this user, to illustrate the **VoIP** tab available for an IP Telephone. To view, select **Extension** from the Navigation pane, and the appropriate extension (**500**) from the Group pane. Select **VoIP** in the Details pane.

- Use the **IP Address** field default value (**0.0.0.0**).
- Note that the same codec list as shown in **Section 5.3.6** is used.
- Use defaults for the remaining fields.

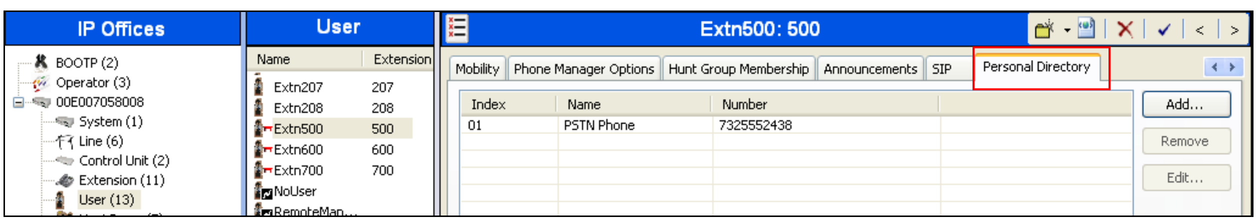
Note that by default the VoIP Silence Suppression box is not checked (the same applies to provisioned SIP phones as shown in **Section 5.5.3**). This disables the use of the G.729B codec. If silence suppression is desired, check this box, and enable the Silence Suppression option on the SIP Line form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



OPTIONAL: As described in **Section 5.3.5**, the option to **Favor Trunk** or **Favor Directory** can be set system-wide, or set uniquely for each user based on the SIP Line setting. The system-wide application of **Favor Trunk** was used in the reference configuration. The following example shows how the **Favor Directory** option can be specified for users.

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

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



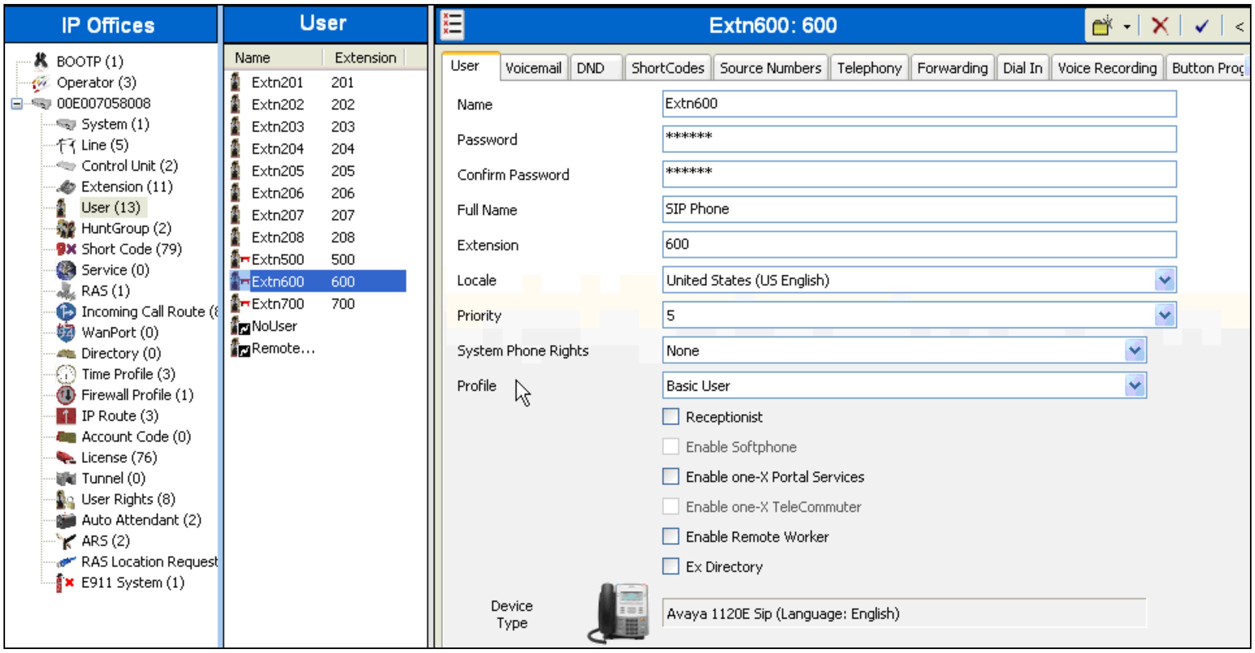
5.5.3. SIP Telephone Users (Avaya 1120E and Avaya IP Office SoftPhone)

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

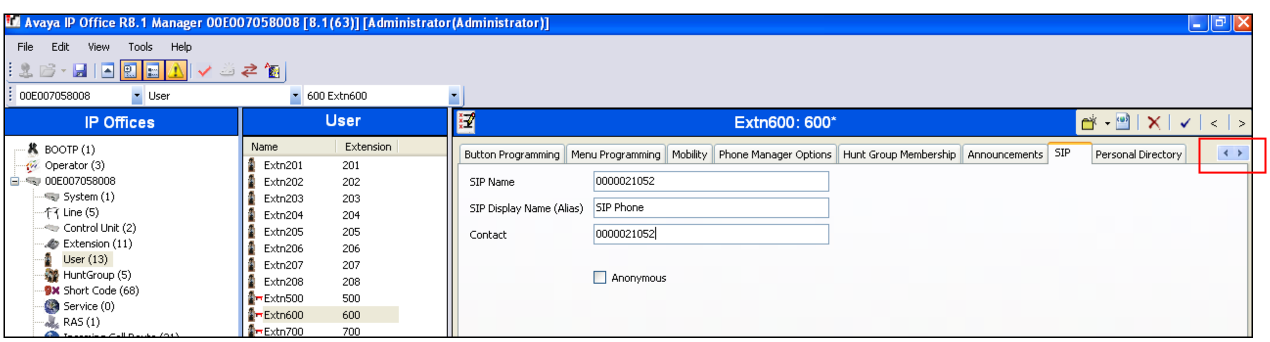
5.5.3.1 SIP Avaya 1120E

Step 1 - The following screen shows an1120E Telephone provisioned in the **User** tab for User **Extn600**.

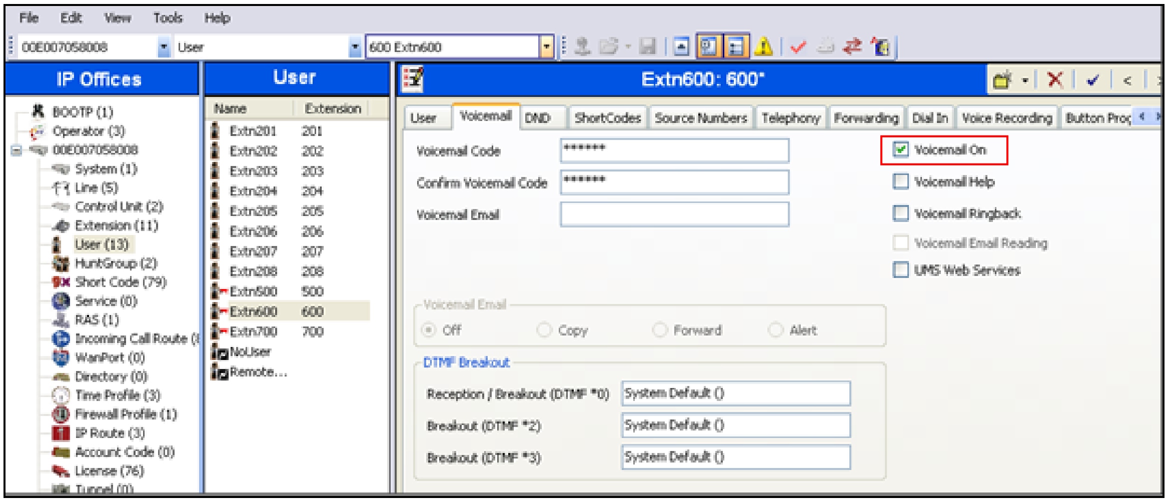
- **Password**: This password is used by user applications such as SoftConsole, Phone Manager and TAPI. It is also used for users with Dial In access. Note that this is ***not*** the user's phone log in code (see the information on the **Telephony** tab → **Supervisor Settings** below), or their Voicemail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



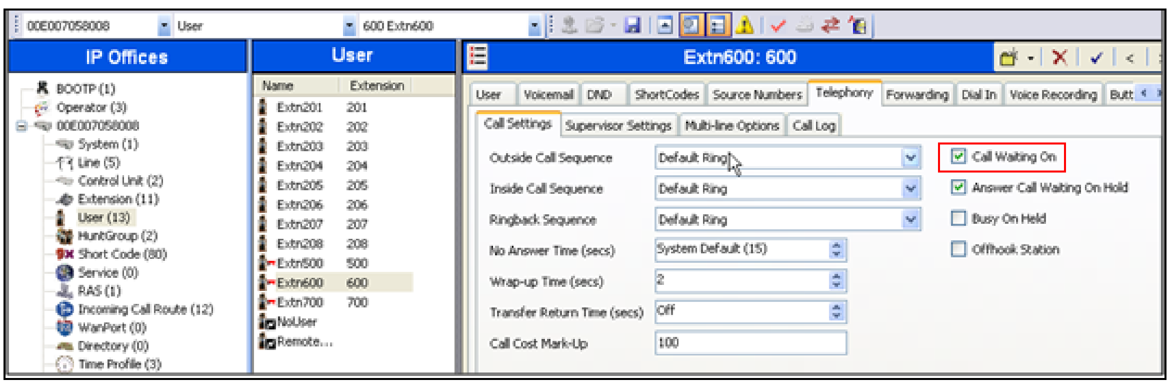
Step 2 - Like the H.323 Extn500 user, the **SIP** tab (use the arrow buttons in the upper right corner to navigate to the SIP tab) for User Extn600 is configured with a **SIP Name** and **Contact** specifying the user's associated AT&T DNIS number (e.g., **0000021052**). Optionally a user can be set to use privacy for all calls by selecting the **Anonymous** option.



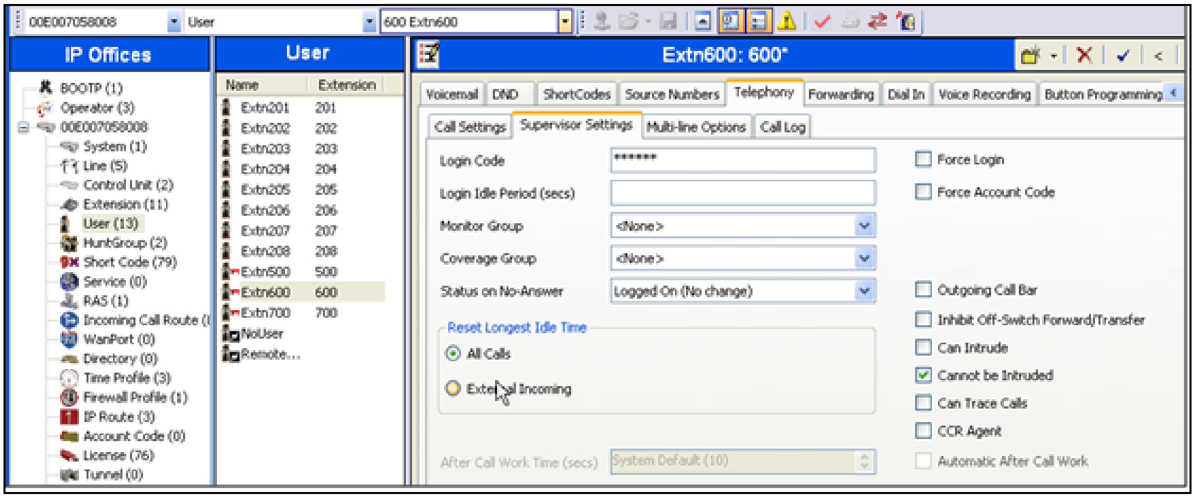
Step 3 - Like the H.323 Extn500 user, Extn600 also utilized the external Voicemail Pro server. The **Voicemail On** box is checked, and a Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters.



Step 4 - Select the **Telephony→ Call Settings** tab as shown below. Check the **Call Waiting On** box to allow multiple call appearances and transfer operations.

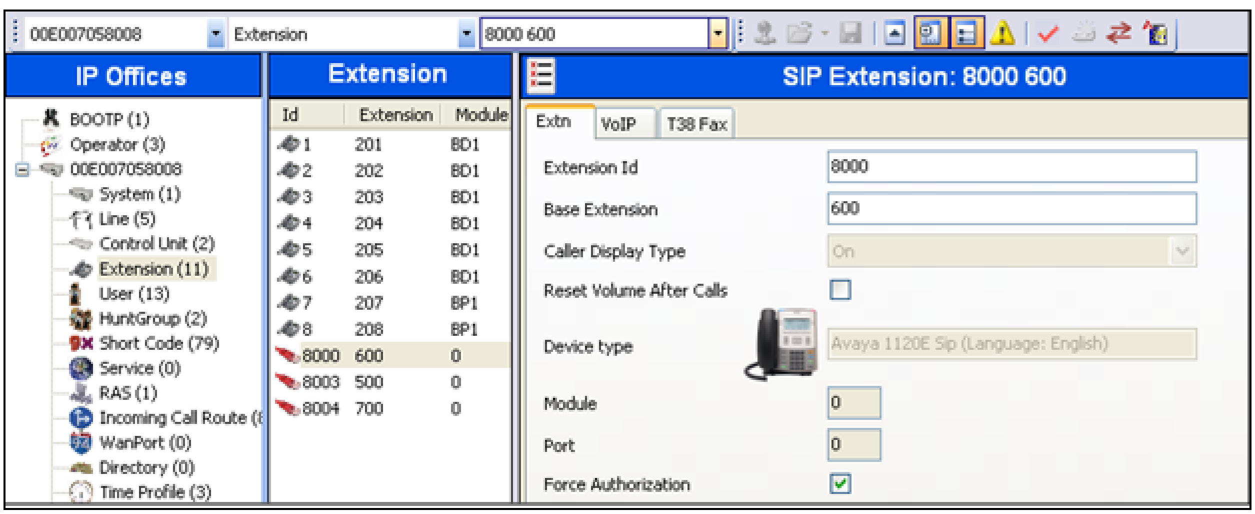


Step 5 - Select the **Telephony→ Supervisor Settings** tab as shown below. The **Login Code** will be used by the telephone user as the phone login password.



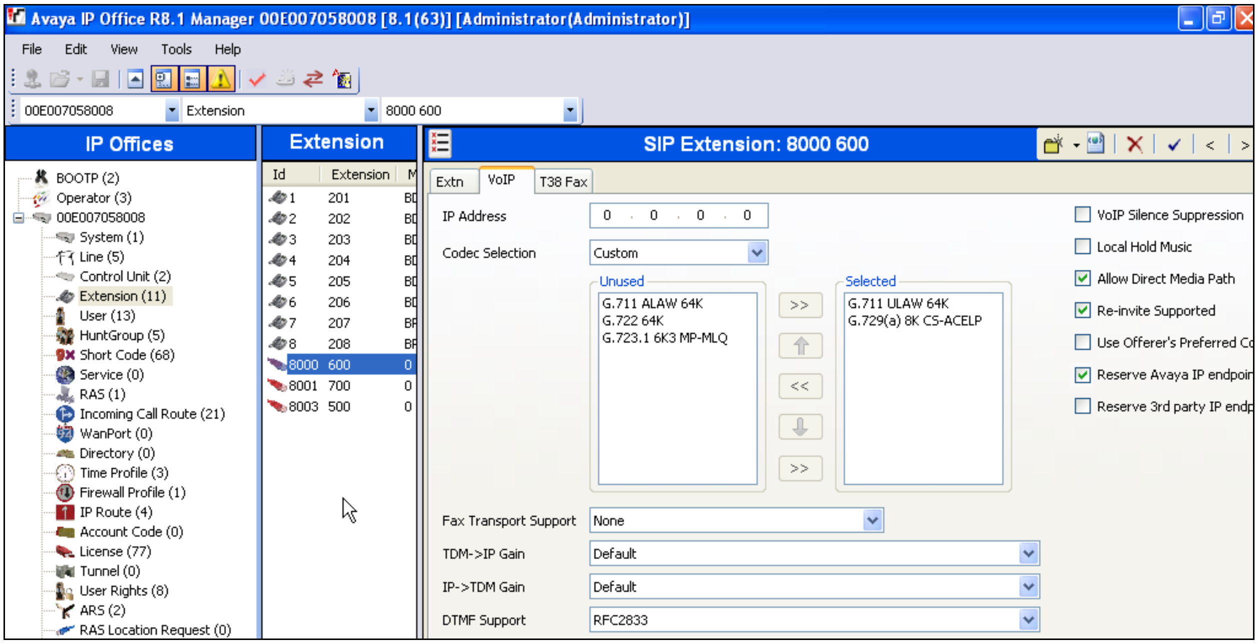
Step 6 - A new SIP extension may be added by right-clicking on **Extension** in the Navigation pane and selecting **New** and **SIP Extension**. Alternatively, an existing SIP extension may be selected in the group pane. The following screen shows the **Extn** tab for the extension corresponding to an Avaya 1120E.

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



Step 7 - The following screen shows the **VoIP** tab for extension 600.

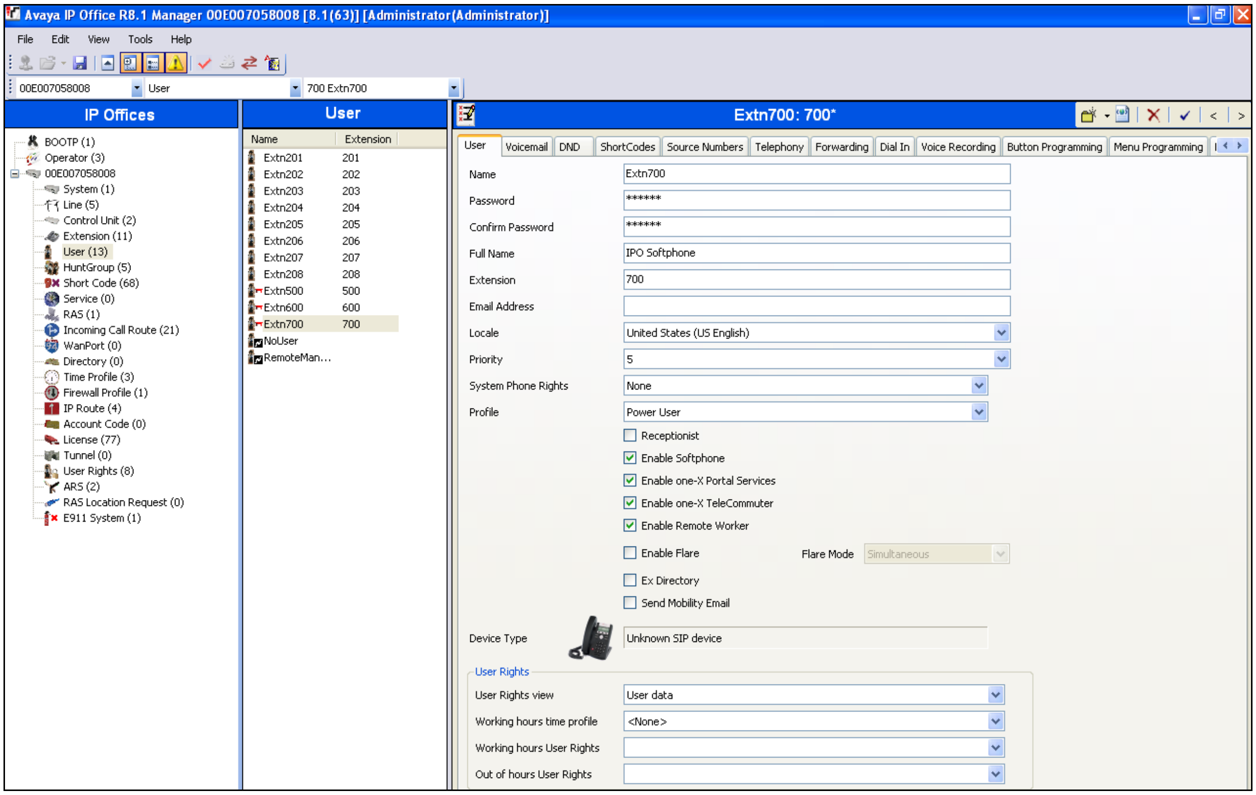
- The **IP Address** default value is used (**0.0.0.0**).
- Check the **Reserve Avaya IP endpoint license** box.
- In the reference configuration the **Codec Selection** parameter is set to **Custom**, and the same codec list used in **Section 5.3.6** is used.
- Other fields may retain default values.



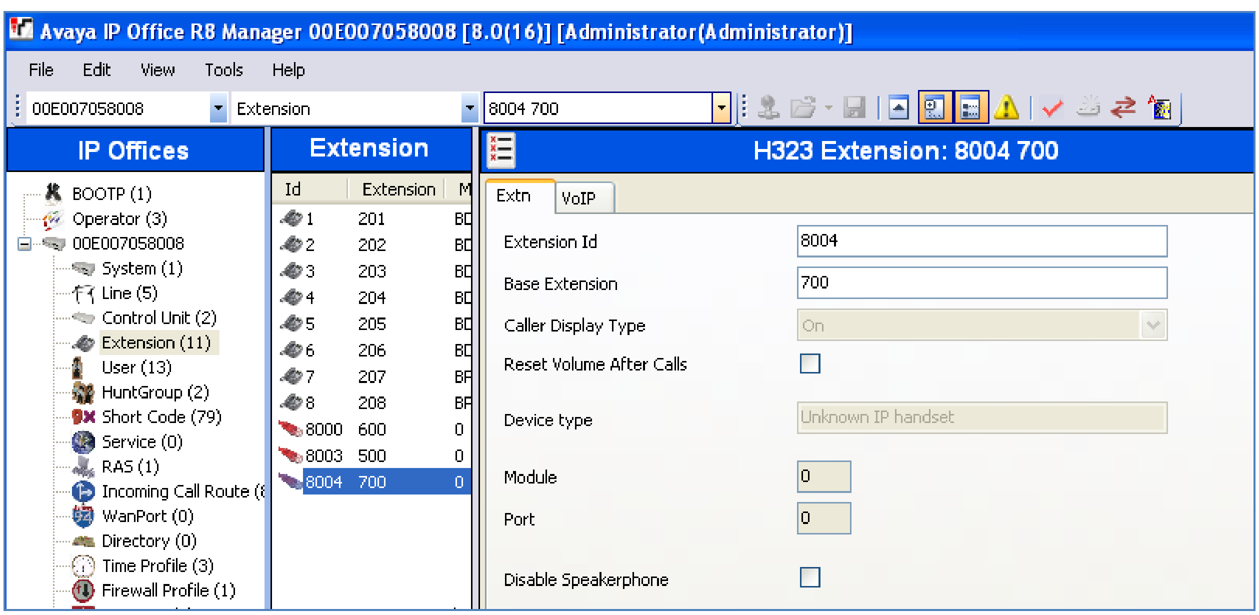
5.5.3.2 SIP Avaya IP Office Softphone

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

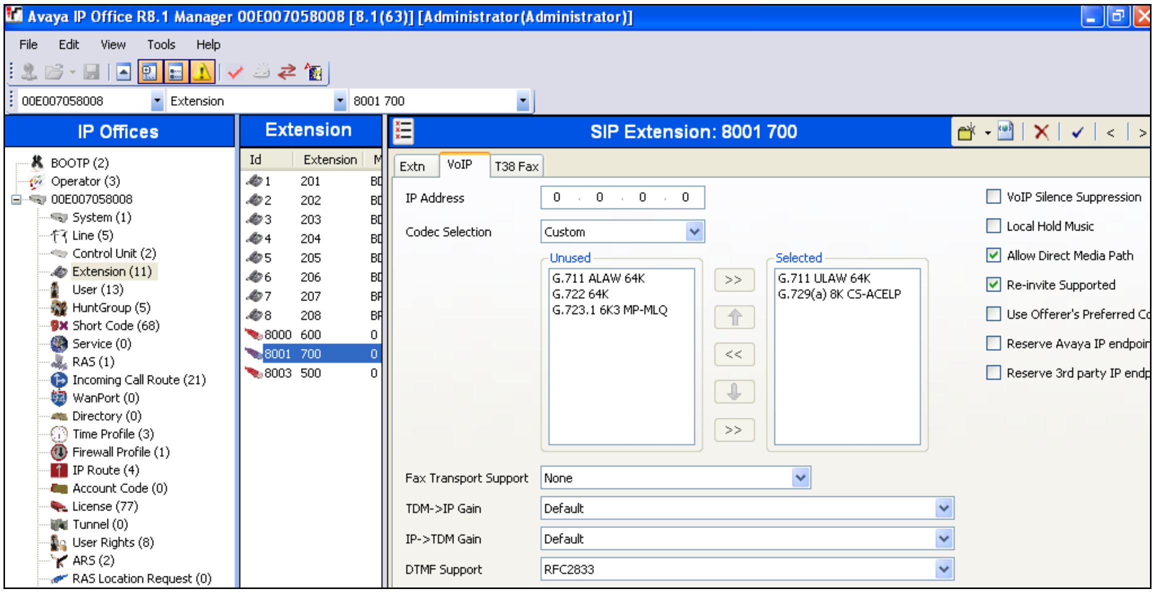
- **Defining a User**
  - **User** tab
    - The Avaya IP Office SoftPhone was provisioned as User **Extn700**.
    - The **Enable Softphone** box is checked, the other enabled advanced capabilities shown (e.g., Power User), are optional.
  - **SIP** tab
    - o **SIP Name** and **Contact** specifying the user's associated AT&T DNIS number (e.g., **0000031054**).
  - **Voicemail** tab
    - o User Extn700 also utilized the embedded Voicemail. The **Voicemail On** box is checked, and a Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters.
  - **Telephony→ Call Settings** tab.
    - o Check the **Call Waiting On** box to allow multiple call appearances and transfer operations.
  - **Telephony→ Supervisor Settings** tab
    - o The **Login Code** is the softphone login password.



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



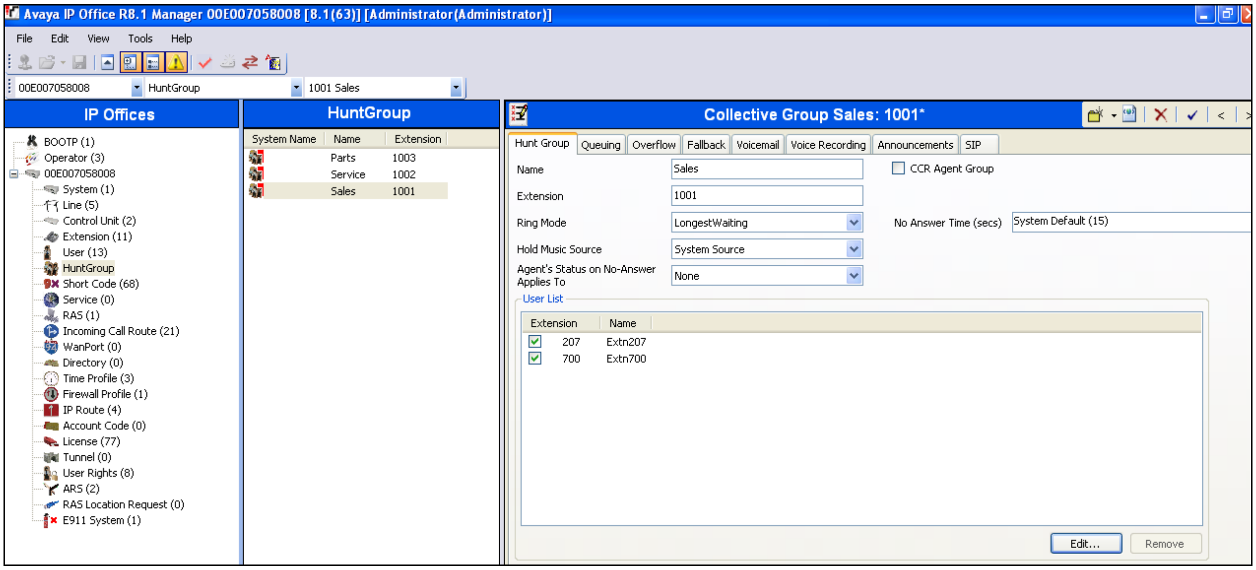
- **SIP Extension → VoIP** tab for extension 700.
    - The **IP Address** default value is used (**0.0.0.0**).
    - In the reference configuration the **Codec Selection** parameter is set to **Custom**, and the same codec list used in **Section 5.3.6** is used.
    - Other fields may retain default values.



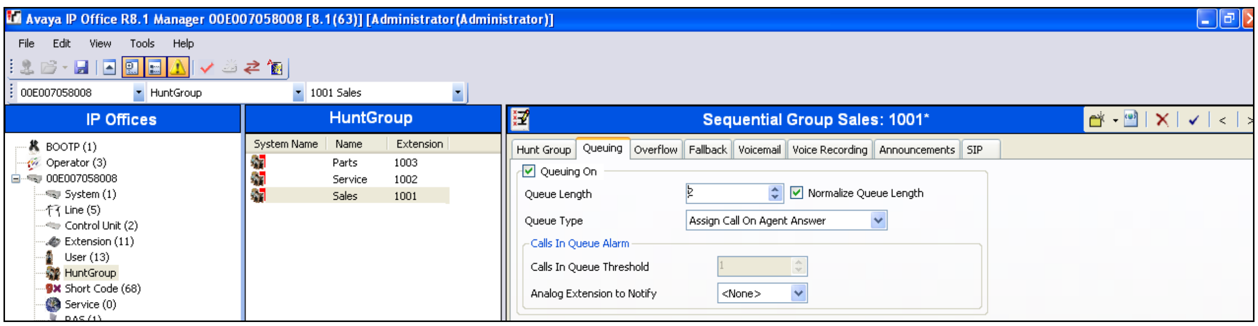
5.5.4. Hunt Groups

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

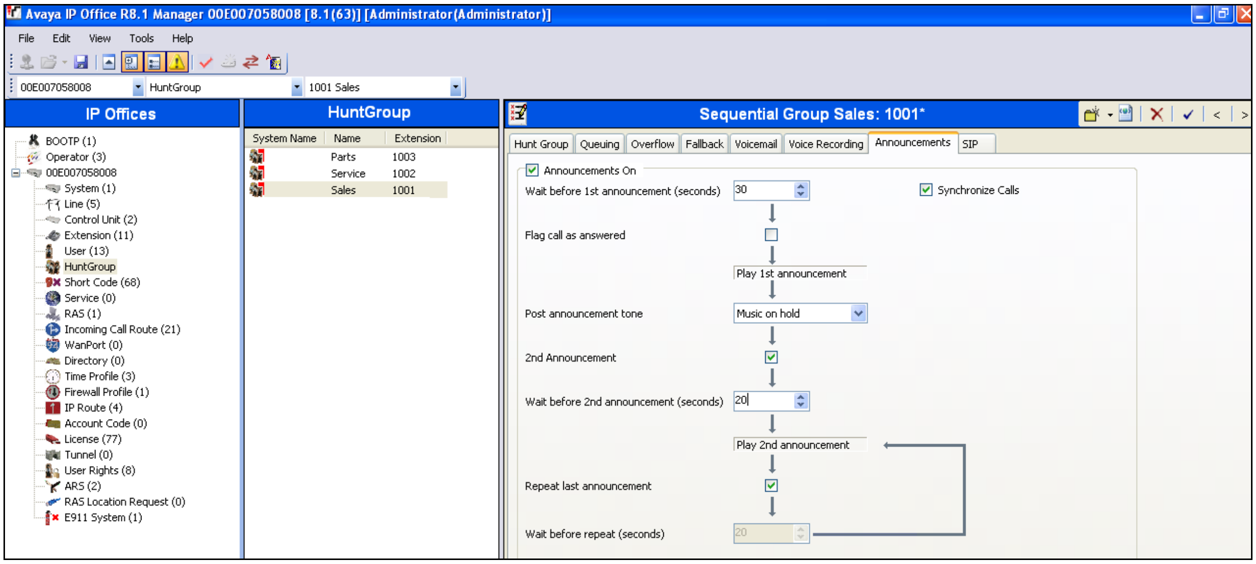
Step 1 - The following screen shows the **Hunt Group** tab for hunt group **Sales**. This hunt group was configured to contain the Analog telephone (Extn207), and the SIP Softphone Extn700. In the reference configuration, these telephones extensions are rung based on idle time, due to the **Ring Mode** setting **LongestWaiting**. Click the **Edit** button to select/deselect from the **User List** included in the Hunt Group from the list of available users.



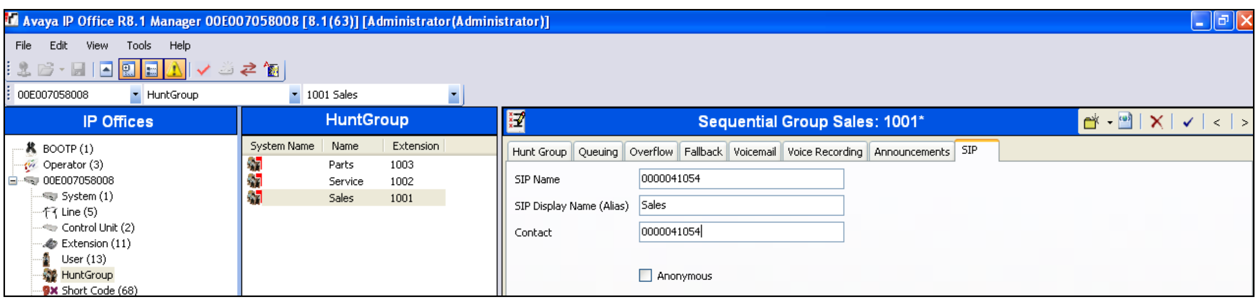
Step 2 - Under the **Queuing** tab, check the **Queuing On** box and set the **Queue Length** field to any desirable value. Use the default values for all the other fields.



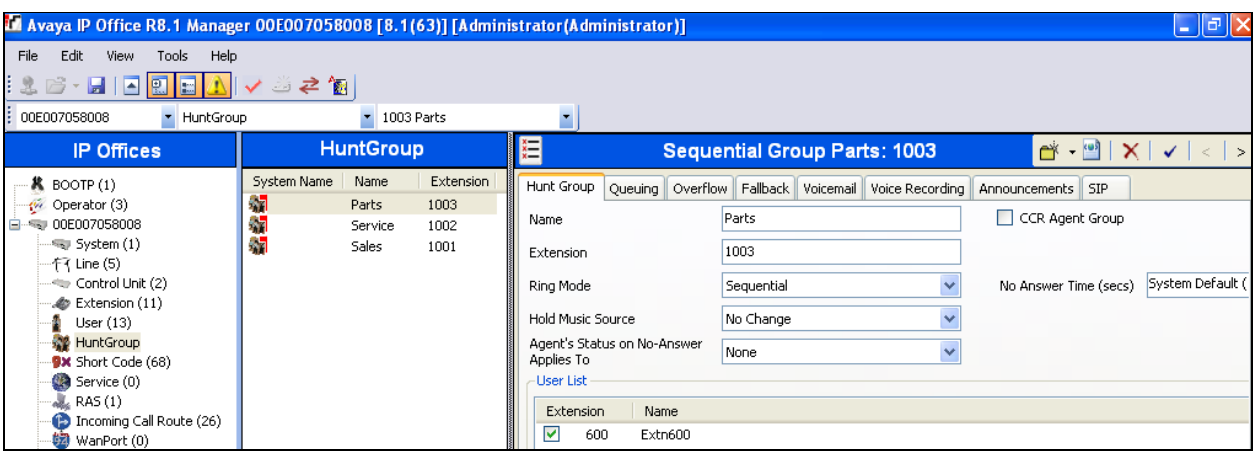
Step 3 - Under the **Announcements** tab, check the **Announcements On** box. The wait time can be set to any desirable value. Make sure that the **Synchronize Calls** box is checked. These announcements are played if an agent for a particular skill is unavailable.

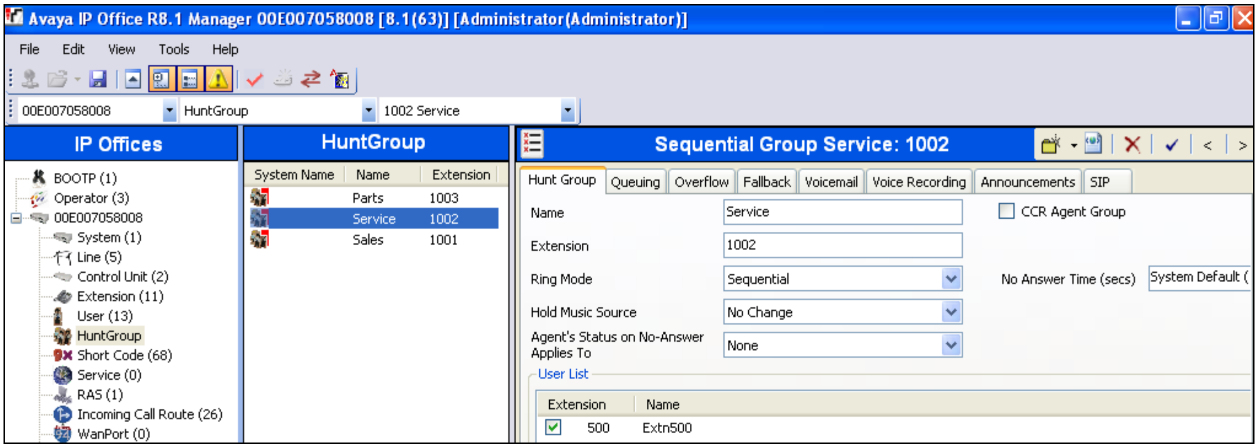


Step 4 - The following screen shows the **SIP** tab for hunt group **Sales**. The **SIP Name** and **Contact** are configured with the AT&T DNIS number **0000041054.** In **Section 5.7**, an **Incoming Call Route** will map **0000041054** to this hunt group.



Similarly, additional hunt groups **Parts** and **Service**, are created by following **Steps 1-4**.



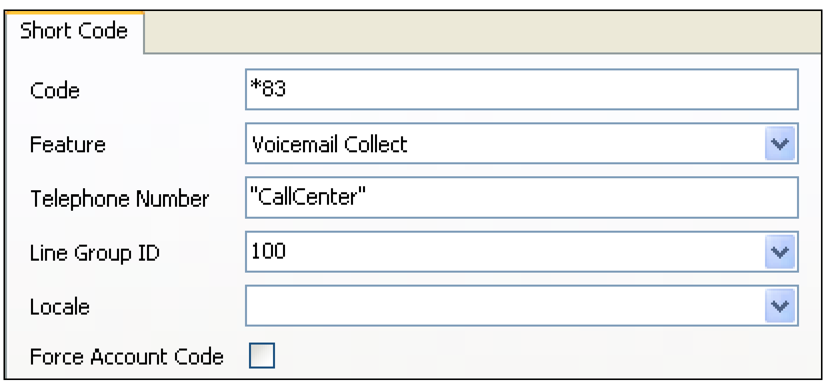


5.6. Short Codes

Avaya IP Office provides predefined Short Codes, however new Short Codes may be defined to match number strings to an action. To add a Short Code, right click on **Short Code** in the Navigation pane, and select **New.** To edit an existing Short Code, click **Short Code** in the Navigation pane, and the Short Code to be configured in the Group pane.

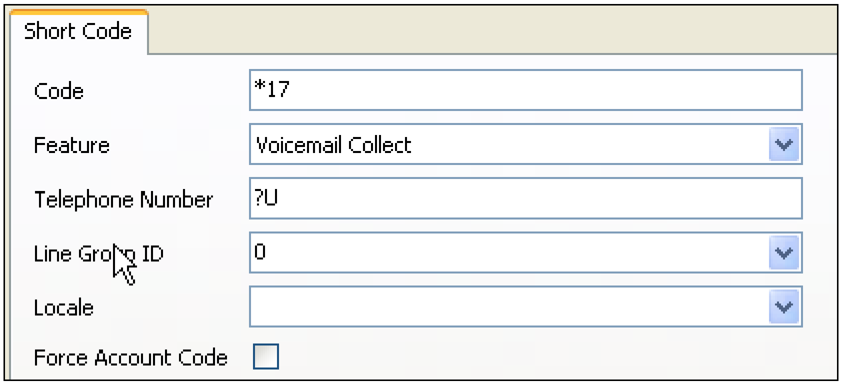
5.6.1. Call Center Access to Voicemail Pro

In the reference configuration, Call Center functionality is configured on Voicemail Pro (see **Section 5.8).** In order to access this functionality, short codes can be used. The following section shows the short code set to access this functionality.



5.6.2. Voicemail Access

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

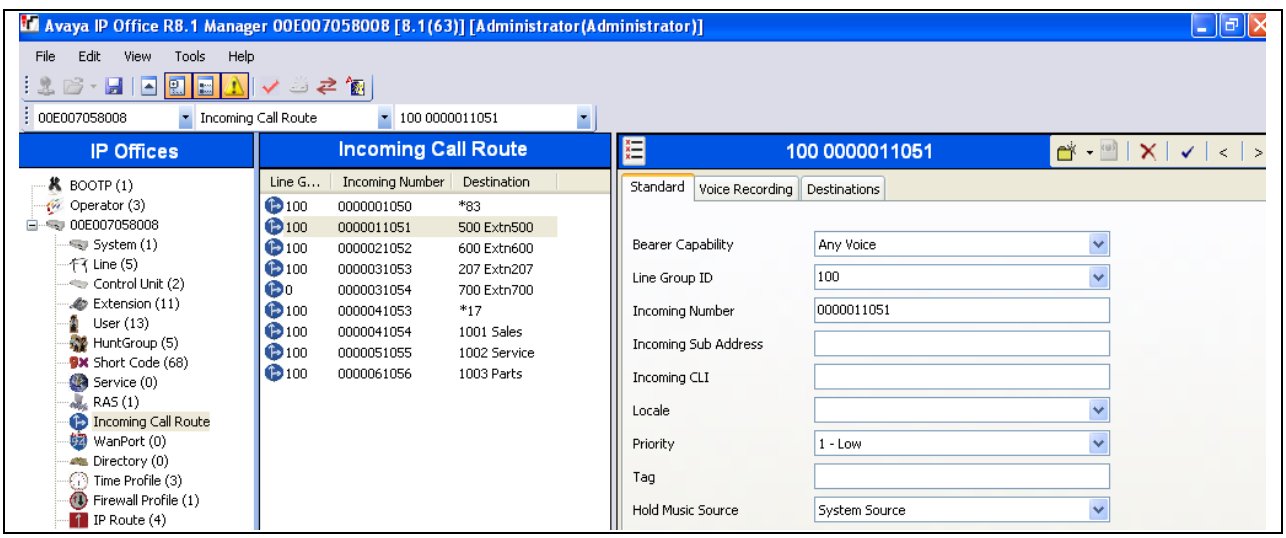


5.7. Incoming Call Routes

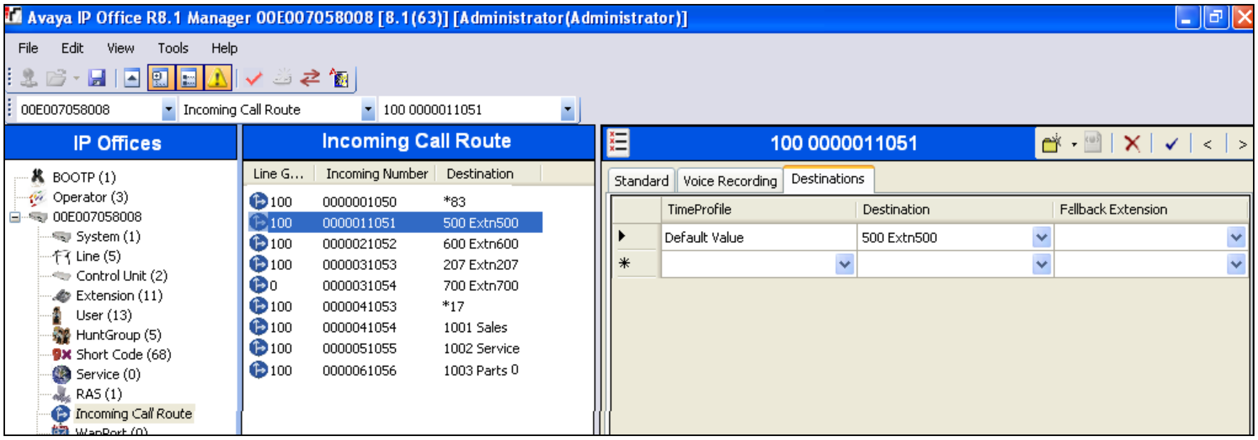
Each Incoming Call Route will map a specific AT&T DNIS number to a destination user, hunt group, or function on Avaya IP Office. To add an incoming call route, right click on **Incoming Call Route** in the Navigation pane, and select **New.** To edit an existing incoming call route, select **Incoming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

In the screen shown below, the incoming call route for **Incoming Number → 0000011051** is illustrated.

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



Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when AT&T delivers DNIS digits 000011051. In the reference configuration DNIS digits are associated with Avaya IP Office User **Extn500** (the 1608 H.323 telephone).



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

- 0000001050 →***83** (Voicemail Pro Call Center access Short Code)
- 0000041053 →***17** (Voicemail access Short Code)
- 0000041054 → **1001 Sales** (Sales Hunt Group)

Note that the **Destination** drop down menu may not contain all desired destinations. In these cases the desired destination may be manually typed into the **Destination** field.

5.8. Call Center Provisioning in Avaya Voicemail Pro

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

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

In the reference configuration a Call Center 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:

Step 1 – Hunt Groups **Sales**, **Service**, and **Parts** are created in IP Office (**Section 5.5.4)**.

Step 2 – Short Code ***83** is created in IP Office for Call Center access (**Section 5.6.1)**.

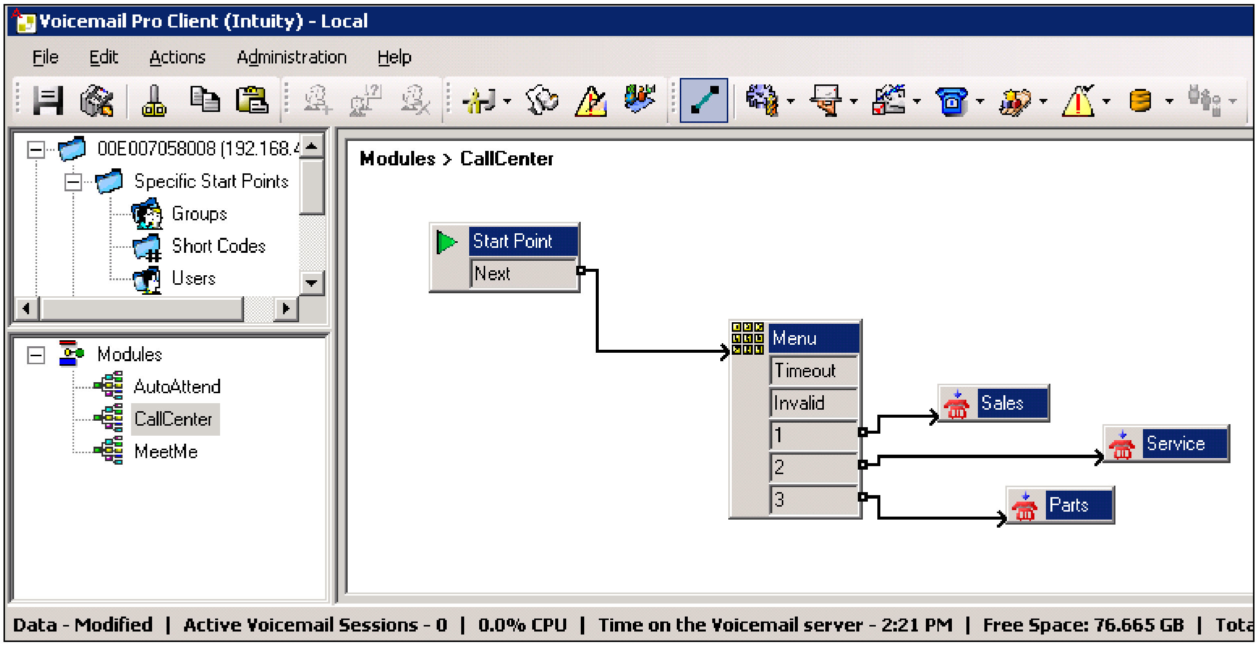
Step 3 - Incoming Call Route for DNIS digits **0000001050** is defined for access to the Call Center prompts (**Section 5.7)**.

Step 4 - Via the Voicemail Pro GUI interface on the Voicemail Pro platform:

- Open the **Voicemail Pro Client** application.
- Create a **Start Point** by right clicking on **Modules** and selecting **Add.**

- Enter a name (e.g., **CallCenter**) and click on **Ok**. The new Start Point "CallCenter" will appear under Modules, and a Start Point icon will appear in the work area. 

- Click on the **Start Point** icon to activate the options bar at the top of the screen. From the options bar, 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. **General** tab → **Token Name** = **Start Point**
        2. Click **Ok**
    ii. Double click the **Menu** icon.
        1. **General** tab, **Token Name** = **Menu**
        2. **Entry Prompts** tab → Create an **Entry Prompt** that will tell the caller what digits to press to reach Sales, Service, and Parts (e.g., **attendant.wav**). To modify an existing recording, double click on the .wav file and rerecord. If no .wav files exist, double click on the iconto open the .wav editor.
        3. **Touch Tone** tab:
            a. Select **1, 2,** and **3** as the possible entry digits.
            b. Select **4** for **No of Retries**.
            c. Check the **Timeout** and **Invalid Entry** options.
        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. Select and place two more Transfer Icons (these will be used for Sales, Service, and Parts).
    i. Double click on the first **Transfer** icon ("**Sales**")
        1. **General** tab → **Token Name** = **Sales**
        2. **Specific** tab → **Destination** → **Mailbox** → **Sales** → **Ok**
    ii. Double click on the second **Transfer** icon **("Service**").
        1. **General** tab → **Token Name** = **Service**
        2. **Specific** tab → **Destination** → **Mailbox** → **Service** → **Ok**
    iii. Double Click on the third **Transfer** icon ("**Parts**").
        1. **General** tab, **Token Name** = **Parts**
        2. **Specific** tab, **Destination** → **Mailbox** → **Parts** → **Ok**
- 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).



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

When the associated AT&T IPTF number is called from PSTN (e.g., 0000001050), 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 Frequency

In the reference configuration, Avaya IP Office periodically checks the health of the SIP Line by sending a SIP OPTIONS message. In **Section 5.4**, the SIP Line to the Avaya SBCE/AT&T is shown with the **Check OOS** box checked. The Avaya SBCE will pass the OPTIONS message on to AT&T. If there is no response, Avaya IP Office can mark the trunk out of service. Once the problem with the SIP Line is resolved, the SIP OPTIONS maintenance will automatically bring the link back to the in-service state. In addition, for secure networks, the periodic sending of OPTIONS by Avaya IP Office *may* serve to keep network Firewall "pinholes" open preventing the blockage of inbound traffic to Avaya IP Office.

In the reference configuration, Avaya IP Office sourced SIP OPTIONS every 120 seconds, (the value configured in the **Binding Refresh Time** provisioned in **Section 5.3.2**). This interval may be adjusted as required.

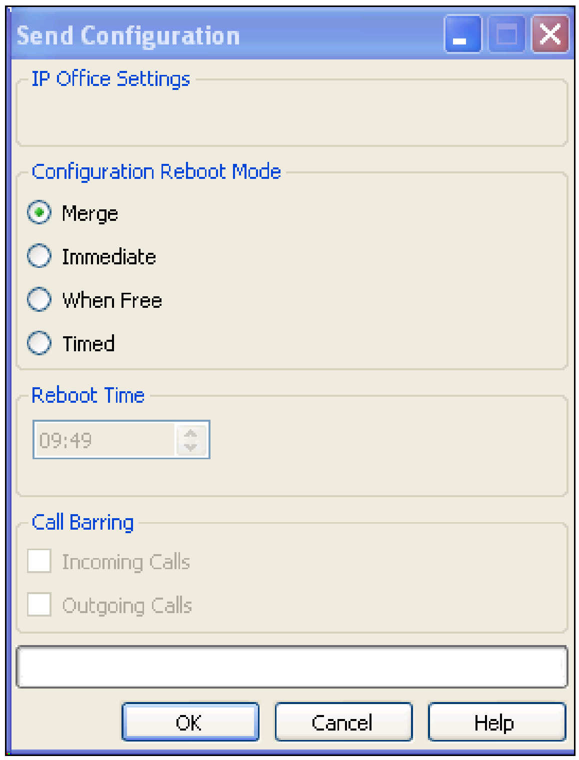
Note – In the reference configuration Avaya IP Office sent OPTIONS to the AT&T IPTF service Border Element and AT&T responded with *405 Method Not Allowed* (which is the expected response). That response is sufficient for Avaya IP Office to consider the connection up.

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

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

Click **OK** to execute the save.



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

6. Configure Avaya Session Border Controller for Enterprise

As described in **Section 3**, the Avaya SBCE used in the reference configuration ran on a CAD-0208 platform. This solution is extensible to other Avaya Session Border Controller for Enterprise platforms as well.

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 **[7]** and **[8]** for additional information.

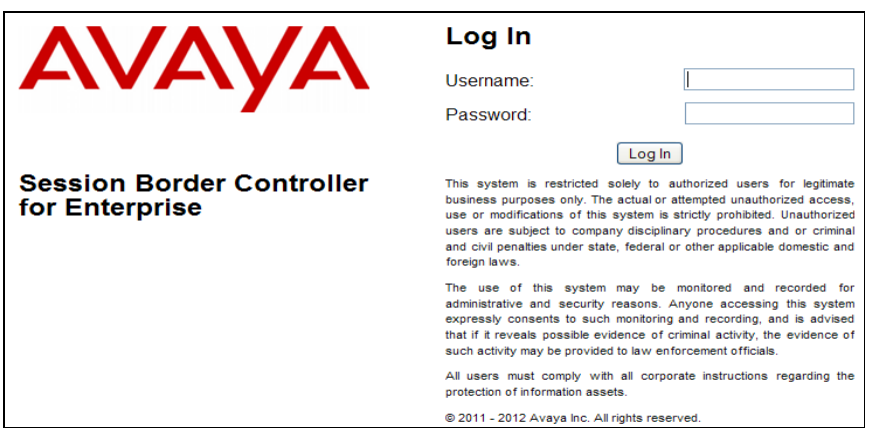
IMPORTANT! – During the Avaya SBCE installation, the Avaya SBCE Management interface, (labeled "E3" on the CAD-0208 platform), must be provisioned on a different subnet than either of the Avaya SBCE private and public network interfaces (e.g., A1 and B1). If this is not the case, contact your Avaya representative 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 IP Office LAN1 interface (192.168.42.1). The connection to AT&T used the Avaya SBCE public interface B1 (IP address 192.168.64.130).

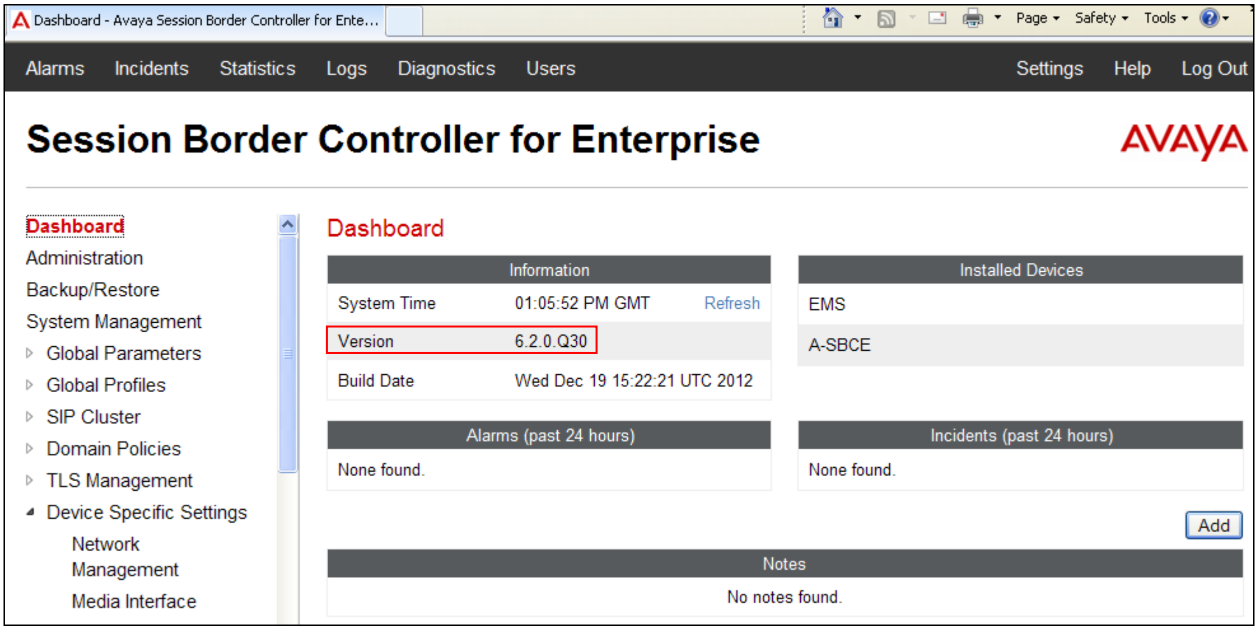
6.2. Log into the Avaya SBCE

The follow provisioning is performed via the Avaya SBCE GUI interface, using the "E3" management LAN connection on the chassis.

A. 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).
B. Enter the login ID and password.



C. The main menu window will open. Note that the installed software version is displayed.



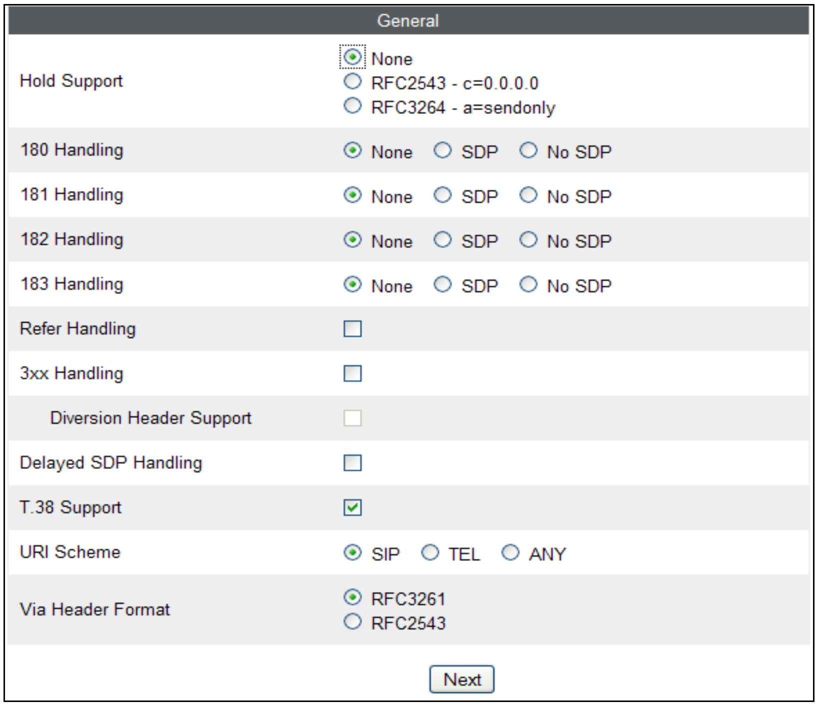
6.3. Global Profiles

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

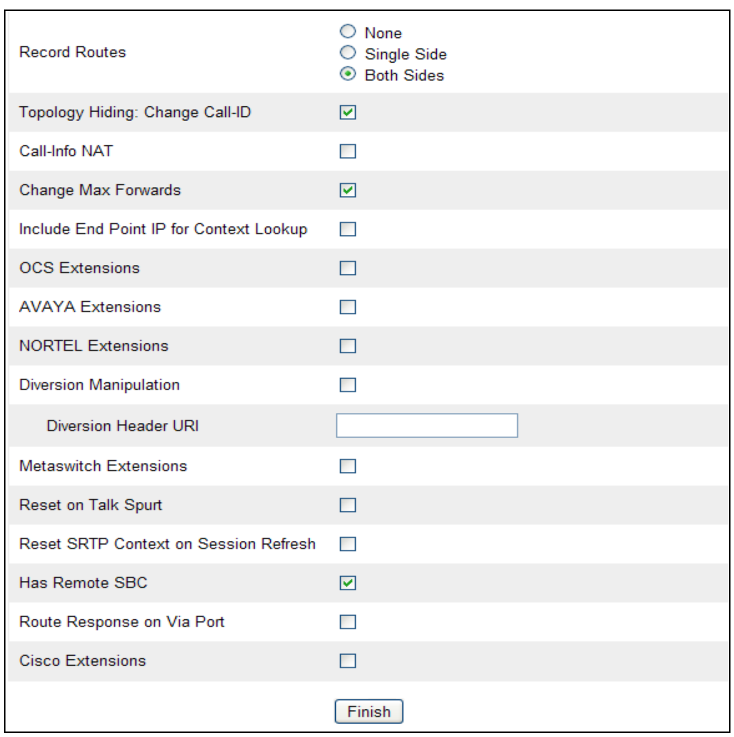
6.3.1. Server Interworking – to Avaya IP Office

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

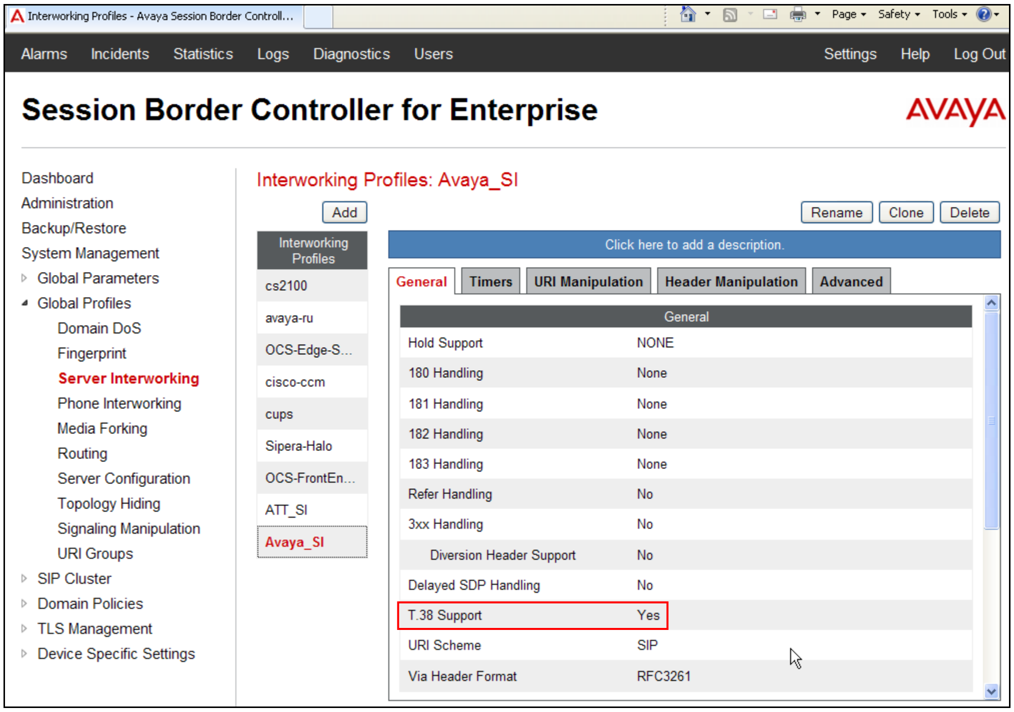
1. Select **Global Profiles** from the menu on the left-hand side.
2. Select **Server Interworking**.
3. Select the **Add** button (not shown) and the **Profile** name window will open (not shown).
4. Enter profile name: (e.g., **Avaya_SI**), and click **Next**.
5. The **General** screen will open.
    a. Check **T38 Support**
    b. All other options can be left at default
    c. Select **Next**



6. On the **Privacy/DTMF** window (not shown), select **Next** to accept default values.
7. On the **Timers** tab →**SIP Timers/Transport Timers** window (not shown), select **Next** to accept default values.
8. On the **Advanced** tab, accept the default values, and click **Finish**.



The following screenshot shows the completed **General** tab form.



6.3.2. Server Interworking – to AT&T

Repeat the steps shown in **Section 6.3.1** to add an Interworking Profile for the connection to AT&T via the public network.

1. Select **Global Profiles** from the menu on the left-hand side.
2. Select **Server Interworking**.
3. Select **Add Profile**.
4. On the **General** Tab (not shown):
    a. Enter a profile name**:** (e.g., **ATT_SI**).
    b. Check **T38 Support.**
    c. All other options can be left at default.
    d. Select **Next.**
5. At the **Privacy** tab (not shown), select **Next** to accept default values.
6. At the **Interworking Profile** tab (not shown), select **Next** to accept default values.
7. On the last screen (**Advanced** options, not shown), accept the default values, and click **Finish**.

6.3.3. Routing – to Avaya IP Office

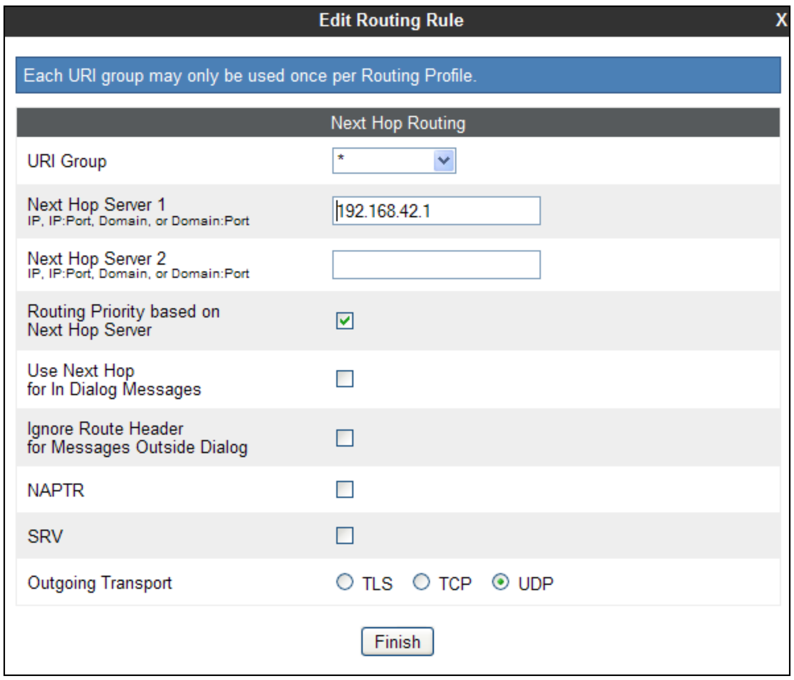
The Routing Profile allows you to manage parameters related to routing SIP signaling messages. This provisioning defines the Routing Profile for the connection to Avaya IP Office.

1. Select **Global Profiles** from the menu on the left-hand side.
2. Select the **Routing** tab (not shown).
3. Select **Add Profile** (not shown).

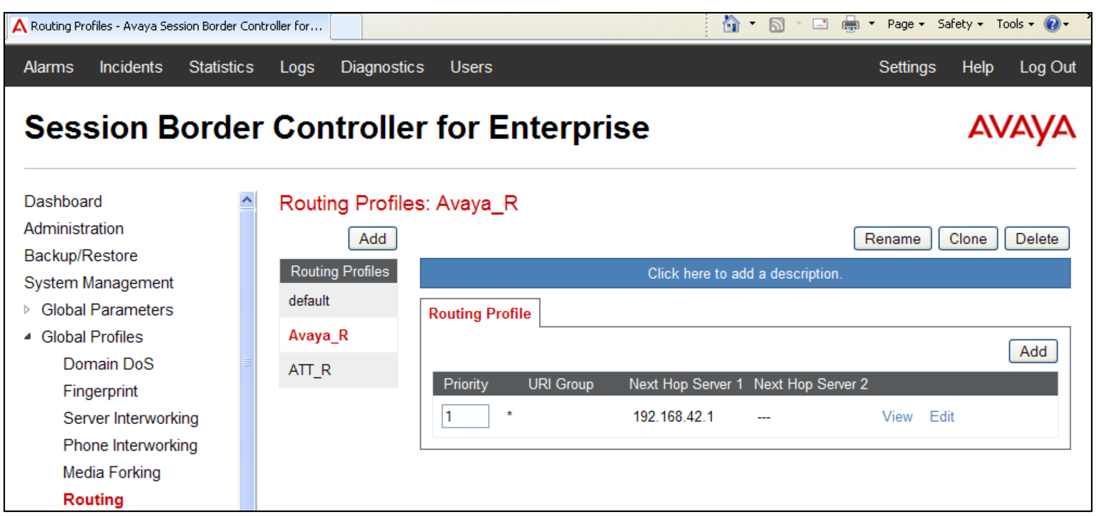
4. Enter **Profile Name**: (e.g., **Avaya_R)**.
5. Click **Next** and enter:
   a. **Next Hop Server 1: 192.168.42.1** (Avaya IP Office LAN1 IP address)
   b. Verify **Routing Priority Based on Next Hop Server** is selected (default).
   c. **Outgoing Transport**: **UDP**

      Note – UDP is the recommended protocol to use on the connection between the Avaya SBCE and IP Office. However TCP may be used if necessary.

   d. Accept remaining default values
6. Click **Finish**.



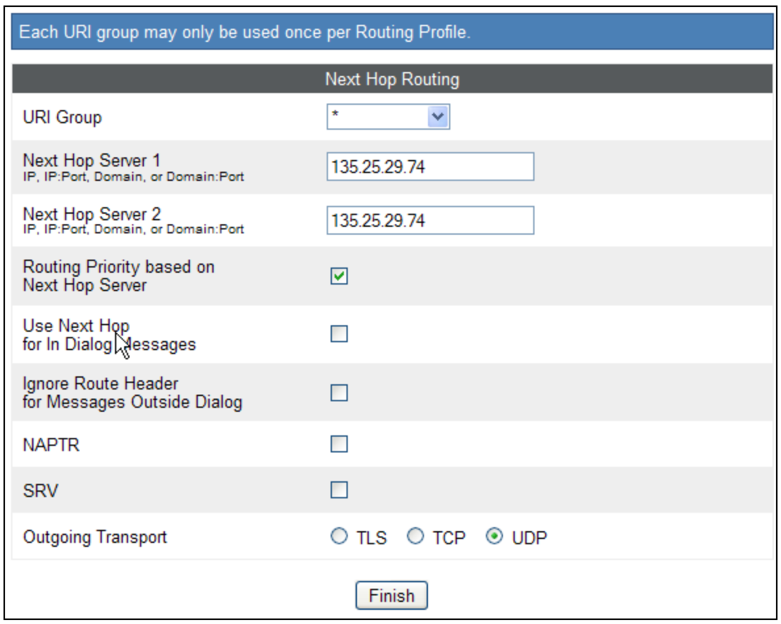
The following screenshot shows the completed **Routing Profiles** form.



6.3.4. Routing – to AT&T

Repeat the steps in **Section 6.3.3** to add a Routing Profile for the connection to AT&T.

1. Select **Global Profiles** from the menu on the left-hand side.
2. Select the **Routing** tab.
3. Select **Add Profile**.
4. Enter Profile Name: (e.g., **ATT_R)**.
5. Click **Next**, then enter the following:
    a. **Next Hop Server 1: 135.25.29.74** (AT&T Border Element IP address)
    b. Verify **Routing Priority Based on Next Hop Server** is selected (default).
    c. **Outgoing Transport**: **UDP**
6. Click **Finish**.



6.3.5. Server Configuration – To Avaya IP Office

This section defines the Server Configuration for the connection to Avaya IP Office. The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow you to configure and manage various SIP call server-specific parameters such as port assignments, IP Server type, heartbeat signaling parameters and some advanced options.

1. Select **Global Profiles** from the menu on the left-hand side.
2. Select **Server Configuration**.
3. Select **Add Profile** and the **Profile Name** window will open (not shown). Enter a Profile Name (e.g., **Avaya_SC**) and select **Next**.
4. The **Add Server Configuration Profile - General** window will Open (not shown).
    a. Select Server Type**: Call Server**
    b. **IP Address: 192.168.42.1** (Avaya IP Office LAN1 IP Address)
    c. **Supported Transports**: Check **UDP**

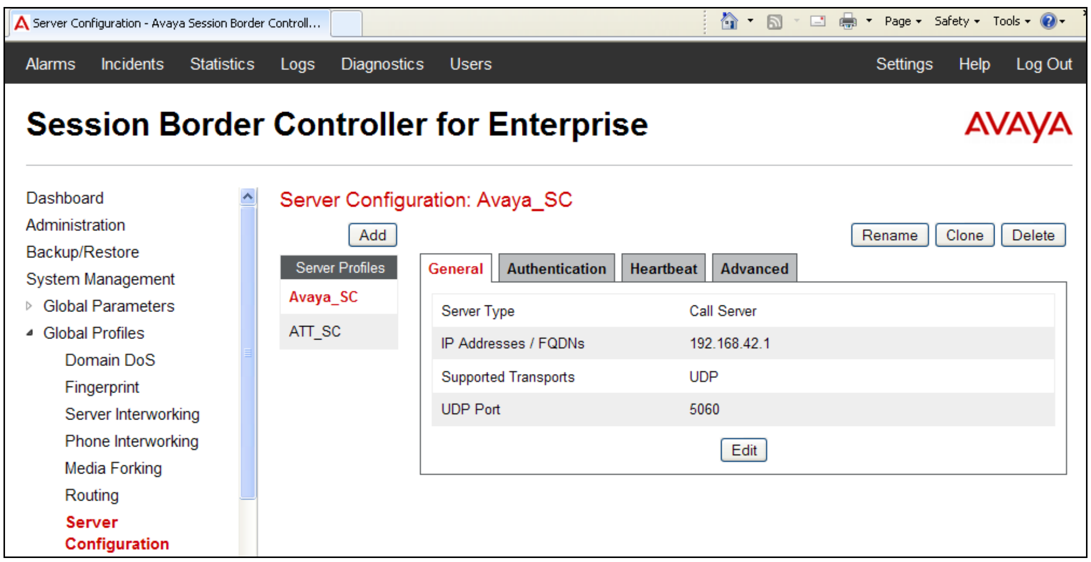
d. **UDP Port: 5060**

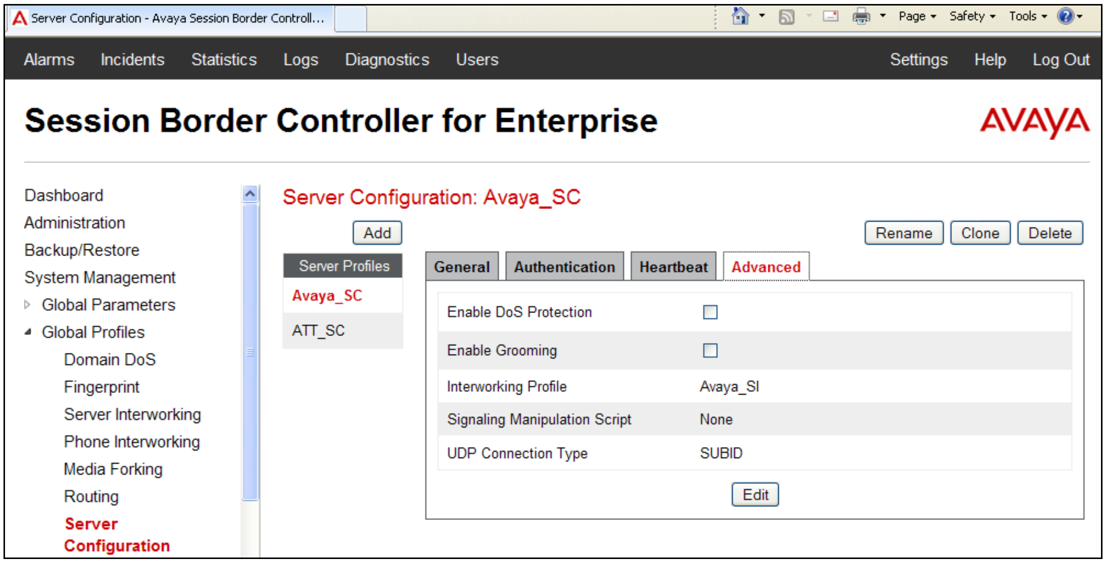
Note – UDP is the recommended protocol to use on the connection between the Avaya SBCE and IP Office. However TCP may be used if necessary.

e. Select **Next**

5. The **Add Server Configuration Profile - Authentication** window will open (not shown).
   a. Select **Next** to accept default values.
6. The **Add Server Configuration Profile - Heartbeat** window will open (not shown).
   a. Select **Next** to accept remaining default values.
7. The **Add Server Configuration Profile - Advanced** window will open.
   a. Select **Avaya_SI** (created in **Section 6.3.1**), for **Interworking Profile**.
   b. In the **Signaling Manipulation Script** field select **None**.
   c. Select **Finish**.

The following screen shots show the completed **General** and **Advanced** tabs.



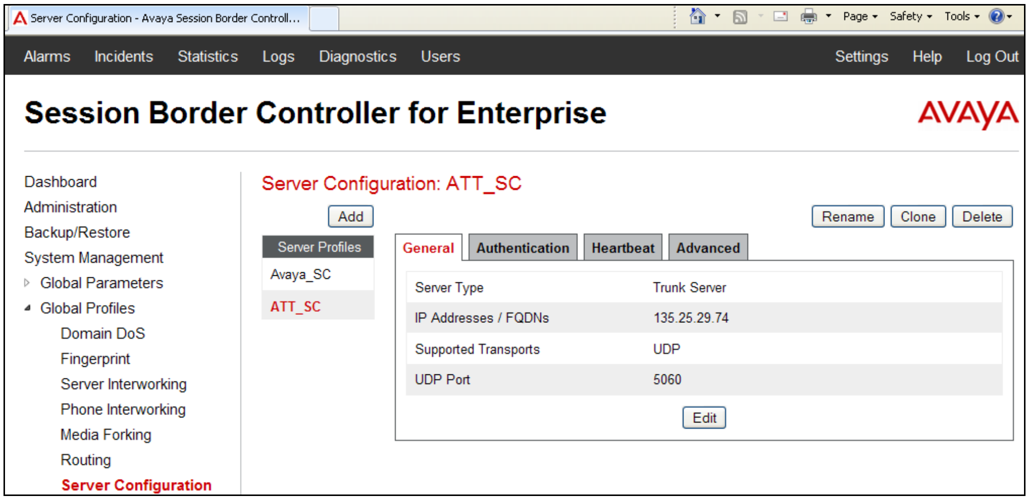


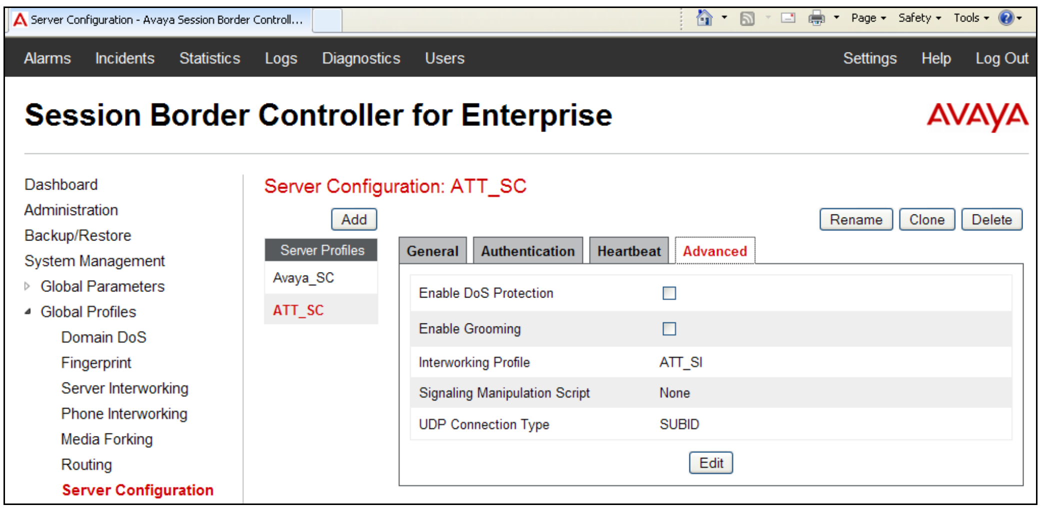
6.3.6. Server Configuration – To AT&T

Repeat the steps in **Section 6.3.5** to create a Server Configuration for the connection to AT&T.

1. Select **Global Profiles** from the menu on the left-hand side.
2. Select **Server Configuration**.
3. Select **Add Profile** and the **Profile Name** window will open (not shown). Enter a Profile Name (e.g., **ATT_SC**) and select **Next**.
4. The **Add Server Configuration Profile - General** window will open (not shown).
    a. Select Server Type**: Trunk Server**
    b. **IP Address: 135.25.29.74** (AT&T Border Element IP Address)
    c. **Supported Transports**: Check **UDP**
    d. **UDP Port: 5060**
    e. Select **Next**.
5. The **Add Server Configuration Profile - Authentication** window will open (not shown).
    a. Select **Next** to accept default values.
6. The **Add Server Configuration Profile - Heartbeat** window will open (not shown).
    a. Select **Next** to accept default values.
7. The **Add Server Configuration Profile - Advanced** window will open.
    d. Select **ATT_SI** (created in **Section 6.3.2**), for **Interworking Profile**.
    e. In the **Signaling Manipulation Script** field select **None**.
    a. Select **Finish**.

The following screen shots show the completed **General** and **Advanced** tabs.



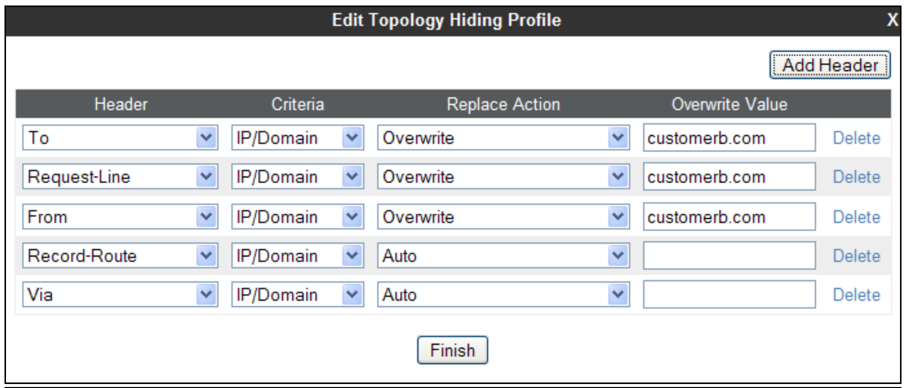


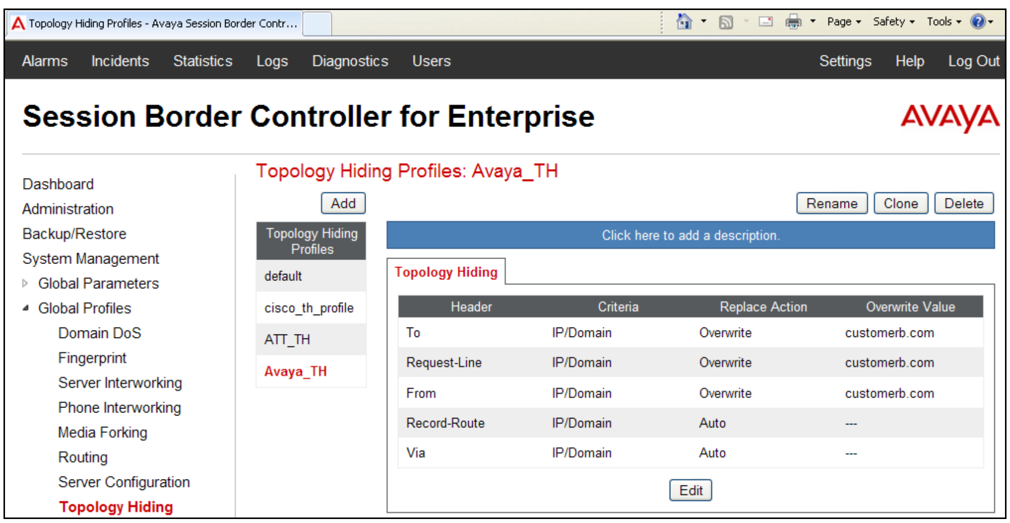
6.3.7. Topology Hiding – Avaya IP Office

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

1. Select **Global Profiles** from the menu on the left-hand side.
2. Select **Topology Hiding**.
3. Click **default** profile and select **Clone Profile**.
4. Enter Profile Name: (e.g., **Avaya_TH**)
5. For the Header **To**,
    a. In the **Criteria** column select **IP/Domain**
    b. In the **Replace Action** column select**: Overwrite**
    c. In the **Overwrite Value** column**: customerb.com**
6. For the Header **Request Line**,
    a. In the **Criteria** column select **IP/Domain**
    b. In the **Replace Action** column select**: Overwrite**
    c. In the **Overwrite Value** column**: customerb.com**

7. For the Header **From**,
    a. In the **Criteria** column select **IP/Domain**
    b. In the **Replace Action** column select**: Overwrite**
    c. In the **Overwrite Value** column**: customerb.com**
8. Let the remaining fields default.
9. Click **Finish**.

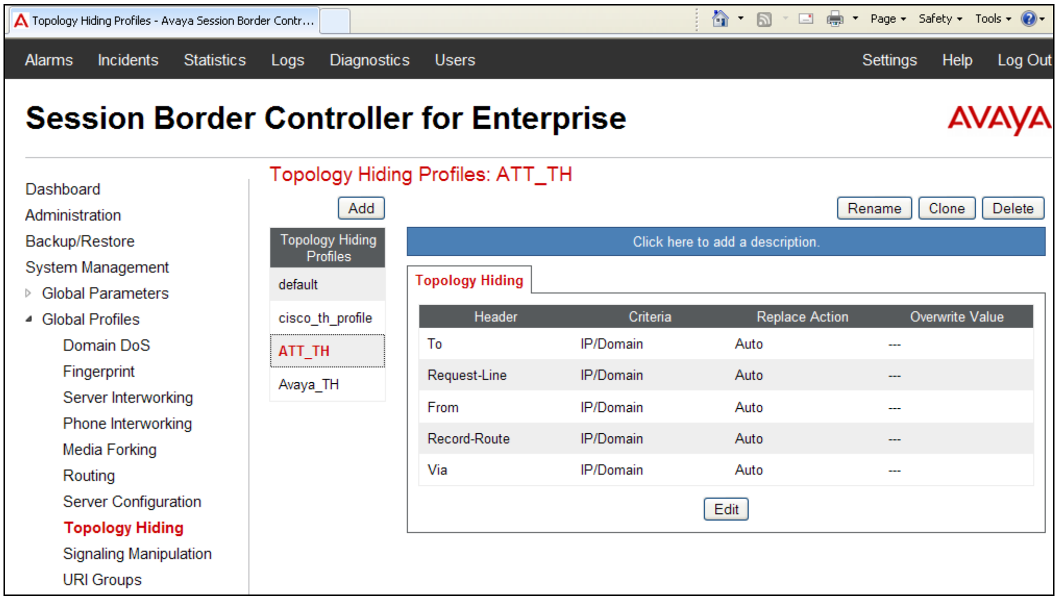




6.3.8. Topology Hiding – AT&T

Repeat the steps in **Section 6.3.7** to create a Topology Hiding Profile for the connection to AT&T.

1. Select **Global Profiles** from the menu on the left-hand side.
2. Select **Topology Hiding**.
3. Click **default** profile and select **Clone Profile**.
4. Enter Profile Name: (e.g., **ATT_TH**).
5. Let all **Replace Action** default to **Auto**.
6. Click **Finish**.



6.3.9. Signaling Manipulation

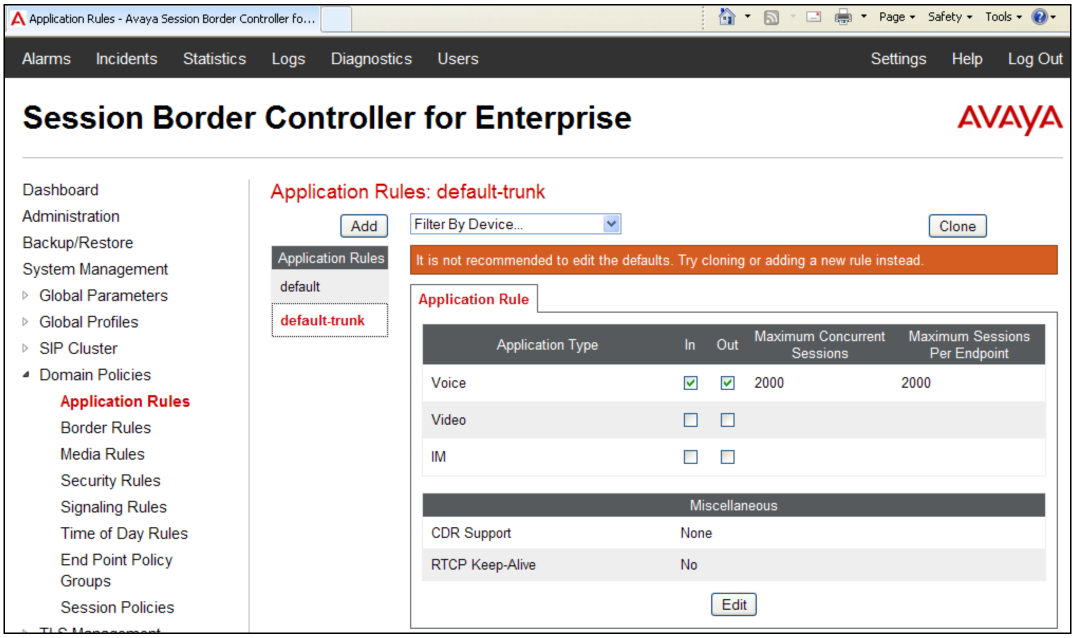
The Avaya SBCE can manipulate inbound and outbound SIP headers. However no SIP header manipulations were required in the reference configuration.

6.4. Domain Policies

The Domain Policies feature allows you 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. These criteria can be used to trigger different policies which will apply on call flows, change the behavior of the call, manipulate SIP headers, and make sure the call does not violate any of the policies.

6.4.1. Application Rules

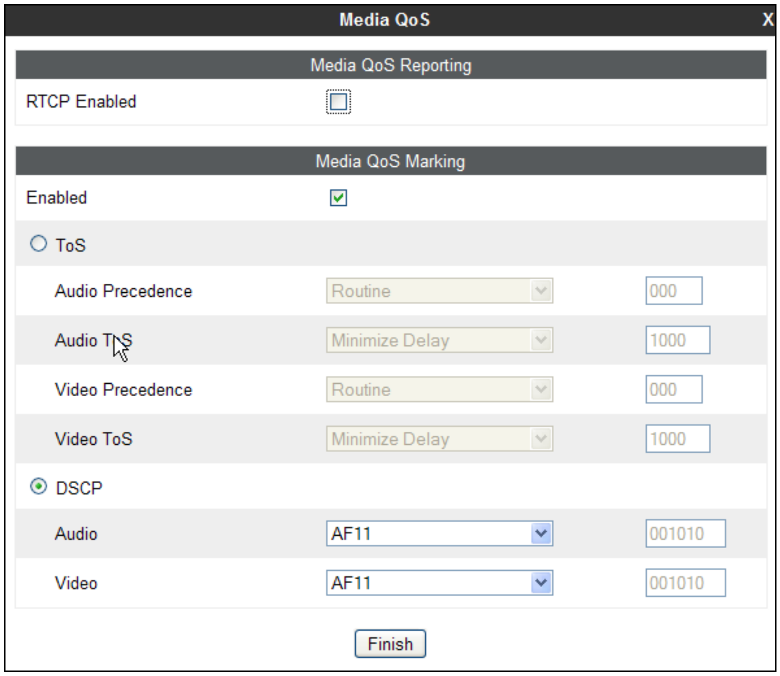
1. Select **Domain Policies** from the menu on the left-hand side
2. Select the **Application Rules**
3. Select the **default** Rule
4. Select **Clone Rule** button
    a. Name**: default-trunk**
    b. Click **Finish**
5. Highlight the rule just created: **default-trunk**
    a. Click the **Edit** button
    b. In the **Voice** row:
        i. Change the **Maximum Concurrent Sessions** to **2000**
        ii. Change the **Maximum Sessions per Endpoint** to **2000**



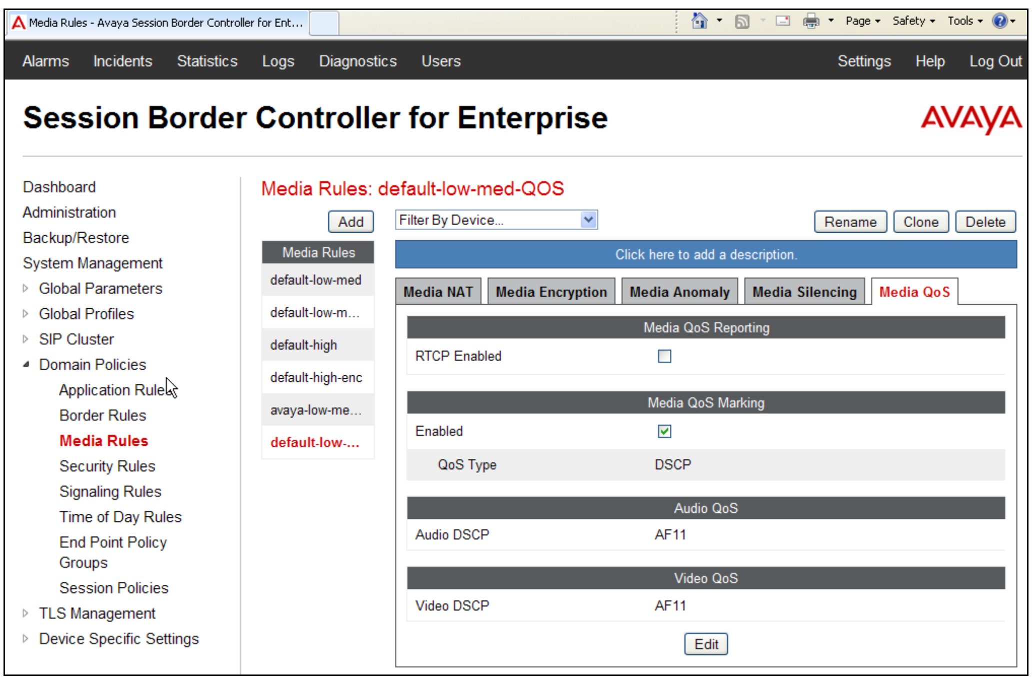
6.4.2. Media Rules

This Media Rule will be applied to both directions and therefore, only one rule is needed.

1. Select **Domain Policies** from the menu on the left-hand side menu (not shown).
2. Select the **Media Rules** (not shown).
3. The Media Rules window will open (not shown). From the Media Rules menu, select the **default-low-med** rule
4. Select **Clone Rule** button
   a. Name**: default-low-med-QOS**
   b. Click **Finish**
5. Highlight the rule just created from the Media Rules menu: **default-low-med-QOS**
   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 - Enabled**
   d. Select the **DSCP** box
   e. **Audio:** Select **AF11** from the drop-down
   f. **Video**: Select **AF11** from the drop-down
6. Click **Finish**



The next screen shot shows the completed **Media Rules** window.

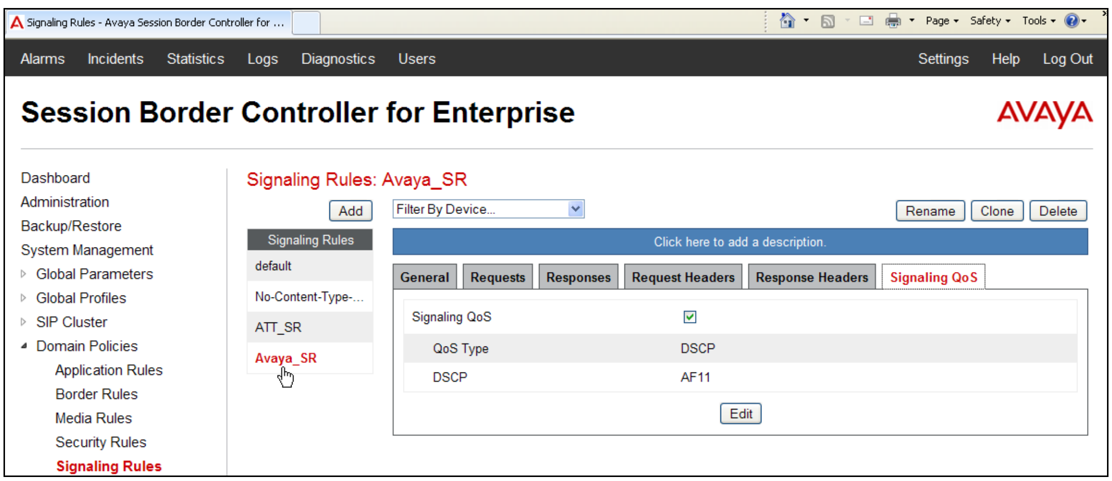


6.4.3. Signaling Rules

Signaling Rules may be used to remove or block various SIP headers. However no SIP header manipulations were required in the reference configuration.

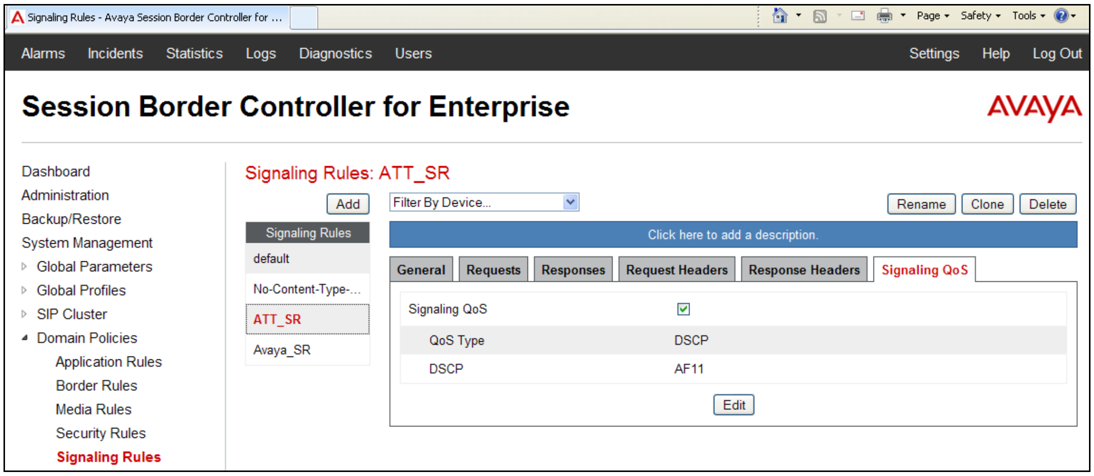
6.4.3.1 Avaya – Signaling QOS

1. Select **Domain Policies** from the menu on the left-hand side menu (not shown).
2. Select the **Signaling Rules** (not shown).
3. The Signaling Rules window will open (not shown). From the Signaling Rules menu, select the **default** rule.
4. Select **Clone Rule** button
   - Enter a name**: Avaya_SR**
   - Click **Finish**
5. Highlight the **Avaya_SR** rule created in **Step 4** and enter the following:
   - Select the **Signaling QOS** tab (not shown).
   - Click the **Edit** button and the **Signaling QOS** window will open.
   - Verify that **Signaling QOS** is selected.
   - Select **DCSP**.
   - Select **Value** = **AF11**.
2. Click **Finish**



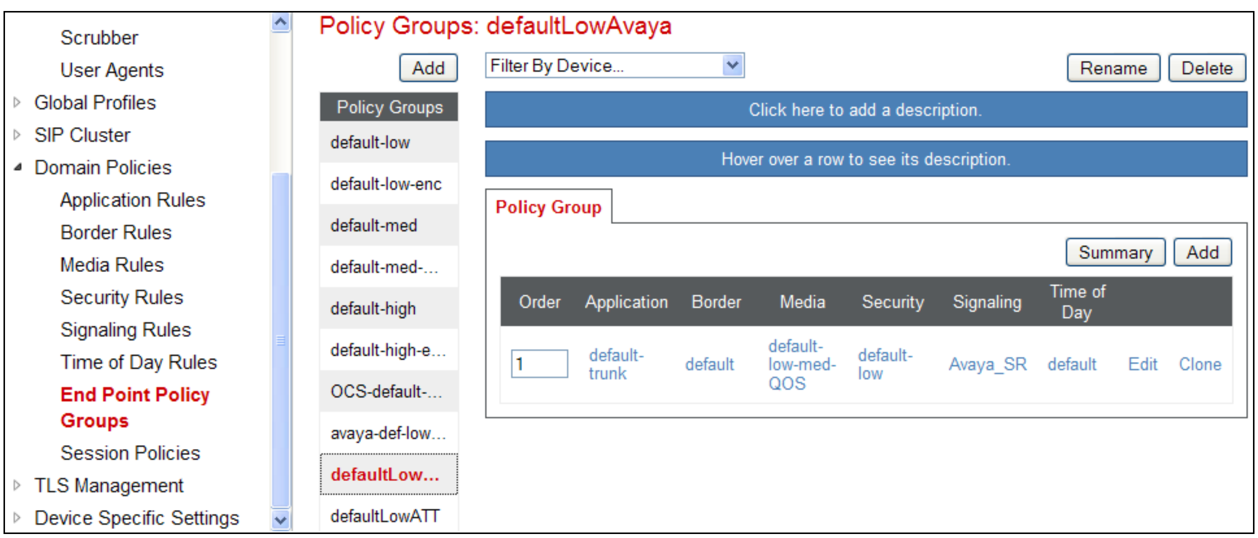
6.4.3.2 AT&T – Signaling QOS

1. Select **Domain Policies** from the menu on the left-hand side menu (not shown).
2. Select the **Signaling Rules** (not shown).
3. The Signaling Rules window will open (not shown). From the Signaling Rules menu, select the **default** rule.
4. Select **Clone Rule** button
   - Enter a name**: ATT_SR**
   - Click **Finish**
5. Highlight the **ATT_SR** rule created in **Step 4** and enter the following:
   - Select the **Signaling QOS** tab (not shown).
   - Click the **Edit** button and the **Signaling QOS** window will open.
   - Verify that **Signaling QOS** is selected.
   - Select **DCSP**.
   - Select **Value** = **AF11**.
3. Click **Finish**



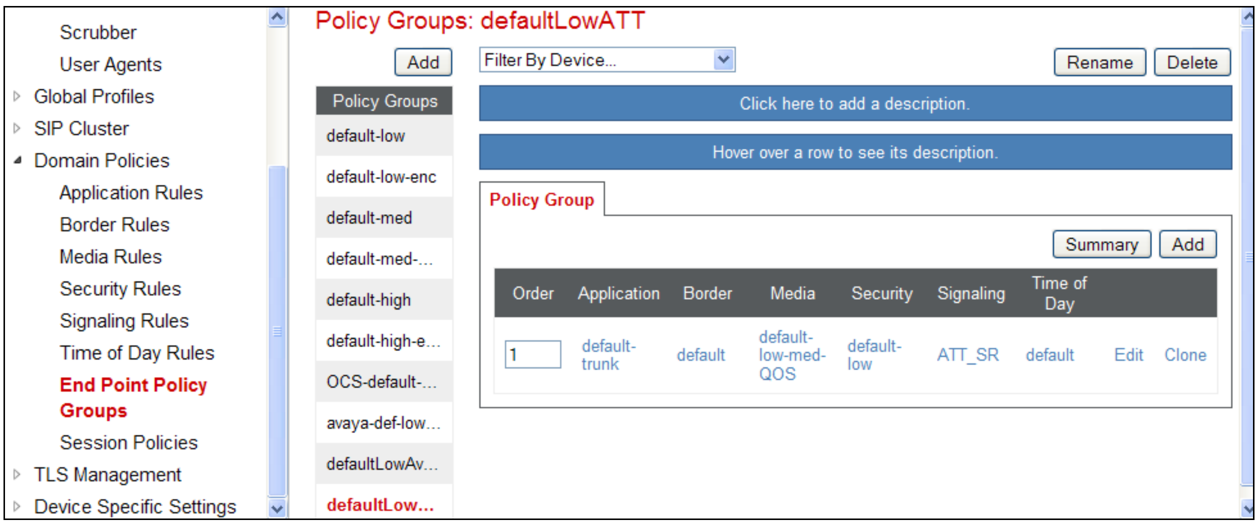
6.4.4. Endpoint Policy Groups – Avaya IP Office

1. Select **Domain Policies** from the menu on the left-hand side
2. Select **End Point Policy Groups**
3. Select **Add Group**
   a) **Name: defaultLowAvaya**
   b) **Application Rule: default-trunk** (created in **Section 6.4.1**)
   c) **Border Rule: default**
   d) **Media Rule: default-low-med-QOS** (created in **Section 6.4.2**)
   e) **Security Rule: default-low**
   f) **Signaling Rule: Avaya_SR** (created in **Section 6.4.3.1**)
   g) **Time of Day: default**
4. Select **Finish** (not shown)



6.4.5. Endpoint Policy Groups – AT&T

1. Select **Domain Policies** from the menu on the left-hand side
2. Select **End Point Policy Groups**
3. Select **Add Group**
    a. **Name: defaultLowATT**
    b. **Application Rule: default-trunk** (created in **Section 6.4.1**)
    c. **Border Rule: default**
    d. **Media Rule: default-low-med-QOS** (created in **Section 6.4.2**)
    e. **Security Rule: default-low**
    f. **Signaling Rule: ATT_SR** (created in **Section 6.4.3.2**)
    g. **Time of Day: default**
4. **Select Finish** (not shown)

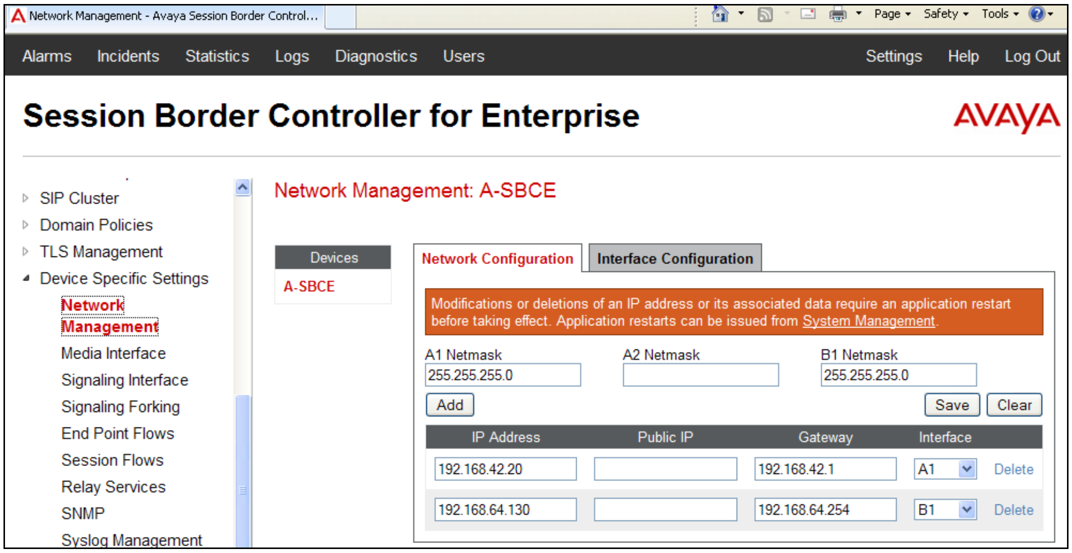


6.5. Device Specific Settings

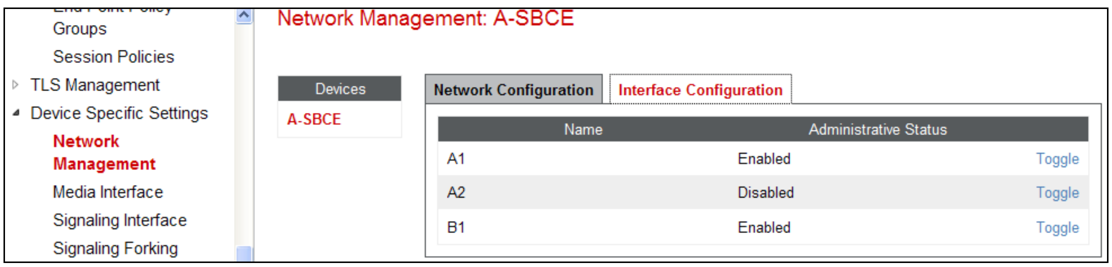
The **Device Specific Settings** feature for SIP allows you to view system information, and manage various device-specific network parameters such as port ranges.

6.5.1. Network Management

1. Select **Device Specific Settings** from the menu on the left-hand side
2. Select **Network Management**
    a) The network interfaces are defined during installation. However if these values need to be modified, do so via this tab.



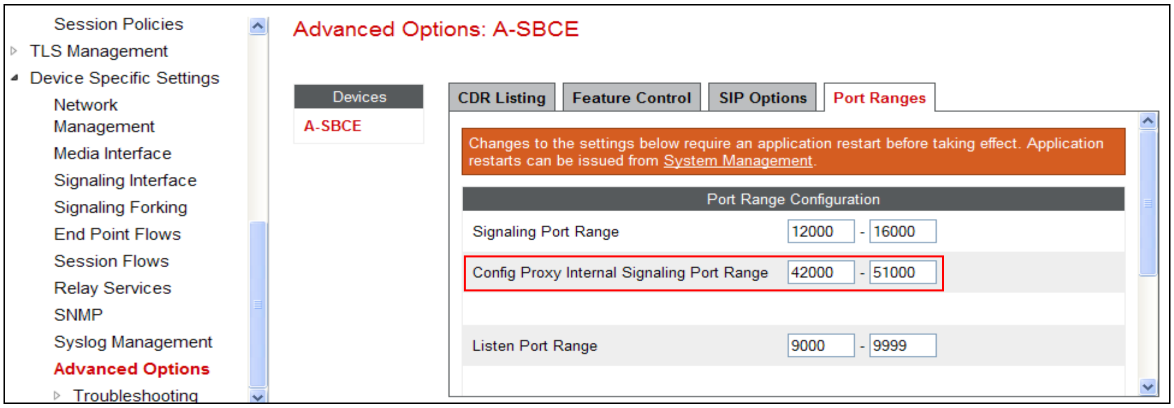
3. In addition, the provisioned interfaces may be enabled/disabled via the **Interface Configuration** tab.



6.5.2. Advanced Options

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

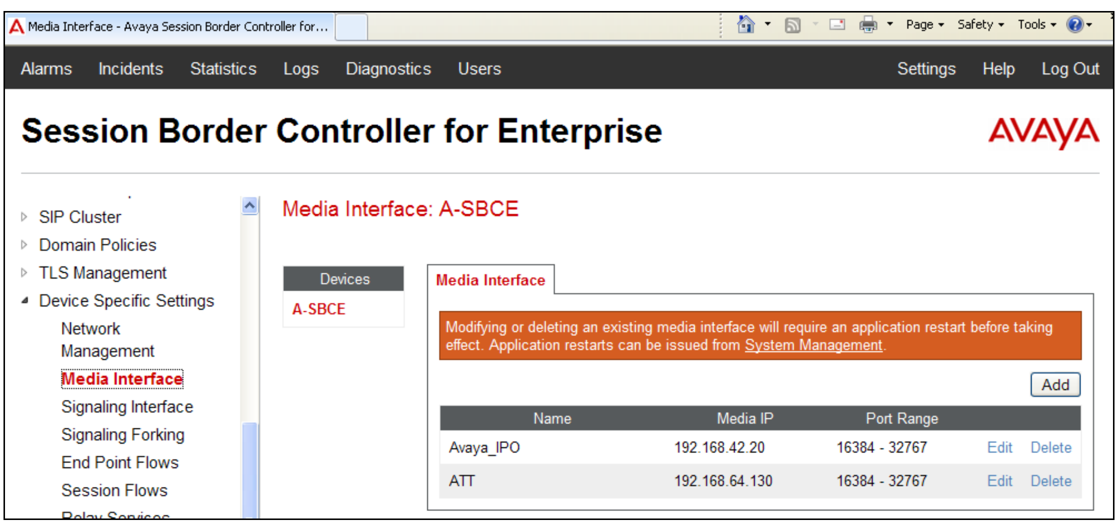
1. Select **Device Specific Settings → Advanced Options** from the menu on the left-hand side.
2. Select the **Port Ranges** tab.
3. In the **Config Proxy Internal Signaling Port Range** row, change the range to **42000 – 51000**.
4. Scroll to the bottom of the window and select **Save** (not shown).



6.5.3. 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 but only the outside is required by the AT&T IPFR-EF service.

1. Select **Device Specific Settings** from the menu on the left-hand side
2. Select **Media Interface**
3. Select **Add Media Interface**
   a) **Name: Avaya_IPO**
   b) **Media IP: 192.168.42.20** (Avaya SBCE A1 address to IP Office)
   c) **Port Range: 16384 - 32767**
4. Click **Finish** (not shown)
5. Select **Add Media Interface**
   a) **Name: ATT**
   b) **Media IP: 192.168.64.130** (Avaya SBCE B1 address toward AT&T)
   c) **Port Range: 16384 - 32767**
6. Click **Finish** (not shown)

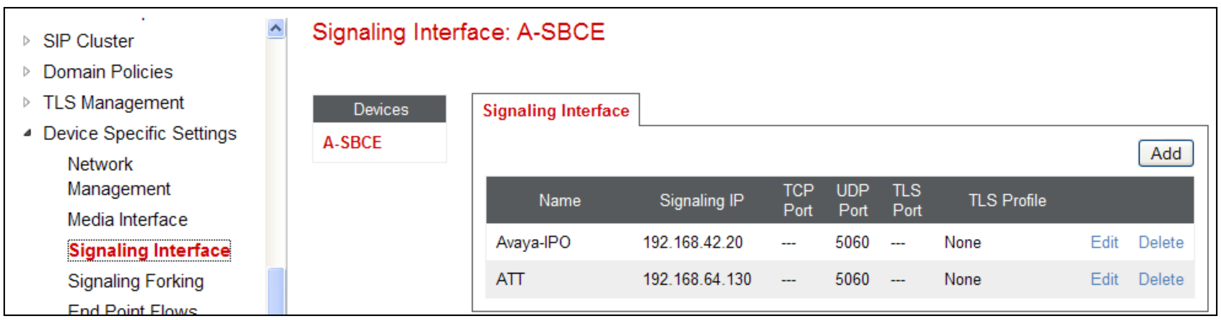


6.5.4. Signaling Interface

1. Select **Device Specific Settings** from the menu on the left-hand side
2. Select **Signaling Interface**
3. Select **Add Signaling Interface**
    a) **Name: Avaya-IPO**
    b) **Media IP: 192.168.42.20** (Avaya SBCE A1 address to IP Office)
    c) **UDP Port: 5060**

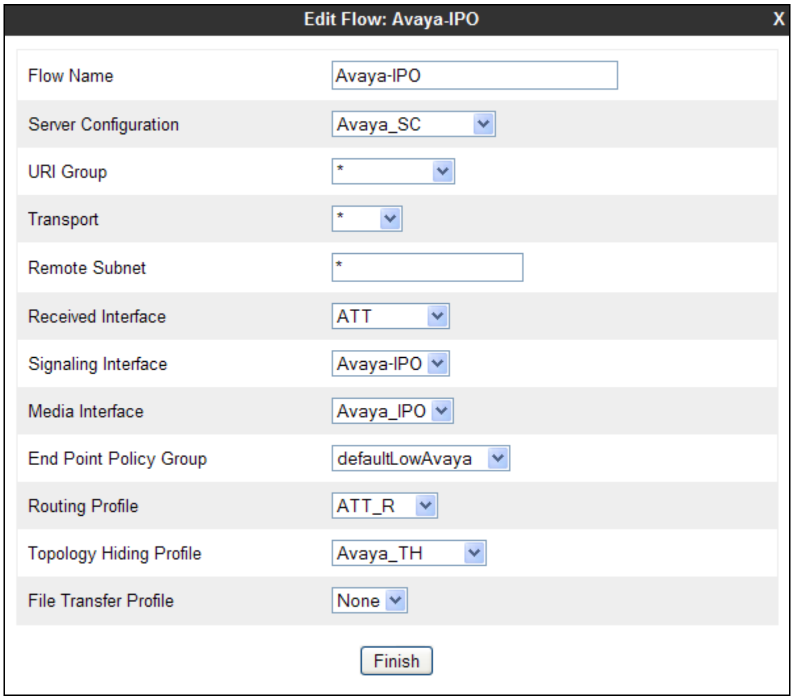
Note – UDP is the recommended protocol to use on the connection between the Avaya SBCE and IP Office. However TCP may be used if necessary.

4. Click **Finish**
5. Select **Add Media Interface**
    a) **Name: ATT**
    b) **Media IP: 192.168.64.130** (Avaya SBCE B1/public address toward AT&T)
    c) **UDP Port: 5060**
6. Click **Finish** (not shown).



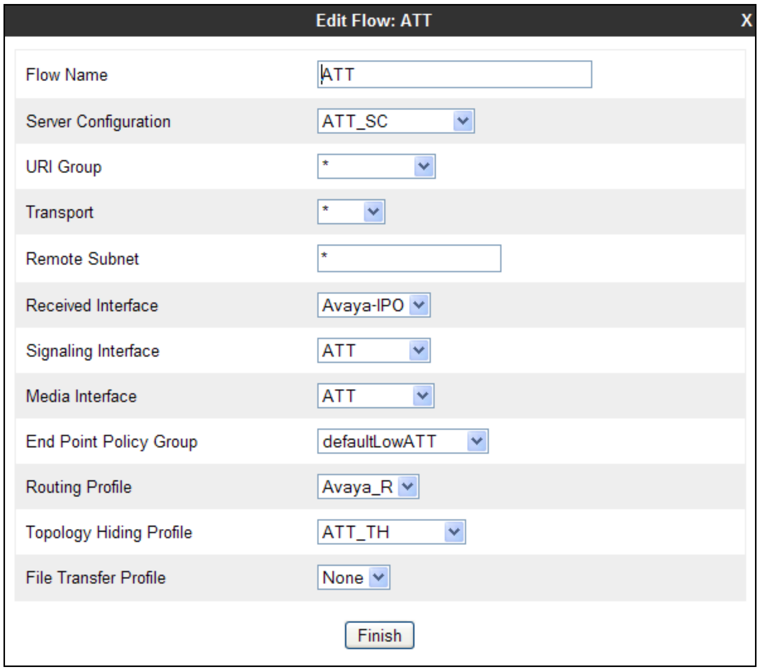
6.5.5. Endpoint Flows – to Avaya IP Office

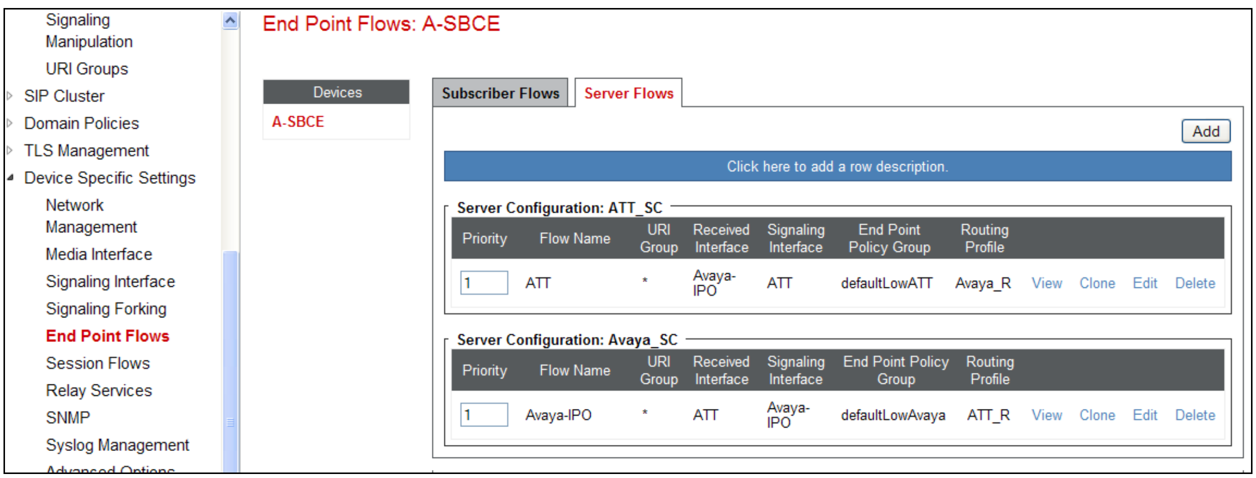
1. Select **Device Specific Settings** from the menu on the left-hand side
2. Select **Endpoint Flows**
3. Select the **Server Flows** tab
4. Select **Add Flow**, and enter the following:
    a) **Name: Avaya-IPO**
    b) **Server Configuration**: **Avaya_SC**
    c) **URI Group: ***
    d) **Transport: ***
    e) **Remote Subnet: ***
    f) **Received Interface**: **ATT**
    g) **Signaling Interface: Avaya-IPO**
    h) **Media Interface**: **Avaya_IPO**
    i) **End Point Policy Group: defaultLowAvaya**
    j) **Routing Profile: ATT_R**
    k) **Topology Hiding Profile: Avaya_TH**
    l) **File Transfer Profile: None**
5. Click **Finish** (not shown)



6.5.6. Endpoint Flows – To AT&T

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**, and enter the following:
    a) **Name: ATT**
    b) **Server Configuration: ATT_SC**
    c) **URI Group: ***
    d) **Transport: ***
    e) **Remote Subnet: ***
    f) **Received Interface: Avaya-IPO**
    g) **Signaling Interface: ATT**
    h) **Media Interface: ATT**
    i) **End Point Policy Group**: **defaultLowATT**
    j) **Routing Profile: Avaya_R**
    k) **Topology Hiding Profile: ATT_TH**
    l) **File Transfer Profile: None**
5. Click **Finish** (not shown)





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 required 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 R8.1 and the Avaya Session Border Controller for Enterprise 6.2, 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 and the Avaya Session Border Controller for Enterprise 6.2, functionality with the AT&T IPTF service:

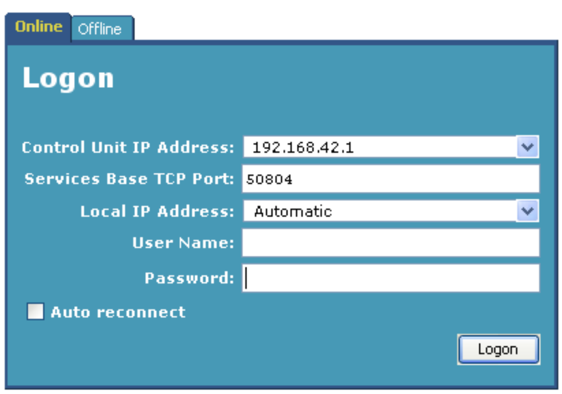
- Place inbound calls, answer the calls, and verify that two-way talk path exists. Verify that the calls remain stable for several minutes and disconnects properly.
- Verify basic call functions such as hold, transfer, and conference.
- Place an inbound call to a telephone, but do not answer the call. Verify that the call covers to Avaya Aura® Messaging voicemail. Retrieve the message from Avaya Aura® Messaging 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. In the reference configuration Avaya IP Office sent OPTIONS to the AT&T IPTF service Border Element, (via the Avaya SBCE), 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.729 (A or B) and G.711 ULAW codecs.

8.2. Avaya IP Office 8.1

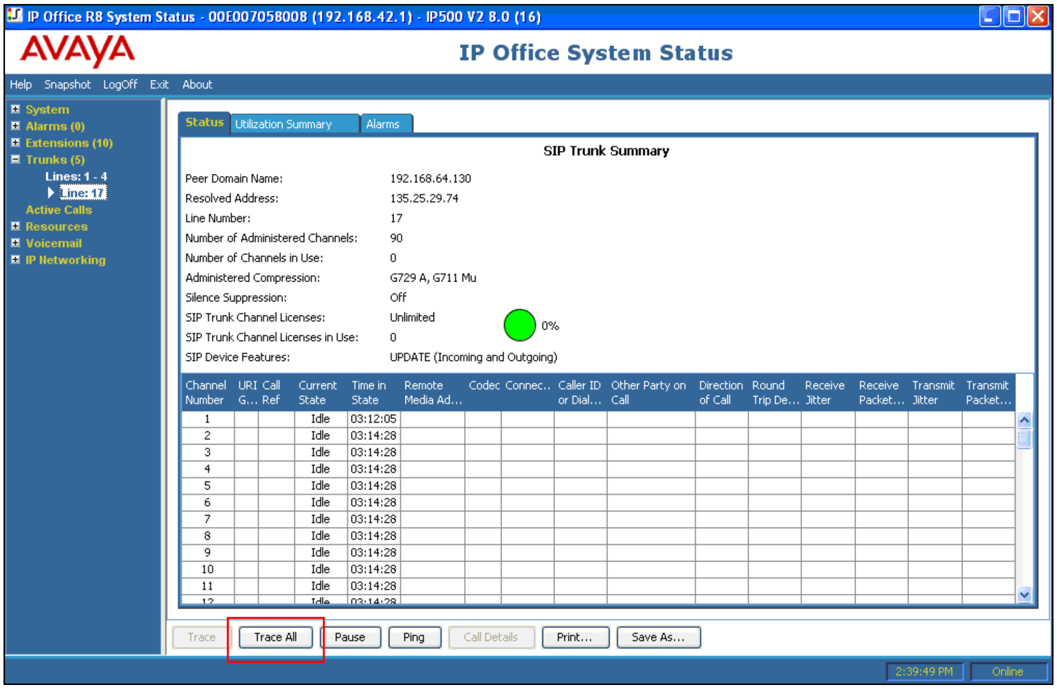
The following items may be used to analyze/troubleshoot Avaya IP Office operations.

8.2.1. System Status Application

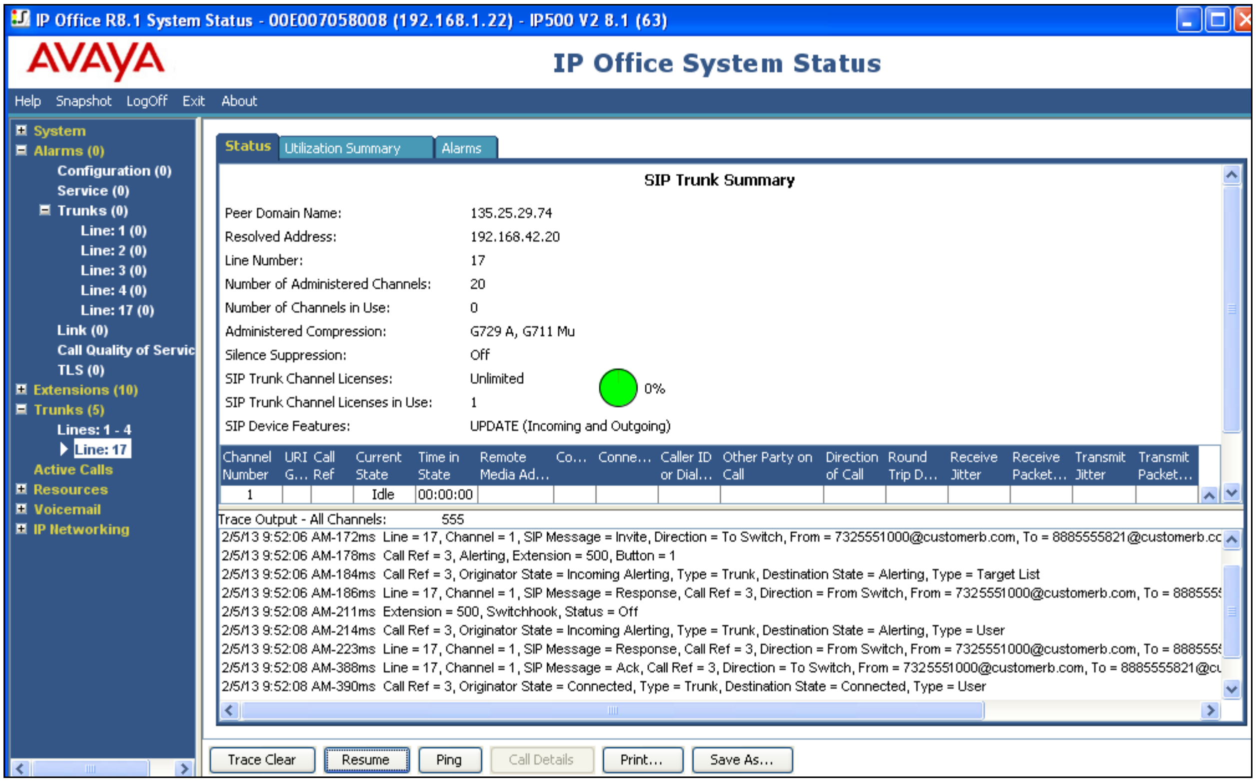
The System Status application can be used to monitor or troubleshoot Avaya IP Office. The System Status application can typically be accessed from **Start → Programs → Avaya IP Office → System Status**. The following screen shows an example **Logon** screen. Enter the Avaya IP Office IP address in the **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 following screen shows an example inbound call where PSTN called Avaya IP Office Hunt Group “Service” (H.323 phone, Extn500).

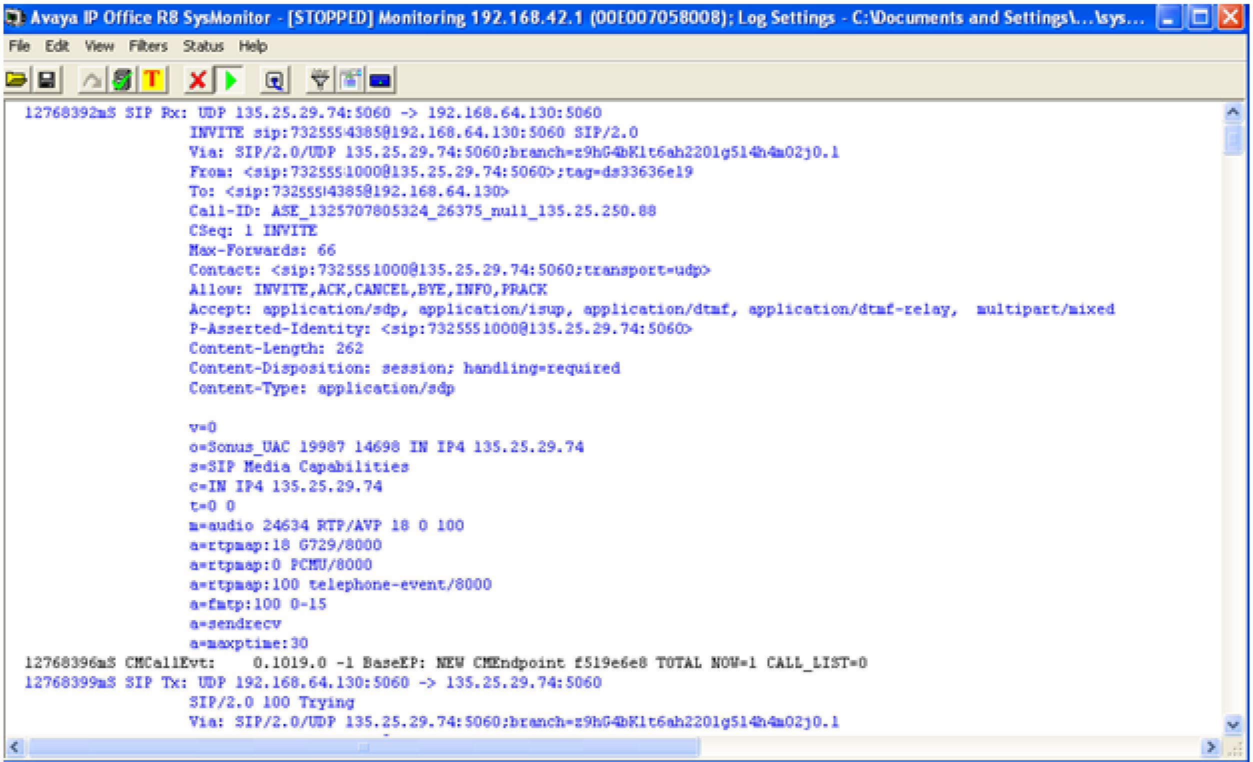


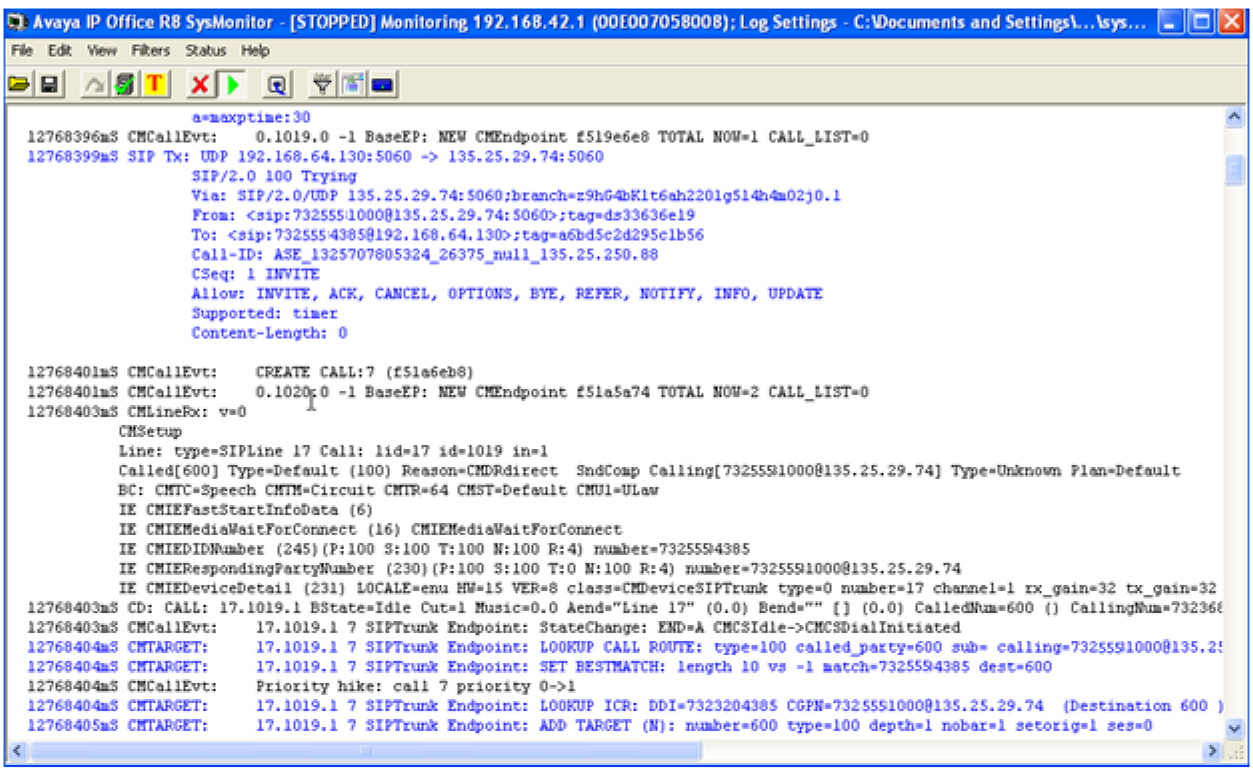
8.2.2. System Monitor Application

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

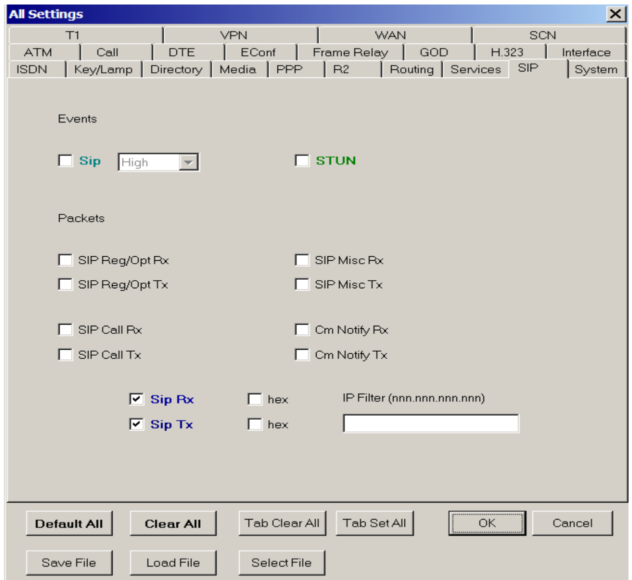
The Monitor will be active at startup. To pause the Monitor press the Pause  button. The pause button will be replaced with the Start  button. Press this button to resume the monitoring. To clear the Monitor display, press the Clear  button.

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


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


Avaya IP Office R8 SysMonitor - [STOPPED] Monitoring 192.168.42.1 (00E007058008); Log Settings - C:\Documents and Settings\...\sys...
File Edit View Filters Status Help
a=maxptime:30
12768396mS CMCallEvt: 0.1019.0 -1 BaseEP: NEW CMEndpoint f519e6e8 TOTAL NOW=1 CALL_LIST=0
12768399mS SIP Tx: UDP 192.168.64.130:5060 -> 135.25.29.74:5060
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 135.25.29.74:5060;branch=z9hG4bK1t6ah2201g514h4m02j0.1
From: <sip:7325551000@135.25.29.74:5060>;tag=ds33636e19
To: <sip:7325554385@192.168.64.130>;tag=a6bd5c2d295c1b56
Call-ID: ASE_1325707805324_26375_null_135.25.250.88
CSeq: 1 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
Supported: timer
Content-Length: 0
12768401mS CMCallEvt: CREATE CALL:7 (f51a6eb8)
12768401mS CMCallEvt: 0.1020.0 -1 BaseEP: NEW CMEndpoint f51a5a74 TOTAL NOW=2 CALL_LIST=0
12768403mS CMLineRx: v=0
CMSetup
Line: type=SIPLine 17 Call: lid=17 id=1019 in=1
Called[600] Type=Default (100) Reason=CMDRdirect SndComp Calling[7325551000@135.25.29.74] Type=Unknown Plan=Default
BC: CMTC=Speech CMTM=Circuit CMTR=64 CMST=Default CMU1=ULaw
IE CMIEFastStartInfoData (6)
IE CMIEMediaWaitForConnect (16) CMIEMediaWaitForConnect
IE CMIEDIDNumber (245)(P:100 S:100 T:100 N:100 R:4) number=7325554385
IE CMIERespondingPartyNumber (230)(P:100 S:100 T:0 N:100 R:4) number=7325551000@135.25.29.74
IE CMIEDeviceDetail (231) LOCALE=enu HW=15 VER=8 class=CMDeviceSIPTrunk type=0 number=17 channel=1 rx_gain=32 tx_gain=32
12768403mS CD: CALL: 17.1019.1 BState=Idle Cut=1 Music=0.0 Aend="Line 17" (0.0) Bend="" [] (0.0) CalledNum=600 () CallingNum=73236
12768403mS CMCallEvt: 17.1019.1 7 SIPTrunk Endpoint: StateChange: END=A CMCSIdle->CMCSDialInitiated
12768404mS CMTARGET: 17.1019.1 7 SIPTrunk Endpoint: LOOKUP CALL ROUTE: type=100 called_party=600 sub= calling=7325551000@135.25
12768404mS CMTARGET: 17.1019.1 7 SIPTrunk Endpoint: SET BESTMATCH: length 10 vs -1 match=7325554385 dest=600
12768404mS CMCallEvt: Priority hike: call 7 priority 0->1
12768404mS CMTARGET: 17.1019.1 7 SIPTrunk Endpoint: LOOKUP ICR: DDI=7323204385 CGPN=7325551000@135.25.29.74 (Destination 600)
12768405mS CMTARGET: 17.1019.1 7 SIPTrunk Endpoint: ADD TARGET (N): number=600 type=100 depth=1 nobar=1 setorig=1 ses=0

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



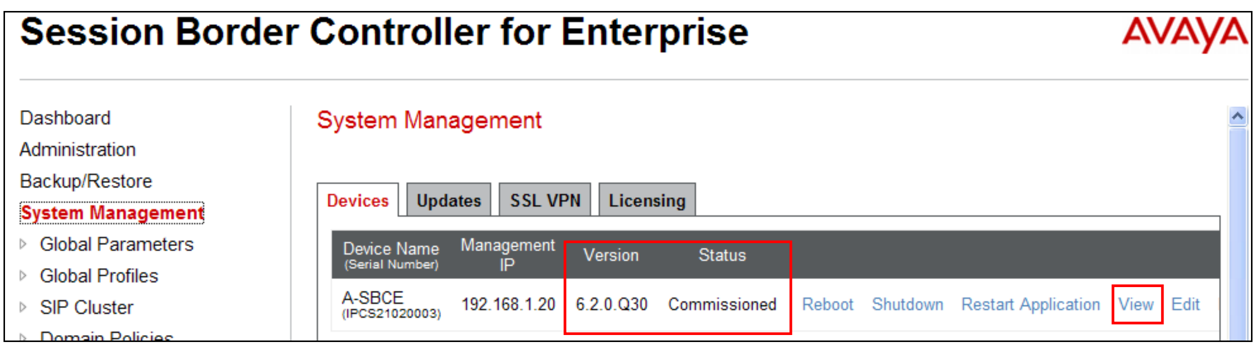
8.3. Avaya Session Border Controller for Enterprise 6.2

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

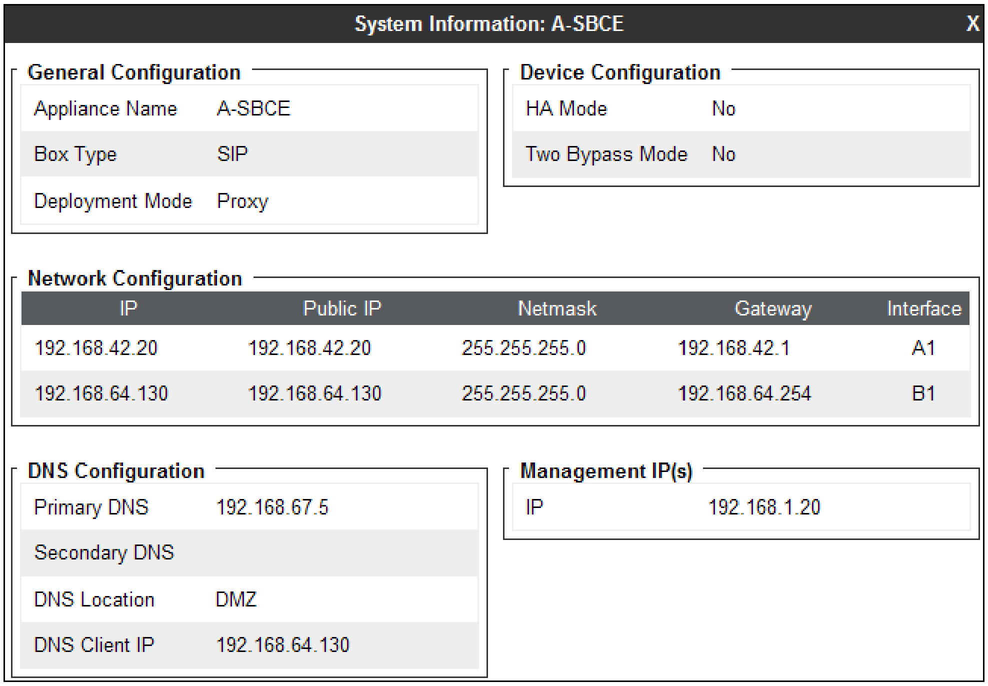
8.3.1. System Information

Step 1 - Navigate to **System Management → Devices** tab.

- The software version is shown in the **Version** column.
- Also verify that the **Status** column says **Commissioned**.



Step 2 – Click on **View** (shown above) to display the system information.

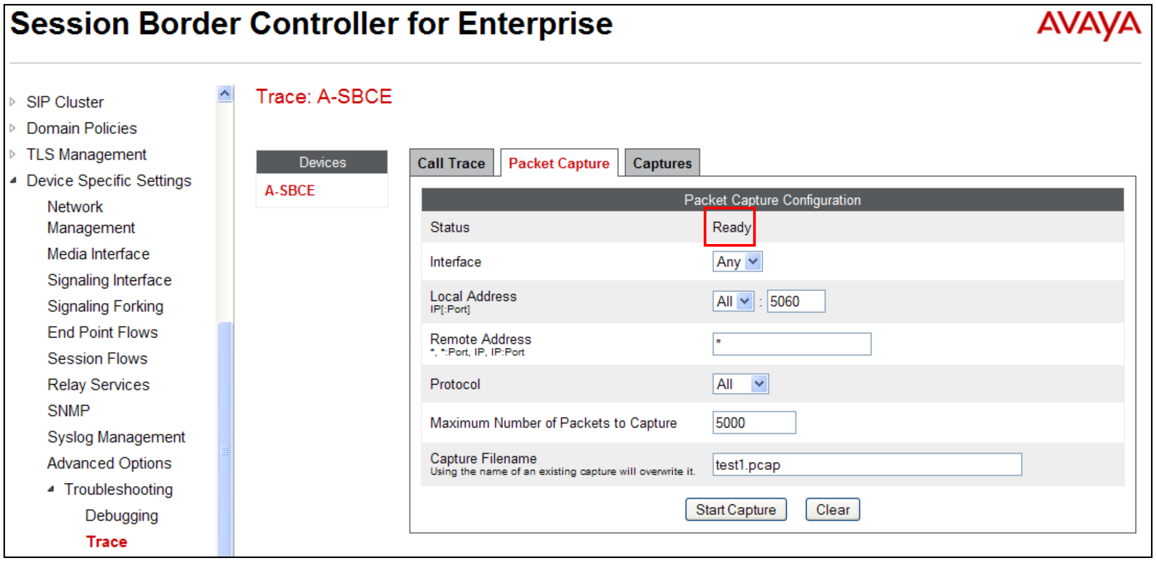


8.3.2. Avaya SBCE Protocol Traces

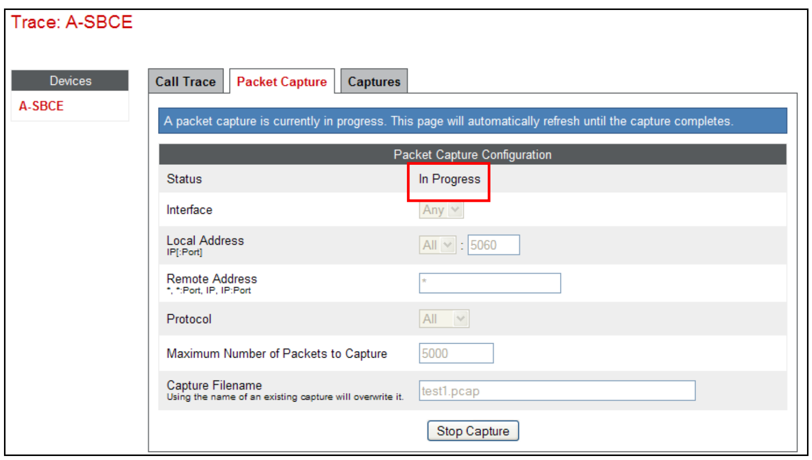
The Avaya SBCE can take internal traces of specified interfaces. In the example below all SIP signaling crossing interfaces A1 and B1 are captured.

Step 1 - Navigate to **Device Specific Settings → Troubleshooting → Trace**
Step 2 - Select the **Packet Capture** tab and select the following:

- **Interface** : Select **A1**, **B1**, or **Any** from the drop down menu. Note that specifying **Any** will capture packets from both the A1 and B1 interfaces used in the reference configuration.
- **Local Address**: Select **All** and **5060** from the dropdown menus. This will capture any packets using port 5060 (e.g., SIP signaling).
- **Remote Address**: Enter *
- **Protocol**: Select **All** from the dropdown menu.
- **Maximum Number of Packets to Capture**: Specify an amount that will include all the messaging required for your trace (e.g., **5000**). Note that specifying amounts greater than 10,000 may impact system performance.
- **Capture Filename**: Specify a name for the trace file. Include a **.pcap** extension if you wish to open the file with Wireshark.
- Click **Start Capture** to begin the trace.



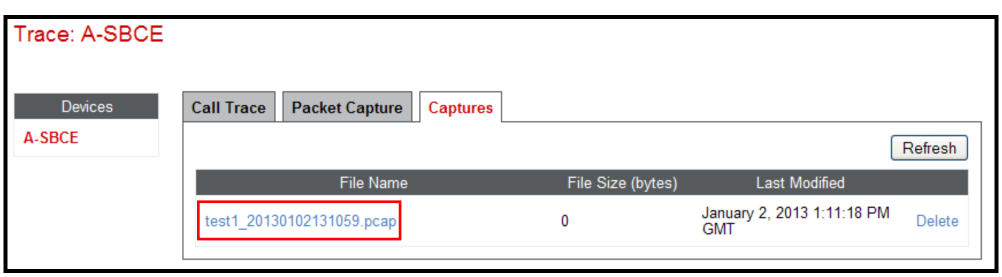
The capture process will initialize and then display the following status window:



Step 3 – Run the test.

Step 4 - Select **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** (e.g., test1). Note that the system will append date/time information to the filename.

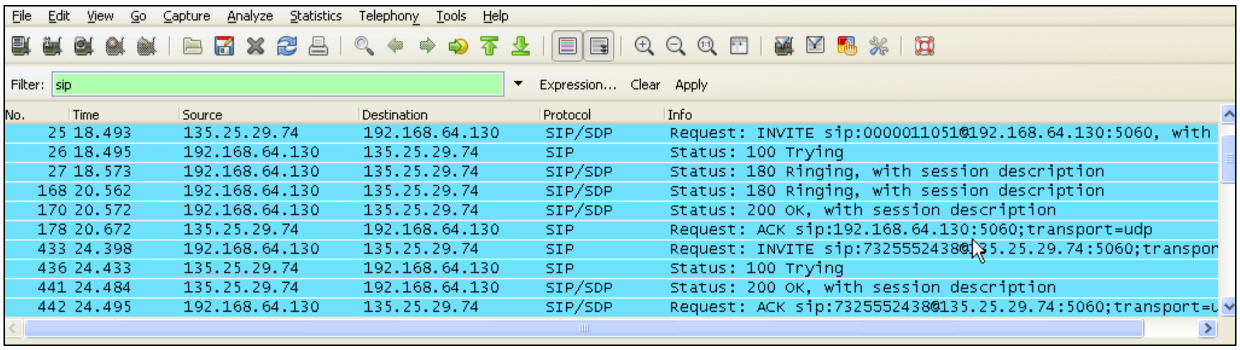


Step 6 - Click on the **File Name** to download the file and use an application such as Wireshark to open the trace.

8.4. Protocol Trace Examples

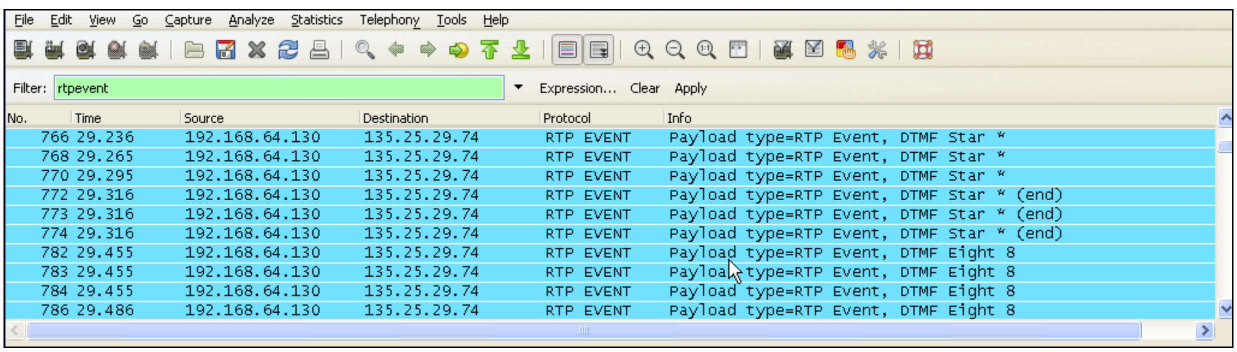
This section shows examples of protocol traces taken at the Avaya SBCE B1 interface (to AT&T).

The following is an example of an inbound call from AT&T, filtering on the SIP protocol.



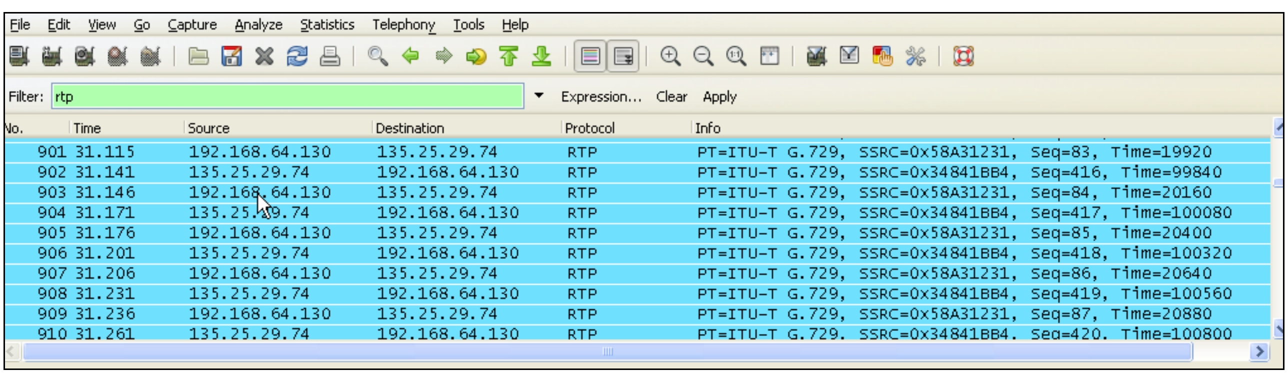
No.	Time	Source	Destination	Protocol	Info
25	18.493	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:0000011051@192.168.64.130:5060, with
26	18.495	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
27	18.573	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
168	20.562	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
170	20.572	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
178	20.672	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
433	24.398	192.168.64.130	135.25.29.74	SIP	Request: INVITE sip:73255524380135.25.29.74:5060;transpor
436	24.433	135.25.29.74	192.168.64.130	SIP	Status: 100 Trying
441	24.484	135.25.29.74	192.168.64.130	SIP/SDP	Status: 200 OK, with session description
442	24.495	192.168.64.130	135.25.29.74	SIP/SDP	Request: ACK sip:7325552438@135.25.29.74:5060;transport=u

The following is an example of a call filtering on DTMF events.



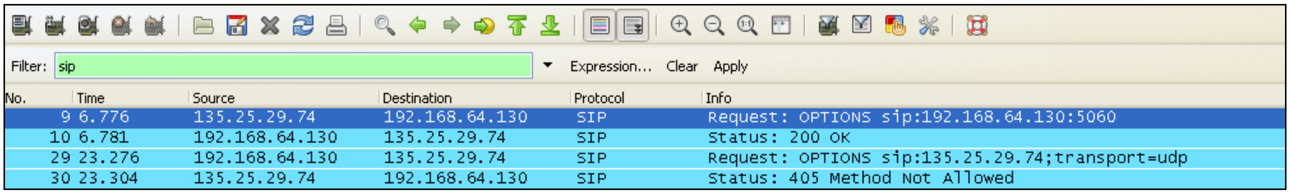
No.	Time	Source	Destination	Protocol	Info
766	29.236	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Star *
768	29.265	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Star *
770	29.295	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Star *
772	29.316	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Star * (end)
773	29.316	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Star * (end)
774	29.316	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Star * (end)
782	29.455	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Eight 8
783	29.455	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Eight 8
784	29.455	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Eight 8
786	29.486	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Eight 8

The following is an example of a call filtering on RTP.



No.	Time	Source	Destination	Protocol	Info
901	31.115	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x58A31231, Seq=83, Time=19920
902	31.141	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0x34841BB4, Seq=416, Time=99840
903	31.146	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x58A31231, Seq=84, Time=20160
904	31.171	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0x34841BB4, Seq=417, Time=100080
905	31.176	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x58A31231, Seq=85, Time=20400
906	31.201	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0x34841BB4, Seq=418, Time=100320
907	31.206	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x58A31231, Seq=86, Time=20640
908	31.231	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0x34841BB4, Seq=419, Time=100560
909	31.236	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x58A31231, Seq=87, Time=20880
910	31.261	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0x34841BB4, Seq=420, Time=100800

A trace of an idle line can verify that your SIP Trunk from the Avaya SBCE B1 interface (192.168.64.130) to the IPTF Service border element (135.25.29.74) is up and communicating with SIP *OPTIONS* messages and response messages. A SIP *405 Method Not Allowed* response is normal for the Avaya SBCE to AT&T test environment.



9. Conclusion

As illustrated in these Application Notes, Avaya IP Office R8.1 and the Avaya Session Border Controller for Enterprise 6.2, can be configured to interoperate successfully with the AT&T IP Toll Free service using **AVPN** or **MIS/PNT** transport connections, within the limitations described in **Section 2.2.1**.

This solution provides users of Avaya IP Office R8.1 the ability to support inbound calls utilizing an AT&T IPTF SIP trunk service connection, via AVPN or MIS/PNT transport, using the platform and service features listed in **Section 2.1**.

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

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

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

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

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

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

[7] Installing Avaya Session Border Controller For Enterprise, Release 6.2, Issue 1, January 2013.

[8] Administering Avaya Session Border Controller, Release 6.2, Issue 1, January 2012

AT&T IPTF Service:

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

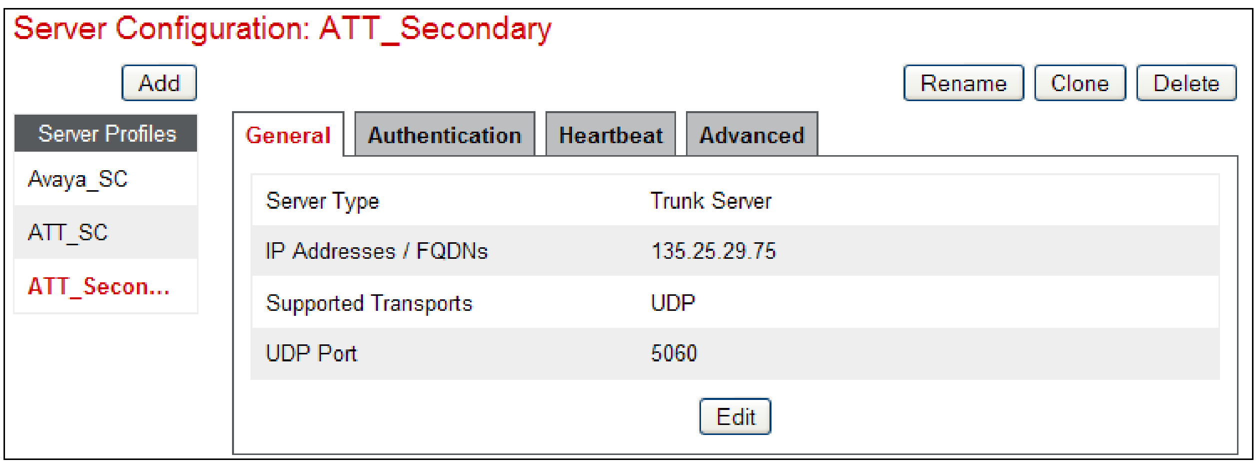
11. Addendum 1 – Redundancy to Multiple AT&T Border Elements

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

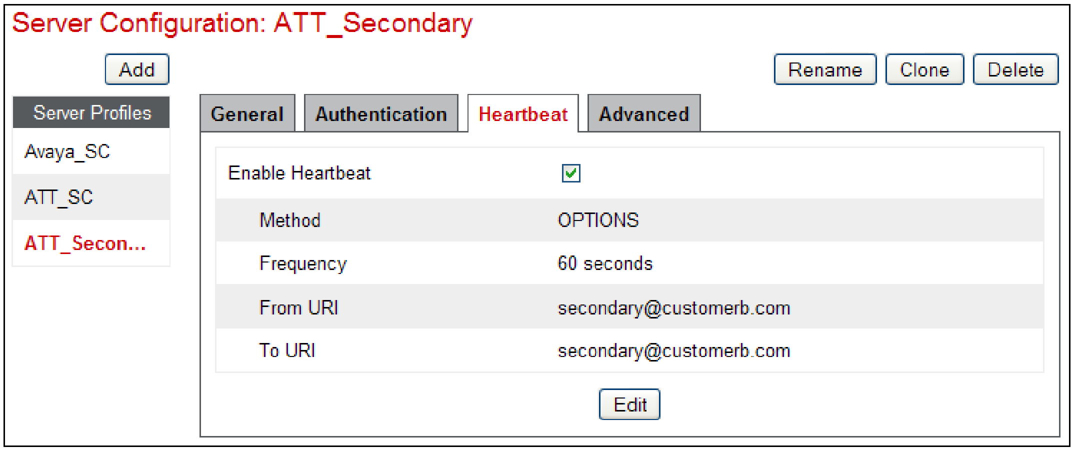
Given two AT&T border elements **135.25.29.74** and **135.25.29.75**, the Avaya SBCE is provisioned as follows to include the backup trunk connection to 135.25.29.75 (the primary AT&T trunk connection to 135.25.29.74 is defined in **Section 6.3.6**).

Step 1: Secondary Location in Server Configuration

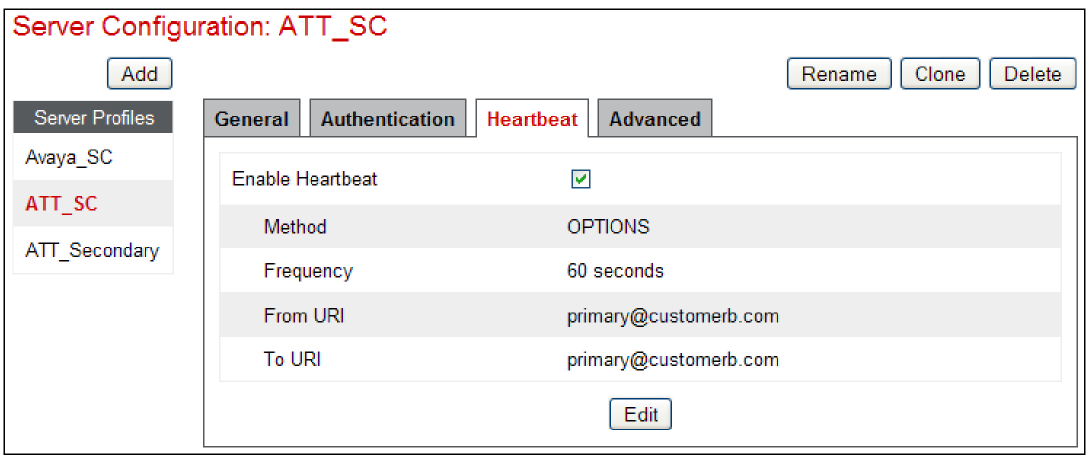
1. Select **Global Profiles** from the menu on the left-hand side
2. Select the **Server Configuration**
3. Select **Add Profile**
    a) **Name: ATT_Secondary**
4. On the **Add Server Configuration Profile – General** tab:
    a) Select **Server Type: Trunk Server**
    b) **IP Address: 135.25.29.75** (Example Address for a secondary location)
    c) **Supported Transports**: Check **UDP**
    d) **UDP Port: 5060**
    e) Select **Next** (not shown)



5. On the **Authentication** tab
    a) Select **Next** to accept defaults (not shown).
6. On the **Heartbeat** tab:
    a) Check **Enable Heartbeat**
    b) **Method: OPTIONS**
    c) **Frequency: 60 seconds**
    d) **From URI: secondary@customerb.com**
    e) **To URI: secondary@customerb.com**
    f) Select **Next** (not shown)

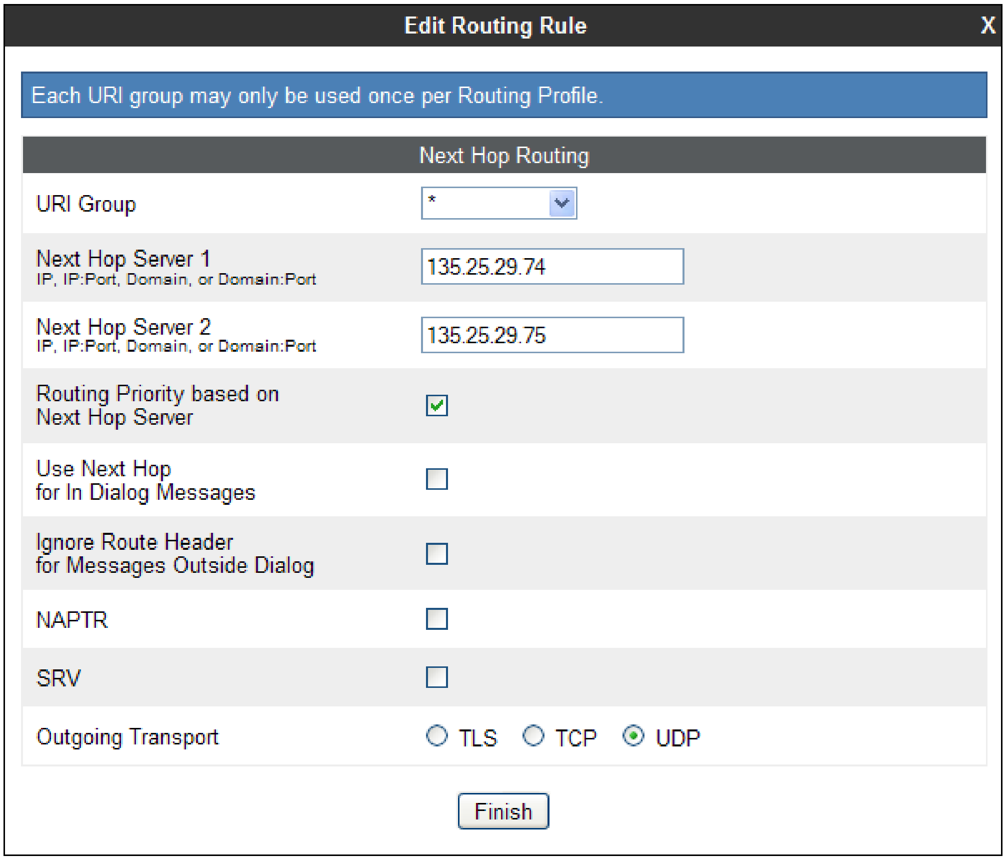


7. On the **Advanced** Tab
   a) Click **Finish** to accept defaults (not shown).
8. Select the Profile created in **Section 6.3.6** (e.g., **ATT_SC**)
9. Select the **Heartbeat Tab**
10. Select **Edit**
11. Repeat **Steps 6 – 7**, but with information for the Primary Trunk as shown below.



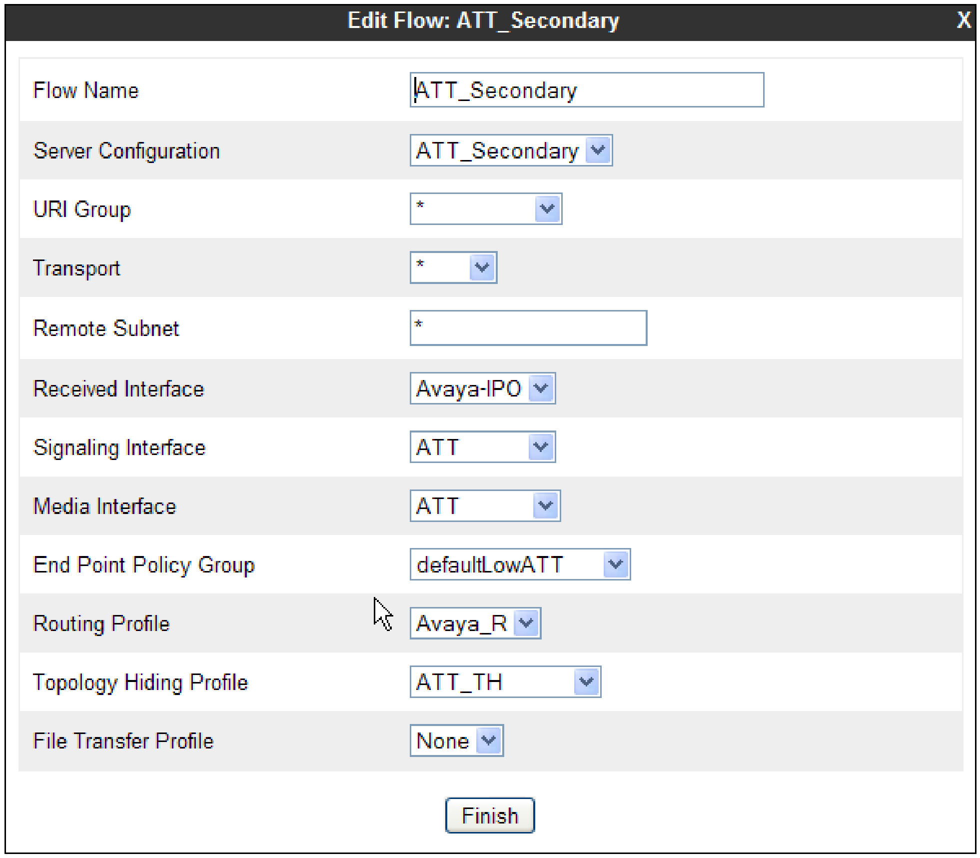
Step 2: Add Secondary IP Address to Routing

1. Select **Global Profiles** from the menu on the left-hand side
2. Select the **Routing**
3. Select the profile created in **Section 6.3.4** (e.g., **ATT_R**)
4. Click **Edit** (not shown)
   a) Enter the IP Address of the secondary location in the **Next Hop Server 2** (e.g., **135.25.29.75**)
5. Click **Finish**



Step 3: Configure End Point Flows – ATT_Secondary

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_Secondary**
   c) **URI Group: ***
   d) **Transport: ***
   e) **Remote Subnet: ***
   f) **Received Interface: Avaya-IPO**
   g) **Signaling Interface: ATT**
   h) **Media Interface: ATT**
   i) **End Point Policy Group**: **defaultLowATT**
   j) **Routing Profile: Avaya_R**
   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 (135.25.29.74) and secondary (135.25.29.75) border elements.

12. Appendix: Avaya IP Office 8.1 SIP Line Template

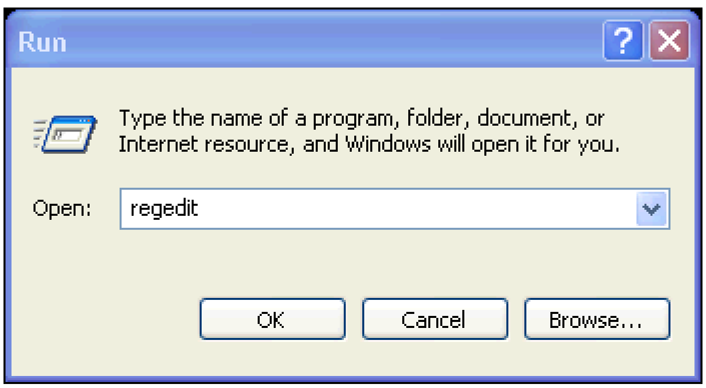
Avaya IP Office Release 8.1 supports a SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

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

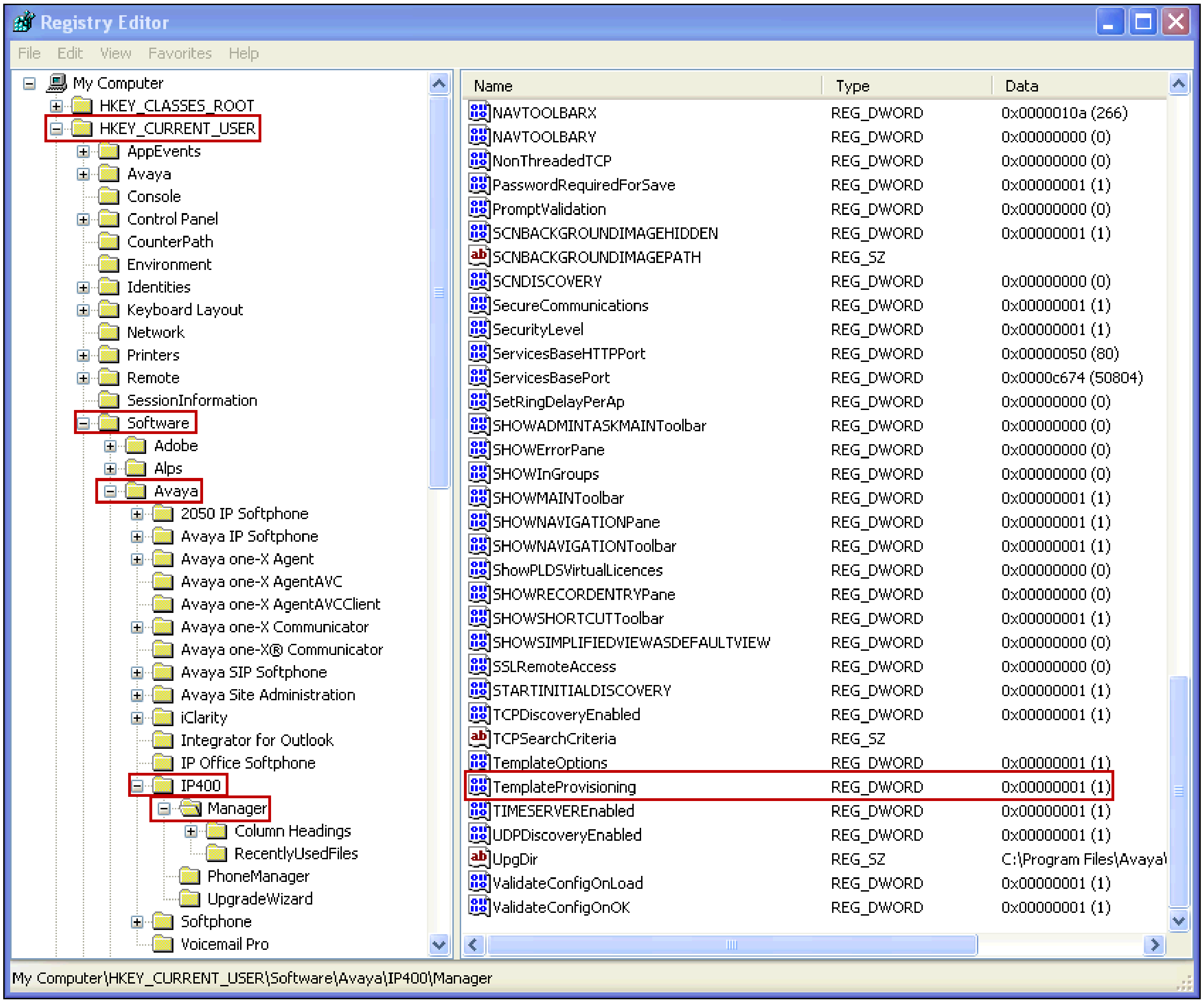
To create a SIP Line Template from the configuration described in these Application Notes, configure the parameters as described below.

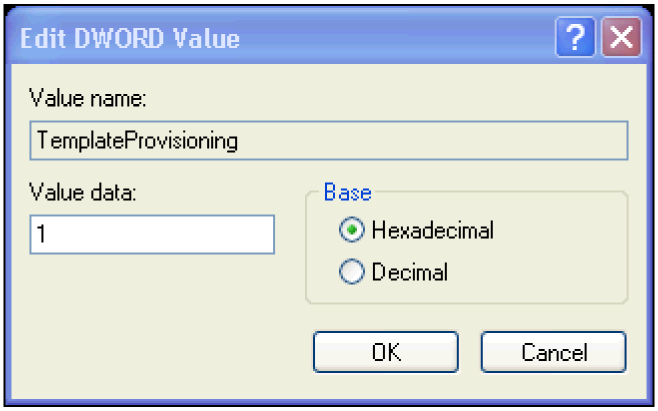
Create a new registry entry called **TemplateProvisioning** and set the **Value data** to **1**, as follows:

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



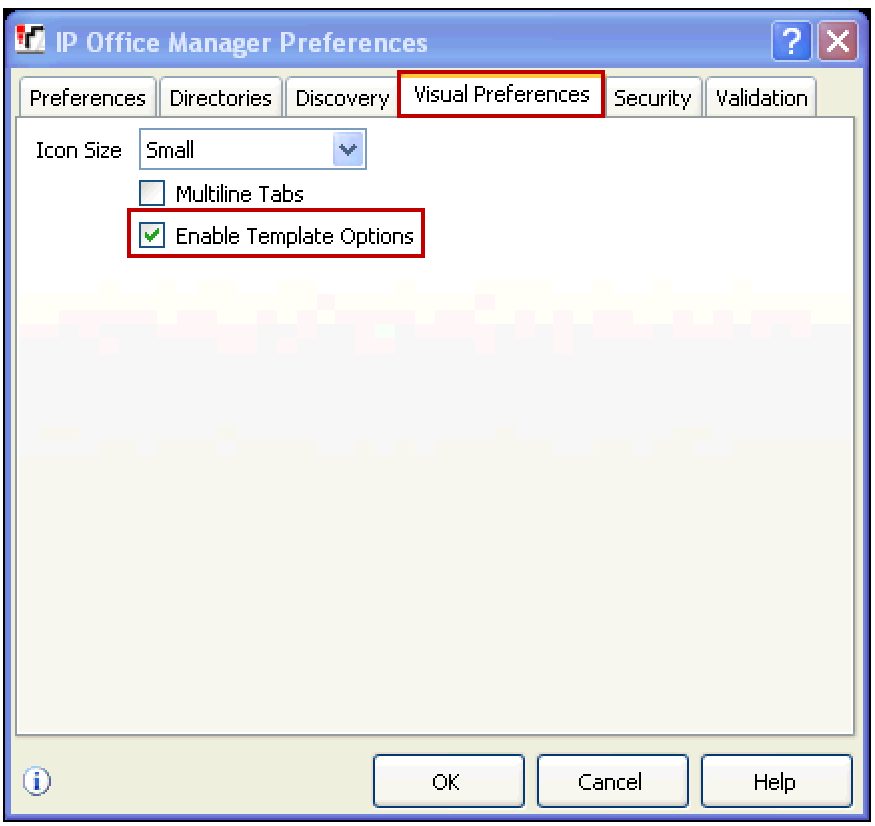
Under **HKEY_CURRENT_USER, Software, Avaya, IP400**, right click on **Manager**, then select **New, DWORD value,** then rename the newly created entry to: **TemplateProvisioning.** Right click on the newly created entry and select **Modify,** change the value under **Value Data** from "**0**" to "**1**".





Reboot the computer.

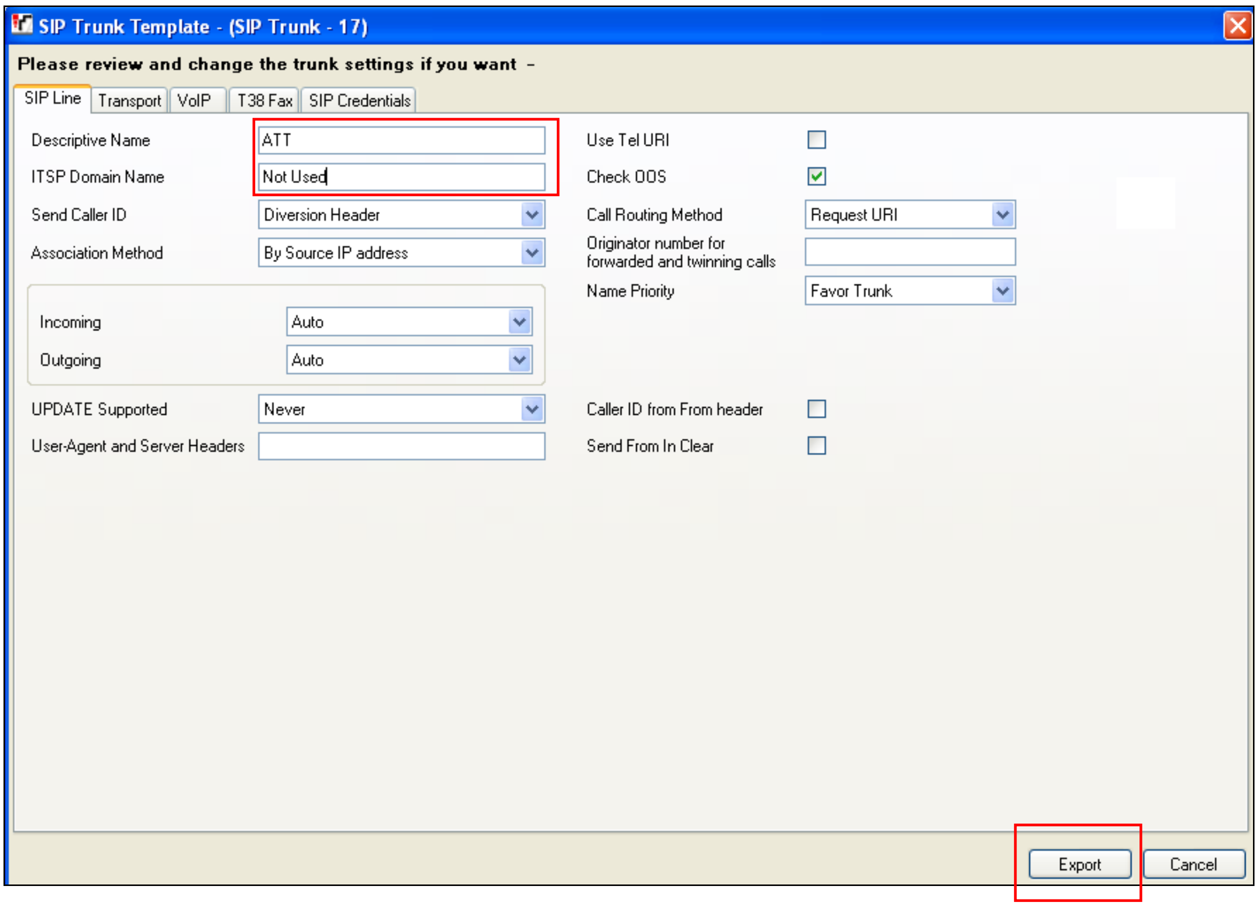
When the computer comes back up, enable the template by opening **IP Office Manager**, select **File**, and then **Preferences**. On the **Visual Preferences** tab, check the **Enable Template Options** box, and click **OK**.



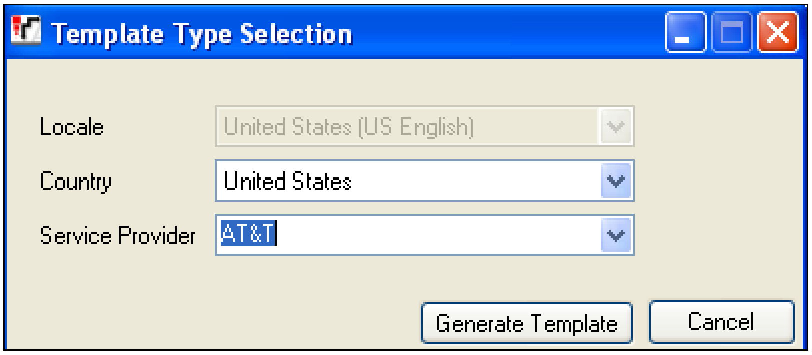
To create a SIP Line Template from the configuration, on the left Navigation Pane, right click on the Sip Line (**17**), and select **Generate SIP Trunk Template** (not shown)

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

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



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



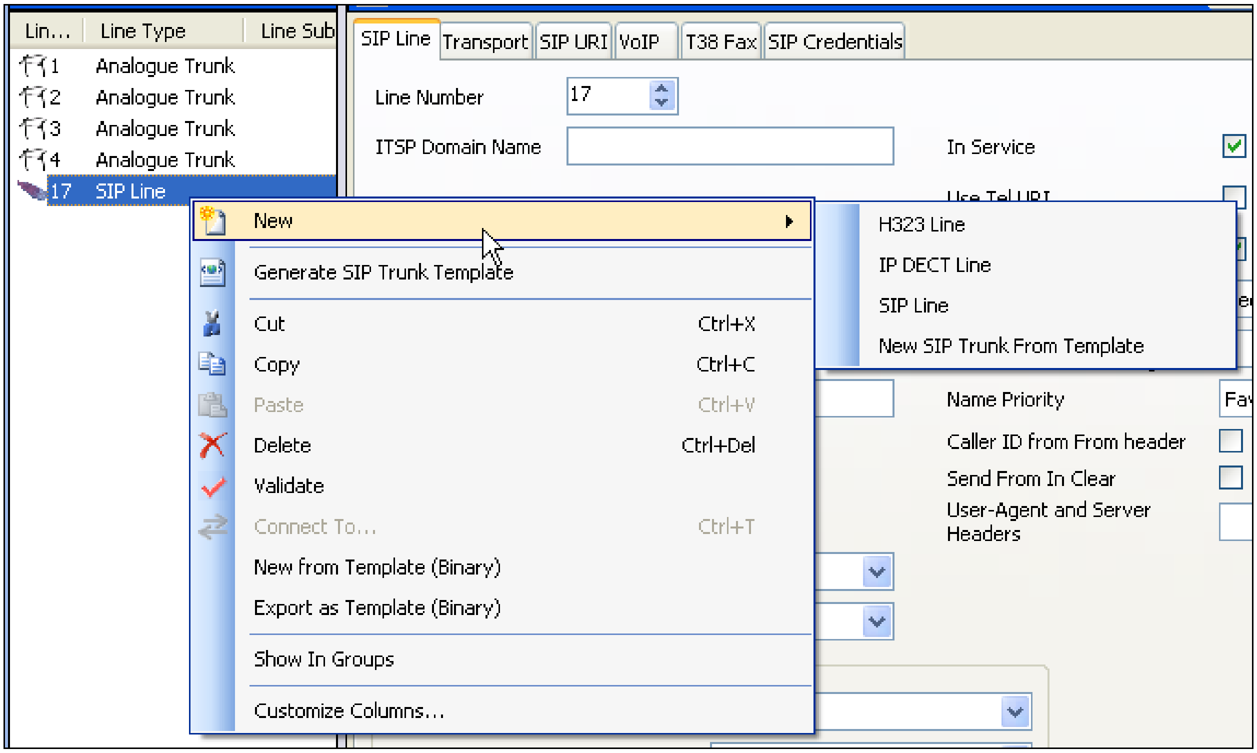
By default the template file is generated to the path **\\Program Files\Avaya\IP Office\Manager\Templates**.

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

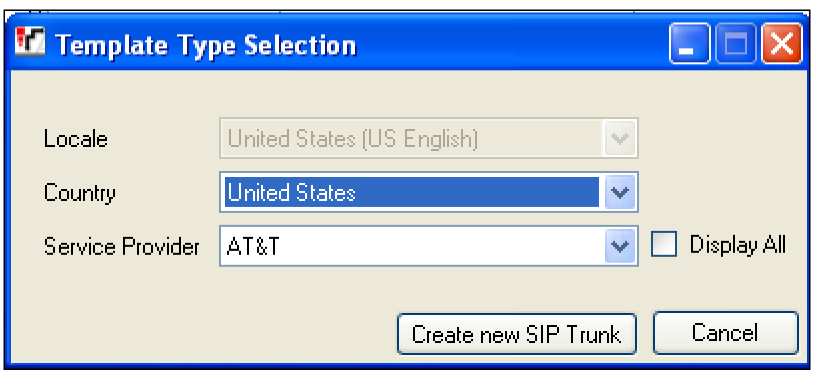

```
 <?xml version="1.0" encoding="utf-8" ?>
- <Template xmlns="urn:SIPTrunk-schema">
<TemplateType>SIPTrunk</TemplateType>
<Version>20121130</Version>
<SystemLocale>enu</SystemLocale>
<DescriptiveName>ATT</DescriptiveName>
<ITSPDomainName>Not Used</ITSPDomainName>
<SendCallerID>CallerIDDIV</SendCallerID>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

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


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



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



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