Avaya Solution & Interoperability Test Lab

Application Notes for Avaya IP Office Release 10.0, Avaya Session Border Controller for Enterprise Release 7.1 with AT&T IP Toll Free Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office Release 10.0 and Avaya Session Border Controller for Enterprise Release 7.1, with the AT&T IP Toll Free service using AVPN or MIS/PNT transport connections.

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

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

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

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1. Introduction

These Application Notes describe the steps for configuring Avaya IP Office Release 10.0 and Avaya Session Border Controller for Enterprise Release 7.1, with the AT&T IP Toll Free service using **AVPN** or **MIS/PNT** transport connections.

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

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

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

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

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

2. General Test Approach and Test Results

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

The interoperability compliance testing focused on verifying inbound and outbound call flows between IPTF and the Customer Premises Equipment (CPE) containing Avaya IP Office Release 10.0 and Avaya SBCE Release 7.1 (see **Section 3.2** for call flow examples).

The test environment described in these Application Notes consisted of:

- A simulated enterprise with Avaya IP Office 10, Avaya SBCE 7.1, Avaya SIP, H.323 and Analog telephones, as well as a fax machine emulator (Ventafax).

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

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- Laboratory versions of the IPTF service, to which the simulated enterprise was connected via AVPN/MIS transport.

2.1. Interoperability Compliance Testing

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

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

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

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

- Incoming calls from PSTN, routed by the IPTF service, to Avaya SBCE and Avaya IP Office. These calls are via the Avaya IP Office SIP Line and may be generated/answered by Avaya SIP telephones/softphones, H.323 telephones, Analog telephones, Analog fax machines or via Hunt Groups. Coverage to Voicemail Pro, and Voicemail Pro auto-attendant applications, were also used.
- Inbound fax using T.38 or G.711, and G3 or SG3 endpoints.
- Proper disconnect when the caller abandons a call before answer, and when the Avaya IP Office party or the PSTN party terminates an active call.
- Proper busy tone heard when an Avaya IP Office user calls a busy PSTN user, or a PSTN user calls a busy Avaya IP Office user (i.e., if no redirection was configured for user busy conditions).
- SIP OPTIONS monitoring the health of the SIP trunk. In the reference configuration Avaya IP Office sent OPTIONS to the IPTF service Border Element and AT&T responded with *405 Method Not Allowed* (which is the expected response). That response is sufficient for Avaya IP Office to consider the connection up.
- Incoming calls using the G.729A and G.711 ULAW codecs.
- Long duration calls.
- DTMF transmission (RFC 2833) for successful voice mail navigation, including navigation of a simple auto-attendant application configured on Voicemail Pro, as well as IPTF DTMF generated features.
- Telephony features such as call waiting, hold, transfer, and conference.
- Verify reception of IPTF SIP Multipart/NSS headers, including SDP and XML content.
- AT&T IP Toll Free features such as Legacy Transfer Connect and Alternate Destination Routing.

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2.2. Test Results

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

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

2. **IP Toll Free ADR Call Redirection feature based on SIP error code response.** Upon receiving an error response, IPTF service can be configured to invoke ADR Call Redirection. The following error codes were producible by the reference configuration and tested successfully; 408 Request Timeout, 480 Temporarily Unavailable, 486 Busy Here, and 500 Server Internal Error. The following error codes are also supported by IPTF service, but were not producible by the reference configuration, and thus not tested; 503 Service Unavailable, 504 Server Timeout, and 600 Busy Everywhere.

3. **Enhanced CID – NSS feature**. The inbound calls to Avaya IP Office are not exercising the Enhanced CID feature. Although Avaya IP Office is accepting SIP Multipart/NSS headers, it is neither passing nor acting upon it. It is simply being ignored.

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.

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3. Reference Configuration

Note – Documents used to provision the test environment are listed in **Section 11**. References to these documents are indicated by the notation **[x]**, where *x* is the document reference number.

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

- An Avaya Server Edition Primary server with an IP500 V2 Expansion System for fax and digital endpoint support. The single Server Edition Primary server provides IP Office Server Edition, Voicemail Pro, Avaya WebRTC Gateway, and Avaya one-X® Portal for IP Office.
- Avaya IP Office Server Edition provides the voice communications services for a particular enterprise site.
- Voicemail Pro provides the voice messaging capabilities in the reference configuration. This solution is extensible to the standalone IP Office 500 V2 embedded voice mail as well.
- Avaya "desk" telephones are represented with an Avaya 1616 H.323 set, an Avaya 9611 H.323 set, an Avaya 9508 Digital set, an Avaya 1140E SIP set, as well as Avaya Communicator for Windows (SIP). Fax endpoints are represented by PCs running Ventafax emulation software connected by modem to an Avaya IP Office 500 V2 analog port.
- The Avaya SBCE provides SIP Session Border Controller (SBC) functionality, including address translation and SIP header manipulation between the IPTF service and the CPE. In the reference configuration, the Avaya SBCE runs on a VMWare platform.
- In the reference configuration, both the Avaya IP Office (interface "LAN1"), and the Avaya SBCE (interface "A1") are connected to the private CPE network. The Avaya SBCE interface "B1" is connected to the AT&T network.
- UDP transport via port 5060 was used between the Avaya IP Office and the Avaya SBCE, as well as between the Avaya SBCE and AT&T.
- The AT&T IPTF service requires RTP port ranges 16384-32767.
- AT&T provided the inbound and outbound access numbers (DID and DNIS) used in the reference configuration. Note that the IPTF service may deliver various digit lengths in the SIP Invite R-URI depending on the circuit order provisioning. In the reference configuration, the IPTF service delivered 15 digits.

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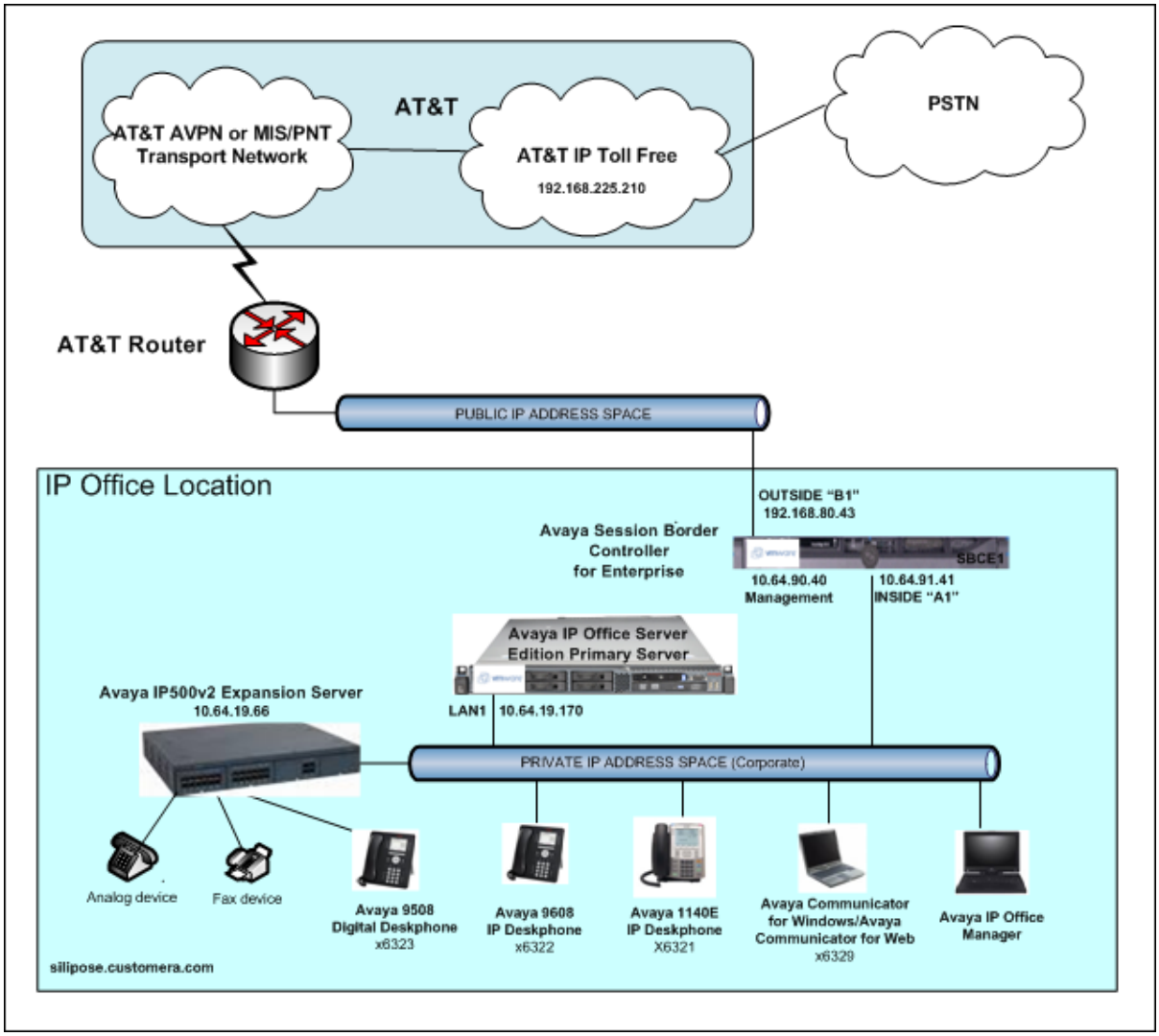

Figure 1: Reference Configuration

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3.1. Illustrative Configuration Information

The specific values listed in **Table 1** below and in subsequent sections are used in the reference configuration described in these Application Notes, and are for illustrative purposes only. Customers must obtain and use the values based on their own specific configurations.

Note – The Avaya SBCE "B1" interface communicates with AT&T Border Elements (BEs) located in the AT&T IPTF network. For security reasons, the IP addresses of the AT&T BEs are not included in this document. However as placeholders in the following configuration sections, the IP addresses **192.168.80.43** (Avaya SBCE "B1"), and **192.168.225.210** (AT&T BE IP address), are specified. In addition, AT&T DID/DNIS numbers shown in this document are examples as well. AT&T Customer Care will provide the actual Border Element IP addresses and DID/DNIS numbers as part of the IPTF provisioning process.

Component	Illustrative Value in these Application Notes
Avaya IP Office Server Edition	
Private network LAN1 interface	10.64.19.170
Avaya IP Office Expansion System (IP500 V2)	
Private network LAN1 interface	10.64.19.66
Avaya SBCE	
Private network "A1" interface	10.64.91.41
Public network "B1" interface	192.168.80.43
AT&T IPTF Service	
Border Element IP Address	192.168.225.210

Table 1: Illustrative Values Used in these Application Notes

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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. Basic 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.

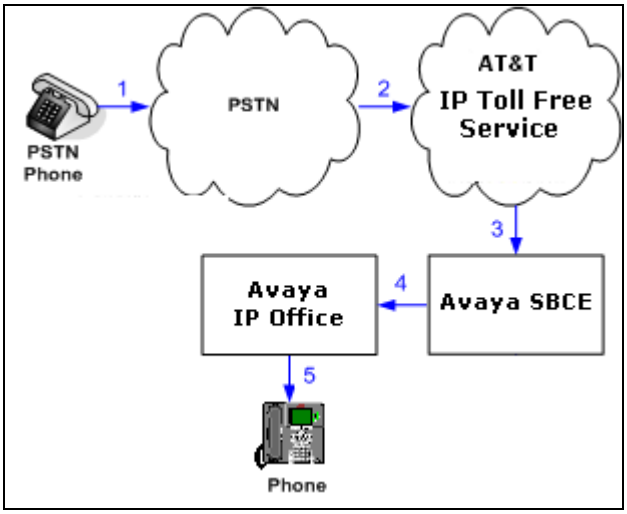

Figure 2: Inbound AT&T IPTF Call

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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 Voicemail Pro, running on Avaya Server Edition Primary 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.

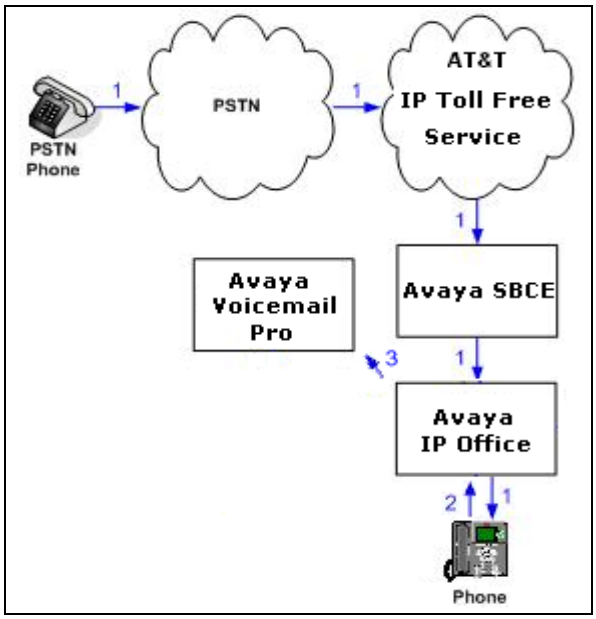

Figure 3: Coverage to Voicemail (Voicemail Pro)

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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 Server Edition ▪ IP Office ▪ Voicemail Pro ▪ Avaya WebRTC Gateway ▪ Avaya one-X® Portal for IP Office	 10.0.0.1.0 build 53 10.0.0.0.0 build 469 10.0.0.1.0 build 3 10.0.0.1.0 build 16
Avaya IP Office 500 V2 ▪ Avaya IP Office TCM 8 ▪ Avaya IP Office COMBO6210/ATM4	 10.0.0.1.0 build 53 10.0.0.1.0 build 53
Avaya IP Office Server Edition Manager	10.0.0.1.0 build 53
Avaya Communicator for Windows (SIP)	2.1.3.80
Avaya 9641G (H.323) IP Deskphone	6.6302
Avaya 1616 (H.323) Telephone	Ha1616ua1_390A.bin
Avaya 1140E (SIP) Telephone	04.04.23
Avaya 9508 Digital Telephone	0.59
Analog Fax device	Ventafax 7.8
Avaya SBCE	7.1.0.0-04-11122

Table 2: Equipment and Software Versions

Note – Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2, and also when deployed with all configurations of IP Office Server Edition. IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

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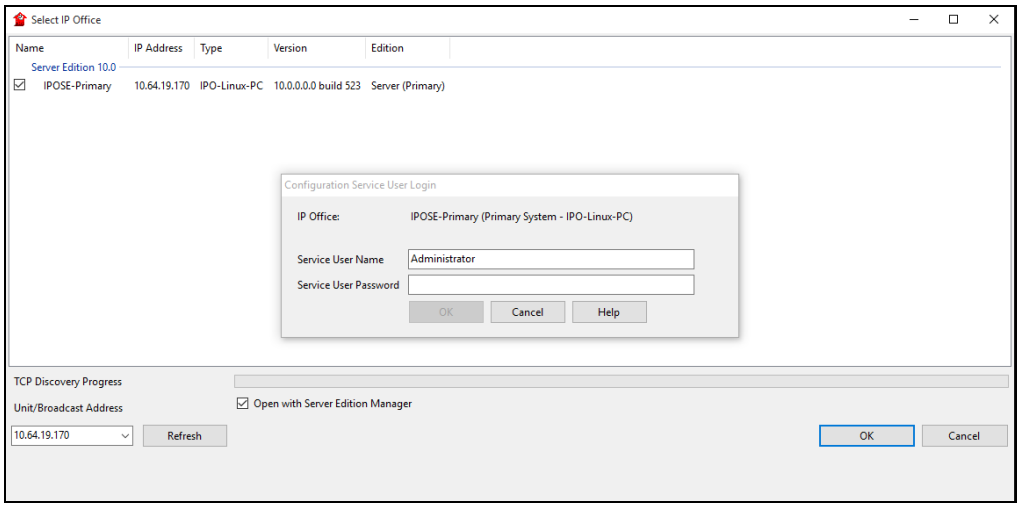
5. Avaya IP Office Primary Configuration

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

IP Office is configured via the IP Office Manager program. For more information on IP Office Manager, consult references **[1]**, **[2]**, and **[3]**. From the IP Office Manager PC, select **Start → All Apps → IP Office → Manager** to launch the Manager application. Navigate to **File → Open Configuration** (not shown), select the proper Avaya IP Office system from the pop-up window, and log in using the appropriate credentials.



On Server Edition systems, the Solution View screen will appear, similar to the one shown on next page. If the left navigation pane does not immediately appear, click on the **Configuration** link as highlighted below. In the reference configuration, IP users registered to the Primary server and failover to the Secondary server. Digital and Analog users are configured on the Expansion System. A SIP trunk to the AT&T IPTF service is configured on the Primary server. Clicking the "plus" sign next to the Primary server system name, e.g., **IPOSE-Primary,** on the left navigation pane will expand the menu on this server.

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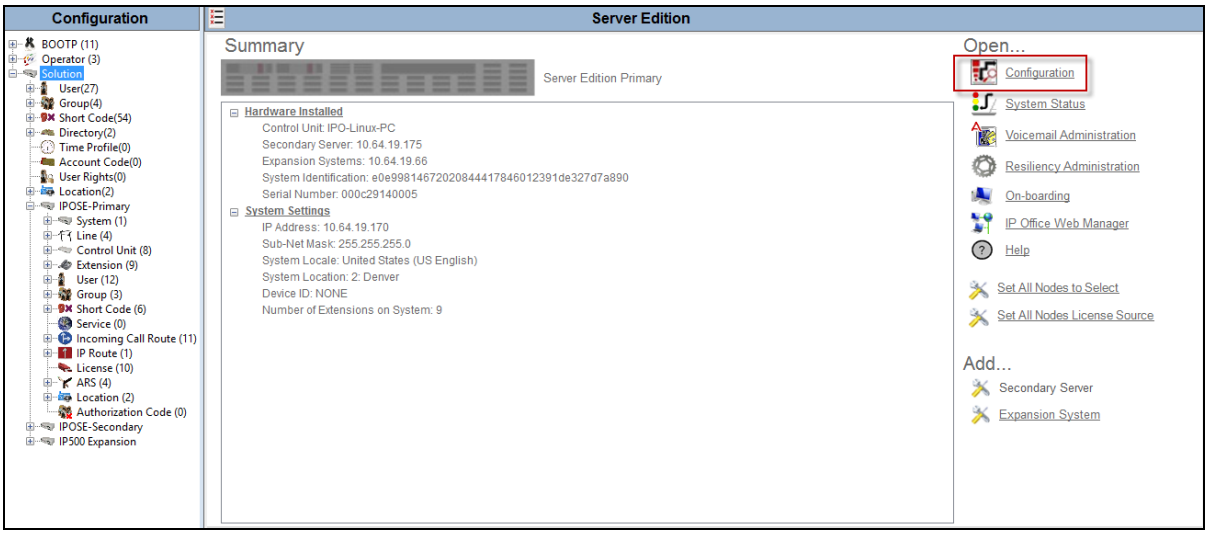
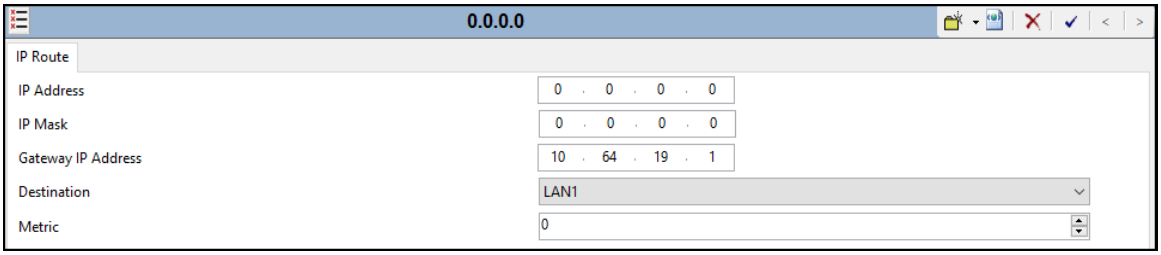


5.1. IP Route

In the reference configuration, the Primary server LAN1 port is physically connected to the CPE network. The default gateway for this network is **10.64.19.1**. To add an IP Route in Avaya IP Office, right-click **IP Route** from the left hand Navigation pane, and select **New** (not shown). To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the relevant default route using **Destination → LAN1**.



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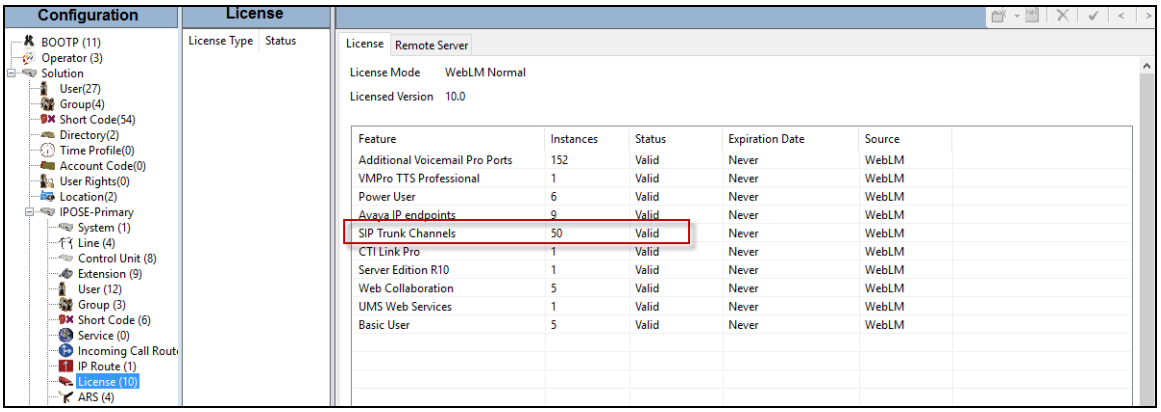
5.2. Licensing

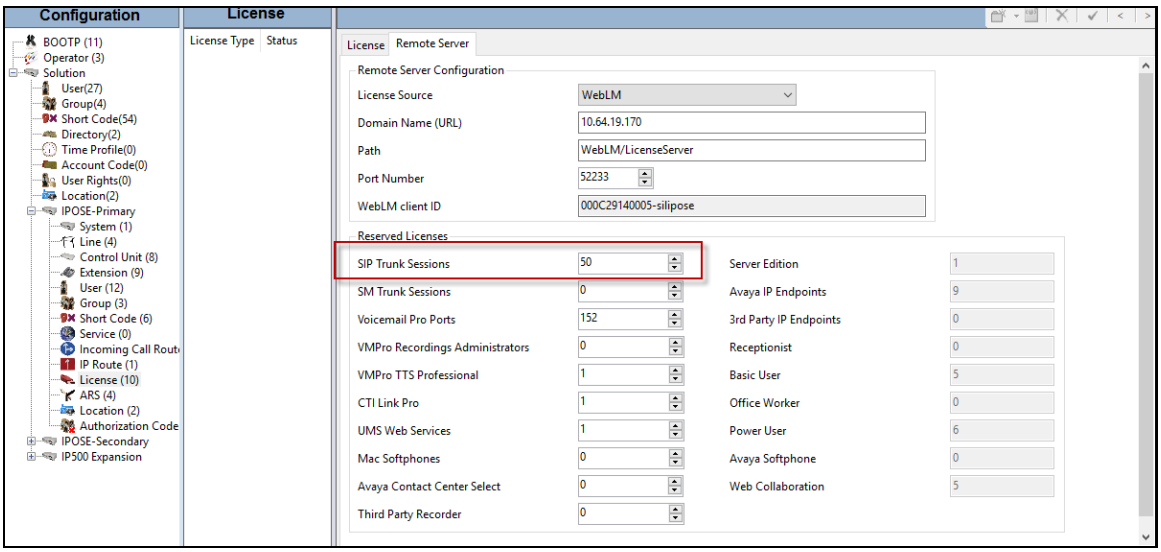
In the reference configuration, **IPOSE-Primary** was used as the system name of the Primary Server, **IPOSE-Secondary** was used as the system name of the Secondary Server and **IP500 Expansion** was used as the system name of the Expansion System. All navigation described in the following sections (e.g., **License**) appears as submenus underneath the system name in the Navigation Pane.

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

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



On the **Remote Server** tab, set the **SIP Trunk Sessions** to a value sufficient for the number of simultaneous SIP trunk calls needed.



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5.3. System Settings

This section illustrates the configuration of system settings. Select **System** in the Navigation pane to configure these settings. The subsection order corresponds to a left to right navigation of the tabs in the Details pane for System settings. For all of the following configuration sections, the **OK** button (not shown) must be selected in order for any changes to be saved.

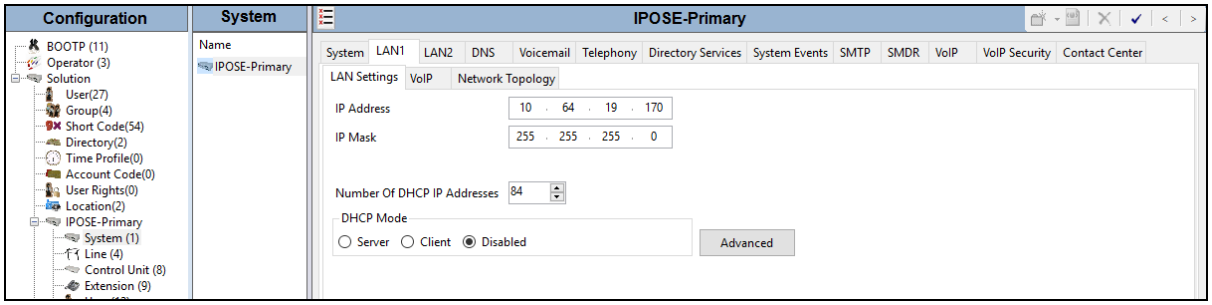
5.3.1. LAN1 Tab

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

5.3.1.1 LAN1 – LAN Settings Tab

To view or configure the LAN1 IP address, select the **LAN1 → LAN Settings** tab, and enter the following:

- **IP Address:** Set to **10.64.19.170** as specified in the reference configuration.
- **IP Mask:** Set to **255.255.255.0** as specified in the reference configuration.



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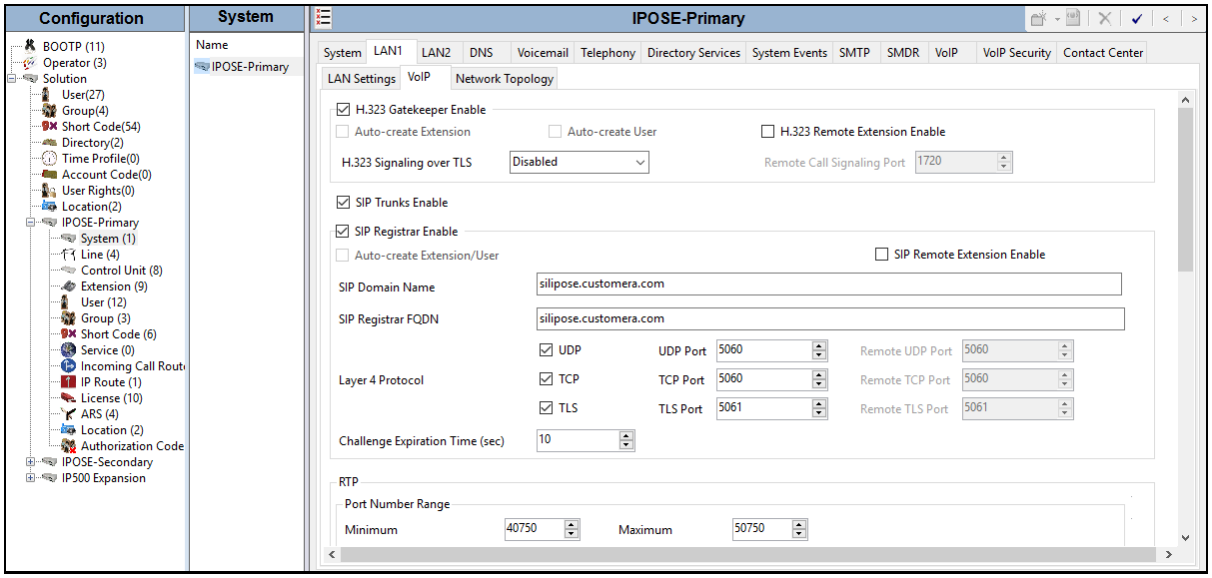
5.3.1.2 LAN1 - VoIP Tab

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

- The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 9600-Series Telephones used in the reference configuration.
- Select the **SIP Trunks Enabled** option.
- The **SIP Registrar Enable** box is checked to allow Avaya 1140E (SIP) and Avaya Communicator for Windows (SIP) usage.
- The **SIP Domain Name** used in the reference configuration is **silipose.customera.com**.
- The **SIP Registrar FQDN** used in the reference configuration is **silipose.customera.com**.
- In the **Layer 4 Protocol** section, select **UDP/5060**, **TCP/5060**, **TLS/5061**.
- Let all other values default.



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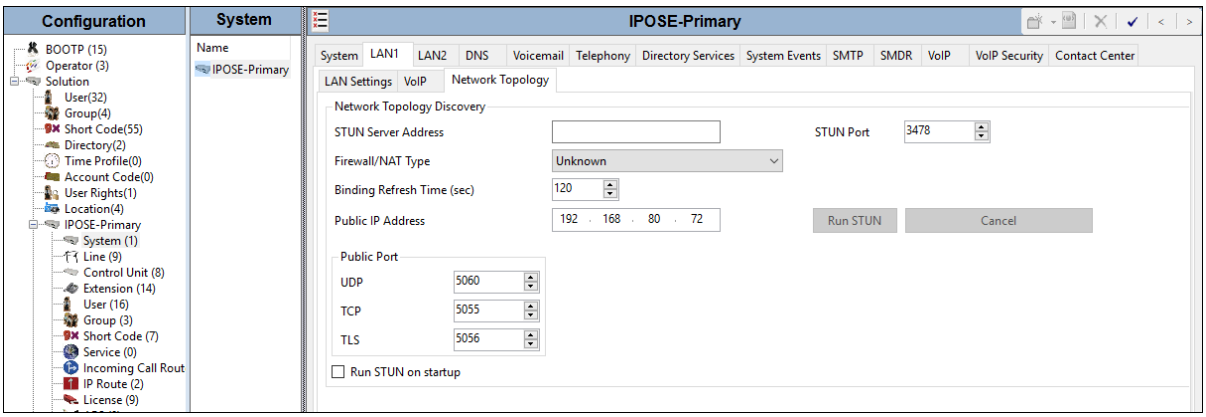
5.3.1.3 LAN1 - Network Topology Tab

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

- **STUN Server Address** is left blank as NAT is not required in this configuration
- **Firewall/NAT Type** is set to **Unknown** in the reference configuration. Note that the **Firewall/NAT Type** parameter may need to be set differently, depending on the type of firewall or Network Address Translation device used at the customer premise.
- **Binding Refresh Time**: Set to an appropriate interval (e.g., **120** seconds) for Avaya IP Office to send "keep alive" OPTIONS messages to the Avaya SBCE/AT&T.
- **Public IP Address** and **Public Port** section relates to remote workers, and is not used for the AT&T IPTF SIP trunk service connection.

Click the **OK** button (not shown).



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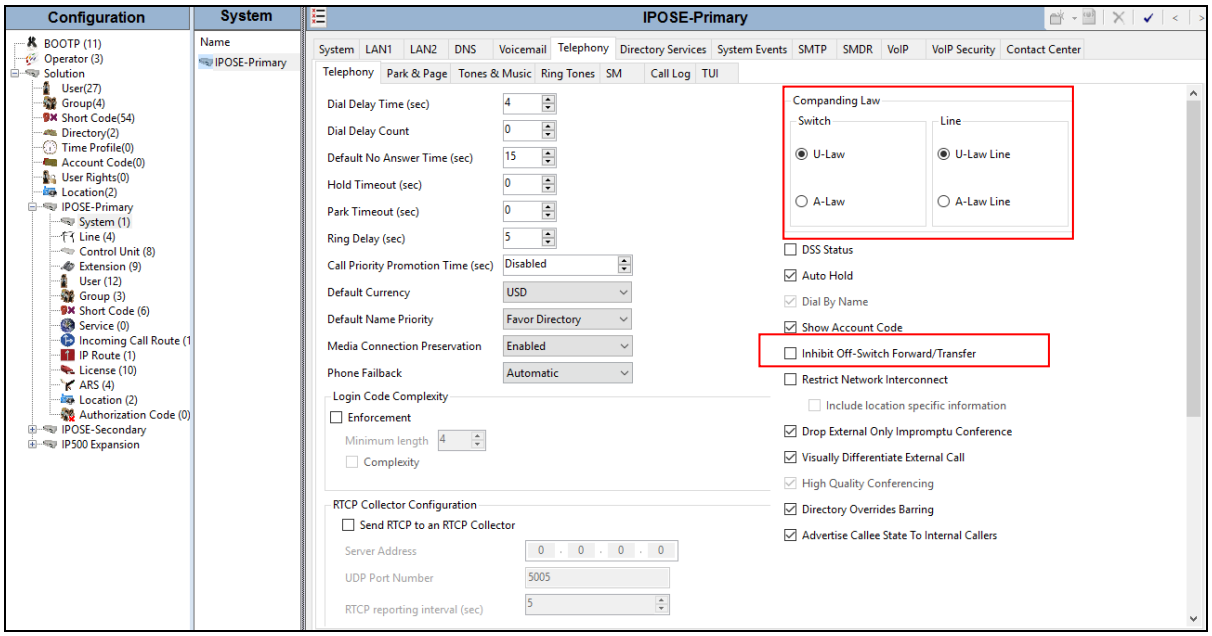
5.3.2. LAN2 Tab

The LAN2 interface is not used in the reference configuration.

5.3.3. Telephony Tab

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

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



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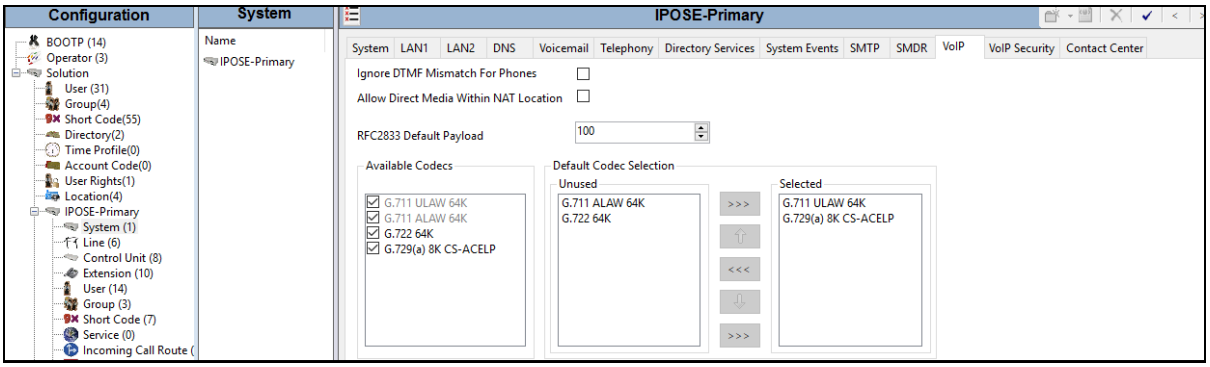
5.3.4. VoIP Tab

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

The up and down arrow buttons are used to order the selected codecs. By default, all IP (SIP and H.323) lines and extensions will assume the system default **Selected** codec list, unless configured otherwise for the specific SIP Line or extension (see the note below).

- Populate the **Selected** column with **G.711 ULAW 64K** as the first codec and **G.729(a) 8K CS-ACELP** as the second codec.
- In the **RFC2833 Default Payload** setting field, specify **100**, which is the recommended value for AT&T interoperability.

Note – In the reference configuration, the Extension codec lists (see **Section 5.6.2**) also specify *G.711ULAW* and *G.729(a)* (in that order), and the SIP Line (see **Section 5.4.6**) offers *G.729(a)* and *G.711ULAW* (in that order). In this manner, local Avaya IP Office calls will offer G.711mu first, and SIP trunk calls will offer G.729A first.



5.4. SIP Line

The following sections describe the configuration of a SIP Line. The SIP Line terminates the CPE end of the SIP trunk to the AT&T IPTF service.

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

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

- IP addresses
- SIP Credentials (if applicable)

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- SIP URI entries
- Setting of the **Use Network Topology Info** field on the Transport tab.

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

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

- SIL Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Requirements
- SIP Advanced Engineering

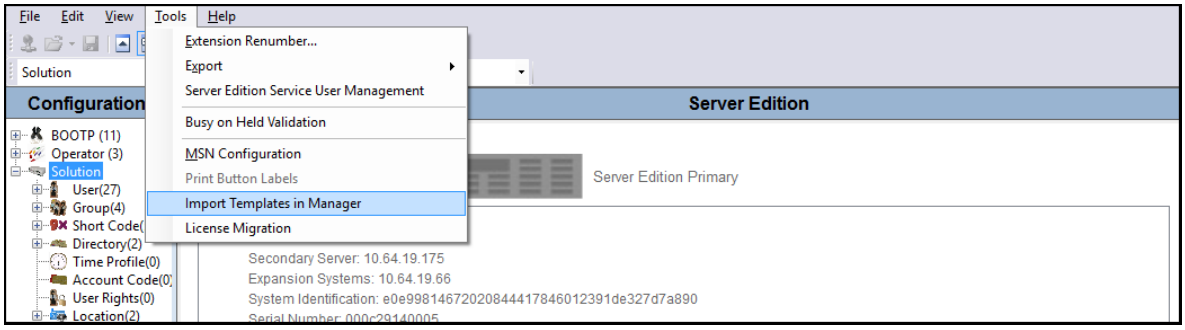
Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.3 – 5.4.8**.

5.4.1. Importing a SIP Line Template

Note – DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (IP500 V2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer's environment.

1. Copy a previously created template file to a location (e.g., *\temp*) on the same computer where IP Office Manager is installed.

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



3. A folder browser will open (not shown). Select the directory used in **step 1** to store the template(s) (e.g., *\temp*). The template files are automatically copied into the IP Office default template location, **C:\Program Files\Avaya\IP Office\Manager\Templates**.

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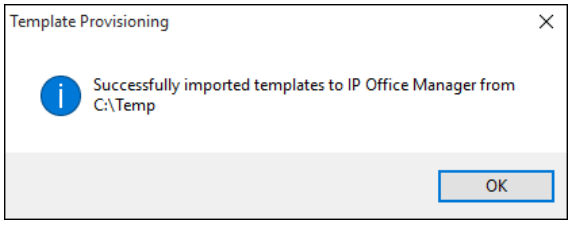
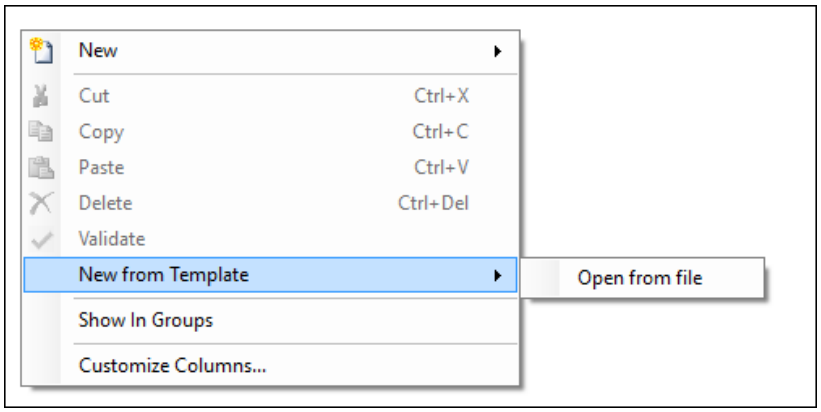
4. After the import is complete, a final import status pop-up window will open stating success or failure.

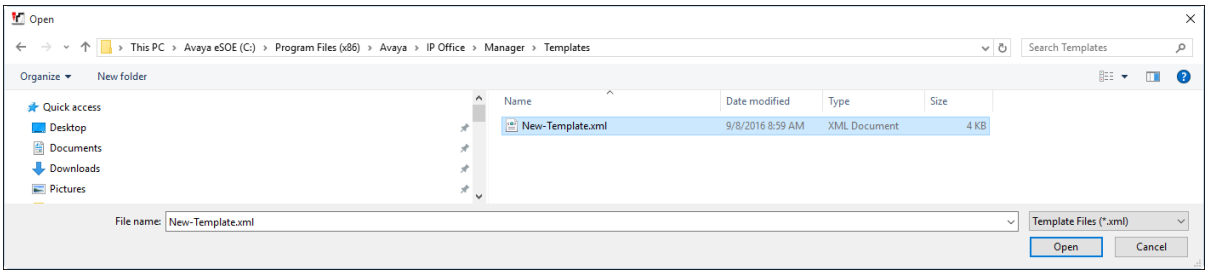


5.4.2. Creating a SIP Trunk from an XML Template

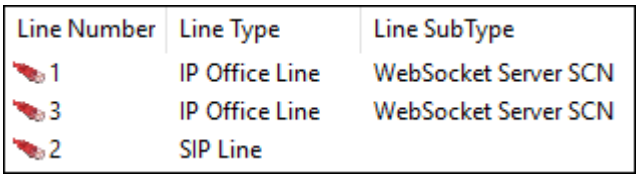
1. To create the SIP Trunk from a template, right-click on **Line** in the Navigation Pane, and hover over **New from Template**, and select **Open from file**.



Navigate to **C:\Program Files\Avaya\IP Office\Manager\Templates**. Select ***.xml** as the file type, find the template, and click **Open**.



The newly created SIP Line will appear in the Navigation pane (e.g., SIP Line **2**).



Line Number	Line Type	Line SubType
1	IP Office Line	WebSocket Server SCN
3	IP Office Line	WebSocket Server SCN
2	SIP Line	

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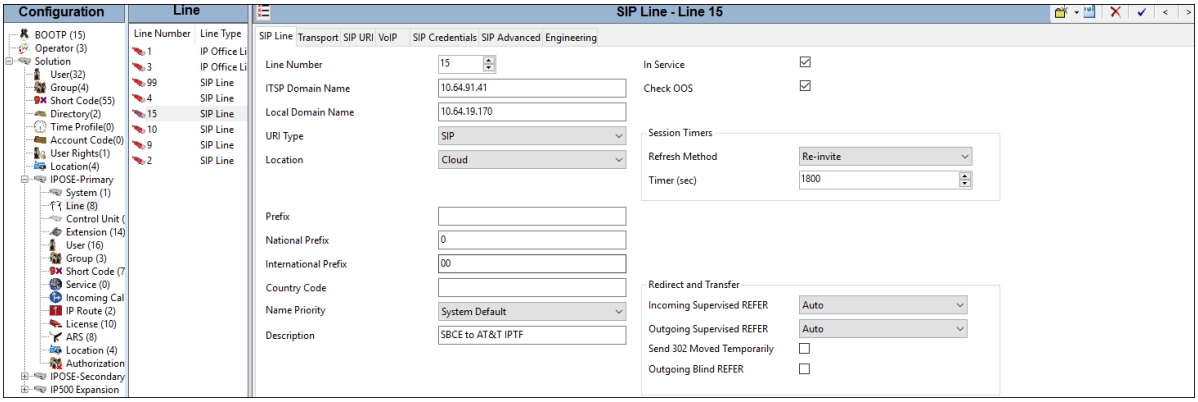
Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.3 – 5.4.8**.

5.4.3. SIP Line – SIP Line tab

The **SIP Line** tab is shown below for **Line Number 15**, used for the SIP Trunk to AT&T. Note, if no SIP Line exists, right click on the **Line** item in the **Navigation** pane and select **New → SIP Line** (not shown). In the reference configuration, SIP Line 15 was created. The SIP Line form is completed as follows:

- **ITSP Domain Name**: Set to the IP address of the Avaya SBCE "A1" interface (e.g., **10.64.91.41**).
- **Local Domain Name:** Set to the IP address of the Avaya IP Office LAN1 interface (e.g., **10.64.19.170**).
- **In Service** and **Check OOS**: These boxes are checked (default).
- **Refresh Method:** Set to **Re-Invite**, as AT&T does not support UPDATE.
- **Incoming Supervised Refer**: Set this field to **Auto** (default).
- **Outgoing Supervised Refer**: Set this field to **Auto** (default).
- **Send 302 Moved Temporarily:** Verify this field is unchecked (default).
- **Outgoing Blind REFER**: Verify this field is unchecked (default).
- Use the default values for the other fields.
- Click **OK** (not shown).



5.4.4. SIP Line - Transport tab

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

- **ITSP Proxy Address:** Set to the Avaya SBCE "A1" interface (e.g., **10.64.91.41**).
- **Network Configuration → Layer 4 Protocol**: Set to **UDP**.
- **Network Configuration → Send Port**: Set to **5060** (default).
- **Network Configuration → Use Network Topology Info**: Set to **None**.
- **Calls Route via Registrar:** Verify this field is checked (default).
- **Click OK** (not shown).

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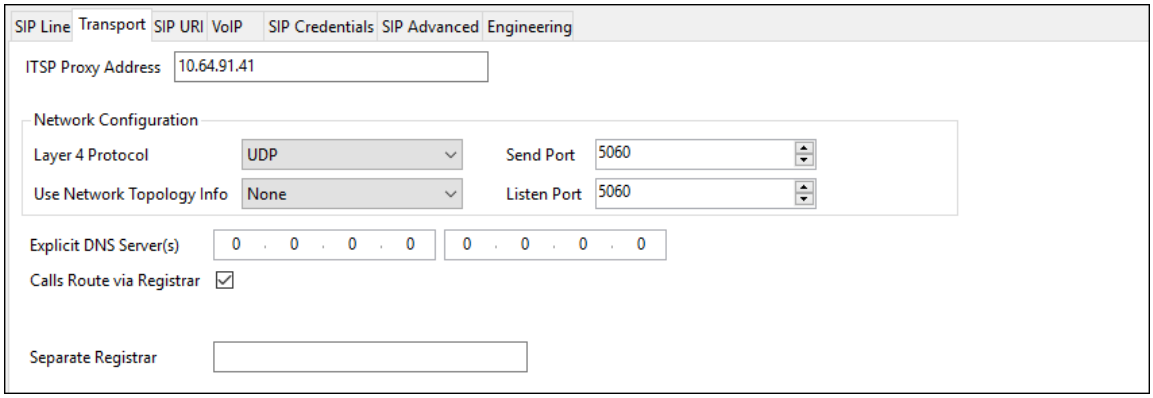
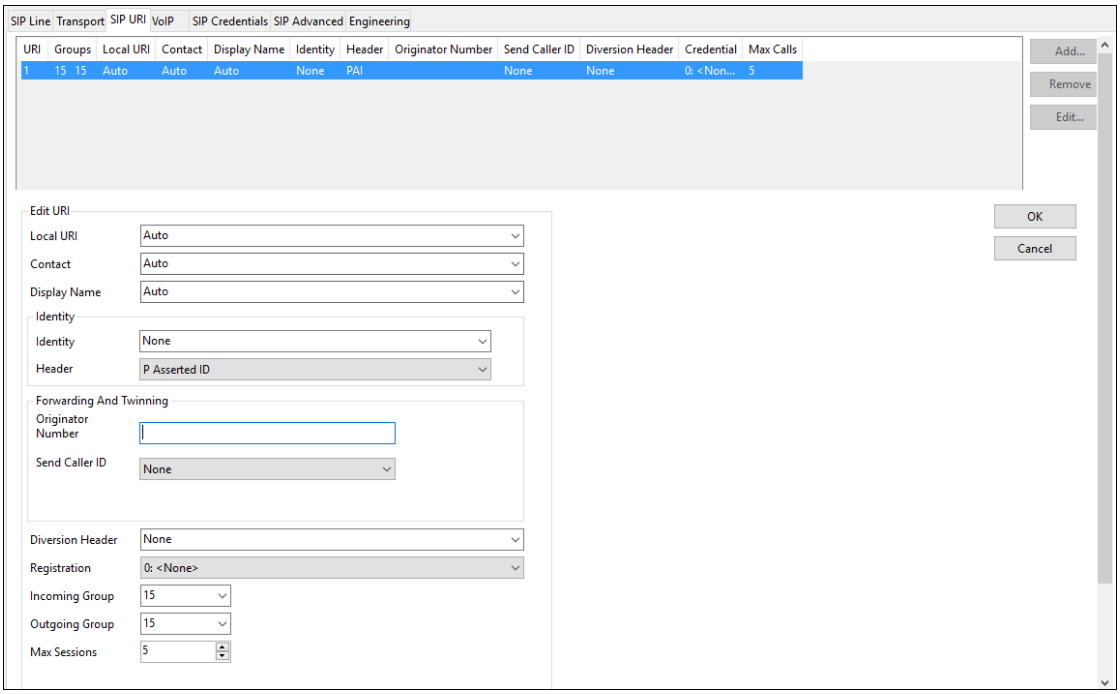


5.4.5. SIP Line - SIP URI tab

Select the **SIP Line → SIP URI** tab. To add a new SIP URI, click the **Add…** button. At the bottom of the screen, a **New Channel** area will be opened. Configure the following:

- **Local URI**, **Contact**, and **Display Name** fields: Set these fields to **Auto**. This setting replaces the wildcard "*" used in previous releases.
- Verify **Identity**, **Send Caller ID**, and **Diversion Header:** Set to the default **None**.
- Verify **Registration:** Set to the default **0: <None>**.
- **Incoming Group:** Set to an unused group number, e.g., **15**. This value references the table created with **Incoming Call Routes** in **Section 5.7**.
- **Outbound Group:** Set to an unused group number, e.g., **15**.
- **Max Sessions:** In the reference configuration, this was set to **5**. 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**.



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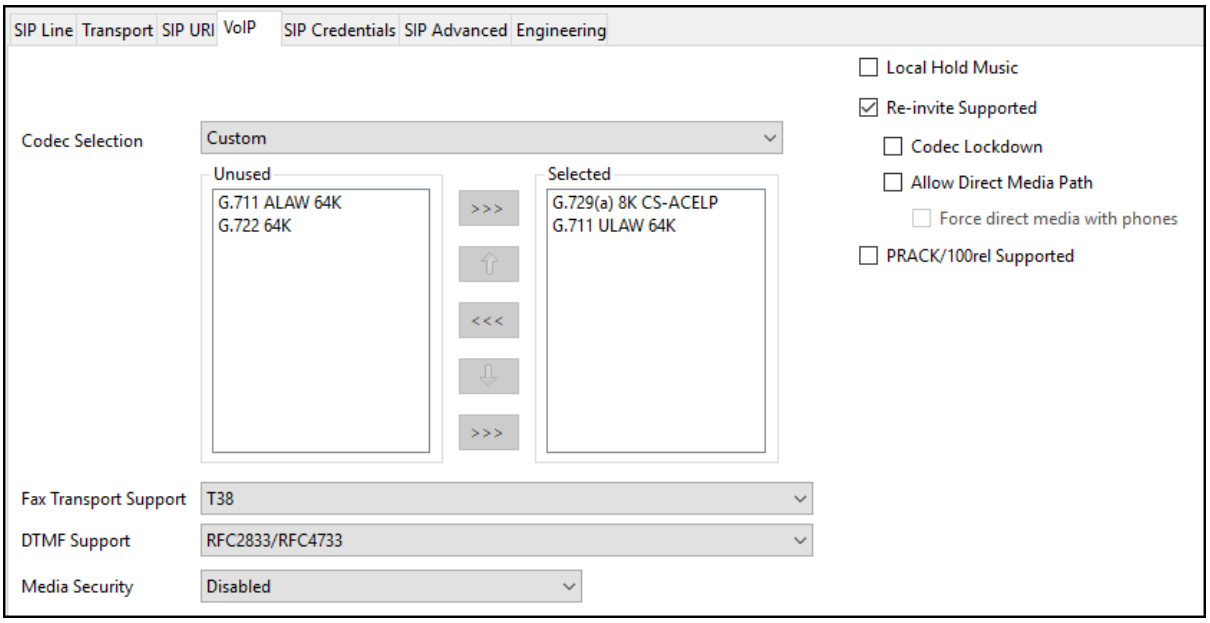
- To edit an existing entry, click an entry in the list and click the **Edit** button.
- When all SIP URI entries have been added or edited, click **OK** at the bottom of the screen (not shown).

5.4.6. SIP Line - VoIP tab

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

- The **Codec Selection** drop-down box → **System Default** will list all available codecs. In the reference configuration, **Custom** was selected and **G729(a) 8K CS-ACELP,** and **G.711 ULAW 64K** were specified. This causes Avaya IP Office to include these codecs in the Session Description Protocol (SDP) offer, and in the order specified. Note that in the reference configuration G.729A is set as the preferred codec on the SIP trunk to the AT&T IPTF network.
- T.38 fax was used in the reference configuration. Set the **Fax Transport Support** drop-down menu to **T38**. G.711 fax also worked in the reference configuration (T.38 option disabled); however, T.38 is the preferred method.
- The **Re-invite Supported** parameter can be checked to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk.
- The **DTMF Support** parameter can remain set to the default value **RFC2833/RFC4733**.
- Click **OK** (not shown).



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5.4.7. SIP Line - T38 Fax Tab

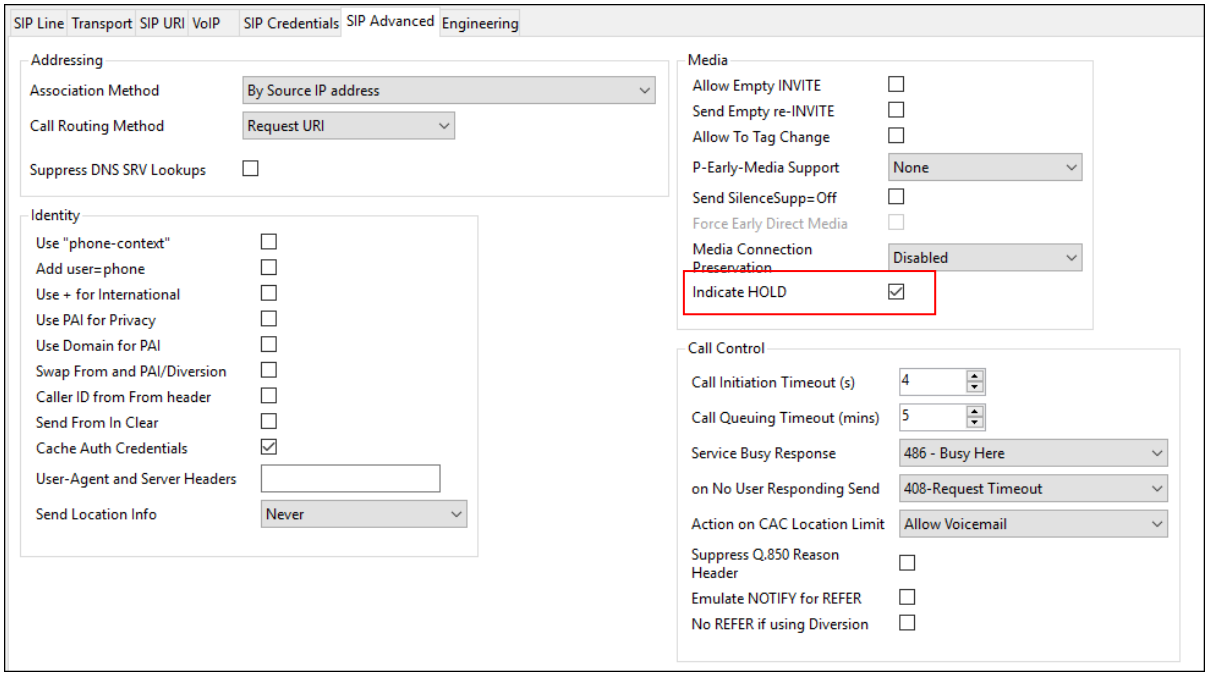
Note – This tab is only available when configuring a SIP line on IP Office 500 V2, and the settings on this tab are only accessible if **Re-invite Supported** and a **Fax Transport Support** option (**T38**) are selected on the **VoIP** tab (**Section 5.4.6**). See **Section 6.4** for T.38 fax settings.

5.4.8. SIP Line – SIP Advanced Tab

IP Office can be configured to signal when a call is placed on hold by sending an INVITE with media attribute "sendonly". AT&T in turn will respond with media attribute "recvonly", and will stop sending RTP media for the duration the call is on hold. When the call is taken off of hold, IP Office will send another INVITE with media attribute "sendrecv" indicating to AT&T to start sending RTP again.

To have Avaya IP Office signal to AT&T when a call is placed on/off hold, select the **SIP Line → SIP Advanced** tab and enter the following:

- Select **Indicate HOLD**.



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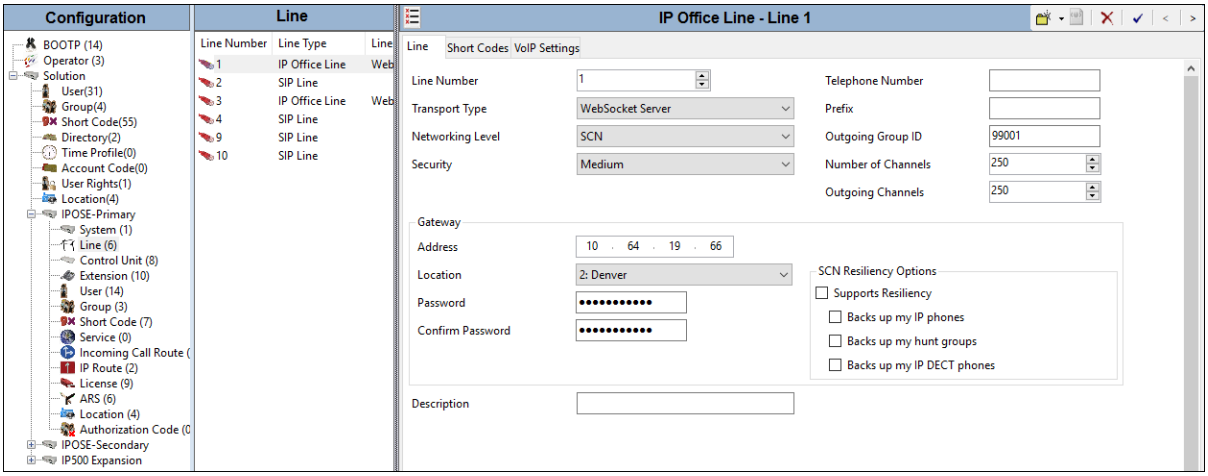
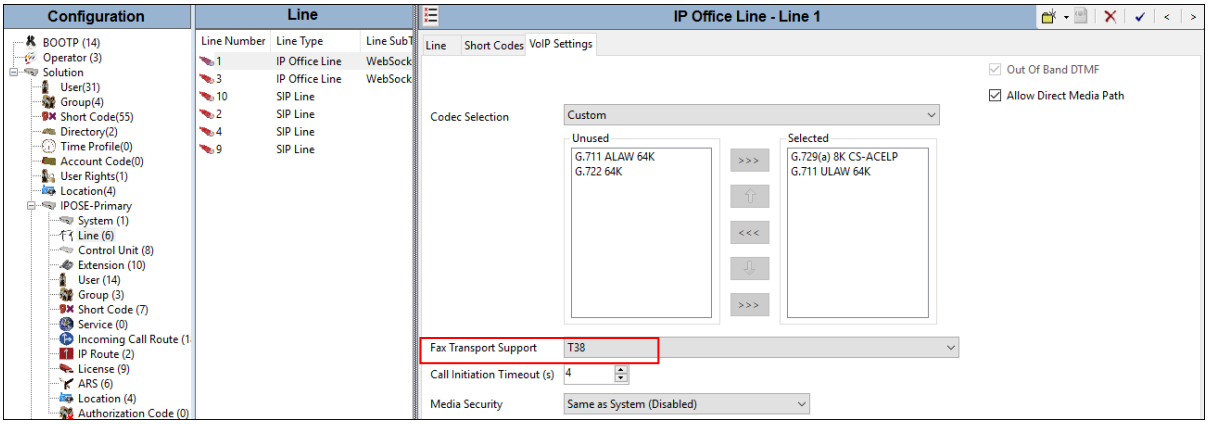
5.5. IP Office Line

IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. To edit an existing IP Office Line, select **Line** in the Navigation pane, and select the appropriate line to be configured in the Group pane. Below is the IP Office Line to the Expansion System.



In the reference configuration, a fax machine is connected to one of the analog ports on the Expansion System. To accommodate T.38 fax, select the **VoIP Settings** tab and configure the following:

- **Fax Transport Support: T38**



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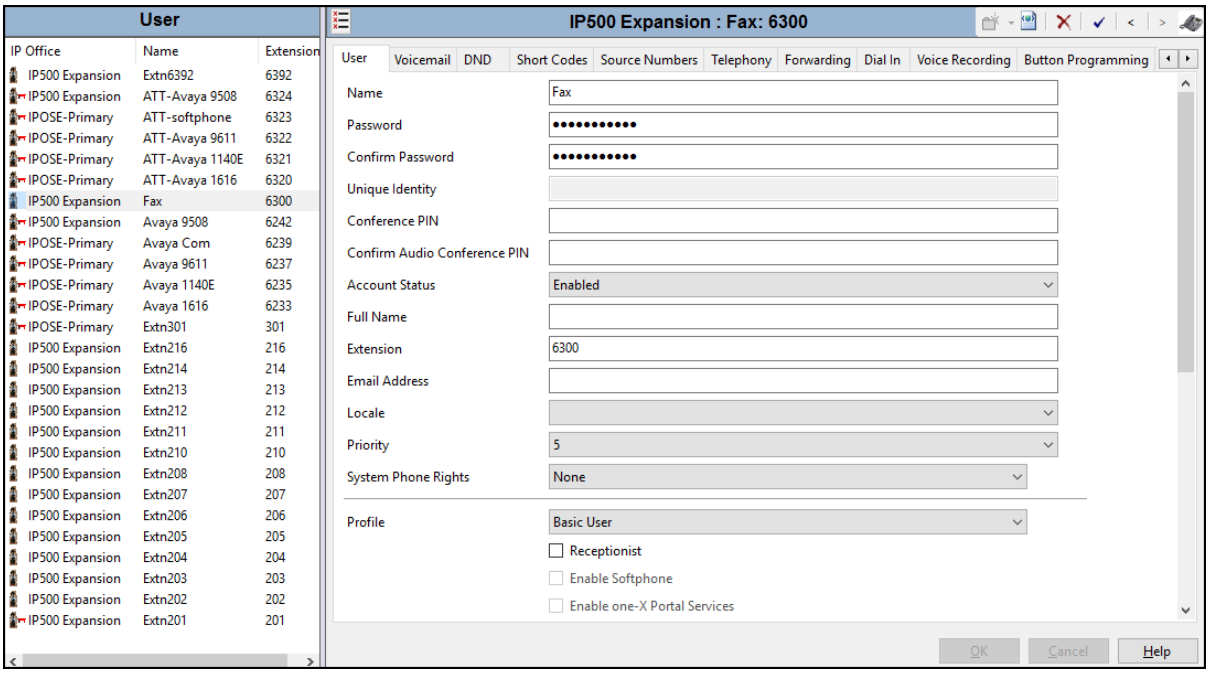
5.6. Users, Extensions, and Hunt Groups

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

5.6.1. Analog User 6300

The following screen shows the **User** tab for analog phone User **6300**. This user corresponds to a fax machine.

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



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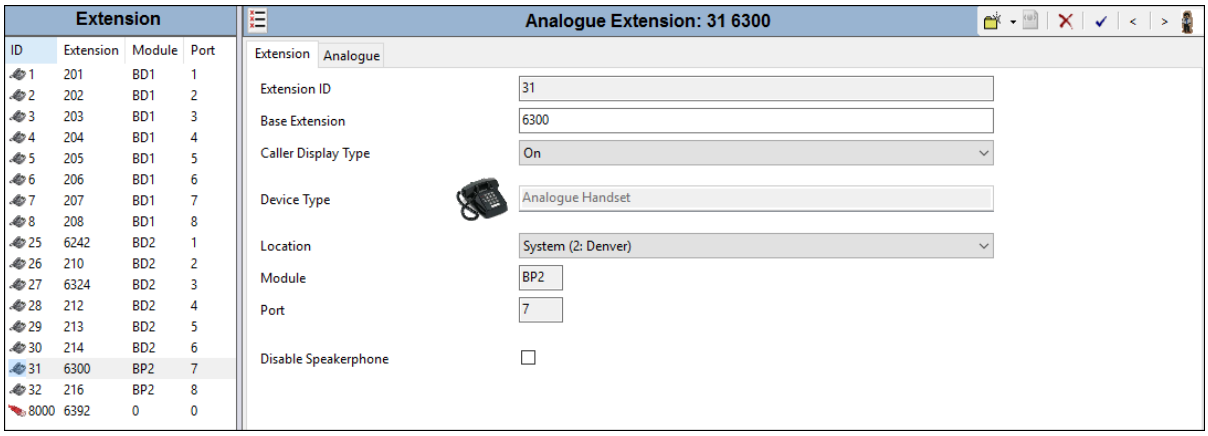
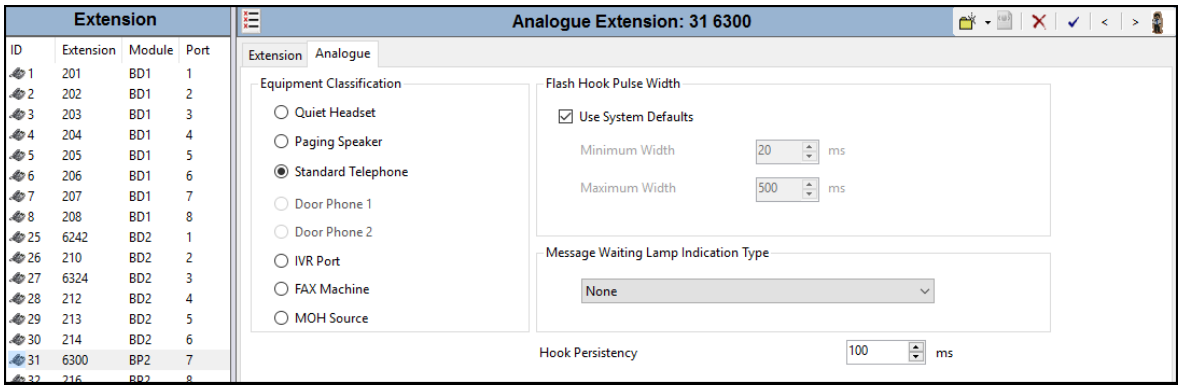
2. Analog (or digital) phone extension ports are either integral to the control unit or added by the installation of an analog or digital phone expansion module. Analog (or digital) extension records are automatically created for each physical extension port within the system. These ports cannot be added or deleted manually. For Server Edition, non-IP extensions are only supported on Expansion System (IP500 V2) units.
   - To edit an existing analog extension, select the appropriate extension to be configured (e.g., **6300**).



   - Select the Analogue tab and verify that **Standard Telephone** is selected. Note that even though a fax machine is connected, it needs to be classified as a standard telephone.
   - Click the **OK** button (not shown).



5.6.2. IP Phone User 6237

1. Following the steps shown in **Section 5.6.1**, create a 9611 H.323 IP phone user (e.g., **6237**).

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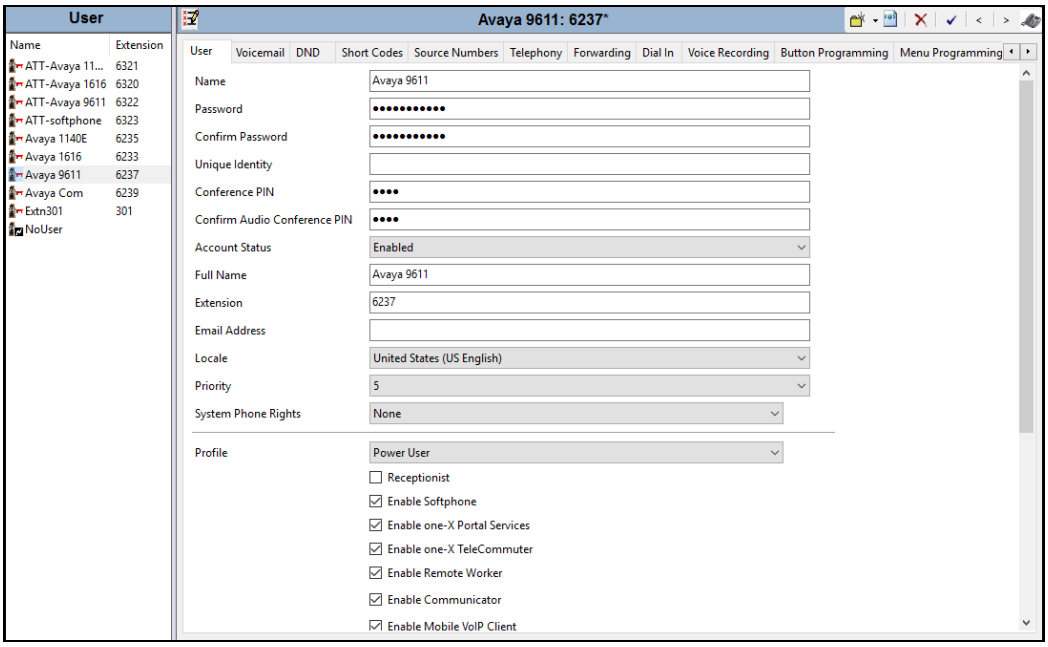
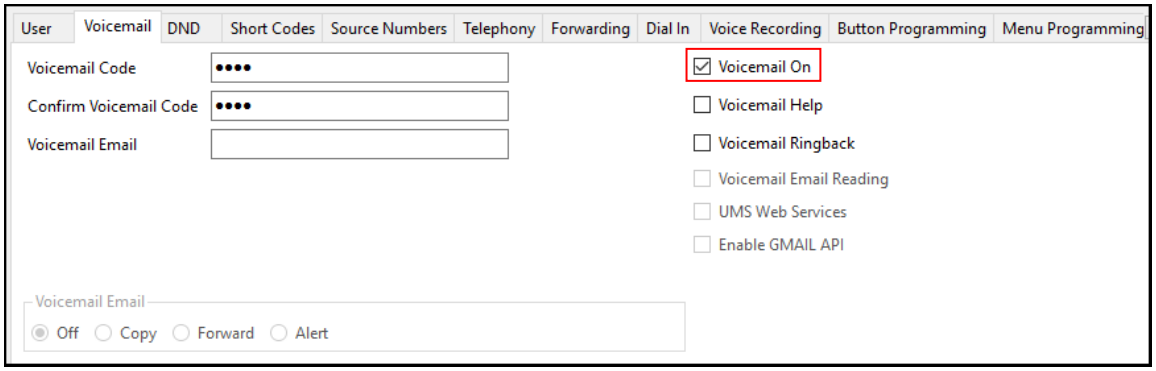
- **Password**: This password is used by user applications such as SoftConsole, one-X® Portal and TAPI, or users with Dial In access. Note that this is ***not*** the user's phone login code (see the information on the **Extension** tab below), or their Voicemail mailbox password (see information on the **Voicemail** tab below).
- **Conference PIN**: This is the pin number used to access the user's meet me conference.
- The **Profile** parameter is set to **Power User**. This gives this user access to additional IP Office features. See **[3]** for more information.



The following screen shows the **Voicemail** tab for user 6237. The **Voicemail On** box is checked and a Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters.



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

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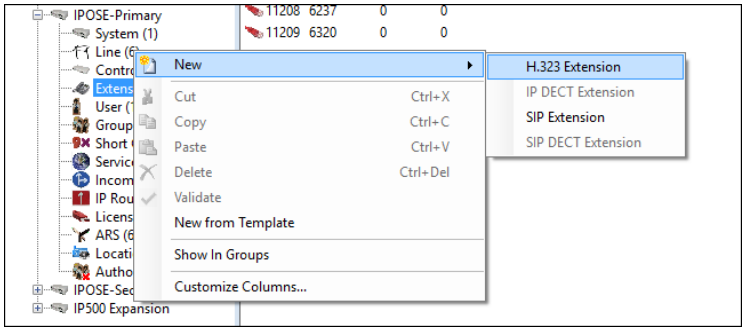
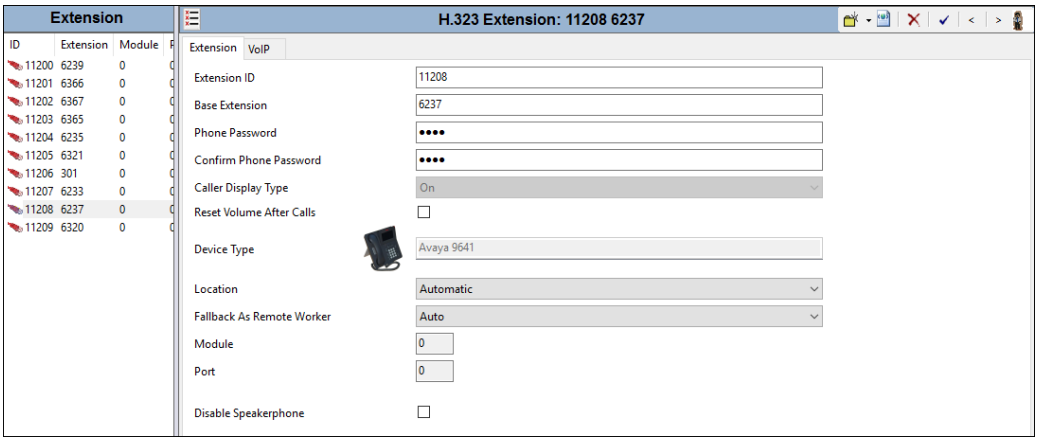
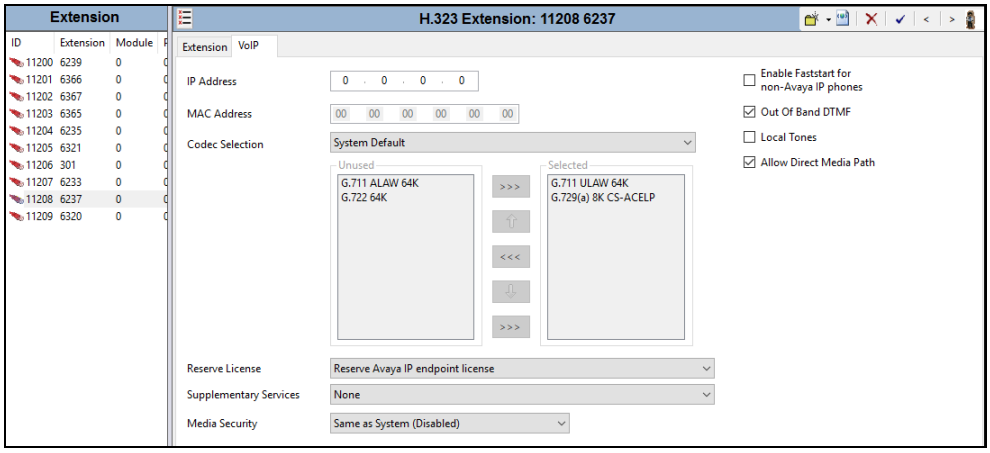


On the **Extension** tab, enter the **Base Extension** (e.g., **6237**). Note that the **Extension ID** field will auto populate. The **Phone Password** will be used by the telephone user as the phone login password.



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

- Keep the **IP Address** field as the default value (**0.0.0.0**).
- Populate the **Selected** column with **G.711 ULAW 64K** as the first codec and **G.729(a) 8K CS-ACELP** as the second codec, (see **Section 5.3.4**).
- Click the **OK** button (not shown).



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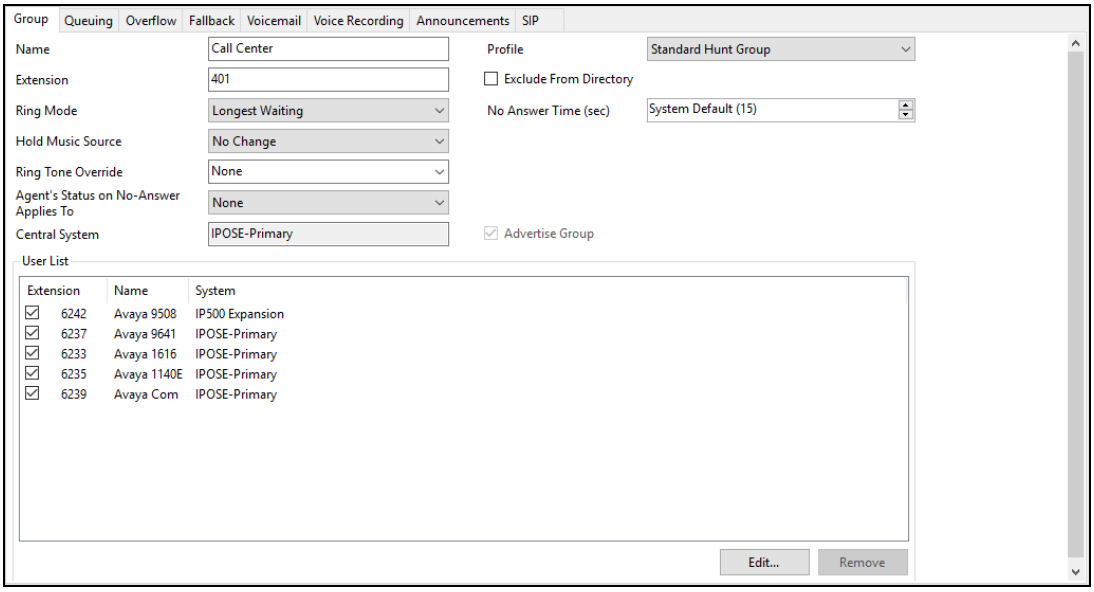
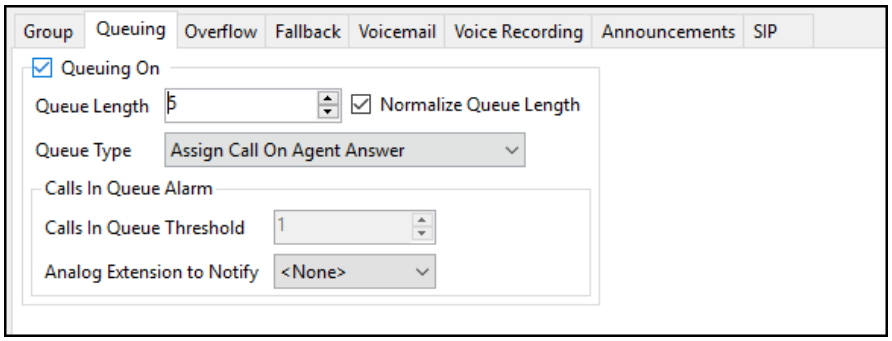
5.6.3. Hunt Groups

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

1. The following screen shows the **Group** tab for hunt group **Call Center**. This hunt group was configured to contain various IP Office extensions. In the reference configuration, these telephones extensions are rung based on idle time, due to the **Ring Mode** setting **Longest Waiting**. Click the **Edit** button to select/deselect from the **User List** included in the Hunt Group from the list of available users.



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



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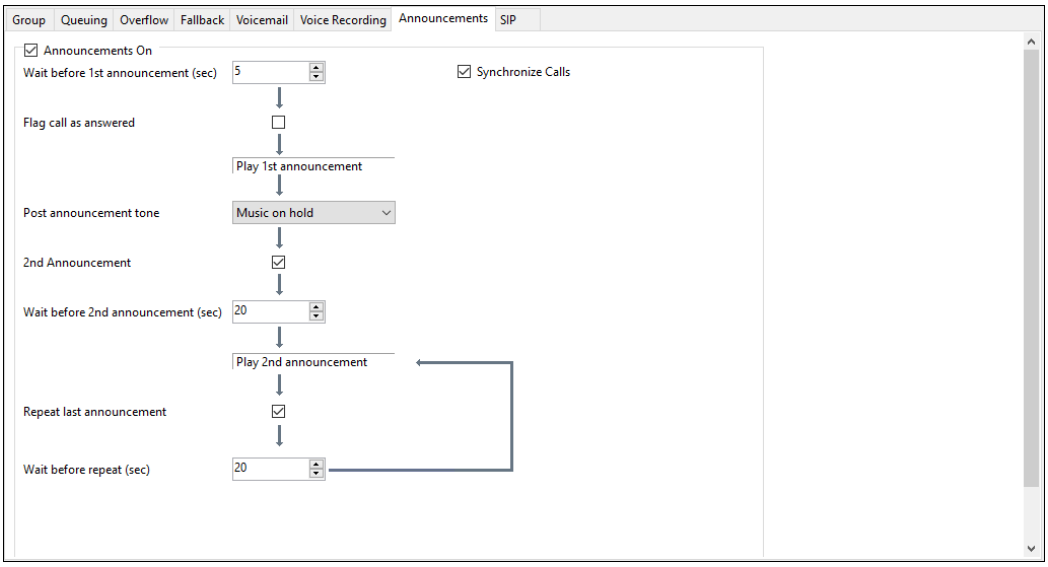
3. Under the **Announcements** tab, check the **Announcements On** box. The wait time can be set to any desirable value. The **Synchronize Calls** box is checked to greatly reduce the number voicemail channels needed to play announcements. These announcements are played if an agent for a particular skill is unavailable.



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

In the reference configuration, these steps were used to create additional Hunt Group "Support" (402).

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5.7. Incoming Call Routes

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

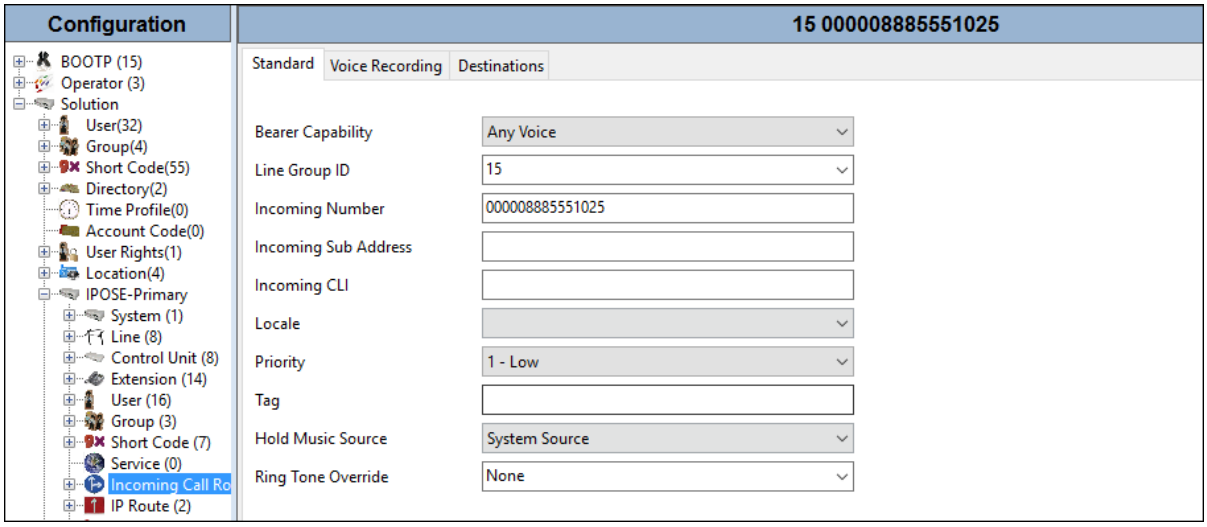
The Incoming Call Route table will map specific AT&T DNIS numbers to an IP Office User, or Hunt Group, as well as to Voicemail Pro scripts.

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

5.7.1. Calls to IP Office Stations and Hunt Groups

In the example below, the incoming number **000008885551025** is directed to H.323 phone 6237.

1. On the **Standard** tab enter the following:
   - **Line Group ID**: Enter the SIP Line defined in **Section 5.4** (e.g., **15**).
   - **Incoming Number**: Enter the associated DNIS digits sent by AT&T (e.g., **000008885551025**).
   - Use default values for the remaining fields and click **OK** (not shown).



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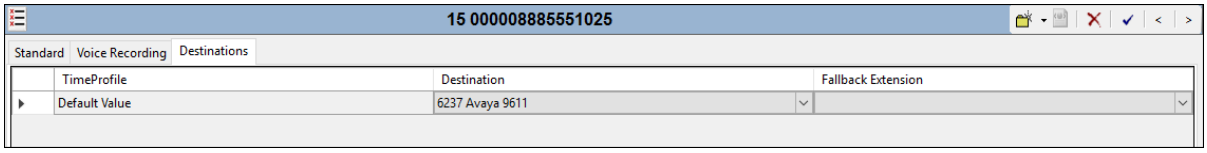
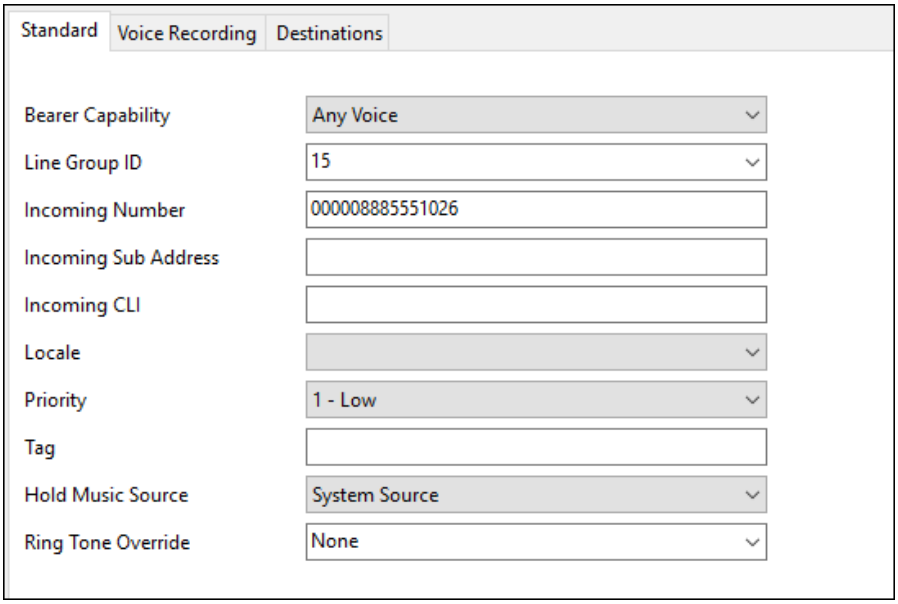
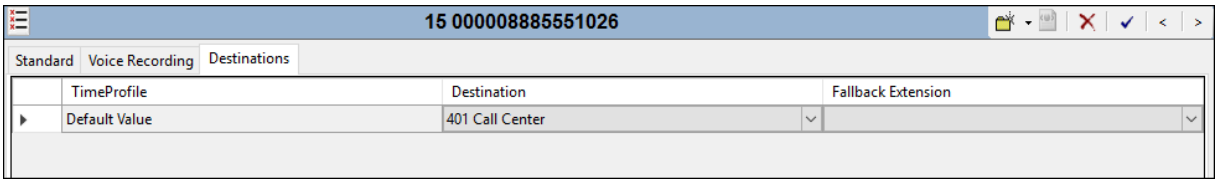
2. On the **Destinations** tab enter the following:
   - In the **Destinations** column, select extension **6237** from the drop down menu.
   - Use default values for the remaining fields and click **OK** (not shown).



Below is an example of a call for **000008885551026** being directed to Hunt Group **401** (Call Center).





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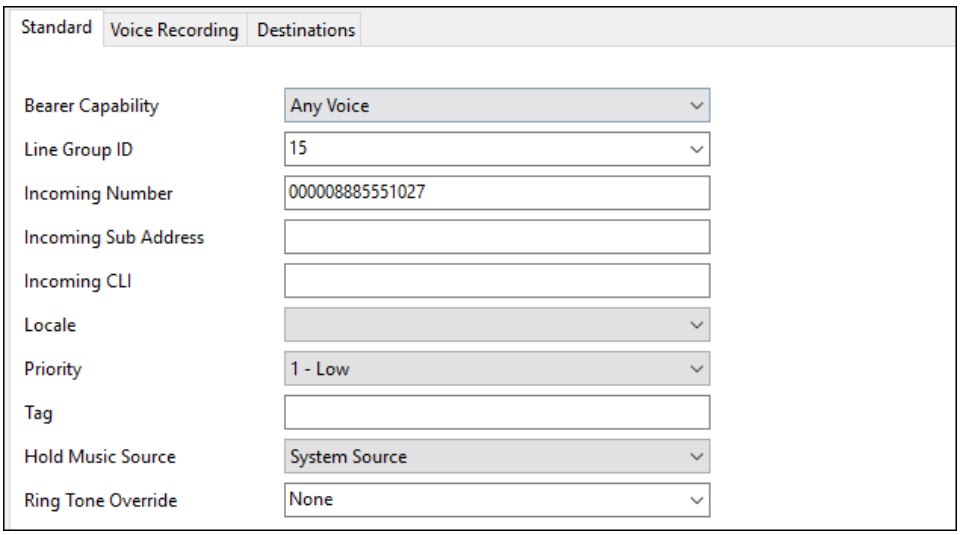
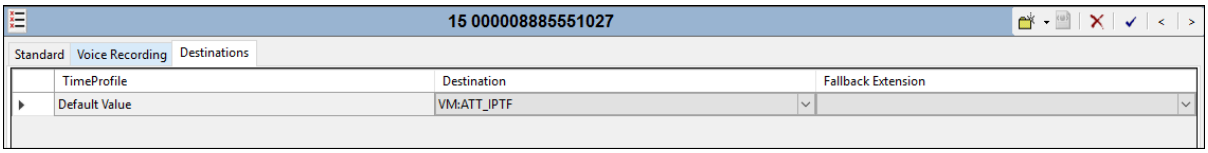
5.7.2. Calls to Voicemail Pro Scripts

As described in **Sections 5.8**, Voicemail Pro scripts are defined with specific names. These script names are specified as destinations in the Incoming Call Route table.
In the example below, incoming number **000008885551027** is directed to the Voicemail Pro Auto-Attendant script **ATT_IPTF**.

1. On the **Standard** tab repeat the steps in **Section 5.7.1**, with the following changes:
   - **Incoming Number**: Enter the associated DNIS digits sent by AT&T (e.g., **000008885551027**).
2. On the **Destinations** tab enter the following:
   - In the **Destinations** column, enter the string **VM:ATT_IPTF** from the drop down menu (note if the voicemail module does not appear in the list, enter the value manually).
   - Use default values for the remaining fields and click **OK** (not shown).





5.8. Call Center Provisioning in Voicemail Pro

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

In the reference configuration, Voicemail Pro is used for Voicemail processing as well as for simulating basic Call Center functionality.

The Auto-Attendant 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 (Call Center, and Support), or user 6237. This is accomplished via the following steps:

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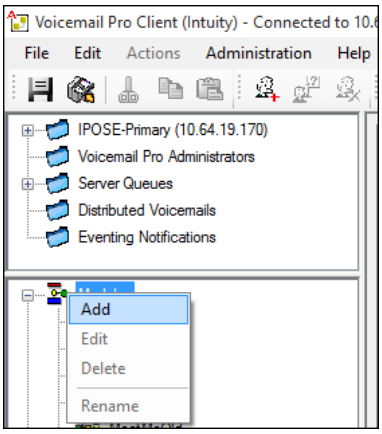
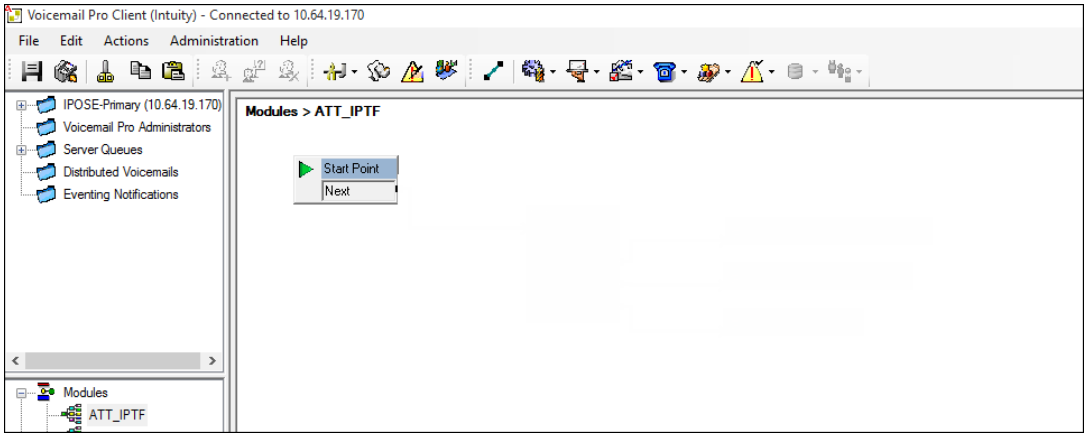
1. Hunt Groups **Call Center** and **Support** are created in IP Office (**Section 5.6.3**).
2. User 6237 is created in IP Office (**Section 5.6.2**).
3. Incoming Call Route for DNIS digits **000008885551027** is defined for access to the Auto-Attendant script (**Section 5.7.2**).
4. Via the Voicemail Pro GUI interface:
    - Open the **Voicemail Pro Client** application and log in to the Voicemail Pro server (not shown).
    - Create a **Start Point** by right clicking on **Modules** and selecting **Add**.



- Enter a name (e.g., **ATT_IPTF**) and click on **OK** (not shown). The new script **ATT_IPTF** will appear under Modules and a Start Point icon will appear in the work area.



- Click on the **Start Point** icon to activate the script options at the top of the screen. From the options, select the **Basic Actions** icon , select the **Menu** icon , and click on the work area to place the **Menu** icon.
    i. Double click the **Menu** icon.
        1. On the **General** tab → **Token Name**, enter **Menu** (not shown).

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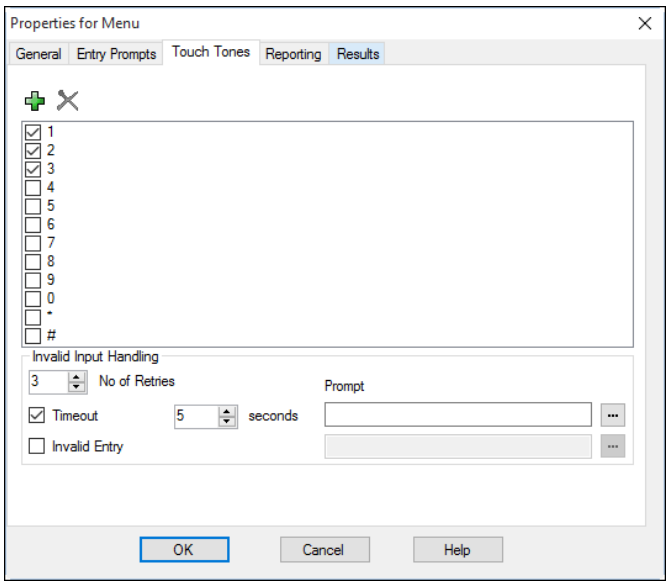
2. On the **Entry Prompts** tab (not shown), select or create an **Entry Prompt** that will tell the caller what digits to press (e.g., **mainmenu.wav**). To modify an existing recording, double click on the .wav file and rerecord. If no .wav files exist, double click on the icon to open the .wav editor.
3. On the **Touch Tone** tab:
    a. Select **1**, **2**, and **3** as the possible entry digits.
    b. Select **3** for **No of Retries**.
4. Click on **OK**.



- Click on the Telephony Actions icon , select the Transfer icon , and click on the work area to place the **Transfer** icon in the work area. This will be used for "Call Center". Select and place two more Transfer Icons (these will be used for "Service" and extension 6237).
    i. Double click on the first **Transfer** icon ("**Call Center**")
        1. On the **General** tab → **Token Name** = **Transfer to 401 - Call Center** (not shown).
        2. On the **Specific** tab → **Destination = 401** (not shown).
    ii. Double click on the second **Transfer** icon ("**Support**").
        1. On the **General** tab → **Token Name** = **Transfer to 402 - Support** (not shown).
        2. On the **Specific** tab → **Destination** = **402** (not shown).
    iii. Double Click on the third **Transfer** icon ("**Ext6237**").
        1. On the **General** tab, **Token Name** = **Transfer to 6237** (not shown).
        2. On the **Specific** tab, **Destination = 6237** (not shown).
- From the options bar, select the Connector icon and:
    i. Drag a connecting flow line from the **Start Point** box to the **Menu** box (see screen shot below).

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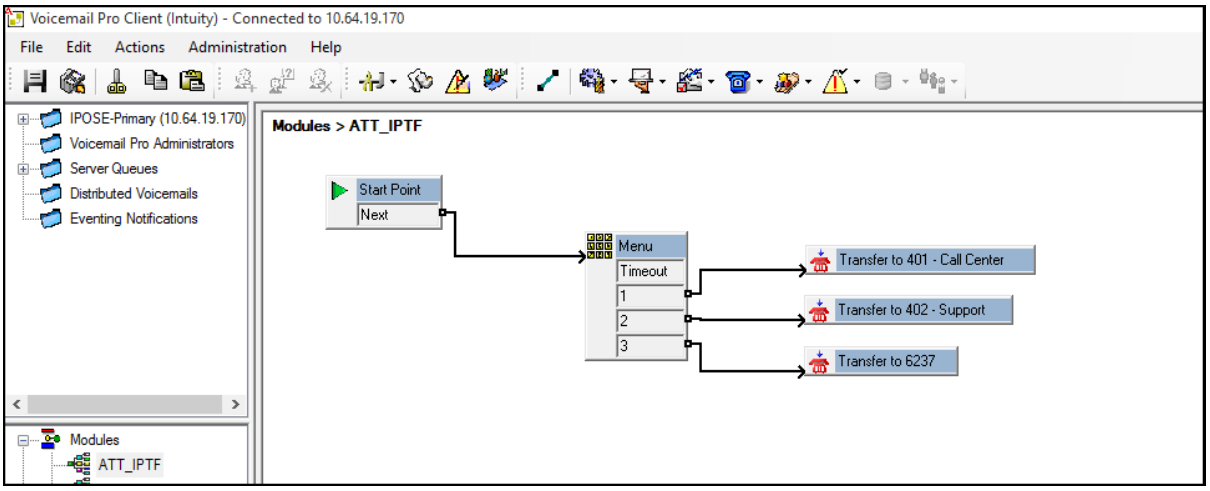
ii. Drag connecting flow lines from each of the **Menu** options to their associated **Transfer** boxes (see screenshot below).



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

When the associated AT&T DNIS number is received (e.g., **000008885551027**), IP Office will send the call to Voicemail Pro. The caller will be prompted to enter 1, 2, or 3 to access Call Center, Support, or user 6237. The associated Avaya IP Office extension (e.g., 401, 402, or 6237) will then ring.

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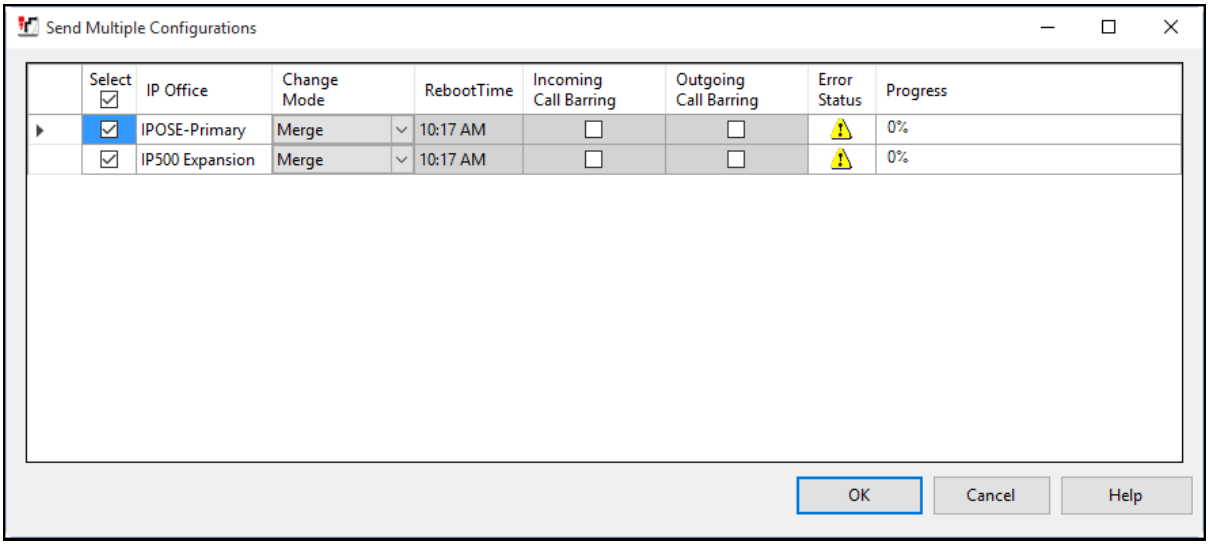
5.9. Saving Configuration Changes to Avaya IP Office

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

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

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

Click **OK** to execute the save.



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

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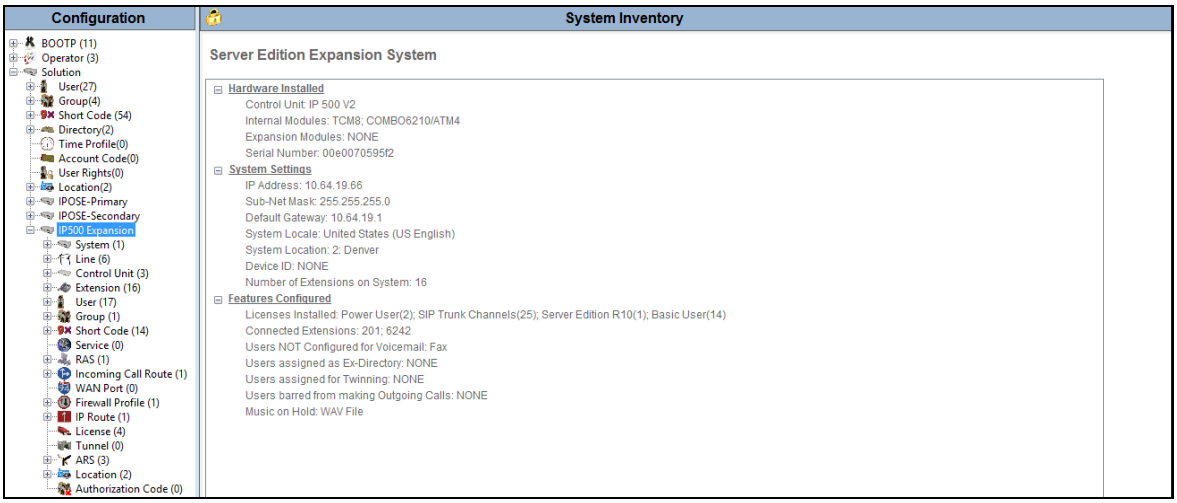
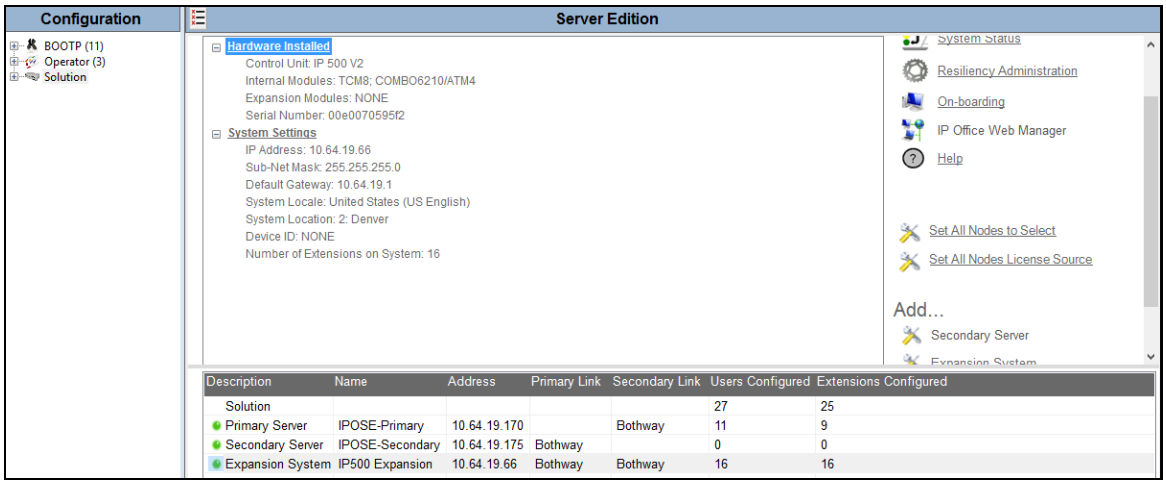
6. Avaya IP Office Expansion Configuration

Navigate to **File → Open Configuration** (not shown), select the proper Avaya IP Office system from the pop-up window, and log in using the appropriate credentials. Clicking the "plus" sign next to **IP500 Expansion** on the left navigation pane will expand the menu on this server.



6.1. Physical Hardware

In the reference configuration, looking at the Expansion System IP500 V2 from left to right, the first module is a **TCM 8** Digital Station Module. This module supports BCM / Norstar T-Series and M-Series telephones. The second module is a **COMBO6210/ATM4** module. This module is used to add a combination of ports to an IP500 V2 control unit and is not supported by IP500 control units. The module supports 10 voice compression channels. Codec support is G.722, G.711, G729A and G.723 with 64ms echo cancellation. The "Combo" card will support 6 Digital Station ports for digital stations in slots 1-6 (except 3800, 4100, 4400, 7400, M and T-Series), 2 Analog Extension ports in slots 7-8, and 4 Analog Trunk ports in slots 9-12.



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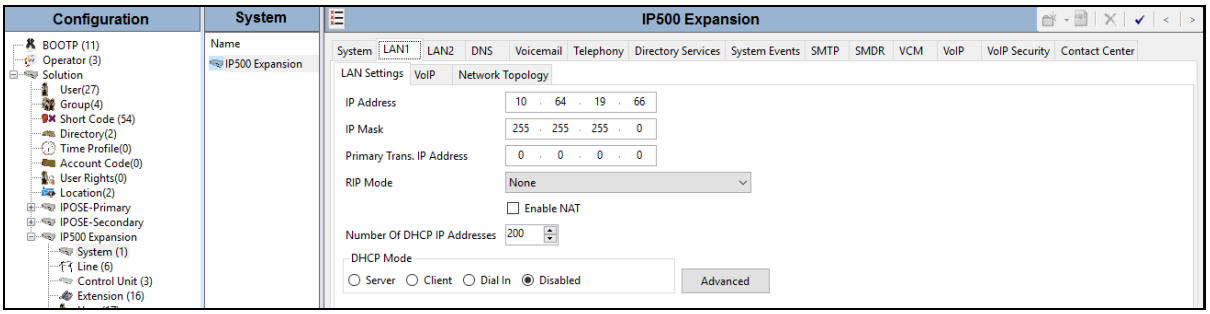
6.2. System Settings

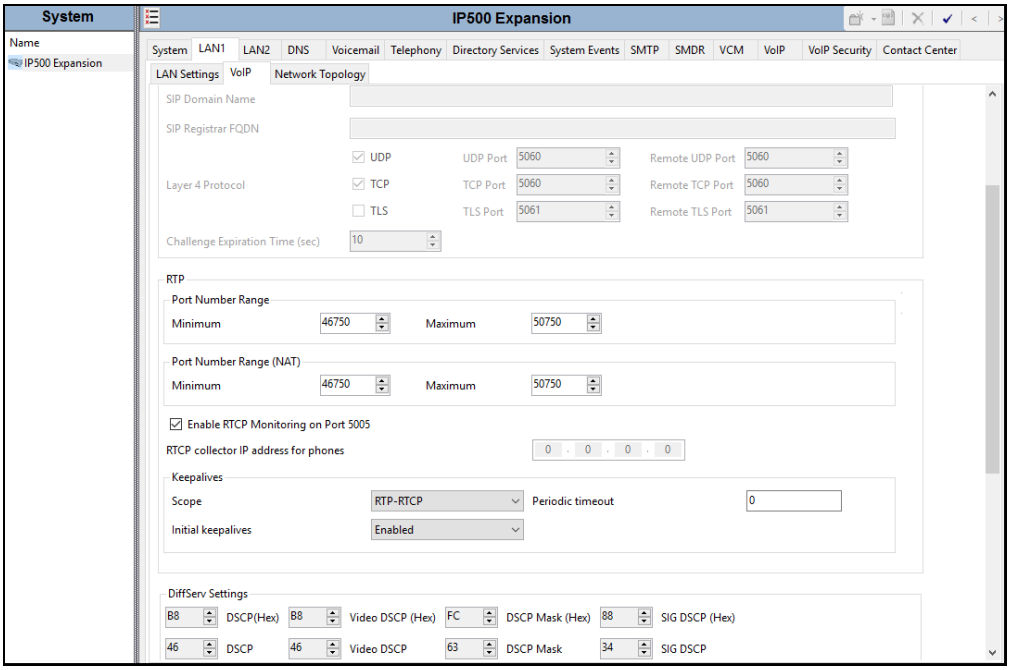
This section illustrates the configuration of system settings. Select **System** in the Navigation pane to configure these settings. The subsection order corresponds to a left to right navigation of the tabs in the Details pane for System settings. For all of the following configuration sections, the **OK** button (not shown) must be selected in order for any changes to be saved.

6.2.1. LAN Settings

In the reference configuration, LAN1 is used to connect the Expansion System to the enterprise network. To view or configure the **IP Address** of LAN1, select the **LAN1** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP Address of the Expansion System is **10.64.19.66**. Other parameters on this screen may be set according to customer requirements.



Select the **VoIP** tab as shown in the following screen. If desired, the **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media paths from Primary server to the Expansion System. The defaults are used here.



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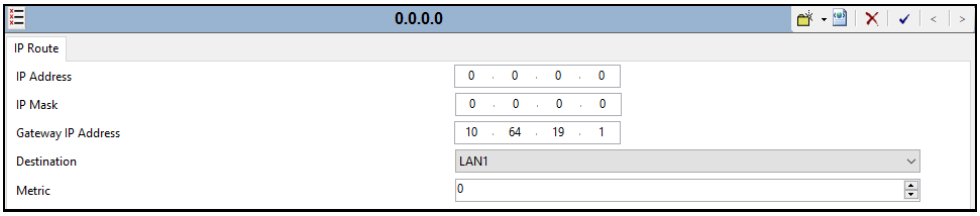
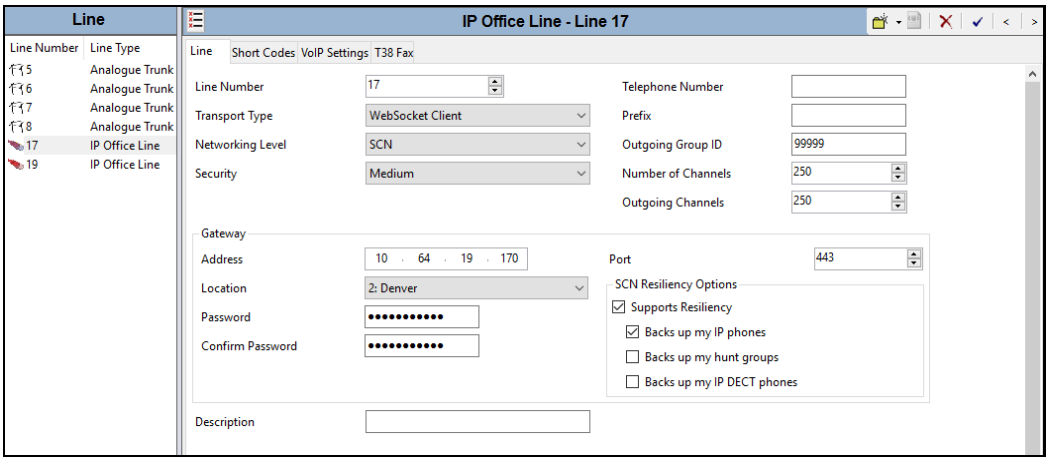
6.3. IP Route

In the reference configuration, the Expansion System LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is **10.64.19.1**.



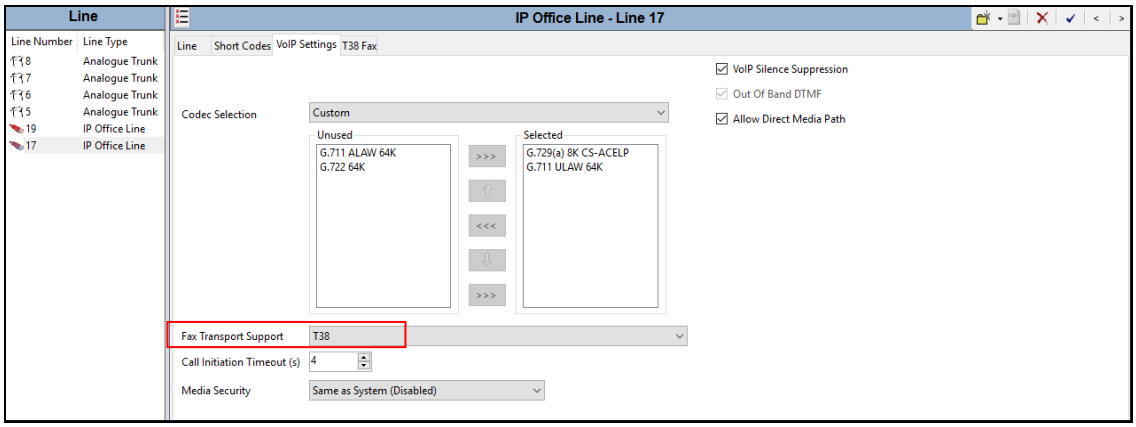
6.4. IP Office Line

The IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. Below is the IP Office Line to the Primary server.



In the reference configuration, a fax machine is connected to one of the analog ports on the Expansion System. To accommodate T.38 fax, select the **VoIP Settings** tab and configure the following:

- **Fax Transport Support: T38**



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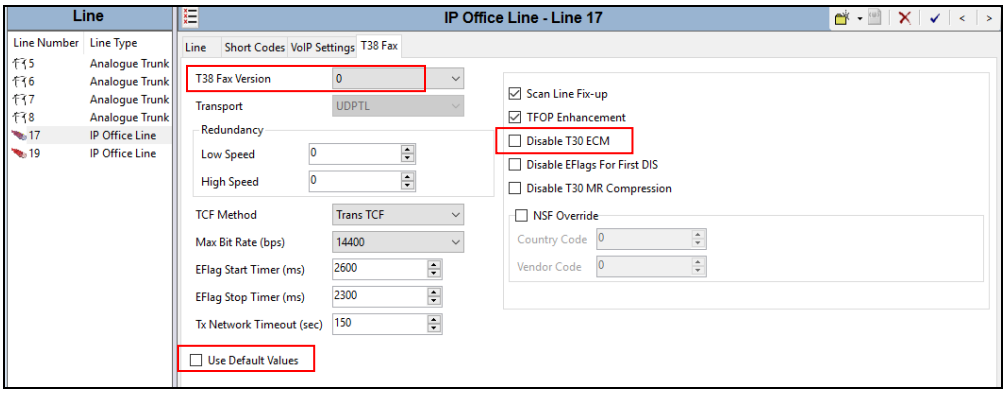
Select the **T38 Fax** tab and enter the following:

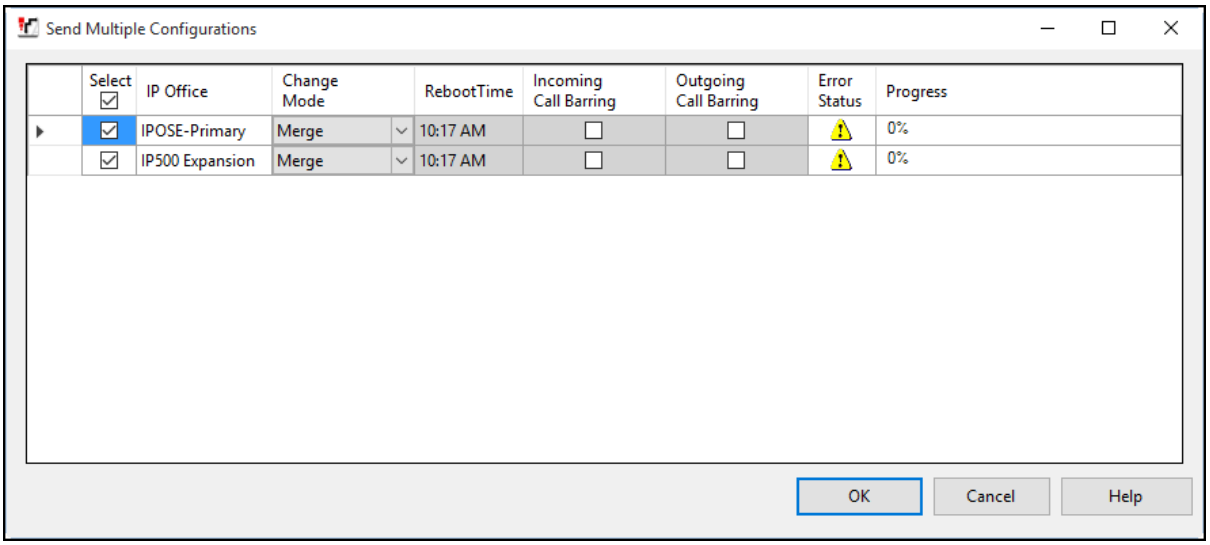
- Unselect the **Use Default Values** option.
- Set the **T38 Fax Version** option to **0** (zero). This matches the version AT&T uses.
- Verify that **Disable T30 ECM** is *not* checked,

Default values are used for the remaining fields. Select **Ok** (not shown).



6.5. Saving Configuration Changes to Avaya IP Office

Similar to **Section 5.9**, navigate to **File → Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.



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

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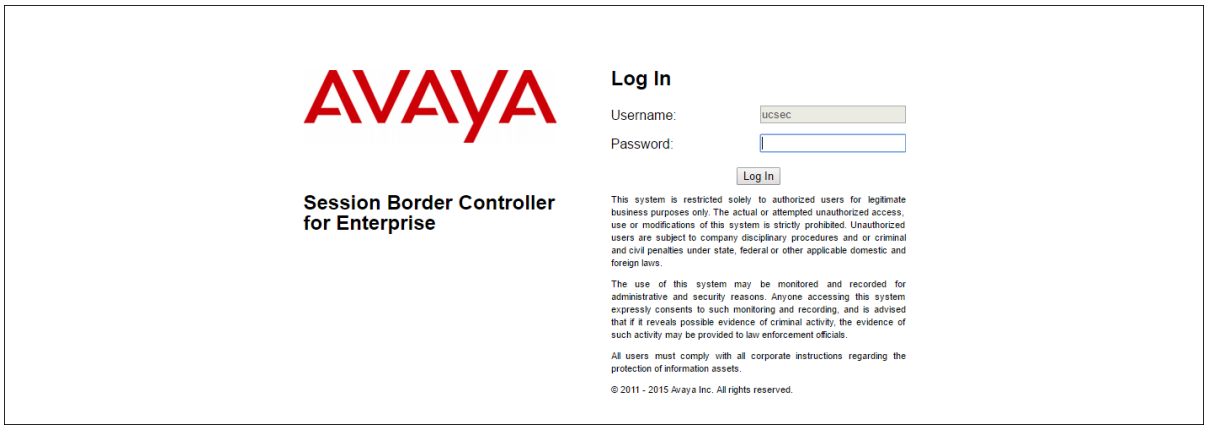
7. Configure Avaya Session Border Controller for Enterprise

In the reference configuration, Avaya SBCE is used as an edge device between the CPE and AT&T.

This section covers the configuration of the Avaya SBCE. It is assumed that the initial provisioning of the Avaya SBCE, including the assignment of the management interface IP Address and license installation have already been completed; hence these tasks are not covered in these Application Notes. For more information on the installation and provisioning of the Avaya SBCE consult references **[5]** and **[6]**.

Use a web browser to access the Element Management Server (EMS) web interface, and enter https://*ipaddress*/sbc in the address field of the web browser, where *ipaddress* is the management LAN IP address of the Avaya SBCE.

Log in with the appropriate credentials. Click **Log In**.



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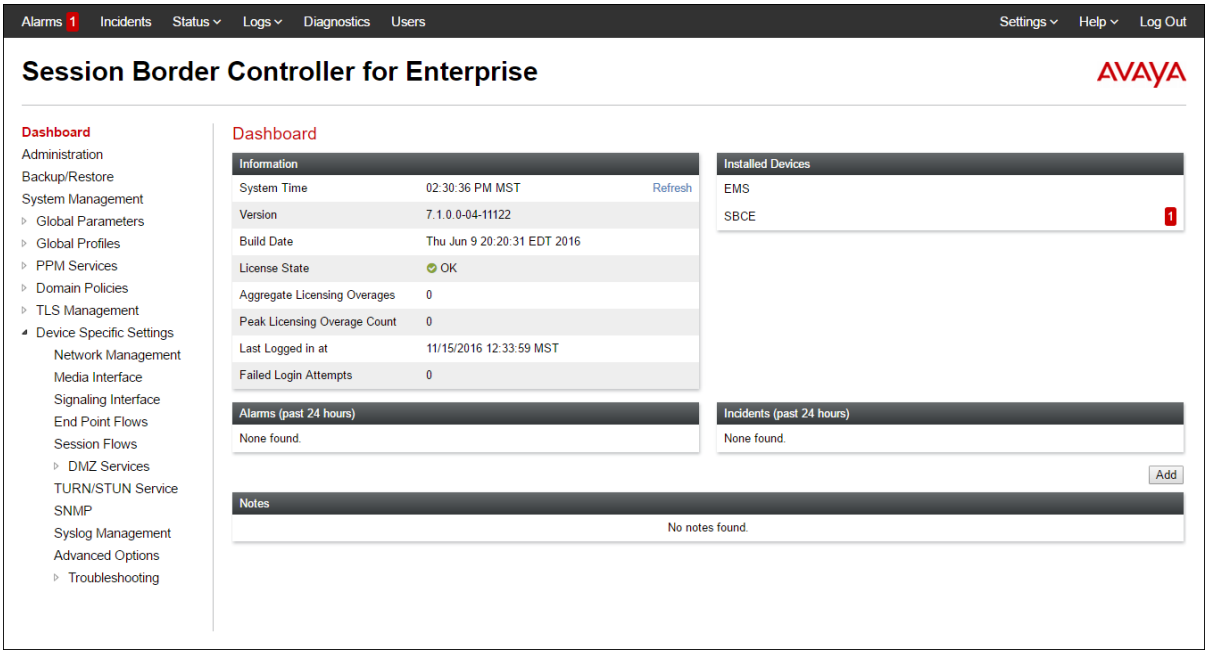
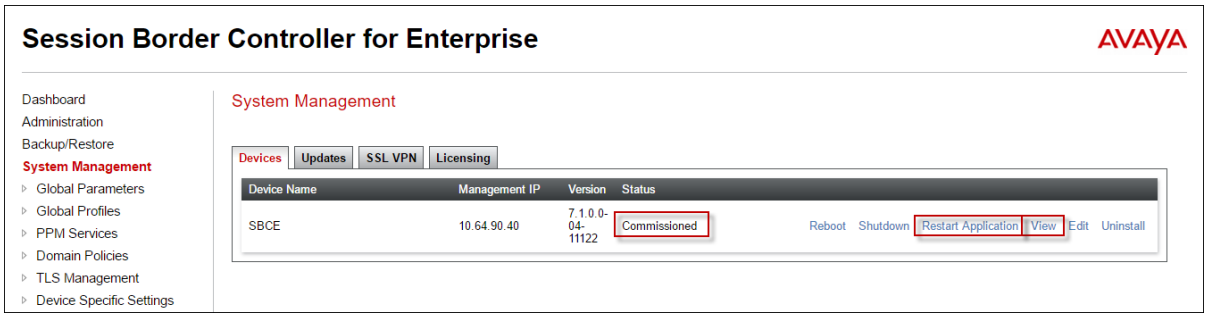
The main page of the Avaya SBCE will appear. Note that the installed software version is displayed. Verify that the **License State** is **OK**. The SBCE will only operate for a short time without a valid license. Contact your Avaya representative to obtain a license.



7.1. System Management – Status

Select **System Management** and verify that the **Status** column says **Commissioned**. If not, contact your Avaya representative. To view system information that was configured during installation, navigate to **System Management**. A list of installed devices is shown in the right pane. In the case of the reference configuration, a single device named **SBCE** is shown. To view the configuration of this device, click **View** as highlighted below.

> **Note** – Certain Avaya SBCE configuration changes require that the underlying application be restarted. To do so, click on **Restart Application** shown below.



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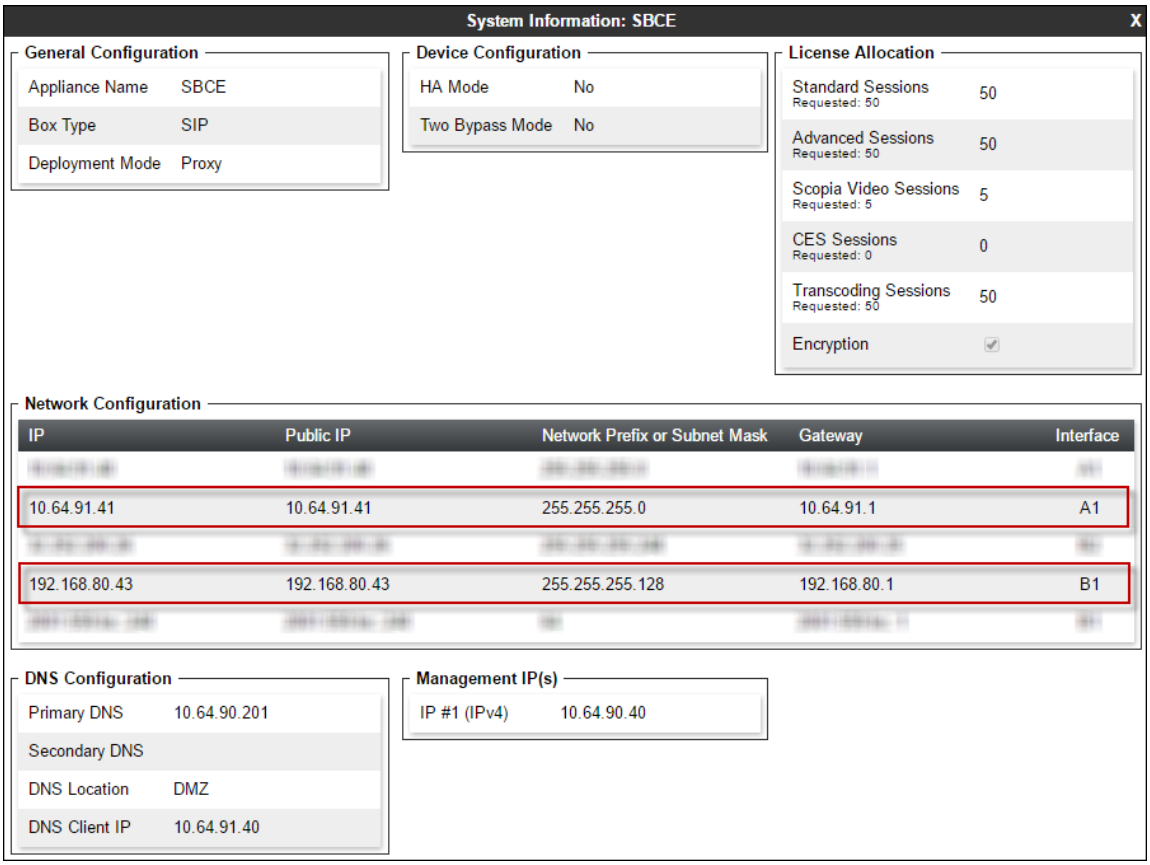
The **System Information** screen shows the **Device Configuration**, **License Allocation**, **Network Configuration**, **DNS Configuration** and **Management IP(s)** information provided during installation and corresponds to **Figure 1**. In the shared test environment, the highlighted **A1** and **B1** IP addresses are the ones relevant to the configuration of the SIP trunk to AT&T.



7.2. Network Management

The Network Management screen is where the network interface settings are configured and enabled. During the installation process of Avaya SBCE, certain network-specific information is defined such as device IP address(es), public IP address(es), netmask, gateway, etc., to interface the device to the network. It is this information that populates the various Network Management tab displays, which can be edited as needed to optimize device performance and network efficiency. Navigate to **Device Specific Settings → Network Management** and verify the IP addresses assigned to the interfaces and that the interfaces are enabled. The following screen shows the enterprise interface is assigned to **A1** and the interface towards AT&T is assigned to **B1**.

The following Avaya SBCE IP addresses and associated interfaces were used in the reference configuration:

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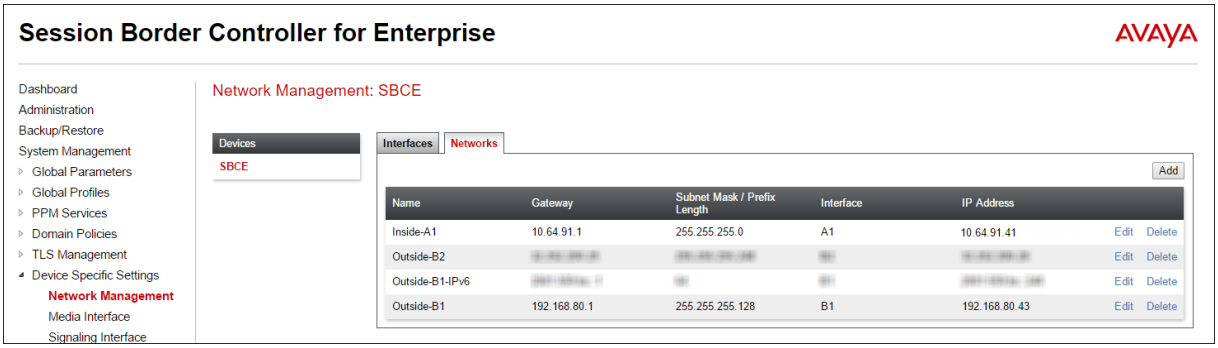
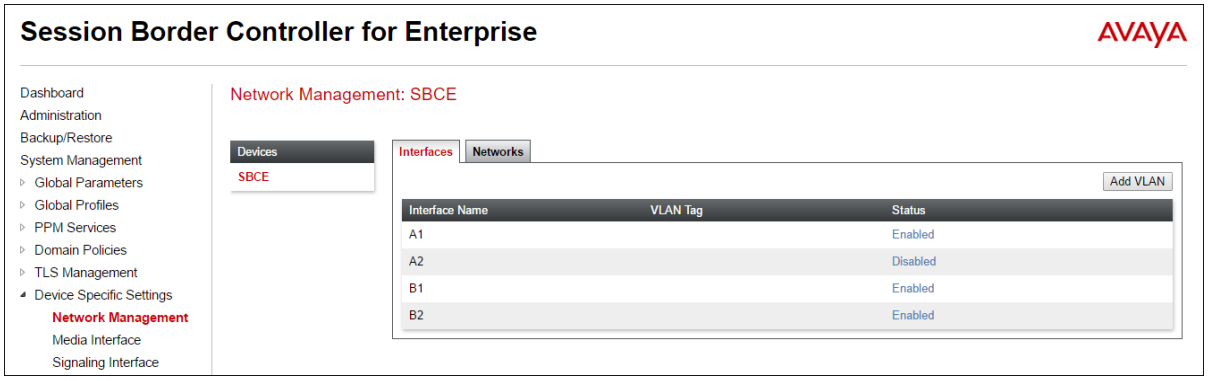
- **B1: 192.168.80.43** – IP address configured for the AT&T IPTF service. This address is known to AT&T. See **Section 3**.
- **A1**: **10.64.91.41** – IP address configured for AT&T IPTF service to IP Office.



The following screen shows interface **A1**, and **B1** are **Enabled**. To enable an interface click the corresponding **Disabled** Status link to change it to **Enabled**.



7.3. Server Interworking Profile

The Server Internetworking profile includes parameters to make the Avaya SBCE function in an enterprise VoIP network using different implementations of the SIP protocol. There are default profiles available that may be used as is, or modified, or new profiles can be configured as described below.

In the reference configuration, separate Server Interworking Profiles were created for IP Office and AT&T IPTF service.

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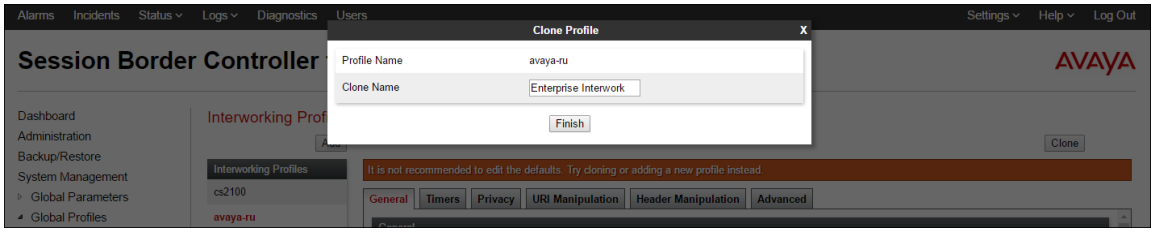
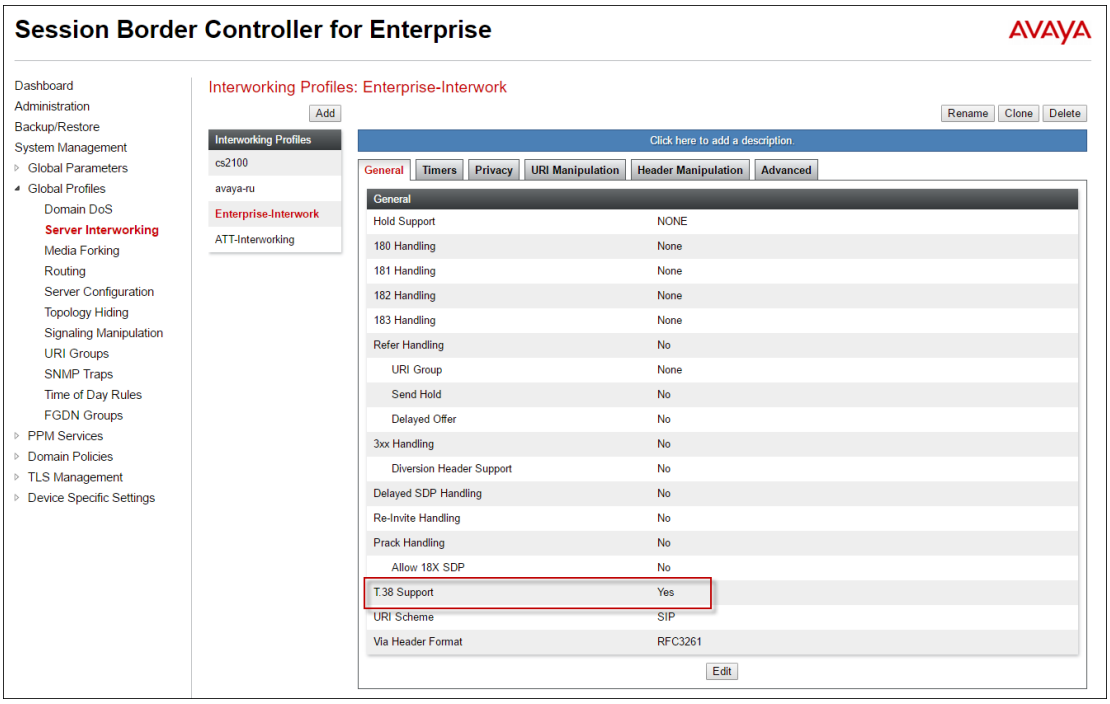
7.3.1. Server Interworking Profile – IP Office

In the reference configuration, the IP Office Server Interworking profile was cloned from the default **avaya-ru** profile. To clone a Server Interworking Profile for IP Office, navigate to **Global Profiles → Server Interworking**, select the **avayu-ru** profile and click the **Clone** button. Enter a **Clone Name** and click **Finish** to continue.



The following screen shows the **Enterprise-Interwork** profile used in the reference configuration, with **T.38 Support** set to **Yes**. To modify the profile, scroll down to the bottom of the screen and click **Edit**. Select the **T.38 Support** parameter and then click **Next** and then **Finish** (not shown). Default values can be used for all other fields.



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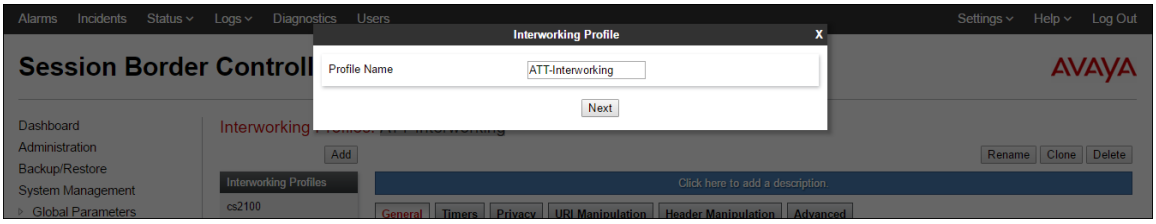
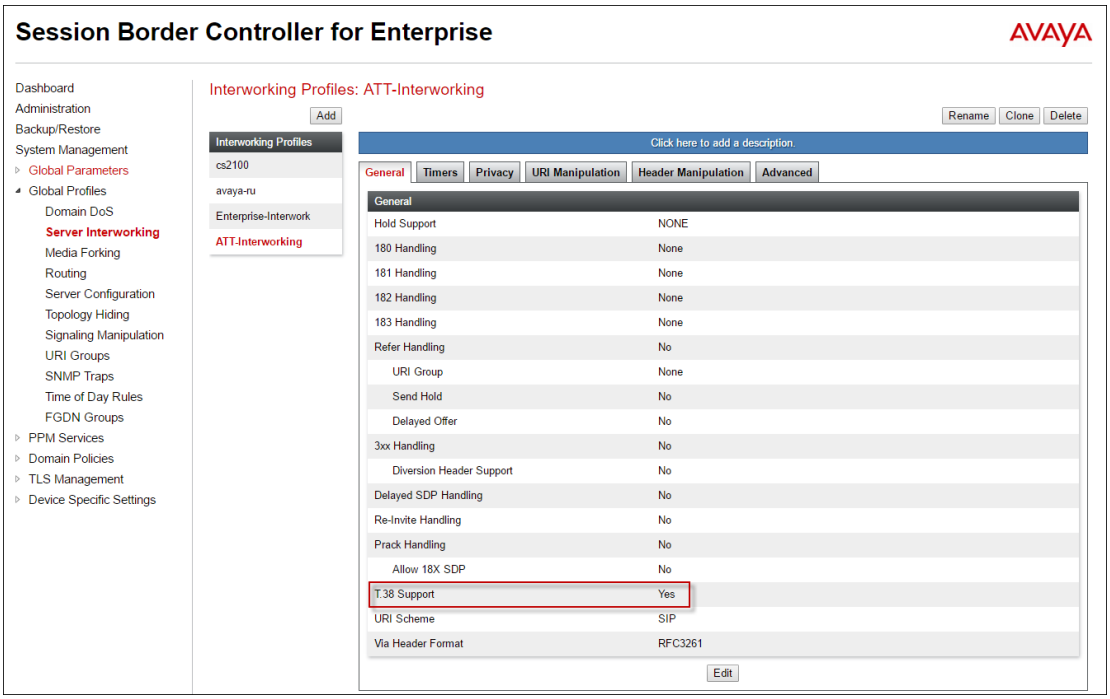
7.3.2. Server Interworking Profile – AT&T

To create a new Server Interworking Profile for AT&T, navigate to **Global Profiles → Server Interworking** and click **Add** as shown below. Enter a **Profile Name** and click **Next**.



The following screens show the **ATT-Interworking** profile used in the reference configuration. On the **General** tab, default values are used with the exception of **T.38 Support** set to **Yes**.



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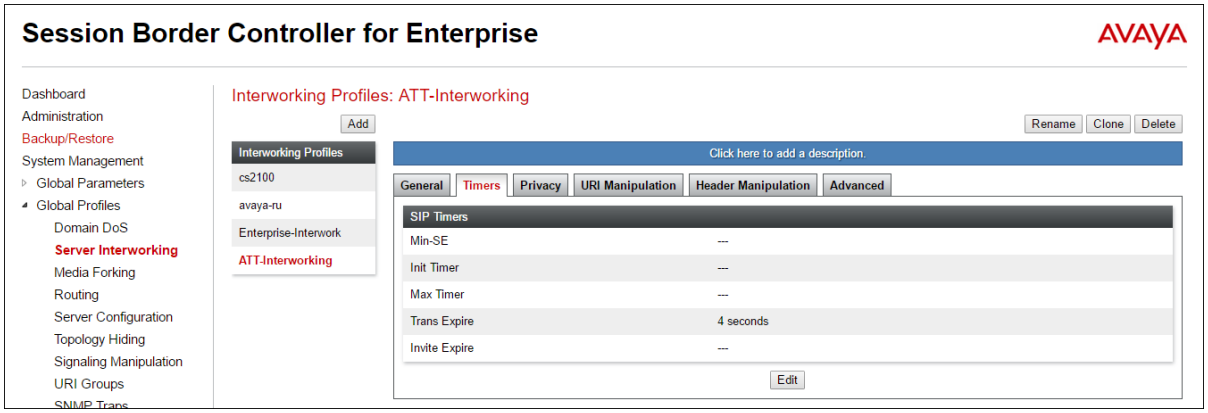
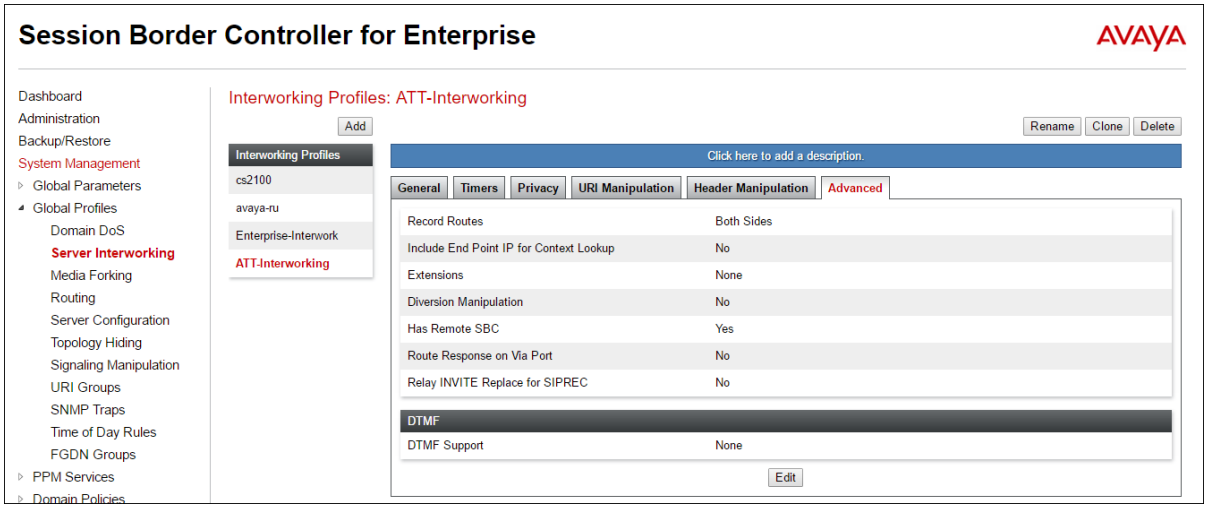
The **Timers** tab shows the values used for compliance testing for the **Trans Expire** field. The **Trans Expire** timer sets the allotted time the Avaya SBCE will try the first primary server before trying the secondary server, if one exists.



Click **Next** to accept default parameters for the **Privacy**, **URI Manipulation**, and **Header Manipulation** tabs (not shown) and advance to the **Advanced** area. **Record Routes** is set to **Both Sides**. Default values can be used for all other fields.



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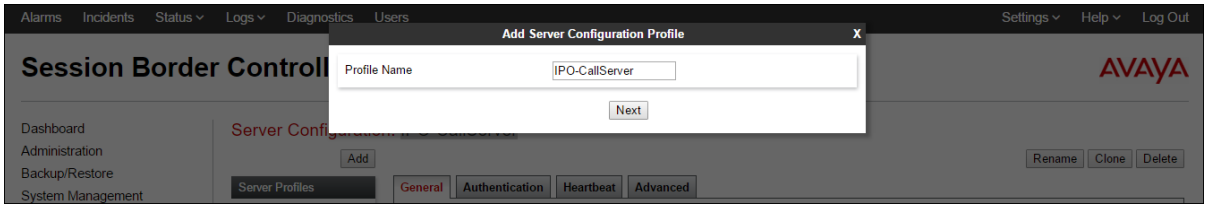
7.4. Server Configuration

The **Server Configuration** contains parameters to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters, DoS security statistics, and trusted domains.

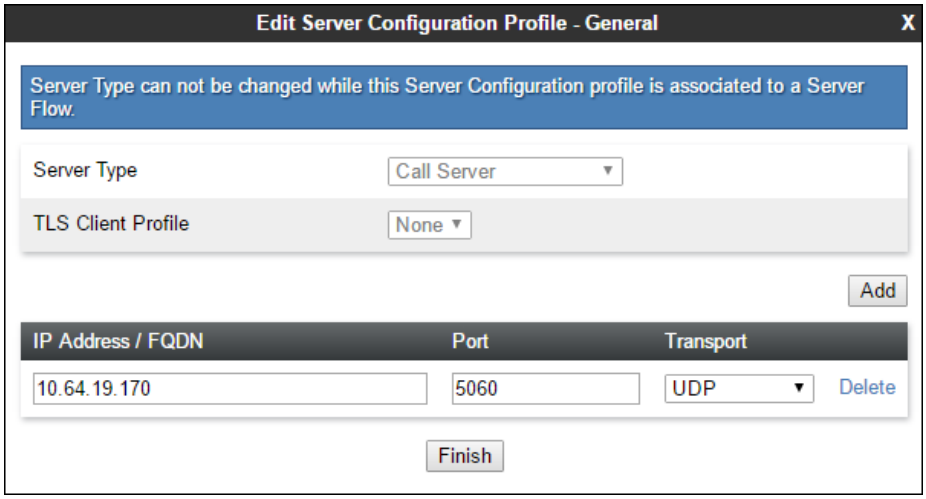
In the reference configuration, separate Server Configurations were created for IP Office and AT&T IPTF service.

7.4.1. Server Configuration – IP Office

To add a Server Configuration Profile for IP Office, navigate to **Global Profiles → Server Configuration** and click **Add**. Enter a descriptive name for the new profile and click **Next**.



The following screens illustrate the Server Configuration for the Profile name **IPO-CallServer**. In the **General** parameters, the **Server Type** is set to **Call Server**. In the **IP Address / FQDN** field, the IP Address of the Primary server LAN1 interface in the reference configuration is entered. This IP address is **10.64.19.170**. Under **Port**, **5060** is entered, and the **Transport** parameter is set to **UDP**. If adding the profile, click **Next** (not shown) to proceed. If editing an existing profile, click **Finish**.



Default values can be used on the **Authentication** tab, click **Next** (not shown) to proceed to the **Heartbeat** tab. The Avaya SBCE can be configured to source "heartbeats" in the form of PINGs or SIP OPTIONS towards IP Office. When remote workers are configured, IP Office

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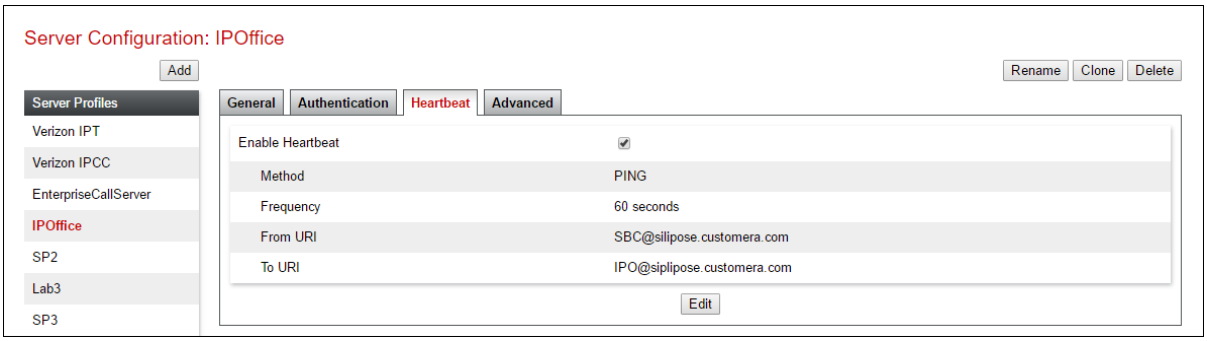
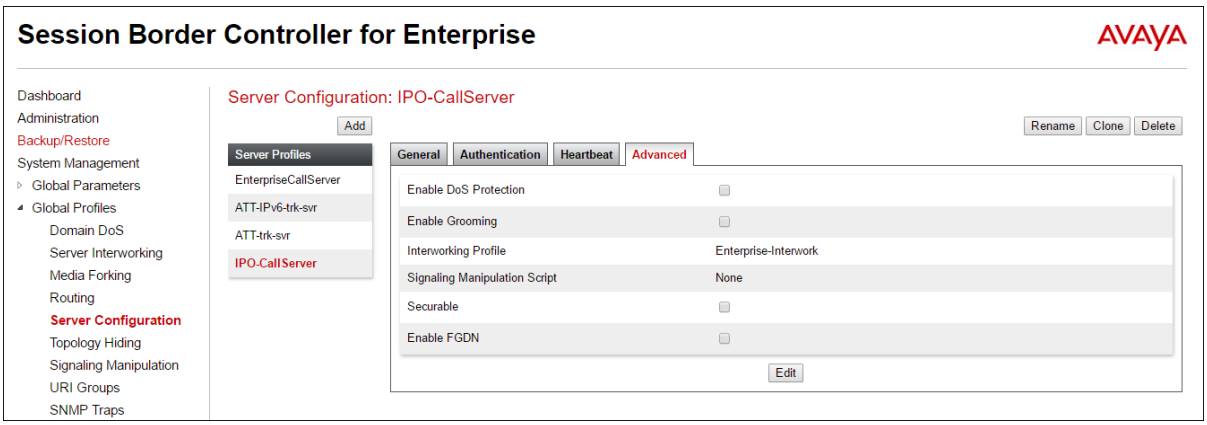
may not respond to SIP OPTIONS from the SBCE IP address designated for remote workers; therefore PING will be used instead.

Select **PING** from the **Method** drop-down menu. Select the desired frequency that the SBCE will source PINGs towards IP Office.

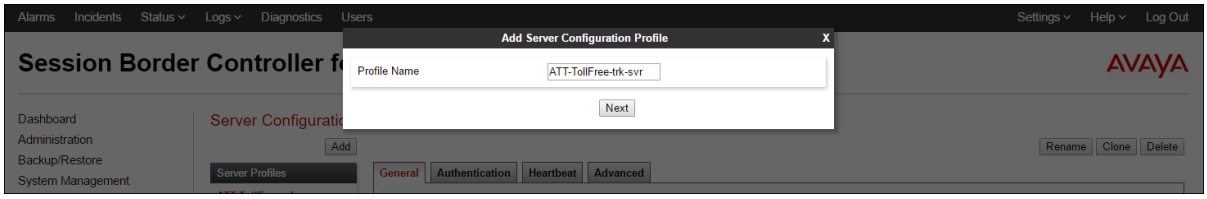


On the **Advanced** tab, the **Interworking Profile** is set to **Enterprise-Interwork** created in **Section 7.3.1** for IP Office.



7.4.2. Server Configuration – AT&T

To add a Server Configuration Profile for AT&T, navigate to **Global Profiles → Server Configuration** and click **Add**. Enter a descriptive name for the new profile and click **Next**.



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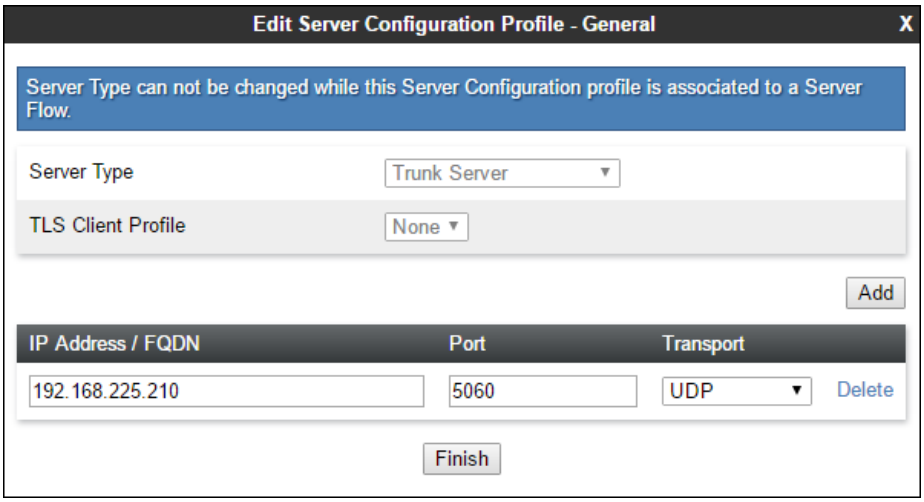
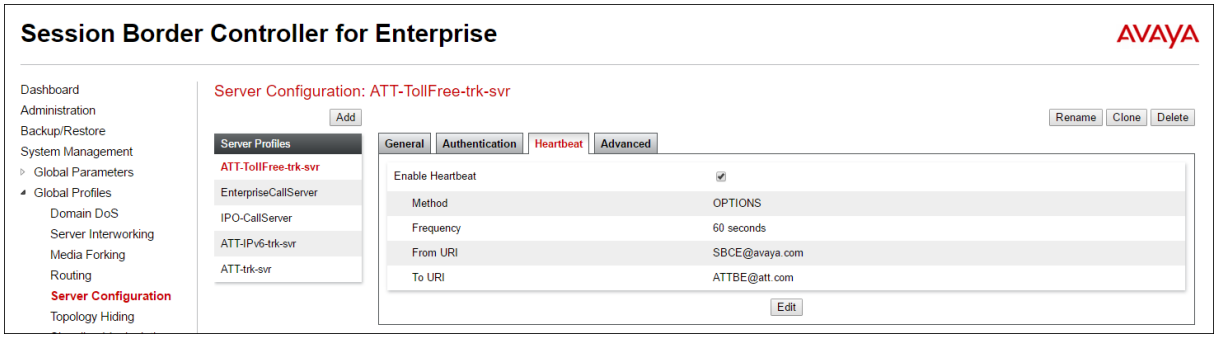
The following screens illustrate the Server Configuration for the Profile name **ATT-TollFree-trk-svr**. In the **General** parameters, the **Server Type** is set to **Trunk Server**. In the **IP Address / FQDN** field, the AT&T-provided IP address is entered. This is **192.168.38.69**. Under **Port**, **5060** is entered, and the **Transport** parameter is set to **UDP**. If adding the profile, click **Next** (not shown) to proceed. If editing an existing profile, click **Finish**.



Default values can be used on the **Authentication** tab, click **Next** (not shown) to proceed to the **Heartbeats** tab. The Avaya SBCE can be configured to source "heartbeats" in the form of SIP OPTIONS towards AT&T. This configuration is optional. Independent of whether the Avaya SBCE is configured to source SIP OPTIONS towards AT&T, AT&T will receive OPTIONS from the IP Office site as a result of the **Check OOS** parameter being enabled on IP Office (see **Section 5.4.3**). When IP Office sends SIP OPTIONS to the inside private IP Address of the Avaya SBCE, the Avaya SBCE will send SIP OPTIONS to AT&T. When AT&T responds, the Avaya SBCE will pass the response to IP Office.

Select **OPTIONS** from the **Method** drop-down menu. Select the desired frequency that the SBCE will source OPTIONS. The **From URI** and **To URI** may be filled in to configure easily identifiable URIs to appear in SIP OPTIONS sourced by the Avaya SBCE. If adding a new profile, click **Next** to continuing to the **Advanced** settings.



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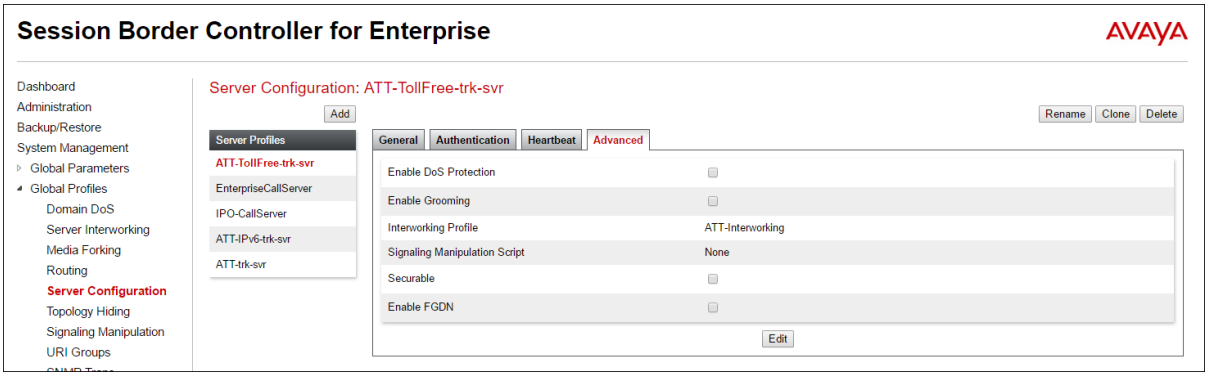
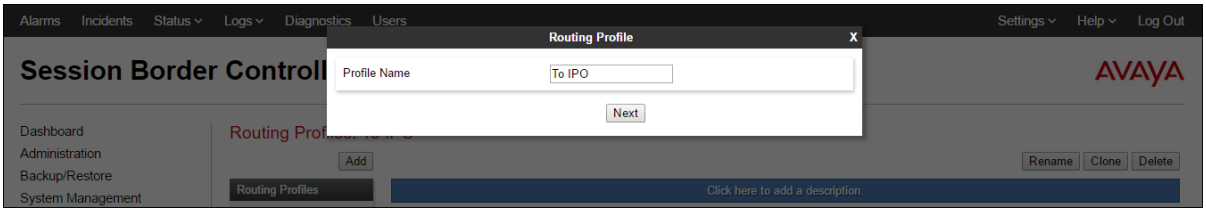
On the **Advanced** tab, **Enable Grooming** is not used for UDP connections and is left unchecked. The **Interworking Profile** is set to **ATT-Interworking** created in **Section 7.3.2** for AT&T.



7.5. Routing Profile

Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport settings, name server addresses and resolution methods, next hop routing information, and packet transport types.

Create a Routing Profile for IP Office and AT&T IPTF service. To add a routing profile, navigate to **Global Profiles → Routing** and select **Add**. Enter a **Profile Name** and click **Next** to continue.



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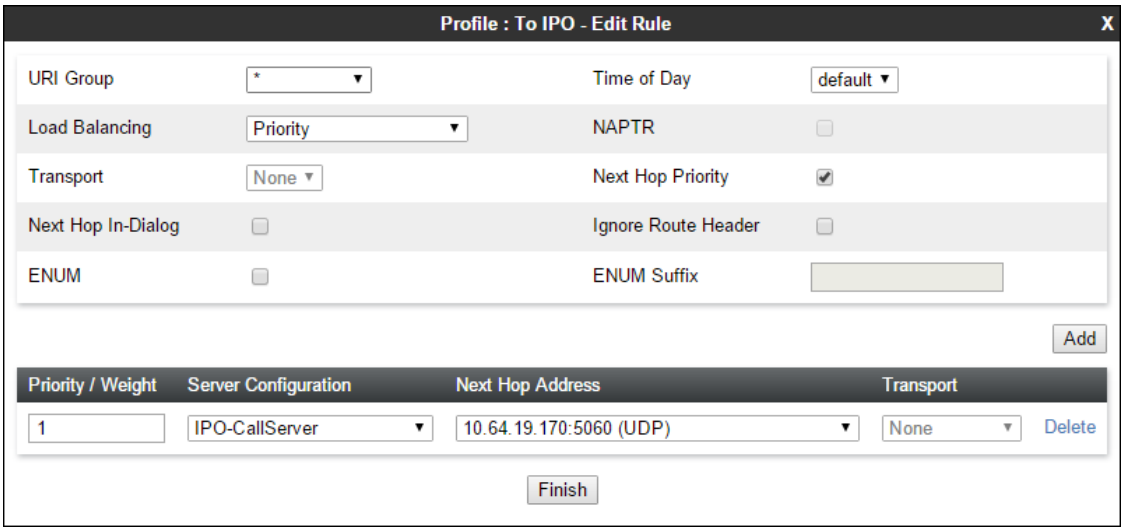
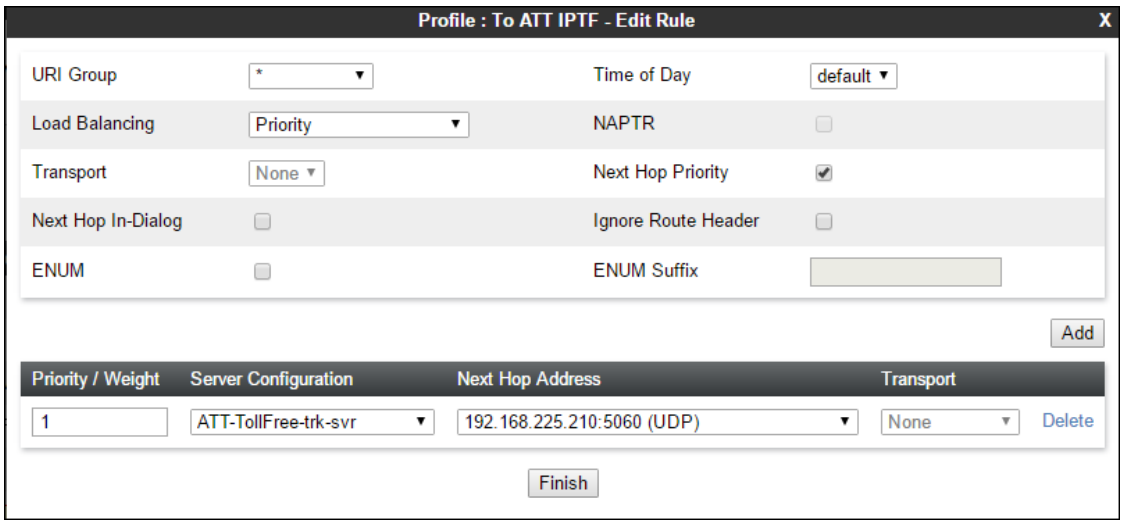
The following screen shows the Routing Profile **To IPO** created in the reference configuration. The parameters in the top portion of the profile are left at their default settings. The **Priority / Weight** parameter is set to **1**, and the IP Office **Server Configuration**, created in **Section 7.4.1**, is selected from the drop-down menu. The **Next Hop Address** is automatically selected with one of the values from the IP Office Server Configuration, and **Transport** becomes greyed out. Select the **UDP** entry from the drop-down menu for the **Next Hop Address**, and select **Finish**.



Similarly add a Routing Profile to AT&T. The following screen shows the Routing Profile **To ATT IPTF** created in the reference configuration. The parameters in the top portion of the profile are left at their default settings. The **Priority / Weight** parameter is set to **1**, and the AT&T **Server Configuration**, created in **Section 7.4.2**, is selected from the drop-down menu. The **Next Hop Address** is automatically selected with the values from the Server Configuration, and **Transport** becomes greyed out. Click **Finish**.



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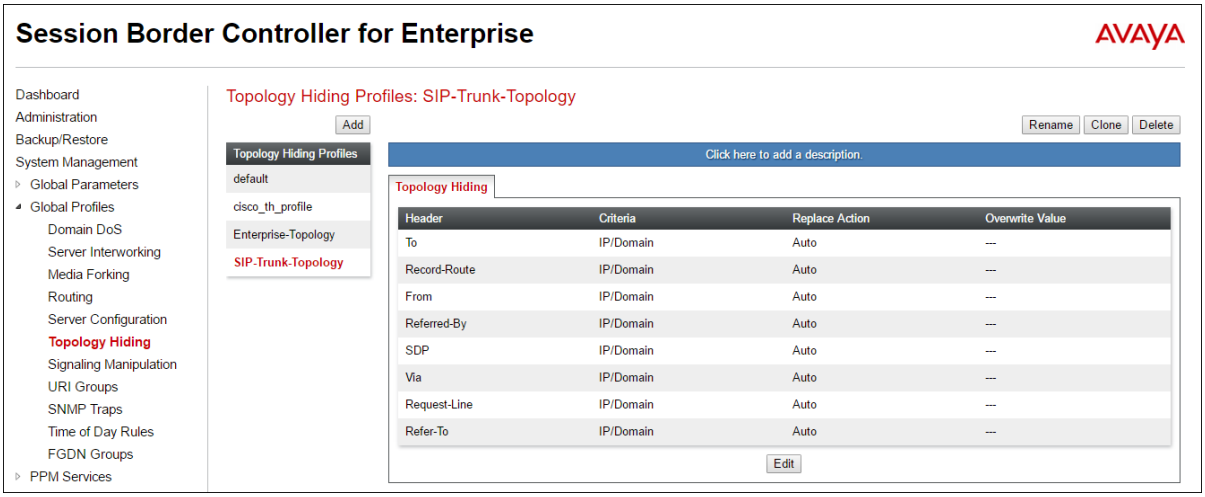
7.6. Topology Hiding Profile

The Topology Hiding profile manages 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.

Click the **Add** button to add a new profile, or select an existing topology hiding profile to edit. In the reference configuration, the **default** profile was cloned for IP Office, and cloned and modified for AT&T.

In the **Replace Action** column an action of **Auto** will replace the header field with the IP address of the Avaya SBCE interface and the **Overwrite** will use the value in the **Overwrite Value**.

In the example shown, **SIP-Trunk-Topology** was cloned from the **default** profile and will later be applied to the Server Flows in **Section 7.14**.



7.7. Application Rule

Application Rules define which types of SIP-based Unified Communications (UC) applications the Avaya SBCE security device will protect: voice, video, and/or Instant Messaging (IM). In addition, you can determine the maximum number of concurrent voice and video sessions the network will process in order to prevent resource exhaustion.

Select **Domain Policies → Application Rules** from the left-side menu as shown below. Click the **Add** button to add a new profile, or select an existing topology hiding profile to edit. In the reference configuration, the **sip-trunk** profile was created for IP Office and AT&T. In an actual customer installation, set the **Maximum Concurrent Sessions** for the **Audio** application to a value slightly larger than the licensed sessions. For example, if licensed for 150 session set the values to **200**. The **Maximum Session Per Endpoint** should match the **Maximum Concurrent Sessions**.

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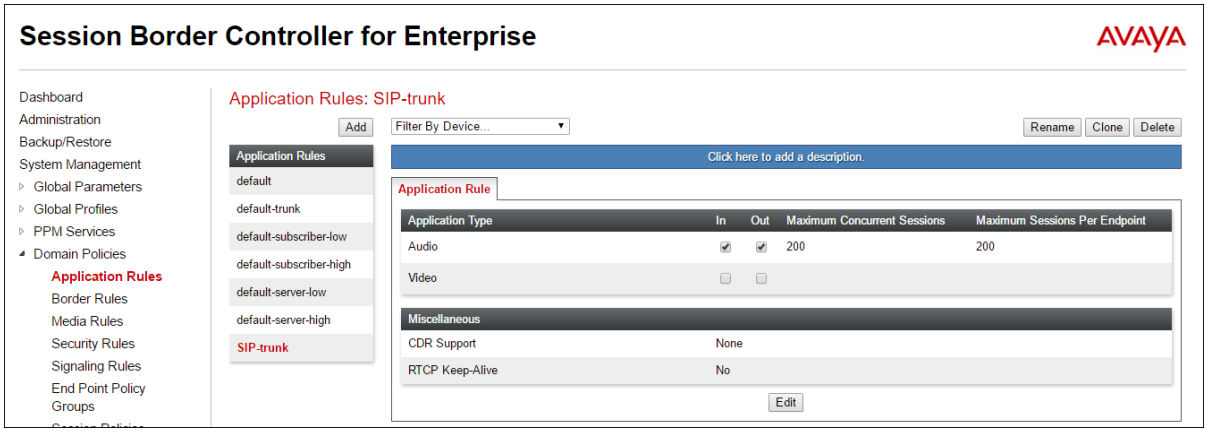
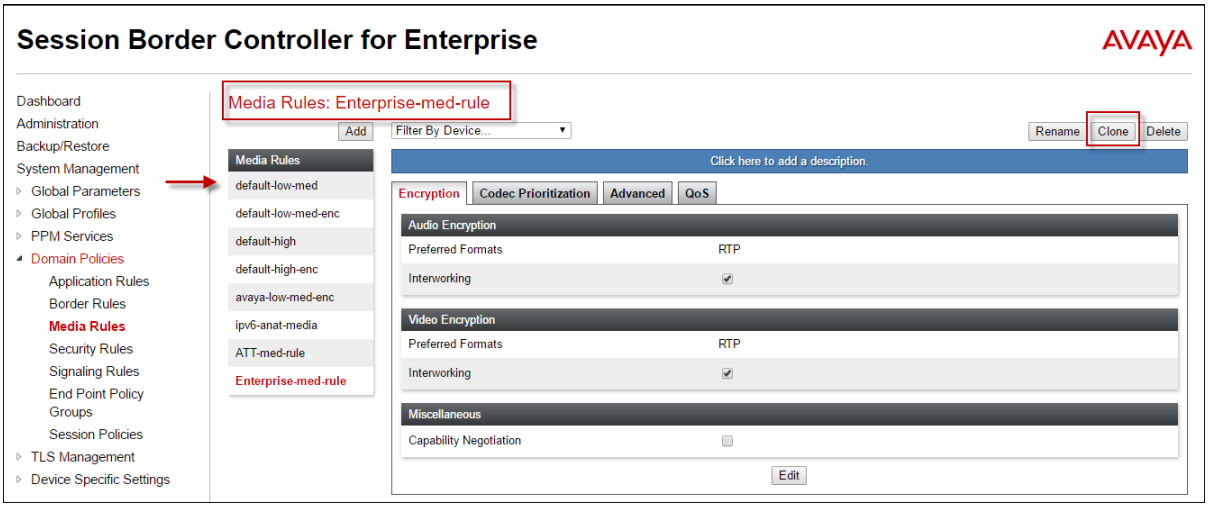


7.8. Media Rule

Media Rules define RTP media packet parameters such as prioritizing encryption techniques and packet encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packets matching these criteria will be handled by the Avaya SBCE security product.

Select **Domain Policies → Media Rules** from the left-side menu as shown below. In the reference configuration, the default media rule **default-low-med** was cloned for IP Office, **Enterprise-med-rule**, and AT&T IPTF, **ATT-med-rule**. With the **default-low-med** rule chosen, click **Clone**. Enter a descriptive name for the new rule and click **Finish** (not shown).

In the reference configuration, media rule **Enterprise-med-rule** was used for IP Office as shown below.



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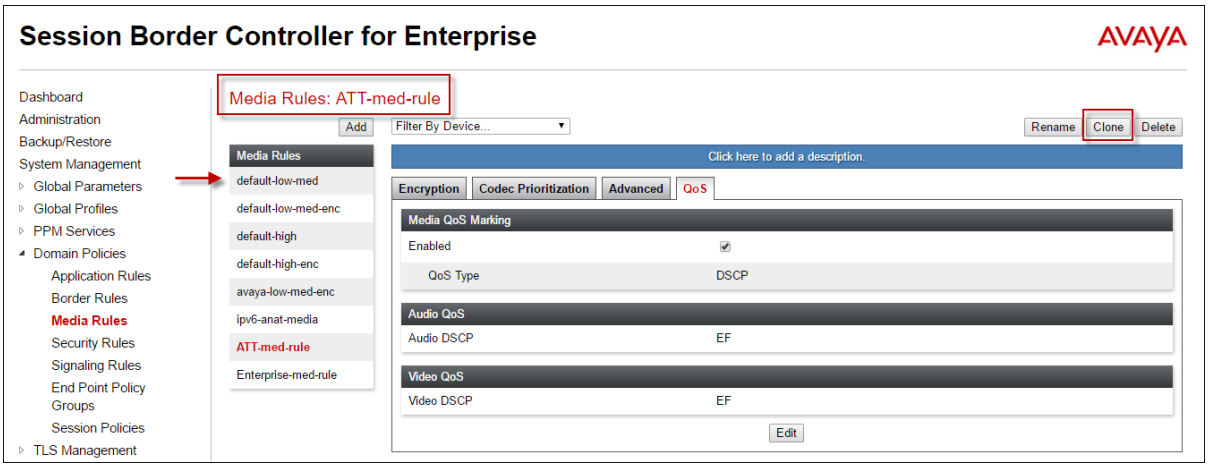
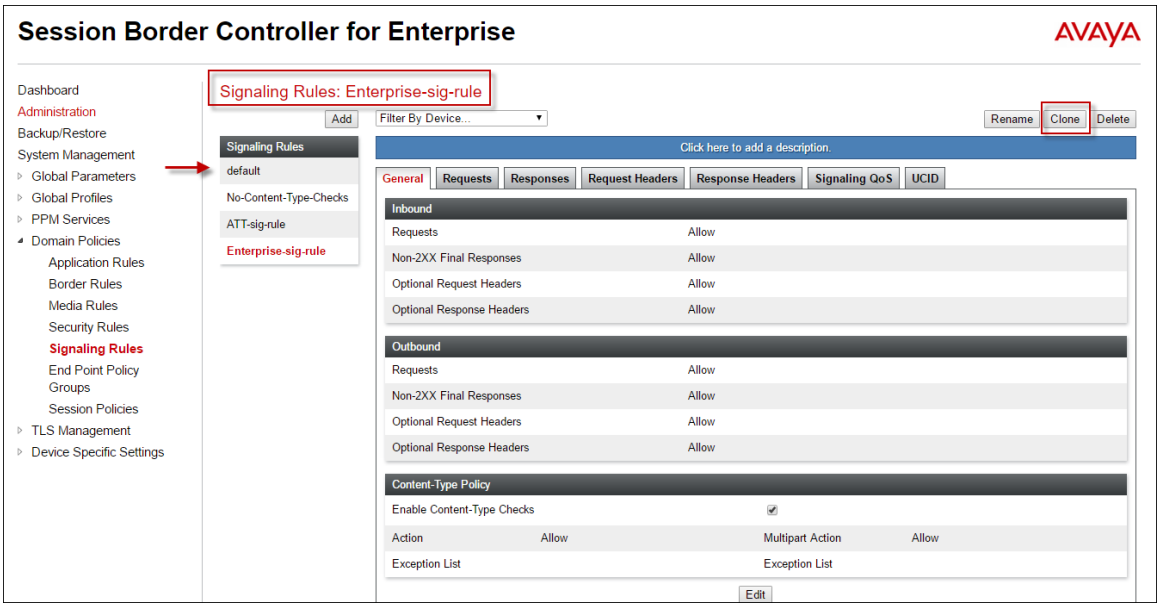
The AT&T Media Rule, **ATT-med-rule** with the DSCP values **EF** for expedited forwarding (default value) for **Media QoS**.



7.9. Signaling Rule

Signaling Rules define the action to be taken (Allow, Block, Block with Response, etc.) for each type of SIP-specific signaling request and response message. When SIP signaling packets are received by Avaya SBCE, they are parsed and pattern-matched against the particular signaling criteria defined by these rules. Packets matching the criteria defined by the Signaling Rules are tagged for further policy matching.

Clone and modify the **default** signaling rule to add the proper quality of service to the SIP signaling. To clone a signaling rule, navigate to **Domain Policies → Signaling Rules**. With the **default** rule chosen, click **Clone**. Enter a descriptive name for the new rule and click **Finish** (not shown). In the reference configuration, signaling rule **Enterprise-sig-rule** is unchanged from the default rule.



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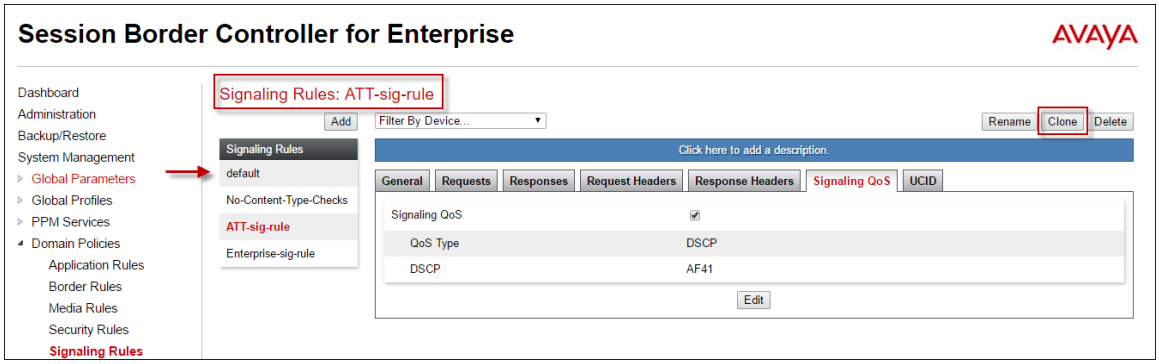
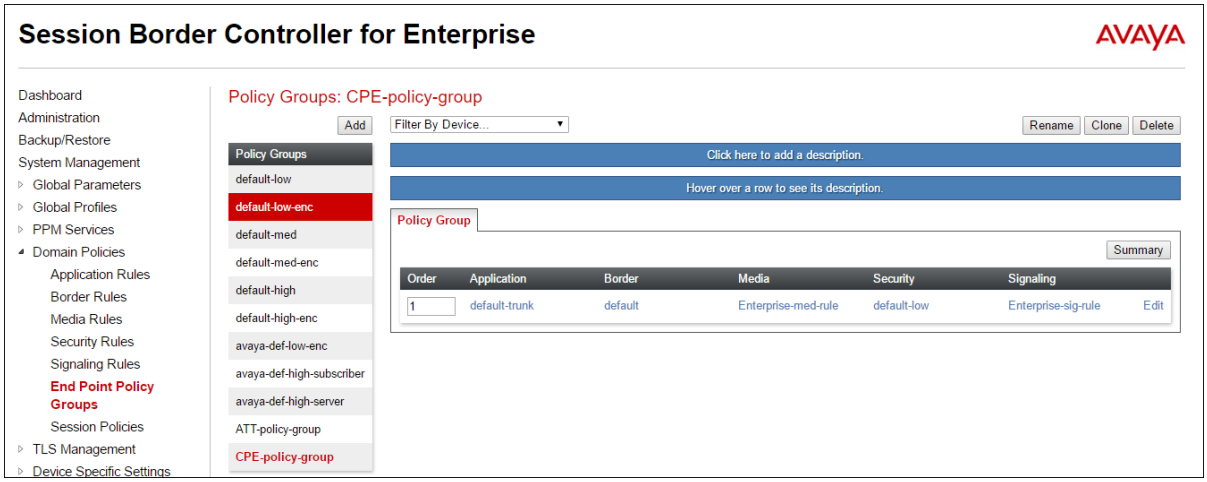
Signaling rule **ATT-sig-rule** was also cloned from the default rule and used for AT&T. The DSCP value **AF41** for assured forwarding (default value) for **Signaling QoS**.



7.10. Endpoint Policy Groups

The rules created within the Domain Policy section are assigned to an Endpoint Policy Group. The Endpoint Policy Group is then applied to a Server Flow in **Section 7.14**.

To create a new policy group, navigate to **Domain Policies → Endpoint Policy Groups** and click on **Add** as shown below. The following screen shows the **CPE-policy-group** created for IP Office. The details of the non-default rules chosen are shown in previous sections.



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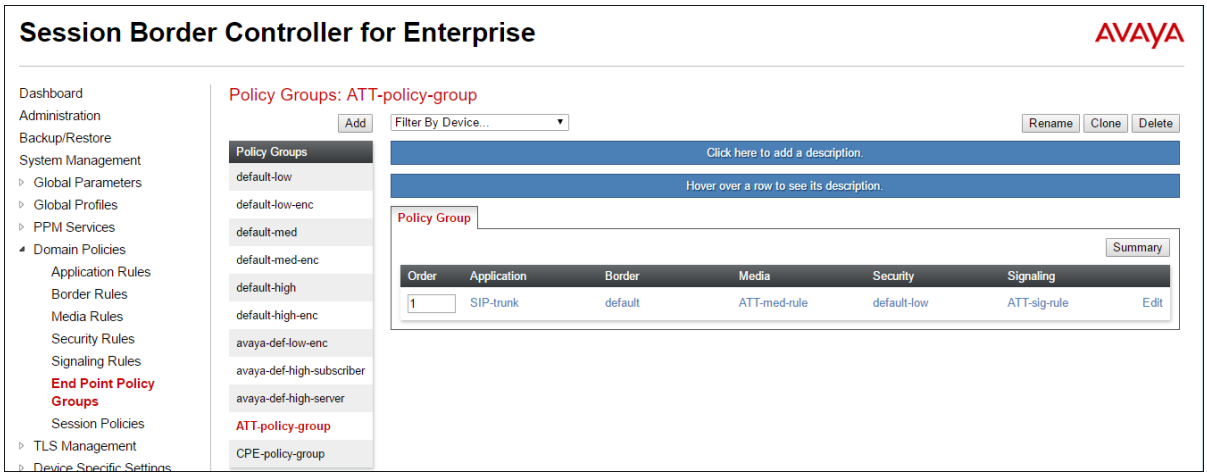
The following screen shows the **ATT-policy-group** created for AT&T. The details of the non-default rules chosen are shown in previous sections.



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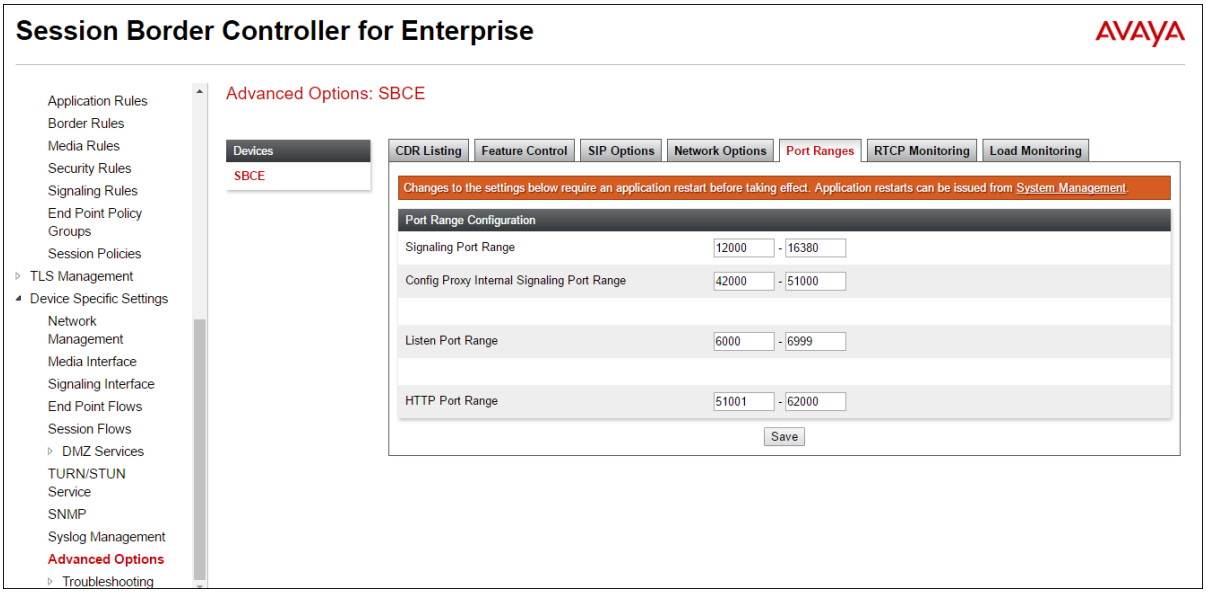
7.11. Advanced Options

In **Section 7.12**, the media UDP port ranges required by AT&T are configured (**16384** – **32767**). However, by default part of this range is already allocated by the Avaya SBCE for internal use (22000 - 31000). The following steps reallocate the port ranges used by the Avaya SBCE so the range required by AT&T can be defined in **Section 7.12**.

1. Select **Device Specific Settings → Advanced Options** from the menu on the left-hand side.
2. Select the **Port Ranges** tab.
3. In the **Signaling Port Range** row, change the range to **12000** – **16380**
4. In the **Config Proxy Internal Signaling Port Range** row, change the range to **42000** – **51000**.
5. In the **Listen Port Range** row, change the range to **6000** – **6999**.
6. In the **HTTP Port Range** row, change the range to **51001** – **62000**.
7. Select **Save**. Note that changes to these values require an application restart (see **Section 7.1**).



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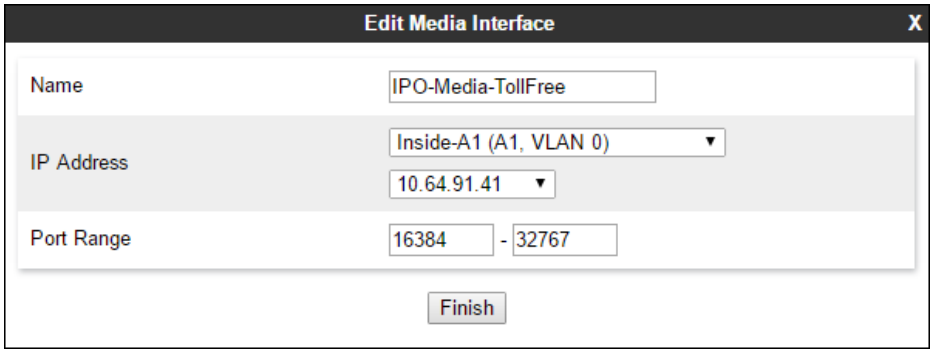
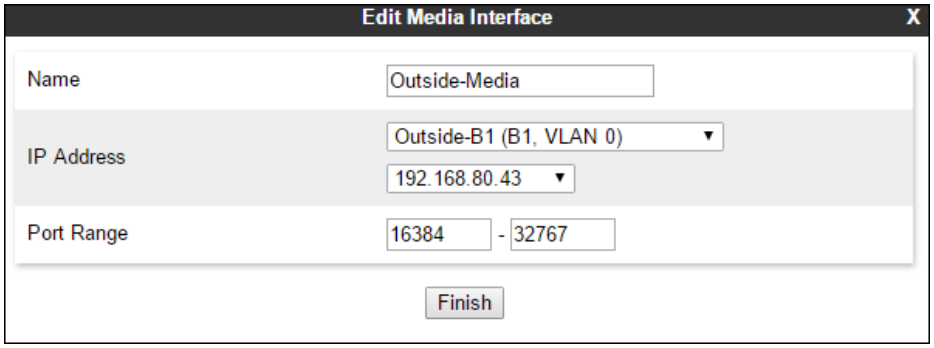
7.12. Media Interface

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

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



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



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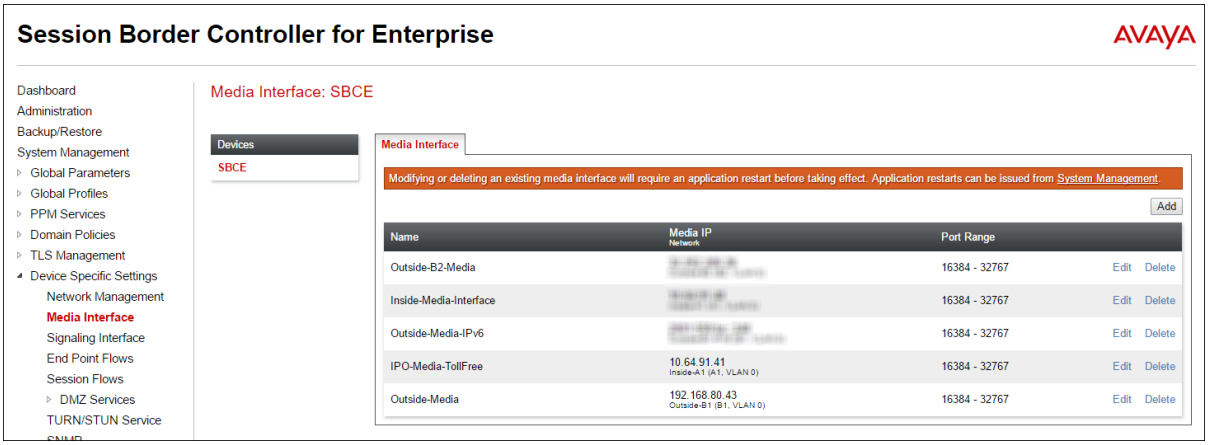
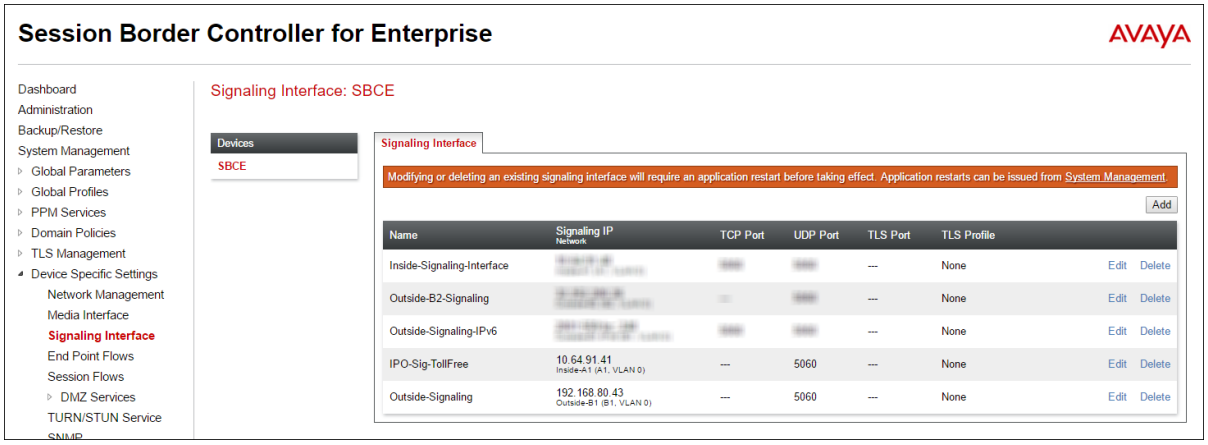
The completed **Media Interface** screen is shown below.



7.13. Signaling Interface

The Signaling Interface screen is where the SIP signaling ports are defined. Avaya SBCE will listen for SIP requests on the defined ports. Create a signaling interface for the inside and outside IP interfaces.

To create a new Signaling Interface, navigate to **Device Specific Settings → Signaling Interface** and click **Add**. The following screen shows the signaling interfaces defined for the reference configuration.



7.14. End Point Flows - Server Flow

When a packet is received by Avaya SBCE, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for this destination endpoint are applied. The context is maintained, so as to be applied to future packets in the same flow.

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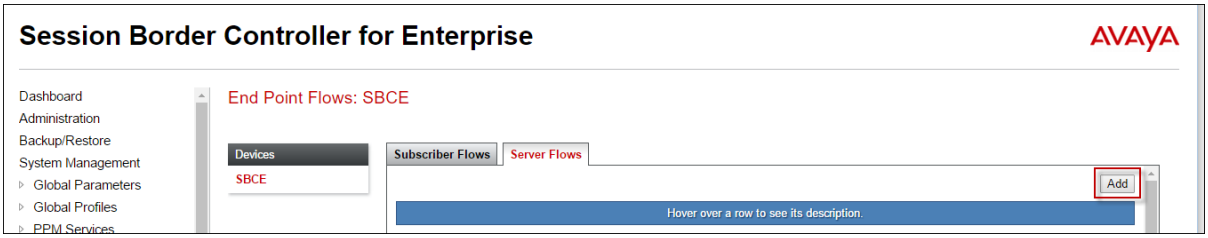
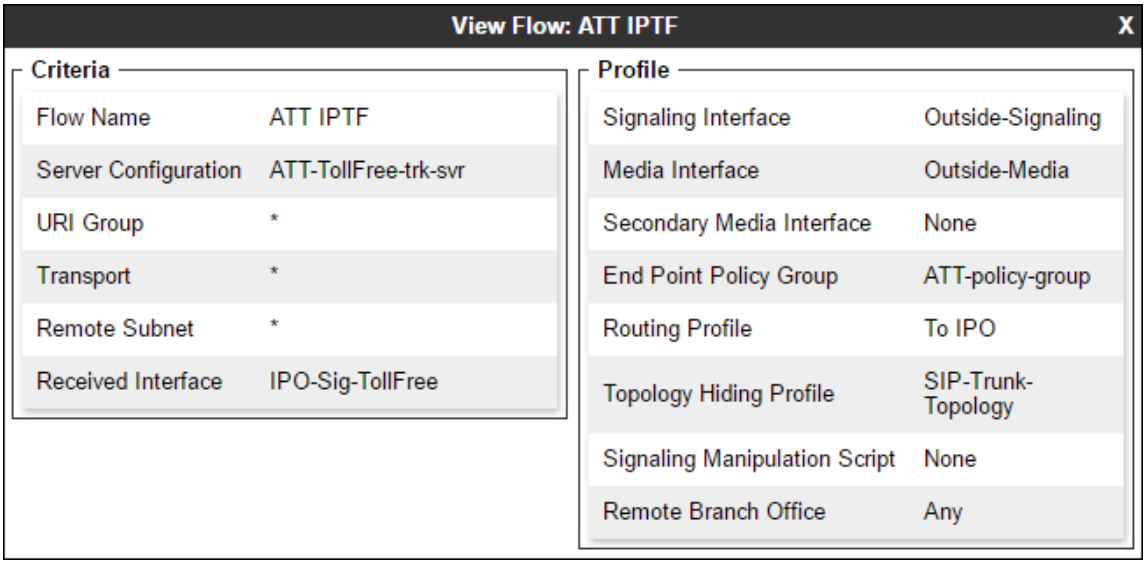
Create a Server Flow for IP Office and AT&T IPTF service. To create a Server Flow, navigate to **Device Specific Settings → End Point Flows**. Select the **Server Flows** tab and click **Add** as highlighted below.



The following screen shows the flow named **ATT IPTF** viewed from the reference configuration. This flow uses the interfaces, polices, and profiles defined in previous sections.



View Flow: ATT IPTF

Criteria

Flow Name	ATT IPTF
Server Configuration	ATT-TollFree-trk-svr
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	IPO-Sig-TollFree

Profile

Signaling Interface	Outside-Signaling
Media Interface	Outside-Media
Secondary Media Interface	None
End Point Policy Group	ATT-policy-group
Routing Profile	To IPO
Topology Hiding Profile	SIP-Trunk-Topology
Signaling Manipulation Script	None
Remote Branch Office	Any

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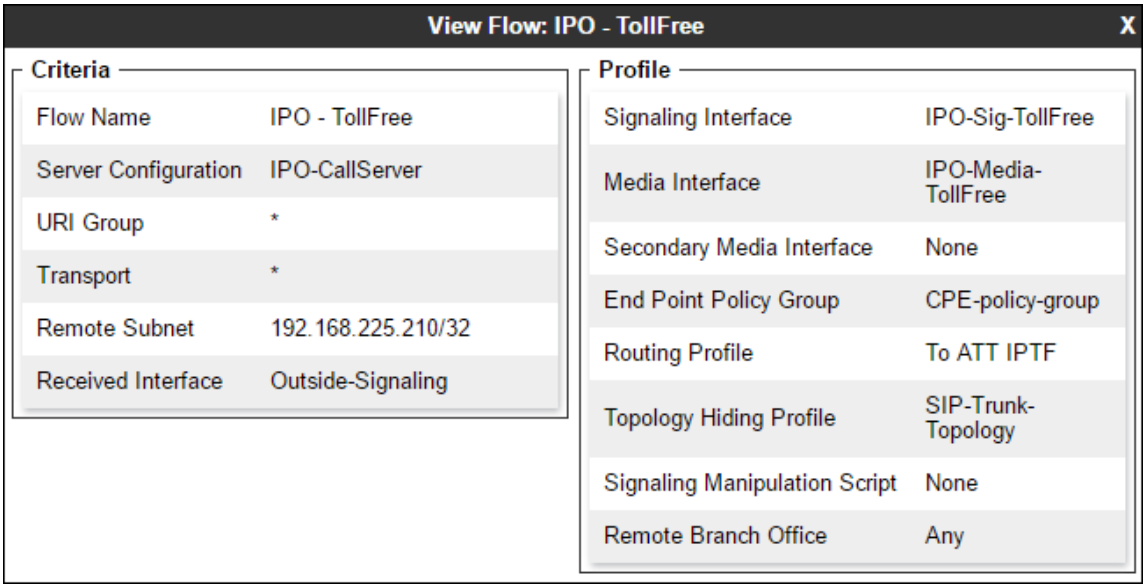
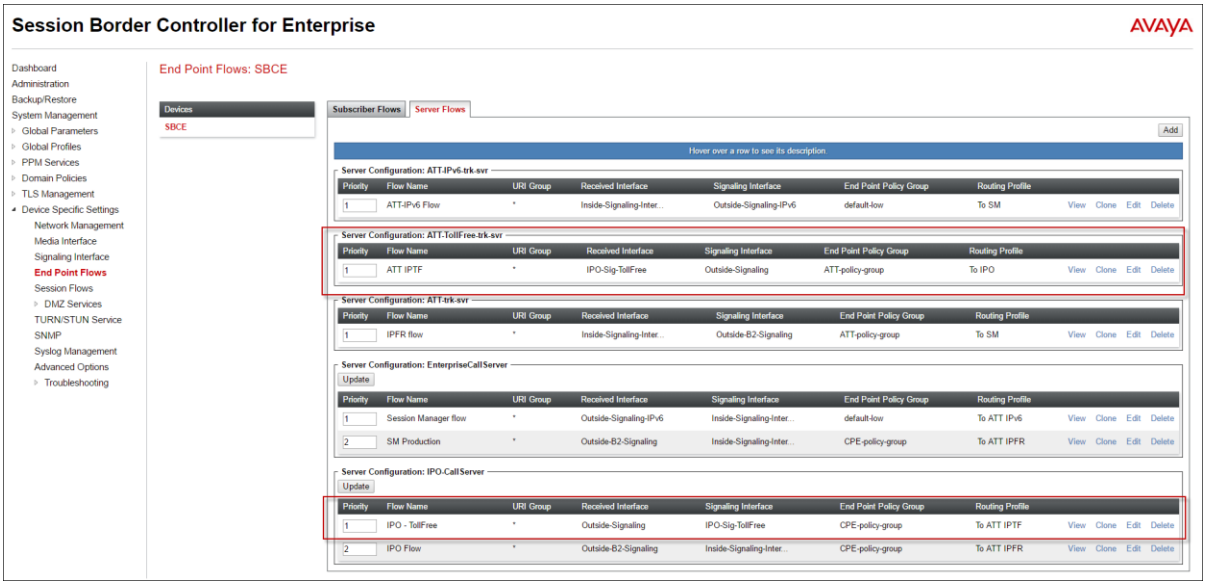
Once again, select the **Server Flows** tab and click **Add**. The following screen shows the flow named **IPO - TollFree** viewed from the reference configuration. This flow uses the interfaces, polices, and profiles defined in previous sections. In addition, the **Remote Subnet** is configured with the AT&T-provided IP address for IPTF service, along with "/32", i.e., **192.168.225.210/32**. Using "/32" for the subnet mask indicates that a single IP address, not a range of addresses, is the criteria to match for this particular IP Office flow.



The following screen summarizes the Server Flows configured in the reference configuration in the shared test environment. The highlighted flows are the ones relevant to the configuration of the SIP trunk to AT&T IPTF.



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8. AT&T IP Toll Free Service Configuration

AT&T provides the IPTF service border element IP address, the access DID numbers, and the associated DNIS digits used in the reference configuration. In addition the AT&T IPTF features, and their associated access numbers, are also assigned by AT&T. AT&T requires that the Avaya SBCE public (B1) IP address be provided to the IPTF service, as part of the provisioning process. For more information, consult reference **[7]**.

9. Verification Steps

The following procedures may be used to verify the Avaya IP Office Release 10.0 and Avaya SBCE Release 7.1 with the AT&T IPTF service configuration.

9.1. AT&T IP Toll Free Service

The following scenarios may be executed to verify Avaya IP Office R10.0 functionality with the AT&T IPTF service:

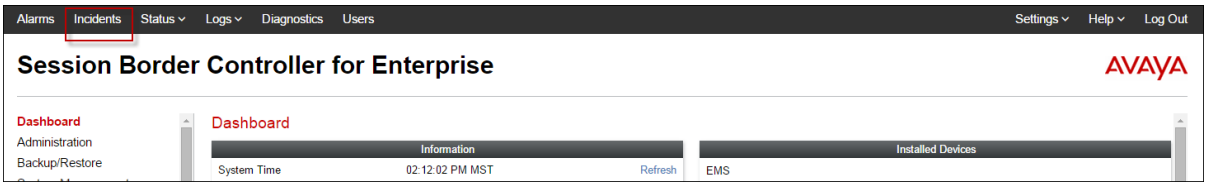
- Place inbound calls, answer the calls, and verify that two-way talk path exists. Verify that the calls remain stable for several minutes and disconnects properly.
- Incoming calls using the G.729A and G.711 ULAW codecs.
- Verify basic call functions such as hold, transfer, and conference.
- Place an inbound call to a telephone, but do not answer the call. Verify that the call covers to voicemail (e.g., Voicemail Pro). Retrieve the message either locally or from PSTN.
- Using the appropriate IPTF access numbers and codes, verify the "Legacy Transfer Connect" DTMF initiated features.
- Inbound fax using T.38 or G.711.
- SIP OPTIONS monitoring of the health of the SIP trunk.

9.2. Avaya SBCE

This section provides verification steps that may be performed with the Avaya SBCE.

9.2.1. Incidents

The Incident Viewer can be accessed from the Avaya SBCE Dashboard as highlighted in the screen shot below.



Use the Incident Viewer to verify Server Heartbeat and to troubleshoot routing failures.

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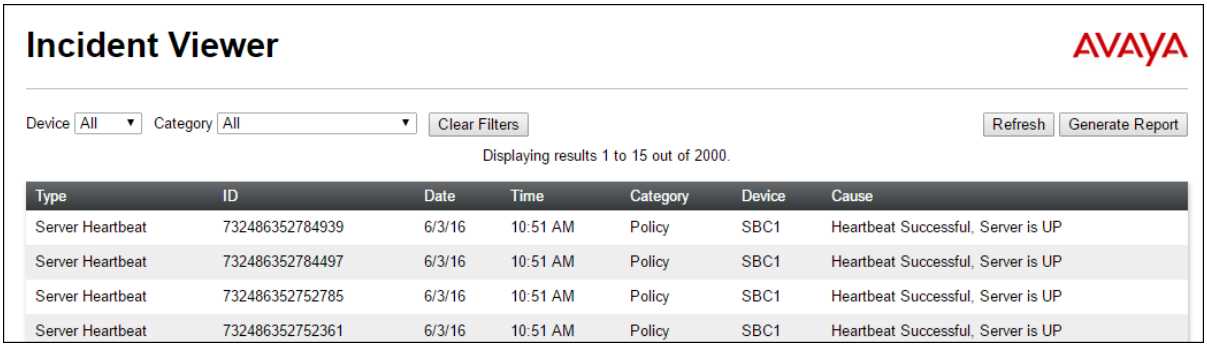
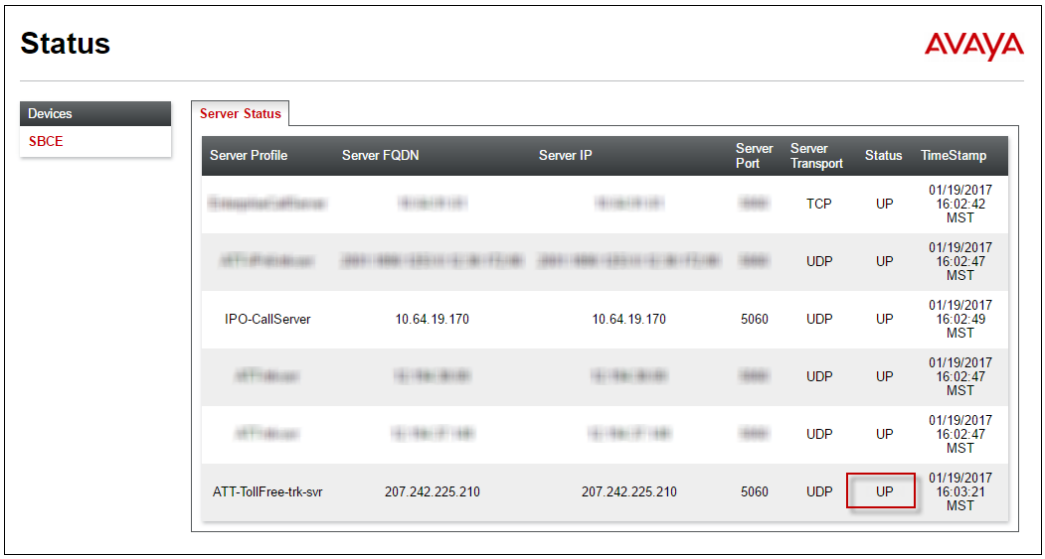


Type	ID	Date	Time	Category	Device	Cause
Server Heartbeat	732486352784939	6/3/16	10:51 AM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	732486352784497	6/3/16	10:51 AM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	732486352752785	6/3/16	10:51 AM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	732486352752361	6/3/16	10:51 AM	Policy	SBC1	Heartbeat Successful, Server is UP

9.2.2. Server Status

The **Server Status** can be access from the Avaya SBCE Dashboard by selecting the **Status** menu, and then **Server Status**.

A pop-up window will appear with the **Status** of **UP** for the AT&T IPTF. The **Server Profile** will only list servers with Server Configuration settings that have Heartbeats enabled, see **Section 7.4.2**.



Server Profile	Server FQDN	Server IP	Server Port	Server Transport	Status	TimeStamp
[illegible]	[illegible]	[illegible]	[illegible]	TCP	UP	01/19/2017 16:02:42 MST
[illegible]	[illegible]	[illegible]	[illegible]	UDP	UP	01/19/2017 16:02:47 MST
IPO-CallServer	10.64.19.170	10.64.19.170	5060	UDP	UP	01/19/2017 16:02:49 MST
[illegible]	[illegible]	[illegible]	[illegible]	UDP	UP	01/19/2017 16:02:47 MST
[illegible]	[illegible]	[illegible]	[illegible]	UDP	UP	01/19/2017 16:02:47 MST
ATT-TollFree-trk-svr	207.242.225.210	207.242.225.210	5060	UDP	UP	01/19/2017 16:03:21 MST

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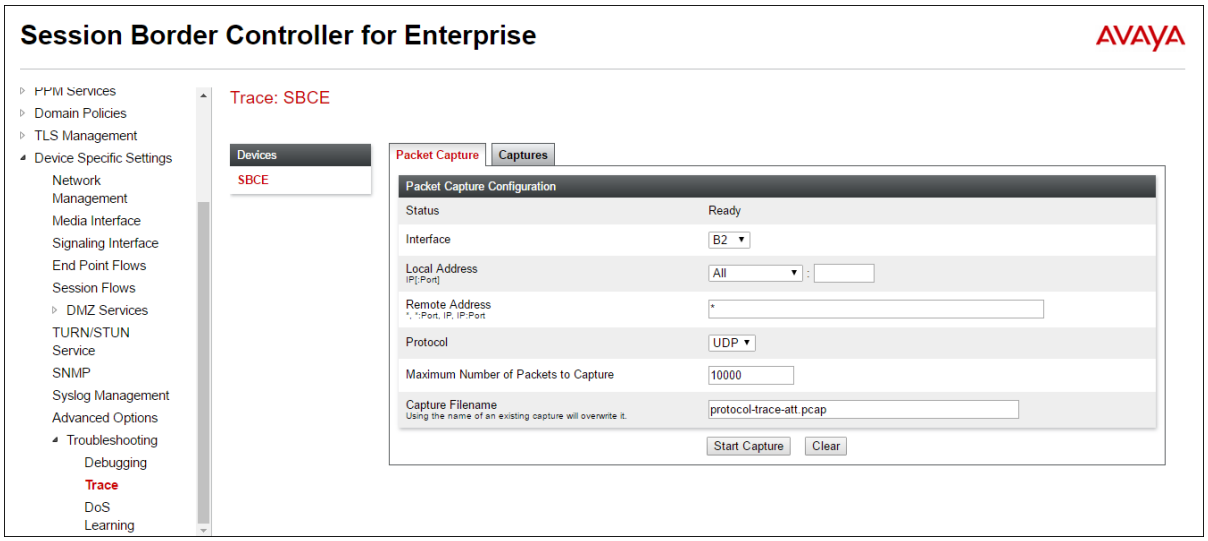
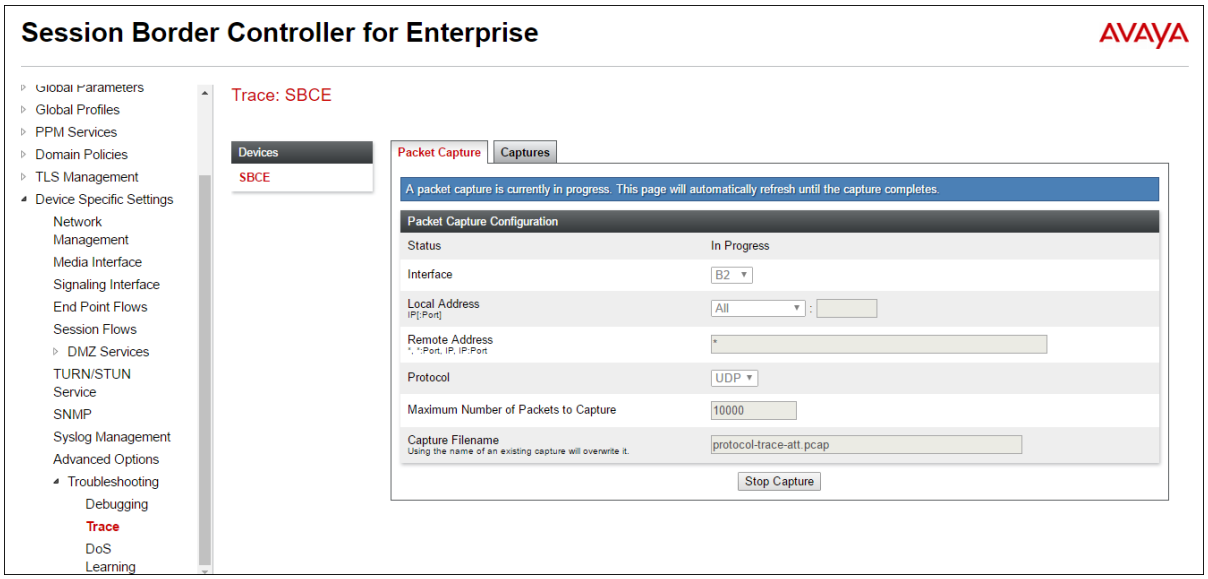
9.2.3. Tracing

To take a call trace, navigate to **Device Specific Settings → Troubleshooting → Trace** and select the **Packet Capture** tab. Populate the fields for the capture parameters and click **Start Capture** as shown below.



When tracing has reached the desired number of packets the trace will stop automatically, or alternatively, hit the **Stop Capture** button at the bottom.



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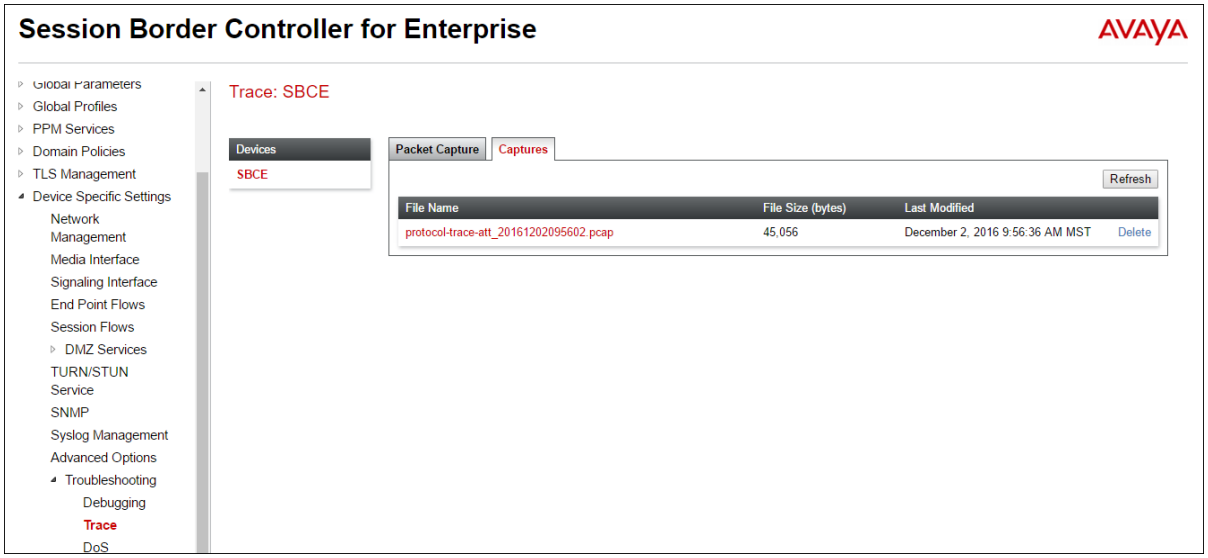
Select the **Captures** tab to view the files created during the packet capture.



The packet capture file can be downloaded and then viewed using a Network Protocol Analyzer like WireShark.

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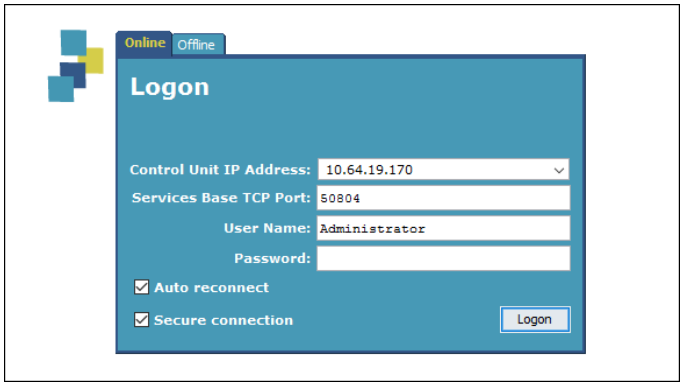
9.3. Avaya IP Office 10

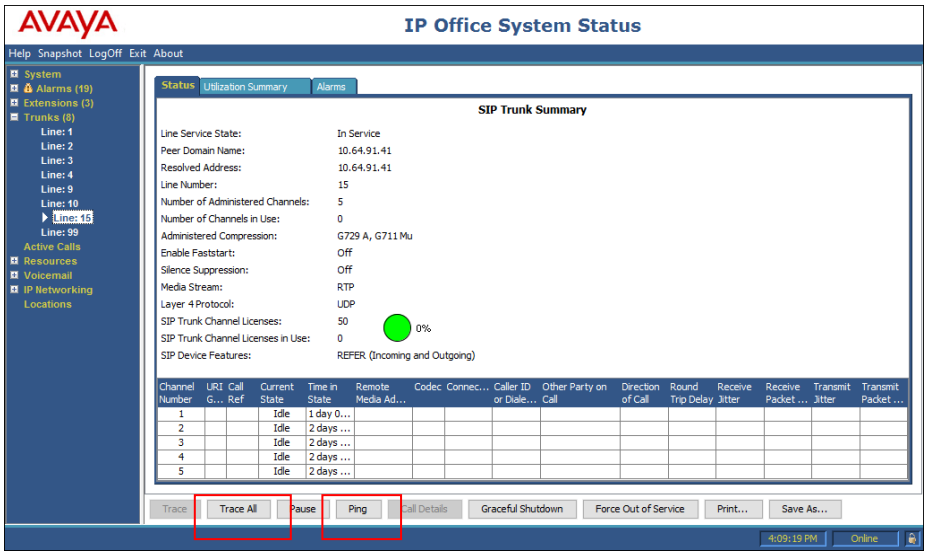
The following items may be used to analyze/troubleshoot Avaya IP Office operations.

9.3.1. 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: 15** from the left navigation menu. (SIP Line 15 is configured in **Section 5.4**). A screen such as the one shown below is displayed. In the lower left, the **Trace All** button may be pressed to display tracing information as calls are made using this SIP Line. The **Ping** button can be used to ping the other end of the SIP trunk (e.g., the Avaya SBCE).



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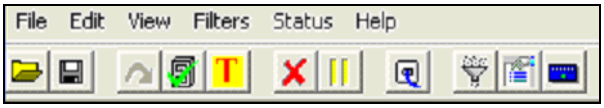
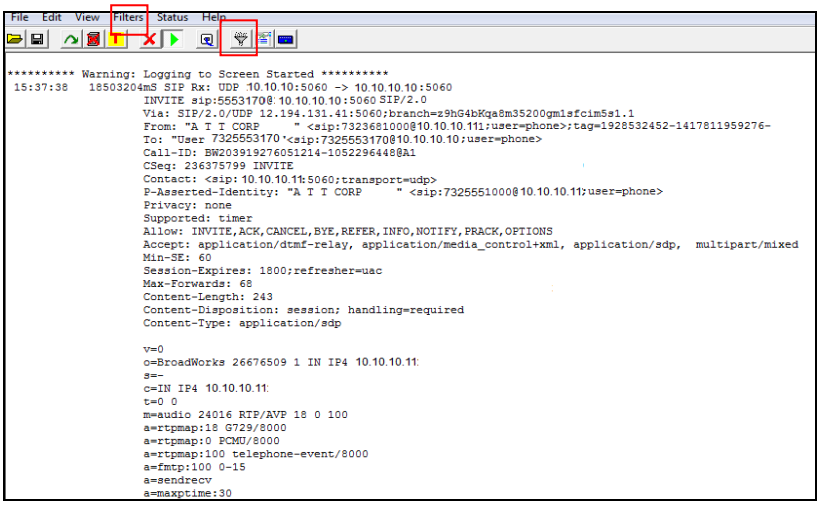
9.3.2. System Monitor Application

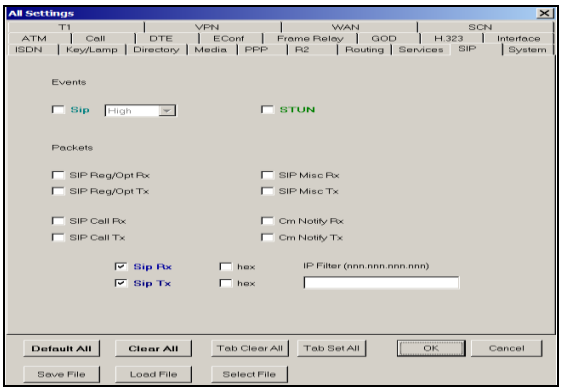
The System Monitor application can also be used to monitor or troubleshoot Avaya IP Office functionality (see reference **[3]**). The System Monitor application can typically be accessed from **Start → Programs → Avaya IP Office → Monitor**.



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 is a sample of a monitored inbound call.



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



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10. Conclusion

As illustrated in these Application Notes, Avaya IP Office Release 10.0 and Avaya Session Border Controller for Enterprise Release 7.1 can be configured to interoperate successfully with the AT&T IP Toll Free service using AVPN or MIS/PNT transport connections, utilizing service features listed in **Section 2.1,** and within the limitations described in **Section 2.2**.

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

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11. References

Avaya:

Avaya product documentation is available at http://support.avaya.com. Additional Avaya IP Office information can be found at: http://marketingtools.avaya.com/knowledgebase/

[1] *IP Office ™ Platform 10.0, Deploying Avaya IP Office ™ Platform Servers as Virtual Machines*, Document Number 15-601011, Issue 04d, July 2016

[2] *IP Office ™ Platform 10.0, Deploying Avaya IP Office ™ Platform IP500 V2*, Document Number 15-601042, Issue 31h, Aug 2016

[3] *Administering Avaya IP Office ™ Platform with Manager*, Release 10.0, August 2016

[4] *Administering Avaya Communicator on IP Office*, February 2016

[5] *Deploying Avaya Session Border Controller for Enterprise*, June 2016

[6] *Administering Avaya Session Border Controller*, June 2016

AT&T IPTF Service:

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

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