

Avaya Solution & Interoperability Test Lab

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

Abstract

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

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

Avaya IP Office 8.1 is a versatile communications solution that combines the reliability and ease of a traditional telephony system with the applications and advantages of an IP telephony solution. This converged communications solution can help businesses reduce costs, increase productivity, and improve customer service.

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

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

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1. Introduction

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

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

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

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 AVPN¹ or MIS/PNT² transport services.

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

2. General Test Approach and Test Results

The test environment (see Figure 1) consists of:

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

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

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

2.1. Interoperability Compliance Testing

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

¹ AVPN uses compressed RTP (cRTP).

² MIS/PNT does not support cRTP.

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

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

- Incoming calls from PSTN, routed by the IPTF service, via the Avaya SBCE, to Avaya IP Office. These calls are via the Avaya IP Office SIP Line and may be generated/answered by Avaya SIP telephones/Softphones, H.323 telephones, Analog telephones, Analog fax machines or via Hunt Groups. Coverage to Avaya IP Office Voicemail Pro, and Voicemail Pro auto-attendant applications, may also be used.
- Inbound fax using T38 or G.711, and G3 or SG3 endpoints.
- Proper disconnect when the caller abandoned a call before answer, and when the Avaya IP Office party or the PSTN party terminated an active call.
- Proper busy tone heard when an Avaya IP Office user called a busy PSTN user, or a PSTN user called a busy Avaya IP Office user (i.e., if no redirection was configured for user busy conditions).
- SIP OPTIONS monitoring of the health of the SIP trunk. In the reference configuration Avaya IP Office sent OPTIONS to the IPTF service Border Element and AT&T responded with 405 Method Not Allowed (which is the expected response). That response is sufficient for Avaya IP Office to consider the connection up.
- Incoming calls using the G.729 and G.711 ULAW codecs.
- Long duration calls.
- DTMF transmission (RFC 2833) for successful voice mail navigation, including navigation of a simple auto-attendant application configured on Avaya IP Office Voicemail Pro as well as IPTF DTMF generated features.
- Telephony features such as call waiting, hold, transfer, and conference.
- AT&T IP Toll Free features such as Legacy Transfer Connect and Alternate Destination Routing were also tested.

2.2. Test Results

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

2.2.1. Known Limitations

- 1. The Avaya IP Office fax feature "T.38 Fallback" (to G.711) is not supported in the reference configuration.
- 2. G.726 codec support G.726 codec is not supported by IP Office.

- 3. Avaya IP Office only supports a packet size of 20 msecs, and therefore does not specify a PTIME value in the SIP SDP (in either requests or responses) Network responses include MAXPTIME=20, and network requests include MAXPTIME=30.
 - Although no issues were found during testing, the AT&T IPTF service recommends a value of 30ms when AVPN transport is used.
- 4. Avaya IP Office uses fixed RFC2833 Telephone Event type 101 in SIP requests Avaya IP Office uses a fixed RFC2833 Telephone Event type of 101, and the AT&T network responses do comply with a value of 101. However, AT&T network SIP requests specify Telephone Event type 100, and Avaya IP Office complies with a value of 100.
 - No issues were found during testing as a result of this behavior.
- 5. IP Trunk shuffling is not supported with Avaya IP Office 8.1 using the 500v2 platform.
 - IP Trunk shuffling *is* supported with Avaya IP Office 8.1 using the server platform.

2.3. Support

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

Avaya customers may obtain documentation and support for Avaya products by visiting <u>http://support.avaya.com</u>. 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 <u>http://support.avaya.com</u>) to directly access specific support and consultation services based upon their Avaya support agreements.

3. Reference Configuration

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

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

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

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

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

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

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

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

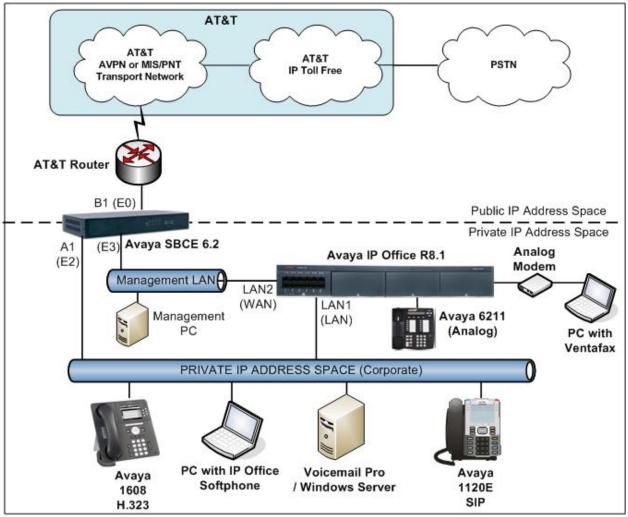


Figure 1: Reference Configuration

3.1. Illustrative Configuration Information

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

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

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

 Table 1: Illustrative Values Used in these Application Notes

3.2. Call Flows

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

3.2.1. Inbound

The first call scenario illustrated in the figure below is an inbound AT&T IPTF service call that arrives on Avaya IP Office, which in turn routes the call to a hunt group, phone or a fax endpoint.

- 1. A PSTN phone originates a call to an IPTF service number.
- 2. The PSTN routes the call to the AT&T IPTF service network.
- 3. The AT&T IPTF service routes the call to the Avaya SBCE.
- 4. The Avaya SBCE performs SIP Network Address Translation (NAT) and any specified SIP header modifications, and routes the call to Avaya IP Office.
- 5. Avaya IP Office applies any necessary digit manipulations based upon the DID and routes the call to a hunt group, phone or a fax endpoint.

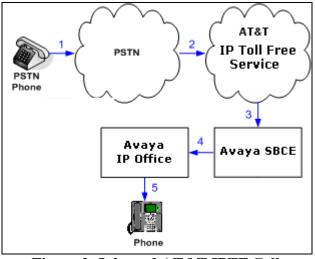


Figure 2: Inbound AT&T IPTF Call

3.2.2. Coverage to Voicemail

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

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

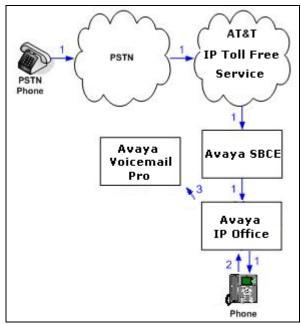


Figure 3: Coverage to Voicemail (Voicemail Pro)

4. Equipment and Software Validated

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

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

Table 2: Equipment and Software Versions

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

5. Avaya IP Office Configuration

Note - This section describes attributes of the reference configuration, but is not meant to be prescriptive. In the following sections, only the parameters that are highlighted in **bold** text are applicable to this document. Other parameter values may or may not match based on local configurations. Additionally, the screen shots referenced in these sections may not be complete at times. For more information on installing Avaya IP Office consult reference [1].

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

Configuration Service User Login									
IP Office :	00E007058008 - IP 500 V2								
<u>S</u> ervice User Name Service User Password	OK Cancel Help								

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5.1. Physical, Network, and Security Configuration

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

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

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

🐮 Avaya IP Office R8.1 Manager					- 7 🛛
File Edit View Tools Help					
🗄 🏖 🖆 - 🖬 🖪 🖪 🔜 🚹 🗸 🗸	r 🐸 🛹 👔 🕴 ooeoo7o58oo8	-	Control Unit	• 2 COMBO6210/ATM4	
IP Offices	Control Unit		6	COMBO6210/ATM4	$\lim_{k \to \infty} \mathbf{X} \leq \mathbf{X} > \mathbf{X} \leq \mathbf{X} > \mathbf{X} < \mathbf{X} > \mathbf{X} < \mathbf{X} < $
BOOTP (1) Operator (3) Ouecorto58008 System (1) -↑↑ Line (5) Control Unit (2) User (13) User (13) User (13) Service (0) RA5 (1) WanPort (0)		/ersion .1 (43) .1 (43)	Unit Device Number Unit Type Version Serial Number Unit IP Address Interconnect Number Module Number	2 COMBO6210/ATM4 8.1 (43) 00e007058008 0.0.0.0 0 Control Unit	

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

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

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

🕼 Avaya IP Office R8.1 Manager 00E007058008 [8.1(63)] [Administrator(Administrator)]									
File Edit View Tools Help									
2 🗁 - 🔄 🗖 💽 🚍 🚹 🖌 🐸 🗢 🔞									
i 00E007058008 🔹 IP Route	1	• 0.0.0.0	•						
IP Offices		IP Route		X	0.0.0.0		📸 • 🔛 🗙 🗸 <		
♣ BOOTP (1) ♀ Operator (3) ● 00E007058008 ● System (1) - - - Control Unit (2) ● Extension (11) ● Extension (11) ● HuntGroup (5) ● Service (0) ● Service (0) ● Incoming call Route (21) ● WanPort (0) ● Directory (0) ● Time Profile (3) ● Firewall Profile (1)	IP Address	IP Mask 255.255.255.240 255.255.255.0	Gateway 192.168.42.20 0.0.0.0	IP Route IP Address IP Mask Gateway IP Address Destination Metric			 0 240 20 ✓ 		

5.2. Licensing

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

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

IP Offices	License	×	SIP Trunk Channels
BOOTP (1)	License Type	Licenses	
System (1)	 IP500 Voice Networking Channels IP500 Voice Networking Channels 	License Key	
一行 Line (5) 一一つ Control Unit (3) 一級 Extension (22)	IPSec Tunnelling Microsoft CRM Integration (users)	License Type License Status	SIP Trunk Channels Valid
User (22)	Nobile Worker	Instances	255
Short Code (62)	Constant of the second	Expiry Date	Never
RAS (1) Incoming Call Route (8)	Phone Manager Pro Pro (per seat) Phone Manager Pro IP Audio Enable		
	Power User		
Firewall Profile (1) IP Route (4)	Referred Edition Additional VoiceMa		
Account Code (0)	RAS LRQ Support (Rapid Response)		
	Receptionist		
Auto Attendant (0)	SIP Trunk Channels		

If Avaya IP Telephones will be used, verify the Avaya IP endpoints license. Click **License** in the Navigation pane and **Avaya IP endpoints** in the Group pane. Confirm a valid license with sufficient "Instances" in the Details pane. Note that in some cases duplicate license entries may

JF; Reviewed: SPOC 3/15/2013 be listed (e.g, Avaya IP endpoints below). One will display a key sequence in the License Key field while the other will display "Virtual".

IP Offices	License	×××	Avaya IP endpoints
BOOTP (1)	License Type	Licenses	
	🛼 1600 Series Phones		
🖃 🤜 Verizon1	👟 3rd Party IP Endpoints	License Key	
	🛼 Advanced Edition		
一行了 Line (5)	Advanced Small Community Network	License Type	Avaya IP endpoints
	🛼 AUDIX Voicemail	License Status	Valid
🛶 🖉 Extension (22)	🔍 Avaya IP endpoints		Yaliu
User (22)	🔍 Avaya IP endpoints	Instances	255
	Rranch Edition		
Short Code (62)	👟 CCC Agent Rostering	Expiry Date	Never

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

IP Offices	License		×××	Power User
	License Type	^	Licenses	
- 👰 Operator (3)	🐜 IP500 Universal PRI (Additional char			
🖃 🤜 Verizon1	👟 IP500 Upgrade Standard to Professi		License Key	
System (1)	👟 IP500 Voice Networking Channels			
一行了 Line (5)	👟 IP500 Voice Networking Channels		License Type	Power User
	🛼 IPSec Tunnelling		License Status	Valid
Extension (22)	🛼 Microsoft CRM Integration (users)		License Dialus	100
User (22)	👟 Mobile Worker		Instances	255
HuntGroup (4)	👟 Mobility Features			Marray
Short Code (62)	🍋 Office Worker		Expiry Date	Never
Service (0)	🍬 one-X Portal for IP Office			
RAS (1)	🍬 Phone Manager Pro			
WanPort (0)	👟 Phone Manager Pro (per seat)			
Directory (1)	👟 Phone Manager Pro IP Audio Enable			
Time Profile (0)	Rower User			
Firewall Profile (1)	🛼 Preferred Edition (VoiceMail Pro)			
1 IP Route (4)	👟 Preferred Edition Additional VoiceMa			
Account Code (0)	🐜 Preferred/Advanced to Branch Editic			
License (76)	Reporting			

5.3. System Settings

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

5.3.1. System Tab

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

🐮 Avaya IP Office R8.1 Manag	er 00E00705800	8 [8.1(63	8)] [Administ	rator(A	dministra	tor)]						
File Edit View Tools Help												
1 2 🖻 - 🖬 I 🔤 🔜 🔝	🗸 🎂 🏞 🗽											
00E007058008 🛛 System		 00E0070 	158008	-								
IP Offices	System	×××				00E0070	58008			Ľ	(- 🗎 🛛 🗙	 ✓ <
BOOTP (1) Operator (3) Operator (3) Operator (3) Operator (3) Operator (3) Operator (1) Operator (1) Operator (1) User (13) Operator (5) Service (0) RA5 (1) Operator (0) Operator (0)	Name	System Name Contact Set con Device II TFTP Ser HTTP Ser	LAN1 LAN2 t Information itact information or rver IP Address le Server Type	DNS	System und	. 168 .	Directory Services	System I	Events SM Locale Branch Pre Local Numi	əfix	Twinning VC United States r	
		Avaya H Enable S Automati	PC IP Address TTP Clients Only oftphone HTTP I ic Backup ting Config Sour	Provisionii	ng V	email Pro/Ma			🗌 Favor F	RIP Routes,	, over static rout	55

5.3.2. LAN 1 Settings

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

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

🌃 Avaya IP Office R8.1 Manage	r 00E00705800	8 [8.1(63)] [Administrator(A	dministrator)]	
File Edit View Tools Help				
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00E007058008 System		 00E007058008 		
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Select the LAN1 \rightarrow VoIP tab as shown in the following screen. The following settings were used in the reference configuration:

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

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Select the LAN1 \rightarrow Network Topology tab as shown in the following screen, and enter the following:

- **Public IP Address**: The **0.0.0** default value is used. This means Avaya IP Office will use the LAN1 IP address specified on the LAN1 **LAN Settings** tab described above (192.168.42.1).
- Public Port to 5060.
- **Firewall/NAT Type** is set to **Open Internet**. With this configuration, STUN will not be used.
- Binding Refresh Time is set to 120 (used for OPTIONS interval, see Section 5.9).
- Click the **OK** button (not shown).

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

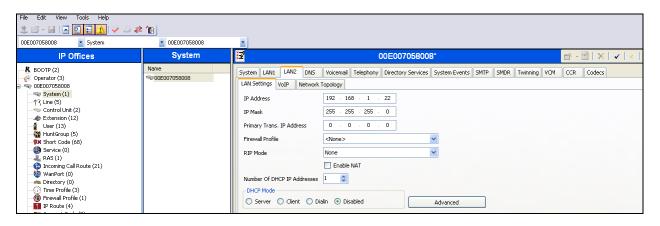
If SIP endpoints are used, select the **SIP Registrar** tab. The following screen shows the settings used in the reference configuration. Note that the **Domain Name** field is set to **customerb.com**, (the CPE domain), otherwise the LAN1 IP address is used for registration.

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		Domain Name customerb.com	
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5.3.3. LAN 2 Settings

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

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



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

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

5.3.4. Voicemail

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

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

• Set Voicemail Type: Voicemail Lite/Pro.

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SPOC 3/15/2013	©2013 Avaya Inc. All Rights Reserved.	IPO81SBCE62TF

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

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5.3.5. System Telephony Configuration

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

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

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

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

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5.3.6. System Codecs Configuration

Navigate to the **System** \rightarrow **Codecs** tab. On the left, observe the list of **Available Codecs**. By selecting codecs in this column, they will appear in the **Default Codec Selection** \rightarrow **Unused** column. Codecs may be selected from the **Unused** list and moved to the **Selected** column by use

of the >> button, thereby making the selected codecs available in other screens where codec configuration may be performed (e.g., Sip Lines and Extensions).

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

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

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5.4. SIP Line

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

- **ITSP Domain Name:** Set to the AT&T border element IP address supplied by AT&T (e.g., **135.25.29.74**).
- In Service and Check OOS: These boxes are checked (default).
 - Note that the Out Of Service (OOS) option is used in conjunction with SIP OPTIONS (see Section 5.9).
- Call Routing Method: Matched values based on the Request URI, or To Header contents, may be selected. In the reference configuration, the default Request URI setting was used.
- **Country Code:** Use the default <blank>.
- Send Caller ID: Set to Diversion Header.
- **REFER Support:** Verify that this open is *not* selected (default).
- Use the default values for the other fields.
- Click **OK**.

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

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♣ BOOTP (2)	Lin Line Type †↑1 Analogue Trunk †↑2 Analogue Trunk ↑↑3 Analogue Trunk ↑↑4 Analogue Trunk ▶17 SIP Line	SIP Line Transport SI Line Number ITSP Domain Name Prefix National Prefix Country Code	17 135.25.29.74 0 0 00 Diversion Header By Source IP address	In Service Use Tel URI Check OOS Call Routing Method Originator number for forwarded and twinning calls. Name Priority Caller ID from From header Send From In Clear User-Agent and Server Headers	Request URI System Default	×

5.4.1. SIP Line - Transport Tab

Select the **SIP Line** \rightarrow **Transport** tab and configure the following:

- **ITSP Proxy Address:** Set to the Avaya SBCE A1 interface IP address **192.168.42.20** (see Section 3).
- Network Configuration → Layer 4 Protocol. Set to UDP.
- Network Configuration → Send Port: Set to 5060 (default).
- Network Configuration \rightarrow Use Network Topology Info: Set to LAN 1.
- Calls Route via Registrar: Enabled (default).
- **Click OK** (not shown).

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5.4.2. SIP Line - SIP URI Tab

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

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

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

In this method the following information is specified:

- The **VIA** field will automatically be populated with the IP address of the system LAN interface (LAN 1) with which the SIP trunk is associated (see **Sections 5.3.2** and **5.4.1**).
- Local URI, Contact, Display Name, and PAI: Set to Use Internal Data. Note – This PAI setting directs Avaya IP Office to send the PAI (P-Asserted-Identity) header instead of the default PPI (P-Preferred-Identity) header when appropriate (e.g., privacy calls). The PAI header will be populated from the data set in the SIP tab of the call initiating User as shown in Section 5.5.
- **Registration:** Set to the default **0: <None>**.
- **Incoming Group:** Set here to **100**. This value references the **Incoming Call Routes** in **Section 5.7**.
- **Outgoing Group:** Set to **100**. Note that although the AT&T IP Toll Free service does not support outbound calls, this value was set to match the Incoming Group.
- Max Calls per Channel: In the reference configuration this was set to 10. This sets the maximum number of simultaneous calls that can use the URI before Avaya IP Office returns busy to any further calls.
- Click **OK** to save the information.

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		Max Calls per Channel	

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

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

In this method the following information is specified:

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

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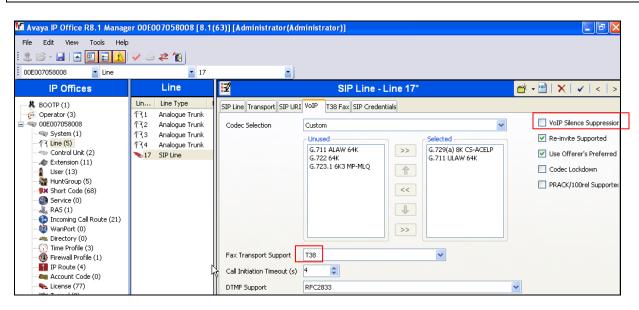
- To edit an existing entry, click an entry in the list and click the **Edit...**button.
- When all SIP URI entries have been added/edited, click **OK** at the bottom of the screen (not shown).

5.4.3. SIP Line - VoIP Tab

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

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

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



5.4.4. SIP Line - T38 Fax

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

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

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

5.5. Users, Extensions, and Hunt Groups

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

5.5.1. Analog User Extn207

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

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

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

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

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- 🖘 System (1)	🛔 Extn203	203						
(F7 Line (5)	🛔 Extn204	204	SIP Display Name (Alias)	Analog Phone				
	🛔 Extn205	205	Contact	0000031053				
- A Extension (11)	Extn206	206						
User (13)	Extn207	207						
HuntGroup (5)	Extn208	208		Anonymous				

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

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5.5.2. IP Phone User Extn500

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

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

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

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 BOOTF (1) Oberator (3) Oberator (3) System (1) T (1) (e (5) Control Unit (2) Extension (11) User (13) HurdGraup (5) Short Code (68) Service (0) ARS (1) Directory (0) Direct	Name Extension Extn201 201 Extn202 202 Extn203 203 Extn204 204 Extn205 205 Extn206 200 Extn207 207 Extn208 500 Extn209 200 Extn200 600 Extn200 600 Extn700 700 RemoteMan France	Voice Type User Rights User Rights User Rights Cut of hours User Rights Cut of hours User Rights	hortCodes Source Numbers Telephony Forwarding Dial In Voice Recording But Extn500	ton Programming Menu Programming Mobility < >

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

IP Offices	U	ser	₽	Extn500: 500* 🛛 🖄 🗸 🗸 🗸 🗸	< >
BOOTP (1)	Name Extn201	Extension 201	Button Programming Me	nu Programming Mobility Phone Manager Options Hunt Group Membership Announcements SIP Personal Directory	< >
	Extn202 Extn203 Extn204	202 203 204	SIP Name SIP Display Name (Alias)	0000011051 H323 Phone	
	Extn205 Extn206 Extn207	205 206 207	Contact	0000011051	
HuntGroup (5) Short Code (68)	Extn208	208		Anonymous	

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

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

00E007058008 Use	r 🛓	500 Extn500	1 🕹 🖉 - 🖬 🖃 💽 🚹 🗸 🖉 🖉 1	6
IP Offices	User	12	Extn500: 500*	≝ - X √ < >
K BOOTP (1) Gerator (3) Gerator (3) Gerator (3) Gerator (3) Gerator (3) Gerator (1) Gerator	Name Extension Extra201 201 Extra202 202 Extra203 203 Extra204 204 Extra205 205 Extra204 204 Extra205 206 Extra206 206 Extra208 208 Extra208 500 Extra700 700 NoUser Remote	User Voicemail DND Voicemail Code Confirm Voicemail Code Voicemail Email O Off DTMF Breakout Reception / Breakout (Breakout (DTMF *2) Breakout (DTMF *3)	Copy Forward Alert	Dial In Voice Recording Button Progra Voicemail On Voicemail Help Voicemail Ringback Voicemail Enail Reading LIMS Web Services

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

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File Edit View Tools Help	⊉ 1						
00E007058008 User	💌 600 Extri	1600	_]				
IP Offices	User	×=			Extn600: 600		📸 • 🕑 🗙 🗸 < >
BOOTP (1) Operator (3) Operator (3) System (1) -73 (1nc (5) -74 (1nc (5) -75 (1nc (5)) -74 (1nc (5)) -75 (1nc (5)) -74 (1nc (5)) -74 (1nc (5)) -75 (1nc (5)) -74 (1nc (5)) -75 (1nc (5)) -74 (1nc (3)) -75 (1nc (3)) -75 (1nc (3)) -75 (1nc (1)) -75 (1nc (1)) <	Extn201 22 Extn202 24 Extn203 22 Extn204 22 Extn204 22 Extn205 22 Extn206 22 Extn207 22 Extn207 22 Extn208 55 Extn208 55 Extn208 55	03 04 05 05 06 07 08 00 00 00		ttings Multi-line Options Call ****** CNone> Logged On (No change)		ding Dial In Voice Recording Force Login Force Account Code Outgoing Call Bar Inhibit Off-Switch Forward Can Intrude Can Intrude Can Intrude Can Trace Calls CCR Agent Automatic After Call Work. Deny Auto Intercom Calls	 Menu Programming Mobility < >

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

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

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

00E007058008 Extension	- 8	001 500) 2 🗈 - 🖬 🖪 💽 🔝 🗸 🗸 🖉	
IP Offices	Extension	E	H323 Extension: 8001 500	🖆 • 🗙 🗸 🖌 🖘
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License (76)		TDM->IP Gain	Default	×
User Rights (8)		IP->TDM Gain	Default	×
 Auto Attendant (2) ARS (2) 		Supplementary Services	None	×

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

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

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

IP Offices	Use	1	XXX XXX		Extn500: 500		💣 + 🖭	× ✓ < >
BOOTP (2)	Name	Extension	Mobility P	hone Manager Options H	unt Group Membership Annour	ncements SIP	Personal Directory	< >
- 💯 Operator (3)	📲 Extn207	207						
00E007058008	🛔 Extn208	208	Index	Name	Number			Add
	2- Extn500	500	01	PSTN Phone	7325552438			
(FT Line (6)	5- Extn600	600						Remove
Control Unit (2)	Extn700	700						Edit
	NoUser							Luit
User (13)	Remote Man.							

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

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

5.5.3.1 SIP Avaya 1120E

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

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

IP Offices	U	ser	×				Extn600	: 600)				ë -	×	🗸 <
★ BOOTP (1) ✓ Operator (3) ● 00E007058008 ● 5ystem (1) ←↑↑ Line (5) ● Control Unit (2) ● Extension (11) ● User (13) ●● HunGroup (2) ●> Short Code (79) ●● Service (0) ●● Incoming Call Route (f ●● Directory (0) ●● Firewall Profile (1) ●● Firewall Profile (1) ●● License (76) ●● Auto Attendant (2) ●● Akto Attendant (2) ●● Akto Attendant (2) ●● Akto Attendant (2) ●● Akto Attendant (2) ●● System (1)	To NoUser	Extension 201 202 203 204 205 206 207 208 500 600 700	Full Na Exten: Locale Priorit Syster Profile	m Password ame sion y y m Phone Riç	Shor	5 None Basic U Rec Enal Enal Enal	* * States (US Er	e rtal Ser Worker	nuter	Forwardin	ng Dial	I In Vo	v Recordi	•	Button Proc

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

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	Name	Extension	Button Programming Me	u Programming Mobility Phone Manager Options	Hunt Group Membership Announcements SI	P Personal Directory
—🕖 Operator (3)	🛔 Extn201	201				
🖻 🖘 00E007058008	🛔 Extn202	202	SIP Name	0000021052		
	🛔 Extn203	203		SIP Phone		
一行了 Line (5)	🛔 Extn204	204	SIP Display Name (Alias)	SIP Phone		
	🛔 Extn205	205	Contact	0000021052		
Æxtension (11)	🛔 Extn206	206		· · · · · · · · · · · · · · · · · · ·		
User (13)	🛔 Extn207	207				
W HuntGroup (5)	🛔 Extn208	208		Anonymous		
Short Code (68)	2- Extn500	500				
Service (0)	Extn600	600				
RAS (1)	Extn700	700				

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

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00E007058008	ser	- 600	Extn600	1 2 🖙 - 🖬 🖬 💽 🖬 🛦	✓ ² / ₂ ≈ ⁴
IP Offices	U	ser	12°	Extn600: 600*	<u>a</u> * • X √ <
BOOTP (1) Generator (3) @ 00E007058008 @ System (1)	Name Extra01 Extra02 Extra03 Extra04 Extra06 Extra06 Extra06 Extra06 Extra070 Extra00 Extra	Extension 201 202 203 204 205 206 207 208 500 600 700	User Voicemail DND Voicemail Code Confirm Voicemail Code Voicemail Email Off Off DTMF Breakout Reception / Breakout (Breakout (DTMF *2) Breakout (DTMF *3)	Copy O Forward	lephony Forwarding Dial In Voice Recording Button Pro; 4

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

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IP Offices	User	E	Extn600: 600	≝ - X √ < :
BOOTP (1) Gerator (3) Gerator (3) Gerator (3) Gerator (3) Gerator (1) Gerator (1	Name Extension Extra201 201 Extra201 201 Extra202 202 Extra203 203 Extra204 204 Extra205 206 Extra206 208 Extra206 500 Extra206 600 Extra207 700 Fxtn700 700 Nol.Ser Remote	Call Settings Supervisor Setti Outside Call Sequence Inside Call Sequence Ringback Sequence No Answer Time (secs) Wrap-up Time (secs)	ortCodes Source Numbers Telephony ings Multi-line Options Call Log Default Ring Default Ring Default Ring System Default (15) 2 Off © 100	Forwarding Dial In Voice Recording Butt 4

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

00E007058008 Use	r	• 600 E	xtn600), izi - 🖬 🖪 💽 🖬 🤳	\ √ ä ₹	2 🔞	
IP Offices	Use	r	2	Extn600: 600'		C	<- × < <
K BOOTP (1) Greater (3) Greater (3) Greater (1) Greater	Extr201 21 Extr202 23 Extr203 21 Extr204 21 Extr205 21 Extr206 22 Extr206 22 Extr206 23 Extr206 23 Extr206 24 Extr206 55 Extr206 55	Extension 101 102 103 104 105 106 107 106 100 100 100 100 100 100 100		es Source Numbers Telephor ttings Multi-Ine Options Call ****** dNone> dNone> Logged On (No change) System Default (10)		Dial In Voice Recording Force Login Force Account C Outgoing Call Ba Inhibit Off-Switcl Can Intrude Cannot be Intruc Can Trace Calls CCR Agent Automatic After	ode , n Forward/Transfer led

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

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

Exte	nsion		- 800	0 600		•	2 🖻	- 🖃 🖪 📰 🗘 🗸 🐸 🛹 🔞
IP Offices	E	xtension	1	Ш			SIP	Extension: 8000 600
BOOTP (1) Operator (3) Operator (3) Operator (3) Operator (3) Operator (3) Operator (3) Operator (1) Ope	\$ 8003	Extension 201 202 203 204 205 206 207 208 600 500 700	Module 8D1 8D1 8D1 8D1 8D1 8D1 8P1 8P1 0 0 0	Extn Extens Base E Caller I Reset 1 Device Module Port	xtension Display T Yolume A type	ype ifter Calls		8000 600 On V Avaya 1120E Sip (Language: English) 0 0 V

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

- The **IP Address** default value is used (**0.0.0.**).
- Check the **Reserve Avaya IP endpoint license** box.
- In the reference configuration the **Codec Selection** parameter is set to **Custom**, and the same codec list used in **Section 5.3.6** is used.
- Other fields may retain default values.

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00E007058008 Extension	- 8000	600 🔹			
IP Offices	Extension	XXX	SIP Extension: 800	00 600	📸 • 🔛 🗙 🗸 < >
BOOTP (2) Operator (3) ODE007058008 System (1) T Line (5) Control Unit (2) Extension (11) User (13) WarPort (0) Marce (0) WanPort (0) WanPort (0)	Id Extension M ▲ 1 201 BI ▲ 2 202 BI ▲ 3 203 BI ▲ 4 204 BI ▲ 5 205 BI ▲ 6 206 BI ▲ 7 207 BF ▲ 8 208 BF ▲ 8000 600 0 ▲ 8003 500 0	IP Address Codec Selection	< 0 . 0 . 0 . 0 Custom Custom G.711 ALAW 64K G.722 64K G.723.1 6K3 MP-MLQ C <lic< li=""> C <lic< li=""> C</lic<></lic<>	G.729(a) 8K C5-ACELP	 VoIP Silence Suppression Local Hold Music ✓ Allow Direct Media Path ✓ Re-invite Supported Use Offerer's Preferred Co ✓ Reserve Avaya IP endpoir Reserve 3rd party IP endp
Directory (0) Time Profile (3) Firewall Profile (1) Fi	ß	Fax Transport Support TDM->IP Gain IP->TDM Gain DTMF Support	None Default Default RFC2833	· · · · · · · · · · · · · · · · · · ·	v v

5.5.3.2 SIP Avaya IP Office Softphone

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

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

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ID 00E007055008 User 700 Extn700 IP Offices User 700 Extn700 IP Offices User Extn200 Operator (3) Extn201 201 Extn202 Extn203 203 System (1) Extn203 203 Control Unit (2) Extn204 204 Extn205 205 Confirm Password Extn204 Extn205 205	
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Mame Extension Operator (3) Extra01 Extra02 201 System (1) Extra03 Extra03 203 Extra04 204 Confort Unit (2) Extra05 Extra05 205 Extra05 205 Extra05 206	
Operator (3) Extn201 201 Outcomail DND ShortCodes Source Numbers Telephony Forwarding Dial In Voice Recording Extn201 202 Name Extn20 Figure 1 Extn202 202 Name Figure 2 Extn204 204 Confirm Password Extn205 205 Extn201 204 Confirm Password	📸 • 🔛 🗙 🗸 < >
Water (L3) Exh.202 207 Water (L3) Exh.202 208 With Short Code (68) Exh.202 208 Water (10) Exh.202 208 Water (11) Exh.202 208 Water (12) Exh.202 Exh.202 Water (12) Exh.2	

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

Extension M Extension M 201 BE 202 BE		â 🖻 - 🖃 🖻 🚉 💁 🗸 😂 🗢 🐐 H323 Extension: 8004 700
Extension Extension M 201 BD	X X X X	,
201 BC	Extn VoIP	
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; ; ; ;; ;;;;;;;;;;;;;;;;;;;;;;;;;;;;;	205 BC 206 BC 207 BF 208 BF 200 600 0 203 500 0	204 BC 205 BC 206 BC 207 BF 208 BF 000 0 003 500 004 700

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- **SIP Extension** \rightarrow **VoIP** tab for extension 700.
 - The **IP Address** default value is used (**0.0.0.**).
 - In the reference configuration the **Codec Selection** parameter is set to **Custom**, and the same codec list used in **Section 5.3.6** is used.
 - Other fields may retain default values.

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BOOTP (2) Operator (3) ODE07058008 System (1)	Id Extension M ▲1 201 BC ▲2 202 BC ▲3 203 BC ▲4 204 BC ▲5 205 BC ▲6 206 BC ▲7 207 BF ▲8000 600 0 ▲8000 600 0 ▲8000 500 0	Extn VoIP T38 Fax IP Address Codec Selection	0 . 0 . 0 . 0 Custom	lected 711 ULAW 64K 729(a) 8K CS-ACELP	VoIP Silence Suppression Local Hold Music Allow Direct Media Path Re-invite Supported Use Offerer's Preferred Co Reserve Avaya IP endpoir Reserve 3rd party IP endp
IP Route (4)		Fax Transport Support	None	~	
 Recourt Code (6) Recourt Code (77) Recourt Code (77) Recourt Code (77) 		TDM->IP Gain	Default	~	
- loser Rights (8)		IP->TDM Gain	Default	~	
→ Y ARS (2) → RAS Location Request (0)		DTMF Support	RFC2833	~	

5.5.4. Hunt Groups

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

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

DOED007050000 Image: HunkGroup Iool Sales Image: Sales Collective Group Sales: 1001* Image: Sales Image:	File Edit View Tools Help	≈ 1		
BOOTP (1) System Name Name Extension Image: System (1) Parts 1003 Source (1) Service 1002 System (1) Service 1002 Source (1) Service 1002 Extension (11) Seles 1001 Extension (11) Fathers (13) Name HuntGroup Agent's Status on No-Answer None Source (0) Agent's Status on No-Answer None RAS (1) User (12) Variable (21) Variable (21) WanPort (0) Directory (0) Extension Name Directory (0) Directory (0) Extension Service (12)	00E007058008 HuntGroup	• 1001 Sales		
Operator (s) Parts 1003 Operator (s) Parts 1003 Operator (s) Service 1002 System (n) Service 1002	IP Offices	HuntGroup	🛛 📴 🔹 Collective Group Sales: 1001*	🖻 🗙 🗸 <
		Parts 1003	Hunk Group Queuing Overflow Fallback. Voice Recording Announcements SIP Name Sales CCR Agent Group Extension 1001 Image: CCR Agent Group Ring Mode LongestWalting No Answer Time (secs) System Default (15) Hold Music Source Agent's Status on No-Answer None Voice List Extension None Voice List Voice List Extension Name Voice List Voice List V 207 Extension Name Voice List	

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

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00E007058008 • HuntGroup	- 10	01 Sales		-		
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Step 3 - Under the **Announcements** tab, check the **Announcements On** box. The wait time can be set to any desirable value. Make sure that the **Synchronize Calls** box is checked. These announcements are played if an agent for a particular skill is unavailable.

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00E007058008 • HuntGroup	▼ 1001 Sales ▼		
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BOOTP (1) Operator (3) Operator (3) Operator (3) Operator (3) Operator (3) Operator (1) True (5) Operator (1) Operator (2) Operato	System Name Extension Parts 1003 Service 1002 Sales 1001	Hunk Group Queuing Overflow Fallback Voicemail Voice Recording Announcements SIP Wak before 1st announcement (seconds) 30 Image: Synchronize Calls Image: Synchronize Calls Flag call as answered Image: Synchronize Calls Image: Synchronize Calls Image: Synchronize Calls Post announcement tone Image: Synchronize Calls Image: Synchronize Calls Image: Synchronize Calls Value of the synchronize Calls Image: Synchronize Calls Image: Synchronize Calls Image: Synchronize Calls Play Ist announcement Image: Synchronize Calls Image: Synchronize Calls Image: Synchronize Calls Value of the synchronize Calls Image: Synchronize Calls Image: Synchronize Calls Image: Synchronize Calls Value of the synchronize Calls Image: Synchronize Calls Image: Synchronize Calls Image: Synchronize Calls Value of the synchronize Calls Image: Synchronize Calls Image: Synchronize Calls Image: Synchronize Calls Value of the synchronize Calls Image: Synchronize Calls Image: Synchronize Calls Image: Synchronize Calls Value of the synchronize Calls Image: Synchronize Calls Image: Synchronize Calls Image: Synchronize Calls <	
		Wait before repeat (seconds)	

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

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IP Offices	HuntGroup	😰 Sequential Group Sales: 1001*	📸 • 🖭 🗙 🗸 < >
BOOTP (1) Operator (3) Operator (3) System (1) TQ ince (5) Control Unit (2) Extension (11) User (13) WhittGroup WhittGroup	System Name Extension Parts 1003 Service 1002 Sales 1001	Hunt Group Queuing Overflow Fallback Voice Recording Announcements SIP SIP Name 0000041054	

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

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00E007058008 HuntGrou	qu	1003	8 Parts	-					
IP Offices	H	IntGrou	ıp	×=	Sequ	ential Group Pa	rts: 1003	📑 - 🖻 🗙	✓ < >
BOOTP (1) Operator (3) Operator (3) ODE007058008 System (1) -↑↑ Line (5) Control Unit (2) Ware (13) User (13) User (13) Short Code (68) Service (0) RAS (1) Trooming Call Route (26)	System Name	Name Parts Service Sales	Extension 1003 1002 1001	Hunt Group Name Extension Ring Mode Hold Music S Agent's Sta Applies To User List – Extensio	Source n Name	ow Fallback Voicemail Parts 1003 Sequential No Change None	Voice Recording	Announcements SIP	System Default (

🌃 Avaya IP Office R8.1 Manage	er 00E007058	008 [8.1	(63)] [Admin	istrator(Adm	inistrato	r)]				- 7 ×
File Edit View Tools Help										
2 🖻 - 🖬 🖪 🔜 🔝	✓ ॐ ₹ [*]									
i 00E007058008 🔹 HuntGrou	p	• 1002	2 Service	-						
IP Offices	Hu	IntGrou	ıp	×	S	equen	tial Group Serv	rice: 1002	📥 - 🔛 🗙	✔ < >
BOOTP (1) Operator (3) ODE007058008 System (1) -↑↑ Line (5) Control Unit (2) Extension (11) User (13)	System Name	Name Parts Service Sales	Extension 1003 1002 1001	Hunt Group Name Extension Ring Mode Hold Music S	iource		Fallback Voicemail Service 1002 Sequential No Change	Voice Recording	Announcements SIP	System Default (
- ● HuntGroup - ● Short Code (68) - ● Service (0) - → ARS (1) - ● Incoming Call Route (26) - ● WanPort (0)				Agent's Stal Applies To User List Extension	n Nam	ne	None	v		

5.6. Short Codes

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

5.6.1. Call Center Access to Voicemail Pro

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

Short Code		
Code	*83	
Feature	Voicemail Collect	
Telephone Number	"CallCenter"	
Line Group ID	100 💌	
Locale	×	
Force Account Code		

5.6.2. Voicemail Access

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

Short Code		
Code	*17	
Feature	Voicemail Collect	
Telephone Number	?U	
Line Gro	0	
Locale	~	
Force Account Code		

5.7. Incoming Call Routes

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

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

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

🌃 Avaya IP Office R8.1 Manage	r 00E007	058008 [8.1(63)] [Administra	tor (Adr	ninistrator)]			
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00E007058008 🛛 Incoming •	Call Route	 100 0000 	011051	-				
IP Offices		Incoming Ca	III Route		10	0 0000011051	📸 - 🔤 🗙 🗸	< >
BOOTP (1) Operator (3) ODE00705008 System (1) (7 (line (5) Control Unit (2) Extension (11) User (13) HuntGroup (5) Short Code (68) Service (0) RAS (1) Oirectory (0) Directory (0) Firewall Profile (3) Firewall Profile (1) P Route (4)	Line G (*) 100 (*) 100 (Incoming Number 0000001050 0000011051 0000021052 0000031053 0000031054 0000041053 0000041054 0000041055 0000061056	Destination *83 500 Extn500 600 Extn600 207 Extn600 207 Extn600 *17 1001 Sales 1002 Service 1003 Parts		Standard Voice Recording Bearer Capability Line Group ID Incoming Number Incoming Sub Address Incoming CLI Locale Priority Tag Hold Music Source	Destinations Any Voice 100 0000011051		

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

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i 🗶 🗁 - 🕞 i 🛋 🔜 🔝 🛝	🗸 🛎 🥏	≥ ^									
00E007058008 🔹 Incoming	Call Route	• 100 0000	011051	•							
IP Offices		Incoming Ca	all Route		×	1	00 00000	011051	ď	- 🔤 🗙 🗸	< >
BOOTP (1)	Line G	Incoming Number	Destination		Standa	rd Voice Recording	Destination	s			
- 🦗 Operator (3)	100	0000001050	*83	_		TimeProfile		Destination		Fallback Extension	
00E007058008 System (1)	100	0000011051	500 Extn500		•	Default Value		500 Extn500	~		~
一行了 Line (5)	(2) 100 (2) 100	0000021052 0000031053	600 Extn600 207 Extn207		*		~	000 EAGIOOD	*		~
Control Unit (2)	0	0000031054	700 Extn700								
	100	0000041053	*17								
HuntGroup (5)	100	0000041054	1001 Sales								
9× Short Code (68)	100	0000051055	1002 Service								
Service (0)	100	0000061056	1003 Parts 0								
				1]]]						
WapPort (0)											

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

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

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

5.8. Call Center Provisioning in Avaya Voicemail Pro

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

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

In the reference configuration a Call Center function was provisioned to prompt callers to select a numeric option (1, 2, or 3), that would forward the call to an associated Avaya IP Office Hunt Group (**Sales, Service**, and **Parts**). This is accomplished via the following steps:

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

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

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

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

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

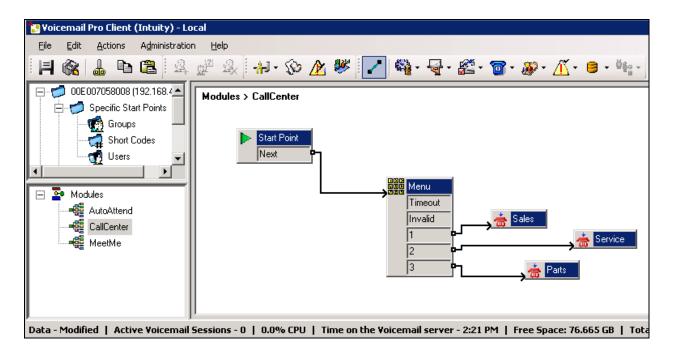
• Enter a name (e.g., **CallCenter**) and click on **Ok**. The new Start Point "CallCenter" will

appear under Modules, and a Start Point icon will appear in the work area.

- Click on the **Start Point** icon to activate the options bar at the top of the screen. From the options bar, select the **Basic Actions** icon **W**, select the **Menu** icon, and click on the work area to place the **Menu** icon.
 - i. Double click the **Start Point** icon.
 - 1. General tab \rightarrow Token Name = Start Point
 - 2. Click Ok
 - ii. Double click the **Menu** icon.
 - 1. General tab, Token Name = Menu
 - 2. Entry Prompts tab → Create an Entry Prompt that will tell the caller what digits to press to reach Sales, Service, and Parts (e.g., attendant.wav). To modify an existing recording, double click on the

.wav file and rerecord. If no .wav files exist, double click on the Eliconto open the .wav editor.

- 3. Touch Tone tab:
 - a. Select **1**, **2**, and **3** as the possible entry digits.
 - b. Select 4 for No of Retries.
 - c. Check the **Timeout** and **Invalid Entry** options.
- 4. Click on **Ok.**
- Click on the Telephony Actions icon , select the Transfer icon , and click on the work area to place the **Transfer** icon. Select and place two more Transfer Icons (these will be used for Sales, Service, and Parts).
 - i. Double click on the first **Transfer** icon ("**Sales**")
 - 1. General tab \rightarrow Token Name = Sales
 - 2. Specific tab \rightarrow Destination \rightarrow Mailbox \rightarrow Sales \rightarrow Ok
 - ii. Double click on the second **Transfer** icon ("Service").
 - 1. General tab \rightarrow Token Name = Service
 - 2. Specific tab \rightarrow Destination \rightarrow Mailbox \rightarrow Service \rightarrow Ok
 - iii. Double Click on the third **Transfer** icon ("**Parts**").
 - 1. General tab, Token Name = Parts
 - 2. Specific tab, Destination \rightarrow Mailbox \rightarrow Parts \rightarrow Ok
 - From the options bar, select the Connector icon 🖌 and:
 - i. Drag a connecting flow line from the **Start Point** box to the **Menu** box (see screen shot below).
 - ii. Drag connecting flow lines from each of the **Menu** options to their associated **Transfer** boxes (see screen shot below).



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

When the associated AT&T IPTF number is called from PSTN (e.g., 0000001050), the caller will be prompted to enter 1, 2, or 3 to access Sales, Service, or Parts. The associated Avaya IP Office extension (e.g., 207, 500, or 600) will then ring.

5.9. SIP Options Frequency

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

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

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

5.10. Saving Configuration Changes to Avaya IP Office

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

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

Click **OK** to execute the save.

Send Configuration	
IP Office Settings	
Configuration Reboot Mode	
💽 Merge	
🔘 Immediate	
🔘 When Free	
O Timed	
Reboot Time	
09:49	
Call Barring	
Incoming Calls	
Outgoing Calls	
OK Cancel	
OK Cancel	Help

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

6. Configure Avaya Session Border Controller for Enterprise

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

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

6.1. Initial Installation/Provisioning

Note: The installation and initial provisioning of the Avaya SBCE is beyond the scope of this document. Refer to reference **[7]** and **[8]** for additional information.

IMPORTANT! – During the Avaya SBCE installation, the Avaya SBCE Management interface, (labeled "E3" on the CAD-0208 platform), <u>must</u> be provisioned on a different subnet than either of the Avaya SBCE private and public network interfaces (e.g., A1 and B1). If this is not the case, contact your Avaya representative to have this resolved.

The Avaya SBCE installation typically defines public and private networks. As described in **Section 3**, the reference configuration defines the Avaya SBCE private interface A1 (IP address 192.168.42.20) on the same CPE network as the IP Office LAN1 interface (192.168.42.1). The connection to AT&T used the Avaya SBCE public interface B1 (IP address 192.168.64.130).

6.2. Log into the Avaya SBCE

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

- A. Access the web interface by typing "**https://x.x.x.**" (where x.x.x.x is the management IP address of the Avaya SBCE).
- B. Enter the login ID and password.



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

🗛 Dashboard - Avaya Session Bord	er Controller	for Ente				- 👌 • 🔊 -	📑 🖶 🔻 Page 🕶	Safety 🔻 To	ols 🕶 🔞 🕶
Alarms Incidents St	tatistics	Logs C	Diagnostics	Users			Setting	s Help	Log Ou
Session Bo	rder	Cont	roller	for Enter	prise			A۱	/AYA
Dashboard	^	Dashbo	ard						
Administration				Information			Installed Device	S	
Backup/Restore		System T	ïme	01:05:52 PM GMT	Refresh	EMS			
System Management		Version		6.2.0.Q30		A-SBCE			
 Global Parameters Global Profiles 	≡	Build Date	e	Wed Dec 19 15:22:2	1 UTC 2012	100002			
SIP Cluster			Alam	ns (past 24 hours)			Incidents (past 24 h	iours)	
 Domain Policies TLS Management 		None four				None found.	monaonito (publici 2 n n	iouroj	
Device Specific Setting	js								Add
Network Management					No	tes			Add
Media Interface					No note	es found.			

6.3. Global Profiles

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

6.3.1. Server Interworking – to Avaya IP Office

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

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

	General
Hold Support	 None RFC2543 - c=0.0.0.0 RFC3264 - a=sendonly
180 Handling	
181 Handling	None ○ SDP ○ No SDP
182 Handling	⊙ None ○ SDP ○ No SDP
183 Handling	None ○ SDP ○ No SDP
Refer Handling	
3xx Handling	
Diversion Header Support	
Delayed SDP Handling	
T.38 Support	
URI Scheme	
Via Header Format	 ● RFC3261 ○ RFC2543
	Next

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

Record Routes	 ○ None ○ Single Side ③ Both Sides
Topology Hiding: Change Call-ID	
Call-Info NAT	
Change Max Forwards	
Include End Point IP for Context Lookup	
OCS Extensions	
AVAYA Extensions	
NORTEL Extensions	
Diversion Manipulation	
Diversion Header URI	
Metaswitch Extensions	
Reset on Talk Spurt	
Reset SRTP Context on Session Refresh	
Has Remote SBC	
Route Response on Via Port	
Cisco Extensions	
	Finish

Solution & Interoperability Test Lab Application Notes ©2013 Avaya Inc. All Rights Reserved. The following screenshot shows the completed **General** tab form.

Alarms Incidents Statistics	s Logs Diagnosti	cs Users		Settings Help Log (
Session Borde	r Controlle	er for Enterprise		AVAY
Dashboard	Interworking P	Profiles: Avaya_SI		
Administration	Add			Rename Clone Delete
Backup/Restore	Interworking		Click here to add a description	n
System Management	Profiles			
Global Parameters	cs2100	General Timers URI Manipu	ulation Header Manipula	ation Advanced
Global Profiles	avaya-ru		General	
Domain DoS	OCS-Edge-S	Hold Support	NONE	
Fingerprint Server Interworking		180 Handling	None	
Phone Interworking	cisco-ccm	181 Handling	None	
Media Forking	cups			
Routing	Sipera-Halo	182 Handling	None	
Server Configuration	OCS-FrontEn	183 Handling	None	
Topology Hiding		Refer Handling	No	
Signaling Manipulation	ATT_SI	3xx Handling	No	
URI Groups	Avaya_SI	Diversion Header Support	No	
SIP Cluster		Delayed SDP Handling	No	
Domain Policies		, ,		
TLS Management		T.38 Support	Yes	
		URI Scheme	SIP	

6.3.2. Server Interworking – to AT&T

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

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

6.3.3. Routing - to Avaya IP Office

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

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

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

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

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

	Edit Routing Rule X
Each URI group may only be used onc	e per Routing Profile.
	Next Hop Routing
URI Group	*
Next Hop Server 1 IP, IP:Port, Domain, or Domain:Port	192.168.42.1
Next Hop Server 2 IP, IP:Port, Domain, or Domain:Port	
Routing Priority based on Next Hop Server	
Use Next Hop for In Dialog Messages	
Ignore Route Header for Messages Outside Dialog	
NAPTR	
SRV	
Outgoing Transport	○ TLS ○ TCP ④ UDP
	Finish

The following screenshot shows the completed Routing Profiles form.

A Routing Profiles - Avaya Session Bor	rder Conti	roller for				6		1 🖶 🔹 Page 🕶 S	iafety 🕶 Tools 👻 🕜	• '
Alarms Incidents Stati	istics	Logs Dia	agnostics	Users				Settings	Help Log O	ut
Session Bor	der	Contr	oller	for Ent	erpris	se			AVAYA	7
Dashboard	^	Routing F	Profiles:	Avaya_R						
Administration			Add					Rename	Clone Delete]
Backup/Restore		Routing Pro	ofiles			Click here to :	add a descripti	on.		
System Management		default			1					-
Global Parameters			R	outing Profile						_
 Global Profiles 		Avaya_R							Add	
Domain DoS	=	ATT_R		Priority U	RI Group	Next Hop Server	1 Next Hop S	oner 2		
Fingerprint					Ki Oloup		T Next hop a			
Server Interworking				1 *		192.168.42.1		View E	dit	
Phone Interworking										
Media Forking										
Routing										

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6.3.4. Routing – to AT&T

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

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

Each URI group may only be used on	ce per Routing Profile.
	Next Hop Routing
URI Group	*
Next Hop Server 1 IP, IP:Port, Domain, or Domain:Port	135.25.29.74
Next Hop Server 2 IP, IP:Port, Domain, or Domain:Port	135.25.29.74
Routing Priority based on Next Hop Server	
Use Next Hop for In Dialog Hessages	
Ignore Route Header for Messages Outside Dialog	
NAPTR	
SRV	
Outgoing Transport	○ TLS ○ TCP ③ UDP
	Finish

6.3.5. Server Configuration – To Avaya IP Office

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

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

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d. UDP Port: 5060

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

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

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

A Server Configuration - Avaya Session Border	Controll		👌 • 🔊 · 🗆	🖶 🔻 Page 🕶 Safety 🕶 Tools 👻 🕢 🖛
Alarms Incidents Statistics	Logs Diagnostics	Users		Settings Help Log Out
Session Border	Controller	for Enterpris	se	AVAYA
Dashboard	Server Configur	ation: Avaya_SC		
Administration Backup/Restore	Add			Rename Clone Delete
System Management	Server Profiles	General Authentication	Heartbeat Advanced	
 Global Parameters 	Avaya_SC	Server Type	Call Server	
 Global Profiles 	ATT_SC	IP Addresses / FQDNs	192.168.42.1	
Domain DoS				
Fingerprint		Supported Transports	UDP	
Server Interworking		UDP Port	5060	
Phone Interworking			Edit	
Media Forking				
Routing				
Server Configuration				

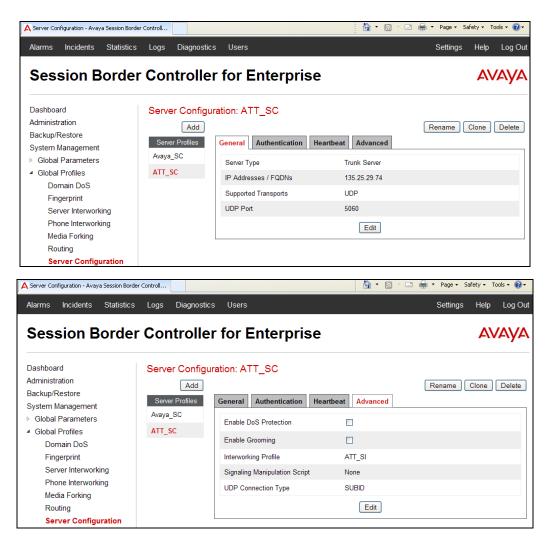
Alarms Incidents Statis	stics	Logs Diagnosti	cs Users		Settings Help Log Ou
Session Bord	ler	Controlle	er for Enterprise	è.	AVAYA
Dashboard	^	Server Config	uration: Avaya_SC		
Administration		Add			Rename Clone Delete
Backup/Restore		Server Profiles	General Authentication He	eartheat Advanced	
System Management			General Authentication no	eanbeat Advanced	
Global Parameters		Avaya_SC	Enable DoS Protection		
 Global Profiles 		ATT_SC		_	
Domain DoS	=		Enable Grooming		
Fingerprint			Interworking Profile	Avaya_SI	
Server Interworking			Signaling Manipulation Script	None	
Phone Interworking			UDP Connection Type	SUBID	
Media Forking			ODP Connection Type	30010	
Routing				Edit	

6.3.6. Server Configuration – To AT&T

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

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

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



6.3.7. Topology Hiding – Avaya IP Office

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

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

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

	l .	Lu	it Topology Hid	-		
						Add Header
Header		Criteria	Rep	lace Action	Overwrit	e Value
То	V IP/D	omain 🔉	Overwrite		 customerb. 	com Delete
Request-Line	V IP/D	omain 💉	Overwrite		 customerb. 	com Delete
From	V IP/D	omain 🔉	Overwrite		 customerb. 	com Delete
Record-Route	V IP/D	omain 🔉	Auto		▼	Delete
Via	V IP/D	omain 🔉	Auto		¥	Delete
ogy Hiding Profiles - Avaya Session	Border Contr				🏠 • 🔊 - 🖻	🖶 🔹 Page 🕶 Safety 🕶 Took
ogy Hiding Profiles - Avaya Session ns Incidents Statistic ession Borde	cs Logs D	Diagnostics	^{Users}	prise		ه • Page + Safety + Took Settings Help المعالم
ns Incidents Statistic	er Conti	roller		-	å • a · □	Settings Help
ns Incidents Statistic	er Conti	roller	for Enter	-		Settings Help
Incidents Statistic	cs Logs D er Conti Topolog Topology	roller 1 gy Hiding F (Add) Hiding	for Enter	_TH	to add a description.	Settings Help
ns Incidents Statistic SSSION BORDE Iboard nistration up/Restore em Management	er Conti Topolog	roller f gy Hiding F (Add) Hiding es	for Enter	_TH		Settings Help
ns Incidents Statistic SSION BORDE Iboard nistration up/Restore	cs Logs D er Conti Topology Profil default	roller 1 gy Hiding F Add Hiding les	for Enter	_TH		Settings Help
ns Incidents Statistic	cs Logs D er Contr Topology Profil default cisco_th_	roller 1 gy Hiding F Add Hiding les	for Enter	_TH Click here	: e to add a description.	Settings Help
Incidents Statistic S	CS Logs D CONTR Topology Topology Profil default cisco_th_ ATT_TH	roller 1 gy Hiding F Add Hiding les	for Enter Profiles: Avaya_ opology Hiding Header	_TH Click here Criteria	: e to add a description. Replace Ac	Settings Help
Incidents Statistic S	cs Logs D er Contr Topology Profil default cisco_th_	roller 1 gy Hiding F Add Hiding les Tr profile	for Enter Profiles: Avaya_ opology Hiding Header To	_TH Click here Criteria IP/Domain	: e to add a description. Replace Ac Overwrite	Settings Help
Incidents Statistic Session Borde aboard nistration up/Restore m Management abal Parameters abal Profiles Domain DoS Fingerprint Server Interworking Phone Interworking	CS Logs D CONTR Topology Topology Profil default cisco_th_ ATT_TH	roller 1 gy Hiding F Add Hiding les Tr profile	for Enter Profiles: Avaya_ opology Hiding Header To Request-Line From	_TH Click here Criteria IP/Domain IP/Domain IP/Domain	: e to add a description. Replace Ac Overwrite Overwrite Overwrite	Settings Help
Incidents Statistic Session Borde aboard nistration up/Restore m Management abal Parameters abal Profiles Domain DoS Fingerprint Server Interworking Phone Interworking Media Forking	CS Logs D CONTR Topology Topology Profil default cisco_th_ ATT_TH	roller 1 gy Hiding F Add Hiding les profile	for Enter Profiles: Avaya opology Hiding Header To Request-Line From Record-Route	_TH Click here IP/Domain IP/Domain IP/Domain IP/Domain	e to add a description. Replace Ac Overwrite Overwrite Overwrite Auto	Settings Help
Incidents Statistic Session Borde aboard nistration up/Restore m Management abal Parameters abal Profiles Domain DoS Fingerprint Server Interworking Phone Interworking	CS Logs D CONTR Topology Topology Profil default cisco_th_ ATT_TH	roller 1 gy Hiding F Add Hiding les profile	for Enter Profiles: Avaya_ opology Hiding Header To Request-Line From	_TH Click here Criteria IP/Domain IP/Domain IP/Domain	: e to add a description. Replace Ac Overwrite Overwrite Overwrite	Settings Help

6.3.8. Topology Hiding – AT&T

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

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

Alarms Incidents Statistics	s Logs Diagnosti	cs Users		:	Settings Help Log
Session Borde	r Controlle	er for Enter	prise		AVAY
Dashboard Administration	Topology Hidir Add	ng Profiles: ATT_	ТН	Re	name Clone Dele
Backup/Restore	Topology Hiding		Click here t	o add a description.	
System Management	Profiles default	Topology Hiding			
Global Profiles	cisco_th_profile	Header	Criteria	Replace Action	Overwrite Value
Domain DoS	ATT TH	То	IP/Domain	Auto	
Fingerprint	- Avaya TH	Request-Line	IP/Domain	Auto	
Server Interworking	/ daya_	From	IP/Domain	Auto	
Phone Interworking		Record-Route	IP/Domain	Auto	
Media Forking Routing		Via	IP/Domain	Auto	
Server Configuration					
Topology Hiding				Edit	
Signaling Manipulation					
URI Groups					

6.3.9. Signaling Manipulation

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

6.4. Domain Policies

The Domain Policies feature allows you to configure, apply, and manage various rule sets (policies) to control unified communications based upon various criteria of communication sessions originating from or terminating in the enterprise. These criteria can be used to trigger different policies which will apply on call flows, change the behavior of the call, manipulate SIP headers, and make sure the call does not violate any of the policies.

6.4.1. Application Rules

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

Application Rules - Avaya Session Bord	er Controller fo		<u>à</u>) ▼ 🗟 > 🖃 🖶 ▼ P	age 🔹 Safety 🔹 Tools 👻 🔞 🕶
Alarms Incidents Statist	cs Logs Diagnostic	s Users		Se	ettings Help Log Ou
Session Bord	er Controlle	r for Enterprise			AVAYA
Dashboard	Application Ru	les: default-trunk			
Administration	Add	Filter By Device			Clone
Backup/Restore	Application Rules		lte Texeleni	an an addian a naw ada in	
System Management		It is not recommended to edit the defau	lits. Try cionii	ng or adding a new rule in	stead.
Global Parameters	default	Application Rule			
Global Profiles	default-trunk			. Maximum Concurrent	Maximum Sessions
SIP Cluster		Application Type	In Ou	t Sessions	Per Endpoint
 Domain Policies 		Voice	~ ~	2000	2000
Application Rules					2000
Border Rules		Video]	
Media Rules		IM]	
Security Rules					
Signaling Rules			Miscell	aneous	
Time of Day Rules		CDR Support	None		
End Point Policy Groups		RTCP Keep-Alive	No		
Session Policies			E	dit	
TIC Management					

6.4.2. Media Rules

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

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

	Media QoS	х
	Media QoS Reporting	
RTCP Enabled		
	Media QoS Marking	
Enabled		
O ToS		
Audio Precedence	Routine	000
Audio T	Minimize Delay 🗸	1000
Video Precedence	Routine	000
Video ToS	Minimize Delay 🗸	1000
• DSCP		
Audio	AF11 💌	001010
Video	AF11 💌	001010
	Finish	

The next screen shot shows the completed Media Rules window.

A Media Rules - Avaya Session Border Contro	ller for Ent			in - 6	🗋 🗉 🖶 🔻 Page	 ✓ Safety	•
Alarms Incidents Statistics	Logs Diagnostic	s Users			Settir	ngs Help Log	Out
Session Borde	r Controlle	r for Ent	erprise			AVAy	Ά/
Dashboard	Media Rules: d	efault-low-me	d-QOS				
Administration	Add	Filter By Device	*		Rename	e Clone Dele	te
Backup/Restore System Management	Media Rules		(Click here to add a de	escription.		
Global Parameters	default-low-med	Media NAT N	ledia Encryption	Media Anomaly	Media Silencing	Media QoS	
Global Profiles	default-low-m			Media QoS Repo	orting		
SIP Cluster	default-high	RTCP Enabled			Jung		- 1
 Domain Policies Application Rule 	default-high-enc						-
Application Rules	avaya-low-me			Media QoS Mar	king		
Media Rules	default-low	Enabled		v			
Security Rules		QoS Type		DSCP			
Signaling Rules			_	Audio QoS			
Time of Day Rules		Audio DSCP		Audio QoS			
End Point Policy Groups		Audio DSCP		ALTI			_
Session Policies				Video QoS			
 TLS Management 		Video DSCP		AF11			
Device Specific Settings				Edit			

6.4.3. Signaling Rules

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

6.4.3.1 Avaya – Signaling QOS

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

A Signaling Rules - Avaya Session Border C	ontroller for				👌 • 🔊 - 🗆	🖶 🔻 Page 🕶 Sa	fety 🔹 Tools 👻 🔞 🕶
Alarms Incidents Statistic	s Logs Diagnostics	Users				Settings	Help Log Out
Session Borde	er Controller	for Enterpr	ise				AVAYA
Dashboard	Signaling Rules:	Avaya_SR					
Administration	Add	Filter By Device	~			Rename	Clone Delete
Backup/Restore	Signaling Rules			Click here to add	a description.		
System Management Global Parameters 	default	General Requests	Responses	Request Headers	Response Headers	Signaling QoS]
Global Profiles	No-Content-Type						
SIP Cluster	ATT_SR	Signaling QoS					
 Domain Policies 	Avava SR	QoS Type		DSCP			
Application Rules	Avaya_SR	DSCP		AF11			
Border Rules Media Rules Security Rules				Ed	it		
Signaling Rules							

6.4.3.2 AT&T – Signaling QOS

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

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A Signaling Rules - Avaya Session Border		Users	<u>∆</u> • ⊠	• E + Page • Safety • Tools • @ Settings Help Log O
Session Bord			9	ΑναγΑ
Dashboard Administration	Signaling Rules:	ATT_SR Filter By Device	*	
Backup/Restore System Management	Add Signaling Rules	Filler by Device	Click here to add a description.	Rename Clone Delete
 Global Parameters Global Profiles 	default No-Content-Type	General Requests Res	sponses Request Headers Response Hea	ders Signaling QoS
SIP Cluster	ATT_SR	Signaling QoS	V	
 Domain Policies Application Rules 	Avaya_SR	QoS Type DSCP	DSCP AF11	
Border Rules Media Rules		2001	Edit	
Security Rules Signaling Rules				

6.4.4. Endpoint Policy Groups – Avaya IP Office

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

Scrubber	^	Policy Groups:	default	LowAvaya							
User Agents		Add	Filter By	Device	*				Ren	ame	Delete
Global Profiles		Policy Groups			(Click here to	add a desci	ription.			
 SIP Cluster 		default-low									
Domain Policies		default-low-enc			Hove	er over a row	to see its d	escription.			
Application Rules		deladit-low-enc	Policy (Group							
Border Rules		default-med	-	•							
Media Rules		default-med							Sum	mary	Add
Security Rules		default-high	Orde	Application	Border	Media	Security	Signaling	Time of Day		
Signaling Rules	=	, in the second s				1.6 10			Day		
Time of Day Rules	_	default-high-e	1	default-	default	default- low-med-	default- low	Avaya SR	default	Edit	Clone
End Point Policy		OCS-default				QOS	IOW				
Groups		avaya-def-low									
Session Policies		· · · · · · · · · · · · · · · · · · ·									
TLS Management		defaultLow									
Device Specific Settings	~	defaultLowATT									

6.4.5. Endpoint Policy Groups – AT&T

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

Scrubber	^	Policy Groups:	defaultl	owATT								^
User Agents		Add	Filter By D	evice	*				Ren	ame	Delete	
Global Profiles		Policy Groups			(Click here to	add a desci	ription.				
SIP Cluster		default-low			11							
 Domain Policies 		default-low-enc			HOVE	er over a row	to see its d	escription.				
Application Rules			Policy G	roup								
Border Rules		default-med										
Media Rules		default-med							Sum	mary	Add	
Security Rules		default-high	Order	Application	Border	Media	Security	Signaling	Time of Day			
Signaling Rules	=					1.6.10			Day			
Time of Day Rules	-	default-high-e	1	default- trunk	default	default- low-med-	default- low	ATT_SR	default	Edit	Clone	
End Point Policy		OCS-default		trunk		QOS	1000					
Groups		avaya-def-low										
Session Policies												
TLS Management		defaultLowAv										
Device Specific Settings	~	defaultLow										~

6.5. Device Specific Settings

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

6.5.1. Network Management

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

Alarms Incidents Statistics	s Logs Diagnost	ics Users		Setting	gs Help Log O
Session Borde	r Controlle	er for Enterp	rise		AVAYA
 SIP Cluster Domain Policies 	Network Mana	igement: A-SBCE			
 TLS Management Device Specific Settings Network Management 	Devices A-SBCE		Interface Configuration s of an IP address or its ass lication restarts can be issu	ociated data require an a	
Media Interface Signaling Interface Signaling Forking		A1 Netmask 255.255.255.0 Add	A2 Netmask	B1 Netmask 255.255.255.0	Save Clear
End Point Flows Session Flows Relay Services		IP Address 192.168.42.20	Public IP	Gateway 192.168.42.1	Interface A1 V Delete
SNMP Syslog Management		192.168.64.130		192.168.64.254	B1 V Delete

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

Groups	^	Network Manag	ement: A-SBCE			
Session Policies						
TLS Management		Devices	Network Configurati	on Interface Configuration		
 Device Specific Settings 		A-SBCE	N	ame	Administrative Status	
Network			N	ame	Administrative Status	
Management			A1	Enabled		Toggle
Media Interface			A2	Disabled		Toggle
Signaling Interface			B1	Enabled		Toggle
Signaling Forking						

6.5.2. Advanced Options

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

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

Session Policies	^	Advanced Options: A-SBCE	
TLS Management			
 Device Specific Settings 			
Network		Devices CDR Listing Feature Control SIP Options Port Ranges	
Management		A-SBCE	-
Media Interface		Changes to the settings below require an application restart before taking effect. Application restarts can be issued from System Management.	
Signaling Interface		Pestako dan be ibadea nom <u>oyatem management</u> .	
Signaling Forking		Port Range Configuration	
End Point Flows		Signaling Port Range 12000 - 16000	
Session Flows			
Relay Services		Config Proxy Internal Signaling Port Range 42000 - 51000	
SNMP	=		
Syslog Management		Listen Port Range 9000 - 9999	
Advanced Options			
Troubleshooting	~		~

6.5.3. Media Interfaces

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

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



6.5.4. Signaling Interface

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

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

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

▷ SIP Cluster	^	Signaling Interf	face: A-SBCE							
Domain Policies										
TLS Management		Devices	Signaling Interfac	e						
Device Specific Settings		A-SBCE								
Network										Add
Management			Name	Signaling IP	TCP	UDP	TLS	TLS Profile		
Media Interface				5 5	Port	Port	Port			
Signaling Interface			Avaya-IPO	192.168.42.20		5060		None	Edit	Delete
Signaling Forking			ATT	192.168.64.130		5060		None	Edit	Delete
End Point Flows										

6.5.5. Endpoint Flows – to Avaya IP Office

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

	Edit Flow: Avaya-IPO X
Flow Name	Avaya-IPO
Server Configuration	Avaya_SC
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	ATT 💌
Signaling Interface	Avaya-IPO 💌
Media Interface	Avaya_IPO 💌
End Point Policy Group	defaultLowAvaya 👻
Routing Profile	ATT_R 💌
Topology Hiding Profile	Avaya_TH
File Transfer Profile	None 💌
	Finish

6.5.6. Endpoint Flows – To AT&T

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

	Edit Flow: ATT
Flow Name	Атт
Server Configuration	ATT_SC 💌
URI Group	* v
Transport	* 💌
Remote Subnet	*
Received Interface	Avaya-IPO 💌
Signaling Interface	ATT
Media Interface	ATT
End Point Policy Group	defaultLowATT 💌
Routing Profile	Avaya_R 💌
Topology Hiding Profile	ATT_TH
File Transfer Profile	None 💌
	Finish

Signaling Manipulation	End Point Flows:	A-SBCE	
URI Groups			
SIP Cluster	Devices	Subscriber Flows Server Flows	
Domain Policies	A-SBCE		Add
TLS Management			
 Device Specific Settings 		Click here to add a row description.	
Network		Server Configuration: ATT_SC	
Management		Priority Flow Name URI Received Signaling End Point Routing	
Media Interface		Group Interface Interface Policy Group Profile	
Signaling Interface		1 ATT * Avaya- ATT defaultLowATT Avaya_R	R View Clone Edit Delete
Signaling Forking			
End Point Flows		┌ Server Configuration: Avaya_SC	
Session Flows		Priority Flow Name URI Received Signaling End Point Policy Routing Group Interface Interface Group Profile	
Relay Services		· · · · · · · · · · · · · · · · · · ·	
SNMP		1 Avaya-IPO * ATT Avaya- defaultLowAvaya ATT_R	R View Clone Edit Delete
Syslog Management			
Advanced Options			

7. AT&T IP Toll Free Service Configuration

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

8. Verification Steps

The following procedures may be used to verify the Avaya IP Office R8.1 and the Avaya Session Border Controller for Enterprise 6.2, with the AT&T IP Toll Free service configuration.

8.1. AT&T IP Toll Free Service

The following scenarios may be executed to verify Avaya IP Office and the Avaya Session Border Controller for Enterprise 6.2, functionality with the AT&T IPTF service:

- Place inbound calls, answer the calls, and verify that two-way talk path exists. Verify that the calls remain stable for several minutes and disconnects properly.
- Verify basic call functions such as hold, transfer, and conference.
- Place an inbound call to a telephone, but do not answer the call. Verify that the call covers to Avaya Aura® Messaging voicemail. Retrieve the message from Avaya Aura® Messaging either locally or from PSTN.
- Using the appropriate IPTF access numbers and codes, verify the "Legacy Transfer Connect" DTMF initiated features.
- Inbound fax using T38 or G.711.
- SIP OPTIONS monitoring of the health of the SIP trunk. In the reference configuration Avaya IP Office sent OPTIONS to the AT&T IPTF service Border Element, (via the Avaya SBCE), and AT&T responded with 405 Method Not Allowed (which is the expected response). That response is sufficient for Avaya IP Office to consider the connection up.
- Incoming calls using the G.729 (A or B) and G.711 ULAW codecs.

8.2. Avaya IP Office 8.1

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

8.2.1. System Status Application

The System Status application can be used to monitor or troubleshoot Avaya IP Office. The System Status application can typically be accessed from Start \rightarrow Programs \rightarrow Avaya IP Office \rightarrow 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.

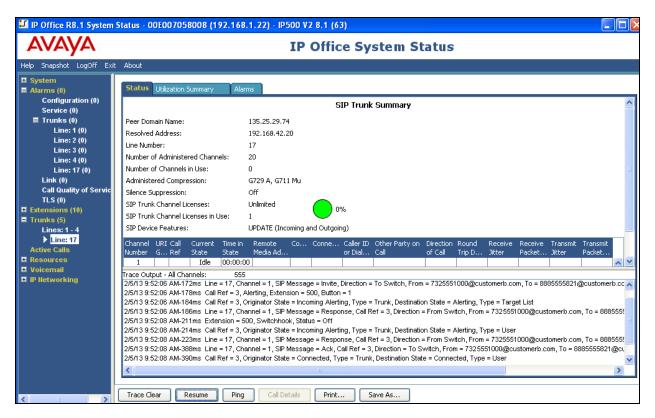


After logging in, select **Trunks** \rightarrow **Line:** 17 from the left navigation menu. (SIP Line 17 is configured in **Section 5.4**). A screen such as the one shown below is displayed. In the lower left,

the **Trace All** button may be pressed to display tracing information as calls are made using this SIP Line.

💵 IP Office R8 System Sta	tus - 00E	0070580	08 (192	.168.42.	1) - IP500	V2 8.0	0 (16)									
AVAYA		IP Office System Status														
Help Snapshot LogOff Exit	About															
Help Snapshot LogOff Exit I System I Alarms (0) I Trans (0) I Tran	Status Peer Dom Resolved Line Num Number of Number of Administer Silence So SIP Trunk SIP Trunk	ber: If Administer If Channels i red Compre uppression: Channel Lic Channel Lic E Features:	red Channi in Use: ssion: enses: enses in U	1: 1: 0 G 0 U U Ise: 0	92.168.64.1(35.25.29.74 7 0 729 A, G711 iff	Mu ming and	۵۹ Outgoing	%	C Summary			Receive		Transmit		
	Number 1 2 3 4 5 6 7 8 9 10 11 12 Trace	G Ref	State Idle Idle Idle Idle Idle Idle Idle Idl	State 03:12:05 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28		Call Deta	ails	or Dial	Cal	of Call	Trip De	. Jitter	Packet	Jitter	Packet	

The following screen shows an example inbound call where PSTN called Avaya IP Office Hunt Group "Service" (H.323 phone, Extn500).



8.2.2. System Monitor Application

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

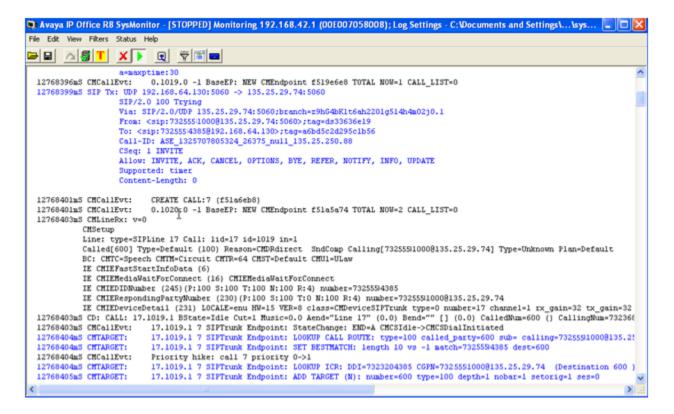


The Monitor will be active at startup. To pause the Monitor press the Pause III button.

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

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

🐘 Avaya IP Office R	3 SysMonitor - [STOPPED] Monitoring 192.168.42.1 (00E007058008); Log Settings - C:\Documents and Settings\\sys 📒	
File Edit View Filters	Status Help	
🛏 🖬 🔤 🔳		
	r: UDP 135,25,29,74:5060 -> 192,168,64,130:5060	~
	INVITE sip:73255543858192.168.64.130:5060 3IP/2.0	
	Via: SIP/2.0/UDP 135.25.29.74:5060;branch=z9hG4bK1t6ah2201g514h4a02j0.1	
	From: <sip:73255510008135.25.29.74:5060>;tag=ds33636e19</sip:73255510008135.25.29.74:5060>	
	To: <sip:732555143858192.168.64.130></sip:732555143858192.168.64.130>	
	Call-ID: ASE_1325707805324_26375_null_135.25.250.88	
	CSeq: 1 INVITE	
	Nax-Forwards: 66	
	Contact: <sip:73255510000135.25.29.74:5060;transport=udp></sip:73255510000135.25.29.74:5060;transport=udp>	
	Allow: INVITE,ACK,CANCEL,BYE,INFO,FRACK Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay, multipart/mixed	
	<pre>Accept: application/sdp, application/isup, application/dtmr, application/dtmr-relay, multipart/mixed P-Asserted-Identity: <sip:73255510000135.25.29.74:5060></sip:73255510000135.25.29.74:5060></pre>	
	Content-Length: 262	
	Content-Disposition: session; handling=required	
	Content-Type: application/sdp	
	v=0	
	o=Sonus UAC 19987 14698 IN IP4 135.25.29.74	
	s=SIP Media Capabilities	
	c=IN IP4 135.25.29.74	
	t=0 0	
	n=audio 24634 RTP/AVP 18 0 100	
	a=rtpmap:18 G729/8000	
	a=rtpmap:0 PCMU/8000	
	a=ttpaap:100 telephone-event/8000	
	a-fmtp:100 0-15	
	a-sendrecv	
12768396mS CMCal	a=maxptime:30 1Evt: 0.1019.0 -1 BaseEP: NEW CMEndpoint f519e6e8 TOTAL NOW=1 CALL LIST=0	
	12V: 0.1019.0 -1 BBSEFF: MLW CHENGPOINT ESISEES IOIAL MOWEL CALL_LIST=0 (: UDD 19.168.64.130:5060 -> 135.25.29.74:5060	
16700335m0 31F 11	: UP 192.100,04,100:000 -> 155.25.29.74:000 SIP/2.0.100 Trying	
	Via: 519/2.0/UPP 135.25.29.74:5060;branch=z9h04bKlt6ah2201g514h4m02j0.1	-
\$		- 1



The Monitor application allows the monitored information to be customized. To customize,

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

All Settings					×
T1		'PN	WAN) s	
ATM Call	DTE		Frame Relay GC		1 1110110100
ISDN Key/Lamp	Directory N	ledia PPP	R2 Routing	Services SIP	System
Events					
Sip High			STUN		
Packets					
SIP Reg/Op	it Bx	_	SIP Misc Bx		
SIP Reg/Op			SIP Misc Tx		
I Sir Regiop			SIF MISC 1X		
🔲 SIP Call Rx			Cm Notify Rx		
SIP Call Tx		Г	Cm Notify Tx		
	Sip Rx	🗖 hex	IP Filter (nnn.nnn	nnn.nnn)	
<u>ح</u> ا	Sip Tx	🗖 hex			
Default All	Clear All	Tab Clear A	II Tab Set All	ОК	Cancel
Save File	Load File	Select File			

8.3. Avaya Session Border Controller for Enterprise 6.2

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

8.3.1. System Information

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

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

Session Bord	er Controller for Enterprise AVAY	Ά
Dashboard	System Management	^
Administration		
Backup/Restore	Devices Updates SSL VPN Licensing	
System Management	Devices opulates 352 VFN Licensing	1
Global Parameters	Device Name Management Version Status	
Global Profiles	(Serial Number) IP	4
SIP Cluster	A-SBCE 192.168.1.20 6.2.0.Q30 Commissioned Reboot Shutdown Restart Application View Edit	
N Domain Policies		

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

	System	ı Inform	nation: A-SBCE)
General Configura Appliance Name Box Type Deployment Mode	A-SBCE SIP		Device Conf HA Mode Two Bypass	No	
Network Configura	tion		Netmask	Gateway	Interface
192.168.42.20	192.168.42.20	258	5.255.255.0	192.168.42.1	A1
192.168.64.130	192.168.64.130	255	5.255.255.0	192.168.64.254	B1
DNS Configuration	I		Managemen	t IP(s)	
Primary DNS	192.168.67.5		IP	192.168.1.20	
Secondary DNS					
DNS Location	DMZ				
DNS Client IP	192.168.64.130				

8.3.2. Avaya SBCE Protocol Traces

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

Step 1 - Navigate to **Device Specific Settings** \rightarrow **Troubleshooting** \rightarrow **Trace**

Step 2 - Select the Packet Capture tab and select the following:

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

Session Borde	er Controller	for Enterprise	AVAYA
SIP Cluster Domain Policies	Trace: A-SBCE		
TLS Management	Devices	Call Trace Packet Capture Captures	
Device Specific Settings	A-SBCE	P	acket Capture Configuration
Network Management		Status	Ready
Media Interface		Interface	Any 🗸
Signaling Interface			
Signaling Forking		Local Address IP[:Port]	All 😪 : 5060
End Point Flows		Remote Address	*
Session Flows		*, *:Port, IP, IP:Port	
Relay Services		Protocol	All 🔽
SNMP		Maximum Number of Packets to Capture	5000
Syslog Management			3000
Advanced Options		Capture Filename Using the name of an existing capture will overwrite it.	test1.pcap
 Troubleshooting 			
Debugging			Start Capture Clear
Trace		L	

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

Trace: A-SBCE		
Devices	Call Trace Packet Capture Captures	
A-SBCE		s page will automatically refresh until the capture completes.
	Pa	cket Capture Configuration
	Status	In Progress
	Interface	Any 🗸
	Local Address IP[:Port]	All v : 5060
	Remote Address *, *:Port, IP, IP:Port	
	Protocol	All
	Maximum Number of Packets to Capture	5000
	Capture Filename Using the name of an existing capture will overwrite it.	test1.pcap
		Stop Capture

- **Step 3** Run the test.
- Step 4 Select Stop Capture button shown above.
- Step 5 Click on the Captures tab and the packet capture is listed as a .pcap file with the date and time added to filename specified in Step 2 (e.g., test1). Note that the system will append date/time information to the filename.

Trace: A-SBCE				
Devices	Call Trace Packet Capture Captures			
A-SBCE			(Refresh
	File Name	File Size (bytes)	Last Modified	
	test1_20130102131059.pcap	0	January 2, 2013 1:11:18 PM GMT	Delete

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

8.4. Protocol Trace Examples

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

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

Til.		18 C-	Contrary Analysis Chattation	Talaabaass Taala Uala		
Eile	<u>E</u> dit	: <u>V</u> iew <u>G</u> o	Capture Analyze Statistics	Telephon <u>y T</u> ools <u>H</u> elp		
	1		(🖻 🖪 🗙 🎜 🗄	@ @ @ 주 쏘		Q. Q. 🖭 🚟 🖾 🎭 💢
Filte	er: sip)		-	Expression Clear	Apply
No.		Time	Source	Destination	Protocol	Info
	25	18.493	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:0000011051@192.168.64.130:5060, with
	26	18.495	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
	- 27	18.573	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
	168	20.562	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
	170	20.572	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
	178	20.672	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
	433	24.398	192.168.64.130	135.25.29.74	SIP	Request: INVITE sip:73255524380 35.25.29.74:5060;transpor
	436	24.433	135.25.29.74	192.168.64.130	SIP	Status: 100 Trying
	441	24.484	135.25.29.74	192.168.64.130	SIP/SDP	Status: 200 OK, with session description
	442	24.495	192.168.64.130	135.25.29.74	SIP/SDP	Request: ACK sip:7325552438@135.25.29.74:5060;transport=u
<					Ш	

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

Eile	<u>E</u> dit <u>V</u> iew <u>G</u> o	<u>Capture</u> <u>Analyze</u> <u>Statistic</u>	s Telephon <u>y T</u> ools <u>H</u> el	p		
		🖻 🔏 🗶 🎜 🔺	🔍 🍬 🔿 🐴		Q. Q. 📅 👹 🕅 畅 💥 💢	
Filte	r: rtpevent			▼ Expression Clea	r Apply	
No.	Time	Source	Destination	Protocol	Info	~
	766 29.236	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Star *	
	768 29.265	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Star *	
	770 29.295	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Star *	
	772 29.316	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Star * (end)	
	773 29.316	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Star * (end)	
	774 29.316	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Star * (end)	
	782 29.455	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Eight 8	
	783 29.455	192.168.64.130	135.25.29.74	RTP EVENT	Payloa type=RTP Event, DTMF Eight 8	
	784 29.455	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Eight 8	
	786 29.486	192.168.64.130	135.25.29.74	RTP EVENT	Payload type=RTP Event, DTMF Eight 8	~
<						>

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

Eile	Edit	<u>V</u> iew <u>G</u> o <u>⊂</u>	apture <u>A</u> nalyze <u>S</u> tatistics	Telephony <u>T</u> ools <u>H</u> elp		
	i		🖻 🟅 🗶 🚨 🛛	🔍 🗢 🛸 🍄 🖉 🕹		Q. Q. 🖾 👪 🖄 🍢 💢
Filter	: rtp	I		•	Expression Clear	r Apply
No.		Time	Source	Destination	Protocol	Info
	901	31.115	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x58A31231, Seq=83, Time=19920
	902	31.141	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0x34841BB4, Seq=416, Time=99840
	903	31.146	192.168,64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x58A31231, Seq=84, Time=20160
	904	31.171	135.25.39.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0x34841BB4, Seq=417, Time=100080
	905	31.176	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x58A31231, Seq=85, Time=20400
	906	31.201	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0x34841BB4, Seq=418, Time=100320
	907	31.206	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x58A31231, Seq=86, Time=20640
	908	31.231	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0x34841BB4, Seq=419, Time=100560
	909	31.236	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x58A31231, Seq=87, Time=20880
	910	31.261	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729. SSRC=0x34841BB4. Sea=420. Time=100800
<						>

JF; Reviewed: SPOC 3/15/2013 Solution & Interoperability Test Lab Application Notes ©2013 Avaya Inc. All Rights Reserved. 74 of 88 IPO81SBCE62TF A trace of an idle line can verify that your SIP Trunk from the Avaya SBCE B1 interface (192.168.64.130) to the IPTF Service border element (135.25.29.74) is up and communicating with SIP *OPTIONS* messages and response messages. A SIP *405 Method Not Allowed* response is normal for the Avaya SBCE to AT&T test environment.

		🖹 🖬 🗶 🎜 🖴	🔍 🗢 🔹 🖓 🚡		€, 0, 0, 17 🗸 🕅 畅 % 🐹
Filte	r: sip			 Expression 	Clear Apply
No.	Time	Source	Destination	Protocol	Info
	9 6.776	135.25.29.74	192.168.64.130	SIP	Request: OPTIONS sip:192.168.64.130:5060
	10 6.781	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
	29 23.276	192.168.64.130	135.25.29.74	SIP	Request: OPTIONS sip:135.25.29.74;transport=udp
	30 23.304	135.25.29.74	192.168.64.130	SIP	Status: 405 Method Not Allowed

9. Conclusion

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

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

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

10. References

Avaya:

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

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

- [2] Avaya IP Office R8.1 Manager, 10.115-601011 Issue 290 (03 August 2012)
- [3] Avaya IP Office System Monitor, Document Number 15-601019

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

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

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

- [7] Installing Avaya Session Border Controller For Enterprise, Release 6.2, Issue 1, January 2013.
- [8] Administering Avaya Session Border Controller, Release 6.2, Issue 1, January 2012

AT&T IPTF Service:

[9] AT&T IP Toll Free Service description -

http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voipenterprise/ip-toll-free-enterprise//

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

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

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

Step 1: Secondary Location in Server Configuration

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

Server Configuration: ATT_Secondary				
Add			Rename Clone Delete	
Server Profiles	General Authentication	Heartbeat Advanced		
Avaya_SC	Server Type	Trunk Server		
ATT_SC	IP Addresses / FQDNs	135.25.29.75		
ATT_Secon	Supported Transports	UDP		
	UDP Port	5060		
		Edit		

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

Server Configuration: ATT_Secondary				
Add		Rename Clone Delete		
Server Profiles	General Authentication	Heartbeat Advanced		
Avaya_SC	Enable Heartbeat			
ATT_SC	Method	OPTIONS		
ATT_Secon	Frequency	60 seconds		
	From URI	secondary@customerb.com		
	To URI	secondary@customerb.com		
		Edit		

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

Server Configu	ration: ATT_SC		
Add			Rename Clone Delete
Server Profiles	General Authentication	Heartbeat Advanced]
Avaya_SC	Enable Heartbeat		
ATT_SC	Method	OPTIONS	
ATT_Secondary	Frequency	60 seconds	
	From URI	primary@customerb.com	
	To URI	primary@customerb.com	
		. ,.	
		Edit	

Step 2: Add Secondary IP Address to Routing

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

	Edit Routing Rule X				
Each URI group may only be used onc	Each URI group may only be used once per Routing Profile.				
	Next Hop Routing				
URI Group	*				
Next Hop Server 1 IP, IP:Port, Domain, or Domain:Port	135.25.29.74				
Next Hop Server 2 IP, IP:Port, Domain, or Domain:Port	135.25.29.75				
Routing Priority based on Next Hop Server					
Use Next Hop for In Dialog Messages					
Ignore Route Header for Messages Outside Dialog					
NAPTR					
SRV					
Outgoing Transport	○ TLS ○ TCP				
	Finish				

Step 3: Configure End Point Flows – ATT_Secondary

- 1. Select Device Specific Settings from the menu on the left-hand side
- 2. Select Endpoint Flows
- 3. Select the **Server Flows** Tab
- 4. Select Add Flow
 - a) Name: ATT_Secondary
 - b) Server Configuration: ATT_Secondary
 - c) URI Group: *
 - d) Transport: *
 - e) Remote Subnet: *
 - f) Received Interface: Avaya-IPO
 - g) Signaling Interface: ATT
 - h) Media Interface: ATT
 - i) End Point Policy Group: defaultLowATT
 - j) Routing Profile: Avaya_R
 - k) Topology Hiding Profile: ATT_TH
 - 1) File Transfer Profile: None
- 5. Click Finish

Edit	Flow: ATT_Secondary X
Flow Name	ATT_Secondary
Server Configuration	ATT_Secondary 💙
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Avaya-IPO 💌
Signaling Interface	ATT 💌
Media Interface	ATT 💌
End Point Policy Group	defaultLowATT 💌
Routing Profile	Avaya_R 💌
Topology Hiding Profile	ATT_TH
File Transfer Profile	None 💌
	Finish

When completed the Avaya SBCE will issue OPTIONS messages to the primary (135.25.29.74) and secondary (135.25.29.75) border elements.

12. Appendix: Avaya IP Office 8.1 SIP Line Template

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

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

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

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

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

Run	? 🗙
-	Type the name of a program, folder, document, or Internet resource, and Windows will open it for you.
Open:	regedit 💌
	OK Cancel Browse

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

Edit View Favorites Help		[
My Computer HKEY_CLASSES_ROOT	<u>^</u>	Name	Туре	Data
		201 MAVTOOLBARX	REG_DWORD	0x0000010a (266)
AppEvents		201 MAVTOOLBARY	REG_DWORD	0×00000000 (0)
		2010 NonThreadedTCP	REG_DWORD	0×00000000 (0)
		- 2010 Password Required For Save	REG_DWORD	0x00000001 (1)
Control Panel		not the second s	REG_DWORD	0x00000000 (0)
CounterPath			REG_DWORD	0x00000001 (1)
			REG_SZ	
🖬 🧰 Identities			REG_DWORD	0×00000000 (0)
🗉 🦲 Keyboard Layout		B SecureCommunications	REG_DWORD	0x00000001 (1)
📄 Network		B SecurityLevel	REG_DWORD	0x00000001 (1)
🗊 🧰 Printers		B ServicesBaseHTTPPort	REG_DWORD	0x00000050 (80)
🕀 🧰 Remote		BervicesBasePort	REG_DWORD	0x0000c674 (50804)
SessionInformation		🕮 SetRingDelayPerAp	REG_DWORD	0×00000000 (0)
😑 🧰 Software			REG_DWORD	0x00000000 (0)
😟 🧰 Adobe		B SHOWErrorPane	REG_DWORD	0x00000000 (0)
😟 🦲 Alps		B SHOWInGroups	REG_DWORD	0×00000000 (0)
😑 🧱 Avaya		3 SHOWMAINToolbar	REG_DWORD	0x00000001 (1)
🗈 🦲 2050 IP Softphone		B SHOWNAVIGATIONPane	REG_DWORD	0x00000001 (1)
🖻 🦲 Avaya IP Softphone			REG_DWORD	0x00000001 (1)
🗈 🦲 Avaya one-X Agent		B ShowPLDSVirtualLicences	REG DWORD	0x00000000 (0)
Avaya one-X AgentAVC			REG_DWORD	0×00000000 (0)
Avaya one-X AgentAVCClient			REG_DWORD	0x00000001 (1)
Avaya one-X Communicator Avaya one-X® Communicator		B SHOWSIMPLIFIEDVIEWASDEFAULTVIEW	REG DWORD	0×00000000 (0)
Avaya one-xig communicator Avaya SIP Softphone		SSLRemoteAccess	REG DWORD	0×00000000 (0)
Avaya Site Administration		B STARTINITIALDISCOVERY	REG DWORD	0×00000001 (1)
			REG_DWORD	0×00000001 (1)
Integrator for Outlook		a) TCPSearchCriteria	REG SZ	
			REG DWORD	0×00000001 (1)
		TemplateProvisioning	REG DWORD	0×00000001 (1)
😑 🔄 Manager			REG_DWORD	0×00000001 (1)
🕀 🦲 Column Headings			REG_DWORD	0×00000001 (1)
RecentlyUsedFiles		a)UpgDir	REG SZ	C:\Program Files\Ava
Phone Manager		WalidateConfigOnLoad	REG_DWORD	0x00000001 (1)
🛄 UpgradeWizard		ValidateConfigOnOK	REG_DWORD	0×00000001 (1)
😟 🧰 Softphone		and the second of the second o	Neg_pmone	0,0000001(1)
🔄 🧰 Voicemail Pro	~	<		

Edit DWORD Value	? 🗙
Value name: TemplateProvisioning	
Value data:	Base
1	 Hexadecimal Decimal
	OK Cancel

Reboot the computer.

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

🔟 IP Office Manager Preference	:es		? 🗙
Preferences Directories Discovery	Visual Preferences	Security	Validation
Icon Size Small			
Multiline Tabs Enable Template Option	ns		
i	ок Са	ancel	Help

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

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

🐮 SIP Trunk Template - (SI	P Trunk - 17)				X
Please review and change the trunk settings if you want –					
SIP Line Transport VoIP T3	38 Fax SIP Credentials				
Descriptive Name	ATT	Use Tel URI			
ITSP Domain Name	Not Used	Check OOS	~		
Send Caller ID	Diversion Header 🛛 🗸	Call Routing Method	Request URI 🗸 🗸		
Association Method	By Source IP address 🗸 🗸	Originator number for forwarded and twinning calls]	
		Name Priority	Favor Trunk 💌]	
Incoming	Auto				
Outgoing	Auto				
UPDATE Supported	Never	Caller ID from From header			
User-Agent and Server Headers		Send From In Clear			
				Export	Cancel

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

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

Template Type Selection							
Locale	United States (US English)	·					
Country	United States	•					
Service Provider	AT&T	•					
	Generate Template	Cancel					

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

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

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

<DNSServerTwo>0.0.0.0</DNSServerTwo>

<CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>

<SeparateRegistrar />

<CompressionMode>**AUTOSELECT**</CompressionMode>

<UseAdvVoiceCodecPrefs>true</UseAdvVoiceCodecPrefs>

<AdvCodecPref>G.711 ULAW 64K,G.729(a) 8K CS-ACELP</AdvCodecPref>

<CallInitiationTimeout>4</CallInitiationTimeout>

<DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>

<VoipSilenceSupression>false</VoipSilenceSupression>

<ReinviteSupported>true</ReinviteSupported>

<FaxTransportSupport>FOIP_T38</FaxTransportSupport>

<UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>

<CodecLockdown>false</CodecLockdown>

<Rel100Supported>**true**</Rel100Supported>

<T38FaxVersion>3</T38FaxVersion>

<Transport>**UDPTL**</Transport>

<LowSpeed>**0**</LowSpeed>

<HighSpeed>**0**</HighSpeed>

<TCFMethod>Trans_TCF</TCFMethod>

<MaxBitRate>FaxRate_14400</MaxBitRate>

```
<EflagStartTimer>2600</EflagStartTimer>
```

```
<EflagStopTimer>2300</EflagStopTimer>
```

<UseDefaultValues>false</UseDefaultValues>

<ScanLineFixup>**true**</ScanLineFixup>

<TFOPEnhancement>**true**</TFOPEnhancement>

<DisableT30ECM>false</DisableT30ECM>

<DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>

<DisableT30MRCompression>false</DisableT30MRCompression>

<NSFOverride>false</NSFOverride>

</Template>

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

Lin Line Type		Line Sub	SIP Line Transport Si	IP URI VoIP T38 Fax SIP C	reden	ntials		
f위1 Analogue T f위2 Analogue T f위3 Analogue T	runk		Line Number	17				
113 Analogue T 114 Analogue T			ITSP Domain Name				In Service	>
🔪 17 - SIP Line 💡								
	<u>1</u>	New		•		H323	} Line	Ĺ
	•	Generate	SIP Trunk Template				ECT Line	
	X	Cut		Ctrl+X		SIP L		e
	Ca l	Сору		Ctrl+C		New	SIP Trunk From Template	
		Paste		Ctrl+V			Name Priority	Fai
	\boldsymbol{x}	Delete		Ctrl+Del			Caller ID from From header	
	1	Validate					Send From In Clear	
	₹2	Connect T	·o	Ctrl+T			User-Agent and Server Headers	
		New from	Template (Binary)			~		
		Export as	Template (Binary)			~		
		Show In G	iroups					
		Customize	Columns				~	

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

🐮 Template Ty		
Locale	United States (US English)	×
Country	United States	~
Service Provider	AT&T	🔽 🔲 Display All
	Create new SIP Tr	unk Cancel

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