



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking Using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 8 with Avaya Session Border Controller for Enterprise Release 6.2 – Issue 1.0

Abstract

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Contact Center VoIP Inbound SIP Trunk Service and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya Session Border Controller Release 6.2, an Avaya IP Office 500 v2 Release 8.1 Essential Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, SIP, digital, and analog endpoints.

The Verizon Business IP Contact Center VoIP Inbound offer referenced within these Application Notes enables a business to receive inbound toll free calls via standards-based SIP trunks, without the need for additional TDM enterprise gateways or TDM cards and the associated maintenance costs.

IP Office Release 8.1 with Avaya Session Border Controller for Enterprise Release 6.2 has not been independently certified by Verizon labs. These Application Notes can be used to facilitate customer engagements via the Verizon field trial process, pending Verizon labs independent certification.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted in the Avaya Solution & Interoperability Test Lab, utilizing a Verizon Business Private IP (PIP) circuit connection to the Verizon Business IP Contact Center service.

Table of Contents

Table of Contents	2
1. Introduction	4
2. General Test Approach and Test Results	4
2.1. Interoperability Compliance Testing	4
2.2. Test Results	6
2.3. Support	7
2.3.1. Avaya	7
2.3.2. Verizon	7
3. Reference Configuration	8
4. Equipment and Software Validated	10
5. Avaya IP Office Configuration	10
5.1. Physical, Network, and Security Configuration	11
5.2. Licensing	13
5.3. System Settings	15
5.3.1. System Tab	15
5.3.2. LAN Settings	16
5.3.3. Voicemail	19
5.3.4. System Telephony Configuration	20
5.3.5. System Twinning Configuration	21
5.3.6. System Codecs Configuration	22
5.4. SIP Line	22
5.4.1. SIP Line - SIP Line Tab	22
5.4.2. SIP Line - Transport Tab	24
5.4.3. SIP Line - SIP URI Tab	25
5.4.4. SIP Line - VoIP Tab	27
5.5. Users, Extensions, and Hunt Groups	28
5.5.1. Digital User 232	28
5.5.2. Avaya Flare Experience User 234 with IP Office Softphone Privileges	31
5.5.3. Hunt Groups	35
5.6. Short Codes	37
5.7. Incoming Call Routes	39
5.7.1. Incoming Call Route to a Specific Telephone Extension	39
5.7.2. Incoming Call Routes to a Hunt Group by Dialed Toll-Free Number	40
5.7.3. Incoming Call Routes Based on Calling Party Number	41
5.7.4. Incoming Call Routes to Various IP Office Features	42
5.8. Saving Configuration Changes to IP Office	43
6. Configure Avaya Session Border Controller for Enterprise	44
6.1. Network Management	46
6.2. Routing Profile	47
6.3. Server Interworking Profile	48
6.3.1. Server Interworking Profile – IP Office	49
6.3.2. Server Interworking Profile – Verizon IPCC	50
6.4. Server Configuration	52

6.4.1.	Server Configuration – IP Office	53
6.4.2.	Server Configuration - Verizon	55
6.5.	Media Rule	57
6.6.	Signaling Rule	58
6.7.	Application Rule	59
6.8.	Endpoint Policy Groups	60
6.9.	Media Interface	61
6.10.	Signaling Interface.....	62
6.11.	End Point Flows - Server Flow.....	63
7.	Verizon Business Configuration	66
8.	Verification	67
8.1.	Illustration of OPTIONS Handling	67
8.2.	Avaya SBCE	69
8.2.1.	Incidents	69
8.2.2.	Tracing	70
8.3.	IP Office	72
8.3.1.	System Status	72
8.3.2.	System Monitor.....	74
9.	Conclusion	76
10.	Additional References.....	76
11.	Appendix A: SIP Line Template.....	78

1. Introduction

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Contact Center VoIP Inbound Service and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya Session Border Controller for Enterprise Release 6.2, and Avaya IP Office 500 v2 Release 8.1 Preferred Edition, Avaya Voicemail Pro, one-X® Portal, Avaya Flare® Experience for Windows, Avaya IP Office Softphone, and Avaya H.323, SIP, digital, and analog endpoints.

IP Office Release 8.1 with Avaya Session Border Controller for Enterprise Release 6.2 has not been independently certified by Verizon labs. These Application Notes can be used to facilitate customer engagements via the Verizon field trial process, pending Verizon labs independent certification.

In the sample configuration, An Avaya Session Border Controller for Enterprise (SBCE) is used as an edge device between the Avaya IP Office and Verizon business. The Avaya SBCE performs SIP header manipulation and provides topology hiding.

Customers using Avaya IP Office with the Verizon Business IP Contact Center SIP Trunk service are able to receive inbound toll-free calls from the PSTN via the SIP protocol. The converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI.

Verizon Business IP Contact Center service can be delivered to the customer premise via either a Private IP (PIP) or Internet Dedicated Access (IDA) IP network terminations. Although the configuration documented in these Application Notes used Verizon's IP Contact Center service terminated via a PIP network connection, the solution validated in this document also applies to IP Contact Center services delivered via IDA service terminations.

For more information on the Verizon Business IP Contact Center service, visit <http://www.verizonbusiness.com/Products/communications/contact-center/>

2. General Test Approach and Test Results

The Avaya IP Office location was connected to the Verizon Business IPCC Service, as depicted in **Figure 1**. The Avaya SBCE and IP Office were configured to use the commercially available IP Toll Free VoIP Inbound solution. This allowed Avaya IP Office to receive inbound toll-free calls from the PSTN via the SIP protocol.

2.1. Interoperability Compliance Testing

The testing included executing the test cases detailed in Reference [VZ-Test-Plan], which contains the Verizon IPCC Interoperability Lab Test Plan. To summarize, the testing included the following successful SIP trunk interoperability compliance testing:

- SIP OPTIONS monitoring of the health of the SIP trunk was verified. Verizon Business, Avaya SBCE, and IP Office can all monitor health using SIP OPTIONS.

- Proper recovery from induced failure conditions such as IP Office reboots, and IP network outages between Verizon and IP Office, of short and long durations.
- Incoming calls from the PSTN were routed to the toll-free numbers assigned by Verizon Business to the Avaya IP Office location. These incoming calls arrived via the SIP Line configured in Section 5.4 and were answered by Avaya H.323 telephones, Avaya SIP telephones, Avaya digital telephones, analog telephones, Avaya IP Office Softphone, Avaya Flare® Experience for Windows, and Avaya IP Office Voicemail Pro.
- Proper disconnect when either party hangs up an active call.
- Proper disconnect when the PSTN caller abandons (i.e., hangs up) a toll-free call before the IP Office party has answered.
- Proper SIP 486 response and busy tone heard by the caller when a PSTN user calls a toll-free number directed to a busy IP Office user, an IP Office user with Do-not-disturb active, or an IP Office user that is logged out (i.e., assuming no redirection is configured for these conditions). Similarly, busy tone is heard when a PSTN user calls a toll-free number whose “SIP URI Max Calls per Channel” has been reached (see Section 5.4). Similarly, busy tone is heard when a PSTN user calls a toll-free number directed to a hunt group whose queue is “full” (i.e. if no redirection is configured for hunt group busy conditions, see Section 5.5.4).
- Proper termination of an inbound IP Toll Free call left in a ringing state for a relatively long duration.
- The display of caller ID on display-equipped Avaya IP Office telephones was verified. The IP Office capability to use the caller ID received from Verizon to look up and display a name from a configurable directory was also exercised successfully.
- Privacy requests for inbound toll-free calls from the PSTN were verified. That is, when privacy is requested by a PSTN caller (e.g., dialing *67), the inbound toll-free call can be successfully completed to an IP Office telephone user while presenting a “WITHHELD” or anonymous display to an IP Office user (i.e., rather than the caller’s telephone number).
- Inbound toll-free long holding time call stability (See Section 2.2. Although long SIP sessions are not refreshed, the media paths remain connected.)
- IP Office complies with RFC 3261 SIP Methods.
- Avaya SBCE can use UDP for SIP transport with Verizon IPCC.
- Avaya SBCE can use a configured UDP port for SIP signaling with Verizon.
- IP Office accepts the full SIP headers sent by Verizon IPCC.
- IP Office sends SIP 180 RINGING (no SDP in 180) for inbound calls and ring back tone is heard by the caller.
- IP Office does not return a SIP 302 to Verizon.
- Telephony features such as hold and resume, transfer of toll-free calls to other IP Office users, and conference of toll-free calls.
- Incoming voice calls using the G.729(a) and G.711 ULAW codecs, and proper protocol procedures related to media.
- DTMF transmission using RFC 2833. Successful IP Office Voicemail Pro menu navigation for incoming toll-free calls. Successful use of IP Office Mobile Call Control, where DTMF sequences can be performed remotely using the SIP Line.

- Incoming toll-free calls directed to the Hunt Groups configured in Section 5.5.3 were verified. Incoming calls could be queued, queued with priority, and be answered by members of the hunt group as members become available.
- Outgoing calls from the Avaya IP Office location to the PSTN were routed via a SIP Line to the Verizon Business IP Trunk service described in reference [VZBIPT-IPO81SBC]. As detailed in reference [VZBIPT-IPO81SBC], these outgoing PSTN calls can be originated from Avaya H.323 telephones, Avaya SIP telephones, Avaya digital telephones, analog endpoints, and Avaya IP Office Softphone. The display of caller ID on display-equipped PSTN telephones was verified. In the context of inbound toll-free calls using Verizon IP Contact Center, inbound toll-free calls arriving via the SIP Line configured in Section 5.4 could be forwarded or twinned out the Verizon IP Trunk Service SIP Line. Inbound toll-free calls from the Verizon IP Contact Center SIP Line could also trigger mobile callback calls that use the Verizon IP Trunk Service SIP Line.
- Call Forwarding of Verizon toll-free calls to PSTN destinations via the Verizon IP Trunk service documented in reference [VZBIPT-IPO81SBC], presenting true calling party information to the mobile phone. See Section 2.2 for additional information.
- Mobile twinning of Verizon toll-free calls to a mobile phone via the Verizon IP Trunk service documented in reference [VZBIPT-IPO81SBC], presenting true calling party information to the mobile phone.
- Inbound mobile call control, mapping a Verizon toll-free number to the mobile call control feature, as shown in Section 5.6. That is, a configured mobile twinning PSTN caller may dial a Verizon toll-free number, receive dial tone from IP Office, and place calls using IP Office, as if the user were calling from their IP Office telephone. Calls to the same toll-free number from calling numbers that are not configured in IP Office for mobile call control receive busy tone.
- Proper DiffServ markings for Avaya SBCE SIP signaling and RTP media

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results. The following observations may be noteworthy:

1. The Verizon IPCC Service does not support fax.
2. When a call is put on hold by an IP Office user, there is no indication sent to Verizon via SIP messaging. This is transparent to the users on the call.
3. Although the Verizon Business IP Contact Center service supports transfer using the SIP REFER method and IP Office supports sending REFER, IP Office did not send REFER to Verizon in the verified configuration.
4. During compliance testing, one Avaya SBCE was used to support both Verizon IPCC service for inbound toll free calls and Verizon IP Trunk service for outbound calls. One SIP line was created on IP Office to connect the Avaya SBCE for both services. When an inbound Verizon IPCC call to an IP Office user is forwarded to a PSTN number using Verizon IP Trunk service through the same IP Office SIP Line with REFER Support

activated, IP Office sent a REFER to Verizon IPCC Service after the call was answered. Verizon IPCC Service responded with a 202 Accepted message followed by 603 Server Internal Error in the NOTIFY message. This caused the call to drop. To prevent IP Office from sending a REFER to Verizon IPCC Service in this scenario a signaling rule was created in the Avaya SBCE, as shown in Section 6.6.

5. The SIP protocol allows sessions to be refreshed for calls that remain active for some time. In the tested configuration, neither Verizon nor IP Office send SIP re-INVITE or UPDATE messages to refresh a session. In the tested configuration, this is transparent to the users that are party to the call in that the media paths remain established.
6. When a user on the PSTN hangs up an active call, Verizon IPCC will send an INVITE with SDP containing 0.0.0.0 before sending the BYE to clear the call. IP Office processes the INVITE with SDP containing 0.0.0.0 as a request to hold the call, and then processes the BYE to disconnect the call. If the IP Office user is still listening after the PSTN user hangs up, the IP Office user may very briefly hear music on hold from IP Office before the BYE is processed and the call appearance is idled.
7. IP Office does not support the receipt of an initial INVITE that does not contain SDP. Therefore, IP Office does not support the Verizon IP Contact Center “enhanced transfer” service, which sends an initial INVITE without SDP to the transfer-to site of an enhanced transfer.

2.3. Support

2.3.1. Avaya

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

2.3.2. Verizon

For technical support on Verizon Business IP Contact Center service, visit online support at <http://www.verizonbusiness.com/us/customer/>

3. Reference Configuration

Figure 1 illustrates an example Avaya IP Office solution connected to the Verizon Business IP Contact Center SIP Trunk service. The Avaya equipment is located on a private IP subnet. An enterprise edge router provides access to the Verizon Business network via a Verizon Business T1 circuit. This circuit is provisioned for the Verizon Business Private IP (PIP) service. Reference [VZBIPT-IPO81SBC] illustrates IP Office and Avaya SBCE interoperability with the Verizon Business IP Trunk service. In the verification testing associated with these Application Notes, both the Verizon IP Trunk service and the Verizon IP Contact Center service were accessible via the same PIP connection.

In the sample configuration, the Avaya SBCE receives traffic from the Verizon Business IP Contact Center service on port 5060 and sends traffic to port 5072, using UDP for network transport, as required by the Verizon Business IP Contact Center service. The Avaya SBCE in turn sends and receives traffic to and from IP Office using TCP port 5060. Verizon provided five toll-free numbers associated with the IP Contact Center service. These toll-free numbers were mapped to IP Office destinations via Incoming Call Routes as summarized in **Table 1**. The Avaya IP Office environment domain known to Verizon was *adevc.avaya.globalipcom.com*.

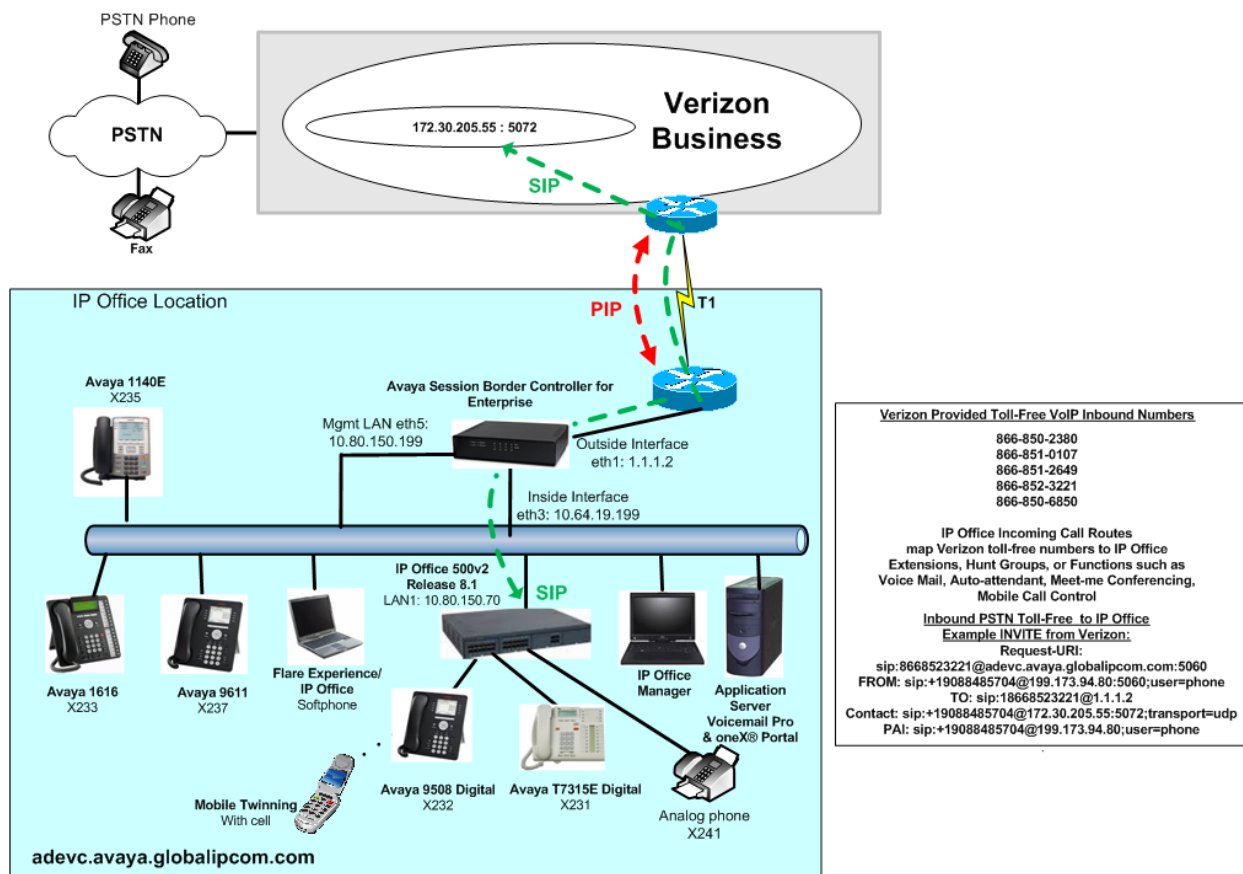


Figure 1: Avaya IP Office with Verizon IP Contact Center Service

Table 1 shows an example mapping of toll-free numbers to IP Office users, groups, or functions. The associated IP Office configuration is shown in Section 5. Since the quantity of toll-free numbers was limited in the test configuration relative to the desired test coverage, the same toll-free number was routed to different IP Office destinations (i.e., IP Office configuration changes were made to the Incoming Call Route destination as needed between successive tests).

Verizon Provided Toll-Free Number	Configured Avaya IP Office Destination(s)	Notes
866-851-0107	X235	Avaya 1140E
866-850-2380	X232, x241, x234	Digital Telephone with Mobile Twinning and Mobile Call Control permission. Also used to test analog telephone and Avaya Flare Experience capabilities.
866-851-2649	X233, x237	Avaya 1616 Telephone, Avaya 9611 Telephone
866-850-6850	Voicemail Collect on Voicemail Pro	Allow external callers to access voice mail toll-free
866-850-6850	Inbound Mobile Call Control	Allow toll-free calls from pre-configured twinning numbers to access mobile call control
866-850-6850	Conference Bridge on Voicemail Pro	Allow external callers to access conference bridge toll-free
866-852-3221 (any caller)	“401 Sales” Hunt Group (with default priority)	Hunt Group with queuing
866-852-3221 (specific callers)	“400 Overdue Account” Hunt Group	Show IP Office destination selection based on caller ID
866-852-3221 (specific priority callers)	“401 Sales” Hunt Group (with High Priority)	Show IP Office priority queuing based on caller ID

Table 1: Example Verizon Toll Free Number to IP Office Destination Mappings

4. Equipment and Software Validated

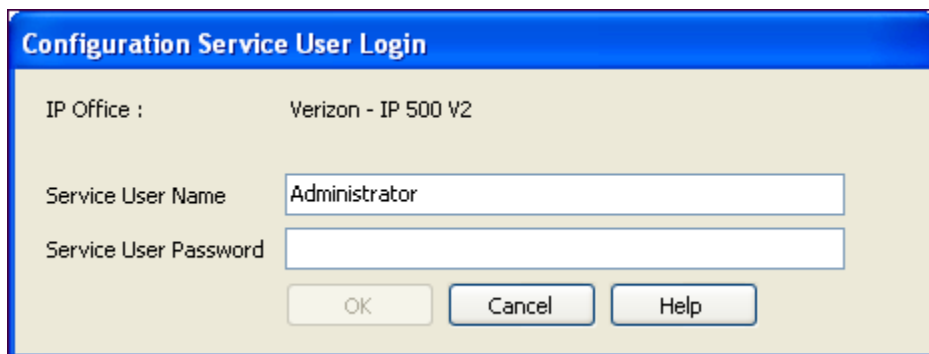
Table 2 shows the equipment and software used in the sample configuration.

Equipment	Software
Avaya Session Border Controller for Enterprise on the Portwell CAD-0208 platform	Release 6.2 (Q33)
Avaya IP Office 500 v2	Release 8.1 (65)
Avaya IP Office Manager	Release 10.1 (65)
Avaya Application Server (Voicemail Pro and IP Office one-X® Portal)	8.1.20-3
Avaya 2500 Analog Telephone	N/A
Avaya 9508 Digital Telephone	N/A
Avaya T7315E Digital Telephone	N/A
Avaya 1616 IP Telephone (H.323)	Release 1.302B
Avaya 9611 IP Telephone (H.323)	Release 6.2209
Avaya 1140E SIP	04.03.12
Avaya IP Office Softphone	Release 3.2.3.20 64770
Avaya Flare® Experience	1.1.0.5

Table 2: Equipment and Software Tested

5. Avaya IP Office Configuration

IP Office is configured via the IP Office Manager program. For more information on IP Office Manager, consult reference [2]. From the IP Office Manager PC, select **Start → Programs → IP Office → Manager** to launch the Manager application. A screen that includes the following in the center may be displayed:



Log in with the appropriate configuration credentials. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu

was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side.

5.1. Physical, Network, and Security Configuration

This section describes attributes of the sample configuration, but is not meant to be prescriptive. Consult reference [1] for more information on the topics in this section.

In the sample configuration, looking at the IP Office 500 from left to right, the first module is a TCM 8 Digital Station Module. This module supports Avaya T-Series and M-Series telephones. The second module is a COMBO6210/ATM4 module. This module is used to add a combination of ports to an IP500 V2 control unit and is not supported by IP500 control units. The module supports 10 voice compression channels. Codec support is G.711, G729a and G.723 with 64ms echo cancellation. G.722 is supported by IP Office Release 8.0 and higher. The “Combo” card will support 6 Digital Station ports for digital stations in slots 1-6 (except 3800, 4100, 4400, 7400, M and T-Series), 2 Analog Extension ports in slots 7-8, and 4 Analog Trunk ports in slots 9-12. Referring to **Figure 1**, the Avaya T7315E telephone with extension 231 is connected to port 1 of the TCM8 module, and the Avaya 9508 telephone with extension 232 is connected to port 1 of the “Combo” card. The analog extension or fax machine is connected to the “Combo” card on port 7

The following screen shows the modules in the IP Office used in the sample configuration. To access such a screen, select **Control Unit** in the Navigation pane. The modules appear in the Group pane. 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.

IP Offices		Control Unit		IP 500 V2	
<ul style="list-style-type: none"> BOOTP (6) Operator (3) Verizon System (1) Line (6) Control Unit (3) Extension (23) User (24) HuntGroup (3) Short Code (67) Service (0) RAS (1) Incoming Call Route (4) WebExt (0) 		Dev No.	Dev Type	Version	Unit
		1	IP 500 V2	8.1 (65)	Device Number
		2	TCM8	8.1 (65)	Unit Type
		3	COMBO6210/ATM4	8.1 (65)	Version
					Serial Number
					Unit IP Address
					Interconnect Number
					Module Number

In the sample configuration, the IP Office LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is 10.80.150.1. The Avaya SBCE resides on a different subnet and requires an IP Route to allow SIP traffic between the two devices. To add an IP Route in 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 Details pane with the relevant route using **Destination** LAN1.

10.64.0.0	
IP Route	
IP Address	10 . 64 . 0 . 0
IP Mask	255 . 255 . 0 . 0
Gateway IP Address	10 . 80 . 150 . 1
Destination	LAN1
Metric	0
<input type="checkbox"/> Proxy ARP	

To facilitate use of Avaya IP Office Softphone, https was enabled in the sample configuration. To check whether https is enabled, navigate to **File → Advanced → Security Settings**. A screen such as the following is presented. Log in with the appropriate credentials.

Security Service User Login	
IP Office :	VerizonIPCC-SBC - IP 500 V2
Service User Name	security
Service User Password
<input type="button" value="OK"/> <input type="button" value="Cancel"/> <input type="button" value="Help"/>	

After logging in, select **Services** from the Navigation pane and **HTTP** from the Group pane. In the Details pane, verify the **Service Security Level** is configured as intended, as shown below.

Security Settings	Services (6)	Service : HTTP																								
<ul style="list-style-type: none"> Security <ul style="list-style-type: none"> General System (1) Services (6) Rights Groups (15) Service Users (8) 	<table border="1"> <thead> <tr> <th>Name</th> <th>Security Level</th> </tr> </thead> <tbody> <tr> <td>Configuration</td> <td>Unsecure Only</td> </tr> <tr> <td>Security Administrati...</td> <td>Unsecure Only</td> </tr> <tr> <td>System Status Interf...</td> <td>Unsecure Only</td> </tr> <tr> <td>Enhanced TSPI</td> <td>Unsecure Only</td> </tr> <tr> <td>HTTP</td> <td>Unsecure + Secure</td> </tr> <tr> <td>Web Services</td> <td>Secure, Medium</td> </tr> </tbody> </table>	Name	Security Level	Configuration	Unsecure Only	Security Administrati...	Unsecure Only	System Status Interf...	Unsecure Only	Enhanced TSPI	Unsecure Only	HTTP	Unsecure + Secure	Web Services	Secure, Medium	<table border="1"> <thead> <tr> <th colspan="2">Service Details</th> </tr> </thead> <tbody> <tr> <td>Name</td> <td>HTTP</td> </tr> <tr> <td>Host System</td> <td>Verizon</td> </tr> <tr> <td>Service Port</td> <td>80, 443</td> </tr> <tr> <td>Service Security Level</td> <td>Unsecure + Secure</td> </tr> </tbody> </table>	Service Details		Name	HTTP	Host System	Verizon	Service Port	80, 443	Service Security Level	Unsecure + Secure
Name	Security Level																									
Configuration	Unsecure Only																									
Security Administrati...	Unsecure Only																									
System Status Interf...	Unsecure Only																									
Enhanced TSPI	Unsecure Only																									
HTTP	Unsecure + Secure																									
Web Services	Secure, Medium																									
Service Details																										
Name	HTTP																									
Host System	Verizon																									
Service Port	80, 443																									
Service Security Level	Unsecure + Secure																									

When complete, select **File → Configuration** to return to configuration activities.

5.2. Licensing

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

To verify that there is a SIP Trunk Channels License with sufficient capacity click **License** in the Navigation pane 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																																	
<ul style="list-style-type: none"> BOOTP (6) Operator (3) Verizon <ul style="list-style-type: none"> System (1) Line (6) Control Unit (3) Extension (23) User (24) HuntGroup (3) Short Code (67) Service (0) RAS (1) Incoming Call Route (4) WanPort (0) Directory (0) Time Profile (0) Firewall Profile (1) IP Route (5) Account Code (0) License (22) Tunnel (0) User Rights (8) 	<table border="1"> <thead> <tr> <th>License Type</th> </tr> </thead> <tbody> <tr><td>Advanced Edition</td></tr> <tr><td>AUDIX Voicemail</td></tr> <tr><td>Avaya IP endpoints</td></tr> <tr><td>CTI Link Pro</td></tr> <tr><td>Customer Service Agent</td></tr> <tr><td>Customer Service Supervis</td></tr> <tr><td>Essential Edition</td></tr> <tr><td>IP500 Voice Networking Cl</td></tr> <tr><td>IPSec Tunnelling</td></tr> <tr><td>Mobile Worker</td></tr> <tr><td>Office Worker</td></tr> <tr><td>Phone Manager Pro (per s</td></tr> <tr><td>Phone Manager Pro IP Aud</td></tr> <tr><td>Power User</td></tr> <tr><td>Preferred Edition (Voicema</td></tr> <tr><td>Receptionist</td></tr> <tr><td>SIP Trunk Channels</td></tr> <tr><td>Software Upgrade 255</td></tr> <tr><td>Teleworker</td></tr> <tr><td>VMPro Networked Message</td></tr> </tbody> </table>	License Type	Advanced Edition	AUDIX Voicemail	Avaya IP endpoints	CTI Link Pro	Customer Service Agent	Customer Service Supervis	Essential Edition	IP500 Voice Networking Cl	IPSec Tunnelling	Mobile Worker	Office Worker	Phone Manager Pro (per s	Phone Manager Pro IP Aud	Power User	Preferred Edition (Voicema	Receptionist	SIP Trunk Channels	Software Upgrade 255	Teleworker	VMPro Networked Message	<table border="1"> <thead> <tr> <th colspan="2">Licenses</th> </tr> </thead> <tbody> <tr> <td>License Key</td> <td>t@HYRX6RAvHOIp8FoCkpxU3K3_Lww4rX</td> </tr> <tr> <td>License Type</td> <td>SIP Trunk Channels</td> </tr> <tr> <td>License Status</td> <td>Valid</td> </tr> <tr> <td>Instances</td> <td>5</td> </tr> <tr> <td>Expiry Date</td> <td>Never</td> </tr> </tbody> </table>	Licenses		License Key	t@HYRX6RAvHOIp8FoCkpxU3K3_Lww4rX	License Type	SIP Trunk Channels	License Status	Valid	Instances	5	Expiry Date	Never
License Type																																			
Advanced Edition																																			
AUDIX Voicemail																																			
Avaya IP endpoints																																			
CTI Link Pro																																			
Customer Service Agent																																			
Customer Service Supervis																																			
Essential Edition																																			
IP500 Voice Networking Cl																																			
IPSec Tunnelling																																			
Mobile Worker																																			
Office Worker																																			
Phone Manager Pro (per s																																			
Phone Manager Pro IP Aud																																			
Power User																																			
Preferred Edition (Voicema																																			
Receptionist																																			
SIP Trunk Channels																																			
Software Upgrade 255																																			
Teleworker																																			
VMPro Networked Message																																			
Licenses																																			
License Key	t@HYRX6RAvHOIp8FoCkpxU3K3_Lww4rX																																		
License Type	SIP Trunk Channels																																		
License Status	Valid																																		
Instances	5																																		
Expiry Date	Never																																		

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.

IP Offices	License	Avaya IP endpoints										
<ul style="list-style-type: none"> BOOTP (6) Operator (3) Verizon <ul style="list-style-type: none"> System (1) Line (6) Control Unit (3) Extension (23) User (24) HuntGroup (3) Short Code (67) Service (0) RAS (1) 	License Type <ul style="list-style-type: none"> Advanced Edition AUDIX Voicemail Avaya IP endpoints CTI Link Pro Customer Service Agent Customer Service Supervisor Essential Edition IP500 Voice Networking Client IPSec Tunnelling Mobile Worker Office Worker 	Licenses <table border="1"> <tr> <td>License Key</td> <td>G2xc7BdNDOa7XnHkzIR01TpZz9dvpG_N</td> </tr> <tr> <td>License Type</td> <td>Avaya IP endpoints</td> </tr> <tr> <td>License Status</td> <td>Valid</td> </tr> <tr> <td>Instances</td> <td>5</td> </tr> <tr> <td>Expiry Date</td> <td>Never</td> </tr> </table>	License Key	G2xc7BdNDOa7XnHkzIR01TpZz9dvpG_N	License Type	Avaya IP endpoints	License Status	Valid	Instances	5	Expiry Date	Never
License Key	G2xc7BdNDOa7XnHkzIR01TpZz9dvpG_N											
License Type	Avaya IP endpoints											
License Status	Valid											
Instances	5											
Expiry Date	Never											

A similar process can be used to check the license status for other desired features. For example, the following screen shows the availability of a valid license for **Power User** features. In the sample configuration, the user with extension 234 will be configured as a “Power User” and will be capable of using the Avaya IP Office Softphone.

IP Offices	License	Power User										
<ul style="list-style-type: none"> BOOTP (6) Operator (3) Verizon <ul style="list-style-type: none"> System (1) Line (6) Control Unit (3) Extension (23) User (24) HuntGroup (3) Short Code (67) Service (0) RAS (1) Incoming Call Route (4) WanPort (0) Directory (0) Time Profile (0) Firewall Profile (1) 	License Type <ul style="list-style-type: none"> Advanced Edition AUDIX Voicemail Avaya IP endpoints CTI Link Pro Customer Service Agent Customer Service Supervisor Essential Edition IP500 Voice Networking Client IPSec Tunnelling Mobile Worker Office Worker Phone Manager Pro (per s Phone Manager Pro IP Aut Power User Preferred Edition (Voicema 	Licenses <table border="1"> <tr> <td>License Key</td> <td>1NWBWbhjX5fs0l4HE4BdHVENy3STAV9O</td> </tr> <tr> <td>License Type</td> <td>Power User</td> </tr> <tr> <td>License Status</td> <td>Valid</td> </tr> <tr> <td>Instances</td> <td>5</td> </tr> <tr> <td>Expiry Date</td> <td>Never</td> </tr> </table>	License Key	1NWBWbhjX5fs0l4HE4BdHVENy3STAV9O	License Type	Power User	License Status	Valid	Instances	5	Expiry Date	Never
License Key	1NWBWbhjX5fs0l4HE4BdHVENy3STAV9O											
License Type	Power User											
License Status	Valid											
Instances	5											
Expiry Date	Never											

5.3. System Settings

This section illustrates the configuration of system settings. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. Select **System** in the Navigation pane to configure these settings. The subsection order corresponds to a left to right navigation of the tabs in the Details pane for System settings.

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, VerizonIPCC-SBC 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.

The screenshot displays the 'VerizonIPCC-SBC' configuration window. On the left is a navigation pane with a tree structure including 'IP Offices', 'System', 'Line', 'Control Unit', 'Extension', 'User', 'HuntGroup', 'Short Code', 'Service', 'RAS', 'Incoming Call', 'WanPort', 'Directory', 'Time Profile', 'Firewall Profile', 'IP Route', 'Account Code', 'License', 'Tunnel', 'User Rights', 'ARS', and 'E911 System'. The 'System' tab is selected. The main area shows the 'System' configuration for 'VerizonIPCC-SBC'. The 'Name' field is set to 'VerizonIPCC-SBC'. Below this is a 'Contact Information' section with a text box for 'Set contact information to place System under special control'. Further down are fields for 'Device ID', 'TFTP Server IP Address' (10.80.150.70), 'HTTP Server IP Address' (10.80.150.70), 'Phone File Server Type' (Memory Card), 'Manager PC IP Address' (10.80.150.38), 'Avaya HTTP Clients Only' (checked), 'Enable Softphone HTTP Provisioning' (checked), and 'Automatic Backup' (checked). On the right side, there are fields for 'Locale', 'Branch Prefix', and 'Local Number Length'. At the bottom right are 'OK', 'Cancel', and 'Help' buttons.

5.3.2. LAN Settings

In the sample configuration, LAN1 was used to connect the IP Office to the enterprise network. To view or configure the **IP Address** of LAN1, select the **LAN1** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP Address of the IP Office is 10.80.150.70. Other parameters on this screen may be set according to customer requirements. In the example screen, the **DHCP Mode** was set to “Server” to allow IP Office to facilitate provisioning for the IP Telephones in the sample configuration.

The screenshot displays the 'LAN Settings' configuration window for 'LAN1'. At the top, there are tabs for 'SMDR', 'Twining', 'VCM', 'CCR', and 'Codecs'. Below these are tabs for 'System', 'LAN1', 'LAN2', 'DNS', 'Voicemail', 'Telephony', 'Directory Services', 'System Events', and 'SMTP'. The 'LAN1' tab is selected, and within it, the 'LAN Settings' sub-tab is active. The configuration fields include: 'IP Address' set to '10 . 80 . 150 . 70', 'IP Mask' set to '255 . 255 . 255 . 0', 'Primary Trans. IP Address' set to '0 . 0 . 0 . 0', 'RIP Mode' set to 'None' with a dropdown arrow, and an unchecked 'Enable NAT' checkbox. The 'Number Of DHCP IP Addresses' is set to '200' with an information icon. The 'DHCP Mode' section shows four radio buttons: 'Server' (selected), 'Client', 'Dialin', and 'Disabled'. An 'Advanced' button is located at the bottom right.

Select the **VoIP** tab as shown in the following screen. 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 and 9600-Series Telephones used in the sample configuration. The **SIP Registrar Enable** box is checked to allow Avaya 1140E, Avaya Flare Experience, and Avaya IP Office Softphone usage. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Verizon Business

RTP Port Number: For each VoIP call, a receive port for incoming Real Time Protocol (RTP) traffic is selected from a defined range of possible ports, using the even numbers in that range. The Real Time Control Protocol (RTCP) traffic for the same call uses the RTP port number plus 1 (i.e., the odd numbers). For control units and Avaya H.323 IP phones, the default port range used is 49152 to 53246. On some installations, it may be a requirement to change or restrict the port range used. It is recommended that only port numbers between 49152 and 65535 are used, that being the range defined by the Internet Assigned Numbers Authority (IANA) for dynamic usage. **Port Range (minimum):** Default = 49152. Range = 1024 to 64510. This sets the lower limit for the RTP port numbers used by the system. **Port Range (maximum):** Default = 53246. Range = 2048 to 65534. This sets the upper limit for the RTP port numbers used by the system. The gap between the minimum and the maximum must be at least 1024.

If desired, 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. In the sample configuration shown below, IP Office will mark SIP signaling with a value associated with “Assured Forwarding” using DSCP decimal 34 (**SIG DSCP** parameter). IP Office will mark the RTP media with a value associated with “Expedited Forwarding” using DSCP decimal 46 (**DSCP** parameter). This screen enables flexibility in IP Office DiffServ markings (RFC 2474) to allow alignment with network routing policies, which are outside the scope of these Application Notes. Other parameters on this screen may be set according to customer requirements.

The screenshot displays the IP Office configuration interface for the VoIP section under the LAN1 tab. The interface is organized into several sections:

- H.323 Settings:**
 - ☒ H.323 Gatekeeper Enable
 - ☒ SIP Trunks Enable
 - ☒ SIP Registrar Enable
 - ☐ H.323 Auto-create Extn
 - ☐ H.323 Auto-create User
 - ☐ H.323 Remote Extn Enable
 - ☒ Enable RTP Monitoring On Port 5005
- RTP Port Number Range:**
 - Port Range (Minimum): 49152
 - Port Range (Maximum): 53246
- DiffServ Settings:**

B8	DSCP(Hex)	FC	DSCP Mask (Hex)	88	SIG DSCP (Hex)
46	DSCP	63	DSCP Mask	34	SIG DSCP
- DHCP Settings:**
 - Primary Site Specific Option Number (SSON): 176
 - Secondary Site Specific Option Number (SSON): 242
 - VLAN: Not Present
 - 1100 Voice VLAN Site Specific Option Number (SSON): 232
 - 1100 Voice VLAN IDs: (empty field)
- RTP Keepalives:**
 - Scope: Disabled
 - Periodic timeout: 0

Select the **Network Topology** tab as shown in the following screen. In the sample configuration, the default settings were used and the **Use Network Topology Info** in the **SIP Line** was set to “None” in Section 5.4.2. The **Binding Refresh Time (seconds)** can still be used to lower the SIP OPTIONS timing from the default of 300 seconds. During the testing, the Binding Refresh Time was varied (e.g., 60 seconds, 120 seconds to test SIP OPTIONS timing).

Network Topology Discovery

STUN Server IP Address: 0 . 0 . 0 . 0 STUN Port: 3478

Firewall/NAT Type: Unknown

Binding Refresh Time (seconds): 120

Public IP Address: 0 . 0 . 0 . 0

Public Port UDP: 0

Run STUN Cancel

☐ Run STUN on startup

If using IP Office as a DHCP server and DHCP Server mode has been selected from the **LAN1** → **Lan Settings** Tab, click the **DHCP Pools** tab. Although beyond the intended scope of these Application Notes, the following screen is shown as a simple example.

☒ Apply to Avaya IP Phones Only

Start Address	Subnet Mask	Default Router	Pool Size
10.80.150.72	255.255.255.0	10.80.150.1	15

Add... Remove Edit...

Optionally, select the **SIP Registrar** tab. The following screen shows the settings used in the sample configuration. The **Domain Name** has been set to the customer premises equipment domain “avayalab.com”. If the **Domain Name** is left at the default blank setting, SIP registrations may use the IP Office LAN 1 IP Address. All other parameters shown are default values.

System	LAN1	LAN2	DNS	Voicemail	Telephony	Directory Services	System Events	SMTP	SMDR	Twinning	VCM	CCR	Co
<div> LAN Settings VoIP Network Topology DHCP Pools SIP Registrar </div>													
Domain Name		avayalab.com											
Layer 4 Protocol		Both TCP & UDP											
TCP Port		5060											
UDP Port		5060											
Challenge Expiry Time (secs)		10											
Auto-create Extn/User		<input type="checkbox"/>											
SIP Remote Extn Enable		<input type="checkbox"/>											

5.3.3. Voicemail

To view or change voicemail settings, select the **Voicemail** tab as shown in the following screen. The **Voicemail Type** in the sample configuration is “Voicemail Lite/Pro”. Other Voicemail types may be used. The **Voicemail IP Address** in the sample configuration is 10.80.150.182, the IP Address of the PC running the Voicemail Pro software.

System	LAN1	LAN2	DNS	Voicemail	Telephony	Directory Services	System Events	SMTP	SMDR	Twinning	VCM	CCR	
<div> Voicemail Type Voicemail Lite/Pro <input checked="" type="checkbox"/> Messages Button Goes To Visual Voice </div>													
Voicemail Destination													
Voicemail IP Address		10 . 80 . 150 . 182											
Backup Voicemail IP Address		0 . 0 . 0 . 0											

Further down on the same **Voicemail** tab the following screen shows the **SIP Settings** configured for Voicemail in the sample configuration. As described in [VZBIPT-IPO81SBC], the “Callback” application of Avaya Voicemail Pro was used to allow Voicemail Pro to call out via the SIP Line to Verizon Business IP Trunk service when a message is left in a voice mailbox.

SIP Settings	
SIP Name	7329450239
SIP Display Name (Alias)	Voicemail
Contact	7329450239
Anonymous	<input type="checkbox"/>

5.3.4. System Telephony Configuration

To view or change telephony settings, select the **Telephony** tab and **Telephony** sub-tab as shown in the following screen. In the sample configuration, the **Inhibit Off-Switch Forward/Transfer** box is unchecked so that call forwarding and call transfer to PSTN destinations via the Verizon Business IP Trunk service can be tested. That is, a call can arrive to IP Office via the Verizon IP Contact Center service, and be forwarded or transferred back to the PSTN with the outbound leg of the call using the Verizon IP Trunk service. The **Companding Law** parameters were set to “ULAW” as is typical in North American locales. Other parameters on this screen may be set according to customer requirements.

The **Default Name Priority** field was introduced in IP Office Release 8 and 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. With the option to “Favor Directory”, 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 will be illustrated further in the context of the SIP Line to Verizon. A user’s personal directory example is shown in Section 5.5.1.

The screenshot displays the 'System Telephony Configuration' window. The 'Telephony' tab is selected, and the 'Telephony' sub-tab is active. The window is divided into several sections:

- Analogue Extensions:** Includes dropdowns for 'Default Outside Call Sequence' (Normal), 'Default Inside Call Sequence' (Ring Type 1), and 'Default Ring Back Sequence' (Ring Type 2). There is also a checkbox for 'Restrict Analogue Extension Ringer Voltage' which is unchecked.
- Companding Law:** Contains two sub-sections: 'Switch' and 'Line'. Both have radio buttons for 'U-Law' (selected) and 'A-Law' (unselected).
- System-wide Settings:** A list of checkboxes including 'DSS Status' (unchecked), 'Auto Hold' (checked), 'Dial By Name' (checked), 'Show Account Code' (checked), 'Inhibit Off-Switch Forward/Transfer' (unchecked), 'Restrict Network Interconnect' (unchecked), 'Drop External Only Impromptu Conference' (unchecked), 'Visually Differentiate External Call' (unchecked), 'Unsupervised Analog Trunk Disconnect Handling' (unchecked), and 'High Quality Conferencing' (checked).
- Time and Delay Settings:** Spinners for 'Dial Delay Time (secs)' (4), 'Dial Delay Count' (0), 'Default No Answer Time (secs)' (15), 'Hold Timeout (secs)' (0), 'Park Timeout (secs)' (300), and 'Ring Delay (secs)' (5).
- Call and Name Settings:** A dropdown for 'Call Priority Promotion Time (secs)' set to 'Disabled', a dropdown for 'Default Currency' set to 'USD', and a dropdown for 'Default Name Priority' set to 'Favor Trunk'.

To view or change settings associated with tones or music, select the **Telephony** tab and **Tones & Music** sub-tab as shown in the following screen. In the sample configuration, music on hold was provided via a WAV file from IP Office. For conferences, entry tone and exit tones are provided by IP Office.

The screenshot shows the 'Telephony' tab selected, with the 'Tones & Music' sub-tab active. The configuration includes:

- Conferencing Tone:** Entry & Exit Tones
- Disconnect Tone:** Default
- Tone Plan:** Tone Plan 1
- CLI Type:** (empty dropdown)
- Local Dial Tone:** ☒
- Local Busy Tone:** ☐
- Beep on listen:** ☒
- GSM Silence Suppression:** ☐
- Busy Tone Detection:** ☐ Mode: System Frequency
- Hold Music:** System Source: WAV File

5.3.5. System Twinning Configuration

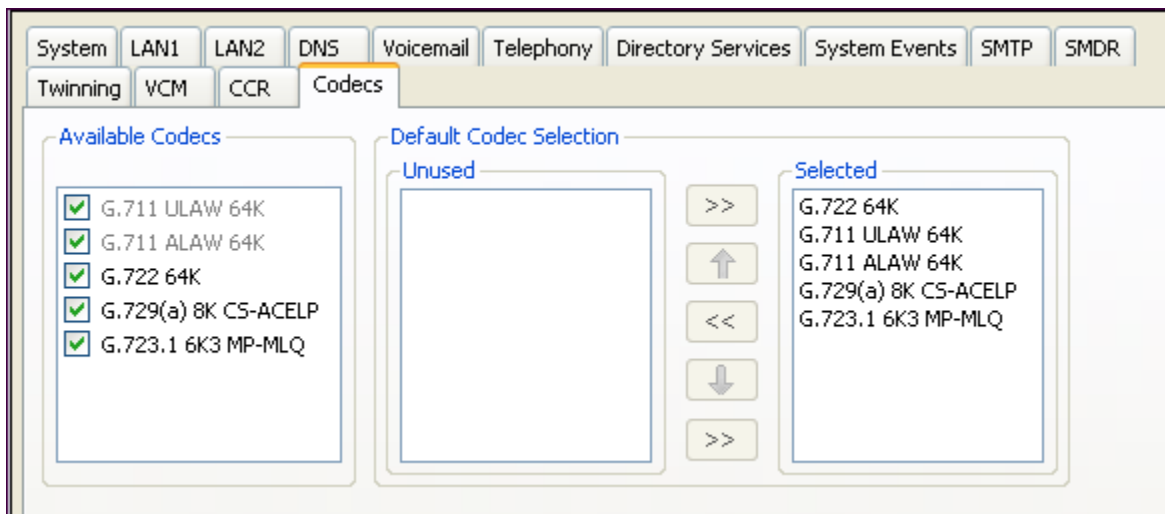
To view or change Twinning settings, select the **Twining** tab as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank. With this configuration, and related configuration of “Diversion header” on the SIP Line to Verizon Business IP Trunk service (Section 5.4.1 of reference [VZBIPT-IPO81SBC]), the true identity of the caller can be presented to the twinning destination (e.g., a user’s mobile phone) when a call is twinned out via the Verizon Business IP Trunk service. That is, a call can arrive via a Verizon IP Contact Center service toll-free number, and be twinned out to a mobile telephone using the Verizon IP Trunk service, with the twinned mobile phone seeing the identity of the caller that dialed the Verizon toll-free number.

The screenshot shows the 'Twining' tab selected. The configuration includes:

- Send original calling party information for Mobile Twinning:** ☐
- Calling party information for Mobile Twinning:** (empty text field)

5.3.6. System Codecs Configuration

The **System → Codecs** tab was introduced in IP Office Release 8. On the left, observe the list of **Available Codecs**. In the example screen below, which is not intended to be prescriptive, the box next to each codec is checked, making all the codecs available in other screens where codec configuration may be performed (such as the SIP Line in Section 5.4). The **Default Codec Selection** area enables the codec preference order to be configured on a system-wide basis, using the up, down, left, and right arrows. By default, all IP (SIP and H.323) lines and extensions will assume the system default codec selection, unless configured otherwise for the specific line or extension.



5.4. SIP Line

This section shows the configuration screens for the SIP Line in IP Office. The Appendix in Section 11 contains an example SIP Trunk template file that was generated from the SIP Line configured in this section.

To add a new SIP Line, right click on **Line** in the Navigation pane, and select **New → SIP Line** (not shown). A new Line Number will be assigned automatically. To edit an existing SIP Line, click **Line** in the Navigation pane, and the SIP Line to be configured in the Group pane.

5.4.1. SIP Line - SIP Line Tab

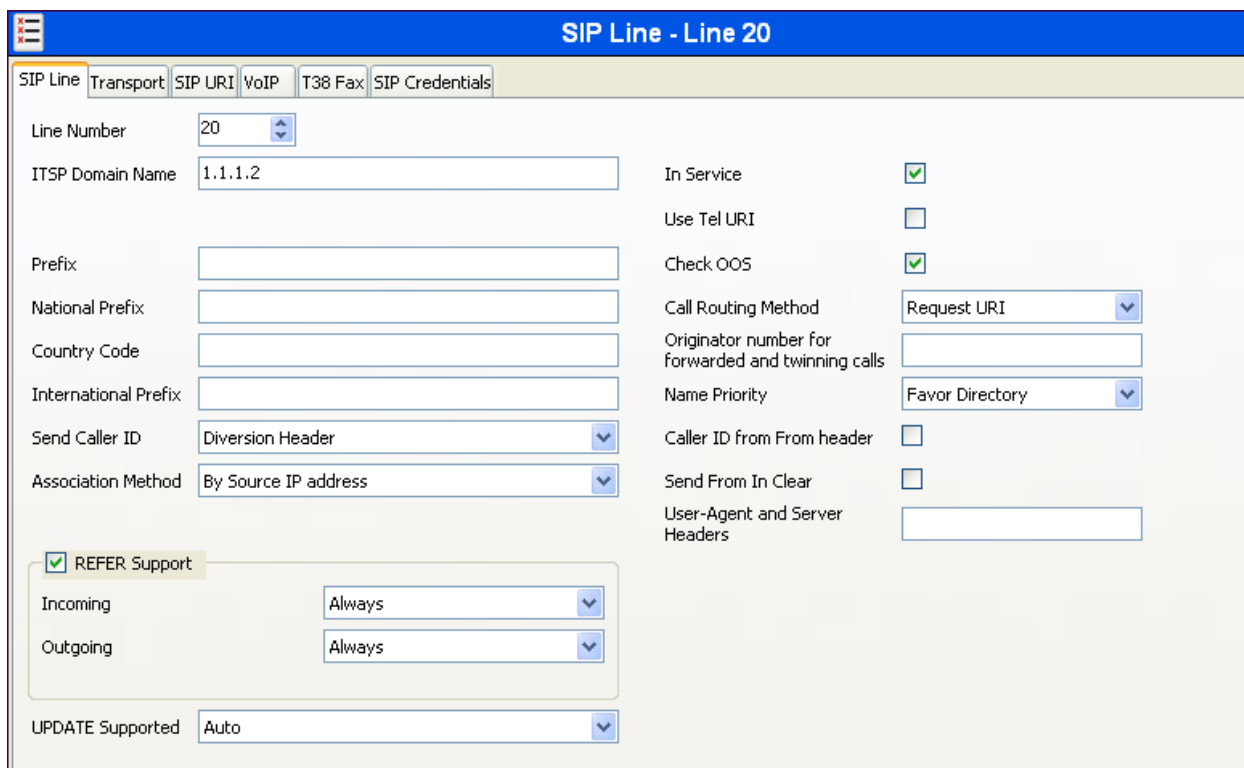
The **SIP Line** tab in the Details pane is shown below for Line Number 20, used for Avaya SBCE. The **ITSP Domain Name** is configured to the IP Office LAN1 address (1.1.1.2). By default, the **In Service** and **Check OOS** boxes are checked.

Although the Verizon IPCC service is inbound only, the **Send Caller ID** parameter is set to “Diversion Header”, as detailed in [VZBIPT-IPO81SBC], for outbound calls through Verizon IP Trunk service through the same Avaya SBCE. The **Call Routing Method** can retain the default “Request URI” setting, or may be changed to “To Header”, to match Incoming Call Routes

based on the contents of the “To Header”. In the sample configuration, the default “Request URI” setting was used.

The **REFER Support** parameter default is “Auto”. Alternatively, the default can be overridden with “Never” to explicitly disable use of REFER, or “Always” to explicitly enable use of REFER. The screen below shows the value “Always” set in the sample configuration. However, a signaling rule was created in the Avaya SBCE in Section 6.6 to prevent REFER messages from going across the Verizon Business IP Contact Center SIP Trunk service. See Section 2.2 for additional information.

The **Name Priority** parameter was introduced in IP Office Release 8.0. The **Name Priority** parameter can retain the default “System Default” setting, or can be configured to “Favor Trunk” or “Favor Directory” as shown in the sample screen below. “System Default” will use the setting displayed on the **System → Telephony → Telephony** tab shown in Section 5.3.4. The “Favor Directory” setting overrides the system setting and enables IP Office to match the caller’s telephone number against available system or personal directories, and display the name obtained from a match in the directory, if any, rather than name information received in the SIP signaling from Verizon.



SIP Line - Line 20

SIP Line | Transport | SIP URI | VoIP | T38 Fax | SIP Credentials

Line Number: 20

ITSP Domain Name: 1.1.1.2

Prefix:

National Prefix:

Country Code:

International Prefix:

Send Caller ID: Diversion Header

Association Method: By Source IP address

In Service: ☒

Use Tel URI: ☐

Check OOS: ☒

Call Routing Method: Request URI

Originator number for forwarded and twinning calls:

Name Priority: Favor Directory

Caller ID from From header: ☐

Send From In Clear: ☐

User-Agent and Server Headers:

☒ REFER Support

Incoming: Always

Outgoing: Always

UPDATE Supported: Auto

5.4.2. SIP Line - Transport Tab

Select the **Transport** tab. The **ITSP Proxy Address** is set to the inside IP address of the Avaya SBCE as shown in **Figure 1**. In the **Network Configuration** area, TCP is selected as the **Layer 4 Protocol**. The **Send Port** and **Listen Port** can retain the default values of 5060. The **Use Network Topology Info** parameter is set to “None”.

The screenshot shows the 'SIP Line - Line 20' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to '10.64.19.199'. In the 'Network Configuration' section, 'Layer 4 Protocol' is set to 'TCP', 'Send Port' is '5060', 'Use Network Topology Info' is 'None', and 'Listen Port' is '5060'. The 'Explicit DNS Server(s)' are set to '0 . 0 . 0 . 0' and '0 . 0 . 0 . 0'. The 'Calls Route via Registrar' checkbox is checked. The 'Separate Registrar' field is empty.

SIP Line - Line 20	
SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials	
ITSP Proxy Address: 10.64.19.199	
Network Configuration	
Layer 4 Protocol: TCP	Send Port: 5060
Use Network Topology Info: None	Listen Port: 5060
Explicit DNS Server(s): 0 . 0 . 0 . 0 0 . 0 . 0 . 0	
Calls Route via Registrar: <input checked="" type="checkbox"/>	
Separate Registrar:	

5.4.3. SIP Line - SIP URI Tab

Select the **SIP URI** tab. To add a new SIP URI, click the **Add...** button. In the bottom of the screen, a New Channel area is opened. To edit an existing entry, select an entry in the list at the top, and click the **Edit...** button. In the bottom of the screen, the Edit Channel area is opened.

In the sample configuration, each of the Verizon-provided toll free numbers are entered as a SIP URI, with the specific number entered in the **Local URI**, **Contact**, and **Display Name** fields. The **PAI** parameter value “None” is shown selected from the drop-down menu. The **Registration** parameter is set to the default “0: <None>” since Verizon Business IP Contact Center service does not require registration. The **Incoming Group** parameter, set here to 20, will be referenced when configuring Incoming Call Routes to map inbound SIP trunk calls to IP Office destinations in Section 5.7. The **Outgoing Group** parameter, set here to 0, is not relevant in that outbound calls in this sample configuration will choose Group 20 configured for Verizon Business IP Trunk Service as illustrated in reference [VZBIPT-IPO81SBC]. Click **OK**.

The screenshot shows the 'SIP Line - Line 20' configuration window. The 'SIP URI' tab is selected. Below the tab are buttons for 'Add...', 'Remove', and 'Edit...'. A table lists 8 channels with columns: Channel, Groups, Via, Local URI, Contact, Display Name, and PAI. The 'Edit Channel' dialog is open, showing fields for Via (<None>), Local URI (8668502380), Contact (8668502380), Display Name (8668502380), PAI (None), Registration (0: <None>), Incoming Group (20), Outgoing Group (0), and Max Calls per Channel (10). OK and Cancel buttons are at the bottom right of the dialog.

Channel	Groups	Via	Local URI	Contact	Display Name	PAI
1	20 20	<...>				N...
2	20 0	<...>	732945...	7329...	7329450240	N...
3	20 0	<...>	732945...	7329...	7329450239	N...
4	20 0	<...>	866850...	8668...	8668502380	N...
5	20 0	<...>	866852...	8668...	8668523221	N...
6	20 0	<...>	866850...	8668...	8668506850	N...
7	20 0	<...>	866851...	8668...	8668512649	N...
8	20 0	<...>	866851...	8668...	8668510107	N...

Edit Channel

Via: <None>

Local URI: 8668502380

Contact: 8668502380

Display Name: 8668502380

PAI: None

Registration: 0: <None>

Incoming Group: 20

Outgoing Group: 0

Max Calls per Channel: 10

OK Cancel

IP Office allows the number of simultaneous calls to a specific SIP URI to be managed using the **Max Calls per Channel** field. In the following screen, note that the **Max Calls per Channel** field has been changed from the default 10 to 2. With this configuration, two simultaneous calls to the number 866-850-6850 will be allowed. Once two calls are active, and a third call is attempted to 866-850-6850, IP Office will return a SIP 4xx response. Calls to other toll-free numbers using this same SIP Line are unaffected by the **Max Calls per Channel** for a different URI. Therefore, this approach could be used to control the maximum number of calls to each of the specific toll-free numbers, preventing a surge of calls to a given toll-free number from monopolizing the available call handling capacity of the access line or IP Office resources. An alternative means to restrict the number of simultaneous calls to a toll-free number that terminates on a hunt group would be to limit the queue size of the destination hunt group. If a non-priority call arrives to IP Office to a hunt group with a fixed size queue, and the queue is full, and there is no voice mail for the hunt group, IP Office returns a **486 Busy Here**. See Section 5.5.3 for hunt group configuration.

Edit Channel	
Via	<None>
Local URI	8668506850
Contact	8668506850
Display Name	8668506850
PAI	None
Registration	0: <None>
Incoming Group	20
Outgoing Group	0
Max Calls per Channel	2

OK
Cancel

5.4.4. SIP Line - VoIP Tab

Select the **VoIP** tab. In the sample configuration, the **Codec Selection** was configured using the “Custom” option, allowing an explicit ordered list of codecs to be specified, different from the system default (see Section 5.3.6). The arrow buttons can be used such that **G.729(a) 8K CS-ACELP** and **G.711 ULAW 64K** codecs are listed in the **Selected** column. This configures IP Office to support either G.729a or G.711MU for this SIP Line. 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. The **Use Offerer’s Preferred Codec** parameter can be left at the default unchecked value. In the sample configuration, Verizon preferred the G.729a codec in SDP, while also allowing the G.711MU codec. The IP Office configuration shown below matches these Verizon preferences. In the course of testing, the IP Office configuration was varied such that G.711MU was the preferred or only codec listed, and G.711MU calls were also successfully verified. The **PRACK/100rel Supported** parameter is new in IP Office Release 8, and should be left at the default unchecked value. Since the Verizon Business IP Contact Center service does not support fax, the **Fax Transport Support** parameter can be set to “None”, and the **T38 Fax** tab need not be visited. In the sample configuration the parameter was set to “T38 Fallback” for use with the Verizon IP Trunk service sharing the same SIP Line. For more information on T.38 fax with Verizon IP Trunk service, see reference [VZBIPT-IPO81SBC]. Since the Avaya SBCE does not require registration, the **SIP Credentials** tab need not be visited. Click **OK** (not shown).

The screenshot shows the 'SIP Line - Line 20' configuration window with the 'VoIP' tab selected. The 'Codec Selection' is set to 'Custom'. The 'Unused' list contains G.711 ALAW 64K, G.722 64K, and G.723.1 6K3 MP-MLQ. The 'Selected' list contains G.729(a) 8K CS-ACELP and G.711 ULAW 64K. The 'Fax Transport Support' is set to 'T38 Fallback', 'Call Initiation Timeout (s)' is 4, and 'DTMF Support' is RFC2833. On the right, 'Re-invite Supported' is checked, while 'VoIP Silence Suppression', 'Use Offerer's Preferred Codec', 'Codec Lockdown', and 'PRACK/100rel Supported' are unchecked.

Unused	Selected
G.711 ALAW 64K	G.729(a) 8K CS-ACELP
G.722 64K	G.711 ULAW 64K
G.723.1 6K3 MP-MLQ	

Parameters:

- Fax Transport Support: T38 Fallback
- Call Initiation Timeout (s): 4
- DTMF Support: RFC2833

Options:

- ☐ VoIP Silence Suppression
- ☒ Re-invite Supported
- ☐ Use Offerer's Preferred Codec
- ☐ Codec Lockdown
- ☐ PRACK/100rel Supported

5.5. Users, Extensions, and Hunt Groups

In this section, examples of IP Office Users, Extensions, and Hunt Groups will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users. To add a User, right click on **User** in the Navigation pane, and select **New** (not shown). To edit an existing User, select **User** in the Navigation pane, and select the appropriate user to be configured in the Group pane.

5.5.1. Digital User 232

The following screen shows the **User** tab for User 232. As shown in **Figure 1**, this user corresponds to the Avaya Digital 9508.

The screenshot displays the Avaya IP Office configuration interface for a specific user. On the left, a navigation pane lists various system elements, with 'User' selected. The main area is titled 'Avaya9508: 232' and contains a series of tabs for configuration: Menu Programming, Mobility, Phone Manager Options, Hunt Group Membership, Announcements, SIP, Personal Directory, User, Voicemail, DND, ShortCodes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, and Button Programming. The 'User' tab is active, showing fields for Name (Avaya9508), Password (masked with ****), Confirm Password (masked with ****), Full Name, Extension (232), Email Address, Locale, Priority (5), System Phone Rights (None), and Profile (Basic User). Below these fields are several checkboxes for user permissions: Receptionist, Enable Softphone, Enable one-X Portal Services, Enable one-X TeleCommuter, Enable Remote Worker, Enable Flare, and Ex Directory. A 'Flare Mode' dropdown is set to 'Standalone'. At the bottom, the 'Device Type' is specified as 'Avaya 9508' with a small phone icon next to the text.

Name	Extension
NoUser	
RemoteMa...	
Extn202	202
Extn203	203
Extn204	204
Extn205	205
Extn206	206
Extn207	207
Extn208	208
Extn210	210
Extn211	211
Extn212	212
Extn213	213
Extn214	214
Extn216	216
T7316E	231
Avaya9508	232
Avaya1616	233
Softphone	234
Avaya1140E	235
Avaya9630	236
Avaya9611	237
Avaya9621	238
Analog	241

User Configuration Fields:

- Name: Avaya9508
- Password: ****
- Confirm Password: ****
- Full Name:
- Extension: 232
- Email Address:
- Locale:
- Priority: 5
- System Phone Rights: None
- Profile: Basic User
- Receptionist: ☐
- Enable Softphone: ☐
- Enable one-X Portal Services: ☐
- Enable one-X TeleCommuter: ☐
- Enable Remote Worker: ☐
- Enable Flare: ☐
- Ex Directory: ☐
- Flare Mode: Standalone
- Device Type: Avaya 9508

The following screen shows the **SIP** tab for User 232. In the sample configuration, the **SIP Name** and **Contact** parameters are configured with a Verizon IP Trunk DID number for the user, 7329450232. As shown in [VZBIPT-IPO81SBC], these parameters configure the user part of the SIP URI in the From header for outgoing SIP trunk calls, and allow matching of the SIP URI for incoming calls from Verizon IP Trunk service, without having to enter this number as an explicit SIP URI for the SIP Line. 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.

User		Avaya9508: 232									
Name	Extension	<div> <div>User</div> <div>Voicemail</div> <div>DND</div> <div>ShortCodes</div> <div>Source Numbers</div> <div>Telephony</div> <div>Forwarding</div> <div>Dial In</div> <div>Voice Recording</div> <div>Button Programming</div> <div>Menu Programming</div> <div>Mobility</div> <div>Phone Manager Options</div> <div>Hunt Group Membership</div> <div>Announcements</div> <div>SIP</div> <div>Personal Directory</div> </div>									
<div> <div>Analog</div> <div>241</div> </div> <div> <div>Avaya1140E</div> <div>235</div> </div> <div> <div>Avaya1616</div> <div>233</div> </div> <div> <div>Avaya9508</div> <div>232</div> </div> <div> <div>Avaya9611</div> <div>237</div> </div> <div> <div>Avaya9621</div> <div>238</div> </div> <div> <div>Avaya9630</div> <div>236</div> </div> <div> <div>Extn202</div> <div>202</div> </div> <div> <div>Extn203</div> <div>203</div> </div> <div> <div>Extn204</div> <div>204</div> </div>	<div> <div>SIP Name</div> <div>7329450232</div> </div> <div> <div>SIP Display Name (Alias)</div> <div>Avaya9508</div> </div> <div> <div>Contact</div> <div>7329450232</div> </div> <div> <div><input type="checkbox"/></div> <div>Anonymous</div> </div>										

From **Figure 1**, note that user 232 will use the Mobile Twinning feature. The following screen shows the **Mobility** tab for User 232. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case 913035387024. Other options can be set according to customer requirements. In the sample configuration, the **Mobile Call Control** and **Mobile Callback** boxes were checked, and both mobile call control feature and mobile callback were tested using a Verizon-provided Toll Free number. In the case of mobile callback, a Verizon provided toll-free number was used to call in to IP Office and hang up. The mobile callback outbound leg used the Verizon IP Trunk service provisioned in [VZBIPT-IPO81SBC].

User

Name	Extension
Analog	241
Avaya1140E	235
Avaya1616	233
Avaya9508	232
Avaya9611	237
Avaya9621	238
Avaya9630	236
Extn202	202
Extn203	203
Extn204	204
Extn205	205
Extn206	206
Extn207	207
Extn208	208
Extn210	210
Extn211	211
Extn212	212
Extn213	213
Extn214	214
Extn216	216
NoUser	
RemoteMa...	
Softphone	234
T7316E	231

Avaya9508: 232

Internal Twinning

Twinned Handset: <None>

Maximum Number of Calls: 1

☐ Twin Bridge Appearances

☐ Twin Coverage Appearances

☐ Twin Line Appearances

☒ Mobility Features

☒ Mobile Twinning

Twinned Mobile Number (including dial access code): 913035387024

Twining Time Profile: <None>

Mobile Dial Delay (secs): 0

Mobile Answer Guard (secs): 0

☐ Hunt group calls eligible for mobile twinning

☐ Forwarded calls eligible for mobile twinning

☐ Twin When Logged Out

☐ one-X Mobile Client

☒ Mobile Call Control

☒ Mobile Callback

As described in Section 5.3.4, names can be entered in directories to allow IP Office to match the caller ID for incoming calls and display the names from the directory. The following screen shows the **Personal Directory** tab for user 232. With the configuration shown below and on the SIP Line in Section 5.4.1 (where “Favor Directory” is selected), if user 232 receives an inbound Verizon IP Toll Free call from the telephone number 13035387006, the phone will display the name “Avaya Lab 1” (along with the number).

User

Name	Extension
Analog	241
Avaya1140E	235
Avaya1616	233
Avaya9508	232
Avaya9611	237
Avaya9621	238
Avaya9630	236
Extn202	202

Avaya9508: 232

Personal Directory

Index	Name	Number
00	Avaya Lab 1	13035387006
01	Avaya Lab 2	13035387022

Add...

Remove

Edit...

5.5.2. Avaya Flare Experience User 234 with IP Office Softphone Privileges

The following screen shows the **User** tab for User 234. This user corresponds to a user that will be granted “Power User”, Flare features and Avaya IP Office Softphone features. The **Profile** parameter is set to “Power User”. The **Enable Softphone** and **Enable Flare** boxes are checked, along with the **Flare Mode** set to “Standalone”.

The screenshot displays the 'Softphone: 234' configuration window. The 'User' tab is selected, showing various configuration fields and checkboxes. The 'Name' field is set to 'Softphone', and the 'Extension' is '234'. The 'Profile' is set to 'Power User'. The 'Enable Softphone' and 'Enable Flare' checkboxes are checked, and the 'Flare Mode' is set to 'Standalone'. The 'Device Type' is 'Unknown SIP device'.

Field	Value
Name	Softphone
Password	****
Confirm Password	****
Full Name	
Extension	234
Email Address	
Locale	
Priority	5
System Phone Rights	None
Profile	Power User
Receptionist	<input type="checkbox"/>
Enable Softphone	<input checked="" type="checkbox"/>
Enable one-X Portal Services	<input checked="" type="checkbox"/>
Enable one-X TeleCommuter	<input checked="" type="checkbox"/>
Enable Remote Worker	<input checked="" type="checkbox"/>
Enable Flare	<input checked="" type="checkbox"/>
Flare Mode	Standalone
Send Mobility Email	<input type="checkbox"/>
Ex Directory	<input type="checkbox"/>
Device Type	Unknown SIP device

User Rights

Field	Value
User Rights view	User data

Like User 232, the **SIP** tab for the User 234 is configured with a **SIP Name** and **Contact** specifying the user's Verizon Business DID number using the Verizon IP Trunk Service, as detailed in [VZBIPT-IPO81SBC].

Softphone: 234	
User	VoiceMail
DND	ShortCodes
Source Numbers	Telephony
Forwarding	Dial In
Voice Recording	Button Programming
Menu Programming	Mobility
Phone Manager	
Hunt Group Membership	Announcements
SIP	Personal Directory
SIP Name	7329450234
SIP Display Name (Alias)	Softphone
Contact	7329450234
<input type="checkbox"/> Anonymous	

The following screen shows the Voicemail tab for User 234. The **Voicemail On** box is checked and a voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters. In the verification of these Application Notes, incoming calls from the Verizon Business IP Contact Center service to this user/agent were redirected to Voicemail Pro after no answer. Voicemail messages were recorded and retrieved successfully. Voice mail navigation and retrieval were performed locally and from PSTN telephones using the IP Contact Center toll-free numbers, to test DTMF using RFC 2833, and to test assignment of a Verizon-provided toll free number to the “Voicemail Collect” feature (e.g., via the *17 short code shown in Section 5.6).

Softphone: 234	
Hunt Group Membership	Announcements
SIP	Personal Directory
User	VoiceMail
DND	ShortCodes
Source Numbers	Telephony
Forwarding	Dial In
Voice Recording	Button Programming
Voicemail Code	*****
Confirm Voicemail Code	*****
Voicemail Email	
<input checked="" type="checkbox"/> Voicemail On	
<input type="checkbox"/> Voicemail Help	
<input type="checkbox"/> Voicemail Ringback	
<input type="checkbox"/> Voicemail Email Reading	
<input type="checkbox"/> UMS Web Services	
Voicemail Email	
<input checked="" type="radio"/> Off <input type="radio"/> Copy <input type="radio"/> Forward <input type="radio"/> Alert	
DTMF Breakout	
Reception / Breakout (DTMF *0/0)	System Default ()
Breakout (DTMF 2)	System Default ()
Breakout (DTMF 3)	System Default ()

Select the **Telephony** tab. Then select the **Supervisor Settings** tab as shown below. The **Login Code** will be used by the Flare Experience and IP Office Softphone user as the login password.

The screenshot shows the 'Softphone: 234' configuration window. The 'Telephony' tab is selected, and within it, the 'Supervisor Settings' sub-tab is active. The interface includes a top navigation bar with tabs like 'Hunt Group Membership', 'Announcements', 'SIP', 'Personal Directory', 'User', 'Voicemail', 'DND', 'ShortCodes', 'Source Numbers', 'Telephony', 'Forwarding', 'Dial In', 'Voice Recording', 'Button Programming', and 'Menu Programming'. Below this, the 'Supervisor Settings' sub-tab contains various configuration options:

- Login Code:** A text field containing '****'.
- Login Idle Period (secs):** An empty text field.
- Monitor Group:** A dropdown menu showing '<None>'.
- Coverage Group:** A dropdown menu showing '<None>'.
- Status on No-Answer:** A dropdown menu showing 'Logged On (No change)'.
- Reset Longest Idle Time:** A section with two radio buttons: 'All Calls' (selected) and 'External Incoming'.
- After Call Work Time (secs):** A dropdown menu showing 'System Default (10)'.
- Checkboxes on the right:**
 - ☐ Force Login
 - ☐ Force Account Code
 - ☐ Outgoing Call Bar
 - ☐ Inhibit Off-Switch Forward/Transfer
 - ☐ Can Intrude
 - ☒ Cannot be Intruded
 - ☐ Can Trace Calls
 - ☐ CCR Agent
 - ☐ Automatic After Call Work
 - ☐ Deny Auto Intercom Calls

Select the **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow an IP Office Softphone logged in as this extension to have multiple call appearances (e.g., necessary for call transfer from IP Office Softphone).

The screenshot shows the 'Softphone: 234' configuration window. The 'Telephony' tab is selected, and within it, the 'Call Settings' sub-tab is active. The interface includes the same top navigation bar as the previous screenshot. Below this, the 'Call Settings' sub-tab contains various configuration options:

- Outside Call Sequence:** A dropdown menu showing 'Default Ring'.
- Inside Call Sequence:** A dropdown menu showing 'Default Ring'.
- Ringback Sequence:** A dropdown menu showing 'Default Ring'.
- No Answer Time (secs):** A dropdown menu showing 'System Default (15)'.
- Wrap-up Time (secs):** A text field containing '2'.
- Transfer Return Time (secs):** A dropdown menu showing 'Off'.
- Call Cost Mark-Up:** A text field containing '100'.
- Checkboxes on the right:**
 - ☒ Call Waiting On
 - ☒ Answer Call Waiting On Hold
 - ☐ Busy On Held
 - ☒ Offhook Station

The following screen shows the Extension information for this user, simply to illustrate the **VoIP** tab available for a SIP Telephone and is not intended to be prescriptive. To view, select **Extension** from the Navigation pane, and the appropriate extension from the Group pane. Select **VoIP** in the Details pane. The new **Codec Selection** parameter may retain the default setting “System Default” to follow the system configuration shown in Section 5.3.6. Alternatively, “Custom” may be selected to allow the codecs to be configured for this extension, using the arrow keys to select and order the codecs.

SIP Extension: 8001 234

Extn VoIP T38 Fax

IP Address: 0 . 0 . 0 . 0

Codec Selection: System Default

Unused: [Empty box]

Selected: G.722 64K, G.711 ULAW 64K, G.711 ALAW 64K, G.729(a) 8K CS-ACELP, G.723.1 6K3 MP-MLQ

Fax Transport Support: None

TDM->IP Gain: Default

IP->TDM Gain: Default

DTMF Support: RFC2833

☐ VoIP Silence Suppression
☐ Local Hold Music
☒ Allow Direct Media Path
☒ Re-invite Supported
☐ Use Offerer's Preferred Codec
☐ Codec Lockdown
☐ Reserve Avaya IP endpoint license
☐ Reserve 3rd party IP endpoint license

5.5.3. Hunt Groups

During the verification of these Application Notes, users could 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** (not shown). To view or edit an existing hunt group, select **HuntGroup** from the Navigation pane, and the appropriate hunt group from the Group pane.

The following screen shows the **Hunt Group** tab for a hunt group with Extension 401 and Name “Sales”. This hunt group was configured to contain various telephones from **Figure 1**. The **Ring Mode** was set to “LongestWaiting” (i.e., “longest waiting”, most idle user receives next call). Click the **Edit** button to change the **User List**.

Extension	Name
235	Avaya1140E
232	Avaya9508
237	Avaya9611
234	Softphone
231	T7316E

The following screen shows the **Queuing** tab for hunt group 401. In the sample configuration, the hunt group was configured to allow queuing so that incoming Verizon toll-free calls could be queued when all the members of the hunt group were busy on calls. In the testing associated with these Application Notes, the **Queue Length** was varied using both “No Limit” and specifically sized queues. For example, if the **Queue Length** was configured to 2, and if two calls were already in queue, a third call to the Verizon toll-free number corresponding to this hunt group would get busy tone because IP Office would send a 486 Busy Here (i.e., if there was no Voicemail for the hunt group). As another example, if the **Queue Length** had a fixed limit of 2, and if two calls were already in queue, a third call to the Verizon toll-free number destined for this hunt group from a priority caller (see Section 5.7.3) would be queued ahead of non-priority callers, temporarily expanding the queue.

The screenshot shows a configuration window titled "Longest Waiting Group Sales: 401". The "Queuing" tab is selected, showing the following settings:

- ☒ Queuing On
- Queue Length: No Limit (dropdown menu)
- ☒ Normalize Queue Length
- Queue Type: Assign Call On Agent Answer (dropdown menu)
- Calls In Queue Alarm** (section header)
- Calls In Queue Threshold: 1 (spin box)
- Analog Extension to Notify: <None> (dropdown menu)

The following screen shows the **Announcements** tab for hunt group 220. In the sample configuration, when a call arrives when all members of the hunt group are busy on calls, the caller will first hear ring back tone. If a member of the hunt group does not become available after 10 seconds, the call will be answered by IP Office (i.e., 200 OK will be sent to Verizon), and the toll-free caller will hear a first announcement. Note that the **Flag call as answered** box is relevant for reporting applications, but does not change the fact that IP Office will answer the call when the first announcement is played. If the call is still not answered after the first announcement completes, the caller will hear music, a repeating second announcement, music, and so on until the call is answered by a member of the hunt group, or answered by voicemail for the hunt group (if configured). If a member of the hunt group becomes available while the caller is listening to ring back, music, or an announcement, the call is de-queued and delivered to the available member.

IP Office supports priority for queuing. For example, if low priority calls are waiting in queue, a higher priority call entering queue can be moved to the front of the queue and serviced before lower priority callers. For an inbound SIP trunk call, the priority can be specified on the Incoming Call Route as shown in Section 5.7.3.

Longest Waiting Group Sales: 401

Hunt Group | Queuing | Overflow | Fallback | Voicemail | Voice Recording | **Announcements** | SIP

☒ Announcements On

Wait before 1st announcement (seconds) 10 ☐ Synchronize Calls

Flag call as answered ☐

Play 1st announcement

Post announcement tone Music on hold

2nd Announcement ☒

Wait before 2nd announcement (seconds) 20

Play 2nd announcement

Repeat last announcement ☒

Wait before repeat (seconds) 20

5.6. Short Codes

In this section, various examples of IP Office short codes will be illustrated. To add a short code, right click on **Short Code** in the Navigation pane, and select **New** (not shown). To edit an existing short code, click **Short Code** in the Navigation pane, and the short code to be configured in the Group pane.

Although Verizon IP Contact Center, the focus of these Application Notes, is used for inbound toll-free numbers, inbound toll-free calls can be twinned, forwarded, or transferred back to the PSTN via the Verizon IP Trunk SIP Line. For more information on outbound calls, short codes, and ARS, see reference [VZBIPT-IPO81SBC].

The following screen illustrates a short code that acts like a feature code rather than a means to access a SIP Line. In this case, the **Code** “*17” is defined for **Feature** “Voicemail Collect”. This short code will be used as one means to allow a Verizon toll-free number to be programmed to route directly to voice messaging, via inclusion of this short code as the destination of an Incoming Call Route. See Section 5.7 for configuration of Incoming Call Routes.

The screenshot shows a configuration window titled "*17: Voicemail Collect". It has a "Short Code" tab selected. The fields are: Code (*17), Feature (Voicemail Collect), Telephone Number (?U), Line Group ID (0), Locale (empty), and Force Account Code (unchecked checkbox).

*17: Voicemail Collect	
Short Code	
Code	*17
Feature	Voicemail Collect
Telephone Number	?U
Line Group ID	0
Locale	
Force Account Code	<input type="checkbox"/>

The following screen illustrates another short code. In this case, the **Code** “FNE31” is defined for **Feature** “FNE Service” to **Telephone Number** “31” (Mobile Call Control). This short code will be used to allow a Verizon DID to be programmed to route directly to this feature, via inclusion of this short code as the destination of an Incoming Call Route (see Section 5.7). This feature is used to provide dial tone to twinned mobile devices (e.g., cell phone) directly from IP Office; once dial tone is received the user can perform dialing actions including making calls and activating Short Codes.

The screenshot shows a configuration window titled "FNE31: FNE Service". It has a "Short Code" tab selected. The fields are: Code (FNE31), Feature (FNE Service), Telephone Number (31), Line Group ID (0), Locale (empty), and Force Account Code (unchecked checkbox).

FNE31: FNE Service	
Short Code	
Code	FNE31
Feature	FNE Service
Telephone Number	31
Line Group ID	0
Locale	
Force Account Code	<input type="checkbox"/>

5.7. Incoming Call Routes

In this section, IP Office Incoming Call Routes are illustrated. Each Incoming Call Route will map a Verizon Business toll-free number to a destination user, group, or function on IP Office. In some cases, the destination will be chosen based on the combination of the toll-free number and the caller id of the caller. Example mappings are summarized in **Table 1** in Section 3. To add an incoming call route, right click on **Incoming Call Route** in the Navigation pane, and select **New** (not shown). To edit an existing incoming call route, select **Incoming Call Route** in the Navigation pane, and the appropriate incoming call route to be edited in the Group pane.

5.7.1. Incoming Call Route to a Specific Telephone Extension

In the screen shown below, the incoming call route for **Incoming Number** “8668502380” is illustrated. The **Line Group Id** is 20, matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to the Verizon Business IP Contact Center Service, in Section 5.4.

20 8668502380	
Standard Voice Recording Destinations	
Bearer Capability	Any Voice
Line Group ID	20
Incoming Number	8668502380
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

Select the **Destinations** tab. From the **Destination** drop-down, select an extension to receive the call when a PSTN user dials 8668502380. As shown in **Table 1**, 8668502380 is the number associated with IP Office user extension 232. (The **Destination** was changed in the course of testing to associate different destinations with the toll-free numbers.)

20 8668502380		
Standard Voice Recording Destinations		
	TimeProfile	Destination
	Default Value	232 Avaya9508

Incoming Call Routes for other direct mappings of toll-free numbers to IP Office users are not presented here, but are configured in the same fashion.

5.7.2. Incoming Call Routes to a Hunt Group by Dialed Toll-Free Number

In the screen shown below, an incoming call route for **Incoming Number** “8668523221” is illustrated. The **Line Group Id** is 20, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 5.4.

20 8668523221	
Standard Voice Recording Destinations	
Bearer Capability	Any Voice
Line Group ID	20
Incoming Number	8668523221
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

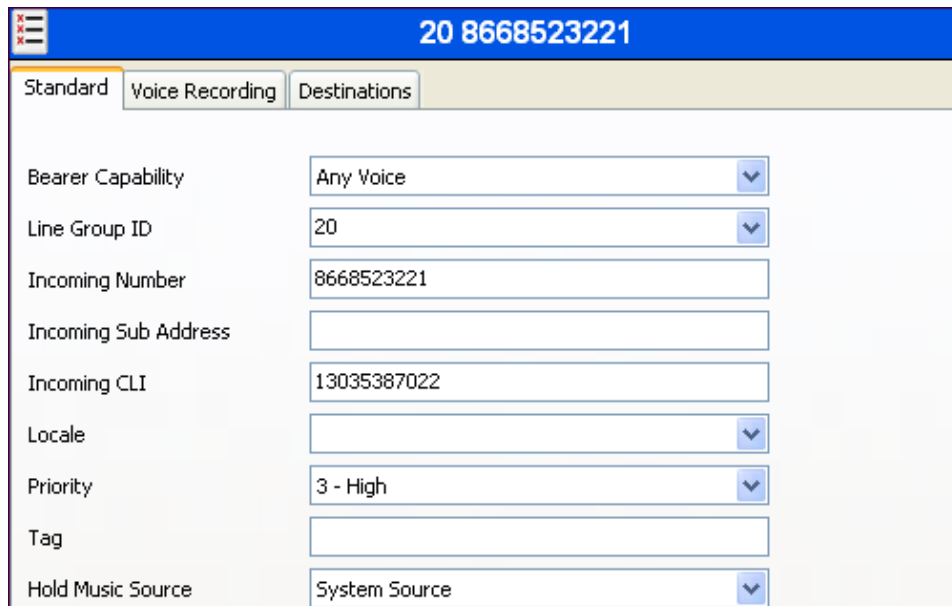
Select the **Destinations** tab. From the **Destination** drop-down, select the destination to receive the call when an arbitrary PSTN user dials 8668523221. As shown in **Table 1**, 8668510107 is the toll-free number associated with IP Office hunt group extension 401, the “Sales” hunt group.

20 8668523221		
Standard Voice Recording Destinations		
TimeProfile	Destination	Fallback Extension
▶ Default Value	401 Sales	

5.7.3. Incoming Call Routes Based on Calling Party Number

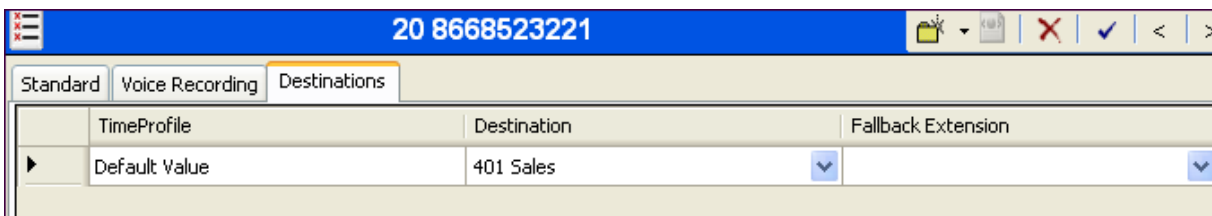
This section presents simple examples showing that IP Office can use the calling party number to distinguish call priority or call destination, for calls to the same toll-free number. While the matching shown here is based on the full calling number, partial matching is also possible (e.g., to match a calling area code for a targeted geographic treatment).

In the screen shown below, the incoming call route for **Incoming Number** “8668523221” and **Incoming CLI** “13035387022” is illustrated. The **Line Group Id** is 20, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 5.4. Note that the **Incoming Number** is the same as the toll-free number configured in the previous section. This route will be used for calls to the toll-free number specifically from a caller with caller ID “13035387022”. In this case, to allow this caller to be treated with priority when calling in for “Sales”, the **Priority** field is set to “3 – High”.



20 8668523221	
Standard	Voice Recording Destinations
Bearer Capability	Any Voice
Line Group ID	20
Incoming Number	8668523221
Incoming Sub Address	
Incoming CLI	13035387022
Locale	
Priority	3 - High
Tag	
Hold Music Source	System Source

Select the **Destinations** tab and from the drop-down, select the extension to receive the call when PSTN user 13035387022 dials 8668523221. In this case, the **Destination** is also the hunt group “401 Sales”, but since high priority has been configured via the **Standard** tab, incoming calls from this caller will move to the front of the queue, and be serviced before calls waiting in queue from other non-priority callers.



20 8668523221		
Standard	Voice Recording Destinations	
TimeProfile	Destination	Fallback Extension
Default Value	401 Sales	

5.7.4. Incoming Call Routes to Various IP Office Features

In the sample configuration, the incoming call route for **Incoming Number** “8668506850” was varied to test different destination features, such as Voice Mail, Mobile Call Control, and conference bridge, as shown in **Table 1** in Section 3. The screen showing the **Standard** tab for this toll-free number is shown below.

The screenshot shows a configuration window for an incoming call route. The title bar is blue and contains a small icon on the left and the text "20 8668506850" on the right. Below the title bar is a tabbed interface with three tabs: "Standard" (selected), "Voice Recording", and "Destinations". The "Standard" tab is active, showing a list of configuration fields on the left and their corresponding values or input fields on the right. The fields are: Bearer Capability (Any Voice), Line Group ID (20), Incoming Number (8668506850), Incoming Sub Address (empty), Incoming CLI (empty), Locale (empty), Priority (1 - Low), Tag (empty), and Hold Music Source (System Source). Each field has a small blue downward arrow icon to its right, indicating a dropdown menu.

Field	Value
Bearer Capability	Any Voice
Line Group ID	20
Incoming Number	8668506850
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

When configuring an Incoming Call Route, the **Destination** field can be manually configured with a number such as a short code, or certain keywords available from the drop-down list. At different times during testing, the **Destinations** tab for 8668506850 was configured to contain the following destinations:

- “*17” (short code “Voicemail Collect”, as shown in Section 5.6). With this destination, an incoming call to 8668506850 will be delivered directly to voice mail, allowing the caller to log-in to voice mail and access messages.
- “FNE31” (short code for accessing the Mobile Call Control feature directly, as shown in Section 5.6) With this destination, an incoming call to 8668506850 from configured mobile callers will be provided dial tone to make calls from the mobile phone as if the user were using their IP Office extension.
- “VM:MeetMeConf” With this destination, an incoming call to 8668506850 will be delivered directly to the Voicemail Pro Module “MeetMeConf” created for use as a conference bridge.

An example screen showing the short code configured for Voicemail Collect is shown below.

The screenshot shows a configuration window for the short code '20 8668506850*'. The 'Destinations' tab is selected. The table below shows the configuration for the 'Default Value'.

TimeProfile	Destination	Fallback Extension
Default Value	*17	

5.8. Saving Configuration Changes to IP Office

Navigate to **File → Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

The following will appear, with either **Merge** or **Immediate** selected, based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a service disruption. Click **OK** if desired.

The 'IP Office Settings' dialog box is shown with the 'Verizon' provider selected. The 'Configuration Reboot Mode' is set to 'Immediate'. The 'Reboot Time' is set to 13:18. The 'Call Barring' section shows 'Incoming Calls' and 'Outgoing Calls' both unchecked. The 'OK', 'Cancel', and 'Help' buttons are at the bottom.

Configuration Reboot Mode
<input type="radio"/> Merge
<input checked="" type="radio"/> Immediate
<input type="radio"/> When Free
<input type="radio"/> Timed

Reboot Time
13:18

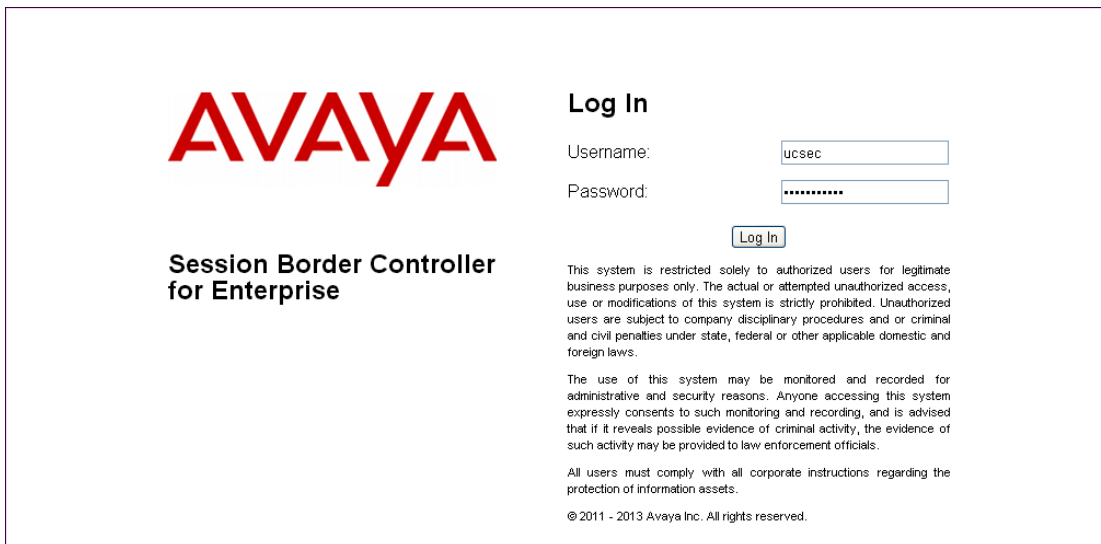
Call Barring
<input type="checkbox"/> Incoming Calls
<input type="checkbox"/> Outgoing Calls

6. Configure Avaya Session Border Controller for Enterprise

This section covers the configuration of the Avaya SBCE. It is assumed that the Avaya SBCE software has already been installed.

Use a WEB browser to access the Element Management Server (EMS) web interface, and enter `https://<ip-addr>/sbc` in the address field of the web browser, where `<ip-addr>` is the management LAN IP address of the Avaya SBCE.

Enter appropriate credentials and click **Log In**.



The login page features the Avaya logo on the left and a login form on the right. The form includes fields for Username (pre-filled with 'ucsec') and Password (masked with dots), followed by a 'Log In' button. Below the form, there is a disclaimer about system restrictions and a consent statement regarding monitoring. At the bottom, it states '© 2011 - 2013 Avaya Inc. All rights reserved.'

AVAYA

Session Border Controller for Enterprise

Log In

Username:

Password:

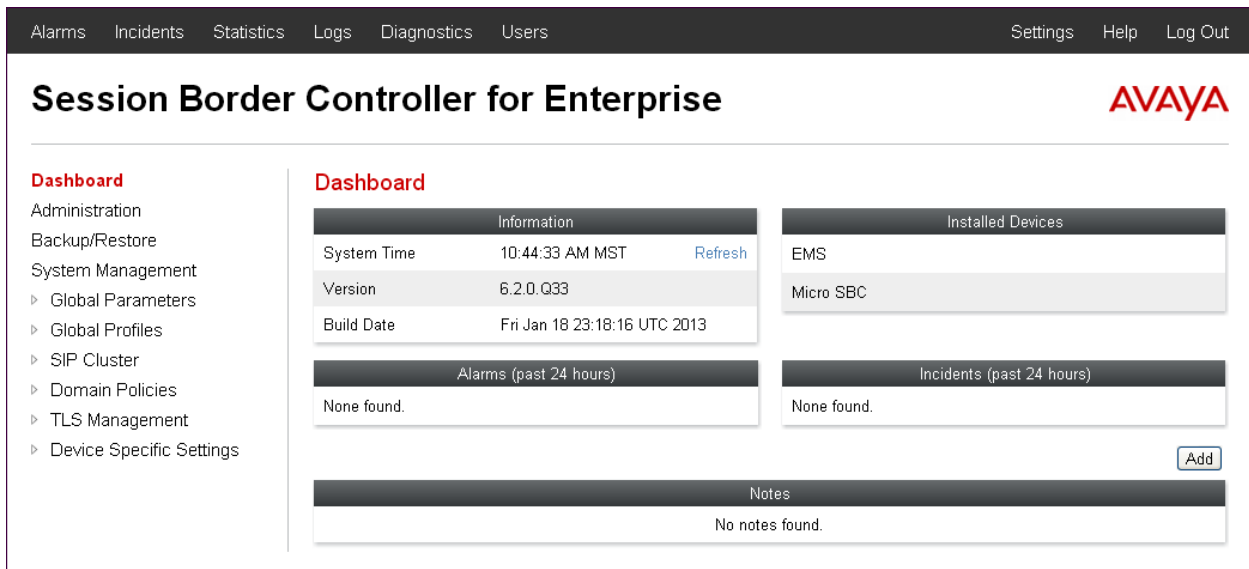
This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use or modifications of this system is strictly prohibited. Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

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The Dashboard for the Avaya SBCE will appear.



The dashboard has a top navigation bar with links: Alarms, Incidents, Statistics, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header shows 'Session Border Controller for Enterprise' and the Avaya logo. A left sidebar lists navigation options under 'Dashboard', 'Administration', 'Backup/Restore', and 'System Management'. The main content area displays several widgets: 'Information' (System Time, Version, Build Date), 'Installed Devices' (EMS, Micro SBC), 'Alarms (past 24 hours)', 'Incidents (past 24 hours)', and 'Notes'.

Alarms Incidents Statistics Logs Diagnostics Users Settings Help Log Out

Session Border Controller for Enterprise **AVAYA**

Dashboard

Administration

Backup/Restore

System Management

- Global Parameters
- Global Profiles
- SIP Cluster
- Domain Policies
- TLS Management
- Device Specific Settings

Dashboard

Information

System Time	10:44:33 AM MST	Refresh
Version	6.2.0.Q33	
Build Date	Fri Jan 18 23:18:16 UTC 2013	

Installed Devices

EMS
Micro SBC

Alarms (past 24 hours)

None found.

Incidents (past 24 hours)

None found.

Notes

No notes found.

To view system information that was configured during installation, click on **System Management**. A list of installed devices is shown in the right pane. In the case of the sample configuration, a single device named **Micro SBC** is shown. To view the configuration of this device, click **View** as highlighted below.

The **System Information** screen shows the **Network Configuration**, **DNS Configuration** and **Management IP(s)** information provided during installation and corresponds to **Figure 1**. The **Box Type** was set to **SIP** and the **Deployment Mode** was set to **Proxy**. Default values were used for all other fields.

System Information: Micro SBC				
General Configuration		Device Configuration		
Appliance Name	Micro SBC	HA Mode	No	
Box Type	SIP	Two Bypass Mode	No	
Deployment Mode	Proxy			
Network Configuration				
IP	Public IP	Netmask	Gateway	Interface
10.64.19.199	10.64.19.199	255.255.255.0	10.64.19.1	A1
1.1.1.2	1.1.1.2	255.255.255.0	1.1.1.1	B1
DNS Configuration		Management IP(s)		
Primary DNS	10.80.150.201	IP	10.80.150.199	
Secondary DNS				
DNS Location	DMZ			
DNS Client IP	10.64.19.199			

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

The following screen shows interface **A1** and **B1** are **Enabled**. To enable an interface click the corresponding **Toggle** button.

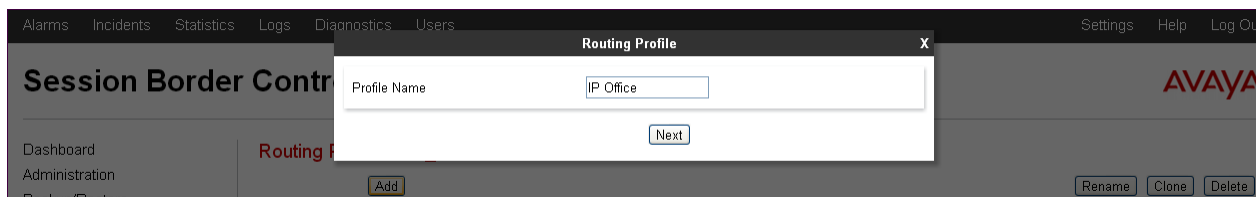
DDT; Reviewed:
SPOC 5/20/2013

6.2. Routing Profile

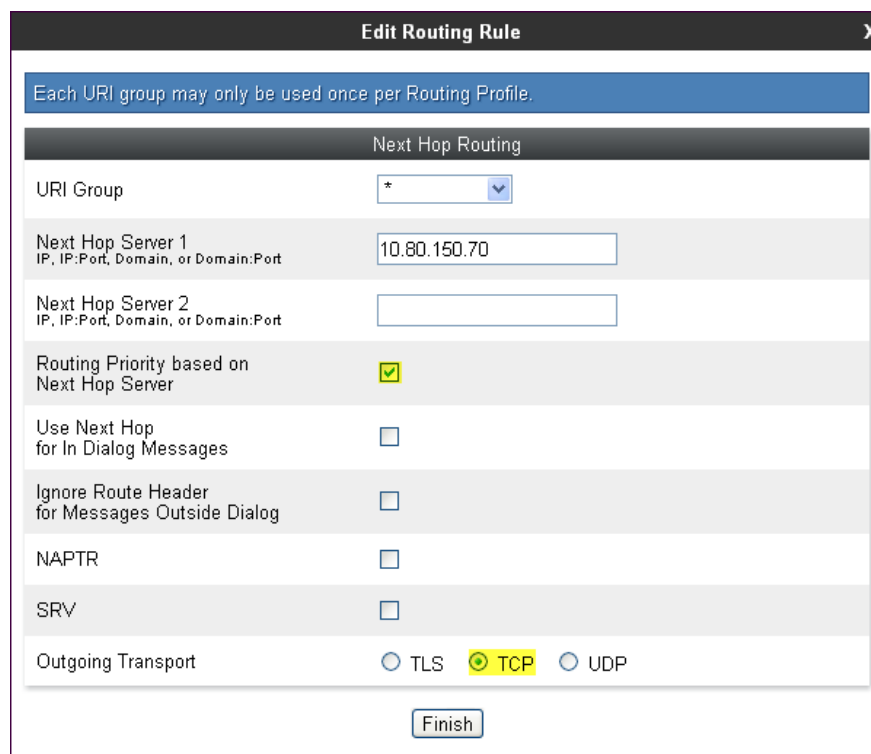
Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport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

To add a routing profile for IP Office, navigate to **Global Profiles → Routing** and select **Add** (not shown). Enter a **Profile Name** and click **Next** to continue.

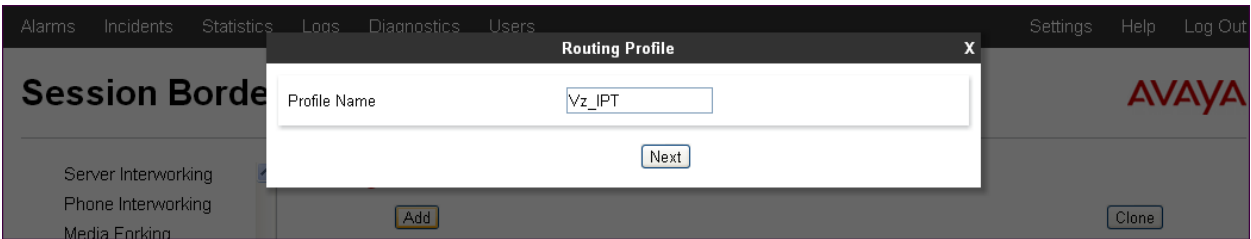
Note: Verizon Business IP Contact Center SIP Trunk service is an inbound only service, so a routing profile for Verizon is not required.



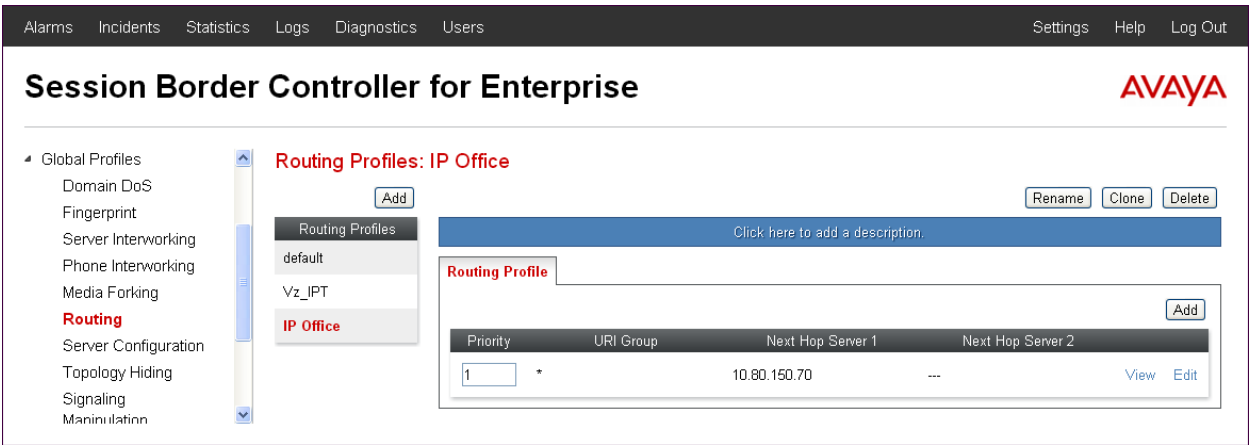
The following screen illustrates the Routing Profile named “IP Office” created in the sample configuration for IP Office. The **Next Hop Server 1** IP address must match the IP address of the IP Office LAN settings entered in Section 5.3.2. Leave the **Routing Priority based on Next Hop Server** box checked and enter **TCP** for the **Outgoing Transport** field matching the **Layer 4 Protocol** set in IP Office **SIP Line → Transport** in Section 5.4.2.



An existing routing profile named “Vz_IPT” created for the Verizon IP Trunk service provisioned in [VZBIPT-IPO81SBC] was used during compliance testing for outbound calls from IP Office.



The complete list of routing profiles used during compliance testing appears as follows.



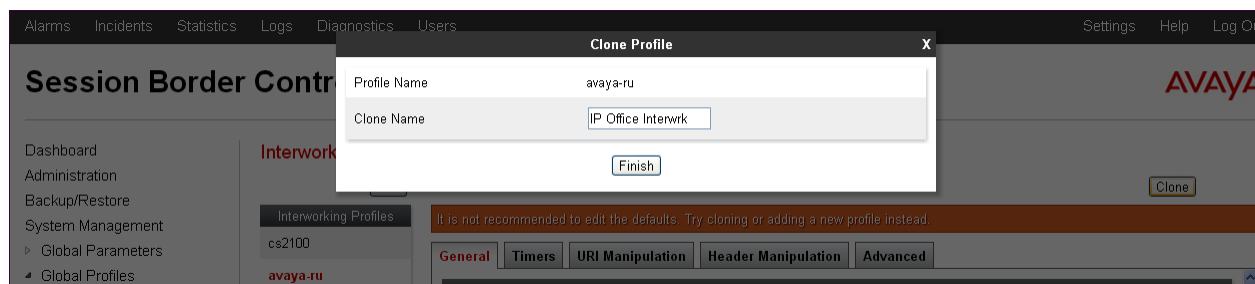
6.3. Server Interworking Profile

The Server Interworking profile configures and manages various SIP call server-specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters (for HA deployments), DoS security statistics, and trusted domains. Interworking Profile features are configured based on different Trunk Servers. There are default profiles available that may be used as is, or new profiles can be configured as described below.

In the sample configuration, separate Server Interworking profiles were created for IP Office and Verizon Business IP Contact Center SIP Trunk service.

6.3.1. Server Interworking Profile – IP Office

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



In the new window that appears, check the **T.38 Support** field. Use default values for all remaining fields and click **Next** to continue.

The screenshot shows the 'Editing Profile: IP Office Interwrk' window with the 'General' tab selected. The following fields are visible:

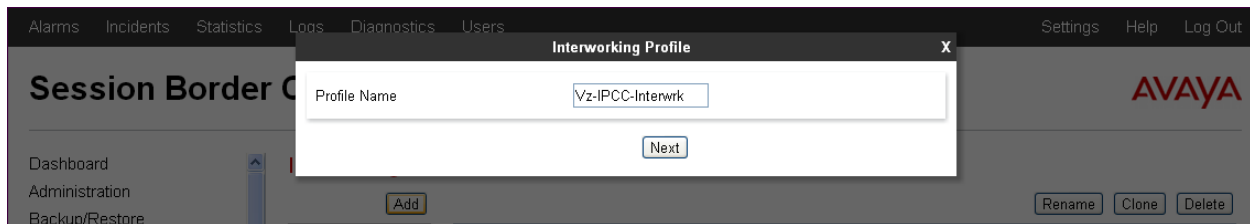
- Hold Support: ☐ None, ☒ RFC2543 - c=0.0.0.0, ☐ RFC3264 - a=sendonly
- 180 Handling: ☒ None, ☐ SDP, ☐ No SDP
- 181 Handling: ☒ None, ☐ SDP, ☐ No SDP
- 182 Handling: ☒ None, ☐ SDP, ☐ No SDP
- 183 Handling: ☒ None, ☐ SDP, ☐ No SDP
- Refer Handling: ☐
- 3xx Handling: ☐
- Diversion Header Support: ☐
- Delayed SDP Handling: ☐
- T.38 Support: ☒
- URI Scheme: ☒ SIP, ☐ TEL, ☐ ANY
- Via Header Format: ☒ RFC3261, ☐ RFC2543

A 'Next' button is located at the bottom right of the window.

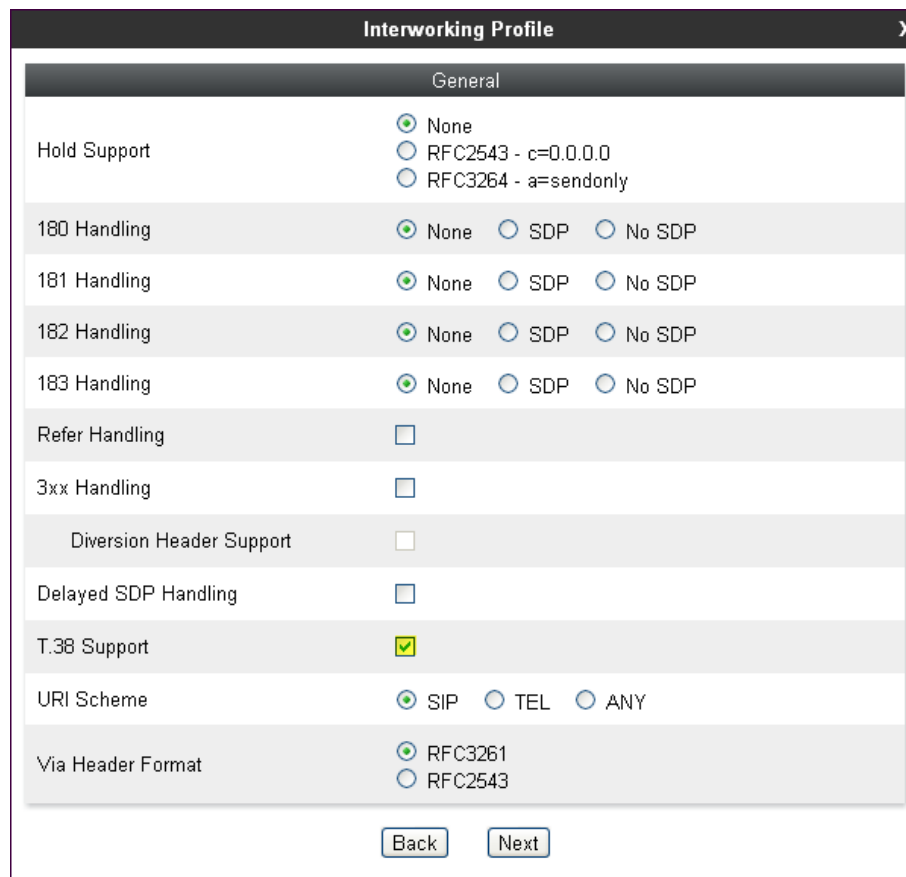
Default values can be used for the next windows that appear. Click **Next** to continue, then **Finish** to save the changes (not shown).

6.3.2. Server Interworking Profile – Verizon IPCC

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



In the new window that appears, check the **T.38 Support** field. Use default values for all remaining fields. Click **Next** to continue.



General	
Hold Support	<input checked="" type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543

Back Next

Default values can be used for the **Privacy** and **DTMF** sections on the following screen. Click **Next** to continue.

The screenshot shows the 'Interworking Profile' configuration window. It has a title bar with 'Interworking Profile' and a close button 'X'. The window is divided into two main sections: 'Privacy' and 'DTMF'. The 'Privacy' section contains five fields: 'Privacy Enabled' (checkbox), 'User Name' (text input), 'P-Asserted-Identity' (checkbox), 'P-Preferred-Identity' (checkbox), and 'Privacy Header' (text input). The 'DTMF' section contains a 'DTMF Support' field with three radio button options: 'None' (selected), 'SIP NOTIFY', and 'SIP INFO'. At the bottom of the window are 'Back' and 'Next' buttons.

Privacy	
Privacy Enabled	<input type="checkbox"/>
User Name	<input type="text"/>
P-Asserted-Identity	<input type="checkbox"/>
P-Preferred-Identity	<input type="checkbox"/>
Privacy Header	<input type="text"/>

DTMF	
DTMF Support	<input checked="" type="radio"/> None <input type="radio"/> SIP NOTIFY <input type="radio"/> SIP INFO

Back Next

Default values can be used for the **SIP Timers** and **Transport Timers** sections on the following screen. Click **Next** to continue.

The screenshot shows the 'Interworking Profile' configuration window, continuing from the previous screen. It has a title bar with 'Interworking Profile' and a close button 'X'. A blue banner at the top states 'All fields are optional.' The window is divided into two main sections: 'SIP Timers' and 'Transport Timers'. The 'SIP Timers' section contains five fields: 'Min-SE' (text input, seconds, [90 - 86400]), 'Init Timer' (text input, milliseconds, [50 - 1000]), 'Max Timer' (text input, milliseconds, [200 - 8000]), 'Trans Expire' (text input, seconds, [1 - 64]), and 'Invite Expire' (text input, seconds, [180 - 300]). The 'Transport Timers' section contains one field: 'TCP Connection Inactive Timer' (text input, seconds, [600 - 3600]). At the bottom of the window are 'Back' and 'Next' buttons.

All fields are optional.

SIP Timers	
Min-SE	<input type="text"/> seconds, [90 - 86400]
Init Timer	<input type="text"/> milliseconds, [50 - 1000]
Max Timer	<input type="text"/> milliseconds, [200 - 8000]
Trans Expire	<input type="text"/> seconds, [1 - 64]
Invite Expire	<input type="text"/> seconds, [180 - 300]

Transport Timers	
TCP Connection Inactive Timer	<input type="text"/> seconds, [600 - 3600]

Back Next

On the next window uncheck the **Topology Hiding: Change Call-ID** and **Change Max Forwards** boxes. Click **Finish** to save changes.

Interworking Profile	
Record Routes	<input type="radio"/> None <input type="radio"/> Single Side <input checked="" type="radio"/> Both Sides
Topology Hiding: Change Call-ID	<input checked="" type="checkbox"/>
Call-Info NAT	<input type="checkbox"/>
Change Max Forwards	<input checked="" type="checkbox"/>
Include End Point IP for Context Lookup	<input type="checkbox"/>
OCS Extensions	<input type="checkbox"/>
AVAYA Extensions	<input type="checkbox"/>
NORTEL Extensions	<input type="checkbox"/>
Diversion Manipulation	<input type="checkbox"/>
Diversion Header URI	<input type="text"/>
Metaswitch Extensions	<input type="checkbox"/>
Reset on Talk Spurt	<input type="checkbox"/>
Reset SRTP Context on Session Refresh	<input type="checkbox"/>
Has Remote SBC	<input checked="" type="checkbox"/>
Route Response on Via Port	<input type="checkbox"/>
Cisco Extensions	<input type="checkbox"/>

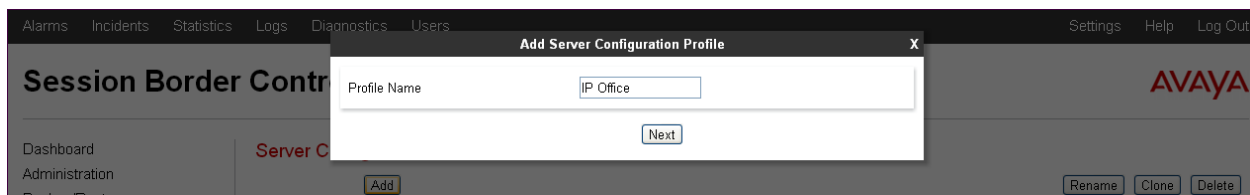
6.4. Server Configuration

The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs are used to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters, DoS security statistics, and trusted domains.

In the sample configuration, separate Server Configurations were created for IP Office and Verizon Business IP Contact Center SIP Trunk service.

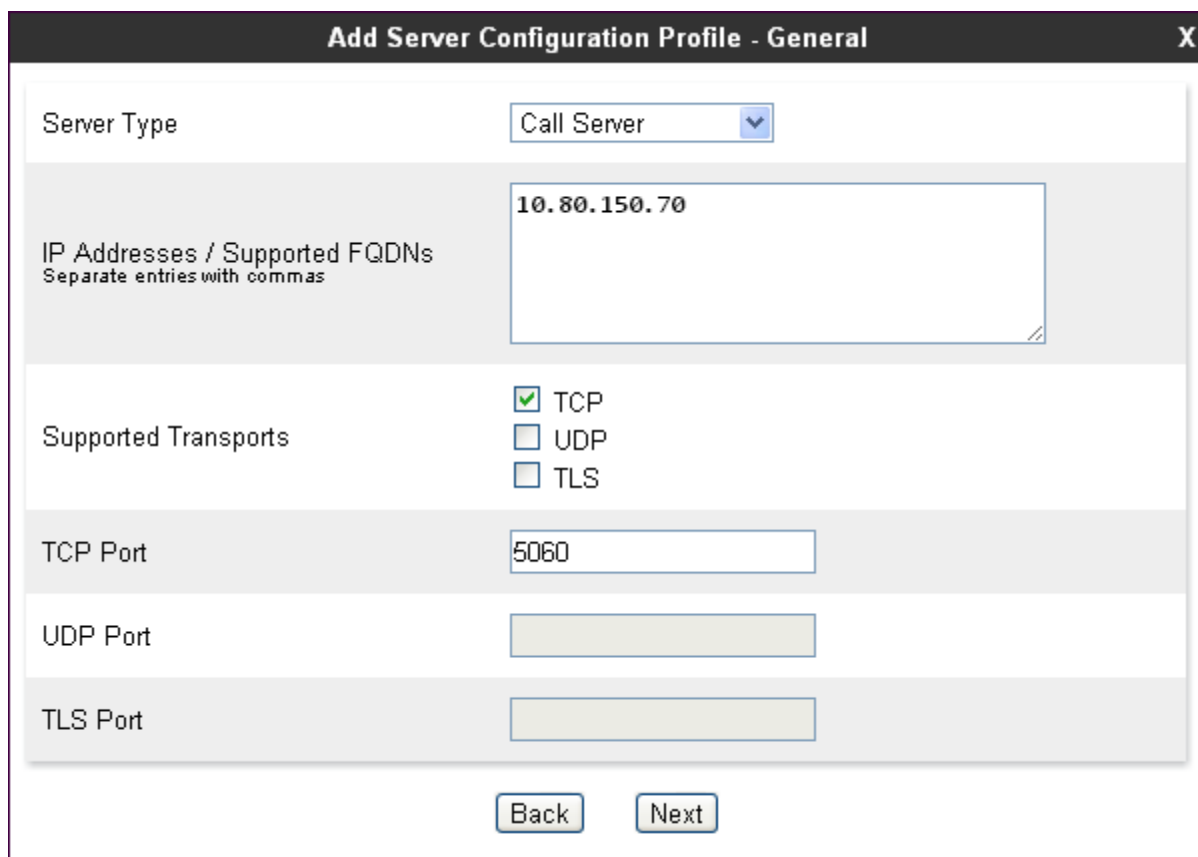
6.4.1. Server Configuration – IP Office

To add a Server Configuration Profile for IP Office, navigate to **Global Profiles → Server Configuration** and click **Add** (not shown). Enter a descriptive name for the **Profile Name** and click **Next**.



The screenshot shows a web interface for a Session Border Controller. A modal dialog titled "Add Server Configuration Profile" is open. It has a "Profile Name" text input field containing "IP Office" and a "Next" button. The background interface shows a sidebar with "Session Border Controller" and "Server Configuration" tabs, and a main area with "Add", "Rename", "Clone", and "Delete" buttons.

The following screens illustrate the Server Configuration for the Profile name “IP Office”. In the **General** parameters, select “Call Server” from the **Server Type** drop-down menu (not shown). In the **IP Addresses / Supported FQDNs** area, the IP Address of the IP Office LAN 1 interface in the sample configuration is entered. In the **Supported Transports** area, “TCP” is selected, and the **TCP Port** is set to “5060”. If adding a new profile, click **Next**. If editing an existing profile, click **Finish** (not shown).



The screenshot shows the "Add Server Configuration Profile - General" configuration screen. It includes the following fields and options:

- Server Type:** A dropdown menu set to "Call Server".
- IP Addresses / Supported FQDNs:** A text area containing "10.80.150.70". Below the text area is the instruction "Separate entries with commas".
- Supported Transports:** A section with three checkboxes: "TCP" (checked), "UDP" (unchecked), and "TLS" (unchecked).
- TCP Port:** A text input field containing "5060".
- UDP Port:** An empty text input field.
- TLS Port:** An empty text input field.
- Buttons:** "Back" and "Next" buttons at the bottom.

In the next two windows that appear, verify **Enable Authentication** and **Enable Heartbeat** are unchecked. IP Office does not require authentication and the Heartbeat feature is not necessary because Avaya SBCE will forward SIP OPTIONS from Verizon to the IP Office. Click **Next** to continue.

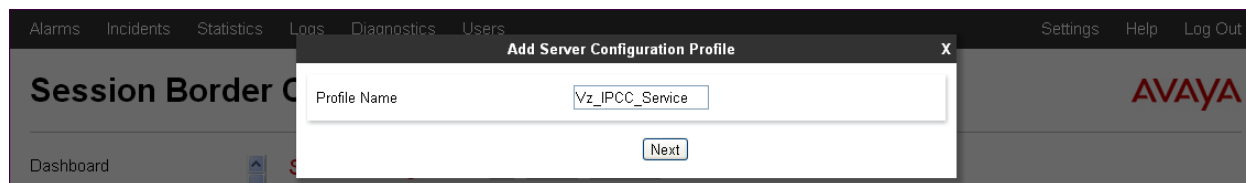
Add Server Configuration Profile - Authentication		Add Server Configuration Profile - Heartbeat	
Enable Authentication	<input type="checkbox"/>	Enable Heartbeat	<input type="checkbox"/>
User Name	<input type="text"/>	Method	<input type="text" value="OPTIONS"/>
Realm (Leave blank to detect from server challenge)	<input type="text"/>	Frequency	<input type="text" value="10"/> seconds
Password	<input type="text"/>	From URI	<input type="text"/>
Confirm Password	<input type="text"/>	To URI	<input type="text"/>
<input type="button" value="Back"/> <input type="button" value="Next"/>		<input type="button" value="Back"/> <input type="button" value="Next"/>	

In the new window that appears, select the **Interworking Profile** created for IP Office in Section 6.3.1. Use default values for all remaining fields. Click **Finish** to save the configuration.

Add Server Configuration Profile - Advanced	
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	<input type="text" value="IP Office Interwrk"/>
Signaling Manipulation Script	<input type="text" value="None"/>
TCP Connection Type	<input checked="" type="radio"/> SUBID <input type="radio"/> PORTID <input type="radio"/> MAPPING
<input type="button" value="Back"/> <input type="button" value="Finish"/>	

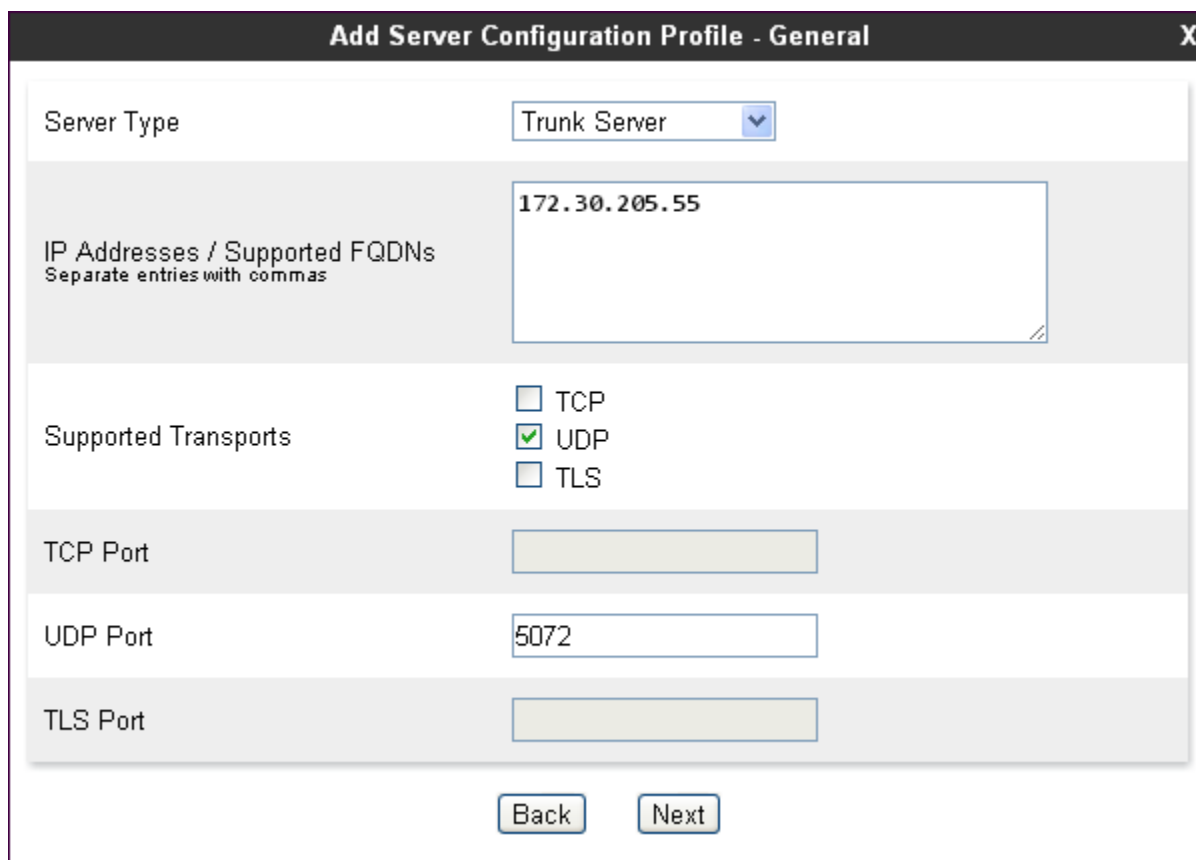
6.4.2. Server Configuration - Verizon

To add a Server Configuration Profile for Verizon, navigate to **Global Profiles → Server Configuration** and click **Add**. Enter a descriptive name for the **Profile Name** and click **Next**.



The screenshot shows a web interface with a top navigation bar containing 'Alarms', 'Incidents', 'Statistics', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main content area is titled 'Session Border Controller'. A modal dialog box titled 'Add Server Configuration Profile' is open, featuring a text input field for 'Profile Name' with the value 'Vz_IPCC_Service' and a 'Next' button.

The following screens illustrate the Server Configuration for the Profile name “Vz_IPCC_Service”. In the **General** parameters, select “Trunk Server” from the **Server Type** drop-down menu. In the **IP Addresses / Supported FQDNs** area, the Verizon-provided IP address is entered. In the sample configuration this is “172.30.205.55”. In the **Supported Transports** area, UDP is selected, and the **UDP Port** is set to “5072”. Click **Next** to continue.



The screenshot displays the 'Add Server Configuration Profile - General' configuration window. It includes the following fields and options:

- Server Type:** A dropdown menu set to 'Trunk Server'.
- IP Addresses / Supported FQDNs:** A text area containing '172.30.205.55' with the instruction 'Separate entries with commas'.
- Supported Transports:** Three checkboxes: 'TCP' (unchecked), 'UDP' (checked), and 'TLS' (unchecked).
- TCP Port:** An empty text input field.
- UDP Port:** A text input field containing '5072'.
- TLS Port:** An empty text input field.

At the bottom of the window are 'Back' and 'Next' buttons.

Verify **Enable Authentication** is unchecked as Verizon does not require authentication. Click **Next** to continue.

The screenshot shows a dialog box titled "Add Server Configuration Profile - Authentication" with a close button (X) in the top right corner. The dialog contains the following fields and controls:

- Enable Authentication:** A checkbox that is currently unchecked.
- User Name:** A text input field.
- Realm:** A text input field with the placeholder text "(Leave blank to detect from server challenge)".
- Password:** A text input field.
- Confirm Password:** A text input field.
- Navigation:** "Back" and "Next" buttons at the bottom.

In the new window that appears, check the **Enable Heartbeat** box. Select "OPTIONS" from the **Method** drop-down menu. Select the desired **Frequency** that the SBC will source OPTIONS. The **From URI** and **TO URI** may be filled in to configure easily identifiable URIs to appear in SIP OPTIONS sourced by the SBC. Click **Next** to continue.

The screenshot shows a dialog box titled "Add Server Configuration Profile - Heartbeat" with a close button (X) in the top right corner. The dialog contains the following fields and controls:

- Enable Heartbeat:** A checkbox that is currently checked.
- Method:** A drop-down menu with "OPTIONS" selected.
- Frequency:** A text input field containing "60", followed by the unit "seconds".
- From URI:** A text input field containing "ping@1.1.1.2".
- To URI:** A text input field containing "ping@172.30.205.55".
- Navigation:** "Back" and "Next" buttons at the bottom.

In the new window that appears, select the **Interworking Profile** “Vz-IPCC-interwrk” created previously in Section 6.3.2. Use default values for all remaining fields. Click **Finish** to save the configuration.

Add Server Configuration Profile - Advanced

Enable DoS Protection ☐

Enable Grooming ☐

Interworking Profile Vz-IPCC-Interwrk

Signaling Manipulation Script None

TCP Connection Type ☒ SUBID ☐ PORTID ☐ MAPPING

Back Finish

6.5. Media Rule

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

Select **Domain Policies** → **Media Rules** from the left-side menu as shown below. In the sample configuration, a single default media rule “default-low-med” was used with the **Audio and Video DSCP** values “EF” (Expedited Forwarding) set for **Media QoS** as shown below.

Session Border Controller for Enterprise AVAYA

Dashboard
Administration
Backup/Restore
System Management
Global Parameters
Global Profiles
SIP Cluster
Domain Policies
Application Rules
Border Rules
Media Rules
Security Rules
Signaling Rules
Time of Day Rules
End Point Policy Groups
Session Policies
TLS Management
Device Specific Settings
Network Management

Media Rules: default-low-med

Add Filter By Device... Clone

It is not recommended to edit the defaults. Try cloning or adding a new rule instead.

Media NAT Media Encryption Media Anomaly Media Silencing **Media QoS**

Media QoS Reporting

RTCP Enabled ☐

Media QoS Marking

Enabled ☒

QoS Type DSCP

Audio QoS

Audio DSCP EF

Video QoS

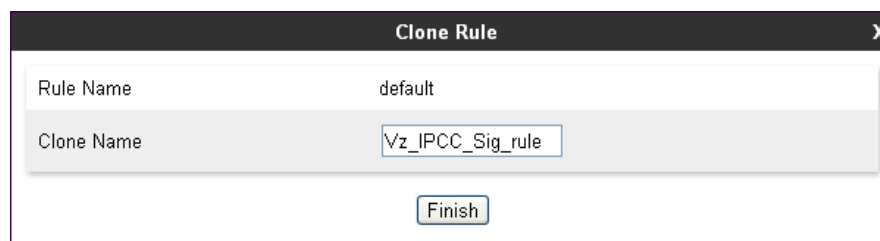
Video DSCP EF

Edit

6.6. Signaling Rule

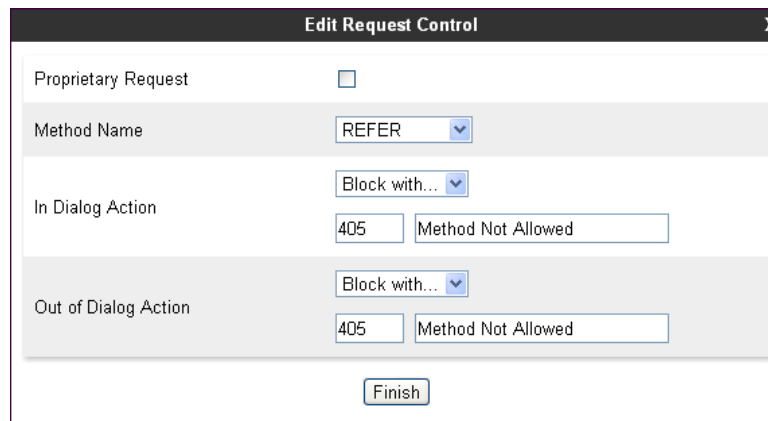
Signaling Rules define the action to be taken (Allow, Block, Block with Response, etc.) for each type of SIP-specific signaling request and response message. When SIP signaling packets are received by Avaya SBCE, they are parsed and “pattern-matched” against the particular signaling criteria defined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

Clone and modify the default signaling rule to block REFER messages from IP Office to Verizon IPCC and to add the proper quality of service to the SIP signaling. To clone a signaling rule, navigate to **Domain Policies → Signaling Rules**. With the **default** rule chosen, click **Clone** (not shown). Enter a descriptive name for the new rule and click **Finish**.



The 'Clone Rule' dialog box has a title bar with 'Clone Rule' and a close button 'X'. It contains two input fields: 'Rule Name' with the value 'default' and 'Clone Name' with the value 'Vz_IPCC_Sig_rule'. At the bottom is a 'Finish' button.

Select the **Requests** tab, and select the **Add Out Request Control** button (not shown). Select “REFER” as the **Method Name**. In the **In Dialog Action** and **Out of Dialog Action**, select “Blocks with...” and type “405” and “Method Not Allowed” and click **Finish**. The intent is to have the SBC return a “405 Method Not Allowed” response whenever a REFER is sent to Verizon IPCC from IP Office. See Section 2.2 for additional information.



The 'Edit Request Control' dialog box has a title bar with 'Edit Request Control' and a close button 'X'. It contains several fields: 'Proprietary Request' with an unchecked checkbox, 'Method Name' with a dropdown menu showing 'REFER', 'In Dialog Action' with a 'Block with...' dropdown, a text input '405', and another text input 'Method Not Allowed'. The 'Out of Dialog Action' section has identical controls. At the bottom is a 'Finish' button.

Once complete, the **Requests** tab appears as follows.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left sidebar lists navigation options: SIP Cluster, Domain Policies (Application Rules, Border Rules, Media Rules, Security Rules, **Signaling Rules**, Time of Day Rules, End Point Policy Groups, Session Policies), and Time of Day Rules. The main content area is titled 'Signaling Rules: Vz_IPCC_Sig_rule'. It includes an 'Add' button, a 'Filter By Device...' dropdown, and buttons for 'Rename', 'Clone', and 'Delete'. Below these is a description field with the text 'Click here to add a description.' The 'Requests' tab is selected, showing a table with the following data:

Row	Method Name	In Dialog Action	Out of Dialog Action	Proprietary	Direction	
1	REFER	Block with "405 Method Not Allowed"	Block with "405 Method Not Allowed"	No	Out	Edit Delete

The following screen shows the **Signaling QoS** set with the DSCP value “AF32” for assured forwarding.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left sidebar lists navigation options: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, SIP Cluster, Domain Policies (Application Rules, Border Rules, Media Rules, Security Rules, **Signaling Rules**, Time of Day Rules), and Time of Day Rules. The main content area is titled 'Signaling Rules: Vz_IPCC_Sig_rule'. It includes an 'Add' button, a 'Filter By Device...' dropdown, and buttons for 'Rename', 'Clone', and 'Delete'. Below these is a description field with the text 'Click here to add a description.' The 'Signaling QoS' tab is selected, showing a table with the following data:

QoS Type	DSCP
DSCP	AF32

6.7. Application Rule

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

Select **Domain Policies** → **Application Rules** from the left-side menu as shown below. In the sample configuration, a single default application rule “default-trunk” was used and will be applied to the Endpoint Policy Group in the next section.

Session Border Controller for Enterprise AVAYA

Dashboard
Administration
Backup/Restore
System Management
Global Parameters
Global Profiles
SIP Cluster
Domain Policies
Application Rules
Border Rules
Media Rules
Security Rules
Signaling Rules
Time of Day Rules
End Point Policy Groups
Session Policies

Application Rules: default-trunk

Add Filter By Device... Clone

It is not recommended to edit the defaults. Try cloning or adding a new rule instead.

Application Rule

Application Type	In	Out	Maximum Concurrent Sessions	Maximum Sessions Per Endpoint
Voice	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	2000	2000
Video	<input type="checkbox"/>	<input type="checkbox"/>		
IM	<input type="checkbox"/>	<input type="checkbox"/>		

Miscellaneous

CDR Support	None
RTCP Keep-Alive	No

Edit

6.8. Endpoint Policy Groups

The rules created within the Domain Policy section are assigned to an Endpoint Policy Group. The Endpoint Policy Group is then applied to a Server Flow in Section 6.11.

To create a new policy group, navigate to **Domain Policies** → **Endpoint Policy Groups** and click on **Add** as shown below. In the sample configuration “SIP-Trunk-Policy” was created for IP Office using defaults selected for all fields, with the exception of **Application** set to “default-trunk” as shown below. The details of the non-default rules chosen are shown in previous sections.

Session Border Controller for Enterprise AVAYA

Alarms Incidents Statistics Logs Diagnostics Users Settings Help Log Out

Backup/Restore
System Management
Global Parameters
Global Profiles
SIP Cluster
Domain Policies
Application Rules
Border Rules
Media Rules
Security Rules
Signaling Rules
Time of Day Rules
End Point Policy Groups
Session Policies
TLS Management
Device Specific Settings

Policy Groups: SIP-Trunk-Policy

Add Filter By Device... Rename Delete

Click here to add a description.

Click here to add a row description.

Policy Group

Order	Application	Border	Media	Security	Signaling	Time of Day
1	default-trunk	default	default-low-med	default-low	default	default

Summary Add Edit Clone

Similarly, a separate profile named “Vz-IPCC-Policy” was created for Verizon Business IP Contact Center SIP Trunk service using defaults selected for all fields, with the exception of **Application** set to “default-trunk” and **Signaling** set to “Vz_IPCC_Sig_rule” as shown below.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left sidebar contains a navigation menu with options like Dashboard, Administration, Backup/Restore, System Management, and Domain Policies. The main content area is titled "Policy Groups: Vz-IPCC-Policy". It features a list of policy groups on the left, including default-low, default-low-enc, default-med, default-med-enc, default-high, default-high-enc, OCS-default-high, avaya-def-low-enc, SIP-Trunk-Policy, and Vz-IPCC-Policy. The Vz-IPCC-Policy group is selected. The right pane shows the configuration for this group, including a table with columns: Order, Application, Border, Media, Security, Signaling, and Time of Day. The table contains one row with the following values: 1, default-trunk, default, default-low-med, default-low, Vz_IPCC_Sig_rule, and default. There are buttons for Edit and Clone next to the row.

6.9. Media Interface

The Media Interface screen is where the SIP media ports are defined. Avaya SBCE will send SIP media on the defined ports. Create a SIP media interface for the inside and outside IP interfaces.

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

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left sidebar contains a navigation menu with options like Dashboard, Administration, Backup/Restore, System Management, and Device Specific Settings. The main content area is titled "Media Interface: Micro SBC". It features a list of media interfaces on the left, including Micro SBC. The right pane shows the configuration for this interface, including a table with columns: Name, Media IP, and Port Range. The table contains two rows with the following values: Media_to_IPO, 10.64.19.199, 35000 - 40000; and Media_to_Vz, 1.1.1.2, 35000 - 40000. There are buttons for Edit and Delete next to each row. A warning message at the top states: "Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be issued from System Management."

After the media interfaces are created, an application restart is necessary before the changes will take effect. Navigate to **System Management** and click **Restart Application** as highlighted below.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes Alarms, Incidents, Statistics, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header displays "Session Border Controller for Enterprise" and the Avaya logo. On the left, a sidebar menu lists various sections: Administration, Backup/Restore, System Management (highlighted), Global Parameters, Global Profiles, SIP Cluster, Domain Policies, TLS Management, and Device Specific Settings. The main content area is titled "System Management" and contains tabs for Devices, Updates, SSL VPN, and Licensing. Under the "Devices" tab, a table lists device information:

Device Name (Serial Number)	Management IP	Version	Status	Reboot	Shutdown	Restart Application	View	Edit	Delete
Micro SBC (PCS11099999)	10.80.150.199	6.2.0.Q33	Commissioned						

The "Restart Application" button for the Micro SBC is highlighted with a red box.

6.10. Signaling Interface

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

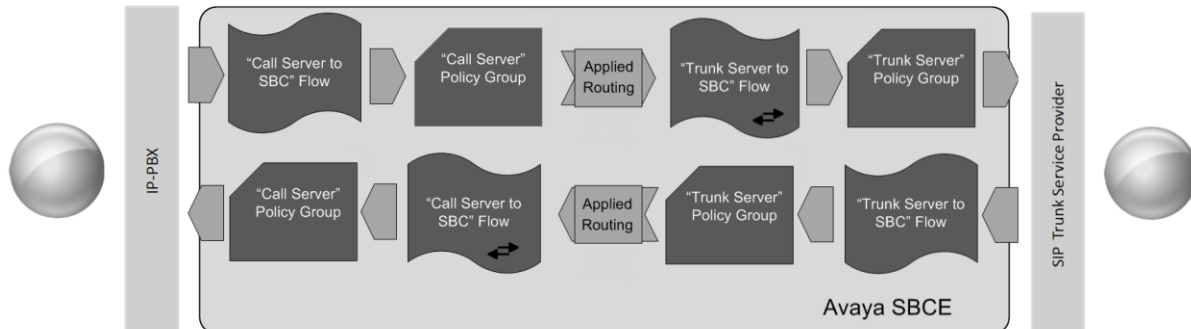
To create a new Signaling Interface, navigate to **Device Specific Settings** → **Signaling Interface** and click **Add**. The following screen shows the signaling interfaces defined for the sample configuration.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface, specifically the "Signaling Interface: Micro SBC" configuration page. The top navigation bar and header are the same as in the previous screenshot. The sidebar menu is expanded to show "Device Specific Settings" and "Signaling Interface" is highlighted. The main content area is titled "Signaling Interface: Micro SBC" and contains a tab for "Signaling Interface". Below the tab, there is an "Add" button and a table listing the configured signaling interfaces:

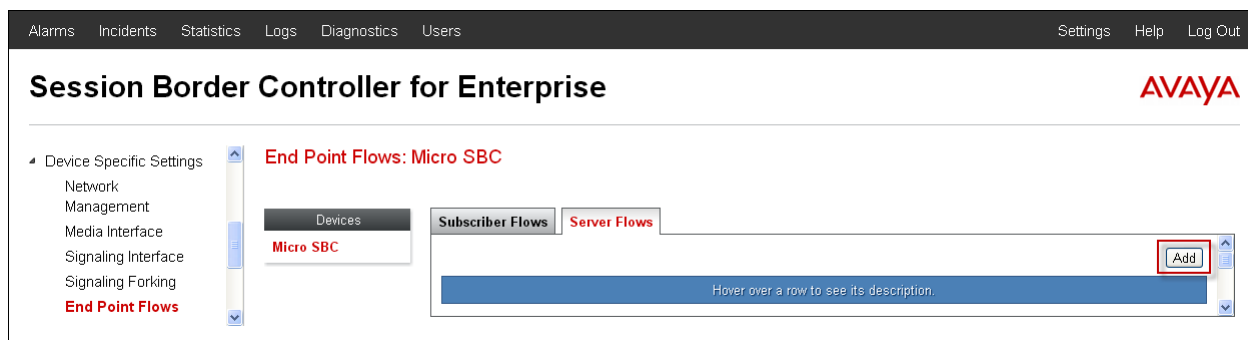
Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile	Edit	Delete
Sig_to_IPO	10.64.19.199	5060	---	---	None		
Sig_to_Vz	1.1.1.2	---	5060	---	None		

6.11. End Point Flows - Server Flow

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



To create a Server Flow for IP Office and Verizon Business IP Contact Center SIP Trunk service, navigate to **Device Specific Settings** → **End Point Flows**. Select the **Server Flows** tab and click **Add** as highlighted below.



The following screen shows the flow named “Vz-IPCC Flow” configured in the sample configuration. This flow uses the interfaces, policies, and profiles defined in previous sections. Click **Finish**.

Edit Flow: IPCC FlowX

Flow Name	Vz-IPCC Flow
Server Configuration	Vz_IPCC_Service
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Sig_to_IPO
Signaling Interface	Sig_to_Vz
Media Interface	Media_to_Vz
End Point Policy Group	Vz-IPCC-Policy
Routing Profile	IP Office
Topology Hiding Profile	default
File Transfer Profile	None

Finish

Similarly, “IP Office Flow” was configured in this sample configuration as shown below.

Edit Flow: IP Office Flow

Flow Name

IP Office Flow

Server Configuration

IP Office

URI Group

*

Transport

*

Remote Subnet

*

Received Interface

Sig_to_Vz

Signaling Interface

Sig_to_IPO

Media Interface

Media_to_IPO

End Point Policy Group

SIP-Trunk-Policy

Routing Profile

Vz_IPT

Topology Hiding Profile

default

File Transfer Profile

None

Finish

7. Verizon Business Configuration

Information regarding Verizon Business IP Contact Center service offer can be found by contacting a Verizon Business sales representative, or by visiting <http://www.verizonbusiness.com/Products/communications/contact-center/>

The configuration described in these Application Notes was located in the Avaya Solutions and Interoperability Lab. The Verizon Business IP Contact Center service was accessed via a Verizon Private IP (PIP) T1 connection as described in Section 1. Verizon Business provided the necessary service provisioning, which included the domain *adevc.avaya.globalipcom.com* for the Avaya IP Office location.

For service provisioning, Verizon will require the customer IP address of the Avaya Session Border Controller for Enterprise. Verizon provided the following information for the compliance testing: the IP address and port used by the Verizon SBC, and the toll-free numbers shown in **Figure 1** and **Table 1**. This information was used to complete the configuration for Avaya Session Border Controller for Enterprise shown in Section 6 and the IP Office shown in Section 5.

8. Verification

This section provides example verifications of the Avaya configuration with Verizon Business IP Contact Center service.

8.1. Illustration of OPTIONS Handling

The following screens from a filtered Wireshark trace illustrate OPTIONS sent by Verizon to the CPE. Verizon IPCC service uses OPTIONS to determine whether the CPE is available to receive inbound calls. Therefore, proper OPTIONS response is necessary. In the trace shown below, taken from the outside interface of the Avaya SBCE, frame 53 is highlighted and expanded to show OPTIONS sent from Verizon IPCC Trunk (172.30.205.55) to the SBC (1.1.1.2). Observe the use of UDP for transport, from source port 5072 (Verizon) to destination port 5060 (Avaya). Verizon sends the FQDN “adevc.avaya.globalipcom.com” in the Request-Line. Note that Max-Forwards is 70.

Filter: sip.tag == "0c4735ba69561f209fa491a5ab74731e000cek3" Expression... Clear Apply					
No.	Time	Source	Destination	Protocol	Info
53	47.620487	172.30.205.55	1.1.1.2	SIP	Request: OPTIONS sip:adevc.avaya.globalipcom.com:5060
54	47.631377	1.1.1.2	172.30.205.55	SIP/SDP	Status: 200 OK, with session description

Frame 53: 414 bytes on wire (3312 bits), 414 bytes captured (3312 bits)
Ethernet II, Src: Cisco_5c:21:41 (00:04:9a:5c:21:41), Dst: Portwell_34:5b:c4 (00:90:fb:34:5b:c4)
Internet Protocol, Src: 172.30.205.55 (172.30.205.55), Dst: 1.1.1.2 (1.1.1.2)
User Datagram Protocol, Src Port: ayiya (5072), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: OPTIONS sip:adevc.avaya.globalipcom.com:5060 SIP/2.0
Message Header
Via: SIP/2.0/UDP 172.30.205.55:5072;branch=z9hG4bKdbs7a330cg102ssnu2p0
Call-ID: c2d969ea76c64f9eeda03954f3f206f7000cek3@172.30.205.55
To: sip:ping@c800026409-pcs-n0001
From: <sip:ping@172.30.205.55>;tag=0c4735ba69561f209fa491a5ab74731e000cek3
Max-Forwards: 70
CSeq: 59112 OPTIONS
Route: <sip:1.1.1.2:5060;lr>

Before Avaya SBCE replies to Verizon, the SBC sends OPTIONS to IP Office on the inside interface. In the trace shown below, taken from the inside interface of the SBC, frame 97 is highlighted and expanded to show OPTIONS sent from the inside interface of the SBC (10.64.19.199) to IP Office (10.80.150.70). Note that Max-Forwards header has been decremented by 1 and is now 69.

Filter: sip.tag == "0c4735ba69561f209fa491a5ab74731e000cek3" Expression... Clear Apply					
No.	Time	Source	Destination	Protocol	Info
97	53.454971	10.64.19.199	10.80.150.70	SIP	Request: OPTIONS sip:10.80.150.70
100	53.461667	10.80.150.70	10.64.19.199	SIP/SDP	Status: 200 OK, with session description

Frame 97: 449 bytes on wire (3592 bits), 449 bytes captured (3592 bits)
Ethernet II, Src: Portwell_34:5b:c6 (00:90:fb:34:5b:c6), Dst: Avaya_a3:a2:1c (90:fb:5b:a3:a2:1c)
Internet Protocol, Src: 10.64.19.199 (10.64.19.199), Dst: 10.80.150.70 (10.80.150.70)
Transmission Control Protocol, Src Port: 14268 (14268), Dst Port: sip (5060), Seq: 2, Ack: 1, Len: 395
Session Initiation Protocol
Request-Line: OPTIONS sip:10.80.150.70 SIP/2.0
Message Header
From: <sip:ping@10.64.19.199:14268>;tag=0c4735ba69561f209fa491a5ab74731e000cek3
To: sip:ping@10.80.150.70
CSeq: 59112 OPTIONS
Call-ID: 36af1121eadee96ed447dc418932ea02
Record-Route: <sip:10.64.19.199:5060;ipcs-line=223;lr;transport=tcp>
Max-Forwards: 69
Via: SIP/2.0/TCP 10.64.19.199:5060;branch=z9hG4bK-s1632-000183850527-1--s1632-
Content-Length: 0

In this same trace, highlighted frame 100 below shows IP Office responding to the OPTIONS with 200 OK.

Filter: sip.tag == "0c4735ba69561f209fa491a5ab74731e000cek3" Expression... Clear Apply					
No.	Time	Source	Destination	Protocol	Info
97	53.454971	10.64.19.199	10.80.150.70	SIP	Request: OPTIONS sip:10.80.150.70
100	53.461667	10.80.150.70	10.64.19.199	SIP/SDP	Status: 200 OK, with session description

Frame 100: 841 bytes on wire (6728 bits), 841 bytes captured (6728 bits)
Ethernet II, Src: Avaya_a3:a2:1c (90:fb:5b:a3:a2:1c), Dst: Portwell_34:5b:c6 (00:90:fb:34:5b:c6)
Internet Protocol, Src: 10.80.150.70 (10.80.150.70), Dst: 10.64.19.199 (10.64.19.199)
Transmission Control Protocol, Src Port: sip (5060), Dst Port: 14268 (14268), Seq: 1, Ack: 397, Len: 787
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Message Header
Via: SIP/2.0/TCP 10.64.19.199:5060;branch=z9hG4bK-s1632-000183850527-1--s1632-
Record-Route: <sip:10.64.19.199:5060;ipcs-line=223;lr;transport=tcp>
From: <sip:ping@10.64.19.199:14268>;tag=0c4735ba69561f209fa491a5ab74731e000cek3
To: <sip:ping@10.80.150.70>;tag=97621777e3d25fa8
Call-ID: 36af1121eadee96ed447dc418932ea02
CSeq: 59112 OPTIONS
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
Supported: timer
Server: IP Office 8.1 (65)
Content-Type: application/sdp
Content-Length: 255
Message Body

Returning to the outside trace, and advancing to frame 54, the 200 OK sent back to the inbound OPTIONS from Verizon is illustrated below. The receipt of a valid OPTIONS response from the CPE is necessary for Verizon to route inbound calls to the CPE. Since the SBC proxies the OPTIONS received from Verizon to IP Office, the end to end path from Verizon through to IP Office must be in service for OPTIONS (and ultimately calls) to be successful.

Filter: sip.tag == "0c4735ba69561f209fa491a5ab74731e000cek3" Expression... Clear Apply					
No.	Time	Source	Destination	Protocol	Info
53	47.620487	172.30.205.55	1.1.1.2	SIP	Request: OPTIONS sip:adevc.avaya.globalipcom.com:5060
54	47.631377	1.1.1.2	172.30.205.55	SIP/SDP	Status: 200 OK, with session description

+	Frame 54: 836 bytes on wire (6688 bits), 836 bytes captured (6688 bits)
+	Ethernet II, Src: Portwell_34:5b:c4 (00:90:fb:34:5b:c4), Dst: Cisco_5c:21:41 (00:04:9a:5c:21:41)
+	Internet Protocol, Src: 1.1.1.2 (1.1.1.2), Dst: 172.30.205.55 (172.30.205.55)
+	User Datagram Protocol, Src Port: sip (5060), Dst Port: ayiya (5072)
+	Session Initiation Protocol
+	Status-Line: SIP/2.0 200 OK
+	Message Header
+	From: <sip:ping@172.30.205.55>;tag=0c4735ba69561f209fa491a5ab74731e000cek3
+	To: <sip:ping@c800026409-pcs-n0001>;tag=97621777e3d25fa8
+	CSeq: 59112 OPTIONS
+	Call-ID: c2d969ea76c64f9eeda03954f3f206f7000cek3@172.30.205.55
+	Record-Route: <sip:1.1.1.2:5060;ipcs-line=223;lr;transport=udp>
+	Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
+	Supported: timer
+	Via: SIP/2.0/UDP 172.30.205.55:5072;branch=z9hG4bKdbs7a330cg102ssnu2p0
+	Server: IP Office 8.1 (65)
+	Content-Type: application/sdp
+	Content-Length: 251
+	Message Body

8.2. Avaya SBCE

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

8.2.1. Incidents

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

Alarms

Incidents

Statistics

Logs

Diagnostics

Users

Settings

Help

Log Out

Session Border Controller for Enterprise

AVAYA

Dashboard

Administration

Backup/Restore

Dashboard

Information

System Time

04:11:00 PM MST

Refresh

Installed Devices

EMS

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

Incident Viewer

AVAYA

Device

All

Category

All

Clear

Refresh

Generate Report

Displaying results 61 to 75 out of 84.

Type	ID	Date	Time	Category	Device	Cause
Routing Failure	680296395608192	2/11/13	7:26 AM	Policy	Micro SBC	Target is neither a server nor a subscriber, Sending 403 Forbidden
Server Heartbeat	680073964826219	2/6/13	3:52 AM	Policy	Micro SBC	Heartbeat Successfull, Server is UP
Server Heartbeat	680073937294193	2/6/13	3:51 AM	Policy	Micro SBC	Heartbeat Failed, Server is Down
Server Heartbeat	680039634906183	2/5/13	8:47 AM	Policy	Micro SBC	Heartbeat Failed, Server is Down

8.2.2. Tracing

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

Alarms	Incidents	Statistics	Logs	Diagnostics	Users	Settings	Help	Log Out					
Session Border Controller for Enterprise							AVAYA						
Trace: Micro SBC													
Devices		Call Trace		Packet Capture		Captures							
Micro SBC		Packet Capture Configuration											
		Status		Ready									
		Interface		A1									
		Local Address IP:Port		All :									
		Remote Address *, *.Port, IP, IP:Port		*									
		Protocol		UDP									
		Maximum Number of Packets to Capture		1000									
		Capture Filename Using the name of an existing capture will overwrite it.		TC56_DSCP_test.pcap									
				Start Capture Clear									

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

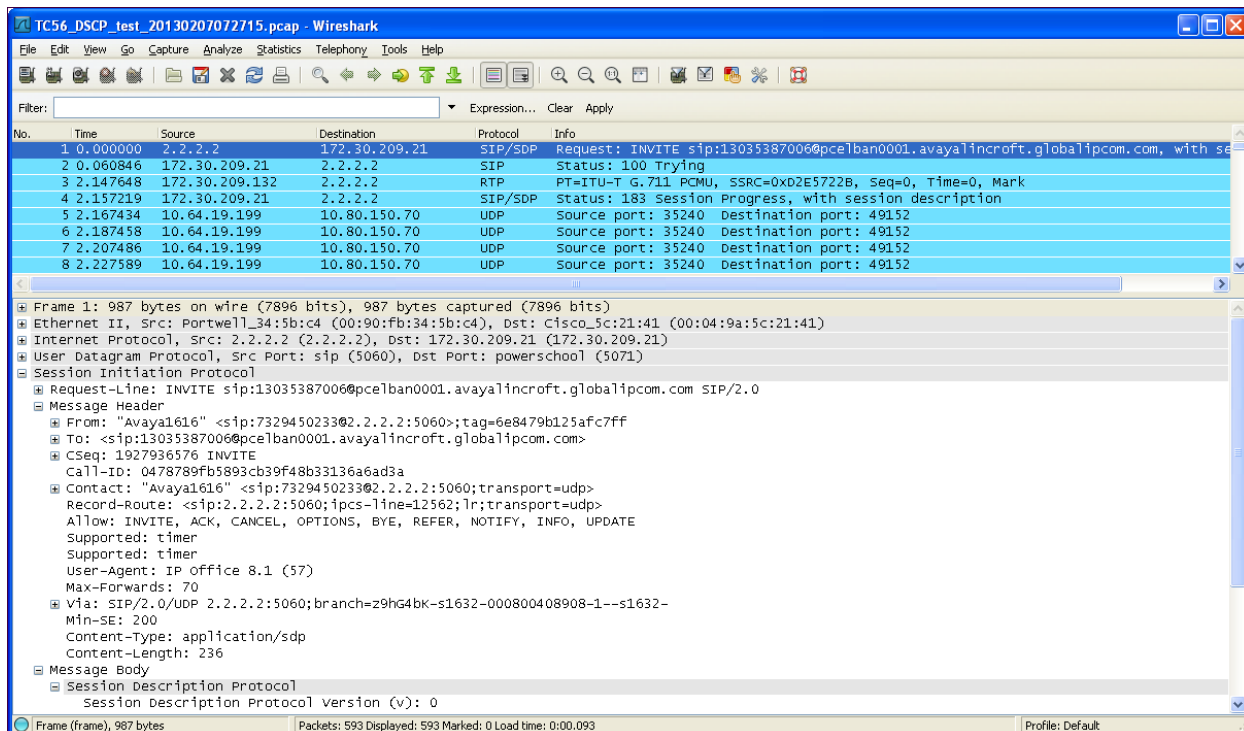
The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes Alarms, Incidents, Statistics, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header displays "Session Border Controller for Enterprise" and the Avaya logo. On the left, a sidebar menu lists various configuration options under "Device Specific Settings", with "Trace" highlighted in red. The main content area is titled "Trace: Micro SBC" and features three tabs: "Call Trace", "Packet Capture", and "Captures". The "Packet Capture" tab is active, showing a configuration form. A blue banner at the top of the form states: "A packet capture is currently in progress. This page will automatically refresh until the capture completes." The configuration form includes fields for Interface (A1), Local Address (All), Remote Address (*.*.*.*), Protocol (UDP), Maximum Number of Packets to Capture (1000), and Capture Filename (TC56_DSCP_test.pcap). A "Stop Capture" button is located at the bottom right of the form.

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

The screenshot shows the same Avaya Session Border Controller for Enterprise web interface, but with the "Captures" tab selected. The sidebar menu remains the same, with "Trace" highlighted. The main content area is titled "Trace: Micro SBC" and features three tabs: "Call Trace", "Packet Capture", and "Captures". The "Captures" tab is active, displaying a table of captured files. Above the table, there are controls for sorting (Last Modified, Descending, Sort, Reset) and a Refresh button. The table has three columns: File Name, File Size (bytes), and Last Modified. Two files are listed: "TC56_DSCP_test_20130207072715.pcap" (139,264 bytes, February 7, 2013 7:27:50 AM MST) and "test-trace_20130204084632.pcap" (4,096 bytes, February 4, 2013 8:47:00 AM MST). Each file has a "Delete" link next to it.

File Name	File Size (bytes)	Last Modified	
TC56_DSCP_test_20130207072715.pcap	139,264	February 7, 2013 7:27:50 AM MST	Delete
test-trace_20130204084632.pcap	4,096	February 4, 2013 8:47:00 AM MST	Delete

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



8.3. IP Office

This section provides verification steps that may be performed with the IP Office.

8.3.1. System Status

The System Status application is used to monitor and troubleshoot IP Office. Use the System Status application to verify the state of the SIP trunk. System Status can be accessed from **Start → Programs → IP Office → System Status** or by opening an Internet browser and type the URL: <http://ipaddress> where *ipaddress* is the IP address of the Avaya IP Office LAN1 interface. See reference [4] for more information. Click on **System Status** to launch the application.



The following screen shows an example **Logon** screen. Enter the IP Office IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.

AVAYA IP Office System Status

Help About

Online Offline

Logon

Control Unit IP Address: 10.80.150.70

Services Base TCP Port: 50804

Local IP Address: Automatic

User Name: Administrator

Password: ●●●●●●●●●●

☒ Auto reconnect

Logon

IP Office System Status Version 8.1(65)

Select the SIP line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current State** is **Idle** for each channel.

AVAYA IP Office System Status

Help Snapshot LogOff About

System
Alarms (3)
Extensions (18)
Trunks (6)
Active Calls
Resources
Voicemail
IP Networking

Lines: 5 - 8
Line: 17
Line: 20

Status Utilization Summary Alarms

SIP Trunk Summary

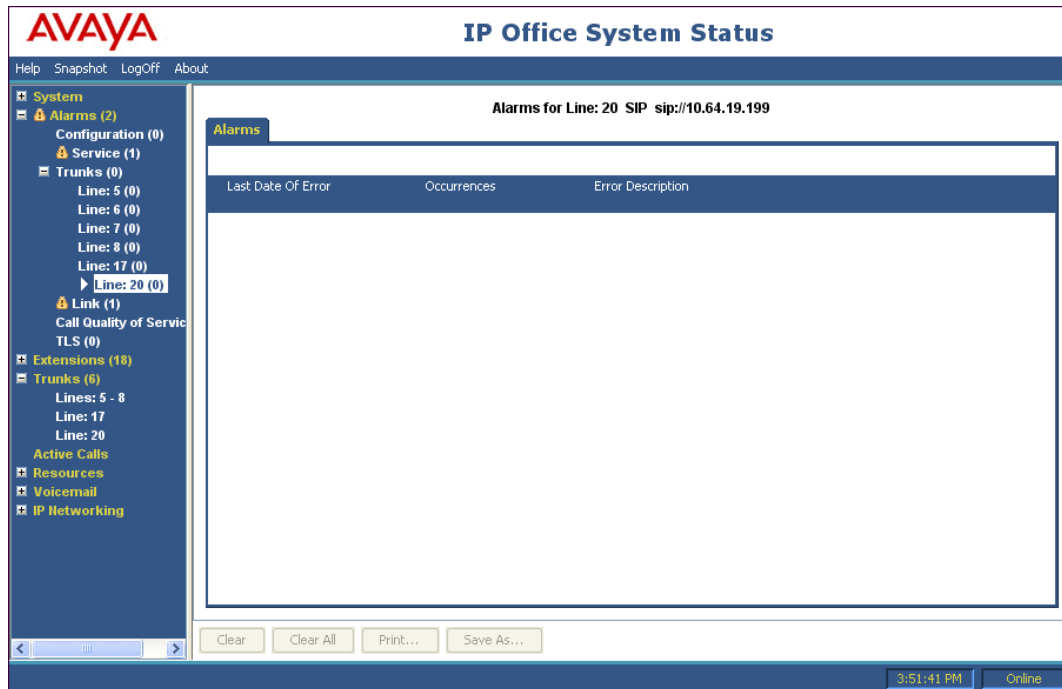
Peer Domain Name: sip://10.64.19.199
Resolved Address: 10.64.19.199
Line Number: 20
Number of Administered Channels: 30
Number of Channels in Use: 0
Administered Compression: G722, G729 A, G711 Mu
Silence Suppression: Off
SIP Trunk Channel Licenses: 5
SIP Trunk Channel Licenses in Use: 0
SIP Device Features: REFER (Incoming and Outgoing)

Channel Number	U...	Call Ref	Current State	Time in State	Remote Media ...	Co...	Conn...	Caller ID or ...	Other Party on Call	Directi...	Round Trip ...	Receive Jitter	Receive Pack...	Trans...	Trans...
1			Idle	05:16...											
2			Idle	07:15...											
3			Idle	07:15...											
4			Idle	07:15...											
5			Idle	07:15...											
6			Idle	07:15...											
7			Idle	07:15...											
8			Idle	07:15...											
9			Idle	07:15...											

Trace Trace All Pause Ping Call Details Print... Save As...

3:50:09 PM Online

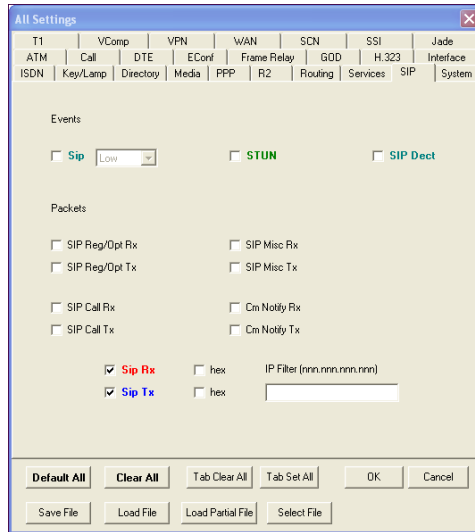
Select the **Alarms** tab and verify that no alarms are active on the SIP line.



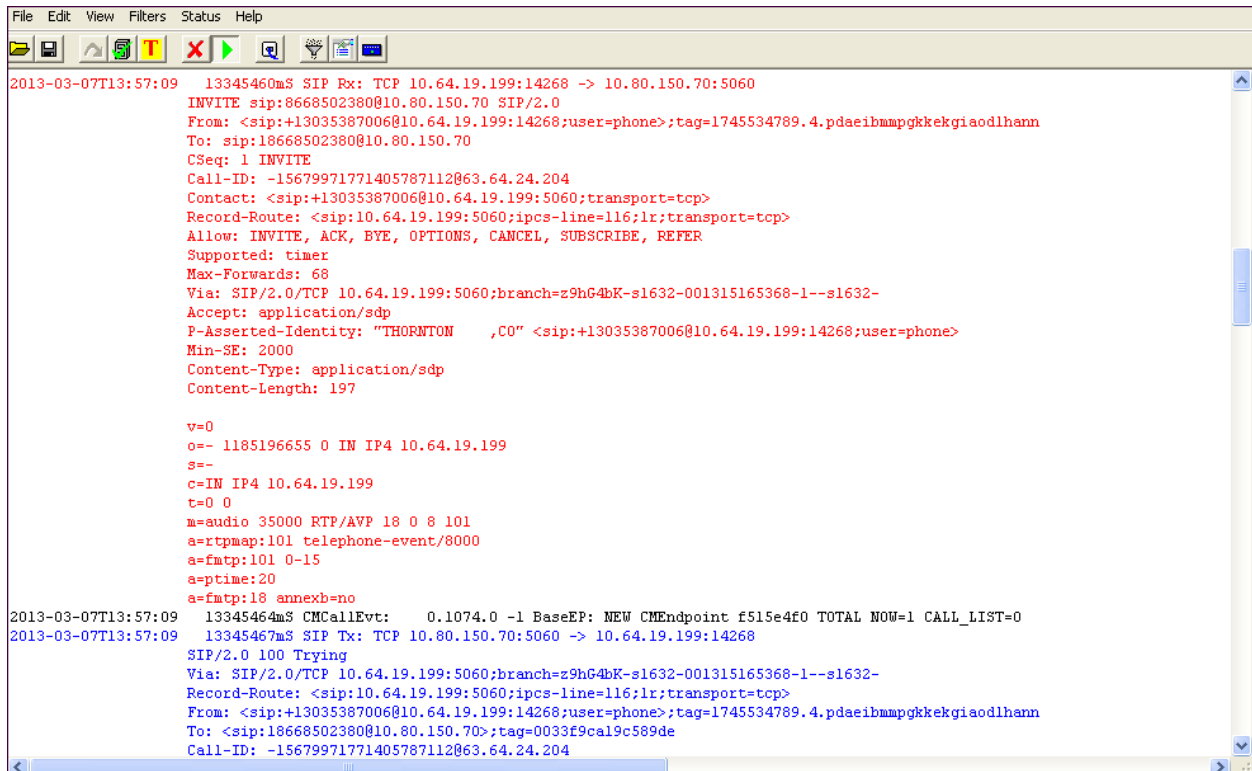
8.3.2. System Monitor

The System Monitor application can also be used to monitor and troubleshoot IP Office. System Monitor can be accessed from **Start → Programs → IP Office → Monitor**. The application allows the monitored information to be customized. To customize, select the button that is third from the right in the screen below, or select **Filters → Trace Options** (not shown).

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.



The following screen shows a portion of the monitor trace of an inbound call. As can be observed, PSTN caller 303-538-7006 dialed Verizon IP Toll Free number 1-866-850-2380. Details of the SIP INVITE message sent by Verizon are shown below. This information matches the configuration in these Application Notes and is not intended to be prescriptive. The intent is to illustrate the INVITE sent by Verizon in the sample configuration, along with the means to retrieve this type of trace information from IP Office.



9. Conclusion

IP Office is a highly modular IP telephone system designed to meet the needs of home offices, standalone businesses, and networked offices for small and medium enterprises.

These Application Notes demonstrated how Avaya Session Border Controller for Enterprise Release 6.2 and IP Office Release 8.1 can be successfully combined with a Verizon Business IP Contact Center SIP Trunk service connection to enable a business to receive toll-free calls. Utilizing this solution, IP Office customers can leverage the operational efficiencies and cost savings associated with SIP trunking while gaining the advanced technical features provided through the marriage of best of breed technologies from Avaya and Verizon.

IP Office Release 8.1 with Avaya Session Border Controller for Enterprise Release 6.2 has not been independently certified by Verizon labs. These Application Notes can be used to facilitate customer engagements via the Verizon field trial process, pending Verizon labs independent certification.

10. Additional References

This section references documentation relevant to these Application Notes. In general, Avaya product documentation is available at <http://support.avaya.com>

- [1] *IP Office 8.1 Installation Manual*, Document Number 15-601042, August 2012
- [2] *IP Office Manager Manual 10.0*, Document Number 15-601011, August 2012
- [3] *IP Office Release 8.1 Implementing Voicemail Pro*, Document Number 15-601064, June 2012
- [4] *IP Office System Status Application*, Document Number 15-601758, November 2011
- [5] *Avaya IP Office Knowledgebase*, <http://marketingtools.avaya.com/knowledgebase>
- [6] *Administering Avaya Session Border Controller*, Document Number 08-604063, Sept. 2012

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 8.0 with Verizon IPCC Service Suite.

[VZB-IPCCIPOR8FT] Application Notes for Configuring SIP Trunking using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 8, Issue 1.0

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 8.1 and Avaya Session Border Controller for Enterprise with Verizon IP Trunk Service Suite.

[VZBIPT-IPO81SBC] Application Notes for SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service and Avaya IP Office Release 8.1 with Avaya Session Border Controller for Enterprise Release 6.2 – Issue 1.0

[RFC-3261] RFC 3261 *SIP: Session Initiation Protocol* <http://www.ietf.org/rfc/rfc3261.txt>

[RFC-2833] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*
<http://www.ietf.org/rfc/rfc2833.txt>

Information in the following Verizon documents was also used for these Application Notes.
Contact a Verizon Business Account Representative for additional information.

- [VZ-Test-Plan] Test Suite for CPE IP Trunking Interoperability v1.3
- [VZ-Spec] Verizon Business IPCC Trunk Interface Network Interface Specification, Document Version 2.2.1.9

11. Appendix A: 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.

Note that not all of the configuration information, particularly items relevant to a specific installation environment, 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 Section 5.4 in these Application Notes as a reference.

The SIP Line Template created from the configuration as documented in these Application Notes is as follows:

```
<?xml version="1.0" encoding="utf-8"?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20130307</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>AvayaSBCE-VerizonIPCC</DescriptiveName>
  <ITSPDomainName>1.1.1.2</ITSPDomainName>
  <SendCallerID>CallerIDDIV</SendCallerID>
  <ReferSupport>true</ReferSupport>
  <ReferSupportIncoming>1</ReferSupportIncoming>
  <ReferSupportOutgoing>1</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>true</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>FavourDirectory</LineNamePriority>
  <UpdateSupport>UpdateAuto</UpdateSupport>
  <UserAgentServerHeader />
  <CallerIDfromFromheader>false</CallerIDfromFromheader>
  <PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
  <ITSPProxy>10.64.19.199</ITSPProxy>
  <LayerFourProtocol>SipTCP</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.729(a) 8K CS-ACELP,G.711 ULAW 64K</AdvCodecPref>
  <CallInitiationTimeout>4</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
  <VoipSilenceSupression>false</VoipSilenceSupression>
```

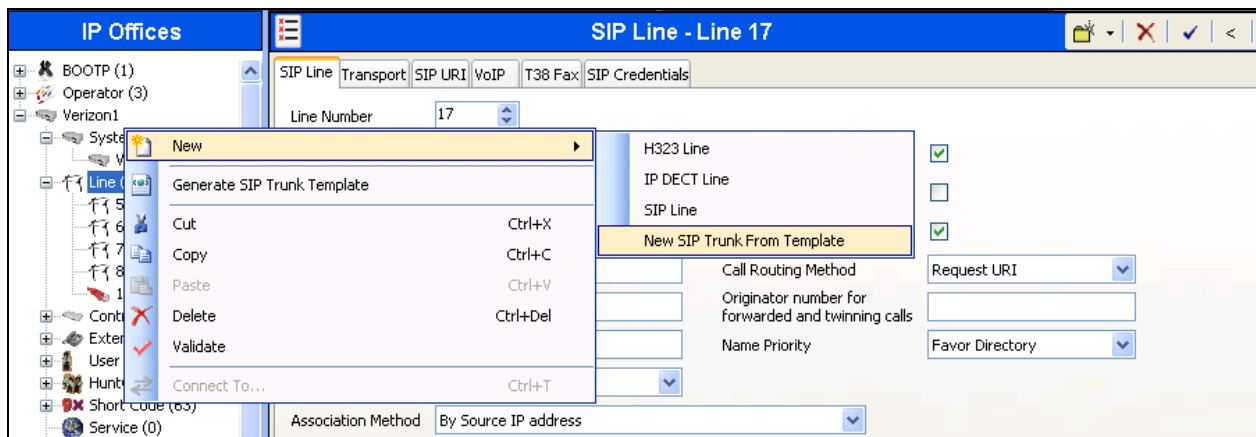
```

<ReinviteSupported>true</ReinviteSupported>
<FaxTransportSupport>FOIP_T38FB</FaxTransportSupport>
<UseOffererPreferredCodec>false</UseOffererPreferredCodec>
<CodecLockdown>false</CodecLockdown>
<Rel100Supported>false</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>true</DisableT30ECM>
<DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
<DisableT30MRCompression>false</DisableT30MRCompression>
<NSFOVERRIDE>false</NSFOVERRIDE>
</Template>

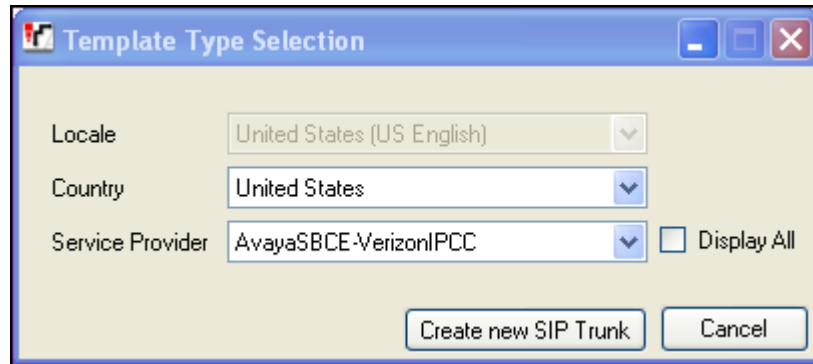
```

To import the above template into a new installation:

1. On the PC where IP Office Manager was installed, copy and paste the above template into a text document named **US_AvayaSBCE-VerizonIPCC_SIPTrunk.xml**. Move the .xml file to the IP Office Manager template directory (C:\Program Files\Avaya\IP Office\Manager\Templates). It may be necessary to create this directory.
2. Import the template into an IP Office installation 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**:



1. Verify that “United States” is automatically populated for **Country** and “AvayaSBCE-VerizonIPCC” is automatically populated for **Service Provider** in the resulting Template Type Selection screen as shown below. Click **Create new SIP Trunk** to finish the importing process.



Template Type Selection

Locale: United States (US English)

Country: United States

Service Provider: AvayaSBCE-VerizonIPCC ☐ Display All

Create new SIP Trunk Cancel

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