



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking Using Verizon Business IP Contact Center VoIP Inbound with Avaya IP Office Release 11.1 and Avaya Session Border Controller for Enterprise Release 8.0.1 – Issue 1.0

Abstract

These Application Notes describe a reference 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 reference configuration, the Avaya IP Office solution consists of Avaya IP Office Server Edition Release 11.1, Avaya Session Border Controller for Enterprise Release 8.1 and Avaya SIP, H.323, digital, and analog endpoints.

These Application Notes complement previously published Application Notes by illustrating the configuration screens and Avaya testing of IP Office Release 11.1 and Avaya Session Border Controller for Enterprise Release 8.1.

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.

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

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.

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1. Introduction

These Application Notes describe a reference configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Contact Center VoIP Inbound Service (Verizon Business IPCC) and an Avaya IP Office solution. In the reference configuration, the Avaya IP Office solution consists of Avaya IP Office Server Edition Primary Server Release 11.1, an IP500 V2 Expansion System Release 11.1, Avaya Session Border Controller for Enterprise Release 8.1 and Avaya SIP, H.323, digital, and analog endpoints.

Customers using Avaya IP Office with the Verizon Business IPCC 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. This service provides toll free inbound calling via standards-based SIP trunks as well as re-routing of inbound toll-free calls to alternate destinations based upon SIP messages (i.e., REFER) generated by the Avaya IP Office.

In the reference configuration, an Avaya Session Border Controller for Enterprise (Avaya 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, as well as a variety of other functions providing security and the presentation of a standardized SIP interface.

Verizon Business IPCC 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 IPCC service terminated via a PIP network connection, the solution validated in this document applies also to IP Contact Center services delivered via IDA service terminations.

For more information on the Verizon Business IPCC service, visit <https://enterprise.verizon.com/products/customer-experience-services/transport-and-intelligent-routing/ip-contact-center/>

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the Verizon Business IPCC service, as depicted in **Figure 1**. The Avaya SBCE and IP Office were configured to use the commercially available SIP Trunking solution provided by the Verizon Business IPCC service. This allowed Avaya IP Office to receive inbound toll-free calls from the PSTN via the SIP protocol.

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

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect

Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and Verizon Business IPCC Services did not include use of any specific encryption features as requested by Verizon.

Encryption (TLS/SRTP) was used internal to the enterprise between Avaya products wherever possible.

2.1. Interoperability Compliance Testing

The testing included executing the test cases detailed in **Section 11**, reference [VZ-Test-Plan], which contains the Verizon Business 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 5.5** and were answered by Avaya H.323 telephones, Avaya SIP telephones, Avaya IX™ Workplace for Windows (SIP) softphones, Avaya digital and analog telephones, as well as 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 directed to a hunt group whose queue is “full” (i.e. if no redirection is configured for hunt group busy conditions, see **Section 5.7.2**).
- 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 from a mobile phone), 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.
- Inbound fax calls using T.38 and G.711.
- 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.
- Incoming toll-free calls directed to the Hunt Groups configured in **Section 5.7.2** 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 [IPT-IPO111SBC81]. As detailed in reference [IPT-IPO111SBC81], these outgoing PSTN calls can be originated from Avaya H.323 telephones, Avaya SIP telephones, Avaya digital telephones, and analog endpoints. The display of caller ID on display-equipped PSTN telephones was verified. In the context of inbound toll-free calls using Verizon Business IPCC, inbound toll-free calls arriving via the SIP Line configured in **Section 5.5** could be forwarded or twinned out the Verizon Business IP Trunk service SIP Line. Inbound toll-free calls from the Verizon Business IPCC SIP Line could also trigger mobile callback calls that use the Verizon Business IP Trunk service SIP Line.
- Call Forwarding of Verizon toll-free calls to PSTN destinations via the Verizon Business IP Trunk service documented in reference [IPT-IPO111SBC81], presenting true calling party information to the mobile phone.
- Mobile twinning of Verizon toll-free calls to a mobile phone via the Verizon Business IP Trunk service documented in reference [IPT-IPO111SBC81], presenting true calling party information to the mobile phone.
- Proper DiffServ markings for Avaya SBCE SIP signaling and RTP media.
- Avaya Remote Worker configuration via the Avaya SBCE.

2.2. Test Results

Interoperability testing of the reference configuration was completed with successful results. The following observations were noted.

- During the compliance test, some inbound toll-free calls are transferred back to the Verizon Business IPCC service to be routed to alternate destinations. Since the Verizon Business IPCC in the reference configuration is an inbound only service, the call transfer is achieved by IP Office sending a SIP REFER message, with a Refer-To without a Replaces header, back to Verizon. This functionality is limited to Voicemail Pro modules and certain SIP endpoints using blind transfer (e.g., Avaya 1140E, and Avaya IX™ Workplace for Windows). Other endpoints like J100 series IP telephones, H.323, digital, or analog endpoints do not support this type of transfer.
- On inbound toll-free calls that are transferred back out to the Verizon Business IPCC service using SIP REFER, Verizon sends a NOTIFY message after the “referred” call has been released from the IP Office. Since IP Office cannot associate this NOTIFY with any active call, it sends a "481 Call/Transaction Does Not Exist" to Verizon in response. No user perceived problem occurs because of this anomaly.
- When transferring an inbound Verizon Business IPCC toll-free call back out the Verizon Business IPCC service using SIP REFER to a third party that is busy, Verizon disconnects the call and no error message or busy tone is heard by the originating caller.
- The Verizon Business IPCC Services suite does not support the SIP 302 Redirect method.

2.3. Support

For technical support on Verizon Business IPCC Services offer, visit online support at <http://www.verizonenterprise.com/support/>

Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

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.

In the reference configuration, the Avaya SBCE receives traffic from the Verizon Business IPCC service on port 5060 and sends traffic to port 5072, using UDP for network transport, as required by the Verizon Business IPCC service. 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 shown in **Section 5.9**. The Avaya CPE environment domain known to Verizon was *adevc.avaya.globalipcom.com*.

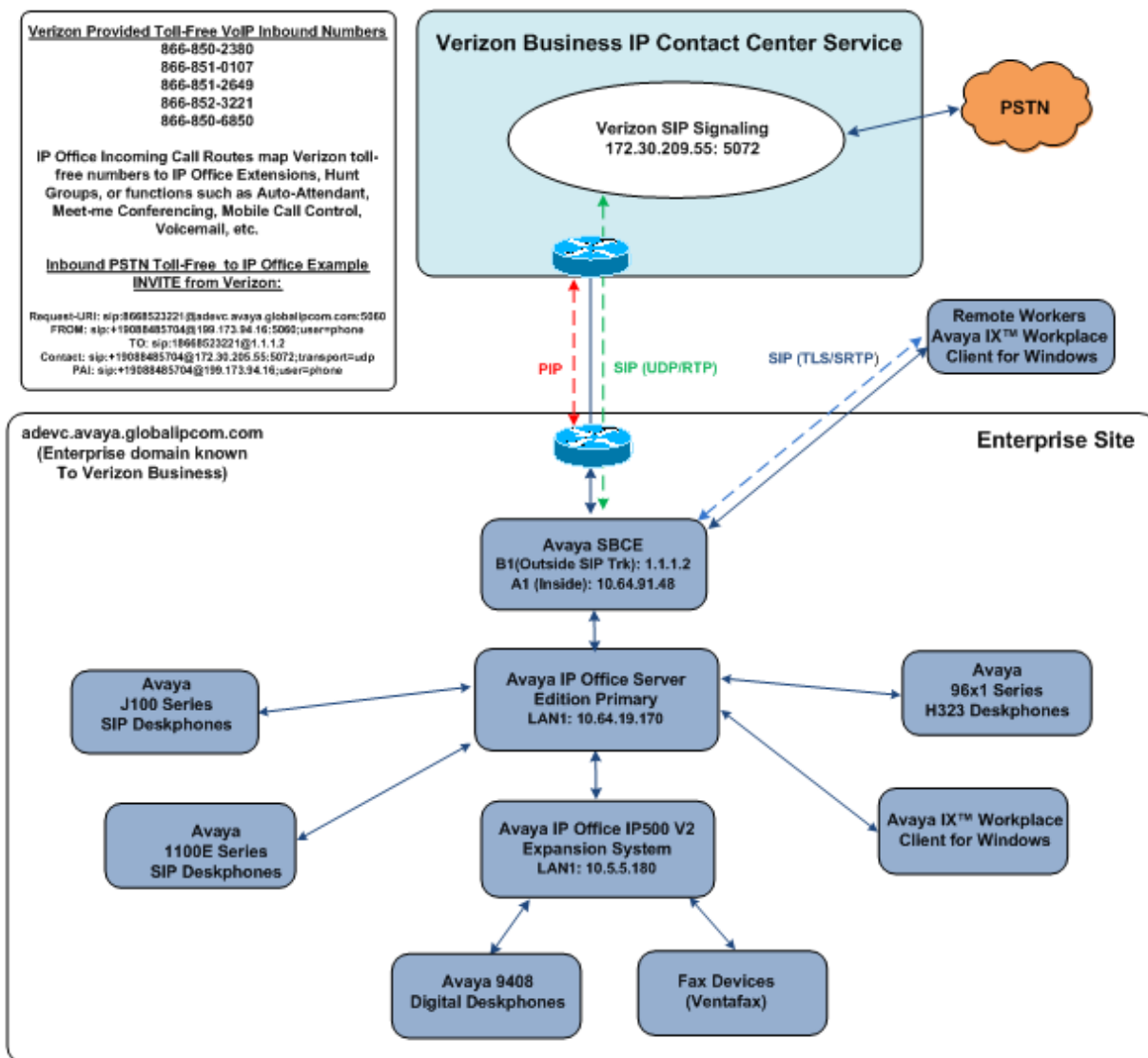


Figure 1: Avaya Interoperability Test Lab Configuration

The Verizon Business IP Trunk service used for outgoing calls, as described in **Section 2.1**, required different SIP line configuration parameters than what were needed for the Verizon Business IPCC service. A new SIP line was created in IP Office towards Avaya SBCE to support the Verizon Business IPCC service. This SIP line is separate from the SIP line previously created towards Avaya SBCE for Verizon Business IP Trunk service as described in reference [IPT-IPO111SBC81]. Having separate SIP lines for each Verizon service will allow for unique parameters to be set on IP Office to accommodate the differences between the two services. In order to support two SIP lines going towards the same Avaya SBCE, the Avaya SBCE was configured with two internal IP addresses designated for SIP trunk traffic.

Additionally, the reference configuration included remote worker functionality. A remote worker is a SIP endpoint that resides in the untrusted network, registered to IP Office via the Avaya SBCE. Remote workers feature the same functionality as any other endpoint within the enterprise. This functionality was successfully tested during the compliance test.

The configuration tasks required to support remote workers are beyond the scope of these Application Notes; hence they are not discussed in this document. For more information on configuring the Avaya SBCE for IP Office remote workers, consult Error! Reference source not found. in the Additional References section.

4. Equipment and Software Validated

Table 1 shows the equipment and software used in the reference configuration.

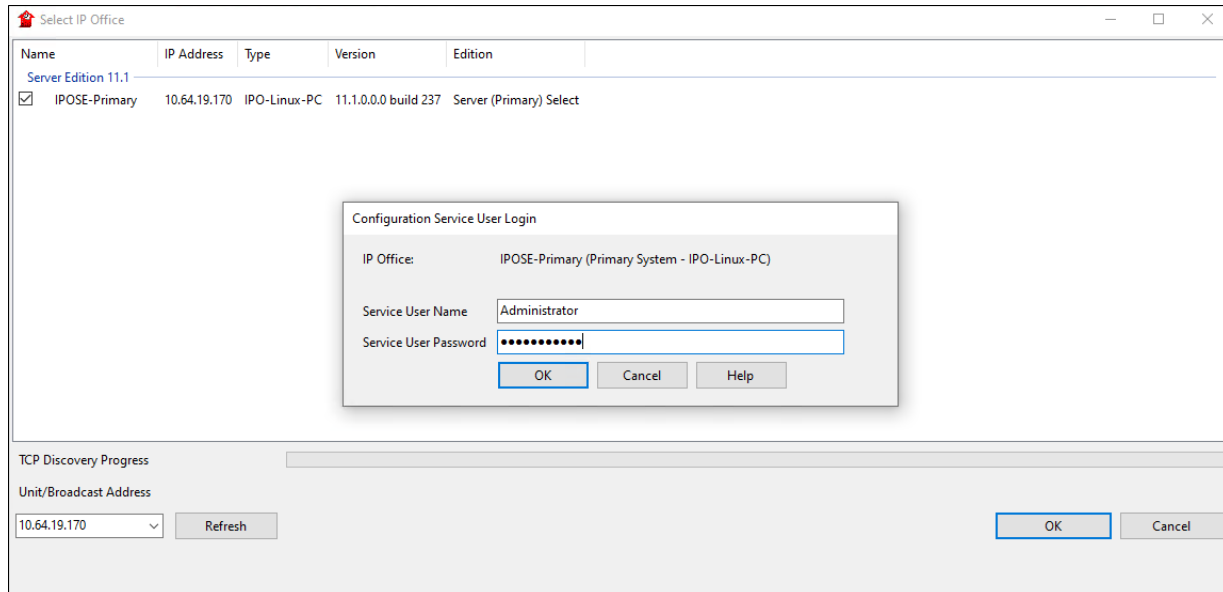
Avaya IP Telephony Solution Components	
Equipment	Software
Avaya IP Office Server Edition	Release 11.1.0.0.0 Build 237
- Avaya IP Office Voicemail Pro	Release 11.1.0.0.0 Build 234
Avaya IP Office 500 V2 Expansion System	Release 11.1.0.0.0 Build 237
Avaya IP Office Manager	Release 11.1.0.0.0 Build 237
Avaya Session Border Controller for Enterprise	8.1.0.0.14-18490
Avaya 96x1 Series IP Deskphone (H.323)	Release 6.8304
Avaya 1140E IP Deskphone (SIP)	Release 04.04.23.00
Avaya J169 IP Deskphone (SIP)	Release 4.0.5.0.10
Avaya IX™ Workplace for Windows (SIP)	Release 3.8.5.41.23
Avaya 9508 Digital Deskphone	Release 0.60

Table 1: Equipment and Software Tested

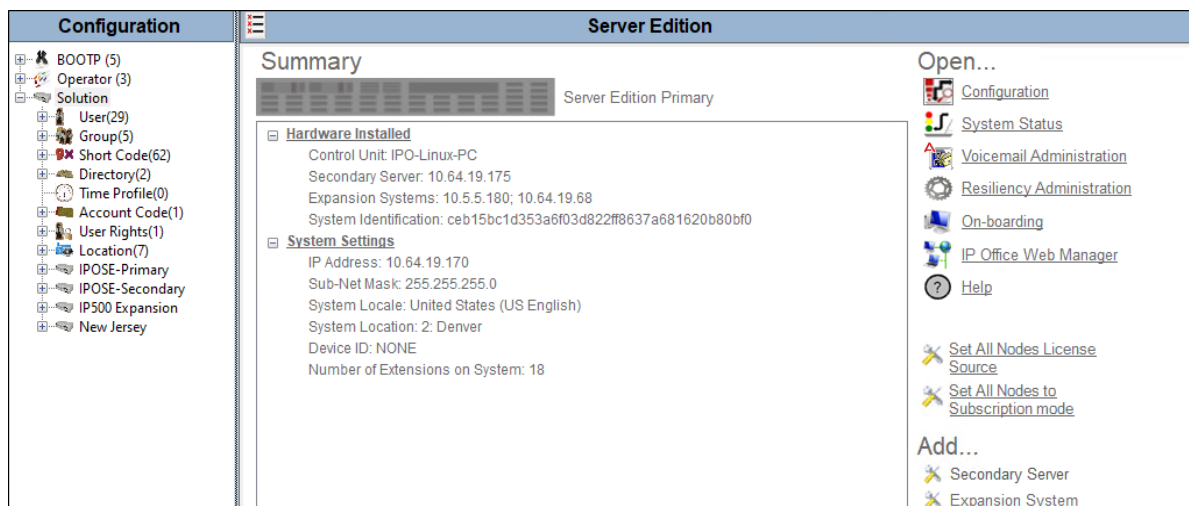
Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2, and also when deployed with all configurations of IP Office Server Edition. Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

5. Avaya IP Office Primary Server Configuration

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



On Server Edition systems, the Solution View screen will appear, similar to the one shown below. All the Avaya IP Office configurable components are shown in the left pane, known as the Navigation Pane. Clicking the “plus” sign next to the Primary server system name, e.g., **IPOSE-Primary**, on the navigation pane will expand the menu on this server.



5.1. Licensing

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

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

Navigate to **License** in the Navigation Pane. In the Details Pane verify that the **License Status** for **SIP Trunk Channels** is Valid and that the number of **Instances** is sufficient to support the number of channels provisioned for the SIP trunk.

Configuration

BOOTP (3)

Operator (3)

Solution

User(48)

Group(7)

Short Code(61)

Directory(2)

Time Profile(0)

Account Code(1)

User Rights(1)

Location(7)

IPOSE-Primary

System (1)

Line (16)

Control Unit (9)

Extension (18)

User (21)

Group (4)

Short Code (11)

Service (0)

Incoming Call Route (75)

IP Route (1)

License (10)

ARS (13)

License

License Type

Status

License

Remote Server

License Mode

WebLM Normal

Licensed Version

11.0

Select Licensing

Valid

Feature	Instances	Status	Expiration Date	Source
Additional Voicemail Pro Ports	152	Valid	Never	WebLM
VMPro TTS Professional	1	Valid	Never	WebLM
Power User	16	Valid	Never	WebLM
Avaya IP endpoints	18	Valid	Never	WebLM
SIP Trunk Channels	100	Valid	Never	WebLM
CTI Link Pro	1	Valid	Never	WebLM
Server Edition	1	Valid	Never	WebLM
Web Collaboration	2	Valid	Never	WebLM
UMS Web Services	1	Valid	Never	WebLM
VM Media Manager	1	Valid	Never	WebLM

5.2. TLS Management

For the compliance test, the signaling on the SIP trunk between IP Office and the Avaya SBCE was secured using TLS. Testing was done using identity certificates signed by a local certificate authority **SystemManager CA**. The generation and installation of these certificates are beyond the scope of these Application Notes. However, once the certificates are available they can be viewed on IP Office in the following manner.

To view the certificates currently installed on IP Office, navigate to **File → Advanced → Security Settings**. Log in with the appropriate security credentials (not shown). In the Security Settings window, navigate to **Security → System** and select the **Certificates** tab.

To verify the identity certificate, locate the **Identity Certificate** section and click **View** to see the details of the certificate.

System: IPOSE-Primary

System Details | Unsecured Interfaces | **Certificates**

Identity Certificate

Offer Certificate ☒

Offer ID Certificate Chain ☐

Issued To: silipose.customer.com

Set View Regenerate

Certificate Expiry Warning Days 60

Use Different Identity Certificate For SIP Telephony None

Received Certificate Checks (Management Interfaces) None

Received Certificate Checks (Telephony Endpoints) None

Trusted Certificate Store

Installed Certificates

- System Manager CA
- Symantec Class 3 Secure Server CA - G4
- VeriSign Class 3 International Server CA - G3
- SIP Product Certificate Authority

5.3. System Settings

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

5.3.1. LAN1 Settings

In the reference configuration, LAN1 is used to connect the Primary server 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 Primary server is **10.64.19.170**. Other parameters on this screen may be set according to customer requirements.

The screenshot displays the configuration interface for the IPOSE-Primary system. On the left, a 'Configuration' tree shows a hierarchy starting with 'Solution', followed by 'User(49)', 'Group(7)', 'Short Code(60)', 'Directory(2)', 'Time Profile(0)', 'Account Code(1)', 'User Rights(1)', 'Location(7)', and finally 'IPOSE-Primary'. Under 'IPOSE-Primary', there is a 'System (1)' entry, which is further expanded to show 'Line (14)' and 'Control Unit (9)'. The main panel is titled 'System' and contains a 'Name' field with the value 'IPOSE-Primary'. To the right, a series of tabs are visible: '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. This sub-tab contains fields for 'IP Address' (set to 10 . 64 . 19 . 170) and 'IP Mask' (set to 255 . 255 . 255 . 0). Below these, there is a 'Number Of DHCP IP Addresses' field set to 84. At the bottom, the 'DHCP Mode' is set to 'Disabled' (indicated by a selected radio button), with options for 'Server' and 'Client' also present. An 'Advanced' button is located at the bottom right of the DHCP Mode section.

Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** parameter is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 96x1 deskphones used in the reference configuration. The **H.323 Signaling over TLS** should be set based on customer needs. In the reference configuration it was set to **Preferred**. The **SIP Trunks Enable** parameter must be checked to enable the configuration of SIP trunks to Verizon Business. The **SIP Registrar Enable** parameter is checked to allow Avaya J169, Avaya 1140E, and Avaya IX™ Workplace for Windows (SIP) usage.

The **SIP Domain Name** and **SIP Registrar FQDN** may be set according to customer requirements. Set the **Layer 4 Protocol** section based on customer needs. In the reference configuration **TCP/5055** and **TLS/5056** were configured.

If desired, the **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media paths from Avaya SBCE to the Primary server. The defaults are used here.

The screenshot displays the Avaya configuration interface with the **VoIP** tab selected. The configuration is organized into several sections:

- H.323 Settings:**
 - ☒ **H.323 Gatekeeper Enable**
 - ☐ Auto-create Extension ☐ Auto-create User ☐ H.323 Remote Extension Enable
 - H.323 Signaling over TLS: **Preferred** (dropdown)
 - Remote Call Signaling Port: **1720** (spin box)
- SIP Settings:**
 - ☒ **SIP Trunks Enable**
 - ☒ **SIP Registrar Enable**
 - ☐ Auto-create Extension/User ☒ SIP Remote Extension Enable
 - Allowed SIP User Agents: **Block blacklist only** (dropdown)
 - SIP Domain Name: **silipose.customer.com** (text box)
 - SIP Registrar FQDN: **silipose.customer.com** (text box)
- Layer 4 Protocol:**
 - ☐ UDP: UDP Port **5060**, Remote UDP Port **5060**
 - ☒ TCP: TCP Port **5055**, Remote TCP Port **5055**
 - ☒ TLS: TLS Port **5056**, Remote TLS Port **5056**
- Challenge Expiration Time (sec):** **10** (spin box)
- RTP Settings:**
 - Port Number Range:** Minimum **40750**, Maximum **50750**
 - Port Number Range (NAT):** Minimum **40750**, Maximum **50750**

Scrolling down the page, on the **Keepalives** section, set the **Scope** to **RTP-RTCP**. Set the **Periodic timeout** to **30** and the **Initial keepalives** parameter to **Enabled**. These settings will cause the Primary server to send RTP and RTCP keepalive packets starting at the time of initial connection and every 30 seconds thereafter if no other RTP or RTCP traffic is present. This facilitates the flow of media in cases where each end of the connection is waiting to see media from the other, as well as helping to keep ports open for the duration of the call.

In the **DiffServ Settings** section, the Primary server can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services (QoS) policies for both signaling and media. The **DSCP** field is the value used for media, while the **SIG DSCP** is the value used for signaling. These settings should be set according to the customer's QoS policies in place. The default values used during the compliance test are shown.

System	LAN1	LAN2	DNS	Voicemail	Telephony	Directory Services	System Events	SMTP	SMDR	VoIP	Contact Center	Avaya Cloud Services
<div> LAN Settings VoIP Network Topology </div>												
<input checked="" type="checkbox"/> Enable RTCP Monitoring on Port 5005 RTCP collector IP address for phones: 0 . 0 . 0 . 0												
<div> Keepalives <div> Scope: RTP-RTCP Periodic timeout: 30 </div> <div> Initial keepalives: Enabled </div> </div>												
<div> DiffServ Settings <div> <div> B8 DSCP(Hex) </div> <div> B8 Video DSCP (Hex) </div> <div> FC DSCP Mask (Hex) </div> <div> 88 SIG DSCP (Hex) </div> </div> <div> <div> 46 DSCP </div> <div> 46 Video DSCP </div> <div> 63 DSCP Mask </div> <div> 34 SIG DSCP </div> </div> </div>												
<div> DHCP Settings <div> Primary Site Specific Option Number (4600/5600): 176 </div> <div> Secondary Site Specific Option Number (1600/9600): 242 </div> <div> VLAN: Not Present </div> <div> 1100 Voice VLAN Site Specific Option Number (SSON): 232 </div> <div> 1100 Voice VLAN IDs: </div> </div>												

Select the **Network Topology** tab as shown in the following screen. The **Firewall/NAT Type** was set to **Unknown** in the reference configuration. **Binding Refresh Time (sec)** was set to **60** seconds. This is used to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages, to periodically check the status of the SIP lines configured on this interface. The **Public IP Address** and **Public Port** sections are not used for the Verizon Business SIP trunk service connection.

The screenshot displays the Avaya IP Office configuration interface. At the top, a series of tabs are visible: System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VoIP, Contact Center, and Avaya Cloud Services. Below these, a sub-menu shows 'LAN Settings', 'VoIP', and 'Network Topology', with 'Network Topology' being the active tab. The main configuration area is titled 'Network Topology Discovery'. It contains several fields: 'STUN Server Address' (an empty text box), 'STUN Port' (a numeric spinner set to 3478), 'Firewall/NAT Type' (a dropdown menu showing 'Unknown'), 'Binding Refresh Time (sec)' (a numeric spinner set to 60), and 'Public IP Address' (a text box showing '0 . 0 . 0 . 0'). To the right of the IP address are 'Run STUN' and 'Cancel' buttons. Below these fields is a 'Public Port' section with three sub-sections: 'UDP' (spinner set to 0), 'TCP' (spinner set to 0), and 'TLS' (spinner set to 0). At the bottom left of the configuration area is a checkbox labeled 'Run STUN on startup' which is currently unchecked.

5.3.2. System Telephony Settings

To view or change telephony settings, select the **Telephony** tab and **Telephony** sub-tab as shown in the following screen. The settings presented here simply illustrate the reference configuration and are not intended to be prescriptive. In the reference configuration, the **Inhibit Off-Switch Forward/Transfer** box is unchecked so that call forwarding and call transfer to PSTN destinations via the 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 are set to **U-Law** as is typical in North American locales. Other parameters on this screen may be set according to customer requirements.

The screenshot displays the 'Telephony' configuration page within the Avaya IP Office management interface. The top navigation bar includes tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony (selected), Directory Services, System Events, SMTP, SMDR, VoIP, Contact Center, and Avaya Cloud Services. The 'Telephony' sub-tab is active, showing various settings organized into sections.

Telephony Settings:

- Dial Delay Time (sec): 4
- Dial Delay Count: 0
- Default No Answer Time (sec): 15
- Hold Timeout (sec): 0
- Park Timeout (sec): 0
- Ring Delay (sec): 5
- Call Priority Promotion Time (sec): Disabled
- Default Currency: USD
- Default Name Priority: Favor Trunk
- Media Connection Preservation: Enabled
- Phone Failback: Automatic

Login Code Complexity:

- ☐ Enforcement
- Minimum length: 6
- ☒ Complexity

RTCP Collector Configuration:

- ☐ Send RTCP to an RTCP Collector
- Server Address: 0 . 0 . 0 . 0
- UDP Port Number: 5005
- RTCP reporting interval (sec): 5

Companding Law:

- Switch:** ☒ U-Law, ☐ A-Law
- Line:** ☒ U-Law Line, ☐ A-Law Line

Other Settings:

- ☐ DSS Status
- ☒ Auto Hold
- ☒ Dial By Name
- ☒ Show Account Code
- ☐ Inhibit Off-Switch Forward/Transfer
- ☐ Restrict Network Interconnect
 - ☐ Include location specific information
- ☐ Drop External Only Impromptu Conference
- ☒ Visually Differentiate External Call
- ☒ High Quality Conferencing
- ☒ Directory Overrides Barring
- ☒ Advertise Callee State To Internal Callers
- ☐ Internal Ring on Transfer

5.3.3. System VoIP Settings

To view or change system codec settings, select the **VoIP → VoIP** tab. The **RFC2833 Default Payload** field allows for the manual configuration of the payload type used on SIP calls that are initiated by the IP Office. During the compliance test, this was set to **101**, the value preferred by Verizon Business. For codec selection, on the left, observe the list of **Available Codecs**. In the example screen below, which is not intended to be prescriptive, the parameter 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 5.5.5**). 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.

The screenshot displays the 'VoIP' configuration page with the 'VoIP' tab selected. The 'VoIP Security' sub-tab is active, showing options for 'Ignore DTMF Mismatch For Phones' and 'Allow Direct Media Within NAT Location', both of which are unchecked. The 'RFC2833 Default Payload' is set to '101'. Below this, the 'Default Codec Selection' section is visible, featuring three columns: 'Available Codecs', 'Unused', and 'Selected'. The 'Available Codecs' column lists four codecs with checked boxes: G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, and G.729(a) 8K CS-ACELP. The 'Unused' column contains 'G.711 ALAW 64K'. The 'Selected' column lists 'G.722 64K', 'G.711 ULAW 64K', and 'G.729(a) 8K CS-ACELP'. Navigation arrows (right, up, down, left) are positioned between the columns to facilitate moving codecs between the lists.

During the compliance test, SRTP was used internal to the enterprise wherever possible. To view or configure the media encryption settings, select the **VoIP → VoIP Security** tab on the Details pane. The **Media Security** drop-down menu is set to **Preferred** to have IP Office attempt use encrypted RTP for devices that support it and fall back to RTP for devices that do not support encryption. Under **Media Security Options**, **RTP** is selected for the **Encryptions** and **Authentication** fields. Under **Crypto Suites**, **SRTP_AES_CM_128_SHA1_80** is selected.

The screenshot shows the 'VoIP Security' configuration page. At the top, there is a navigation bar with tabs: System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VoIP, Contact Center, and Avaya Cloud Services. Below this, there are sub-tabs: VoIP, VoIP Security (selected), and Access Control Lists. The main content area includes fields for 'Default Extension Password' and 'Confirm Default Extension Password'. Below these is the 'Media Security' section, which has a dropdown menu set to 'Preferred' and a checkbox for 'Strict SIPs'. Under 'Media Security Options', there are three sections: 'Encryptions' with checkboxes for 'RTP' (checked) and 'RTCP' (unchecked); 'Authentication' with checkboxes for 'RTP' (checked) and 'RTCP' (checked); and 'Replay Protection'. Below these is the 'SRTP Window Size' field set to '64'. Finally, the 'Crypto Suites' section has a list box with 'SRTP_AES_CM_128_SHA1_80' (checked) and 'SRTP_AES_CM_128_SHA1_32' (unchecked).

5.4. IP Route

In the reference configuration, the Primary server LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is 10.64.19.1. 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 the Primary server, right-click **IP Route** from the Navigation pane, and select **New** (not shown). To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the Details pane with the relevant route using **Destination LAN1**.

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'Configuration' tree, which includes a hierarchy of system components such as BOOTP, Operator, Solution, User, Group, Short Code, Directory, Time Profile, Account Code, User Rights, Location, IPOSE-Primary, System, Line, Control Unit, Extension, User, Group, Short Code, Service, Incoming Call Route, IP Route (3), Licence, and ARS. The 'IP Route (3)' item is selected. The main area on the right is titled '0.0.0.0' and shows the configuration details for an IP Route. The fields are as follows:

Field	Value
IP Address	0 . 0 . 0 . 0
IP Mask	0 . 0 . 0 . 0
Gateway IP Address	10 . 64 . 19 . 1
Destination	LAN1
Metric	0

5.5. SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 11.1. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.5.1** to create the SIP Line from the template.

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

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

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.5.2 – 5.5.6**.

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

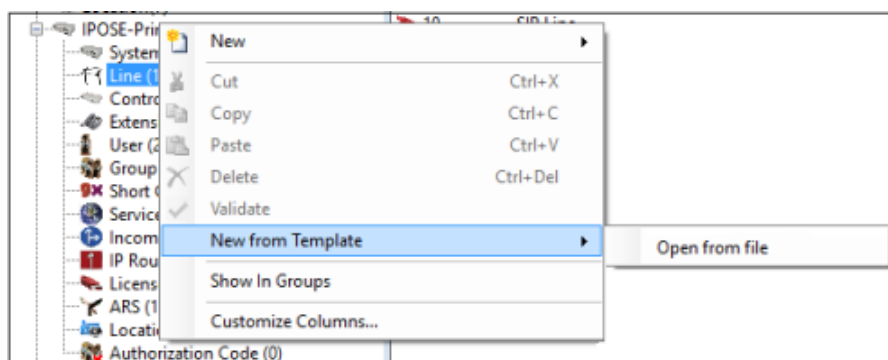
- SIP Line – Originator number for forwarded and twinning calls.
- Transport – Second Explicit DNS Server.
- SIP Credentials – Registration Requirement.
- SIP Advanced Engineering.

Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.5.2 – 5.5.6**.

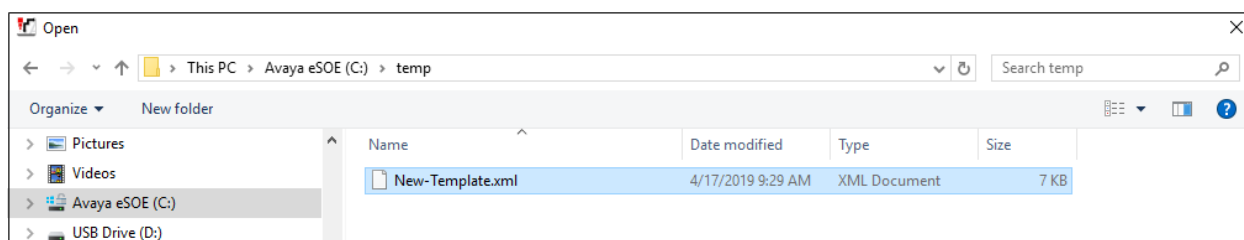
5.5.1. Importing a SIP Line Template

Note – DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500v2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer's environment.

Copy a previously created template file to the computer where IP Office Manager is installed. To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then navigate to **New → New from Template**. Select **Open from file**.



Navigate to the directory where the template was copied on the local computer (e.g., `\temp`) and select it. Click **Open** (not shown).



The new SIP Line is created, and it will appear on the **Navigation** pane (e.g., SIP Line **10**). The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.5.2** to **5.5.6**.

Line		
Line Number	Line Type	Line SubType
7	IP Office Line	WebSocket Server SCN
8	IP Office Line	WebSocket Server SCN
9	SIP Line	
10	SIP Line	
15	SIP Line	

5.5.2. SIP Line – SIP Line Tab

The **SIP Line** tab in the Details pane is shown below for Line Number 10, used for Avaya SBCE to the Verizon Business IPCC service. The **ITSP Domain Name** is left blank. Local Domain Name is set to the IP address of the Avaya IP Office LAN1 interface (e.g., **10.64.19.170**). By default, the **In Service** and **Check OOS** boxes are checked. With these settings, IP Office will use the SIP OPTIONS method to periodically check the SIP Line.

In the reference configuration, the IP Office **Country Code** is set to **1**. The From and PAI headers received from Verizon for calls from U.S. PSTN numbers contain “+1” before the calling PSTN number. By configuring the IP Office **Country Code** to 1, the caller ID display presented to IP Office users will be the PSTN number without any codes or prefixes. For example, a call from 3035387006 would display 3035387006. If the **Country Code** does not match the value following the “+” from Verizon, the IP Office user display would show the contents of the **International Prefix** field, followed by the value following the “+”, followed by the PSTN number. For example, if the Country Code parameter were left blank, the IP Office user would see a display such as “01113035387006”. Aside from display implications, if the **Country Code** is not configured, other patterns may also fail to match as expected, such as a match on the **Incoming CLI** field of the Incoming Call Route. See **Section 5.9.3** for configuration of incoming call routing based on the calling number.

Under **Session Timers**, the **Refresh Method** is set to **Re-invite** and the **Timer (seconds)** is set to **1800**. With this configuration, IP Office will send re-INVITEs every 15 minutes (half of the set value) to keep the active session alive.

SIP Line - Line 10	
Line Number	10
ITSP Domain Name	
Local Domain Name	10.64.19.170
URI Type	SIP URI
Location	Cloud
Prefix	
National Prefix	
International Prefix	011
Country Code	1
Name Priority	System Default
Description	SBCE-90-Vz IPCC
In Service	<input checked="" type="checkbox"/>
Check OOS	<input checked="" type="checkbox"/>
Session Timers	
Refresh Method	Re-invite
Timer (sec)	1800
Redirect and Transfer	
Incoming Supervised REFER	Auto
Outgoing Supervised REFER	Auto
Send 302 Moved Temporarily	<input type="checkbox"/>
Outgoing Blind REFER	<input checked="" type="checkbox"/>

Under **Redirect and Transfer**, the default automatic determination of **Incoming Supervised REFER** and **Outgoing Supervised REFER** is **Auto**. A supervised transfer occurs when a consultation call is made and the REFER contains a Replaces header. The Verizon Business

IPCC service does not support supervised REFER, and with this setting, IP Office will not use REFER for supervised transfers. The **Send 302 Moved Temporarily** setting is unchecked, as Verizon does not support receiving a 302 Moved Temporarily message. The **Outgoing Blind REFER** box can be optionally checked to enable use of REFER for blind transfers. In the reference configuration, this parameter is checked. See **Section 2.2** for limitations.

5.5.3. 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, **TLS** is selected as the **Layer 4 Protocol**. The **Send Port** and **Listen Port** can retain the default value 5061. The **Use Network Topology Info** parameter is set to **None**.

The screenshot shows the 'SIP Line' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to '10.64.91.48'. Under the 'Network Configuration' section, 'Layer 4 Protocol' is set to 'TLS', 'Send Port' is '5061', 'Use Network Topology Info' is set to 'None', and 'Listen Port' is '5061'. Below this, 'Explicit DNS Server(s)' are set to '0 . 0 . 0 . 0' and '0 . 0 . 0 . 0'. The 'Calls Route via Registrar' checkbox is checked. At the bottom, there is a 'Separate Registrar' field.

5.5.4. SIP Line – Call Details Tab

Select the **Call Details** tab. To add a new SIP URI, click the **Add...** button. A New URI area will be opened. To edit an existing entry, click an entry in the list at the top, and click the **Edit...** button.

The screenshot shows the 'SIP Line' configuration window with the 'Call Details' tab selected. It displays a table of SIP URIs with columns: URI, Groups, Credential, Local URI, Contact, P Asserted ID, P Preferred ID, Diversion Header, and Remote Party ID. There is one entry with ID 1, URI 10 10, Credential 0: <None>, Local URI Auto, and Contact Auto. To the right of the table are three buttons: 'Add...', 'Remove', and 'Edit...'. A scrollbar is visible at the bottom of the table area.

In the example screen below, a previously configured entry is edited. The **Incoming Group** parameter, set here to **10**, will be referenced when configuring Incoming Call Routes to map inbound SIP trunk calls to IP Office destinations in **Section 5.9**. The **Outgoing Group** parameter, also set to **10**, is relevant when using the SIP REFER method to transfer an inbound toll-free call back out the Verizon Business IPCC service. The **Max Sessions** parameter was set to **10**. This value sets the maximum number of simultaneous calls that can use the URI before IP Office returns busy to any further calls.

Auto is selected for the **Local URI** and **Contact** parameters. With this configuration, information in the Incoming Call Route (**Section 5.9**) is used to determine what call is accepted on the SIP Line. Set the **Field meaning** section to the values shown in the screenshot below.

SIP Line - 10 | Call Details | SIP URI

New URI

Incoming Group: 10 Max Sessions: 10

Outgoing Group: 10

Credentials: 0: <None>

	Display	Content
Local URI	Auto	Auto
Contact	Auto	Auto
P Asserted ID	<input type="checkbox"/> None	None
P Preferred ID	<input type="checkbox"/> None	None
Diversion Header	<input type="checkbox"/> None	None
Remote Party ID	<input type="checkbox"/> None	None

Field meaning		
Outgoing Calls	Forwarding/Twinning	Incoming Calls
Caller	Caller	Called
Caller	Caller	Called
None	None	None
None	None	None
None	None	None
None	None	None

OK Cancel Help

5.5.5. SIP Line – VoIP Tab

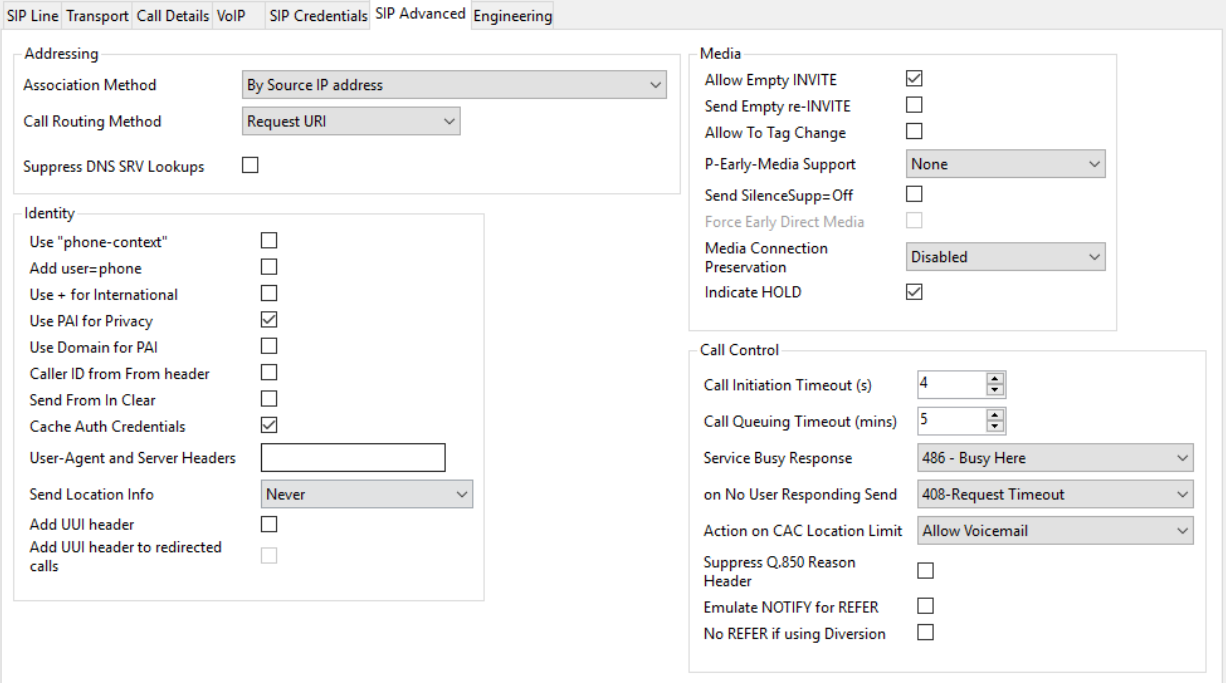
Select the **VoIP** tab. The **Codec Selection** drop-down parameter **System Default** (default) will match the codecs set in the system wide Default Selection list (**System** → **Codecs**). In the reference configuration, **Custom** is selected and the codecs **G729(a) 8K CS-ACELP** and **G.711 ULAW 64K** preferred by Verizon are specified. This will cause IP Office to include G.729a and G.711MU in the Session Description Protocol (SDP) offer, in that order. The **Fax Transport Support** drop-down is set to **T38 Fallback**. This enables T.38 to be used if supported, and fall back to G.711 if not. The **DTMF Support** parameter can remain set to the default value **RFC2833/RFC4733**. The **Media Security** drop-down menu is set to **Same as System** (**Preferred**) to have IP Office use the system setting for media security set in **Section 5.3.3** to encrypted RTP toward Avaya SBCE. The **Re-invite Supported** parameter is 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 **Allow Direct Media Path** parameter can be checked to allow for direct media between IP endpoints and the internal interface of the Avaya SBCE, if possible within the network infrastructure, freeing up DSP resources on the Primary server.

For PSTN originations, Verizon preferred the G.729a codec in the SDP, while also allowing the G.711MU codec. During 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 screenshot shows the 'SIP Line' configuration window with the 'VoIP' tab selected. The 'Codec Selection' dropdown is set to 'Custom'. Below it, there are two lists: 'Unused' and 'Selected'. The 'Unused' list contains 'G.711 ALAW 64K' and 'G.722 64K'. The 'Selected' list contains 'G.729(a) 8K CS-ACELP' and 'G.711 ULAW 64K'. There are arrows between the lists for moving items. To the right of the codec lists are several checkboxes: 'Local Hold Music' (unchecked), 'Re-invite Supported' (checked), 'Codec Lockdown' (unchecked), 'Allow Direct Media Path' (checked), 'Force direct media with phones' (unchecked), and 'PRACK/100rel Supported' (unchecked). Below these are three dropdown menus: 'Fax Transport Support' set to 'T38 Fallback', 'DTMF Support' set to 'RFC2833/RFC4733', and 'Media Security' set to 'Same as System (Preferred)'. At the bottom, there is an 'Advanced Media Security Options' section with a 'Same As System' checkbox checked. This section includes checkboxes for 'Encryptions' (RTP checked, RTCP unchecked), 'Authentication' (RTP checked, RTCP checked), 'Replay Protection' (SRTP Window Size set to 64), and 'Crypto Suites' (SRTP_AES_CM_128_SHA1_80 checked, SRTP_AES_CM_128_SHA1_32 unchecked).

5.5.6. SIP Line – SIP Advanced Tab

Select the **SIP Advanced** tab. In the **Media** area, the **Allow Empty INVITE** box is checked to allow IP Office to be the recipient of a Verizon Business IPCC enhanced transfer where the initial INVITE may not have SDP information. The **Indicate HOLD** box is checked to have IP Office signal to Verizon when a call is placed on/off hold. Other parameters may be left at default values.



Addressing	
Association Method	By Source IP address
Call Routing Method	Request URI
Suppress DNS SRV Lookups	<input type="checkbox"/>

Identity	
Use "phone-context"	<input type="checkbox"/>
Add user=phone	<input type="checkbox"/>
Use + for International	<input type="checkbox"/>
Use PAI for Privacy	<input checked="" type="checkbox"/>
Use Domain for PAI	<input type="checkbox"/>
Caller ID from From header	<input type="checkbox"/>
Send From In Clear	<input type="checkbox"/>
Cache Auth Credentials	<input checked="" type="checkbox"/>
User-Agent and Server Headers	
Send Location Info	Never
Add UUI header	<input type="checkbox"/>
Add UUI header to redirected calls	<input type="checkbox"/>

Media	
Allow Empty INVITE	<input checked="" type="checkbox"/>
Send Empty re-INVITE	<input type="checkbox"/>
Allow To Tag Change	<input type="checkbox"/>
P-Early-Media Support	None
Send SilenceSup=Off	<input type="checkbox"/>
Force Early Direct Media	<input type="checkbox"/>
Media Connection Preservation	Disabled
Indicate HOLD	<input checked="" type="checkbox"/>

Call Control	
Call Initiation Timeout (s)	4
Call Queuing Timeout (mins)	5
Service Busy Response	486 - Busy Here
on No User Responding Send	408-Request Timeout
Action on CAC Location Limit	Allow Voicemail
Suppress Q.850 Reason Header	<input type="checkbox"/>
Emulate NOTIFY for REFER	<input type="checkbox"/>
No REFER if using Diversion	<input type="checkbox"/>

5.6. IP Office Line

In IP Office Server Edition systems, IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. To edit an existing IP Office Line, select **Line** in the Navigation pane, and select the appropriate line to be configured in the Group pane. The screen below shows the IP Office Line to the IP500 V2 Expansion System.

Configuration	Line	IP Office Line - Line 8																																																															
<ul style="list-style-type: none"> BOOTP (3) Operator (3) Solution User (49) Group (7) Short Code (50) Directory (2) Time Profile (0) Account Code (1) User Rights (1) Location (7) IPOSE-Primary System (1) Line (14) Control Unit (9) Extension (18) User (22) Group (4) Short Code (10) Service (0) Incoming Call Route (6) IP Route (3) License (10) 	<table border="1"> <thead> <tr> <th>Line Number</th> <th>Line Type</th> <th>Line SubType</th> </tr> </thead> <tbody> <tr><td>1</td><td>IP Office Line</td><td>Websocket Serv</td></tr> <tr><td>2</td><td>SIP Line</td><td></td></tr> <tr><td>3</td><td>IP Office Line</td><td>Websocket Serv</td></tr> <tr><td>4</td><td>SIP Line</td><td></td></tr> <tr><td>5</td><td>IP Office Line</td><td>Websocket Serv</td></tr> <tr><td>6</td><td>SIP Line</td><td></td></tr> <tr><td>7</td><td>IP Office Line</td><td>Websocket Serv</td></tr> <tr><td>8</td><td>IP Office Line</td><td>Websocket Serv</td></tr> <tr><td>10</td><td>SIP Line</td><td></td></tr> <tr><td>15</td><td>SIP Line</td><td></td></tr> <tr><td>16</td><td>SIP Line</td><td></td></tr> <tr><td>21</td><td>SIP Line</td><td></td></tr> <tr><td>22</td><td>SIP Line</td><td></td></tr> <tr><td>25</td><td>SIP Line</td><td></td></tr> </tbody> </table>	Line Number	Line Type	Line SubType	1	IP Office Line	Websocket Serv	2	SIP Line		3	IP Office Line	Websocket Serv	4	SIP Line		5	IP Office Line	Websocket Serv	6	SIP Line		7	IP Office Line	Websocket Serv	8	IP Office Line	Websocket Serv	10	SIP Line		15	SIP Line		16	SIP Line		21	SIP Line		22	SIP Line		25	SIP Line		<table border="1"> <thead> <tr> <th>Line</th> <th>Short Codes</th> <th>VoIP Settings</th> </tr> </thead> <tbody> <tr> <td>Line Number: 8</td> <td>Telephone Number: </td> <td>Transport Type: Websocket Server</td> </tr> <tr> <td>Networking Level: SCN</td> <td>Prefix: </td> <td>Outgoing Group ID: 99007</td> </tr> <tr> <td>Security: Medium</td> <td>Number of Channels: 500</td> <td>Outgoing Channels: 500</td> </tr> <tr> <td colspan="3"> Gateway: Address: 10 . 5 . 5 . 180 Location: 8: Miami Password: ***** Confirm Password: ***** </td> </tr> <tr> <td colspan="3"> SCN Resiliency Options: <input type="checkbox"/> Supports Resiliency <input type="checkbox"/> Backs up my IP phones <input type="checkbox"/> Backs up my hunt groups <input type="checkbox"/> Backs up my IP DECT phones </td> </tr> </tbody> </table>	Line	Short Codes	VoIP Settings	Line Number: 8	Telephone Number:	Transport Type: Websocket Server	Networking Level: SCN	Prefix:	Outgoing Group ID: 99007	Security: Medium	Number of Channels: 500	Outgoing Channels: 500	Gateway: Address: 10 . 5 . 5 . 180 Location: 8: Miami Password: ***** Confirm Password: *****			SCN Resiliency Options: <input type="checkbox"/> Supports Resiliency <input type="checkbox"/> Backs up my IP phones <input type="checkbox"/> Backs up my hunt groups <input type="checkbox"/> Backs up my IP DECT phones		
Line Number	Line Type	Line SubType																																																															
1	IP Office Line	Websocket Serv																																																															
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The screen below shows the IP Office Line, **VoIP Settings** tab. In the reference configuration, a fax machine is connected to one of the analog ports on the Expansion System. **Fax Transport Support** is set to **T.38 Fallback**. The **Media Security** is set to **Same as System (Preferred)** to have IP Office use the system setting for media security set in **Section 5.3.3** to encrypted RTP. Default values were used for all other parameters.

Line	Short Codes	VoIP Settings
<div> <input checked="" type="checkbox"/> Out Of Band DTMF <input checked="" type="checkbox"/> Allow Direct Media Path </div> <div> Codec Selection: System Default <div> <div>Unused</div> <div>Selected</div> <div> <div>G.711 ALAW 64K</div> <div>>>></div> <div>G.722 64K</div> <div><<<</div> <div>G.711 ULAW 64K</div> <div>>>></div> <div>G.729(a) 8K CS-ACELP</div> </div> </div> </div> <div> Fax Transport Support: T38 Fallback </div> <div> Call Initiation Timeout (s): 4 </div> <div> Media Security: Same as System (Preferred) <div> Advanced Media Security Options: <input checked="" type="checkbox"/> Same As System </div> </div>		

5.7. 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**. To edit an existing User, select **User** in the Navigation pane, and select the appropriate user to be configured in the Group pane.

5.7.1. SIP User 6241

The following screen shows the **User** tab for user 6241. This user corresponds to an Avaya J169 SIP endpoint.

The screenshot shows the 'User' configuration window for user 'aj169: 6241'. The window has a title bar with the user name and a standard toolbar. Below the title bar is a tabbed interface with the 'User' tab selected. The 'User' tab contains various configuration fields and a list of checkboxes for user profile settings.

Field	Value
Name	aj169
Password	••••••••
Confirm Password	••••••••
Unique Identity	
Conference PIN	••••
Confirm Audio Conference PIN	••••
Account Status	Enabled
Full Name	Avaya J169
Extension	6241
Email Address	aj169@customera.com
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 Desktop/Tablet VoIP client
- ☐ Enable Mobile VoIP Client
- ☐ Send Mobility Email

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

SIP Extension: 11210 6241

Extension ID: 11210


Base Extension: 6241

Phone Password: •••••

Confirm Phone Password: •••••

Caller Display Type: On

Reset Volume After Calls: ☐

Device Type:  Avaya J169 (SIP Feature)

Location: Automatic

Fallback As Remote Worker: Auto

Module: 0

Port: 0

Disable Speakerphone: ☐

The following screen shows the **VoIP** tab for the extension. The **IP Address** field may be left blank. Check the **Reserve Avaya IP endpoint license** box. The **Codec Selection** parameter may retain the default setting **System Default** to follow the system configuration shown in **Section 5.3.3**. The Media Security parameter may also retain the default setting **Same as System (Preferred)** to follow the system configuring shown in **Section 5.3.3**.

IP Address: 0 . 0 . 0 . 0

Codec Selection: System Default

Unused: G.711 ALAW 64K

Selected: G.722 64K, G.711 ULAW 64K, G.729(a) 8K CS-ACELP

Reserve License: Reserve Avaya IP endpoint license

Fax Transport Support: None

DTMF Support: RFC2833/RFC4733

3rd Party Auto Answer: None

Media Security: Same as System (Preferred)

Advanced Media Security Options: ☒ Same As System

☐ Local Hold Music

☒ Re-invite Supported

☐ Codec Lockdown

☒ Allow Direct Media Path

5.7.2. 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 **Group** (not shown) from the Navigation pane and select **New**. To view or edit an existing hunt group, select **Group** from the Navigation pane, and the appropriate hunt group from the Group pane.

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

The screenshot displays the configuration window for a hunt group titled "Longest Waiting Group Call Center: 401". The window is divided into several sections:

- Group Pane:** A list on the left showing the selected group "Call Center" with extension "401".
- Configuration Fields:**
 - Name:** Call Center
 - Extension:** 401
 - Ring Mode:** Longest Waiting (dropdown)
 - Hold Music Source:** No Change (dropdown)
 - Ring Tone Override:** None (dropdown)
 - Agent's Status on No-Answer Applies To:** None (dropdown)
 - Central System:** IPOSE-Primary
 - Profile:** Standard Hunt Group (dropdown)
 - Exclude From Directory:** ☐
 - No Answer Time (sec):** System Default (15) (dropdown)
 - Advertise Group:** ☒
- User List:** A table listing members of the hunt group.

Extension	Name	System
<input checked="" type="checkbox"/> 6242	Avaya 9508	IP500 Expansion
<input checked="" type="checkbox"/> 6237	Avaya 9611	IPOSE-Primary
- Buttons:** "Edit..." and "Remove" buttons at the bottom right.

The following screen shows the **Queuing** tab for hunt group 401. In the reference 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** is configured to 2, and if two calls are already in queue, a third call to the Verizon toll-free number corresponding to this hunt group will get busy tone because IP Office will send a 486 Busy Here (i.e., if there is no Voicemail for the hunt group). As another example, if the **Queue Length** has a fixed limit of 2, and if two calls are already in queue, a third call to the Verizon toll-free number destined for this hunt group from a priority caller (see **Section 5.9.3**) will be queued ahead of non-priority callers, temporarily expanding the queue.

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 5.9.3**.

The screenshot displays the 'IPOSE-Primary : Longest Waiting Group Call Center: 401' configuration window. The 'Queuing' tab is selected, showing the following settings:

- ☒ Queuing On
- Queue Length: No Limit
- ☒ Normalize Queue Length
- Queue Type: Assign Call On Agent Answer
- Calls In Queue Alarm: ☐
- Calls In Queue Threshold: 1
- Analog Extension to Notify: <None>

The following screen shows the **Announcements** tab for hunt group 401. In the reference 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.

IPPOSE-Primary : Longest Waiting Group Call Center: 401

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

☒ Announcements On

Wait before 1st announcement (sec)

☐ Synchronize Calls

Flag call as answered ☐

↓

Play 1st announcement

↓

Post announcement tone

↓

2nd Announcement ☒

↓

Wait before 2nd announcement (sec)

↓

Play 2nd announcement

↓

Repeat last announcement ☒

↓

Wait before repeat (sec)

← (Loop back to Play 2nd announcement)

5.8. 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** (not shown) in the Navigation pane and select **New**. To edit an existing short code, click **Short Code** in the Navigation pane, and the short code to be configured in the Group pane.

Verizon Business IPCC service allows for blind transfers out their service using the SIP REFER Method. In the screen shown below, the short code **8N;** is illustrated. The **Code** parameter is set to **8N;**. The **Feature** parameter is set to **Dial**. The **Telephone Number** parameter is set to **N**. The **Telephone Number** field is used to construct the Request URI and To Header in the outgoing SIP INVITE message. The **Line Group ID** parameter is set to **60: SBCE to Vz IPCC**, configurable via ARS. See **Section 5.11** for example ARS route configuration for “60: SBCE to Vz IPCC”.

The screenshot shows a configuration window titled "8N;: Dial". It contains the following fields and values:

Field	Value
Short Code	8N;
Code	8N;
Feature	Dial
Telephone Number	N
Line Group ID	60: SBCE to Vz IPCC
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

The following screen illustrates a short code that acts like a feature access 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 5.9** for configuration of Incoming Call Routes.

The screenshot shows a configuration window titled "*17: Voicemail Collect". It contains the following fields and values:

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

The following screen illustrates another short code. In this case, the **Code** “55N;” is defined for **Feature** “**Conference Meet Me**”. The **Telephone Number** parameter is set to **N**. In the verification of these Application Notes, this short code was used in conjunction with a Voicemail Pro module named “MeetMeConf”. Although the Voicemail Pro configuration is beyond the

scope of these Application Notes, the module enabled a PSTN caller to dial a Verizon toll-free number, be prompted to enter a conference ID and PIN by Voicemail Pro, and then be transferred to the appropriate meet-me conference based on the ID entered by the caller. Local IP Office callers could also dial 556xxx to join the corresponding conference ID.

5.9. 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. To add an incoming call route, right click on **Incoming Call Route** (not shown) in the Navigation pane and select **New**. To edit an existing incoming call route, select **Incoming Call Route** in the Navigation pane, and the appropriate incoming call route to be configured in the Group pane.

5.9.1. 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 **10**, matching the **Incoming Group** field configured in the **Call Details** tab for the SIP Line to the Verizon Business IPCC service, in **Section 5.5.4**.

Select the **Destinations** tab. From the **Destination** drop-down, select an extension to receive the call when a PSTN user dials 8668502380. This number is associated with IP Office user extension 6241. (The **Destination** was changed in the course of testing to associate different destinations with the toll-free numbers.)

The screenshot shows the configuration window for the incoming number 10 8668502380. The 'Destinations' tab is selected. The table below shows the configuration for the default value.

TimeProfile	Destination	Fallback Extension
Default Value	6241 aj169	

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.

5.9.2. 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 **10**, matching the Incoming Group field configured in the Call Details tab for the SIP Line to Verizon Business in **Section 5.5.4** Optionally, the **Tag** parameter can be populated with a descriptive name that will associate the call with this incoming call route.

The screenshot shows the configuration window for the incoming number 10 8668523221. The 'Standard' tab is selected. The configuration parameters are as follows:

Bearer Capability	Any Voice
Line Group ID	10
Incoming Number	8668523221
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	IPCC-4
Hold Music Source	System Source
Ring Tone Override	None

Select the **Destinations** tab. From the **Destination** drop-down, select the destination to receive the call when an arbitrary PSTN user dials 8668523221. This toll-free number is associated with IP Office hunt group extension 401, the “Call Center” hunt group.

The screenshot shows the configuration window for the incoming number 10 8668523221. The 'Destinations' tab is selected. The table below shows the configuration for the default value.

TimeProfile	Destination	Fallback Extension
Default Value	401 Call Center	

5.9.3. Incoming Call Routes Based on Calling Party Number

This section presents a simple example 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. Although 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 3035382177** is illustrated. The **Line Group Id** is **10**, matching the Incoming Group field configured in the Call Details tab for the SIP Line to Verizon Business in **Section 5.5.4**. 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 “3035382177”. In this case, to allow this caller to be treated with priority when calling in, the **Priority** field is set to **3 - High**. Optionally, the **Tag** parameter can be populated with a descriptive name that will associate the call with this incoming call route.

The screenshot shows the configuration window for Incoming Call Route 10 8668523221. The 'Standard' tab is selected. The configuration fields are as follows:

Field	Value
Bearer Capability	Any Voice
Line Group ID	10
Incoming Number	8668523221
Incoming Sub Address	
Incoming CLI	3035382177
Locale	
Priority	3 - High
Tag	Priority Caller
Hold Music Source	System Source
Ring Tone Override	None

Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when PSTN user 3035382177 dials 8668523221. In this case, the **Destination** is also the hunt group **401 Call Center**, 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.

The screenshot shows the configuration window for Incoming Call Route 10 8668523221, with the 'Destinations' tab selected. The configuration table is as follows:

TimeProfile	Destination	Fallback Extension
Default Value	401 Call Center	

5.9.4. Incoming Call Routes to Various IP Office Features

In the reference configuration, the incoming call route for **Incoming Number 8668506850** was varied to test different destination features, such as Voice Mail, Mobile Call Control, Refer Call Redirection, and Conference Bridge. The screen showing the **Standard** tab for this toll-free number is shown below.

The screenshot shows the configuration window for the incoming call route 10 8668506850. The 'Standard' tab is selected, showing various configuration fields:

Bearer Capability	Any Voice
Line Group ID	10
Incoming Number	8668506850
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	IPCC-5
Hold Music Source	System Source
Ring Tone Override	None

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 5.8**). 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.
- **VM:MeetMe**. With this destination, an incoming call to 8668506850 will be delivered directly to the Voicemail Pro module “MeetMe” created for use as a conference bridge.
- **VM:Refer**. With this destination, an incoming call to 8668506850 will be delivered directly to the Voicemail Pro module “Refer” created for use as a Refer Call Redirection example. See **Section 5.10** for an illustration of this Voicemail Pro module.

An example screen showing the short code configured for a Voicemail Pro module is shown below.

The screenshot shows the 'Destinations' tab for the incoming call route 10 8668506850. It displays a table with the following data:

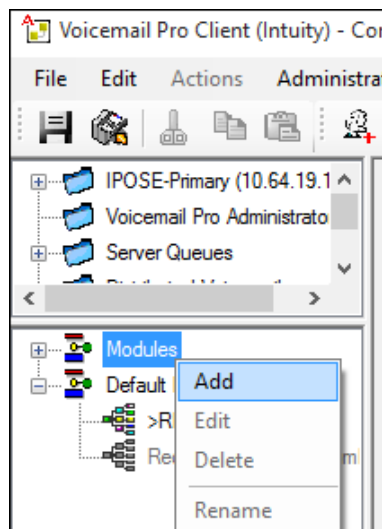
TimeProfile	Destination	Fallback Extension
Default Value	VM:MeetMe	

5.10. Voicemail Pro Refer Module



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

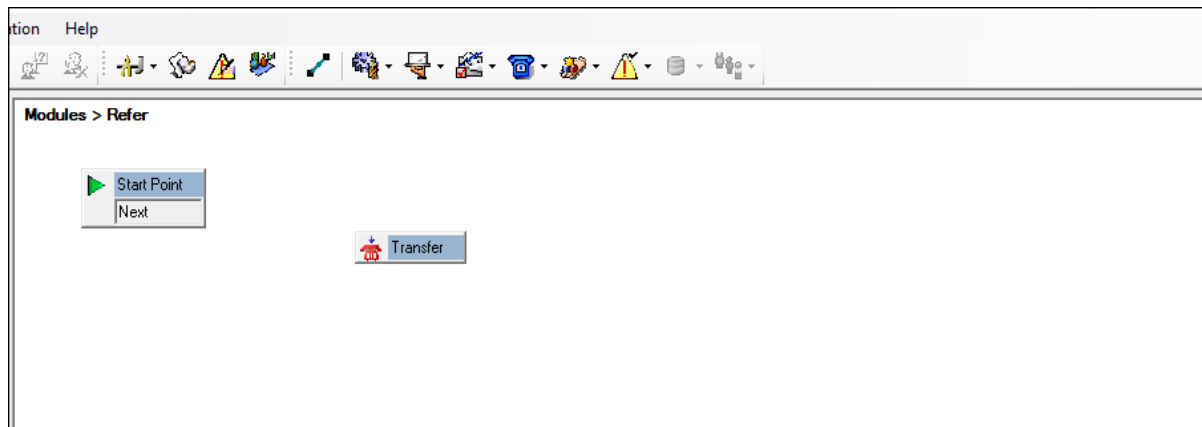
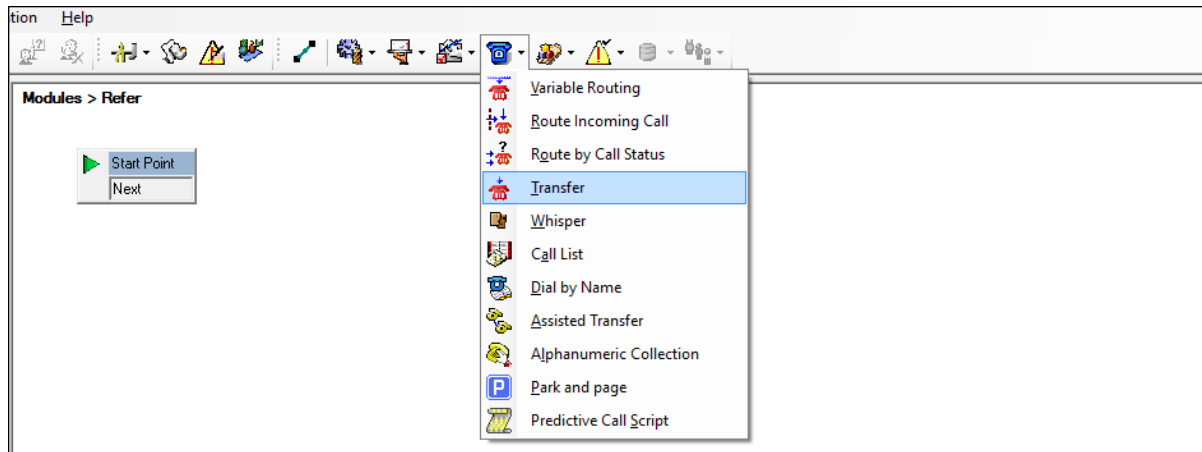
The Refer module is provisioned to play an announcement to the caller, and then generate a Refer (without Replaces) back to the Verizon Business IPCC service. This is accomplished via the Voicemail Pro Client interface.


From the IP Office Manager PC, select **Start → All Apps → IP Office → Voicemail Pro Client** to launch the Voicemail Pro Client interface. Navigate to **File → Login**, select the proper Voicemail Pro system, and log in using the appropriate credentials (not shown). Create a **Start Point** by right clicking on **Modules** and selecting **Add**.

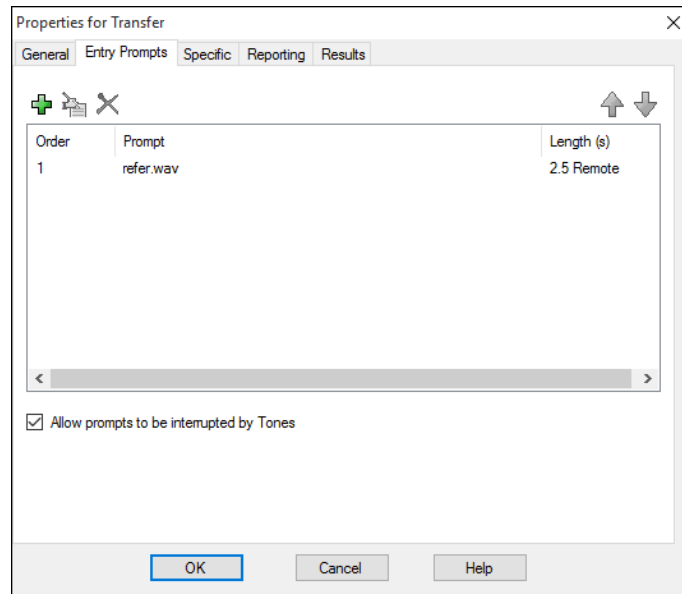


Enter a name (e.g., **Refer**) and click on **OK** (not shown). The new module “Refer” will appear, and a **Start Point** icon will appear in the work area.

Click on the **Telephony Actions** icon , select the **Transfer** icon , and click on the work area to place the **Transfer** icon in the work area.



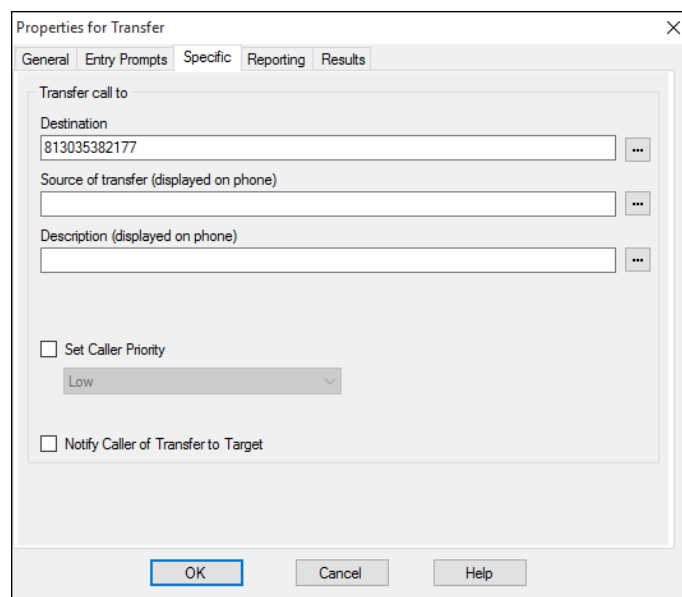
Double click on the **Transfer** icon, select the **Entry Prompts** tab and select or create an announcement to be played to the caller prior to the Refer (e.g., **refer.wav**). To modify an existing recording, double click on the .wav file and rerecord. If no .wav files exist, click on the  icon to open the .wav file editor.




The 'Properties for Transfer' dialog box is shown with the 'Entry Prompts' tab selected. It features a table with columns 'Order', 'Prompt', and 'Length (s)'. The first row shows '1' in the Order column, 'refer.wav' in the Prompt column, and '2.5 Remote' in the Length (s) column. Below the table is a checkbox labeled 'Allow prompts to be interrupted by Tones' which is checked. At the bottom are 'OK', 'Cancel', and 'Help' buttons.

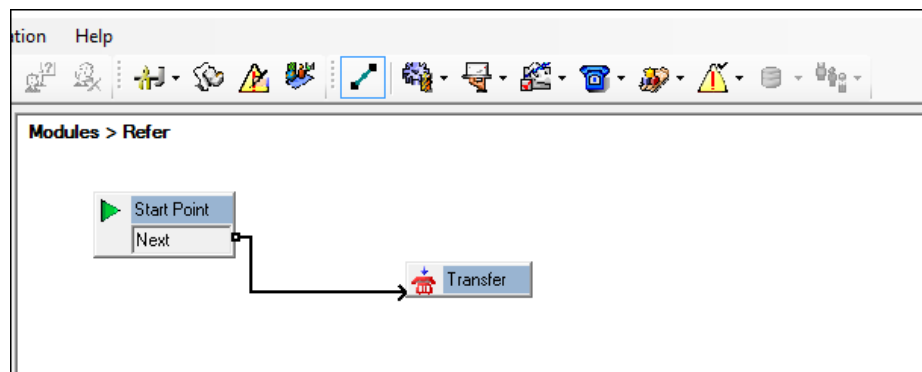
Order	Prompt	Length (s)
1	refer.wav	2.5 Remote

On the **Specific** tab enter the destination, including the outbound Short Code (e.g., **813035382177**). Click on **OK**.



The 'Properties for Transfer' dialog box is shown with the 'Specific' tab selected. It contains fields for 'Destination' (with the value '813035382177'), 'Source of transfer (displayed on phone)', and 'Description (displayed on phone)'. There is also a checkbox for 'Set Caller Priority' with a dropdown menu set to 'Low', and a checkbox for 'Notify Caller of Transfer to Target'. At the bottom are 'OK', 'Cancel', and 'Help' buttons.

From the options bar, select the **Connector** icon  and drag a connecting flow line from the **Start Point** box to the **Transfer** box.



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

When the associated Verizon Business Toll-Free number is received, IP Office will send the call to Voicemail Pro (see **Section 5.9.4**). The caller will hear the announcement (e.g., **refer.wav**), and Voicemail Pro/Avaya IP Office will send a REFER back to the Verizon Business IPCC service, specifying “13035382177” in the Refer-To header. The Verizon Business IPCC service will then send a new Invite to the 1-303-538-2177 destination.

5.11. Alternate Route Selection (ARS)

Alternate Route Selection (ARS) is used to route outbound traffic to the SIP line. To define a new ARS route, right-click **ARS** in the Navigation pane and select **New**. In the Details pane that appears, a collection of matching patterns (similar to short codes) can be entered to route calls as shown below.

To add a new ARS route, right-click **ARS** in the Navigation pane, and select **New**. To view or edit an existing ARS route, select **ARS** in the Navigation pane, and select the appropriate route name in the Group pane.

The following screen shows an example ARS configuration for the route named **SBCE to Vz IPCC**. Verizon Business IPCC service allows for blind transfers using the SIP REFER Method. The sequence of **Xs** used in the **Code** column of the entries to specify the exact number of digits to be expected following the access code. The entry below shows that for calls to area codes in the North American Numbering Plan, the user dials 8, followed by 11 digits. The **Telephone Number** is set to “+.”. This prepends a plus sign (+) to the beginning of the number dialed, denoting a global E.164 number. This is the format preferred by Verizon Business IPCC service for the destination number specified in the Refer-To header. The **Line Group ID** is set to **10** matching the number of the **Outgoing Group** configured on the **Call Details** tab of SIP Line 10 to Verizon Business (Section 5.5.4).

The screenshot shows the ARS configuration window for the route named "SBCE to Vz IPCC". The window has a title bar with the route name and standard window controls. The configuration is organized into several sections:

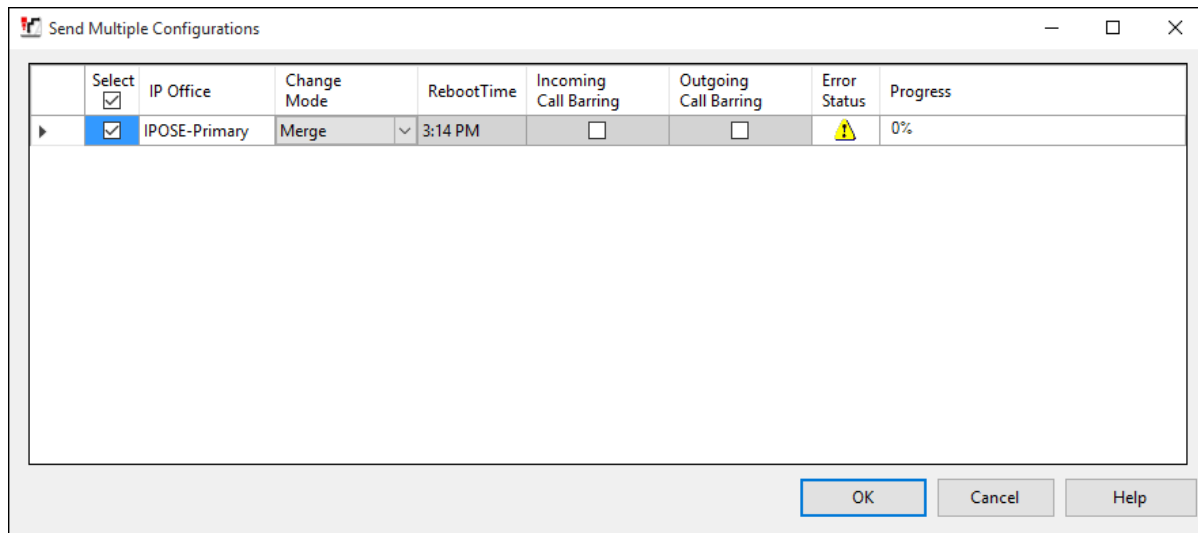
- ARS Section:** Contains fields for "ARS Route ID" (60), "Route Name" (SBCE to Vz IPCC), "Dial Delay Time" (System Default (4)), and "Description" (empty). There are checkboxes for "Secondary Dial tone" (checked) and "Check User Call Barring" (checked). A "SystemTone" dropdown is also present.
- In Service Section:** Includes an "In Service" checkbox (checked) and a "Time Profile" dropdown (set to "<None>").
- Out of Service/Hours Section:** Contains "Out of Service Route" and "Out of Hours Route" dropdowns, both set to "<None>".
- Code Table:** A table with columns: Code, Telephone Number, Feature, and Line Group ID. It contains one entry: Code "1xxxxxxxx", Telephone Number "+.", Feature "Dial", and Line Group ID "10". To the right of the table are buttons for "Add...", "Remove", and "Edit...".
- Alternate Route Section:** Includes "Alternate Route Priority Level" (dropdown set to "1") and "Alternate Route Wait Time" (dropdown set to "5").
- Alternate Route:** A dropdown menu set to "<None>".

Arrows indicate the flow of configuration from the "In Service" checkbox down to the "Alternate Route" dropdown.

5.12. Save Configuration

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 **Reboot** selected for the **Change Mode**, 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.



6. Avaya IP Office Expansion Configuration

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

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'Configuration' tree with 'IP500 Expansion' selected. The main area is titled 'System Inventory' and shows the 'Server Edition Expansion System' configuration. It includes sections for 'Hardware Installed', 'System Settings', and 'Features Configured'.

Section	Details
Hardware Installed	Control Unit: IP 500 V2 Internal Modules: VCM64/PRID U; PHONE8; COMBO6210/ATM4 Expansion Modules: NONE
System Settings	IP Address: 10.5.5.180 Sub-Net Mask: 255.255.255.0 Default Gateway: 10.5.5.2 System Locale: United States (US English) System Location: 8: Miami Device ID: NONE Number of Extensions on System: 19
Features Configured	Licenses Installed: Power User(1); Avaya IP endpoints(3); SIP Trunk Channels(10); Server Edition(1); IP Office Select(1); And more... Connected Extensions: 6723 Users NOT Configured for Voicemail: Analog6727; Digital6723; Fax6728 Users assigned as Ex-Directory: NONE Users assigned for Twinning: NONE Users barred from making Outgoing Calls: NONE Music on Hold: Tone

6.1. Expansion System - Physical Hardware

In the sample configuration, the IP500 V2 Expansion System contained a PHONE8 analog card and a COMBO6210 card, for the support of analog and digital stations. Also included is a VCM64 (Voice Compression Module). Both the VCM64 and the COMBO6210 cards provide voice compression channels to the control unit. Voice compression channels are needed to support VoIP calls, including IP extensions and or IP trunks.

The screenshot displays the Avaya IP Office configuration interface, specifically the 'IP 500 V2' control unit details. The left pane shows the 'Configuration' tree with 'Control Unit (4)' selected. The main area is titled 'IP 500 V2' and shows a table of installed modules and their details.

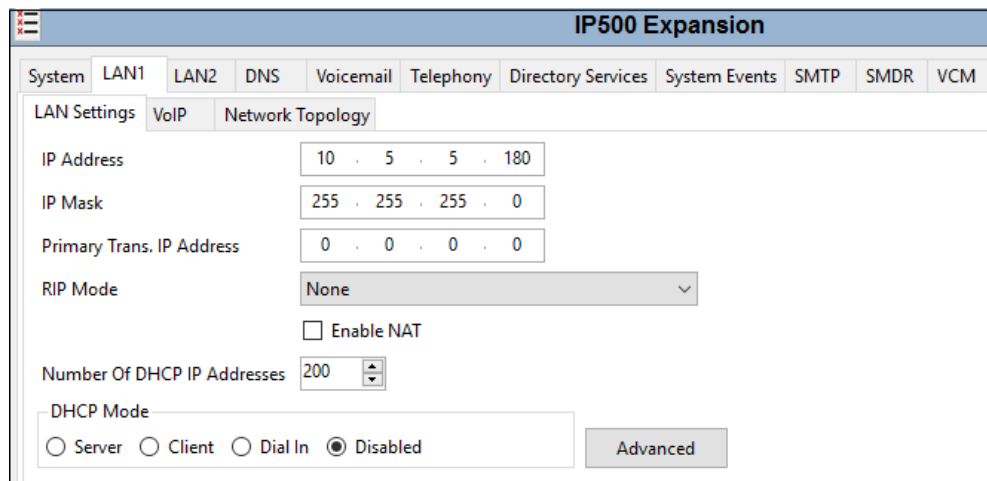
Dev No.	Dev Type	Version
1	IP 500 V2	11.1.0.0.0 build 237
2	VCM64/PRID U	11.1.0.0.0 build 237
3	PHONE8	11.1.0.0.0 build 237
4	COMBO6210/ATM4	11.1.0.0.0 build 237

Below the table, the 'Unit' details are shown:

Unit	
Device Number	1
Unit Type	IP 500 V2
Version	11.1.0.0.0 build 237
Serial Number	00e00706ebf2
Unit IP Address	10.5.5.180
Interconnect Number	0
Module Number	Control Unit

6.2. Expansion System - LAN Settings

In the sample configuration, LAN1 is used to connect the Expansion System to the enterprise network. To view or configure the LAN1 IP address, select **System** on the Navigation pane. Select the **LAN1 → LAN Settings** tab on the Details pane. As shown in **Figure 1**, the IP Address of the Expansion System is **10.5.5.180**. Other parameters on this screen may be set according to customer requirements.



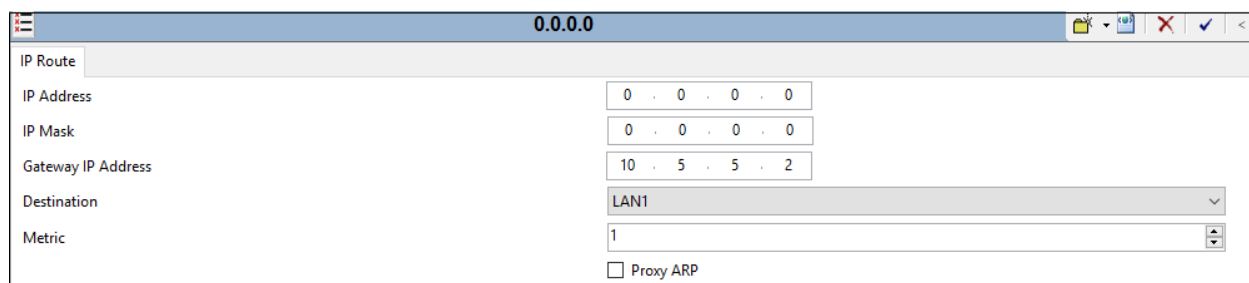
The screenshot shows the 'IP500 Expansion' configuration window. The 'System' tab is selected, and the 'LAN1' sub-tab is active. The 'LAN Settings' section is expanded, showing the following configuration:

- IP Address: 10 . 5 . 5 . 180
- IP Mask: 255 . 255 . 255 . 0
- Primary Trans. IP Address: 0 . 0 . 0 . 0
- RIP Mode: None (dropdown menu)
- ☐ Enable NAT
- Number Of DHCP IP Addresses: 200 (spinner)
- DHCP Mode: ☐ Server ☐ Client ☐ Dial In ☒ Disabled
- Advanced button

Defaults were used on the **VoIP** and **Network Topology** tabs (not shown).

6.3. Expansion System - IP Route

To create an IP route for the Expansion System, right-click on **IP Route** on the left Navigation pane. Select **New** (not shown). The configuration is similar to the one on the Primary server (**Section 5.4**), with the difference that in the reference configuration, the default gateway for the Expansion System is **10.5.5.2**.



The screenshot shows the 'IP Route' configuration window. The 'IP Route' tab is selected, and the 'New' sub-tab is active. The configuration is as follows:

- IP Address: 0 . 0 . 0 . 0
- IP Mask: 0 . 0 . 0 . 0
- Gateway IP Address: 10 . 5 . 5 . 2
- Destination: LAN1 (dropdown menu)
- Metric: 1 (spinner)
- ☐ Proxy ARP

6.4. Expansion System - IP Office Line

The IP Office Lines are automatically created on each server when the Expansion System is added to the solution. Below is the IP Office Line (**Line Number 17**) to the Primary server.

The screenshot shows the 'IP Office Line - Line 17' configuration window. The 'Line' tab is selected. The configuration includes:

- Line Number:** 17
- Transport Type:** WebSocket Client
- Networking Level:** SCN
- Security:** Medium
- Telephone Number:** (empty)
- Prefix:** (empty)
- Outgoing Group ID:** 99999
- Number of Channels:** 250
- Outgoing Channels:** 250
- Gateway:**
 - Address:** 10 . 64 . 19 . 170
 - Location:** 2: Denver
 - Password:** (masked)
 - Confirm Password:** (masked)
 - Port:** 443
- SCN Resiliency Options:**
 - ☐ Supports Resiliency
 - ☐ Backs up my IP phones
 - ☐ Backs up my hunt groups
 - ☐ Backs up my IP DECT phones
- Description:** (empty)

In the reference configuration, a fax machine is connected to one of the analog ports on the Expansion System. To accommodate T.38 fax, select the **VoIP Settings** tab and set **Fax Transport Support** to **T38 Fallback**. The **Media Security** drop-down menu is set to “**Same as System (Preferred)**” to have IP Office use the system setting for media security set in **Section 5.3.3** to encrypted RTP.

The screenshot shows the 'IP Office Line - Line 17' configuration window with the 'VoIP Settings' tab selected. The configuration includes:

- Codec Selection:** System Default
- Unused Codecs:** G.711 ALAW 64K, G.723.1 6K3 MP-MLQ
- Selected Codecs:** G.722 64K, G.711 ULAW 64K, G.729(a) 8K CS-ACELP
- Fax Transport Support:** T38 Fallback
- Call Initiation Timeout (s):** 4
- Media Security:** Same as System (Preferred)
- Advanced Media Security Options:**
 - ☒ Same As System
- Other Options:**
 - ☐ VoIP Silence Suppression
 - ☒ Out Of Band DTMF
 - ☒ Allow Direct Media Path

Select the **T38 Fax** tab. The **Use Default Values** box is unchecked, and the **T38 Fax Version** is set to **0**. In the **Redundancy** area, the **Low Speed** and **High Speed** parameters are set to **2**. All other values are left at default.

The screenshot shows the configuration page for 'IP Office Line - Line 17'. The 'T38 Fax' tab is selected. The 'T38 Fax Version' is set to 0. The 'Transport' is set to UDPTL. In the 'Redundancy' section, 'Low Speed' and 'High Speed' are both set to 2. The 'TCF Method' is set to Trans TCF. The 'Max Bit Rate (bps)' is set to 14400. The 'EFlag Start Timer (ms)' is set to 2600, and the 'EFlag Stop Timer (ms)' is set to 2300. The 'Tx Network Timeout (sec)' is set to 150. On the right, several checkboxes are visible: 'Scan Line Fix-up' (checked), 'TFOP Enhancement' (checked), 'Disable T30 ECM' (unchecked), 'Disable EFlags For First DIS' (unchecked), 'Disable T30 MR Compression' (unchecked), and 'NSF Override' (unchecked). Below these, 'Country Code' and 'Vendor Code' are both set to 0. At the bottom left, the 'Use Default Values' checkbox is unchecked.

6.5. Expansion System - Short Codes

Similar to the configuration of the Primary server in **Section 5.8**, a Short Code is created to access ARS. In the reference configuration, the **Line Group ID** is set to an ARS route illustrated in the next section.

The screenshot shows the configuration page for a Short Code named '9N: Dial'. The 'Short Code' tab is selected. The 'Code' is set to 9N. The 'Feature' is set to Dial. The 'Telephone Number' is set to N. The 'Line Group ID' is set to 53: To-Primary. The 'Locale' is set to a default value. The 'Force Account Code' and 'Force Authorization Code' checkboxes are both unchecked.

6.6. Expansion System - Automatic Route Selection – ARS

The following screen shows an example ARS configuration for the route named **To-Primary** on the Expansion System. The **Telephone Number** is set to **9N**. The **Line Group ID** is set to **99999** matching the number of the **Outgoing Group ID** configured on the IP Office Line 17 to the Primary server (Section 6.4).

Code	Telephone Number	Feature	Line Group ID
N	9N	Dial	99999

6.7. Save IP Office Expansion System Configuration

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 **Reboot** 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** to proceed.

Select	IP Office	Change Mode	RebootTime	Incoming Call Barring	Outgoing Call Barring	Error Status	Progress
<input checked="" type="checkbox"/>	IP500 Expansion	Merge	12:12 PM	<input type="checkbox"/>	<input type="checkbox"/>		0%

7. Configure Avaya Session Border Controller for Enterprise

In the reference configuration, Avaya SBCE is used as an edge device between the CPE and Verizon Business.

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

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



The screenshot displays the login interface for the Avaya Session Border Controller for Enterprise. On the left, the Avaya logo is shown in red, with the text "Session Border Controller for Enterprise" below it. To the right, under the heading "Log In", there are input fields for "Username:" (containing "ucsec") and "Password:" (masked with dots). A "Log In" button is positioned below the password field. Further down, a "WELCOME TO AVAYA SBC" message is followed by a disclaimer: "Unauthorized access to this machine is prohibited. This system is for the use authorized users only. Usage of this system may be monitored and recorded by system personnel." Below this is a consent statement: "Anyone using this system expressly consents to such monitoring and is advised that if such monitoring reveals possible evidence of criminal activity, system personnel may provide the evidence from such monitoring to law enforcement officials." At the bottom, the copyright notice "© 2011 - 2020 Avaya Inc. All rights reserved." is displayed.

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

Device: EMSAlarmsIncidentsStatusLogsDiagnosticsUsersSettingsHelpLog Out

Session Border Controller for Enterprise

AVAYA

EMS Dashboard

Device Management

- System Administration
- Backup/Restore
- Monitoring & Logging

Dashboard

Information

System Time	06:18:31 AM MDT	Refresh
Version	8.1.0.0-14-18490	
GUI Version	8.1.0.0-18490	
Build Date	Mon Feb 03 17:23:09 UTC 2020	
License State	OK	
Aggregate Licensing Overages	0	
Peak Licensing Overage Count	0	
Last Logged in at	06/23/2020 06:14:44 MDT	
Failed Login Attempts	0	

Active Alarms (past 24 hours)
None found.

Incidents (past 24 hours)
None found.

Notes
No notes found.

Installed Devices

EMS
SBCE8-90

Add

7.1. Device Management – Status

Select **Device Management** on the left-hand menu. A list of installed devices is shown on the **Devices** tab on the right pane. In the case of the sample configuration, a single device named **SBCE8-90** is shown. Verify that the **Status** column shows **Commissioned**. If not, contact your Avaya representative. To view the configuration of this device, click **View** on the screen below.

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

Device: EMSAlarmsIncidentsStatusLogsDiagnosticsUsersSettingsHelpLog Out

Session Border Controller for Enterprise

AVAYA

EMS Dashboard

Device Management

- System Administration
- Backup/Restore
- Monitoring & Logging

Device Management

Devices

Updates

SSL VPN

Licensing

Key Bundles

Device Name	Management IP	Version	Status	
SBCE8-90	10.64.90.90	8.1.0.0-14-18490	Commissioned	RebootShutdownRestart ApplicationViewEditUninstall

The **System Information** screen shows the **Network Configuration**, **DNS Configuration** and **Management IP(s)** information provided during installation, corresponding to **Figure 1**. In the shared test environment, the highlighted **A1** and **B1** IP addresses are the ones relevant to the configuration of the SIP trunk to Verizon. Other IP addresses assigned to interfaces **A1** and **B2** on the screen below are used to support remote workers and are not the focus of these Application Notes. Note that the **Management IP** must be on a separate subnet from the IP interfaces designated for SIP traffic.

System Information: SBCE8-90

General Configuration

Appliance Name

SBCE8-90

Box Type

SIP

Deployment Mode

Proxy

Device Configuration

HA Mode

No

Two Bypass Mode

No

Dynamic License Allocation

	Min License Allocation	Max License Allocation
Standard Sessions	10	100
Advanced Sessions	10	100
Scopia Video Sessions	10	100
CES Sessions	10	100
Transcoding Sessions	10	100
CLID	---	
Encryption	<input checked="" type="checkbox"/>	
Available:	Yes	

Network Configuration

IP	Public IP	Network Prefix or Subnet Mask	Gateway	Interface
10.64.91.48	10.64.91.48	255.255.255.0	10.64.91.1	A1
10.64.91.49	10.64.91.49	255.255.255.0	10.64.91.1	A1
10.64.91.50	10.64.91.50	255.255.255.0	10.64.91.1	A1
1.1.1.2	1.1.1.2	255.255.255.0	1.1.1.1	B1
		255.255.255.128		B2
		255.255.255.128		B2

DNS Configuration

Primary DNS

172.30.209.4

Secondary DNS

DNS Location

DMZ

DNS Client IP

1.1.1.2

Management IP(s)

IP #1 (IPv4)

10.64.90.90

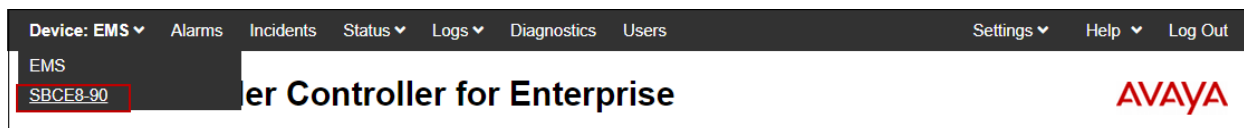
7.2. TLS Management

Note – Testing was done using identity certificates signed by a local certificate authority. The procedure to create and obtain these certificates is outside the scope of these Application Notes.

In the reference configuration, TLS transport is used for the communication between IP Office and Avaya SBCE. The following procedures show how to view the certificates and configure the profiles to support the TLS connection.

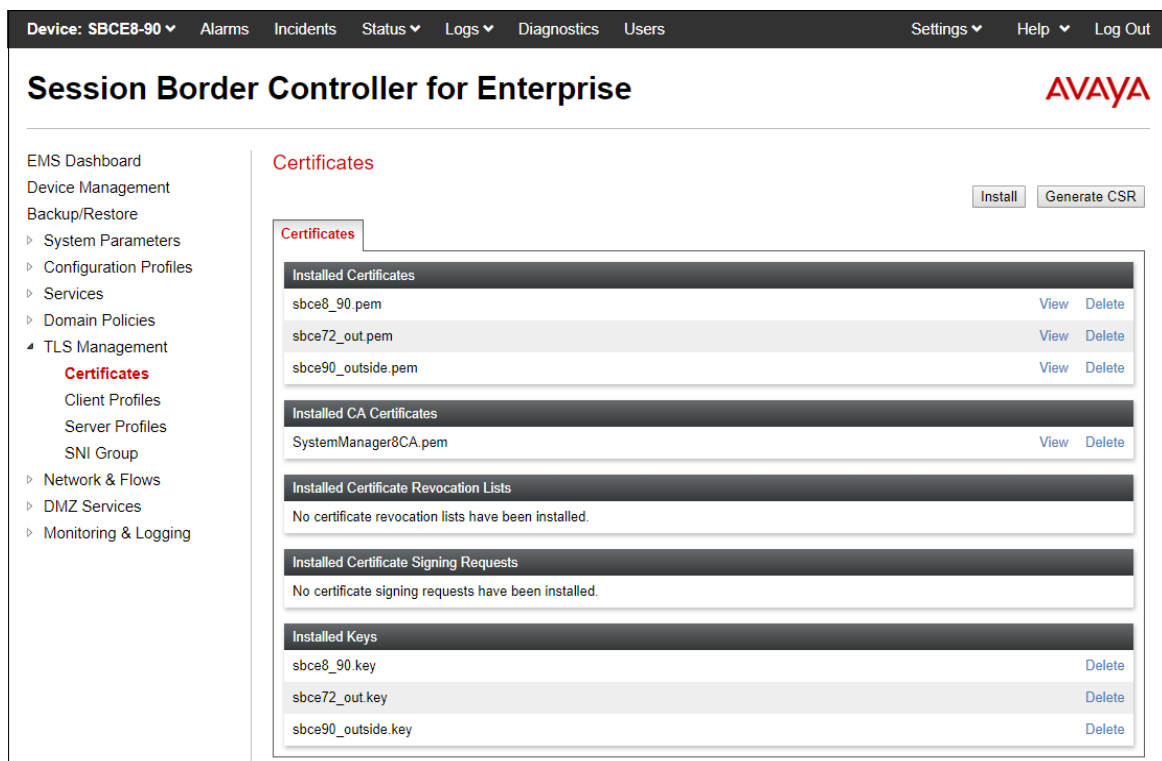
7.2.1. Verify TLS Certificates – Avaya Session Border Controller for Enterprise

To access the SBCE configuration menus, select the SBCE device from the top navigation menu.



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

Select **TLS Management** → **Certificates** from the left-hand menu. Verify the root CA certificate is present in the **Installed CA Certificates** area. The signed identity certificate is present in the **Installed Certificates** area. The private key associated with the identity certificate is present in the **Installed Keys** area.



7.2.2. Server Profiles

Navigate to **TLS Management** → **Server Profiles** and click the **Add** button to add a new profile or select an existing profile. Enter a descriptive **Profile Name** such as **Inside_Server** show below. Select the Avaya SBCE identity certificate for the inside interface from the **Certificate** drop-down menu. In the reference configuration this is **sbce8_90.pem**. Select **None** from the **Peer Verification** drop-down menu. Click **Next** and accept default values for the next screen, then click **Finish** (not shown).

Edit Profile

WARNING: Due to the way OpenSSL handles cipher checking, Cipher Suite validation will pass even if one or more of the ciphers are invalid as long as at least one cipher is valid. Make sure to carefully check your entry as invalid or incorrectly entered Cipher Suite custom values may cause catastrophic problems.

Changing the certificate in a TLS Profile which has SNI enabled may cause existing Reverse Proxy entries which utilize this TLS Profile to become invalid.

TLS Profile

Profile Name:

Certificate:

SNI Options:

SNI Group:

Certificate Verification

Peer Verification:

Peer Certificate Authorities:

Peer Certificate Revocation Lists:

Verification Depth:

The following screen shows the completed TLS **Server Profile** form:

Session Border Controller for Enterprise AVAYA

EMS Dashboard
Device Management
Backup/Restore
System Parameters
Configuration Profiles
Services
Domain Policies
TLS Management
Certificates
Client Profiles
Server Profiles
SNI Group
Network & Flows
DMZ Services
Monitoring & Logging

Server Profiles: Inside_Server

Server Profile

TLS Profile

Profile Name:

Certificate:

SNI Options:

Certificate Verification

Peer Verification:

Extended Hostname Verification: ☐

Renegotiation Parameters

Renegotiation Time:

Renegotiation Byte Count:

Handshake Options

Version: ☒ TLS 1.2 ☐ TLS 1.1 ☐ TLS 1.0

Ciphers: ☒ Default ☐ FIPS ☐ Custom

Value:

7.2.3. Client Profiles

Navigate to **TLS Management** → **Client Profiles** and click the **Add** button to add a new profile or select an existing profile. Enter a descriptive **Profile Name**, such as **Inside_Client** show below. Select the identity certificate from the **Certificate** drop-down menu. In the reference configuration this is **sbce8_90.pem**. The **Peer Certificate Authorities** field is set to the root certificate used to verify the IP Office certificate, e.g., **SystemManager8CA.pem**. The **Verification Depth** field is set to **1**. Click **Next** and accept default values for the next screen and click **Finish** (not shown).

The 'Edit Profile' dialog box contains a warning message at the top: 'WARNING: Due to the way OpenSSL handles cipher checking, Cipher Suite validation will pass even if one or more of the ciphers are invalid as long as at least one cipher is valid. Make sure to carefully check your entry as invalid or incorrectly entered Cipher Suite custom values may cause catastrophic problems. Changing the certificate in a TLS Profile which has SNI enabled may cause existing Reverse Proxy entries which utilize this TLS Profile to become invalid.'

The form is divided into two main sections: 'TLS Profile' and 'Certificate Verification'.

TLS Profile Section:

- Profile Name:** Inside_Client
- Certificate:** sbce8_90.pem
- SNI:** ☐ Enabled

Certificate Verification Section:

- Peer Verification:** Required
- Peer Certificate Authorities:** AvayaDeviceEnrollmentCAchain.crt, SystemManager8CA.pem
- Peer Certificate Revocation Lists:** (Empty field)
- Verification Depth:** 1
- Extended Hostname Verification:** ☐
- Server Hostname:** (Empty field)

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

The following screen shows the completed TLS **Client Profile** form:

The screenshot shows the 'Session Border Controller for Enterprise' interface. On the left is a navigation menu with options like EMS Dashboard, Device Management, Backup/Restore, System Parameters, Configuration Profiles, Services, Domain Policies, TLS Management, Certificates, Client Profiles, Server Profiles, SNI Group, Network & Flows, DMZ Services, and Monitoring & Logging. The 'Client Profiles' section is selected.

The main area displays 'Client Profiles: Inside_Client' with an 'Add' button and a 'Delete' button. Below this is a list of client profiles: 'Client Profiles', 'Inside_Client', and 'Outside_Client'. The 'Inside_Client' profile is selected.

The 'Client Profile' form for 'Inside_Client' is shown, containing the following fields:

- Profile Name:** Inside_Client
- Certificate:** sbce8_90.pem
- SNI:** ☐ Enabled
- Certificate Verification:** Required
- Peer Certificate Authorities:** SystemManager8CA.pem
- Peer Certificate Revocation Lists:** ---
- Verification Depth:** 1
- Extended Hostname Verification:** ☐
- Renegotiation Parameters:** Renegotiation Time: 0, Renegotiation Byte Count: 0
- Handshake Options:** Version: ☒ TLS 1.2, ☐ TLS 1.1, ☐ TLS 1.0; Ciphers: ☒ Default, ☐ FIPS, ☐ Custom; Value: HIGH IDH IADH IMD5 IaNULL IeNULL @STRENGTH

An 'Edit' button is located at the bottom right of the form.

7.3. Network Management

The Network Management screen is where the network interface settings are configured and enabled. During the installation process of Avaya SBCE, certain network-specific information is defined such as device IP address(es), public IP address(es), netmask, gateway, etc. to interface the device to the network. It is this information that populates the various Network Management tab displays, which can be edited as needed to optimize device performance and network efficiency. Navigate to **Networks & Flows** → **Network Management** from the menu on the left-hand side. The **Interfaces** tab displays the enabled/disabled interfaces. In the reference configuration, interfaces A1 and B1 are used.

The screenshot shows the 'Session Border Controller for Enterprise' interface. On the left is a navigation menu with options like EMS Dashboard, Device Management, Backup/Restore, System Parameters, Configuration Profiles, Services, Domain Policies, TLS Management, and Network & Flows. Under 'Network & Flows', 'Network Management' is selected. The main area is titled 'Network Management' and has two tabs: 'Interfaces' (active) and 'Networks'. An 'Add VLAN' button is in the top right. A table lists interfaces:

Interface Name	VLAN Tag	Status
A1		Enabled
A2		Disabled
B1		Enabled
B2		Enabled

Select the **Networks** tab to display the IP provisioning for the A1 and B1 interfaces. These values are normally specified during installation. These can be modified by selecting **Edit**; however, some of these values may not be changed if associated provisioning is in use.

- **A1: 10.64.91.48** – “Inside” IP address, configured for Verizon Business IPCC VoIP Inbound Service to IP Office.
- **A1: 10.64.91.50** – “Inside” IP address configured for Verizon Business IP Trunk SIP Trunk Service to IP Office. This address is used to connect IP Office to the Verizon Business IP Trunk service as illustrated in reference [IPT-IPO111SBC81].
- **B1: 1.1.1.2** – “Outside” IP address toward the Verizon SIP trunk. This address is known to Verizon and is associated with the FQDN *adevc.avaya.globalipcom.com*.

The screenshot shows the 'Session Border Controller for Enterprise' interface, same as above but with the 'Networks' tab selected. An 'Add' button is in the top right. A table lists network configurations:

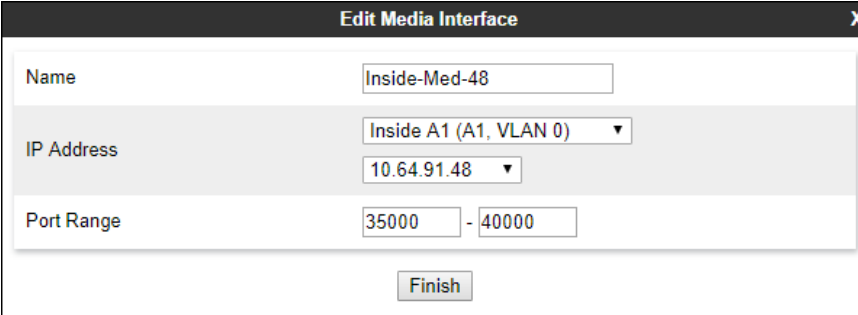
Name	Gateway	Subnet Mask / Prefix Length	Interface	IP Address	
Inside A1	10.64.91.1	255.255.255.0	A1	10.64.91.48, 10.64.91.49, 10.64.91.50	Edit Delete
Verizon B1	1.1.1.1	255.255.255.0	B1	1.1.1.2	Edit Delete
Public B2		255.255.255.128	B2		Edit Delete

7.4. Media Interfaces

Media Interfaces are created to specify the IP address and port range in which the Avaya SBCE will accept media streams on each interface. Packets leaving the interfaces of the Avaya SBCE will advertise this IP address, and one of the ports in this range as the listening IP address and port in which the SBCE will accept media from the connected server. Create a SIP Media Interface for both the inside and outside IP interfaces.

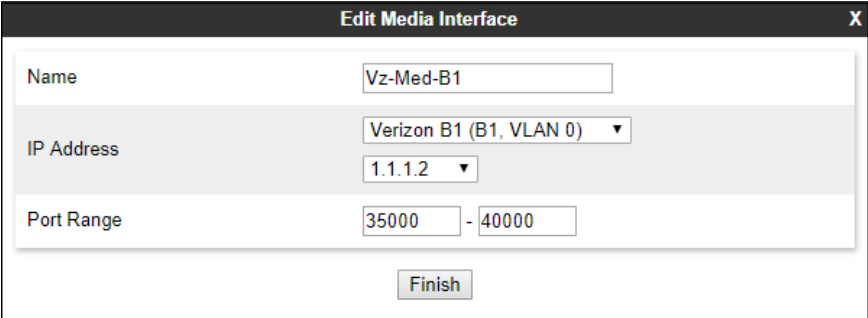
To create a new Media Interface, navigate to Select **Network & Flows → Media Interface** from the menu on the left-hand side and select **Add** (not shown).

The screen below shows the **Inside-Med-48** Media Interface created toward the IP Office. On the **IP Address** drop-down menus, **Inside-A1 (A1,VLAN0)** and **10.64.91.48** are selected. Default **Port Range** values are used.



The screenshot shows a dialog box titled "Edit Media Interface" with a close button (X) in the top right corner. The dialog contains three main sections: "Name" with a text input field containing "Inside-Med-48"; "IP Address" with a dropdown menu showing "Inside A1 (A1, VLAN 0)" and a sub-dropdown showing "10.64.91.48"; and "Port Range" with two input fields containing "35000" and "40000" separated by a hyphen. A "Finish" button is located at the bottom center of the dialog.

The screen below shows the **Vz-Med-B1** Media Interface created toward Verizon. On the **IP Address** drop-down menus, **Verizon B1 (B1,VLAN0)** and **1.1.1.2** are selected. Default **Port Range** values are used.



The screenshot shows a dialog box titled "Edit Media Interface" with a close button (X) in the top right corner. The dialog contains three main sections: "Name" with a text input field containing "Vz-Med-B1"; "IP Address" with a dropdown menu showing "Verizon B1 (B1, VLAN 0)" and a sub-dropdown showing "1.1.1.2"; and "Port Range" with two input fields containing "35000" and "40000" separated by a hyphen. A "Finish" button is located at the bottom center of the dialog.

7.5. Signaling Interfaces

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 both the inside and outside IP interfaces.

To create a new Signaling Interface, navigate to Select **Network & Flows** → **Media Interface** from the menu on the left-hand side and select **Add** (not shown).

The screen below shows the **Inside-Sig-48** Signaling Interface created toward the IP Office. On the **IP Address** drop-down menus, **Inside-A1 (A1,VLAN0)** and **10.64.91.48** are selected. **TLS Port 5061** is used. The TLS server profile created in **Section 7.2.2** (e.g., **Inside_Server**) is selected on the TLS Profile drop-down menu.

The screenshot shows the 'Edit Signaling Interface' window for 'Inside-Sig-48'. The form includes the following fields and values:

Field	Value
Name	Inside-Sig-48
IP Address	Inside A1 (A1, VLAN 0) (dropdown) 10.64.91.48 (dropdown)
TCP Port	(empty)
UDP Port	(empty)
TLS Port	5061
TLS Profile	Inside_Server (dropdown)
Enable Shared Control	<input type="checkbox"/>
Shared Control Port	(empty)

A 'Finish' button is located at the bottom right of the form.

The screen below shows the **Vz-Sig-B1** Signaling Interface created toward Verizon. On the **IP Address** drop-down menus, **Verizon-B1 (B1,VLAN0)** and **1.1.1.2** are selected. **UDP Port 5060** is used.

The screenshot shows the 'Edit Signaling Interface' window for 'Vz-Sig-B1'. The form includes the following fields and values:

Field	Value
Name	Vz-Sig-B1
IP Address	Verizon B1 (B1, VLAN 0) (dropdown) 1.1.1.2 (dropdown)
TCP Port	(empty)
UDP Port	5060
TLS Port	(empty)
TLS Profile	None (dropdown)
Enable Shared Control	<input type="checkbox"/>
Shared Control Port	(empty)

A 'Finish' button is located at the bottom right of the form.

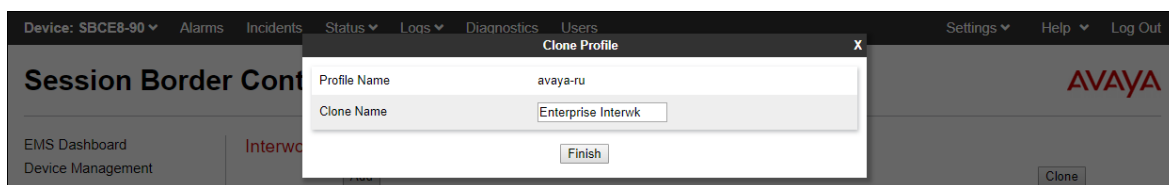
7.6. Server Interworking Profile

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

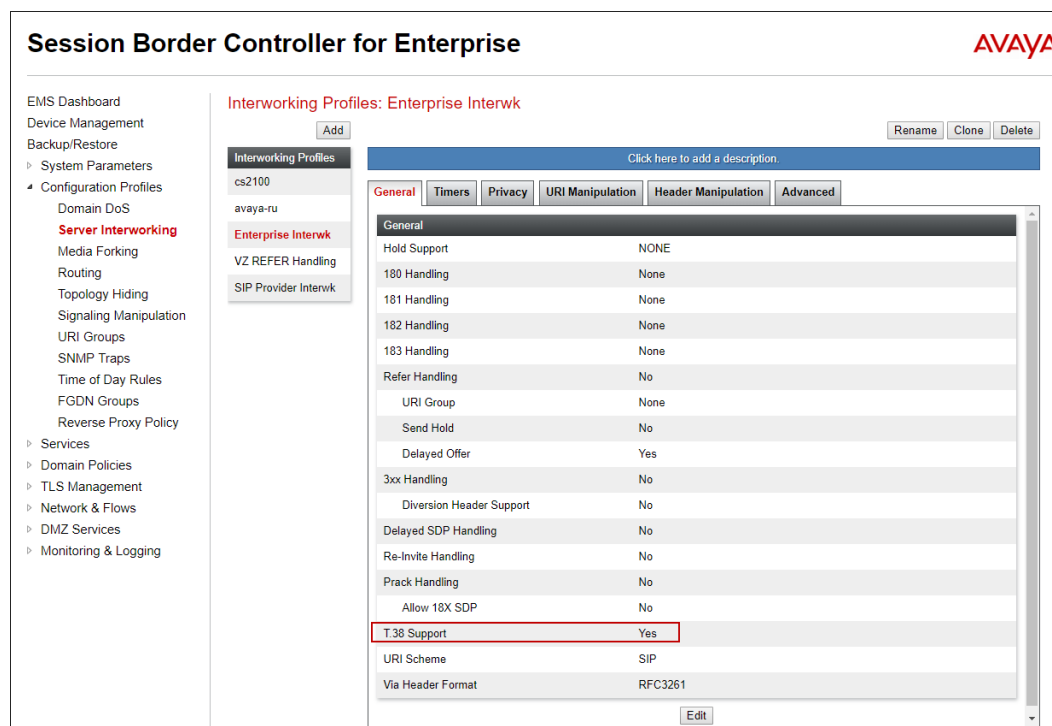
In the reference configuration, separate Server Interworking Profiles were created for IP Office and Verizon Business IPCC service.

7.6.1. Server Interworking Profile – IP Office

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

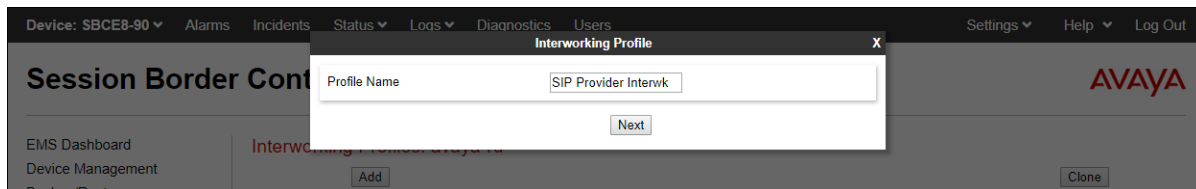


The following screen shows the **Enterprise Interwk** profile used in the reference configuration, with **T.38 Support** set to **Yes**. To modify the profile, scroll down to the bottom of the screen and click **Edit**. Select the **T.38 Support** parameter and then click **Next** and then **Finish**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



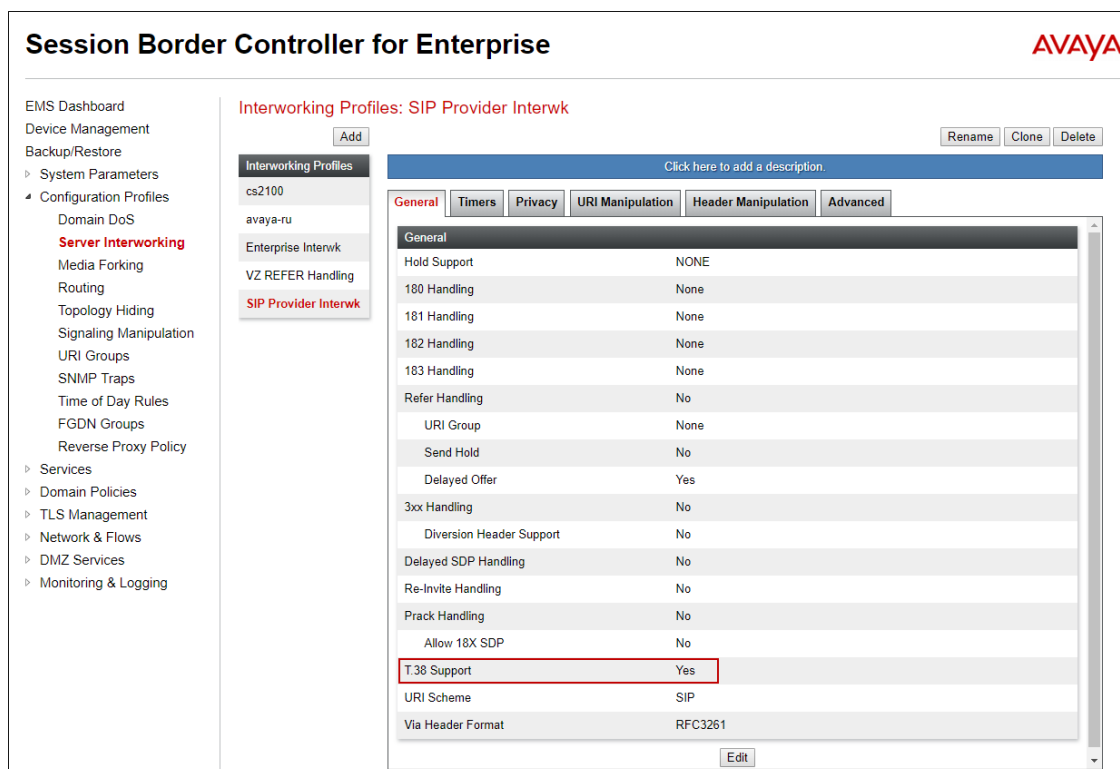
7.6.2. Server Interworking Profile – Verizon

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



The screenshot shows a web interface for a Session Border Controller. A modal dialog titled 'Interworking Profile' is open. It has a 'Profile Name' input field containing the text 'SIP Provider Interwk' and a 'Next' button. The background shows the 'Session Border Controller for Enterprise' dashboard with a sidebar menu and a top navigation bar.

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



The screenshot displays the 'Interworking Profiles: SIP Provider Interwk' configuration page. The left sidebar shows the navigation menu with 'Server Interworking' highlighted. The main content area shows the 'General' tab selected, displaying a list of settings. The 'T.38 Support' setting is highlighted with a red box and set to 'Yes'.

Setting	Value
Hold Support	NONE
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
URI Group	None
Send Hold	No
Delayed Offer	Yes
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
Re-Invite Handling	No
Prack Handling	No
Allow 18X SDP	No
T.38 Support	Yes
URI Scheme	SIP
Via Header Format	RFC3261

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

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, Global Parameters, Global Profiles, Domain DoS, Server Interworking, Media Forking, Routing, Server Configuration, Topology Hiding, Signaling Manipulation, and Manipulation. The main content area is titled 'Interworking Profiles: SIP Provider Interwk'. It features a list of profiles on the left: cs2100, avaya-ru, Enterprise Interwk, and SIP Provider Interwk (selected). The main panel shows the 'Timers' tab for the selected profile. The 'SIP Timers' section includes a table with the following data:

SIP Timers	Value
Min-SE	---
Init Timer	---
Max Timer	---
Trans Expire	4 seconds
Invite Expire	---

Buttons for 'Rename', 'Clone', 'Delete', and 'Edit' are visible.

Default parameters were used for the **Privacy**, **URI Manipulation**, and **Header Manipulation** tabs (not shown). On the **Advanced** tab, **Record Routes** is set to **Both Sides**. Default values can be used for all other fields.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface, specifically the 'Advanced' tab for the 'SIP Provider Interwk' profile. The left sidebar is similar to the previous screenshot but includes additional options like EMS Dashboard, Device Management, Configuration Profiles, URI Groups, SNMP Traps, Time of Day Rules, FGDN Groups, Reverse Proxy Policy, Services, Domain Policies, TLS Management, and Network & Flows. The main content area shows the 'Advanced' tab selected. The 'Record Routes' section includes a table with the following data:

Record Routes	Value
Record Routes	Both Sides
Include End Point IP for Context Lookup	No
Extensions	None
Diversion Manipulation	No
Has Remote SBC	Yes
Route Response on Via Port	No
Relay INVITE Replace for SIPREC	No
MOBX Re-INVITE Handling	No

The 'DTMF' section includes a table with the following data:

DTMF	Value
DTMF Support	None

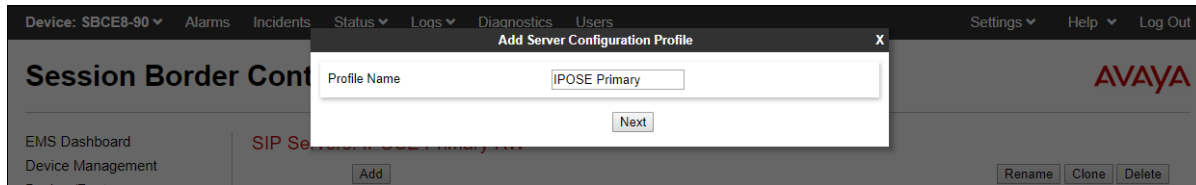
Buttons for 'Rename', 'Clone', 'Delete', and 'Edit' are visible.

7.7. SIP Servers Profiles

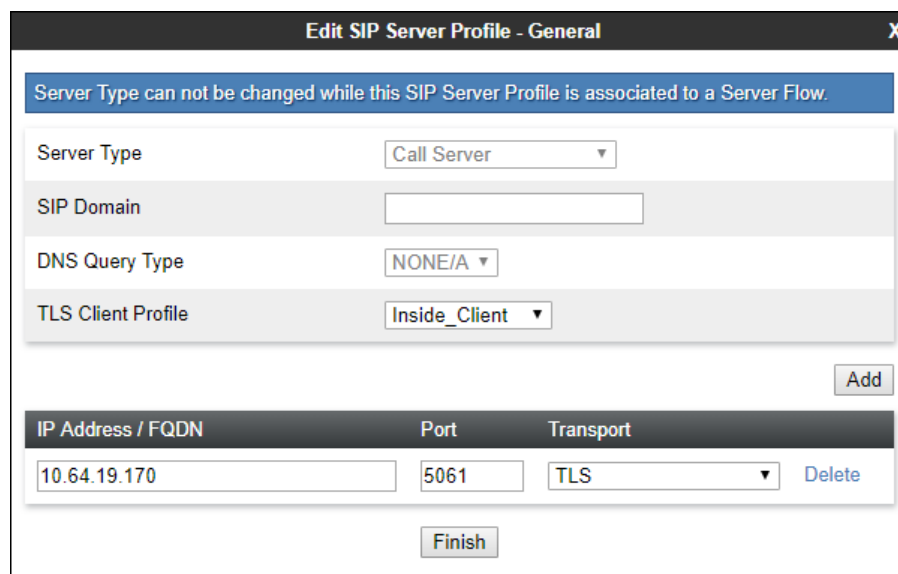
The **SIP Server Profile** contains parameters to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters, DoS security statistics, and trusted domains.

7.7.1. SIP Server Profile – IP Office

To add a SIP Server Profile for IP Office, navigate to **Services** → **SIP Servers** on the left-hand menu and click **Add**. Enter a descriptive name for the new profile and click **Next**.



The following screen illustrates the SIP Server Profile named **IPOSE Primary**. In the **General** parameters, the **Server Type** is set to **Call Server**. In the **IP Address / FQDN** field, the IP Address of IP Office LAN 1 interface in the sample configuration is entered. This IP address is **10.64.19.170**. Under **Port**, **5061** is entered, and the **Transport** parameter is set to **TLS**. The TLS profile **Inside_Client** created in **Section 7.2.3** is selected for **TLS Client Profile**. If adding the profile, click **Next** (not shown) to proceed. If editing an existing profile, click **Finish**.



IP Address / FQDN	Port	Transport
10.64.19.170	5061	TLS

Default values can be used on the **Authentication** tab, click **Next** (not shown) to proceed to the **Heartbeats** tab. The Avaya SBCE can be configured to source “heartbeats” in the form of PINGs or SIP OPTIONS towards IP Office. Check the **Enable Heartbeat** box and select **OPTIONS** from the **Method** drop-down menu. Select the desired frequency that the SBCE will source OPTIONS. The **From URI** and **To URI** may be filled in to configure easily identifiable URIs to appear in SIP OPTIONS sourced by the Avaya SBCE.

SIP Servers: IPOSE Primary

Rename Clone Delete

General Authentication **Heartbeat** Registration Ping Advanced

Enable Heartbeat	<input checked="" type="checkbox"/>
Method	OPTIONS
Frequency	60 seconds
From URI	sbce@silipose.customera.com
To URI	IPOSE@silipose.customera.com

Edit

On the **Advanced** tab, select the **Enable Grooming** checkbox. The **Interworking Profile** is set to the **Enterprise Interwk** profile created in **Section 7.6.1** for IP Office.

SIP Servers: IPOSE Primary

Rename Clone Delete

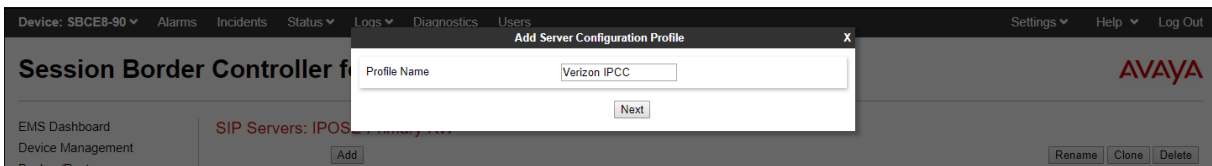
General Authentication Heartbeat Registration Ping **Advanced**

Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input checked="" type="checkbox"/>
Interworking Profile	Enterprise Interwk
Signaling Manipulation Script	None
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>
Tolerant	<input type="checkbox"/>
URI Group	None

Edit

7.7.2. SIP Server Profile - Verizon

To add a SIP Server Profile for Verizon, navigate to **Services → SIP Servers** and click **Add**. Enter a descriptive name for the new profile and click **Next**.



The following screens illustrate the SIP Server Profile named **Verizon IPCC**. In the **General** parameters, the **Server Type** is set to **Trunk Server**. The **DNS Query Type** is set to **NONE/A**. In the **IP Address / FQDN** field, the Verizon-provided IP address is entered. This is **172.30.205.55**. Under **Port**, **5072** is entered, and the **Transport** parameter is set to **UDP**. If adding the profile, click **Next** (not shown) to proceed. If editing an existing profile, click **Finish**.

IP Address / FQDN	Port	Transport
172.30.205.55	5072	UDP

Default values can be used on the **Authentication** tab, click **Next** (not shown) to proceed to the **Heartbeats** tab. The Avaya SBCE can be configured to source “heartbeats” in the form of SIP OPTIONS towards Verizon. This configuration is optional. Independent of whether the Avaya SBCE is configured to source SIP OPTIONS towards Verizon, Verizon will receive OPTIONS from the IP Office site as a result of the **Check OOS** parameter being enabled on IP Office (see **Section 5.5.2**). When IP Office sends SIP OPTIONS to the inside IP Address of the Avaya SBCE, the Avaya SBCE will send SIP OPTIONS to Verizon Business. When Verizon Business responds, the Avaya SBCE will pass the response to IP Office.

Check the **Enable Heartbeat** box and select **OPTIONS** from the **Method** drop-down menu. Select the desired frequency that the SBCE will source OPTIONS. The **From URI** and **To URI** may be filled in to configure easily identifiable URIs to appear in SIP OPTIONS sourced by the Avaya SBCE.

SIP Servers: Verizon IPCC

Rename Clone Delete

General Authentication **Heartbeat** Registration Ping Advanced

Enable Heartbeat	<input checked="" type="checkbox"/>
Method	OPTIONS
Frequency	60 seconds
From URI	SBCE@adevc.avaya.globalipcom.com
To URI	VzIPCC@172.30.205.55

Edit

On the **Advanced** tab, **Enable Grooming** is not used for UDP connections and left unchecked. The **Interworking Profile** is set to **SIP Provider Interwk** created in **Section 7.6.2** for Verizon.

SIP Servers: Verizon IPCC

Rename Clone Delete

General Authentication Heartbeat Registration Ping **Advanced**

Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	SIP Provider Interwk
Signaling Manipulation Script	None
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>
Tolerant	<input type="checkbox"/>
URI Group	None

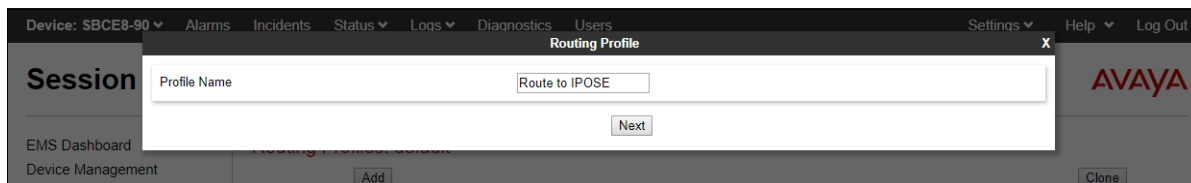
Edit

7.8. Routing Profiles

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. Separate Routing Profiles were created in the reference configuration for IP Office and the Verizon Business IPCC service.

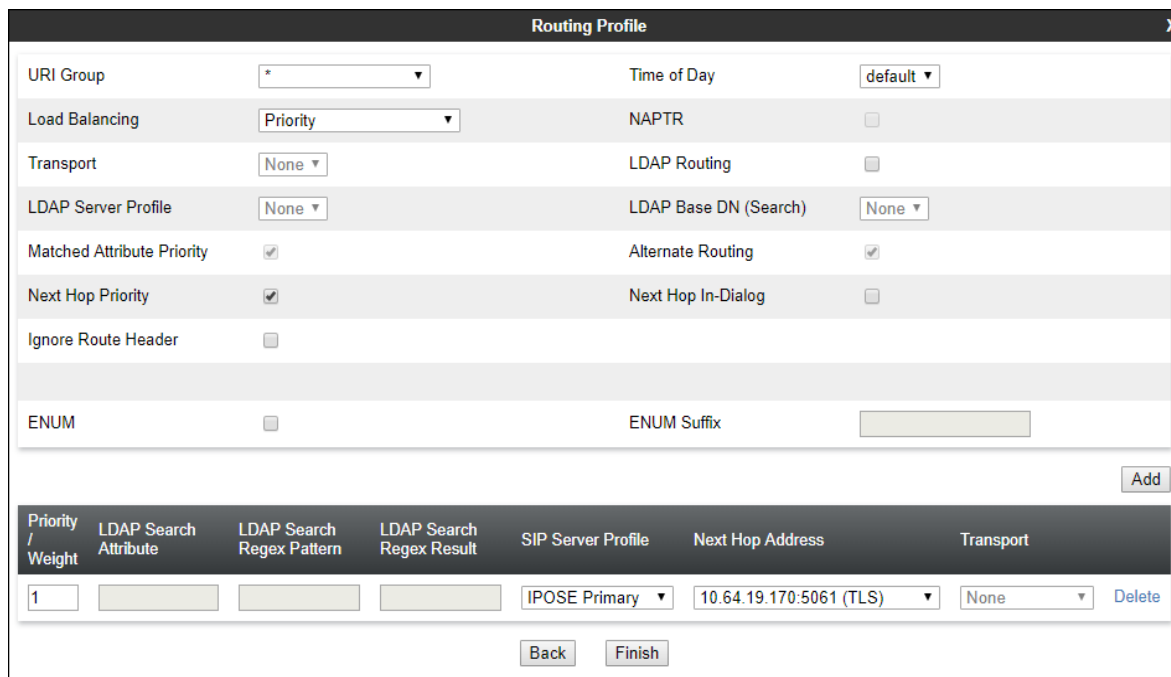
7.8.1. Routing Profile – IP Office

To add a routing profile for the IP Office, navigate to **Configuration Profiles → Routing** and select **Add**. Enter a **Profile Name** and click **Next** to continue.



The screenshot shows the 'Routing Profile' configuration window. The 'Profile Name' field is populated with 'Route to IPOSE'. The 'Next' button is visible. The window has a dark header with navigation links like 'Device: SBCE8-90', 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The left sidebar shows 'Session', 'EMS Dashboard', and 'Device Management'. The AVAYA logo is in the top right corner.

The following screen shows the Routing Profile **Route to IPOSE** created in the reference configuration. The parameters in the top portion of the profile are left at their default settings. The **Priority / Weight** parameter is set to **1**, and the IP Office **SIP Server Profile**, created in **Section 7.7.1**, is selected from the drop-down menu. The **Next Hop Address** is automatically selected with the values from the SIP Server Profile, and **Transport** becomes greyed out. Click **Finish**.



The screenshot shows the 'Routing Profile' configuration window with the 'Route to IPOSE' profile. The 'Priority / Weight' is set to 1, and the 'SIP Server Profile' is set to 'IPOSE Primary'. The 'Next Hop Address' is '10.64.19.170:5061 (TLS)'. The 'Transport' is set to 'None' and is greyed out. The 'Finish' button is visible.

URI Group	Time of Day
*	default

Load Balancing	NAPTR
Priority	<input type="checkbox"/>

Transport	LDAP Routing
None	<input type="checkbox"/>

LDAP Server Profile	LDAP Base DN (Search)
None	None

Matched Attribute Priority	Alternate Routing
<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>

Next Hop Priority	Next Hop In-Dialog
<input checked="" type="checkbox"/>	<input type="checkbox"/>

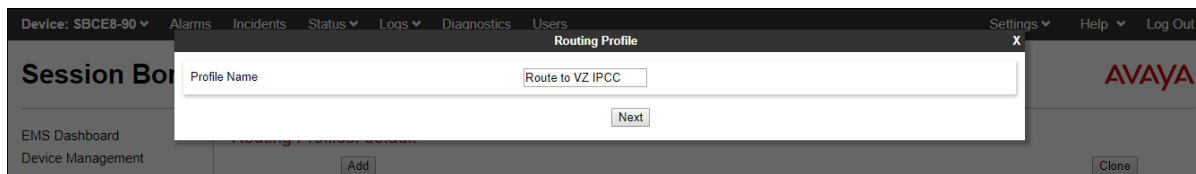
Ignore Route Header
<input type="checkbox"/>

ENUM	ENUM Suffix
<input type="checkbox"/>	

Priority / Weight	LDAP Search Attribute	LDAP Search Regex Pattern	LDAP Search Regex Result	SIP Server Profile	Next Hop Address	Transport
1				IPOSE Primary	10.64.19.170:5061 (TLS)	None

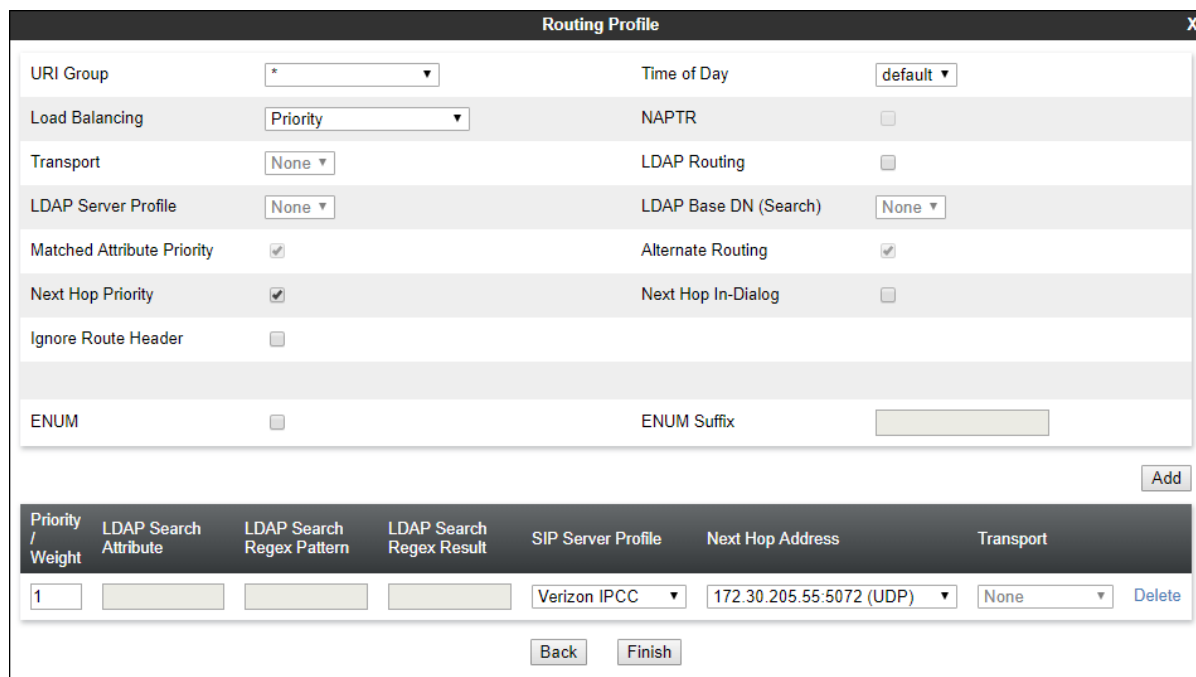
7.8.2. Routing Profile – Verizon

Similarly add a Routing Profile to the Verizon Business IPCC service.



The screenshot shows the Avaya EMS Dashboard with the 'Routing Profile' configuration window open. The 'Profile Name' field is set to 'Route to VZ IPCC'. The 'Next' button is visible.

The following screen shows the Routing Profile **Route to VZ IPCC** created in the reference configuration. The parameters in the top portion of the profile are left at their default settings. The **Priority / Weight** parameter is set to **1**, and the Verizon **Server Configuration**, created in **Section 7.7.2**, is selected from the drop-down menu. The **Next Hop Address** is automatically selected with the values from the Server Configuration, and the Transport becomes greyed out. Click **Finish**.



The screenshot shows the Avaya EMS Dashboard with the 'Routing Profile' configuration window open. The 'Profile Name' field is set to 'Route to VZ IPCC'. The 'Next' button is visible.

URI Group	Time of Day
*	default

Load Balancing	NAPTR
Priority	<input type="checkbox"/>

Transport	LDAP Routing
None	<input type="checkbox"/>

LDAP Server Profile	LDAP Base DN (Search)
None	None

Matched Attribute Priority	Alternate Routing
<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>

Next Hop Priority	Next Hop In-Dialog
<input checked="" type="checkbox"/>	<input type="checkbox"/>

Ignore Route Header
<input type="checkbox"/>

ENUM	ENUM Suffix
<input type="checkbox"/>	

Priority / Weight	LDAP Search Attribute	LDAP Search Regex Pattern	LDAP Search Regex Result	SIP Server Profile	Next Hop Address	Transport
1				Verizon IPCC	172.30.205.55:5072 (UDP)	None

Buttons: Back, Finish, Add, Delete

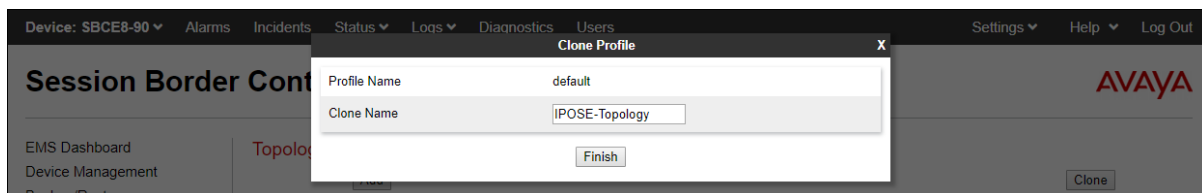
7.9. Topology Hiding Profiles

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

Topology Hiding can also be used as an interoperability tool to adapt the host portion of the SIP headers, to the IP addresses or domains expected on the service provider and the enterprise networks.

7.9.1. Topology Hiding – IP Office

In the sample configuration, the IP Office Topology Hiding profile was cloned from the **default** profile and then modified. Select **Configuration Profiles → Topology Hiding** from the left-hand menu. Select the pre-defined **default** profile and click the **Clone** button. Enter profile name (e.g., **IPOSE-Topology**) and click **Finish** to continue.



On the newly created profile, in the **Replace Action** column, an action of **Auto** will replace the header field with the IP address of the Avaya SBCE interface, while **Overwrite** will use the value in the **Overwrite Value**.

The screen below shows the **IPOSE-Topology** used in the reference configuration. For the **Request-Line**, **To** and **From** headers, **Overwrite** is selected under the **Replace Action** column. The domain of the enterprise (e.g., **silipose.customer.com**) is entered on the **Overwrite Value** field.

Session Border Controller for Enterprise

EMS Dashboard
Device Management
Backup/Restore
System Parameters
Configuration Profiles
Domain DoS
Server Interworking
Media Forking
Routing
Topology Hiding
Signaling Manipulation
URI Groups
SNMP Traps
Time of Day Rules
FGDN Groups
Reverse Proxy Policy
Services
Domain Policies

Topology Hiding Profiles: IPOSE-Topology

Add

Topology Hiding Profiles

- default
- cisco_th_profile
- IPOSE-Topology**
- Enterprise-Topology
- VZ IPT Topology

Click here to add a description.

Header	Criteria	Replace Action	Overwrite Value
Record-Route	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---
Request-Line	IP/Domain	Overwrite	silipose.customer.com
To	IP/Domain	Overwrite	silipose.customer.com
From	IP/Domain	Overwrite	silipose.customer.com
Referred-By	IP/Domain	Auto	---
Refer-To	IP/Domain	Auto	---
Via	IP/Domain	Auto	---

Edit

Rename Clone Delete

7.9.2. Topology Hiding – Verizon

Similarly create a Topology Hiding profile for the Avaya SBCE connection to Verizon. Enter a Profile Name (e.g., **VZ IPCC Topology**). Overwrite the **From** and **Referred-By** headers with the FQDNs known by Verizon, as shown on the screen below.

The screenshot shows the 'Session Border Controller for Enterprise' interface. The left sidebar contains a navigation menu with options like EMS Dashboard, Device Management, Backup/Restore, System Parameters, Configuration Profiles, Domain DoS, Server Interworking, Media Forking, Routing, Topology Hiding (selected), Signaling Manipulation, URI Groups, SNMP Traps, Time of Day Rules, FQDN Groups, Reverse Proxy Policy, Services, and Domain Policies. The main content area is titled 'Topology Hiding Profiles: VZ IPCC Topology'. It features a list of profiles on the left: default, cisco_th_profile, IPOSE-Topology, VZ IPCC Topology (selected), Enterprise-Topology, and VZ IPT Topology. An 'Add' button is next to the list. The main panel shows the 'Topology Hiding' configuration for the selected profile. It includes a description field and a table with the following data:

Header	Criteria	Replace Action	Overwrite Value
Record-Route	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---
Request-Line	IP/Domain	Auto	---
To	IP/Domain	Auto	---
From	IP/Domain	Overwrite	adevc.avaya.globalipcom.com
Referred-By	IP/Domain	Overwrite	adevc.avaya.globalipcom.com
Refer-To	IP/Domain	Auto	---
Via	IP/Domain	Auto	---

An 'Edit' button is located at the bottom right of the table.

7.10. Application Rule

Application Rules define which types of SIP-based Unified Communications applications the Avaya SBCE security device will protect. In addition, the maximum number of concurrent voice and video sessions the network will process are set, in order to prevent resource exhaustion.

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

The screenshot shows the 'Session Border Controller for Enterprise' interface. The left sidebar contains a navigation menu with options like EMS Dashboard, Device Management, Backup/Restore, System Parameters, Configuration Profiles, Domain Policies, Application Rules (selected), Border Rules, Media Rules, Security Rules, Signaling Rules, Charging Rules, End Point Policy Groups, and Session Policies. The main content area is titled 'Application Rules: sip-trunk'. It features a list of application rules on the left: default, default-trunk, default-subscriber-low, default-subscriber-high, default-server-low, default-server-high, sip-trunk (selected), and rv-app-rule. An 'Add' button is next to the list. The main panel shows the 'Application Rule' configuration for the selected profile. It includes a description field and a table with the following data:

Application Type	In	Out	Maximum Concurrent Sessions	Maximum Sessions Per Endpoint
Audio	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	200	200
Video	<input type="checkbox"/>	<input type="checkbox"/>		

Below the table is a 'Miscellaneous' section with the following data:

Miscellaneous	
CDR Support	Off
RTCP Keep-Alive	No

An 'Edit' button is located at the bottom right of the miscellaneous section.

7.11. Media Rules

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.

To create a Media Rule for the IP Office, select **Domain Policies** → **Media Rules** from the left-side menu. In the sample configuration, the default **avaya-low-med-enc** rule was cloned for IP Office, and then modified as shown on the screen below. With the **avaya-low-med-enc** rule chosen, click **Clone**. Enter a descriptive name for the new rule and click **Finish** (not shown).

The Media Rule **enterprise-med-rule** created for the IP Office is shown below. The **Preferred Formats** are changed to include **SRTP_AES_CM_128_HMAC_SHA1_80** as the first choice and **RTP** as second. In the **Miscellaneous** section, **Capability Negotiation** is checked. All other fields retained their default cloned value.

The screenshot displays the 'Session Border Controller for Enterprise' web interface. On the left is a navigation menu with categories like EMS Dashboard, Device Management, and Domain Policies. Under 'Domain Policies', 'Media Rules' is selected. The main panel shows the configuration for 'enterprise-med-rule'. It includes tabs for Encryption, Codec Prioritization, Advanced, and QoS. The 'Encryption' tab is active, showing 'Audio Encryption' and 'Video Encryption' sections. Both sections have 'Preferred Formats' set to 'SRTP_AES_CM_128_HMAC_SHA1_80' and 'RTP'. Other options like 'SRTP Context Reset on SSRC Change', 'Encrypted RTCP', 'MKI', 'Lifetime', and 'Interworking' are also visible. A 'Miscellaneous' section at the bottom has 'Capability Negotiation' checked. Buttons for 'Rename', 'Clone', 'Delete', and 'Edit' are present.

Audio Encryption	
Preferred Formats	SRTP_AES_CM_128_HMAC_SHA1_80 RTP
SRTP Context Reset on SSRC Change	<input type="checkbox"/>
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime	Any
Interworking	<input checked="" type="checkbox"/>

Video Encryption	
Preferred Formats	SRTP_AES_CM_128_HMAC_SHA1_80 RTP
SRTP Context Reset on SSRC Change	<input type="checkbox"/>
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime	Any
Interworking	<input checked="" type="checkbox"/>

Miscellaneous	
Capability Negotiation	<input checked="" type="checkbox"/>

Similarly, a Media Rule is created for Verizon. In this case, the **default-low-med** profile was cloned. With the **default-low-med** rule chosen, click **Clone**. Enter a descriptive name for the new rule and click **Finish** (not shown).

The Media Rule named **Vz-trk-med-rule**, used for Verizon in the sample configuration is shown below.

The screenshot shows the 'Session Border Controller for Enterprise' web interface. On the left is a navigation menu with options like EMS Dashboard, Device Management, Backup/Restore, System Parameters, Configuration Profiles, Services, Domain Policies, Application Rules, Border Rules, Media Rules (highlighted), Security Rules, Signaling Rules, Charging Rules, End Point Policy Groups, Session Policies, TLS Management, Network & Flows, DMZ Services, and Monitoring & Logging. The main area is titled 'Media Rules: Vz-trk-med-rule' and includes an 'Add' button and 'Rename', 'Clone', and 'Delete' buttons. Below this is a list of Media Rules: default-low-med, default-low-med-enc, default-high, default-high-enc, avaya-low-med-enc, enterprise-med-rule, rw-med-rule, and Vz-trk-med-rule (highlighted). The configuration details for 'Vz-trk-med-rule' are shown on the right, with tabs for Encryption, Codec Prioritization, Advanced, and QoS. The 'Encryption' tab is active, showing 'Audio Encryption' (Preferred Formats: RTP, Interworking: checked) and 'Video Encryption' (Preferred Formats: RTP, Interworking: checked). There is also a 'Miscellaneous' section with 'Capability Negotiation' (unchecked). An 'Edit' button is at the bottom right.

Note the DSCP values **EF** for expedited forwarding (default value) used for Media **QoS**, as specified by Verizon.

This screenshot shows the 'QoS' tab in the configuration interface. It contains three sections: 'Media QoS Marking' (Enabled: checked, QoS Type: DSCP), 'Audio QoS' (Audio DSCP: EF), and 'Video QoS' (Video DSCP: EF). An 'Edit' button is located at the bottom right.

7.12. Signaling Rules

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. Signaling Rules are also used to define QoS parameters for the SIP signaling packets.

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

The screenshot shows the 'Session Border Controller for Enterprise' web interface. The left sidebar contains a navigation menu with 'Signaling Rules' highlighted. The main content area is titled 'Signaling Rules: enterprise-sig-rule'. It features a list of signaling rules on the left: 'default', 'No-Content-Type-Checks', 'Vz-trk-sig-rule', 'rw-sig-rule', and 'enterprise-sig-rule' (which is selected and highlighted in red). To the right of this list is an 'Add' button. The main configuration area for the selected rule has tabs for 'General', 'Requests', 'Responses', 'Request Headers', 'Response Headers', 'Signaling QoS', and 'UCID'. The 'General' tab is active, showing a description field and a table of inbound and outbound rules. The inbound rules table has columns for the rule name and its action, with all actions set to 'Allow'. The outbound rules table also has all actions set to 'Allow'. Below these tables is a 'Content-Type Policy' section with a checkbox for 'Enable Content-Type Checks' (checked), an 'Action' dropdown set to 'Allow', a 'Multipart Action' dropdown set to 'Allow', and an 'Exception List' field. An 'Edit' button is at the bottom right of the configuration area.

Inbound	
Requests	Allow
Non-2XX Final Responses	Allow
Optional Request Headers	Allow
Optional Response Headers	Allow

Outbound	
Requests	Allow
Non-2XX Final Responses	Allow
Optional Request Headers	Allow
Optional Response Headers	Allow

Content-Type Policy			
Enable Content-Type Checks <input checked="" type="checkbox"/>			
Action	Allow	Multipart Action	Allow
Exception List			

Signaling rule **Vz-trk-sig-rule** was also cloned from the default rule and used for Verizon. The settings for **Signaling QoS** are changed from the default values to **DSCP** value **AF32** for assured forwarding, as specified by Verizon, shown below.

The screenshot shows the 'Session Border Controller for Enterprise' web interface for the 'Vz-trk-sig-rule' signaling rule. The left sidebar shows 'Signaling Rules' highlighted. The main content area is titled 'Signaling Rules: Vz-trk-sig-rule'. The list of signaling rules on the left includes 'default', 'No-Content-Type-Checks', 'Vz-trk-sig-rule' (selected and highlighted in red), 'rw-sig-rule', and 'enterprise-sig-rule'. The 'Signaling QoS' tab is active in the configuration area, showing a checkbox for 'Signaling QoS' (checked). Below this, the 'QoS Type' is set to 'DSCP' and the 'DSCP' value is set to 'AF32'. An 'Edit' button is at the bottom right of the configuration area.

Signaling QoS	
QoS Type	DSCP
DSCP	AF32

7.13. Endpoint Policy Groups

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

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

The screenshot shows the 'Session Border Controller for Enterprise' interface. On the left is a navigation menu with 'End Point Policy Groups' highlighted. The main area is titled 'Policy Groups: enterpr-trk-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', 'avaya-def-low-enc', 'avaya-def-high-subscriber', 'avaya-def-high-server', and 'enterpr-trk-policy'. The 'enterpr-trk-policy' group is selected. To the right, there is a table with columns: Order, Application, Border, Media, Security, Signaling, Charging, and RTCP Mon Gen. The table contains one row with the following values: Order 1, Application sip-trunk, Border default, Media enterprise-med-rule, Security default-low, Signaling enterprise-sig-rule, Charging None, and RTCP Mon Gen Off. There are also buttons for 'Add', 'Rename', 'Clone', and 'Delete'.

Order	Application	Border	Media	Security	Signaling	Charging	RTCP Mon Gen
1	sip-trunk	default	enterprise-med-rule	default-low	enterprise-sig-rule	None	Off

The following screen shows the **Vz-policy-grp** created for Verizon Business IP Trunking service. The details of the non-default rules chosen are shown in previous sections.

The screenshot shows the 'Session Border Controller for Enterprise' interface. On the left is a navigation menu with 'End Point Policy Groups' highlighted. The main area is titled 'Policy Groups: Vz-policy-grp'. 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', 'avaya-def-low-enc', 'avaya-def-high-subscriber', 'avaya-def-high-server', 'enterpr-trk-policy', 'RW-policy-grp', and 'Vz-policy-grp'. The 'Vz-policy-grp' group is selected. To the right, there is a table with columns: Order, Application, Border, Media, Security, Signaling, Charging, and RTCP Mon Gen. The table contains one row with the following values: Order 1, Application sip-trunk, Border default, Media Vz-trk-med-rule, Security default-low, Signaling Vz-trk-sig-rule, Charging None, and RTCP Mon Gen Off. There are also buttons for 'Add', 'Rename', 'Clone', and 'Delete'.

Order	Application	Border	Media	Security	Signaling	Charging	RTCP Mon Gen
1	sip-trunk	default	Vz-trk-med-rule	default-low	Vz-trk-sig-rule	None	Off

7.14. End Point Flows - Server Flows

Server Flows combine the interfaces, polices, and profiles defined in the previous sections into inbound and outbound flows. 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.

Create a Server Flow for IP Office and Verizon Business IPCC service. To create a Server Flow, navigate to **Networks & Flows → End Point Flows**. Select the **Server Flows** tab and click **Add** (not shown).

The following screen shows the flow named **Verizon IPCC Flow for IPOSE** viewed from the reference configuration. This flow uses the interfaces, polices, and profiles defined in previous sections.

View Flow: Verizon IPCC Flow for IPOSE

Criteria	
Flow Name	Verizon IPCC Flow for IPOSE
Server Configuration	Verizon IPCC
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Inside-Sig-48

Profile	
Signaling Interface	Vz-Sig-B1
Media Interface	Vz-Med-B1
Secondary Media Interface	None
End Point Policy Group	Vz-policy-grp
Routing Profile	Route to IPOSE
Topology Hiding Profile	Vz IPCC Topology
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	<input type="checkbox"/>

Once again, select the **Server Flows** tab and click **Add** (not shown). The following screen shows the flow named **IPOSE Flow for Vz IPCC** viewed from the reference configuration.

This flow uses the interfaces, policies, and profiles defined in previous sections. In addition, the **Remote Subnet** is configured with the Verizon-provided IP address/mask of the subnet for the IPCC service, i.e., **172.30.205.0/24**.

View Flow: IPOSE Flow for Vz IPCC		X	
Criteria		Profile	
Flow Name	IPOSE Flow for Vz IPCC	Signaling Interface	Inside-Sig-48
Server Configuration	IPOSE Primary	Media Interface	Inside-Med-48
URI Group	*	Secondary Media Interface	None
Transport	*	End Point Policy Group	enterpr-trk-policy
Remote Subnet	172.30.205.0/24	Routing Profile	Route to VZ IPCC
Received Interface	Vz-Sig-B1	Topology Hiding Profile	IPOSE-Topology
		Signaling Manipulation Script	None
		Remote Branch Office	Any
		Link Monitoring from Peer	<input type="checkbox"/>

For illustration, the following screen shows the flow named **IPOSE Flow for Vz IPT** viewed from the reference configuration. This flow was originally created for use with Verizon Business IP Trunk service as shown in [IPT-IPO111SBC81]. Similar to the **IPOSE Flow for Vz IPCC** flow shown above, the **Remote Subnet** is also configured. However, it is configured with the Verizon-provided IP address/mask of the subnet for IP Trunk service, rather than the IPCC service. This is shown here to illustrate how the Avaya SBCE can send SIP traffic out on different internal signaling and media interfaces based on the **Criteria** specified in the flow. In the reference configuration provided, a Verizon Business IP Trunk SIP call from IP address 172.30.209.21 will use internal IP address 10.64.91.50 towards IP Office, while a Verizon Business IPCC SIP call from IP address 172.30.205.55 will use internal IP address 10.64.91.48. This will allow IP Office to have separate unique SIP Line configurations to the Avaya SBCE for each Verizon service.

View Flow: IPOSE Flow for Vz IPT		X	
Criteria		Profile	
Flow Name	IPOSE Flow for Vz IPT	Signaling Interface	Inside-Sig-50
Server Configuration	IPOSE Primary	Media Interface	Inside-Med-50
URI Group	*	Secondary Media Interface	None
Transport	*	End Point Policy Group	enterpr-trk-policy
Remote Subnet	172.30.209.0/24	Routing Profile	Route to VZ IPT
Received Interface	Vz-Sig-B1	Topology Hiding Profile	IPOSE-Topology
		Signaling Manipulation Script	None
		Remote Branch Office	Any
		Link Monitoring from Peer	<input type="checkbox"/>

8. 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 <https://enterprise.verizon.com/products/customer-experience-services/transport-and-intelligent-routing/ip-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. For the compliance testing, Verizon provided the IP address and port used by the Verizon SBC, as well as the toll-free numbers. This information was used to complete the configuration of the Avaya IP Office and the Avaya Session Border Controller for Enterprise shown in the previous sections.

9. Verification Steps

This section provides example verifications of the Avaya configuration with Verizon Business Private IP (PIP) Trunk service.

9.1. Avaya SBCE

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

9.1.1. Incidents

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

Device: SBCE8-90 ▾ Alarms **Incidents** Status ▾ Logs ▾ Diagnostics Users Settings ▾ Help ▾ Log Out

Session Border Controller for Enterprise

EMS Dashboard

- Device Management
- Backup/Restore
 - System Parameters
 - Configuration Profiles
 - Services
 - Domain Policies

Dashboard

Information	
System Time	09:04:21 AM MDT Refresh
Version	8.1.0.0-14-18490
GUI Version	8.1.0.0-18490
Build Date	Mon Feb 03 17:23:09 UTC 2020

Installed Devices

EMS
SBCE8-90

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

Incident Viewer

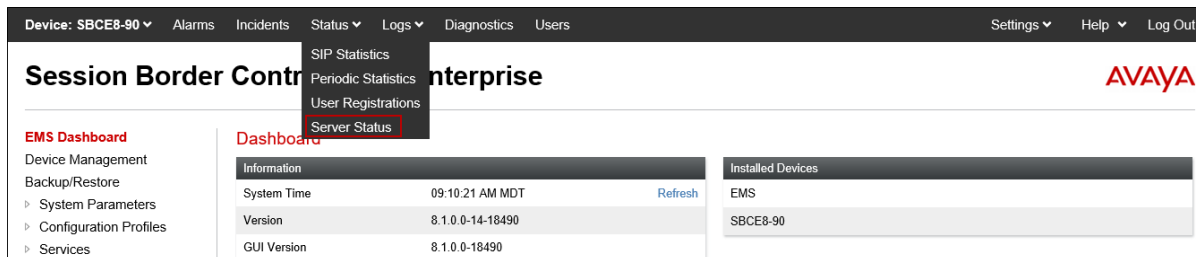
Device: All ▾ Category: All ▾ [Clear Filters](#) [Refresh](#) [Generate Report](#)

Displaying results 1 to 15 out of 2000.

ID	Device	Date & Time	Category	Type	Cause
796315286396678	SBCE8-90	Jun 19, 2020, 11:22:52 PM	TLS Certificate	TLS Handshake Failed	error:1408F10B:SSL routines:SSL3_GET_RECORD:wrong version number
796271085015624	SBCE8-90	Jun 18, 2020, 10:49:30 PM	Policy	Server Heartbeat	Heartbeat Successful, Server is UP
796271071477705	SBCE8-90	Jun 18, 2020, 10:49:02 PM	Policy	Server Heartbeat	Heartbeat Failed, Server is Down
796245726550420	SBCE8-90	Jun 18, 2020, 8:44:13 AM	Scrubbing	Message Detected	Scrubber Anomaly

9.1.2. Server Status

The **Server Status** screen can be accessed from the Avaya SBCE top navigation menu by selecting the **Status** menu, and then **Server Status**.



The **Server Status** screen provides information about the condition of the connection to the connected SIP Servers. This functionality requires Heartbeat to be enabled on the SIP Server Configuration profiles, as configured in **Section 7.7**.

The screenshot shows the 'Status' screen with the 'Server Status' tab selected. The table lists the following data:

Server Profile	Server FQDN	Server IP	Server Port	Server Transport	Heartbeat Status	Registration Status	TimeStamp
Verizon IPCC	172.30.205.55	172.30.205.55	5072	UDP	UP	UNKNOWN	07/08/2020 08:00:04 MDT
Verizon IPT	172.30.209.21	172.30.209.21	5071	UDP	UP	UNKNOWN	07/08/2020 08:00:01 MDT
IPOSE Primary	10.64.19.170	10.64.19.170	5061	TLS	UP	UNKNOWN	07/08/2020 07:59:58 MDT

9.1.3. Tracing

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

The screenshot shows the 'Session Border Controller for Enterprise' interface with the 'Trace: SBCE8-90' screen. The 'Packet Capture' tab is selected, and the configuration form is displayed. The form includes the following fields:

- Status: Ready
- Interface: Any
- Local Address (IP:Port): All
- Remote Address: *
- Protocol: All
- Maximum Number of Packets to Capture: 10000
- Capture Filename: Test.pcap

The 'Start Capture' button is highlighted.

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

EMS Dashboard

Device Management

Backup/Restore

System Parameters

Configuration Profiles

Services

Domain Policies

TLS Management

Network & Flows

DMZ Services

Monitoring & Logging

SNMP

Syslog Management

Debugging

Trace

Log Collection

DoS Learning

CDR Adjunct

Trace: SBCE8-90

Packet Capture

Captures

A packet capture is currently in progress. This page will automatically refresh until the capture completes.

Packet Capture Configuration

Status

In Progress

Interface

Any

Local Address

IP:Port

All

Remote Address

* *Port, IP, IP:Port

*

Protocol

All

Maximum Number of Packets to Capture

10000

Capture Filename

Using the name of an existing capture will overwrite it.

Test.pcap

Stop Capture

Select the **Captures** tab at the top and the capture will be listed; select the **File Name** and choose to open it with an application like Wireshark.

EMS Dashboard

Device Management

Backup/Restore

System Parameters

Configuration Profiles

Services

Domain Policies

TLS Management

Network & Flows

DMZ Services

Monitoring & Logging

SNMP

Syslog Management

Debugging

Trace

Trace: SBCE8-90

Packet Capture

Captures

Refresh

File Name	File Size (bytes)	Last Modified	
Test_20190801093220.pcap	2,558,166	August 1, 2019 9:32:58 AM MDT	Delete

9.2. Avaya IP Office

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

9.2.1. System Status Application

The Avaya IP Office System Status application can be used to verify the service state of the SIP line. From the IP Office Manager application, select **File → Advanced → System Status**.

Under **Control Unit IP Address** select the IP address of the IP Office system under verification. Log in using the appropriate credentials.

The screenshot shows the 'Logon' dialog box for the Avaya IP Office System Status application. It has tabs for 'Online' and 'Offline'. The 'Online' tab is selected. The dialog contains the following fields and options:

- Control Unit Address: 10.64.19.170
- Proxy Server Address: <None>
- Services Base TCP Port: 50804
- Local IP Address: Automatic
- User Name: Administrator
- Password: (masked with dots)
- ☐ Auto reconnect
- ☒ Secure connection
- ☐ Websocket connection
- Logon button

Select the SIP line from the left pane (**Line 10** in the reference configuration). On the **Status** tab in the right pane, verify that the **Current State** is *Idle* for each channel (assuming no active calls at present time).

The screenshot shows the Avaya IP Office System Status application interface. The left pane shows a tree view with 'System' selected, and 'Line: 10' highlighted. The right pane shows the 'Status' tab for 'Line: 10'. The 'SIP Trunk Summary' section displays the following information:

- Line Service State: In Service
- Peer Domain Name: 10.64.91.48
- Resolved Address: 10.64.91.48
- Line Number: 10
- Number of Administered Channels: 10
- Number of Channels in Use: 0
- Administered Compression: G729 A, G711 Mu
- Enable Faststart: Off
- Silence Suppression: Off
- Media Stream: RTP
- Layer 4 Protocol: TLS
- SIP Trunk Channel Licenses: 10
- SIP Trunk Channel Licenses in Use: 0
- SIP Device Features: 0%

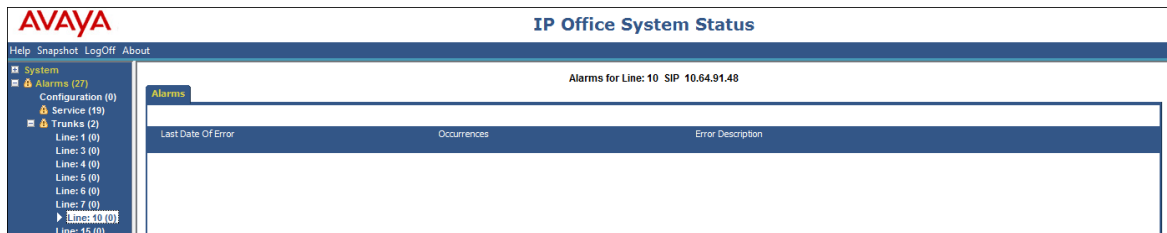
The 'Channel' table shows the following data:

Channel Number	URI Gr...	Call Ref	Current State	Time in State	Remote Media Address	Codec	Connection Type	Caller ID or Dialed Digits	Other Party on Call	Direction of Call	Round Trip Delay	Receive Jitter	Receive Packet Loss ...	Transmit Jitter	Transmit Packet Loss ...
1			Idle	02:29:23											
2			Idle	18:48:09											
3			Idle	1 day 17:45...											
4			Idle	1 day 17:45...											
5			Idle	1 day 17:45...											

The bottom of the interface shows a status bar with the time '8:45:56 AM' and the status 'Online'.

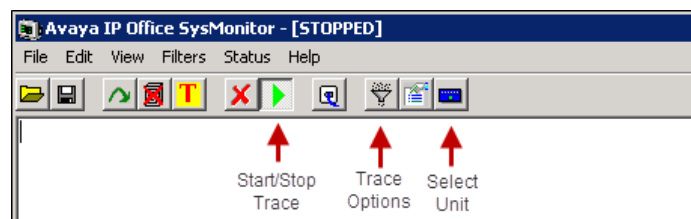
In the lower part of the screen, the **Trace All** button may be pressed to display real time tracing information as calls are made using this SIP Line. The **Ping** button can be used to ping the other end of the SIP trunk (e.g., Avaya SBCE).

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

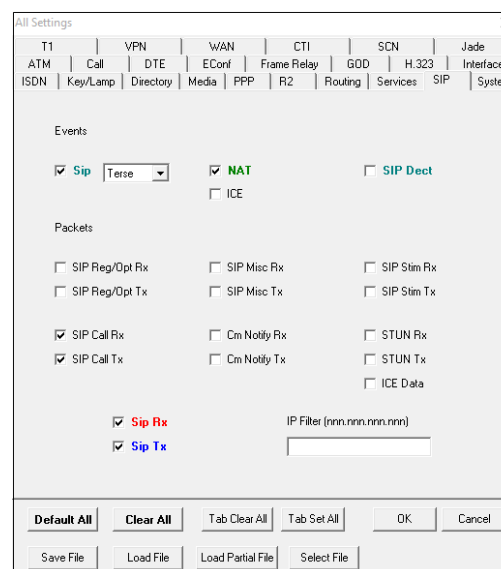


9.2.2. System Monitor

The Avaya IP Office Monitor application can be used to monitor and troubleshoot signaling messaging on the SIP trunk. Launch the application from **Start → Programs → IP Office → Monitor** on the PC where IP Office Manager was installed. Click the **Select Unit** icon on the taskbar and Select the IP address of the IP Office system under verification.



Clicking the **Trace Options** icon on the taskbar and selecting the **SIP** tab allows modifying the threshold used for capturing events, types of packets to be captured, filters, etc. Additionally, the color used to represent the packets in the trace can be customized by right clicking on the type of packet and selecting to the desired color.



As an example, the following shows a portion of the monitoring window for an inbound call to 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 reference configuration, along with the means to retrieve this type of trace information from IP Office.

```
Avaya IP Office SysMonitor - [STOPPED] Monitoring 10.64.19.170 (IPOSE-Primary (Server Edition(P)) (Select)); Log Settings - C:\Users\...\sysmonitorsettings.ini
File Edit View Filters Status Help

08:27:21 58495743mS SIP Rx: TLS 10.64.91.48:42626 -> 10.64.19.170:5061
INVITE sip:8668502380@silipose.customer.com SIP/2.0
From: <sips:+17863310799@silipose.customer.com;user=phone>;tag=00balf9c
To: sips:18668502380@silipose.customer.com
CSeq: 1 INVITE
Call-ID: 5c12b026aabl7572cde8c863fb842376
Contact: <sips:+17863310799@10.64.91.48:5061;sipappsessionid=app-atvkwfghcx7j;wlsfscid=sip-ng4wpzvm5att;transport=tls>
Record-Route: <sip:10.64.91.48:5061;ipcs-line=96628;lr;transport=tls>
Allow: INVITE, ACK, BYE, OPTIONS, CANCEL, SUBSCRIBE, REFER
Supported: replaces
User-Agent: CS2000_NGSS/8.0
Max-Forwards: 68
Via: SIP/2.0/TLS 10.64.91.48:5061;branch=z9hG4bK-s1632-001694560819-1--s1632-
P-Asserted-Identity: "AVAYA INC" <sips:+17863310799@silipose.customer.com;user=phone>
c: application/sdp
Max-Breadth: 60
Content-Length: 458

v=0
o=genband 300855296 1594218441 IN IP4 10.64.91.48
s=-
c=IN IP4 10.64.91.48
t=0 0
m=audio 35294 RTP/AVP 18 0 8 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sgn: 0
a=csrc:1 image udptl t38
a=ptime:20
a=tcap:1 RTP/SAVP
a=acap:1 crypto:1 AES_CM_128_HMAC_SHA1_80 inline:A3KsCOVWwW0P8tRPUw7GqP8xTcJH167WHwF+QrnA UNENCRYPTED_SRTP
a=pcfg:1 t=1 a=1

08:27:21 58495743mS CMCallEvt: 0000000000000000 0.1022.0 -1 BaseEP: NEW CMEndpoint f6c82e28 TOTAL NOW=1 CALL_LIST=0
08:27:21 58495743mS Sip: SIP Line (10): sip_trunk_config_items 50021009, sip_trunk_config_items_2 00000000, voip.flags 81000949
08:27:21 58495743mS Sip: SIPDialog f6c46838 created, dialogs 2 txn_keys 1 video 1 presentation 1 camera 1 unsupp audio 0
08:27:21 58495743mS CMMap: IP::SetCodec pcp[20]b0r0 0 -> f6c48510
08:27:21 58495744mS Sip: SipICPUser 12 has 1 dialog open (AttachDialogToSipICPUser)
08:27:21 58495744mS Sip: SIP Line (10): License, Valid 1, Available 100, Consumed 0
08:27:21 58495744mS Sip: SIPTrunkEndpointDialogOwner::SetRemoteAddressForRequest from 10.64.91.48:5061 to 10.64.91.48:5061
08:27:21 58495744mS Sip: SIPTrunkEndpointDialogOwner::SetRemoteAddressForResponse from 10.64.91.48:5061 to 10.64.91.48:5061
08:27:21 58495744mS SIP Tx: TLS 10.64.19.170:5061 -> 10.64.91.48:42626
SIP/2.0 100 Trying
Via: SIP/2.0/TLS 10.64.91.48:5061;branch=z9hG4bK-s1632-001694560819-1--s1632-
Record-Route: <sip:10.64.91.48:5061;ipcs-line=96628;lr;transport=tls>
From: <sips:+17863310799@silipose.customer.com;user=phone>;tag=00balf9c
Call-ID: 5c12b026aabl7572cde8c863fb842376
CSeq: 1 INVITE
Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,INFO,NOTIFY
Supported: timer
Server: IP Office 11.1.0.0.0 build 237
Content-Length: 0
To: <sips:18668502380@silipose.customer.com>;tag=385e7f0f47d98d9d
```

10. 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 IP Office Release 11.1 with Avaya Session Border Controller for Enterprise Release 8.1 can be successfully combined with a Verizon Business IP Contact Center VoIP Inbound 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.

11. 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, Deploying IP Office Server Edition*, Release 11.1, Issue 14, April 2020.
- [2] *IP Office™ Platform 11.0, Deploying Avaya IP Office Servers as Virtual Machines*, June 2020.
- [3] *IP Office™ Platform 11.1, Deploying an IP500 V2 IP Office Essential Edition*, June 2020.
- [4] *Administering Avaya IP Office™ Platform with Manager*, Release 11.1, Issue 2, May 2020.
- [5] *Administering Avaya IP Office™ Platform with Web Manager*, Release 11.1, Issue 2, May 2020.
- [6] *Planning for and Administering Avaya IX™ Workplace Client for Android, iOS, Mac and Windows*, Release 3.8, Issue 1, March 2020.
- [7] *Using Avaya IX™ Workplace Client for Android, iOS, Mac and Windows*, Release 3.8, Issue 1, March 2020.
- [8] *IP Office Platform 11.1, IP Office SIP Phones with ASBCE, Issue 04c*, April 2020.
- [9] *Deploying Avaya Session Border Controller in Virtualized Environment*, Release 8.1, April 2020.
- [10] *Administering Avaya Session Border Controller for Enterprise*, Release 8.1, Issue 2, April 2020.
- [11] *RFC 3261 SIP: Session Initiation Protocol*. <https://www.ietf.org/rfc/rfc3261.txt>

Additional IP Office documentation can be found at:

<https://ipofficekb.avaya.com/>

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 11.1 and Avaya Session Border Controller for Enterprise Release 8.1 with Verizon IP Trunk Service Suite:

[IPT-IPO111SBC81] Application Notes for SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service with Avaya IP Office Release 11.0 and Avaya Session Border Controller for Enterprise Release 8.0.1, Issue 1.0

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 v1.6
- [VZ-Spec] Verizon Business IPCC Trunk Interface Network Interface Specification, Document Version 2.2.1.9

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