Interoperability Guide

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ADTRAN SBC and Avaya IP Office PBX SIP Trunk Interoperability

This guide describes an example configuration used in testing the interoperability of an ADTRAN session border controller (SBC) and the Avaya IP Office private branch exchange (PBX) using a Session Initiation Protocol (SIP) trunk to provide a SIP trunk gateway to the service provider network. This guide includes the description of the network application, verification summary, and example individual device configurations for the ADTRAN SBC and the Avaya IP PBX products.

For additional information on configuration of the ADTRAN products, please visit the ADTRAN Support Community at https://supportforums.adtran.com

This guide consists of the following sections:

Overview of Application

Service providers are increasingly using SIP trunks to provide Voice over IP (VoIP) services to customers. ADTRAN SBCs terminate the SIP trunk from the service provider and interoperate with the customer's IP PBX system. A second SIP trunk from the gateway connects to the IP PBX. The SBC operates as a SIP back-to-back user agent (B2BUA). The ADTRAN SBC features normalize the SIP signaling and media between the service provider and the customer IP PBX. *Figure 1* illustrates the use of the ADTRAN IP business gateway in a typical network deployment.

There is additional information available online at ADTRAN's Support Community, https://supportforums.adtran.com. Specific resources are listed in *Additional Resources on page 19*.

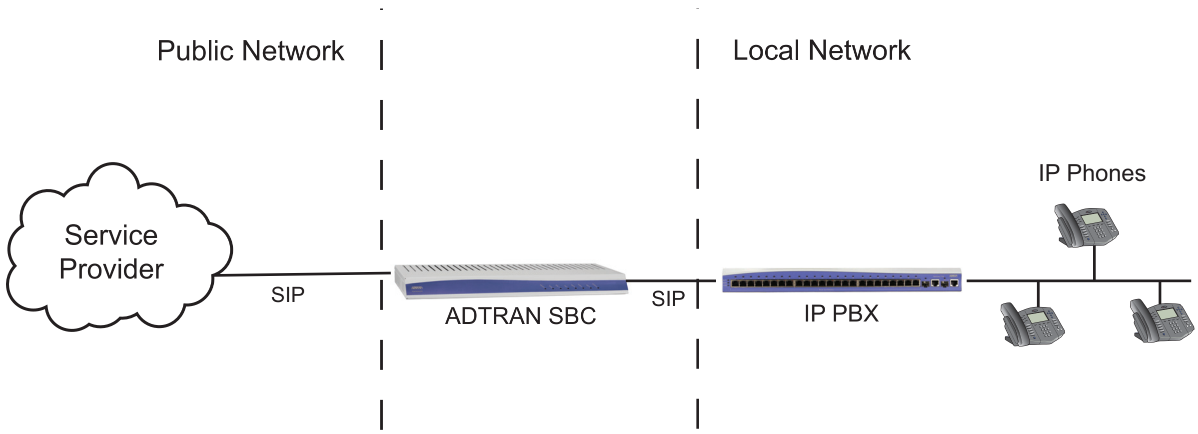


Figure 1. ADTRAN SBC in the Network

Interoperability

The network topology shown in *Figure 2 on page 3* was used for interoperability verification between the ADTRAN SBC and the Avaya IP Office PBX. The configuration is a typical SIP trunking application, where the ADTRAN gateway Ethernet interface provides the Ethernet wide area network (WAN) connection to the service provider network. A second Ethernet interface connects to the customer local area network (LAN). The Avaya IP Office PBX LAN interface connects to the customer LAN. Two SIP trunks are configured on the ADTRAN SBC gateway: one to the service provider and the second to the Avaya IP Office PBX. The ADTRAN SBC gateway operates as a SIP B2BUA, and outbound and inbound calls to the public switched telephone network (PSTN) are routed through the ADTRAN SBC.

The ADTRAN SBC provides SIP trunk registration to the service provider if required. Some service providers have different requirements. Consult your service provider for specific SIP trunking configuration information.

The Avaya IP Office PBX supports various phone types (including digital, H.323, and SIP IP phones). The phones register locally to the Avaya IP PBX. Dial plan configuration routes external calls through the SIP trunk to the ADTRAN SBC gateway.

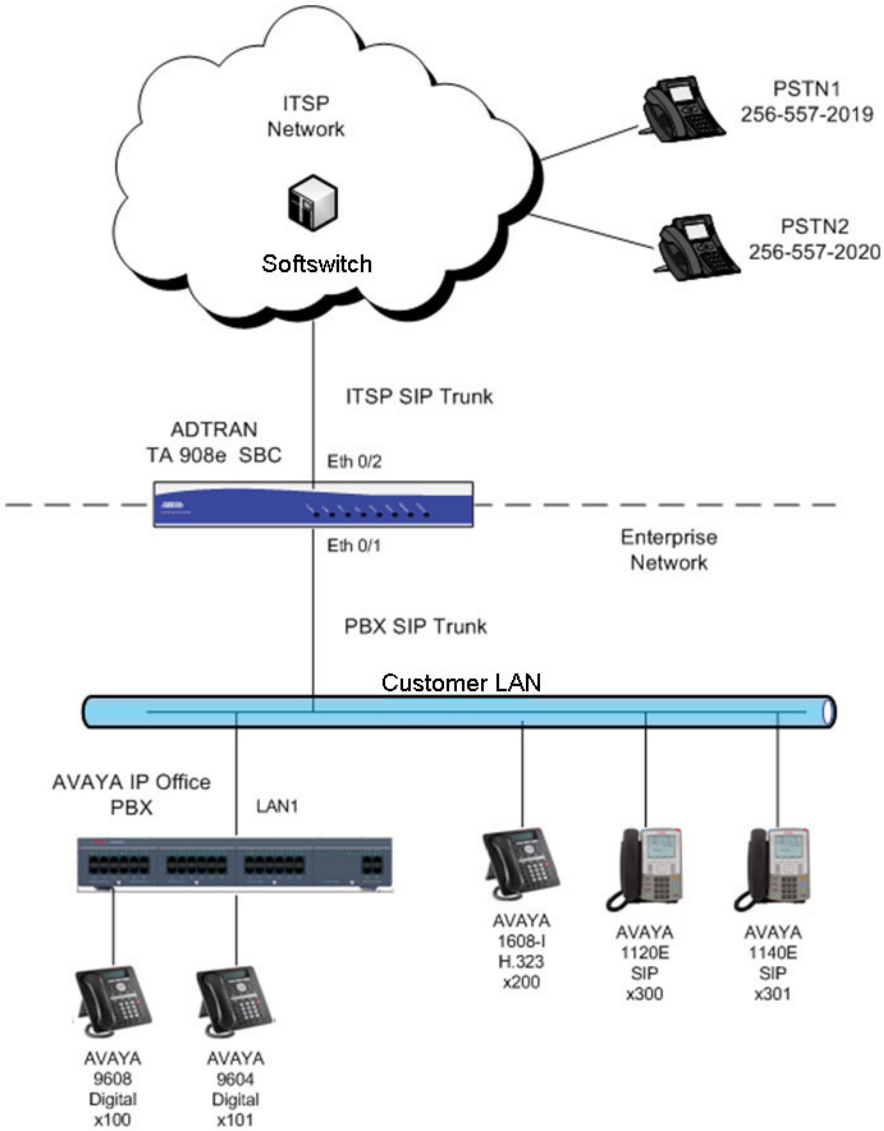


Figure 2. Network Topology for Verification

Hardware and Software Requirements and Limitations

Interoperability with the Avaya IP Office IP PBX is available on ADTRAN products with the SBC feature code as outlined in the *AOS Feature Matrix*, available online at ADTRAN's Support Forum, https://supportforums.adtran.com. The test equipment, testing parameters, and associated caveats are described in the following sections.

Equipment and Versions

The following table outlines the equipment and firmware versions used in verification testing.

Table 1. Verification Test Equipment and Firmware Versions

Product	Firmware Version
ADTRAN Total Access 908e IP Business Gateway SBC P/N 424908L1SBC	R10.3.1
Avaya IP Office 500 V2 Control Unit, IP PBX	7.0 (5)
Avaya IP Office Manager software	9.0 (12)
Avaya 1120E SIP IP Phone	04.01.13.00
Avaya 1140E SIP IP Phone	04.01.13.00
Avaya 1608-I H.323 IP Phone	Hal608ua1_3000.bin
Avaya 9504, 9508 Digital Phones	0.27

Verification Performed

Interoperability verification testing focused on SIP trunk operations between the ADTRAN SBC gateway and the Avaya IP Office PBX. Other PBX features not specific to basic SIP trunking were not included in this verification. Verification testing included the following features:

- Avaya SIP trunk operation with the ADTRAN SBC gateway.
- Avaya SIP OPTIONS message for SIP trunk keepalive.
- Basic inbound and outbound calling with the PSTN using SIP trunking.
- Dial plan operation with the PSTN.
- Dual tone multifrequency (DTMF) operation (both RFC 2833 and in-band signaling).
- Coder-decoder (CODEC) negotiation (using both G.711u and G.729).
- Call forwarding (local and external) with the PSTN.
- Call hold and retrieve with the PSTN.
- Call transfers (consultative and unassisted) with the PSTN.
- Three-way conferencing with the PSTN.
- Caller ID presentation and privacy with the PSTN.
- Voicemail operation with the PSTN.

Supported Features and Exceptions

The SIP OPTIONS message keepalive and user privacy features require special consideration during configuration. The following section summarizes the caveats associated with these features in the interoperability between the ADTRAN SBC and the Avaya IP Office IP PBX.

SIP OPTIONS Message

The Avaya IP Office PBX will send SIP OPTIONS messages to the SIP server as a keepalive to verify the SIP trunk is in service. If the Avaya IP PBX configuration is set to perform out of service (OOS) check on the SIP trunk, and no response is received, then the SIP trunk is disabled and the SIP OPTIONS message is sent to **Unknown**. The ADTRAN SBC responds to the SIP OPTIONS message if a voice user with the SIP identity of **Unknown** is configured. You can choose to configure a voice user with a SIP identity of **Unknown** user in the ADTRAN SBC configuration, or you can disable the OOS check in the Avaya IP PBX configuration. The commands for creating the voice user with the SIP identity of **Unknown** are shown in the SBC configuration section *Step 11: Configuring SIP OPTIONS Support (Optional) on page 10*.

User Privacy

The Avaya IP Office PBX supports configuring user privacy for outbound calls. By default, the Avaya IP PBX uses P-Preferred-Identity headers. The ADTRAN SBC supports P-Asserted-Identity. Therefore, the Avaya IP PBX should be configured to use P-Asserted-Identity headers using the **Source Number** option, as shown in *Step 11: Configuring the Avaya IP PBX P-Asserted-Identity Option on page 19*.

Configuring the ADTRAN SBC Using the CLI

The SBC can be configured using either the command line interface (CLI) or the web-based graphical user interface (GUI). The following sections describe the key configuration settings required for this solution using the CLI. Refer to *Additional Resources on page 19* for more information about SBC GUI configuration.

To configure the SBC for interoperability with the Avaya IP PBX, follow these steps:

Step 1: Accessing the SBC CLI

To access the CLI on your AOS unit, follow these steps:

1. Boot up the unit.
2. Telnet to the unit (**telnet** <*ip address*>), for example:

   telnet 10.10.10.1.

> NOTE: *If during the unit's setup process you have changed the default IP address (10.10.10.1), use the configured IP address.*

3. Enter your user name and password at the prompt.

> NOTE: *The AOS default user name is **admin** and the default password is **password**. If your product no longer has the default user name and password, contact your system administrator for the appropriate user name and password.*

4. Enable your unit by entering **enable** at the prompt as follows:

   >enable
5. If configured, enter your Enable mode password at the prompt.
6. Enter the unit's Global Configuration mode as follows:

   #configure terminal
   (config)#

Step 2: Configuring the Basic Network Settings

Basic network configuration includes setting up two Ethernet interfaces, one for the Ethernet LAN interface to the Avaya IP PBX, and the second for the Ethernet WAN interface to the ISP. Both interfaces are configured using the **ip address** <*ipv4 address*> <*subnet mask*> and **media-gateway ip primary** commands. The **ip address** command configures a static IP address for the interface, and the **media-gateway** command is required on the interface for SIP and Realtime Transport Protocol (RTP) media traffic. Enter the commands from the Ethernet interface configuration mode as follows:

For the LAN interface:

(config)#**interface ethernet 0/1**
(config-eth 0/1)#**description CUSTOMER LAN**
(config-eth 0/1)#**ip address 10.22.227.14 255.255.255.0**
(config-eth 0/1)#**media-gateway ip primary**

For the WAN interface:

(config)#**interface ethernet 0/2**
(config-eth 0/2)#**description PROVIDER WAN**
(config-eth 0/2)#**ip address 192.0.2.3 255.255.255.248**
(config-eth 0/2)#**media-gateway ip primary**
(config-eth 0/2)#**no shutdown**

Step 3: Configuring Global Voice Modes for Local Handling

Configure the ADTRAN SBC to use the local mode for call forwarding and transfer handling. By default, both of these functions are handled by the network. To change these settings, use the **voice transfer-mode local** and **voice forward-mode local** commands. Enter these commands from the Global Configuration mode. By using the **local** parameter, both commands specify allowing the unit to handle call forwarding and transfers locally.

Enter the commands as follows:

(config)#**voice transfer-mode local**
(config)#**voice forward-mode local**

Step 4: Configuring the Service Provider SIP Trunk

The first of two voice trunks that must be configured is the SIP trunk to the service provider from the ADTRAN SBC. The minimum amount of configuration is provided in this document; however, your application may require additional settings (depending on your service provider requirements). Check with your service provider for any specific requirements beyond those listed in this document.

Use the **voice trunk** <*txx*> **type sip** command to define a new SIP trunk and activate the Voice Trunk Configuration mode for the individual trunk. From the Voice Trunk Configuration mode, you can provide a descriptive name for the trunk and define the SIP server's primary IPv4 address (or host name). Use the **description** <*text*> command to label the trunk. Use the **sip-server primary** <*ipv4 address* | *hostname*> command to define the host name or IPv4 address of the primary server to which the trunk sends call-related SIP messages.

Enter the commands as follows:

(config)#**voice trunk T01 type sip**
(config-T01)#**description Provider**
(config-T01)#**sip-server primary 198.51.100.2**

Step 5: Configuring the Avaya IP Office PBX SIP Trunk

The second of two voice trunks that must be configured is the SIP trunk to the Avaya IP Office PBX from the ADTRAN SBC. The trunk is also configured using the **voice trunk** <*txx*> **type sip**, **description** <*text*>, and **sip-server-primary** <*ipv4 address* | *hostname*> commands. Use the **sip-server-primary** <*ipv4 address* | *hostname*> command to set the server address to the Avaya IP PBX LAN1 IP address. In addition, the Avaya IP PBX will control call transfers, so enter the **transfer-mode-network** command in the trunk's configuration. Use the **grammar from host local** command to specify that the IP address of the interface is used in the SIP FROM field for outbound messages.

Enter the commands as follows:

(config)#**voice trunk T02 type sip**
(config-T02)#**description PBX**
(config-T02)#**sip-server primary 10.22.227.1**
(config-T02)#**transfer-mode network**
(config-T02)#**grammar from host local**

Step 6: Configuring a Trunk Group for the Service Provider

After configuring the two SIP trunks, configure an individual trunk group for the service provider trunk account. The previously created trunks are added to the trunk group, which is then used to assign outbound call destinations (local calls, long distance calls, etc.). A cost is also assigned to each **accept** template in the trunk group.

Use the **voice grouped-trunk** <*name*> command to create a trunk group and to enter the Voice Trunk Group Configuration mode. The **trunk** <*txx*> command adds an existing trunk to the trunk group, so that outbound calls can be placed out of that particular trunk. The <*txx*> parameter specifies the trunk identity where *xx* is the trunk ID number.

Use the **accept** <*pattern*> command to specify number patterns that are accepted for routing calls out of the trunk. Use the **no** form of this command to remove a configured dial pattern. The <*pattern*> parameter is specified by entering a complete phone number or using wildcards to help define accepted numbers.

Valid characters for templates are as follows:

0 - 9	Match the exact digit(s) only
X	Match any single digit 0 through 9
N	Match any single digit 2 through 9
M	Match any single digit 1 through 8
$	Match any number string dialed
[]	Match any digit in the list within the brackets (for example, [1,4,6])
,()	Formatting characters that are ignored but allowed
-	Use within brackets to specify a range, otherwise ignored

The following are example template entries using wildcards:

1) NXX-XXXX	Match any 7-digit number beginning with 2 through 9
2) 1-NXX-NXX-XXXX	Match any number with a leading 1, then 2 through 9, then any 2 digits, then 2 through 9, then any 6 digits
3) 555-XXXX	Match any 7-digit number beginning with 555
4) XXXX$	Match any number with at least 5 digits
5) [7,8]$	Match any number beginning with 7 or 8
6) 1234	Match exactly 1234

Some template number rules:

1. All brackets must be closed with no nesting of brackets and no wildcards within the brackets.
2. All brackets can hold digits and commas, for example: [1239]; [1,2,3,9]. Commas are implied between numbers within brackets and are ignored.
3. Brackets can contain a range of numbers using a hyphen, for example: [1-39]; [1-3,9].
4. The $ wildcard is only allowed at the end of the template, for example: 91256$; XXXX$.

Enter the commands as follows:

(config)#**voice grouped-trunk PROVIDER**
(config-PROVIDER)#**trunk T01**
(config-PROVIDER)#**accept N11 cost 0**
(config-PROVIDER)#**accept NXX-XXXX cost 0**
(config-PROVIDER)#**accept NXX-NXX-XXXX cost 0**
(config-PROVIDER)#**accept 1-NXX-NXX-XXXX cost 0**
(config-PROVIDER)#**accept 011-X$ cost 0**

Step 7: Configuring a Trunk Group for the PBX

After configuring a trunk group for the service provider, create a trunk group for the Avaya IP PBX trunk account. Create the trunk group using the **voice grouped-trunk** <*name*> command. Add an existing trunk to the trunk group using the **trunk** <*txx*> command. The outbound allowed calls are defined using the **accept** <*pattern*> command, and are assigned a cost using the **cost** parameter, as described in *Step 6: Configuring a Trunk Group for the Service Provider on page 8*. Enter the commands from the Global Configuration mode as follows:

(config)#**voice grouped-trunk PBX**
(config-PBX)#**trunk T02**
(config-PBX)#**accept 256-555-01XX cost 0**

Step 8: Enabling Media Anchoring

Media anchoring is an SBC feature that routes RTP traffic through the ADTRAN SBC gateway. Minimum configuration for media anchoring includes enabling the feature using the **ip rtp media-anchoring** command from the Global Configuration mode. The RTP symmetric filter works in conjunction with media anchoring to filter nonsymmetric RTP packets. Enable RTP symmetric filtering using the **ip rtp symmetric-filter** command. Enter the commands as follows:

(config)#**ip rtp media-anchoring**
(config)#**ip rtp symmetric-filter**

> NOTE
>
> *For more information about configuring additional media anchoring settings, refer to the configuration guide Configuring Media Anchoring in AOS, available online at* http://supportforums.adtran.com.

Step 9: Configuring the Double reINVITE Preference

After configuring the trunks, trunk groups, and any media anchoring settings, determine whether a double reINVITE is preferred globally for all calls in the system using the **ip sip prefer double-reinvite** command. Calls that typically require a double reINVITE are forwarded calls and attended transfers. When these calls connect, a double reINVITE is initiated.

By default, the system is configured so that double reINVITEs are preferred. If a transfer call involves a SIP trunk operating in the local transfer mode, a double reINVITE is executed regardless of this preference setting. To avoid extra SIP messaging in situations where it is not necessary, set this feature to not prefer double reINVITEs by entering the **no** version of the **ip sip prefer double-reinvite** command from the Global Configuration mode.

Enter the command as follows:

(config)#**no ip sip prefer double-reinvite**

Step 10: Configuring SIP Privacy (Optional)

The ADTRAN SBC supports SIP user privacy by using the P-Asserted-Identity SIP header. Enable P-Asserted-Identity (PAI) and SIP privacy support by entering the **ip sip privacy** command from the Global Configuration mode and by entering the **trust-domain** command for voice trunks (to add PAI). Enter the commands as follows:

(config)#**ip sip privacy**
(config)#**voice trunk T01 type sip**
(config-T01)#**trust-domain**
(config-T01)#**exit**
(config)#**voice trunk T02 type sip**
(config-T02)#**trust-domain**

Step 11: Configuring SIP OPTIONS Support (Optional)

The Avaya IP PBX periodically sends a SIP OPTIONS request as a keepalive to check the status of the SIP trunk and the service provider SIP server. The SIP OPTIONS request is sent to a user of **Unknown**. A voice user with the SIP identity of **Unknown** can be configured on the ADTRAN SBC to enable a response to the SIP OPTIONS request. If there is no **Unknown** voice user configured on the ADTRAN SBC, the Avaya IP PBX will ignore a non-response if the **Check OOS** option is disabled in the SIP line configuration (refer to *Step 4: Configuring the Avaya IP PBX LAN1 VoIP Settings on page 14*).

To configure a voice user with the SIP identity of **Unknown** on the ADTRAN SBC, enter the Voice User Configuration mode by creating a virtual user (**0000**), and entering the **sip-identity** *<station>* *<txx>* command. Enter the commands as follows:

(config)#**voice user 0000**
(config-0000)#**sip-identity Unknown T02**

ADTRAN SBC Sample Configuration

The following example configuration is for a typical installation of an ADTRAN SBC gateway or router with SIP trunking configured to the service provider and the Avaya IP Office PBX. This configuration was used to validate the interoperability between the ADTRAN SBC and the Avaya IP PBX. Only the commands relevant to the interoperability configuration are shown.

> NOTE
>
> *The configuration parameters entered in this example are sample configurations only, and only pertain to the configuration of the SIP trunking gateway functionality. This application should be configured in a manner consistent with the needs of your particular network. CLI prompts have been removed from the configuration example to provide a method of copying and pasting configurations directly from this guide into the CLI. This configuration should not be copied without first making the necessary adjustments to ensure it will function properly in your network.*

```
!
interface eth 0/1
  description CUSTOMER LAN
  ip address 10.22.227.14 255.255.255.0
  media-gateway ip primary
  no shutdown
!
!
interface eth 0/2
  description PROVIDER WAN
  ip address 192.0.2.3 255.255.255.248
  media-gateway ip primary
  no shutdown
!
!
voice transfer-mode local
voice forward-mode local
!
voice trunk T01 type sip
  description service provider
  sip-server primary 198.51.100.2
  trust-domain
!
!
voice trunk T02 type sip
  description PBX
  sip-server primary 10.22.227.1
  trust-domain
  grammar from host local
  transfer-mode network
!
!
```



```
voice grouped-trunk PROVIDER
  trunk T01
  accept N11 cost 0
  accept NXX-XXXX cost 0
  accept NXX-NXX-XXXX cost 0
  accept 1-NXX-NXX-XXXX cost 0
  accept 011-X$ cost 0
!
!
voice grouped-trunk PBX
  trunk T02
  accept 256-555-01XX cost 0
!
!
voice user 0000
  sip-identity Unknown T02
!
!
ip sip privacy
!
!
no ip sip prefer double-reinvite
!
!
ip rtp media-anchoring
ip rtp symmetric-filter
!
end
```

Configuring the Avaya IP Office PBX

The Avaya IP Office PBX system supports many features. The following sections describe the minimum configuration required for SIP trunking interoperability with the ADTRAN SBC. The Avaya IP PBX is configured using the Avaya product's GUI (Avaya IP Office Manager software). Refer to the Avaya documentation for detailed instructions about accessing the GUI. To configure the Avaya IP PBX using the GUI, follow these steps:

- *Step 1: Connecting to the Avaya Product GUI on page 13*
- *Step 2: Verifying Installation of a SIP Trunk License Key on page 13*
- *Step 3: Configuring the Avaya IP PBX LAN1 Interface on page 13*
- *Step 4: Configuring the Avaya IP PBX LAN1 VoIP Settings on page 14*
- *Step 5: Configuring the SIP Line to the ADTRAN SBC on page 15*
- *Step 6: Configuring SIP Line Transport Settings on page 16*
- *Step 7: Configuring the SIP Line URI Channel on page 16*
- *Step 8: Configuring Call Routing to the SIP Trunk on page 17*
- *Step 9: Configuring Incoming Call Routing on page 18*

- *Step 10: Configuring Avaya IP PBX Users on page 18*
- *Step 11: Configuring the Avaya IP PBX P-Asserted-Identity Option on page 19*

Step 1: Connecting to the Avaya Product GUI

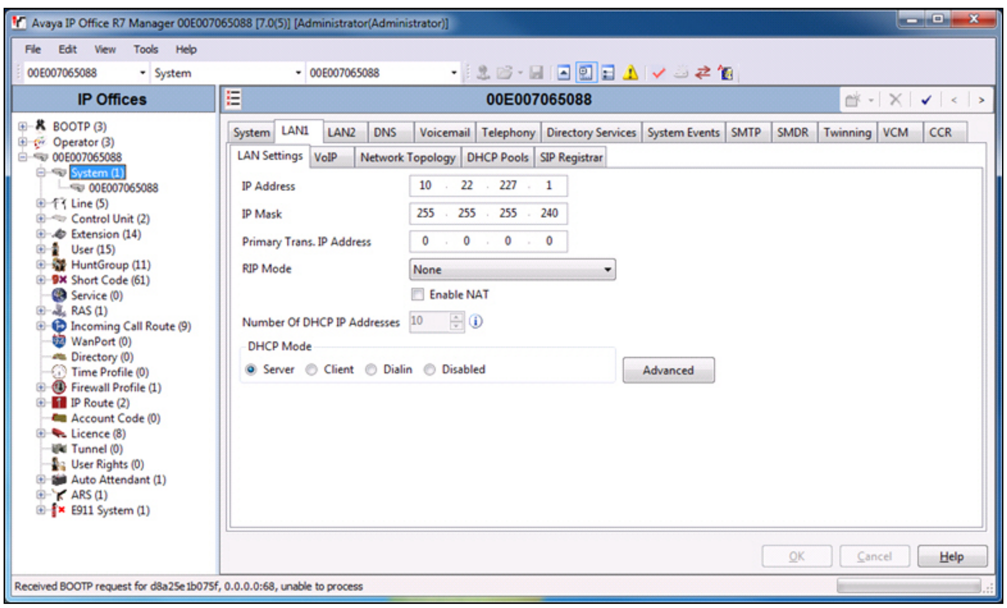
The Avaya IP Office IP PBX system is configured using the Avaya IP Office Manager software. Refer to the Avaya documentation for detailed instructions about accessing the GUI.

Step 2: Verifying Installation of a SIP Trunk License Key

Once you have accessed the Avaya IP Office Manager, verify the installation of a SIP trunk license key. The Avaya IP PBX requires that a SIP trunk license key is installed to enable SIP trunking. Use the Avaya IP Office Manager software to view the license. Select the Avaya IP Office 500V2 unit, then select **License** and **SIP Trunk Channels.**

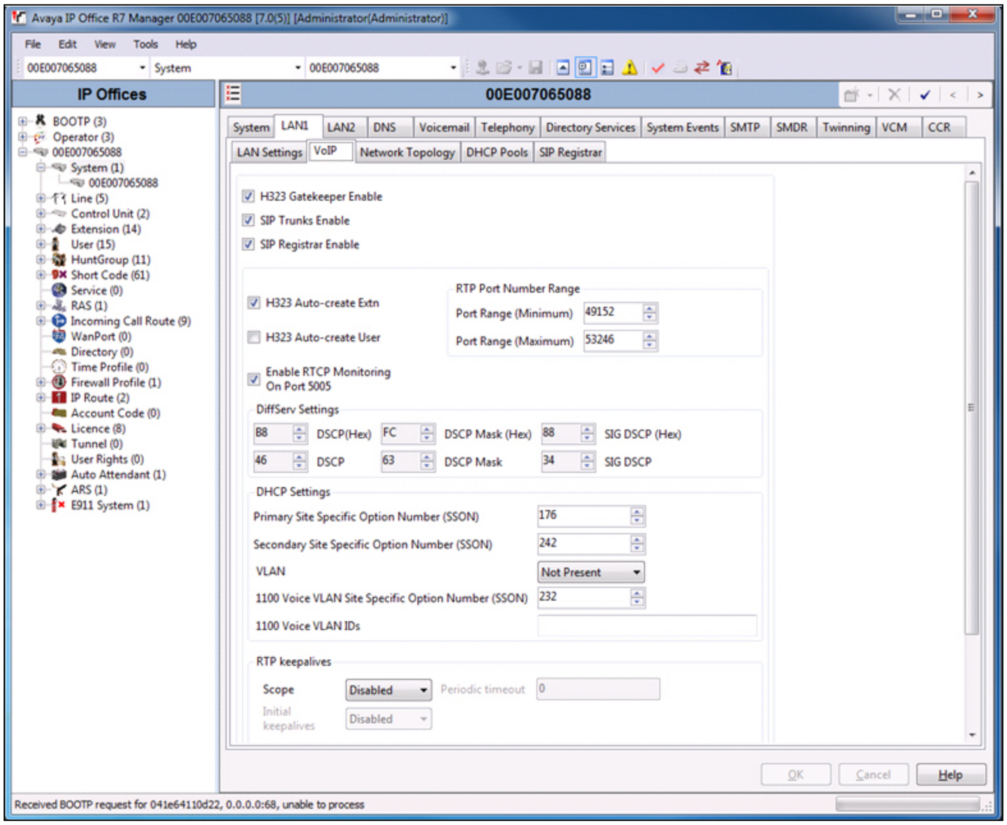
Step 3: Configuring the Avaya IP PBX LAN1 Interface

After verifying the SIP trunk license key, begin configuring the LAN1 interface settings for the Avaya IP PBX. The LAN1 interface connects to the LAN and the ADTRAN SBC gateway. Enter the LAN1 IP address and subnet IP mask.



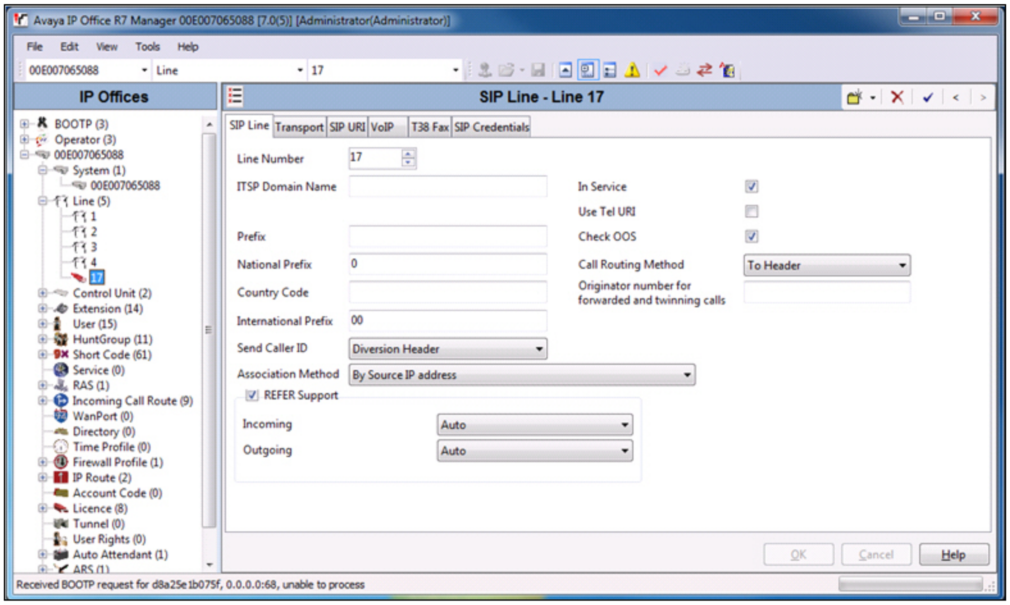
Step 4: Configuring the Avaya IP PBX LAN1 VoIP Settings

After configuring the Avaya IP PBX LAN1 IP information, enable the settings for **H.23 Gatekeeper**, **SIP Trunks**, and **SIP Registrar**. In addition, you must configure the other LAN1 network topology and DHCP pool settings as required for your application. Specific configuration of these settings is not required for the SIP trunking solution.



Step 5: Configuring the SIP Line to the ADTRAN SBC

Create the SIP trunk from the Avaya IP PBX to the ADTRAN SBC gateway by selecting **Line** and **New** > **SIP Line**. Enable **In Service** and **Check OOS**. Only enable **Check OOS** if you configured the ADTRAN SBC to support SIP OPTIONS as described in *Step 11: Configuring SIP OPTIONS Support (Optional) on page 10*. Specify the **Call Routing Method** as the **To Header**, the **Send Caller ID** option as **Diversion Header**, and the **Association Method** as **By Source IP address**. Enable REFER support by selecting the **REFER Support** check box.

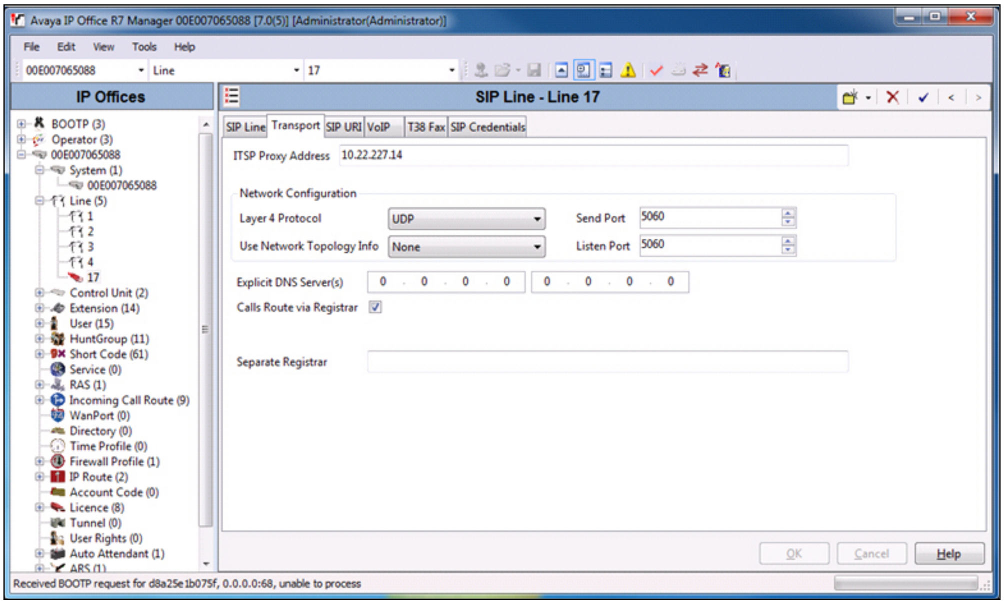


NOTE

These settings may vary depending on the service provider requirements.

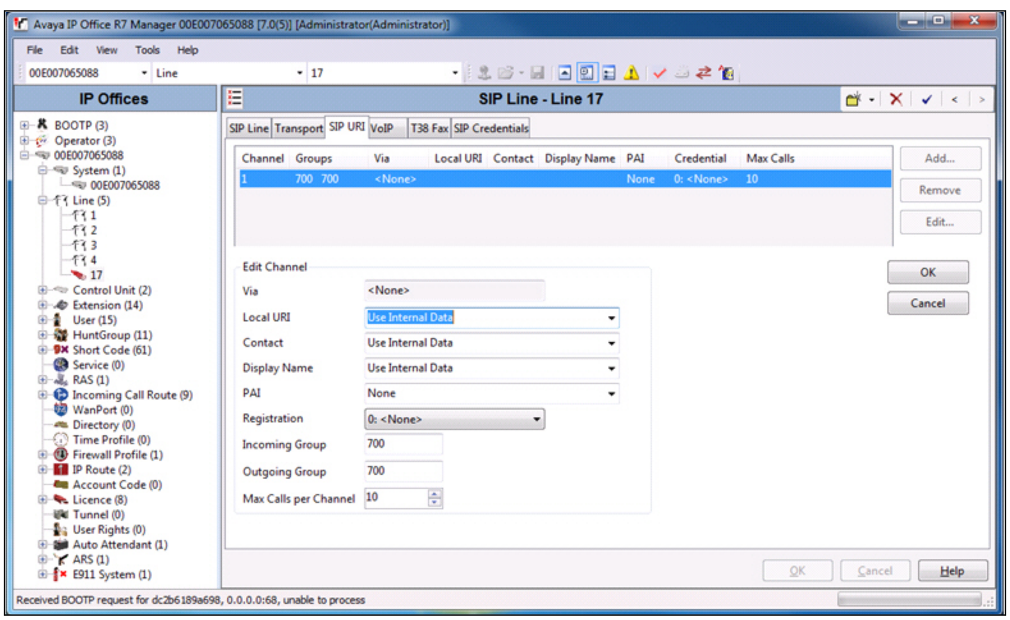
Step 6: Configuring SIP Line Transport Settings

Configure the Avaya IP PBX SIP line **Transport** settings by specifying the **ITSP Proxy Address** as the the ADTRAN SBC Ethernet LAN IP address (**10.22.227.14** in the example below).



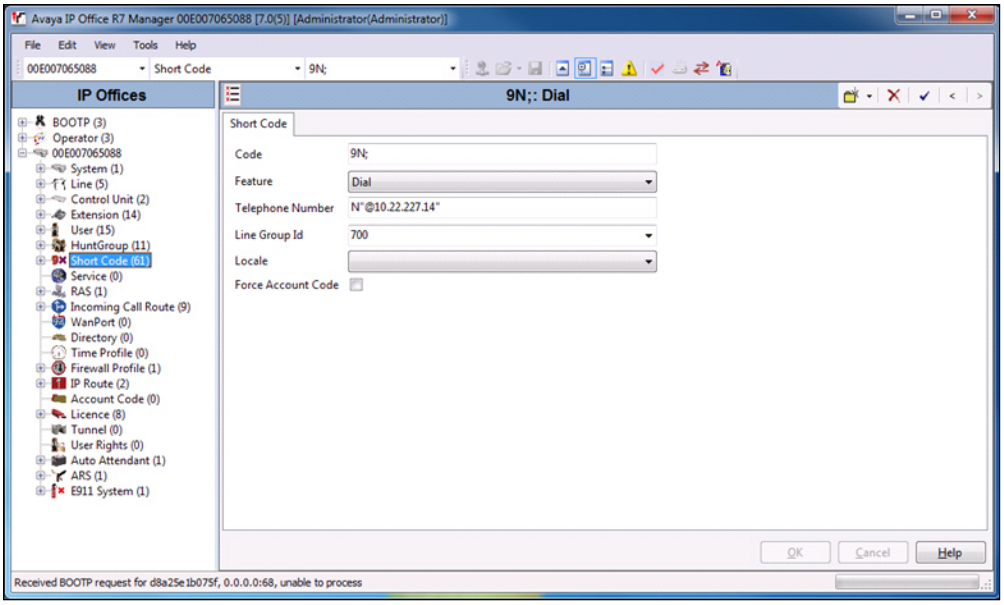
Step 7: Configuring the SIP Line URI Channel

Configure the **SIP URI** channel settings for an incoming and outgoing group number. Select the **Use Internal Data** option from the **Local URI**, **Contact**, and **Display Name** drop-down menus. Registration is not required on this trunk, so select **None** from the **Registration** drop-down menu. If registration is required by the service provider, the ADTRAN SBC gateway will provide the registration.



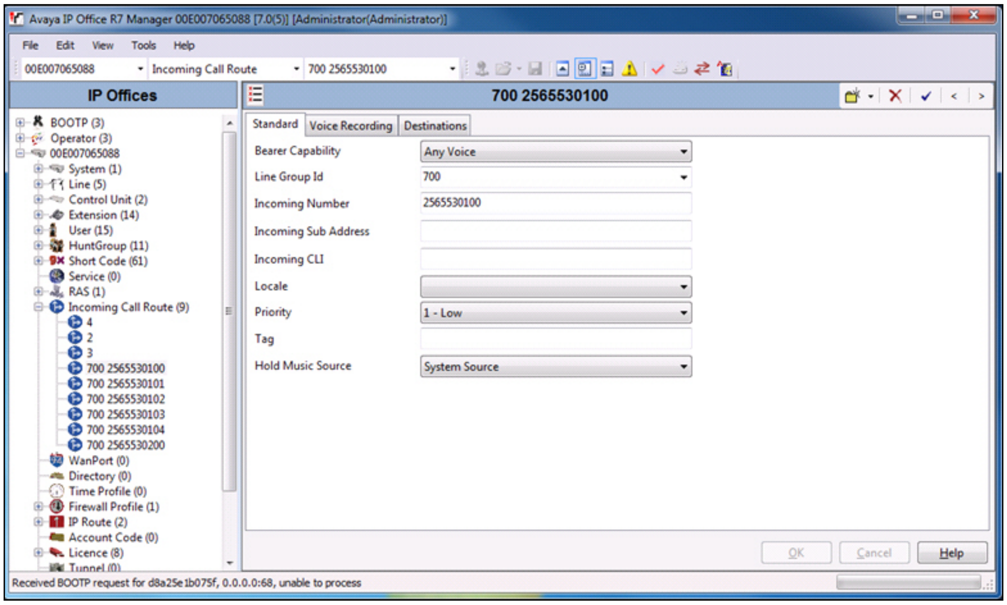
Step 8: Configuring Call Routing to the SIP Trunk

The Avaya IP PBX uses short codes to match user-dialed digits for various functions, including call routing. To configure the short codes for call routing, specify the **Code** as **9N** (to match when a user dials **9** followed by **N** digits), the **Feature** as **Dial**, the **Telephone Number** as **N”@10.22.227.14”**, and the **Line Group Id** as **700**. These settings will route external calls (dial 9 + number) out of the Avaya’s line 17 SIP trunk to the ADTRAN SBC gateway.



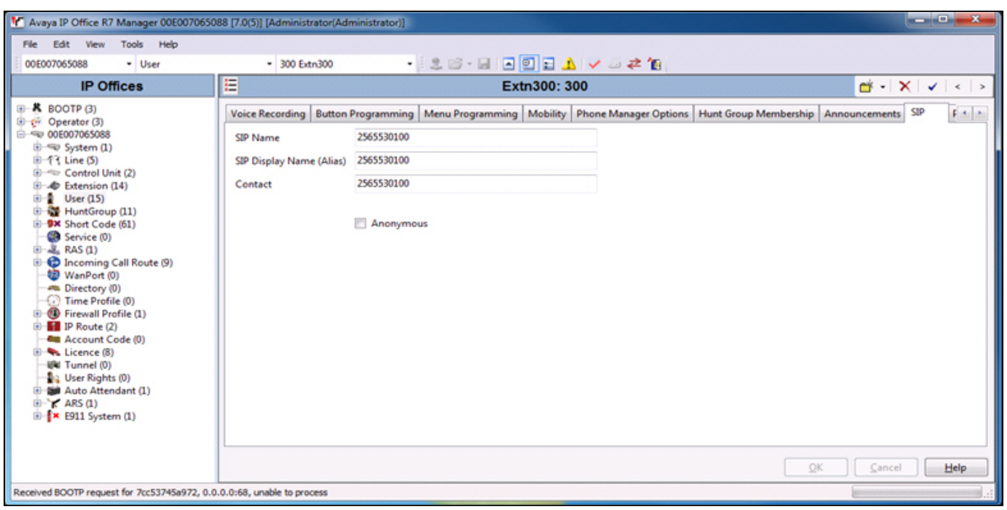
Step 9: Configuring Incoming Call Routing

The Avaya IP PBX routes incoming calls based on the configured **Incoming Call Routes**. Select **Incoming Call Route** > **New** to create routes for the main business telephone number (**2565330100**) and other direct inward dial (DID) telephone numbers. Then assign the **Destinations** to the phone extension from the **Destinations** tab.



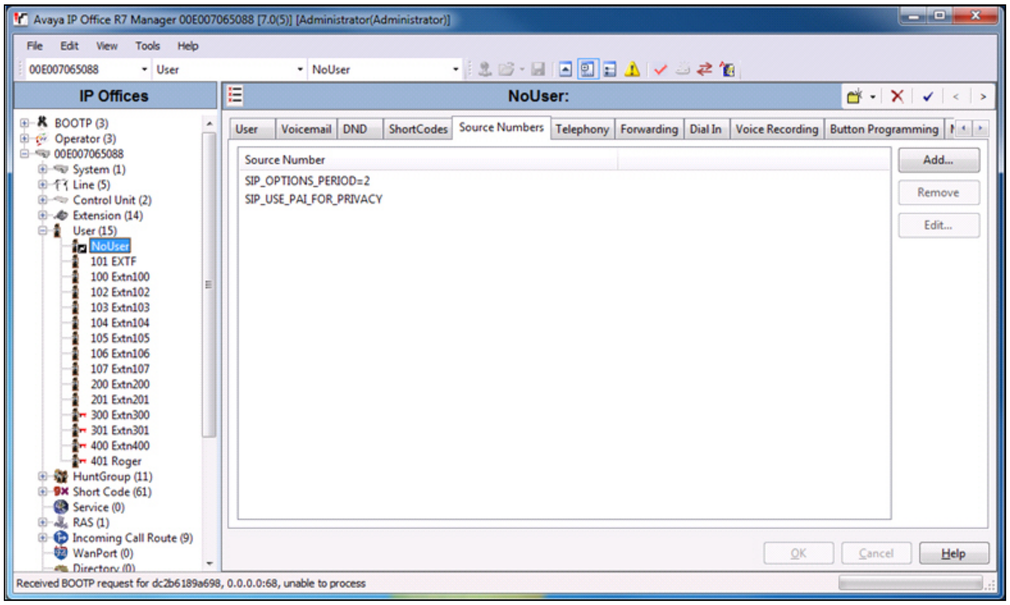
Step 10: Configuring Avaya IP PBX Users

Create user accounts and assign them to IP phone extensions by selecting **User** > **Extension Number** > **SIP**. Assign the DID telephone numbers to the **SIP Name**, **SIP Display**, and **Contact** fields. Optionally, select the **Anonymous** setting to enable privacy.



Step 11: Configuring the Avaya IP PBX P-Asserted-Identity Option

The Avaya IP PBX supports sending P-Asserted-Identity headers for user privacy. This configuration requires the creation of a **Source Number** using the string **SIP_USE_PAI_FOR_PRIVACY**. Specify this string in the **User** > **NoUser** > **Source Numbers** tab.



The Avaya IP Office PBX is now configured for interoperability with the ADTRAN SBC gateway.

Additional Resources

There are additional resources available to aid in configuring your ADTRAN SBC unit. Many of the topics discussed in this guide are complex and require additional understanding, such as using the CLI, SBC in AOS, and ANI/DNIS substitution. The documents listed in *Table 2* are available online at ADTRAN's Support Forum at https://supportforums.adtran.com.

Table 2. Additional ADTRAN Documentation

Feature	Document Title
All AOS Commands Using the CLI	*AOS Command Reference Guide*
ANI and DNIS Substitution	*Enhanced ANI/DNIS Substitution in AOS*
SBC Product Overview	*Session Border Controllers in AOS*
Media Anchoring	*Configuring Media Anchoring in AOS*
Configuring SIP Trunks on a Total Access 900 Series Using the GUI	*Total Access 900 900e SIP Trunk Quick Configuration Guide*