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Interoperability Guide

ADTRAN SBC and ShoreTel SIP Trunk Interoperability

This interoperability guide provides instructions for integrating an ADTRAN session border controller (SBC) and a ShoreTel® IP phone system using a Session Initiation Protocol (SIP) trunk to provide a connection to the service provider network. This guide includes the description of the network application, verification summary, and individual device configurations for the ADTRAN SBC and the ShoreTel 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

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 private branch exchange (PBX) system. A second SIP trunk from the gateway connects to the IP PBX. The ADTRAN SBC operates as a SIP back-to-back user agent (B2BUA) and acts as a gateway to the service provider for SIP trunking. 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 SBC in a typical network deployment.

Additional information is available online at ADTRAN's Support Community, https://supportforums.adtran.com. Specific resources are listed in *Additional Resources on page 26*.

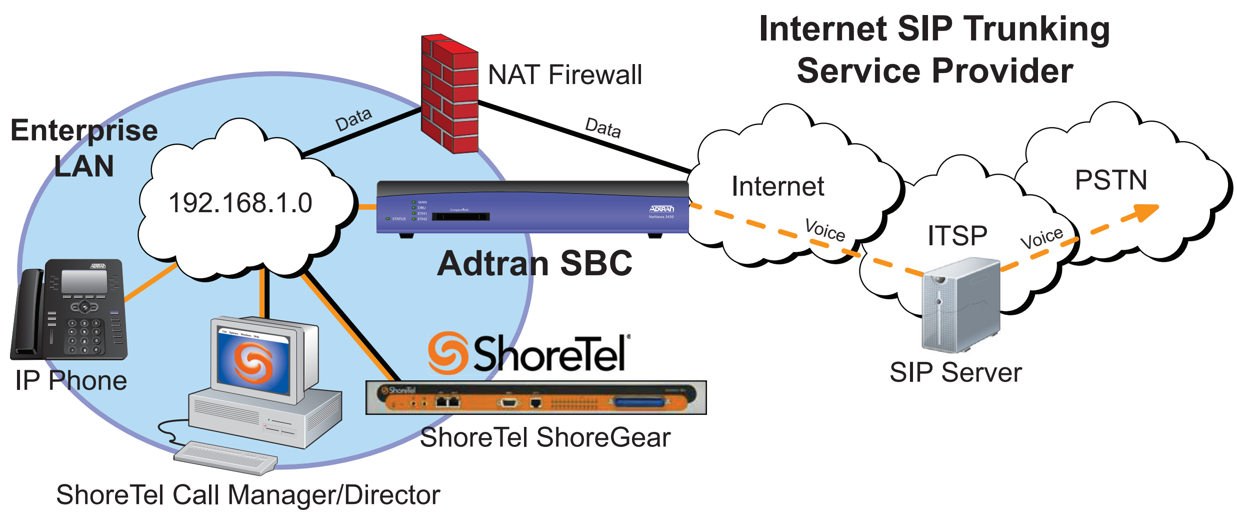


Figure 1. ADTRAN SBC in the Network

Hardware and Software Requirements and Limitations

Interoperability with the ShoreTel IP phone system 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

Products are certified via the Technology Partner Validation Process for the ShoreTel IP phone system. The table below lists the firmware releases certified for both the AOS SBC and the ShoreTel IP phone system.

Table 1. Verification Test Firmware Versions

Product	Firmware Version
AOS SBC	R10.5.3
ShoreTel	14.1

Verification Performed

Interoperability verification testing focused on SIP trunk operations between the ADTRAN SBC IP business gateway and the ShoreTel IP phone system. PBX features not specific to basic SIP trunking were not included in this verification.

Table 2. Initialization and Basic Calls

ID	Name	Description	Notes
1.1	Setup and Initialization	Verify successful set up and initialization of the system under test (SUT).	PASS
1.2	Outbound Call (Domestic)	Verify calls outbound placed through the SUT reach the external destination.	PASS
1.3	Inbound Call (Domestic)	Verify calls received by the SUT are routed to the default trunk group destination.	PASS
1.4	Device Restart – Power Loss	Verify that the SUT recovers after power loss to the SUT.	PASS
1.5	Device Restart – Network Loss	Verify the SUT recovers after loss of network link to the SUT.	PASS
1.6	All Trunks Busy – Inbound Callers	Verify an inbound caller hears the busy tone when all channels/trunks are in use.	PASS
1.7	All Trunks Busy – Outbound Callers	Verify an outbound caller hears the busy tone when all channels/trunks are in use.	PASS
1.8	Incomplete Inbound Calls	Verify proper call progress tones are provided and proper call teardown for incomplete inbound calls.	PASS

Table 3. Media and Dual-Tone Multi-Frequency (DTMF) Support

ID	Name	Description	Notes
2.1	Media Support – ShoreTel Phone to SUT	Verify call connection and audio path from a ShoreTel phone to an external destination through the service provider using all supported coder-decoders (CODECs) with both sides set to a common CODEC.	PASS
2.1	Media Support – SIP Reference to SUT	Verify call connection and audio path from SIP reference phones to an external destination through the service provider using all supported CODECs with both sides set to a common CODEC.	PASS
2.2	CODEC Negotiation	Verify CODEC negotiation between the SUT and the calling device with each side configured for a different CODEC.	PASS
2.3	DTMF Transmission – Out-of-Band/Inband	Verify transmission of inband and out-of-band digits per RFC 2833 for various devices connected to the SUT.	PASS

Table 3. Media and Dual-Tone Multi-Frequency (DTMF) Support *(Continued)*

2.4	Auto Attendant Menu	Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that calls can be transferred to the desired extension.	PASS
2.5	Auto Attendant Menu - Dial by Name	Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that calls can be transferred to the desired extension using the Dial by Name feature.	PASS
2.6	Auto Attendant Menu Checking Voice Mailbox	Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and calls can be transferred to the Voicemail Login Extension.	PASS

Table 4. Performance and Quality of Service

ID	Name	Description	Notes
3.1	Voice Quality Service Levels	Verify the SUT can provide a voice quality service level agreement (SLA) across the WAN from the customer premises to the SUT SIP gateway.	PASS
3.2	Capacity Test	Verify the service provider interface can sustain services through periods of heavy outbound and inbound load.	PASS
3.3	Post Dial Delay	Verify that the post dial delay is within acceptable limits.	PASS
3.4	Billing Accuracy	Verify that all test calls made are accurately reflected in the SUT's Call Detail Record (CDR) and billing reports.	PASS

Table 5. Enhanced Services and Features

ID	Name	Description	Notes
4.1	Caller ID Name and Number - Inbound	Verify that the caller ID name and number are received from the SIP endpoint device.	PASS
4.2	Caller ID Name and Number - Outbound	Verify that caller ID name and number are sent from SIP endpoint device.	PASS
4.3	Hold from SUT to SIP Reference	Verify connected call can be successfully put on hold and resumed.	PASS
4.4	Call Forward - SUT	Verify outbound calls that are being forwarded by the SUT are redirected and connected to the appropriate destination.	PASS
4.5	Call Transfer - Blind	Verify a call connected from the SUT to the ShoreTel phone can be blind transferred to an alternate destination.	PASS

Table 5. Enhanced Services and Features *(Continued)*

4.6	Call Transfer - Consultative	Verify a call connected from the SUT to the ShoreTel phone can be transferred with consultation to an alternate destination.	PASS
4.7	Conference - Ad Hoc	Verify successful ad hoc conference of three parties.	PASS
4.8	Inbound Direct Inward Dialing/Dialed Number Identification Service (DID/DNIS)	Verify the SUT provides inbound dialed number information and is correctly routed to the configured destination.	PASS
4.9	Outbound 911	Verify that outbound calls to 911 are routed to the correct Public Safety Answering Point (PSAP) for the calling location and that caller ID information is delivered. *Note: If **9911** is dialed, the call goes out immediately. If **911** is dialed, there is a 5 second delay. This is documented in the **ShoreTel System Administration Guide**. "If the user forgets to dial an access code before dialing 911, the system waits five seconds before routing the call to a 911-capable trunk. This pause has been introduced to eliminate accidental calls to 911."*	Conditional Pass Note: This was tested in a controlled environment without actual circuits, just to verify call placement.
4.10	Operator Assisted	Verify that 0+ calls are routed to an operator for calling assistance.	PASS
4.11	Inbound/Outbound Call with Blocked Caller ID	Verify that calls with blocked caller ID route properly, and the answering phone does not display any caller ID information.	PASS
4.12	Inbound Call to a Hunt Group	Verify that calls route to the proper hunt group and are answered by an available hunt group member with audio in both directions using G.729 and G.711 CODECs.	PASS
4.13	Inbound Call to a Workgroup	Verify that calls route to the proper workgroup and are answered successfully by an available workgroup agent with audio in both directions using G.729 and G.711 CODECs.	PASS
4.14	Inbound Call to DNIS/DID and Leave a Voicemail Message	Verify that inbound calls to a user, via DID/DNIS, routes to the proper user mailbox and a message can be left with proper audio.	PASS
4.15	Call Forward - FindMe	Verify that inbound calls are forwarded to a user's FindMe destination.	PASS
4.16	Call Forward Always	Verify that inbound calls are immediately and automatically forwarded to a user's external destination.	PASS
4.17	Inbound/Outbound Fax Calls	Verify that inbound/outbound fax calls complete successfully.	PASS

Table 5. Enhanced Services and Features *(Continued)*

4.18	ShoreTel UCB	Verify that inbound calls are properly forwarded to the ShoreTel UCB, that it properly accepts the access code, and that the conference bridge functions properly.	PASS
4.19	Inbound Call to Bridged Call Appearance (BCA) Extension	Verify that inbound calls are properly presented to all phones that have BCA configured and that the call can be answered, placed on hold, and then transferred.	PASS
4.20	Inbound Call to a Pickup Group	Verify that inbound calls are properly presented to all of the phones that have a pickup group configured and that the call can be answered, placed on hold, and then transferred.	PASS

Table 6. Security

ID	Name	Description	Notes
5.1	Digest Authentication	Verify the SUT supports the use of digest authentication for service access for both inbound and outbound calls.	N/A

ShoreTel Features and Exceptions

The configuration information provided in this document offers example configurations for the ShoreTel IP phone system and the ADTRAN SBC unit. Even though configuration requirements can vary among installations, the information provided in these steps, along with documentation provided by ADTRAN and the service provider, should prove to be sufficient. However, every design can vary and some may require more planning than others. Refer to the *ShoreTel Planning and Installation Guide*, available from the ShoreTel support website: http://support.shoretel.com/.

ShoreTel Supported Features

For more information on ShoreTel supported features of SIP trunks, refer to the *ShoreTel Administration Guide* available from the ShoreTel support website: http://support.shoretel.com/.

Some feature limitations may occur when using SIP trunks. For more information on feature limitations, refer to the *ShoreTel Administration Guide* and *ShoreTel Partner Guide,* available from the ShoreTel support website: http://support.shoretel.com/.

Configuring the ADTRAN SBC

The SBC can be configured using either the AOS 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.

To configure the SBC for interoperability with the ShoreTel IP phone system, follow these steps:

Step 1: Accessing the SBC CLI

The AOS unit can be managed by console port, http, https, telnet and SSH. Most of the initial configuration is performed through the console port or Telnet session. Accessing the AOS unit is described in this step.

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**. The default enable password is **password**. If your product no longer has the default user name and passwords, contact your system administrator for the appropriate user name and passwords.*

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 ShoreTel unit, and the second for the Ethernet WAN interface to the service provider. 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:
(config)#**interface ethernet 0/1**
(config-eth 0/1)#**description LAN**
(config-eth 0/1)#**ip address 192.168.1.1 255.255.255.0**
(config-eth 0/1)#**media-gateway ip primary**
(config-eth 0/1)#**no shutdown**

For the WAN:
(config)#**interface ethernet 0/2**
(config-eth 0/2)#**description WAN**
(config-eth 0/2)#**ip address 203.0.113.2 255.255.255.252**
(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: 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 the RTP symmetric filter using the **ip rtp symmetric-filter** command. Enter the commands as follows:

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

Step 5: Configuring Header Manipulation Rules

The following configuration example is specific to the ShoreTel IP phone system. For a generic example of a basic SBC configuration, refer to *SBC SIP Trunking Sample Configuration*.

A header manipulation rule (HMR) is created to convert a 183 Session Progress with Session Description Protocol (SDP) to a 180 Ringing with SDP. The ShoreTel unit expects a 180 Ringing, but some service providers send a 183 Session Progress instead. If your service provider never sends a 183 Session Progress, then the **Shoretel_Early_Media_Workaround** rule set and the **Shoretel_Outbound** HMR policy are unnecessary. However, they will not negatively affect the operation of the SBC if they are configured and not needed.

Refer to *Manipulating SIP Headers and Messages in AOS* for a more complete discussion of SIP HMR policies and rule sets.

Enter the commands as follows:

(config)#**hmr policy ShoreTel_Outbound**
(config-policy-ShorTel_Outbound)#**rule-set ShoreTel_Early_Media_Workaround 10**
!
(config)#**hmr rule-set ShoreTel_Early_Media_Workaround**
(config-rule-set-ShoreTel_Early_Media_Workaround)#**message-rule Convert_183_to_180 message-type response 10**
(config-msg-rule-Convert_183_to_180)#**modify header sip-status-line position first match-value "/183 Session Progress/" new-value "/180 Ringing/" 10**

Step 6: 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. Check with your service provider for any specific requirements beyond those listed in this document. Your service provider will provide you with the IP addresses or FQDN and possibly the port numbers for their SIP server. They may also provide a backup or secondary SIP server. This example uses one SIP server with the default port number (5060).

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 SIP messages.

Enter the commands as follows:

(config)#**voice trunk T01 type sip**
(config-T01)#**description PROVIDER**
(config-T01)#**sip-server primary 192.0.2.2**
(config-T01)#**trust-domain**

Step 7: Configuring the ShoreTel PBX SIP Trunk

The second voice trunk that must be configured is the SIP trunk to the ShoreTel PBX from the ADTRAN SBC. The trunk is 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 ShoreTel PBX LAN1 IP address. In addition, the ShoreTel PBX must control call transfers. This is accomplished using 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 header for outbound messages.

Enter the commands as follows:

(config)#**voice trunk T11 type sip**
(config-T11)#**description SHORETEL**
(config-T11)#**sip-server primary 192.168.1.2**
(config-T11)#**hmr Shoretel_Outbound out**
(config-T11)#**trust-domain**
(config-T11)#**grammar from host local**
(config-T11)#**transfer-mode network**

Step 8: 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 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 NXX-NXX-XXXX cost 0**
(config-PROVIDER)#**accept 1-NXX-NXX-XXXX cost 0**
(config-PROVIDER)#**accept 011-$ cost 0**
(config-PROVIDER)#**accept 411 cost 0**
(config-PROVIDER)#**accept 611 cost 0**
(config-PROVIDER)#**accept 911 cost 0**

Step 9: Configuring a Trunk Group for the ShoreTel PBX

After configuring a trunk group for the service provider, create a trunk group for the ShoreTel 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 8: Configuring a Trunk Group for the Service Provider on page 10*.

Enter the commands as follows:

(config)#**voice grouped-trunk SHORETEL**
(config-SHORETEL)#**trunk T11**
(config-SHORETEL)#**accept $ cost 0**

ADTRAN SBC Sample Configuration

The following example configuration is for a typical installation of an ADTRAN SBC IP business gateway or router with SIP trunking configured to the service provider and the ShoreTel PBX. This is the configuration that was used to validate the interoperability between the ADTRAN SBC and the ShoreTel 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 LAN
  ip address 192.168.1.1 255.255.255.0
  media-gateway ip primary
  no shutdown
!
interface eth 0/2
  description WAN
  ip address 203.0.113.2 255.255.255.252
  media-gateway ip primary
  no shutdown
!
!
ip route 0.0.0.0 0.0.0.0 203.0.113.1
!
!
voice transfer-mode local
voice forward-mode local
!
hmr policy ShoreTel_Outbound
  rule-set ShoreTel_Early_Media_Workaround 10
!
hmr rule-set ShoreTel_Early_Media_Workaround
  message-rule Convert_183_to_180 message-type response 10
    modify header sip-status-line position first match-value "/183 Session Progress/" new-value "/180
      Ringing/" 10
!
voice trunk T01 type sip
  description PROVIDER
  sip-server primary 192.0.2.2
  trust-domain
!
```



```
voice trunk T11 type sip
 description SHORETEL
 sip-server primary 192.168.1.2
 hmr Shoretel_Outbound out
 trust-domain
 grammar from host local
 transfer-mode network
!

voice grouped-trunk PROVIDER
 trunk T01
 accept NXX-NXX-XXXX cost 0
 accept 1-NXX-NXX-XXXX cost 0
 accept 011-$ cost 0
 accept 411 cost 0
 accept 611 cost 0
 accept 911 cost 0
!
voice grouped-trunk SHORETEL
 trunk T11
 accept $ cost 0
!
!
ip rtp symmetric-filter
ip rtp media-anchoring
!
end
```

Configuring the ShoreTel PBX

The ShoreTel PBX is configured using the ShoreTel product's GUI (ShoreWare Director). Refer to the ShoreTel documentation for detailed instructions about accessing the GUI.

The following section describes ShoreTel system configuration to support SIP trunking interoperability with the ADTRAN SBC. The section is divided into general system settings and trunk configurations (both group and individual) needed to support SIP trunking.

The first configuration section pertains to general system settings and includes call control, site administration, and switch administration. If these items have already been configured on the system, skip this section and continue to *Configuring System Settings on page 18*.

NOTE	*ShoreTel points its individual SIP trunks to the ADTRAN SBC.*

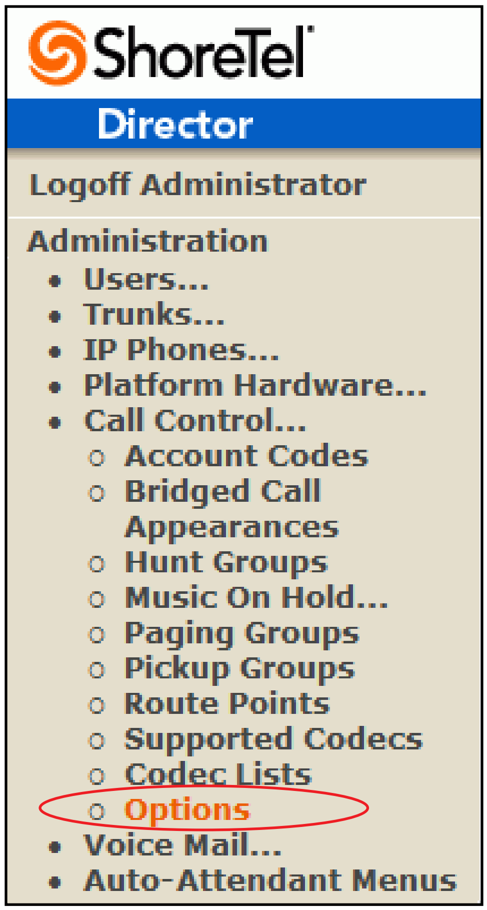
Configuring General Settings

To configure the General Settings for the ShoreTel PBX, follow these steps:

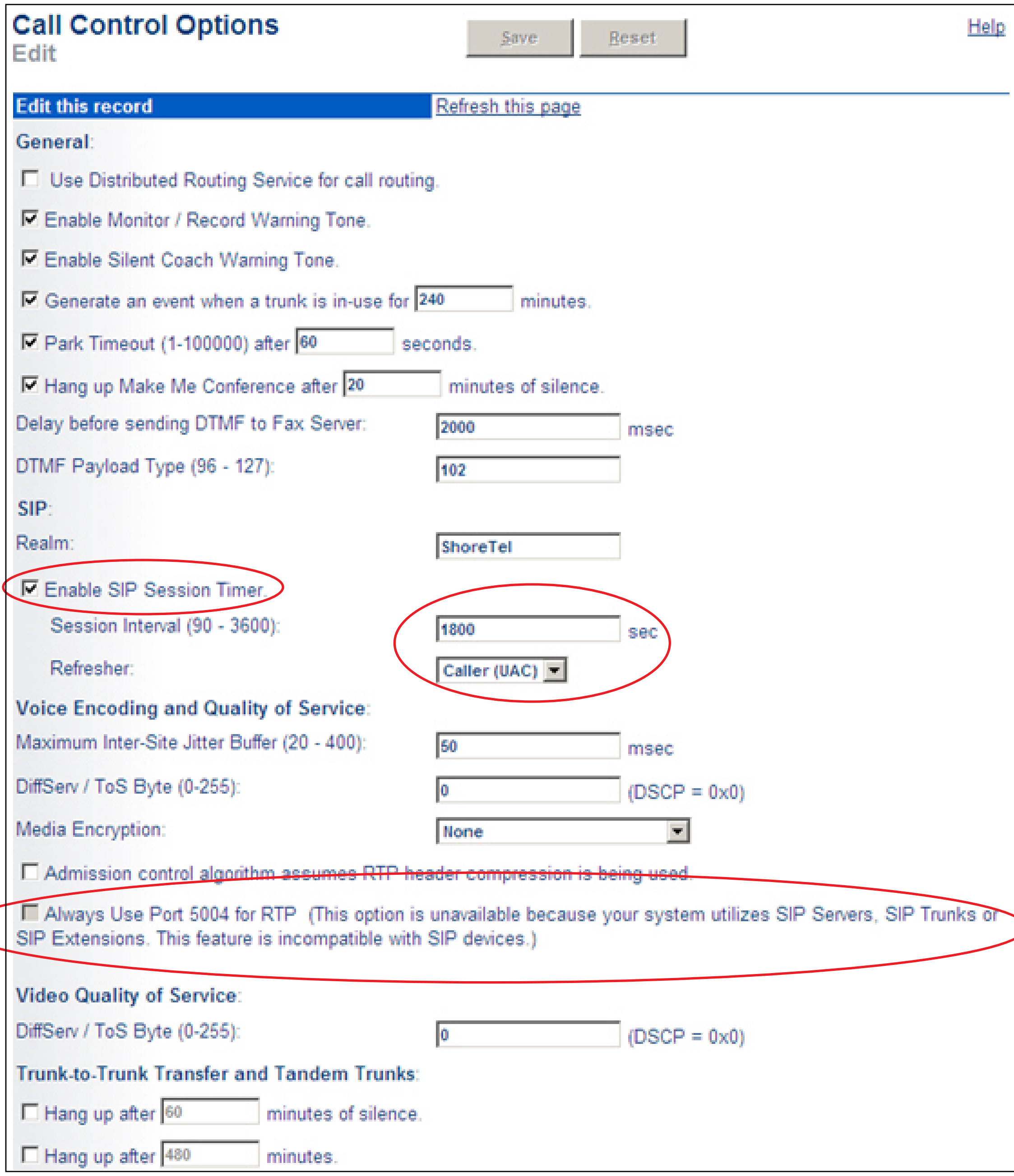
- *Configure Call Control on page 14*
- *Configure Sites Administration on page 16*
- *Configure Switch Administration on page 17*

Configure Call Control

1. To configure the Call Control settings, log into ShoreWare Director and navigate to **Administration** > **Call Control** > **Options**.



2. The **Call Control Options** menu will appear.



3. Ensure the **Enable SIP Session Timer** option is checked.
4. Set the **Session Interval** timer. The recommended session interval is **1800** seconds.
5. Select the appropriate refresher (from the drop-down menu) for the **SIP Session Timer**. Set the **Refresher** field to either **Caller (UAC)** [User Agent Client] or **Callee (UAS)** [User Agent Server]. If **Caller (UAC)** is selected, the caller's device will be in control of the session timer refresh. If **Callee (UAS)** is selected, the device of the person called will control the session timer refresh.

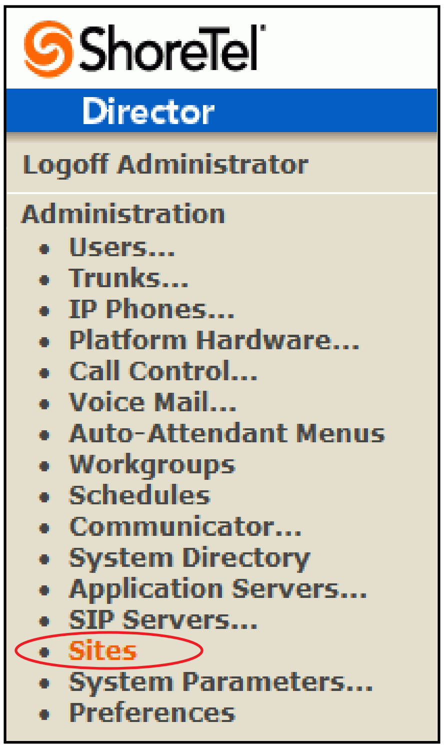
6. Uncheck the box for **Always Use Port 5004 for RTP**. For SIP configurations, Dynamic User Datagram Protocol (UDP) must be used for RTP traffic. If the box is unchecked, Media Gateway Control Protocol (MGCP) will no longer use UDP port 5004; MGCP and SIP traffic will use dynamic UDP ports.

> NOTE *The option **Always Use Port 5004 for RTP** will be grayed out by default if SIP servers, SIP trunks, or SIP extensions are configured.*

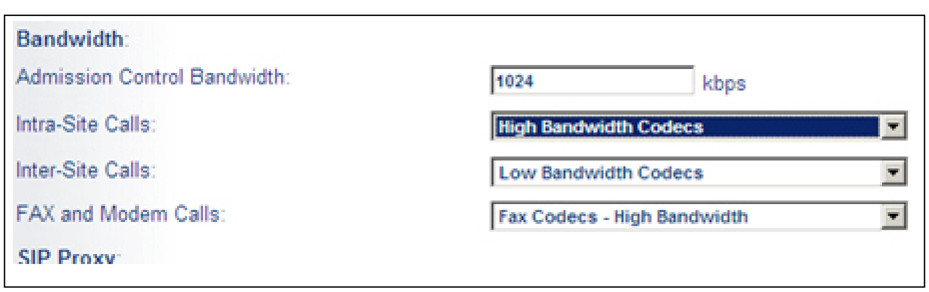
7. Reboot the following items: IP Phones, ShoreGear Switches, ShoreWare Director, Distributed Voice Services/Remote Servers, Conference Bridges, and Contact Centers. If a full system reboot is not performed, one-way audio will occur during initial testing.

Configure Sites Administration

1. To configure the sites settings (related to the administration of sites) navigate to **Administration** > **Sites**.



2. In the **Bandwidth** section of the **Sites Edit** menu that appears, set the **Admission Control Bandwidth**.



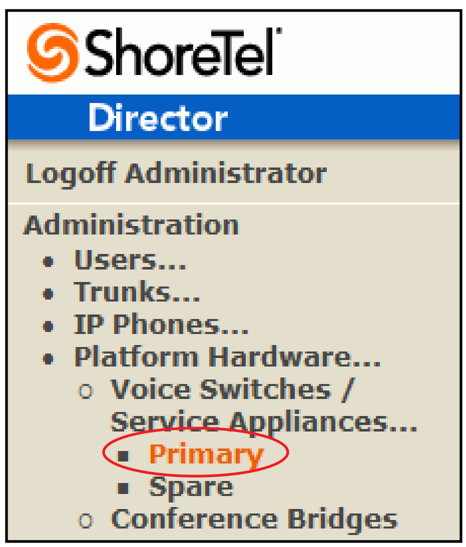
3. The **Admission Control Bandwidth** setting defines the bandwidth available to and from the site. This is important because SIP devices will be counted against the site bandwidth. Bandwidth must be set appropriately based on the site's configuration with the service provider SIP trunking. Refer to the *ShoreTel Planning and Installation Guide* for more information.
4. Set the **Intra-Site Calls** and the **Inter-Site Calls** settings next. For the **Intra-Site Calls**, verify that the desired audio bandwidth is selected from the CODEC list for calls within the system. The settings should then be confirmed for the desired audio bandwidth CODEC list for **Inter-Site Calls** (calls between sites).

> NOTE *The CODEC list selection is used as an example. Refer to the **ShoreTel Planning and Installation Guide** for additional information on CODEC list selections.*

> NOTE *SIP uses both G.711 and G.729 CODECs. The CODEC setting will be negotiated to the highest CODEC supported (fax requires G.711 at minimum).*

Configure Switch Administration

1. The final general setting to configure is allocating ports for the switch settings. To make these changes, navigate to **Administration** > **Platform Hardware > Voice Switches / Service Appliances > Primary**.



2. From the **Voice Switches** menu that appears, select the name of the switch to configure. The **Edit ShoreGear Switch** screen will display.
3. From the **Edit ShoreGear Switch** menu, select the desired number of SIP trunks from the ports available.
4. Each port designated as a SIP trunk enables the support for five individual trunks.
5. Select the check box for **Enable Jack Based Music On Hold** if MOH is expected on the SIP trunk.

> NOTE
>
> *For more information on configuring file based MOH, refer to the **ShoreTel Administration Guide** available from the ShoreTel support website:* http://support.shoretel.com/.

Configuring System Settings

To configure the System Settings for the ShoreTel PBX, follow these steps:

- *Configure SIP Trunk Groups on page 19*
- *Configure Trunk Services on page 22*
- *Configure Individual Trunks on page 24*

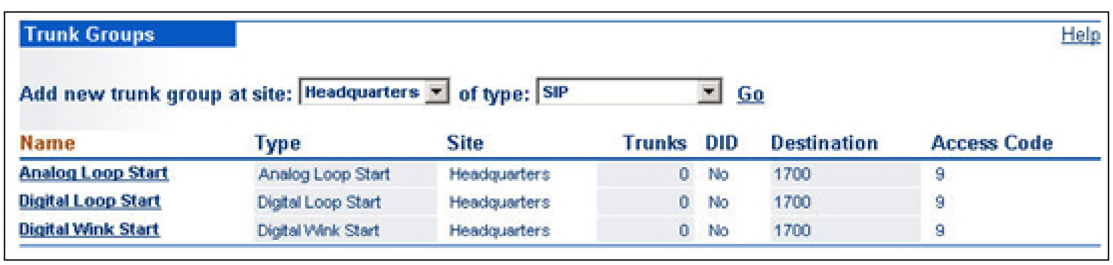
Configure SIP Trunk Groups

If the SIP trunk groups have already been configured on the system, proceed to the section *Configure Individual Trunks on page 24*.

> **NOTE** *ShoreTel trunk groups support only static SIP endpoint individual trunks.*

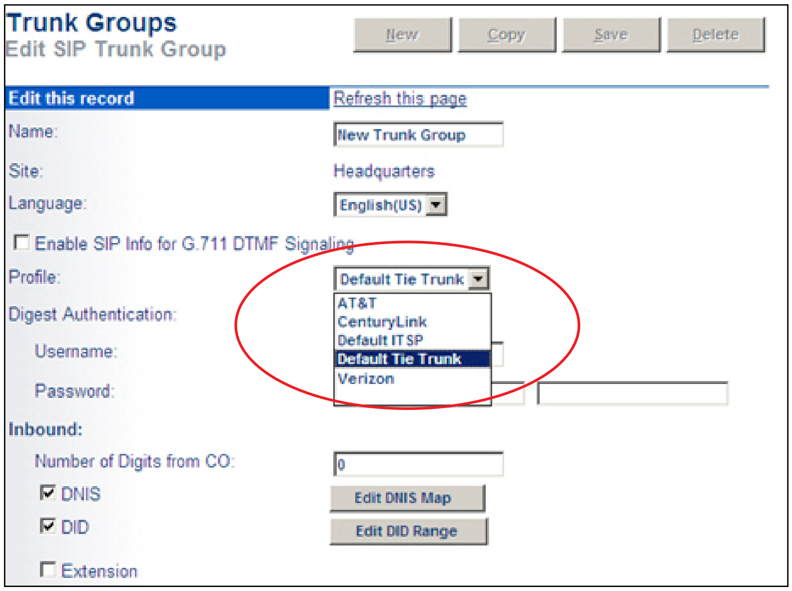
1. To modify the SIP trunk groups settings, navigate to **Administration** > **Trunks** > **Trunk Groups**.

2. From the drop-down menus on the **Trunk Groups** menu, select the desired site and select the **SIP** trunk type to configure.

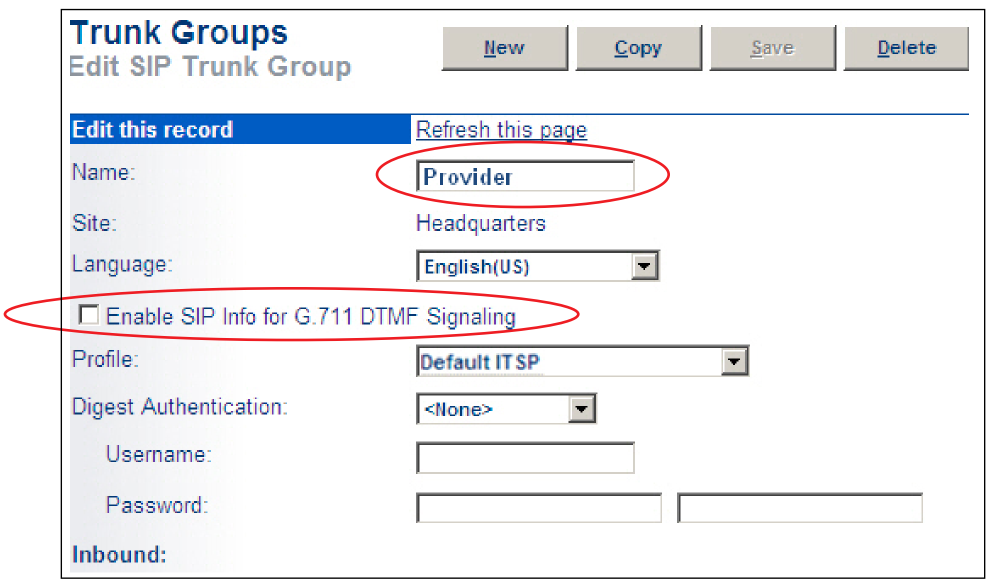


3. Select the **Go** link to the right of the **Add new trunk group at site** option. The **Edit SIP Trunk Group** menu will appear.

4. Select your service provider or **Default ITSP** profile from the drop-down list.



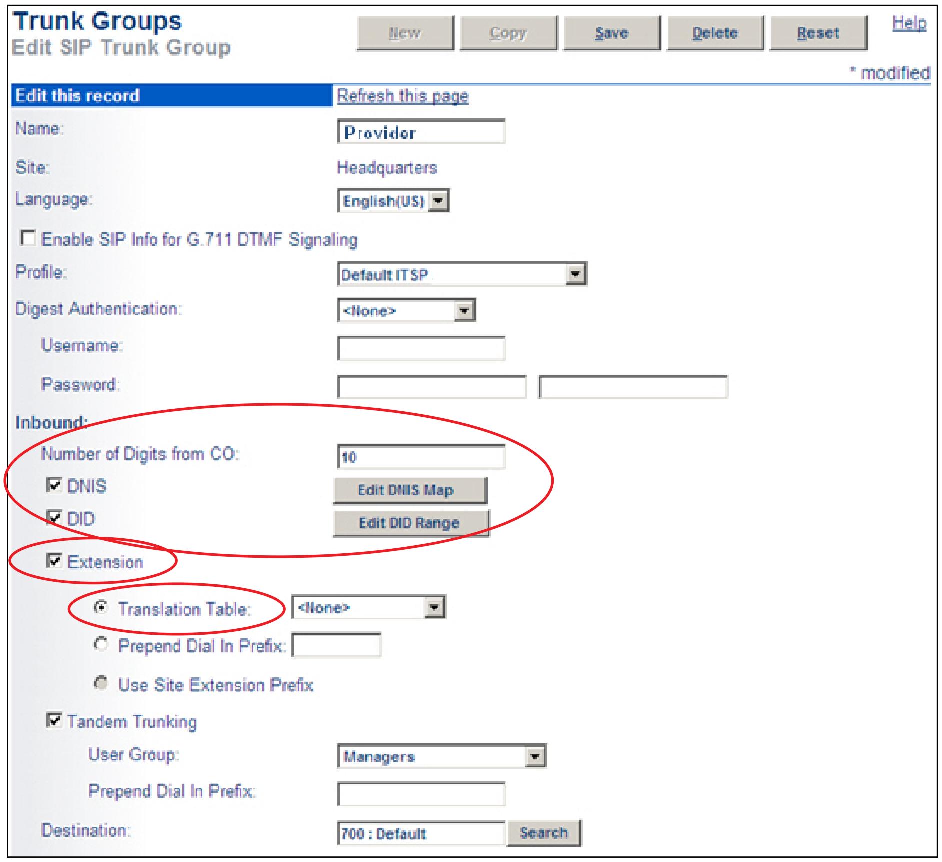
5. From the **Edit SIP Trunks Group** menu, enter a name for the trunk group. In the example below, the name **Provider** has been created.



6. Ensure **Enable SIP Info for G.711 DTMF Signaling** is not checked (disabled).

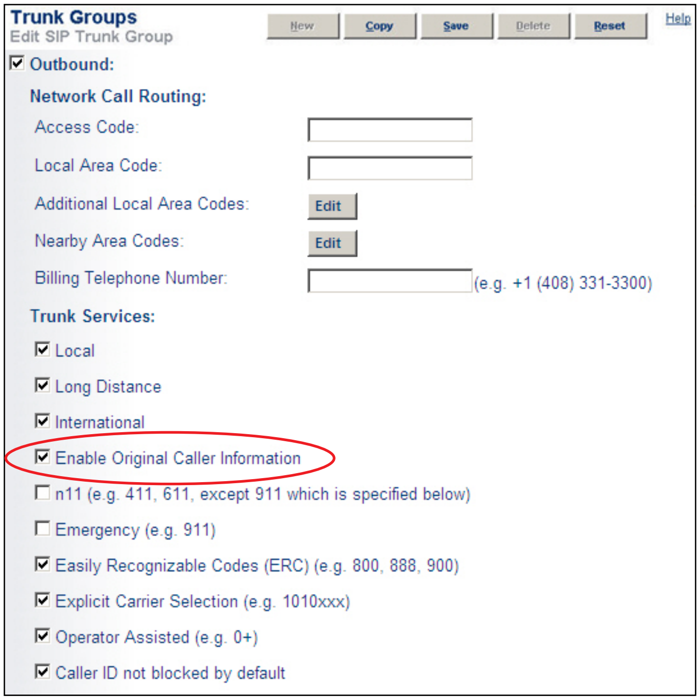
> **NOTE** *The* ***Digest Authentication*** *field is not required when connecting to an ADTRAN SBC.*

7. From the **Inbound** settings at the bottom of the **Edit SIP Trunks Group** menu, ensure the **Number of Digits from CO** is set to **10**. Enable the **DNIS** or **DID** parameters as necessary for your installation. It is not necessary to enable the **Extension** parameter for SIP Trunks, because it defaults to disabled, but it can be enabled if desired (refer to the *ShoreTel Planning and Installation Guide* for further information).
8. Enable **Tandem Trunking** if you plan on transferring calls to external parties via the SIP trunk.

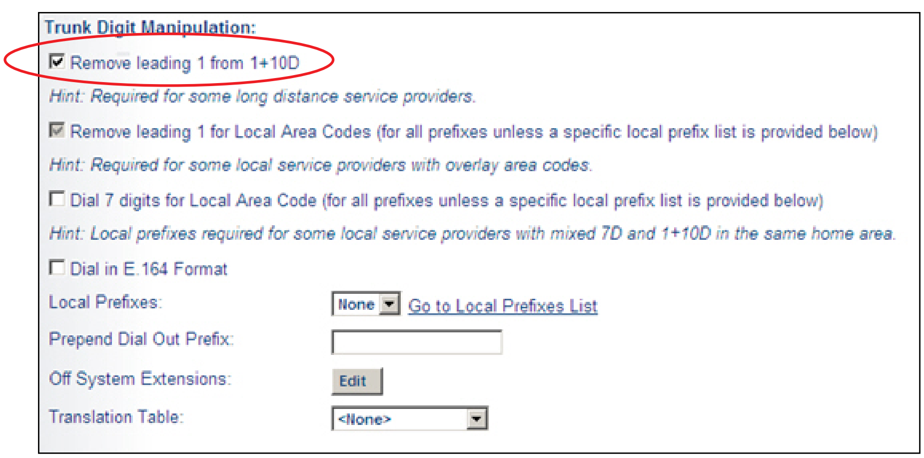


Configure Trunk Services

1. From the **Trunk Groups** menu in the **Trunk Services** section, enable or disable the appropriate services based on what the service provider supports and which features are required from this trunk group. The parameter **Enable Original Caller Information** should be enabled.



2. From the **Trunk Digit Manipulation** section, make sure the **Remove leading 1 from 1+10D** parameter is enabled (checked).

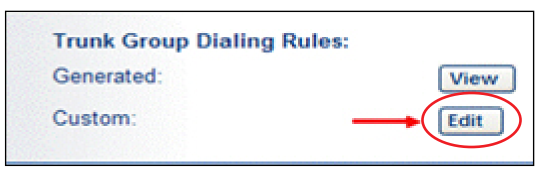


3. After these settings have been configured in the **Edit SIP Trunk Group** menu, select **Save** to accept the changes.

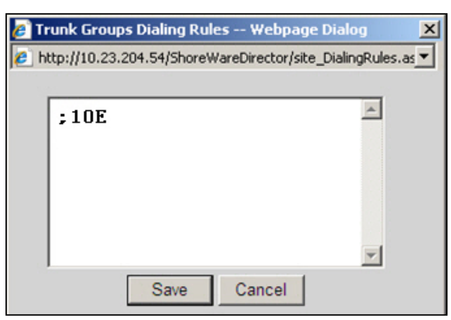
4. This completes the settings necessary to configure the trunk groups on the ShoreTel system. Log out of ShoreTel Director.

5. You will then be presented with the ShoreTel Director login page. From this page, you will enable the **Support Entry** mode of the ShoreTel Director. On your keyboard, hold down the <CTRL> and <Shift> keys and with the mouse pointer click on the **U** of **Username**.

6. Log into ShoreTel Director with your usual administration user credentials.
7. Navigate to the **Edit SIP Trunk Group** menu, by selecting **Administration** > **Trunks** > **Trunk Groups**.
8. From the **Trunk Groups** menu, select the trunk group you created for your service provider (in the previous steps). The **Edit SIP Trunk Group** menu will appear.
9. Scroll down to the bottom of the menu to the **Trunk Group Dialing Rules** parameter section. Select the **Edit** button to the right of the **Custom** parameter.



10. The **Trunk Groups Dialing Rules – Webpage Dialog** will appear. In the blank area of the webpage dialog, enter, **;10E** and select **Save**. Be sure to enter the exact syntax shown here. Include the semicolon, one, and zero, followed by an uppercase E. This syntax is case sensitive, so verify that it matches the screen below.

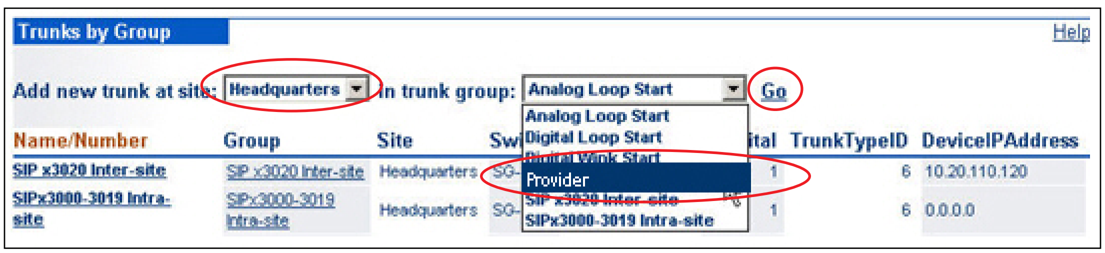


This completes the settings necessary to set up the trunk groups on the ShoreTel system.

Configure Individual Trunks

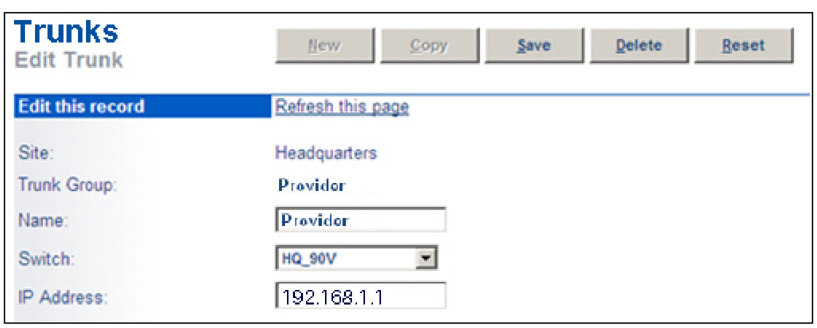
1. To configure the individual trunks, navigate to **Administration > Trunks > Individual Trunks**.

2. From the **Trunks by Group** menu that appears, select the site for the new individual trunk(s) to be added. Select the appropriate trunk group from the drop-down list in the **Add new trunk at site** area. In this example, the site is **Headquarters** and the trunk group is **SIP**.



3. Select the **Go** link to the right of the **Add new trunk at site** option to bring up the **Edit Trunk** screen.

4. From the individual trunks **Edit Trunk** menu, enter a name for the individual trunk. Select the appropriate switch, SIP trunk type, and enter the number of trunks. When selecting a name, it is recommended that you name the individual trunks the same as the name of the trunk group so that the trunk type can be tracked easily.



5. Select the switch upon which the individual trunk will be created. For the service provider trunk, select **Use IP Address** and enter the IP address of the ADTRAN SBC device.

6. Select the number of individual trunks desired, each one supports **one** audio path. For example, if 5 is entered, then 5 audio paths can be up at one time. Once these changes are complete, press **Save** to make the changes.

> NOTE: *Individual SIP trunks cannot span networks. SIP trunks can only terminate on the switch selected. There is no failover to another switch. For redundancy, two trunk groups will be needed with each pointing to another ADTRAN SBC.*

After setting up the trunk groups and individual trunks, refer to the *ShoreTel Product Installation Guide* to make the appropriate changes for the User Group settings. The ShoreTel PBX is now configured for interoperability with the ADTRAN SBC gateway.

Troubleshooting

The ADTRAN SBC unit has several **show** and **debug** commands that display information on the console session. The **show** commands display a snapshot of the activity on the ADTRAN SBC unit. The **debug** commands display real time activity. Some useful **show** and **debug** commands are listed below.

Use the **show ip rtp media sessions** command to display all of the anchored RTP flow associations and the number of relayed packets per association currently active in an anchored RTP flow. In addition, the and the session type (digital signal processing (DSP), media-anchored, or transcoded) for the association is displayed.

#show ip rtp media sessions

Use the **debug voice switchboard** command to display the decision making process as it routes incoming and outgoing calls. The debug output of this command displays the trunk from which the call was received and the trunk to which the call was routed.

#debug voice switchboard

Use the **debug sip stack messages** command to display the SIP messages that are received by or sent from the ADTRAN SBC. This command displays the full SIP headers as well as the SDP body and will tell whether the ADTRAN SBC received or transmitted the message.

#debug sip stack messages

Packet Capture

The Packet Capture capability of the ADTRAN SBC allows the capture and export of all traffic on any one or all interfaces simultaneously. It can then be exported to your PC where it can be viewed in Wireshark or Ethereal. For more information about configuring packet capture, refer to the configuration guide *Configuring Packet Capture in AOS*.

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 HMR. The documents listed in *Table 7* are available online at ADTRAN's Support Forum at https://supportforums.adtran.com.

Table 7. Additional ADTRAN Documentation

Feature	Document Title
All AOS Commands Using the CLI	*AOS Command Reference Guide*
Sample Configuration for SIP Trunking on ADTRAN SBC	*SBC SIP Trunking Sample Configuration*
SBC Product Overview	*Session Border Controllers in AOS*
Media Anchoring	*Configuring Media Anchoring in AOS*
Firewall (IPv4)	*Configuring the Firewall (IPv4) in AOS*
Security Settings	*Security Best Practices for AOS Products*
Access Control Lists	*Configuring IP Access Control Lists (ACLs) in AOS*
Access Control Policies	*Configuring Access Policies in AOS*
HMR	*Manipulating SIP Headers and Messages in AOS*

