



## Configuration Guide

# Configuring a SIP Trunking Gateway for Use with NetVanta ECS

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This configuration guide outlines the steps necessary to configure an ADTRAN session border controller (SBC) as a Session Initiation Protocol (SIP) trunking gateway to use with the NetVanta Enterprise Communications System (ECS). The guide includes an overview of the ADTRAN SBC, an overview of the SIP trunking gateway, outlines the steps necessary to configure the AOS device using the command line interface (CLI) and the Web-based graphical user interface (GUI), as well as troubleshooting information.

This guide consists of the following sections:

- *ADTRAN SBC Overview on page 2*
- *Hardware and Software Requirements and Limitations on page 2*
- *Configuring the SIP Trunking Gateway Using the CLI on page 2*
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- *Configuration Example on page 16*
- *Command Summary on page 18*
- *Troubleshooting on page 20*
- *Additional Resources on page 21*

## ADTRAN SBC Overview

ADTRAN added functionality to their IP business gateways and NetVanta routers with the introduction of the SBC feature pack (available with AOS release R10.1.0). The SBC feature pack is an additional firmware package that can be installed on specific ADTRAN products as outlined in the *Product Feature Matrix*, available online at <https://supportforums.adtran.com>. The features described in this document do not apply to any prior firmware release.

Among the many features included with the SBC feature pack is the enhanced ability for the AOS device to serve as a SIP trunking gateway with the NetVanta ECS as a unified communications solution. By incorporating this function into the AOS device, the IP business gateway or router can provide Voice over Internet Protocol (VoIP) telephony service for public switched telephone networks (PSTNs) access for enterprise networks. The AOS SBC device is an integrated all-in-one solution that providers can use for all levels of their VoIP requirements.

There is additional information available online at ADTRAN's Support Forum, <https://supportforums.adtran.com>. Specific resources are listed in *Additional Resources on page 21*.

## SIP Trunking

SIP trunking is a packet-based voice service that routes calls over an IP network to an IP private branch exchange (PBX) or voice switch using SIP signaling to place and receive calls. The typical SIP trunk service provider offers extensive cost savings compared to conventional trunk services. The IP connection to the provider carries all traffic, such as local, long distance, and toll free calls, video, email, Internet, data, and other media over a single circuit. Calls into PSTNs are also handled by the SIP service provider by passing the calls off to a media gateway that connects to the PSTN for users not using VoIP service. SIP trunks are configured over data connections, they can be configured over an Ethernet or a T1 connection.

## Hardware and Software Requirements and Limitations

The SIP trunking gateway feature is only available on ADTRAN products running the SBC feature pack as outlined in the *Product Feature Matrix*, available online at ADTRAN's Support Forum, <https://supportforums.adtran.com>.

## Configuring the SIP Trunking Gateway Using the CLI

To configure the SIP trunking gateway functionality on an AOS product using the CLI, use the following steps:

1. Configure global voice modes for local handling.
2. Configure the voice trunk to the service provider.
3. Configure voice trunk to NetVanta ECS.
4. Configure a trunk group for the service provider.
5. Configure a trunk group for NetVanta ECS.
6. Enable the RTP symmetric filter.

7. Enable media anchoring.
8. Configure double reINVITE preference.

## Accessing the 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.
```



*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.



*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 1: Configure Global Voice Modes for Local Handling

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

The following example demonstrates these commands entered in sequence, starting with **voice transfer-mode local** command at the Global Configuration mode prompt:

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

## Step 2: Configure the Voice Trunk to the Service Provider

The first of two voice trunks is configured in this step. The parameters for this trunk configure the SIP trunk to the service provider from the ADTRAN SBC. The minimum amount of configuration is provided in this example but your application may require additional settings depending on your service provider requirements. Check with your service provider for any specific requirements beyond those listed here.

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. Once in the Voice Trunk Configuration mode, you can provide a descriptive name for the trunk and define the SIP server's primary IPv4 address (or hostname). Use the **description** command to label the trunk. Use the **sip-server primary** command to define the hostname or IPv4 address of the primary server to which the trunk will send call-related SIP messages.

The following example demonstrates these commands entered in sequence, starting with the **voice trunk type sip** command at the Global Configuration mode prompt:

```
(config)#voice trunk T01 type sip
(config-T01)#description PROVIDER
(config-T01)#sip-server primary 198.51.100.2
```

## Step 3: Configure Voice Trunk to NetVanta ECS

A second voice trunk is configured defining the connection from the ADTRAN SBC to the NetVanta ECS. Use the same commands as in *Step 2: Configure the Voice Trunk to the Service Provider on page 4*, only change the trunk identifier, trunk description, and IPv4 address (or hostname). In addition, this SIP trunk requires the NetVanta ECS to control call transfers, which is configured by issuing the **transfer-mode network** command.

The following example demonstrates these commands entered in sequence, starting with the **voice trunk type sip** command at the Global Configuration mode prompt:

```
(config)#voice trunk T11 type sip
(config-T11)#description UC SERVER
(config-T11)#sip-server primary 192.168.2.200
(config-T11)#transfer-mode network
```

## Step 4: Configure a Trunk Group for the Service Provider

In this step, an individual trunk group is created for the service provider trunk account. The trunk group is used to assign outbound call destinations (local calls, long distance calls, etc.) to the group, which in turn is applied to the trunk account once the SIP trunk is added to the trunk group. 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 outbound calls may 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 out the trunk. This command controls the type of outbound calls users can place on the system. 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.

The available wildcards for this command are:

- 0-9** = Match exact digit only.
- M** = Any digit 1 to 8.
- X** = Any single digit (0 to 9).
- N** = Any digit 2 to 9.
- [123]** = Any digit contained in the bracketed list.



*The following special characters are always ignored; (), +, commas, and spaces. The dash character is used inside brackets to imply a range of numbers. For example, [1-4] defines numbers 1, 2, 3, or 4. Commas are implied between numbers in the brackets.*

The following are some examples of outbound call templates:

1. 555-81xx matches 555-8100 to 555-8199.
2. 555-812[012] matches 555-8120 to 555-8122.
3. 1-800\$ matches any 1-800 calls.
4. NXX-XXXX matches 7 digit local.
5. 1-NXX-NXX-XXXX matches long distance calls in North America.

The **cost** parameter specifies the cost value for the trunk if a call is accepted by several trunks. The call will be routed to the trunk with the lowest cost value. The valid range is 0 to **499**.

The following CLI example demonstrates these commands entered in sequence, starting with the **voice grouped-trunk** command at the Global Configuration mode prompt:

```
(config)#voice grouped-trunk PROVIDER
(config-PROVIDER)#trunk T01
(config-PROVIDER)#accept N11
(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 5: Configure a Trunk Group for NetVanta ECS

In this step, an individual trunk group is created for the NetVanta ECS trunk account. An existing trunk is added to the trunk group using the **trunk <Txx>** command. The allowed outbound calls are defined using the **accept <pattern>** command and assigned a cost using the **cost** parameter. The valid ranges for each of the command in this step are explained in detail in the previous step, *Step 4: Configure a Trunk Group for the Service Provider on page 4*.

The following CLI example demonstrates these commands entered in sequence, starting with the **voice grouped-trunk** command at the Global Configuration mode prompt:

```
(config)#voice grouped-trunk UC_SERVER
(config-UC_SERVER)#trunk T11
(config-UC_SERVER)#accept XXXX cost 0
(config-UC_SERVER)#accept 256-555-XXXX cost 0
```

## Step 6: Enable the RTP Symmetric Filter

The Realtime Transport Protocol (RTP) symmetric filter works in conjunction with media anchoring to filter nonsymmetric RTP packets. It is enabled by default on some platforms and disabled on others. If this feature is enabled, it will drop RTP packets destined to a particular port that are sourced from an IP address and port that does not match what was received in Session Description Protocol (SDP). To enable the RTP symmetric filter, use the **ip rtp symmetric-filter** command. Use the **no** form of this command to disable the feature.

To enable the RTP symmetric filter, enter the command as follows from the Global Configuration mode:

```
(config)#ip rtp symmetric-filter
```

## Step 7: Enable Media Anchoring

Media anchoring honors the setting for the RTP symmetric filter which was configured in *Step 6: Enable the RTP Symmetric Filter*. Additionally, media anchoring must be enabled. The media anchoring settings described in this guide provide only the minimal configuration. Media Anchoring is a complex feature and includes many optional settings. Additional settings and configuration may be necessary for your particular application. Refer to the configuration guide *Configuring Media Anchoring in AOS* available online at <https://supportforums.adtran.com> for more details.

Enable media anchoring using the **ip rtp media-anchoring** command from the Global Configuration mode. This command enables media anchoring on a global basis for all RTP-to-RTP calls. By default, media anchoring is disabled. Using the **no** form of this command disables the media anchoring feature.

To enable media anchoring, enter the command as follows from the Global Configuration mode:

```
(config)#ip rtp media-anchoring
```

## Step 8: Configure Double ReINVITE Preference

The **ip sip prefer double-reinvite** command is used in the Global Configuration mode to determine whether a double reINVITE is preferred globally for all calls in the system. Calls that typically require a double reINVITE are forwarded calls and any attended transfer. When these calls connect, a double reINVITE is initiated.

By default, the system is configured so that double reINVITES are preferred. If a transfer involves a SIP trunk operating in local transfer mode, a double reINVITE will be 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. Using the **no** form of this command indicates that double reINVITES are not globally preferred.

To specify that SIP double reINVITES are not preferred in the system, enter the command as follows from the Global Configuration mode:

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

## Configuring the SIP Trunking Gateway Using the GUI

The GUI is an especially useful tool for those who are less familiar with CLI configuration. AOS products ship with a user-friendly GUI that can be used to perform many basic management and configuration functions on the AOS product.

Most of the configuration steps explained previously in *Configuring the SIP Trunking Gateway Using the CLI on page 2* can also be performed using the GUI, with exception of the final two steps. This section provides all of the steps that can be completed using the GUI, with the final two steps provided using the CLI. This section will explain how to access the GUI to perform the steps necessary to configure the SIP Trunking Gateway functionality using the GUI. It is written in a manner which allows you to complete the steps without referring to *Configuring the SIP Trunking Gateway Using the CLI on page 2*.

To configure the SIP trunking gateway functionality on an AOS product using the GUI, use the following steps:

1. Configure global voice modes for local handling.
2. Configure the voice trunk to the service provider.
3. Configure the voice trunk to NetVanta ECS.
4. Configure a trunk group for the service provider.
5. Configure a trunk group for NetVanta ECS.
6. Enable the RTP symmetric filter.
7. Enable media anchoring.
8. Configure double reINVITE preference.

### Accessing the GUI

To begin configuring the SIP trunking gateway through the GUI, follow these steps to access the GUI:

1. Open a new Web page in your Internet browser.
2. Enter your AOS product's IP address in the browser's address field, **http://<ip address>**, for example:  
**http://10.10.10.1**
3. At the prompt, enter your user name and password and select **OK**.



*The default user name is **admin** and the default password is **password**.*

### Step 1: Configure Global Voice Modes for Local Handling

Configure the ADTRAN SBC to use local mode for forwarding and call transfer handling. By default, both of these functions are handled by the network. To change this setting, navigate to **Voice > System Setup > System Parameters**. In the System Parameter menu, select **Local** from the drop-down box for both **Transfer Mode** and **Forward Mode**. Select **Apply** at the bottom of the menu to accept the changes.

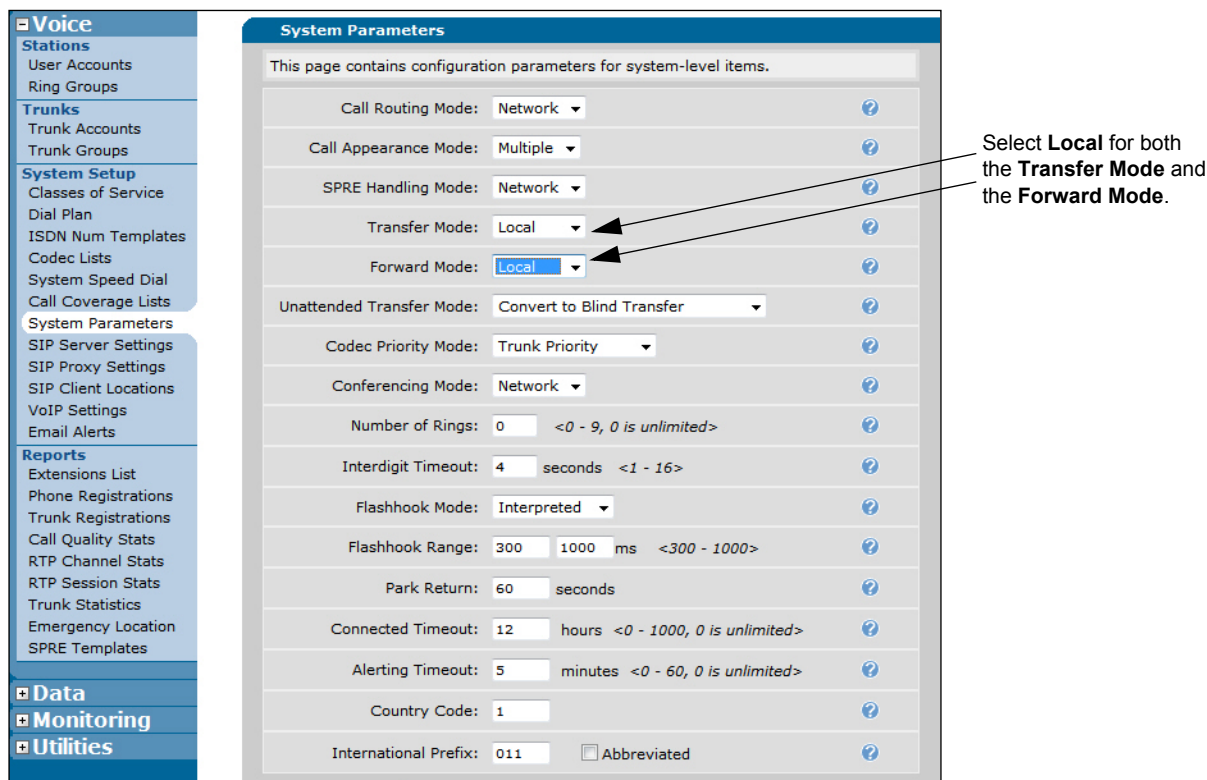


Figure 1. Setting System Parameters

### Step 2: Configure the Voice Trunk to the Service Provider

The first of two voice trunks is configured in this step. The parameters for this trunk configure the SIP trunk to the service provider from the ADTRAN SBC. The minimum configuration is covered in this example, but your application may require additional settings depending on your service provider requirements. Check with your service provider for any specific requirements beyond those listed here.

To create the voice trunk, navigate to **Voice > Trunks > Trunk Accounts**. Enter the **Trunk Name**, and select **SIP** for the **Type** from the drop down menu. Select **Add** to create the voice trunk.

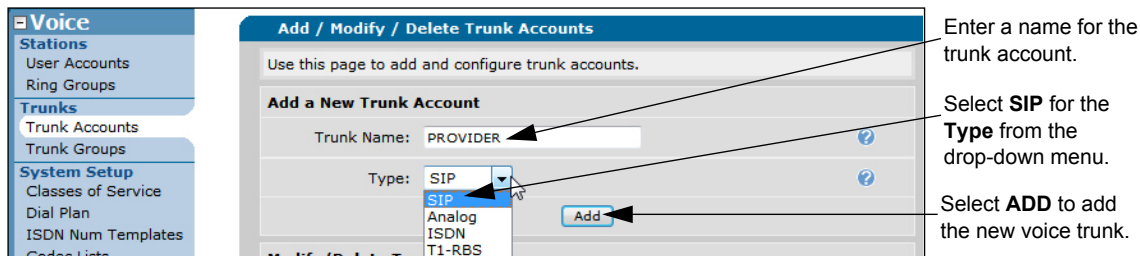
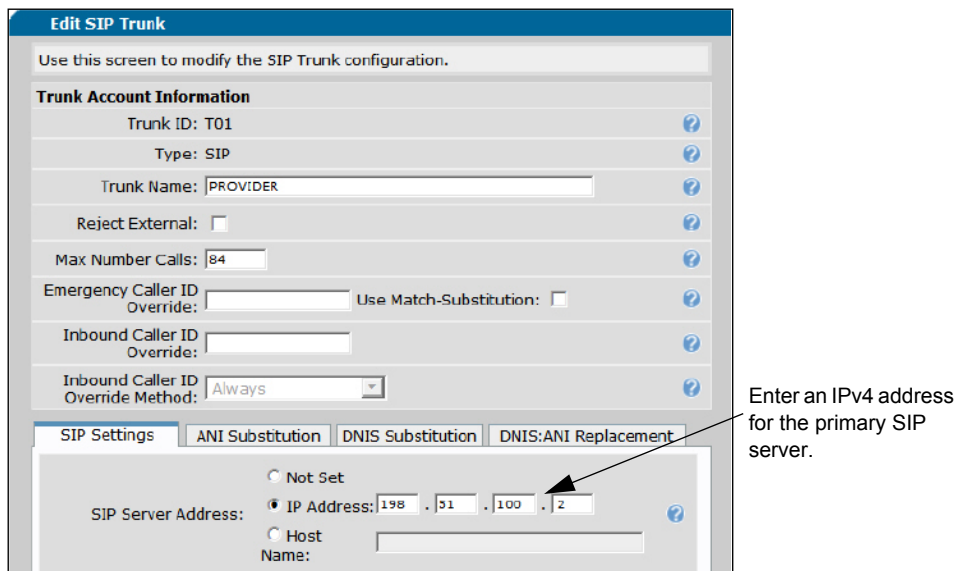


Figure 2. Creating a Voice Trunk to the Service Provider



Once the new trunk account is created, select it from the list to further configure the voice trunk.

Add the primary SIP server IPv4 address or hostname under the **SIP Settings** tab as shown in *Figure 3*. Select **Apply** at the bottom of the **Edit SIP Trunk** menu to accept the changes.

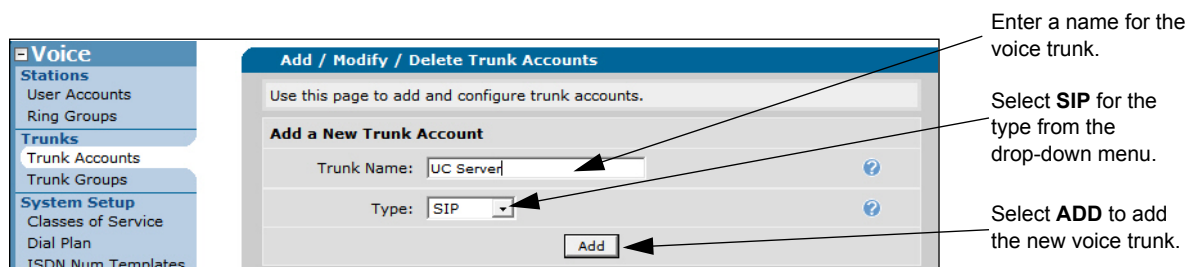


**Figure 3. Configuring a Voice Trunk to the Service Provider**

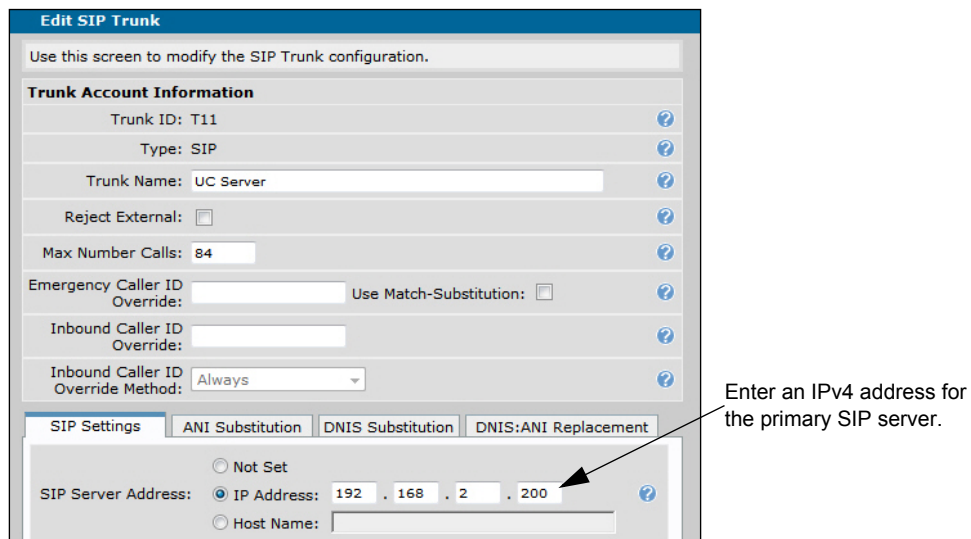
### Step 3: Configure the Voice Trunk to the NetVanta ECS

A second voice trunk is configured defining the connection from the ADTRAN SBC to the NetVanta ECS. Use the same method as described in *Step 2: Configure the Voice Trunk to the Service Provider*.

To create the voice trunk, navigate to **Voice > Trunks > Trunk Accounts**. Enter the **Trunk Name**, and select **SIP** for the **Type** from the drop down menu. Create the SIP trunk using a different name and primary IPv4 address or hostname. For this configuration example, we used the name **UC SERVER**. Select **Add** to create the voice trunk.

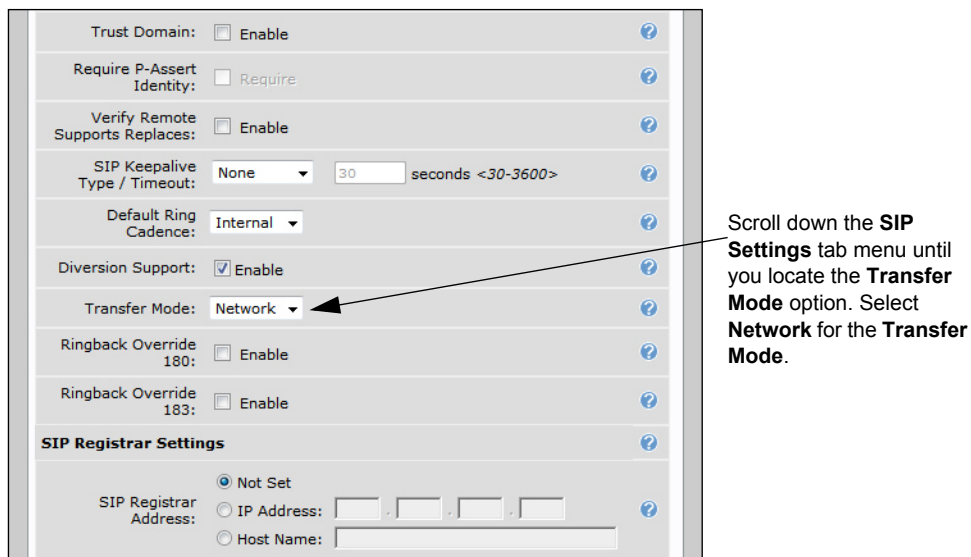


**Figure 4. Create a Voice Trunk to the NetVanta ECS**



**Figure 5. Configuring a Voice Trunk to the NetVanta ECS**

In addition, this SIP trunk requires the NetVanta ECS to control call transfers, which is configured by setting the **Transfer Mode** to **Network**. The **Transfer Mode** setting is located halfway down the menu on the **SIP Settings** tab (shown in *Figure 6*).



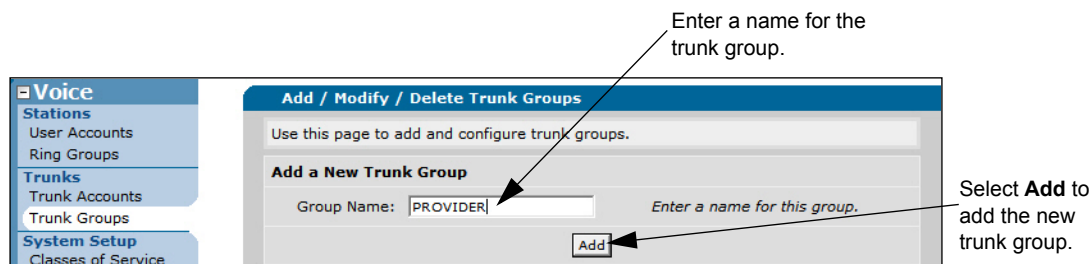
**Figure 6. Setting the Transfer Mode for this Voice Trunk Only**

Select **Apply** at the bottom of the **Edit SIP Trunk** menu to accept the changes.

### Step 4: Configure a Trunk Group for the Service Provider

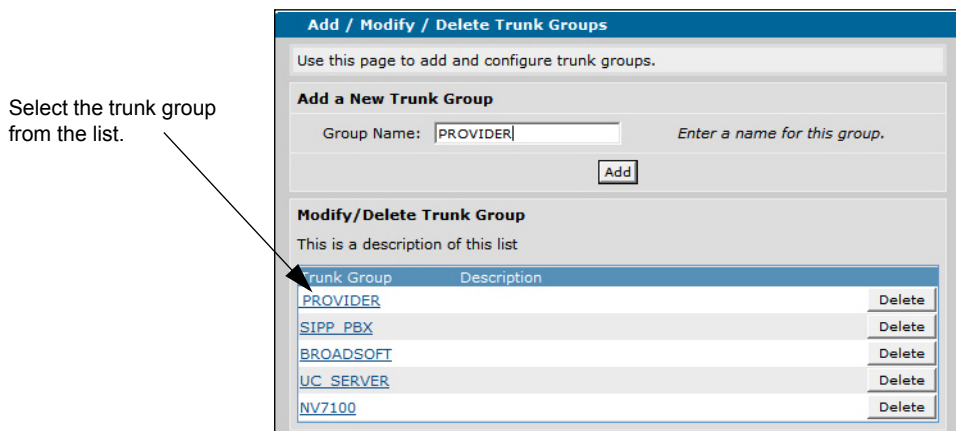
In this step, an individual trunk group is created for the service provider trunk account. The trunk group is used to assign outbound call destinations (local calls, long distance calls, etc.) to the group, which in turn is applied to the trunk account once the SIP trunk is added to the trunk group. A cost is also assigned to each **accept** template in the trunk group.

Navigate to **Voice > Trunks > Trunk Groups** to create a grouped trunk. Enter a name for the trunk group and select **Add** as shown in *Figure 7 on page 11*.



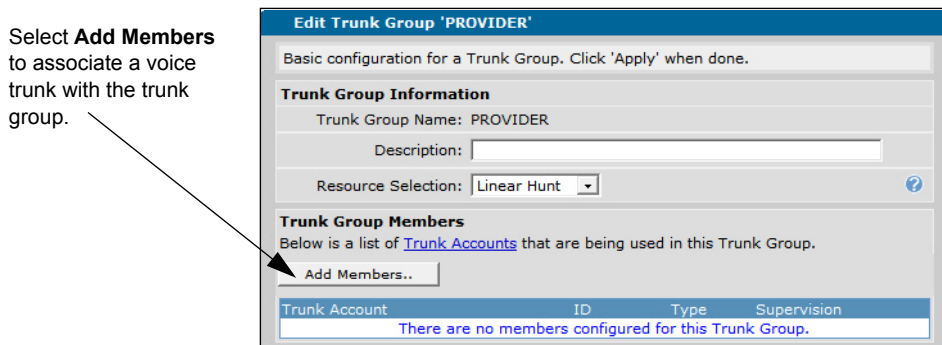
**Figure 7. Creating a Trunk Group for the Service Provider**

Select the newly created trunk group from the **Add/Modify/Delete Trunk Groups** list for further configuration.



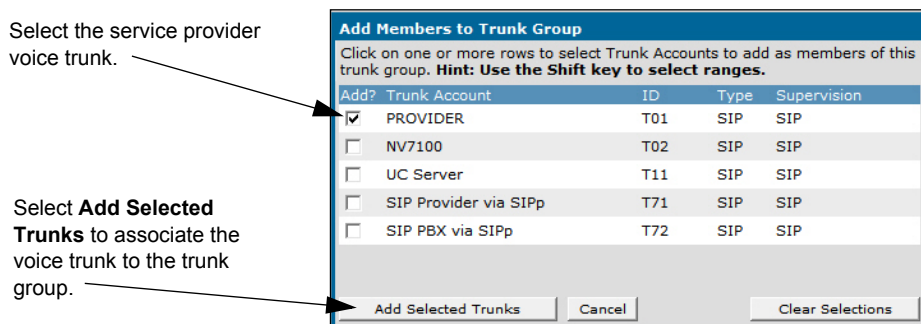
**Figure 8. Configuring the Trunk Group for the Service Provider**

To add an existing trunk to the trunk group so outbound calls can be placed out that particular trunk, select **Add Members** from the **Edit Trunk Group** menu as shown in *Figure 9*.



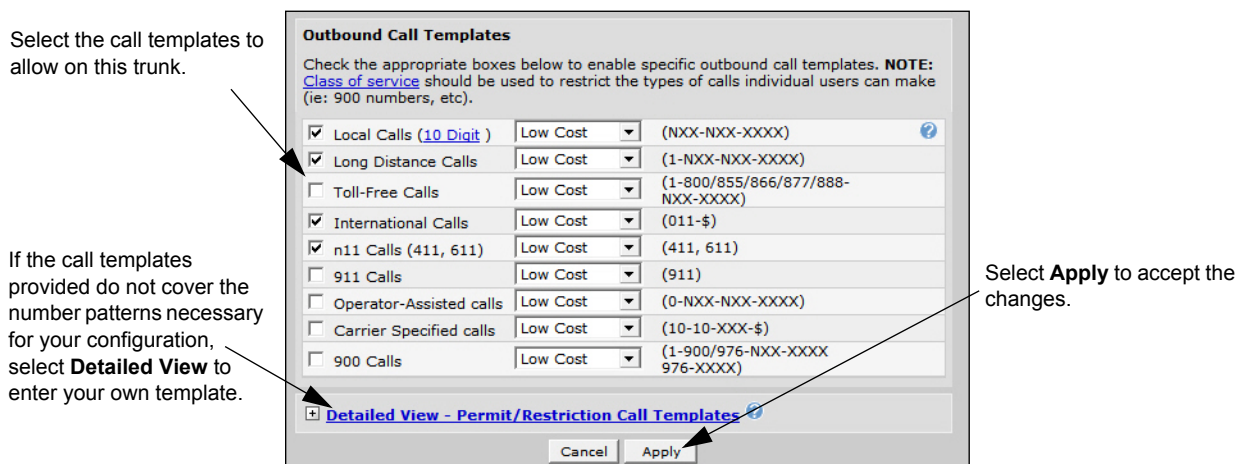
**Figure 9. Associating a Voice Trunk with the Trunk Group**

Select the voice trunk for the service provider from the **Add Members to Trunk Group** menu. Select **Add Selected Trunks** to accept the association as shown in *Figure 10*.



**Figure 10. Associating a Voice Trunk with the Trunk Group**

Next, specify number patterns that are accepted for routing out the trunk from the **Outbound Call Templates** menu. This controls the type of outbound calls users can place on the system. Select from the existing call templates shown in *Figure 11* by checking the box next to the template.



**Figure 11. Configuring Outbound Call Templates**

If the call templates provided do not suit your needs, refer to *Defining Advanced Call Templates* below for instructions for creating custom call templates.

Select **Apply** at the bottom of the **Edit Trunk Groups** menu to accept the changes.

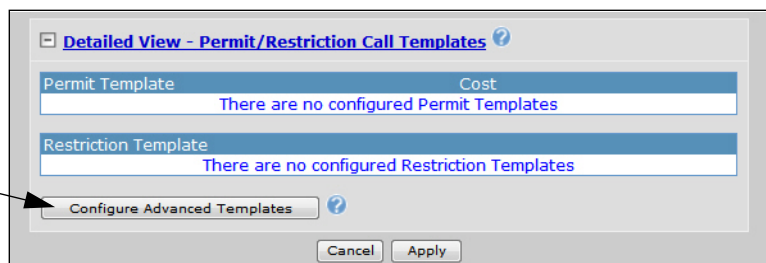
## Step 5: Configure a Trunk Group for NetVanta ECS

Repeat the same steps as described in *Step 4: Configure a Trunk Group for the Service Provider on page 11* to create and configure a trunk group for the NetVanta ECS. For this step, name the trunk group **UC\_SERVER** and associate it with the trunk account **UC Server** (created in *Step 3: Configure the Voice Trunk to the NetVanta ECS on page 9*).

### Defining Advanced Call Templates

If the provided call templates (shown in *Figure 11*) are not sufficient for this trunk group, your application could require advanced configuration. Select **The Detailed View - Permit/Restriction Call Templates** option from the **Outbound Call Templates** menu, and select **Configure Advanced Templates** as shown in *Figure 12*.

Select **Configure Advanced Templates** to access the **Add/Delete Permit Templates** menu.



**Figure 12. Accessing Advanced Templates**

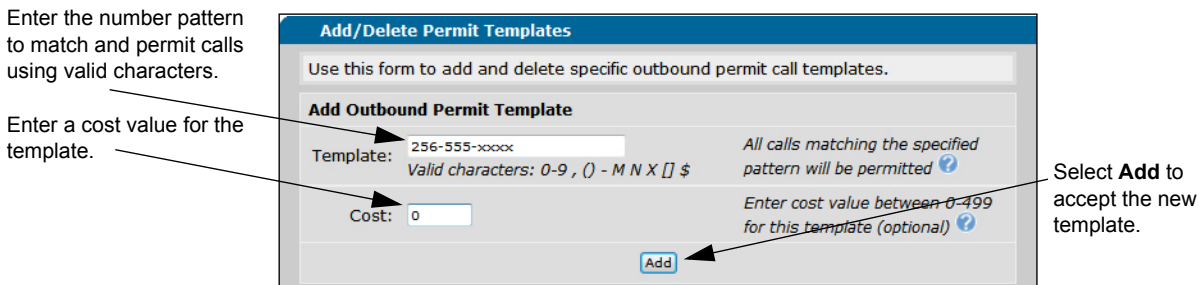
The advanced templates allow entering a complete phone number or using wildcards to help define accepted numbers. Valid characters for entering number patterns are:

- 0-9** = Match exact digit only.
- M** = Any digit 1 to 8.
- X** = Any single digit (0 to 9).
- N** = Any digit 2 to 9.
- [123]** = Any digit contained in the bracketed list.

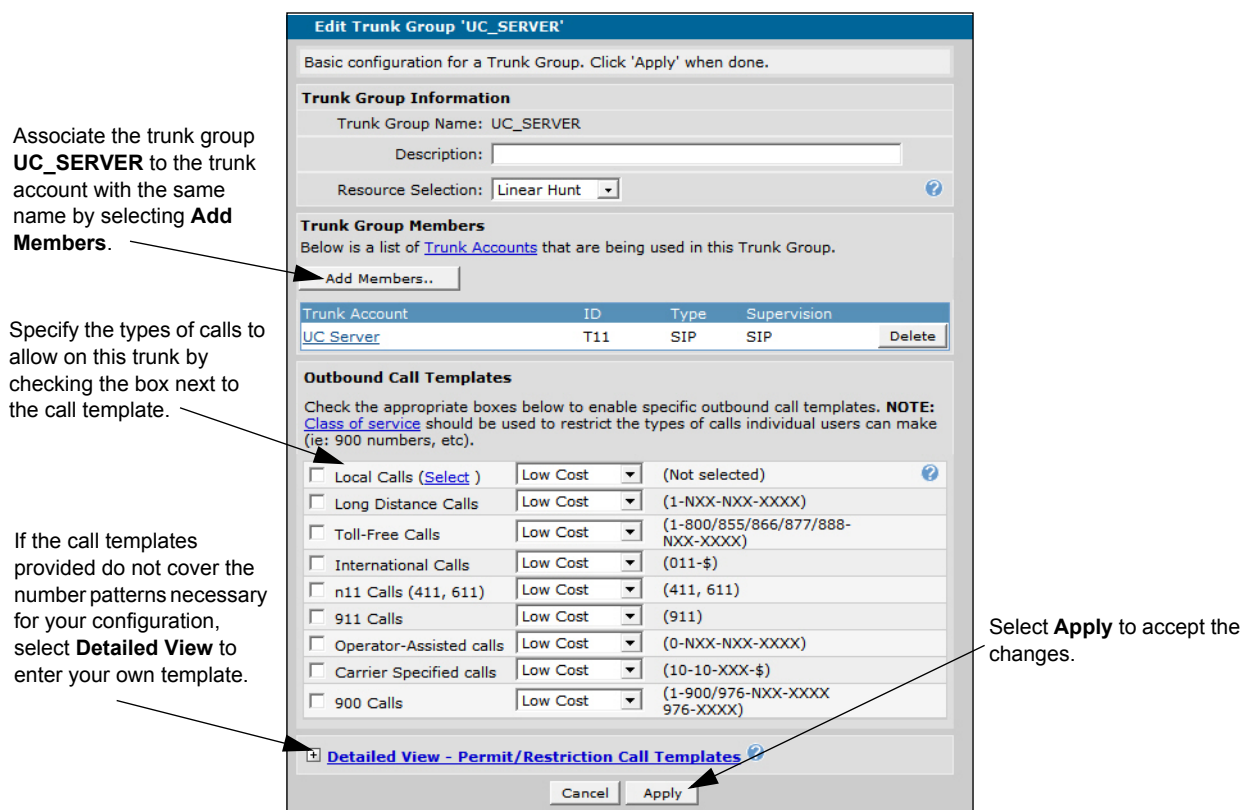


*The following special characters are always ignored; ( ), +, commas, and spaces. The dash character is used inside brackets to imply a range of numbers. For example, [1-4] defines numbers 1, 2, 3, or 4. Commas are implied between numbers in the brackets.*

Use the **Add/Delete Permit Templates** menu to define custom call templates allowing outbound calls as needed for this trunk group.



**Figure 13. Adding Custom Templates**



**Figure 14. Configuring the Trunk Group for the NetVanta ECS**

Select **Apply** at the bottom of the **Edit Trunk Groups** menu to accept the changes.

**Step 6: Enable the RTP Symmetric Filter**

The last step to be performed through the GUI is setting the RTP symmetric filter. This functionality filters nonsymmetric RTP packets and should be enabled on the ADTRAN SBC for media anchoring. When this setting is enabled, it will drop RTP packets destined to a particular port that are sourced from an IP address and port that does not match what was received in SDP.

Navigate to **Voice > System Setup > VoIP Settings**. From the **RTP Settings** tab, select the checkbox to enable **RTP Symmetric Filter**. Select **Apply** to accept the changes.

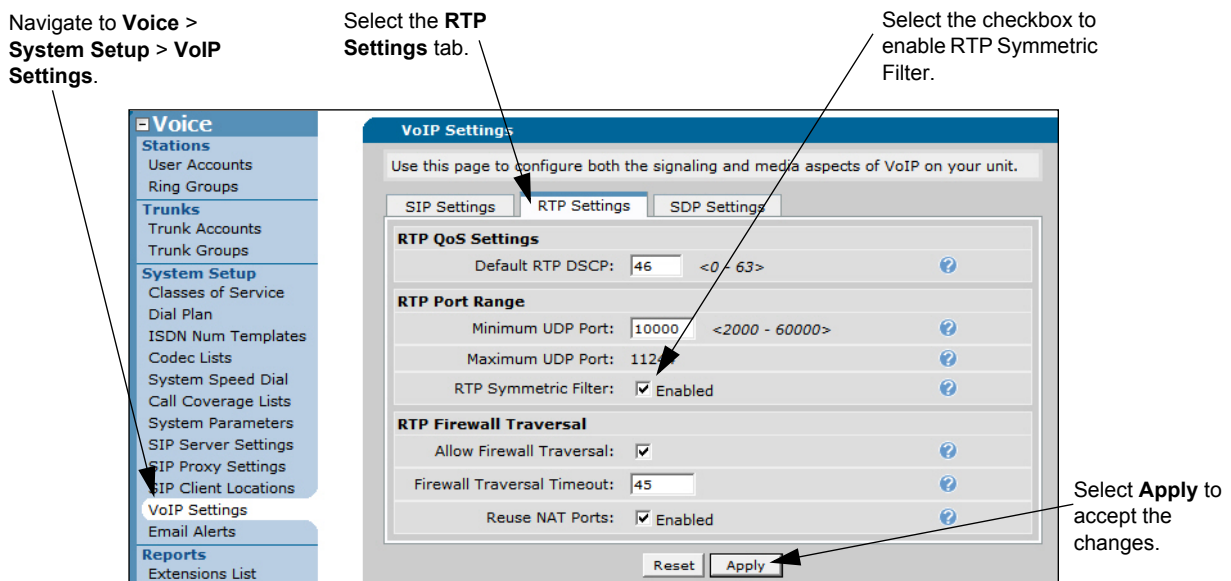


Figure 15. Enabling RTP Symmetric Filter

## Step 7: Enable Media Anchoring Through the CLI

Step 7 and Step 8 cannot be completed through the GUI and must be performed using the CLI. Access the CLI using the steps explained previously in *Accessing the CLI on page 3*. If you are unfamiliar with using the CLI, more detailed information is provided in the *AOS Command Reference Guide* available online from <https://supportforums.adtran.com>.

Media anchoring honors the settings for the RTP symmetric filter which were configured through the GUI in *Step 6: Enable the RTP Symmetric Filter on page 14*. Additionally, media anchoring must be enabled, which is the last step in configuring this feature.

The media anchoring settings described in this guide provide only the minimum configuration. Additional settings and configuration may be necessary for your particular application. Refer to the configuration guide *Configuring Media Anchoring in AOS* available online at <https://supportforums.adtran.com> for further details.

Enable media anchoring using the **ip rtp media-anchoring** command from the Global Configuration mode. This command enables media anchoring on a global basis for all RTP-to-RTP calls. By default, media anchoring is disabled. Using the **no** form of this command disables the media anchoring feature.

To enable media anchoring, enter the command as follows from the Global Configuration mode:

```
(config)#ip rtp media-anchoring
```

## Step 8: Configure Double ReINVITE Preferences Through the CLI

The `ip sip prefer double-reinvite` command is used in the Global Configuration mode to determine whether a double reINVITE is preferred globally for all calls in the system. Typically, calls that require a double reINVITE are forwarded calls and any attended transfer. When these calls connect, a double reINVITE message is initiated.

By default, the system is configured so that double reINVITES are preferred. If a transfer involves a SIP trunk operating in local transfer mode, a double reINVITE will be 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. Using the `no` form of this command indicates that double reINVITES are not globally preferred.

To specify that SIP double reINVITES are not preferred in the system, enter the command as follows from the Global Configuration mode:

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

## Configuration Example

The following example demonstrates a typical installation of an ADTRAN SBC configured as the SIP trunking gateway between a NetVanta ECS and a service provider. All of the following configuration in this example occurs on the ADTRAN SBC shown in [Figure 16](#).

Two SIP trunks are configured, T01 and T11. T01 is configured as the SIP trunk to the service provider with the IPv4 address of 198.51.100.2. T11 is configured as the SIP trunk to the NetVanta ECS with the IPv4 address of 192.168.2.200. Two voice trunk groups are created and configured with outbound call templates. Additionally, a cost is assigned to each accept template in the trunk group.

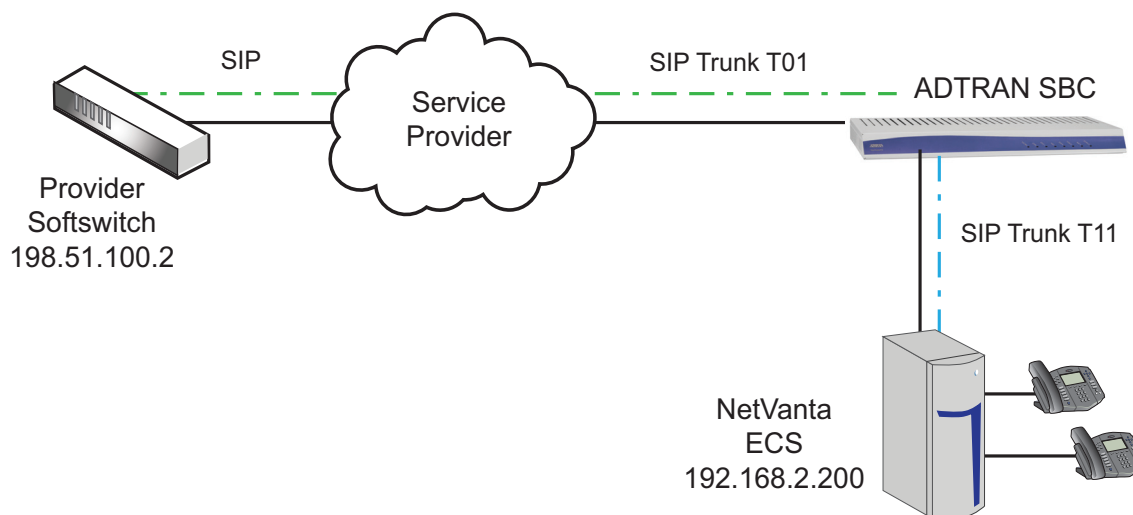


Figure 16. SIP Trunking Gateway with NetVanta ECS





*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 configuration 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.*

```
!  
voice transfer-mode local  
voice forward-mode local  
!  
!  
voice trunk T01 type sip  
  description PROVIDER"  
  sip-server primary 198.51.100.2  
!  
!  
voice trunk T11 type sip  
  description UC Server  
  sip-server primary 192.168.2.200  
  transfer-mode network  
!  
!  
!  
voice grouped-trunk PROVIDER  
  trunk T01  
  accept N11  
  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 UC_SERVER  
  trunk T11  
  accept XXXX cost 0  
  accept 256-555-XXXX cost 0  
!  
!  
no ip sip prefer double-reinvite  
!  
!  
ip rtp symmetric-filter  
ip rtp media-anchoring  
!  
end
```

## Command Summary

The following table summarizes the commands used to configure simple remote phone in AOS products.

**Table 1. Remote Phone Command Summary**

Step	Command	Description
<b>Step 1</b>	Configure global voice modes for local handling.	
	(config)# <b>voice transfer-mode local</b>	Specifies the local unit will control call transfers.
	(config)# <b>voice forward-mode local</b>	Specifies the local unit will control forwarding of calls.
<b>Step 2</b>	Configure the voice trunk to the service provider.	
	(config)# <b>voice trunk &lt;Txx&gt; type sip</b>	Creates a SIP trunk and enters the SIP Trunk Configuration mode.
	(config-Txx)# <b>description &lt;text&gt;</b> (config-Txx)# <b>sip-server primary &lt;value&gt;</b>	Use the <b>description</b> command to provide a descriptive label to the trunk. Use the <b>sip-server primary</b> command to define the hostname or IPv4 address of the primary server to which the trunk will send SIP messages. The <i>&lt;value&gt;</i> parameter is specified using the fully qualified domain name (FQDN) or IPv4 address of the SIP server. IPv4 addresses should be expressed in dotted decimal notation (for example, 10.10.10.1).
<b>Step 3</b>	Configure voice trunk to NetVanta ECS.	
	(config)# <b>voice trunk &lt;Txx&gt; type sip</b>	Creates a SIP trunk and enters the SIP Trunk Configuration mode.
	(config-Txx)# <b>description &lt;text&gt;</b> (config-Txx)# <b>sip-server primary &lt;value&gt;</b>	Use the <b>description</b> command to provide a descriptive label to the trunk. Use the <b>sip-server primary</b> command to define the hostname or IPv4 address of the primary server to which the trunk will send SIP messages. The <i>&lt;value&gt;</i> parameter is specified using the fully qualified domain name (FQDN) or IPv4 address of the SIP server. IPv4 addresses should be expressed in dotted decimal notation (for example, 10.10.10.1).
	(config-Txx)# <b>transfer-mode network</b>	Specifies the behavior of the SIP trunk to allow the network to control call transfers. This setting is only necessary on the SIP trunk to the NetVanta ECS and not on the SIP trunk to the service provider.

Table 1. Remote Phone Command Summary (Continued)

Step	Command	Description
<b>Step 4</b>	Configure a trunk group for the service provider.	
	(config)# <b>voice grouped-trunk</b> <name>	Creates a trunk group and enters the Voice Trunk Group Configuration mode.
	(config-grouped-trunk-name)# <b>trunk</b> <Txx>	Adds a trunk to the trunk group. The <Txx> parameter specifies a 2-digit identifier in the format Txx where <b>xx</b> is the trunk ID number. Enter a trunk ID between <b>1</b> and <b>99</b> .
	(config-grouped-trunk-name)# <b>accept</b> <pattern> [ <b>cost</b> <value>]	Specifies the numbers allowed for routing on the trunk using outbound call templates. The <pattern> variable specifies a number pattern using complete phone numbers or wildcards. The <b>cost</b> <value> parameter is optional and specifies the cost value associated with the number pattern for the trunk. The call is routed to the trunk with the lowest cost. Valid range for <value> is <b>0</b> to <b>499</b> .
<b>Step 5</b>	Configure a trunk group for NetVanta ECS.	
	(config)# <b>voice grouped-trunk</b> <name>	Creates a trunk group and enters the Voice Trunk Group Configuration mode.
	(config-grouped-trunk-name)# <b>trunk</b> <Txx>	Adds a trunk to the trunk group for outbound call capability. The <Txx> parameter specifies a 2-digit identifier in the format Txx where <b>xx</b> is the trunk ID number. Enter a trunk ID between <b>1</b> and <b>99</b> .
	(config-grouped-trunk-name)# <b>accept</b> <pattern> [ <b>cost</b> <value>]	Specifies the numbers allowed for routing on the trunk using outbound call templates. The <pattern> variable specifies a number pattern using complete phone numbers or wildcards. The <b>cost</b> <value> parameter is optional and specifies the cost value associated with the number pattern for the trunk. The call is routed to the trunk with the lowest cost. Valid range for <value> is <b>0</b> to <b>499</b> .
<b>Step 6</b>	Enable the RTP symmetric filter.	
	(config)# <b>ip rtp symmetric-filter</b>	Enables filtering of received nonsymmetric RTP packets. When enabled, the RTP symmetric-filter drops packets destined to a particular port that are sourced from an IP address and port that was not specified in received SDP. By default, the RTP symmetric filter is disabled.

**Table 1. Remote Phone Command Summary (Continued)**


Step	Command	Description
<b>Step 7</b>	Enable media anchoring.	
	(config)# <b>ip rtp media-anchoring</b>	Enable media anchoring on a global basis for all RTP to RTP calls. By default, media anchoring is disabled.
<b>Step 8</b>	Configure double reINVITE preferences.	
	(config)# <b>[no] ip sip prefer double-reinvite</b>	Globally specifies SIP double reINVITEs are preferred for certain call flows. Using the <b>no</b> form of this command indicates that double reINVITEs are not preferred.

## Troubleshooting

After configuring SIP trunking on the ADTRAN SBC, several commands can be issued from the Enable mode in the CLI to assist in troubleshooting. The following section explains the **show** and **debug** commands that can be useful.

### Show Commands

The following **show** commands can be used to display and troubleshoot specific portions of the configuration.



**NOTE**

*The output of all **show** commands can be limited by appending the following modifiers to the end of the command: | **begin** <text>, | **exclude** <text>, and | **include** <text>. The **include** modifier limits output to lines that contain the specified text, the **exclude** modifier excludes any lines with the specified text, and the **begin** modifier displays the first line of output with the specified text and all lines thereafter.*

Use the **show ip rtp media-anchoring 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 time to live (TTL) for the association is displayed. To display media-anchoring information, enter the command as follows:

>enable

#show ip rtp media-anchoring sessions

CallID	Anchored Address	Remote Address	TTL	Packets	Ovrd	Type
7	10.10.10.1:40008	10.10.10.2:2230	45	108062	No	Audio
7	10.17.250.12:40010	10.17.250.14:10262	45	108063	No	Audio
7	10.10.10.1:40009	10.10.10.2:2231	44	432	No	Audio
7	10.17.250.12:40011	10.17.250.14:10263	44	432	No	Audio

## Debug Commands

There are several debug messages that can be enabled to assist in troubleshooting your SIP trunking gateways configuration. Debug messages are displayed in real time. You can activate multiple debug messages simultaneously.



*Turning on a large amount of debug information can adversely affect the performance of your unit.*

Use the **debug ip rtp media-anchoring** command to enable debug messaging for media anchoring activity. Debug messages are generated for media anchoring functions, such as the beginning and ending of anchoring sessions, and the creation and destruction of associations. To enable media anchoring debug messages, enter the command as follows:

```
>enable
#debug ip rtp media-anchoring
```

Use the **debug voice verbose** command to activate debug messages associated with voice functionality. To enable voice debug messages, enter the command as follows:

```
>enable
#debug voice verbose
```

Use the **debug sip cldu** command to activate debug messages associated with SIP call leg distribution unit (CLDU) events. To enable **sip cldu** debug messages, enter the command as follows:

```
>enable
#debug sip cldu
```

Use the **debug sip stack messages** command to activate debug messages associated with SIP messaging. Use the **no** form of this command to disable the debug messages. To enable **sip stack message** debug messages, enter the command as follows:

```
>enable
#debug sip stack message
```

## Additional Resources

There are additional resources available to aid in configuring your ADTRAN unit. Many of the topics discussed in this guide are complex and require additional understanding, such as using the CLI, SBC in AOS, configuring media anchoring, and implementing remote phones for AOS SIP Gateways. The documents listed in *Table 2* are available online at ADTRAN's Support Forum at <https://supportforums.adtran.com>.

**Table 2. Additional ADTRAN Documentation**

<b>Feature</b>	<b>Document Title</b>
All AOS Commands	<i><a href="#">AOS Command Reference Guide</a></i>
Media Anchoring	<i><a href="#">Configuring Media Anchoring in AOS</a></i>
Remote Phones	<i><a href="#">Remote Phone Configuration for AOS SIP Gateway</a></i>
SBC Product Overview	<i><a href="#">Session Border Controllers in AOS</a></i>