



## Configuration Guide

# Configuring a PRI Gateway for Use with NetVanta ECS

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This configuration guide outlines the steps necessary to configure an ADTRAN IP business gateway product as a Primary Rate Interface (PRI) gateway to use with the NetVanta Enterprise Communications System (ECS). The guide includes an overview of the PRI gateway, provides 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:

- *ISDN PRI Gateway Overview on page 2*
- *Hardware and Software Requirements and Limitations on page 2*
- *Configuring the PRI Gateway Using the CLI on page 2*
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## ISDN PRI Gateway Overview

PRI is an Integrated Digital Service Network (ISDN) circuit. On T1 circuits, it is composed of 23 bearer-channels (B-channels) and 1 data-channel (D-channel) and on E1 circuits, it is composed of 30 B-channels and 1 D-channel. ISDN PRI is an international standard for digital communications, allowing a full range of enhanced services supporting voice and data. The B-channels are used to transmit voice and data over an all-digital public switched telephone network. The D-channel is used to transmit out-of-band signaling for the B-channels that controls dialing numbers.

## ADTRAN IP Business Gateway Solution

The ADTRAN IP business gateway products are used in NetVanta ECS installations to provide interworking between internal Session Initiation Protocol (SIP) phone calls and the public switch telephone network (PSTN). The IP business gateway products interwork SIP phones on the local area network (LAN) and the traditional time division multiplexing (TDM) voice network. Supporting one or more T1/E1 spans, the IP business gateway products are an ideal gateway for these installations.

An IP business gateway works in conjunction with the SIP functionality that is part of the NetVanta Unified Communications (UC) Server. All telephony services are provided through the mutual cooperation of the SIP gateway, SIP telephones, and NetVanta ECS.

This configuration guide explains how to configure an ADTRAN IP business gateway product with a PRI trunk on a T1 port allowing it to connect to the NetVanta ECS as the private branch exchange (PBX). For additional details about the difference between T1 and E1 installations, refer to the *International Configuration Guide*.

## Hardware and Software Requirements and Limitations

This configuration guide is applicable to ADTRAN IP business gateway products that support user role ISDN PRI termination as outlined in the *Product Feature Matrix*, available online at ADTRAN's Support Forum, <https://supportforums.adtran.com>.

## Configuring the PRI Gateway Using the CLI

To configure the PRI gateway functionality on an AOS IP business gateway 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 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 Realtime Transport Protocol (RTP) symmetric filter.
7. Enable voice fax tones globally.
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 AOS unit 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 at the Global Configuration mode prompt, beginning with **voice transfer-mode local** command:

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

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

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.

If the voice trunk to the service provider is provided through a PRI, the T1 interface must be configured and a PRI interface created. To configure the T1 interface, set the primary timing source to the T1 interface before configuring the T1 interface.

Use the **timing-source** *<interface id>* command to set the timing source to the T1 from the provider. The following example sets the T1 0/3 interface as the primary timing source:

```
(config)#timing-source t1 0/3
```



*For information on how to properly configure timing on the NetVanta 6240 and NetVanta 644, refer to [Independent T1 Timing on the NetVanta 640 and NetVanta 6240](#).*

To configure the T1 interface to the provider, create a group of contiguous channels using the **tdm-group** *<number>* command where *<number>* identifies the group. Valid range for *<number>* is **1** to **255**. The parameter **timeslots** *<value>* specifies the channels to be used. Valid range for *<value>* is **1** to **31**. To create the PRI interface, enter the **interface pri** *<interface id>* command. The following example demonstrates these commands entered in sequence:

```
(config)#interface t1 0/3  
(config-t1 0/3)#tdm-group 1 timeslots 1-24  
(config-t1 0/3)#no shutdown
```

```
(config)#interface pri 1  
(config-pri 1)#connect t1 0/3 tdm-group 1  
(config-pri 1)#role user  
(config-pri 1)#no shutdown
```

The PRI must be connected to an ISDN group before it can be connected to a voice trunk. Use the **isdn-group** *<number>* command to create a new ISDN group where *<number>* uniquely identifies the ISDN group. Valid range for *<number>* is **1** to **255**. Associate it with the PRI interface using the **connect pri** *<interface id>* command.

The following example demonstrates these commands entered in sequence:

```
(config)#isdn-group 1  
(config-isdn-group 1)#connect pri 1  
(config-isdn-group 1)#exit
```

After configuring the ISDN group, configure a voice trunk to the service provider from the AOS unit and associate it with the ISDN group. It is also necessary, while in the Voice Trunk Configuration mode, to enable T.38 and modem passthrough.

Use the **voice trunk** *<Txx>* **type isdn** command to define a new ISDN trunk and activate the Voice Trunk Configuration mode for the individual trunk. Once in the Voice Trunk Configuration mode, you can associate the trunk with an ISDN group using the **connect isdn-group** *<number>* command.

The following example specifies that trunk **T01** will use ISDN group **1**:

```
(config)#voice trunk T01 type isdn  
(config-T01)#connect isdn-group 1
```

Use the **modem-passthrough** command to enable fax/modem tone detection:

```
(config-T01)#modem-passthrough
```

Use the **t38** command to enable T.38 fax operation:

```
(config-T01)#t38
```



*ADTRAN recommends enabling **t38 cng-relay-selective** (which was added in AOS R10.4.0) for NetVanta UC Server installations. Enabling this option causes the AOS unit to ignore incoming calling tones (CNG) packets from the NetVanta UC Server once it begins to receive data from the terminating fax machine. If this feature is not enabled, it could cause negotiation failures and terminate the V.21 messages in mid-transmission. (Enabling **t38 cng-relay-selective** is not applicable to the NetVanta 6240 or NetVanta 644.)*

Following the recommendation in the note above, use the **t38 cng-relay-selective** command to enable fax selective relay only when V.21 messages are not being transmitted. Return to the Global Configuration mode once the voice trunk has been configured using the **exit** command. The following example enables T.38 CNG selective relay and exits to the Global Configuration mode:

```
(config-T01)#t38 cng-relay-selective
(config-T01)#exit
(config)#
```

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

A second voice trunk is configured defining the connection from the AOS unit to the NetVanta ECS.

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 use the **description** command to provide a descriptive name for the trunk, and use the **sip-server primary <ipv4 address | hostname>** command to define the hostname or IPv4 address of the primary server to which the trunk will send call-related SIP messages. This SIP trunk also 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, beginning 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, a 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 ISDN 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 can be placed out that particular trunk. The *<Txx>* parameter specifies the 2-digit identifier in the format Txx where *xx* is the trunk ID number.

Use the **accept** *<template>* 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 *<template>* 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 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\$.

The **cost** *<value>* 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 example demonstrates these commands entered in sequence, beginning 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 SIP trunk is added to the trunk group using the **trunk** <Txx> command. The allowed outbound calls are defined using the **accept** <template> command and assigned a cost using the **cost** <value> 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 5](#).

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

```
(config)#voice grouped-trunk UC_SERVER
(config-UC_SERVER)#trunk T02
(config-UC_SERVER)#accept XXXX cost 0
(config-UC_SERVER)#accept 256-555-XXXX cost 0
```

## Step 6: Enable the RTP Symmetric Filter

The RTP symmetric filter blocks nonsymmetric RTP packets. It is enabled by default on certain AOS 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 Voice Fax Tones Globally

From the Global Configuration mode, globally allow T.30 calling tones to trigger a reINVITE to T.38. Use the **voice fax-tone t38 t30-cng** command as follows:

```
(config)#voice fax-tone t38 t30-cng
```

## 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 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 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 PRI 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 PRI Gateway Using the CLI on page 2* can also be performed using the GUI, with exception of the final two steps. This section explains how to access the GUI to perform the steps necessary to configure the PRI Trunking Gateway functionality using the GUI, with the final two steps accomplished using the CLI. It is written in a manner which allows you to complete the steps without referring to *Configuring the PRI 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 voice fax tones globally.
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:

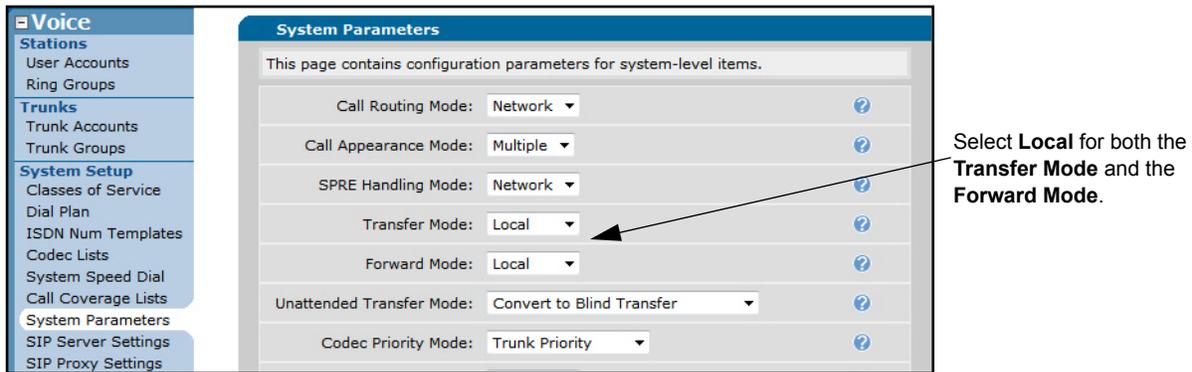
1. Open a new web page in your Internet browser.
2. Enter your AOS product's IP address in the browser's address field, **http://<ip address>**, for example:  
**http://10.10.10.1**
3. At the prompt, enter your user name and password and select **OK**.



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

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

Configure the AOS unit 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 menu for both **Transfer Mode** and **Forward Mode**. Select **Apply** at the bottom of the menu to accept the changes.



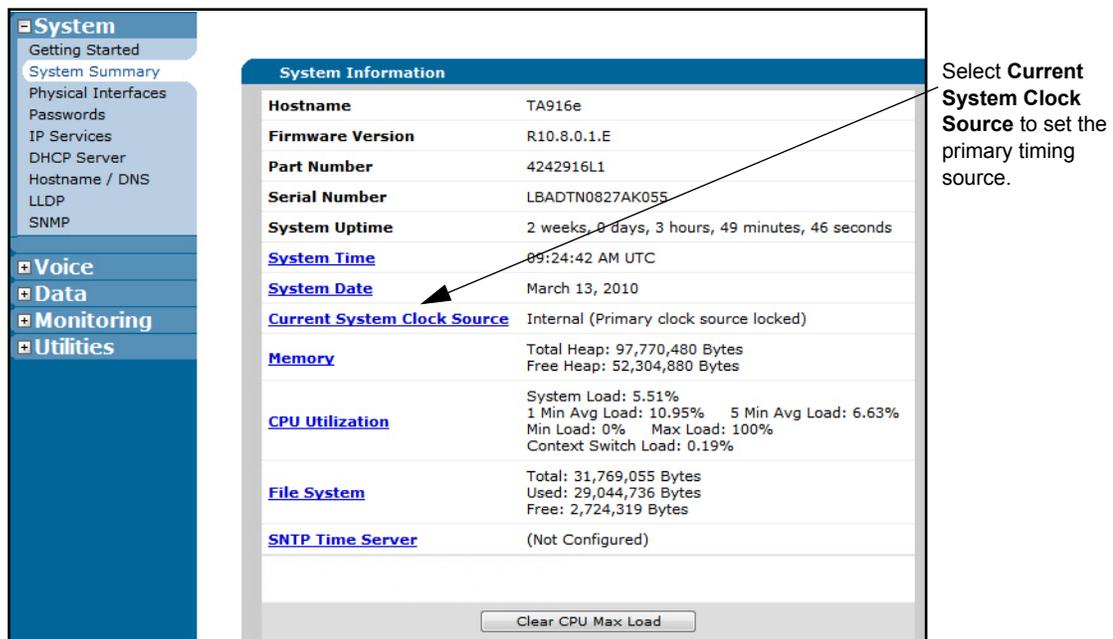
Select **Local** for both the **Transfer Mode** and the **Forward Mode**.

### 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 ISDN trunk to the service provider from the AOS unit. 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.

If the voice trunk to the service provider is provided through a PRI, the T1 interface must be configured and a PRI interface created. To configure the T1 interface, set the primary timing source to the T1 interface.

Navigate to **System > System Summary**. Select **Current System Clock Source**.



Select **Current System Clock Source** to set the primary timing source.

**NOTE** For information on how to properly configure timing on the NetVanta 6240 and NetVanta 644, refer to *Independent T1 Timing on the NetVanta 640 and NetVanta 6240*.

Select the T1 interface from the provider as the primary timing source from the drop-down menu (for example, **t1 0/3**). Select **Apply** to accept the changes.

The dialog box contains the following text and controls:

The Total Access should have a Primary Clock or Timing source set. A backup source can also be selected if more than one source exists, otherwise, Internal timing will be used as a backup.

Primary Clock Source: **t1 0/3** (dropdown menu)  
 Backup Clock Source: **t1 0/2** (dropdown menu)

Buttons: Cancel, Apply

Select the T1 interface from the provider from the drop-down menu.

Select **Apply** to accept the changes.

To create the PRI interface, navigate to **System > Physical Interfaces**. Select the T1 interface from the list (for example, **t1 0/3**).

The 'Physical Interfaces' page shows a table with the following columns: Name, Logical Interface, Line Status, and Type.

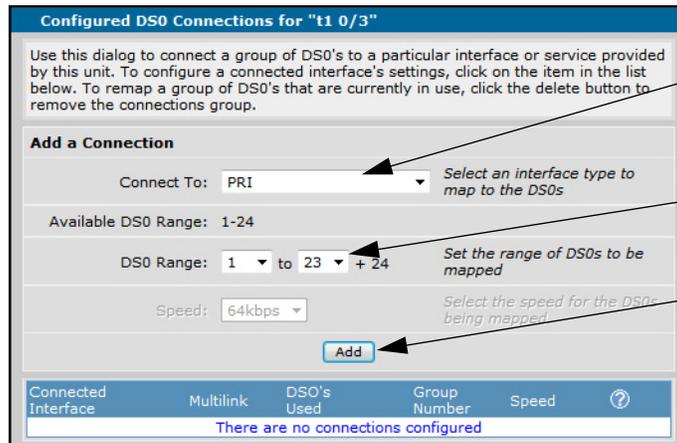
Name	Logical Interface	Line Status	Type
t1 0/1	none	TxYellow, Red, LOS	WAN-T1
t1 0/2	none	TxYellow, Red, LOS	WAN-T1
t1 0/3	none	TxYellow, Red, LOS	WAN-T1
t1 0/4	none	TxYellow, Red, LOS	WAN-T1
eth 0/1	none	Up	Ethernet
eth 0/2	none	Down	Ethernet
fxo 0/0	none	Down	FXO
fxs 0/1	none	OnHook	FXS
fxs 0/2	none	OnHook	FXS
fxs 0/3	none	OnHook	FXS
fxs 0/4	none	OnHook	FXS
fxs 0/5	none	OnHook	FXS
fxs 0/6	none	OnHook	FXS
fxs 0/7	none	OnHook	FXS
fxs 0/8	none	OnHook	FXS
fxs 0/9	none	OnHook	FXS
fxs 0/10	none	OnHook	FXS
fxs 0/11	none	OnHook	FXS
fxs 0/12	none	OnHook	FXS
fxs 0/13	none	OnHook	FXS
fxs 0/14	none	OnHook	FXS
fxs 0/15	none	OnHook	FXS
fxs 0/16	none	OnHook	FXS

Statistics Rate Interval: 300 (dropdown) Statistics Rate Interval (in seconds)

Apply button

Select the T1 interface to create a PRI interface.

To specify a PRI interface to map to the DSOs, select **PRI** from the **Connect To** drop-down menu in the **Configured DSO Connections** menu. Set the DSO range by selecting **1 to 23** for the **DSO Range**. Select **Add** to add the new PRI interface.

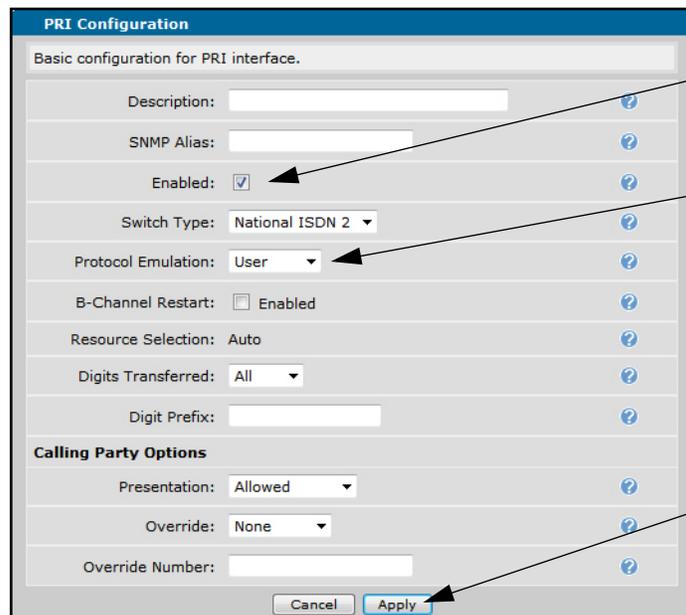


Select **PRI** from the drop down menu to connect to a PRI interface.

Select **23** from the drop down menu to set the DSO range.

Select **Add** to add the new PRI interface.

From the **PRI Configuration** menu, select the **Enabled** check box, and select **User** from the **Protocol Emulation** drop-down menu. To accept the changes, select **Apply**.

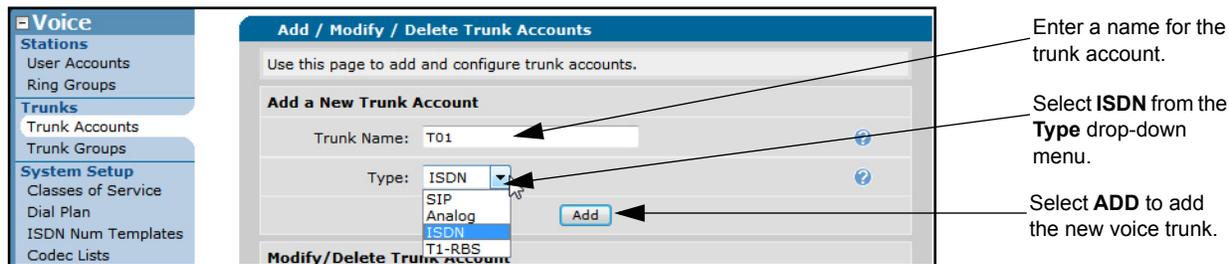


Select the check box to enable the PRI interface.

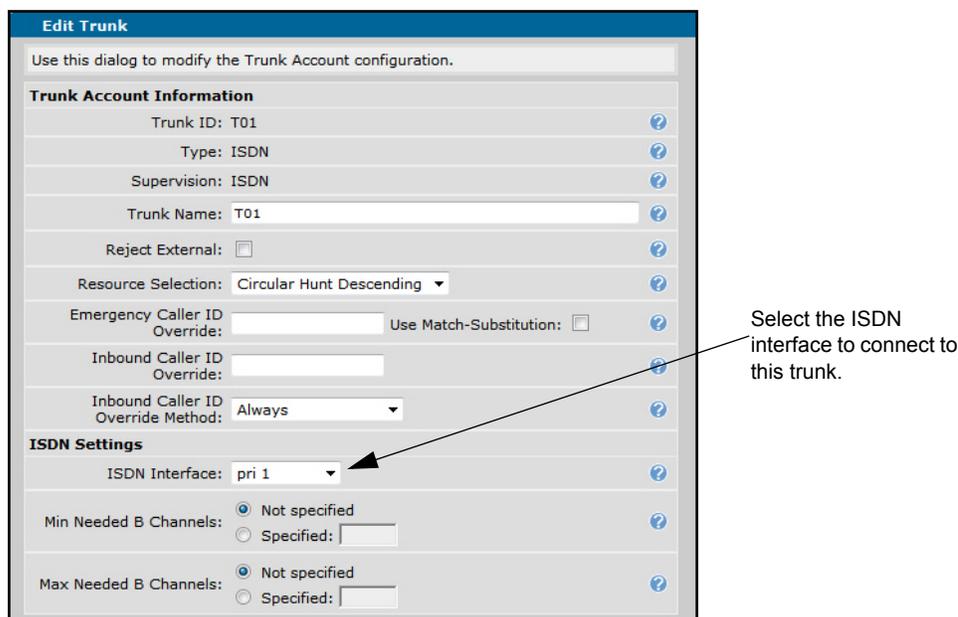
Select **User** for the Protocol Emulation.

Select **Apply** to accept the changes.

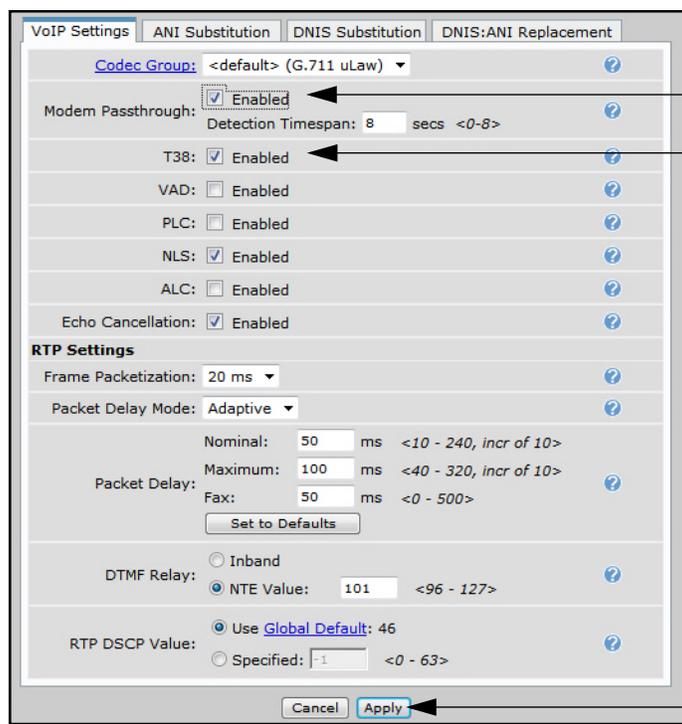
To create the voice trunk, navigate to **Voice > Trunks > Trunk Accounts**. Enter the **Trunk Name**, and select **ISDN** from the **Type** drop-down menu. Select **Add** to create the voice trunk.



Once the new trunk account is created, the **Edit Trunk** menu displays allowing you to further configure the voice trunk. Associate the PRI interface to this ISDN trunk for voice calls. Select **pri 1** from the **ISDN Interface** drop-down menu.



To enable T.38 and modem passthrough, locate the **VoIP Settings** tab at the bottom of the **Edit Trunk** menu. Select the **Enabled** check box next to **Modem Passthrough** and **T. 38**. Select **Apply** at the bottom of the **Edit Trunk** menu to accept the changes.



The screenshot shows the VoIP Settings configuration page. The 'Modem Passthrough' section has a checked 'Enabled' checkbox and a 'Detection Timespan' of 8 seconds. The 'T.38' section also has a checked 'Enabled' checkbox. Other settings include VAD, PLC, NLS, ALC, and Echo Cancellation, all with checkboxes. The 'RTP Settings' section includes 'Frame Packetization' (20 ms), 'Packet Delay Mode' (Adaptive), and 'Packet Delay' (Nominal: 50 ms, Maximum: 100 ms, Fax: 50 ms). The 'DTMF Relay' section has 'NTE Value' set to 101. The 'RTP DSCP Value' section has 'Global Default' selected with a value of 46. At the bottom, there are 'Cancel' and 'Apply' buttons. Arrows point from the text annotations to the 'Enabled' checkboxes and the 'Apply' button.

Select the **Enabled** check box next to **Modem Passthrough** and **T.38**.

Select **Apply** to accept the changes.



*ADTRAN recommends enabling **t38 cng-relay-selective** (which was added in AOS R10.4.0) for NetVanta UC Server installations. Enabling this option causes the AOS unit to ignore incoming calling tones (CNG) packets from the NetVanta UC Server once it begins to receive data from the terminating fax machine. If this feature is not enabled, it could cause negotiation failures and terminate the V.21 messages in mid-transmission. (Enabling **t38 cng-relay-selective** is not applicable to the NetVanta 6240 or NetVanta 644.)*

Following the recommendation in the note above, use the **t38 cng-relay-selective** command to enable fax CNG selective relay only when V.21 messages are not being transmitted. This function can only be completed using the CLI and requires accessing the CLI as explained in [Accessing the CLI on page 3](#).

Once the voice trunk has been configured as explained above, navigate to the Global Configuration mode in the CLI, and use the **voice trunk <Tx> type isdn** command to access the Voice Trunk Configuration mode for the individual trunk. The following example accesses the trunk **T01** Voice Trunk Configuration mode:

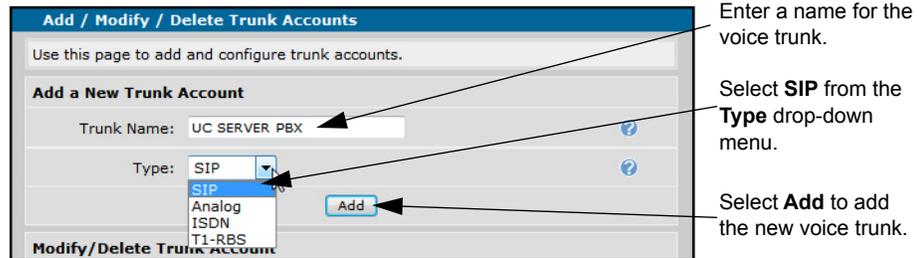
```
(config)#voice trunk T01 type isdn
(config-T01)#
```

Next, enable the T.38 CNG selective relay. The following example enables T.38 CNG selective relay and exits to the Global Configuration mode:

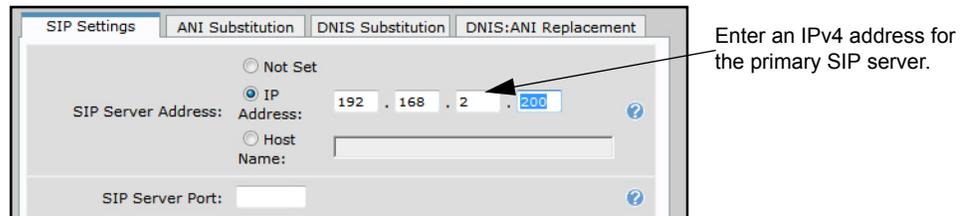
```
(config-T01)#t38 cng-relay-selective
(config-T01)#exit
(config)#
```

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

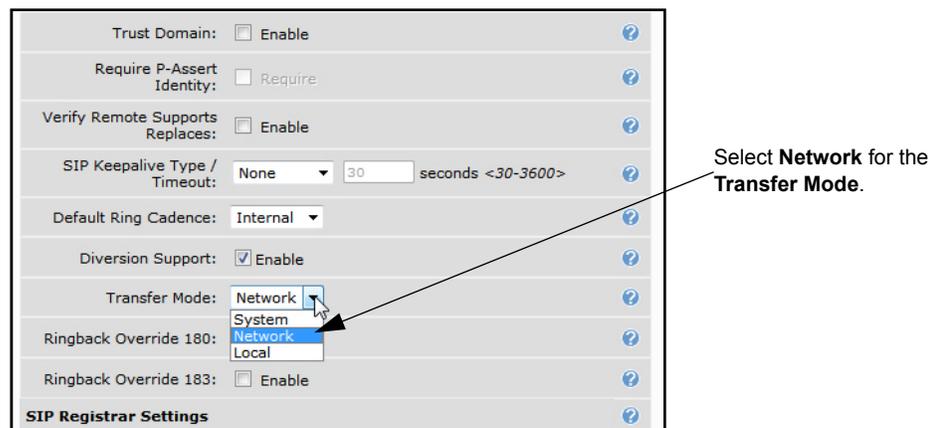
Configure a second voice trunk that defines the connection from the AOS unit to the NetVanta ECS. To create the voice trunk, navigate to **Voice > Trunks > Trunk Accounts**. Enter the **Trunk Name**, and select **SIP** from the **Type** drop down menu. Select **Add** to create the voice trunk.



The **Edit SIP Trunk** menu displays, allowing additional configuration of the SIP trunk. From the **SIP Settings** tab at the bottom of the **Edit SIP Trunk** menu, enter primary IPv4 address or hostname for the **SIP Server Address**.



In addition, this SIP trunk requires the NetVanta ECS to control call transfers. Scroll down the **SIP Settings** tab menu until you locate the **Transfer Mode** option. Select **Network**.

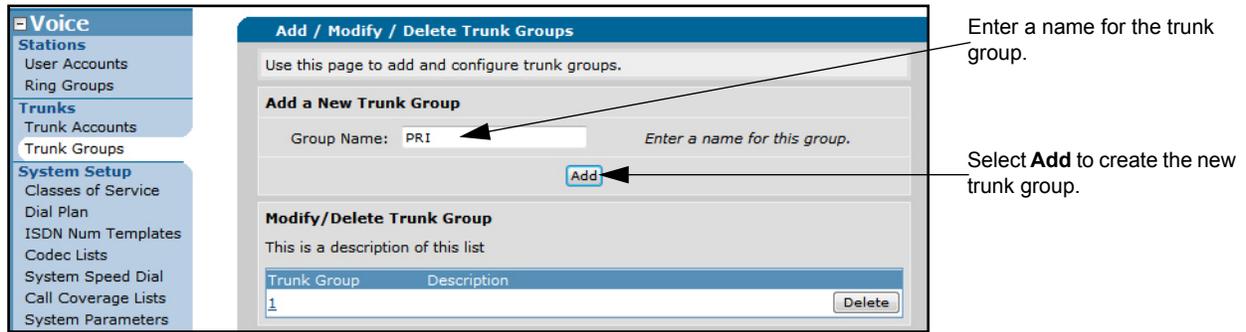


Select **Apply** at the bottom of the **Edit SIP Trunk** menu to accept the changes.

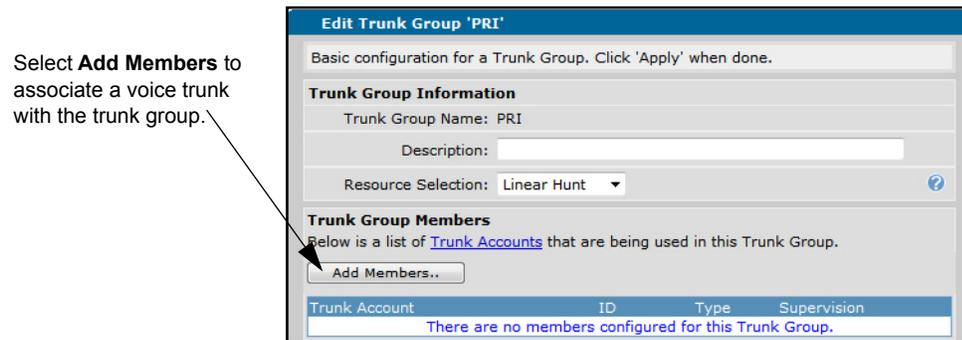
### Step 4: Configure a Trunk Group for the Service Provider

In this step, a 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 ISDN trunk is added to the trunk group. A cost is also assigned to each **accept** template in the trunk group.

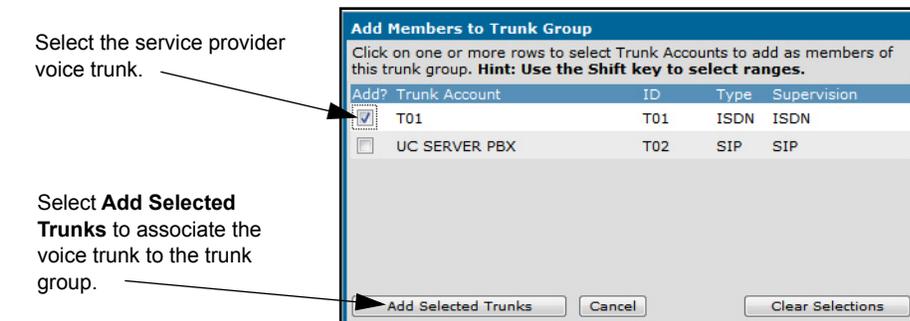
Navigate to **Voice > Trunks > Trunk Groups**. Enter a name for the trunk group and select **Add** as shown below.



The **Edit Trunk Group** menu displays for additional configuration settings. To add an existing trunk to the trunk group so outbound calls can be placed from that particular trunk, select **Add Members**.



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 below.



Next, specify call templates that are accepted for routing out the trunk from the **Outbound Call Templates** menu. Call templates control the type of outbound calls users can place on the system. Select from the existing call templates by checking the box next to the template. If the call templates provided do not cover the number patterns necessary for your configuration, select **Detailed View** to enter your own template. Refer to *Defining Advanced Call Templates (Optional) on page 17* for instructions for creating custom call templates.

Select the call templates to allow on this trunk.

Select **Detailed View** to enter your own template.

Select **Apply** to accept the changes.

Template	Cost	Pattern
<input checked="" type="checkbox"/> Local Calls (7 Digit)	Low Cost	(NXX-XXXX)
<input checked="" type="checkbox"/> Long Distance Calls	Low Cost	(1-NXX-NXX-XXXX)
<input checked="" type="checkbox"/> Toll-Free Calls	Low Cost	(1-800/855/866/877/888-NXX-XXXX)
<input checked="" type="checkbox"/> International Calls	Low Cost	(011-\$)
<input checked="" type="checkbox"/> n11 Calls (411, 611)	Low Cost	(411, 611)
<input checked="" type="checkbox"/> 911 Calls	Low Cost	(911)
<input checked="" type="checkbox"/> Operator-Assisted calls	Low Cost	(0-NXX-NXX-XXXX)
<input type="checkbox"/> Carrier Specified calls	Low Cost	(10-10-XXX-\$)
<input type="checkbox"/> 900 Calls	Low Cost	(1-900/976-NXX-XXXX 976-XXXX)
<b>Detailed View - Permit/Restriction Call Templates</b>		

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

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

To create a second trunk group for the NetVanta ECS (named **SIP**) and associate it with the trunk account T02, navigate to **Voice > Trunks > Trunk Groups**. Enter a name for the trunk group and select **Add**.

Enter a name for the trunk group.

Select **Add** to create the new trunk group.

The **Edit Trunk Group** menu displays for additional configuration settings. To add an existing trunk to the trunk group so outbound calls can be placed from that particular trunk, select **Add Members** from the **Edit Trunk Group** menu.

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 below.

Select the service provider voice trunk.

Select **Add Selected Trunks** to associate the voice trunk to the trunk group.

Add?	Trunk Account	ID	Type	Supervision
<input type="checkbox"/>	T01	T01	ISDN	ISDN
<input checked="" type="checkbox"/>	UC SERVER PBX	T02	SIP	SIP

Next, specify call templates that are accepted for routing out the trunk from the **Outbound Call Templates** menu. Call templates control the type of outbound calls users can place on the system. Select from the existing call templates by checking the box next to the template. If the call templates provided do not cover the number patterns necessary for your configuration, select **Detailed View** to enter your own template.

Specify the types of calls to allow on this trunk by checking the box next to the call template.

If the call templates provided do not cover the number patterns necessary for your configuration, select **Detailed View** to enter your own template. Refer to [Defining Advanced Call Templates \(Optional\)](#) on page 17 for more information

Select **Apply** to accept the changes.

Trunk Account	ID	Type	Supervision
UC_SERVER_PBX	T02	SIP	SIP

**Outbound Call Templates**

Check the appropriate boxes below to enable specific outbound call templates. **NOTE:** *Class of service* should be used to restrict the types of calls individual users can make (ie: 900 numbers, etc).

<input type="checkbox"/>	Local Calls (7 Digit)	Low Cost	(NXX-XXXX)
<input type="checkbox"/>	Long Distance Calls	Low Cost	(1-NXX-NXX-XXXX)
<input type="checkbox"/>	Toll-Free Calls	Low Cost	(1-800/855/866/877/888-NXX-XXXX)
<input type="checkbox"/>	International Calls	Low Cost	(011-\$)
<input type="checkbox"/>	n11 Calls (411, 611)	Low Cost	(411, 611)
<input type="checkbox"/>	911 Calls	Low Cost	(911)
<input type="checkbox"/>	Operator-Assisted calls	Low Cost	(0-NXX-NXX-XXXX)
<input type="checkbox"/>	Carrier Specified calls	Low Cost	(10-10-XXX-)\$
<input type="checkbox"/>	900 Calls	Low Cost	(1-900/976-NXX-XXXX 976-XXXX)
<input checked="" type="checkbox"/>	<b>Detailed View - Permit/Restriction Call Templates</b>		

Permit Template Cost  
There are no configured Permit Templates

Restriction Template  
There are no configured Restriction Templates

Configure Advanced Templates

Cancel Apply

### Defining Advanced Call Templates (Optional)

If the provided call templates 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 below.

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

**Detailed View - Permit/Restriction Call Templates**

Permit Template Cost  
There are no configured Permit Templates

Restriction Template  
There are no configured Restriction Templates

Configure Advanced Templates

Cancel Apply

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

The available wildcards for this command are:

- 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

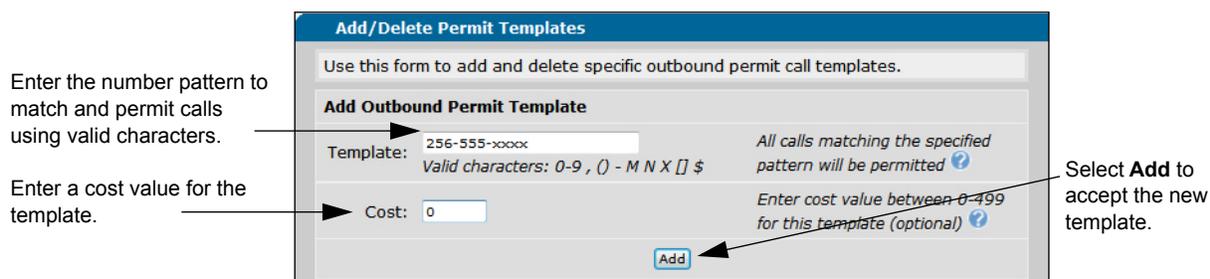
The following are example template entries using wildcards:

- 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\$.

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



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 enabling the RTP symmetric filter. The RTP symmetric filter blocks nonsymmetric RTP packets. It is enabled by default on certain AOS 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 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.

The screenshot shows the 'VoIP Settings' configuration page. On the left is a navigation menu with categories: 'Voice Stations', 'Trunks', 'System Setup', and 'Reports'. The 'VoIP Settings' option is selected. The main content area has three tabs: 'SIP Settings', 'RTP Settings', and 'SDP Settings'. The 'RTP Settings' tab is active, showing sections for 'RTP QoS Settings', 'RTP Port Range', and 'RTP Firewall Traversal'. In the 'RTP Port Range' section, the 'RTP Symmetric Filter' checkbox is checked and labeled 'Enabled'. At the bottom of the page, the 'Apply' button is highlighted with an arrow. Three callout boxes with arrows point to specific elements: one to the 'RTP Settings' tab, one to the 'RTP Symmetric Filter' checkbox, and one to the 'Apply' button.

## Step 7: Enable Voice Fax Tones Globally Through the CLI

This step cannot be completed through the GUI and therefore requires you to access the CLI.

From the Global Configuration mode, globally allow T.30 calling tones to trigger a reINVITE to T.38. Use the **voice fax-tone t38 t30-cng** command as follows:

```
(config)#voice fax-tone t38 t30-cng
```

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

This step cannot be completed through the GUI and therefore requires you to access 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 transfers. When these calls connect, a double reINVITE message is initiated.

By default, the system is configured so that double reINVITES are preferred. 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.

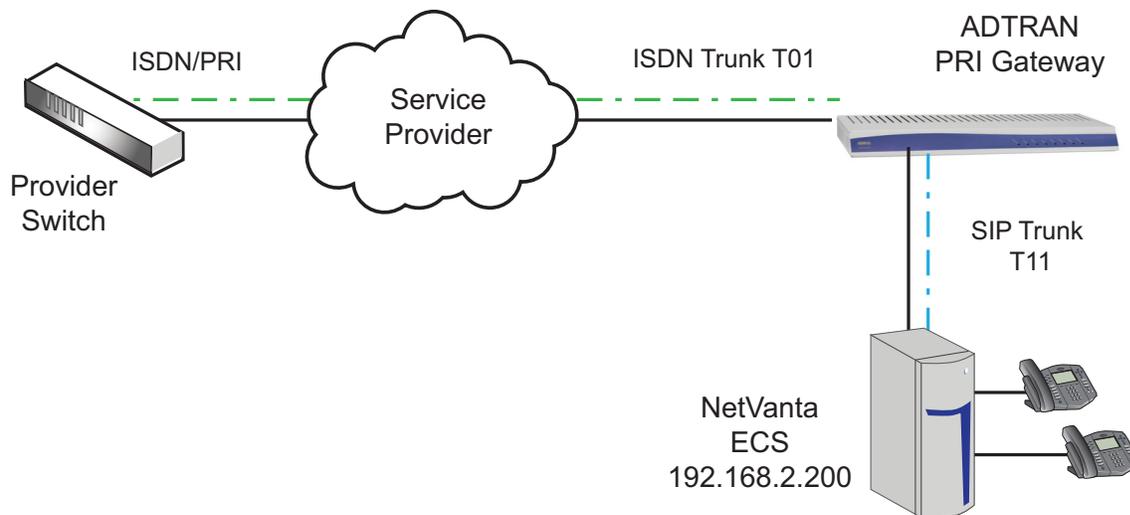
To specify that SIP double reINVITES are not preferred in the system, enter the command as follows from the Global Configuration mode:

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

## Configuration Example

The following example demonstrates a typical installation of an ADTRAN IP business gateway configured as the PRI gateway between a NetVanta ECS and a service provider. The configuration in this example occurs on the ADTRAN IP business gateway shown in *Figure 1 on page 20*.

Two voice trunks are configured, T01 and T11. T01 is configured as the ISDN PRI trunk to the service provider. 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 1. PRI Gateway with NetVanta ECS**



*The configuration parameters entered in this example are sample configurations only, and only pertain to the configuration of the PRI 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.*

```
!
configure terminal
!
timing-source t1 0/3
!
interface t1 0/3
 tdm-group 1 timeslots 1-24
 no shutdown
!
interface pri 1
 connect t1 0/3 tdm-group 1
 role user
 no shutdown
!
voice transfer-mode local
voice forward-mode local
!
```

```
isdn-group 1
  connect pri 1
!
voice trunk T01 type isdn
  connect isdn-group 1
  t38
  modem-passthrough
  match ani "<XXXX>" substitute "<256-555-6666>"
  t38 cng-relay-selective
!
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 T02
  accept XXXX cost 0
  accept 256-555-XXXX cost 0
!
ip rtp symmetric-filter
!
voice fax-tone t38 t30-cng
!
no ip sip prefer double-reinvite
!
end
write
```

## Command Summary

The following table summarizes the commands used to configure PRI gateway in AOS products, using the CLI. This table is provided for reference for configurations using the CLI only and may not be presented in the same order as the GUI steps

**Table 1. AOS CLI 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>timing-source T1</b> <interface id>	Sets the T1 interface from the provider as the primary timing source.
	(config)# <b>interface</b> <Txx> <interface id> (config-t1 0/3)# <b>tdm-group</b> <number> <b>timeslots</b> <value> (config-t1 0/3)# <b>no shutdown</b>	Configures the T1 interface to the provider. Creates a group of contiguous channels using the <b>tdm-group</b> <number> command where <number> identifies the group. Valid range is <b>1</b> to <b>255</b> . The parameter <b>timeslots</b> <value> specifies the channels to be used. Valid range is <b>1</b> to <b>24</b> on a <b>T1</b> , and <b>1</b> to <b>31</b> on an E1.
	(config)# <b>interface pri</b> <id> (config-pri 1)# <b>connect t1</b> <slot/port> <b>tdm-group</b> <number> (config-pri 1)# <b>role user</b> (config-pri 1)# <b>no shutdown</b>	Creates a PRI interface and maps it to a TDM group of the T1. The valid range for <b>pri</b> <id> is <b>1</b> through <b>255</b> . To configure the TDM group number, the valid range for <number> is <b>1</b> to <b>255</b> . The final two commands set the role to <b>user</b> and enables the interface.
	(config)# <b>isdn-group</b> <number> (config-isdn-group 1)# <b>connect pri</b> <id>	Creates an ISDN group with a unique ID number. Valid range for <b>isdn-group</b> <number> is <b>1</b> to <b>255</b> . Associates the ISDN group with an already configured PRI interface. The valid range for <b>pri</b> <id> is <b>1</b> through <b>255</b> .
	(config-isdn-group 1)# <b>exit</b>	Exits the ISDN Group Configuration mode and returns to the Global Configuration mode.
	(config)# <b>voice trunk</b> <Txx> <b>type isdn</b>	Creates an ISDN trunk and enters the Voice Trunk Configuration mode. 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> .

Table 1. AOS CLI Command Summary (*Continued*)

Step	Command	Description
<b>Step 2 Cont'd</b>	(config-Txx)# <b>connect isdn-group</b> <number>	Associates the ISDN trunk with an ISDN group. Valid range for <b>isdn-group</b> <number> is <b>1</b> to <b>255</b> .
	(config-Txx)# <b>t38</b>	Enables T.38 fax operation on this trunk.
	(config-Txx)# <b>modem-passthrough</b>	Enables fax and modem tone detection for this trunk.
	(config-Txx)# <b>t38 cng-relay-selective</b>	Enables fax CNG relay only when V.21 messages are not being transmitted.
	(config-Txx)# <b>exit</b>	Exits the Voice Trunk Configuration mode and returns to the Global Configuration mode.
<b>Step 3</b>	Configure the voice trunk to NetVanta ECS.	
	(config)# <b>voice trunk</b> <Txx> <b>type sip</b>	Creates a SIP trunk and enters the Voice Trunk Configuration mode. The <Txx> parameter specifies a 2-digit identifier in the format Txx where <b>xx</b> is the trunk ID number.
	(config-Txx)# <b>description</b> <text> (config-Txx)# <b>sip-server primary</b> <value>	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 <value> 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.

Table 1. AOS CLI 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 previously configured in Step 2.
	(config-grouped-trunk-name)# <b>accept</b> <template> [ <b>cost</b> <value>]	Specifies the numbers allowed for routing on the trunk using outbound call templates. The <template> 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 to 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 the trunk ID previously configured in Step 3.
	(config-grouped-trunk-name)# <b>accept</b> <template> [ <b>cost</b> <value>]	Specifies the numbers allowed for routing on the trunk using outbound call templates. The <template> 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 to 499</b> .

Table 1. AOS CLI Command Summary (*Continued*)

Step	Command	Description
<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 on certain products, and enabled on others.
<b>Step 7</b>	Enable voice fax tones globally.	
	(config)# <b>voice fax-tone t38 t30-cng</b>	Allows T.30 calling tones to trigger a reINVITE to T.38 globally on the AOS unit.
<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 the PRI gateway on the ADTRAN IP business gateway, several commands can be issued from the Enable mode in the CLI to assist in troubleshooting. The following section explains the **debug** commands that can be useful.

### 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 isdn I2-formatted** command to enable debug messaging for ISDN Layer 2 formatted messages. To enable ISDN Layer 2 formatted debug messages, enter the command as follows:

```
>enable
#debug isdn I2-formatted
```

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 messages** debug messages, enter the command as follows:

```
>enable
#debug sip stack messages
```

## Additional Resources

There are additional resources available to aid in configuring your AOS unit. Many of the topics discussed in this guide are complex and require additional understanding. The documents listed below are available online at ADTRAN's Support Forum at <https://supportforums.adtran.com>.

- *AOS Command Reference Guide*
- *Configuring IP Interfaces for SIP in AOS IPBGs*
- *Configuring the Switchboard and Dial Plan in AOS*
- *Enhanced ANI and DNIS Substitution in AOS Voice Products*
- *Independent T1 Timing on the NetVanta 640 and NetVanta 6240*
- *International Configuration Guide*