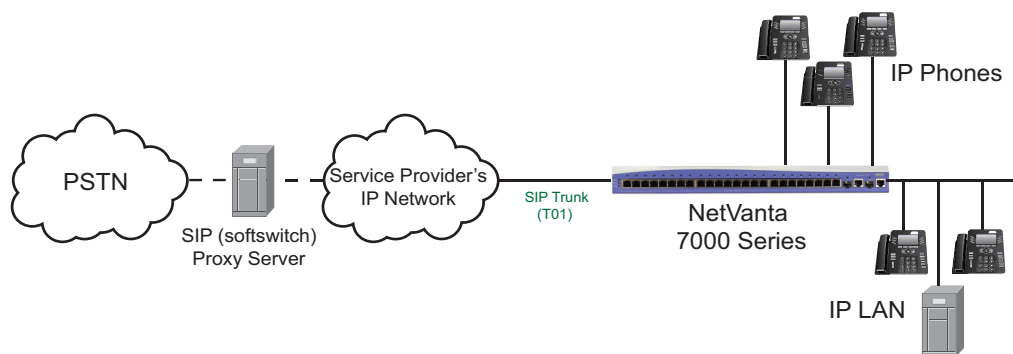


Introduction

Session Initiation Protocol (SIP) trunking is a packet-based voice service that routes calls over an IP network to an IP-compatible private branch exchange (PBX) or voice switch using SIP signaling to place and receive calls. ADTRAN's NetVanta 7000 Series IP PBXs support SIP trunks delivered from your service provider. The following sections in this document explain application and configuration implementation of SIP trunking using the ADTRAN NetVanta 7000 Series IP PBXs.



ADTRAN Operating System (AOS) firmware version A2.02.00 or later is required on your NetVanta 7000 Series product in order to support SIP trunking.

The following ADTRAN documents are prerequisites to configuring a new SIP trunk on your system:

- *NetVanta 7000 Series Quick Start Guide*
- *NetVanta 7000 Series Web GUI Quick Configuration Guide*
- *NetVanta 7000 Administrator's Guide*
- *NetVanta 7000 Series SIP Trunking*

The following ADTRAN configuration guides are also available to assist with implementing configuration and application relating to SIP trunks and SIP networks:

- *Source and ANI Based Routing*
- *Switchboard and Dial Plan*
- *Transparent Proxy*
- *Voice Traffic over SIP Trunks*
- *Voice Quality Monitoring*
- *User Accounts (additional information available in the NetVanta 7000 Series Admin Guide)*
- *Voicemail (additional information available in the NetVanta 7000 Series Admin Guide)*

Configuration guides are located on the *AOS Documentation* CD shipped with your AOS unit or on the website at <http://kb.adtran.com>.

Follow the steps below to configure the incoming SIP trunk:

- Create a SIP trunk account.
 - Set the SIP server address.
 - Set the SIP registrar address.
- Create trunk group(s).

Configuring the SIP Trunk

1. Open a Web-based graphical user interface (GUI) session. If you need assistance, follow the steps in the *NetVanta 7000 Series Web GUI Quick Configuration Guide* available on the *AOS Documentation* CD shipped with your unit.
2. Navigate to the **Voice > System Setup > System Parameters** menu. Verify that the **International Prefix Abbreviated** check box is selected to support E.164 dialing.

Codecs Priority Mode:	Trunk Priority	?
Number of Rings:	4 <0 - 9, 0 is unlimited>	?
Interdigit Timeout:	4 seconds <1 - 16>	?
Flashhook Mode:	Interpreted	?
Flashhook Range:	300 - 1000 ms <300 - 1000>	?
Hold Reminder Timeout:	30 seconds	?
Park Return:	60 seconds	?
Connected Timeout:	12 hours <0 - 1000, 0 is unlimited>	?
Alerting Timeout:	5 minutes <0 - 60, 0 is unlimited>	?
Country Code:	1	?
International Prefix:	011 <input checked="" type="checkbox"/> Abbreviated	?
Prompt Language:	English	?
Companding Type:	U-Law	?

Verify that the **International Prefix Abbreviated** check box is selected.

3. Navigate to the **Voice > Trunks > Trunk Accounts** menu. Enter the desired name for the SIP trunk in the **Trunk Name** field and select **SIP** as the trunk **Type**. Select **Add** to create the new trunk account.

Enter a name for the **SIP** trunk.

Select **SIP** as the **Type** of trunk.

Add / Modify / Delete Trunk Accounts

Use this page to add and configure trunk accounts.

Add a New Trunk Account

Trunk Name: ?

Type: ?

Modify/Delete Trunk Account

Click on a name to edit that trunk's settings.

Trunk Name	ID	Type	Supervision	Role	
ISDN PRI Trunk	T02	ISDN	ISDN	User	<input type="button" value="Delete"/>
T1-RBS Trunk	T01	RBS	Wink	User	<input type="button" value="Delete"/>

- Continue the SIP trunk configuration on the **SIP Settings** tab. Set the SIP server and SIP proxy addresses using the information provided by your service provider.

Configure the **SIP Server Address** and **SIP Proxy Address**.

The screenshot shows the 'SIP Settings' configuration page. The 'SIP Settings' tab is selected. The page contains the following fields and options:

- SIP Server Address:** Radio buttons for 'Not Set', 'IP Address' (with four input boxes), and 'Host Name' (with one input box).
- SIP Server Port:** Input field.
- SIP Proxy Address:** Radio buttons for 'Not Set', 'IP Address' (with four input boxes), and 'Host Name' (with one input box).
- SIP Proxy Port:** Input field.
- SIP Conferencing URI:** Input field.
- Force Host Resolve:** Checkboxes for 'Override' and 'Enable'.
- FROM Header User Formatting:** Checkboxes for 'Override' and a dropdown menu set to 'Domestic'.
- FROM Header Host Type:** Checkboxes for 'Override' and a dropdown menu set to 'SIP Server'.
- TO Header Host Type:** Checkboxes for 'Override' and a dropdown menu set to 'SIP Server'.
- P-Asserted Identity Host Type:** Checkboxes for 'Override' and a dropdown menu set to 'SIP Server'.
- Request URI Header Host Type:** Checkboxes for 'Override' and a dropdown menu set to 'SIP Server'.
- Alert Info URL:** Checkboxes for 'Override', 'Default', and 'Custom' (with one input box).
- Supports 100rel:** Checkboxes for 'Override' and 'Enable' (checked).
- Require 100rel:** Checkboxes for 'Override' and 'Enable'.
- Dial String Source:** Dropdown menu set to 'Request URI'.
- Trust Domain:** Checkboxes for 'Enable'.

5. Still on the **SIP Settings** tab, scroll down and configure the **SIP Registrar Settings** using the information obtained from your service provider.

Configure the SIP Registrar Address.

The screenshot shows the following configuration options:

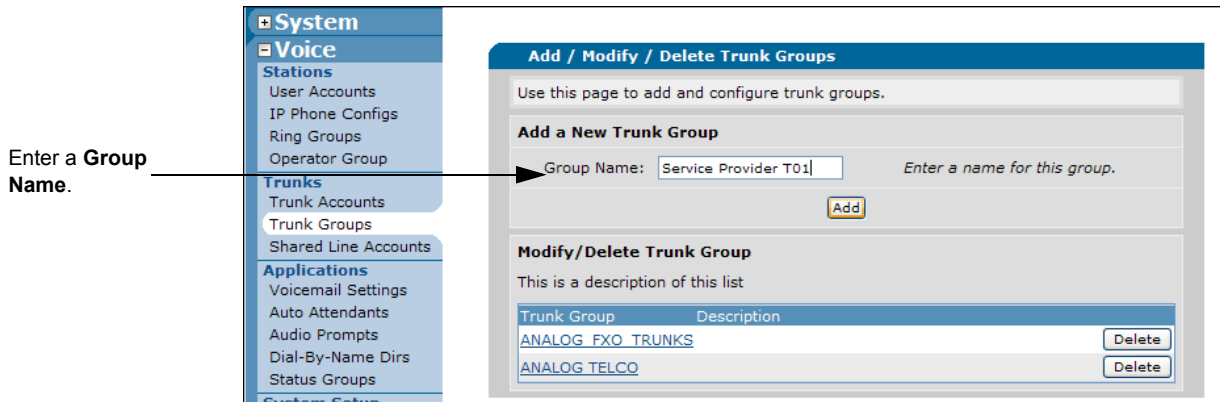
- SIP Registrar Address:** Not Set, IP Address: [] [] [] [] [], Host Name: []
- SIP Registrar Port:** []
- Requires Expires:** Enable
- Registration Expire Time:** Server Default, Request an Expire Time: [3600] seconds
- Max Concurrent Registrations:** [32] <0-32>
- Registrar Threshold:** Absolute: <30 secs - 7 days>, [0] days [0] hours [5] min. [0] sec., Percentage: [] % <0 - 90%>
- Default Authentication:** Not Set, Set, User: [], Password: []
- Domain Address:** Server Default, Use this domain: []
- Codec Group:** <default> (G.711 uLaw)
- Registration Settings:** Table with columns: Register value, End (if range), Authname. Content: There are no Register entries for this Trunk.
- Buttons:** Add Register Entry, Cancel, Apply

Select **Apply** to append the settings to create the new SIP identity number(s) registration.

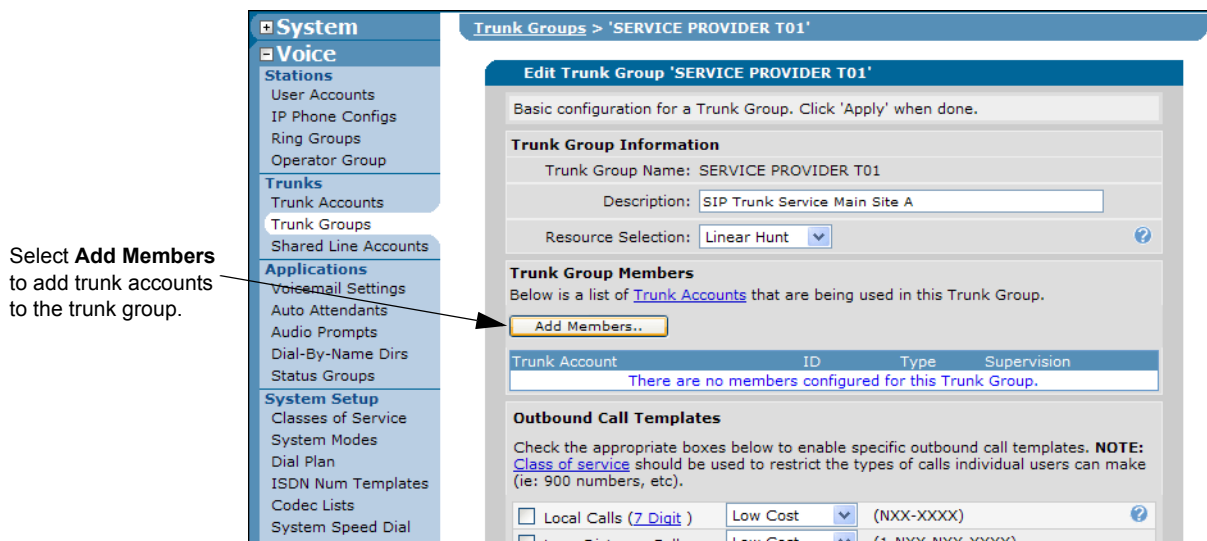
Configuring the Trunk Group

Trunk groups combine one or more trunk accounts and assign outbound call characteristics. The trunk group is assigned outbound call capabilities (local calls, long distance calls, etc.). Additionally, a cost is assigned to each attribute in the outbound call template. Use this section to create the trunk group, add the trunk account members to the group, and define the outbound call templates and costs.

1. Navigate to the **Voice > Trunks > Trunk Groups** menu, enter a new **Group Name** and select **Add**. To edit an existing trunk, select the link for the desired trunk from the list under **Modify/Delete Trunk Group**.



2. To add members to the trunk group, select the **Add Members** button. The **Add Members to Trunk Group** menu will appear.



3. Add members by selecting the name you entered for the SIP trunk.

Add?	Trunk Account	ID	Type	Supervision
<input checked="" type="checkbox"/>	Service Provider	T01	SIP	SIP
<input type="checkbox"/>	test	T02	SIP	SIP
<input type="checkbox"/>	T1 RBS Trunk	T05	RBS	Ground Start
<input type="checkbox"/>	Feature Grp D	T10	RBS	Feature Group D

Buttons: Add Selected Trunks, Cancel, Clear Selections

4. Select **Add Selected Trunks** to append the new member selection(s) and return to the **Edit Trunk Group** menu.
5. Select the appropriate check boxes under **Outbound Call Templates** to enable specific outbound call templates. Outbound call templates are the types of calls to allow from this trunk.

Select the boxes to enable specific outbound call templates for this trunk group. Select the cost for each template.

Edit Trunk Group 'SERVICE PROVIDER T01'

Basic configuration for a Trunk Group. Click 'Apply' when done.

Trunk Group Information

Trunk Group Name: SERVICE PROVIDER T01

Description:

Resource Selection:

Trunk Group Members

Below is a list of [Trunk Accounts](#) that are being used in this Trunk Group.

Trunk Account	ID	Type	Supervision
There are no members configured for this Trunk Group.			

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 checked="" type="checkbox"/>	Local Calls (7 Digit)	Medium 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 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 checked="" type="checkbox"/>	Carrier Specified calls	High Cost	(10-10-XXX-)
<input type="checkbox"/>	900 Calls	Low Cost	(1-900/976-NXX-XXXX 976-XXXX)

Detailed View - Permit/Restriction Call Templates

Permit Template	Cost
There are no configured Permit Templates	

Restriction Template
There are no configured Restriction Templates

Buttons: Cancel, Apply

6. Select **Apply** at the bottom of the menu to append the new settings and return to the **Add/Modify/Delete Trunk Groups** menu.

7. Verify the addition of the new trunk group.

Verify the newly added trunk group. Look for the name you entered for the SIP trunk in this area.

Use this page to add and configure trunk groups.

Add a New Trunk Group

Group Name: *Enter a name for this group.*

Modify/Delete Trunk Group

This is a description of this list

Trunk Group	Description	
ANALOG FXO TRUNKS		<input type="button" value="Delete"/>
ANALOG TELCO		<input type="button" value="Delete"/>
SERVICE PROVIDER		<input type="button" value="Delete"/>
T01		
LOCAL	Local calls for Main Site	<input type="button" value="Delete"/>

Special Configuration Notes

This section contains Bandwidth.com-specific SIP trunk configurations when interfacing with ADTRAN's NetVanta 7000 Series products.

E.164 Dialing Configuration

To support E.164 dialing, navigate to **Voice > VoIP Settings** and change the **FROM Header User Formatting** option to **International**.

The screenshot shows the 'VoIP Settings' page in a web interface. The left sidebar contains a navigation menu with categories: System, Voice, Trunks, Applications, System Setup, Reports, Data, Monitoring, and Utilities. The 'Voice' section is expanded, showing 'VoIP Settings' as the selected option. The main content area is titled 'VoIP Settings' and includes a sub-header 'SIP Configuration Parameters'. The 'FROM Header User Formatting' dropdown menu is set to 'International'. The 'Apply' button is highlighted. Annotations with arrows point to the 'International' dropdown and the 'Apply' button.

Select **International** for the **FROM Header User Formatting**.

Select **Apply** to save the settings.