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Interoperability Guide

Integrating the NetVanta 7100 with a BroadVox GO! SIP Trunk

This interoperability guide provides instructions for integrating the NetVanta 7100 with a BroadVox GO! SIP trunk. This guide provides an overview and instructions for the integration as well as a list of equipment used for testing the integration, the features supported by the integration, and the verified functionality of the integration.

This guide consists of the following sections:

Solution Overview

The NetVanta 7100 Communication System provides small and medium-sized businesses with a complete voice and data networking solution. The NetVanta 7100 includes an IP private branch exchange (PBX), voicemail, multilevel auto attendant, full-featured IP router, firewall, Virtual Private Network (VPN), 24-port Power over Ethernet (PoE) (802.3af) Fast Ethernet switch with Gigabit uplinks, 2 foreign exchange service (FXS) ports, 2 foreign exchange office (FXO) ports, music on hold, and two expansion slots for network interface modules (NIMs) and voice interface modules (VIMs).

BroadVox is a SIP trunk service provider for VoIP communication to small to medium sized businesses. The solution in this interoperability guide provides SIP trunking between the NetVanta 7100 and BroadVox.

The configuration used in this solution is a typical SIP trunking application. All SIP phones are registered and controlled by the NetVanta 7100, which has a SIP trunk registered to the BroadVox SIP Server. Incoming PSTN calls are routed to the NetVanta 7100 from BroadVox, and outgoing calls are sent from the NetVanta 7100 to BroadVox and out to the PSTN.

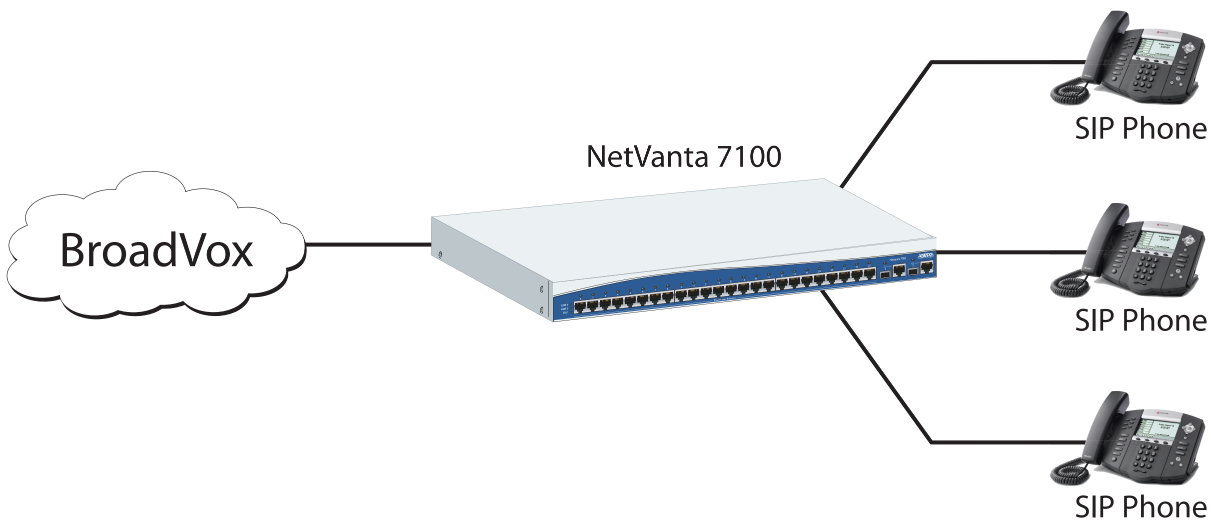


Figure 1. Network Topology Used for Interoperability Verification

Hardware and Software Requirements and Limitations

The following table outlines the equipment and software/firmware versions used during verification testing:

Table 1. Hardware Components Tested

Provider	Hardware Component	Version
ADTRAN	NetVanta 7100	R10.8.1.E

Supported Features and Exceptions

The NetVanta 7100 successfully passed the BroadVox Interoperability test for the BroadVox GO! SIP trunking. The following sections provide information on the feature verification performed and issues discovered during interoperability verification. The features listed in the *Supported Features* section below are the features verified to work with the testing equipment. These are the only features you can expect to function with the configuration provided in this guide. For a full summary of the interoperability testing performed, refer to *Verification Testing Summary on page 17*

Supported Features

The focus of the interoperability verification for this solution was SIP trunking. The following features were tested and are supported by the integration:

- NetVanta 7100 registration with BroadVox SIP server
- Inbound and outbound calling using G7.11 u-law and G.729a
- Inbound and outbound faxing using G7.11 u-law and G.729a
- RFC 2833 dual-tone multi-frequency (DTMF) operation
- Call hold and resume
- Call transfers
- Direct inward dialing (DID)
- Calling party number presentation
- Network Address Translation (NAT)

Exceptions

The following issues were discovered during interoperability verification:

- Calls using Transport Control Protocol (TCP), Transport Layer Security (TLS), and Secure Real-time Transport Protocol (SRTP) are not supported.
- Outbound T.38 faxing using G.711 u-law is not supported.

Configuring the NetVanta 7100 for BroadVox Using the GUI

To configure the NetVanta 7100 for a BroadVox SIP trunk using the GUI, follow these steps:

- *Step 1: Access the GUI on page 3*
- *Step 2: Create a SIP Trunk Account to the BroadVox SIP Server on page 4*
- *Step 3: Configure a Trunk Group for the BroadVox SIP Trunk on page 8*
- *Step 4: Configure 10-digit Dialing on page 10*
- *Step 5: Save the NetVanta 7100 Configuration on page 10*

Step 1: Access the GUI

To access the GUI on the NetVanta 7100, follow these steps:

1. Open a new page in your Internet browser.
2. Enter your unit's IP address in the browser's address field in the following form:

   http://<*ip address*>**/admin**, for example:

http://10.10.10.1/admin

3. At the prompt, enter your user name and password and select **OK**.

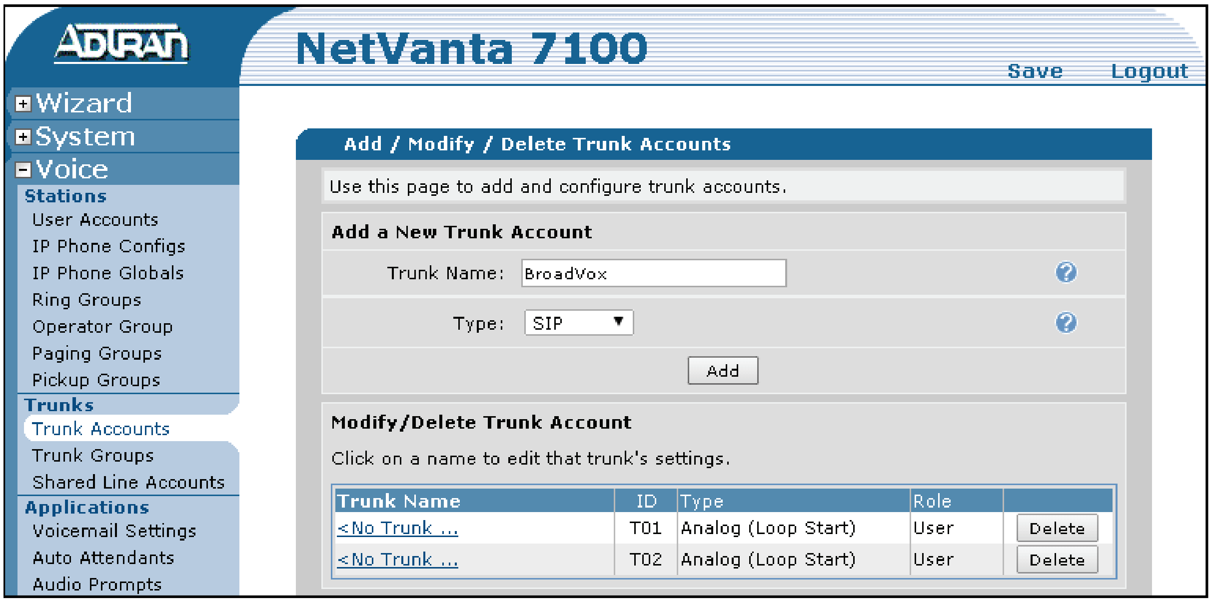
NOTE *The default user name is **admin** and the default password is **password**.*

Step 2: Create a SIP Trunk Account to the BroadVox SIP Server

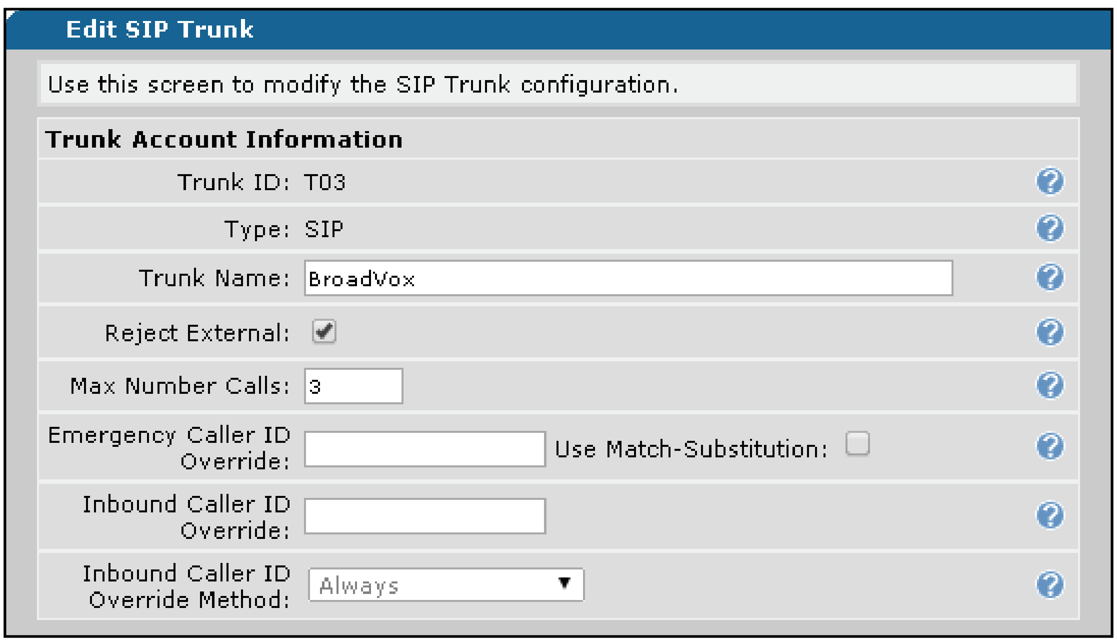
To create a SIP trunk account to the BroadVox SIP server using the NetVanta 7100 GUI, follow these steps:

1. In the NetVanta 7100 GUI, use the navigation bar on the left side of the page to navigate to **Voice** > **Trunks** > **Trunk Accounts**. The **Add/Modify/Delete Trunk Accounts** menu will appear.

2. In the **Trunk Name** field of the **Add/Modify/Delete Trunk Accounts** menu, enter the desired name for the BroadVox SIP trunk account. Use the **Type** drop-down menu to select **SIP**, then select **Add**. The **Edit SIP Trunk** menu will appear.



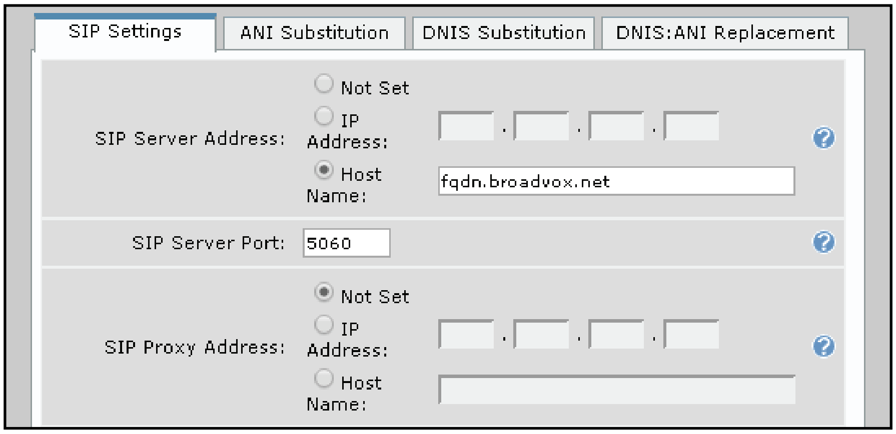
3. In the **Edit SIP Trunk** menu, enter the maximum number of concurrent SIP calls allowed by the BroadVox SIP trunk in the **Max Number Calls** field.



4. In the **Edit SIP Trunk** menu, select the **SIP Settings** tab.

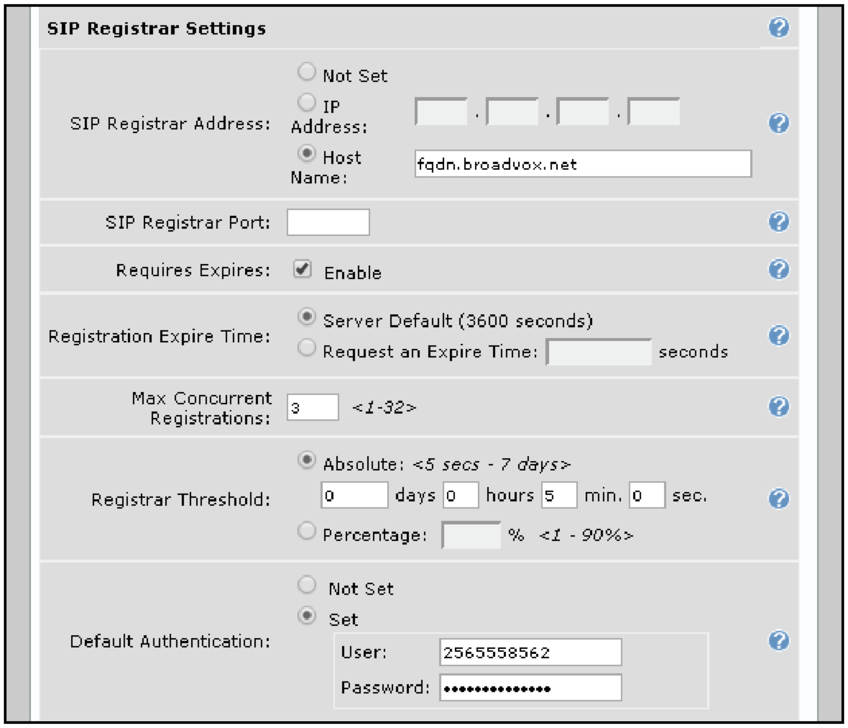
5. In the **SIP Settings** tab, select the **Host Name** radio button next to **SIP Server Address**, and enter the fully qualified domain name (FQDN) of the BroadVox SIP server (provided by BroadVox) in the

adjacent field. In the **SIP Server Port** field, enter **5060**. If required, use the **Dial String Source** drop-down menu to select **To Header**.

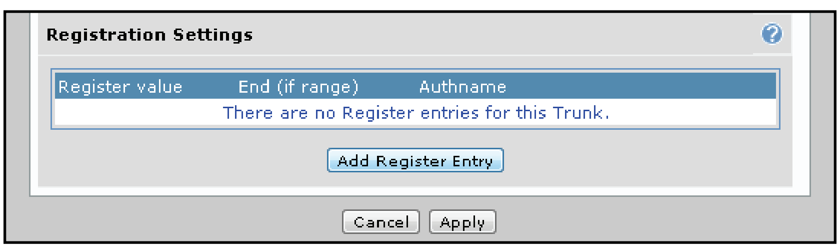


The DNS servers on the NV7100 must be configured to resolve the FQDN of the BroadVox SIP server.

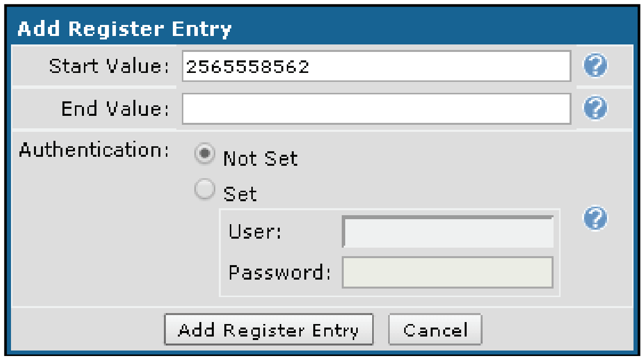
6. In the **SIP Registrar Settings** section, select the **Host Name** radio button next to **SIP Registrar Address**, and enter the FQDN of the BroadVox SIP server in the adjacent field. In the **Max Concurrent Registrations** field, specify the maximum number of simultaneous registration requests to the SIP registrar to limit congestion. Select the **Set** button next to **Default Authentication** to specify a SIP authentication username and password. In the **User** field enter the main business telephone number used to register with the BroadVox SIP server. In the **Password** field, enter the account password for the main business telephone number.



7. In the **Registration Settings** section, select the **Add Register Entry** button to add a register entry. The **Add Register Entry** menu appears.



8. In the **Start Value** field of the **Add Register Entry** menu, enter the main business telephone number used to register with the BroadVox SIP server. Then, select **Add Register Entry**.

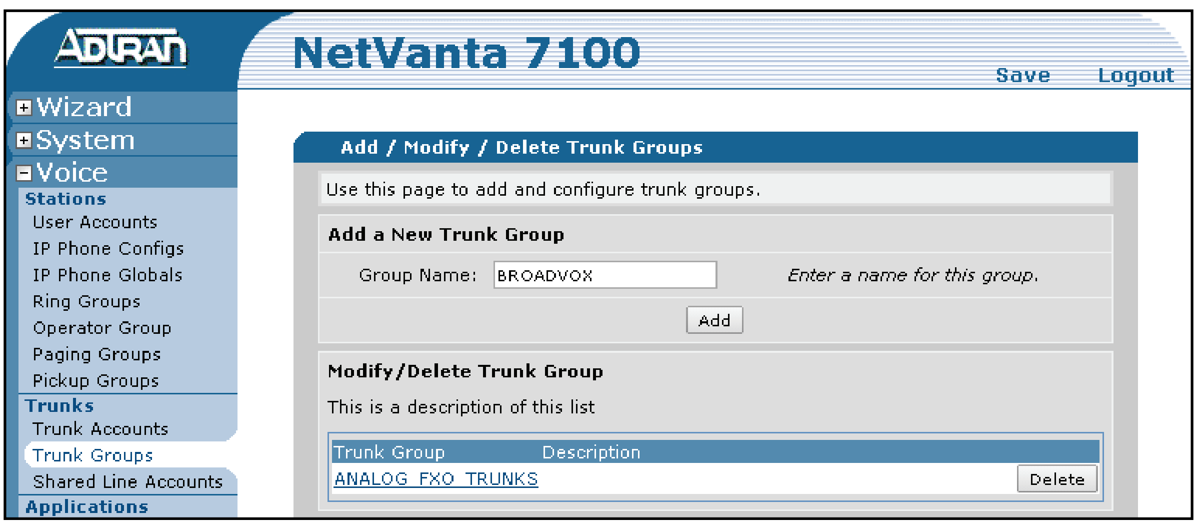


9. In the **Edit SIP Trunk** menu, select the **Apply** button to apply and save the settings to the SIP trunk account.

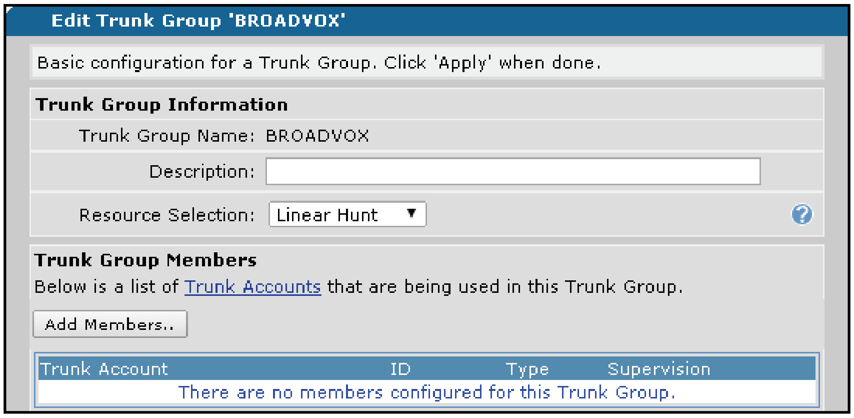
Step 3: Configure a Trunk Group for the BroadVox SIP Trunk

To create a trunk group for the BroadVox SIP trunk, add the BroadVox SIP trunk account as a member of the trunk group, and configure the outbound call templates for the trunk group, follow these steps:

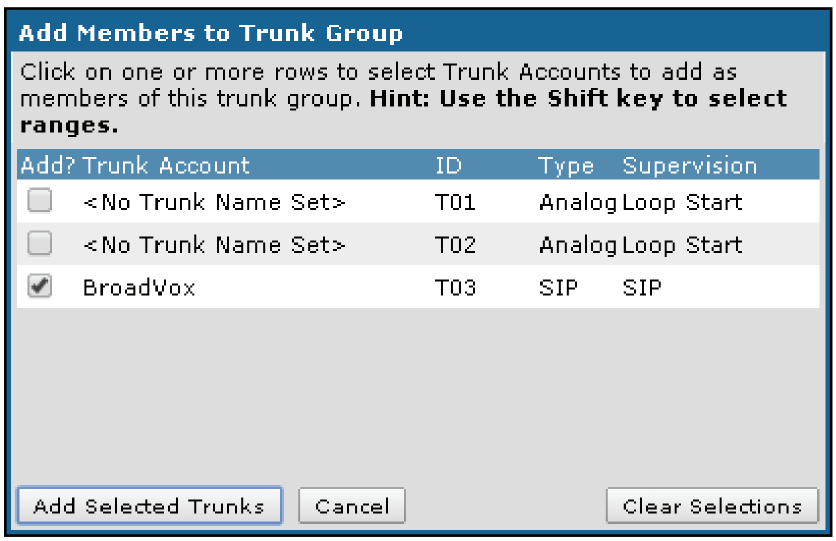
1. In the NetVanta 7100 GUI, use the navigation bar on the left side of the page to navigate to **Voice** > **Trunks** > **Trunk Groups**. The **Add/Modify/Delete Trunk Groups** menu will appear.
2. In the **Group Name** field of the **Add/Modify/Delete Trunk Groups** menu, enter the desired name for the trunk group. Then, select the **Add** button to add the trunk group. The **Edit Trunk Group** menu will appear.



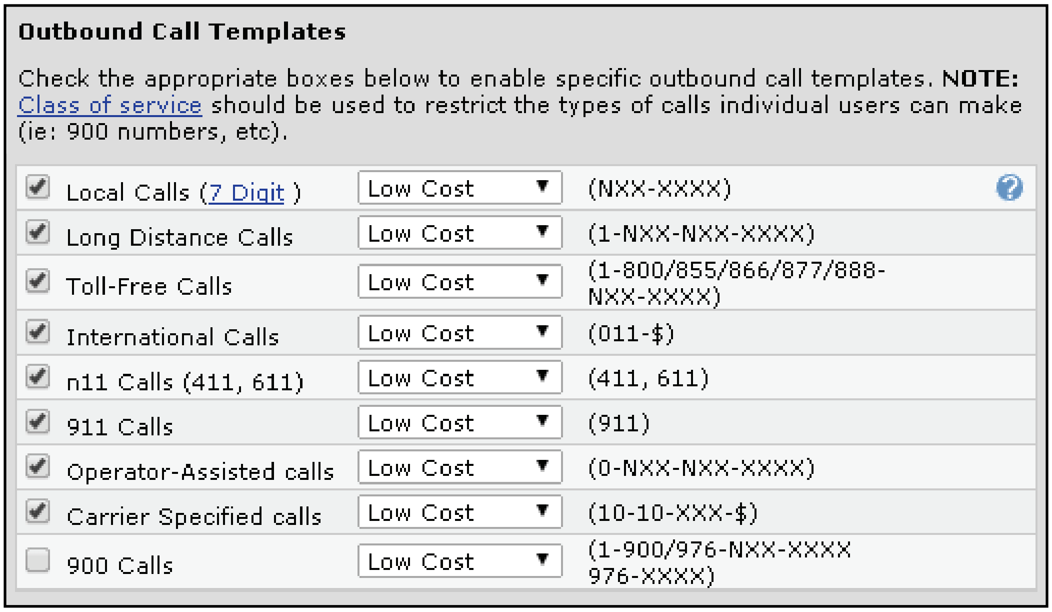
3. In the **Trunk Group Member** section of the **Edit Trunk Group** menu, select the **Add Members** button. The **Add Members to Trunk Group** menu will appear.



4. In the **Add Members to Trunk Group** menu, select the check box next to the BroadVox SIP trunk account that you created in *Create a SIP Trunk Account to the BroadVox SIP Server on page 4*. Then select **Add Selected Trunks**. The SIP trunk account will be added as a member of the trunk group.



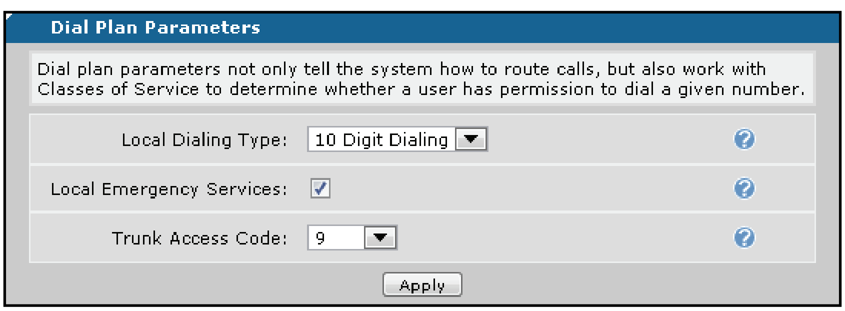
5. In the **Outbound Call Templates** section of the **Edit Trunk Group** menu, use the check boxes to enable the desired outbound call templates for the trunk group. Then select the **Apply** button.



Step 4: Configure 10-digit Dialing

To configure 10-digit dialing on the NetVanta 7100, follow these steps:

1. In the NetVanta 7100 GUI, use the navigation bar on the left side of the page to navigate to **Voice** > **System Setup** > **Dial Plan**. The **Dial Plan Parameters** menu will appear.
2. In the **Dial Plan Parameters** menu, use the **Local Dialing Type** drop-down menu to select **10 Digit Dialing**. Then select **Apply**.



Step 5: Save the NetVanta 7100 Configuration

To save the configurations made on the NetVanta 7100 so that they will persist after the unit is rebooted, select the **Save** button located at the top-right corner of the GUI.

Configuring the NetVanta 7100 for BroadVox Using the CLI

To configure the NetVanta 7100 for a BroadVox SIP trunk using the CLI, follow these steps:

- *Step 1: Access the CLI on page 11*
- *Step 2: Create a SIP Trunk Account to the BroadVox SIP Server on page 11*
- *Step 3: Configure a Trunk Group for the BroadVox SIP Trunk on page 13*
- *Step 4: Configure 10-digit Local Dialing on page 15*
- *Step 5: Save the NetVanta 7100 Configuration on page 16*

Step 1: Access the CLI

To access the command line interface (CLI) on the ADTRAN SBC, follow these steps:

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

   telnet 10.10.10.1

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

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

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

4. Enter Enable mode by entering **enable** at the prompt as follows:

   >enable

5. Enter your Enable mode password at the prompt.

   > NOTE: *The default Enable mode password is **password**. If your product no longer has the default Enable password, contact your system administrator for the appropriate password.*

6. Enter the unit's Global Configuration mode as follows:

   #configure terminal
   (config)#

Step 2: Create a SIP Trunk Account to the BroadVox SIP Server

To create a SIP trunk account to the BroadVox SIP server on the NetVanta7100 using the CLI, follow these steps:

1. From the Global Configuration mode, use the **voice trunk** <*Txx*> **type sip** command to create a SIP trunk and enter the Voice SIP Trunk Configuration mode.

   (config)#**voice trunk** <*Txx*> **type sip**.

Syntax	Description
<*Txx*>	Specifies the trunk identity in the format Txx, where xx is the trunk ID number between **01** and **99** (for example, **T01**)

   The following example creates SIP trunk **T03** and enters the Voice SIP Trunk Configuration mode:

   (config)#**voice trunk T03 type sip**
   (config-T03)#

2. From the Voice SIP Trunk Configuration mode, use the **sip-server primary** command to specify the fully qualified domain name (FQDN) or IP address of the SIP server to which the trunk will send SIP messages. The value used should be the FQDN or IP address of the BroadVox SIP server.

   (config-T03)#**sip-server primary** <*FQDN or IP address*>

Syntax	Description
<*FQDN or IP address*>	Specifies the FQDN or IP address of the SIP server. IP addresses should be expressed in dotted decimal notation (for example, **10.10.10.1**).

   The following example sets the SIP server FQDN to **fqdn.broadvox.net**:

   (config-T03)#**sip-server fqdn.broadvox.net**

3. Use the **registrar primary** command to specify the fully qualified domain name (FQDN) or IP address of the primary SIP registrar to which the trunk will send SIP register messages. The value used should be the FQDN or IP address of the BroadVox SIP server.

   (config-T03)#**registrar primary** <*FQDN or IP address*>

Syntax	Description
<*FQDN or IP address*>	Specifies the FQDN or IP address of the SIP registrar serer. IP addresses should be expressed in dotted decimal notation (for example, **10.10.10.1**).

   The following example sets the primary SIP registrar server FQDN to **fqdn.broadvox.net**:

   (config-T03)#**registrar primary fqdn.broadvox.net**

4. Use the **registrar max-concurrent-reg** command to specify the maximum number of simultaneous registration requests allowed to the SIP registrar. This is used to limit congestion caused by too many concurrent registration requests.

   (config-T03)#**registrar max-concurrent-reg** <*value*>

Syntax	Description
<*value*>	Specifies the maximum number of concurrent registrations. Valid range is **1** to **32** registrations.

The following example sets the maximum number of concurrent registrations to **3**:

(config-T03)#**registrar max-concurrent-reg 3**

5. Use the **register** command to specify the number used to register with the BroadVox SIP server.

   (config-T03)#**register** <*name*>

Syntax	Description
<*name*>	Specifies a value to register with the SIP server.

The following example registers the number **2565558562** with the BroadVox SIP server:

(config-T03)#**register 2565558562**

6. Use the **authentication username password** command to specify the SIP authentication username and password used to register with the BroadVox SIP server.

   (config-T03)#**authentication username** <*username*> **password** <*password*>

Syntax	Description
<*username*>	Specifies a string to be sent as the user name in authentication
<*password*>	Specifies a string to be sent as the password in authentication.

The following example configures an authentication username of **2565558562** and password of **PASSWORD**:

(config-T03)#**authentication username 2565558562 password PASSWORD**

Step 3: Configure a Trunk Group for the BroadVox SIP Trunk

To create a trunk group for the BroadVox SIP trunk, add the BroadVox SIP trunk account as a member of the trunk group, and configure the outbound call templates for the trunk group using the CLI, follow these steps:

1. From the Global Configuration mode, use the **voice grouped-trunk** <*name*> command to create a trunk group and enter the Voice Trunk Group Configuration mode.

   (config)#**voice grouped-trunk** <*name*>

Syntax	Description
<*name*>	Specifies the name of the trunk.

The following example creates a trunk group named **BROADVOX** and enters the Voice Trunk Group Configuration mode:

(config)#**voice grouped-trunk BROADVOX**
(config-BROADVOX)#

2. From the Voice Trunk Group Configuration mode, use the **trunk** <*Txx*> command to add to the trunk group the SIP trunk configured to the BroadVox SIP server.

(config-BROADVOX)#**trunk** <*Txx*>

Syntax	Description
<*Txx*>	Specifies the trunk identity in the format Txx, where xx is the trunk ID number between **01** and **99** (for example, **T01**).

The following example adds trunk **T03** to the **BROADVOX** trunk group.

(config-BROADVOX)#**trunk T03**

3. Use the **accept** command to specify the numbers that can be dialed on the SIP trunk.

(config-BROADVOX)#**accept** <*template*> **cost** <*value*>

Syntax	Description
<*template*>	Specifies the patterns users can dial on the trunk. You can enter a complete phone number or wildcards can be used to help define accepted numbers.
cost <*value*>	Specifies the cost value for the trunk. This option is used 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**.

Valid characters for templates are as follows:

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

The following are example template entries using wildcards:

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

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 following example allows users on the trunk to dial 10-digit numbers, 11-digit numbers, international numbers, 411, 611, and 911:

(config-BROADVOX)#**accept NXX-NXX-XXXX cost 0**
(config-BROADVOX)#**accept 1-NXX-NXX-XXXX cost 0**
(config-BROADVOX)#**accept 011-$ cost 0**
(config-BROADVOX)#**accept 411 cost 0**
(config-BROADVOX)#**accept 611 cost 0**
(config-BROADVOX)#**accept 911 cost 0**

4. Use the **reject** *<template>* command to specify the numbers that cannot be dialed on the trunk.

(config-BROADVOX)#**reject** *<template>*

Syntax	Description
<template>	Specifies the patterns users cannot dial on the trunk. You can enter a complete phone number or wildcards can be used to help define rejected numbers.

The following example blocks calls to any 900 number on the trunk group:

(config-BROADVOX)#**reject 900-NXX-XXXX**
(config-BROADVOX)#**reject 1-900-NXX-XXXX**

Step 4: Configure 10-digit Local Dialing

To configure 10-digit local dialing on the NetVanta 7100 using the CLI, follow these steps:

1. From the Global Configuration mode, use the **voice dial-plan local** command to add 10-digit local dialing to the local dial plan group.

(config)#**voice dial-plan** *<pattern id>* **local** *<pattern>*

Syntax	Description
<pattern id>	Specifies dial pattern identification. Valid range is **1** to **255**.
<pattern>	Specifies a dialing pattern. You can enter a complete phone number, or wildcards can be used to define the dialing pattern. Refer to Step 3 of *Configure a Trunk Group for the BroadVox SIP Trunk on page 13* for more information on using wild cards.

The following example adds 10-digit dialing to the local dial plan group:

(config)#**voice dial-plan NXX-NXX-XXXX**

Step 5: Save the NetVanta 7100 Configuration

To save the configurations made on the NetVanta 7100 so that they will persist after the unit is rebooted, use the **write** command from the Enable mode.

>**enable**
#**write**

Verification Testing Summary

Interoperability verification testing focused on SIP trunk operations between the NetVanta 7100 and BroadVox. A summary of the testing performed and the results from the testing is provided in the tables below...

Table 2. Outbound Calling

Name	Description	Results
SUT-PSTN - G.711 μ-Law - Party B Answers - Party A Clears	Party A places a call to Party B from a phone through the system under test (SUT) and BroadVox. Party B answers the call. Party A hangs up the call.	PASS
SUT-PSTN - G.711 μ-Law - Party B Answers - Party B Clears	Party A places a call to Party B from a phone through SUT and BroadVox. Party B answers the call. Party B hangs up the call.	PASS
SUT-PSTN - G.729 Annex A - Party B Answers - Party A Clears	Party A places a call to Party B from a phone through SUT and BroadVox. Party B answers the call. Party A hangs up the call.	PASS
SUT-PSTN - G.729 Annex A - Party B Answers - Party B Clears	Party A places a call to Party B from a phone through SUT and BroadVox. Party B answers the call. Party B hangs up the call.	PASS
SUT-PSTN - G.729 Annex A - Party A Cancels	Party A places a call to Party B from a phone through SUT and BroadVox. Party A hangs up the call before Party B answers.	PASS
SUT-PSTN - G.711 μ-Law - Party B No Answer	Party A places a call to Party B from a phone through SUT and BroadVox. Party B does not answer the call. A timeout occurs.	PASS
SUT-PSTN - G.711 μ-Law - Party B Busy	Party A places a call to Party B from a phone through SUT and BroadVox. Party B is already on the phone. A busy condition occurs.	PASS
SUT-PSTN - G.711 μ-Law - CLID Delivery	Party A places a call to Party B from a phone through SUT and BroadVox.	PASS

Table 3. Inbound Calling

Name	Description	Results
PSTN-SUT - G.711 μ-Law - Party A Answers - Party B Clears	Party B places a call to Party A from a phone through BroadVox and SUT. Party A answers the call. Party B hangs up the call.	PASS
PSTN-SUT - G.711 μ-Law - Party A Answers - Party A Clears	Party B places a call to Party A from a phone through BroadVox and SUT. Party A answers the call. Party A hangs up the call.	PASS

Table 3. Inbound Calling *(Continued)*

Name	Description	Results
PSTN-SUT - G.711 μ-Law - Party B Cancels	Party B places a call to Party A from a phone through SUT and BroadVox. Party B hangs up the call before Party A answers.	PASS
PSTN-SUT - G.729 Annex A - Party A Answers - Party B Clears	Party B places a call to Party A from a phone through SUT and BroadVox. Party A answers the call. Party B hangs up the call.	PASS
PSTN-SUT - G.729 Annex A - Party A Answers - Party A Clears	Party B places a call to Party A from a phone through SUT and BroadVox. Party A answers the call. Party A hangs up the call.	PASS
PSTN-SUT - G.729 Annex A - Party B Cancels	Party B places a call to Party A from a phone through SUT and BroadVox. Party B hangs up the call before Party A answers.	PASS
PSTN-SUT - G.711 μ-Law - Party A No Answer	Party B places a call to Party A from a phone through BroadVox to the SUT. Party A does provide a progress message, but no connection is made. Call ends when expiration timer is exceeded.	N/A NV7100 forces Call Coverage rules to either Voice Mail or transfer to another extension.
PSTN-SUT - G.711 μ-Law - Party A Busy	Party B places a call to Party A from a phone through SUT and BroadVox. Party A is already on the phone. A busy condition occurs.	PASS
PSTN-SUT - G.711 μ-Law - No Route Available	Party B places a call to Party A from a phone through SUT and BroadVox. No route exists to reach Party A. This test ensures that DIDs which are not built in routing on the PBX are not looped back to BroadVox.	PASS
PSTN-SUT - G.711 μ-Law - No Circuits Available	Party B places a call to Party A from a phone through SUT and BroadVox. All circuits to reach Party A are in use. This test ensures that the PBX returns a route-advance-able code so that BroadVox may hunt to an alternate endpoint if all channels to the user on this PBX are used. This only applies if the user has multiple channels by which the call may traverse (typically across multiple PBX systems) and all of them on this PBX are in use. For example, if the PBX is out of licenses or at capacity but the user is not busy, the PBX should return a 503.	SUT responds with a 480 instead of a 503. 16 total INVITEs received.
PSTN-SUT - G.711 μ-Law - CLID Delivery - From Field	Party B places a call to Party A from a phone through SUT to BroadVox. Caller ID Number (CLID) is confirmed on Party A display.	PASS
PSTN-SUT - G.711 μ-Law - CNAM Delivery - From Field	Party B places a call to Party A from a phone through SUT to BroadVox. Caller ID Name (CNAM) is confirmed on Party A display.	PASS

Table 4. Outbound Faxing

Name	Description	Results
SUT-PSTN - G.711 μ-Law - Fax	Party A places a fax call to Party B from a fax machine through SUT to BroadVox.	PASS
SUT-PSTN - G.711 μ-Law - T.38 Fax	Party A places a fax call to Party B from a fax machine through SUT to BroadVox.	FAIL BroadVox does not send back a CSI & DIS from the far end as required.
SUT-PSTN - G.711 μ-Law - T.38 - Fallback - No Silence Supp - Fax	Party A places a fax call to Party B from a fax machine through SUT to BroadVox. This use case is required if the PBX supports fax over G.711.	PASS
SUT-PSTN - G.711 μ-Law - T.38 - Fallback - W/Silence Supp - Fax	Party A places a fax call to Party B from a fax machine through SUT to BroadVox. This use case is required if the PBX supports fax over G.711.	PASS
SUT-PSTN - G.729 Annex A - T.38 Fax	Party A places a fax call to Party B from a fax machine through SUT to BroadVox.	PASS
SUT-PSTN - G.729 Annex A - T.38 - Fallback - Fax	Party A places a fax call to Party B from a fax machine through SUT to BroadVox. This use case is required if the PBX supports fax over G.711.	PASS

Table 5. Inbound Faxing

Name	Description	Results
PSTN-SUT - G.711 μ-Law Fax	Party B places a fax call to Party A from a fax machine through BroadVox to SUT.	PASS
PSTN-SUT - G.711 μ-Law - T.38 Fax	Party B places a fax call to Party A from a fax machine through BroadVox to SUT.	PASS
PSTN-SUT - G.729 Annex A - T.38 Fax	Party B places a fax call to Party A from a fax machine through BroadVox to SUT.	PASS

Table 6. Outbound DTMF

Name	Description	Results
SUT-PSTN - G.711 μ-Law - DTMF	Party A places a call to Party B through SUT to BroadVox. This test is required if the PBX supports sending DTMF over G.711.	PASS
SUT-PSTN - G.711 μ-Law - RFC 2833	Party A places a call to Party B through SUT to BroadVox. This test is required if the PBX supports sending DTMF using RFC 2833.	PASS
SUT-PSTN - G.711 μ-Law - SIP INFO	Party A places a call to Party B through SUT to BroadVox. This test is required if RFC 2833 does not work AND the PBX supports sending DTMF using the SIP INFO method.	PASS

Table 7. Inbound DTMF

Name	Description	Results
PSTN-SUT - G.711 μ-Law - DTMF	Party B places a call to Party A through BroadVox to SUT. This test is required if the PBX supports DTMF over G.711.	PASS
PSTN-SUT - G.711 μ-Law - RFC 2833	Party B places a call to Party A through BroadVox to SUT. This test is required if the PBX supports DTMF using RFC 2833.	PASS
PSTN-SUT - G.711 μ-Law - SIP INFO	Party B places a call to Party A through BroadVox to SUT. This test is required if RFC 2833 does not work AND the PBX supports DTMF using the SIP INFO method.	N/A

Table 8. Outbound Feature Interworking

Name	Description	Results
SUT-PSTN - G.711 μ-Law - Party B Answers - Party A Hold - Party A Clears	Party A places a call to Party B from a phone through SUT and BroadVox. Party B answers the call. Party A places the call on hold, takes the call off hold, then hangs up the call.	PASS
SUT-PSTN - G.711 μ-Law - Party B Answers - Party A Hold - Party B Clears	Party A places a call to Party B from a phone through SUT and BroadVox. Party B answers the call. Party A places the call on hold, takes the call off hold; then Party B hangs up the call.	PASS
SUT-PSTN - G.711 μ-Law - B1 Answers - B2 Answers - A1 Clears - A2 Clears	Party A1 places a call to Party B1 from a phone through SUT to BroadVox. Party B1 answers the call. Party A2 places a call to Party B2 from a phone through SUT to BroadVox. Party B2 answers the call. Party A1 hangs up the call. Party A2 hangs up the call.	PASS
SUT-PSTN - G.711 μ-Law - Call Transfer - Party B Clears	Party A places a call to Party B from a phone through SUT to BroadVox. Party B answers the call. Party A transfers the call to Party C.	PASS
SUT-PSTN - G.711 μ-Law - TCP	Party A places a call to Party B from a phone through SUT to BroadVox using Transport Control Protocol.	NOT SUPPORTED
SUT-PSTN - G.711 μ-Law - TLS	Party A places a call to Party B from a phone through SUT to BroadVox using Transport Layer Security.	NOT SUPPORTED
SUT-PSTN - G.711 μ-Law - SRTP	BroadVox places a call to Party A from a phone through SUT and BroadVox.	NOT SUPPORTED

Table 9. Inbound Feature Interworking

Name	Description	Results
PSTN-SUT - BTN Routing - Static	Party B places a call to Party A (BTN DID) from a phone through BroadVox to SUT. Party A answers the call. By default (unless specified otherwise) dialed number in a static configuration is sent in the Request Line URI as well as the TO-Header.	PASS
PSTN-SUT - DID Routing - Static	Party B places a call to Party A from a phone through BroadVox and SUT. Party A answers the call. By default (unless specified otherwise) dialed number in a static configuration is sent in the Request Line URI as well as the TO-Header.	PASS
SUT-PSTN - Registration	SUT registers to BroadVox.	PASS
PSTN-SUT - BTN Routing - Dynamic	Party B places a call to Party A (BTN DID) from a phone through BroadVox to SUT. Party A answers the call. By default (unless otherwise specified) dialed number is sent in the To-Header.	PASS
PSTN-SUT - DID Routing - Dynamic	Party B places a call to Party A (DID) from a phone through BroadVox to SUT. Party A answers the call. By default (unless otherwise specified) dialed number is sent in the To-Header.	PASS
PSTN-SUT - G.711 μ-Law - TCP	BroadVox places a call to Party A from a phone through SUT to BroadVox using Transport Control Protocol.	Not tested per BroadVox. Note when UDP is disabled no Station to Station calling is allowed
PSTN-SUT - G.711 μ-Law - TLS	BroadVox places a call to Party A from a phone through SUT to BroadVox using Transport Layer Security.	NOT SUPPORTED
PSTN-SUT - G.711 μ-Law - SRTP	BroadVox places a call to Party A from a phone through SUT to BroadVox.	NOT SUPPORTED

Table 10. Outbound Call Hunting

Name	Description	Results
SUT-PSTN - G.711 μ-Law - No Circuits Available	Party A places a call to Party B from a phone through SUT and BroadVox. All circuits to reach Party B are in use. This test case verifies that your system properly supports alternate routes and call hunting, which is the basis for the greater availability achieved on this platform.	PASS
SUT-PSTN - G.711 μ-Law - Timeout Call Hunting	Party A places a call to Party B from a phone through SUT and BroadVox. All circuits to reach Party B are in use. This test case verifies that your system properly supports alternate routes and call hunting, which is the basis for the greater availability achieved on this platform. It also verifies that your system provides a configuration option for the maximum number of re-INVITEs allowed before moving to the next route.	PASS

Table 11. Inbound Call Hunting

Name	Description	Results
PSTN-SUT - Inbound Routing - Primary	Party B places a call to Party A from a phone through BroadVox to the SUT. The call should have routed from Primary IP address in BroadVox DNS hostname. Party A answers the call.	PASS
PSTN-SUT - Inbound Routing - Back-up	Party B places a call to Party A from a phone through BroadVox to the SUT. The call should have routed from Back-up IP address in BroadVox DNS hostname. Party A answers the call.	PASS

Table 12. Dynamic NAT/Firewall Traversal

Name	Description	Results
SUT-PSTN - G.711 μ-Law - NAT - Party B Answers - Party A Clears	Party A places a call to Party B from a phone through SUT to BroadVox. Party B answers the call. Party A hangs up the call. SUT must be connected to BroadVox through a SIP Trunk running over a dynamic NAT IP connection.	PASS
PSTN-SUT - G.711 μ-Law - NAT - Party A Answers - Party B Clears	Party B places a call to Party A from a phone through SUT and BroadVox. Party A answers the call. Party B hangs up the call. SUT must be connected to BroadVox through a SIP Trunk running over a dynamic NAT IP connection.	PASS

Table 13. Static NAT/Firewall Traversal

Name	Description	Results
SUT-PSTN - G.711 μ-Law - NAT - Party B Answers - Party A Clears	Party A places a call to Party B from a phone through SUT and BroadVox. Party B answers the call. Party A hangs up the call. SUT must be connected to BroadVox through a SIP Trunk running over a static NAT IP connection.	PASS
PSTN-SUT - G.711 μ-Law - NAT - Party A Answers - Party B Clears	Party B places a call to Party A from a phone through SUT and BroadVox. Party A answers the call. Party B hangs up the call. SUT must be connected to BroadVox through a SIP Trunk running over a static NAT IP connection.	PASS