



Interoperability Guide

Integrating an ADTRAN NetVanta Switch and ShoreTel ShoreGear PBX

This interoperability guide provides instructions for integrating an ADTRAN NetVanta switch and a ShoreTel® ShoreGear PBX. In addition to basic telephony compatibility, this solution provides verified compatibility for Power over Ethernet (PoE), link aggregation, Rapid Spanning Tree Protocol (RTSP), Quality of Service (QoS) and Link-Layer Discovery Protocol-Media Endpoint Discovery (LLDP-MED) This guide includes the description of the network application, verification summary, and individual phone configurations for the NetVanta switch and the ShoreTel products.

This guide consists of the following sections:

- *Overview on page 2*
- *Hardware and Software Requirements and Limitations on page 2*
- *Supported Features and Exceptions on page 3*
- *Verification Performed on page 4*
- *Configuring the NetVanta Switch Using the VoIP Setup Wizard on page 7*
- *Configuring the NetVanta Switch Using the CLI on page 7*
- *NetVanta Switch Sample Configuration on page 12*
- *Configuring the ShoreTel ShoreGear PBX on page 14*
- *Additional Resources on page 26*

Overview

The purpose of this solution is to provide an easy-to-deploy, secure, reliable, and redundant telephony solution for enterprise businesses by integrating the ADTRAN NetVanta switch line with the ShoreTel ShoreGear PBX. In addition to basic telephony, PoE, link aggregation, RSTP, QoS, and LLDP-MED are supported by this integration. *Figure 1* below shows the network topology used to test the solution. In the test network, multiple NetVanta switches were used to test the RSTP and link aggregation features; however, only one NetVanta switch is required.

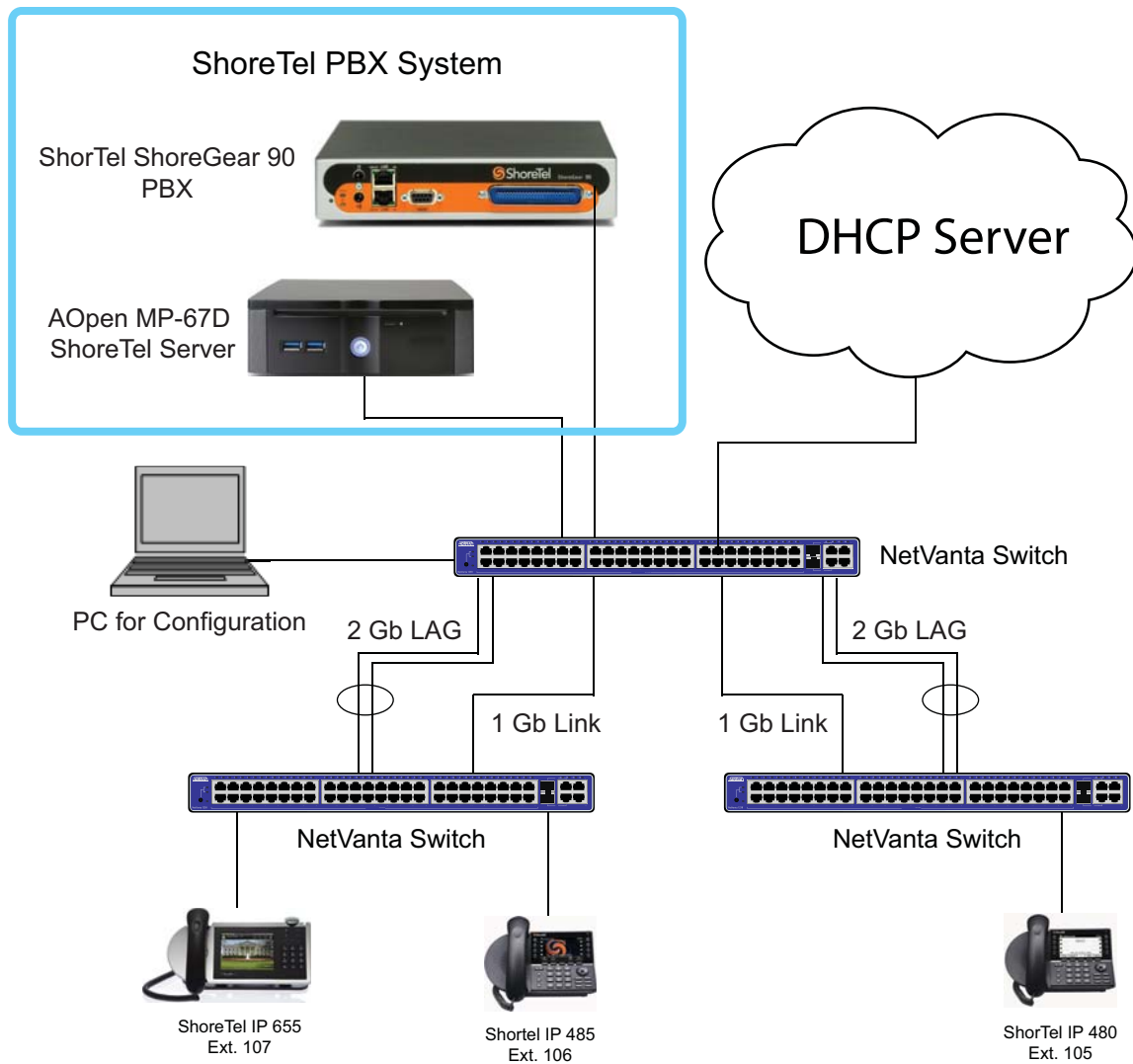


Figure 1. Network Topology Used for Verification

Hardware and Software Requirements and Limitations

Interoperability with the ShoreTel ShoreGear PBX is available on ADTRAN NetVanta switch products running R11.4.1 or later. The test equipment, testing parameters, and associated caveats are described in the following sections.

ActivReach is supported for this solution on the NetVanta 1235P and NetVanta 1535P. A NetVanta ActivReach media converter is required at the far end when employing ActivReach with these switches.

Equipment and Versions

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

Table 1. Verification Test Equipment and Firmware Versions

Product	Firmware Version
ADTRAN NetVanta 1234P (FE L3-Lite)	R 11.4.1
ADTRAN NetVanta 1531P (GE L3-Lite)	R 11.4.1
ADTRAN NetVanta 1535P (Activreach)	R 11.4.1
ADTRAN NetVanta 1638P (GE Full L3)	R 11.4.1
ShoreTel ShoreGear 90 Voice Switch	v. 14.2
AOpen MP-67D Server w/VMware ESXi	4.1
ShoreWare Director Management Software	19.6.6705.0
ShoreTel IP 485g Phone	
ShoreTel IP 655 Phone	
ShoreTel IP 480 Phone	

Supported Features and Exceptions

The following features are supported by the solution:

- ShoreTel telephony compatibility with NetVanta switches
- PoE interoperability
- Link Aggregation Group based on IEEE 802.3ad
- Rapid Spanning Tree Protocol (RSTP)
- Quality of Service (QoS)
- LLDP-MED
- ActivReach

Verification Performed

The following tables describe the feature verifications performed during interoperability testing and the results of the tests:

Table 2. Power over Ethernet Interoperability

Test Case	Description	Notes
5.1.1	Power up the switch. Connect the ShoreTel IP telephone to the NetVanta switch. Verify that the telephone powers up and boots properly. Verify that the switch can disable/enable the inline power on the interface. Administer the phone on the ShoreTel Communication Server and verify that it registers and can complete calls to another phone. Note power level supplied by NetVanta switch (for on hook, off hook, volume high).	Pass
5.1.2	Repeat Test Case 5.1.1 with additional ShoreTel IP telephones.	Pass
5.1.4	Disable inline power from a port. Verify that the port is no longer supplying power although it is still active for data traffic.	Pass
5.1.5	Power off and on the NetVanta switch with all PoE equipment attached. Verify that all PoE equipment boots up correctly after the NetVanta switch is powered on again.	Pass
5.1.1	Power up the switch. Connect the ShoreTel IP telephone to the NetVanta switch. Verify that the telephone powers up and boots properly. Verify that the switch can disable/enable the inline power on the interface. Administer the phone on the ShoreTel Communication Server and verify that it registers and can complete calls to another phone. Note power level supplied by the NetVanta switch (for on hook, off hook, volume high).	Pass
5.1.2	Repeat Test Case 5.1.1 with additional ShoreTel IP telephones.	Pass
5.1.4	Disable inline power from a port. Verify that the port is no longer supplying power although it is still active for data traffic.	Pass

Table 3. DHCP Relay and VLAN Support

Test Case	Description	Notes
Reconnect LAG. Disconnect the Gigabit link between the two NetVanta switches.		
5.2.1	Verify that all IP telephones and workstations are able to obtain an IP address from the Dynamic Host Configuration Protocol (DHCP) server from the correct virtual LAN (VLAN).	Pass

Table 3. DHCP Relay and VLAN Support (Continued)

Test Case	Description	Notes
5.2.2	Mirror the trunk port to another port. Verify that the trunk port is passing 802.1/p and DiffServ information correctly.	Pass
5.2.3	Verify that there is network connectivity between all telephones and workstations via ping.	Pass
5.2.4	Place calls among the different telephones. Verify that calls are completed successfully and voice quality is good.	Pass
5.2.6	Make a phone call between Shoretel IP phones. Verify that dual-tone multi-frequency (DTMF) tones can be heard on both phones when either party presses the numbers on the dial pad.	Pass

Table 4. Link Aggregation Group (LAG) Based on IEEE 802.3ad

Test Case	Description	Notes
Disconnect the Gigabit link between the two NetVanta switches.		
5.3.1	Disconnect and reconnect all telephones from the switches. Verify that all telephones and PCs are able to obtain a correct IP address and VLAN information, and there is network connectivity between them via ping.	Pass
5.3.2	Place a call from Shoretel IP phone on one switch to a Shoretel IP phone on another. Verify that the call has been completed successfully and voice quality is good.	Pass
5.3.4	Make a phone call from Shoretel IP phone to another Shoretel IP phone. Verify that dual-tone multi-frequency (DTMF) tones can be heard on both phones when either party presses the numbers on the dial pad.	Pass
5.3.5	Place a call from Shoretel IP phone on one switch to a Shoretel IP phone on another. Disconnect one of the members from the LAG group. Verify that the call is uninterrupted and voice quality is good.	Pass
5.3.9	Continuing from above with the call established, reconnect all other member(s). Verify that the call remains uninterrupted and voice quality is good.	Pass
5.3.10	Verify that traffic is traversing through all members of the LAG.	Pass

Table 5. Rapid Spanning Tree Protocol (RSTP) or IEEE802.1W

Test Case	Description	Notes
Replace the LAG group with a 1 Gigabit link between the two switches, thereby resulting in two (2) separate trunk links between Site A and Site B.		
5.4.1	Place a call from Shoretel IP phone on one switch to a Shoretel IP phone on another with Rapid Spanning Tree Protocol (RSTP) disabled, then cause a re-span by disconnecting the forwarding link. Observe the time it took to re-span and verify that the call is dropped due to re-span.	Pass
5.4.2	Place a call from Shoretel IP phone on one switch to a Shoretel IP phone on another. Reconnect the link. Observe the time it took to re-span and verify that the call is dropped due to re-span.	Pass
5.4.3	Enable RSTP on both switches. Place a call from Shoretel IP phone on one switch to a Shoretel IP phone on another. Cause a re-span by disconnecting the forwarding link. Observe the time it took to re-span and verify that the call remains uninterrupted.	Pass
5.4.4	Place a call from Shoretel IP phone on one switch to a Shoretel IP phone on another. Reconnect the link back. Observe the time it took to re-span and verify that the call remains uninterrupted.	Pass

Table 6. Quality of Service (QoS)

Test Case	Description	Notes
5.5.5	Place a call from Shoretel IP phone on one switch to a Shoretel IP phone on another. Verify that calls complete successfully and all 802.1p and Diffserv priorities are respected and or observed. (ToS 5 and EF 46)	Pass
5.5.6	Ensure that signaling is sent with Diffserv code point 31 (Assured Forwarding)	Pass

Table 7. LLDP Functionality Between NetVanta Switch and ShoreTel ShoreGear

Test Case	Description	Notes
5.6.1	Connect the Shoretel phone to the NetVanta switch and verify the phone appears in the LLDP switch neighbors table	Pass
5.6.2	Verify the Shoretel phone receives the proper voice VLAN	Pass
5.6.3	Verify the Shoretel phone receives the proper QoS settings	Pass
5.6.4	Verify POE was assigned to the proper class and using the correct wattage	Pass

Table 8. Power over Ethernet Interoperability

Test Case	Description	Notes
5.7.1	Connect a ShoreTel IP phone to the network via 1000+ feet of CAT3 cable using ADTRAN's Activereach Technology. Verify phone powers up via POE successfully	Pass
5.7.2	Place a call to another phone and verify call connects and voice quality is good.	Pass

Configuring the NetVanta Switch Using the VoIP Setup Wizard

The VoIP Setup Wizard can be used to configure most NetVanta switches with best practices settings for deployment in a VoIP network. Although the recommended settings provided by this wizard are general, best practice recommendations, they may not be applicable for every VoIP network. When running the wizard, you are given the option of applying the ADTRAN recommended settings or specifying your own settings. Additional features such as link aggregation and LLDP-MED must be configured manually. For more information on using the VoIP Setup Wizard, refer to *Configuring NetVanta Switches for a VoIP Network* available online at <https://supportforums.adtran.com>.

Configuring the NetVanta Switch Using the CLI


To configure the NetVanta switch for interoperability with the ShoreTel ShoreGear PBX using the CLI, follow these steps:

- *Access the CLI on page 7*
- *Configure the Basic Interoperability Settings on page 8*
- *Optional. Configure Link Aggregation on page 10*
- *Optional. Configure Link Aggregation on page 10*
- *Optional. Configure Rapid Spanning Tree Protocol on page 11*
- *Optional. Configure Quality of Service on page 11*


Access the CLI

To access the CLI on your NetVanta switch, 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**. The default enable password is **password**. If your product no longer has the default user name and passwords, contact your system administrator for the 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)#

Configure the Basic Interoperability Settings

Basic interoperability configuration for the NetVanta switch includes configuring an uplink interface to the ShoreGear PBX, creating a voice virtual local area network (VLAN), and interfaces, one for the Ethernet LAN interface to the ShoreTel ShoreGear PBX, and the second for the Ethernet WAN interface to the service provider. Both interfaces are configured using the **ip address** <ipv4 address> <subnet mask> and **media-gateway ip primary** commands. The **ip address** command configures a static IP address for the interface, and the **media-gateway** command is required on the interface for SIP and Realtime Transport Protocol (RTP) media traffic.

Create an Uplink to the ShoreTel ShoreGear PBX

A gigabit switchport uplink interface to the ShoreGear PBX should be configured on the NetVanta switch as interface Both interfaces are configured using the **ip address** <ipv4 address> <subnet mask> and **media-gateway ip primary** commands. The **ip address** command configures a static IP address for the interface, and the **media-gateway** command is required on the interface for SIP and Realtime Transport Protocol (RTP) media traffic.

Create a Voice VLAN

To configure a voice VLAN, follow these steps

1. From the Global Configuration mode, use the **interface vlan** <vlan id> command to create a voice VLAN and enter the VLAN Interface Configuration mode.

Syntax	Description
<vlan id>	Specifies a valid VLAN interface ID. Range is 1 to 4094 .

The following example creates VLAN 2 and enters the VLAN Interface Configuration mode:


```
(config)#interface vlan 2
(config-intf-vlan 2)#
```

2. Use the **ip route-cache express** command to enable Layer 3 switching on the VLAN interface. The following example enables Layer 3 switching on VLAN 2:

```
(config-intf-vlan 2)#ip route-cache express
```

3. Use the **no shutdown** command to enable the VLAN interface. The following example enables the VLAN 2 interface:

```
(config-intf-vlan 2)#no shutdown
```

Configure an Uplink Interface to the ShoreTel ShoreGear PBX

To configure the uplink interface to the ShoreTel ShoreGear PBX, follow these steps:

1. From the Global Configuration mode, use the **interface gigabit-switchport <slot/port>** command to enter the Gigabit Switchport Interface Configuration mode of the gigabit switchport connected to the ShoreGear PBX.

Syntax	Description
<slot/port>	Specifies the slot and port of the interface.

The following example enters the Gigabit Switchport Interface Configuration mode of gigabit switchport 0/2:

```
(config)#interface gigabit-switchport 0/2
(config-giga-swx 0/2)#
```

2. Use the **spanning-tree edgeport** command to enable the interface as an edgeport:

```
(config-giga-swx 0/2)#spanning-tree edgeport
```
3. Use the **switchport access vlan <vlan id>** command to set the port to be a member of the voice VLAN when in access mode.

Syntax	Description
<vlan id>	Specifies a valid VLAN interface ID. Range is 1 to 4094 .

The following example sets gigabit switchport **0/2** to be a member of VLAN 2 when in access mode:

```
(config-giga-swx 0/2)#switchport access vlan 2
```

4. Use the **no shutdown** command to enable the interface. The following example enables the gigabit switchport **0/2** interface:

```
(config-giga-swx 0/2)#no shutdown
```

Optional. Configure LLDP-MED

LLDP-MED is an extension to the base LLDP protocol that provides support for needs specific to VoIP. Since the protocol is a published standard, LLDP-MED support enables VoIP products from multiple vendors to interoperate with each other on the local network. The LLDP-MED capabilities allows AOS switches that support LLDP-MED to discover phones on the local network that support LLDP-MED. The NetVanta products that support LLDP-MED will advertise themselves as a network connectivity phone with support for LLDP-MED capabilities and network policy. LLDP-MED is configured on a per-port basis for each port connected to an LLDP-MED-capable phone or switch. For more information on configuring LLDP-MED, refer to *Configuring LLDP and LLDP-MED in AOS* available from <https://supportforums.adtran.com>.

To configure a port to specify that attached LLDP-MED-capable phones should use a specific VLAN for voice traffic, use the **switchport voice vlan** <vlan id> command from the desired interface's configuration mode. The <vlan id> variable specifies the voice VLAN ID. The following example specifies VLAN 2 for voice traffic.

```
(config)#interface gigabit-switchport 0/8
(config-giga-swx 0/8)#switchport voice vlan 2
```

To configure a port to specify that attached LLDP-MED-capable phones should use a specific VLAN for voice signaling traffic, use the **switchport voice-signaling vlan** <vlan id> command from the desired interface's configuration mode. The <vlan id> variable specifies the voice VLAN ID. The following example specifies VLAN 2 for voice signaling traffic.

```
(config)#interface gigabit-switchport 0/8
(config-giga-swx 0/8)#switchport voice-signaling vlan 2
```

Optional. Configure Link Aggregation

Link aggregation allows you to bundle multiple Ethernet ports to form a single logical channel. A NetVanta switch can support up to 6 channel groups, each of which can contain up to 8 ports per channel. This is beneficial in two ways: increased link capacity and providing redundancy. To configure a LAG, a port channel is first created, and two or more Ethernet ports are assigned to it. For more information on configuring link aggregation, refer to *Link Aggregation Control Protocol (LACP) in AOS* available from <https://supportforums.adtran.com>.

To create a port channel, use the **interface port-channel** <interface id> command from the Global Configuration mode. The <number> variable specifies the identifier for the port channel. The following example creates port channel 1.

```
(config)#interface port-channel 1
(config-p-chan 1)#
```

To add a port to the channel group, use the **channel-group** <interface id> **mode on** command from the interface's configuration mode. The <interface id> variable specifies the identifier for the port channel. The following example adds gigabit switchports 0/2 and 0/3 to port channel 1.

```
(config)#interface gigabit-switchport 0/2
(config-giga-swx 0/2)#channel-group 1 mode on
(config-giga-swx 0/2)#exit
(config)#interface gigabit-switchport 0/3
(config-giga-swx 0/3)#channel-group 1 mode on
```

Optional. Configure Rapid Spanning Tree Protocol

RSTP identifies and blocks redundant paths, preventing loops from forming in a switch network. RSTP is enabled by default, and most applications can use the default RSTP settings. However, if you would like to customize the RSTP configuration, please refer to *Spanning Tree Protocol* available from <https://supportforums.adtran.com>.

Optional. Configure Quality of Service

QoS is used to appropriately allocate bandwidth, reduce packet delay, and ensure reliability for each data packet on the network. To configure basic QoS, follow these steps:

- *Map Class of Service Values to QoS Queues on page 11*
- *Specify the QoS Queue Type on page 11*
- *Enable DSCP-to-CoS Mapping on page 11*
- *Specify the CoS Value on page 12*

For more information on configuring QoS, refer to *Configuring Ethernet Switch QoS and CoS* available from <https://supportforums.adtran.com>.

Map Class of Service Values to QoS Queues

To map class of service (CoS) values to QoS queues, use the `qos cos-map <cos queue id> <cos value>` command from the Global Configuration mode. The `<cos queue id>` variable specifies the queue number to which CoS values are being assigned, and the `<cos value>` variable associates one or more CoS values with the specified priority queue. The following example maps queue 1 to CoS 0 and 1, queue 2 to CoS 2 and 3, queue 3 to CoS 4 and 5, and queue 4 to CoS 6 and 7:

```
(config)#qos cos-map 1 0 1
(config)#qos cos-map 2 2 3
(config)#qos cos-map 3 4 5
(config)#qos cos-map 4 6 7
```

Specify the QoS Queue Type

To specify the QoS queue type as weighted round robin (WRR) and specify the weight of each queue, use the `qos queue-type wrr <weight1> <weight2> <weight3> expedite` command from the Global Configuration mode. The `<weight 1-4>` variable sets the weight of each queue, and the `expedite` command sets the forth queue as high-priority, causing all outbound traffic to be transmitted out of the queue prior to any other traffic queues. The following example specifies the QoS queue type as **WRR**, configures weights for queues 1 through 3, and sets queue 4 to **expedite**:

```
(config)#qos queue-type wrr 1 2 3 expedite
```

Enable DSCP-to-CoS Mapping

To enable differentiated services code point (DSCP) to CoS mapping using the default AOS values, use the `qos dscp-cos default` command from the Global Configuration mode:

```
(config)#qos dscp-cos default
```

Specify the CoS Value

If you are not using LLDP-MED, you can hard-set inbound packets with a specific CoS value used for placing the packets in the priority queue using the **qos default-cos** *<value>* command from the interface configuration mode of each port connected to a phone. The *<value>* variable specifies the default CoS value for untrusted ports and all untagged packets. The following example sets the default CoS value to **7**:

```
(config)#interface gigabit-switchport 0/9
(config-giga-swx 0/9)#qos default-cos 7
```

If you are using LLDP-MED, you can set the switch to trust the existing CoS values on inbound traffic using the **qos trust cos** command from the interface configuration mode of each port connected to an LLDP-MED-capable phone. Additionally, you can specify CoS and DSCP values to the voice and voice signaling traffic using the **switchport [voice | voice-signaling] vlan** *<vlan id>* [**cos** *<value>* | **dscp** *<value>* | **cos** *<value>* **dscp** *<value>*] command. The *<vlan id>* variable specifies the voice VLAN ID, and the *<value>* variable specifies the CoS or DSCP value assigned to the traffic. The following example enables **trust** mode and specifies a CoS value of **7** and DSCP value of **56** for both voice and voice signaling traffic:

```
(config)#interface gigabit-switchport 0/9
(config-giga-swx 0/9)#qos trust cos
(config-giga-swx 0/9)#switchport voice vlan 2 cos 7 dscp 56
(config-giga-swx 0/9)#switchport voice-signaling vlan 2 cos 7 dscp 56
```

NetVanta Switch Sample Configuration

The following example configuration was used to validate the interoperability between the NetVanta switches and the ShoreTel ShoreGear PBX. Only the commands relevant to the interoperability configuration are shown.



The configuration parameters entered in this example are sample configurations only, and only pertain to the configuration of the SIP trunking gateway functionality. This application should be configured in a manner consistent with the needs of your particular network. CLI prompts have been removed from the configuration example to provide a method of copying and pasting configurations directly from this guide into the CLI. This configuration should not be copied without first making the necessary adjustments to ensure it will function properly in your network.

```
!
ip route-cache express
!
vlan 1
  name "Default"
!
vlan 2
  name "voicevlan"
!
vlan 99
  name "management"
```

```
!  
interface port-channel 2  
  no shutdown  
  switchport mode trunk  
  switchport trunk allowed vlan 2  
  qos trust cos  
!  
interface gigabit-switchport 0/1  
  description voicevlan connected to IP655  
  spanning-tree edgeport  
  no shutdown  
  switchport voice vlan 2 cos 3 dscp 31  
  switchport voice-signaling vlan 2 cos 3 dscp 31  
!  
interface gigabit-switchport 0/3  
  description voicevlan connected to IP480 phone  
  spanning-tree edgeport  
  no shutdown  
  switchport voice vlan 2  
  switchport voice-signaling vlan 2  
!  
interface gigabit-switchport 0/5  
  description trunk link to Dist sw port 10  
  no shutdown  
  switchport mode trunk  
  switchport trunk allowed vlan 2  
  qos trust cos  
!  
interface gigabit-switchport 0/7  
  description Port-Channel 2 connected to Dist sw port 19  
  no shutdown  
  qos trust cos  
  channel-group 2 mode on  
!  
interface gigabit-switchport 0/8  
  description Port-Channel 2 connected to Dist sw port 20  
  no shutdown  
  qos trust cos  
  channel-group 2 mode on  
!  
interface gigabit-switchport 0/9  
  no shutdown  
  switchport mode trunk  
  switchport trunk allowed vlan 2  
!  
interface gigabit-switchport 0/17  
  spanning-tree edgeport
```

```
no shutdown
switchport voice vlan 2
switchport voice-signaling vlan 2 cos 3 dscp 31
!
interface gigabit-switchport 0/24
no shutdown
switchport access vlan 99
!
interface vlan 1
no ip address
ip route-cache express
no shutdown
!
interface vlan 2
ip address 10.99.0.100 255.255.255.0
ip route-cache express
no shutdown
!
interface vlan 99
ip address 10.10.10.2 255.255.255.0
ip route-cache express
no shutdown
!
ip route 10.99.1.0 255.255.255.0 10.99.0.1
!
end
```

Configuring the ShoreTel ShoreGear PBX

The ShoreTel ShoreGear PBX is configured using the ShoreTel product's GUI (ShoreWare Director).

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

The first configuration section pertains to general system settings and includes call control, site administration, and switch administration. If these items have already been configured on the system, skip this section and continue to [Configuring System Settings on page 21](#).

Logging in to ShoreWare Director

To log in to the ShoreWare Director, follow these steps:

1. Open a new Web page in your Internet browser.
2. Enter the host name or IP address of the ShoreTel server in the Internet browser's address field in the following form: **http://<ShoreTel server name> | <IP address>/shorewaredirector**. For example:

http://10.90.0.51/shorewaredirector

3. Enter the administrator **Username** and **Password** in the provided fields. Then, select **Login**. I



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

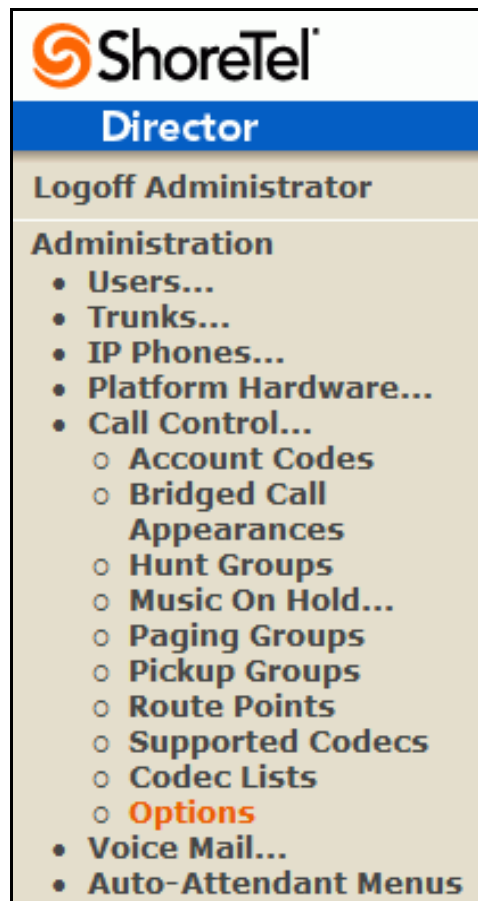
Configuring General Settings

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

- [Configure Call Control on page 16](#)
- [Configure Sites Administration on page 18](#)
- [Configure Switch Administration on page 19](#)

Configure Call Control

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



2. The **Call Control Options** menu In the **Call Control Options** menu, perform the following:
 - a. Ensure the **Enable SIP Session Timer** option is checked.

Call Control Options

[Help](#)

Edit

Edit this record [Refresh this page](#)

General:

Use Distributed Routing Service for call routing.

Enable Monitor / Record Warning Tone.

Enable Silent Coach Warning Tone.

Generate an event when a trunk is in-use for minutes.

Park Timeout (1-100000) after seconds.

Hang up Make Me Conference after minutes of silence.

Delay before sending DTMF to Fax Server: msec

DTMF Payload Type (96 - 127):

SIP:

Realm:

Enable SIP Session Timer.

Session Interval (90 - 3600): sec

Refresher:

Voice Encoding and Quality of Service:

Maximum Inter-Site Jitter Buffer (20 - 400): msec

DiffServ / ToS Byte (0-255): (DSCP = 0x0)

Media Encryption:

Admission control algorithm assumes RTP header compression is being used.

Always Use Port 5004 for RTP (This option is unavailable because your system utilizes SIP Servers, SIP Trunks or SIP Extensions. This feature is incompatible with SIP devices.)

Video Quality of Service:

DiffServ / ToS Byte (0-255): (DSCP = 0x0)

Trunk-to-Trunk Transfer and Tandem Trunks:

Hang up after minutes of silence.

Hang up after minutes.

- b. Set the **Session Interval** timer. The recommended session interval is **1800** seconds.
- c. Select the appropriate refresher (from the drop-down menu) for the **SIP Session Timer**. Set the **Refresher** field to either **Caller (UAC)** [User Agent Client] or **Callee (UAS)** [User Agent Server]. If **Caller (UAC)** is selected, the caller's phone will be in control of the session timer refresh. If **Callee (UAS)** is selected, the phone of the person called will control the session timer refresh.
- d. Uncheck the box for **Always Use Port 5004 for RTP**. For SIP configurations, Dynamic User Datagram Protocol (UDP) must be used for RTP traffic. If the box is unchecked, Media Gateway

Control Protocol (MGCP) will no longer use UDP port 5004; MGCP and SIP traffic will use dynamic UDP ports.



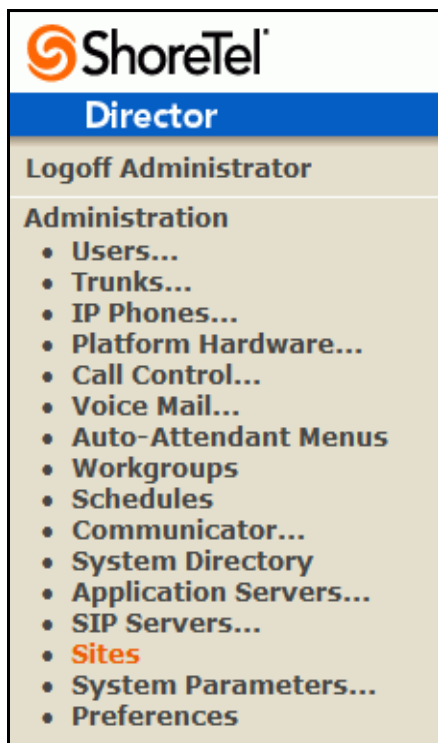
NOTE

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

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

Configure Sites Administration

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



2. In the **Bandwidth** section of the **Sites Edit** menu that appears, perform the following.
 - a. Set the **Admission Control Bandwidth**. The **Admission Control Bandwidth** setting defines the bandwidth available to and from the site. This is important because SIP phones will be counted against the site bandwidth. Bandwidth must be set appropriately based on the site's configuration

with the service provider SIP trunking. Refer to the *ShoreTel Planning and Installation Guide* for more information.

Bandwidth:	
Admission Control Bandwidth:	<input type="text" value="1024"/> kbps
Intra-Site Calls:	<input type="text" value="High Bandwidth Codecs"/>
Inter-Site Calls:	<input type="text" value="Low Bandwidth Codecs"/>
FAX and Modem Calls:	<input type="text" value="Fax Codecs - High Bandwidth"/>
SIP Proxy	

- b. Set the **Intra-Site Calls** and the **Inter-Site Calls** settings next. For the **Intra-Site Calls**, verify that the desired audio bandwidth is selected from the CODEC list for calls within the system. The settings should then be confirmed for the desired audio bandwidth CODEC list for **Inter-Site Calls** (calls between sites).

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

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


Configure Switch Administration

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



2. From the **Voice Switches** menu that appears, select the name of the switch to configure. The **Edit ShoreGear Switch** screen will display.

3. From the **Edit ShoreGear Switch** menu, perform the following:
 - a. Allocate the required ports for the system from the ports available. Each port designated for SIP trunks supports five individual trunks. Each port designated for IP phones supports five SIP phones. At a minimum, one port should be designated for SIP trunks (**5 SIP Trunks**), one port should be designated for IP phones (**5 IP Phones**), one port should be designated for **SIP Trunk with Media Proxy**, and four ports should be designated for **Conference**.
 - b. Select the check box for **Enable Jack Based Music On Hold** if MOH is expected on the SIP trunk.

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

Voice Switches

Edit ShoreGear 90 Switch

New Copy Save Delete Reset

Edit this record Refresh this page

Name:

Description:

Site: Headquarters

IP Address: Find Switches

Ethernet Address:

Server to Manage Switch: Headquarters

Caller's Emergency Service Identification (CESID): (e.g. +1 (408) 331-3300)

Built-in Capacity: IP Phone + SIP Trunk = Total
15 + 15 = 30 of 30 (0 SIP proxy ports)

Enable Jack Based Music On Hold
 Jack Based Music On Hold Gain (-49 to 13): 0 dB

Use Analog Extension Ports as DID Trunks



SG90-HQ

Port	Port Type	Trunk Group	Description	Jack Number
1	Available		P01	
2	Available		P02	
3	SIP Trunk with Media Proxy		P03	
4	5 SIP Trunks		P04	
5	SIP Trunk with Media Proxy		P05	
6	SIP Trunk with Media Proxy		P06	
7	Conference		P07	
8	Conference		P08	

Configuring System Settings

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

- [Configure SIP Trunk Groups on page 21](#)
- [Configure Trunk Services on page 23](#)
- [Configure Individual Trunks on page 25](#)

Configure SIP Trunk Groups

If the SIP trunk groups have already been configured on the system, proceed to the section [Configure Individual Trunks on page 25](#).

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

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



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

Trunk Groups						
Add new trunk group at site: <input type="text" value="Headquarters"/> of type: <input type="text" value="SIP"/> Go						
Name	Type	Site	Trunks	DID	Destination	
Analog Loop Start	Analog Loop Start	Headquarters	0	No	818-5777	
ATT DID Incoming	Digital Wink Start	Headquarters	8	Yes	818-5777	
ATT Pri1 Trk Grp	PRI	Headquarters	46	Yes	818-5777	
ATT T1 2way E&M	Digital Wink Start	Headquarters	8	Yes	818-5777	
bla T1	Digital Wink Start	Headquarters	0	Yes	818-5777	
Digital Loop Start	Digital Loop Start	Headquarters	0	No	818-5777	
Digital Wink Start	Digital Wink Start	Headquarters	8	No	818-5777	

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

4. From the top section of the **Edit SIP Trunks Group** menu, perform the following:
 - a. Enter a name for the trunk group. In the example below, the name **Provider** has been created.

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

*The **Digest Authentication** field is not required when connecting to a NetVanta switch.*

- c. Select your service provider or **Default ITSP** profile from the **Profile** drop-down menu.
5. From the **Inbound** settings at the bottom of the **Edit SIP Trunks Group** menu, perform the following:
 - a. Ensure the **Number of Digits from CO** is set to **10**. Enable the **DNIS** or **DID** parameters as necessary for your installation.

- b. It is not necessary to enable the **Extension** parameter for SIP Trunks, because it defaults to disabled, but it can be enabled if desired (refer to the *ShoreTel Planning and Installation Guide* for further information).
- c. Enable **Tandem Trunking** if you plan to transfer calls to external parties via the SIP trunk.

Configure Trunk Services

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

The screenshot shows the 'Trunk Groups' configuration window for editing a SIP Trunk Group. The window has a title bar with 'Trunk Groups' and a subtitle 'Edit SIP Trunk Group'. At the top right are buttons for 'New', 'Copy', 'Save', 'Delete', 'Reset', and 'Help'. The main content is divided into two sections:

- Outbound:** This section is checked. It includes:
 - Network Call Routing:** Fields for 'Access Code', 'Local Area Code', 'Additional Local Area Codes' (with an 'Edit' button), and 'Nearby Area Codes' (with an 'Edit' button').
 - Billing Telephone Number:** A text field with a placeholder '(e.g. +1 (408) 331-3300)'.
- Trunk Services:** This section contains several checked options:
 - Local
 - Long Distance
 - International
 - Enable Original Caller Information
 - n11 (e.g. 411, 611, except 911 which is specified below)
 - Emergency (e.g. 911)
 - Easily Recognizable Codes (ERC) (e.g. 800, 888, 900)
 - Explicit Carrier Selection (e.g. 1010xxx)
 - Operator Assisted (e.g. 0+)
 - Caller ID not blocked by default

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

Trunk Digit Manipulation:

Remove leading 1 from 1+10D
Hint: Required for some long distance service providers.

Remove leading 1 for Local Area Codes (for all prefixes unless a specific local prefix list is provided below)
Hint: Required for some local service providers with overlay area codes.

Dial 7 digits for Local Area Code (for all prefixes unless a specific local prefix list is provided below)
Hint: Local prefixes required for some local service providers with mixed 7D and 1+10D in the same home area.

Dial in E.164 Format

Local Prefixes: [Go to Local Prefixes List](#)

Prepend Dial Out Prefix:

Off System Extensions:

Translation Table:

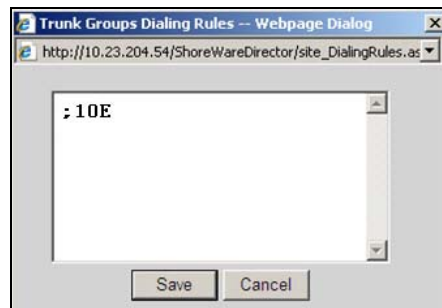
3. After these settings have been configured in the **Edit SIP Trunk Group** menu, select **Save** to accept the changes.
4. This completes the settings necessary to configure the trunk groups on the ShoreTel system. Log out of ShoreTel Director.
5. You will then be presented with the ShoreTel Director login page. From this page, you will enable the **Support Entry** mode of the ShoreTel Director. On your keyboard, hold down the <CTRL> and <Shift> keys and with the mouse pointer click on the **U** of **Username**.



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



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



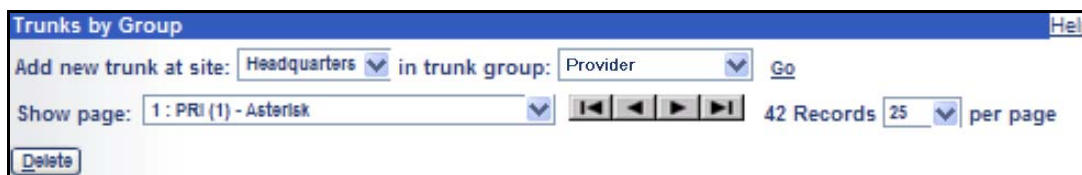
This completes the settings necessary to set up the trunk groups on the ShoreTel system.

Configure Individual Trunks

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



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



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

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

- Select the switch upon which the individual trunk will be created. For the service provider trunk, select **Use IP Address** and enter the IP address of the NetVanta switch.
- Select the number of individual trunks desired, each one supports **one** audio path. For example, if 5 is entered, then 5 audio paths can be up at one time. Once these changes are complete, press **Save** to make the changes.

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

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

Additional Resources

There are additional resources available to aid in configuring your NetVanta switch. Many of the topics discussed in this guide are complex and require additional understanding. The documents listed in *Table 9* are available online at ADTRAN’s Support Forum at <https://supportforums.adtran.com>.

Table 9. Additional ADTRAN Documentation

Feature	Document Title
All AOS Commands Using the CLI	AOS Command Reference Guide
VoIP Setup Wizard	Configuring NetVanta Switches for a VoIP Network
Link Layer Discovery Protocol and Media Endpoint Discovery	Configuring LLDP and LLDP-MED in AOS

Table 9. Additional ADTRAN Documentation

Feature	Document Title
Link Aggregation Control Protocol	<i>Link Aggregation Control Protocol (LACP) in AOS</i>
Rapid Spanning Tree Protocol	<i>Spanning Tree Protocol</i>
Quality of Service and Class of Service on NetVanta Ethernet Switches	<i>Configuring Ethernet Switch QoS and CoS</i>
ActivReach (Extends the reach of Ethernet over existing wiring infrastructure)	<i>ADTRAN NetVanta 1235P/1535P ActivReach Technology Setup and Configuration guide</i>