

Configuration Guide

Configuring Shared Line Appearances over Analog Trunks

This configuration guide explains how to configure shared line appearances (SLAs) on AOS voice products using the web-based graphical user interface (GUI). Steps are also included for creating line keys on an IP phone to present the SLA. Although the steps for creating a shared call appearance (SCA) are similar to creating an SLA, SCA configuration is explained in *Configuring Shared Call Appearances* guide available online at <https://supportforums.adtran.com>. Refer to *Additional Resources on page 27* for links to other documents mentioned throughout this configuration guide.

This guide consists of the following sections:

- *SLA Overview on page 2*
- *Hardware and Software Requirements and Limitations on page 4*
- *Analog Trunk and SLA Configuration on page 4*
- *Additional Resources on page 27*

SLA Overview

An SLA refers to an appearance on a Session Initiation Protocol (SIP) phone that is mapped to an analog trunk on the AOS product. The appearance can be used by multiple phones so that each phone can map to a specific analog trunk. The SIP phone LCD display provides visual indication of the appearance state as it is shared between the phones.

SLAs allows system administrators to enable the key system mode on the unit. An SLA is linked to the IP phone's line keys and functions similar to a key system where the system enables multiple phone users to share the same analog trunk lines. SLAs allow businesses to reduce the cost of providing individual analog phone lines from the carrier to each analog phone station in their facility.

For example, company XYZ has 16 sales employees that need a secondary extension on their desktop ADTRAN IP phones. Instead of ordering an additional 16 trunk lines, company XYZ orders three analog trunk lines and shares the lines between the sales team phone users.

SLAs offer features, such as public hold, retrieve, line status display on subscribers' phones, and the ability to place an outbound call from a selected trunk.

Outlined below are key aspects of SLAs:

- SLAs can only be associated with analog trunks.
- Inbound calls on an SLA notify every SIP-based IP phone that is registered to it.
- SLAs can be seized by pressing the appropriate line key(s) on the phone.
- The status of an SLA is updated on all other phones registered to that line. Status conditions include idle, ringing, busy, and hold.
- A busy SLA cannot be seized by other SIP-based IP phones. Barge or monitoring of the SLA is NOT supported.
- SLAs can have coverage to auto attendant, voicemail, operator, an extension, and an external number.
- Support for special prefix (SPRE) codes on an SLA were added with the release of AOS R10.6.0. All network SPRE codes are supported and sent to the network (if the system is configured to allow network SPRE codes). The local SPRE code *86 (send user to voice mail) is the only local SPRE code supported.



*For more information on configuring network and local SPRE codes (using the **voice spre-mode** command) on AOS products, refer to the *AOS Command Reference Guide* available online at <https://supportforums.adtran.com>.*

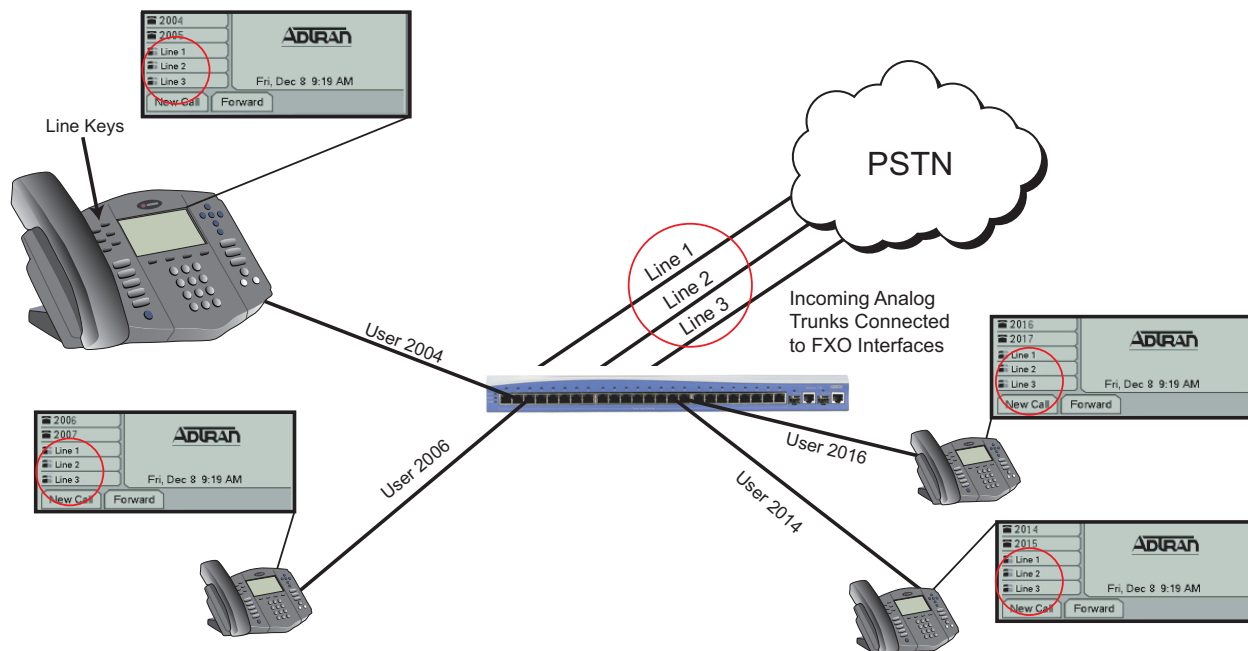


Figure 1. Example SLA Application

SLAs and Analog Trunk Lines

In order for analog trunk lines to appear and be accessible for incoming and outgoing calls for multiple IP phone users, the trunk line(s) must be configured and linked to an SLA. Once the SLA is associated with the trunk line, the SLA can be linked to individual phones. Prior to the introduction of SLAs, all calls were routed out the trunk groups and were only subjected to the permit/deny templates assigned under the trunk group settings. Now, SLAs contain their own set of permit/deny templates. This application allows trunk accounts that are used as SLAs to also be included in an outbound trunk group. Therefore, when a user presses the corresponding line key on a phone that only has SLA extensions programmed, they are subjected to the SLA's accept/reject templates. When running AOS A1.00 firmware or later, the SLA accept/reject templates are not applied to inbound and outbound calls if the phone has a private line (user account) programmed. Instead, the individual user-assigned class of service (CoS) permission settings are applied to all outbound calls.

Using Public Hold

When using SLAs, calls that are put on hold are referred to as being on public hold, which means that every user on that particular SLA has access to the call placed on hold. Also, any phone registered to that SLA will be able to see the hold status and retrieve the call by pressing the line key that corresponds to the call.



Using SLAs and park zones in status groups on the same phone is not supported.

Hardware and Software Requirements and Limitations

The SLA feature was introduced in AOS 15.6 and emulates key system functionality. It is available on AOS products as outlined in the *AOS Feature Matrix*, available online at <http://supportforums.adtran.com>.

AOS firmware version 15.6 or later is required on your system in order to use the SLA trunk features.

AOS firmware version A1.00 or later includes the updated SLA user-based permissions and support for attended and unattended (blind) transfers. In previous versions of the firmware, users could only execute blind transfers. When using firmware version A1.00 or later, users can accompany a call during a transfer (attended transfer).

Support for SLA primary line seizure is only supported on the following IP phone models and the firmware version noted in parenthesis:

- Polycom SoundPoint IP 450 (version 4.0.x)
- ADTRAN/Polycom IP 550 (version 4.0.x)
- ADTRAN/Polycom IP 650 (version 4.0.x)
- Polycom SoundPoint IP 670 (version 4.0.x)
- ADTRAN IP 706 (version 2.3.0)
- ADTRAN IP 712 (version 2.3.0)

The configurable ring options, such as silent ring, delay ring, or ring immediate, are only supported on Polycom IP phone models.



*The configuration parameters used in the examples are for instructional purposes only. Please replace all example settings with specific parameters to configure your application. For detailed information on specific settings, refer to the **Settings and Description** sections following each menu.*



Default router configuration is loaded in the unit prior to factory release. Confirm that your router's Ethernet interfaces and IP addresses are configured properly. Standard router configuration must be completed before continuing with SIP voice configuration.

Analog Trunk and SLA Configuration


When configuring an analog trunk and an SLA, steps 1 and 2 are required. Step 3 is optional since ring options are only available on Polycom phone models. Select either Step 4 or 5, depending on your IP phone model. Finally, Step 6 is required for either phone model. Use the following steps to implement SLA:

- *Step 1: Configuring Analog Trunks on page 5*
- *Step 2: Configuring SLAs on page 14*
- *Step 3: Creating Ring Options (Optional) (Polycom Phones Only) on page 19*
- *Step 4: Assigning an SLA to a Polycom Phone Line Key(s) on page 21*
- *Step 5: Assigning an SLA to an ADTRAN Phone Line Key(s) on page 24*
- *Step 6: Confirming Shared Line Key Configuration on page 27*

Accessing the GUI

To access the GUI and begin configuring this feature, follow these steps:

1. Enter your AOS product’s IPv4 address in the browser’s address field in the format **http://<ip address>/admin**, for example:
http://10.10.10.1/admin
2. At the prompt, enter your user name and password and select **OK**.



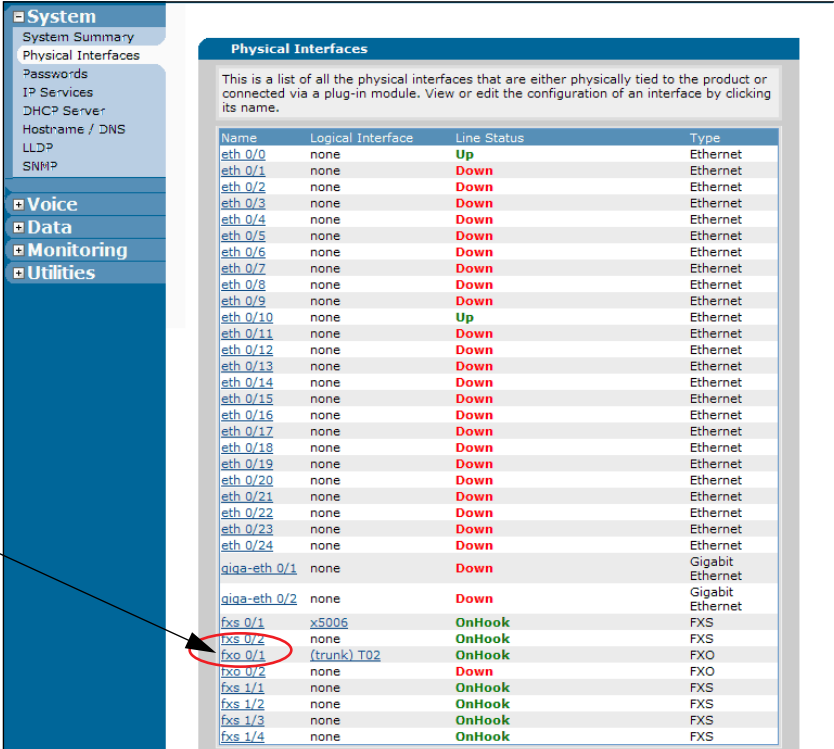
*The default user name is **admin** and the default password is **password**.*

Step 1: Configuring Analog Trunks

First, configure the physical interface. Foreign exchange office (FXO) interfaces are enabled by default, but the interfaces can be verified in the **Physical Interfaces** menu. Once the physical interface is configured, a trunk account must be created in order to make and receive calls. Create the trunk account and assign the FXO port(s), making sure the analog FXO settings (trunk number, supervision, etc.) match the parameters set by your service provider in this configuration.

Configuring the FXO Physical Interface

1. Navigate to the **System > Physical Interfaces** menu to display a list of physical interfaces. Select the FXO interface that provides a trunk line to the system.



This is a list of all the physical interfaces that are either physically tied to the product or connected via a plug-in module. View or edit the configuration of an interface by clicking its name.

Name	Logical Interface	Line Status	Type
eth 0/0	none	Up	Ethernet
eth 0/1	none	Down	Ethernet
eth 0/2	none	Down	Ethernet
eth 0/3	none	Down	Ethernet
eth 0/4	none	Down	Ethernet
eth 0/5	none	Down	Ethernet
eth 0/6	none	Down	Ethernet
eth 0/7	none	Down	Ethernet
eth 0/8	none	Down	Ethernet
eth 0/9	none	Down	Ethernet
eth 0/10	none	Up	Ethernet
eth 0/11	none	Down	Ethernet
eth 0/12	none	Down	Ethernet
eth 0/13	none	Down	Ethernet
eth 0/14	none	Down	Ethernet
eth 0/15	none	Down	Ethernet
eth 0/16	none	Down	Ethernet
eth 0/17	none	Down	Ethernet
eth 0/18	none	Down	Ethernet
eth 0/19	none	Down	Ethernet
eth 0/20	none	Down	Ethernet
eth 0/21	none	Down	Ethernet
eth 0/22	none	Down	Ethernet
eth 0/23	none	Down	Ethernet
eth 0/24	none	Down	Ethernet
giga-eth 0/1	none	Down	Gigabit Ethernet
giga-eth 0/2	none	Down	Gigabit Ethernet
fxs 0/1	x5006	OnHook	FXS
fxs 0/2	none	OnHook	FXS
fxo 0/1	(trunk) T02	OnHook	FXO
fxo 0/2	none	Down	FXO
fxs 1/1	none	OnHook	FXS
fxs 1/2	none	OnHook	FXS
fxs 1/3	none	OnHook	FXS
fxs 1/4	none	OnHook	FXS

Settings and Descriptions

Name displays the identification of the physical interface, for example, **fxo 0/1**. Selecting the name of the physical interface accesses the interface's configuration mode.

Logical Interface displays a description of the logical interface that is linked to the physical interface. For example, the physical interface **fxo 0/1** has a logical interface of **(trunk) T02**.

Line Status displays the network status of the physical interface.

Type displays the technical type of the physical interface.

2. Configure the FXO interface by applying the appropriate settings and selecting **Apply**. After selecting **Apply**, the FXO interface is configured. Next, the trunk account for the interface should be created and configured.

The screenshot shows a configuration window titled "Configuration for fxo 0/1". It contains the following elements:

- A header bar with the title "Configuration for fxo 0/1".
- Instructional text: "Basic configuration for the fxo ports. Use the select boxes below to quickly apply this port's settings to multiple ports."
- Selection controls: Two checkboxes, the first for "0/1" (checked) and the second for "0/2" (unchecked). Below them are links for "Select All" and "Unselect All".
- Configuration fields:
 - Description:** A text input field.
 - Enable:** A checked checkbox.
 - Receive Gain:** A numeric input field set to "+0.0".
 - Transmit Gain:** A numeric input field set to "+0.0".
 - Impedance:** A dropdown menu currently showing "600 Ohm".
- Help text for each field:
 - Description label (optional)
 - Enable or disable this interface
 - Set receive gain (-6.0 to 10.0)
 - Set transmit gain (-6.0 to 10.0)
 - Set the terminating impedance for the interface
- Buttons: "Reset" and "Apply" at the bottom.

Settings and Descriptions

Description (optional) specifies a descriptive label for this interface up to 80 alphanumeric characters.

Enable this interface using the check box.

Receive Gain specifies the receive gain (-6.0 to 10.0 dB) to adjust the receiver sensitivity. Receive gain determines the amplification of the received signal before transmitting out the FXO interface. By default, the receive gain is +0.0 dB.

Transmit Gain specifies the transmit gain (-6.0 to 10.0 dB). Transmit gain determines the amplification of the received signal before transmitting from the FXO interface toward the network. By default, the transmit gain is +0.0 dB. Increasing the value will increase the volume of the signal being transmitted.

Impedance specifies the terminating impedance for the interface. Select the appropriate setting from the drop-down menu.



Before changing **Impedance** values, or for more information about **Impedance** settings, refer to *Analog Configuration and Trunk Troubleshooting* guide available online at <https://supportforums.adtran.com>.

Configuring the Analog Trunk Account

- From the main menu, navigate to **Voice > Trunks > Trunk Accounts** to access the **Add/Modify/Delete Trunk Accounts** menu. To create a new trunk account, enter the desired **Trunk Name** in the appropriate field, and select **Analog** from the trunk **Type** drop-down menu. Select the appropriate **Supervision** setting from the drop-down menu. This setting is provided by your service provider. Next, specify the trunk **Role**. By default, the trunk role is **User** (**Network** is not valid for analog trunks). Select the **Add** button to add the trunk account to the unit, and the **Edit Trunk** menu will appear.

Trunk Name	ID	Type	Supervision	Role	
T01	T01	Analog	Loop Start	User	Delete

Settings and Descriptions

Trunk Name specifies a name for this trunk.

Type specifies the type of trunk to create.

Supervision specifies the type of signaling on the incoming trunk. Obtain this setting from the service provider of the analog trunk.

Role specifies the type of line termination for this trunk. Only the **User** role is valid on analog trunks.

- User** terminates the line at the provider (i.e., the unit acts as a user). Analog trunks can only be terminated in the **User** role.
- Network** terminates the line at the unit (i.e., the unit acts as the network). Use **Network** role for configuring trunks that transmit out to a DSX port (not valid for analog trunks).

- In the **Edit Trunk** menu, edit the trunk account information to match the settings given by your service provider.

Use this dialog to modify the Trunk Account configuration.

Trunk Account Information

Trunk ID: T01

Type: Analog

Supervision: Loop Start

Role: User

Trunk Name: Analog Trunk T01

System Mode	Trunk Number
Default	<None>
Night	<Same as Default>
Lunch	<Same as Default>
Weekend	<Same as Default>
Override	<Same as Default>
Custom1	<Same as Default>
Custom2	<Same as Default>
Custom3	<Same as Default>

Trunk Number:

Blind Dialing:

Caller ID:

Reject External:

Disconnect Tone Busy: Delay: 10

Disconnect Release Delay: 12

DID(Direct Inward Dialing):

DID Digits: 1

DID Prefix:

Resource Selection: Circular Hunt Descending

Emergency Caller ID Override: Use Match-Substitution:

Inbound Caller ID Override:

Inbound Caller ID Override Method: Always

Settings and Descriptions

Trunk Number specifies the number to which inbound calls are routed when direct inward dialing (DID) is disabled on the trunk. Trunk numbers and inbound call behaviors are defined by system modes. For more information about system modes and their operation or configuration, refer to quick configuration guide *Configuring System Modes in the NetVanta 7000 Series* online at <https://supportforums.adtran.com>.

Blind Dialing enables calls to be dialed even without the presence of a dial tone.

Caller ID displays inbound caller ID on this trunk.

Reject External rejects trunk-to-trunk calls.

Disconnect Tone Busy enables and specifies the delay in seconds the trunk will wait to disconnect after release tones have been detected. Valid range is **1** to **120**.

DID enables direct inward dialing (DID) for the SLA trunk.

- DID Digits** (optional) specifies the number of inbound DID digits expected on this trunk.
- DID Prefix** (optional) specifies the prefix added to the DID digits received on this trunk. Valid DID prefix characters are digits **0** to **9**, with no special characters.

Resource Selection specifies how resources will be used by the switchboard during call routing. By default, call routing uses **Circular Hunt Ascending** as the resource selection method. This method divides calls evenly between unused ports, starting with the lowest port first.



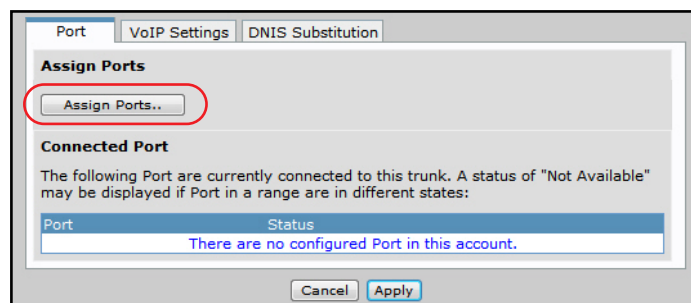
***Resource Selection** is **not** a valid setting for SLA trunks. Each SLA uses one port on an FXO trunk, so this setting should remain at the default.*

Emergency Caller ID Override overrides the caller ID number on emergency calls that are outbound on this trunk with the specified value.

Inbound Caller ID Override overrides the caller ID number on inbound calls on this trunk with the specified value.

Inbound Caller ID Override Method specifies the behavior of the **Inbound Caller ID Override** as one of the two following options:

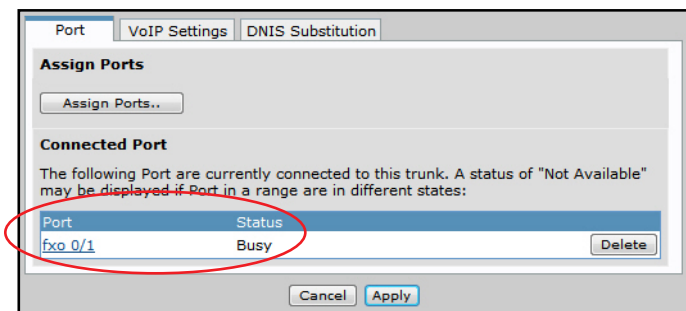
- **Only If Not Present** inserts the configured caller ID override value only if there is no caller ID information present in the call information.
 - **Always** replaces caller ID information with the configured caller ID override value at all times.
3. After specifying the trunk's configuration, associate the trunk account with a port. To do this, select the **Port** tab at the bottom of the **Edit Trunk** menu and select the **Assign Ports** button.



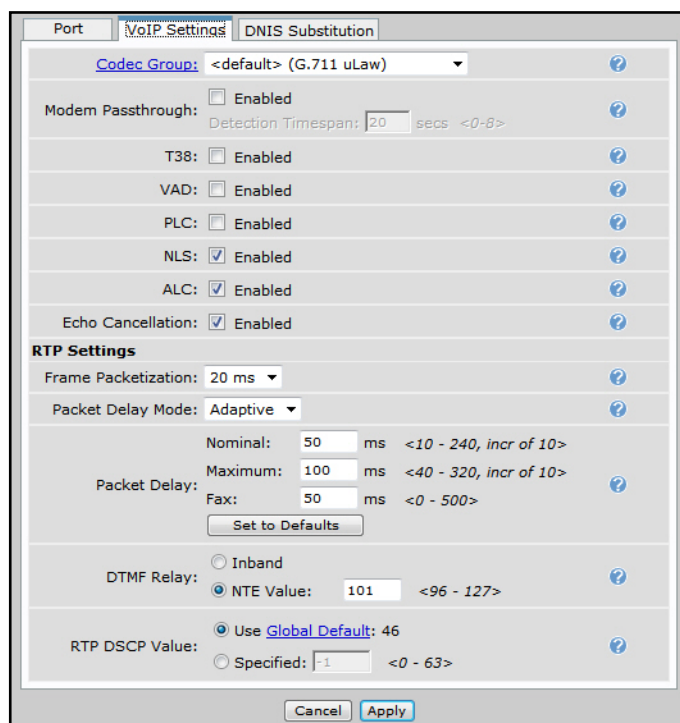
4. The **Add Ports to Trunk Account** menu appears. From this menu, select one FXO interface to add to the trunk account. Adding ports from this menu associates the logical interface of the trunk with the physical interface of the port. Select **Add Ports** to apply the setting and return to the **Edit Trunk** menu. If you need to clear a selection, select **Clear Selections**.



5. Once the ports are added, they appear in the **Port** tab of the **Edit Trunk** menu.



6. After specifying the ports associated with the trunk, you can optionally configure the trunk user's Voice over IP (VoIP) and Realtime Transport Protocol (RTP) settings. To change these settings, select the **VoIP Settings** tab at the bottom of the **Edit Trunk** menu.



Settings and Descriptions

Codec Group specifies the coder-decoder (CODEC) group to use for the trunk. This setting is normally used for SIP trunks.

Modem Passthrough enables automatic renegotiation with the far end to be modem compatible if the unit detects a modem or fax tone (switch to G.711, all voice improvements turned off, packet delay set to Fax).

- **Detection Timespan** specifies the time (in seconds) that the unit attempts to detect a modem or fax tone. Timespan range is **0** to **8** seconds.

T.38 enables automatic renegotiation with the far end to be T.38 compatible if an existing call on the trunk detects a fax tone.

VAD voice activity detection (VAD) transmits only audible speech over the network (not silence). VAD slightly degrades the sound quality, but the connection uses much less bandwidth.

PLC enables packet loss concealment (PLC). When enabled, the unit will try to reconstruct sound lost from dropped packets.

NLS enables the echo canceller's non-linear suppression (NLS). When enabled, acoustic echo should be reduced.

ALC enables the automatic leveling control (ALC). When enabled, ALC reduces received RTP signals to a predefined level.

Echo Cancellation cancels reflected noise from the transmitted voice signal. Normally, echo cancellation should only be disabled if the voice station is connected to a fax machine or modem.

Frame Packetization selects the number of audio samples in milliseconds (1 frame/sample is 10 ms) included in a single RTP packet. Select the frame packetization rate from the drop-down menu, choosing either **10**, **20**, or **30** ms.

Packet Delay Mode configures the operation mode of the jitter buffer for VoIP calls on this trunk account. You can set the packet delay mode by choosing either **Adaptive** or **Fixed** from the drop-down menu.

- **Adaptive** starts the buffer's delay at the **Nominal** delay setting and increases it up to the **Maximum** delay setting if an unacceptable number of packets are being discarded due to jitter. The buffer decreases the amount of delay when possible.
- **Fixed** causes the buffer's delay to remain at the **Nominal** setting at all times.

Packet Delay specifies the **Nominal**, **Maximum**, and **Fax** packet delay settings.

- **Nominal** specifies the packet delay mode. For voice calls, the nominal delay value represents the desired amount of packet delay. In **Adaptive** mode, the buffer may increase this value up to the maximum delay. In **Fixed** mode, the delay is constantly set at this value. By default, the **Nominal** packet delay is set to **50** ms. The **Nominal** delay range is **10** to **240** ms, in increments of **10** ms.
- **Maximum** specifies the maximum packet delay mode. For voice calls, the maximum delay value represents the maximum delay to which the adaptive jitter buffer can increase. By default, the **Maximum** packet delay is set to **100** ms. The **Maximum** delay range is **40** to **320** ms, in increments of **10** ms.
- **Fax** specifies the fax packet delay mode. If **Modem Passthrough** is enabled and modem/fax tones are detected, the packet delay setting is switched to this value. By default, the **Fax** packet delay is set to **50** ms. The **Fax** delay range is **0** to **500** ms.

DTMF Relay specifies how dual-tone multi frequency (DTMF) tones are to be transmitted over RTP. Select either **Inband** or out-of-band (**NTE Value**). The **NTE Value** range is **96** to **127**.

RTP DSCP Value specifies the differentiated services (DiffServ) code point (DSCP) for this station's RTP packets. Either use the global default (which is subject to change as the global default setting changes) or specify a static value for this station only. The valid range for the static value is **0** to **63**.

7. Once the trunk's VoIP settings have been configured, select **Apply** to apply the settings.

8. After configuring the trunk's VoIP settings, you can also optionally specify dialed number identification service (DNIS) substitution settings for the trunk by selecting the **DNIS Substitution** tab at the bottom of the **Edit Trunk** menu. DNIS substitution is used to replace a dialed number with a specific number of your choice, and if it is supported by the unit, the name of the called party can be replaced with a name of your choice. Multiple **DNIS Substitution** entries can be added to each trunk, and each configured entry appears in a list under **Current DNIS Substitution Entries**. The order of entry input is important because the first valid match that is found for outbound numbers is used and no other entries are processed.



For more information about DNIS configuration, refer to *Enhanced ANI and DNIS Substitution in AOS Voice Products* available online at <https://supportforums.adtran.com>.

Settings and Descriptions

Match Number specifies the dialed number that you want to match.

Substitution Number specifies the number that will be sent in place of the number that was matched.

Example Substitutions

Function: Format a call for 10-digit dialing.

Match Number: NXX-XXXX

Substitution Number: 256-NXX-XXXX

Function: Format a long distance call for 10-digit dialing.

Match Number: 1-NXX-XXX-XXXX

Substitution Number: NXX-XXX-XXXX

Function: Insert a long distance call PIC code for a particular carrier.

Match Number: 1-NXX-NXX-XXXX

Substitution Number: 10-10-220-NXX-NXX-XXXX

Function: Redirect 411 information calls.

Match Number: 411

Substitution Number: 256-555-1212

Examples of Wildcard Characters

0 - 9 Match the exact digit(s) only.

X Match any single digit 0 through 9.

N Match any single digit 2 through 9.

\$ Match any number string dialed. This wildcard must occur at the end of the pattern.

- [] Match any digit in the list within the brackets. For example, [1,4,6] matches 1, 4, and 6 only, while [1-3,5] matches 1 through 3 and 5.
- , () Formatting characters that are ignored but allowed.
- Use within brackets to specify a range, otherwise ignored.



All wildcards in the substitution pattern must have the same relative position in the match pattern.

9. Once you have specified the DNIS information, select **Add Substitution**. The substitution entry appears in the entry list.
10. After you have created all the DNIS substitution entries necessary, select **Apply** to apply the DNIS settings to the trunk.

Step 2: Configuring SLAs

1. Navigate to **Voice > Trunks > Shared Line Accounts** to create an SLA for each connected analog trunk account (FXO interface). Enter the **Name** and select the **Associated Trunk** from the drop-down menu. Next, select the **Add New Shared Line Account** button to create the SLA, and begin SLA configuration in the **Shared Line Account** menu.

Navigate to **Shared Line Accounts** to create an SLA.

Enter a name for the new SLA.

Select **Add New Shared Line Account** to create the SLA.

Settings and Descriptions

Name specifies a name for this SLA. The **Name** cannot be changed once the line is created. The SLA must be deleted and recreated to change the **Name**.

Associated Trunk specifies the trunk to be associated with this line. This selection is limited to analog trunks, and the same trunk cannot be used with multiple SLAs. Trunks mapped to a shared line account will not participate in trunk group selections. In addition, selecting the **SCA** check box will deselect a trunk for creating the SLA. For more information about shared call appearances (SCAs), refer to *Configuring Shared Call Appearances* available online at <https://supportforums.adtran.com>.

2. Once you have selected **Add New Shared Line Account**, you can configure the basic information for the SLA from the **Shared Line Account** menu. The **Name** and **Associated Trunk** fields include the

information entered in the previous menu. In this menu, you can enter a description for the SLA, specify the **Line Seize Timeout** period, specify the **SIP Auth Password**, and provide an extension for the **Held Call Pickup Number**.

The screenshot shows the 'Shared Line Account "SalesGroup"' configuration page. It includes a header with the account name and a sub-header 'Basic Information'. Below this, there are several fields: 'Name' (SalesGroup), 'Description' (Line 2), 'Associated Trunk' (T02 - Analog Trunk T02), 'Line Seize Timeout' (15 seconds), 'SIP Auth Password' (masked with dots), and 'Held Call Pickup Number' (6000). Each field has a help icon to its right.

Settings and Descriptions

Description optionally specifies a short text description for this SLA.

Line Seize Timeout specifies the maximum time the trunk can be seized before a call is placed. By default, the line seize timeout is set to **30** seconds. The valid line seize timeout range is **5** to **120** seconds.

SIP Auth Password specifies the four-digit password used to authenticate SIP phone registrations. If this password is changed after the phone configuration files have been created, those phones will not be able to register until they are updated with the new password value. By default, the SIP authentication password is **1234**.



*If the **SIP Auth Password** is changed after phone configuration files have been created, those phones will not be able to register until they are updated with the new password value.*

Held Call Pickup Number specifies a number for private extensions to use to pick up calls held on this shared line.

3. After specifying the basic SLA information, you can optionally specify the permitted call types for this shared line using the **Accept/Reject Templates** tab at the bottom of the **Shared Line Account** menu. Select the permitted call types for this shared line by selecting the check box next to the specific call type.

Accept/Reject Templates | Call Coverage | VoIP Settings

Check the appropriate boxes below to allow specific call types to be dialed using this shared line account.

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

[+ Detailed View - Accept/Reject Call Templates](#) [?](#)

Cancel Apply



To allow local calls on this SLA, the local dialing type must be specified in the dial plan settings as shown above. When the local dialing type has been defined, it is displayed next to the **Local Calls** option as either 7 or 10 digits. For more information about defining the local dialing type and configuring dial plans, refer to the [Configuring the Switchboard and Dial Plan in AOS](#) available online at <https://supportforums.adtran.com>.



When running AOS A1.00 firmware or later, the SLA accept/reject templates are not applied to inbound and outbound calls if the phone has a private line (user account) programmed. Instead, the individual users assigned CoS permission settings are applied to all outbound calls.

- The details of the configured accept and reject templates can be viewed by selecting **Detailed View-Accept/Reject Call Templates**.

Detailed View - Accept/Reject Call Templates [?](#)

Accept Templates [?](#)

<input checked="" type="checkbox"/>	Accept Template
<input type="checkbox"/>	1800NXXXXXX
<input type="checkbox"/>	1855NXXXXXX
<input type="checkbox"/>	1866NXXXXXX
<input type="checkbox"/>	1877NXXXXXX
<input type="checkbox"/>	1888NXXXXXX
<input type="checkbox"/>	911
<input type="checkbox"/>	NXXXXXX

Reject Templates [?](#)

<input checked="" type="checkbox"/>	Reject Template
<input type="checkbox"/>	976XXXX

Add Accept Template Add Reject Template

Delete Templates

- After you have configured the accept and reject templates for the SLA, you can configure the call coverage for the SLA. The call coverage specifies how calls are handled when they go unanswered after

a specified number of rings. To configure the call coverage, select the **Call Coverage** tab at the bottom of the **Shared Line Account** menu.

Settings and Descriptions

Action specifies the coverage for calls. The action items are evaluated in the order selected and displayed on the **Call Coverage** tab. The following actions are available to define the call coverage:

- **Busy Signal** sends a busy signal to the calling party after ringing the SLA.
- **Go to Auto Attendant** routes calling party to the auto attendant. A list box will appear to the right of this selection displaying the available auto attendants.
- **Ring Extension** rings the extension of your choice. Enter the extension in the text box that appears to the right of this selection. Specify the number of times to ring the extension in the **Ring the Extension** column.
- **Ring Operator** rings the operator. Specify the number of times to ring the operator in the **Ring the Extension** column.
- **Ring External Number** rings the external number of your choice. Enter the external number in the text box that appears to the right of this selection.

Ring the Extension column specifies the number of times to ring the extension for the specified action. If there is no response after the specified number of rings (or the extension is busy), the next action in the call coverage list is attempted. If a value of **0** is used, the call coverage list is only processed if the extension is busy. Otherwise, the phone will ring indefinitely.



For additional information about configuring call coverage and understanding how call coverage works with system modes, refer to [Configuring User Accounts on the NetVanta 7000 Series](https://supportforums.adtran.com) available online at <https://supportforums.adtran.com>.

- After specifying the call coverage for the SLA, you can optionally configure the VoIP settings for the SLA. To configure VoIP settings, select the **VoIP Settings** tab at the bottom of the **Shared Line Account** menu.

Settings and Descriptions

Codec Group specifies the coder-decoder (CODEC) group to use for the SLA.

VAD voice activity detection (VAD) transmits only audible speech over the network (not silence). VAD slightly degrades the sound quality, but the connection uses much less bandwidth.

PLC enables packet loss concealment (PLC). When enabled, the unit will try to reconstruct sound lost from dropped packets.

NLS enables the echo canceller's non-linear suppression (NLS). When enabled, acoustic echo should be reduced.

ALC enables the automatic leveling control (ALC). When enabled, ALC reduces received RTP signals to a predefined level.

Echo Cancellation cancels reflected noise from the transmitted voice signal. Normally, echo cancellation should only be disabled if the voice station is connected to a fax machine or modem.

Frame Packetization selects the number of audio samples in milliseconds (1 frame/sample is 10 ms) included in a single RTP packet. Select the frame packetization rate from the drop-down menu, choosing either **10**, **20**, or **30** ms.

Packet Delay Mode configures the operation mode of the jitter buffer for VoIP calls on this SLA. You can set the packet delay mode by choosing either **Adaptive** or **Fixed** from the drop-down menu.

- **Adaptive** starts the buffer's delay at the **Nominal** delay setting and increases it up to the **Maximum** delay setting if an unacceptable number of packets are being discarded due to jitter. The buffer decreases the amount of delay when possible.
- **Fixed** causes the buffer's delay to remain at the **Nominal** setting at all times.

Packet Delay specifies the **Nominal**, **Maximum**, and **Fax** packet delay.

- **Nominal** specifies the packet delay mode. For voice calls, the nominal delay value represents the desired amount of packet delay. In **Adaptive** mode, the buffer may increase this value up to the maximum delay. In **Fixed** mode, the delay is constantly set at this value. By default, the **Nominal** packet delay is set to **50** ms. The **Nominal** delay range is **10** to **240** ms, in increments of **10** ms.
- **Maximum** specifies the maximum packet delay mode. For voice calls, the maximum delay value represents the maximum delay to which the adaptive jitter buffer can increase. By default, the **Maximum** packet delay is set to **100** ms. The **Maximum** delay range is **40** to **320** ms, in increments of **10** ms.
- **Fax** specifies the fax packet delay mode. If **Modem Passthrough** is enabled and modem/fax tones are detected, the packet delay setting will be switched to this value. By default, the **Fax** packet delay is set to **50** ms. The **Fax** delay range is **0** to **500** ms.

DTMF Relay specifies how dual-tone multi frequency (DTMF) tones are to be transmitted over RTP. Select either **Inband** or out-of-band (**NTE Value**). The **NTE Value** range is **96** to **127**.

RTP DSCP Value specifies the DiffServ code point (DSCP) for this station's RTP packets. Either use the global default (which is subject to change as the global default setting changes) or specify a static value for this station only. The valid range for the static value is **0** to **63**.

7. After configuring the SLA's basic settings, accept and reject templates, call coverage, and VoIP settings, select **Apply** at the bottom of the **Shared Line Account** menu to append new settings.
8. Repeat the steps outlined in *Step 2: Configuring SLAs on page 14* to add additional SLAs to the system.

Step 3: Creating Ring Options (Optional)

(Polycom IP Phones Only)

Ring options can be configured and applied to Polycom phones to ring with a different behavior depending upon the system mode.

1. From the main menu, navigate to **Voice > System Setup > Ring Options**.
2. Enter a name for the ring option and a description. Select **Add** to create the ring option.

Enter the name of the ring option.

Enter a description.

Select **Add** to create the ring option.

3. Select a ring type next to a **System Mode** using the drop-down menu. When the system mode is entered (either manually or by the schedule), the ring type becomes active. The default ring type is **immediate**. If no other ring type is selected, the default is used. Select **Apply** to accept the changes.

Select the drop-down menu to choose a ring type for the system mode for this ring option.

Select **Apply** to accept the changes.

Menu Settings and Descriptions

same as default uses the default setting.

immediate rings immediately.

silence provides visual ring indication only, no sound.

delay-12-second rings after a 12 second delay.

delay-24-second rings after a 24 second delay.

4. The newly created ring option appears in the **Add / Modify / Delete Ring Options** menu. It can now be applied to an SLA on a Polycom phone using the instructions below.

The new ring option appears in the **Add / Modify / Delete Ring Options** menu.

Step 4: Assigning an SLA to a Polycom Phone Line Key(s)

The following instructions pertain only to Polycom or ADTRAN/Polycom phones. If you are assigning an SLA to an ADTRAN IP 700 Series phone, refer to the instructions in *Step 5: Assigning an SLA to an ADTRAN Phone Line Key(s) on page 24*.

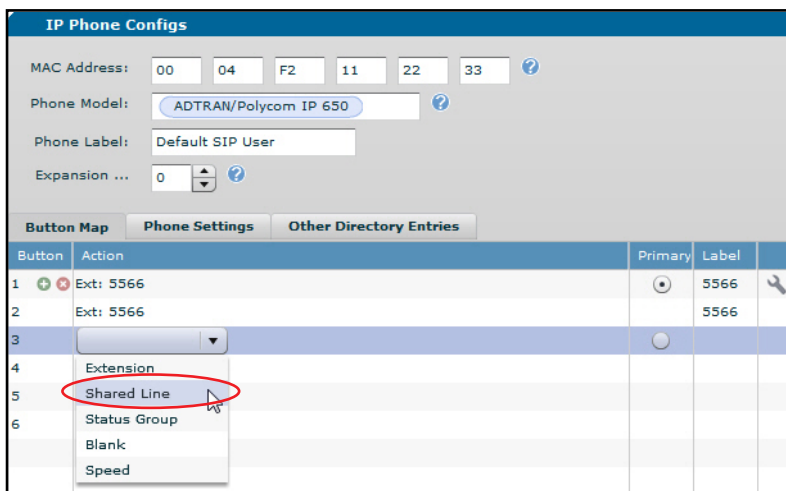
1. From the main menu, navigate to **Voice > IP Phone Configs**. Select the configuration you want to edit by selecting the check box next to the phone in the list and selecting **Edit** or by double-clicking on the appropriate row in the table.

	MAC Address	Associated Account	Registered IP	Phone Model
<input checked="" type="checkbox"/>	00:04:F2:11:22:33	5566	<Not Registered>	ADTRAN/Polycom IP 650
<input type="checkbox"/>	00:A0:C8:00:99:A5	2031 6850	<Not Registered>	ADTRAN IP 706
<input type="checkbox"/>	00:A0:C8:00:99:A9	2003 6850	<Not Registered>	ADTRAN IP 706

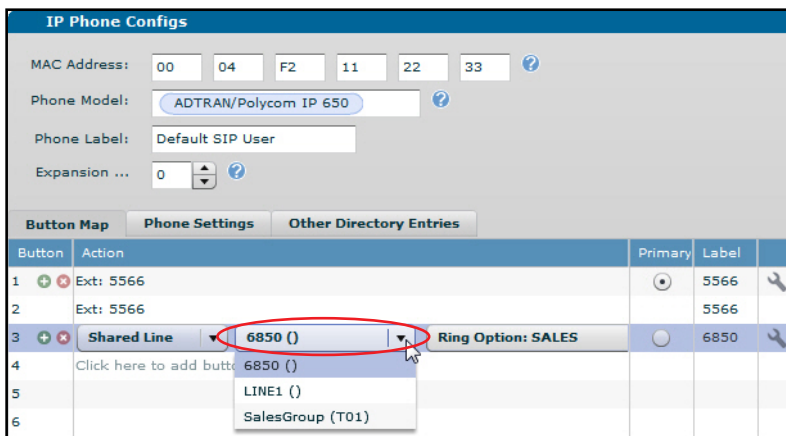
2. Highlight the **Click here to add button** row, and double-click in the **Action** column.

Button	Action	Primary	Label
1	Ext: 5566	<input checked="" type="radio"/>	5566
2	Ext: 5566	<input type="radio"/>	5566
3	Click here to add button	<input type="radio"/>	
4		<input type="radio"/>	
5		<input type="radio"/>	

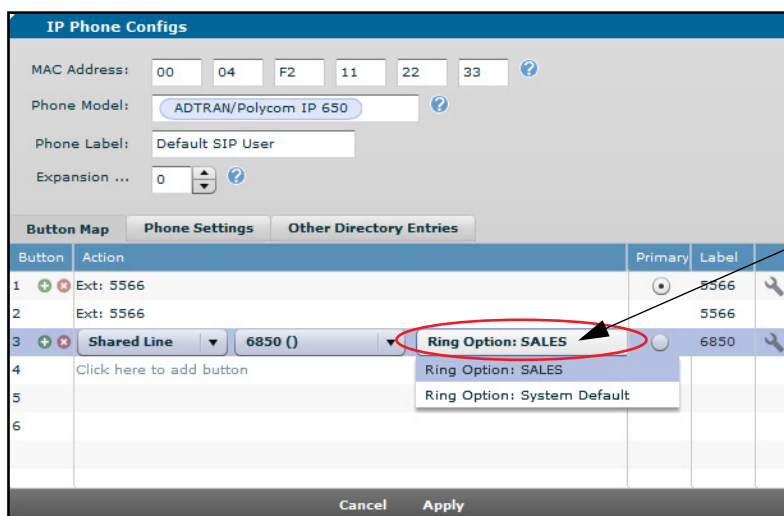
- From the drop-down menu, select **Shared Line**.



- Select the shared line from the drop-down menu.

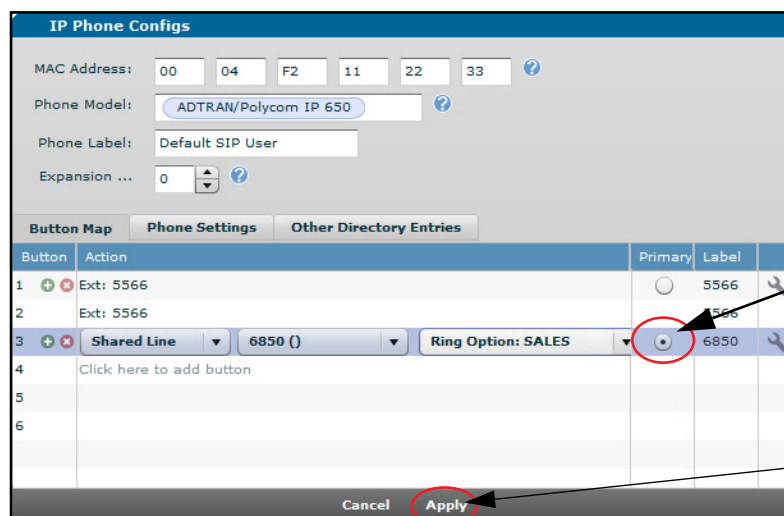


- Select the ring option from the final drop-down menu to apply to the SLA. These choices depend on the ring options configured and are only available on Polycom phones. Refer to *Hardware and Software Requirements and Limitations on page 4* for a complete list of supported phone models.



Select ring option from the drop-down menu to apply to the SLA.

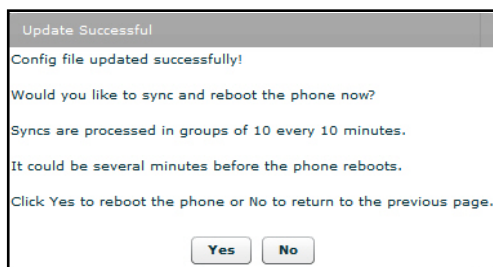
- After assigning an SLA to a line key, select one of the line keys as **Primary**. Designating an SLA line as **Primary** will seize the SLA first (if available) when the phone goes off hook. The default is for the private line to be **Primary**. Select the **Primary** radio button on the appropriate line. Select **Apply** when finished. (For IP phone firmware versions that support this feature, refer to *Hardware and Software Requirements and Limitations on page 4.*)



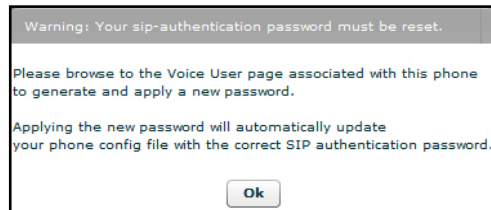
Select the **Primary** radio button to seize the associated line when the phone goes off hook.

Select **Apply** to accept the changes.

- Once you have selected **Apply**, the system will prompt you to reboot the phone. Select **Yes** to reboot the phone. The new SLAs will appear on the phone after the reboot process.



You may receive a warning from the system instead, advising you to reset the SIP authentication password. This occurs if you changed the SIP authentication password and requires you to follow the instructions. Select **OK** to continue. Then the prompt to reboot the phone is displayed. Select **Yes** to reboot the phone.



- Repeat *Step 4: Assigning an SLA to a Polycom Phone Line Key(s) on page 21* to add additional SLAs to a phone. Most phone models allow multiple line registrations on a single phone. The maximum number of line registrations depends upon the phone model.

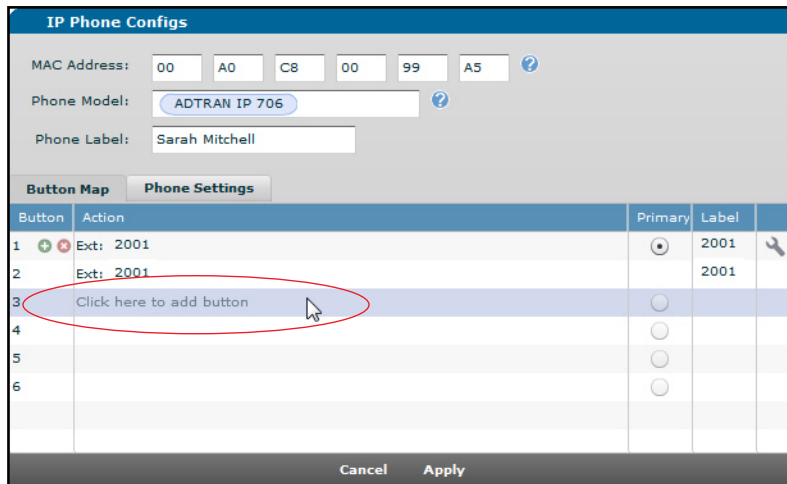
Step 5: Assigning an SLA to an ADTRAN Phone Line Key(s)

The following instructions pertain only to ADTRAN IP 700 Series phones. They do not support the ring option feature. If you are assigning an SCA to a Polycom or ADTRAN/Polycom model phone, refer to the instructions in *Step 4: Assigning an SLA to a Polycom Phone Line Key(s) on page 21*.

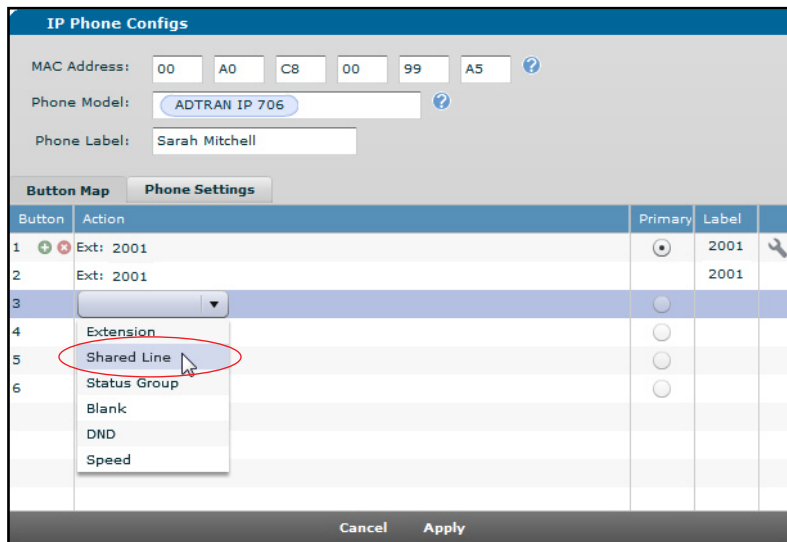
- From the main menu, navigate to **Voice > IP Phone Configs**. Select the configuration you want to edit by selecting the check box next to the phone in the list and selecting **Edit** or by double-clicking on the appropriate row in the table.

System		IP Phone Configs				
Voice		New	Edit	Delete	Refresh	Other Actions
Stations		<input type="checkbox"/>	MAC Address	Associated Accounts	Registered IP	Phone Model
User Accounts		<input type="checkbox"/>	00:A0:C8:25:53:28	2002	<Not Registered>	ADTRAN IP 706
IP Phone Configs		<input checked="" type="checkbox"/>	00:A0:C8:31:58:27	2001	10.10.20.5	ADTRAN IP 706
IP Phone Globals		<input type="checkbox"/>	00:A0:C8:4A:5A:70	2000	10.10.20.4	ADTRAN IP 706
Ring Groups						
Operator Group						
Paging Groups						
Pickup Groups						
Trunks						
Trunk Accounts						
Trunk Groups						
Shared Line Accounts						
3 phone configs						Columns

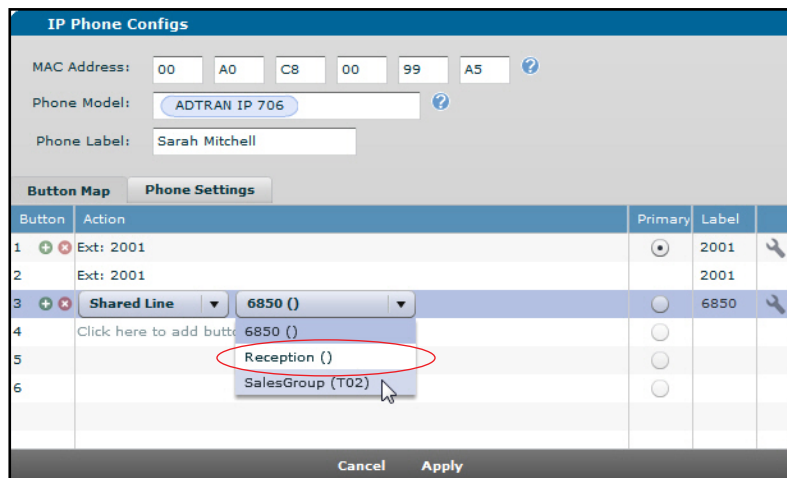
- Highlight the **Click here to add button** row, and double-click in the **Action** column.



- From the drop-down menu, select **Shared Line**.



- Select the shared line from the drop-down menu.



5. After assigning an SLA to a line key, select one of the line keys as **Primary**. Designating an SLA line as **Primary** will seize the SLA first (if available) when the phone goes off hook. The default is for the private line to be **Primary**. Select the **Primary** radio button on the appropriate line. Select **Apply** when finished. (For IP phone firmware versions that support this feature, refer to *Hardware and Software Requirements and Limitations on page 4.*)

The screenshot shows the 'IP Phone Configs' interface. At the top, there are fields for 'MAC Address' (00 A0 C8 00 99 A5), 'Phone Model' (ADTRAN IP 706), and 'Phone Label' (Sarah Mitchell). Below these is a 'Button Map' table with columns for 'Button', 'Action', 'Primary', and 'Label'. Row 3 is selected, showing '3 Shared: SalesGroup' with the 'Primary' radio button selected. The 'Apply' button at the bottom is circled in red. Annotations with arrows point to the Primary radio button and the Apply button.

Select the **Primary** radio button to seize the associated line when the phone goes off hook.

Select **Apply** to accept the changes.

6. Once you have selected **Apply**, the system will prompt you to reboot the phone. Select **Yes** to reboot the phone. The new SLAs will appear on the phone after the reboot process.

Update Successful
 Config file updated successfully!
 Would you like to sync and reboot the phone now?
 Syncs are processed in groups of 10 every 10 minutes.
 It could be several minutes before the phone reboots.
 Click Yes to reboot the phone or No to return to the previous page.

Yes No

You may receive a warning from the system instead, advising you to reset the SIP authentication password. This occurs if you changed the SIP authentication password and requires you to follow the instructions. Select **OK** to continue. Then the prompt to reboot the phone is displayed. Select **Yes** to reboot the phone.

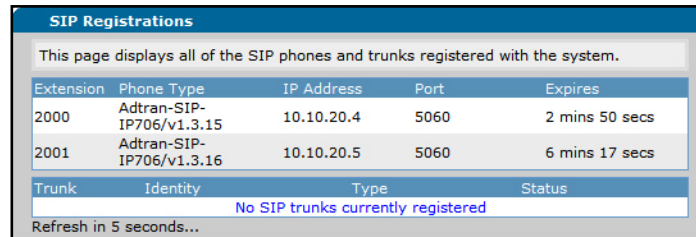
Warning: Your sip-authentication password must be reset.
 Please browse to the Voice User page associated with this phone to generate and apply a new password.
 Applying the new password will automatically update your phone config file with the correct SIP authentication password.

Ok

7. Repeat *Step 5: Assigning an SLA to an ADTRAN Phone Line Key(s) on page 24* to add additional SLAs to a phone. Most phone models allow multiple line registrations on a single phone. The maximum number of line registrations depends upon the phone model.

Step 6: Confirming Shared Line Key Configuration

From the main menu, navigate to **Voice > Reports > Phone Registrations** to view a list of registered IP phones. If your SLA is configured properly and has registered with the specified SIP phone(s) on the system, it will appear in this list. There should be an SLA registration for each phone that is configured to use the SLA.



The screenshot shows a web interface titled "SIP Registrations". It contains a message: "This page displays all of the SIP phones and trunks registered with the system." Below this is a table with the following data:

Extension	Phone Type	IP Address	Port	Expires
2000	Adtran-SIP-IP706/v1.3.15	10.10.20.4	5060	2 mins 50 secs
2001	Adtran-SIP-IP706/v1.3.16	10.10.20.5	5060	6 mins 17 secs

Below the table is another section for trunks:

Trunk	Identity	Type	Status
No SIP trunks currently registered			

At the bottom of the interface, it says "Refresh in 5 seconds..."

Additional Resources

There are additional resources available to aid in configuring your ADTRAN unit. The documents listed below are available online at ADTRAN's Support Forum at <https://supportforums.adtran.com>.

- *Configuring Shared Call Appearances*
- *Enhanced ANI and DNIS Substitution in AOS Voice Products*
- *Configuring User Accounts on the NetVanta 7000 Series*