		T P P A P P N O T E
		TPP-10096 Date: December, 2008
Product: ShoreTel | EtherSpeak SureTrunk™		**System version:** ShoreTel 8.1

Abstract

SIP Trunking allows the use of Session Initiation Protocol (SIP) communications from an Internet Telephony Service Provider (ITSP) instead of the typical analog, Basic Rate Interface (BRI), T1 or E1 trunk connections. Having the pure IP trunk to the ITSP allows for more control and options over the communication link. This application note provides the details on connecting the ShoreTel® IP phone system natively to EtherSpeak's SureTrunk™ SIP trunking services. SureTrunk offers enterprises of any size access to SIP trunking, and is designed to easily connect an IP PBX platform like ShoreTel to SIP trunking to realize all of the benefits and cost savings without additional hardware, software and licensing purchases or costly installation and set-up fees.

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Overview

This document provides details for connecting the ShoreTel IP phone system natively (i.e., without the need for any additional firewall / gateway / hardware) to SureTrunk (a process referred to collectively hereafter as SureTrunking), a SIP trunking service offering developed by EtherSpeak Communications for users of the ShoreTel IP PBX. This document specifically focuses on the configuration procedures used to set up ShoreTel systems on customer networks to interoperate with EtherSpeak's SureTrunk platform.

SureTrunk Features and Benefits

Feature:

- Fully managed service offering provides reliable alternative to CPE-based SIP Trunking
- Dedicated VPN connection from the customer's ShoreTel Switch via IPSec encryption
- Nationwide Quality-of-Service (QoS) within network core and all last mile connections with ADSL / SDSL / T1 MPLS
- Utilize telephone numbers available from any area code in the USA
- Advanced redundancy, recording and encryption features
- *'BustaTrunk'* right-sizing enables line provisioning based on monthly usage rates

Benefit:

- A risk-free approach to connecting SIP networks with traditional fixed-line PSTN
- Reliable, secure way to boost productivity, freely enabling coworkers to talk/collaborate
- High-quality communications with no local PSTN gateways, costly ISDN Basic Rate Interfaces (BRI) or Primary Rate Interfaces (PRI)
- 30 to 40% savings over POTs or PRI
- Auto-failover with auto email notification ensures business continuity / disaster recovery
- Added flexibility to reduce OpEx and total cost of ownership

EtherSpeak Overview and Contact

EtherSpeak, a "communications-as-a-service" provider, is pleased to offer a new option for ShoreTel customers to natively enable SIP trunking (aptly named SureTrunking) to EtherSpeak's SureTrunk service platform. The EtherSpeak SureTrunk service platform is a customized mix of processes and open-standard tools optimally configured to enable Media Gateway Control Protocol (MGCP)-based systems to seamlessly access an ITSP over the Internet. EtherSpeak's SureTrunk service allows the ShoreTel-enabled enterprise to adopt SIP "on the edge" without requiring the replacement of its existing firewall that has IPSec tunneling capabilities. EtherSpeak's SureTrunking includes free setup and a full range of unique features for customers seeking to leverage SIP trunking to augment the capability and connectivity options of their investment in the industry-leading ShoreTel IP PBX platform.

North America

For general sales questions, please contact EtherSpeak directly at:

Neil Darling
703-649-0025
ndarling@etherspeak.us
www.etherspeak.us

Resellers who want to start selling this solution should contact:

Terry Elton
703-221-5641
telton@etherspeak.us
www.etherspeak.us

EtherSpeak Product Information

EtherSpeak provides SIP trunking services for leading IP PBX platforms in the U.S. and Canada. EtherSpeak's SureTrunk service provides a straightforward setup process and currently has no setup costs. For more information and to sign up for a free trial, go to www.SureTrunk.com.

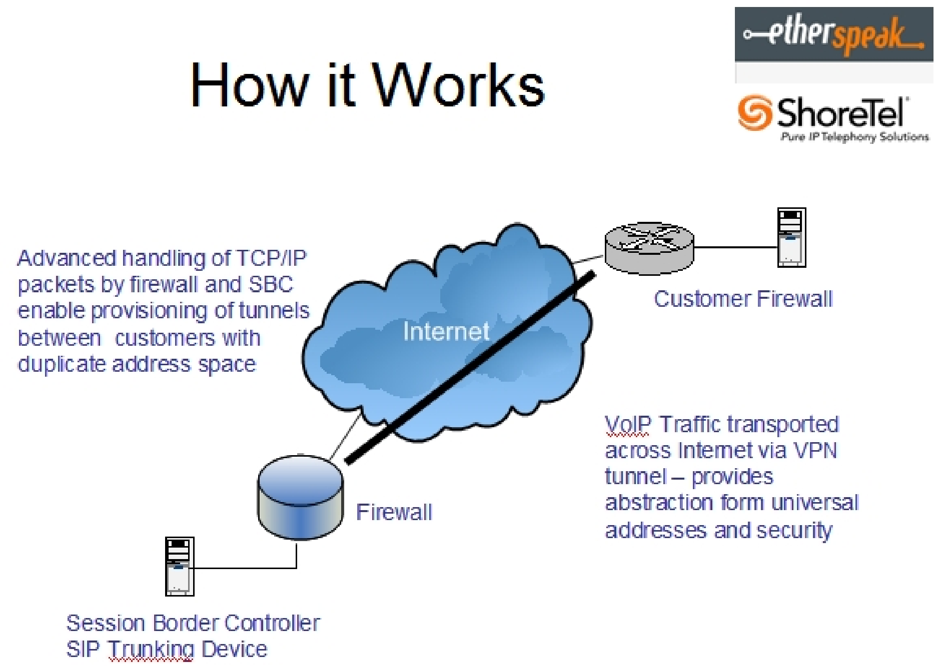


Figure 1 – SureTrunk SIP Trunk Service via IPSec-based Signaling

Architecture Overview

Providing easy access to SIP trunks is important. SIP trunks bring a number of key benefits to customers that include:

- Consolidating voice and data traffic, circuits and connections
- Eliminating overkill … never again will you have to pay for an over-subscribed bundle of voice channels
- Accessing direct dial numbers for virtually any local access and transport area (LATA) in the U.S.
- Having a bridge to the local PSTN to save on long distance costs
- Having support for all forms of IP-enabled communications

The image below shows a high-level drawing of a basic ShoreTel-to-SureTrunk design. This drawing only represents a high-level overview of the architecture. The next section of this application note covers actual deployment design options.

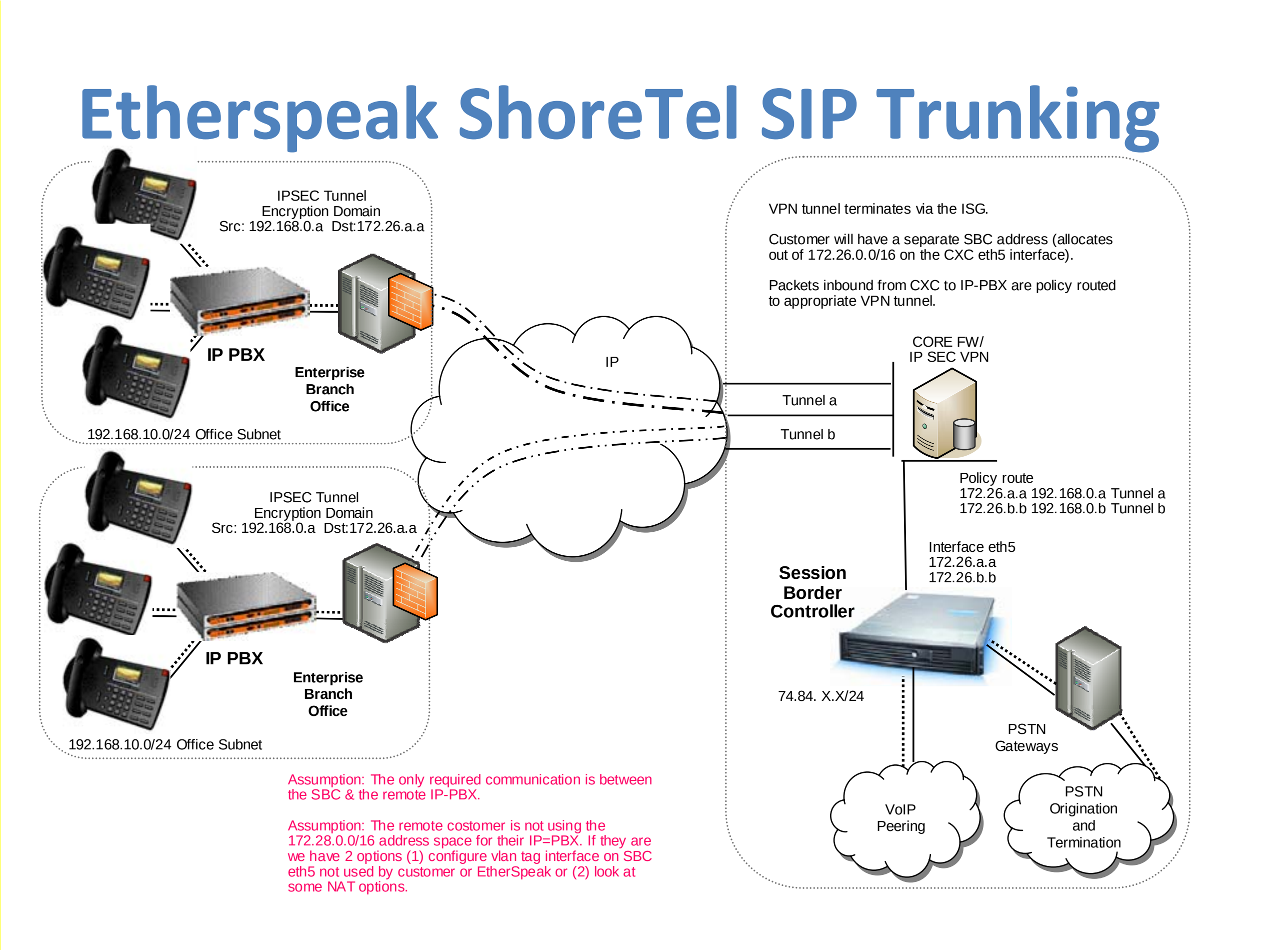


Figure 2 – Architectural Overview

In designing a solution to enable native SIP trunking from the ShoreTel IP PBX, customers simply need to establish a VPN tunnel specifically for SIP signaling. The customer will submit a configuration and provisioning request from EtherSpeak's SureTrunk.com Web site. Configuration and testing of a new SIP trunk will take approximately two (2) hours.

The routing of SIP traffic to EtherSpeak's SureTrunk service can be accomplished by providing three primary categories of information. These categories of information are related to the following:

- **Customer information and service option selections:** Billing information, technical and functional contact information and service options selections
- **Technical information:** Customer's ShoreTel technical information, LAN subnet information, firewall information
- **DID provisioning information:** Customer's address information for local inbound, toll-free and enhanced local service, Authorization for Local Number Portability (LNP)

Requirements, Certification and Limitations

Problem Statement

- If ShoreTel customers wish to connect to inbound or outbound SIP trunks, a dedicated hardware device is required to establish connectivity to any ITSP.

Solution

- By leveraging advanced knowledge of Internet security protocols and voice over IP (VoIP), EtherSpeak is providing a full range of managed SIP trunking services with or without dedicated hardware prerequisites, enabling ShoreTel customers to have greater flexibility and scale SIP trunking services according to growth needs.

A VPN is required for connection to the SureTrunk service. EtherSpeak's solution requires a VPN tunnel with access from the customer's ShoreTel switch (where trunks are configured) to a virtual IP assigned by EtherSpeak to that customer's ShoreTel switch. Therefore, customer firewall should support industry standard IPSec encryption with availability of one-tunnel VPN license. Please consult your firewall vendor to determine if your product supports industry standard IPSec and you are licensed to establish a single IPSec tunnel to enable the EtherSpeak SureTrunk service.

ShoreTel requires licensing for SIP trunking to be enabled. Please contact your ShoreTel reseller for information on purchasing the ShoreTel SIP trunk license.

SureTrunk service can also be configured to work with an Ingate SIParator, which traditionally has been the solution for providing SIP services to ShoreTel customers.

Version Support

Products are certified via the Technology Partner Certification Process for the ShoreTel system.

- EtherSpeak currently supports ShoreTel version 8.0 only.

		EtherSpeak SureTrunk
ShoreTel Release	8.0	✓
	8.1	✓

EtherSpeak SureTrunk Certification Testing Results

Table 1: Basic Feature Test Cases

ID	Name	Description	Results
1.1	Device initialization with static IP address	Verify successful startup and initialization of the device up to a READY/IDLE state using a static IP address	Pass
1.2	Device reset – idle (for static configurations)	Verify successful re-initialization of device after power loss while device is idle	Pass
1.3	Device initialization with DHCP	Verify successful startup and initialization of the device up to a READY/IDLE state using Dynamic Host Configuration Protocol (DHCP)	Not Tested
1.4	Device reset – idle (for dynamic configurations)	Verify successful re-initialization of device after power loss while device is idle	Not Tested
1.5	Verify DiffServ Code Point support	Verify the ability to set DiffServ Code Point from SIP DUT	Not Tested
1.6	Verify Date and Time Update support	Verify setting of Date and Time Update on SIP Device Under Test (DUT)	Pass
1.7	Place call	Verify successful call placement with normal dialing to a variety of terminating phones	Pass
1.8	Receive call	Verify successful reception of calls with normal dialing from a variety of calling phones	Pass
1.9	CODEC support – common (from DUT to ShoreTel Phone, REF-x)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	Pass

ID	Name	Description	Results
1.10	CODEC support – common (from DUT to SIP Reference Phone, SIP-Ref)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	Pass
1.13	CODEC support – negotiated	Verify successful negotiation between devices configured with different default CODECs (G.711-Ulaw and G.729)	Pass
1.15	Hold from ShoreTel Phone	Verify successful hold and resume of connected call	Pass
1.16	Forward	Verify successful forwarding of incoming calls	Pass
1.18	Mute	Verify device's mute function	Pass
1.19	Out-of-band/In-band Dual-tone Multi-frequency (DTMF) Transmission	Verify successful transmission of in-band and out-of-band digits (RFC2833) for calls placed to and from the DUT with a variety of other devices	Pass
1.20	Missed call notification	Verify that device notifies the user about missed calls	Pass
1.21	Volume	Verify the device's volume adjustment function	Pass
1.22	Auto Attendant DTMF detection G.711	Verify successful Auto Attendant transfers between devices configured with CODECs G.711-Ulaw	Pass
1.23	Auto Attendant DTMF detection G.729	Verify successful Auto Attendant transfers between devices configured with CODECs G.729	Pass

Table 2: Extended Feature Test Cases

ID	Name	Description	Notes
2.1	Call waiting	Verify appropriate notification and successful connection of incoming call while busy with another party	Pass
2.2	Park	Verify successful park and retrieval of connected call	Pass
2.3	Extended forward	Verify extended call forwarding options – busy forwarding, no-answer forwarding	Pass
2.5	Transfer – blind	Verify successful blind transfer of connected call	Pass
2.6	Transfer – monitored	Verify successful monitored transfer of connected call	Pass
2.7	Conference – ad hoc	Verify successful ad hoc conference of three parties	Pass
2.8	Place call – secondary line	Verify successful call placement using secondary line	Pass
2.9	Receive call – secondary line	Verify successful connection of incoming call on secondary line	Pass
2.10	Callback	Verify successful connection of a call using the missed-call callback feature of the device	Pass
2.11	Headset	Verify the device's support for external headsets (using headsets supplied by the 3P phone vendor)	N/A
2.12	Ring selection	Verify the device's ability to change the ring type	N/A
2.13	Caller ID Name and Number	Verify that Caller ID name and number are sent and received from SIP endpoint device	Pass
2.14	SIP Device Generates Busy Tone	Verify that SIP DUT generates busy tone when calling a busy extension	Pass
2.15	Verify handling of "911"	Verify dialing "911" on DUT can connect with "911" services	Pass
2.16	Verify Fax Handling	Verify that fax can be sent and received through DUT	Not Tested
2.17	Caller ID Name and Number Variations	Verify that Caller ID name and number is sent and received	Pass

Configuration Overview

The information below shows examples for configuring the ShoreTel-to-SureTrunk service. While configuration requirements vary depending on firewall and network settings (among other things), the information provided in these steps, along with the ShoreTel Planning and Installation Guide as well as documentation provided by EtherSpeak, should be sufficient to get SureTrunk running smoothly. However, if questions arise, inquiries should be directed to EtherSpeak SureTrunk support at 866-384-3747, Option 3 or via email at support@etherspeak.us.

ShoreTel Unsupported Features

At the time of this writing, EtherSpeak's SureTrunk service does not support the following ShoreTel features. However, support will be available with a future releases:

- Fax redirect via SIP trunks.
  (Workaround: Direct Inward Dialing (DID) to fax endpoint is currently supported.)
- Office Anywhere

ShoreTel Configuration

This section provides the general system settings and trunk configurations (both group and individual) required for a ShoreTel system to support SIP trunking.

ShoreTel System Settings – General

General system settings include settings for Call Control, the Site and the Switch. If you confirm that the settings have already been configured as described in this section, proceed to the section titled, "ShoreTel System Settings – Trunk Groups". Otherwise, follow the instruction below.

Call Control Settings:

Configure the settings for Call Control by logging into ShoreWare® Director. After logging in, select "Administration" then "Call Control" followed by "Options" (See Figure 3 below).

Figure 3 – Administration Call Control Options

Upon selection of "Options," the "Call Control Options" screen will appear as shown in Figure 4 below.

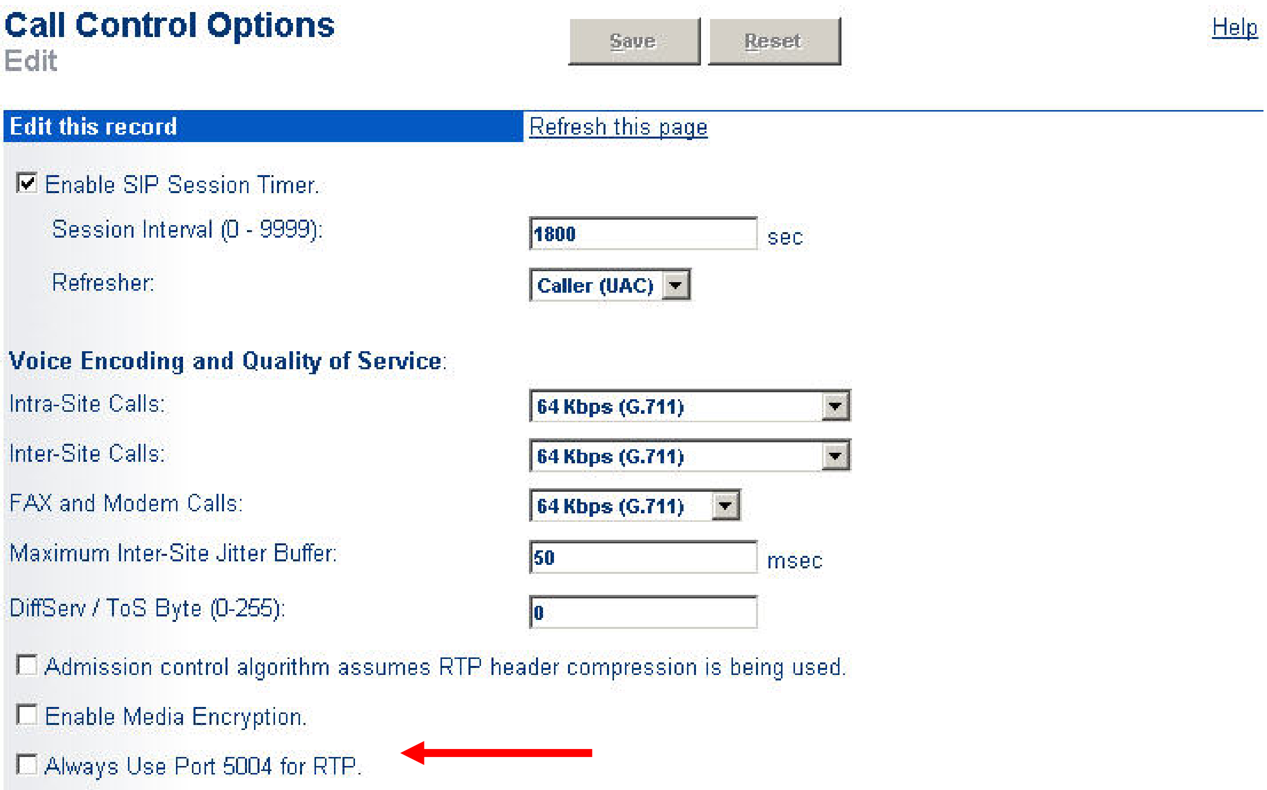


Figure 4 – Call Control Options

Within the "Call Control Options" screen, the following settings are required:

- **Enable SIP Session Timer:** This box must be checked. If it is not, check it.
- **Session Interval (0-9999):** Enter 1800 seconds.
- **Refresher:** Select either "Caller UAC" or "Callee (UAS). If the "Refresher" field is set to "Caller (UAC)" [User Agent Client], the Caller's device will be in control of the session timer refresh. If "Refresher" is set to "Callee (UAS)" [User Agent Server], the device of the person called will control the session timer refresh.
- Selecting "Caller UAC" [User Agent Client] from the pull list confirms that the appropriate settings are made for the "Enable SIP Session Timer," "Intra-Site Calls," "Inter-Site Calls" and "Always Use Port 5004 for RTP" fields.

The next settings to verify are the "Intra-Site Calls" and the "Inter-Site Calls" settings under the "Voice Encoding and Quality of Service" prompt. For the Intra-Site Calls, verify that the desired audio bandwidth is selected for the CODEC for calls within the system. The settings should then be confirmed for the desired audio bandwidth CODEC for Inter-Site calls (calls between sites).

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).

Unchecking the box for "Always Use Port 5004 for RTP" is required for implementing SIP on the ShoreTel system. For SIP configurations, dynamic User Diagram Protocol (UDP) must be used for Real-time Transport Protocol (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. Once this parameter is unchecked, make sure that all devices (IP Phones, ShoreGear® Switches, ShoreWare Director, Distributed Voice Services/Remote Servers, Conference Bridges and Contact Centers) are "fully" rebooted – this is a "one time only" item. By not performing a full system reboot, one way audio will probably occur during initial testing.

Sites Settings:

The next settings to address are the administration of sites. These settings are modified under the ShoreWare Director by selecting "Administration," then "Sites" (Figure 5).

Figure 5 – Administration Site

This selection brings up the "Sites" screen. Within the "Sites" screen, select the name of the site to configure. The "Edit Site" screen will then appear. The only change required to the "Edit Site" screen is to the "Admission Control Bandwidth" field.

Note: Bandwidth of 1024 is just an example. Please see the Planning and Installation Guide for additional information on setting Admission Control Bandwidth.

Sites Edit Screen – Admission Control Bandwidth

The Admission Control Bandwidth defines the bandwidth available to and from the site. This is important as SIP devices will be counted against the site bandwidth. Bandwidth needs to be set appropriately based on site setup and configuration with the EtherSpeak SureTrunk SIP Trunking. See the ShoreTel Planning and Installation Guide for more information on this topic.

Switch Settings - Allocating Ports for SIP Trunks

The final general settings to input are the ShoreGear switch settings. These changes are modified by selecting "Administration," then "Switches" in ShoreWare Director (Figure 6).

Figure 6 – Administration Switches

This action brings up the "Switches" screen. From the "Switches" screen, simply select the name of the switch to configure and the "Edit ShoreGear …Switch" screen will be displayed. Within the "Edit ShoreGear …Switch" screen, select the desired number of SIP Trunks from the ports available (Figure 7).

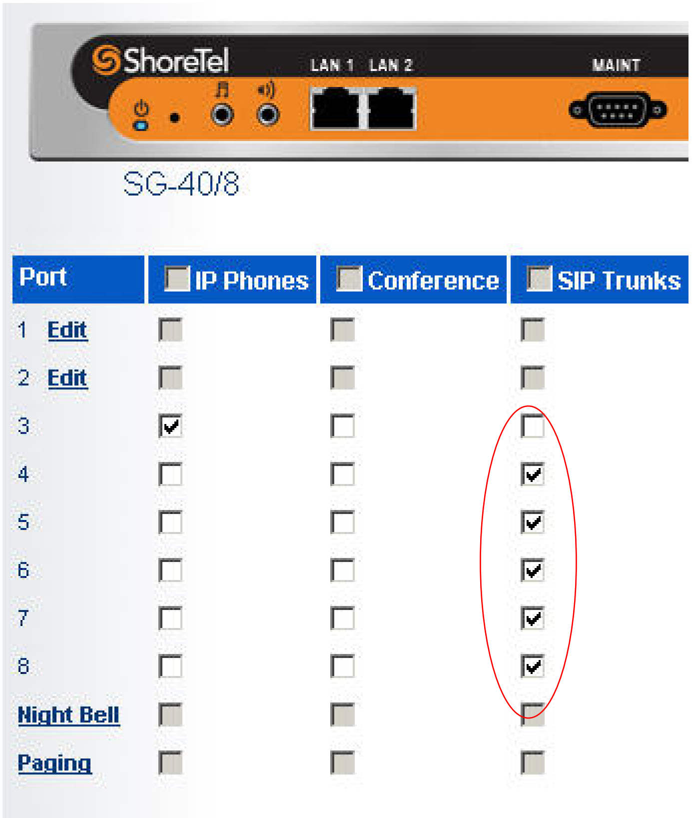


Figure 7 – ShoreGear Switch Settings

Each port designated as an SIP Trunk enables the support for 5 individual trunks.

ShoreTel System Settings – Trunk Groups

ShoreTel Trunk Groups support both Dynamic and Static SIP endpoint individual trunks.

Note: A ShoreGear switch can only support one trunk group with dynamic IP addressing.

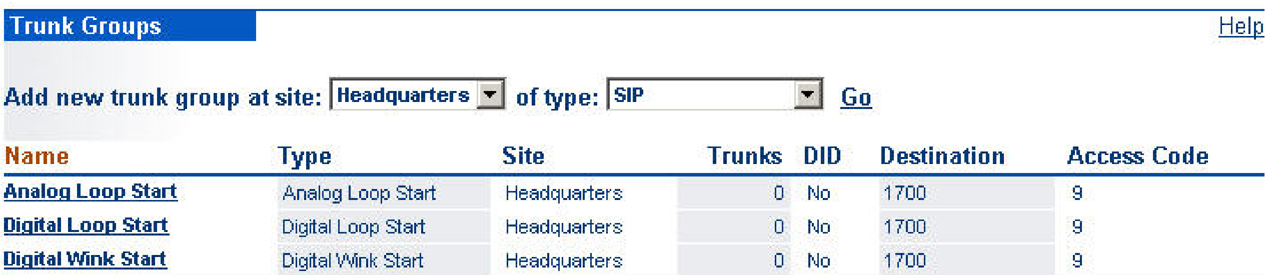
In trunk planning, the following need to be considered:

1. Are the SIP devices using Dynamic Host Configuration Protocol (DHCP) or Static IP?
2. Are the SIP devices endpoints (like Attached Technology Attachments or ATAs, Conference Phones or WiFi handsets) or non-endpoint devices like an ITSP?

If the SIP trunk groups have already been configured on the system, skip down to the "ShoreTel System Settings - Individual Trunks" section. The settings for trunk groups are changed by selecting "Administration," then "Trunks" followed by "Trunk Groups" within ShoreWare Director (Figure 8).

Figure 8 – Administration Trunk Groups

This selection brings up the "Trunk Groups" screen (Figure 9).


Trunk Groups Help

Add new trunk group at site: Headquarters of type: SIP Go

Name	Type	Site	Trunks	DID	Destination	Access Code
Analog Loop Start	Analog Loop Start	Headquarters	0	No	1700	9
Digital Loop Start	Digital Loop Start	Headquarters	0	No	1700	9
Digital Wink Start	Digital Wink Start	Headquarters	0	No	1700	9

Figure 9 – Trunk Groups Settings

From the pull down menus on the "Trunk Groups" screen, select the site desired, then select the "SIP" trunk type to configure and click on the "Go" link from "Add new trunk group at site:" The "Edit SIP Trunk Group" screen will appear (Figure 10).

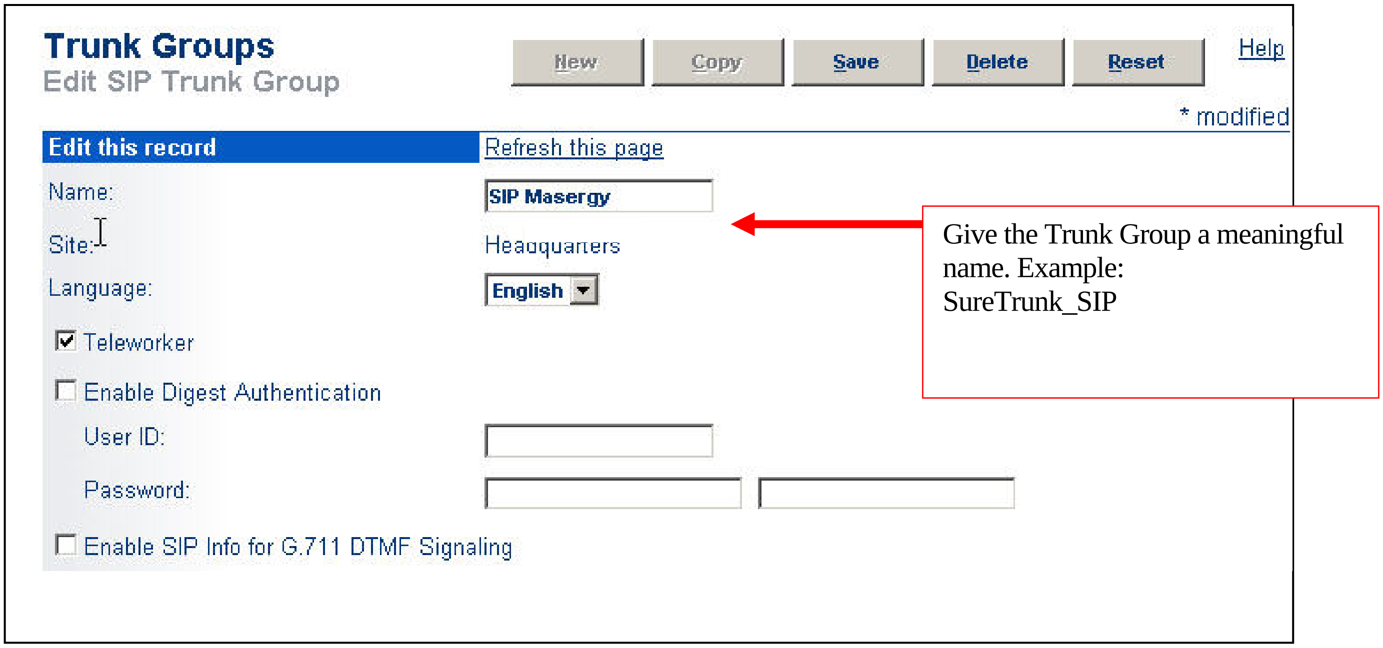


Figure 10 – SIP Trunk Group Settings

For the SureTrunks, the trunks need to be configured as inter-site trunks (trunks between sites). The trunks will also be configured as static.

The next step within the "Edit SIP Trunks Group" screen is to input the name for the trunk group. In the example in Figure 10, the name "SIP" has been created. The next step is to verify the setting of the "Teleworker" check box. The "Teleworker" check box needs to be checked since the trunk groups have been configured as inter-site. Once this box is checked, it will count against the site bandwidth.

The "Enable Digest Authentication" field is not required.

The "Enable SIP Info for G.711 DTMF Signaling" box should not be checked. Enabling SIP info is currently only used with tie trunks between ShoreTel systems.

The next item to change in the "Edit SIP Trunks Group" screen is to make the appropriate settings for the "Inbound:" fields (Figure 11).

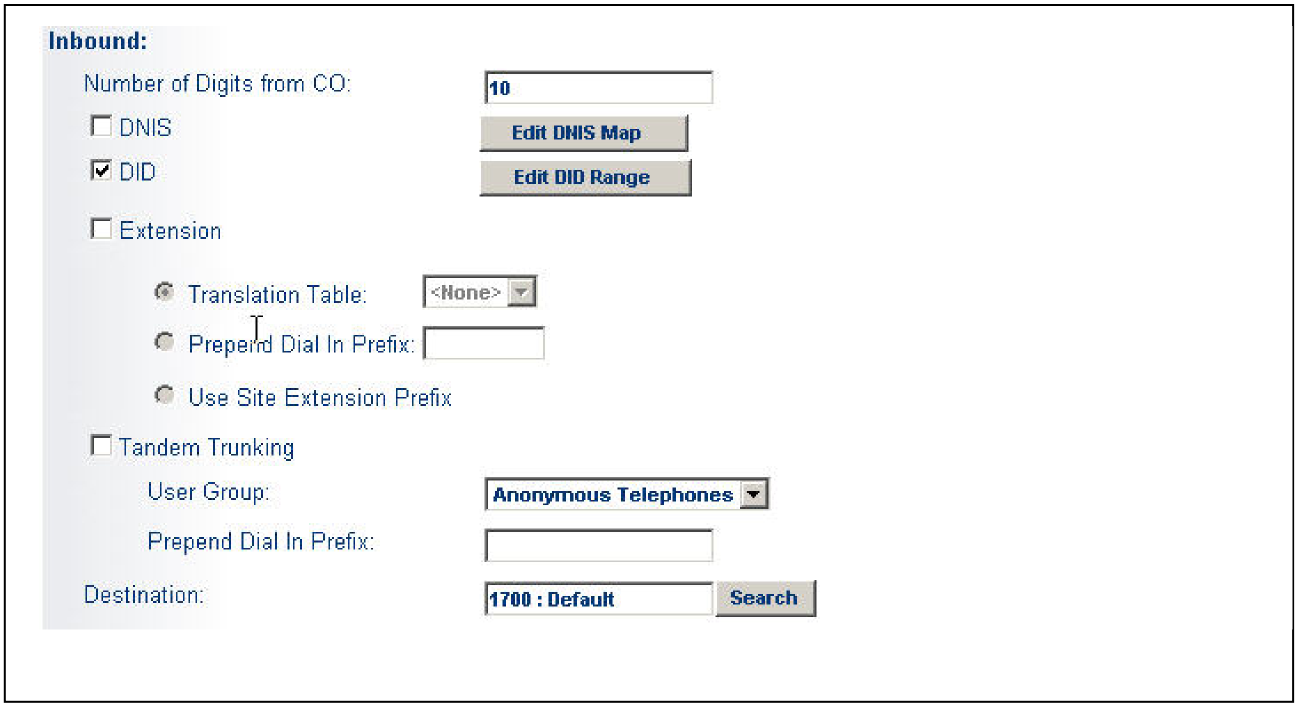


Figure 11 – Inbound:

Within the "Inbound:" settings, ensure that the "Number of Digits from CO is set to 10 and ensure that the "DNIS" or "DID" box is checked, along with the Extension parameter (see Planning and Installation Guide for further information on configuration).

Tandem Trunking is required if you plan on routing incoming SIP trunk calls out other ShoreTel trunks, for example, for three-way calls over SIP trunks.

Note: This section is configured in the same way as any normal trunk group.

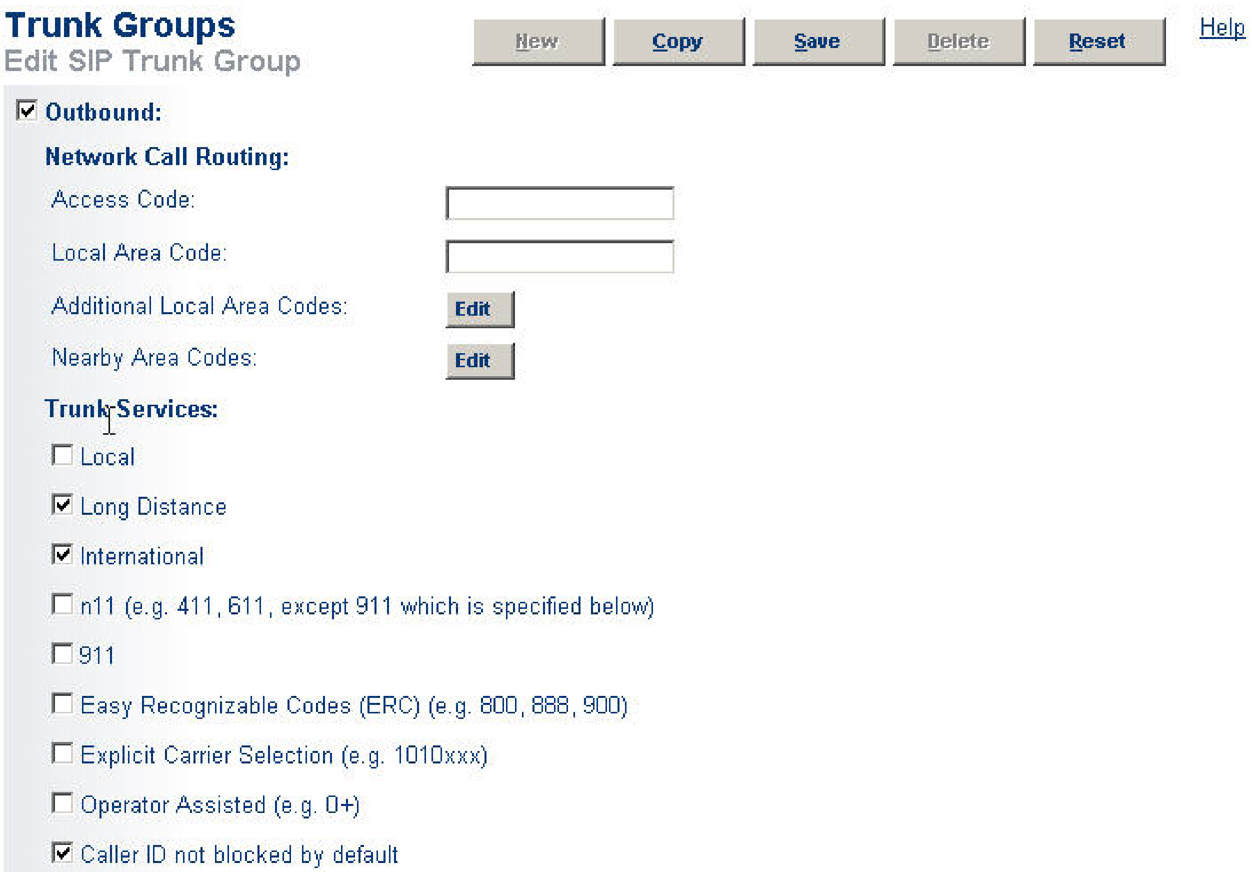


Figure 12 – Trunk Services

On the "Trunk Services:" screen, make sure that the appropriate services are checked or unchecked based on what features are needed from this trunk group.

The last checkbox determines if the call is sent out as <unknown> or with caller information (Caller ID). User DID and other settings will effect how information is passed out to the SIP trunk group.

After these settings are made to the "Edit SIP Trunk Group" screen, press the "Save" button to input the changes.

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

ShoreTel System Settings – Individual Trunks

This section covers the configuration of the individual trunks. Select "Administration," then "Trunks" followed by "Individual Trunks" to configure the individual trunks (Figure 13).

Figure 13 – Individual Trunks

The "Trunks by Group" screen that is used to change the individual trunk settings then appears (Figure 14).

Figure 14 – Trunks by Group

Select the site for the new individual trunk(s) to be added and select the appropriate trunk group from the pull down menu in the "Add new trunk at site" area. In this example, the site is "Headquarters" and the trunk group is "SIP". Click on the "Go" button to bring up the "Edit Trunk" screen (Figure 15).

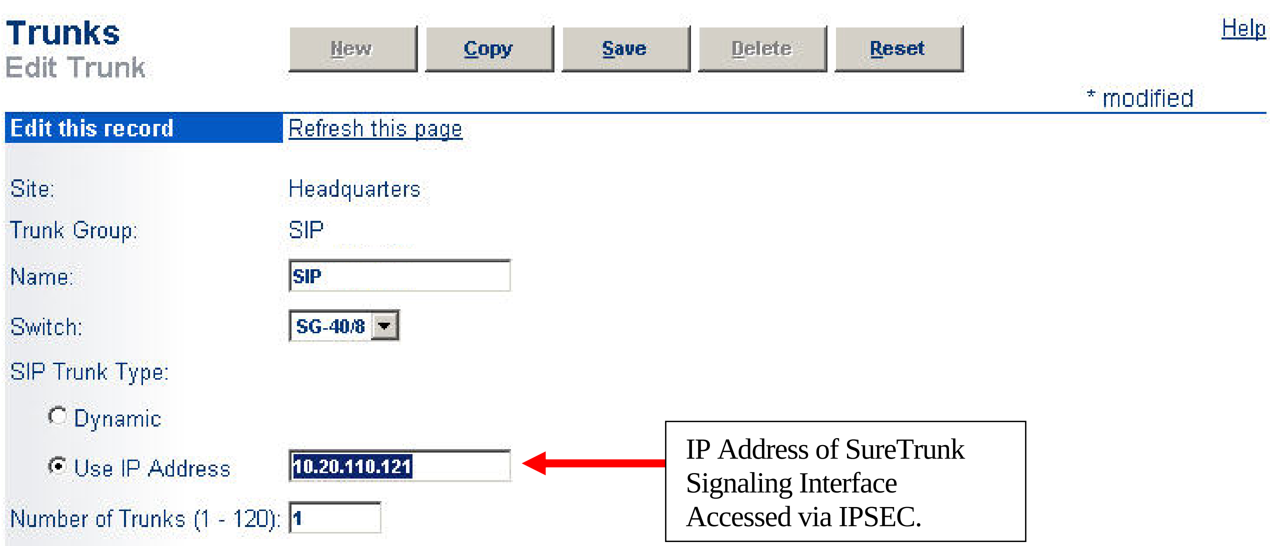


Figure 15 – Edit Trunks Screen for Individual Trunks

From the individual trunk's "Edit Trunk" screen, input a name for the individual trunks, select the appropriate switch, select the SIP trunk type and input the number of trunks. When selecting a name, the recommendation is to name the individual trunks the same as the name of the trunk group so that the trunk type can easily be tracked. Select the switch upon which the individual trunk will be created. For the SureTrunk, select the "Use IP Address" button and input an IP address provided by the EtherSpeak engineering and fulfillment team (as provided after submission on www.SureTrunk.com). The last step is to select the number of individual trunks desired (each one supports "one" audio path – for example, if 5 is input, then 5 audio paths can be up at one time). Once these changes are complete, press the "Save" button to input changes.

Note: Individual SIP trunks cannot span networks. SIP trunks can only terminate on the switch selected. There is no failover to another switch. If redundancy is required, each SIP trunk group will require a VPN tunnel to the switch IP address. EtherSpeak's VPN tunnel may only access one ShoreTel switch IP per tunnel (slash /32). If two trunk groups will be needed, a second tunnel may need to be set up. Please contact EtherSpeak support at (866) 987-8643.

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. This completes the settings for the ShoreTel system side.

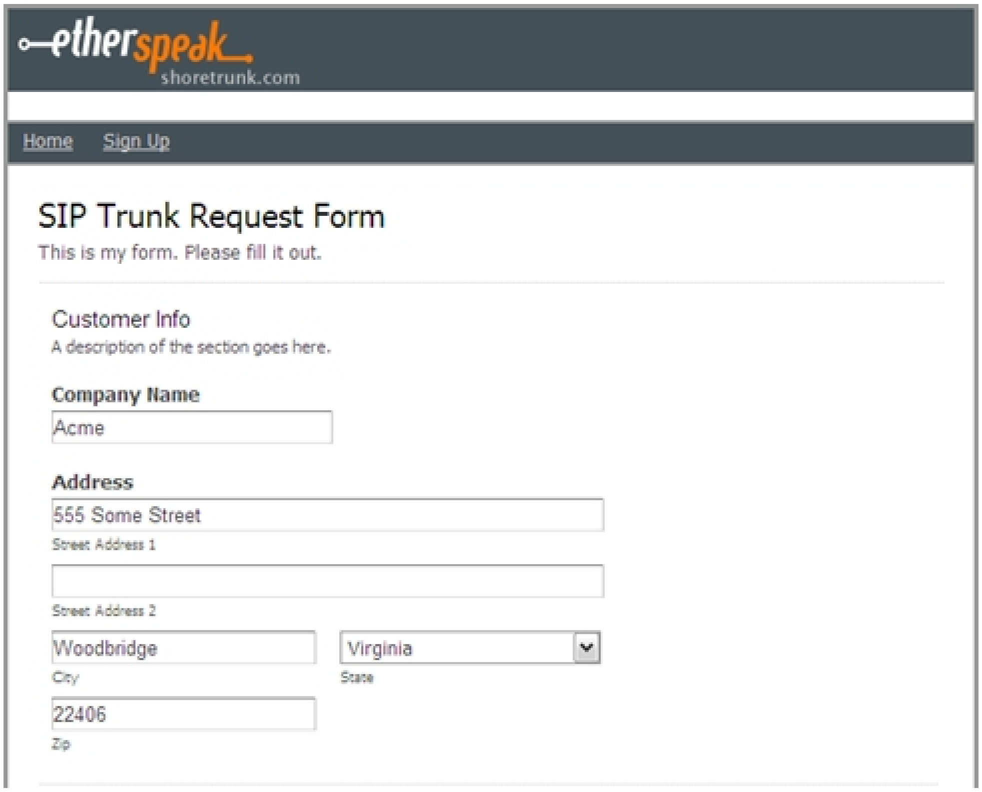
EtherSpeak SureTrunk Provisioning

Signing up for SureTrunking with EtherSpeak is easy.

Simply go to www.SureTrunk.com and register for service with our online provisioning tool. Or simply call toll free (866) 987-8643.

Step I – Customer Information:

Begin the setup process by submitting pertinent information to EtherSpeak engineering. Note that you will need information on the technical contact, business contact and certain LAN information to initiate a new SureTrunk provisioning request.



Step II – LAN Information to establish SIP Signalling IPSec Tunnel:

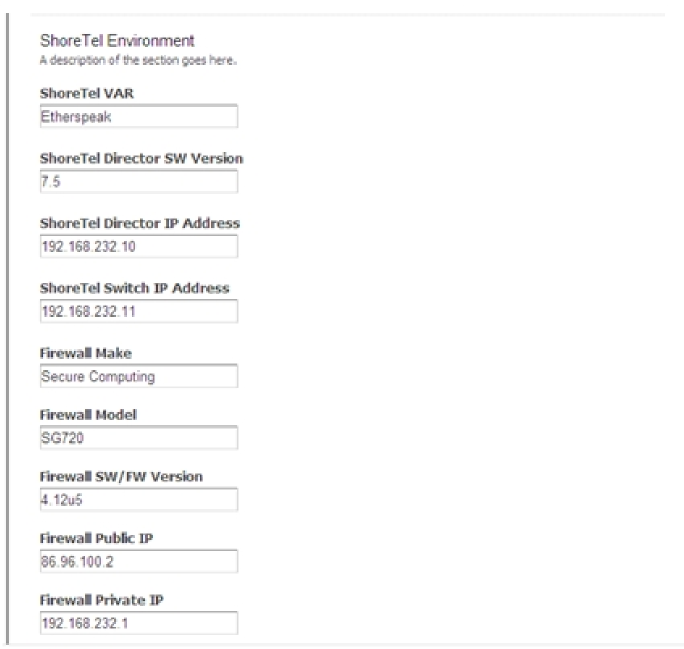
Enter LAN, ShorTel IP PBX and Firewall Info

A. Reference Your ShoreTel Reseller

C. ShoreTel Director and Switch Info

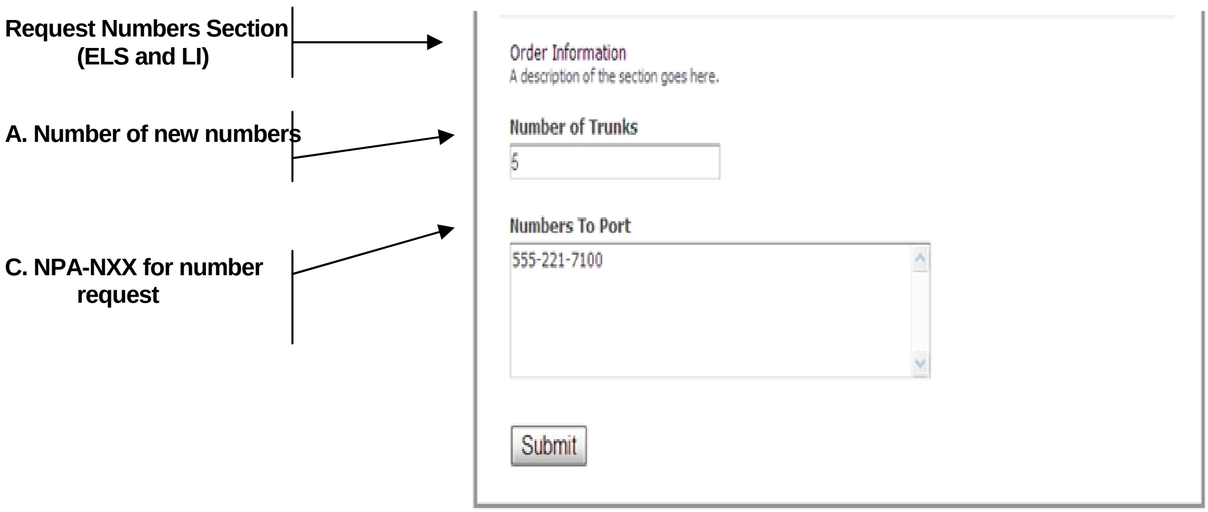
C. Firewall Make, Model and IP Info

D. Scroll to Next Section



Step III – SIP Trunk Configuration:

In the next step, request the quantity of numbers you wish provisioned with the SureTrunk service. EtherSpeak provides some innovative options for bursting concurrent calls dynamically and auto-redundancy options for failing over to a customer PSTN connection in the event of any Internet outages. Please contact EtherSpeak sales for more information on RedundaTrunk and BurstaTrunk options.



D. Click Submit.

Step IV –Confirmation of Provisioning:

Once "Submit" is selected, we will begin the provisioning process and an engineer will contact you to confirm your entries and provide step-by-step guidance for concluding the VPN setup process; provisioning your ShoreTel for SureTrunk service and assigning telephone numbers to your SureTrunk account.

SureTrunk Troubleshooting

EtherSpeak is available to troubleshoot any problems or issues that may occur regarding SureTrunk service. The contact information for EtherSpeak Technical Support is listed below.

EtherSpeak's SureTrunk Technical Support

Worldwide Customer Support: 866-384-3747

support@etherspeak.us

Disclaimer

To be "ShoreTel Certified" means that Technology Partner's product will interoperate with the ShoreTel system, but ShoreTel does not certify that the features or functionality of Technology Partner's product will perform as specified by Technology Partner nor that Technology Partner's product will meet your specific application needs or requirements. To inter-operate means that Technology Partner's product is able to exchange, use and share information with the ShoreTel system.

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