

		Innovation Network App Note TPP: 10096 Date: September, 2011
Product: ShoreTel EtherSpeak		System version: ShoreTel 11.x

Abstract

In 2008, EtherSpeak certified the SureTrunk for ShoreTel connection method to enable an innovative “native” SIP connectivity option for ShoreTel. SIP Trunking provides the ShoreTel users a value-aligned method for public telephone network by-pass. These connection options are enabled when a ShoreTel customer elects to use a SIP trunking provider like EtherSpeak (using the Session Initiation Protocol (SIP)) for communications beyond the customer’s IP network edge.

In the current version of EtherSpeak’s SureTrunk for ShoreTel offering, EtherSpeak now provides the ShoreTel administrator numerous options for connectivity of the SureTrunk for ShoreTel service - to virtually any version of ShoreTel (this application note covers only ShoreTel versions 9.x through versions 11.x).

In addition, EtherSpeak now provides options for enhancing security of SIP, providing new Quality of Service (QoS) options, and Class of Service (COS) guarantees by broadening the EtherSpeak solution to include cross-connections to most Tier-One carrier data networks. This evolution is an improvement to how the service may be delivered, and as a result, has enabled the ShoreTel Administrator to consider leveraging EtherSpeak SIP with inter-connects to carrier networks in an unprecedented way. New carrier supported MPLS cross-connects are available currently with AT&T / ACC Business, Level 3 Communications, Qwest Communications, Verizon Business and others. Please speak with EtherSpeak sales to verify the availability of connecting to your carrier’s network.

They include a wide array of private wide-area-network (WAN) options (MPLS, T1’s, metro-Ethernet connectivity, Asynchronous Transfer Mode (ATM)), quality public network broadband services and most recently, Wireless Wide Area Network (WWAN) options (including WAN (802.11n or WiMax)). In three short years, EtherSpeak SIP is steadily moving to become the de-facto standard for connecting customer-premise based IP PBXs to carrier networks. With these options taking shape, it is an exciting time for ShoreTel and SIP. Although SIP connectivity is a “newer” ShoreTel connection method, it is typically regarded as less expensive, increasingly reliable, and the key driver for reducing customer monthly recurring costs verse legacy Public Switched Telephone Network (PSTN) connections. SIP is replacing Plain-Old-Telephone-Service (POTs), Basic Rate Interface (BRI), Primary Rate Interface (PRI) or T1 / E1 trunk connections. This application note provides a guide for you in selecting and implementing an EtherSpeak SureTrunk SIP connectivity option that is right for your customer. We will help you do this by providing an understanding of the three ShoreTel Certified methods to connect a ShoreGear system to EtherSpeak as a Communications-as-a-Service (CaaS) provider.

1. Connecting natively to EtherSpeak over a single existing Internet connection
2. Connecting natively to EtherSpeak over a redundant Internet connection
3. Connecting natively to EtherSpeak over a private network connection

Although we maintain that providing ShoreTel customers the option for native connectivity for SIP allows for more flexibility, additional deployment options over the customer’s private Wide-Area-Network, or public Internet connection have become part of our experience at EtherSpeak since the first release of the service in June 2008. This application note provides the details on connecting the ShoreTel® IP phone system natively to EtherSpeak’s

SureTrunk SIP trunking services, connecting through an Ingate Siparator, or connecting through a SIP IAD. We will detail some scenarios where, due to limitations with SIP some ShoreTel administrators may want to fully understand what method is best given client requirements.

First announced and deployed over three years ago, EtherSpeak has become an essential partner within the ShoreTel Technology Partner Program eco-system by providing Value-Added Resellers and end-user customers the option of leveraging SIP as a primary, secondary or tertiary connection option to PSTN network by-pass. EtherSpeak supports ShoreTel version 7.0 through 8.2 systems (please refer to previous application note). Please note that this application note is specifically addressing versions 9.x through 11.x. SureTrunk continues to provide options and innovation 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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ShoreTel tests and validates the interoperability of the Member's solution with ShoreTel's published software interfaces. ShoreTel does not test, nor vouch for the Member's development and/or quality assurance process, nor the overall feature functionality of the Member's solution(s). ShoreTel does not test the Member's solution under load or assess the scalability of the Member's solution. It is the responsibility of the Member to ensure their solution is current with ShoreTel's published interfaces.

The ShoreTel Technical Support organization will provide Customers with support of ShoreTel's published software interfaces. This does not imply any support for the Member's solution directly. Customers or reseller partners will need to work directly with the Member to obtain support for their solution.

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, Inc. 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 for ShoreTel platform.



Features and Benefits

Feature:

- Fully managed service offering provides reliable alternative to CPE-based SIP Trunking available as-a-service
- Dedicated VPN connection from the customer's ShoreTel Switch via IPSec encryption
- Options now for nationwide cross-connectivity to many Tier 1 service provider networks providing customer options for Security, Quality-of-Service (QoS) within network core and all last mile connections with ADSL / SDSL / T1 MPLS / Metro-Ethernet
- Utilize telephone numbers available from virtually any area code in the USA / CAN
- Advanced redundancy, recording and encryption features
- *'BustaTrunk'* right-sizing enables line provisioning based on monthly usage rates
- New SureTrunk bundle options provide from 50 to 150 concurrent call paths and from 7,500 minutes to 100,000 minutes of included domestic talk-time
- Provide seamless options for PSTN backup
- EtherSpeak supports any connection method to SIP: Native, On-Premise Session Border Controller CPE; PRI to SIP hand-off

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) and reliability of carrier provided SIP over MPLS with QOS, COS
- Eliminate expensive metro-numbers and remote call forwarding fees
- Auto-failover with auto email notification ensures business continuity / disaster recovery
- Added flexibility to provide right plan for customers – based on their unique requirements
- EtherSpeak SIP may be used as a seamless option for primary, secondary or tertiary trunk replacement and will fit with any customer's disaster recovery plan

Etherspeak Overview and Contact

EtherSpeak, a "communications-as-a-service" provider, is pleased to offer 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 easy 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 or partner information, please contact EtherSpeak directly at:

Ph: 703-221-9999

Email: sales@ietherspeak.com

Website: www.suretrunk.com

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 has no requirement for costly customer premise equipment (CPE). For more information and to sign up for a free trial, go to www.suretrunk.com.

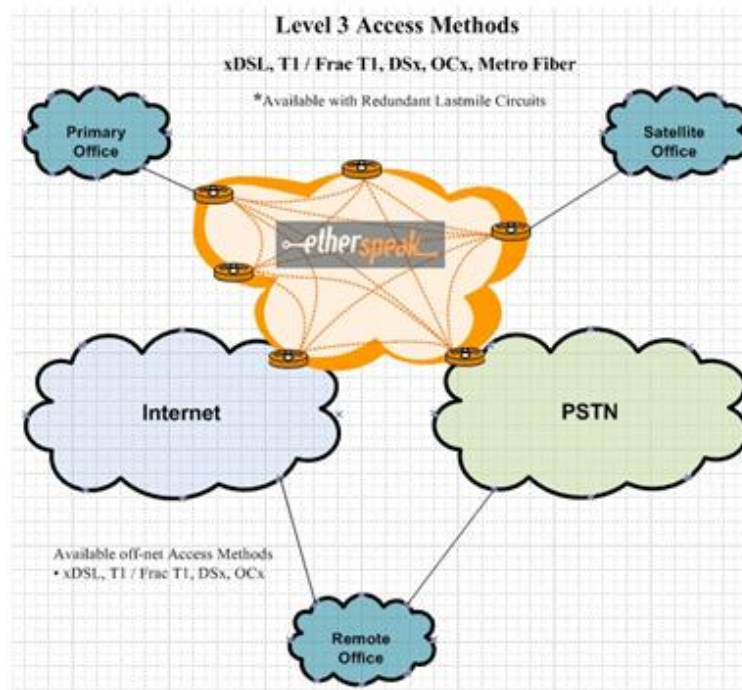


Figure 1 – SureTrunk SIP Trunk Service Private / Public Network Connectivity Options

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
- Utilizing Native SIP connectivity for primary, secondary and tertiary service availability options
- 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.

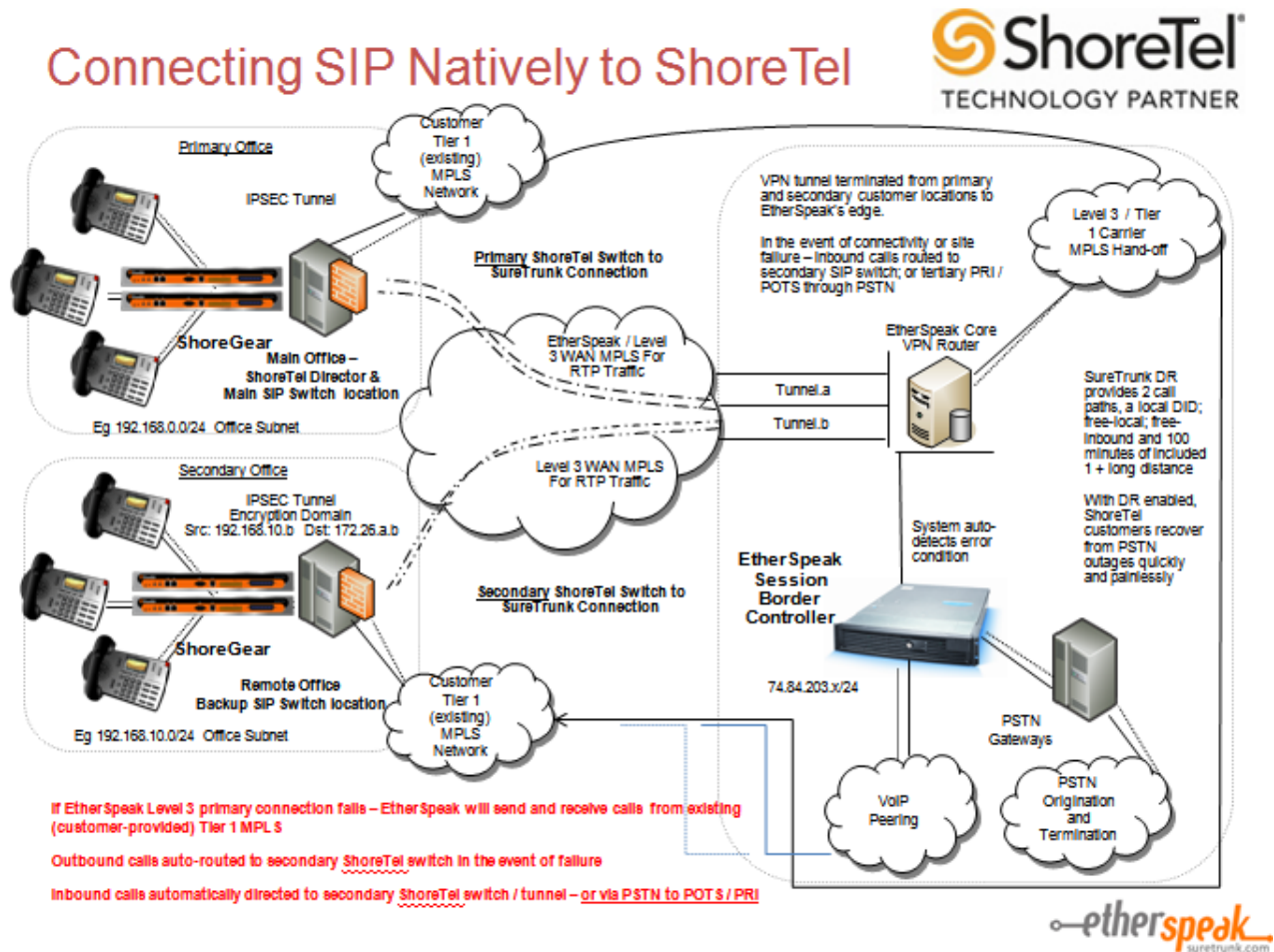


Figure 2 – Architectural Overview Illustrated with Redundant MPLS Connectivity Options

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

EtherSpeak requires an IPSEC tunnel connected over the Internet. However, the customer may choose to connect EtherSpeak SIP based via their existing Internet connection, a redundant Internet connection (a spare just used for the media / RTP) or via a full range of Wide Area Network Interconnection options for customers. As illustrated in Figure 2, a customer may elect to connect to EtherSpeak over a primary MPLS with Level 3 Communications (where EtherSpeak will connect to customer network from a network location as an additional MPLS site), or we may simply plug into the customer's existing MPLS network. In the illustration above, the customer has two MPLS sites, we simply became the third site on the customer MPLS network for receiving SIP and RTP packets with full support of Quality of Service and Class of Service guarantees.

The routing of SIP traffic to EtherSpeak's SureTrunk service can be accomplished by providing three primary categories of information on our web site to initiate a free-trial of the service. These categories of information are:

- **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 requirements, enabling ShoreTel customers to have greater flexibility and scale SIP trunking services according to growth needs. EtherSpeak supports native connectivity options; customer provided Session Border Controller as Customer Premise Equipment; or connection as a SIP to PRI hand-off via a FXS or PRI Internet Access Device (IAD)
- Natively using EtherSpeak SureTrunk Service:
 - Pros: Unlimited call capacity, no capital expense with a 30 minute customer quick turn-up process
 - Cons: Requires ShoreTel SIP Trunk License; SIP Call recording not supported on ShoreTel; No ring-back generated by ShoreTel phones after call is connected through Automated Attendant
- Utilizing a PRI to SIP hand-off:
 - Pros: No ShoreTel SIP license; ShoreTel Director "thinks" it is using a PRI; All features supported
 - Cons: Customer capital expenditure is significant and limited to 23 channels (PRI / T1 switch & SIP / PRI IAD); Limited to capacity of 23 concurrent calls per non-recurring equipment charge necessary for PRI hand-off; Trunks are tied to expensive hardware with limited capacity
- Utilizing a customer-premise-based Session Border Controller



- Pros: Provides ability to use any SIP carrier approved by ShoreTel; Normalizes SIP to carrier requirements / standards
- Cons: In addition to SIP license fee, SBC requires a license fee for concurrent calls; Not all features are supported including call recording, Office Anywhere; Park and Pickup; Limited support for SIP Info; May become possible single point of failure on customer network

A VPN is required for connection to the native 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, the 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 SIPParator, which traditionally has been the solution for providing SIP services to ShoreTel customers. EtherSpeak is an approved ITSP for Ingate on all ShoreTel versions.

ShoreTel Defects effecting SureTrunk for ShoreTel

Defect#: 1-37888019 - One way audio from SG50V switch caused by sip profile ATTBVOIP

Affects all ShoreTel builds that have ATTBVOIP SIP trunk profiles.

- Fixed in ShoreTel 9.2 build 14.42.5606.0 (or greater)
- Fixed in ShoreTel 10.1 build 15.21.5702.0 (or greater)

Defect#: 1-39105858 - No Music-On-Hold via SIP Trunks

Affects all ShoreTel builds that have ATTBVOIP SIP trunk profiles.

- Fixed in ShoreTel 9.2 build 14.42.5606.0 (or greater)
- Fixed in ShoreTel 10.1 build 15.21.7600.0 (or greater), this build is not currently available for download by customers / partners

Defect#: 1-40024807 - Call transfers from AA menu fail to Office Anywhere External users when using ATTBVOIP SIP trunk profile

- Affects all ShoreTel builds that have ATTBVOIP SIP trunk profiles.
- Not fixed in any available ShoreTel builds, will only be fixed in ShoreTel 10.2 build 14.41.4300.0 (or greater).

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

- Fax redirect via SIP trunks
(Workaround: Direct Inward Dialing (DID) to fax endpoint is currently supported.)
- Calls sent to Automated Attendant; dial by name or extension, no ring-back is generated on any configurations utilizing the ATTBVOIP profile (also known as the SIP Re-Invite profile)



Version Support

		SureTrunk Services
ShoreTel Release	7.x	✓
	8.x	✓
	9.x	✓
	10.x	✓
	11.x	✓

Validation Testing Results Summary

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
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

ID	Name	Description	Results
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 SureTrunk for ShoreTel 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@ietherspeak.com.



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).

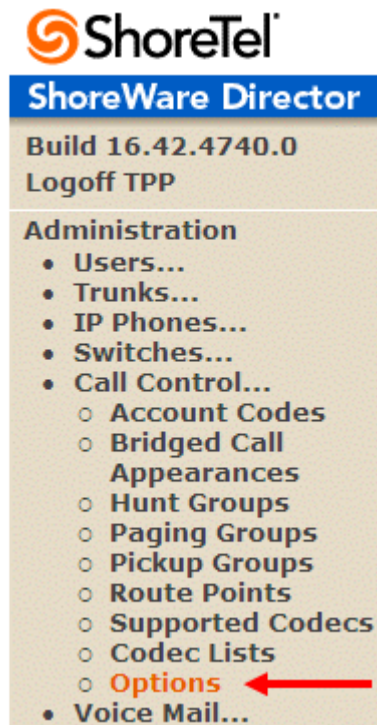


Figure 3 – Administration Call Control Options

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

Call Control Options

Edit

Save

Reset

Help

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: 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.

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" , and "Always Use Port 5004 for RTP" fields.

Unchecking the box for "Always Use Port 5004 for RTP" is required for implementing SIP on the ShoreTel system. For SIP configurations, dynamic User Datagram 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 changes required to the "Edit Site" screen are to the "Admission Control Bandwidth" and "Intra-Site Calls / Inter-Site Calls" parameters.

Bandwidth:

Admission Control Bandwidth: kbps

Intra-Site Calls:

Inter-Site Calls:

FAX and Modem Calls:

Figure 6 – Site Bandwidth settings

Note: Bandwidth of 2046 is just an example. Please see the Planning and Installation Guide for additional information on setting Admission Control Bandwidth and Intra / Inter-Site Calls parameters.

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.

Sites Edit screen – Intra / Inter-Site Calls

By default ShoreTel Version 11 has 11 built-in codecs, these codecs can be grouped as "Codec Lists" and defined in the sites page for "Inter-site" and "Intra-site" calls. Configure the "Inter-Site Calls" option for "Very Low Bandwidth Codecs" and save the change. By default "Very Low Bandwidth Codecs" contains two codecs, G.729 and G.711u, with G.729 being the primary codec of choice.

Note: Please do not modify the "Very Low Bandwidth Codecs" "Codec List".

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" followed by "Primary" in ShoreWare Director (Figure 7).

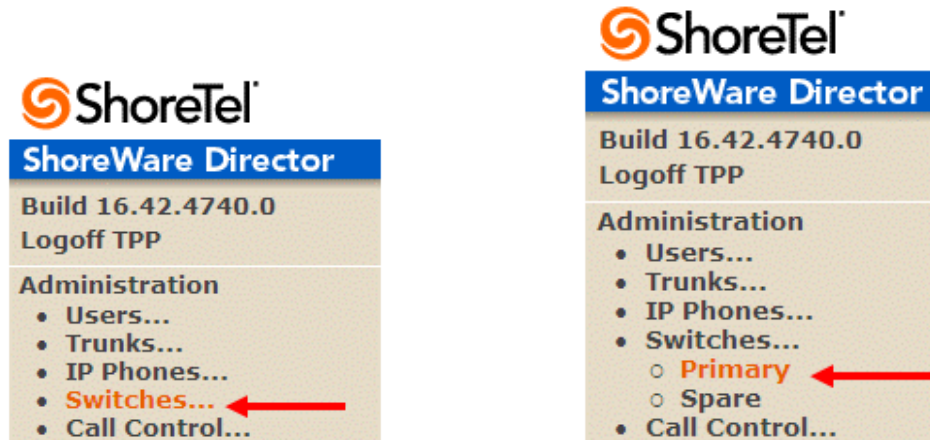


Figure 7 – Administration Switches

This action brings up the "Switches" screen. From the "Switches" screen, simply select the name of the switch to configure. This switch will be the switch that you have determined has enough spare resources for the SIP Trunks and available "make-me" conference ports. Within the "Edit ShoreGear ...Switch" screen, select the desired number of SIP Trunks from the ports available (Figure 8). Please make sure that you also enable four "make-me" conference port resources, if at least four are not configured elsewhere on the ShoreTel system. Conference resources are required on the switch for call-transfer and 3-way conference calls to function as expected. These resources may be on any switch that has spare ports and supports "make-me" conference resources.

Switches
Edit ShoreGear 60/12 Switch

New Copy Save Delete Reset

Edit this record Refresh this page

Name: TPP-SG60-1

Description:

Site: Sunnyvale TPP Lab

IP Address: 10.40.12.32 Find Switches

Ethernet Address: 00-10-49-00-47-D0

Server to Manage Switch: Headquarters

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

Music On Hold Source

Port	Port Type	Trunk Group	Description	Jack Number
1	Conference		p01	
2	Conference		p02	
3	Conference		p03	
4	Conference		p04	
5	5 SIP Trunks		p05	
6	5 SIP Trunks		p06	

Figure 8 – ShoreGear Switch Settings

Each port designated as a 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 9).



Figure 9 – Administration Trunk Groups

This selection brings up the "Trunk Groups" screen (Figure 10).

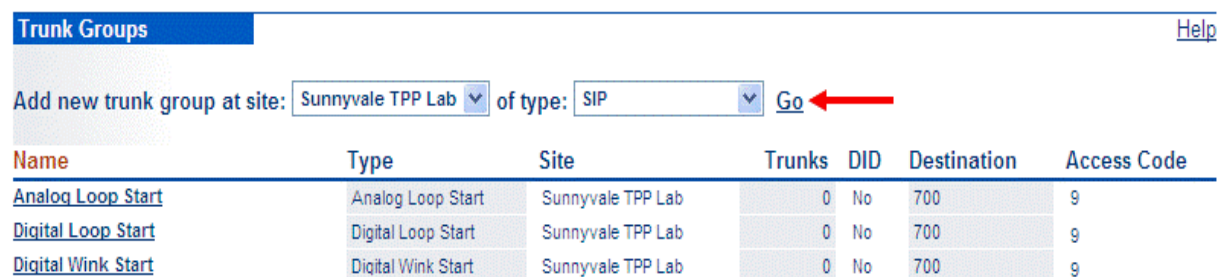


Figure 10 – Trunk Groups Settings

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 11).

Trunk Groups

Edit SIP Trunk Group

[New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#) [Help](#)

Edit this record		Refresh this page	
Name:	<input type="text" value="SureTrunk_SIP"/>	←	
Site:	Sunnyvale TPP Lab		
Language:	<input type="text" value="English(US)"/>	▼	
<input type="checkbox"/> Teleworkers			
<input checked="" type="checkbox"/> Enable SIP Info for G.711 DTMF Signaling			
Profile:	<input type="text" value="_ATTBVOIP"/>	▼ ←	
Digest Authentication:	<input type="text" value="<None>"/>	▼	
User ID:	<input type="text"/>		
Password:	<input type="text"/>		
Inbound:			
Number of Digits from CO:	<input type="text" value="10"/>	←	
<input checked="" type="checkbox"/> DNIS	<input type="text" value="Edit DNIS Map"/>		
<input checked="" type="checkbox"/> DID	<input type="text" value="Edit DID Range"/>		
<input checked="" type="checkbox"/> Extension			
<input checked="" type="radio"/> Translation Table:	<input type="text" value="<None>"/>		
<input type="radio"/> Prepend Dial In Prefix:	<input type="text"/>		
<input type="radio"/> Use Site Extension Prefix			
<input checked="" type="checkbox"/> Tandem Trunking			
User Group:	<input type="text" value="Executives"/>		
Prepend Dial In Prefix:	<input type="text"/>		
Destination:	<input type="text" value="700 : Default"/>	<input type="button" value="Search"/>	

Figure 11 – 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 with a fixed IP address provided for each of the individual trunks.

The next step within the "Edit SIP Trunks Group" screen is to input the name for the trunk group. In the example in Figure 11, the name "SureTrunk_SIP" has been created. The next step is to verify the setting of the "Teleworker" check box. The "Teleworker" check box should not be checked.

The "Enable Digest Authentication" field is not required.

The "Enable SIP Info for G.711 DTMF Signaling" box should be checked. Enabling SIP info is currently for some advanced features such as ShoreTel Office Anywhere.

Ensure that "Profile:" has "ATTBVOIP" selected. ATTBVOIP is a Reinvite profile that enables a newer way for EtherSpeak SIP trunks to communicate within ShoreTel version 9.x, 10.x, 11.x.

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.

The next item to change in the “**Edit SIP Trunks Group**” screen is to make the appropriate settings for the “**Outbound:**” fields (Figure 12).

Outbound: ←

Network Call Routing:

Access Code: ←

Local Area Code: ←

Additional Local Area Codes:

Nearby Area Codes:

Billing Telephone Number: (e.g. +1 (408) 331-3300)

Figure 12 – SureTrunk Trunk Group Setup:

Trunk Services:

Local

Long Distance

International

Enable Original Called 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

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:

Figure 13 – Trunk Services and Trunk Digit Manipulation:

On the “**Trunk Services:**” and “**Trunk Digit Manipulation:**” 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 affect 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 14).



Figure 14 – Individual Trunks

The "Trunks by Group" screen that is used to change the individual trunk settings will appear (Figure 15).

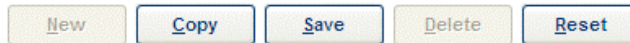


Figure 15 – Trunks by Group

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 16).

Trunks

Edit Trunk



[Help](#)

Edit this record Refresh this page

Site:	Sunnyvale TPP Lab
Trunk Group:	SureTrunk_SIP
Name:	<input type="text" value="SureTrunk_SIP"/>
Switch:	<input type="text" value="TPP-SG120-3"/> ▼
IP Address:	<input type="text" value="10.20.110.121"/>
Number of Trunks (1 - 220):	<input type="text" value="1"/>

IP Address of SureTrunk
Signaling Interface
Accessed via IPSEC.




Figure 16 – 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 first 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 1 (866) Ether-IP.

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.

ShoreTel SSL Phone Settings

This section covers the configuration of the ShoreTel SSL Phone for use with SIP. The ShoreTel SSL Phone provides a great way to connect remote workers to headquarters. By utilizing a SSL VPN Concentrator, the ShoreTel Administrator had an effective way to securely connect small-office-home-office (SOHO) workers back to the headquarters ShoreTel server.

For EtherSpeak SIP to work properly, an additional static route needs to be added to the ShoreTel VPN Concentrator to enable hair-pinning of the RTP or media packets through the VPN concentrator and out to the Internet or MPLS gateway.

Configuration Menu

- ◆ Network
- ◆ Stunnel
- ◆ System
 - ◆ Network Information
 - ◆ Network Restart
 - ◆ Network Test Tools
 - ◆ Reboot System
 - ◆ Route
 - ◆ Services Configuration
 - ◆ Set Link
 - ◆ System Information
 - ◆ System Time
 - ◆ Upgrade Firmware
 - ◆ VLAN Configuration

[Help](#)

Route

The Route page is used to setup static routes to hosts or networks.

Update Route:

IP Network:

Netmask:

Gateway:

Delete Route:

Currently Configured Routes:

IP Network	Netmask	Gateway
10.0.0.0	255.0.0.0	10.2.100.245
74.84.203.90	255.255.255.255	10.2.100.245

Figure 17 – SSL VPN Concentrator Static Route Settings

In this illustration, the unit is default routed to the firewall, with static routes pointing to an internal router and EtherSpeak’s media / RTP IP Address.

The gateway was on subnet A, and the voice switch is on subnet B. Seeing as the firewall either doesn’t issue icmp redirects, or the gateway might ignore them, a static route is needed to point the box at the EtherSpeak media / RTP IP address of 74.84.203.90 and conversely, to the switch that leads to the voice VLAN.

Adding this configuration to an SSL phone VPN concentrator will allow use of the EtherSpeak SIP trunks with remote SSL phone based users.

EtherSpeak Configuration

Signing up for SureTrunk with EtherSpeak is easy.

Simply go to www.suretrunk.com and register for service with our online provisioning form. Or simply call toll free (866) 384-3747.

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.

SIP Trunk Request Form

Please provide us with the information below so we can assist you with your inquiry.

Company Information

Please provide us with general information about your company.

Company Name *

Company Address

Street Address

Address Line 2

City

Postal / Zip Code

State / Province / Region

Country

Company Phone

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

Enter LAN, ShoreTel IP PBX and Firewall Info

ShoreTel Environment

Please provide us with information about the environment.

A. Reference Your
ShoreTel Reseller

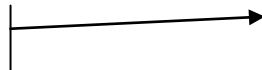


ShoreTel VAR:

ShoreTel Director SW Version

ShoreTel Director IP Address

B. ShoreTel Director and
Switch Info



ShoreTel Switch IP Address

C. Firewall Make, Model
and IP Info



Firewall Make

Firewall Model

Firewall SW/FW Version

Firewall Public IP

D. Scroll to Next Section

Firewall Private IP

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.

**Request Numbers Section
(ELS and LI)**

A. Number of new numbers

B. NPA-NXX for number request

C. Click Submit.

Order Information
Please provide us with information about the number of trunks desired and the numbers to port.

Number of Trunks

Numbers to Port

Comments/Notes:

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.

EtherSpeak 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 Technical Support

Worldwide Customer Support: +1 (703) 221-9999

Email: support@ietherspeak.com

Website: www.suretrunk.com

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