

ShoreTel, Ingate & BandTel for SIP Trunking

SIP Trunking allows the use of Session Initiation Protocol (SIP) communications from an Internet Telephony Service Provider (ITSP) such as BandTel instead of the typical analog, Basic Rate Interface (BRI), T1 or E1 trunk connections. Having the pure IP trunk to the Internet Telephone Service Provider allows for more control and options over the communication link. This application note provides the details on connecting the ShoreTel® IP phone system through an Ingate box which is connected to both the LAN and WAN and acts as a gateway to BandTel for SIP Trunking.

Table of Contents

Overview	2
Vendor Overview and Contact	2
Contact Information	2
Ingate Systems	2
North America	3
ЕМЕА	3
Vendor Product Information	4
Figure 1 - Ingate Firewalls and SIParators	4
Architecture Overview	4
Figure 2 - Architectural Overview	5
Figure 3 - Ingate, 3 Possible Options	6
Requirements, Certification and Limitations	6
Version Support	6
BandTel Certification Testing Results Summary	7
Table 1: Basic Feature Test Cases	7
Table 2: Extended Feature Test Cases	8
Configuration Overview	8
ShoreTel Unsupported Features	8
ShoreTel Configuration	9
ShoreTel System Settings - General	9
Figure 4 -Administration Call Control Options	9
Figure 5 - Call Control Options	. 10
Figure 6 - Administration Site	. 11
Sites Edit Screen - Admission Control Bandwidth .	. 11
Switch Settings - Allocating Ports for SIP Trunks	. 12
Figure / -Administration Switches	. 12

Figure 8 - ShoreGear Switch Settings	13
ShoreTel System Settings - Trunk Groups	13
Figure 9 - Administration Trunk Groups	14
Figure 10 - Trunk Groups Settings	14
Figure 11 - SIP Trunk Group Settings	15
Figure 12 - Inbound:	16
Figure 13 - Trunk Services	17
ShoreTel System Settings - Individual Trunks	17
Figure 14 - Individual Trunks	18
Figure 15 - Trunks by Group	18
Figure 16 - Edit Trunks Screen for Individual Trunk	s19
Ingate Configuration	19
Alternative A: Configuration using the Ingate Start	up
Tool	19
Alternative B: Configuration through the GUI	22
Ingate Troubleshooting	25
Troubleshooting Outbound Calls	25
Troubleshooting Inbound Calls	27
Ingate Technical Support	30
BandTel Configuration & Support	31
BandTel Special Configuration Parameters	31
BandTel Support information	31
Document and Software Copyrights	31
Trademarks	31
Disclaimer	32
Company Information	22
	JZ

Overview

This document provides details for connecting the ShoreTel system though the Ingate SIParator[®] to BandTel for SIP Trunking to enable audio communications. The document specifically focuses on the configuration procedures needed to set up these systems to interoperate.

Vendor Overview and Contact

BandTel Overview

BandTel is a leading global provider of VoIP termination to the Public Switched Telephone Network (PSTN) using SIP. BandTel's SIP Trunk solutions are uniquely designed for Value-Added Resellers and Managed Service Providers for your SMB and enterprise end customers. BandTel SIP Trunks provide VoIP origination and termination and global PSTN interconnection for enterprise IP telephony solutions from ShoreTel.

BandTel services include:

- Local inbound/outbound calling
- Interstate and international long distance calling
- Direct Inward Dial (DID) numbers
- 800 numbers
- Toll free calling
- Local number porting
- Emergency 911 support
- 411 directory assistance
- Caller ID
- Off-premise extensions for mobility
- Standard codec support: G.711 and G.729A
- Customer service portal for online service management
- Mobility and remote worker solutions
- Disaster recovery

<u>Bottom line</u>: BandTel SIP Trunking solutions provide all of the features of traditional PSTN telecom services and more at up to 40% less than the cost for legacy services. Combined with superior quality and ease of use, ShoreTel and BandTel are ready to deliver today!

Contact Information

Sales & Marketing Office Tel: 949-640-9700 13 Corporate Plaza Suite 200 Newport Beach, CA 92660

Ingate Systems offers the only fully SIP-capable security products offering features important to enterprise adoption of SIP Trunking. The Ingate Firewall® offers a single device to protect the network and manage SIP traffic. The Ingate SIParator® allows the enterprise to adopt SIP without replacing its existing firewall. Both products include an SIP Application Layer Gateway (ALG), proxy and registrar that enable SIP signaling to traverse the firewall. They also provide support for dynamic media port management to keep the network safe,



960 Stewart Drive Sunnyvale, CA 94085 USA Phone +1.408.331.3300 +1.877.80SHORE Fax +1.408.331.3333 www.ShoreTel.com

encryption for privacy, added routing capabilities to make the installation of SIP Trunks simple and inexpensive, and remote SIP connectivity so that the enterprise can offer SIP services to its remote workers.

North America

For general sales questions, please contact reseller or contact Ingate directly at: Steven Johnson 603-883-6569 or <u>Steve@ingate.com</u> www.ingate.com

Resellers who want to start selling this solution should contact: Steven Johnson 603-883-6569 or <u>Steve@ingate.com</u> www.ingate.com

EMEA

For general sales questions, please contact reseller or contact Ingate directly at: Ingate Systems HQ +46 86007750 or <u>sales@ingate.com</u> www.ingate.com

Resellers who want to start selling this solution should contact: Ingate Systems HQ +46 86007750 or <u>sales@ingate.com</u> www.ingate.com

Vendor Product Information



Figure 1 - Ingate Firewalls and SIParators

Architecture Overview

ShoreTel, Ingate, and BandTel have collaborated to build a solid solution for businesses. ShoreTel provides the IP PBX which resides on the LAN, interfaces with the Ingate SIParator / firewall, and connects to the SIP trunks provided by BandTel. The Ingate connection to the LAN and WAN provides standard firewall capabilities as well as intelligent SIP routing and the following SIP features:

- Registration
- Digest Authentication
- Dial Plan Modification
- Back-to-back User Agent (Terminates SIP messaging on both the LAN and WAN side)
- Transfer conversion of SIP REFER to SIP reINVITE messaging (critical)
- Quick configuration templates for BandTel



The image below shows a high level drawing of a basic ShoreTel / Ingate / BandTel design. This drawing only represents SIP and Real-time Transfer Protocol (RTP) traffic. The next section of this application note covers actual deployment design options.



Figure 2 - Architectural Overview

Ingate has two products for this solution, the Ingate Firewall and the Ingate SIParator. From an SIP functionality point of view, they are basically the same. The Ingate Firewall also provides normal data firewall functionality and is recommended if the enterprise wants to replace their existing firewall. The Ingate SIParator is the solution for those who want to keep an existing firewall when adopting SIP. In this case, the Ingate SIParator will co-exist in parallel with the normal data firewall.

The routing of SIP traffic to the Ingate SIParator can be accomplished in three primary ways. The first is the most commonly deployed though each configuration offers its own advantages for the enterprise:

- Configuration 1: Single leg/DMZ only, firewall logs all activity
- Configuration 2: DMZ/LAN, reduced load on firewall

• Configuration 3: Two legged/standalone, SIP traffic separate from data traffic



Figure 3 - Ingate, 3 Possible Options

Requirements, Certification and Limitations

Any Ingate SIParator or Ingate Firewall model will work in this configuration. In a Trunking scenario, it is required to have the Ingate SIP Trunking module installed.

A few traversal licenses are included with the Ingate unit at delivery. Typically one traversal license will be needed for each expected concurrent phone call on the SIP Trunk. Additional licenses can be bought via your Ingate reseller.

Version Support

Products are certified via the Technology Partner Certification Process for the ShoreTel system. The table below contains the matrix of Ingate Firewall and Ingate SIParator versions firmware releases certified on the identified ShoreTel software releases.

	Ingate Firewall and Ingate SIParator version						
	4.5.1 with the patch ig-patch- 4-5-1-shoretel- 2 applied	4.5.2	4.6.0	4.6.1			
ShoreTel 7.0	\checkmark	\checkmark	\checkmark	~			
ShoreTel 7.5	\checkmark	\checkmark	\checkmark	\checkmark			

BandTel Certification Testing Results Summary

ID	Name	Description	Results
1.1	Device initialization with	Verify successful startup and initialization of the device	PASS
	static IP address	up to a READY/IDLE state using a static IP address	
1.2	Device reset – idle (for	Verify successful re-initialization of device after power	PASS
	static configurations)	loss while device is idle	
1.3	Device initialization with	Verify successful startup and initialization of the device	PASS
	DHCP	up to a READY/IDLE state using Dynamic Host	
		Configuration Protocol (DHCP)	
1.4	Device reset – idle (for	Verify successful re-initialization of device after power	PASS
	dynamic configurations)	loss while device is idle	
1.5	Verify DiffServ Code	Verify the ability to set DiffServ Code Point from SIP	PASS
	Point support	DUT	
1.6	Verify Date and Time	Verify setting of Date and Time Update on SIP Device	PASS
	Update support	Under Test (DUT)	
1.7	Place call	Verify successful call placement with normal dialing to a	PASS
		variety of terminating phones	
1.8	Receive call	Verify successful reception of calls with normal dialing	PASS
		from a variety of calling phones	
1.9	CODEC support –	Verify successful call connection and audio path using all	PASS
	common (from DUT to	supported CODECs (G.711-Ulaw and G.729)	
	ShoreTel Phone, REF-x)		
1.10	CODEC support –	Verify successful call connection and audio path using all	PASS
	common (from DUT to	supported CODECs (G.711-Ulaw and G.729)	
	SIP Reference Phone,		
	SIP-Ref)		
1.13	CODEC support –	Verify successful negotiation between devices configured	PASS
	negotiated	with different default CODECs (G.711-Ulaw and G.729)	
1.15	Hold from ShoreTel	Verify successful hold and resume of connected call	PASS
	Phone		
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	Verify successful transmission of in-band and out-of-	PASS
	Dual-tone Multi-	band digits (RFC2833) for calls placed to and from the	
	frequency (DTMF)	DUT with a variety of other devices	
	Transmission		
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	Verify successful Auto Attendant transfers between	PASS
	detection G.711	devices configured with CODECs G.711-Ulaw	
1.23	Auto Attendant DTMF	Verify successful Auto Attendant transfers between	PASS
	detection G.729	devices configured with CODECs G.729	
* must b	e on Ingate 4.5.2 or later	· · · · · · · · · · · · · · · · · · ·	•

Table 1: Basic Feature Test Cases

960 Stewart Drive Sunnyvale, CA 94085 USA Phone +1.408.331.3300 +1.877.80SHORE Fax +1.408.331.3333 www.ShoreTel.com

ID	Name	Description	Notes
2.1	Call waiting	Verify appropriate notification and successful connection	PASS
		of incoming call while busy with another party	
2.2	Park	Verify successful park and retrieval of connected call	PASS
2.3	Extended forward	Verify extended call forwarding options – busy	PASS
		forwarding, no-answer forwarding	
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	Verify successful connection of incoming call on	PASS
	line	secondary line	
2.10	Callback	Verify successful connection of a call using the missed-	PASS
		call callback feature of the device	
2.11	Headset	Verify the device's support for external headsets (using	PASS
		headsets supplied by the 3P phone vendor)	
2.12	Ring selection	Verify the device's ability to change the ring type	PASS
2.13	Caller ID Name and	Verify that Caller ID name and number are sent and	PASS
	Number	received from SIP endpoint device	
2.14	SIP Device Generates	Verify that SIP DUT generates busy tone when calling a	PASS
	Busy Tone	busy extension	
2.15	Verify handling of "911"	Verify dialing "911" on DUT can connect with "911"	PASS
		services	
2.16	Verify Fax Handling	Verify that fax can be sent and received through DUT	PASS
2.17	Caller ID Name and	Verify that Caller ID name and number is sent and	PASS
	Number Variations	received	

Table 2: Extended Feature Test Cases

Configuration Overview

The configuration information below shows examples for configuring the ShoreTel, Ingate and BandTel. Even though configuration requirements can vary from setup to setup, the information provided in these steps, along with the Planning and Installation Guide and documentation provided by Ingate and BandTel should prove to be sufficient. However, every design can vary and some may require more planning then others.

ShoreTel Unsupported Features

At the time of this writing, the following features are not supported, though support will be added in an upcoming future release:

- Fax redirect not supported today via SIP Trunks (though direct Direct Inward Dialing (DID) to fax endpoint is supported)
- Office Anywhere



ShoreTel Configuration

This section describes the ShoreTel system configuration to support SIP Trunking and is divided into the general system settings and trunk configurations (both group and individual) needed to support SIP Trunking.

Note: ShoreTel basically just points its Individual SIP Trunks to the Ingate SIParator.

ShoreTel System Settings - General

The first settings to address within the ShoreTel system are the general system settings. These configurations include the Call Control, the Site and the Switch Settings. If these items have already been configured on the system, skip this section and go directly to the "ShoreTel System Settings – Trunk Groups" section below.

Call Control Settings:

The first settings to configure within ShoreWare® Director are the Call Control Options. To configure these settings for the ShoreTel system, log into ShoreWare Director and select "Administration" then "Call Control" followed by "Options" (**Figure 4**).



Figure 4 -Administration Call Control Options

The "Call Control Options" screen will then appear (Figure 5).



Call Control Options Edit	Save <u>R</u> eset	Help
Edit this record	Refresh this page	
Enable SIP Session Timer.		
Session Interval (0 - 9999):	1800 sec	
Refresher:	Caller (UAC)	
Voice Encoding and Quality of Service	e:	
Intra-Site Calls:	64 Kbps (G.711)	
Inter-Site Calls:	64 Kbps (G.711)	
FAX and Modem Calls:	64 Kbps (G.711)	
Maximum Inter-Site Jitter Buffer:	50 msec	
DiffServ / ToS Byte (0-255):	0	
C Admission control algorithm assumes	RTP header compression is being used.	
Enable Media Encryption.		
Always Use Port 5004 for RTP.		

Figure 5 - Call Control Options

Within the "Call Control Options" screen, confirm 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 first step is to make sure that the "Enable SIP Session Timer" box is checked. Next the Session Interval Timer needs to be set. The recommended setting for "Session Interval" is 1800 seconds. The last item to select is the appropriate refresher (from the pull down menu) for the SIP Session Timer. The "Refresher" field will be set either to "Caller (UAC)" [User Agent Client] or to "Callee (UAS)" [User Agent Server]. If the "Refresher" field is set to "Caller (UAC)", the Caller's device will be in control of the session timer refresh. If "Refresher" is set to "Callee (UAS)", the device of the person called will control the session timer refresh.

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 RTP Traffic. If the box is unchecked, Media Gateway Control Protocol (MGCP) will no longer use UDP port 5004; MGCP and SIP traffic



960 Stewart Drive Sunnyvale, CA 94085 USA Phone +1.408.331.3300 +1.877.80SHORE Fax +1.408.331.3333 www.ShoreTel.com

will use dynamic UDP ports. Once this parameter is unchecked, make sure that "everything" (IP Phones, ShoreGear® Switches, ShoreWare Director, Distributed Voice Services / Remote Servers, Conference Bridges and Contact Centers) is "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 6**).



Figure 6 - 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 BandTel 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 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 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 8**).

65		AN 1 LAN 2	
9	6G-40/8		
Port	IP Phones	Conference	SIP Trunks
1 Edit	П	п	П
2 Edit	П	п	Π
3			
4			
5			1
6			V
7			•
8			
Night Bell			F
Paging		Π	п

Figure 8 - 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 (ATAs), Conference Phone or WiFi handset), 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).

Trunk Groups						<u>He</u>
Add new trunk grou	p at site: Headquarters	of type: SIP		• <u>G</u>	<u>o</u>	
Name	Туре	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 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**).

Edit SIP Trunk Group	<u>N</u> ew <u>C</u> opy	Save	Delete	Reset	<u>Help</u>
Edit this record	Defrech this news			* m	nodified
Name: Site:I Language: IZ Teleworker	Headquarters		Give the Tr name. Exar HQ-SIP-In	runk Group a mple: gate- BandTe	meaningfu
 Enable Digest Authentication User ID: Password: Enable SIP Info for G.711 DTMF S 	ignaling				

Figure 11 - SIP Trunk Group Settings

For the Ingate SIP Trunking, 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 9, 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 when connecting to an Ingate box.

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 12**).



valuer of Digits from CO.	10	
	Edit DNIS Map	
	Edit DID Range	
Extension		
Translation Table:	<none> 🔻</none>	
Prepend Dial In Prefix:		
Use Site Extension President Site	fix	
Use Site Extension Pri Tandem Trunking	fix	
Use Site Extension Pri Tandem Trunking User Group:	fix Anonymous Telephones 💌	
Use Site Extension Pro Tandem Trunking User Group: Prepend Dial In Prefix:	fix Anonymous Telephones 💌	

Figure 12 - Inbound:

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

Tandem Trunking is not required unless you plan on routing incoming SIP trunk calls out other ShoreTel trunks.

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

Trunk Groups Edit SIP Trunk Group	New	Сору	<u>S</u> ave	Delete	Reset	Help
✓ Outbound:						
Network Call Routing:						
Access Code:						
Local Area Code:						
Additional Local Area Codes:	Edit					
Nearby Area Codes:	Edit					
Trunk Services:						
Local						
🔽 Long Distance						
🔽 International						
🗖 n11 (e.g. 411, 611, except 911 whi	ich is specified	below)				
911						
Easy Recognizable Codes (ERC) ((e.g. 800, 888, 9	900)				
Explicit Carrier Selection (e.g. 101	Оххх)					
Dperator Assisted (e.g. 0+)						
Caller ID not blocked by default						

Figure 13 - Trunk Services

On the "Trunk Services:" screen, make sure that the appropriate services are checked or unchecked based on what BandTel supports and 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 etc. 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 14**).





Figure 14 - Individual Trunks

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

Trunks by Group							Help
Add new trunk at sit	e: Headquarters 💌	in trunk gro	oup:	Analog Loop Start	G	1	
Name/Number	Group	Site	Sw	Analog Loop Start Digital Loop Start	ital	TrunkTypeID	DeviceIPAddress
SIP x3020 Inter-site	SIP x3020 Inter-site	Headquarters	SG-	Digital Wink Start	1	6	10.20.110.120
<u>SIPx3000-3019 Intra-</u> site	<u>SIPx3000-3019</u> Intra-site	Headquarters	SG-	SIP x3020 Inter-site SIPx3000-3019 Intra-site	1	6	0.0.0.0

Figure 15 - Trunks by Group

ShoreTel

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	New	Сору	Save	Delete	Reset		<u>Help</u>
Edit this record	Defrech this pe	20	×			* modified	_
Eunt unis record	Reliesh this pa	<u>ue</u>					
Site:	Headquarters						
Trunk Group:	SIP						
Name:	SIP						
Switch:	SG-40/8 💌						
SIP Trunk Type:							
C Dynamic			Г			1	
Use IP Address	10.20.110.121			IP Address of interface	of Ingate LAN		
Number of Trunks (1 - 12	0): 1			internation.			

Figure 16 - Edit Trunks Screen for Individual Trunks

From the individual trunks "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 ITSP Trunk, select "Use IP Address" button and input an IP address of the Ingate SIParator product. The last step is to select the number of individual trunks desired (each one supports "one" audio path – 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. For redundancy, two trunk groups will be needed with each pointing to another Ingate SIParator – in exactly the same way as if primary rate interface (PRI) were being used.

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.

Ingate Configuration

The Ingate product can be configured using two alternative methods: using the Ingate Startup Tool, a wizard for a complete first time configuration, or the traditional configuration via the GUI. The latter is more suitable if you already have your Ingate configured and operational in your network.

Select one of these methods for configuration of the Ingate unit (the option most suitable for you).

Alternative A: Configuration using the Ingate Startup Tool

When you have received your Ingate device, unpack it and connect it to the network according to the picture below (in the screen shot). Install the Startup Tool on a Windows PC and start the tool. Make sure that your Ingate device is turned on.

Note: The configuration tool is available now as a free download for all Ingate Firewalls and SIParators. It can be found at <u>www.ingate.com/SIPtrunkingconfigtool.php</u>

If you don't have the Trunking module installed, the tool will prompt you for license information and automatically install the Trunking module before continuing to the trunk configuration.

Step I - IP provisioning:

Accomplish basic network setup by following steps A-D in the picture below. Note that if you already provided your Ingate unit an IP address, you don't need to do the MAC address part, but *all your old settings will be replaced by the tool.*





Step II - Status Information provide:



Step III - SIP Trunk configuration:

In the next III, the Product Type and net work information is configured. Follow steps A-D in the picture below. In this example, Standalone SIParator was selected as the Product Type.



1	💽 Ingate Startup Tool - Network Topology	×	
A. Configure Product	Preduct T: Inside (Interface Etho) IP address: 10 , 3 , 0 , 2	Internet	
B. Configure Netmask for internal network	Netmask: 255 255 0 Outside (Interface Eth1)	Exising freewalt	
C. Configure external interface using DHCP or static IP	Gateway: 0 , 0 , 0 , 0 - Remote SIP Connectivity	Help Help SIP Tranking F Enable Next step - load configuration Launch	
	Status Advanced SIP Routing Remote SIP Connectivity VPN The application has succesfully contacted your ur Logged out from the web interface of the Ingate	it, press the launch button for next step.	D. Click Launch again.

Step IV - Tool Configuration:

Once "Yes" is selected, the Ingate Startup Tool will remove your old configuration data.

StartupT	Tool	X
1	Running the last step of the Startup Tool will remove all configuration you previously may have manually entered through the will ingate unit. Are you sure you want to continue?	eb interface of the

Step V -SIP Trunk Provider Configuration:

In step V, the SIP Trunk itself is configured. Follow steps A-E in the picture below. In this example, BandTel is the ITSP provider.



	⑥Ingate Startup Tool - Create and Load Configuration	
Select BandTel rom the drop-down nenu and provide ecessary account nformation.	Trurking Provider	D. dr ac Co
3. Configure Provider IP Address, this will be provided by BandTel	Password: Help Discusse account on incoming call Help Disclaimer While trapped has made every effort to ensure element activation and/or contract on	
C. Configure DNS server	Stats Advanced SP Broking Results SP Construct, which is successfully contacted your unit, press the learch button for next steeping of from the web interface of the ingate unit. E. When all settings are entered, the tool will generate a configuration based on your input, and you will automatically	
C. Configure DNS server	E. When all settings are entered, the tool will generate a configuration based on your input, and you will automatically be redirected to the Ingate. You only need to apply the	_

configuration and then start using it!

D. Select ShoreTel from the drop-down menu and IP address of ShoreGear Switch Configured for SIP Trunks.

Step VI - Success - SIP Trunk Configured:



Alternative B: Configuration through the GUI

Configure your Ingate Firewall or Ingate SIParator to get basic network connectivity on all applicable interfaces. Please refer to the Reference Guide and other documentation as needed.

Remember to configure the following:

- Assign IP addresses on the inside and outside interface. For DMZ SIParators, use one interface only. (Network -> All Interfaces)
- Assign a default gateway. (Network -> Default Gateway)
- Assign a DNS server address. (Basic Configuration -> Basic Configuration)

- Define the IP subnet allowed to configure the Ingate and the interfaces to use for configuration. (Basic Configuration -> Access Control)

First make these basic settings and apply the configuration to have the unit working in your network environment. Then proceed with the following settings to get SIP Trunking to work with your service provider.

NETWORK - NETWORK AND COMPUTERS

- Add a network for the Service Provider (ITSP IP). If you don't know the IP addresses used, you can put in 0.0.0.0 as lower limit and 255.255.255.255 as upper limit. In this way, requests from any IP address will be accepted.
- Add a network for the LAN (inside IP range).

Networks and Co	omputers						
	G 1	Lower limit		Upper (for IP ra	limit Inges)		Delete
Ivame	Subgroup	DNS name or IP address	IP address	DNS name or IP address	IP address	Interface/VLAIN	Row
+ LAN	- 🗸	10.100.0.0	10.100.0.0	10.100.0.255	10.100.0.255	inside (ethO untagged) 🛛 👻	
+ ITSP_IP	- 🗸	0.0.0.0	0.0.0.0	255.255.255.255	255.255.255.255	outside (eth1 untagged) 🛛 👻	

BASIC CONFIGURATION - SIParator TYPE (SIPARATOR ONLY)

Use the appropriate SIParator configuration for your deployment.

SIP SERVICE - BASIC

• SIP Module: On.

SIP Traffic – Filtering

Under Proxy Rules, change the Default Policy for SIP Requests to "Process All".

Proxy Rules (Help)	
No From Action Delete	Default Policy For SIP Requests
network Row	⊙ Process all
	🔿 Local only
Add new rows	🔿 Reject all

SIP TRAFFIC – USER DATABASE

Configure an account with details as provided from the ITSP.

SIP TRAFFIC - DIAL PLAN

Configure the Dial Plan according to the picture below.



Basic figuration	istration Network Lo	gging SIP Services T	SIP Virtual Private Networks	Quality of Service	About							
P Methods Filtering	User Authentico Database and Accou	ation Dial Plan Re	outing SIP Status									
Use Dial Plar © On © Off © Fallback	1 <u>(Help)</u> En 911	iergency Num	ıber <u>(Help)</u>								The ir	nternal
Matching Fro	om Header (Helt	<u>a)</u>									LAN	
_	Use t	his	or this					_	/		I	
Name	Username	Domain	Reg Exp	Transp	ort Net	work]	Delete R	.ow				
ITSP	*	*		UDP	ITSP							
LAN	*	*		UDP			T	_			IP or o	domai
Add new rows	1 rows.	u)	,	,	,						name Ingate exterr	of the anal
		2	II this					4h:1-			Interra	ace
Name	Prefix	Head	Tail	Min T	'ail	Domain	n	Rev Exn	Delete Row			
Inhound	+1		any character 🔻			172 118		Joe Tub				
Outhound			lany character			0.0						
		J	ally character	<u>µ</u>	10.3	.0.2		IP or c Ingate	domain na e internal	ame of the interface		
		<u>,</u>	ally Character	,	10.3	.U.2		IP or d Ingate IP or de Shore	domain na internal omain nai Gear switc	ame of the interface me of the ch		
Forward To	(<u>Heb</u>)	J	ally Character	,	110.3	.0.2		IP or o Ingate IP or do ShoreO	domain na e internal omain na Gear switc	ame of the interface me of the ch		
Forward To	(Help)	Use this	jaiy Character	or jka	10.3	.0.2		IP or do Ingate IP or do Shore O	domain na internal omain na Gear switc	ame of the interface me of the ch		
Forward To Name	(Help) Subno.	Use this	Replacement L	or the	ji0.3	ranspo	nt F	IP or de Ingate IP or de Shored	domain na internal omain na Gear switc	ame of the interface me of the ch		
Forward To Name + IP-PBX	(Help) Subno.	Use this Account	Replacement L	or that Just F	jio.3	ranspo	ort F	IP or de Ingate IP or de Shore or this Reg Exp	domain na internal omain na Gear switc	ame of the interface me of the ch		
Forward To Name + IP-PBX + ITSP	(Help) Subno.	Use this Account	Replacement [[10.3.0.39	or the	s Fort TJ			IP or de Ingate IP or de Shore or this Reg Exp	domain na internal omain na Gear switc	ame of the interface me of the ch		
Forward To Name + IP-PBX + ITSP	(Help) Subno. 1 1	Use this Account	Replacement I 10.3.0.39 4.79.212.236	or the JR1 F 5060	s Port Tr D U	ranspo DP •	nrt F	IP or de Ingate IP or de Shore or this Reg Exp	domain na internal omain na Gear switc	ame of the interface me of the ch		
Forward To Name + IP-PBX + ITSP	(Help) Subno.	Use this Account	Replacement U 10.3.0.39 4.79.212.236 216.82.224.202	or the Jet F 5060 5060	Sort Dort D D D D U D D U	anspo DP DP DP		IP or de Ingate IP or de Shore C	domain na internal Domain na Gear switc	ame of the interface me of the ch		
Forward To Name + IP-PBX + ITSP Add new rows	(Help) Subno. 1 2 1 2 1 2	Use this Account	Replacement U 10.3.0.39 4.79.212.236 216.82.224.202 rows per group.	or the Jet F 5060 5060	10.3	anspo DP • DP •		IP or de Ingate IP or de Shore C	domain na internal Delete R	ame of the interface me of the ch		
Forward To Name + IP-PBX + ITSP Add new rows Dial Plan ((Help) Subno. 1 2 1 2 Help)	Use this Account	Replacement U 10.3.0.39 4.79.212.236 216.82.224.202 rows per group.	or the	Sort T Oort U D U D U	anspo DP • DP •		IP or do Ingate IP or do Shore O	domain na internal Delete R	ame of the interface me of the ch		
Forward To Name + IP-PBX + ITSP Add new rows Dial Plan (No. 1 H	(Help) Subno. 1 1 2 group Help) From Reque URI	Use this Account	Replacement I 10.3.0.39 4.79.212.236 216.82.224.202 rows per group. Action	or this Ref F 5060 5060	s Port TJ D U D U D U D U D U D U	ranspo DP • DP •	nrt F	IP or do Ingate IP or do ShoreO	domain na internal Delete R	ame of the interface me of the ch ow ENUM Root	Comment	Deleta Row
Forward To Name + IP-PBX + ITSP Add new rows Dial Plan (No. 1 H	(Help) Subno. 1 1 2 1 2 1 5 From Reque URI SP V Inhaund	Use this Account	Replacement I 10.3.0.39 4.79.212.236 216.82.224.202 rows per group. Action	or the JRA F 5061 5066	Forward To IP-PBX		ort F	IP or do Ingate IP or do ShoreO	domain na internal Delete R	ame of the interface me of the ch ew ENUM Root	Comment	Deleta Row
Forward To Name + IP-PBX + ITSP Add new rows Dial Plan (No. 1 H 1 1	(Help) Subno. 1 1 2 1 2 From group Help) From Reque URI SP Inbound AN Cuttors	Use this Account	Replacement [[10.3.0.39] [4.79.212.236] [216.82.224.202] rows per group. Action	or the	10.3		ort F	IP or do Ingate IP or do ShoreO	domain na e internal Dear switc Delete R	ame of the interface me of the ch ew ENUM Root	Comment	Delet Row
Forward To Name F IP-PBX F ITSP Add new rows Dial Plan (No. 1 H 1 1 1 2 1	(Help) Subno. 1 1 2 1 2 4 4 4 5 7 5 7 1 5 7 1 8 6 4 7 7 1 9 7 9 7 1 9 7 9 7 1 9 7 9 7 9 7 1 9 7 9 7	Use this Account	Replacement [10.3.0.39 4.79.212.236 216.82.224.202 rows per group. Action	or the	10.3 Is Port Tr D U U U U U U U U U U U U U U U U U </td <td>ranspo DP v DP v</td> <td>ort F</td> <td>IP or de Ingate IP or de Shore a. or this Reg Exp</td> <td>domain na pinternal Delete R</td> <td>ame of the interface me of the ch ow ENUM Root</td> <td>Comment</td> <td>Delet Row</td>	ranspo DP v DP v	ort F	IP or de Ingate IP or de Shore a. or this Reg Exp	domain na pinternal Delete R	ame of the interface me of the ch ow ENUM Root	Comment	Delet Row

SIP TRAFFIC – ROUTING

• Local REFER handling: check Always handle REFER locally.

The configuration of the Ingate is now done and the changes must be applied on the Administration page to take effect.



Ingate Troubleshooting

Troubleshooting Outbound Calls

Symptom: When trying to make a call from an internal ShoreTel extension to PSTN, there is no ringing signal on the PSTN phone.

Note: If you get a ringing signal on the PSTN phone, these troubleshooting steps will not help you to find the problem. Please contact your sales representative for support.



Outbound traffic troubleshooting overview

Get a log for the failing call:

First try to make a call to a PSTN number from a ShoreTel phone and notice the behavior on the ShoreTel phone as well as on the PSTN phone.

Next step is to search the log on the Ingate. Log into the Ingate box and navigate to the Display Log page. Make necessary settings on this page according to the picture below. Especially make sure that you have the highlighted checkboxes in the correct state.

stet selection: only those packets that meet the search criteria in the three tions below will be selected. This selection will only have effect on the ${f IP}$	Show newest at top
kets as selected choice.	Time Limits
Packet Type Selection	Show log from: (clear)
Il packets	MM-DD) (HH MM SS
P Address Selection (Help)	Show log until: (clear)
🔲 not this address	date (YYYY- time
🔲 not this address	MIM-DD) (HH:MM;SS
A src A dst A any A to B B to A Between A&B In ot this combination.	Show This
	Configuration server logins
9 An ir protocols	🗌 Administration and configuratio
○ TCP ○ All ports ○ UDP ⊙ Selected ports: (Help)	□ Manual reconfigurations and reboots
A: D not this port	Time changes
B: not this port	DHCP/PPPoE client
⊙A src ⊙A dst ⊙A any	RADIUS errors
○ A to B ○ B to A ○ Between A&B □ not this combination	SIMP problems
OTCMP Select twoe/code: (Helm)	🔲 Hardware errors
Type:	🔲 Mail errors
Code:	🗌 Negotiated IPsec tunnels
	🔲 IPsec key negotiations
DESP	🔲 IPsec user authentication
Protocol number (Help)	PPTP negotiations
TALEY NEW YEAR AND AND A TRADUCT	SP errors

Then press "Display log" further down on the same page.

You will now see a log of all SIP packets received and sent by the Ingate, with the newest log entry on the top.

V

Ensure the signaling is received from the ShoreTel:

Localize the call initiation from the ShoreTel by searching for "invite sip" in your browser. You should look for the first packet coming from the ShoreTel system that starts with a "recv from <IP address of the ShoreGear switch>" as you can see in the example (only the first lines of the log messages are shown here).

>>> Info: sipfw: recv from 10.100.0.40:5060 via UDP connection 12746: INVITE sip:16037914522@10.100.0.13:5060 SIP/2.0



If you cannot find a packet like the one above, the problem is in the communication from ShoreGear to the Ingate. Follow these steps:

- 1. Make sure the Ingate SIP module is turned on, SIP Services SIP Module On. Retest if you change any settings.
- 2. Make sure the ShoreTel configuration is correct. Check the IP address pointing at Ingate one extra time. Retest if you change any settings.
- 3. Make sure there is IP connectivity between the ShoreTel and Ingate. Contact your network administrator for assistance if needed.

If none of the steps above solves the problem, you can contact your sales representative for support.

Ensure that the signaling to the ITSP works:

If you find the incoming packet, you should find a similar packet leaving the Ingate just above (just after in time) the incoming packet. The first rows of the outgoing packet will look something like this:

```
>>> Info: sipfw: send sf (0x8422820) to 208.49.124.49:5060 via UDP connection 12748:
INVITE sip:16037914522@208.49.124.49:5060;transport=udp SIP/2.0
```

If you don't see the outgoing packet, something is probably wrong with the Ingate configuration or you lack Internet connectivity.

- 1. Make sure the Ingate is configured correctly.
- 2. Make sure that IP connectivity between the Ingate and the ITSP is working. Contact your network administrator for assistance if needed.

If you see a packet sent from the Ingate, verify that it was sent to the IP address provided by the ITSP. If not, correct your configuration and retest.

If none of the steps above solves the problem, please contact your sales representative for support.

Troubleshooting Inbound Calls

Symptom: When trying to make an inbound call to a ShoreTel phone via the SIP Trunk, there is no ringing signal on the ShoreTel phone.

Note: If you get a ringing signal on the ShoreTel phone, these troubleshooting steps will not help you to find the problem. Please contact your sales representative for support.



960 Stewart Drive Sunnyvale, CA 94085 USA Phone +1.408.331.3300 +1.877.80SHORE Fax +1.408.331.3333 www.ShoreTel.com



Inbound troubleshooting overview

Get a log for the failing call:

First try to make a call to a ShoreTel phone from a PSTN phone and notice the behavior on the ShoreTel phone as well as on the PSTN phone.

Next step is to search the log on the Ingate. Log into the Ingate box and navigate to the Display Log page. Make necessary settings on the logging page have been made according to the picture below. Especially make sure that you have the highlighted checkboxes in the correct state.



Packet Type Selection Show log from: (clear) All packets date (YYYY: fine IP Address Selection (Help) Show log until: (clear) All on this address ate (YYYY: fine All on this address MM-DD) (HH:MM:SS A src (A dst (A any)) not this tombination. A to B (B to A (B etween A&B)) not this tombination. Protocol Port Selection IP packets as selected (A to B (B to A (B etween A&B))) IP packets as selected (DTCP) All ports IP portocols (DTCP) All ports IP on this port (DTCP) All ports In ot this port (DA sec (A dst (O A any)) In ot this combination (DTCM) Select type/code: (Help) In ot (DTCM) Select type/code: (Help) In ot (DES	cket selection: only those packets that meet the search criteria in the three ctions below will be selected. This selection will only have effect on the IP cleets as selected choice.	Time Limits
All packets date (YYY)- mae All packets MM-DD) (HH-MM-SS IP Address Selection (Help) show log until (clear) ate (YYY)- time MIN-DD) (HH-MM-SS) Astc A dst A stc A dst A to B B to A Between A&B not this combination. Protocol Port Selection P packets as selected OTCP All ports OTDP © Selected ports: (Help) Manual reconfigurations and reboots A src A dst A to B O B to A Between A&B not this combination OTCP All ports OTDP © Selected ports: (Help) Nanual reconfigurations and reboots A src A dst A any A to B O B to A Between A&B not this combination OTCMP Select type/code: (Help) True changes Type: not Code not D ESP not Protocol number not Protocol number not	Packet Type Selection	Show log from: (clear)
IP Address Selection (Help) Show log until: (clear) Image: Ima	All packets	MM-DD) (HH-MM:SS)
Image:	IP Address Selection (Help)	Show log until: (clear)
MM-DD) A src A dst A to B B to A B to B CTCP All IP protocols OTCP All ports OTCP B to A on Between A&B Code In not this port A to B B to A B to A B to A dst A src A dst A to B B to A B to A dst B to A dst C A dst A to B B to A B to A B to A dst A src A dst A to B B to A B to A any In not this port A to B A to B B to A B to A <td>not this address</td> <td>date (YYYY- time</td>	not this address	date (YYYY- time
A src A dst A any Protocol Port Selection P packets as selected • All IP protocols Configuration server logins • TCP All ports • UDP • Selected ports: (Hslp) • A to B • B to A • Between A&B not this port B: not this port • A to B • B to A • Between A&B not this combination OTCP All or b • B to A • Between A&B • UDP • Selected ports: (Hslp) A: not this port B: not this port • A to B • B to A • Between A&B • ICMP Select type/code: (Hslp) Type: not Code: not • ESP • Protocol number: (Hslp) • Protocol number: (Hslp)	not this address	MM-DD) (HH:MM;SS)
A to B B to A Between A&B not this combination. Protocol Port Selection P packets as selected O All IP protocols Configuration server logins O TCP All ports Administration and configurations and configurations and configurations and configurations and reboots O TDP O Selected ports: In not this port B: not this port C A dst O A any not this combination O ICMP Select type/code: Hath Type: not Code not O ESP In sec user authentication O Protocol number: Implied in not	A src. A dst A anv	
Protocol Port Selection Image: Configuration server logns Image: Configuration server logns Image: Configuration server logns Image: Configuration server logns <td>A to B \bigcirc B to A \bigcirc Between A&B \square not this combination</td> <td>Show This</td>	A to B \bigcirc B to A \bigcirc Between A&B \square not this combination	Show This
 All IP protocols Configuration server logins Administration and configurations and config	Protocol/Port Selection	P parkets as selected
• All IP protocols • Administration and configurations andexected andexected andexected andexected and configurat	rotocorron selection	Configuration server logins
OTCP All ports Manual reconfigurations and rebots OTDP Selected ports: (Help) The changes A: not this port DHCP/PPPoE client B: not this port DHCP/PPPoE client A src A dst A any A to B OB to A OBetween A&B not this combination STUDP problems OICMP Select type/code: Inot Mail errors Type: not Mail errors Code not Psec tunnels Fsec key negotiations Fsec key negotiations PPTP negotiations STP errors	All IP protocols A	Administration and configuration
A: Inot this port Images B: not this port Images A src A dst A any A to B B to A Between A&B Inot this combination OICMP Select type/code: (Hshp) Images Type: Images Images Code Images Images OICMP Select type/code: Images Images Type: Images Images Images Images Images	OTCP O All ports OUDP ⊙ Selected ports: (Help)	Nanual reconfigurations and
B: not this port D HCP/PPPoE chent A src A dst A any not this combination R ADIUS errors A to B O B to A O Between A&B Inot this combination S MP problems OICMP Select type/code: Inot Hardware errors Type: not Mail errors Code: not Inot Isec type/code: O ESP Inot Isec type rors Isec type rors O Protocol number: Inot PTP negotiations PTP negotiations	A: not this port	Time changes
A src A dst A any Image: Constraint of this combination Image: Constraint of this combination OICMP Select type/code: Image: Constraint of this combination Image: Constraint of this combination Image: Constraint of this combination Type: Image: Constraint of this combination OiCMP Select type/code: Image: Constraint of this combination Image: Constraint of this combination Image: Constraint of this combination Type: Image: Constraint of this combination Image: Constraint of this combination Image: Constraint of this combination Code: Image: Constraint of this combination Image: Constraint of this combination Image: Constraint of this combination OiCMP Select type/code: Image: Constraint of this combination Image: Constraint of this combination OiCMP Select type/code: Image: Constraint of this constraint of this combination Image: Constraint of this combination OiCMP Select type/code: Image: Constraint of this combination Image: Constraint of this combination OiCMP Select type/code: Image: Constraint of this combination Image: Constraint of this combination OiCMP Select type/code: Image: Constraint of this com	B not this port	DHCP/PPPoE client
O A to B O B to A Between A&B not this combination SNMP problems O ICMP Select type/code: Hardware errors Type: not Mail errors Code: not It sec tunnels O ESP It sec user authentication PPTP negotiations O Protocol number: It sec user ST Protocol number:	⊙A src ⊙A dst ⊙A any	RADIUS errors
O ICMP Select type/code: (Help) Type: not Code not O ESP Isec key negotiations O Protocol number: (Help) not SP errors O Protocol number: (Help) O Protocol number: (Help)	○ A to B ○ B to A ○ Between A&B	SIMP problems
Type: not Negotiated IPsec tunnels IFsec key negotiations IFsec user authentication PPTP negotiations SP errors SP errors PTTP negotiations	OTCMP Select two/code: (Help)	Hardware errors
Code: Inot Insections © ESP If sec key negotiations O Protocol number: Inot PPTP negotiations Image: Supervise of the section of	Type:	🔲 Mail errors
© ESP © Protocol number: (Help) not © SUP errors	Code D pat	🗆 Negotiated IPsec tunnels
© ESP □ IF sec user authentication ○ Protocol number: (Help) □ not □ PPTP negotiations ☑ SIP errors		🔲 II sec key negotiations
()Protocol number: (Help) □ not □ PTP negotiations STP errors	O ESP	🗆 🏽 sec user authentication
C) Frotocol number: (Hemp)	And the second sec	PPTP negotiations
	D'Frotocol number: (Help)	SIP errors
🔄 🖉 🖓 🗠 signaing		SIP signaling
	○ ESP ○Protocol number: (Help) □ not	If Sec key negotiations If Sec user authentication PPTP negotiations SIP errors SIP signaling
V DLP DACKETS	The second s	

Then press "Display log" further down on the same page.

You will now see a log of all SIP packets received and sent by the Ingate, with the newest log entry on the top.

Ensure the signaling is received from the ITSP:

Localize the call initiation from the Trunking provider by searching for "invite sip" in your browser. (use Ctrl-F). You should look for the first packet coming from the ITSP system that starts with a "recv from <IP address of the ITSP>" as you can see in the example (only the first lines of the log are shown below).

```
>>> Info: sipfw: recv from 208.49.124.49:5060 via UDP connection 12748:
INVITE sip:6023574058;npdi=yes@193.12.253.37:5060 SIP/2.0
```

If you cannot find a packet like the one above, the problem is in the communication from the ITSP to the Ingate. Follow these steps:



- 1. Make sure you have IP connectivity between the Ingate and your ITSP. Contact your network administrator for assistance, if needed
- 2. Make sure the Ingate SIP module is turned on, SIP Services SIP Module On. Retest if you change any settings.

If you still don't see any packets in the log, contact your ITSP for further troubleshooting.

Ensure correct signaling to the ShoreTel PBX:

If you find the incoming packet, you should find a similar packet leaving the Ingate just above (just after in time) the incoming packet. The first lines of the outgoing packet will look something like this:

>>> Info: sipfw: send sf (0x8419848) to 10.100.0.40:5060 via UDP connection 12746:

INVITE sip:6023574058;npdi=yes@10.100.0.40:5060;transport=udp SIP/2.0

If you don't see the outgoing packet, something is probably wrong with the Ingate configuration or you might lack a connection to your LAN where the ShoreTel is located.

- 1. Make sure that you have IP connectivity between ShoreTel and the Ingate. Contact your network administrator for assistance, if needed.
- 2. Make sure your Ingate is configured correctly.

If you see the outgoing packet, make sure the IP address to which it was sent is the one used by the ShoreGear switch.

If the call still fails after executing the steps described above, please contact your sales representative for support.

Ingate Technical Support

North America Customers:

Contact your reseller for support.

If you don't work with an Ingate Authorized Reseller, you may purchase an Annual Support Agreement from Ingate Systems. For pricing of an Annual Support Agreement, please email <u>sales@ingate.com</u>, or phone Steve Johnson at 603-883-6569.

All support questions and issues should be directed to us_support@ingte.com

Customers outside North America:

Contact your reseller for support

If you don't work with an Ingate Authorized Reseller, you may purchase an Annual Support Agreement from Ingate Systems. For pricing of an Annual Support Agreement, please email <u>sales@ingate.com</u>, or phone +4686007750.

All support questions and issues should be directed to support@ingte.com

BandTel Configuration & Support

BandTel Special Configuration Parameters

- Your SIP User Name: Your SIP Password: Your SIP ANI:
- DNS Server Registration:
 - o Device must use our DNS server in order to resolve the below names.
 - DNS Server Addresses:
 - <u>65.175.129.149</u>
 - <u>66.237.65.90</u>
- Registration: Registration must be sent to: proxyx.bandtel.com
- Outbound Calls: Outbound calls must be sent to the following proxy addresses:
 - Primary: proxyx.bandtel.com

BandTel Support information

Technical Support Office Tel: 866-511-9400 401 Gilford Ave. Suite 220 Gilford, New Hampshire 03249 E-mail: <u>support@bandtel.com</u>

Document and Software Copyrights

Copyright © 2005 by ShoreTel, Inc., Sunnyvale, California, U.S.A. All rights reserved. Printed in the United States of America. Contents of this publication may not be reproduced or transmitted in any form or by any means, electronic or mechanical, for any purpose, without prior written authorization of ShoreTel Communications, Inc.

ShoreTel, Inc. reserves the right to make changes without notice to the specifications and materials contained herein and shall not be responsible for any damage (including consequential) caused by reliance on the materials presented, including, but not limited to typographical, arithmetic or listing errors.

Trademarks

The ShoreTel logo, ShoreTel, ShoreCare, ShoreGear, ShoreWare and ControlPoint are registered trademarks of ShoreTel, Inc. in the United States and/or other countries. ShorePhone is a trademark of ShoreTel, Inc. in the United States and/or other countries. All other copyrights and trademarks herein are the property of their respective owners.



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.

Company Information

ShoreTel, Inc. 960 Stewart Drive Sunnyvale, California 94085 USA +1.408.331.3300 +1.408.331.3333 fax