		TPP APP NOTE
ShoreTel	BandTel Global SIP Trunking Leader	TPP-10066 Date: 1/25/08
Product: ShoreTel | Ingate | BandTel		System version: ShoreTel 7.0 & 7.5

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

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

Bottom line: 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, 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 Steve@ingate.com
www.ingate.com

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

EMEA

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

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

Vendor Product Information

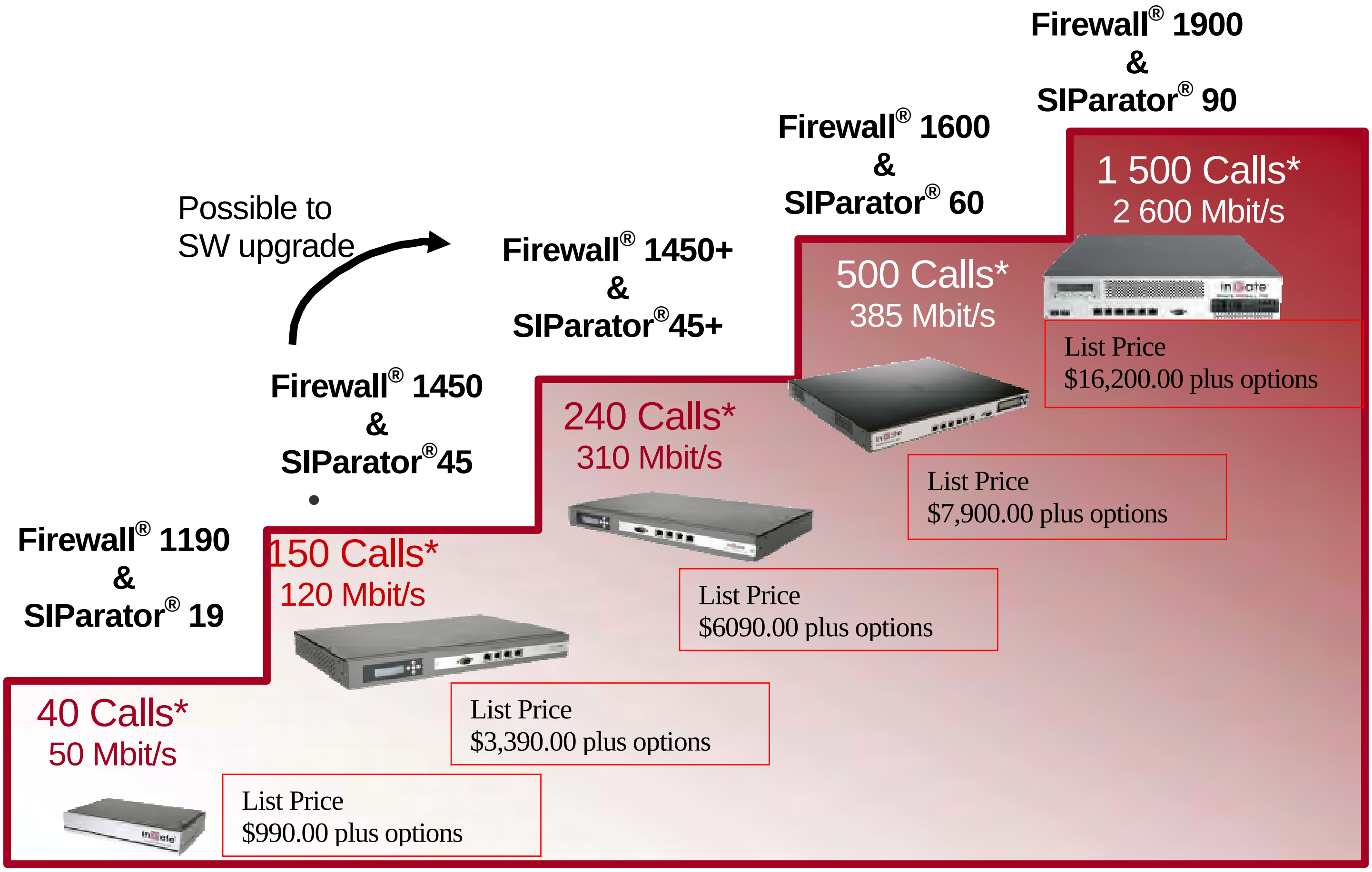


Figure 1 - Ingate Firewalls and SIParators

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.

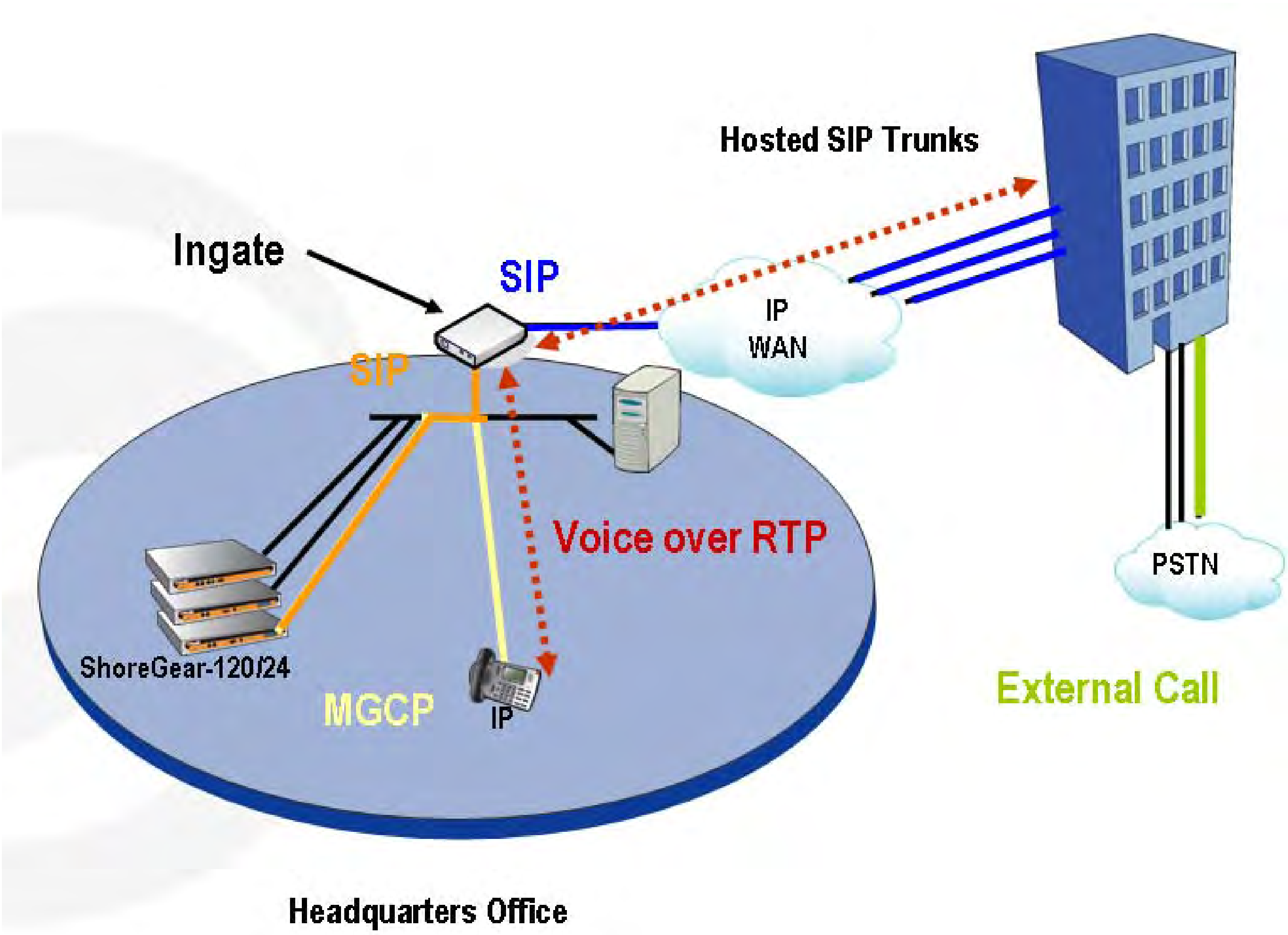


Figure 2 - Architectural Overview

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	✓	✓	✓	✓
ShoreTel 7.5	✓	✓	✓	✓

BandTel Certification 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)	PASS
1.4	Device reset – idle (for dynamic configurations)	Verify successful re-initialization of device after power loss while device is idle	PASS
1.5	Verify DiffServ Code Point support	Verify the ability to set DiffServ Code Point from SIP DUT	PASS
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
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
* must be on Ingate 4.5.2 or later			

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)	PASS
2.12	Ring selection	Verify the device's ability to change the ring type	PASS
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	PASS
2.17	Caller ID Name and Number Variations	Verify that Caller ID name and number is sent and received	PASS

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**).

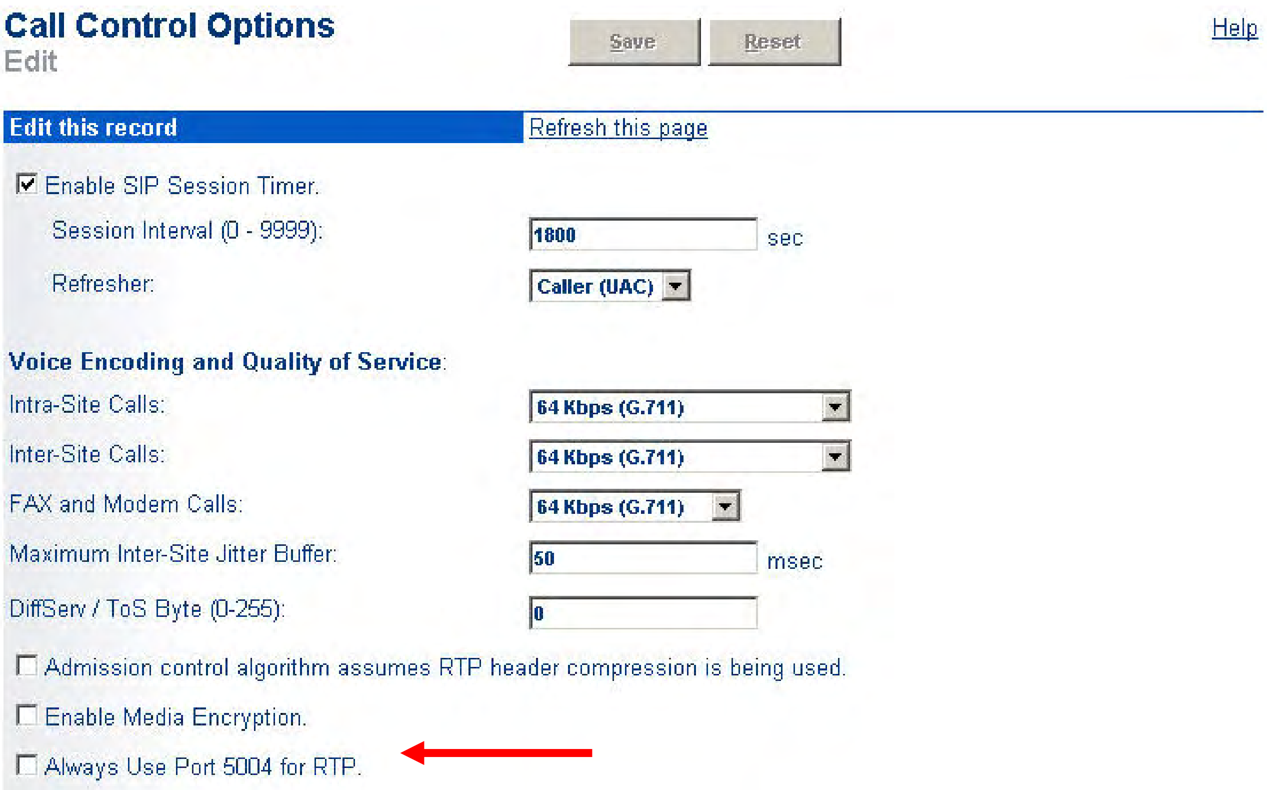


Figure 5 - Call Control Options

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 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**).

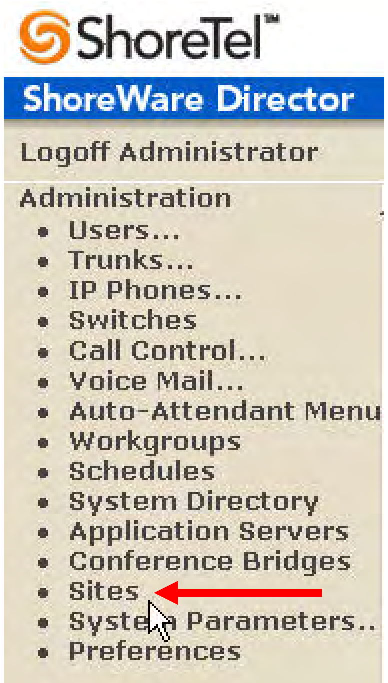


Figure 6 -Administration Site

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**).

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**).

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**).

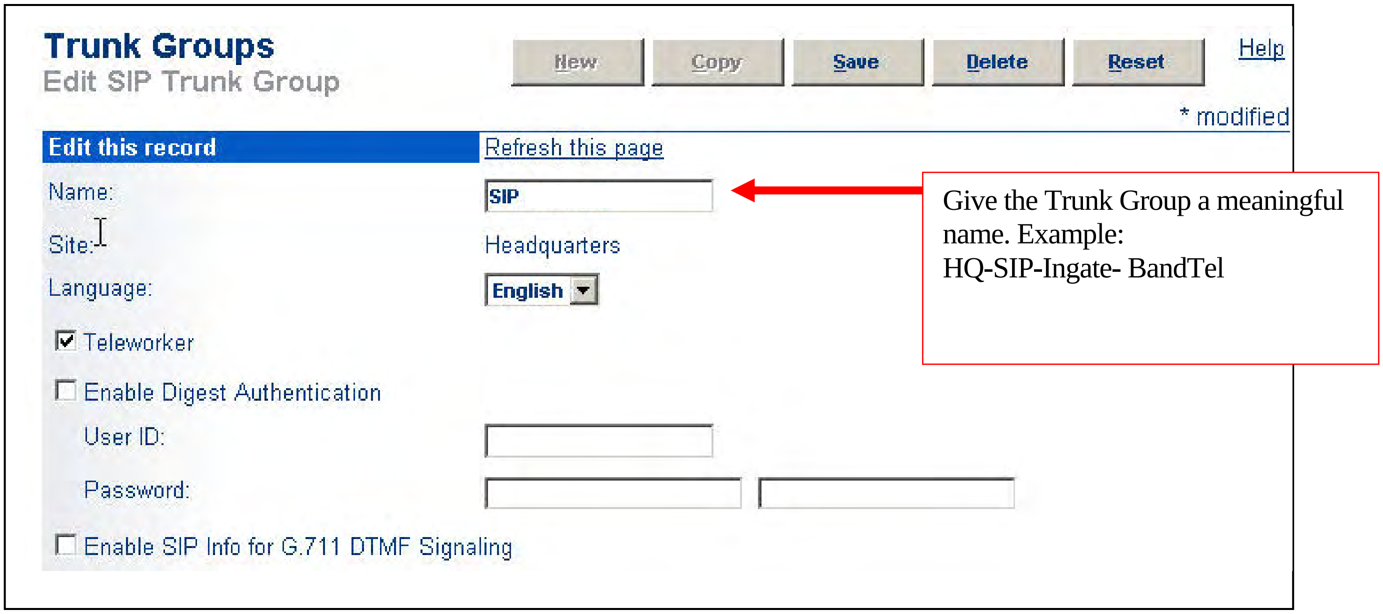


Figure 11 - SIP Trunk Group Settings

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**).

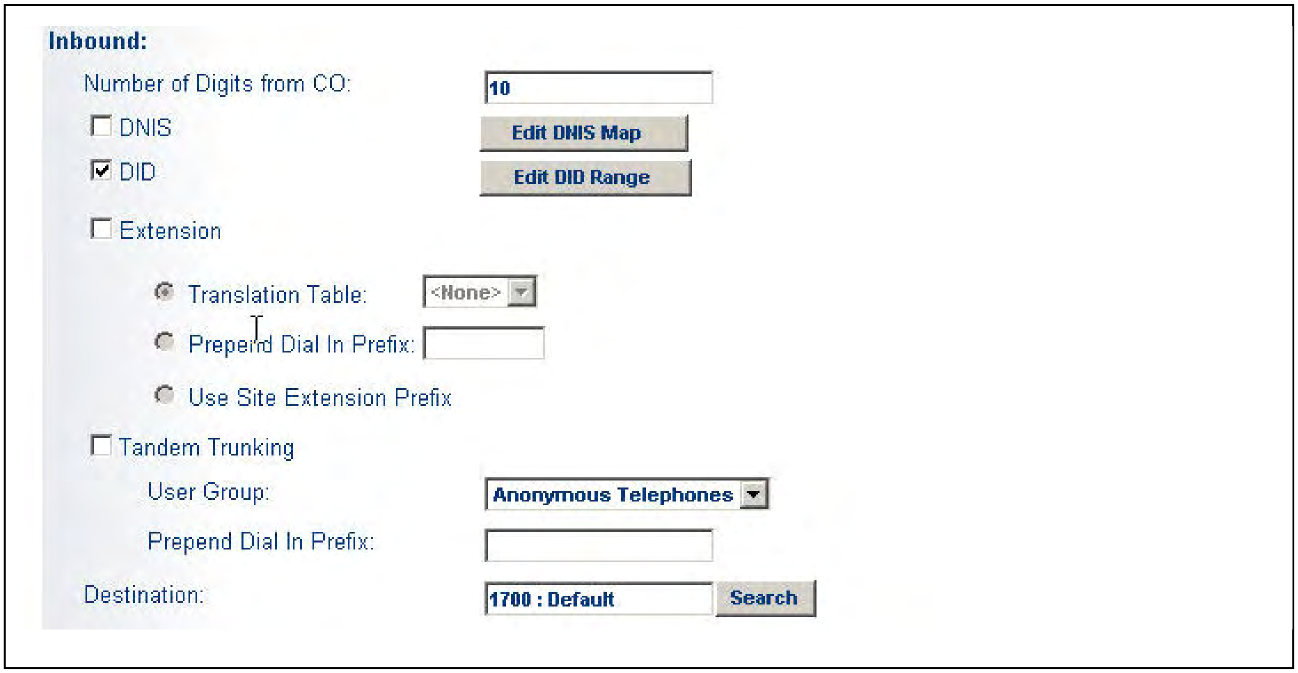


Figure 12 – Inbound:

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.

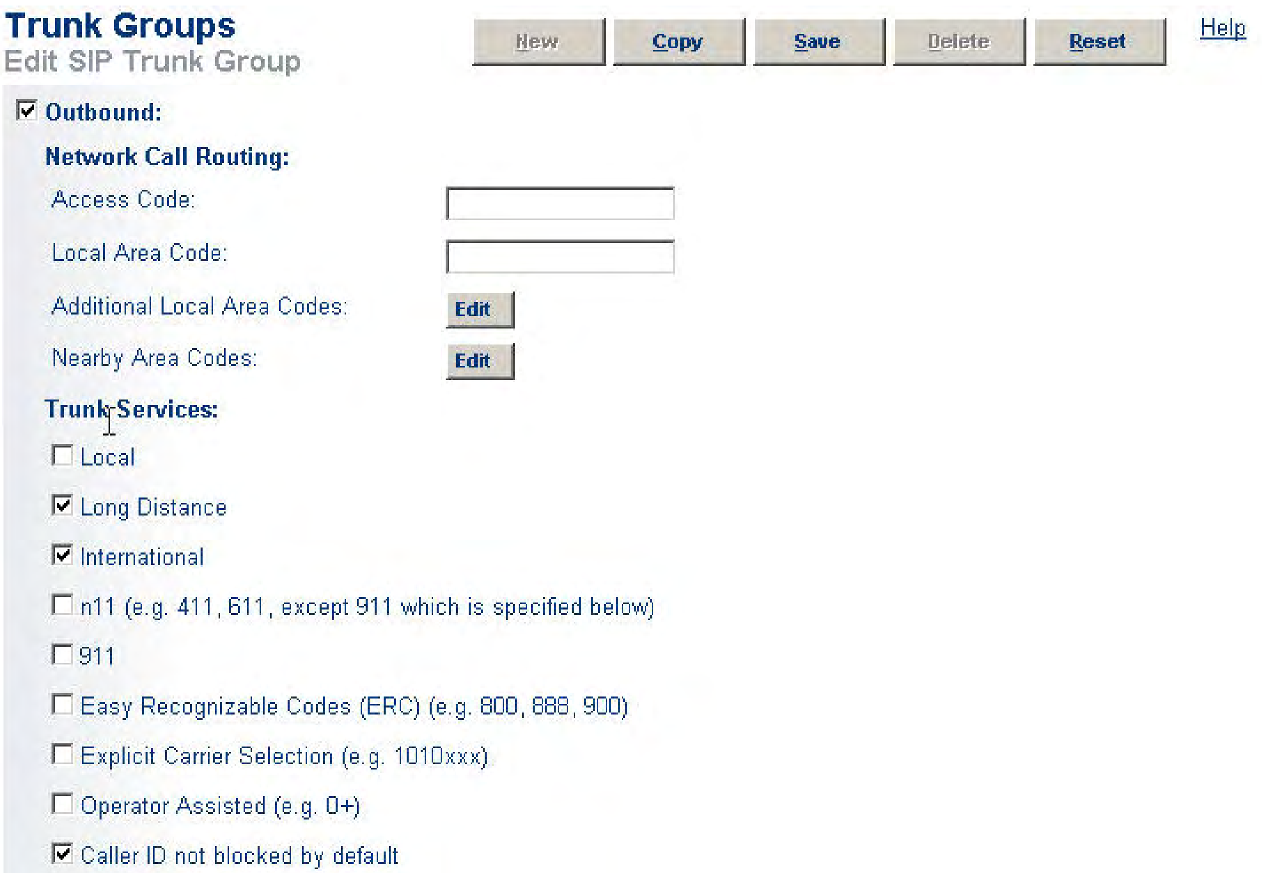


Figure 13 – Trunk Services

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**).

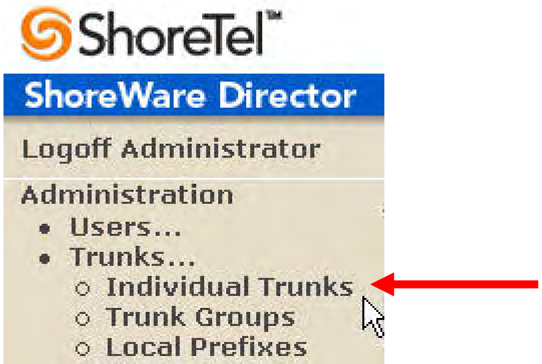


Figure 14 – Individual Trunks

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

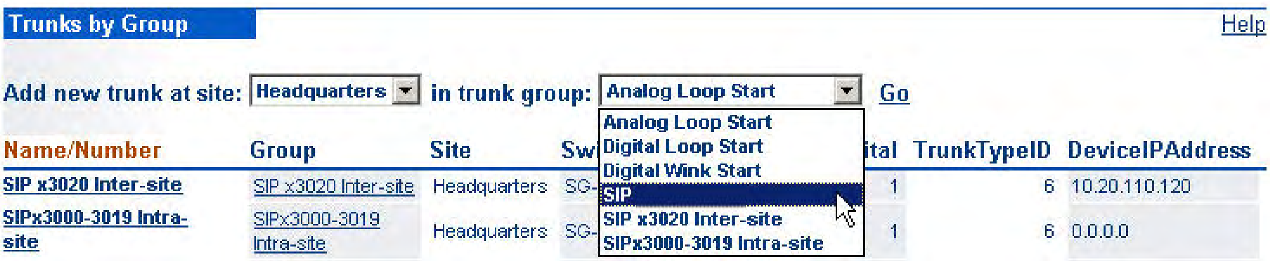


Figure 15 – Trunks by Group

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**).

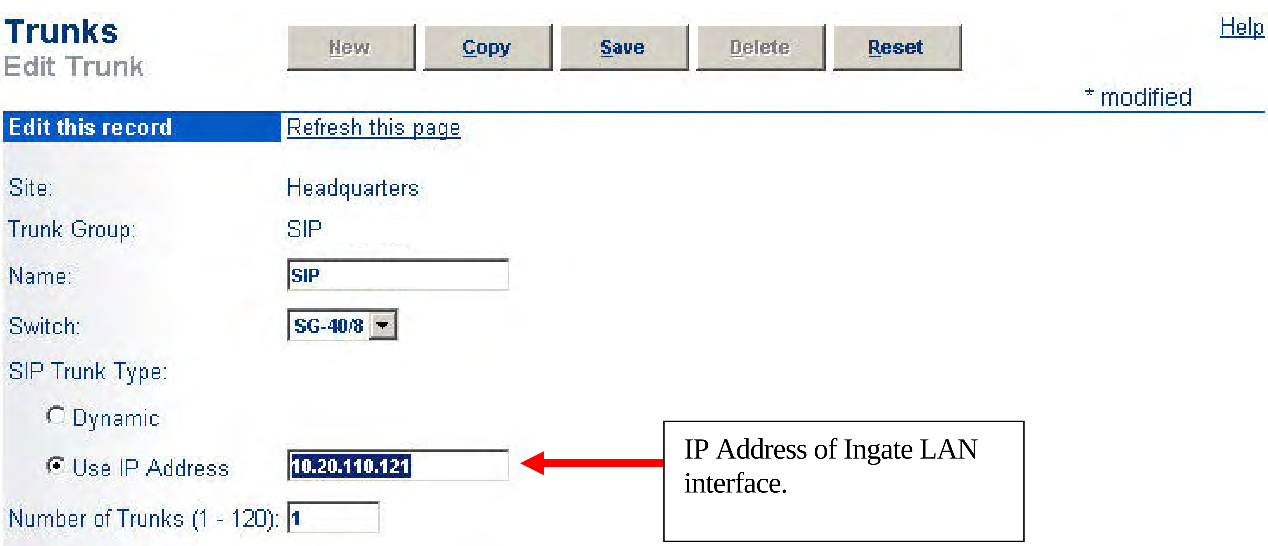


Figure 16 - Edit Trunks Screen for Individual Trunks

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

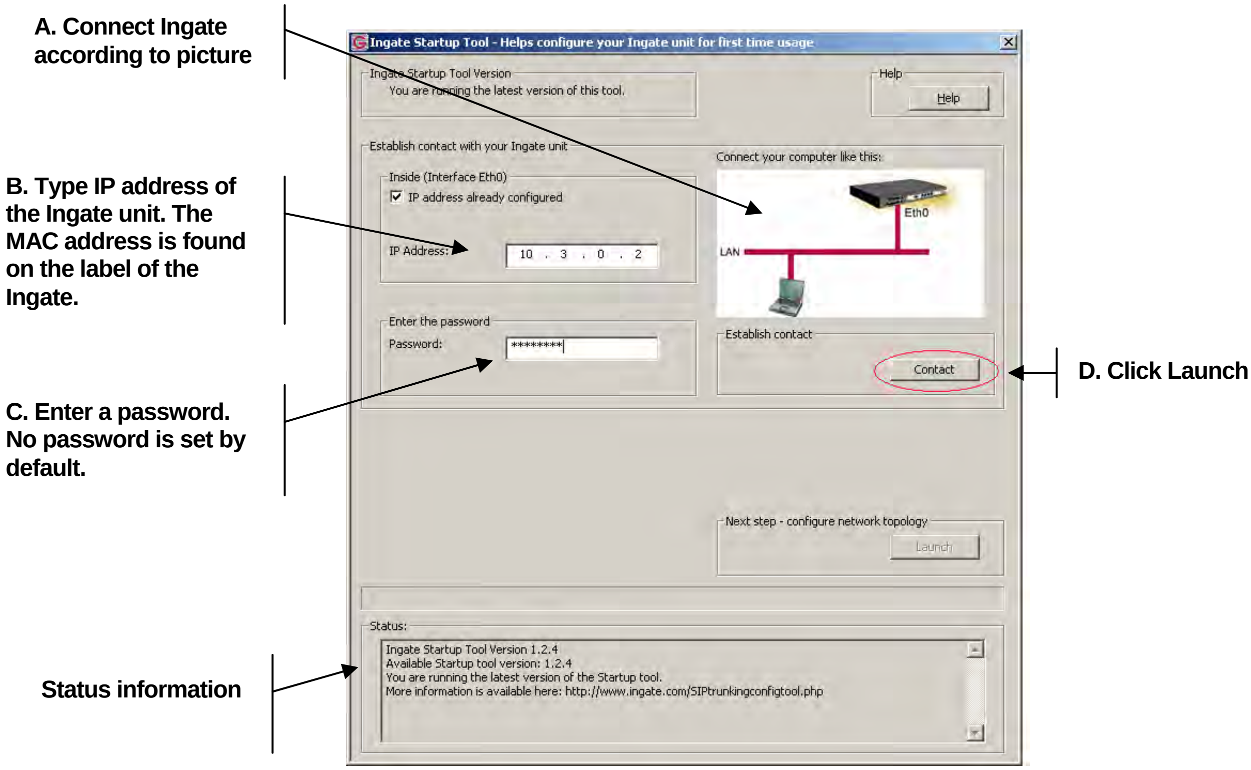
When you have received your Ingate device, unpack it and connect it to the network according to the picture below (in the screen shot). Install the Startup Tool on a Windows PC and start the tool. Make sure that your Ingate device is turned on.

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

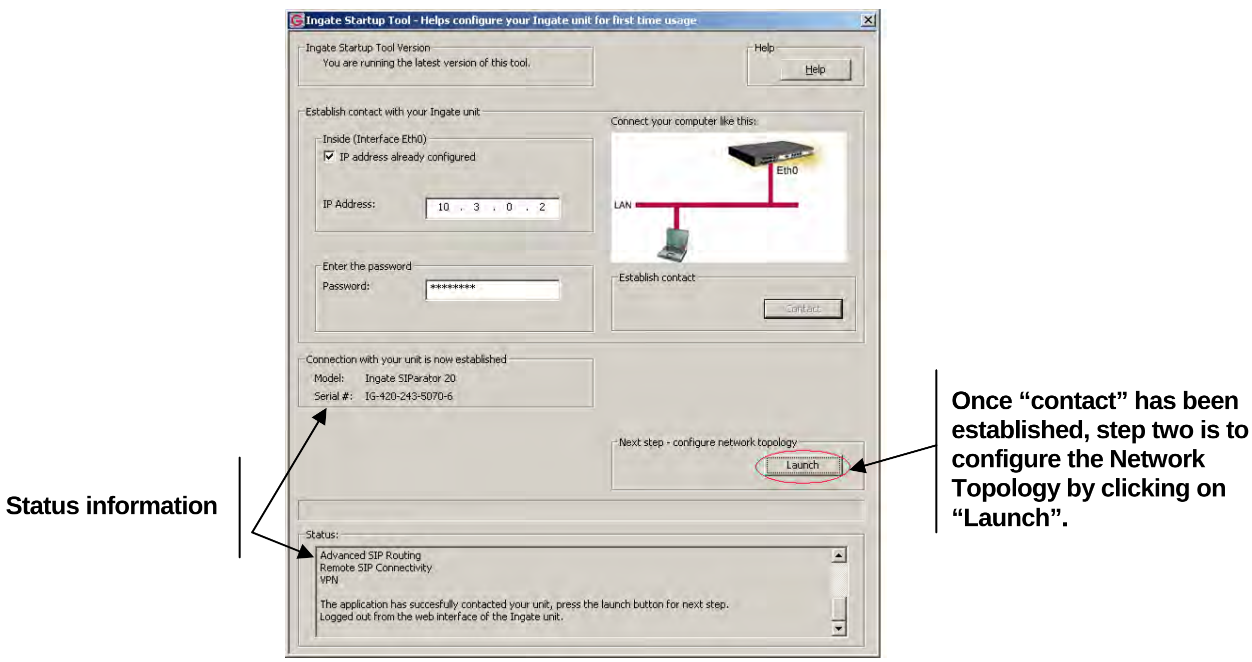
If you don't have the Trunking module installed, the tool will prompt you for license information and automatically install the Trunking module before continuing to the trunk configuration.

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*.

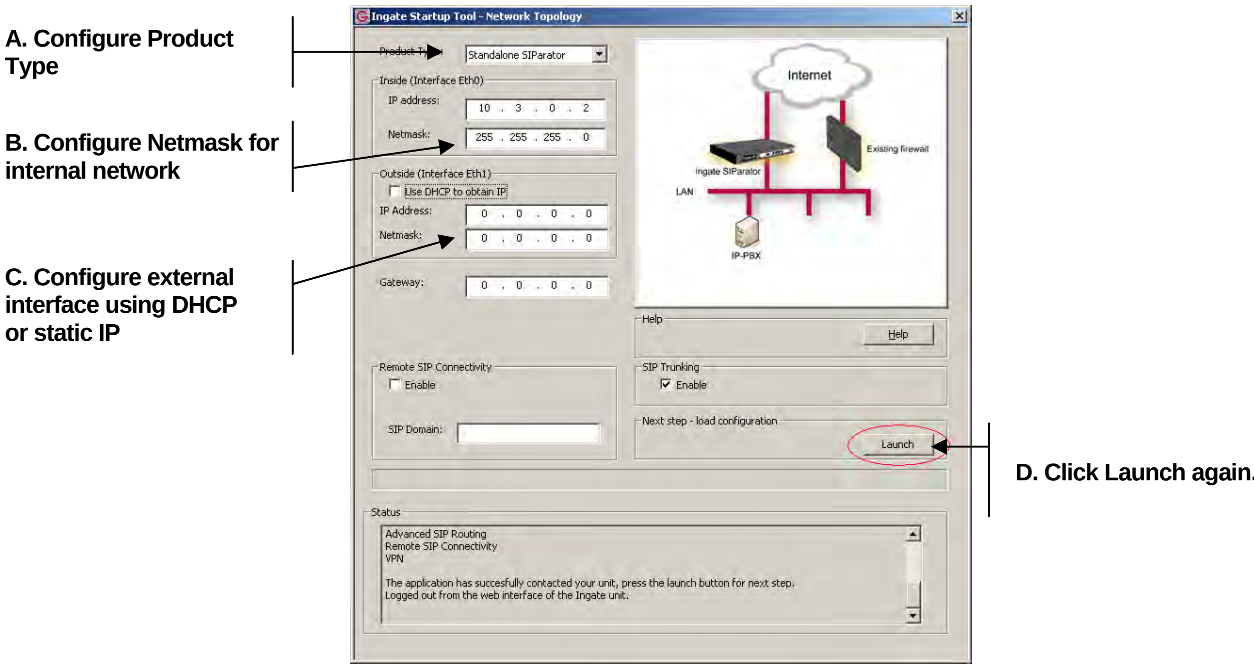


Step II – Status Information provide:



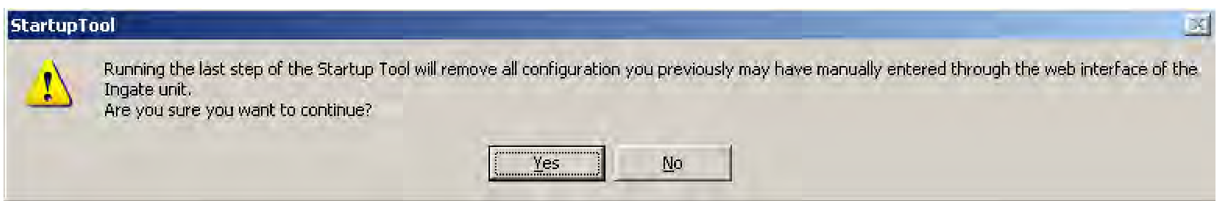
Step III – SIP Trunk configuration:

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.



Step IV –Tool Configuration:

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



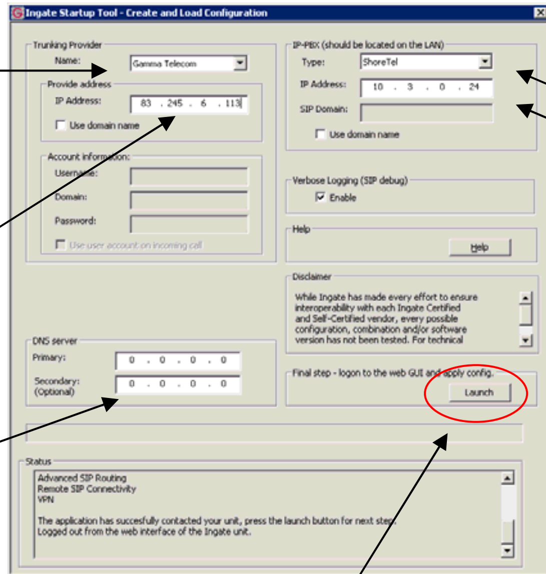
Step V –SIP Trunk Provider Configuration:

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.

A. Select BandTel from the drop-down menu and provide necessary account information.

B. Configure Provider IP Address, this will be provided by BandTel

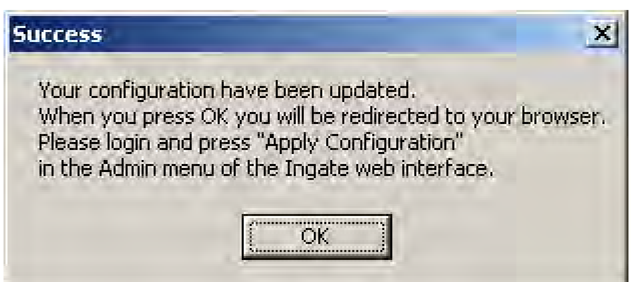
C. Configure DNS server



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

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!

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 Computers

Name	Subgroup	Lower limit		Upper limit (for IP ranges)		Interface/VLAN	Delete Row
		DNS name or IP address	IP address	DNS name or IP address	IP address		
LAN	-	10.100.0.0	10.100.0.0	10.100.0.255	10.100.0.255	inside (eth0 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 network	Action	Delete Row

Add new rows 1 rows.

Default Policy For SIP Requests

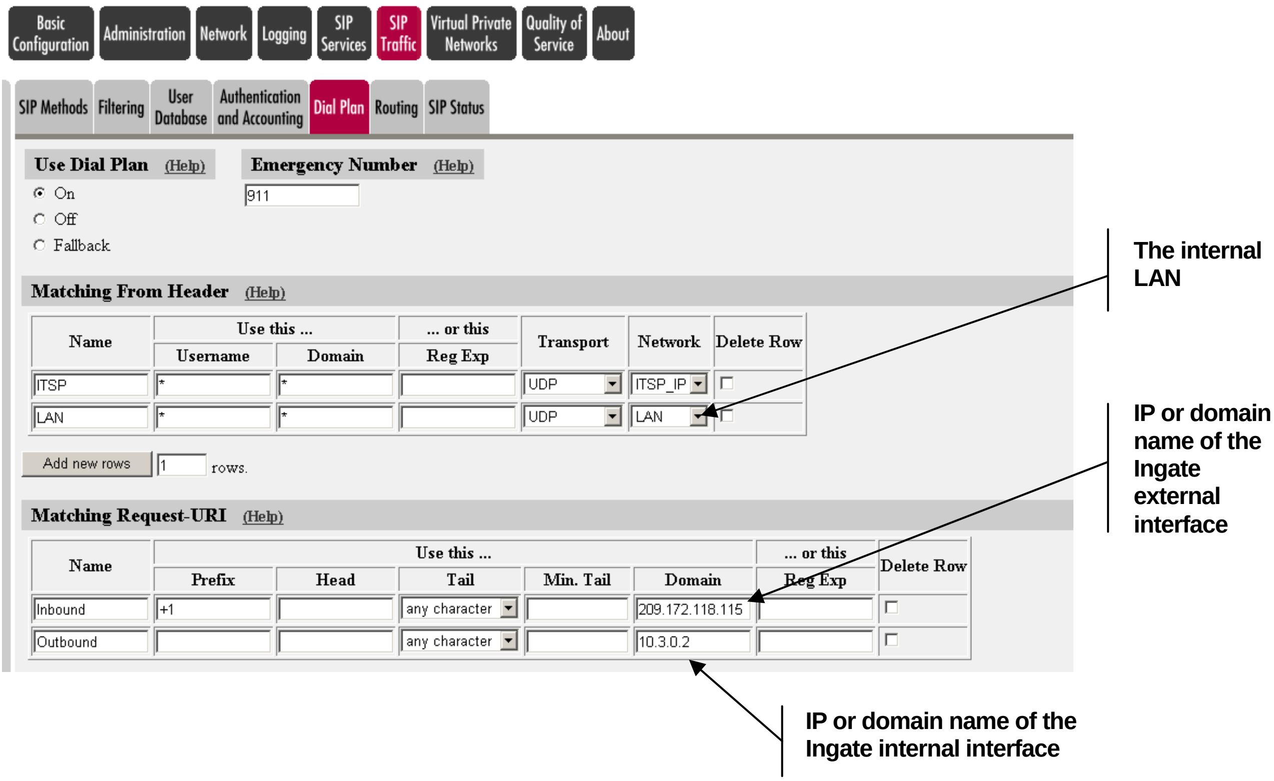
- (●) Process all
- ( ) Local only
- ( ) Reject all

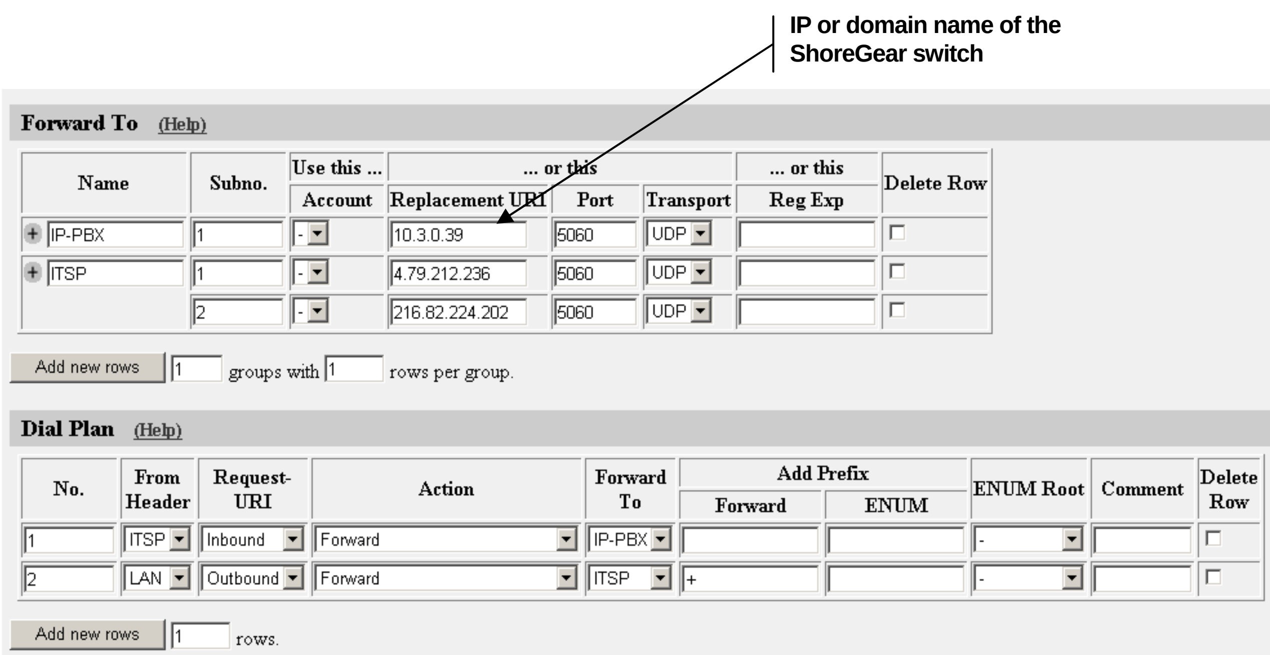
SIP TRAFFIC – USER DATABASE

Configure an account with details as provided from the ITSP.

SIP TRAFFIC – DIAL PLAN

Configure the Dial Plan according to the picture below.





SIP TRAFFIC – ROUTING

- Local REFER handling: check Always handle REFER locally.

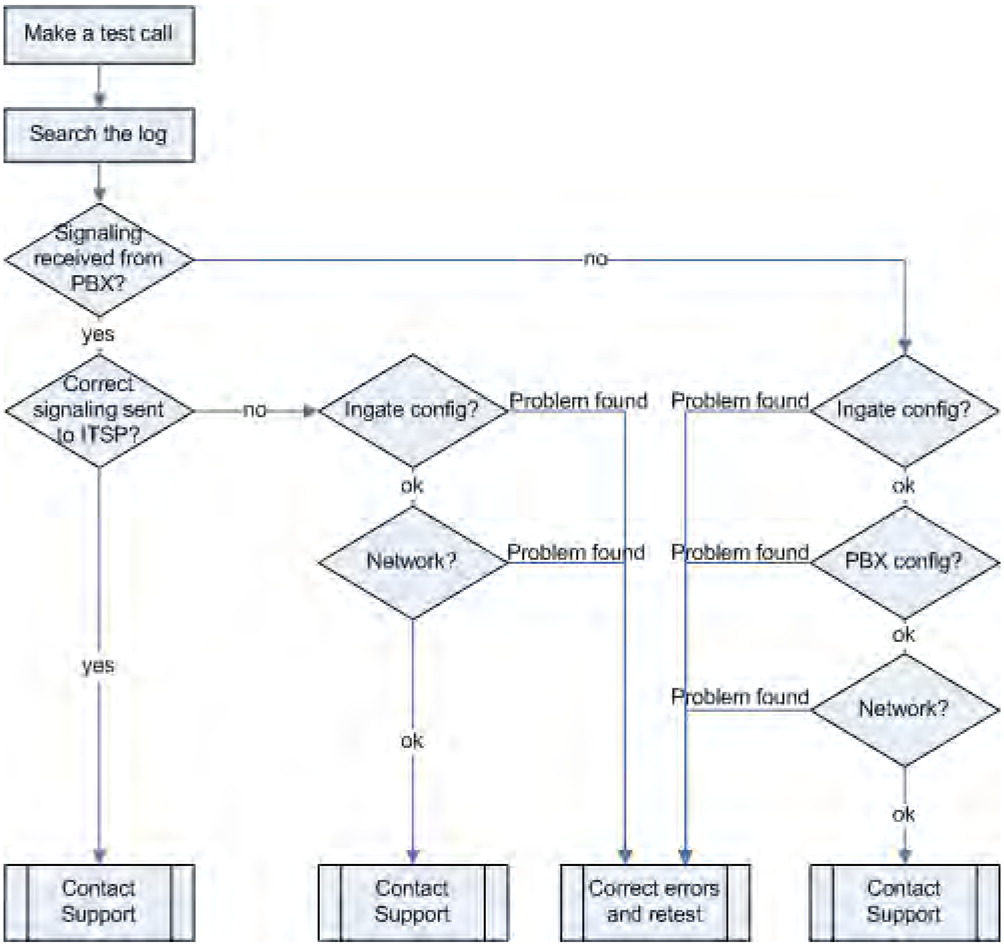
The configuration of the Ingate is now done and the changes must be applied on the Administration page to take effect.

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.

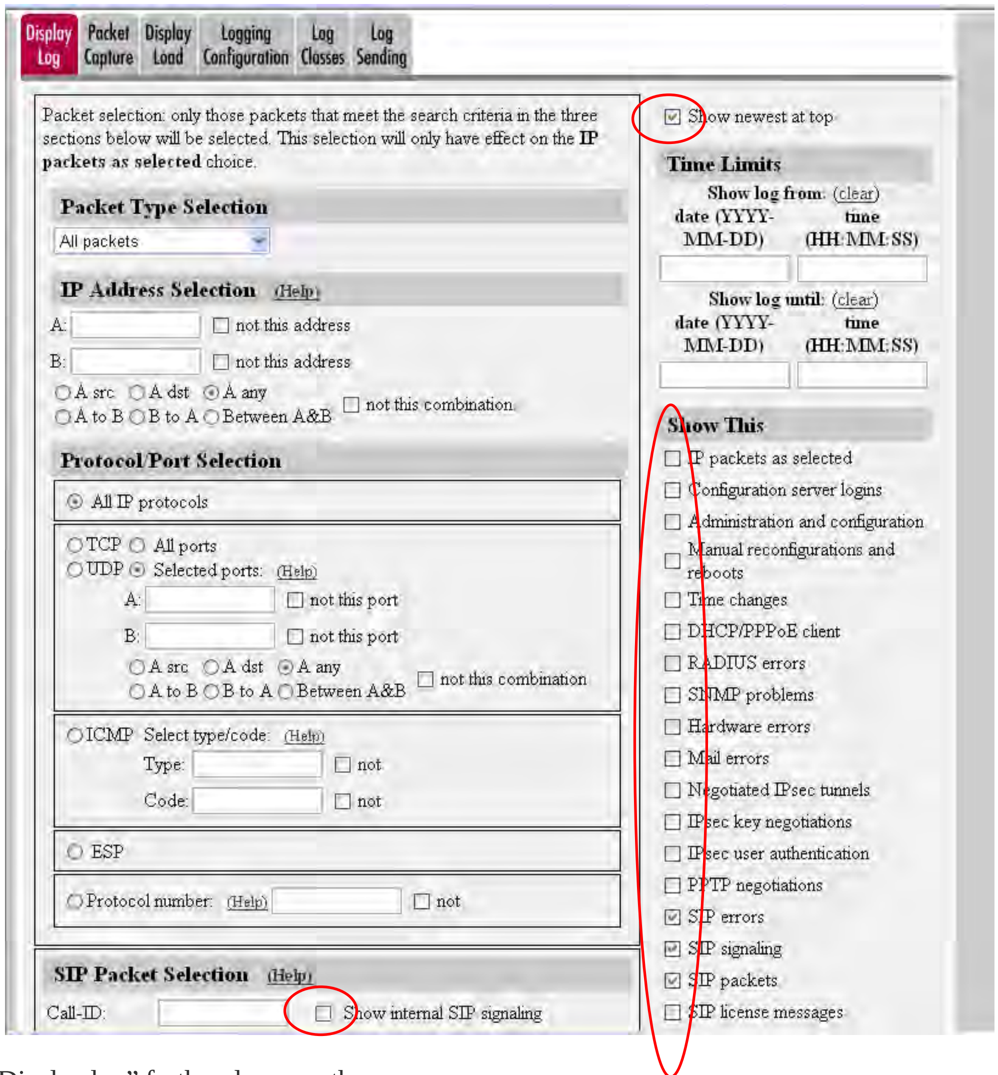


Outbound traffic troubleshooting overview

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.



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 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.

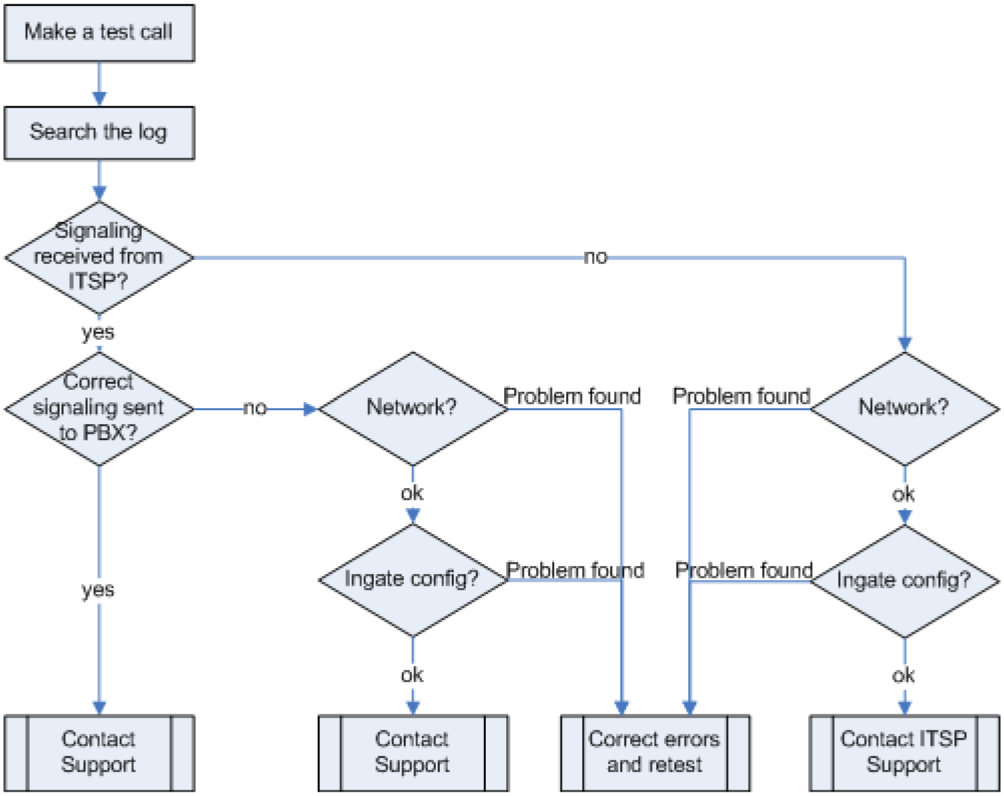
If you see a packet sent from the Ingate, verify that it was sent to the IP address provided by the ITSP. If not, correct your configuration and retest.

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.

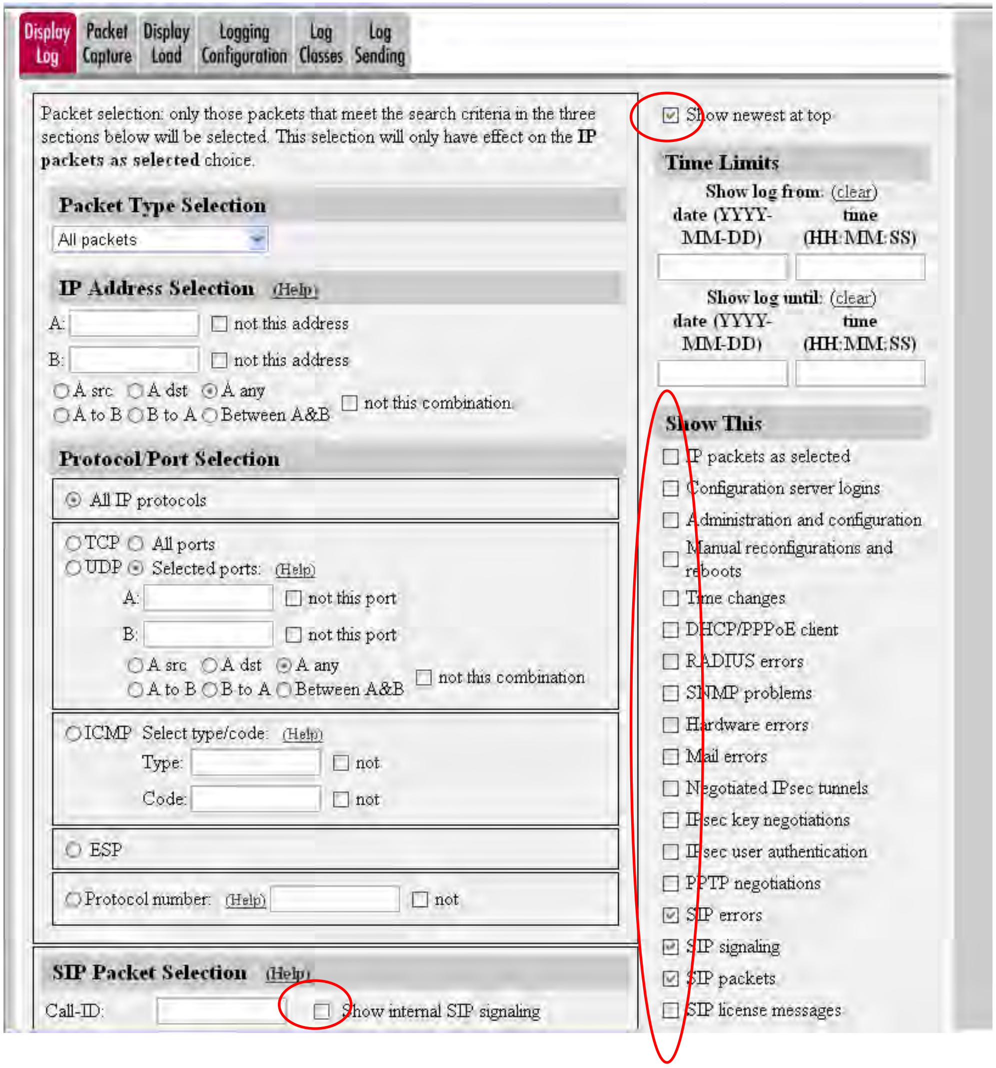


Inbound troubleshooting overview

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.



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 sales@ingate.com, 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 sales@ingate.com, 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:**
  - Device must use our DNS server in order to resolve the below names.
    - DNS Server Addresses:
      - 65.175.129.149
      - 66.237.65.90
- **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

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Gilford, New Hampshire 03249
E-mail: support@bandtel.com

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Trademarks

Disclaimer

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

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