ShoreTel		TPP APP NOTE TPP-10102 Date: September 2008
Product: ShoreTel | Ingate | AireSpring		System version: ShoreTel 8.0

ShoreTel, Ingate & AireSpring for SIP Trunking

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

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Overview

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

AireSpring Overview and Contact

AireSpring SIP Trunking delivers on the voice over IP (VoIP)/SIP promise of powerful features, simplicity of use, low operating costs, fast turn up and efficient use of existing hardware. Our SIP trunking product enables full scalability when used with ShoreTel IP telephony solutions, and addresses the evolving needs of growing companies from the SMB to the Enterprise business. With AireSpring SIP Trunking, everything between the customer and the AireSpring network is pure SIP, and this gives us the ability to least-cost-route public switched telephone network (PSTN) interconnection for every call, globally, passing on the savings to the customer.

North America

For general sales questions, please contact AireSpring directly at:

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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
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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

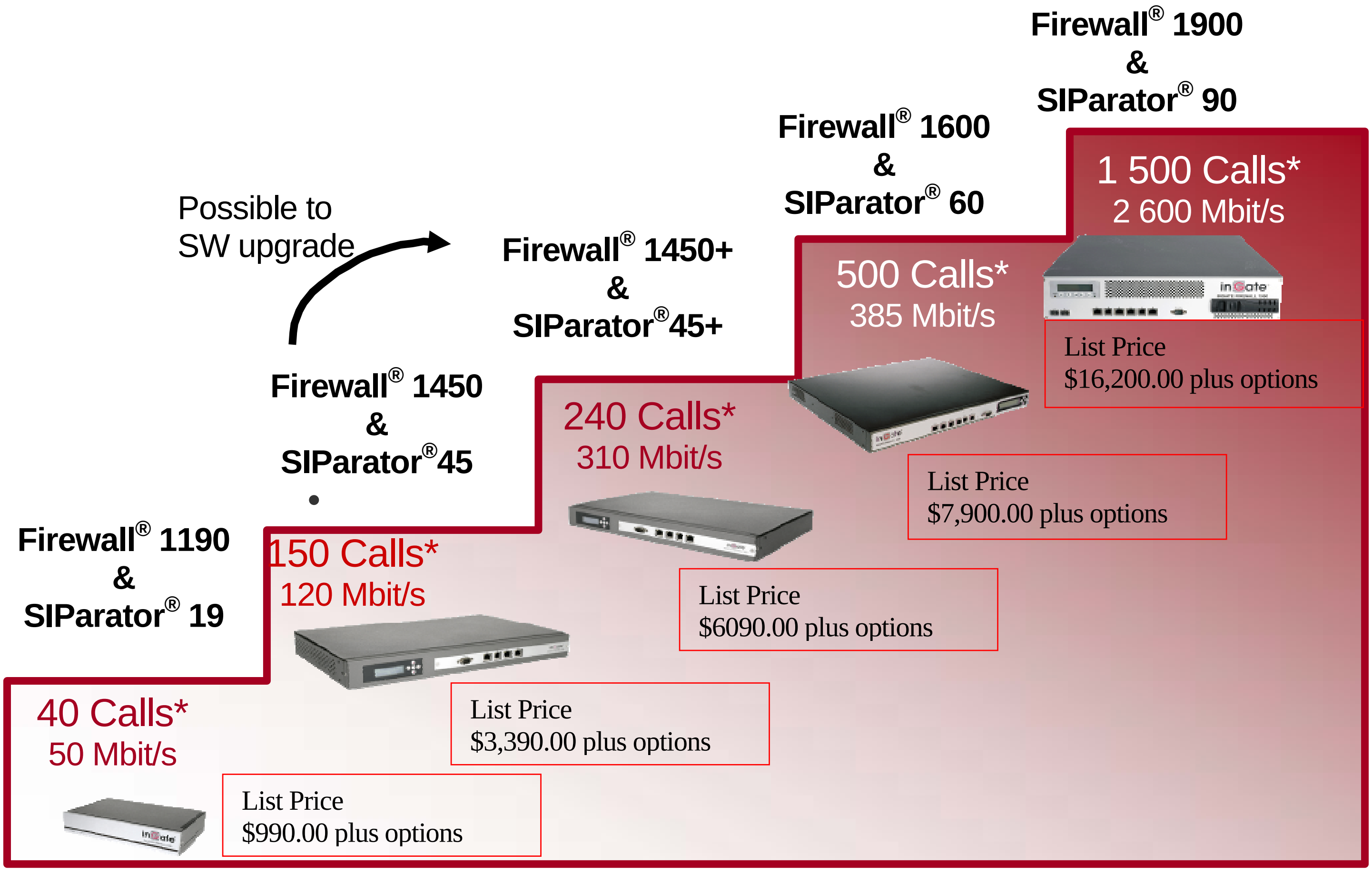


Figure 1 - Ingate Firewalls and SIParators

Architecture Overview

Providing a good solution to allow customers the ability to connect to SIP trunks offered by different ITSPs is important. ShoreTel, Ingate and AireSpring have teamed up to build a solid solution, ShoreTel being the IP PBX which sits on the LAN and connects to the SIP trunks provided by AireSpring via the Ingate SIParator/firewall. The Ingate is then connected to the LAN and also the WAN, providing not only typical firewall capabilities but also intelligent SIP routing and such SIP features as:

- 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 each of the certified ITSPs

The image below shows a high level drawing of a basic ShoreTel/Ingate /ITSP 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.

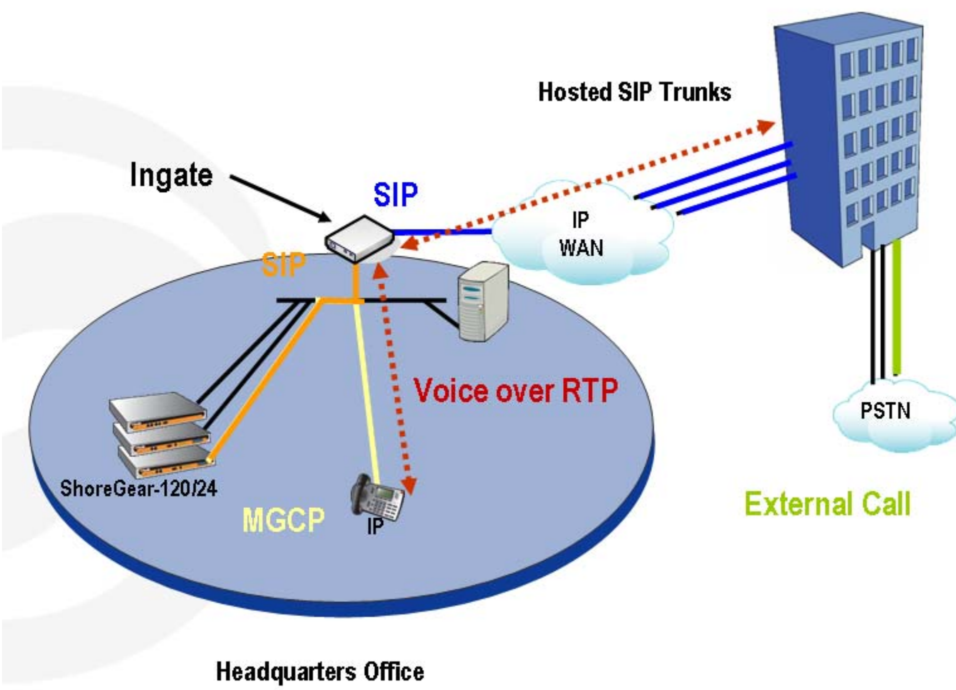


Figure 2 - Architectural Overview

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/demilitarized zone (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	4.6.2
ShoreTel 7.0	✓	✓	✓	✓	✓
ShoreTel 7.5	✓	✓	✓	✓	✓
ShoreTel 8.0	✓	✓	✓	✓	✓

AireSpring Certification Testing Results Summary

Table 1: Initialization and Basic Calls

ID	Name	Description	Notes
1.1	Setup and initialization	Verify successful setup and initialization of the system under test (SUT)	Pass
1.2	Outbound Call (Domestic)	Verify calls outbound placed through the SUT reach the external destination.	Pass
1.3	Inbound Call (Domestic)	Verify calls received by the SUT are routed to the default trunk group destination.	Pass
1.4	Device Restart – Power Loss	Verify that the SUT recovers after power loss to the SUT	Pass
1.5	Device Restart – Network Loss	Verify the SUT recovers after loss of network link to the SUT.	Pass
1.6	All Trunks Busy – Inbound Callers	Verify an inbound caller hears busy tone when all channels/trunks are in use	Pass
1.7	All Trunks Busy – Outbound Callers	Verify an outbound caller hears busy tone when all channels/trunks are in use	Pass
1.8	Incomplete Inbound Calls	Verify proper call progress tones are provided and proper call teardown for incomplete inbound calls.	Pass

Table 2: Media and Dual-Tone Multi-Frequency (DTMF) Support

ID	Name	Description	Notes
2.1	**Error! Reference source not found.**	**Error! Reference source not found.**	Pass
2.2	Media Support – SIP Reference to SUT	Verify call connection and audio path from SIP Reference phones to an external destination through the service provider using all supported codes with both sides set to a common codec.	Pass
2.3	Codec Negotiation	Verify codec negotiation between the SUT and the calling device with each side configured for a different codec.	Pass
2.4	DTMF Transmission – Out of Band / In Band	Verify transmission of in-band and out-of-band digits per RFC 2833 for various devices connected to the SUT.	Pass
2.5	Auto Attendant Menu	Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the desired extension.	Pass
2.6	Auto Attendant Menu “Dial by Name”	Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the desired extension using the “Dial by Name” feature.	Pass
2.7	Auto Attendant Menu Checking Voice Mail Mailbox	Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the Voice Mail Login Extension.	Pass

Table 3: Performance & Quality of Service

ID	Name	Description	Notes
3.1	Voice Quality Service Levels	Verify the SUT can provide a voice quality service-level agreement (SLA) across the WAN from the customer premises to the SUT SIP gateway.	Pass
3.2	Capacity Test	Verify the service provider interface can sustain services through period of heavy outbound and inbound load.	Pass
3.3	Post Dial Delay	Verify that post dial delay is within acceptable limits.	Pass
3.4	Billing Accuracy	Verify that all test calls made are accurately reflected in the SUT's Call Detail Record (CDR) and billing reports.	Pass

Table 4: Enhanced Services and Features

ID	Name	Description	Notes
4.1	Caller ID Name and Number - Inbound	Verify that Caller ID name and number is received from SIP endpoint device	Pass
4.2	Caller ID Name and Number - Outbound	Verify that Caller ID name and number is sent from SIP endpoint device	Pass
4.3	Hold from SUT to SIP Reference	Verify successful hold and resume of connected call	Pass
4.4	Call Forward - SUT	Verify outbound calls that are being forwarded by the SUT are redirected and connected to the appropriate destination.	Pass
4.5	Call Transfer – Blind	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination.	Pass
4.6	Call Transfer – Consultative	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination.	Pass
4.7	Conference – Ad Hoc	Verify successful ad hoc conference of three parties	Pass
4.8	Inbound Direct Inward Dialing/Dialed Number Identification Service (DID/DNIS)	Verify the SUT provides inbound "dialed number information" and is correctly routed to the configured destination.	Pass
4.9	Outbound 911	Verify that outbound calls to 911 are routed to the correct Public Safety Answering Point (PSAP) for the calling location and that caller ID information is delivered.	Not tested
4.10	Operator Assisted	Verify that 0+ calls are routed to an operator for calling assistance.	Not tested
4.11	Inbound / Outbound call with Blocked Caller ID	Verify that calls with Blocked Caller ID route properly and the answering phone does not display any Caller ID information.	Pass
4.12	Inbound call to a Hunt Group	Verify that calls route to the proper hunt group and are answered by an available hunt group member with audio in both directions using G.729 and G.711 codecs.	Pass

ID	Name	Description	Notes
4.13	Inbound Call to a Workgroup	Verify that calls route to the proper workgroup and are answered successfully by an available workgroup agent with audio in both directions using G.729 and G.711 codecs.	Pass
4.14	Inbound Call to DNIS/DID and Leave a Voice Mail Message	Verify that inbound calls to a user, via DID/DNIS, routes to the proper user mailbox and a message can be left with proper audio.	Pass
4.15	Call Forward – "FindMe"	Verify that inbound calls are forwarded to a user's "FindMe" destination.	Pass
4.16	Call Forward Always	Verify that inbound calls are immediately automatically forwarded to a user's external destination.	Pass
4.17	Inbound / Outbound Fax Calls	Verify that inbound/outbound fax calls complete successfully.	Not tested
4.18	ShoreTel Converged Conferencing Server	Verify that inbound calls are properly forwarded to the ShoreTel Converged Conferencing Server, that it properly accepts the access code, and you're able to participate in the conference bridge.	Pass
4.19	Inbound Call to Bridged Call Appearance (BCA) Extension	Verify that inbound calls properly presented to all of the phones that have BCA configured and that the call can be answered, placed on-hold and then transferred.	Pass
4.20	Inbound Call to a Group Pickup Extension	Verify that inbound calls are properly presented to all of the phones that have Group Pickup configured and that the call can be answered, placed on-hold and then transferred.	Pass

Table 5: Security

ID	Name	Description	Notes
5.1	Digest Authentication	Verify the SUT supports the use of digest authentication for service access for inbound and outbound calls.	N/A

Configuration Overview

The configuration information below shows examples for configuring the ShoreTel, Ingate and AireSpring. 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 AireSpring 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 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**).

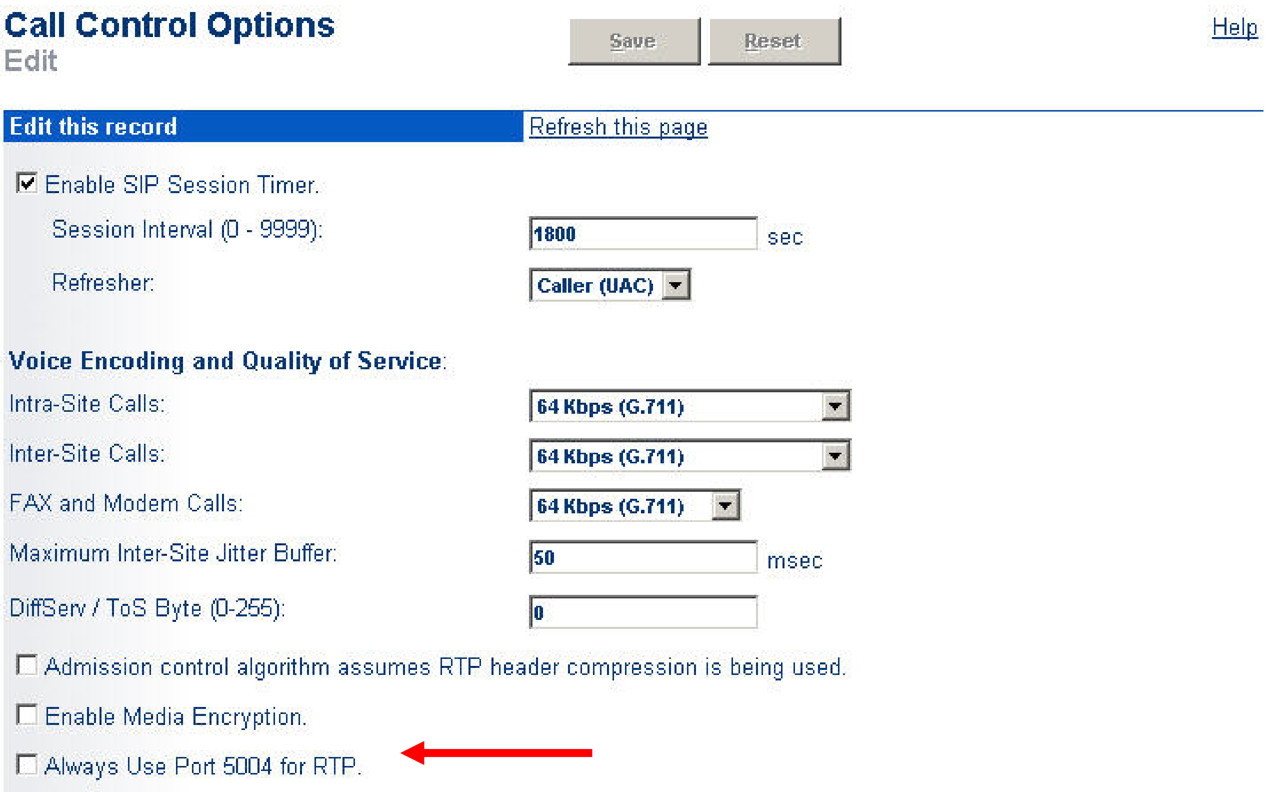


Figure 5 – Call Control Options

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 AireSpring 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**).

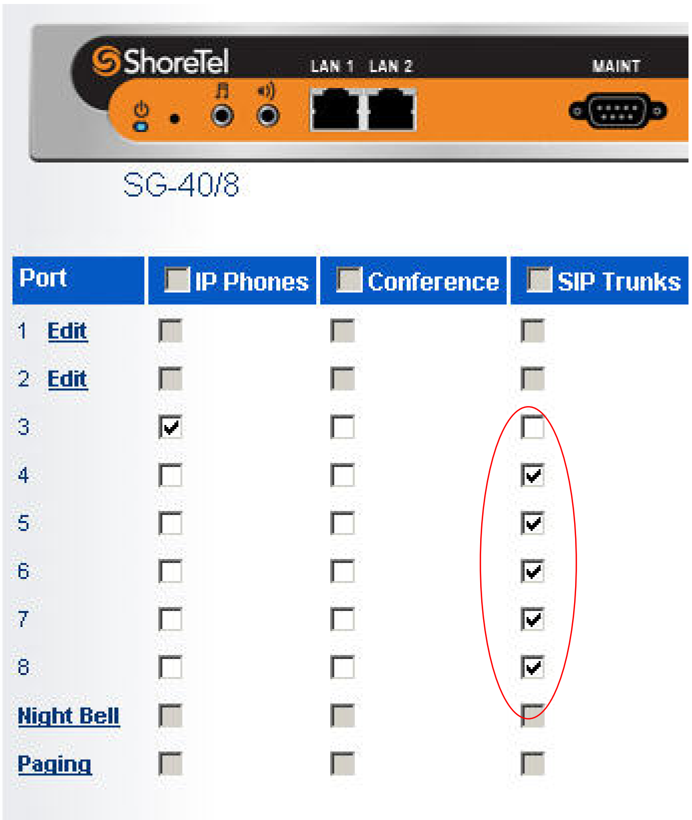


Figure 8 - ShoreGear Switch Settings

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.

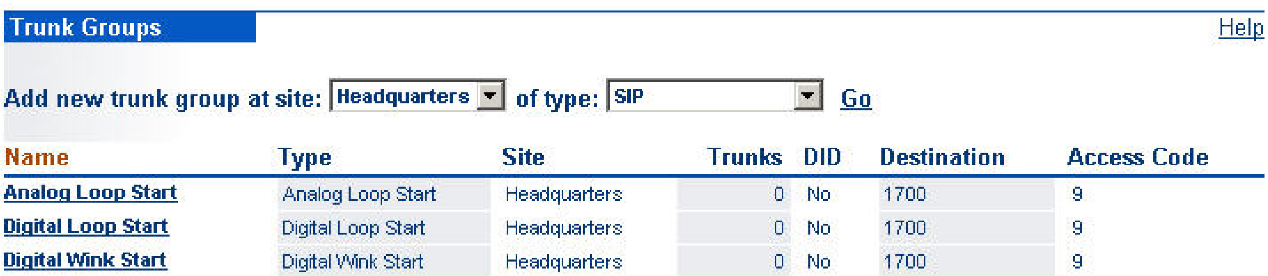
In trunk planning, the following need to be considered:

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**).



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

Figure 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**).

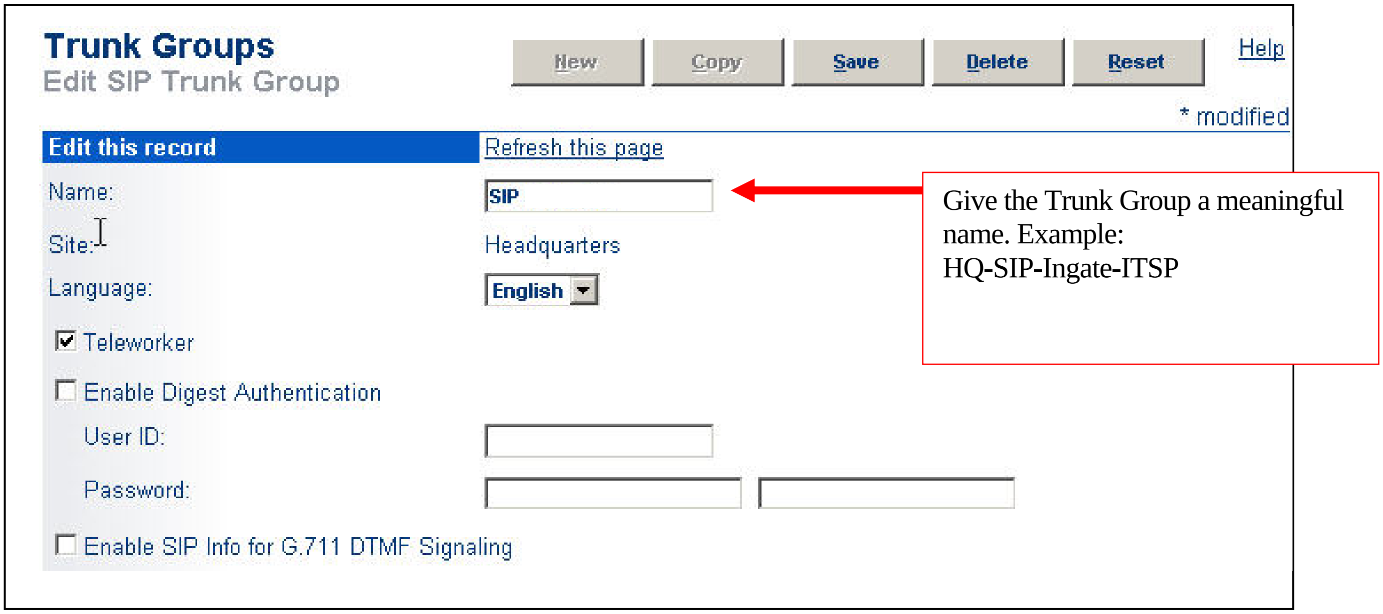


Figure 11 – SIP Trunk Group Settings

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**).

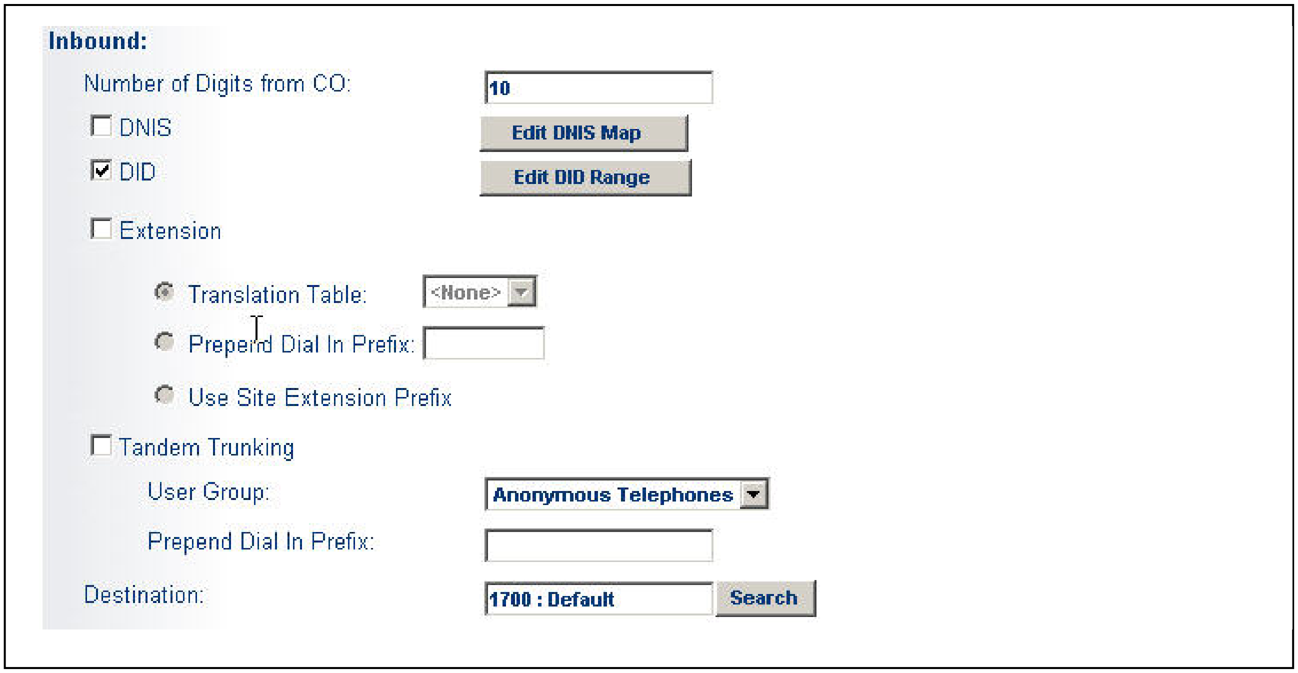


Figure 12 – Inbound:

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

Tandem Trunking is 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.

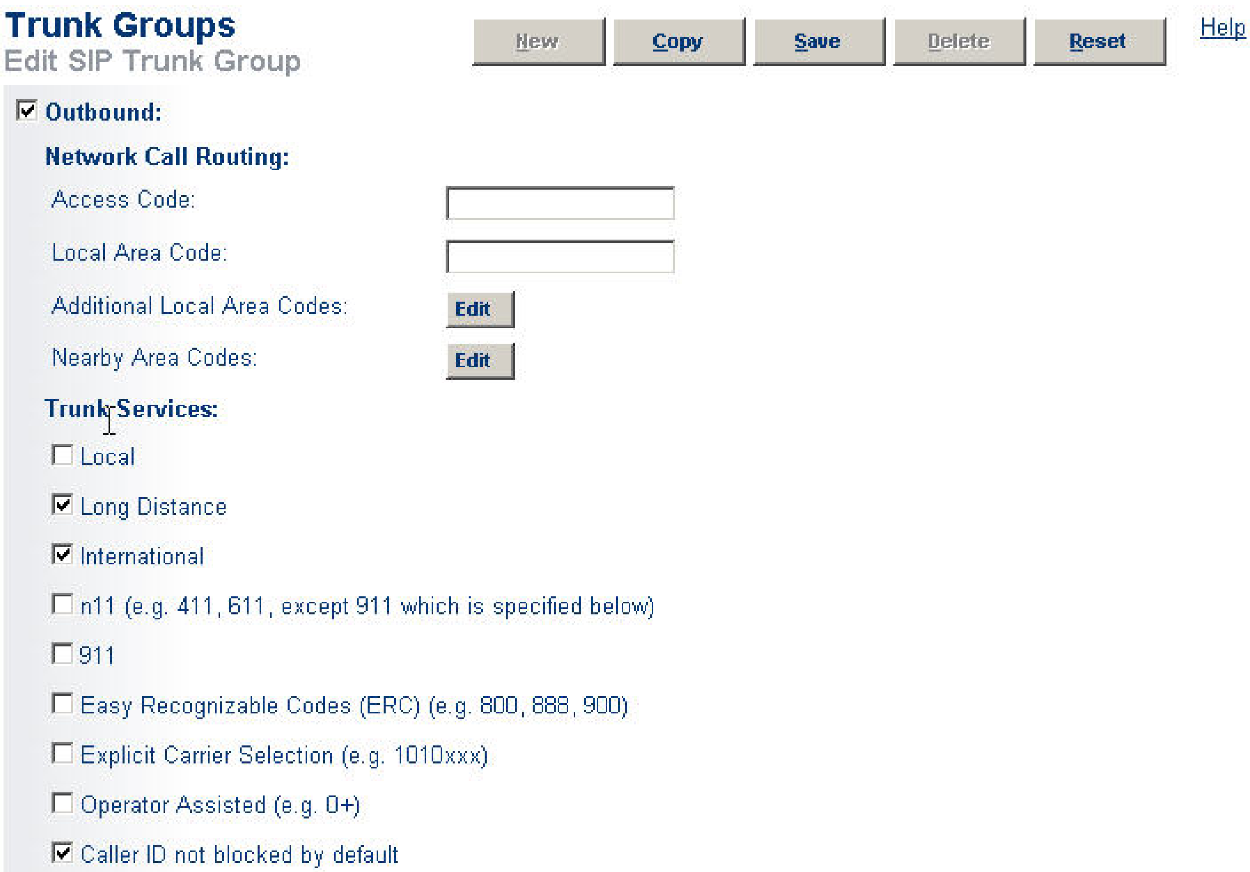


Figure 13 – Trunk Services

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

l. Local

2. LD

3. INT

4. TFN/8XX

5. 411

6. 911

- Note: Caller information (Caller ID) must be 11 digits

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**).

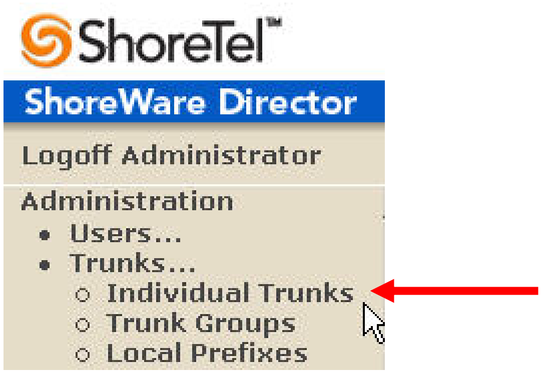


Figure 14 – Individual Trunks

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**).

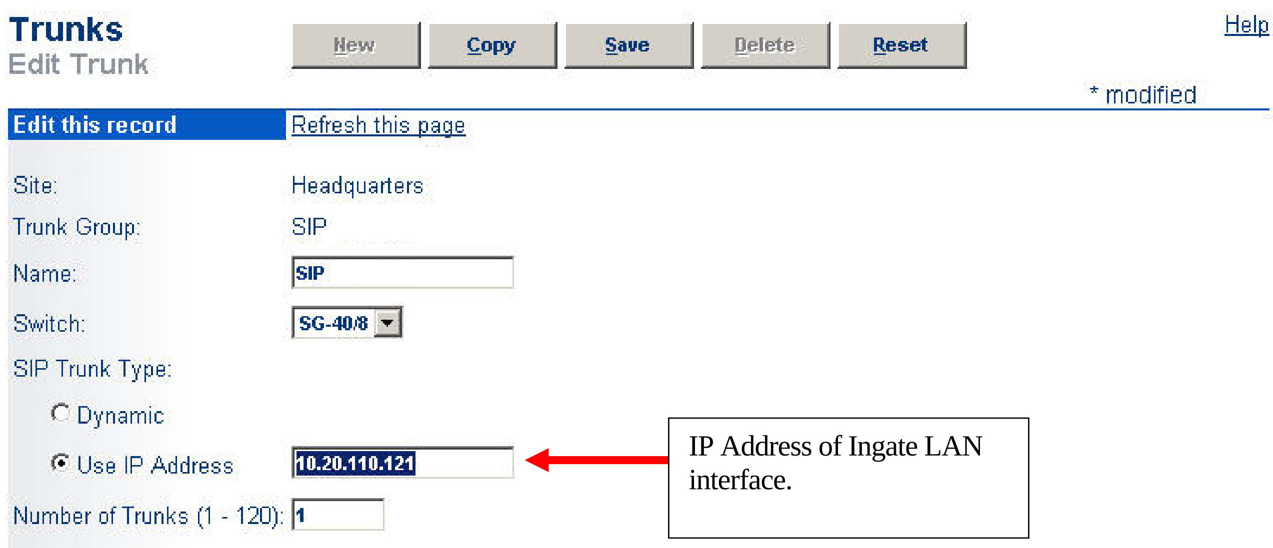


Figure 16 - Edit Trunks Screen for Individual Trunks

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 graphical user interface (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

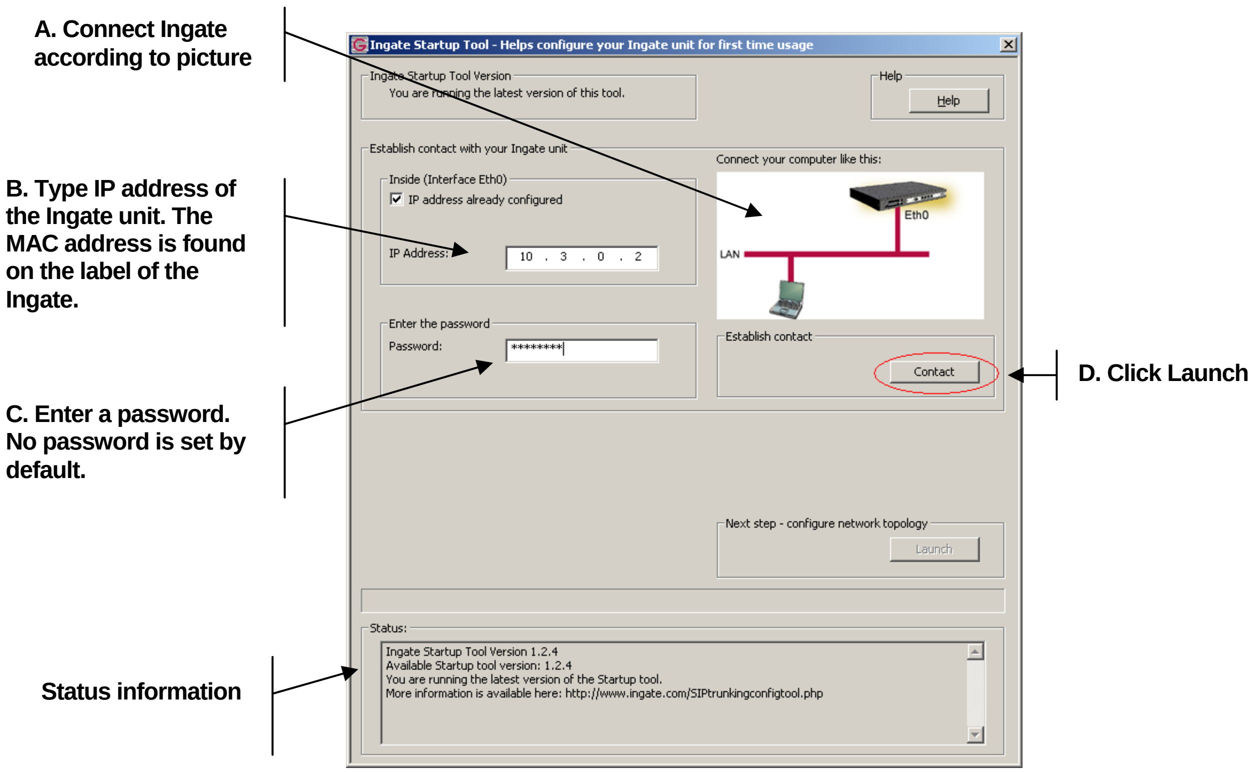
When you have received your Ingate device, unpack it and connect it to the network according to the picture below (in the screen shot). Install the Startup Tool on a Windows PC and start the tool. Make sure that your Ingate device is turned on.

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

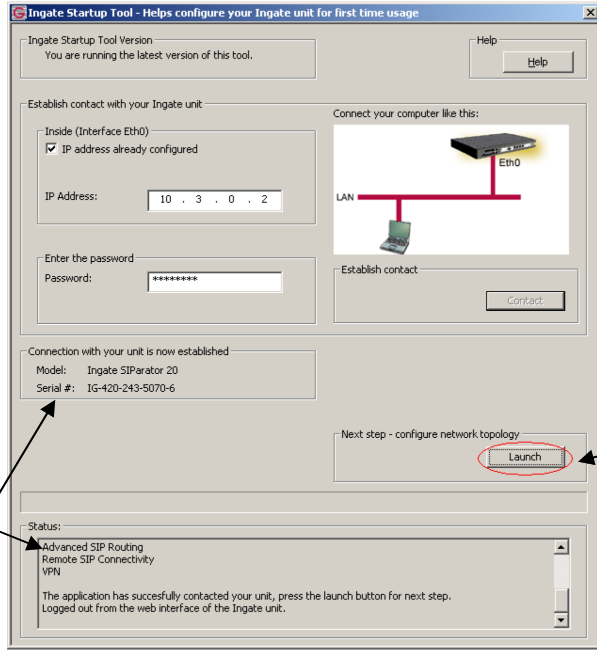
If you don't have the Trunking module installed, the tool will prompt you for license information and automatically install the Trunking module before continuing to the trunk configuration.

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:

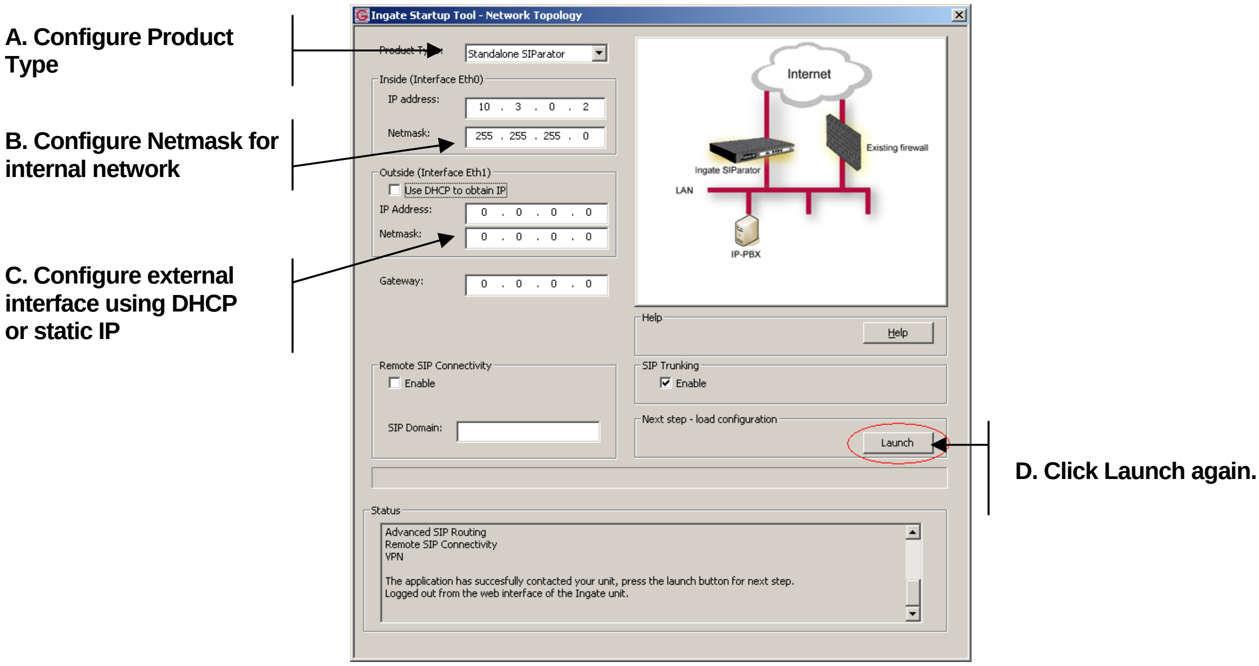


Status information

Once "contact" has been established, step two is to configure the Network Topology by clicking on "Launch."

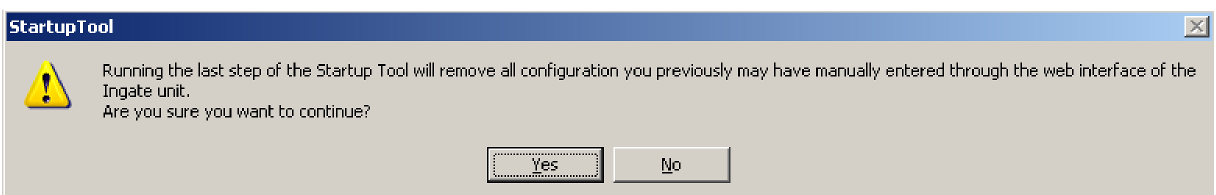
Step III – SIP Trunk configuration:

In Step III, the Product Type and network 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.



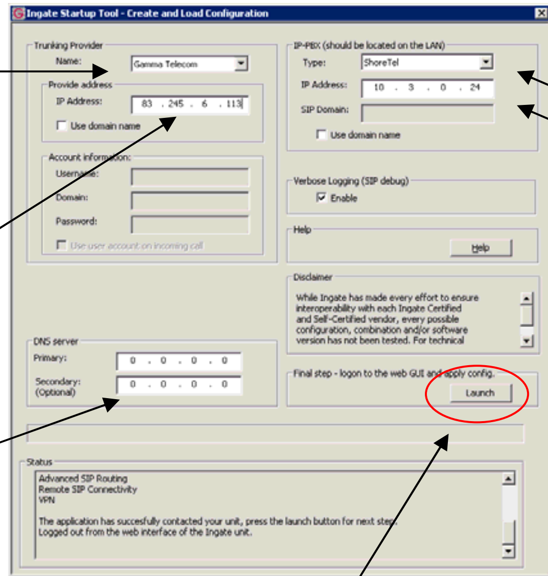
Step V –SIP Trunk Provider Configuration:

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

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

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

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.

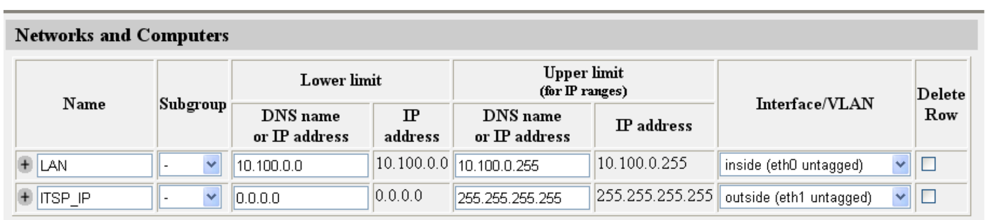
Remember to configure the following:

- 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)

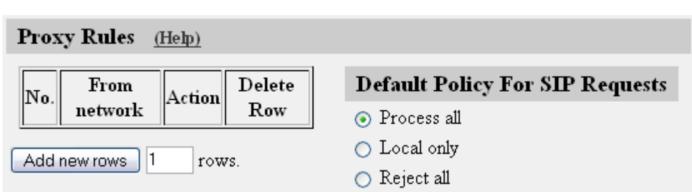
Use the appropriate SIParator configuration for your deployment.

SIP SERVICE – BASIC

- SIP Module: On.

SIP Traffic – Filtering

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

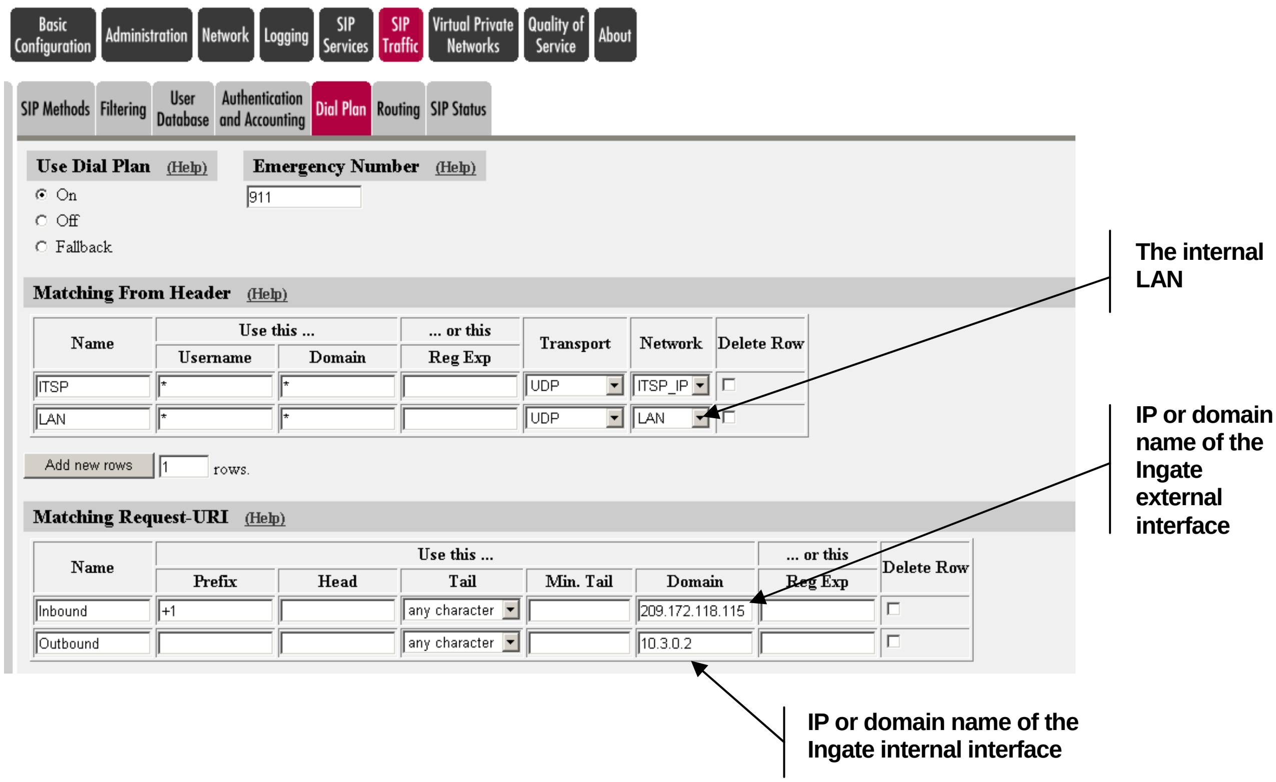


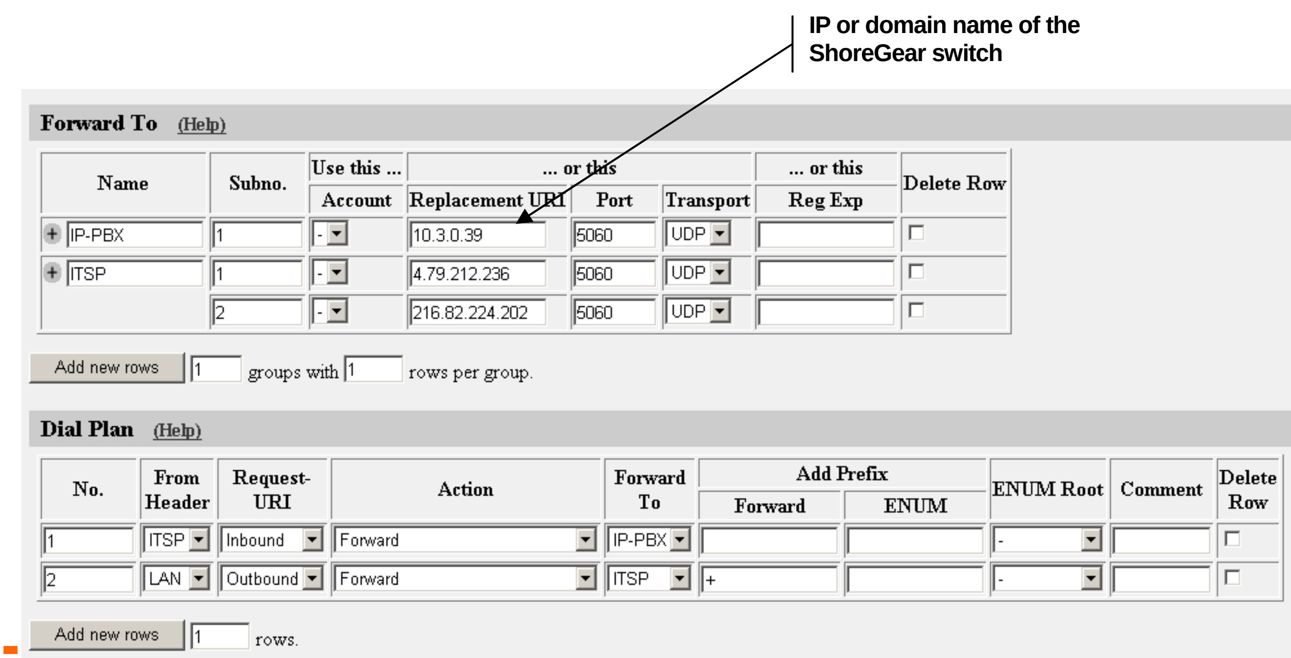
SIP TRAFFIC – USER DATABASE

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.

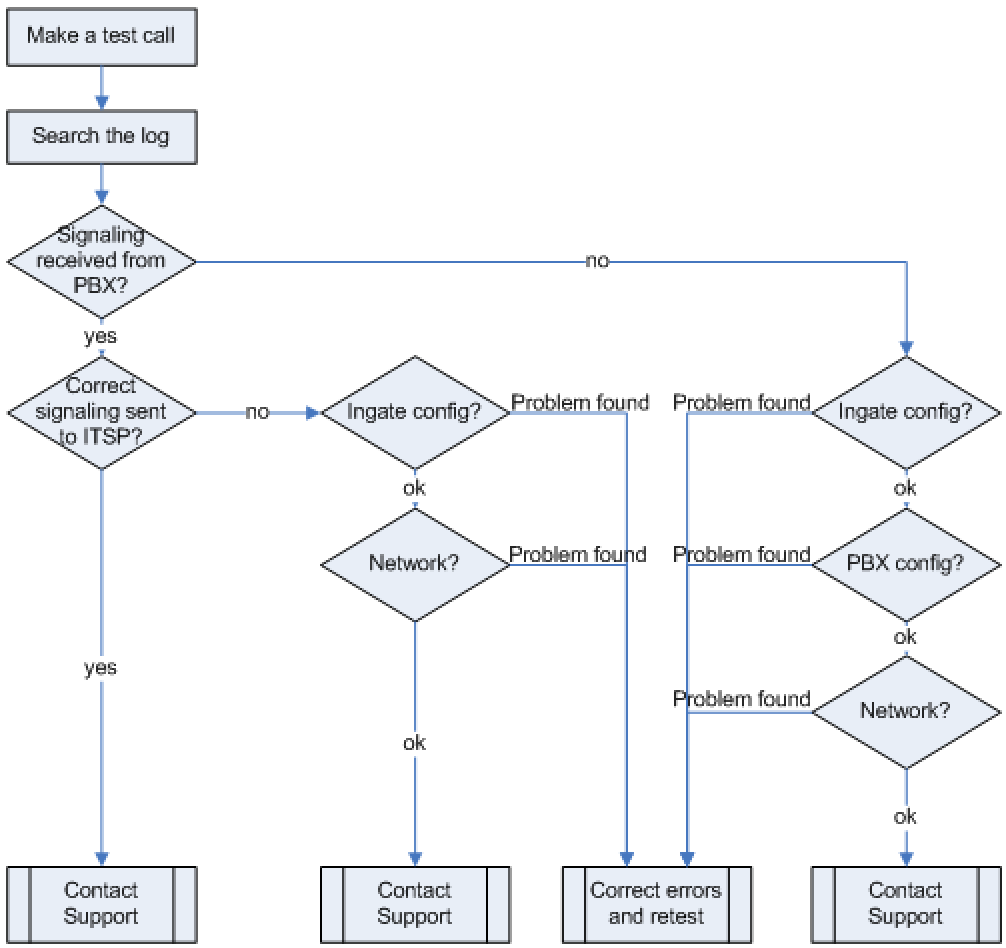
The configuration of the Ingate is now done and the changes must be applied on the Administration page to take effect.

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.

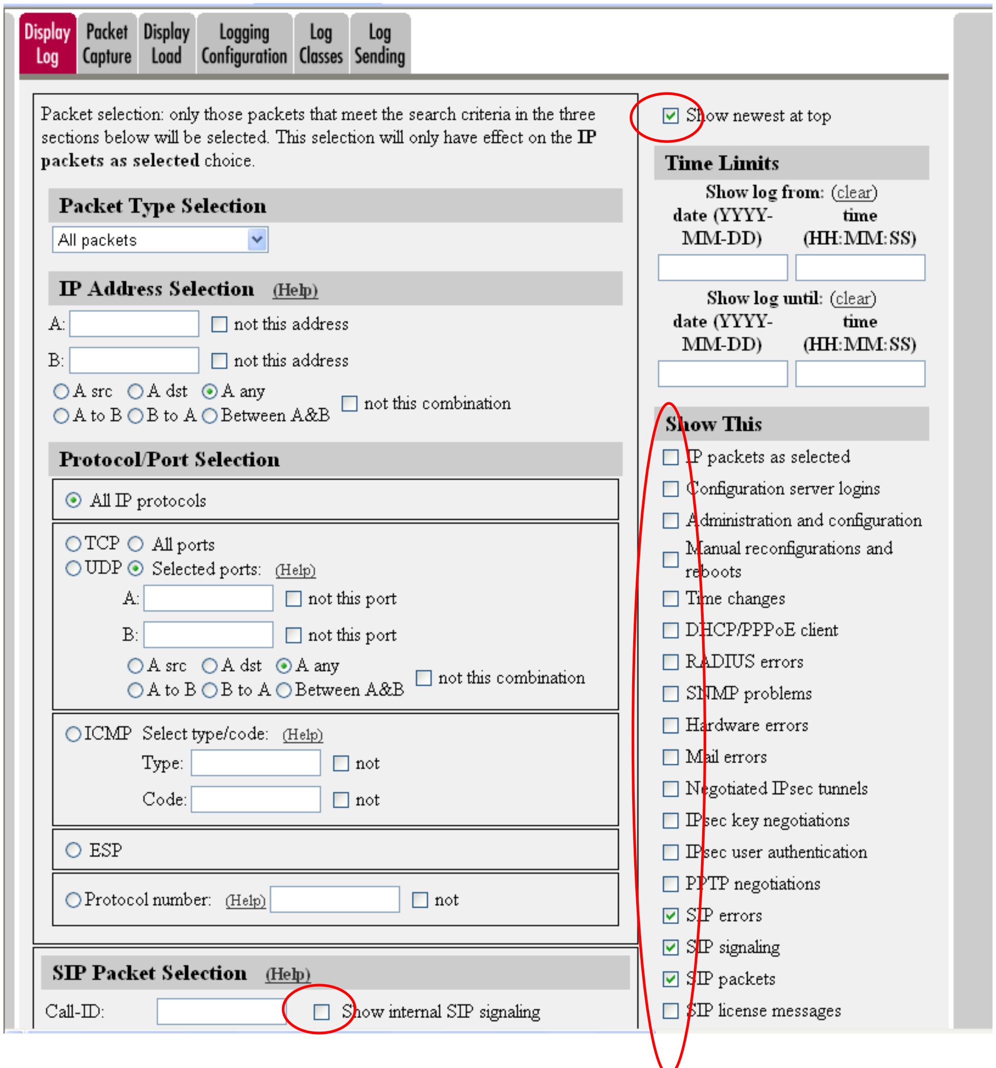


Outbound traffic troubleshooting overview

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.

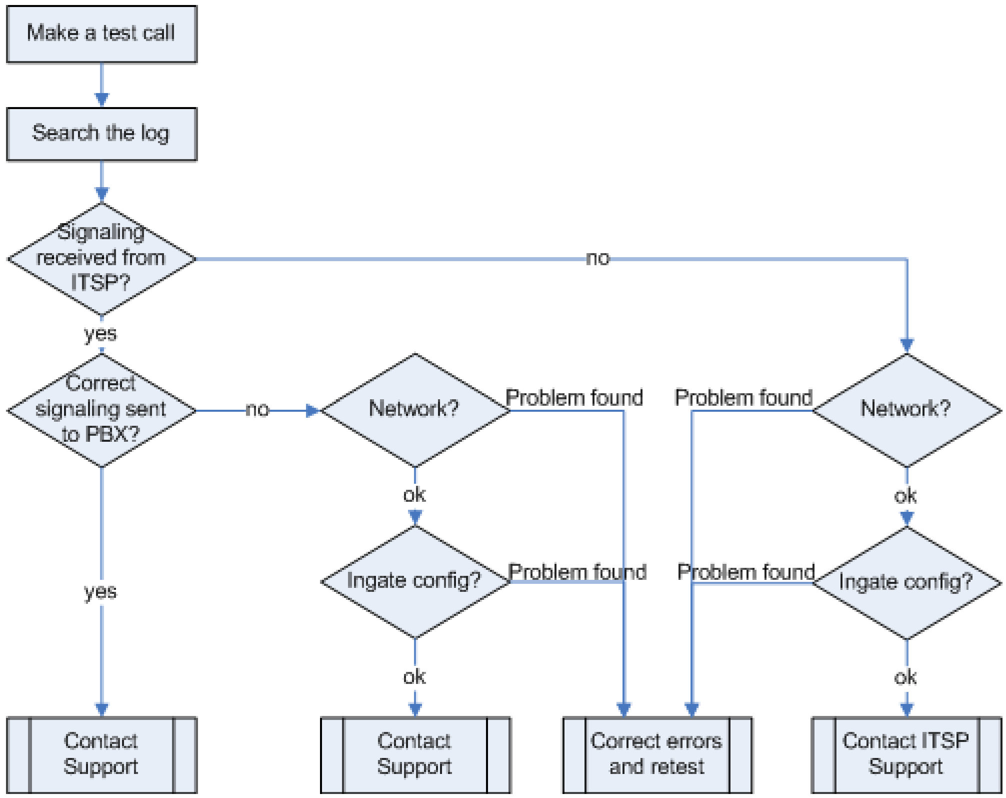
If you see a packet sent from the Ingate, verify that it was sent to the IP address provided by the ITSP. If not, correct your configuration and retest.

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.

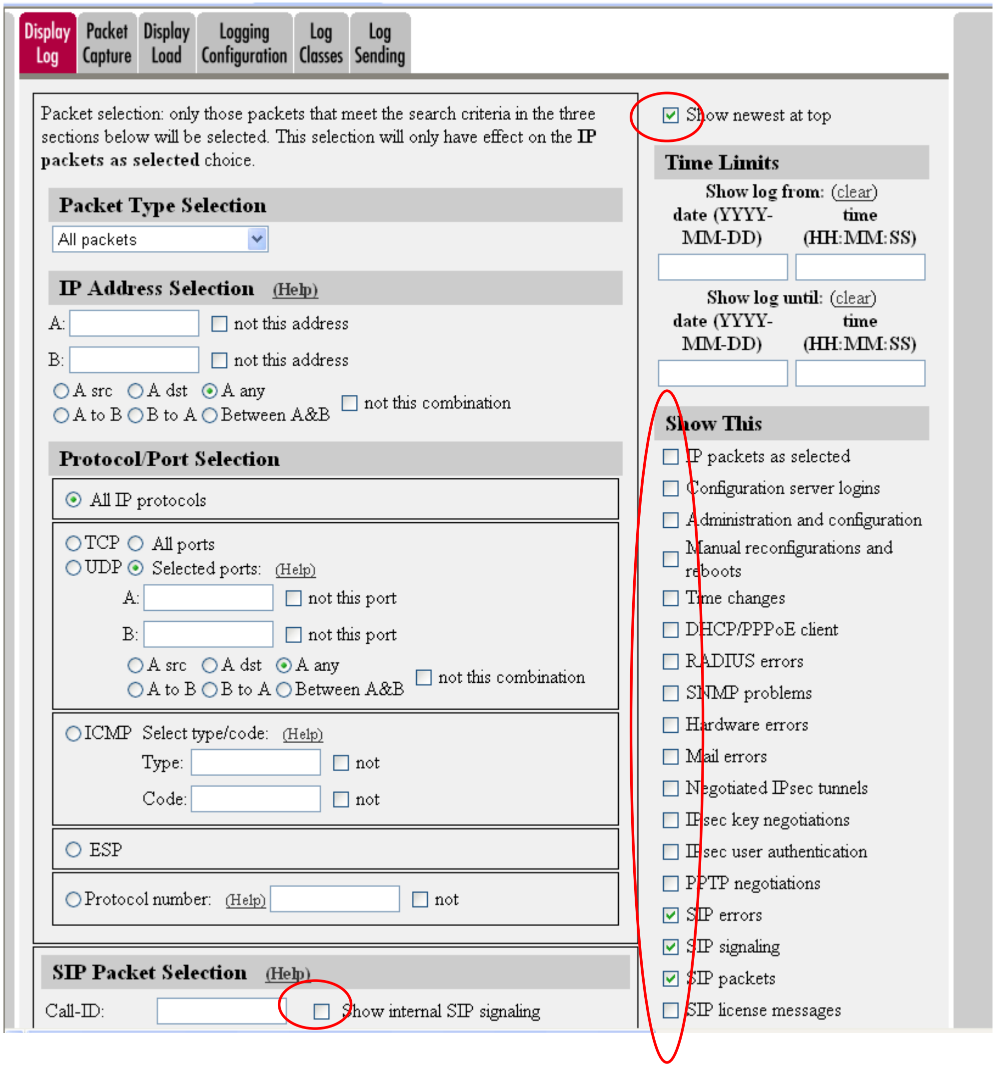


Inbound troubleshooting overview

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 sure that 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

AireSpring Configuration & Support

AireSpring Special Configuration Parameters

Outbound digits delivery: (ShoreTel – -- > AireSpring)

1. ANI Delivery = 11 digits

2. DN Delivery = 10 digits

3. International Termination = 011CC

4. Codecs supported = G711 /G729

AireSpring Support Information.

Network Operations Center (NOC): 800 825 -1055 (OPT1/1)

Product: VoIP/SIP

Brent Shaw – VoIP Engineer : 888 899-2789 x245

Dumitru Borsan – Dir. of Network Engineering : 888 899-2789 x250

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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