Configuration Guide
For Use with tIPicall’s
SIP Trunking Service

(Physical & Virtual ShoreGear Switches with Ingate SIParator SBC)

Version 1.0
February 2015

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ShoreTel tests and validates the interoperability of the Member’s solution with ShoreTel's published software interfaces. ShoreTel does not test, nor vouch for the Member's development and/or quality assurance process, nor the overall feature functionality of the Member's solution(s). ShoreTel does not test the Member's solution under load or assess the scalability of the Member’s solution. It is the responsibility of the Member to ensure their solution is current with ShoreTel's published interfaces.

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

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1. Introduction

1.1 Pre ShoreTel IP PBX & SIParator SBC Configuration Activity

This guide assumes that the administrator is knowledgeable in configuring and administering the ShoreTel IP PBX and the SIParator SBC. An important tool that administrators should have at their disposal prior to testing their ShoreTel IP PBX with tIPicall SIP is a network protocol analyzer. Such software can be used to run traces on problem calls so the information can be shared with equipment and network engineers. There is a free version of such software that can be obtained at [http://www.wireshark.org/](http://www.wireshark.org/). A second alternative that customers may use is TCPDUMP which can be found on most UNIX and Linux systems. To use this software the customer should have Wireshark or TCPDUMP loaded on a server that is connected to a LAN switch or hub that can monitor both the signaling and media packets on any calls between the customer PBX and the edge router. The SIParator SBC has tools available that can also be utilized for this; refer to Ingate documentation for further information. Please note, however, that tIPicall does not offer, warrant, or support this software, and any use of the Wireshark or TCPDUMP software is entirely at the customer’s own risk.

1.2 Customer Questions

Section 5 of this guide provides screen shots and instructions for the configuration of the ShoreTel IP PBX and Ingate SBC. Should you have questions regarding these instructions, please call ShoreTel at 1-800-742-2348. When calling this number please have the following information available:

- Company Name
- Company Location
- Administrator Name & phone number
- ShoreTel release and build number
- Ingate SBC type and version
- Customer Configuration Guide - Issue number & date

1.3 Trouble Reporting

In the event that you experience problems with the ShoreTel system you may contact ShoreTel Technical Assistance Center at +1 (800) 742-2348 (Toll Free) or +1 (408) 331-3313 (International). A support contract must be in place before any assistance will be provided, for contract / account questions please send an email to shorecare_admin@shoretel.com. tIPicall can be contacted for support with the SIP service on +44 (0) 20 3328 4500.

ShoreTel, Ingate and tIPicall will make every effort to quickly resolve reported troubles. The time required for trouble shooting can be reduced if the customer has the necessary detailed
information available when reporting a problem. Please consider taking a Wireshark or TCPDUMP trace of the failed call and provide it to support along with other requested logs.

### 1.4 Document Feedback

ShoreTel IP PBX administrators who would like to provide feedback on the contents of this document should send it to INFeedback@ShoreTel.com.

### 1.5 Document Change History

<table>
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<th>Date</th>
<th>Description</th>
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<td>December 2014; Initial draft</td>
<td></td>
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<tr>
<td>0.2</td>
<td>February 2015; First draft for review</td>
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### 2. Version Information

<table>
<thead>
<tr>
<th>ShoreTel Release</th>
<th>Ingate SIParator</th>
</tr>
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<tbody>
<tr>
<td>Version 5.0.3</td>
<td></td>
</tr>
<tr>
<td>14.2 Build 19.44.2503.0</td>
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Following are screen shots of the versions of ShoreTel and Ingate SIParator utilized for testing interoperability with tIPicall’s SIP Service.

**Figure 1 – ShoreTel Director Login Page**

**Figure 2 – Ingate SIParator Login Page**
3. Special Notes

1.6 Emergency 999 Services Limitations and Restrictions

Although tIPicall provides 999 calling capabilities, tIPicall does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer configuration guide will properly operate with tIPicall’s SIP service to complete 999 calls; therefore, it is the Customer's responsibility to ensure proper operation with its equipment/software vendor.

It is also the responsibility of the customer to provide tIPicall with accurate location information in order for the 999 services to know the location of the premises. If the customer fails to provide accurate location information to tIPicall it cannot be provided to the Emergency Services.

While tIPicall’s SIP service supports 999 calling capabilities, there are circumstances when that 999 service may not be available. Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the Customer's location in the automatic location information database. Please review the tIPicall SIP Service Guide in detail to understand the limitations and restrictions.

1.7 ShoreTel Virtual Switch Support

Starting with ShoreTel 14.2, ShoreTel added support for Virtual Trunk and Virtual Phone switches. The testing carried out for tIPicall’s SIP service currently includes both physical and virtual switches.

1.8 Support for ShoreTel Mobility and 3rd Party SIP handsets

Note that this version of the application note does not include any testing of the ShoreTel Mobility Router or any 3rd party SIP handsets. These will be tested and included in a later revision of the document.

1.9 Support for originating calling line ID on forwarded calls

In this revision of the application note there is no support for the forwarding of originating caller’s CLID on calls that are forwarded externally using features such as Find Me, External Assignment and Call Handling Modes.

1.10 Support for the sending and receiving of fax transmissions

The testing for the tIPicall service included sending and receiving fax transmissions, however tIPicall only support pass through at the time of writing this document, not T38. The testing proved positive when using G711a as the codec, the fax machine’s user set to ‘Fax Server’ and the site fax codecs using a custom list based on the Fax Codec Low Bandwidth Passthrough list with G711a at the highest priority.
1.11 The tIPicall SIP Trunking Service

tIPicall has grown from a group of companies with over 12 years of strong telecoms history in the UK. They own and operate their own SIP network and control a UK based proprietary hosted PBX platform.

On their state of the art network tIPicall provide many value added services including international numbers, fraud protection, disaster recovery and smart diverts. tIPicall are a channel only business and do not sell direct into the market. Everything they do exists to look after our partners and their end customers.

tIPicall have a very experienced team who run the business and this team not only have decades of telecoms experience but they also know what it is like to be you as they have run and operated reseller and dealer businesses of their own. The team of highly experienced and helpful technical staff are regarded as some of the best in the industry and we take pride in assisting our partners at every level.

tIPicall’s international services are one of the many reasons their partners work with them. tIPicall have the ability to provide local numbers in dozens of countries and Freephone numbers in over 100 countries. They also supply their SIP and hosted services throughout the world. Partnerships with leading UK data providers allow end users to connect with quality guarantees and peace of mind.

The tIPicall private SIP product is designed for voice at the highest quality and they can provide private SIP from a single seat to thousands of seats and trunks.

tIPicall’s next generation network is fully resilient and spread across multiple locations. The network is built to the highest specifications and the architecture is designed around scalability and resilience. For more information on tIPicall's services you can visit their website on http://www.tipicall.co.uk.

1.12 ShoreTel Unsupported Features and Limitations

Please refer to the ShoreTel Administration Guide, Chapter 18 – Session Initiation Protocol, for supported and unsupported features via SIP Trunks. Following are some feature limitations via SIP Trunks:

- Fax redirect not supported via SIP Trunks using G.711 (though Direct Inward Dialing (DID) to fax endpoint is supported)
- ShoreTel supports Music On Hold (MOH) over SIP trunks. The maximum number of music on hold (MOH) streams that a SIP-enabled switch can support varies with the switch model. The range of such streams across all the voice switch models is 14–60. Limitation: MOH source needs be on SIP trunk switch.
- If the ShoreTel server has a conference bridge 4.2 installed, you should not enable SIP. The conference bridge is not compatible with a ShoreTel system that has SIP enabled due to the dynamic RTP port required for SIP.
• ShoreTel supports the Service Appliance (SA-100) conferencing / IM system from Release -12. SIP trunk calls from / to the SA-100 is supported. The SA-100 accepts access codes in DTMF RFC2833 only.
• 6 party conferences, when a SIP trunk is involved, utilize Make Me conference ports.
• Silent Monitoring, Barge-In, Silent Coach, Park/Unpark , Call recording features are supported on a SIP trunk call only if SIP trunk is configured with SIP profile supporting media hairpinning and the trunk is on a half-width switch.
• Silence detection on trunk-to-trunk transfers is not supported, it requires a physical trunk.
• The ShoreTel system does not initiate calls with a 30ms payload; all calls are initiated with a 20ms payload.
• IP 400 series Phone might exhibit one way audio issue while using Network based Account Codes due to ShoreTel Defect ENG-041503. This issue will be addressed in a future ShoreTel Release.
• At the time of writing this document there is an open defect where CLID is not displayed correctly when hunt group or workgroup overflow calls are transferred to IP400 series handsets. This issue has been identified and is currently pending patch release.

At this time we are unable to provide additional information on a resolution to the issues mentioned above, but suggest to periodically refer to the ShoreTel 14.2 Software Release Notice (Build Notes) for updates, which can be found at the following location:

http://support.shoretel.com

There may be other feature limitations when using SIP Trunks. Please refer to Chapter 18 of the ShoreTel Administration Guide.

With Physical Switches, “SIP Media Proxy” resources are not allocated by default and must be configured as per requirement. Please refer to the ShoreTel Partner guide for additional details about SIP Media Proxy and SIP Trunk capacity at the following location http://partners.shoretel.com/product_sales_tools/ip_phone_system/shoretel_13/downloads/shoretel_13_partner_guide.pdf.

This same guide is also applicable for half width physical switches in 14.x release.
4. Configuration Component Overview

This section provides a more detailed description of the ShoreTel / Ingate requirements and configuration.

The ShoreTel / Ingate environment is shown next.

![Configuration Component Overview Diagram](image)

**Figure 3 – Configuration Component Overview**

The ShoreTel / Ingate customer premises site shall consist of the following components:

- **ShoreTel IP Phones** – The ShoreTel 2xx/5xx/6xx series IP phones run MGCP and exchange MGCP messages with the ShoreGear switches. The 4xx series IP phones run SIP.

- **ShoreTel PBX** – This PBX connects to the IP phones using MGCP. It connects to the Ingate Session Border Controller using SIP. The ShoreTel PBX implements PBX functionality including phone features, calling routing, voice mail, etc. The test environment uses virtualized phone switches for handset call control, and an SG90 for connectivity to the SIP trunks.

- **vCollaboration** – This is a virtualized ShoreTel service appliance, providing web and audio conferencing services and instant messaging functionality to the ShoreTel users.

- **Ingate SIParator Session Border Controller (SBC)** – The Ingate SBC exchanges SIP with the ShoreGear switches, in this case a single SG90 switch. This SBC then exchanges the SIP messages with the tIPicall network. The Ingate SBC performs some SIP
conversion functions to resolve incompatibilities between the ShoreGear switches and the tIPicall network. In particular, Ingate is performing header manipulation to ensure the SIP headers are in a format compatible with the tIPicall service. The SBC should be configured with a single interface connecting into a private subnet accessible by the ShoreTel platform, and a further interfaces connecting to an untrusted network / DMZ.

- **Edge Router** – This is the router at the edge of the LAN providing connectivity to the tIPicall network. In this example connectivity was via the internet, but other options do exist from tIPicall. A static public IP address is required which must be noted down and provided to tIPicall when the service is provisioned. It will also be used during the configuration process. The edge router must provide an interface for the Ingate SBC to connect into the DMZ and then traffic to the chosen public IP address for the RTP range of ports and SIP ports must be forwarded to the DMZ IP address of the Ingate using NAT translations. Guidelines on which IP addresses should be allowed through the firewall from tIPicall will be provided by tIPicall.

As shown in the diagram above, MGCP & SIP signaling is used between the IP phones and the ShoreGear switches. The ShoreGear switch uses SIP to communicate to the Ingate SBC. The Ingate SBC then uses SIP to communicate to the tIPicall network. The RTP voice traffic flows from the IP phones to the Ingate SBC and then to the tIPicall network, in other words, the Ingate SBC is acting as a back to back user agent (B2BUA).

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

1.13 ShoreTel Configuration

This section describes the ShoreTel system configuration to support SIP Trunking. The section is divided into 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 SBC which is acting as a B2BUA.

1.13.1 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 on to the “ShoreTel System Settings – Trunk Groups” section below.

1.13.2 Call Control Settings

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

![ShoreTel Director](image)

Figure 4 – Administration Call Control Options
The “Call Control Options” screen will then appear (Figure 5).

**Figure 5 - Call Control Options**

In the “General” parameters, the “DTMF Payload Type (96 – 127)” defaults to a value of “102”, and no modification is necessary to interoperate with tIPicall. Within the “SIP” parameters; confirm that the appropriate settings are made for the “Realm” “Enable SIP Session Timer” and “Always Use Port 5004 for RTP” parameters.

The “Realm” parameter is used in authenticating all SIP devices. It is typically a description of the computer or system being accessed. Changing this value will require a reboot of all ShoreGear switches serving SIP extensions. It is not necessary to modify this parameter to get the ShoreTel IP PBX system functional with tIPicall. Verify that the “Enable SIP Session Timer” box is checked (enabled). Next the Session Interval Timer needs to be set. 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 “Voice Encoding and Quality of Service”, specifically the “Media Encryption” parameter, make sure this parameter is set to “None”, otherwise you may experience one-way audio issues. Please refer to ShoreTel’s Administration Guide for additional details on media encryption and the other parameters in the “Voice Encoding and Quality of Service” area.

The ShoreTel legacy parameter “Always Use Port 5004 for RTP” should be disabled by default, if it’s enabled you will need to disable it. It is required for implementing SIP on the ShoreTel system. For SIP configurations, Dynamic User Datagram Protocol (UDP) must be used for RTP Traffic. If the parameter is disabled, 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 disabled (unchecked), make sure that “everything” (IP Phones, ShoreGear® Switches, ShoreTel Server, Distributed Voice Mail Servers / 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.

1.13.3 Sites Settings

The next settings to address are the administration of sites. These settings are modified under the ShoreTel Director by selecting “Administration” then “Sites” (Figure 6).

Figure 6 – Site Administration
This selection brings up the “Sites” screen. Within the “Sites” screen select the name of the site to configure. The “Edit Site” screen will then appear. The only changes required to the “Edit Site” screen are to the “Admission Control Bandwidth” and “Intra-Site / Inter-Site Calls” parameters (Figure 7).

**Figure 7 – Site Bandwidth settings**

*Note:* Bandwidth of 2048 is just an example. Please refer to the *ShoreTel Planning and Installation Guide* for additional information on setting Admission Control Bandwidth.

### 1.13.4 Sites Edit screen – Admission Control Bandwidth

The Admission Control Bandwidth defines the bandwidth available to and from the site. This is important as SIP trunk calls may be counted against the site bandwidth. Bandwidth needs to be set appropriately based on site setup and configuration with tIPicall’s SIP Service. Please refer to the *ShoreTel Planning and Installation Guide* for additional information.

**Figure 8 – Codec List Setup**

### 1.13.5 Sites Edit screen – Intra / Inter-Site Calls

By default, ShoreTel 14.x has 12 built-in codecs; these codecs can be grouped as “Codec Lists” and defined in the sites page for “Inter-site” and “Intra-site” calls. Configure the "Intra-Site Calls" option to a “Codec List” that contains the desired codecs and save the change. When establishing a call with tIPicall’s SIP Service the preferred codec choice is G.711a, although it is possible to use G729 and G711u. The site that the SIP Trunk Group belongs to will determine
which “Intra-Site” Codec List will be utilized so be sure to move the desired codec up the list for higher priority. Our suggestion is to create a new codec list named “G711” (or similar) and then add G.711a, G729 and G.711u as a minimum but do modify this list depending on your requirements for IP phone codecs on the LAN. Please refer to the *ShoreTel Planning and Installation Guide* for additional information.

### 1.13.6 Switch Settings - Allocating Ports for SIP Trunks

The final general settings to configure are the ShoreGear switch settings. These changes are modified by selecting “Administration” then “Platform Hardware…”, then “Voice Switches / Service Appliances…” followed by “Primary” in ShoreTel Director (Figure 8).

![Figure 9 – Administration Switches](image)

This action brings up the “Primary Voice Switches / Service Appliances” screen. From that screen simply select the name of the switch to configure. 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 10). Note that this step is not required for the virtualized trunk switches as the resources on these appliances are automatically available when the switch is added in Director.
Each port designated as a SIP Trunk enables the support for 5 individual trunks.

Starting with ShoreTel 13, the additional option of “Port Type” was added for half-width ShoreGear switches. The new selection is called “SIP Trunk with Media Proxy”. It ensures that the ShoreTel system that is being used for SIP Trunks will provide feature parity similar to PRI trunks. These feature include RFC 2833 DTMF detection for Office Anywhere, External or Simultaneous Ring calls, three party Mesh Conferencing (without needing to configure “MakeMe” conference ports), Call Recording, Silent Monitoring, Barge-In, Whisper Page, Invites with no SDP and when there’s no common codec between ITSP and the local extension. Many of the features included in the test plan require SIP media proxy resources. It is therefore recommended that these are included in your design.

For further information on “SIP Trunk with Media Proxy” please refer to Chapter 18 of the ShoreTel 14.x System Administration Guide.

**Figure 10 – ShoreGear Switch Settings**
1.13.7 **ShoreTel System Settings – SIP Profiles**

ShoreTel’s default ITSP SIP trunk profile can be used as a basis for a custom profile used with the tIPicall trunks. A particular parameter is required for interoperability with tIPicall therefore we recommend creating a new profile specifically for use with the tIPicall trunks.

These changes are made by selecting “Trunks” then “SIP Profiles”, then click on the “Default ITSP” profile to enter its settings. Click on the ‘Copy” button to create a copy of this profile. Rename it to “TIPICALL” and add the parameter “IgnoreEarlyMedia=1” (case sensitive) and save the settings (Figure 11).

<table>
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<th>Name:</th>
<th>Tipicall</th>
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<td>User Agent:</td>
<td>.*</td>
</tr>
<tr>
<td>Priority:</td>
<td>100</td>
</tr>
</tbody>
</table>

**System Parameters:**
- OptionsPing=1
- OptionsPeriod=60
- StripVideoCodec=1
- DoniFwdRefer=1
- SendMach911CallSetup=1
- HistoryInfo=diversion
- EnableP.AssortedIdentity=1
- AddGT23AnnexB_NO=1
- Hairpin=1
- Register=0
- RegisterUser=BTN
- RegisterExpiration=3600
- CustomRules=0
- OverwriteFrontUser=0

**Custom Parameters:**
- IgnoreEarlyMedia=1

*Figure 11 – SIP Profile Setup*
1.13.8  **ShoreTel System Settings – Trunk Groups**

ShoreTel Trunk Groups only support Static IP Addresses for Individual Trunks. In trunk planning, the following needs to be considered.

- The Ingate SBC’s LAN and DMZ interfaces should always be configured to use a “Static” IP Address.

The settings for Trunk Groups are changed by selecting “Administration”, then “Trunks” followed by “Trunk Groups” within ShoreTel Director (**Figure 10**).

![ShoreTel Director](image)

**Figure 12 – Administration Trunk Groups**

This selection brings up the “Trunk Groups” screen (**Figure 11**).

![Trunk Groups](image)

**Figure 13 – Trunk Groups Settings**

From the pull down menus on the “Trunk Groups” screen, select the site desired and select the “SIP” trunk type to configure. Then click on the “Go” link from “Add new trunk group at site”. The “Edit SIP Trunk Group” screen will appear (**Figure 13**).
Trunk Groups
Edit SIP Trunk Group

[Image of the Trunk Groups page]

Enable SIP Info for G.711 DTMF Signaling
Profile:
Digest Authentication:
Username:
Password:

**Figure 14 – SIP Trunk Group Settings**

The “Enable SIP Info for G.711 DTMF Signaling” parameter should not be enabled (checked). Enabling SIP info is currently only used with SIP tie trunks between ShoreTel systems.

The “Profile:” parameter should be changed to the SIP profile created in the previous section. The “Enable Digest Authentication” parameter defaults to “<None>” and modification is not required when connecting to tPicall SIP Trunking.

The next item to change in the “Edit SIP Trunks Group” screen is to make the appropriate settings for the “Inbound:” parameters. (Figure 13).

**Inbound:**

Number of Digits from CO:
- [ ] DNIS
- [ ] DID
- [ ] Extension

- Translation Table: <None>
- Prepend Dial In Prefix:
- Use Site Extension Prefix

- Tandem Trunking
  - User Group:
  - Prepend Dial In Prefix:
  - Destination:

**Figure 15 – Inbound Trunk Group Settings**
Within the “Inbound:” settings, ensure the “Number of Digits from CO:” is configured to a value of “6”, this is the number of digits that the ShoreGear SIP trunk switch will be receiving from the Ingate SBC which will be configured to receive 12 digits from tIPicall but will then strip the first 6 digits leaving only the last 6 digits of the DID numbers. Enable (check) the “DNIS” or “DID” parameters as needed. For additional information on these parameters please refer to the ShoreTel Administration Guide.

The following section is configured in the same way as any normal Trunk Group.

Figure 16 – Outbound and Trunk Services

Enable (check) the “Outbound” parameter and define a Trunk “Access Code” and “Local Area Code” as appropriate. In the “Billing Telephone Number:” be sure to specify the main telephone number provided by tIPicall for your account.

In the “Trunk Services:” area, make sure the appropriate services are enabled or disabled based on what tIPicall’s service supports and what features are needed from this Trunk Group. You will need to disable (uncheck) the “Enable Original Called Information” parameter.

The parameter “Caller ID not blocked by default” determines if the call is sent out as <unknown> or with caller information (Caller ID), be sure to enable (check) this parameter. User DID will impact how information is passed out to the SIP Trunk group.
After these settings are made to the “Edit SIP Trunk Group” screen, select the “Save” button to input the changes.

The final parameters for configuration in the Trunk Group are “Trunk Digit Manipulation”, which in the build tested could be left at their default values (figure 16):

![Trunk Digit Manipulation](image)

**Figure 17 – Trunk Digit Manipulation**

No changes are required from the default to interoperable with tIPicall’s service. The option to dial local numbers in national form must be left enabled to ensure locally dialed numbers are formatted correctly before being sent to the SBC.

### 1.13.9 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 19).

![Director](image)

**Figure 18 – Individual Trunks**

The “Trunks by Group” screen is used to change the individual trunks settings that appear (Figure 18).
Figure 19 – 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 “tIPicall”, as created above, see Figure 20. Click on the “Go” button to bring up the “Edit Trunk” screen (Figure 21).

Figure 20 - Edit Trunks Screen for Individual Trunks

From the individual trunks “Edit Trunk” screen, 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 “IP Address”, define the IP address of the LAN interface on the Ingate SBC. The last step is to select the number of individual trunks desired (each one supports “one” audio path – example if 10 is configured, then 10 audio paths can be established at one time). Once these changes are complete, select the “Save” button to commit changes.

Note: The configuration and design presented within this document assumes a single SBC and ShoreGear switch. The configuration steps required to span trunks over multiple switches has not been tested in this revision and is therefore not included in the scope of this document.

After setting up the trunk groups and individual trunks, refer to the ShoreTel Planning and Installation Guide to make the appropriate changes for the User Group settings. This completes the settings for the ShoreTel system side.
1.14 ShoreTel Technical Support

In the event that you have problems with the ShoreTel system you may contact ShoreTel Technical Assistance Center at +1 (800) 742-2348 (Toll Free) or +1 (408) 331-3313 (International). A support contract must be in place before any assistance will be provided, for contract / account questions please send an email to shorecare_admin@shoretel.com.
1.15 Ingate SBC Configuration

1.15.1 Ingate Setup Introduction

When the Ingate SBC is first powered on, an initial setup process must be followed before any tIPicall specific configuration is applied.

Before moving on, please make a note of the IP addresses and usernames that have been used through this example so you can modify them according to your environment.

- Ingate SIParator LAN interface: 172.16.0.40
- Ingate SIParator DMZ interface: 172.16.1.40
- Ingate SIParator LAN MAC: 00-0C-29-07-3B-8B
- Gateway address in the DMZ: 172.16.1.254
- ShoreTel Trunk Switch: 172.16.0.38
- tIPicall SIP Platform: 91.146.112.10
- Firewall public IP address: 195.162.111.46

1.15.2 Running the Ingate Startup Tool (SUT)

a. Download and install the Ingate SUT from www.ingate.com. When you run the tool you’ll be asked to specify the SBC type.

![Select Product Type](image)

*Figure 21 – Selecting the SBC Type within the SUT*
b. Choose ‘Ingate Firewall/SIParator’ from the drop down list and click Next.

c. In the next screen you must specify the MAC address of the LAN interface of the SIParator device. On the physical SIParators this can be found on a label attached to the device itself. The virtual appliance’s MAC address can be found by looking at the virtual machine’s properties within the VSphere client.

d. In the IP address box choose an IP address that will be assigned to the LAN interface of the unit.

e. Choose a password to assign to the administration interface of the device.

f. At a minimum, select the options to ‘Configure the unit for the first time’ and check the box for ‘Configure SIP trunking’.

g. Click on the ‘Connect’ button.
h. In the ‘Network Topology’ tab configure the following items:
   a. **Product Type:** DMZ-LAN SIParator
   b. **Eth0 IP address / Netmask:** This is the IP address and mask of the internal interface.
   c. **Eth1 IP address / Netmask:** This is the IP address and mask of the DMZ interface.
   d. **Gateway:** This is the default gateway that the SIParator will use to leave the network.
   e. **Use NATing firewall:** Set this to the public IP address of the firewall that the SIParator sits behind.
   f. **DNS Server:** The primary and secondary DNS server addresses for your network.
i. Click on the IP-PBX tab and configure the following items:
   a. Type: Select ‘ShoreTel Shoregear’ from the drop down menu.
   b. IP Address: Enter the LAN IP address of the ShoreGear trunk switch.

![Setting the IP-PBX parameters](image)

**Figure 23 – Setting the IP-PBX parameters**

j. Click on the ITSP tab and configure the following items:
   a. Name: Select ‘Generic (no-register)” from the drop down menu.
   b. Provider IP Address: Enter the SIP address provided by tIPicall.
Figure 24 – Setting the provider’s parameters

k. Click on the ‘Upload Configuration’ tab, make sure ‘Logon to web GUI and apply settings’ is checked and then click on the ‘Upload’ button.
Figure 25 – Uploading the settings

The SIParator will now be configured with a set of defaults according to the SUT. The next step is to ensure the various networks are correctly defined for use in other parts of the configuration.

Make sure you save your changes, but you do not yet have to commit the changes as this can be done after all of the configuration has been completed.

1.15.3 Configure Media Parameters

Navigate to SIP Services / Sessions and Media and find the ‘Limitation of RTP Codecs’ section. Choose the option to Allow All Codecs and save the changes.
Figure 28 – Allowing all Codecs in Sessions and Media

### 1.15.4 Configure the SIP Trunk Parameters

Go into SIP Trunks, choose the Generic (no register) trunk and click on the ‘Goto SIP Trunk Page’ button.

- Changes have been made to the preliminary configuration, but have not been applied.

Figure 29 – Selecting the SIP Trunk

You must now configure a number of parameters from the default settings created by the SUT. Set the parameters as follows:

- **Service Provider Domain / Outbound Proxy**: Ensure the SIP platform IP address provided by tIPicall is in these fields.
- **Main Trunk Line, Outgoing Calls, Display Name**: Set this to the main billing telephone number without the leading 0.
- **Main Trunk Line, Incoming Calls, Trunk Match**: Remove the contents.
- **Main Trunk Line, Incoming Calls, Forward To**: Remove the contents.
- **PBX Lines, Outgoing Calls, Line 2, Display Name & User Name**: Set these both to $1.
- **PBX Lines, Incoming Calls, Incoming Trunk Match**: tIPicall send inbound calls using canonical format (the UK number with the country code including the +). This simple example shows how to strip the first part of the received number and send the remaining...
6 digits to the ShoreTel. In this field enter \+44 and the first 4 digits of your assigned number range followed by (.*). There are three examples in the presented dial plan to show different ways this can be achieved, although a discussion on Regular Expressions is beyond the scope of this document.

- In this example the inbound numbers are 020 3540 6872, 020 3750 1175-9, 020 3750 1180-5
- tIPicall will send +442035406872, for example, to the SBC
- $1 is referring to the section within parentheses. This is what is sent to ShoreTel.
- Add more lines if you have additional ranges with different area codes.

**PBX Lines, Outgoing Calls:** These entries ensure that only valid CLID’s are sent on outbound calls to the tIPicall service. There are 5 entries in this example which must be tailored to suit the environment of the reader. Each of the presented lines are explained here. Use this as a guide.

- Anonymous / Anonymous.invalid – This is for when the CLID is blocked within the ShoreTel trunk group configuration. In this case ShoreTel will send “Anonymous” in the FROM field. This line ensures that the INVITE sent to tIPicall contains the correct information in the FROM field to ensure the CLI is blocked. The contents of the Identity column should contain one of your tIPicall DDI numbers followed by your public IP address. For example, +442037501176@195.162.111.46.
- The following four entries match valid CLI’s and send them untouched. These lines should include all valid provisioned telephone numbers and you can use Regular Expressions to define these if you wish.
- The final line catches anything other than valid CLI’s. If you attempt to send an invalid CLI from ShoreTel the call will fail. This line ensures that if you do attempt to send an invalid CLI the SIParator will replace it with a number of your choosing in both the username column of the FROM header (the Username field) and the number in the P-Asserted ID (the Identity field).

**PBX Lines, Incoming Calls, Forward To:** Enter $1 to send the 6 digits to the IP PBX defined in the SUT.
Figure 30 – Modifying the SIP Trunk (Service Parameters)

Figure 31 – Modifying the SIP Trunk (Main Trunk & PBX Line Parameters)
1.15.5  **Configure the Dial Plan**

Navigate to SIP Traffic / Dial Plan and follow these steps to complete the dial plan. Pay special attention to the IP addresses used in this example as these must be substituted for the correct IP addresses for your environment. The highlighted areas show the addresses that must be modified.

a. Set the Emergency number to 999.

b. Modify the ‘Matching Request URI’ section of the Dial Plan.
   a. Set the RegExp field of the ‘Outbound’ entry to `sip:(.*)@172.16.0.40|sip:172.16.0.40:5060`. This is the LAN interface IP of the SIParator.
   b. Add a new row.
      i. Set the name to ‘ST_OPTIONS’, the Tail to – and the RegExp to `sip:172.16.0.40|sip:@172.16.0.40`. This is the LAN interface IP of the SIParator.

![Table showing Matching Request URI settings](image)

**Figure 32 – Modifying the Request URI section of the Dial Plan**

c. Modify the ‘Forward To’ section of the Dial Plan.
   a. Add a new row.
      i. Set the name to tIPicall OPTIONS and the Replacement Domain to 91.146.112.10. This is the platform IP address provided by tIPicall (the same one used within the SIP trunk configuration in the previous section).

![Table showing Forward To settings](image)

**Figure 33 – Modifying the Forward To section of the Dial Plan**

d. Modify the Dial Plan section towards the bottom.
   a. Add a new row.
      i. Select the From Header to be ShoreTel ShoreGear, the Request URI to be ST_OPTIONS, the Action to be Forward and the Forward To to be tIPicall OPTIONS.
   b. Reorder the rows as per the screenshot below. This is accomplished by manually typing a new position (1-3) into the ‘No.’ column for all of the rows and then saving your changes.
### Figure 34 – Modifying the Dial Plan section of the Dial Plan

The order within this part of the dial plan is very important as dial plan matching is completed in the order configured here. Once a match is found the table is no longer searched. A summary of the rows to be added / modified are shown in the table below.

#### Matching Request URI:

<table>
<thead>
<tr>
<th>Name</th>
<th>Tail</th>
<th>RegExp</th>
</tr>
</thead>
<tbody>
<tr>
<td>Outbound</td>
<td>-</td>
<td>sip:(.*)@172.16.0.40</td>
</tr>
<tr>
<td>ST_OPTIONS</td>
<td>-</td>
<td>sip:172.16.0.40</td>
</tr>
</tbody>
</table>

#### Forward To:

<table>
<thead>
<tr>
<th>Name</th>
<th>Replacement Domain</th>
<th>RegExp</th>
</tr>
</thead>
<tbody>
<tr>
<td>Outbound</td>
<td>-</td>
<td>SIP Trunk 1: Generic (no register)</td>
</tr>
<tr>
<td>tIPicall OPTIONS</td>
<td>91.146.112.10</td>
<td>-</td>
</tr>
</tbody>
</table>

#### Dial Plan:

<table>
<thead>
<tr>
<th>No.</th>
<th>From Header</th>
<th>Request URI</th>
<th>Action</th>
<th>Forward To</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>ShoreTel ShoreGear</td>
<td>ST_OPTIONS</td>
<td>Forward</td>
<td>tIPicall OPTIONS</td>
</tr>
<tr>
<td>2</td>
<td>ShoreTel ShoreGear</td>
<td>Outbound</td>
<td>Forward</td>
<td>Generic (no register)</td>
</tr>
<tr>
<td>4</td>
<td>WAN</td>
<td>-</td>
<td>Reject</td>
<td>-</td>
</tr>
</tbody>
</table>

### 1.15.6 Commit the Configuration Changes

Navigate to Administration / Save Load Configuration. Click on the button to Apply Configuration. In the page that follows click on the Save Configuration Button to complete the process.
Figure 35 – Committing the Configuration Changes

This concludes the Ingate SIParator configuration. All being well you should now be able to make and receive calls to and from the tIPicall SIP service.

6. Troubleshooting Conferencing Failures

In the event that you experience issues with conferencing calls, verify that you have configured Conference resources on the ShoreGear switch. You can also do so by selecting “Administration” then “Platform Hardware…”, then “Voice Switches / Service Appliances…” followed by “Primary” in ShoreTel Director.

Figure 40 – Navigating to Voice Switch Configuration

This action brings up the “Switches” screen. From the “Switches” screen simply select the name of the switch to configure. The “Edit ShoreGear Switch” screen will be displayed. Within the “Edit ShoreGear Switch” screen, select the desired number of Conference resources from the ports available.
Figure 41 – Adding Conferencing Resources

You must enable a minimum of four ports for Conference, otherwise the ShoreTel system will complain. For additional information on Conference ports please refer to the ShoreTel Administration Guide.

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