



**SIP Trunking Configuration Guide
for
ShoreTel ShoreWare
ShoreTel 13.2**

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1 Audience

This document is intended for the SIP trunk customer's technical staff and Value Added Retailer (VAR) having installation and operational responsibilities.

1.1 tekVizion Labs

tekVizion Labs™ is an independent testing and Verification facility offered by tekVizion PVS, Inc. ("tekVizion"). tekVizion Labs offers several types of testing services including:

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The tekVizion team brings together experience from the leading service providers and vendors in telecom. Our unique expertise includes legacy switching services and platforms, and unparalleled product knowledge, interoperability and integration experience on a vast array of VoIP and other next-generation products. We rely on this combined experience to do what we do best: help our clients advance the rollout of services that excite customers and result in new revenues for the bottom line. tekVizion leverages this real-world, multi-vendor integration and test experience and proven processes to offer services to vendors, network operators, enhanced service providers, large enterprises and other professional services firms. tekVizion's headquarters, along with a state-of-the-art test lab and Executive Briefing Center, is located in the Telecom Corridor® in Richardson, Texas.

(For more information on tekVizion and its practice areas, please visit tekVizion Labs's web site at www.tekVizionlabs.com.)

2 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representative of a ShoreTel ShoreWare configuration.

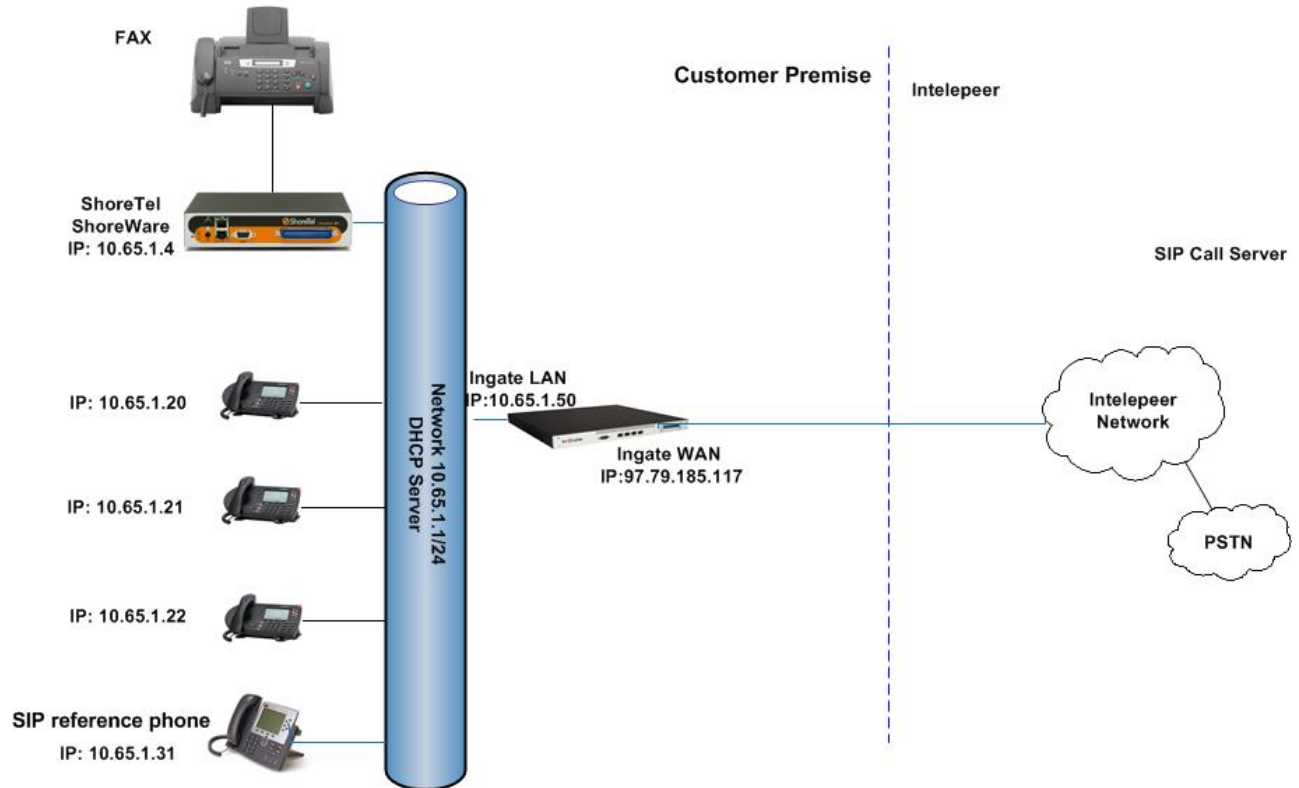


Figure 1 - SIP Trunk Lab Reference Network

The lab network consists of the following components:

- ShoreTel ShoreWare PBX for voice features, SIP proxy and SIP trunk termination.
- Shoregear 90 switch.
- Various MGCP and SIP phones on the local LAN.
- inGate.

3 Shoreware Configuration

3.1 Create SIP Trunk

1. Navigate to Trunks > Trunk Groups.
2. Set **Add new trunk group at site** to Headquarters.
3. Set **of type** to SIP.
4. Click **Go**.



Figure 2: Create Trunk group

5. Set **Name**: Intelepeer Trunk group is used for example.
6. Set **Enable SIP Info for G.711 DTMF Signaling**: checked.
7. Set **Profile**: Default ITSP
8. Set **Digest Authentication**: None
9. Set **Number of Digits from CO**: 10
10. Set **DNIS**: Checked
11. Set **DID**: Checked
12. Click **Edit DID Range**: Enter the Base Phone Number and number of Phone Numbers and click on “**Add this record**” to add the DID Numbers.
13. Click **Save**.
14. Click on “back” button in the top left corner of the browser to get back to trunk group configuration.
15. Confirm **Extension**: unchecked.
16. Set **Tandem Trunking**: Unchecked.
17. Set **Outbound**: Checked.
18. Set **Access Code**: 9 is used for this example.
19. Set Local **Area Code**: This code depends on the given DID range. 347 is used for this example.

Trunk Groups

Edit SIP Trunk Group

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

Edit this record

[Refresh this page](#)

Name:

Intelepeer Trunk Group

Site:

Headquarters

Language:

English(US) ▼

 Enable SIP Info for G.711 DTMF Signaling

Profile:

Default ITSP ▼

Digest Authentication:

<None> ▼

Username:

Password:

Inbound:

Number of Digits from CO:

10

 DNIS[Edit DNIS Map](#) DID[Edit DID Range](#) Extension Translation Table:

<None> ▼

 Prepend Dial In Prefix: Use Site Extension Prefix Tandem Trunking

User Group:

Intelepeer User Group ▼

Prepend Dial In Prefix:

Destination:

7777 : Auto Attendent

[Search](#) Outbound:

Network Call Routing:

Access Code:

9

Local Area Code:

347

Additional Local Area Codes:

[Edit](#)

Nearby Area Codes:

[Edit](#)

Billing Telephone Number:

+1 (347) 542-5760 (e.g. +1 (408) 331-3300)

Trunk Services:

 Local

Figure 3: Trunk group

20. Set **Billing Telephone Number**: "+1(347)542-5760" is used for this example.
21. Set **Trunk Services**:
22. Set **Local**: Checked.
23. Set **Long Distance**: Checked.
24. Set **International**: Checked.

25. Set **Enable Original Caller Information**: Checked.
26. Set **n11 (e.g. 411, 611, except 911 which is specified below)**: Checked
27. Set **Emergency(e.g. 911)** : Checked.
28. Set **Easily Recognizable Codes (ERC) (e.g. 800, 888, 900)**: Checked.
29. Set **Explicit Carrier Selection (e.g. 1010xxx)**: Checked.
30. Set **Operator Assisted(e.g. 0+)**: Checked.
31. Set **Caller ID not blocked by default**: Checked.
32. Confirm **Enable Caller ID**: Unchecked.
33. Set **Trunk Digit** Manipulation:
34. Set **Remove leading 1 from 1+10D**: Unchecked.
35. Confirm **Remove leading 1 for Local Area Codes**: Unchecked.
36. Confirm **Dial 7 digits for Local Area Code**: Checked.
37. Confirm **Dial in E.164 Format**: Unchecked.
38. Set **Local Prefixes**: None.
39. Set **Translation Table**: None.
40. Click **Save**.

(977) 572-9100 (e.g. T1 (400) 331-3300)

Trunk Services:

- Local
- Long Distance
- International
- Enable Original Caller Information
- n11 (e.g. 411, 611, except 911 which is specified below)
- Emergency (e.g. 911)
- Easily Recognizable Codes (ERC) (e.g. 800, 888, 900)
- Explicit Carrier Selection (e.g. 1010xxx)
- Operator Assisted (e.g. 0+)
- Caller ID not blocked by default
- Enable Caller ID (Please confirm with the Carrier(s) or the Service Provider(s) on how the end-to-end caller name is delivered. When Site Name is used for the Caller ID, overwrite it with:

Trunk Digit Manipulation:

- Remove leading 1 from 1+10D
Hint: Required for some long distance service providers.
- Remove leading 1 for Local Area Codes (for all prefixes unless a specific local prefix list is provided below)
Hint: Required for some local service providers with overlay area codes.
- Dial 7 digits for Local Area Code (for all prefixes unless a specific local prefix list is provided below)
Hint: Local prefixes required for some local service providers with mixed 7D and 1+10D in the same home area.
- Dial in E.164 Format

Local Prefixes: [Go to Local Prefixes List](#)

Prepend Dial Out Prefix:

Off System Extensions:

Translation Table:

Figure 4: Trunk group conti.

3.2 Create User Group

1. Navigate to **Users > User Groups > Add new**

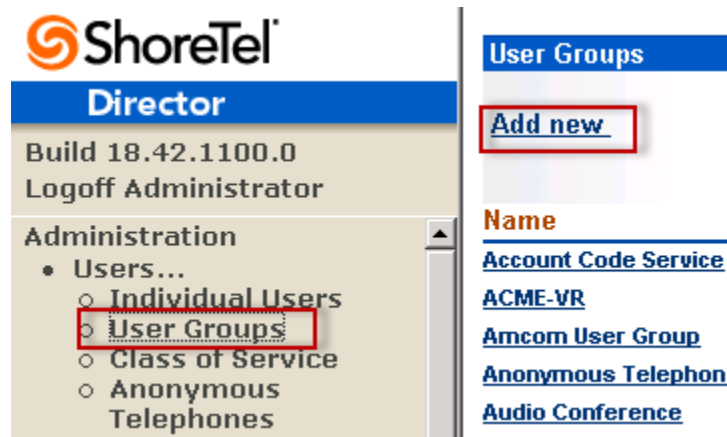


Figure 5 Create User Group

2. Set Name: Intelpeeer User group is used for this example.
3. Set **COS-Telephony**: Fully Featured is selected from the pull down menu.
4. Set **COS-Call Permissions**: No Restrictions is selected from the pull down menu.
5. Set **COS-Voice Mail**: Large Mail Box is selected from the pull down menu.
6. Check **Send Caller ID** as Caller's Emergency Service Identification(CESID)
7. Check **Send DID as Caller's Emergency Service Identification(CESID)**
8. Set **Account Code Collection**: Disabled is selected from the pull down menu.
9. Set **Outgoing Trunk Groups (access Codes)**: Intelpeeer Trunk Group(9) is checked.
10. Confirm all the rest of the options in "User Profile", "Phone Application", "Ring tone are: None which are set by default.

Figure 6 Create User Group cont.

3.3 Create Individual Trunks

1. Navigate to **Trunks > Individual Trunks**.
2. Set **Add new trunk at site**: Headquarters from the pull down menu.
3. Set in **trunk group**: Intelpeer Trunk Group from the pull down menu.
4. Click **Go**.

Figure 7 Individual Trunk

5. Set **Name**: is given for this example.
6. Select **Switch**: ShoreGear90 is selected for this example from the pull down menu.
7. Set **IP Address**: This is the IP address of LAN side of Ingate. Please use the actual IP address of Ingate for your network. The IP Address used in this configuration is 10.65.1.50. The Ingate IP address may/will be different from this example.
8. Select **Number of Trunks**: 3 is selected for this example.
9. Click **Save**.

Figure 8 Individual Trunk cont.

3.4 Add a New User

1. Navigate to **Users > Individual users**.
2. Set **Add a new user at site**. Headquarters is selected for this example.
3. Click **Go**.

First Name	Last Name	Site	User Group
Dave	Phone3	Headquarters	Intelepeer User Group
Intelepeer1	Phone1	Headquarters	Intelepeer User Group
ipc test	chennai	Headquarters	Executives

Figure 9 Add a New User

4. Set **First Name**: For this example Dave is given as first name.
5. Set **Last Name**: For this example Phone3 is used.
6. Set **Number**: This 4 digit extension number should be in the DID Range. For this example 5761 is used.
7. Set **License Type**: External and Mailbox is selected from the pull down menu.
8. Set **Access License**: personal is selected from the pull down menu.
9. Set **Caller ID**: The DID number which is assigned is given in the format as shown in figure.

10. Check **DID Range**.
11. Select the DID Range from the pull down menu.
12. Set **DID number**: This will be the number assigned to the phone. This number should be in the DID Range selected.
13. Set **PSTN Failover**: Select None from the pull down menu.
14. Set **user groups**: For this example Intelepeer is selected from the pull down menu.
15. Set **Primary Phone** Port:
16. Check **IP Phones**: Select the MAC address of the appropriate phone connecting to the user.

Users

Edit User

New

Copy

Save

Delete

Reset

▼ General

▶ Personal Options

▶ Distribution Lists

▶ Workgroups

[Refresh this](#)

First Name:	<input type="text" value="Dave"/>
Last Name:	<input type="text" value="Phone3"/>
Number:	<input type="text" value="5761"/>
License Type:	<input type="text" value="Extension and Mailbox"/>
Access License:	<input type="text" value="Personal"/> <input type="checkbox"/> Enable Contact Center Integration
Caller ID:	<input type="text" value="+1 (347) 542-5761"/> (e.g. +1 (408) 331-3300)
<input checked="" type="checkbox"/> DID Range:	<input type="text" value="+13475425761 (0 of 1 available) IntelepeerTrunk Group"/> View System Directory
DID Number:	<input type="text" value="+1 3475425761"/> (Range: +13475425761 - 13475425761)
PSTN Failover:	<input type="text" value="None"/>
User Group:	<input type="text" value="Intelepeer User Group"/> Go to this User Group
Site:	<input type="text" value="Headquarters"/>
Language:	<input type="text" value="English(US)"/>
Primary Phone Port:	<input checked="" type="radio"/> IP Phones <input type="text" value="00-10-49-07-04-CF"/> <input type="radio"/> Ports <input type="text" value="SG220T1A - 9"/> <input type="radio"/> SoftSwitch <input type="text" value="SoftSwitch"/>
Current Port:	<input type="text" value="00-10-49-07-04-CF"/> <input type="button" value="Go Primary Phone"/>
Jack #:	<input type="text"/>
Mailbox Server:	<input type="text" value="Headquarters"/> Escalation Profiles and Other Mailbox Options
<input checked="" type="checkbox"/> Accept Broadcast Messages	
<input checked="" type="checkbox"/> Include in System Dial By Name Directory	
<input type="checkbox"/> Make Number Private	
Fax Support:	<input type="text" value="User - Redirect"/>
Allow Video Calls:	<input type="text" value="None"/>
<input checked="" type="checkbox"/> Allow Telephony Presence	
<input type="checkbox"/> Shared Call Appearances	
Associated BCA:	<input type="text"/>

Figure 10 Add a New User cont.

1. Set **Client User ID**: dPhone3 is given for this example.
2. Set **Email Address**: dphone3@tekvision.com is given for this example
3. Click **Save**.

4 InGate Configuration

4.1 Select Model

1. Click **Start up Tool TG Tool**.
2. Set **Please Select Model**: Ingate Firewall/SIParator is selected.
3. Click **Next**.

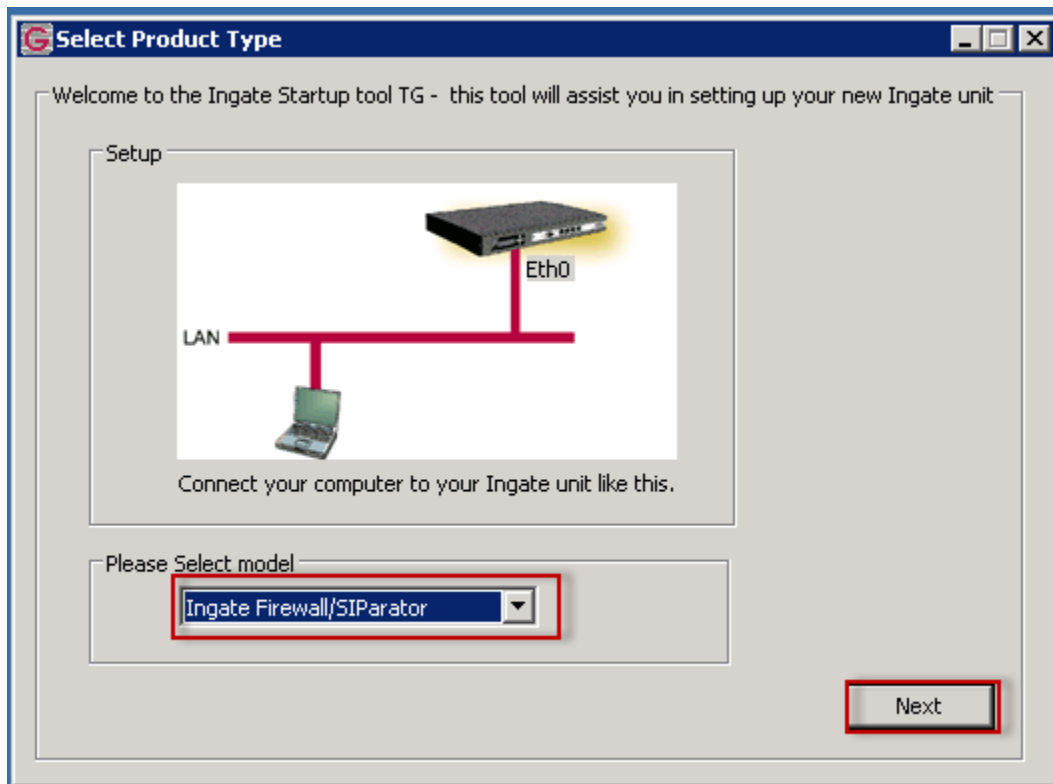


Figure 11 Select Model

4.2 Connect Ingate

1. Under **First select what you would like to do**:
2. Check **Change or update configuration of the unit**.
3. Check **Configure SIP trunking**.
4. Under **Establish contact**:
5. Set **IP Address**: This is the IP address of LAN side of Ingate. Please use the actual IP address of Ingate for your network. The IP Address used in this configuration is 10.65.1.50. The Ingate address may/will be different from this example.
6. Enter **Password**: *****.
7. Click **Contact**.

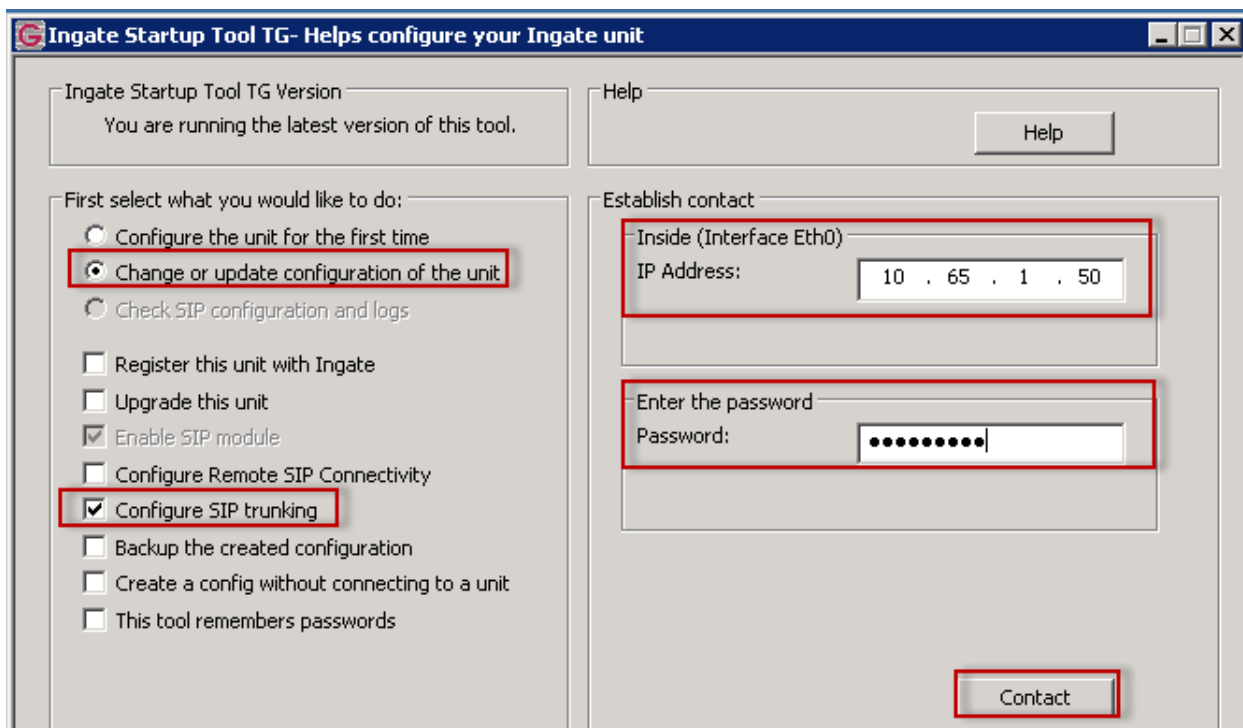


Figure 12 Connect Ingate

4.3 Network Topology

1. Set **Product Type**: Standalone SIParator is selected from the pull down menu.
2. Under **Inside(Interface Eth0)**:
3. Set **IP address**: This is the IP address of LAN side of Ingate. Please use the actual IP address of Ingate for your network. The IP Address used in this configuration is 10.65.1.50. The Ingate address may/will be different from this example.
4. Set **Netmask**: This is the Netmask for the IP address of "Inside (Interface Eth0)" entered in Network Topology
5. Under **Outside (Interface Eth1)**:
6. Confirm **Use DHCP to obtain IP**: Unchecked
7. Set **IP Address**: Set IP address: This is the IP address of WAN side of Ingate. Please use the actual IP address of Ingate for your network. The Ingate WAN address may/will be different from this example.
8. Set **Netmask**: This is the Netmask for the IP address of Outside (Interface Eth1) entered in Network Topology.
9. Confirm **Allow https access to web interface from Internet** is Unchecked.
10. Set **Gateway**: Please use the actual Gateway IP address for your network. IP address may/will be different from this example.
11. Set **DNS Server(Primary)**: The primary DNS Server used here is 10.64.1.3. The IP address may/will be different from this example.

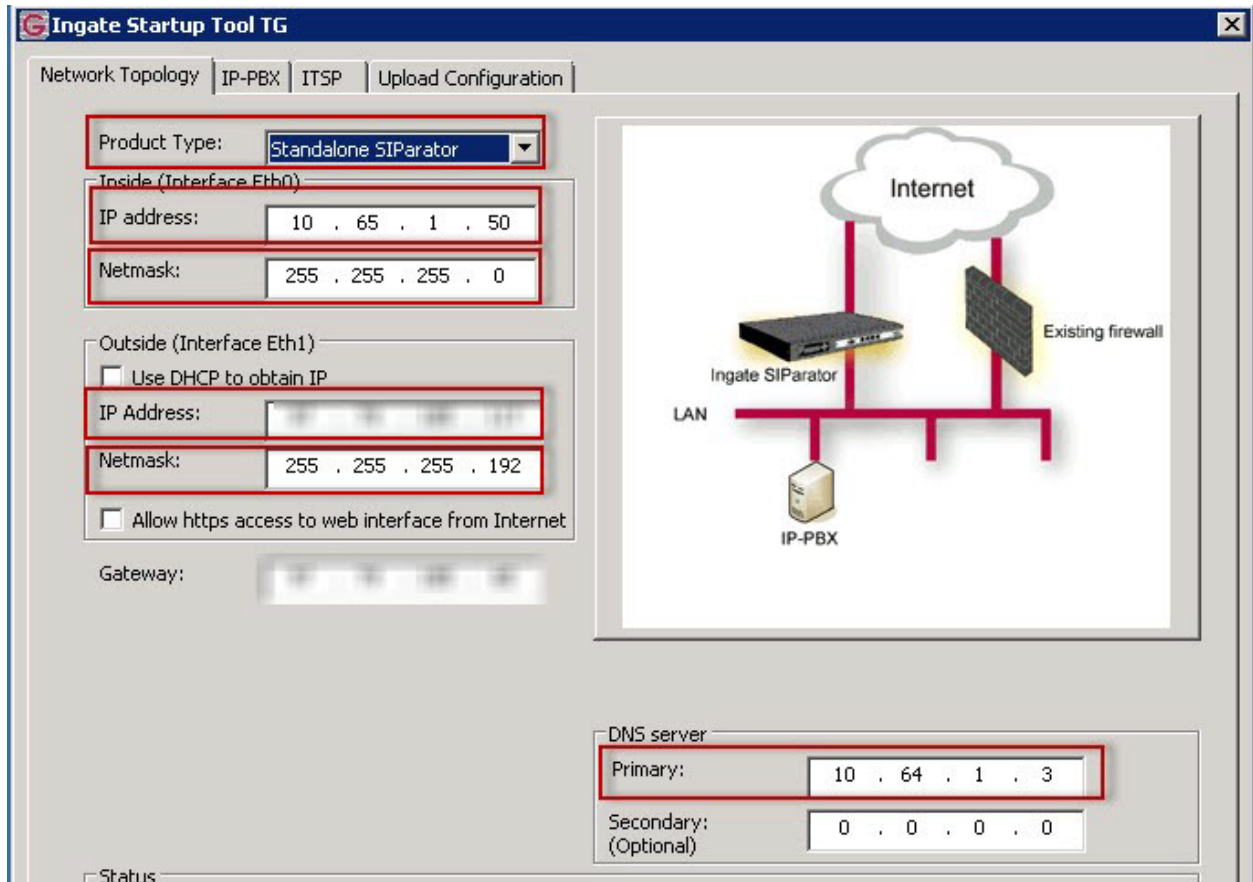


Figure 13 Network Topology

4.4 IP-PBX

1. Set **Type:** ShoreTel ShoreGear.
2. Set **IP Address:** Please the actual IP address of Shoretel PBX for your network. The IP Address used in this configuration is 10.65.1.4. The PBX IP address may/will be different from this example.
3. Conform **Use Domain Name** is Unchecked.

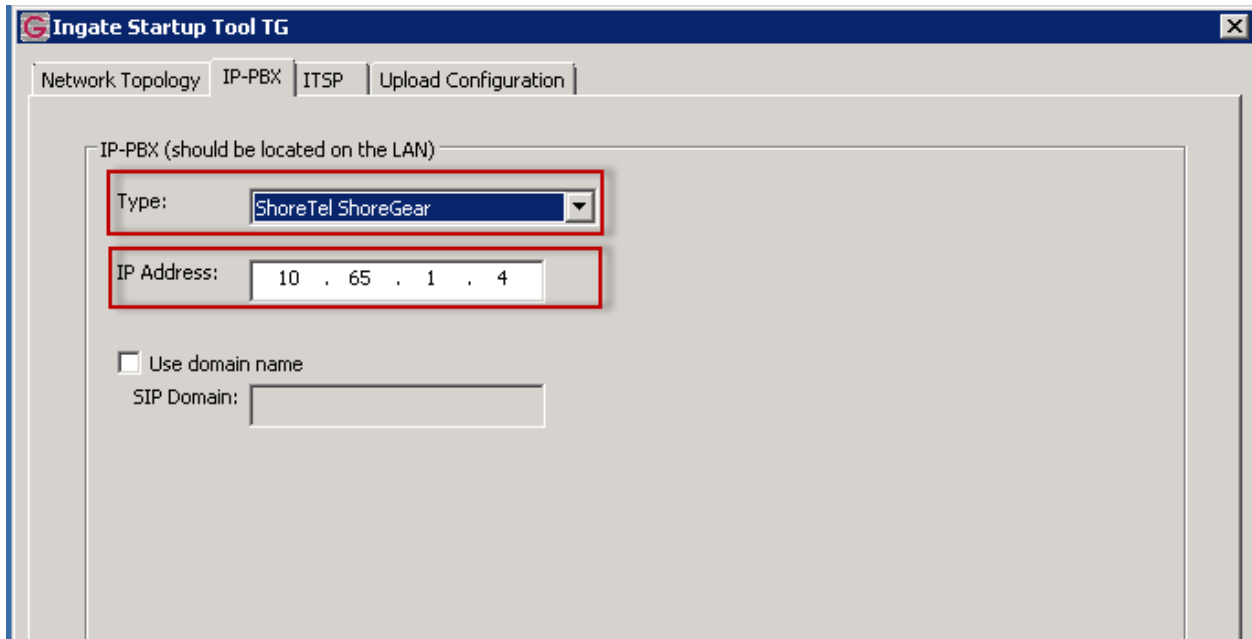


Figure 14 IP-PBX

4.5 Service Provider

1. Select **Name**: Generic(register main)
2. Under **Provider Address**
3. Set **IP Address**: Please use the actual IP address of signaling IP that is provided by the service provider. The IP address may/will be different from this example
4. Confirm **Use domain name** is Unchecked.
5. Uncheck **Authentication**.

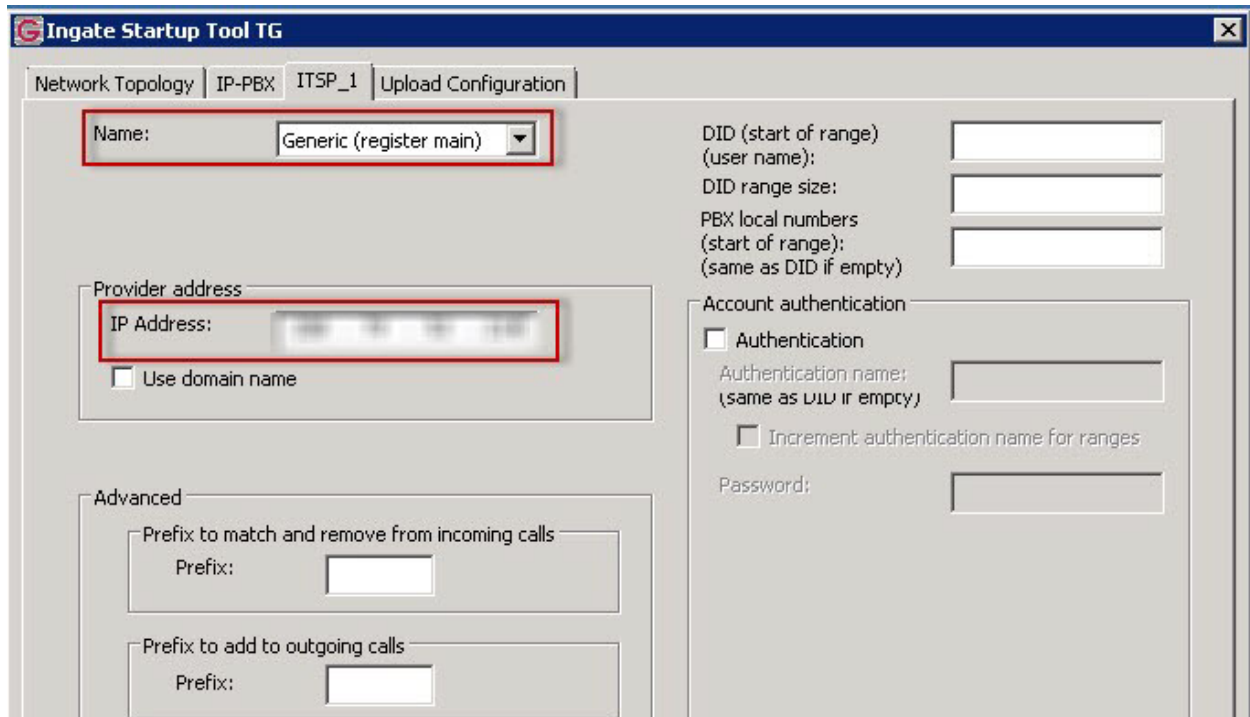


Figure 15 Service Provider

4.6 Upload Configuration

1. Under **Verbose Logging**(SIP debug):gs
2. Check **Enable**.
3. Under **Final step**:
4. Check **Logon to web GUI and apply settings**.
5. Click **Upload**.

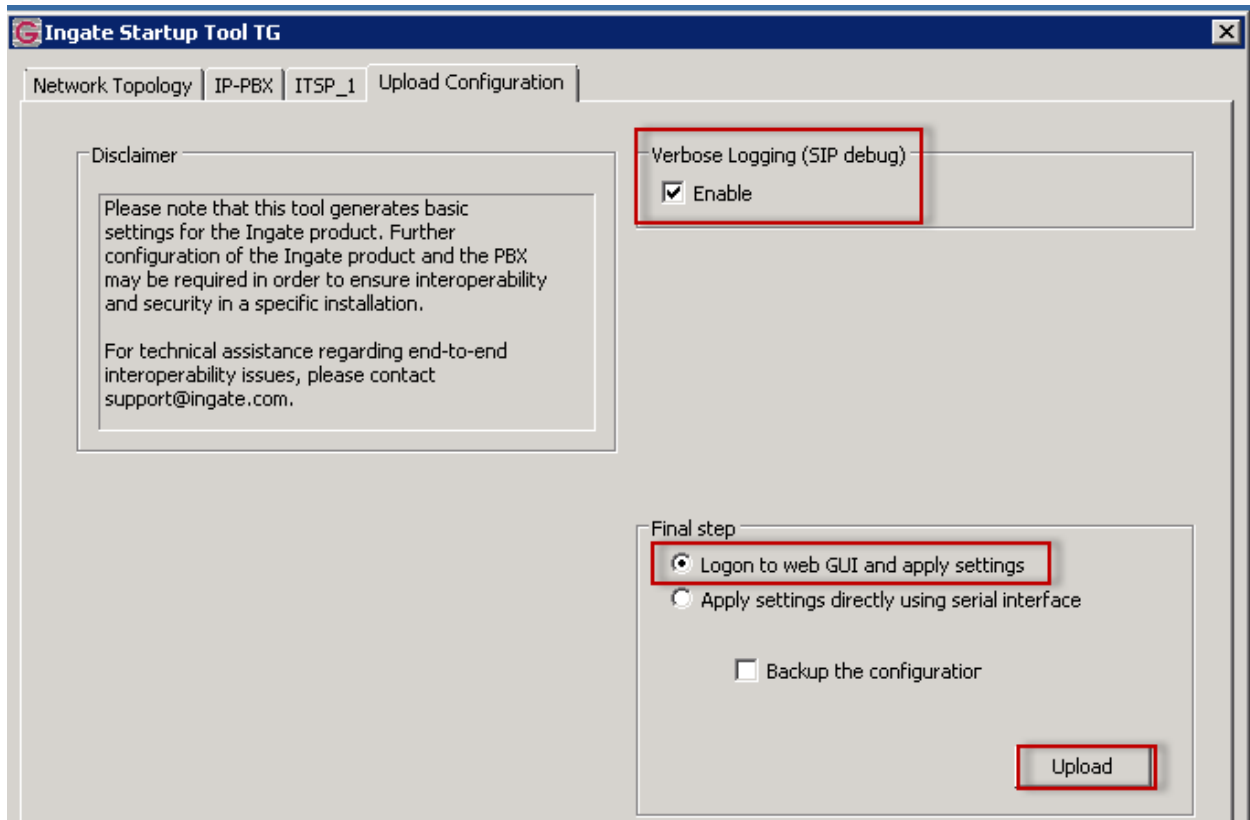


Figure 16 Upload Configuration

4.7 SIP Trunk

1. In Ingate Web GUI:
2. Navigate to **SIP Trunks > Trunk1**.
3. Under **SIP Trunk1**.
4. Check **Enable SIP Trunk**.
5. Under **SIP Trucking Service**:
6. Check **Define SIP Trunk Parameters**.
7. Set **Service name**: Generic (Register main).
6. Set **Service Provider Domain**: Signaling IP address provided by Service Provider. Please use the actual IP address. The IP Address used in this configuration is 208.79.53.214. The IP address may/will be different from this example.
8. Set **Restrict to calls from**: WAN.
9. Set **From header domain**:
10. Check **as entered**.
11. Set **From Domain**: Please use the actual IP address of Ingate WAN for your network. The Ingate WAN IP address may/will be different from this example.

- Administration
- Basic Configuration
- Network
- SIP Services
- SIP Traffic
- SIP Trunks
- Failover
- Virtual Private Networks
- Quality of Service
- Logging and Tools
- About

View trunk: SIP Trunk 1: Generic (register main);ShoreTel ShoreGear [Goto SIP Trunk page](#)

SIP Trunk 1 [\(Help\)](#)

- Enable SIP Trunk
- Disable SIP Trunk

SIP Trunking Service [\(Help\)](#)

- Use parameters from other SIP trunk
- Define SIP trunk parameters

Service name:	<input type="text" value="Generic (register main)"/>	<i>(Descriptive name)</i>
Service Provider Domain:	<input type="text" value="192.168.1.100"/>	<i>(FQDN or IP address)</i>
Restrict to calls from:	<input type="text" value="WAN"/>	<i>('.' = No restriction)</i>
Outbound Proxy:	<input type="text"/>	<i>(FQDN or IP address)</i>
Use alias IP address:	<input type="text" value="-"/>	<i>(Forces this source address from our side)</i>
Outbound Gateway:	<input type="text" value="-"/>	<i>('.' = Use Default Gateway)</i>
Signaling Transport:	<input type="text" value="-"/>	<i>('.' = Automatic)</i>
Port number:	<input type="text"/>	
From header domain:	<input type="radio"/> Provider domain <input type="radio"/> Enterprise domain <input type="radio"/> External IP address <input checked="" type="radio"/> as entered:	
From domain:	<input type="text" value="192.168.1.100"/>	

Figure 17 SIP Trunk

Host name in Request-URI of incoming calls: (Trunk ID - Domain name)

Remote Trunk Group Parameters (RFC 4904):

Used as: ('-' = Don't use TGP)

Local Trunk Group Parameters (RFC 4904):

Used as: ('-' = Don't use TGP)

Preserve Max-Forwards: No

Relay media for remote users: No

Exactly one Via header: No

'gn' registration (RFC 6140): No

Hide Record-Route: No

Show only one To tag: No

SIP 3xx redirection to provider domain: No

SIP 3xx redirection to caller domain: No

Route incoming based on: Request-URI

Service Provider domain is trusted: No (For P-Asserted-Identity)

Use P-Preferred-Identity: No (Instead of P-Asserted-Identity)

Max simultaneous calls: (Call Admission Control)

Max simultaneous calls per Trunk Line:

Main Trunk Line (Help)

No.	Reg	Outgoing Calls			Authentication		Incoming Calls	
		Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to
1	Yes		NA			Change Password	(*)	\$1

PBX Lines (Help)

No.	Reg	Outgoing Calls			Authentication		Incoming Calls		Delete Row
		From PBX Number/User	Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	
1	No	anonymous		anonymous@anonymou		Change Password			<input type="checkbox"/>
2	No	(*)		\$1		Change Password	(*)	\$1	<input type="checkbox"/>

Figure 18: SIP Trunk conti..

Add new rows rows.

SIP Lines [\(Help\)](#)

No.	Reg	Outgoing Calls				Authentication		Incoming Calls		Delete Row
		From SIP Number/User	Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to SIP Account	

Add new rows rows.

Setup for the PBX [\(Help\)](#)

- Use PBX from other SIP trunk
- Define PBX settings

PBX Name: *(Descriptive name)*

Use alias IP address: *(Forces this source address from our side)*

PBX Registration SIP Address	Authentication		PBX IP Address		PBX Domain Name
	User ID	Password	DNS Name or IP Address	IP Address	
<input type="text"/>	<input type="text"/>	<input type="button" value="Change Password"/>	<input type="text" value="10.65.1.4"/>	<input type="text" value="10.65.1.4"/>	<input type="text"/>

(At least one of PBX Registration, IP address or Domain Name is required to locate the PBX)

PBX Network:

Signaling transport: *('-' = Automatic)*

Port number:

Match From Number/User in field:

- To header field:
- Same as Request-URI
 - Copy from Trunk
 - Initial Request-URI
 - as entered.

Remote Trunk Group Parameters usage: *('-' = Don't use TGP)*

Local Trunk Group Parameters usage: *('-' = Don't use TGP)*

Figure 19: SIP Trunk

The End.