Nexmo SIP Trunking Configuration Guide

ShoreTel Director 14.2 and InGate SIParator 5.0.11

July 2017
1 Audience

This document is intended for the SIP trunk customer’s technical staff and Value Added Retailer (VAR) having installation and operational responsibilities. This configuration guide provides steps for configuring ShoreTel Director 14.2 and InGate SIParator 5.0.11 to Nexmo SIP Trunking services.

2 SIP Trunking Network Components

The network for the SIP trunk reference configuration shown below is representative of a ShoreTel 14.2 Director and an InGate SIParator configuration to Nexmo SIP trunking.

Figure 1: Topology Diagram

2.1 Hardware Components

- ShoreTel Director 14.2
- ShoreTel ShoreGear 90
2.2 Software Requirements

- ShoreTel Director, Version 14.2, Build 19.48.2600.0
- InGate SIParator, Version: 5.0.11

3 Features

3.1.1 Features Supported

- Incoming and outgoing off-net calls using G711ULAW & G711ALAW voice codecs
- Calling Line (number) Identification Presentation
- Call Hold and Resume
- Call Transfer (unattended and attended)
- Call Conference
- Call Forward (all, no answer)
- DTMF relay both directions (RFC2833)
- Media flow-through on InGate SIParator

3.1.2 Features Not Supported by PBX

- None

3.1.3 Features Not Tested

- None

3.1.4 Caveats and Limitations

- International calls were responded to with a 404 Not Found message from Nexmo
4 Configuration

4.1 IP Address Worksheet

The specific values listed in the table below and in subsequent sections used in the lab configuration described in this document, and are for **illustrative purposes only**. The customer must obtain and use the values for your deployment.

<table>
<thead>
<tr>
<th>Component</th>
<th>Lab Value</th>
<th>Customer Value</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>InGate SIParator</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>LAN IP Address</td>
<td>10.65.1.200</td>
<td></td>
</tr>
<tr>
<td>LAN Subnet Mask</td>
<td>255.255.255.0</td>
<td></td>
</tr>
<tr>
<td>WAN IP Address</td>
<td>192.65.79.XXX</td>
<td></td>
</tr>
<tr>
<td>WAN Subnet Mask</td>
<td>255.255.255.128</td>
<td></td>
</tr>
<tr>
<td><strong>ShoreTel 14.2 Director IP</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>System IP Address</td>
<td>10.65.1.9</td>
<td></td>
</tr>
</tbody>
</table>
5 ShoreTel Configuration: ShoreWare Director 14.2

![Figure 2: ShoreTel configuration](image-url)
5.1 ShoreTel Configuration: Configure Site

1. Navigate to Administration → Sites
2. Choose site location in drop-down
3. Click Go
1. Set **Name**: Headquarters is used for this example.
2. Set **Local Area Code**
3. Set **Time Zone**
4. Set **Bandwidth**
5. Set **Codecs**
6. Set **Proxy Switch**
7. Click **Save**

![Figure 4: ShoreTel Configuration: Configure Site – Cont.](image-url)
5.2 ShoreTel Configuration: Codec Lists

1. Navigate to Administration → Call Control → Codec Lists

2. Move the Codec you want to use to the top of the list

3. Click Save

Figure 5: ShoreTel Configuration: Configuring Codecs

Figure 6: ShoreTel Configuration: Configuring Codecs – Cont.
5.3 ShoreTel Configuration: SIP Profiles

1. Navigate to **Trunks → SIP Profiles**
2. Click **New**

Click on **AT&T (ShoreTel default SIP Profile)**

Click **Copy** at the top of the page and rename the SIP Profile

Change **System Parameters** in the **Custom Parameters** box as needed

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**Figure 7: ShoreTel Configuration: Creating SIP Profile**

**Figure 8: ShoreTel Configuration: Creating SIP Profile – Cont.**
5.4 ShoreTel Configuration: Configure Trunk Groups

1. Navigate to Trunks → Trunk Groups
2. After adding site and of type, click Go

3. Add Name
4. Choose Profile (created in previous step)
5. You will need to navigate back to this page and select the User Group created in the next step
6. Add Access Code
7. Add Local Area Code for outbound calls to the PSTN

NOTE: For more than one local area code click Edit under Outbound → Additional Local Area
8. Codes and enter the additional area codes in the dialog box then click OK
9. Other selections will depend on user needs
10. Click Save
11. To edit DNIS or DID, click on the appropriate tab
Figure 10: ShoreTel Configuration: Configuring Trunk Group – Cont.
12. Click on **Edit DID Range**
13. Enter **Base Phone Number**
14. Enter **# Phone Numbers**
15. Click **Add this record**
16. Click **Save**
5.5 ShoreTel Configuration: Create User Groups

1. Navigate to Users → User Groups
2. Click Add new

3. Add Name
4. Choose COS selections in the drop-down boxes
5. Choose the Outgoing Trunk Group(s) you created in the previous step
6. Click Save

![Figure 13: ShoreTel Configuration: Configuring User Group](image)

![Figure 14: ShoreTel Configuration: Configuring User Group – Cont.](image)
5.6 ShoreTel Configuration: Create Individual Trunks

1. Navigate to **Trunks → Individual Trunks**
2. In the drop-down boxes at the top of the page, choose **site** and the **trunk group** you created
3. Click **Go**

![Figure 15: ShoreTel Configuration: Creating Individual Trunks](image)

4. Add **Name**
5. Choose **Switch**
6. Enter **IP Address** (IP address of the PBX to which you are connecting. In this test, Ingate was used as a SBC so trunks were created between ShoreTel and Ingate.)
7. Enter **Number of Trunks** (Number of trunks is determined by user needs. Each trunk requires a license.)
8. Click **Save**
9. Follow this process for each individual trunk in the trunk group

![Figure 16: ShoreTel Configuration: Creating Individual Trunks – Cont.](image)

5.7 ShoreTel Configuration: Create Individual Users

1. Navigate to **Users → Individual Users**
2. Choose your site in the drop-down box at the top of the page and click Go.

3. Enter First Name
4. Enter Last Name
5. Under License Type choose to create the extension with or without a voicemail box
6. Choose the DID Range you created in Trunk Groups
7. Choose your User Group
8. Choose Site
<table>
<thead>
<tr>
<th>Mailbox Server:</th>
<th>Headquarters</th>
<th>Escalation Profiles and Other Mailbox Options:</th>
</tr>
</thead>
<tbody>
<tr>
<td>☑ Accept Broadcast Messages</td>
<td></td>
<td></td>
</tr>
<tr>
<td>☑ Include In System Dial By Name Directory</td>
<td></td>
<td></td>
</tr>
<tr>
<td>☐ Make Number Private</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Fax Support:</td>
<td>E-mail - Redirect</td>
<td></td>
</tr>
<tr>
<td>Allow Video Calls:</td>
<td>E-mail</td>
<td></td>
</tr>
<tr>
<td>☑ Allow Telephony Presence</td>
<td></td>
<td></td>
</tr>
<tr>
<td>☐ Shared Call Appearances</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Associated BCA:</td>
<td></td>
<td></td>
</tr>
<tr>
<td>☐ Allow Use of Soft Phone</td>
<td></td>
<td></td>
</tr>
<tr>
<td>☐ Allow Phone API</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Mobility Options:**

| ☑ Allow Mobile Access |             |                                               |
| ☐ Allow Enhanced Mobility with Extension |             |                                               |

| ☑ Delayed Ringdown |             |                                               |
| ☑ Extension: |             | Screen                                       |
| ☑ External Number: |             | (e.g. 9×1 (408) 313-3300)                    |
| Ringdown Delay: |             | sec                                           |
| Client Username: |             | unseven                                      |
| Client Password: |             | *******                                      |
| Voice Mail Password: |             | *******                                      |
| ☐ Must Change On Next Login |             |                                               |
| SIP Password: |             | *******                                      |
| Email Address: |             | unseven@technion.com                          |
| Conferencing Settings: |             |                                               |
| Appliance: |             | <None>                                       |
| Instant Messaging Settings: |             |                                               |
| Server / Appliance: |             | <None>                                       |
| Edit System Directory Record |             |                                               |

**Figure 19: ShoreTel Configuration: Creating Individual Users – Cont.**
6  Ingate Configuration

6.1  Ingate Configuration: Ingate Startup Tool

To launch Ingate:

1. Download and click on the icon
2. The Select Product Type window appears
3. Select your Ingate Model from the drop-down box
4. Click Next

![Select Product Type window](image)

Figure 20: Ingate Configuration: Ingate Startup Tool

5. The Ingate Startup Tool window appears.
6. Enter IP Address
7. Enter Password
8. Click Contact

![Ingate Startup Tool](image)

Figure 21: Ingate Configuration: Ingate Startup Tool – Cont.
9. Enter the correct **Network Topology** information

![Figure 22: Ingate Configuration: Ingate Startup Tool – Cont.](image)

10. Select the **IP-PBX** tab
11. Choose **Type**
12. Enter **IP Address**

![Figure 23: Ingate Configuration: Ingate Startup Tool – Cont.](image)
13. Select the **ITSP_1** tab
14. Choose **Name**
15. Enter **IP Address**

![Network Topology | IP-PBX | ITSP_1 | Upload Configuration](image)

**Please note that this tool generates basic settings for the Ingate product. Further configuration of the Ingate product and the PBX may be required in order to ensure interoperability and security in a specific installation.**

For technical assistance regarding end-to-end interoperability issues, please contact support@ingate.com.

**Figure 24: Ingate Configuration: Ingate Startup Tool – Cont.**

16. Select the **Upload Configuration** tab
17. Click **Upload**

![Network Topology | IP-PBX | ITSP_1 | Upload Configuration](image)

**Figure 25: Ingate Configuration: Ingate Startup Tool – Cont.**
6.2 **Ingate Configuration: Launching Ingate from a Browser**

1. [http://< IP Address>](http://< IP Address>)
2. Enter **Username**
3. Enter **Password**

![Ingate Configuration: Launching Ingate from a Browser](image)

*Figure 26: Ingate Configuration: Launching Ingate from a Browser*
6.3 Ingate Configuration: Basic Configuration

1. Add DNS Name or IP Address

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

![SIParator Configuration Interface]

**General**
- Name of this SIParator:
- Default domain:

**Version of Software SIParator/Firewall**
- Check for new versions of Software SIParator/Firewall:
- Date of last successful version check: Not available
- Software version in use: 5.0.11

**Policy For Ping To the SIParator**
- Never reply to ping
- Only reply to ping to the same interface
- Reply to ping to all IP addresses

**DNS Servers**

<table>
<thead>
<tr>
<th>No.</th>
<th>Dynamic</th>
<th>DNS Name or IP Address</th>
<th>IP Address</th>
<th>Delete Row</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>✔️</td>
<td>8.8.8.8</td>
<td>8.8.8.8</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>✔️</td>
<td>10.85.0.12</td>
<td>10.85.0.12</td>
<td></td>
</tr>
</tbody>
</table>

Add new rows: 1 rows.
6.4 Ingate Configuration: Dial Plan

1. Configure Dial Plan as necessary
2. SIP Traffic > Dial Plan

![Ingate Configuration: Dial Plan](image)

**Figure 28: Ingate Configuration: Dial Plan**

![Ingate Configuration: Dial Plan – Cont.](image)

**Figure 29: Ingate Configuration: Dial Plan – Cont.**

![Ingate Configuration: Dial Plan (3/3)](image)

**Figure 30 “Ingate Configuration: Dial Plan (3/3)”**
6.5 Ingate Configuration: SIP Trunks

1. Click **Go to SIP Trunk page**
2. Select **WAN** in the dropdown box next to **Restrict to calls from**:
3. Add the correct domain names

![Figure 31: Ingate Configuration: SIP Trunks](image)

![Figure 32: Ingate Configuration: SIP Trunks – Cont.](image)
Figure 33: Ingate Configuration: SIP Trunks – Cont.

Setup for the PBX

- Use PBX from other SIP trunk
- Define PBX settings

PBX Name: ShoreTel ShoreGear (Unique descriptive name)
Use alias IP address: [ ] (Forces this source address from our side)

<table>
<thead>
<tr>
<th>PBX Registration SIP Address</th>
<th>Authentication</th>
<th>PBX IP Address</th>
<th>PBX Domain Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>User ID</td>
<td>Password</td>
<td>DNS Name or IP Address</td>
<td>IP Address</td>
</tr>
<tr>
<td>Change Password</td>
<td>10.65.1.9</td>
<td>10.65.1.9</td>
<td></td>
</tr>
</tbody>
</table>

PBX Network: ShoreTel ShoreGear

Signaling transport: [ ] (Default = Automatic)
Port number:
Match From Number/User in field: From URI

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

Remote Trunk Group Parameters usage: [ ] (Default = Don’t use TGP)
Local Trunk Group Parameters usage: [ ] (Default = Don’t use TGP)

Figure 34 *Ingate Configuration: SIP Trunks – Cont.*
6.6 Configure Numbers in Nexmo Account

1. Login to the Nexmo account using the credentials provided at the time of registration. A Key and Secret is displayed on the dashboard and can be use as the username and password for Registration SIP Trunks.

![Nexmo Dashboard](image1)

Figure 35: Nexmo Dashboard

2. In order to provide the URL to which the call has to be routed from Nexmo, navigate to the Numbers tab

3. Click Edit against each number as shown below

![Nexmo DID Numbers](image2)

Figure 36: Nexmo DID Numbers
4. A pop-up will be displayed
5. Select **Forward to** and provide the URL to which the calls route
6. Click **Update** to save the changes

![Settings for 12014647035](image)

*Figure 37: Your Numbers – Cont.*

7 **Summary of Tests and Results**

*N/S = Not Supported   N/T= Not Tested   N/A= Not Applicable*
<table>
<thead>
<tr>
<th>Test Case #</th>
<th>Test Case Description</th>
<th>Result</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Calling Party Disconnects Before Answer</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>Calling Party Disconnects After Answer</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>Called Party Disconnects After Answer</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>Three Way Calling</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>5</td>
<td>Calling Party Presentation Restricted</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>6</td>
<td>Calling Party Disconnect Before Answer</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>7</td>
<td>Calling Party Disconnects after Answer</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>8</td>
<td>Called Party Disconnects after Answer</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>9</td>
<td>Calling Party Receives Busy</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>10</td>
<td>International Outbound Dialing</td>
<td>Failed</td>
<td>Nexmo responds to the Invite with a 404 not found</td>
</tr>
<tr>
<td>11</td>
<td>Outbound Call Forward Always</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>12</td>
<td>Outbound Call Forward Not Available (Ring No Answer)</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>13</td>
<td>Outbound Consultative Call Transfer</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>14</td>
<td>Outbound Semi-Attended/Blind Call Transfer</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>15</td>
<td>Outbound Call Hold</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>16</td>
<td>Terminate Early Media Outbound Call Before Answer</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>17</td>
<td>Early Media Forward Call</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>18</td>
<td>Outbound, Wait for Session Audit</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>19</td>
<td>Inbound, Wait for Session Audit</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>20</td>
<td>Outbound DTMF (RTPevent)</td>
<td>PASS</td>
<td></td>
</tr>
<tr>
<td>21</td>
<td>Inbound DTMF(RTPevent)</td>
<td>PASS</td>
<td></td>
</tr>
</tbody>
</table>