

**Configuring the Synapse® SB67070 SIP Gateway from AT&T  
for Clearfly SIP Trunking**

**January 2013**

## 1 Introduction

This guide was created to assist Synapse partners with configuring the Synapse SB67070 SIP Gateway with Clearly's SIP trunking services. It provides example data for the required fields. The actual account settings are provided by Clearly when service is activated. Questions about software and hardware installation or other PBX configuration issues should be directed to AT&T Synapse technical support at 1-888-916-2007 or refer to the AT&T Synapse Administration Guide on our website <http://smbtelephones.att.com>.

## 2 System and Network Components Required for SIP Trunking Service

- An AT&T Synapse SB67070 SIP Gateway.
- Internet connection with at least 100kb/s of download and upload bandwidth per line of service.
- Synapse firmware version 3.0 or higher installed on all Synapse devices.
- A Clearly SIP trunk account.

**Note:** AT&T recommends all hardware be updated to the latest software version available, and all information be gathered before programming.

## 3 Limitations and Known Issues

- SIP trunk faxing is not supported. We recommend if you require fax support to use an analog line or other efax services. For complete fax setup with analog lines, please refer to the AT&T Synapse Administration Guide found on our website.
- Static registration is not supported at this time.
- During an incoming call if a Synapse phone user hangs up before the originating caller hangs up, the line will still be active and in use. Until Clearly receives a disconnect signal from the originating caller or 5 minutes pass without the originating caller hanging up the line will be active and in use. In most cases the originating caller will hang up within seconds of the Synapse users. Clearly also blocks any kind of robo-calling where the far end may not hang up.

For other limitations of the AT&T Synapse SB67070 SIP Gateway, please refer to the Synapse Administration Guide on our website.

## 4 Clearfly Account Setup

When setting up an account with Clearfly, please request a registered or dynamic account when speaking to a Clearfly representative. This type of account will allow you to register at any location and with any IP address and requires a username and password, which Clearfly will generate for you.

After the account has been set up with the number of lines and DIDs needed, Clearfly will send you the account information, which will look something like this:

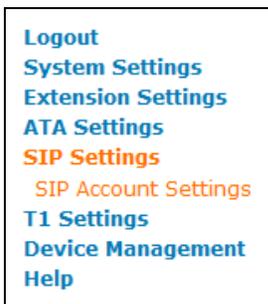
- Clearfly SIP Server/Domain: sc.cfly.co
- Username: 7823333333
- Password: md15quam
- Transport: UDP/5060
- Codec: G.711u
- RTP Payload Type: 101
- DTMF Relay: RFC2833
- Digits Transferred: Default is 10.
- DIDs: 782.333.3333–782.333.3340

The SIP Server/Domain, Username, Password, and DIDs are the only settings we will need to set up the Clearfly account in the AT&T Synapse SB67070 SIP Gateway. The other information is not needed because the gateway already knows which settings to use, and there are no fields to enter this data.

### 4.1 Configuring the SIP Account

To configure the SIP Account:

1. Log in to a Synapse device on the network.
2. Navigate to the **SIP Settings** page.



3. Enter the **Basic Settings**:
  - a. Enable the account and fill out all fields except **Primary Number**.
  - b. Enter a name for this SIP account in the **Account Name** field. The account name is not provided by Clearfly; you just make one up.

- c. Enter the max number of lines you ordered with your account in the **Max Calls** field.
- d. Enter the caller ID name you wish you transmit in the **Display Name** field.
- e. Enter the username Clearfly gave you in the **User Name** and **Auth User Name** field. Both fields need to have the Clearfly Username in them.
- f. Enter the password Clearfly gave you in the **Auth User Password** field.

**Basic Settings:**

Account Enabled:  Disabled  Enabled

Account Name:

Max Calls:

Display Name:

User Name:

Auth User Name:

Auth User Password:

Primary Number:

- 4. Leave the **Registration Settings** at the default values.

**Registration Settings:**

Registration Type:  Static Registration  Dynamic Registration

Registration Expires:

Registration Status:

- 5. Under **Server Settings**, enter the SIP Server/Domain that Clearfly gave you into the SIP Server **Address or URL** field.

**Server Settings:**

**SIP Server:**

Address or Url:

Port:

**Registrar Server:**

Address or Url:

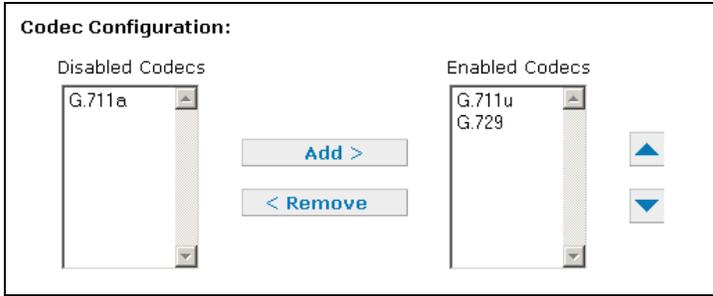
Port:

**Outbound Proxy Server:**

Address or Url:

Port:

- 6. Leave the **Codec Configuration** at default.



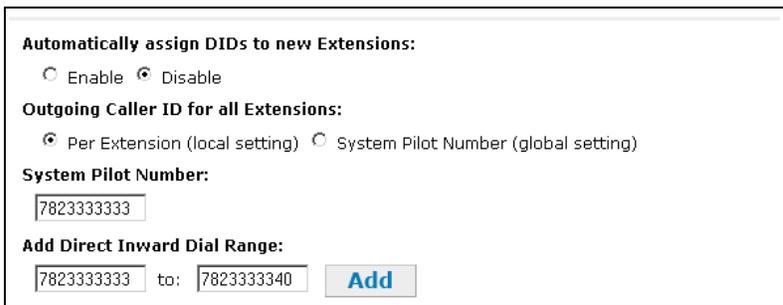
7. Click **Apply** at the bottom of the page to save the settings and activate the account. The page will refresh and the **Registration Status** should change to **Registered**.

## 4.2 DID Configuration

1. Navigate to the **Direct Inward Dial** page.



2. Enter your primary number into the **System Pilot Number** field and add the range of DIDs that Clearly provided to the **Add Direct Inward Dial Range** fields.



3. Click **Add** next to the DID range for each separate range, and then click **Apply** at the bottom of the page to save all the settings.
4. Next, navigate to the **DID Assignments** page to point all DIDs to whatever locations in the system you wish.

For more information about DID setup, please refer to the AT&T Synapse Administration Guide on our website.

## 5 Troubleshooting

Symptom	Suggested Action
<p><b>Registration Status</b> does not change to <b>Registered</b>.</p>	<p>Check to make sure you can ping the SIP server from a PC on the same network as the phones. If you can ping the server, but are still not registering, please verify your account information. If the account information is correct you may need to set up port forwarding for SIP and RTP. SIP runs on port 5060 and should be forwarded to the AT&amp;T Synapse SB67070 SIP Gateway. RTP runs on ports 10000–20000, which should also be forwarded to the AT&amp;T Synapse SB67070 SIP Gateway. If you are not sure how to port forward these ports on the router, please check the router’s manual or go to <a href="http://portforward.com/">http://portforward.com/</a>.</p> <p>If the ports are forwarded and you are still not registering, please contact Clearly at 1-866-652-7520 to verify your account information and to verify they can reach the AT&amp;T Synapse SB67070 SIP Gateway. You can also contact AT&amp;T Synapse Technical Support at 1-888-916-2007 for further assistance.</p>
<p>The Desksets display “All lines are busy.”</p>	<p>All the lines are in use at the time you are attempting to call out, or the gateway is not registered. If you know the lines are not in use, log in to the Synapse WebUI and check the status of the SIP account. If it is not registered, check to make sure you have Internet access by going to a web page on your PC. If you have Internet access and the account still will not register, please contact Clearly at 1-866-652-7520 to check for outages. If there are no outages or service issues, contact AT&amp;T Synapse Technical Support at 1-888-916-2007 for further assistance.</p>
<p>One way audio issues.</p>	<p>If you are experiencing one way audio on all calls, please port forward ports 5060 and 10000–20000 to the AT&amp;T Synapse SB67070 SIP Gateway. If you are not sure how to port forward these ports on the router, please check the routers manual or go to <a href="http://portforward.com/">http://portforward.com/</a>. If you are still experiencing one way audio after port forward, please contact Clearly at 1-866-652-7520 for further assistance.</p>
<p>DIDs are not routing properly or phones are not receiving calls.</p>	<p>If you are experiencing any issues with the DIDs routing properly, please verify that the DIDs are programmed without any dots or dashes in the numbers and that they are pointing to the correct location. If you are still experiencing DID issues, please contact Clearly at 1-866-652-7520 to check for any problems with the phone numbers.</p>