# Configuring a SIP Trunking Account on the Synapse® SB67070 SIP Gateway

Synapse supports up to ten SIP accounts per SIP gateway. There are two possible SIP account types: SIP trunking accounts and remote sites. It is possible to use any combination of SIP trunking accounts and remote sites, but the most any one gateway can have is ten. For example, a SIP gateway can have one SIP trunking account and nine remote sites.

This tech tip covers how to set up a new SIP trunking account on the Synapse SB67070 SIP gateway. Future tech tips will cover remote site configuration.

To configure a SIP trunking account from the provider:

1. Gather your account information from your SIP provider.

2. Log in to the WebUI of any Synapse device, then click on **SIP Settings**.
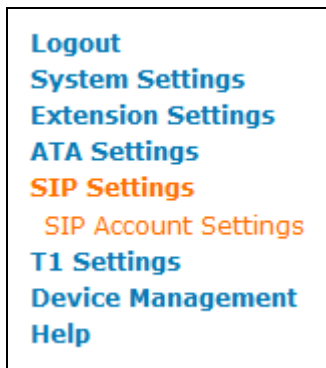
3. From **Select Account to Edit**, select **Create New Account**.

4. There are two types of accounts: SIP Trunking and Remote Site. For SIP account setup, select **SIP Trunking**.
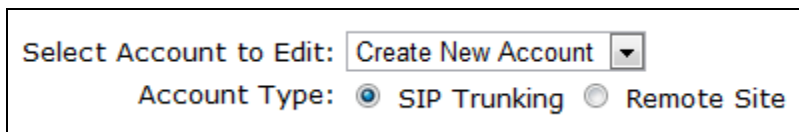
5. Under **Basic Settings**, select **Enable** to turn on the account after you apply the setting on this page. If you miss this step, the account information will be saved in Synapse, but the gateway will not try to register the SIP provider's server.

6. Next, assign all the basic settings for the account:

   — **Account Name**: The account name does not have to be what the SIP provider gave you and is only an identifier in Synapse.

- **Max Calls**: The maximum number of concurrent calls the SIP gateway is allowed to make. The default option of 16 is the maximum number of calls you can make at a time with the SIP gateway. As long as the number is equal to or greater than the number of lines the provider set up, it will work.
- **Display Name**: The name that will appear on the outgoing caller ID. This will only work if the SIP provider allows the PBX to modify the caller ID name. Most SIP providers do not allow this to happen or have controls for this in their own web-based account settings.
- **User Name** and **Auth User Name**: Enter the username the SIP provider gave you with the account. In almost all cases you will need to enter the SIP provider's username into both fields. If you have any questions, please refer to the specific SIP provider's interop guide on our website.
- **Auth User Password**: Enter the password that your provider has assigned to the SIP account. If you were not given a password, then you do not need to fill in this field.
- **Primary Number**: Some SIP providers that do not use a number for the username require that the primary number on the account be sent with the username. This field allows you to enter it. Please leave this field blank unless the interop guide for your specific SIP provider requires it.

7. Enter the registration settings for the account:

- **Registration Type**: Select Static Registration only if the SIP provider requires it and you do not have a username and password for your SIP account.
- **Registration Expires**: The length of time that that the SIP gateway will check back with the SIP provider to ensure the gateway is registered.
- **Registration Status**: Indicates if the link to the SIP provider is up or not. The Registration light on the SIP gateway will turn green when registered to the SIP provider as well.

2

8. Enter the server settings for the account:

— **SIP Server**: Enter the URL of the server given by the SIP provider. This server is usually the inbound, outbound, and registrar server all in one for the SIP provider. Unless your SIP provider specifies different servers for each of these, you will not need to fill in the registrar or outbound proxy server. As always, though, if you are in doubt please check the interop guide for your specific SIP provider.

— **Registrar Server**: This is a server that only handles the SIP registration requests and does not handle calls. Please check the interop guide for your specific SIP provider to see if this required. In most cases it is not.

— **Outbound Proxy Server**: This is a server that handles outbound calls. Please check the interop guide for your specific SIP provider to see if this required. In most cases it is not.

Server Settings:

SIP Server:

Address or Url: [   ]

Port: [5060]

Registrar Server:

Address or Url: [   ]

Port: [   ]

Outbound Proxy Server:

Address or Url: [   ]

Port: [   ]

9. Select the codec configuration for the account. Click a codec and use the **Add >** or **< Remove** buttons to enable or disable codecs. Click a codec and use the ▲ or ▼ buttons to move it up or down in the **Enabled Codecs** list.

We support three codecs for voice encoding of a call: G.711u, G.729, and G.711a. Synapse first tries whichever codec is listed first. If it cannot negotiate a call on that codec, Synapse will try the next one.

— **G.711a** is disabled by default because it is a rarely used codec and is not as efficient as G.729 and G.711u.

— **G.711u** is the most common codec used over all telephone networks in the U.S. today. Almost all calls over T1 or analog lines use this codec. G.711u uses between 64 Kbps and 87.2 Kbps per direction per call. It uses very little compression, so speech quality is very high.

— **G.729** is a very compression-heavy codec that degrades audio quality slightly, but saves significant bandwidth per call. G.729 uses between 8 Kbps and 31.2 Kbps per direction per call. Also G.729 is not a free codec like G.711u, requiring the SIP provider to pay a license fee to use it. G.729 may not be available from your SIP provider.

Synapse®

Codec Configuration:

Disabled Codecs

Enabled Codecs
- G.711u
- G.729
- G.711a

Add >

< Remove

10. Once you have verified that all the settings are correct, click **Apply** at the bottom of the page to save the settings and enable the account.

at&t