SIP Trunking Test Results for Synapse® SB67070 SIP Gateway from AT&T

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1 Executive Summary

This report provides the test results found to date for the Synapse SB67070 SIP Gateway evaluation. The following is a summary of the issues and limitations found while performing the test.

Issues/Limitations

- Outbound Caller ID Blocking not supported
- Call Forward Busy to PSTN from PSTN not supported
- Call Forward Busy to Ext from PSTN not supported
- Call Transfer to PSTN—blind and consult not supported using SIP REFER, but it works based on re-INVITE.

Unsupported SIP Trunk Features

- G.722 and G.722.1 codecs
- RTCP Sender Reports
- Configurable ptime for G.729 (RFC 3264)
- CoS tagging
- Compressed RTP (cRTP)
- Passing Calling Number marked private
- Outbound Call Hold & Resume
- 408 error fail-over
- T.38 Fax is not supported, but fax works with G.711 pass-through

Miscellaneous Unsupported SIP Features

- SIP Asserted Identity (RFC 3325)
- SIP Diversion information
- SIP Session timers (RFC 4028)
- SIP REDIRECT
- SIP REFER (RFC 3515)
- SIP Referred-By header (RFC 3892)
- SIP SUBSCRIBE/NOTIFY (RFC 3265)
- SIP proprietary header for data forwarding
- Presentation-Restricted status (RFC 3325)

- SIP Options keep-alives

1.1 Registration Method

Static registration is utilized between Synapse and XO.

1.2 XO SIP Service Packages Supported

See "Issues/Limitations" above.

Pkg	Codec	DTMF	Fax
1	G.711	RFC2833 (In-band RTP DTMF)	No T.38; G.711 pass-through
2	G.729a	RFC2833	No T.38; G.711 pass-through

2 Software and Hardware Equipment Requirements for Testing

All Synapse devices must have software version 2.0.0 or later installed. All devices must be running identical software versions.

- SB67070 SIP Gateway
- SB67030 Deskset
- SB67031 Deskset
- SB67020 Deskset
- Cisco 1921 used as the LAN router (NAT mode with SIP ALG)

3 Test Configurations

Figure 1 shows the configuration used during lab testing.

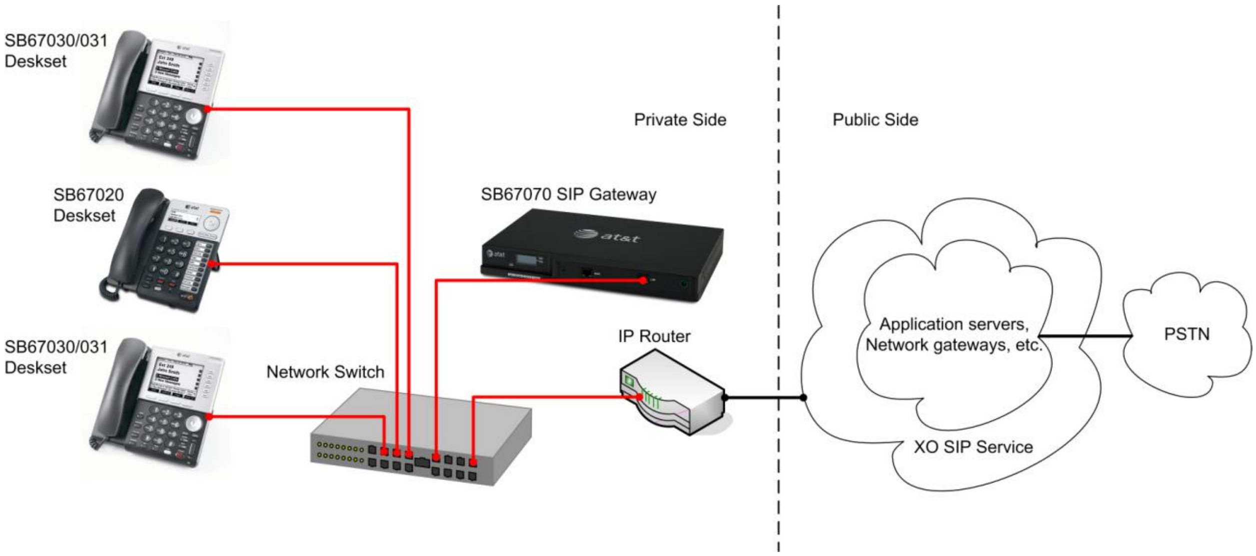


Figure 1 Test Configuration

Note: The above setup only shows main lab network elements.

4 Synapse SIP Gateway Configuration Guide

This configuration guide provides instructions for configuring the Synapse Business Phone System for XO. Specifically, this guide describes how to configure the Synapse SB67070 SIP Gateway for an XO account.

Documents related to this configuration guide include:

- Synapse Installation Guide i17 or later
- Synapse Administrator's Guide i15 or later

You can view and download these documents from www.telephones.att.com/synapseguides.

This configuration guide assumes that the Synapse system is installed and that users are able to make internal (Deskset to Deskset) calls. A Synapse System must include one of more of the following devices:

- Desksets (SB67030/031 and/or SB67020 Desksets). The system supports up to 100 Desksets.
- Gateways. At least one Gateway must be installed. The system supports up to four SB67010 PSTN Gateways, one SB67060 T1 Gateway, and one SB67070 SIP Gateway. You can install any combination of Gateway models in a system, as long as you do not exceed the supported number of each Gateway model.

For more information, see Figure 2: Sample Synapse System.

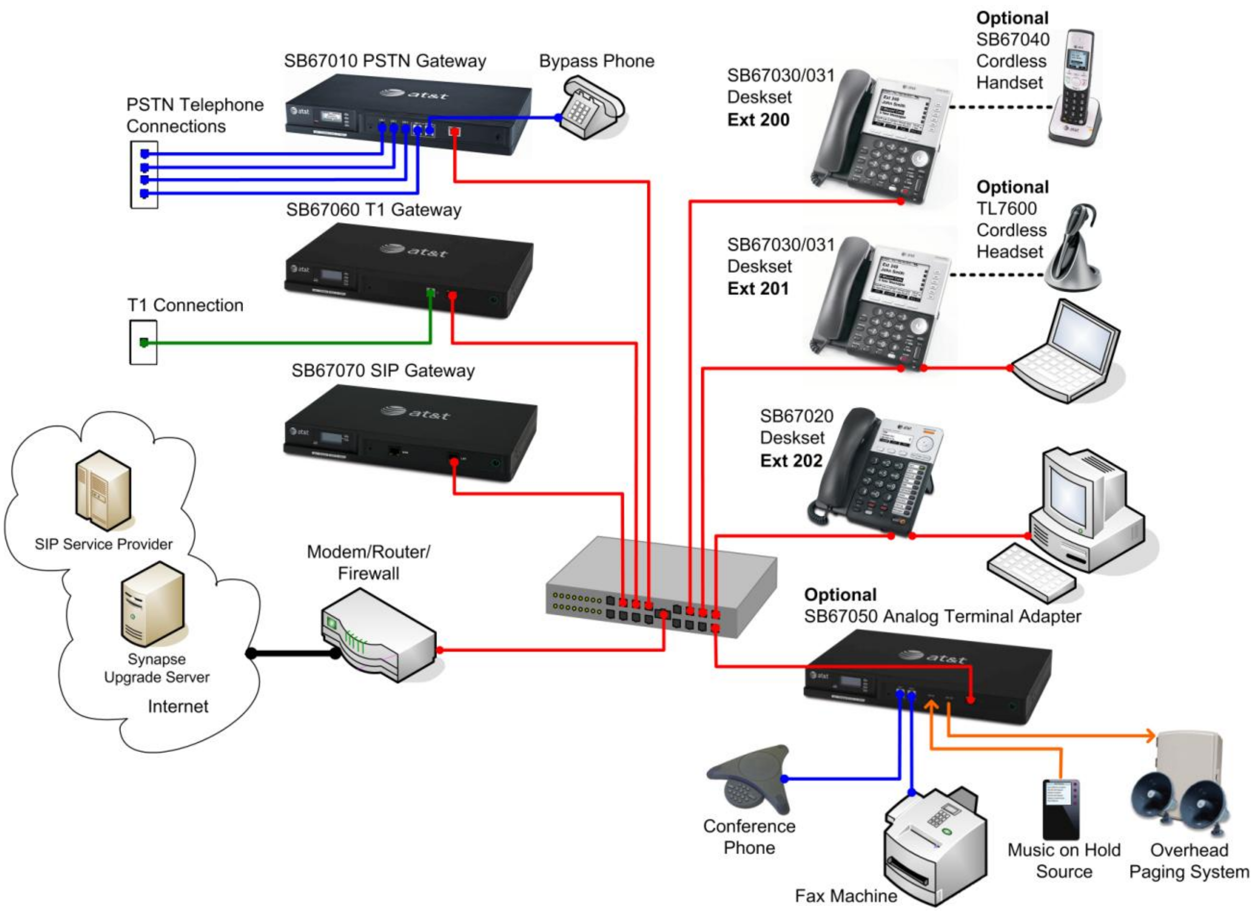


Figure 2: Sample Synapse System

Note: The available system configuration settings will vary according to the Synapse devices installed. For more complete system configuration information, see the Synapse Administrator's Guide i15 or later.

4.1 SB67070 SIP Gateway Features

- A LAN 10/100 BaseT Ethernet port connects to the existing Synapse network.
- A four-line LCD display with four hard keys provides status information, such as the device IP address, and allows basic network configuration and firmware upgrades.
- Device configuration and system configuration, such as SIP Account information, available through the Web User Interface (WebUI).
- Supports up to 16 simultaneous voice calls.
- Additional System features, including Direct Inward Dial (DID) and configurable dial plan.

For more information about Synapse, visit http://telephones.att.com/smb.

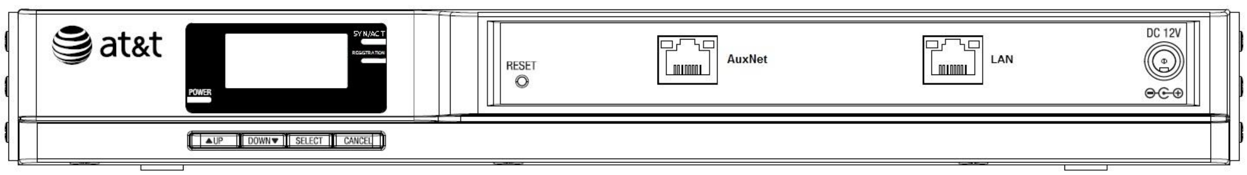
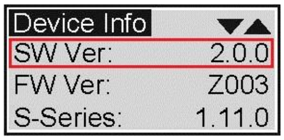


Figure 3: SB67070 SIP Gateway Front Panel

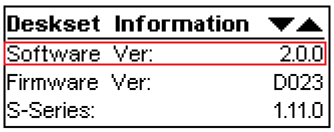
4.2 Viewing the Software Version

Systems with software versions 2.0.0 and later support the features described in this guide. All Gateways and Desksets must have identical software versions installed.

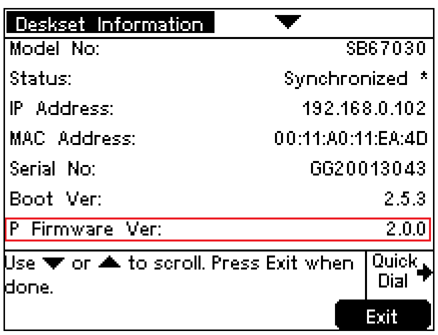
To determine the software version of the SB67070 SIP Gateway from the device front panel, press **SELECT**, **SELECT**, and then **DOWN ▼**. The software version appears.



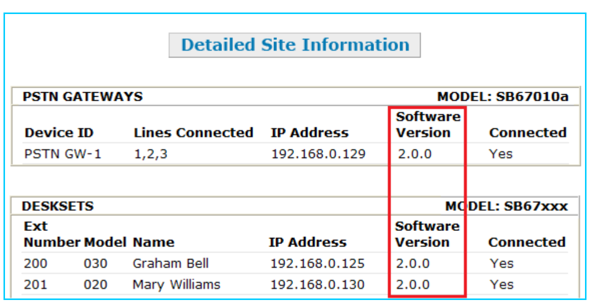
To determine the SB67020 Deskset software version, press **MENU**, then **4**, and then ▼ to display the software version.



To determine the SB67030/031 Deskset software version, press **MENU**, then **4.** See the P Firmware version.



To determine the software version of all installed devices, log in as administrator (see "Using the WebUI" on page 8). Then click **Detailed Site Information** to see the software versions and other information. There may be a delay as the system gathers this information.



Detailed Site Information

PSTN GATEWAYS				MODEL: SB67010a
Device ID	Lines Connected	IP Address	Software Version	Connected
PSTN GW-1	1,2,3	192.168.0.129	2.0.0	Yes

DESKSETS					MODEL: SB67xxx
Ext Number	Model	Name	IP Address	Software Version	Connected
200	030	Graham Bell	192.168.0.125	2.0.0	Yes
201	020	Mary Williams	192.168.0.130	2.0.0	Yes

To update device software, see "Updating Devices" in the Synapse Administrator's Guide, available at www.telephones.att.com/synapseguides.

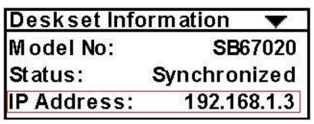
4.3 Using the WebUI

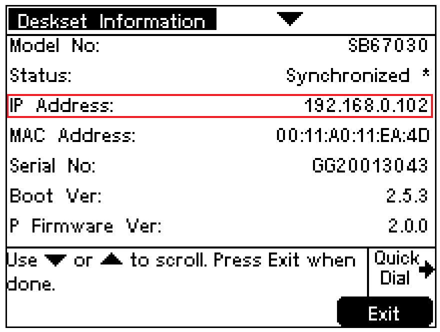
The Synapse WebUI allows you to configure the SIP Gateway for XO SIP Service.

To access the Synapse WebUI and log in:

1. Connect your computer to the same IP subnet as the Synapse system, or ensure that devices on different subnets are able to communicate. For example, you can connect your computer to the PC port on the back of a Deskset.

2. On the Deskset, press **MENU** then **4**. The Deskset Information screen appears.

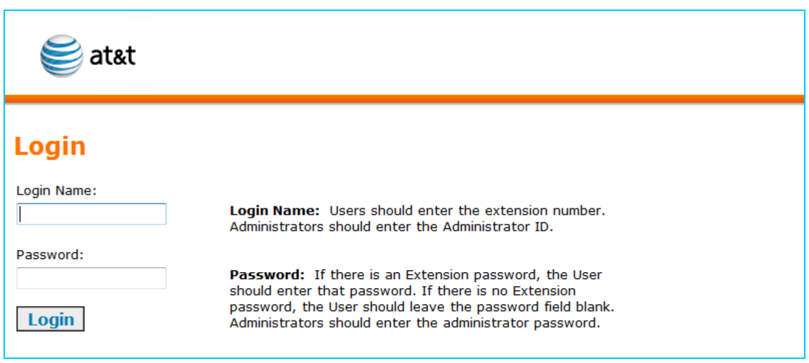




3. Find the IP address on the Deskset Information screen.

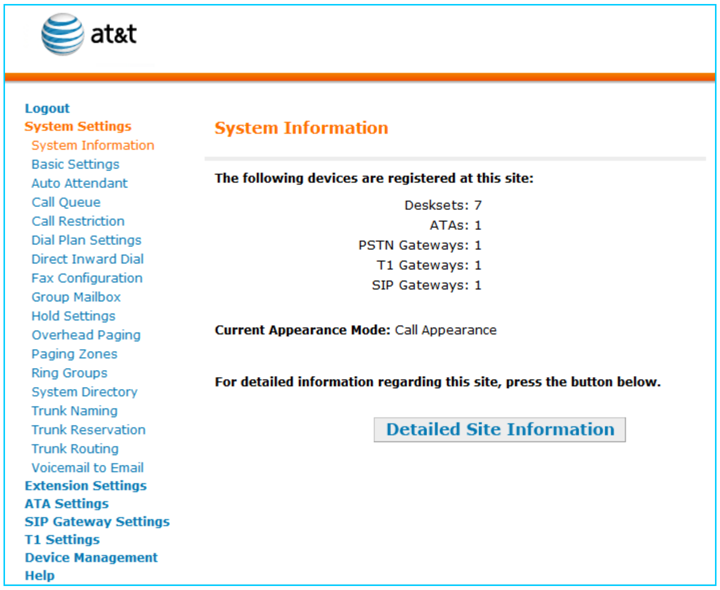
4. Open a browser. Depending on your browser, some of the pages presented here may look different and have different controls.

5. Type the Deskset IP Address in the browser address bar and press **Enter**. The Login page appears.



6. Enter your login credentials. If logging in for the first time, enter **admin** in the Login Name field and **12345** in the Password field, then click Login. You can change your Admin ID and password once you are logged in.

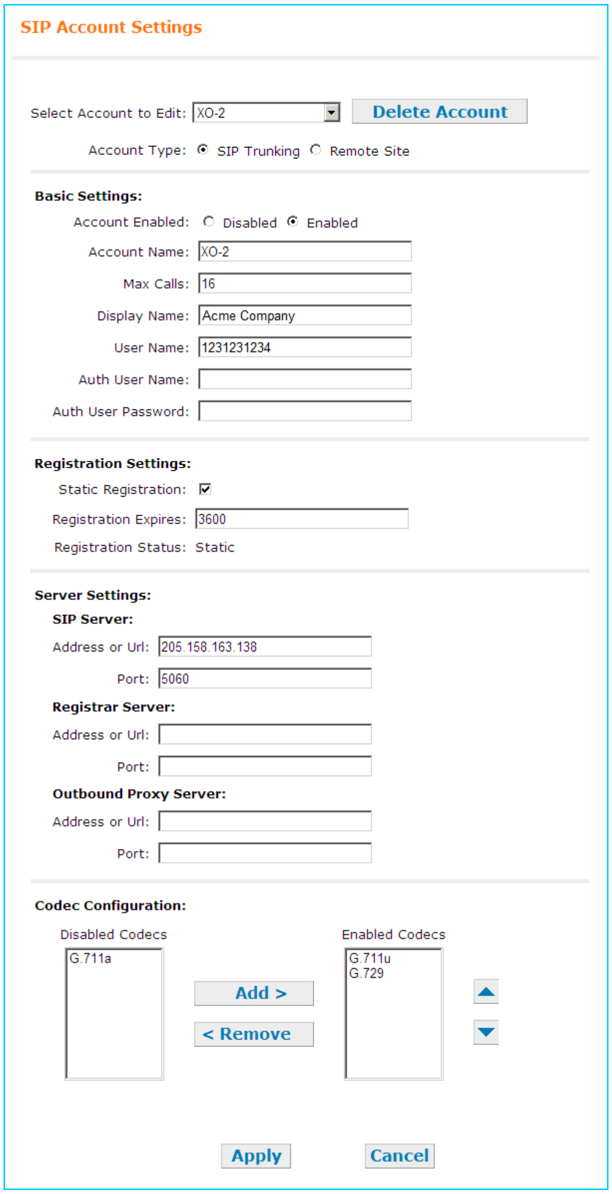
7. Click topics from the navigation list on the left side of the WebUI to see them. For your security, the WebUI times out after being idle for 10 minutes, after which you must log in again.


at&t
Logout
System Settings
System Information
Basic Settings
Auto Attendant
Call Queue
Call Restriction
Dial Plan Settings
Direct Inward Dial
Fax Configuration
Group Mailbox
Hold Settings
Overhead Paging
Paging Zones
Ring Groups
System Directory
Trunk Naming
Trunk Reservation
Trunk Routing
Voicemail to Email
Extension Settings
ATA Settings
SIP Gateway Settings
T1 Settings
Device Management
Help
System Information
The following devices are registered at this site:
Desksets: 7
ATAs: 1
PSTN Gateways: 1
T1 Gateways: 1
SIP Gateways: 1
Current Appearance Mode: Call Appearance
For detailed information regarding this site, press the button below.
Detailed Site Information

4.4 Configuring SIP Account Settings

To configure SIP Account Settings:

1. In the navigation menu at left, click **SIP Gateway Settings**. The SIP Account Settings page appears.



2. Select **SIP Trunking** as the Account Type.

3. Select **Create New Account**, or select an account to edit. If you have already created an account, a Delete Account button appears. Clicking **Delete Account** deletes the account and loads an empty account page.

4. Enter the SIP Gateway Basic Settings.

   a. Enable or Disable the account. You must enable the account before it can be used. Disabling the account does not erase the settings associated with the account.

   b. Enter the Account Name. The SIP account name appears on the Dial Plan Settings page and the Trunk Reservation page.

   c. For Max Calls, enter the number of simultaneous call sessions you purchased. The maximum value is 16. Setting the Max Calls to a value that is less than the current number of Trunk Reservations for the SIP Account will generate an error.

   d. Enter the Display Name. The Display Name is the text portion of the Caller ID that is displayed for outgoing calls.

   e. Enter the User Name as provided by XO. The User Name, also known as the Account ID, is usually the company's main number. Synapse will only accept digits for a User Name.

   f. Leave the **Auth User Name** and **Auth User Password** fields blank.

5. Enter the SIP Gateway Account Registration Settings.

   a. Select **Static Registration**.

   b. Enter the number of seconds for Registration Expires. This setting applies to dynamic registration. It is a re-registration timeout value sent to the SIP Provider. This is usually overriden by a re-registration interval determined by the service provider’s response. The default setting is 3600 seconds and should only be changed on the advice of your service provider.

6. Enter the SIP Gateway Account Server Settings as provided by XO.

   a. Enter the SIP Server Address or URL. Ensure that you do not enter any spaces before or after the address or URL.

   b. If necessary, enter the SIP Server Port. Port 5060, the default setting, is typically used for SIP transmission.

   c. Registrar Server Address or URL and Registrar Server Port should be left blank.

   d. Outbound Proxy Server Address or URL and Outbound Proxy Server Port should be left blank.

7. Configure the Codec Configuration.

   a. Enable or disable audio codecs. You can click **Add >** to add the selected codec to the Enabled Codecs list, or click **< Remove** to add the selected codec to the Disabled Codecs list. **Note:** XO does not support the G.711a codec.

   b. Arrange the enabled audio codecs. Select a codec, then click ▲ or ▼ to change the order.

      The SIP Gateway uses the audio codecs in the order they are listed on a per call basis. You can choose codecs based on the speed versus audio performance required.

8. Click **Apply** to save your changes.

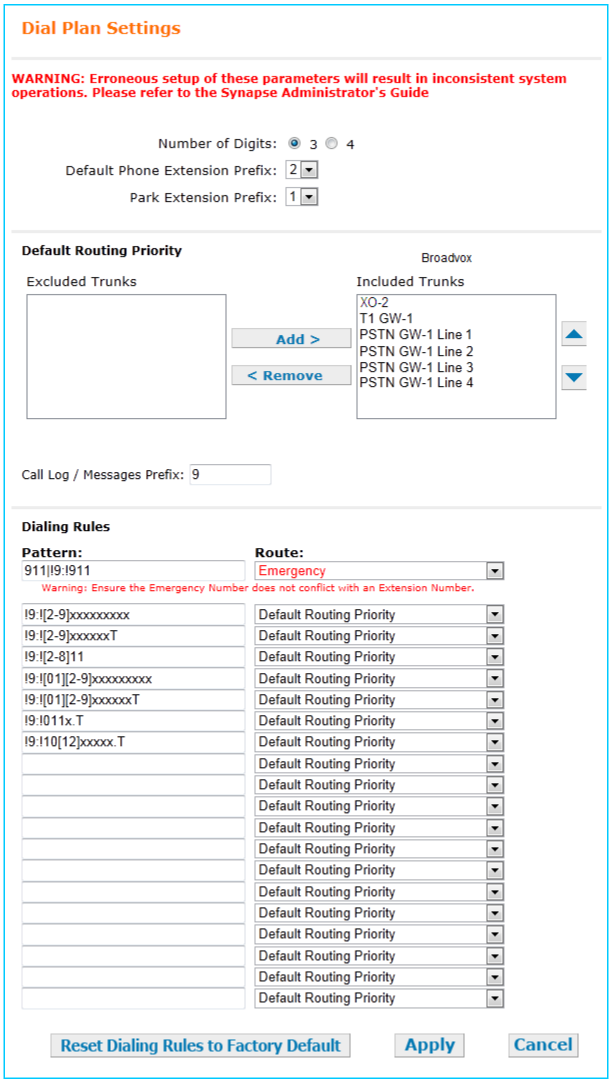
The SIP Gateway Registration LED should turn green when the SIP Gateway has successfully registered to XO.

4.5 Configuring the Dial Plan

Before configuring the Dial Plan, see "Dial Plan Settings" in the Synapse Administrator's Guide i15 or later available at http://telephones.att.com/synapseguides.

To configure Synapse Dial Plan settings:

1. In the navigation menu at left, click **Dial Plan Settings**. The Dial Plan Settings page appears.



2. Set the Default Routing Priority. Ensure that the XO Account that you configured is included in the priority list. You can:

   - exclude a trunk by selecting it, then clicking **< Remove**. You may want to exclude a trunk if it is being used for a fax line or a door phone line, for example.

   - move a trunk up or down the priority list by selecting it, then clicking ▲ or ▼.

3. Set the Call Log/Messages Prefix according to your Dialing Rules. You can enter a maximum of eight digits using only the characters 0–9, #, *, or P. Leave the field blank if the Dialing Rules do not use a number for external line access.

4. Enter or modify Dialing Rule patterns. For more information, see "About Dialing Rules" in the Synapse System Administrator's Guide.

5. Select a Route for each Dialing Rule pattern as required. The list of routes includes all available trunks in the system, as well as the Default Routing Priority. You can route a Dialing Rule pattern to use the SIP Gateway only, or to use the Default Routing Priority.

6. Click **Apply** to save these settings.