nexVortex Setup Guide

AT&T SYNAPSE

May 2012

510 SPRING STREET | HERNDON VA 20170 | +1 855.639.8888

Introduction

This document is intended only for nexVortex customers and resellers as an aid to setting up the Synapse Small & Medium Business Phone System from AT&T to connect to the nexVortex Business Grade SIP Trunking Service.

- Related Synapse documents 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.

- Further AT&T Synapse information can be found at telephones.att.com/smb or call 1 (888) 916-2007. In Canada dial 1 (888) 883-2474.

- Further help may be obtained by emailing support@nexvortex.com.

If you find any errors in this document or have any suggestions, please email us at support@nexvortex.com so that we can make updates to this document.

Important! Your DNS Address
Your specific DNS address was provided in the Account Set Up email you received the day you opened your account. Your Authentication User ID and password are also in this email. If you need assistance locating this information, please contact support@nexvortex.com.

Note: For all instructions throughout this Guide, you must substitute your DNS address wherever xx.xx.xxx.xxx is referenced.

System and Network Components Required for SIP Trunking Service

The necessary components and setup for Synapse using NexVortex SIP Trunking are shown in Figure 1. All Synapse Desksets and Gateways must be running the same software versions (2.2.0 or later).

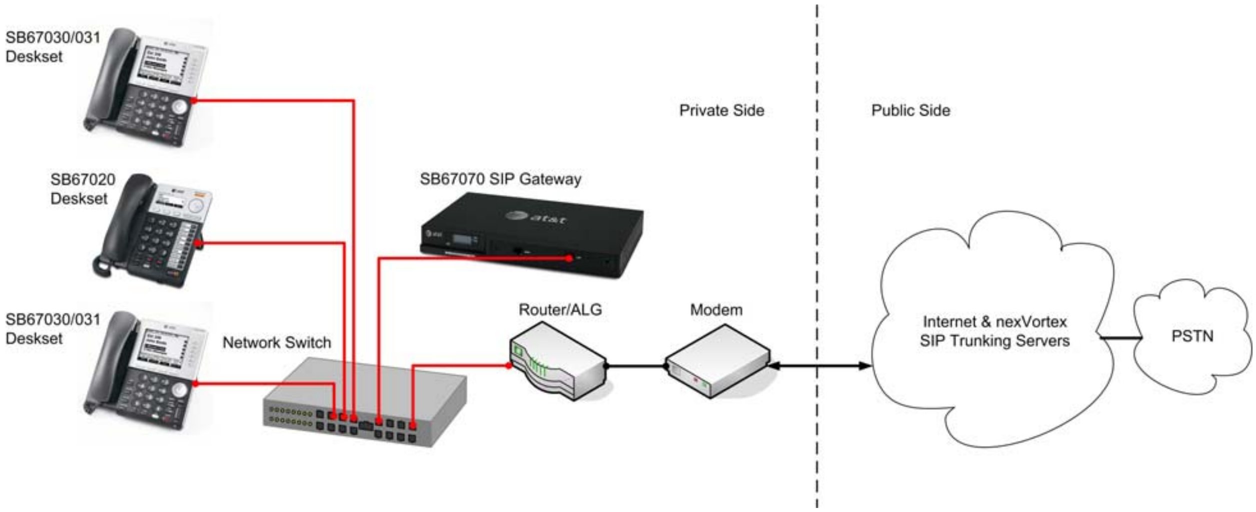

Figure 1 Synapse with SIP Trunking

General network requirements

Router

To provide NAT traversal, use a SIP ALG (Application Gateway Layer)-capable router.

Two ALG-capable routers have been tested for use with the Synapse SIP Gateway.

1. Cisco RV180 SW V1.0.0.30 or later with ALG enabled. Note: Enable SIP ALG on the router before you connect the SIP Gateway to the network.

2. Cisco 1921 (ISR) SW V15.0 or later with ALG enabled. Please consult your IT engineer and make sure that SIP ALG is enabled.

Internet Connection

It is recommended to order Static IPs from your Internet provider.

Ensure that your Internet connection has sufficient bandwidth to handle your projected VOIP call traffic. In general, each simultaneous call requires 100 kb/s on the uplink.

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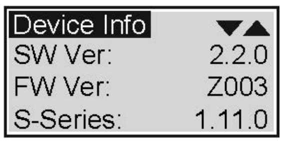
Configuration

This section describes how to configure the SIP Gateway Account Settings to support nexVortex SIP Trunking.

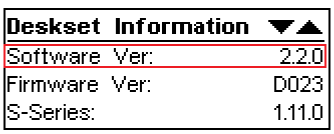
Software Version Compatibility

Synapse systems with software versions 2.2.0 and later support the features described in this guide. All Gateways and Desksets must have the same software versions installed.

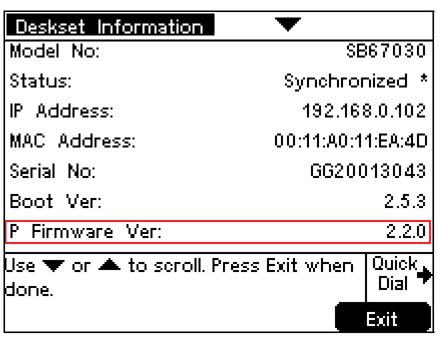
- To determine the software version of the SB67070 SIP Gateway from the device front panel, press SELECT, SELECT, and then DOWN. The software version appears.



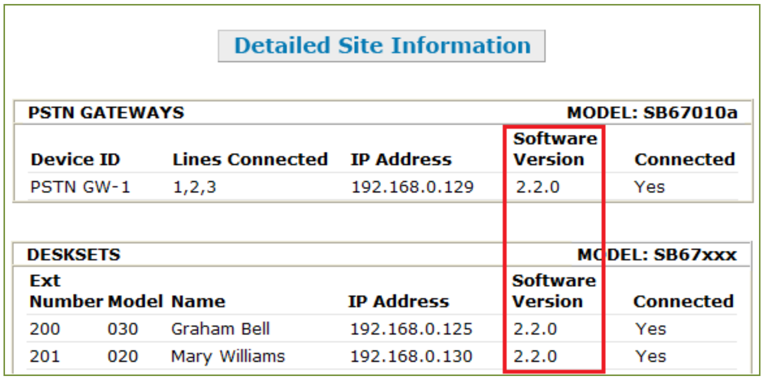
- To determine the SB67020 Deskset software version, press MENU, then 4, and then ▯ to display the software version.



- 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 Logging In to Synapse as Administrator on page 5), 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	2.2.0	Yes

DESKSETS					MODEL: SB67xxx
Ext Number	Model	Name	IP Address	Software Version	Connected
200	030	Graham Bell	192.168.0.125	2.2.0	Yes
201	020	Mary Williams	192.168.0.130	2.2.0	Yes

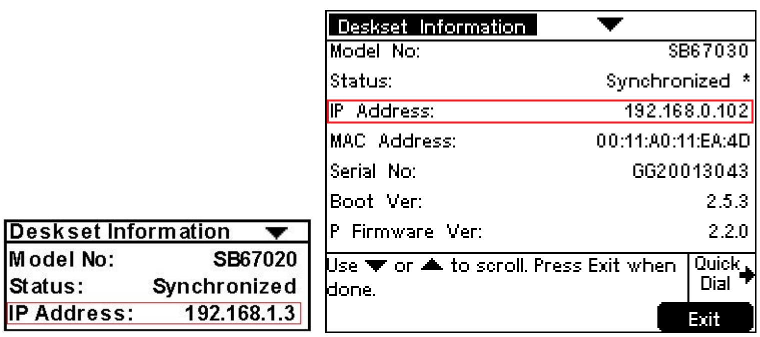
To update device software, see “Updating Devices” in the Synapse Administrator’s Guide, available at www.telephones.att.com/synapseguides.

Logging In to Synapse as Administrator

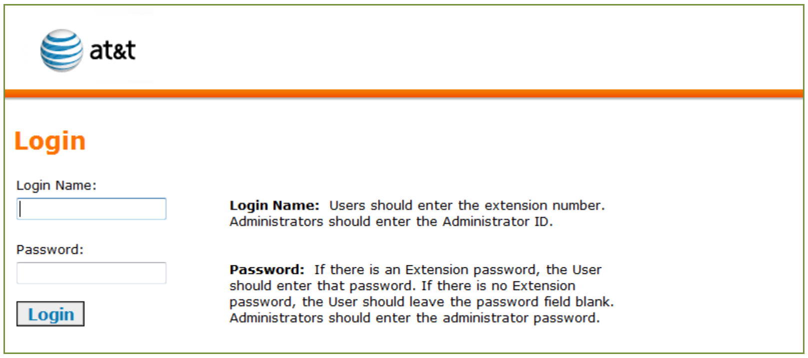
The Synapse WebUI allows you to configure the SIP Gateway for nexVortex SIP Trunking.

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.

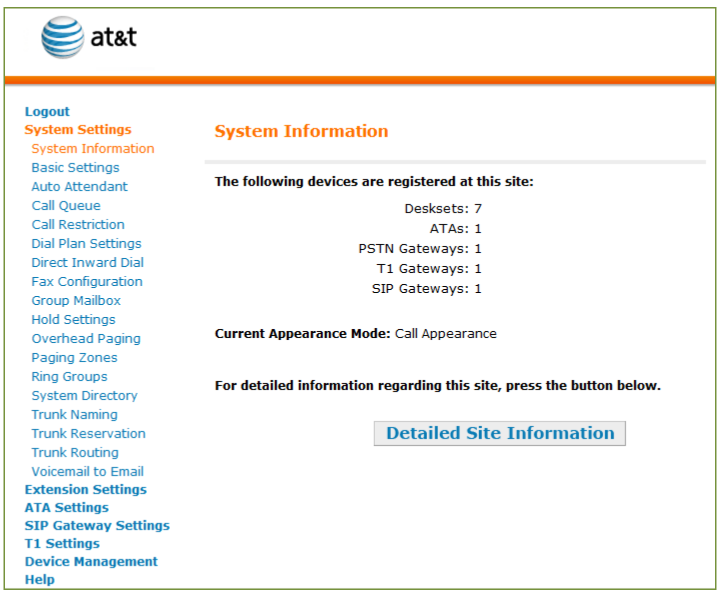


3. Find the IP address on the Deskset Information screen.
4. Open a browser. Depending on your browser, some of the pages presented here may look different and have different controls.
5. Type the Deskset IP Address in the browser address bar and press **Enter**. The Login page appears.



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.



Configuring the SIP Gateway

To configure SIP Account Settings:

1. In the navigation menu at left, click **SIP Gateway Settings**. The **SIP Account Settings** page appears as shown in Figure 2.

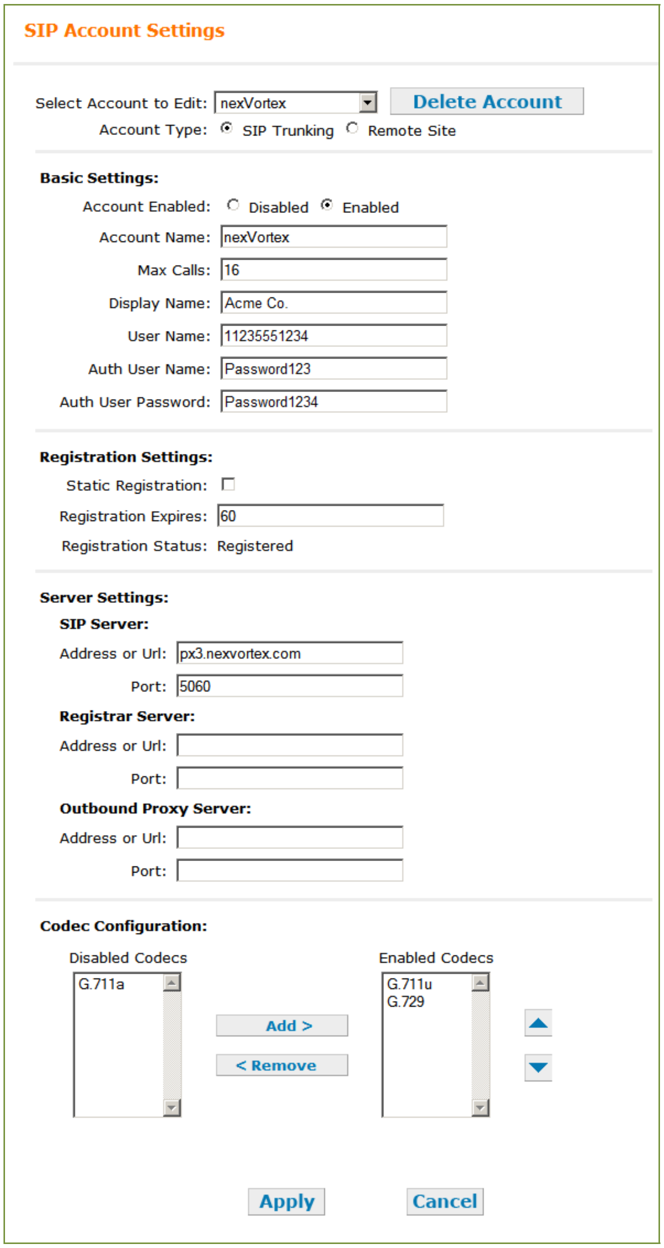


Figure 2 nexVortex SIP Account Settings

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

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

4. Enter the rest of the SIP Account Settings. Refer to the example shown in Figure 2 for the location of each setting.

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Basic Settings		
Menu Item	**Description**	**Recommended setting**
Account Enabled	Enables or Disables the account. You must enable the account before it can be used. Disabling the account does not erase the settings associated with the account.	Enabled
Account Name	The SIP account name appears on the Dial Plan Settings page and the Trunk Reservation page.	As desired
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.	As purchased
Display Name	The Display Name is the text portion of the Caller ID that is displayed for outgoing calls.	As desired
User Name	User Name as provided by nexVortex. The User Name, also known as the Account ID, is usually the assigned primary telephone number that nexVortex has provided to you. Synapse will only accept digits for a User Name.	As specified by nexVortex
Auth User Name	Provided by nexVortex.	As specified by nexVortex
Auth User Password	Provided by nexVortex.	As specified by nexVortex
Registration Settings		
Menu Item	**Description**	**Recommended setting**
Static Registration		Blank (unchecked)
Registration Expires	Applies to dynamic registration. It is a re-registration timeout value sent to the SIP Provider. This is usually overridden by a re-registration interval determined by the service provider's response. The default setting is 3600 seconds.	60
Server Settings		
Menu Item	**Description**	**Recommended setting**
SIP Server Address or URL		As specified by nexVortex
SIP Server Port	Port 5060, the default setting, is typically used for SIP transmission.	5060

Codec Configuration		
Menu Item	**Description**	**Recommended setting**
Disabled Codecs	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.	G.711a
Enabled Codecs		G.711u, G.729

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

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

Configuring the Dial Plan

Before configuring the Dial Plan, see "Dial Plan Settings" in the Synapse System 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.

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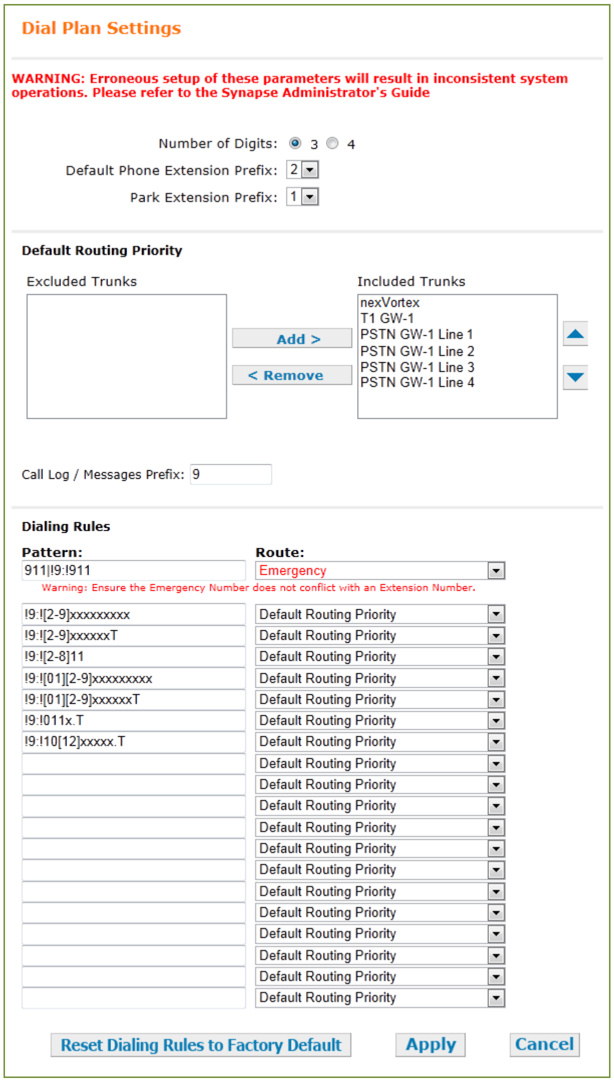


2. Set the **Default Routing Priority**. Ensure that the nexVortex SIP Trunking Account that you configured is included in the priority list. You can:

- 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.

Configuring the nexVortex Service

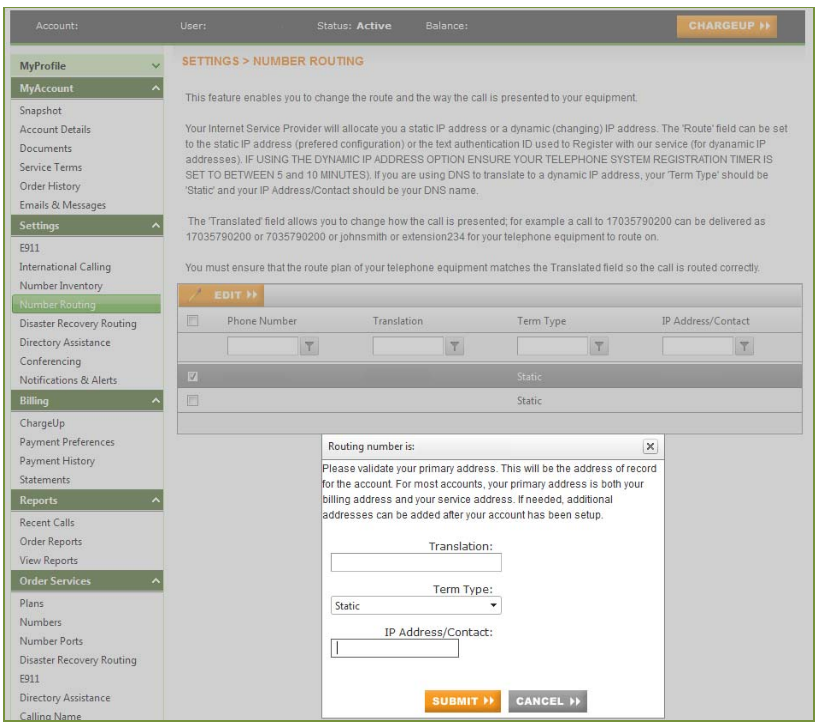
After ordering your nexVortex service and configuring the Synapse system, you must configure number routing in your nexVortex account.

1. From the nexVortex Account Setup email, obtain the 'WEBLOGIN UserName' and log in to your nexVortex account at www.nexvortex.com.

2. If you have a static IP address:

   - Click **Settings** on the left-hand side and then select **Number Routing**. The **Settings > Number Routing** page appears, showing the phone numbers that you ordered from nexVortex.

   - For each phone number, click the check box next to the number, then click the **EDIT** button. A number routing edit window appears.



   - In the **Translation** box, enter how you want the call presented to the Synapse system (e.g. 8015551234 can be presented as 8015551234 or 1234567890 or johnsmith).

   - In the **Term Type** box, select **Static**.

   - In the **IP Address/Contact** box, enter the static IP address you obtained from your Internet service provider.

3. If you have a dynamic IP address, please email support@nexvortex.com

4. Charge up your account by clicking the **CHARGEUP** button on the top right of the screen. nexVortex is a prepay service and once a positive balance is on your account it will be placed in the 'Active' state and you can make and receive calls.

Troubleshooting

Note: For customer service, repair, replacement, or warranty service for Synapse products, visit www.telephones.att.com/smb or call 1 (888) 916-2007. In Canada dial 1 (888) 883-2474.

Device Log

If you have trouble with your system and you require customer service, they may need the device log for troubleshooting purposes. You can generate a device log on the **Device Log** page.

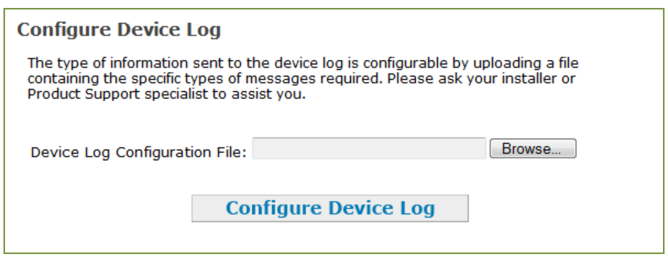
You can also configure a device log prior to generating the log. Your Synapse Product Support specialist may want to see specific information in the device log. If so, you must configure the device log using a configuration file that your Synapse support person provides.

To configure the Device Log:

1. In the navigation menu at left, click **Device Management**, then **Device Log**.

2. In the **Device Log** list, select the desired device.

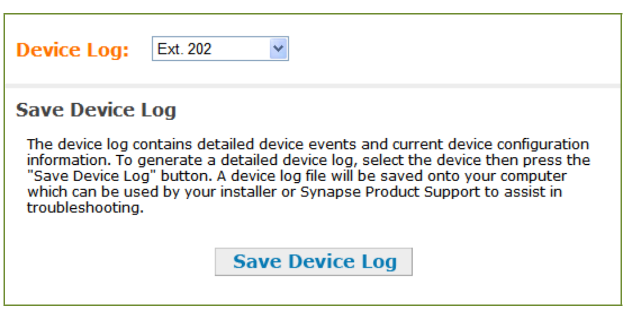
3. Under **Configure Device Log**, click **Browse...** and select the Device Log Configuration file.



4. Click **Configure Device Log**. After configuration is complete, you can proceed with generating the Device Log, as described below.

To generate the Device Log:

1. In the navigation menu at left, click **Device Management**, then **Device Log**. The **Device Log** page appears.



2. In the **Device Log** list, select the desired device and click **Save Device Log**. It takes a minute for the file to generate.

3. When the file is complete, save the file on your computer.

After the download is complete you should provide the file to the installer or customer service.

General Troubleshooting Topics

General problems with setup, registration and calls.

Cause	Action
Causes can vary, but several issues can be corrected by restarting the router and the Gateway.	1. Disconnect the SIP Gateway from the network by unplugging the cable from the Gateway LAN port. 2. Power cycle the router and the SIP Gateway. 3. Reconnect the LAN cable to the SIP Gateway.

SIP Gateway REG LED is RED or SIP Registration status on the WebUI is Unknown.

Cause	Action
Incorrect SIP account settings.	Make sure your SIP Account settings are set as per your SIP service provider's requirements. On the WebUI SIP Account Settings page, ensure Static Registration is not selected.
Network connection problems.	Make sure you have a reliable Internet connection for the SIP Gateway. Make sure the SIP service is not down.

SIP Gateway is not working.

Cause	Action
No power to the Gateway.	Check the front panel LEDs on the Gateway: Make sure the Power LED is on. If not, connect power to the Gateway. Make sure that the SYN/ACT and REGISTRATION LEDs are both on solid GREEN.
Incorrect SIP account settings.	Make sure your SIP account settings are set as per your SIP service provider's requirements.
Network connection problems.	Make sure the SIP Gateway is synchronized to the system. Check the WebUI Detailed System Information page. Make sure you have a reliable Internet connection for the SIP Gateway. Make sure the SIP service is not down.

Calls on the SIP Gateway terminate unexpectedly.

Cause	Action
Issues with SIP account settings.	Make sure that your SIP account is registered. The REGISTRATION LED on the SIP Gateway front panel should be solid GREEN and the Registration Status on the WebUI SIP Account Settings should show Registered. On the WebUI SIP Account Settings page, ensure Static Registration is not selected.
Incorrect SIP account settings.	Make sure your SIP account settings are set as per your SIP service provider's requirements.
Network connection problems.	Make sure the SIP Gateway is synchronized to the system. Check the WebUI Detailed System Information page. Make sure you have a reliable Internet connection for the SIP Gateway. Try port forwarding on the router, the port specified for the SIP Server to the SIP Gateway LAN address. Make sure the SIP service is not down.

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Calls on the SIP Gateway have intermittent audio.

Cause	Action
Too many simultaneous calls approaching or exceeding available network bandwidth.	Enable G.729 codec, or give G.729 priority over G.711. Increase network bandwidth to allow for the maximum number of simultaneous calls you have purchased.

Calls on the SIP Gateway have one-way audio.

Cause	Action
Incorrect LAN configuration.	Check your LAN configuration for the Gateway. We recommend using the IP subnet 192.x.x.x.

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