Teletics Application Note

Using the Feature Server with SIP Trunks

Rev 1.0 – June 4 2012

Executive Summary

One of the key advantages of the Teletics w*intercom system architecture is the ability for the customer to integrate the w*intercom into larger networks. For example, the customer is operating a VSAT network, and wants a more seamless integration with their network. Or, where a third party VoIP provider is providing long distance services to the remote sites.

The purpose of this application note is to give a very simple example of how to integrate the Feature Server, or Feature Server RM, with a third party SIP trunking service. The intent of this document is to give a simple “where to get started” overview of how this is done. It is expected that from this simple example, someone with familiarity with SIP and IP networking could integrate a custom system with multiple accounts into a larger, working system.

SIP Trunking Overview

A SIP based VoIP provider will typically create an account with a login and password. The provider will also create a regular phone number that is serviced by this SIP account.

The Feature Server needs to incorporate four changes to use this account:

1. The Feature Server has to know about the ***customers*** IP network. The Feature Server that is shipped with a standard configuration does not assume that there is an external network of any kind. It is designed to only operate within the Teletics w*intercom network default design.

2. The Feature Server needs to know what to do with calls coming in via the new SIP account, and what phone on the system can access it. (Call Routing)

3. The Feature Server needs to have support for the CODEC used by the SIP provider.

4. The Feature Server needs to know how to log on to the SIP providers account (credentials, port specifications, account details).

Additional Resources

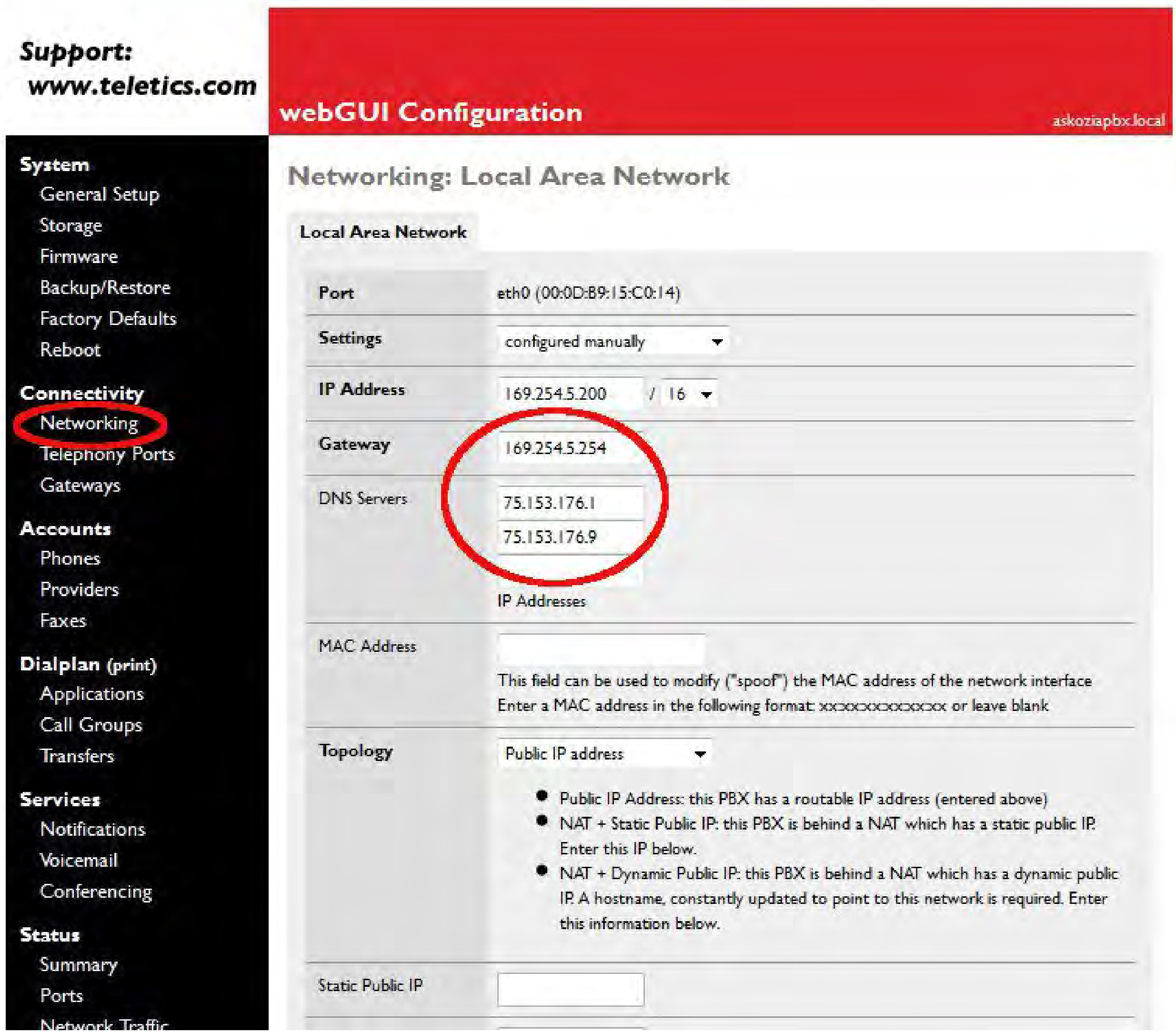
Customers attempting to integrate third party SIP provider services with the Teletics Feature Server or Feature Server RM should obtain the Teletics Feature Server Handbook.

The Teletics Feature Server Handbook is available through Teletics Support.

Should you require integration assistance from Teletics, this is available on a contracted basis, and is dependent on the amount of assistance required. Please contact your Teletics Distributor, or sales representative, for details on custom integration services.

Customer Network Settings

For the sake of our example, we will assume that the Feature Server and w*intercom 11 are connected to the LAN side of a router. The LAN side of the router must be on the network of 169.254.XXX.YYY, since the Teletics network is a zeroconfig network and uses the IP addresses of range 169.254.1.YYY to 169.254.5.YYY. The LAN side of our router has it's gateway at 169.254.5.254 in our example, but this is settable in the Feature Server. You also need to know the final DNS addresses in the network, and enter then into the Network settings page in the Feature Server:

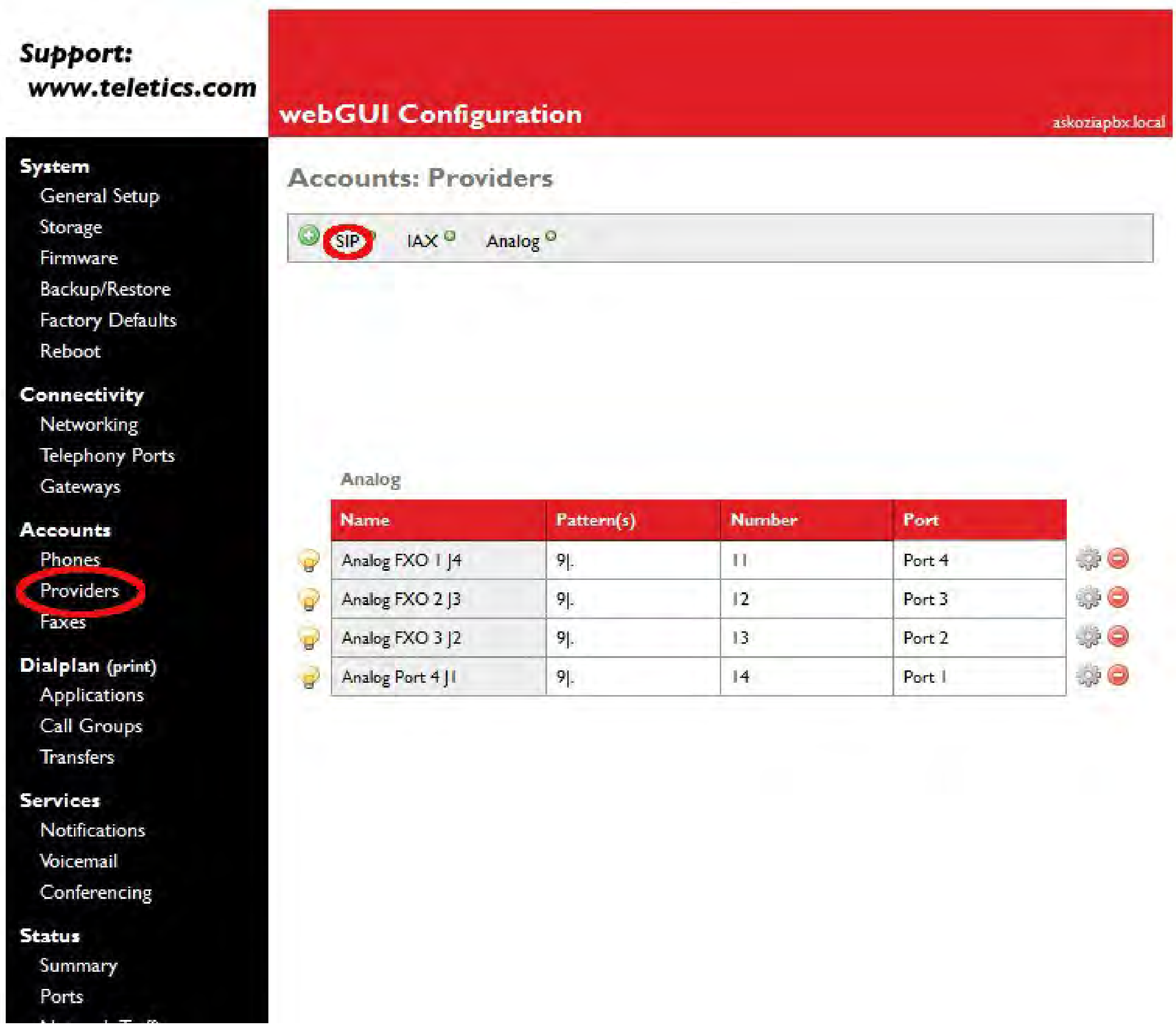


SIP Call Routing

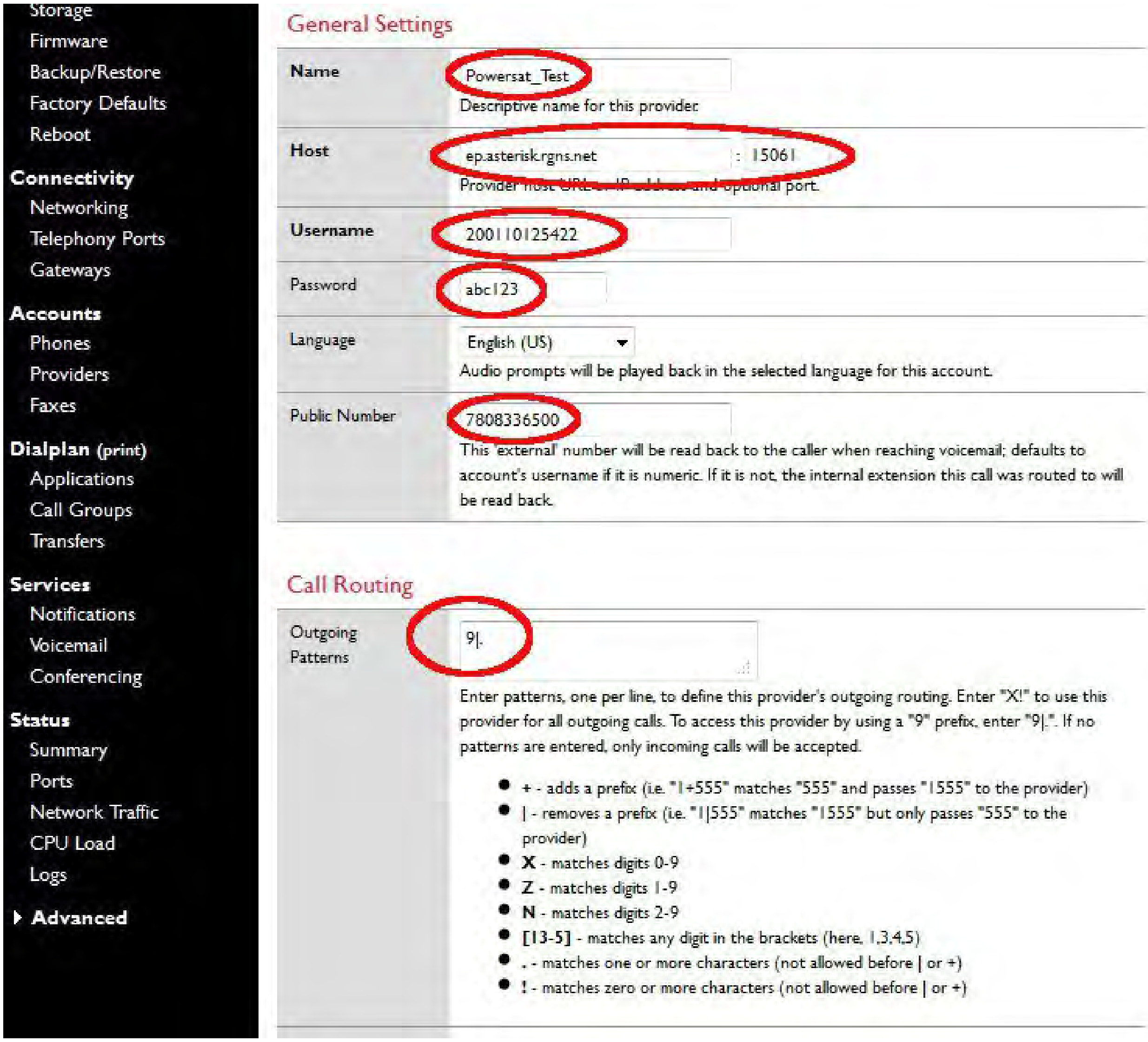
The standard Feature Server and Feature Server RM configuration image provides 4 FXO provider ports that are configured on a “one dedicated line in/out” basis for phones 11,12,13 and 14.

In our example, we have set up the one SIP trunk provided by the third party VoIP carrier to allow the person at phone 15 to dial out using this account, and to receive calls from the number associated to this account by the third part SIP provider’s servers.

Create a new SIP provider:



A new SIP provider menu window will open. Here is an example of typical SIP credentials that you would need to enter. First, the Main Account information and Routing:

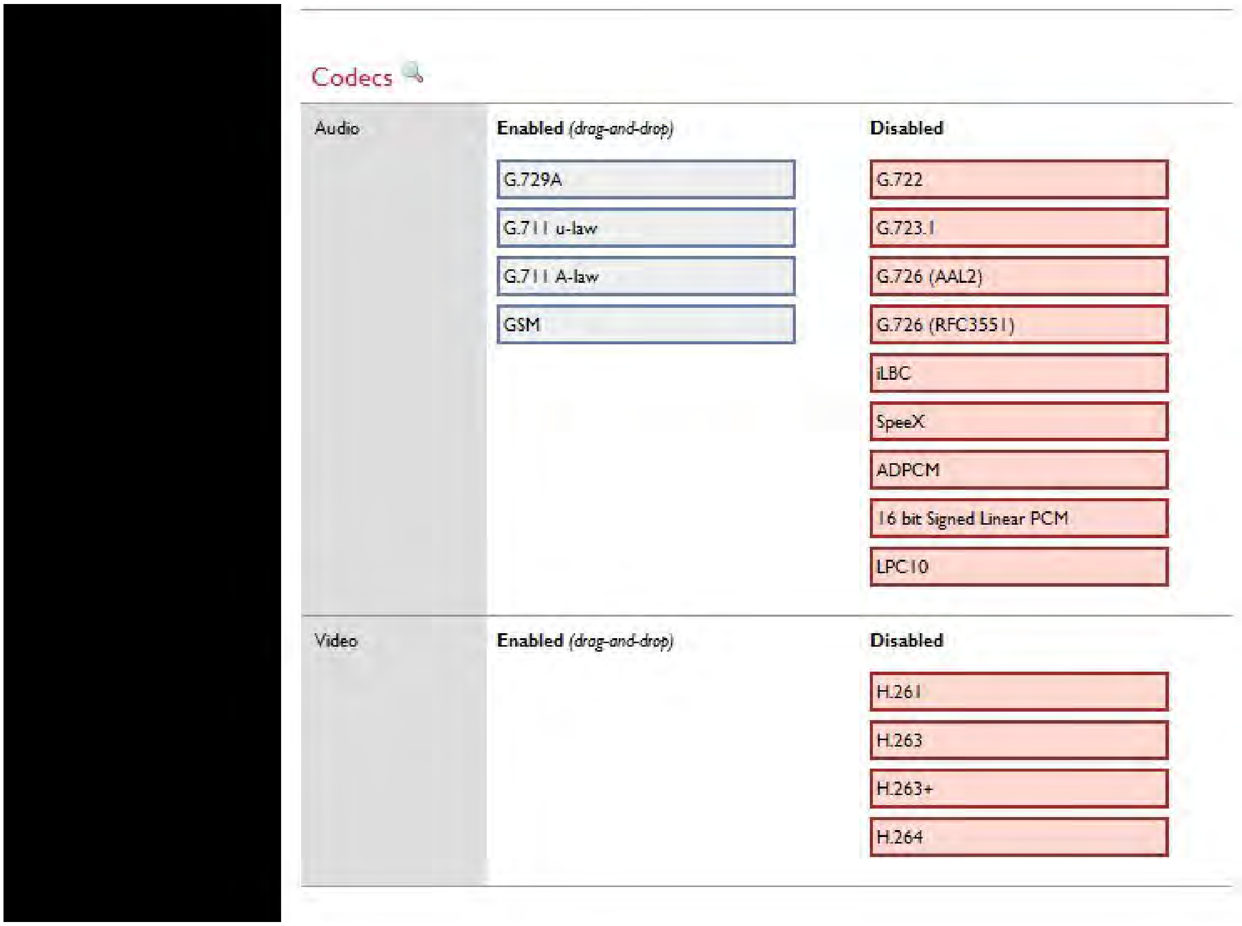


In this example:

- The **Name** setting is whatever you want to call this SIP provider/account.
- The **Host** URL (and port if needed), **Username**, **Password**, and **Public Number** is provided by your SIP provider/carrier for the account you are using.
- The Call Routing example specifies that when processing the call, we use 9 as the lead digit to indicate a SIP call, and then drop the lead 9 before passing the dial string to the SIP provider for further routing.

Setting Codecs

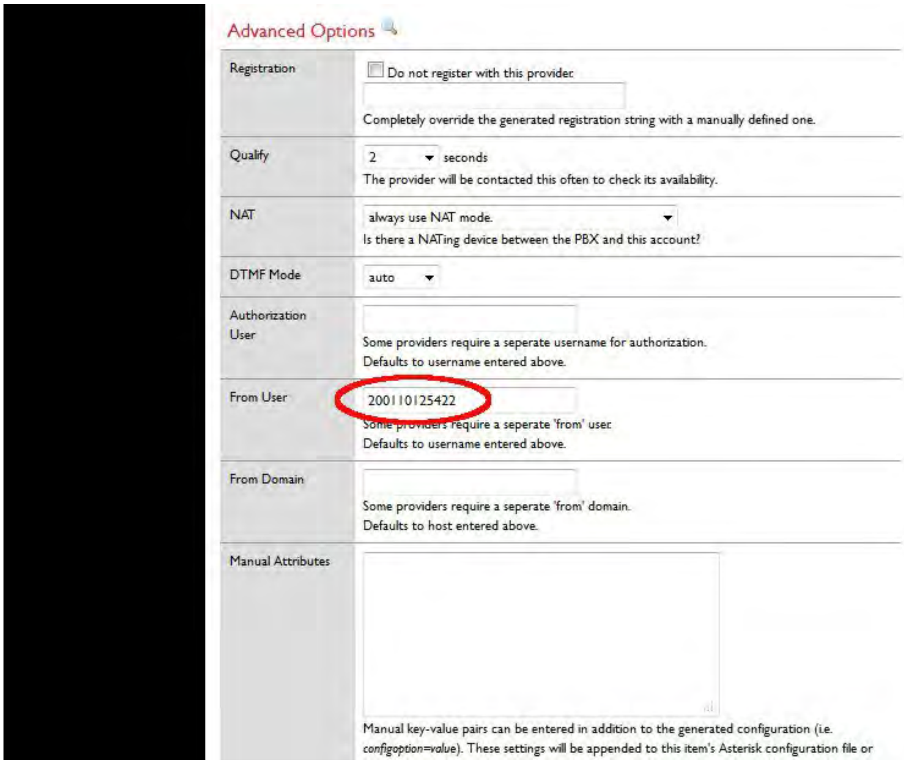
Your SIP provider will specify what CODEC(s) they use. You have to ensure that these are enabled in the Feature Server:



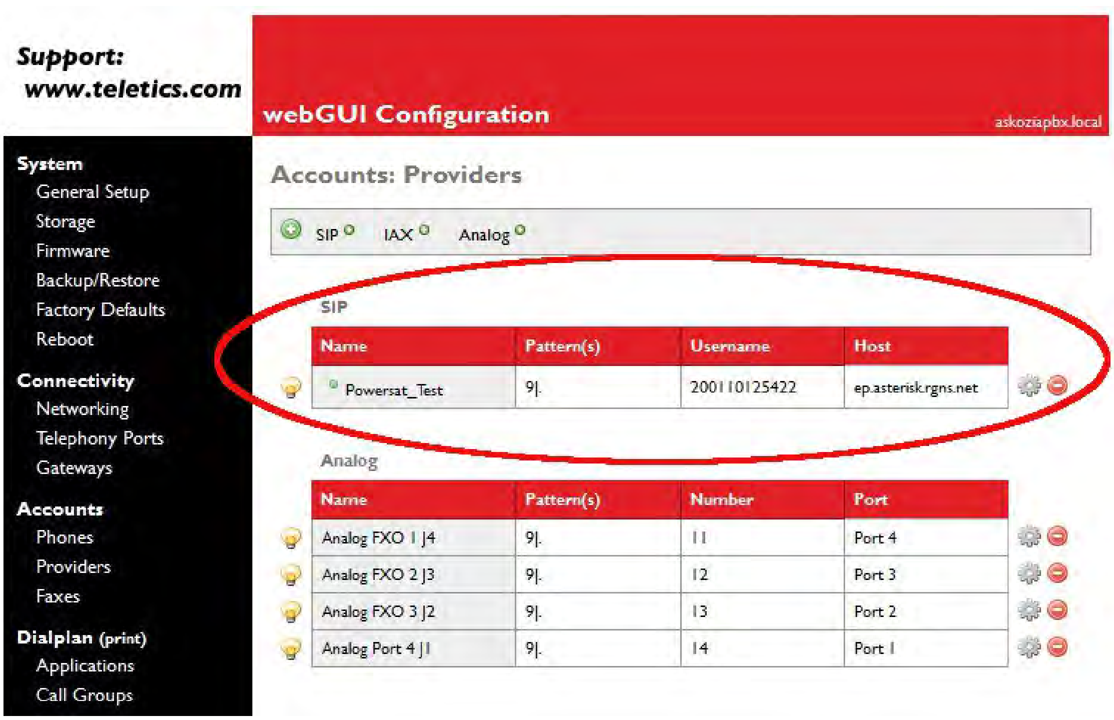
Please note that the Codec G.711 u-law has to be left in the enabled menu to ensure that the w*intercom system works properly.

Advanced SIP Account Settings

Some SIP providers require additional security information from the user when registering. An account name/identifier is a typical requirement. You can have the Teletics Feature Server send this information automatically on logon by entering the account number in the **From User** field in the **Advanced** settings:



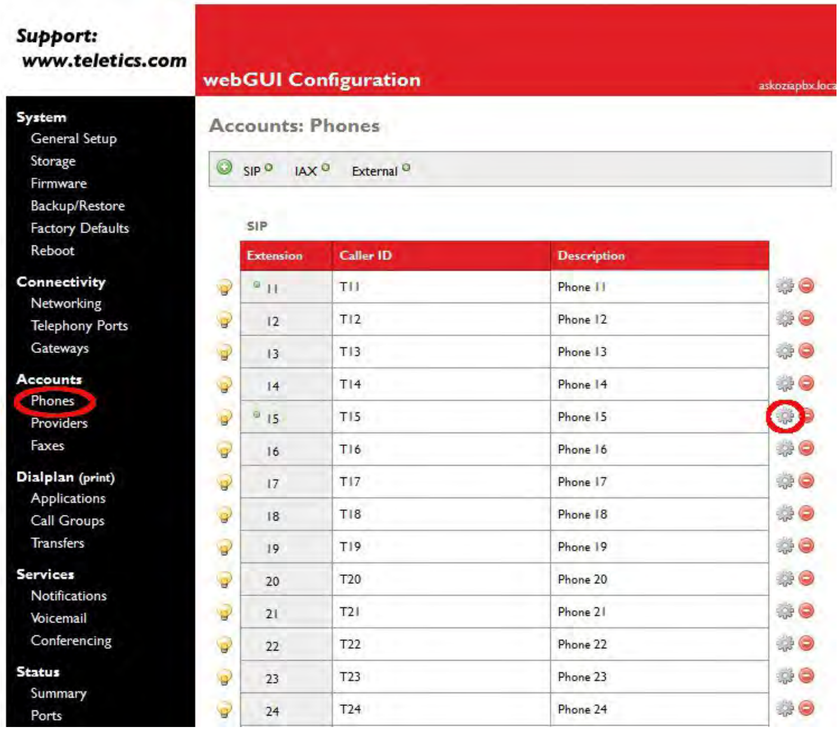
Once you have these settings entered, you will notice a SIP provider has been added on the providers main screen. You can go back and edit these settings at any time later:



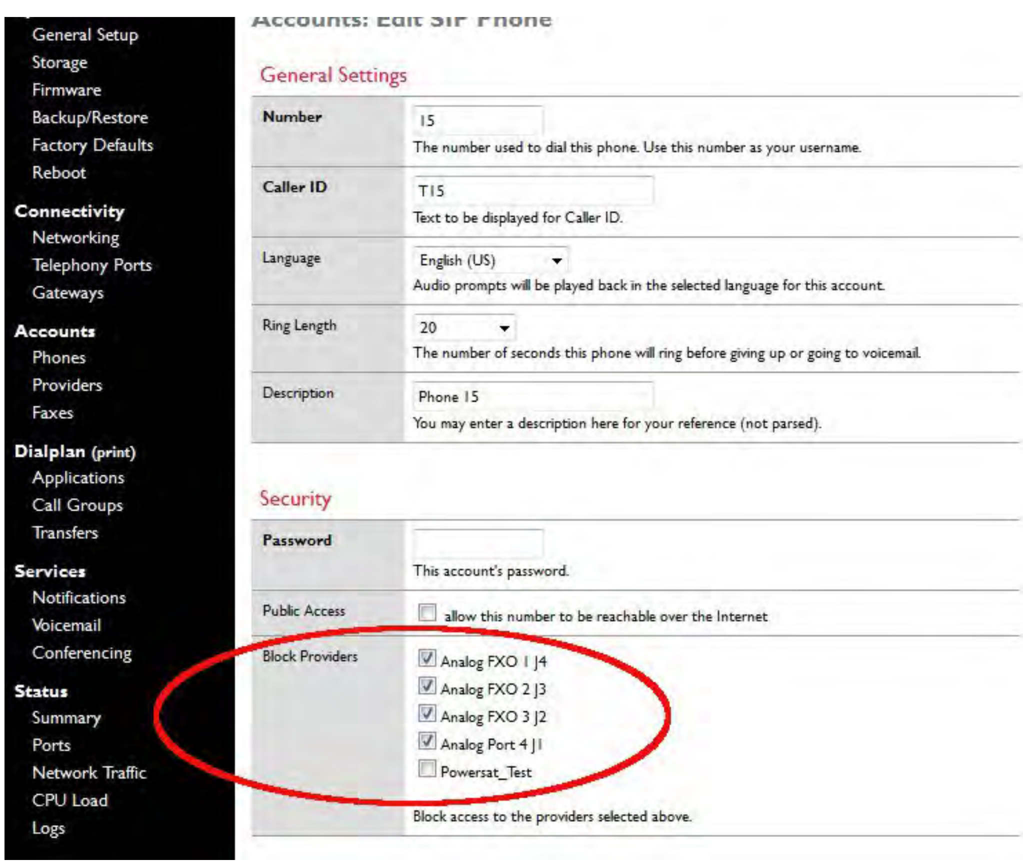
Enabling SIP Provider Account

All phones on the system can be told to use (or not use) certain providers. You need to check that the w*intercom phone you want to have use the account you just created is allowed to do so in the w*intercom phone settings.

From the main menu, select Phones, and then click on the little gear symbol next to the phone you want to use the account you just created. In our example, this will be w*intercom phone 15:

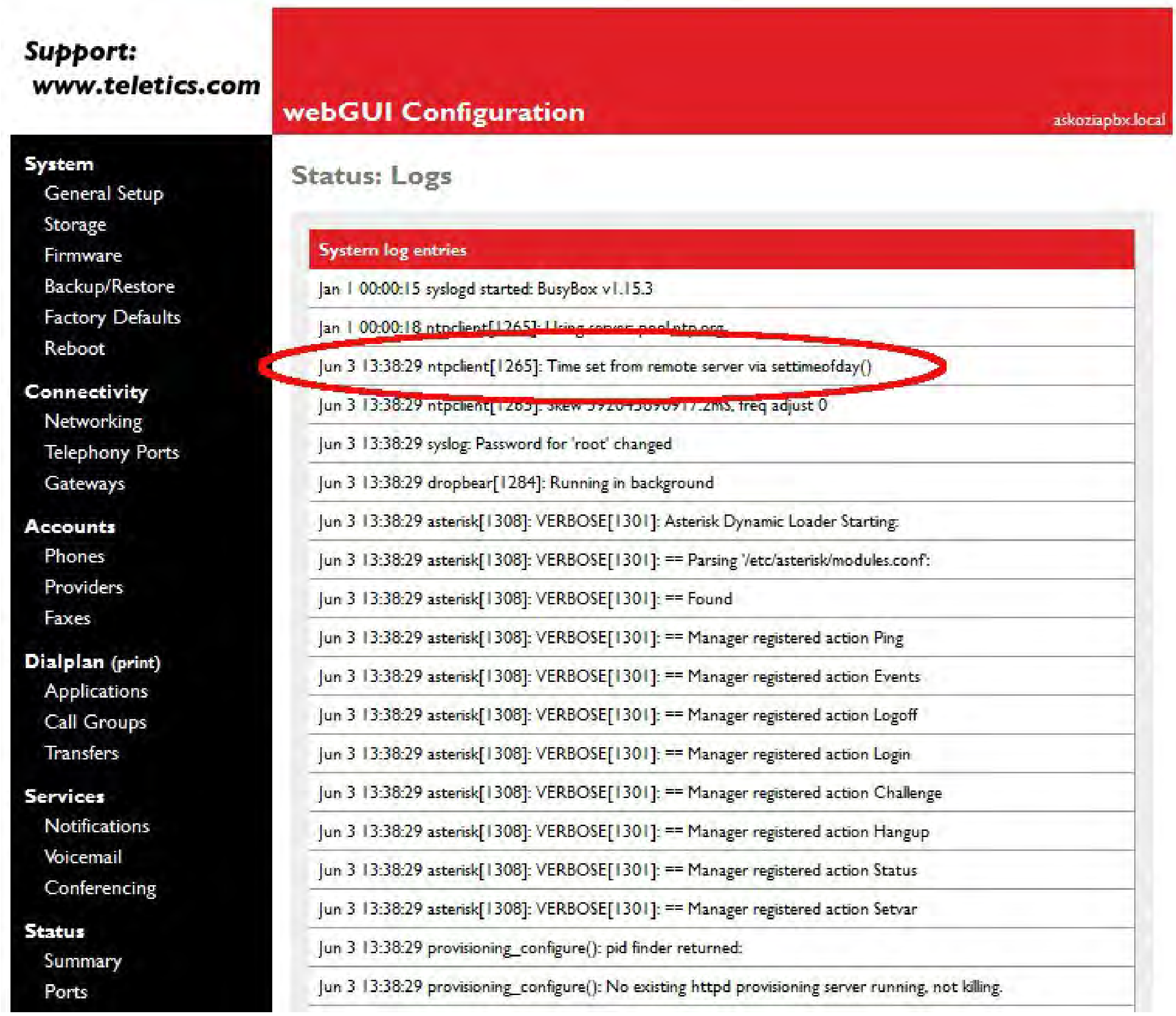


Check that the provider you just created is not blocked:



Basic Testing and Troubleshooting

First, if the Feature Server is properly getting a time setting, the **Network** settings are correct. If the date stays on January 1, you MUST fix your **Network** settings prior to anything else. Here is an example of the first few lines of the log file following a reboot. Notice that the time is correctly set in the first few lines:



When connecting to your SIP provider, you will also see a log entry that will show if an attempted connection fails. The Feature Server will also retry on an ongoing basis. Failed attempts will show the number of failed tries as well.

If SIP logins continue to fail and all your settings appear to be correct, it is recommended that you contact your SIP provider and ask for assistance.