QuestBlue Systems SIP Trunk Setup with FreePBX version 13.0.27 / Dated DEC -2015

One of the major differences in this version will be the addition of the FreePBX Firewall and PJSIP.

At QuestBlue Systems we will support you in both firewall options as noted below:

A. Built in Sangoma Firewall

B. IPTABLES firewall Access Control List /etc/sysconfig/iptables

QuestBlue Systems will not support the SIPStation Free Trial that you *may* have selected Start Trial during your install. If you selected Not Now, then you can continue below.

First steps after creating your login to the GUI will be a question of optional add-on products.

These products are completely up to you, however they are supported by Sangoma.

They are not products that QuestBlue Systems will support.

Your next screen will be the Firewall option.

All new in the version FreePBX 13 is the built-in Sangoma Firewall.

We recommend to our clients that have network / CentOS knowledge to avoid this firewall and use IPTABLES as your firewall. This is an option that you can make at this time.

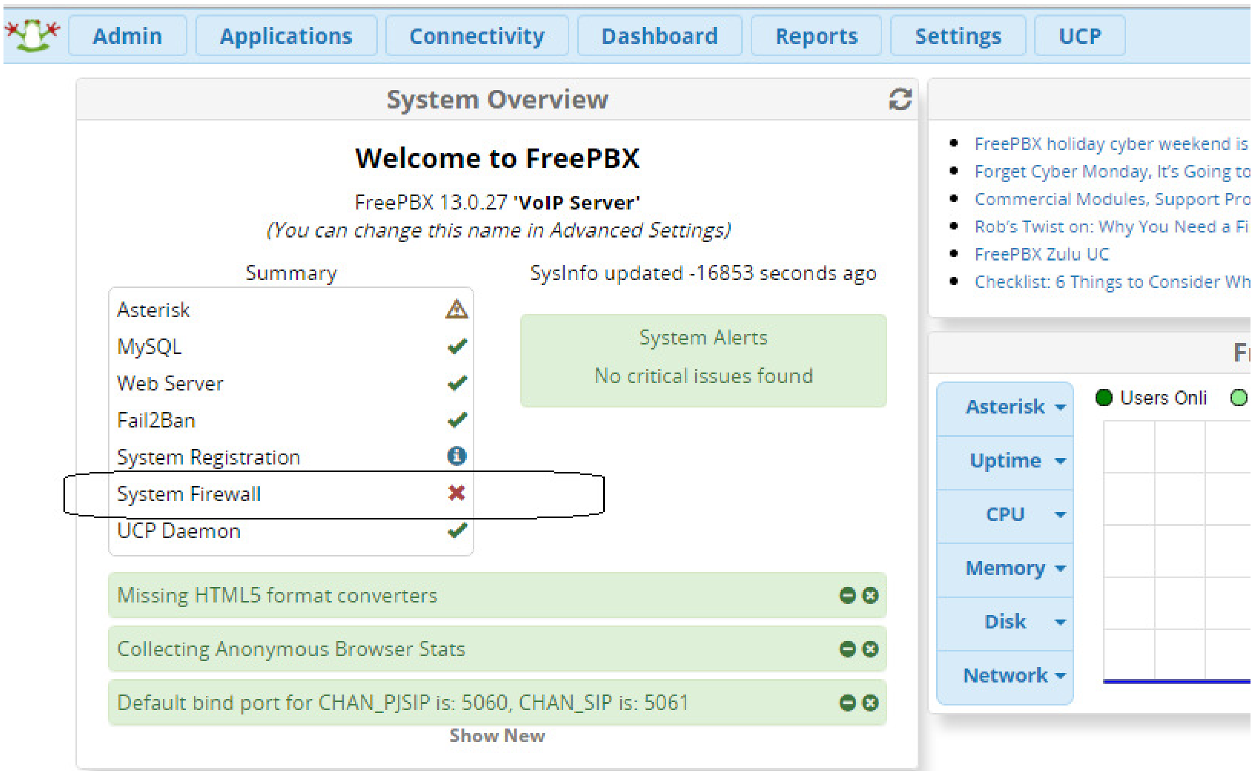
We have selected to ABORT the built in Firewall from Sangoma.

Once you have navigated to the next screen of the iso installation you will notice a sales question asking you to purchase SIPStation Free Trial.

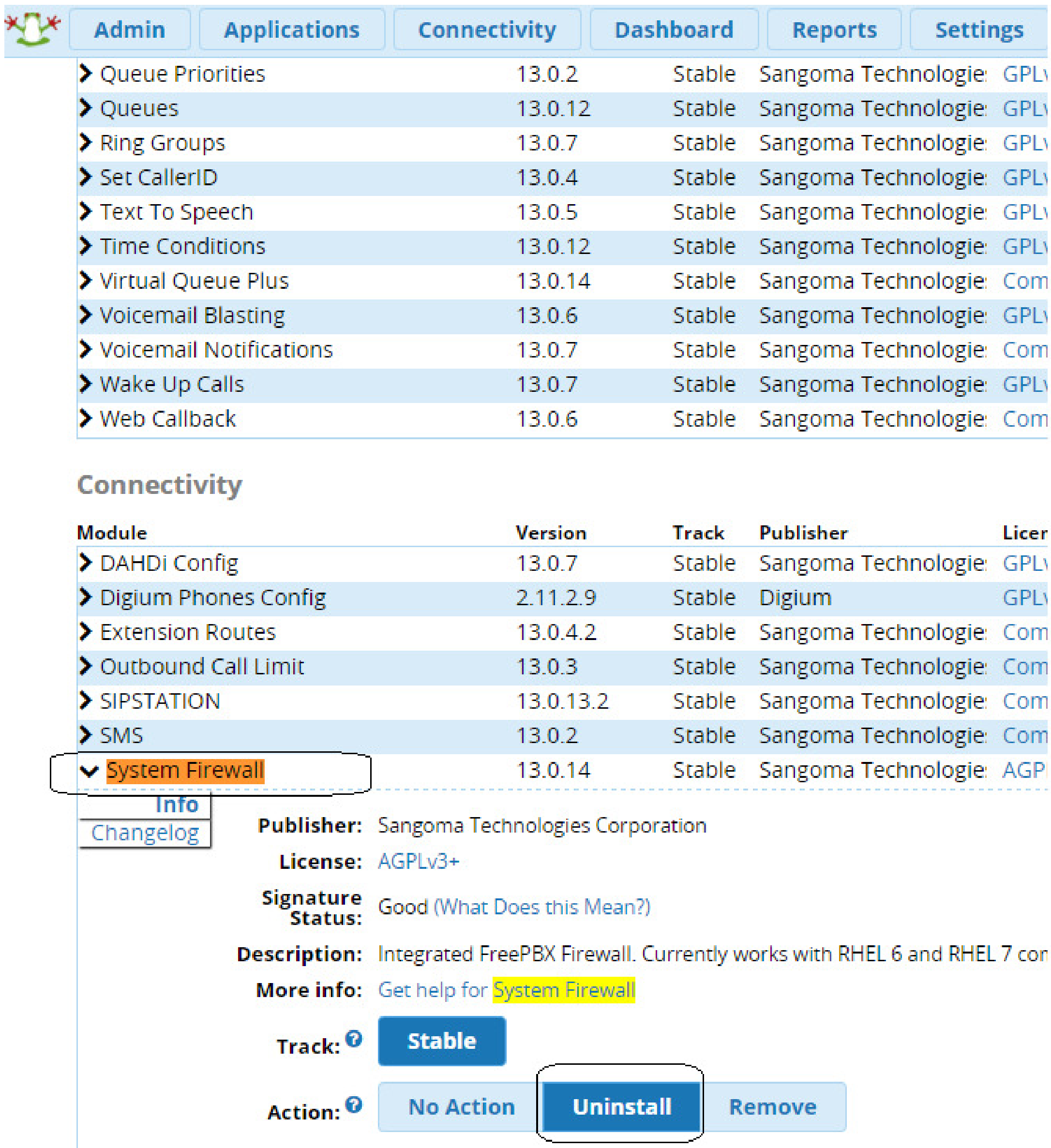
At QuestBlue Systems we do not support SIPStation in your FreePBX, we only support the SIP Trunks associated to QuestBlue Systems and interconnected systems over iax2.

Our suggestion is to select abort when asked.

Because of our selection of the Firewall we will need to address an error that is on the main dashboard.

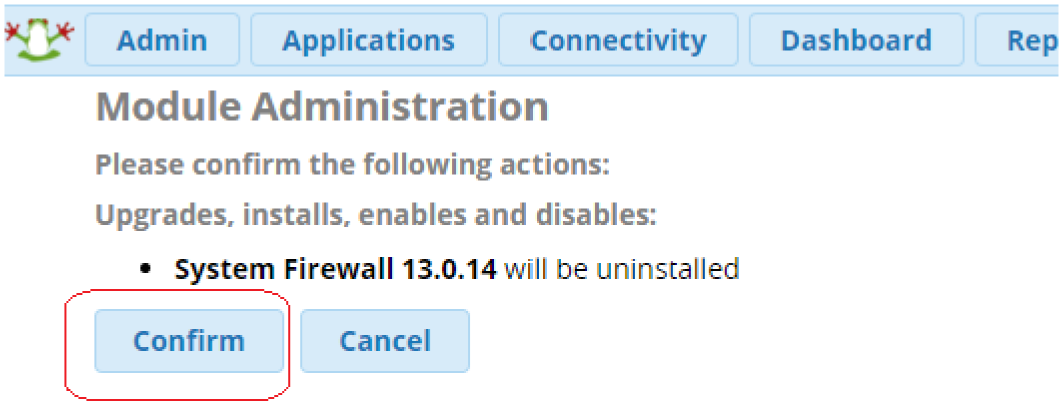


To correct this error, please go to Admin -- Module Admin and locate System Firewall under Connectivity

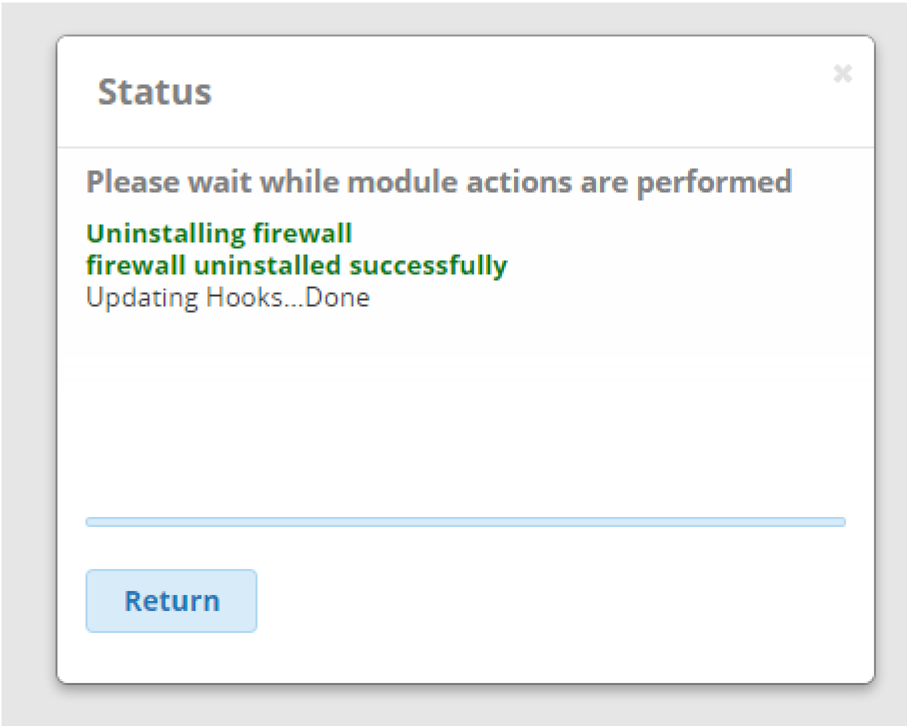


After making your selection above you will need to process your selection

Please confirm your selection



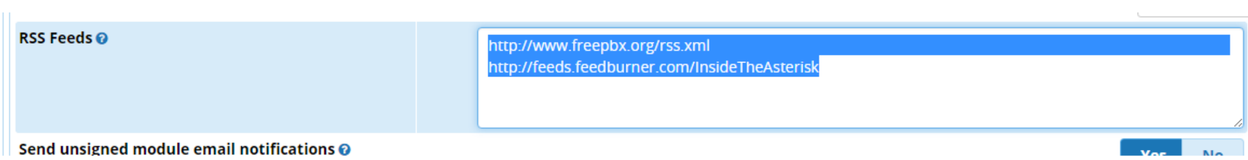
Next you will select Return and then Apply Config in the top right hand corner.



We will now move on to the Setting Advanced Setting section.

Here we will only make one modification to this area.

We want to remove the RSS Feeds from the main dashboard to avoid any bandwidth being used for anything not VoIP related.



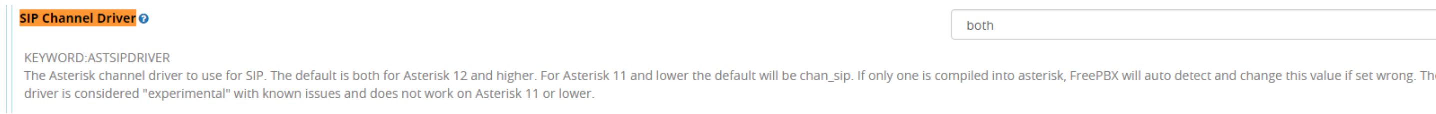
Remove all the text in this box and select No

Then Select SUBMIT on the bottom right of your screen along with Apply Config in the Top Right when prompted.

Now if you click on Dashboard across the top you will return to a clean Dashboard.

The first task we need to address is setting the PBX to CHAN_SIP only, not BOTH.

Navigate to Settings - Advanced Settings and locate SIP CHANNEL Driver and change it from both to CHAN_SIP

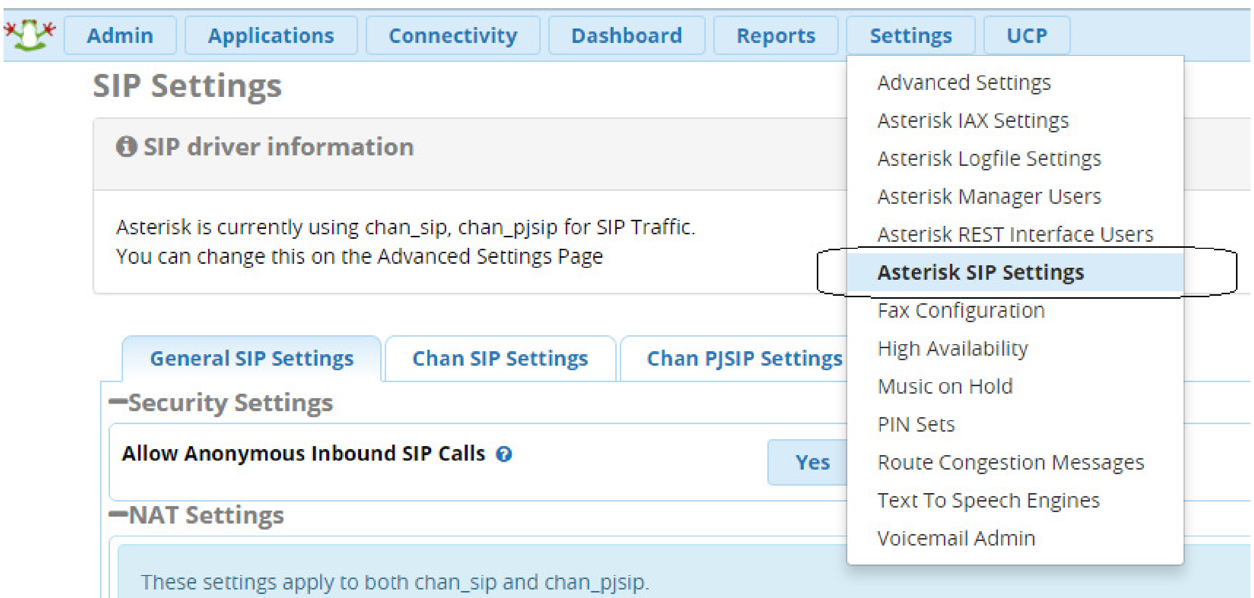


The chan_pjsip channel driver is considered "experimental" with known issues and does not work on Asterisk 11 or lower.

Please make your selection to CHAN_SIP now in the drop down.

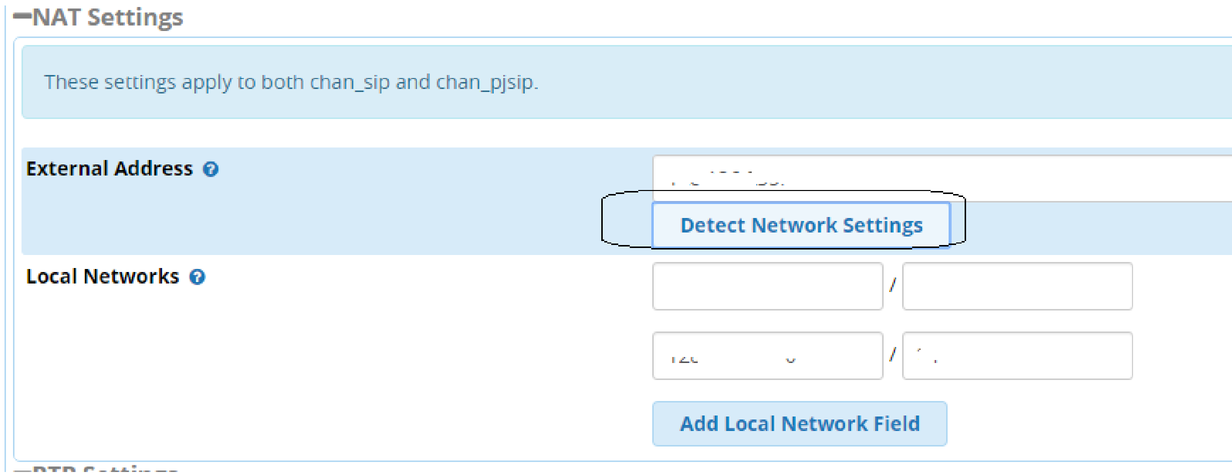
On the bottom of your screen you will need to press Submit then APPLY CONFIG up top in RED.

Please navigate to Settings - Asterisk SIP Settings.



Here we will make the following updates.

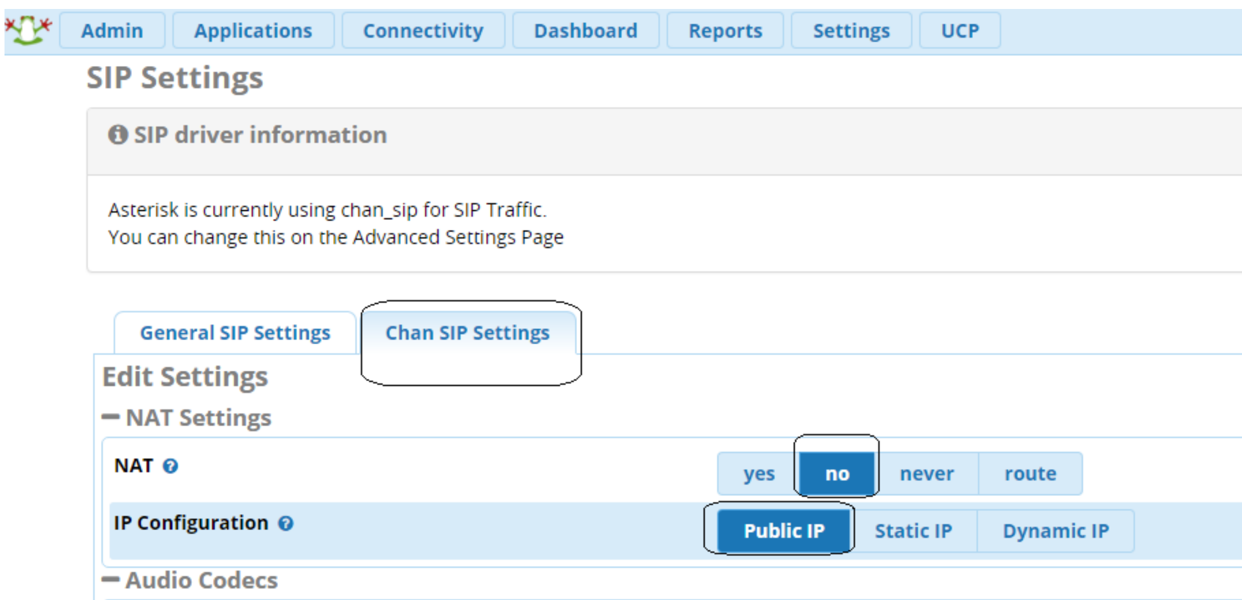
On this page the first item we want to address is the NAT Settings. In our setup we are on a public IP Address that we are protecting with iptables firewall.



In the NAT Settings please select Detect Network Settings.

** It is important that these fields fill in automatically because that will tell you that your network is setup properly at this point of the setup.

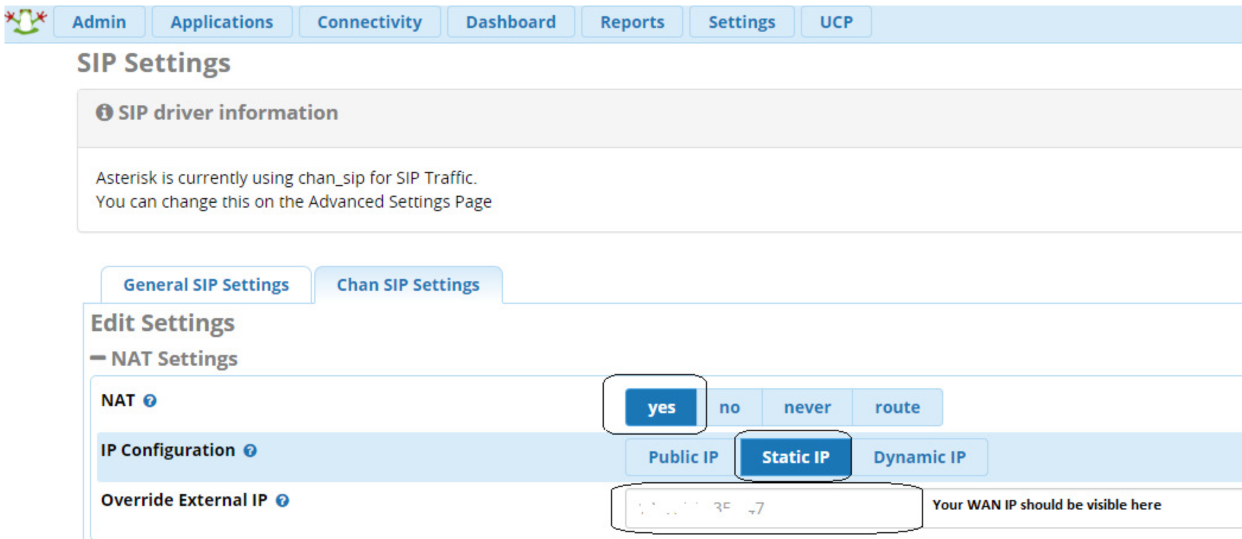
After Submitting your changes and Applying Config please select Chan SIP Settings Tab.



If you are configuring on a Public IP then you can easily make the following choice along with the settings above.

Navigate to the bottom right and select Submit, then Apply Config

If you are running your system on an Internal IP that is using NAT then your selection will be as following:

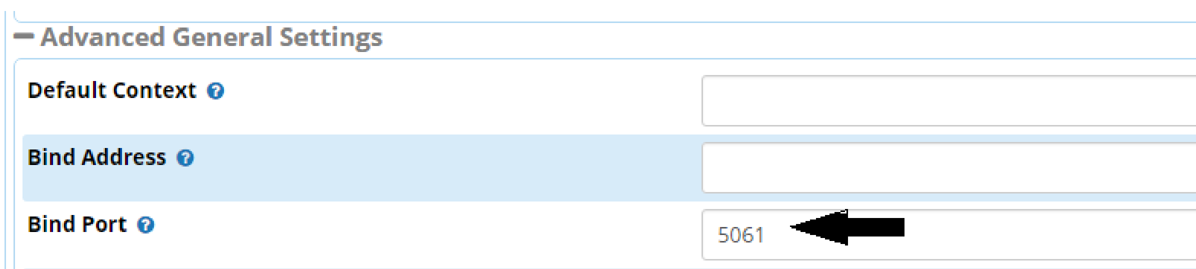


You will need to TYPE IN YOUR IP> It is hinted in the box already, simply retype it.

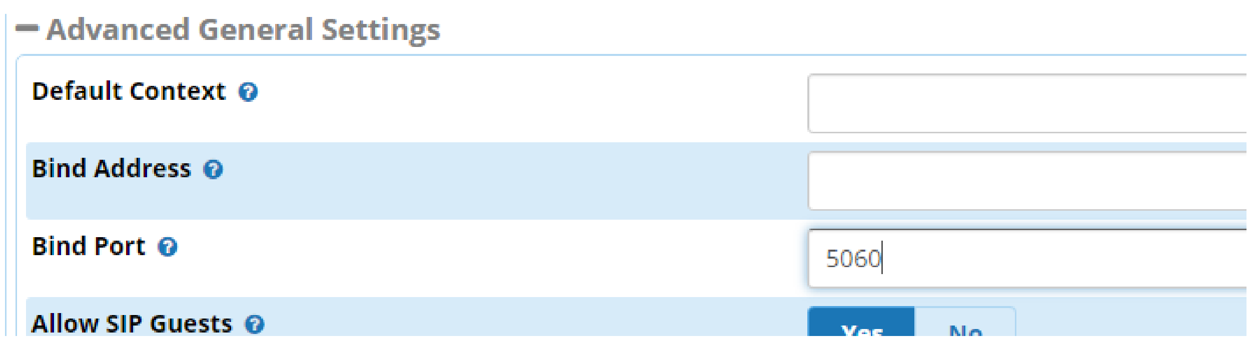
Next the most important step in the CHAN_SIP Driver Setting is making sure that your system is on port 5060 and not 5061 that the default will be set to.

On this same page you are working with already scroll down until you locate:

Advanced General Settings

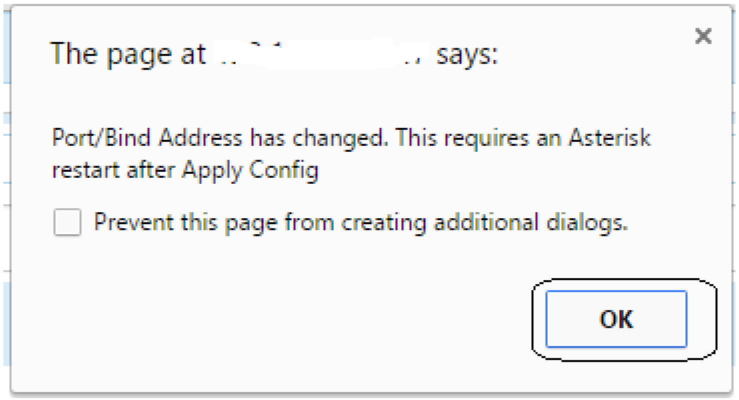


You are required to change this setting FROM 5061 to 5060 as shown below



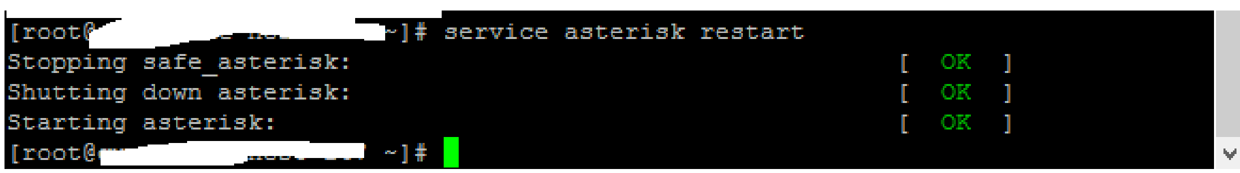
Please Submit Changes. You will notice a message box pop up with a very important message.

The message box clearly states that you must restart Asterisk after Apply Config is pressed.

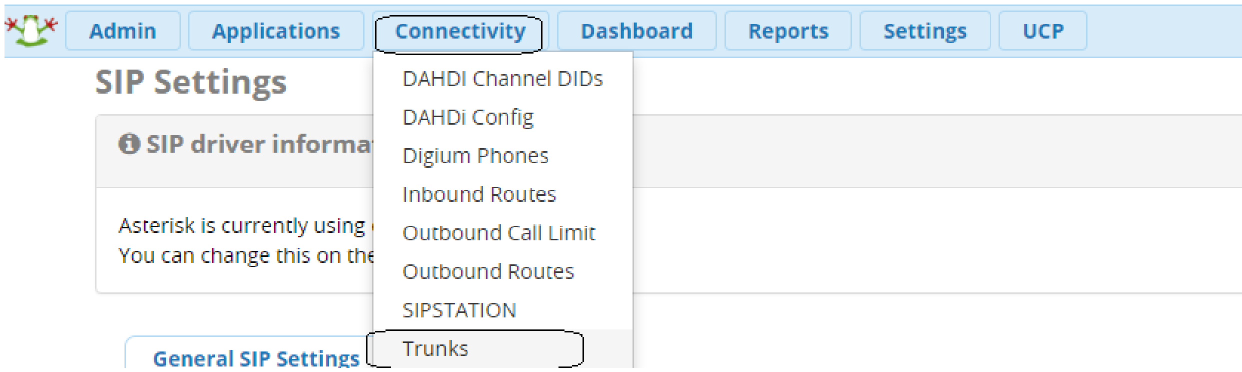


Please select OK and then continue on to Apply Config

Now, from your command line on the system run the command: service asterisk restart

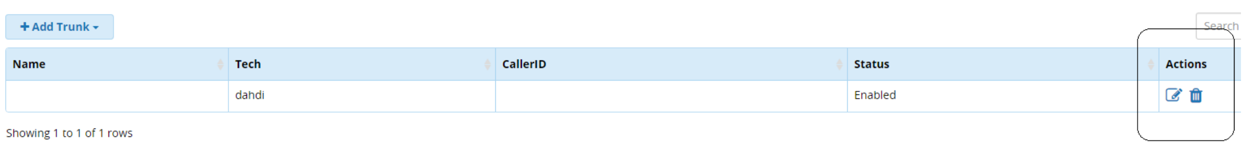


We are now ready to create the SIP Trunk.

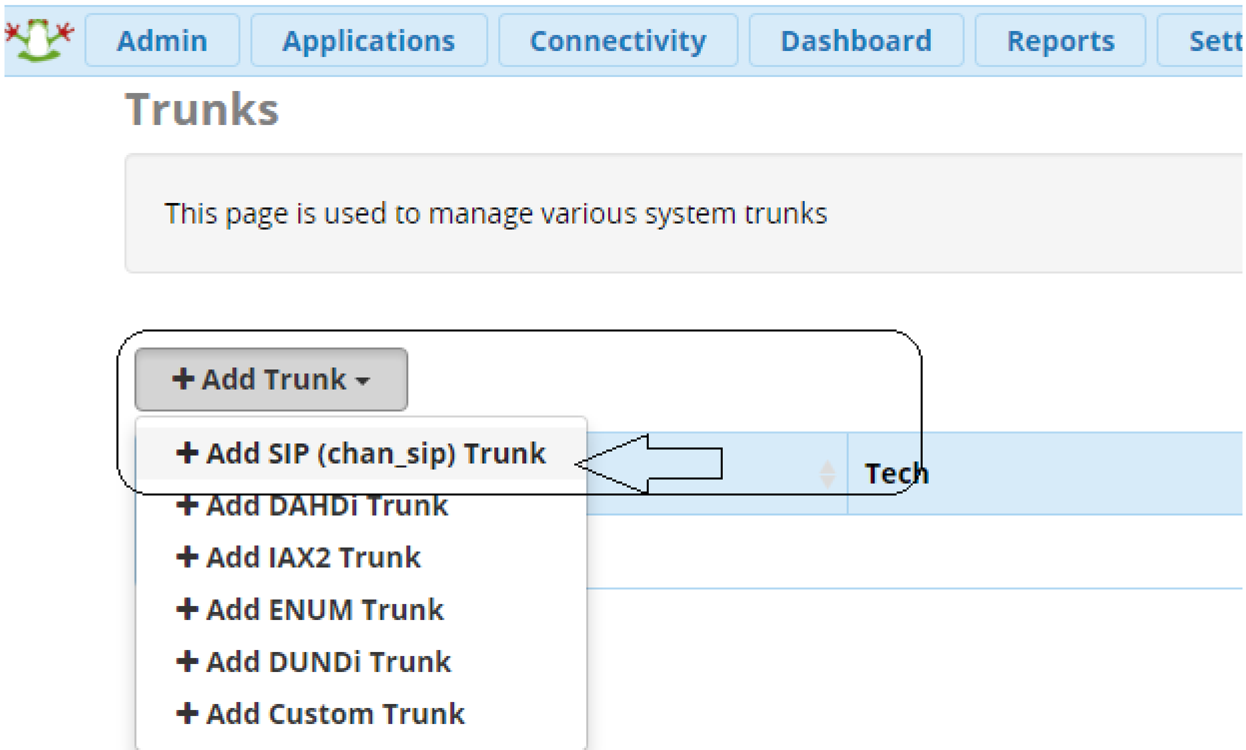


Navigate to Connectivity then Trunks.

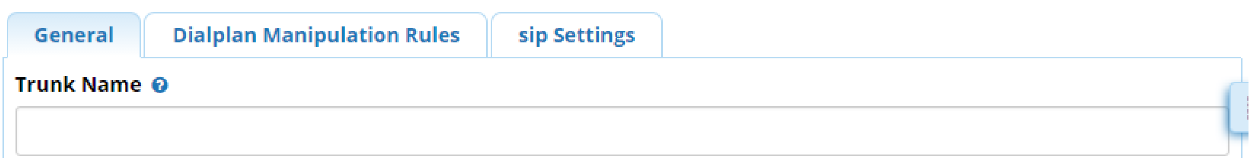
Feel free to delete / trash can the dahdi trunk. It will not be needed for your SIP Trunk.



When you select Add Trunk please make the following choice. Add Trunk - Add SIP (chan_sip)

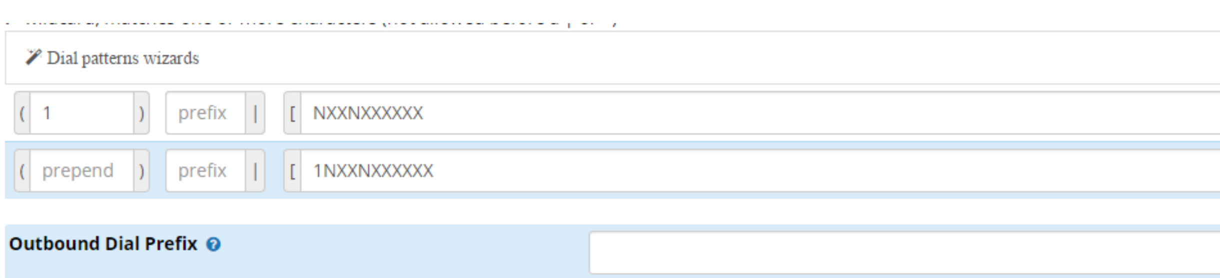


Under the General tab fill in the Trunk Name



Click on Dialplan Manipulation Rules

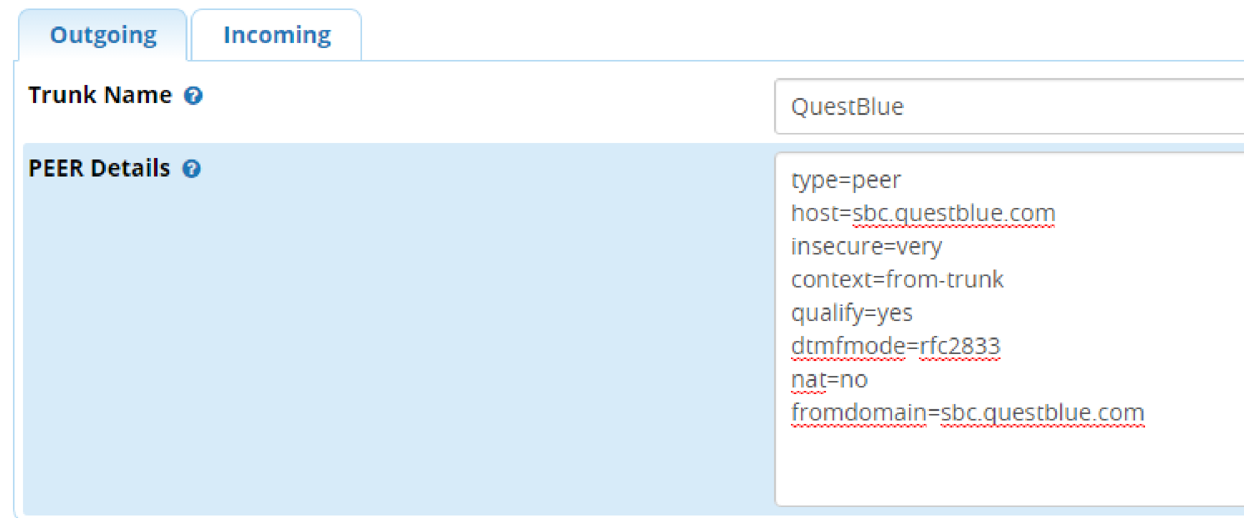
Enter the following rules:



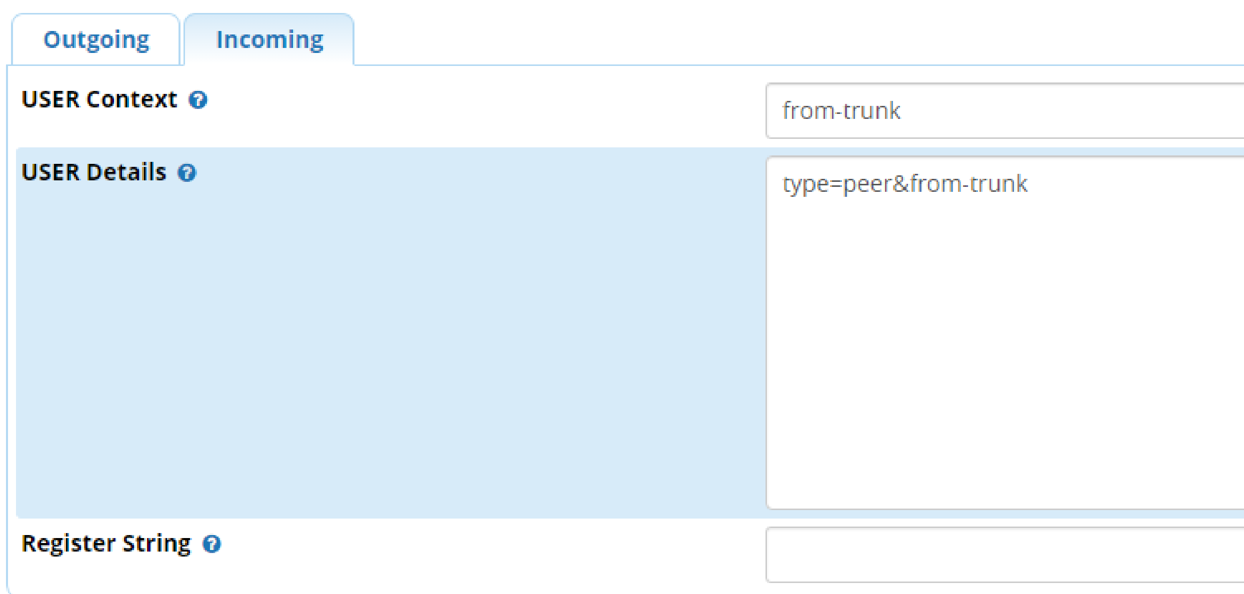
Feel free to add rules that you need for your PBX

Note: In the above dial patterns you will cover 011. (International Dialing) 1NXXNXXXXXX Dialing is equal to you dialing a number with the 1 + area code plus subscriber number and finally a prepended 1 then NXXNXXXXXX and in this case if you do not put the 1 in the outbound call the system will add it for you.

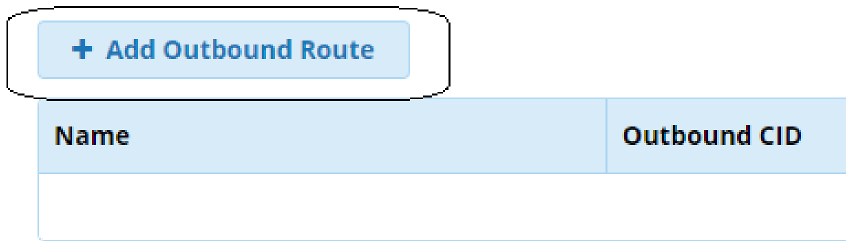
SIP Settings for the trunk to be able to connect to the SBC should be listed as the following:



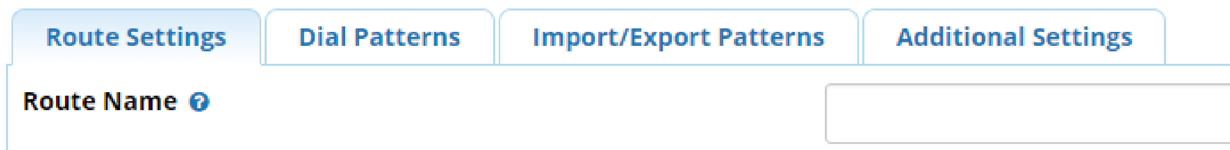
Note NAT settings are based on whether you are behind a firewall



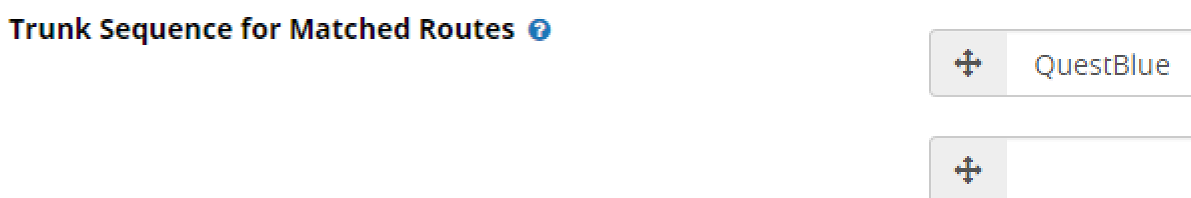
The next step will be to add an outbound route under the connectivity tab



Enter a route name



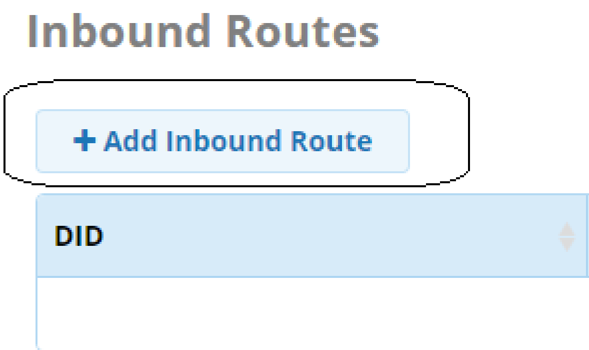
In the same tab, go down to Trunk Sequence for Matched Routes and select the trunk you made



Enter the dial patterns

Now Submit the changes

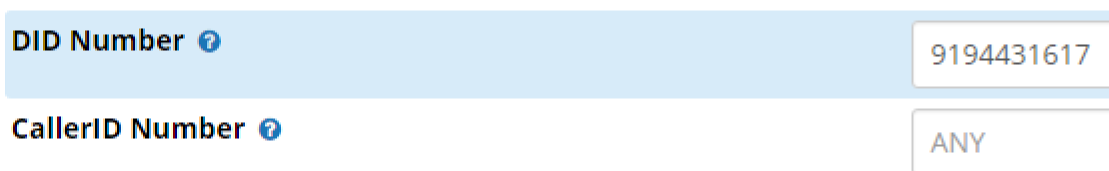
Now we can add an inbound route



*Please note, you should have an extension created prior to creating your first inbound route

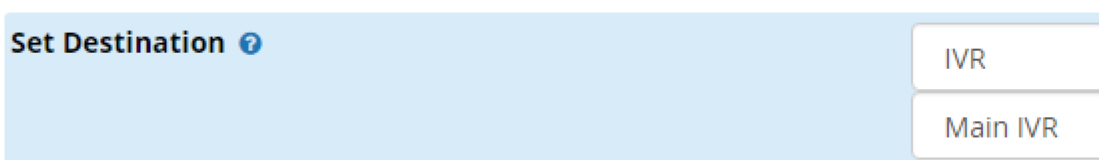
It is recommended to put a description for the DID for management reasons

Enter the 10 digit number with no spaces or characters in the DID Number field



Before creating your first inbound route you should have an extension or destination in the PBX

The next step is to add a destination for the inbound route. The destination will be where the call goes when someone calls inbound to your number



Now Submit the changes

You should now be able to make and receive calls!

The important steps to setting up your PBX with the basic set up are now complete

However, there are a few important steps left to setting up the PBX

Go to the Asterisk SIP Settings page

Under CHAN SIP Settings enter the following:

Now Submit the changes