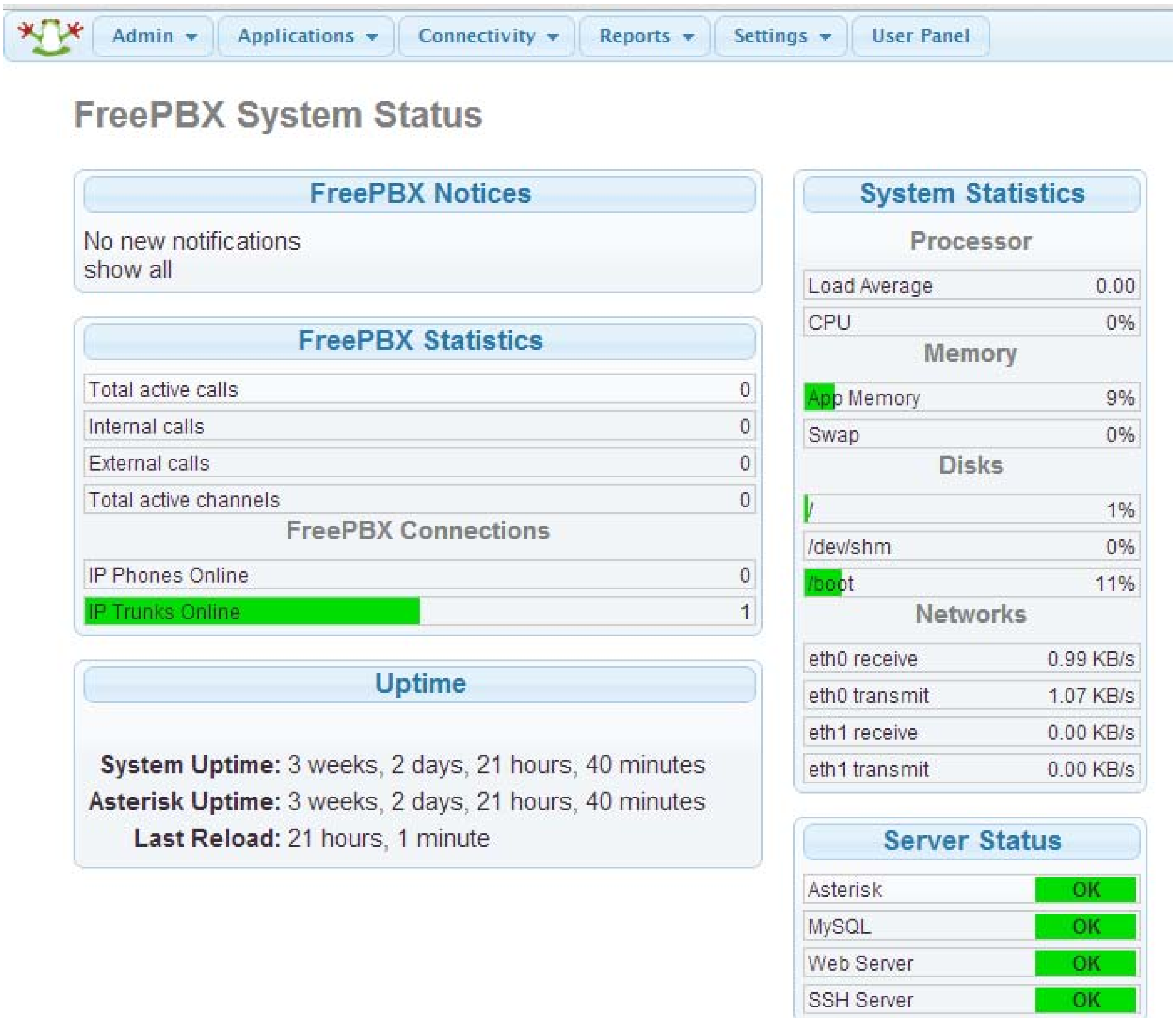
FreePBX SIP Trunk Setup

Once you have installed and created your login to the FreePBX GUI you should see a screen similar to this:

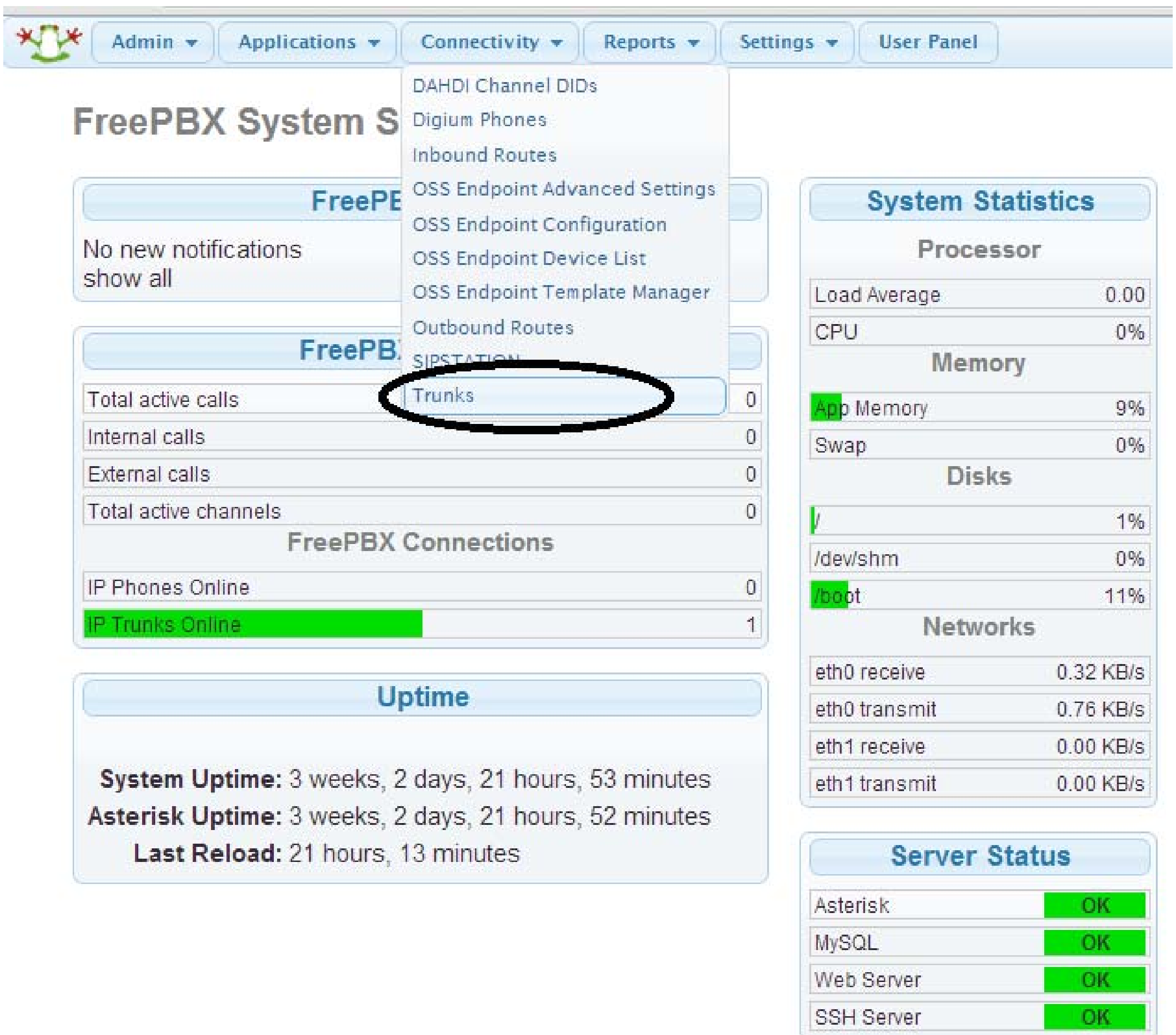


In order to process call to the PSTN network via our SIP Trunk you will need to address a few items first.

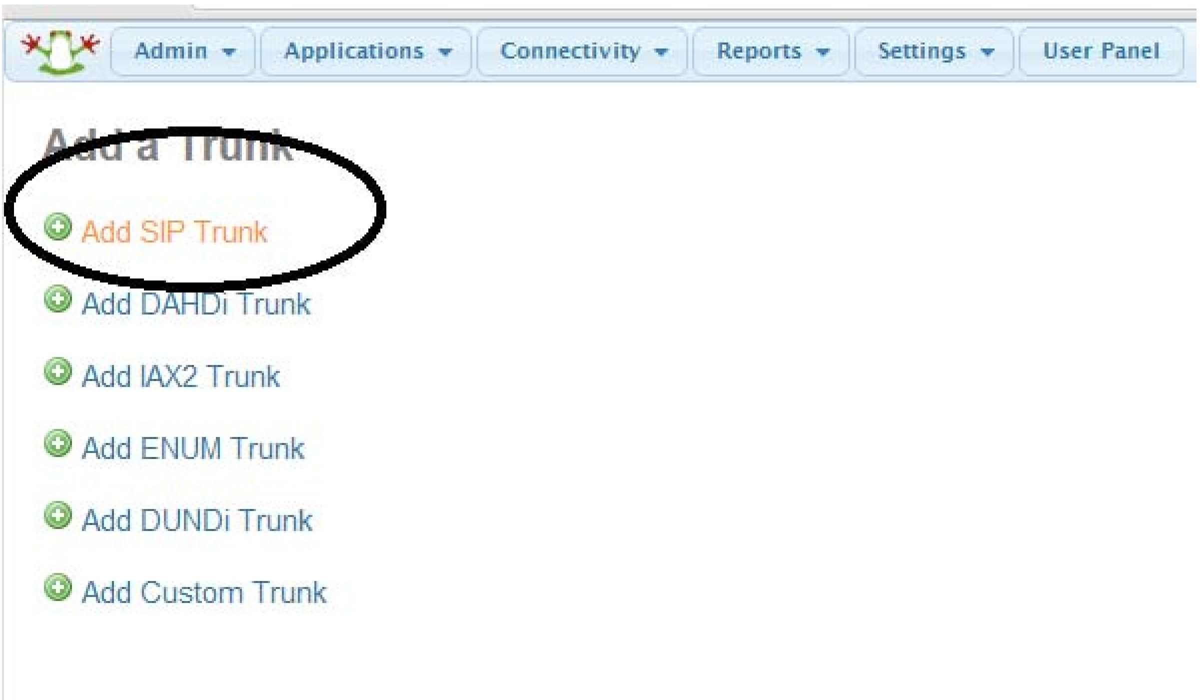
1. **Setup your SIP Trunk in FreePBX**
2. **Create Outbound Routes**
3. **Create Inbound Routes for purchased or ported in phone numbers to the QuestBlue Network.**

<u>First we will cover the setting the SIP Trunk</u>.

1. Navigate to Connectivity then Trunks as show in the illustration.



2.Once you select Trunks you will be able to create a SIP Trunk from the following page:



Next you will be required to fill in the settings for the QuestBlue SIP Trunk.

We will cover this in 4 steps.

A. Naming the SIP Trunk

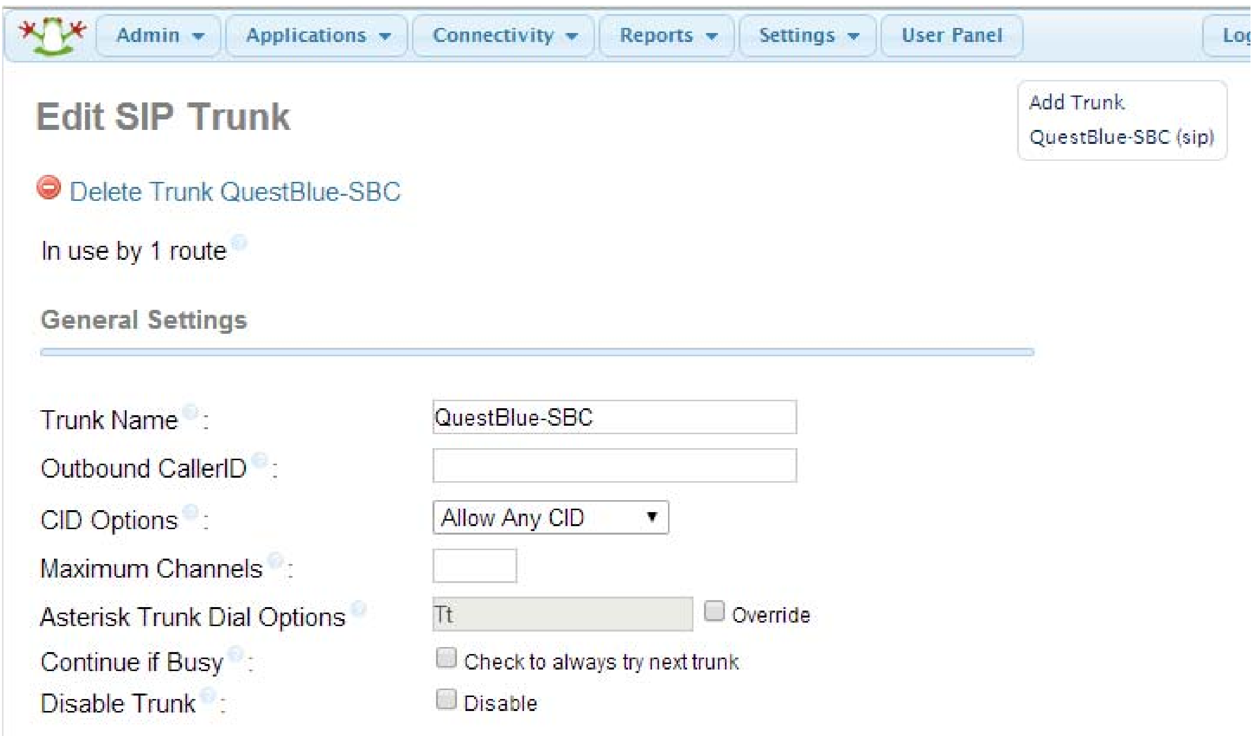
B. Dialed Number Manipulation Rules

C. Outgoing Settings

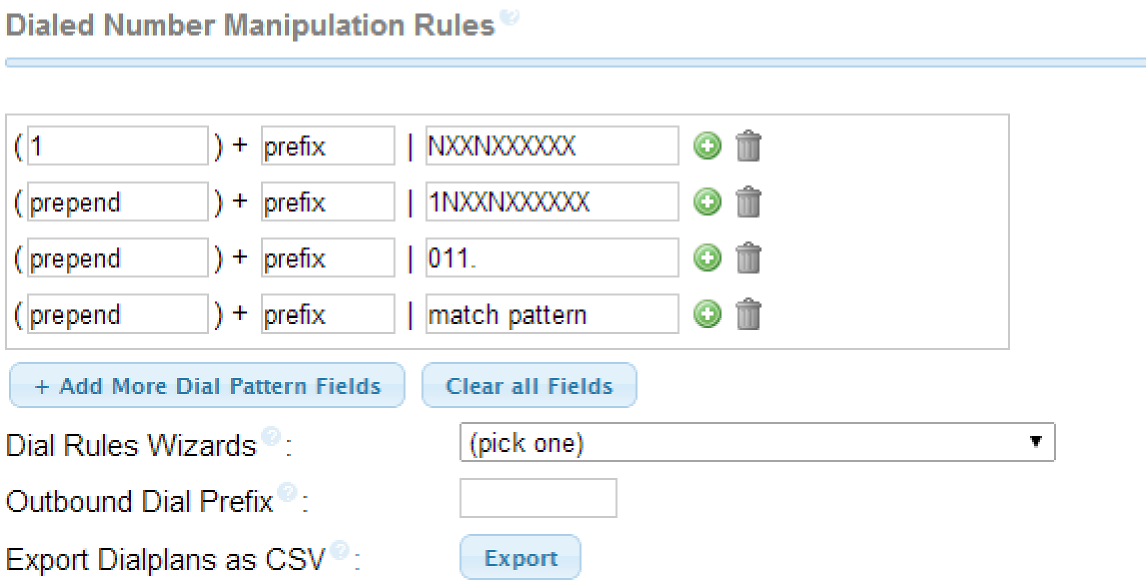
D. Incoming Settings

Setting up the Trunk: From the TOP

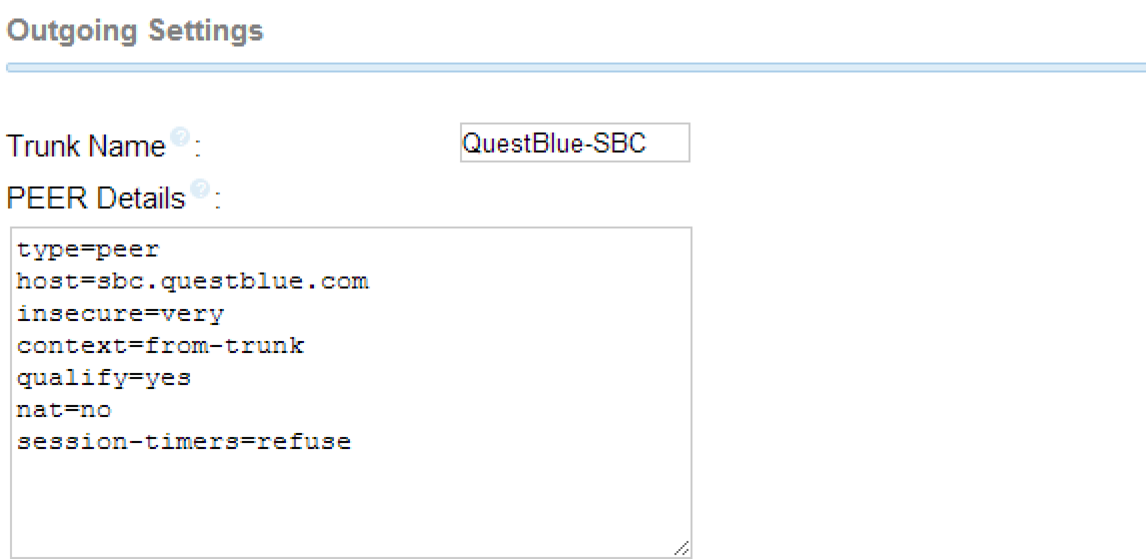
A. First we will name the trunk and provide some information. Name the trunk QuestBlue-SBC



B. Next we will provide the rules for what will match a pattern and the trunk will process the call based on the matches below



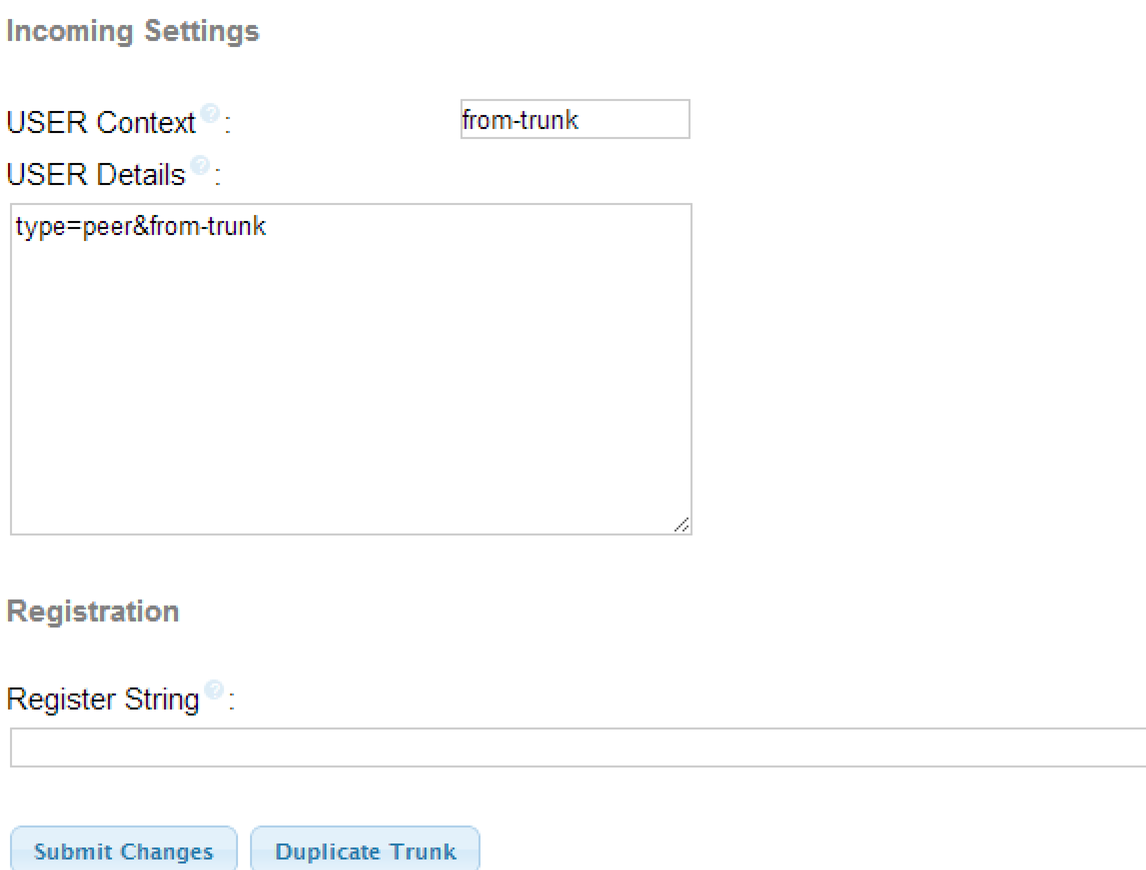
C. Next we will cover the Outgoing settings for the SIP Trunk



It is very import to make the correct setting on the nat=no or nat=yes in the above example.

In the example above the FreePBX is on a Public IP address with a very tight iptables setting for security. IP Tables are covered in the DOCs area of the User Portal

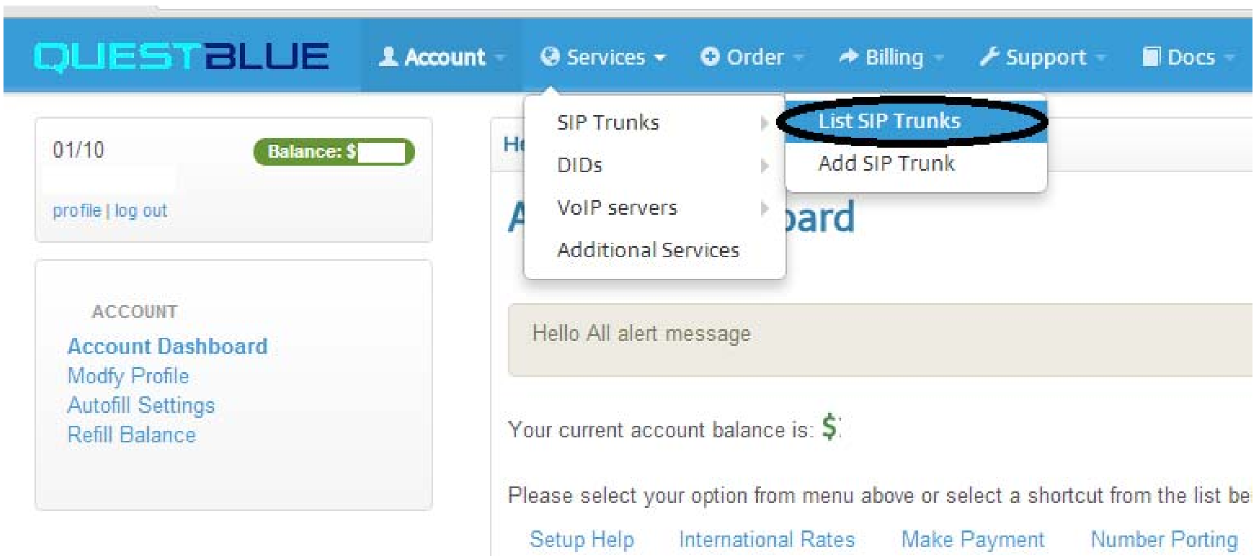
D. Next we will cover the Incoming settings for the SIP Trunk



Press Apply config in red below .

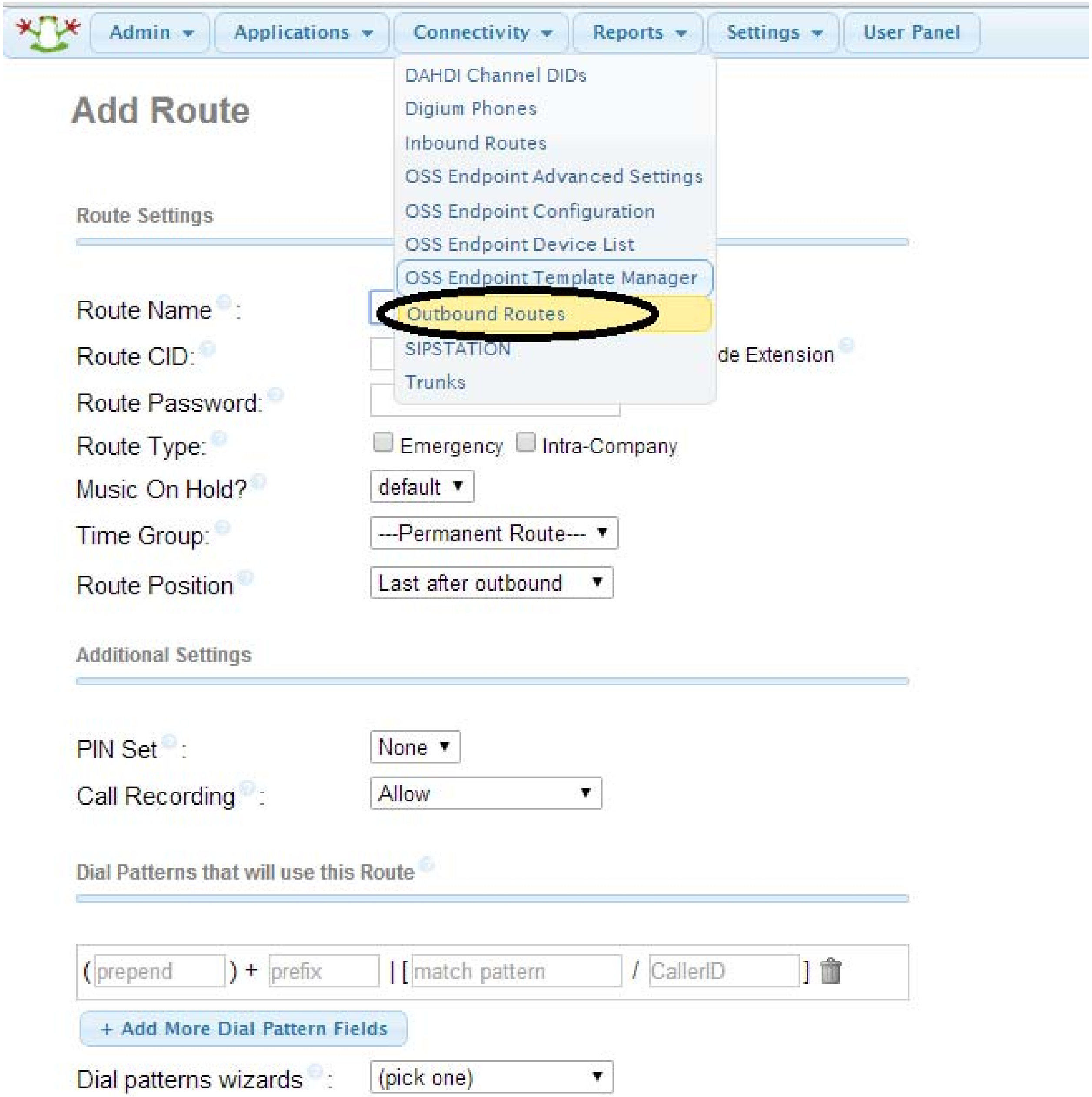
You will see above that we do not register our SIP Trunks. For added security we only allow IP Authentication.

The servers WAN IP Address needs to match what you entered in the user portal under the Services Tab

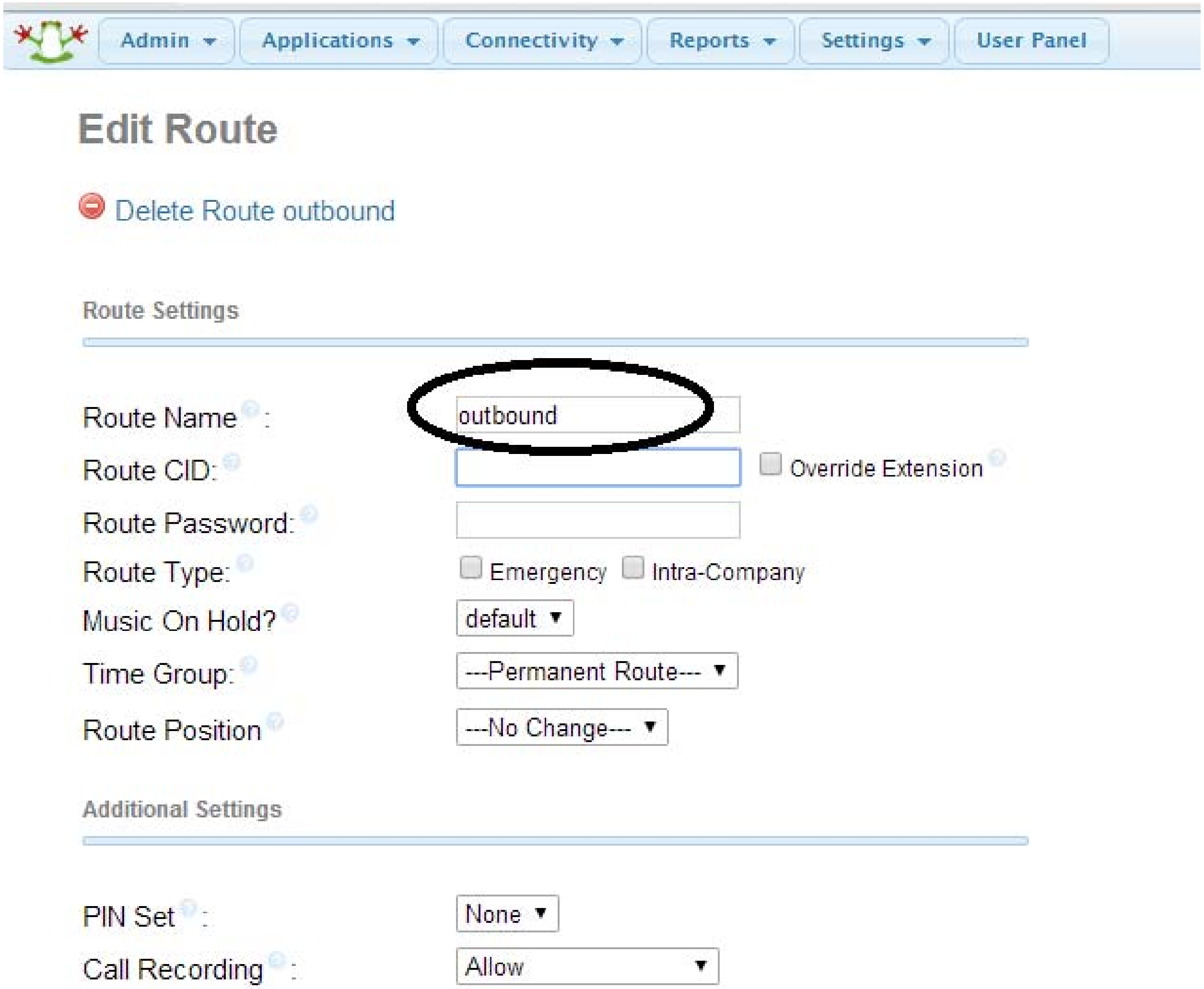


<u>**Outbound Routes**</u>

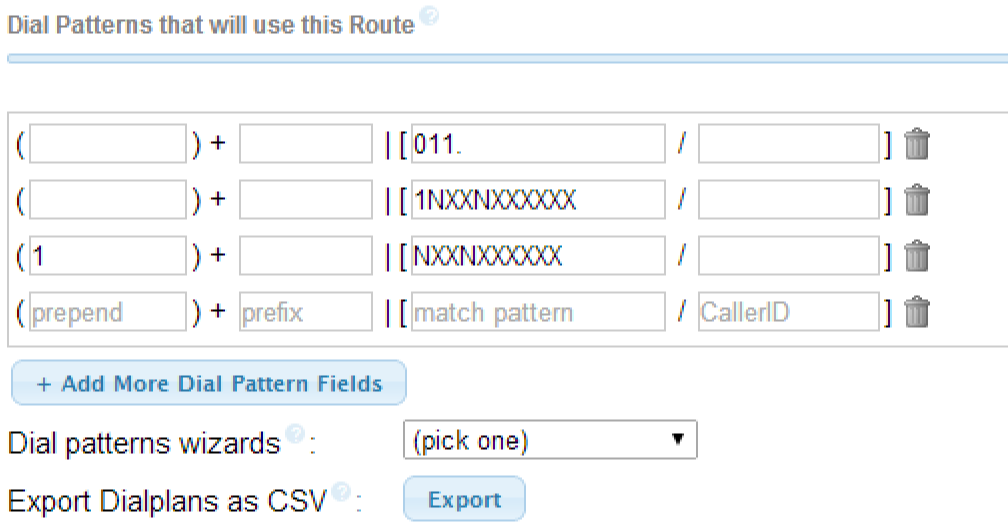
2. Create Outbound Routes by navigating to Outbound Routes from the Connectivity Tab

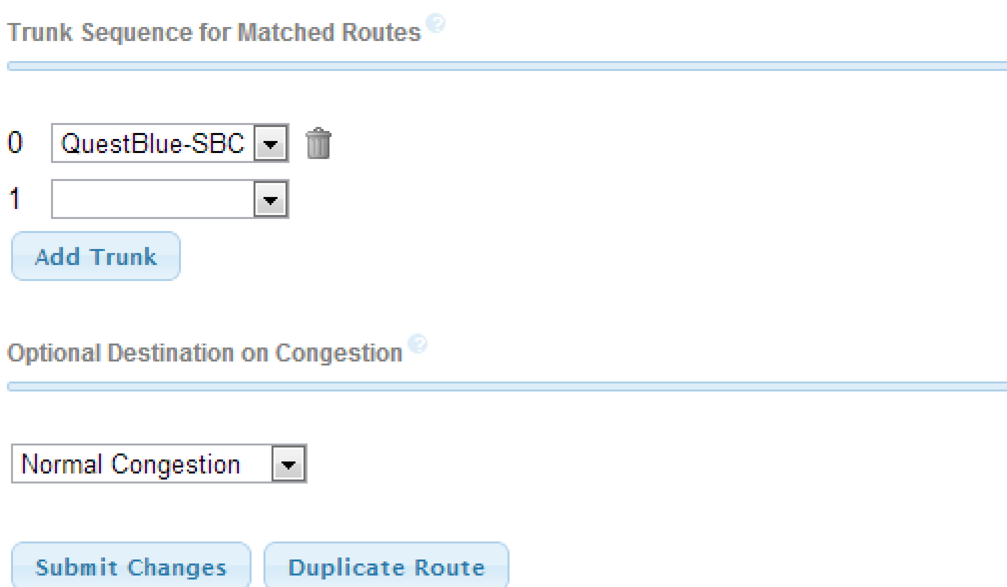


Click on Outbound Routes to navigate to the following screen. Name the Route Name: Outbound under Route Settings



Next you will create the patterns that need to be matched





Once you hit the submit button you will be prompted to the next page with apply config on red press the apply config and your outbound route will be created.

Note:

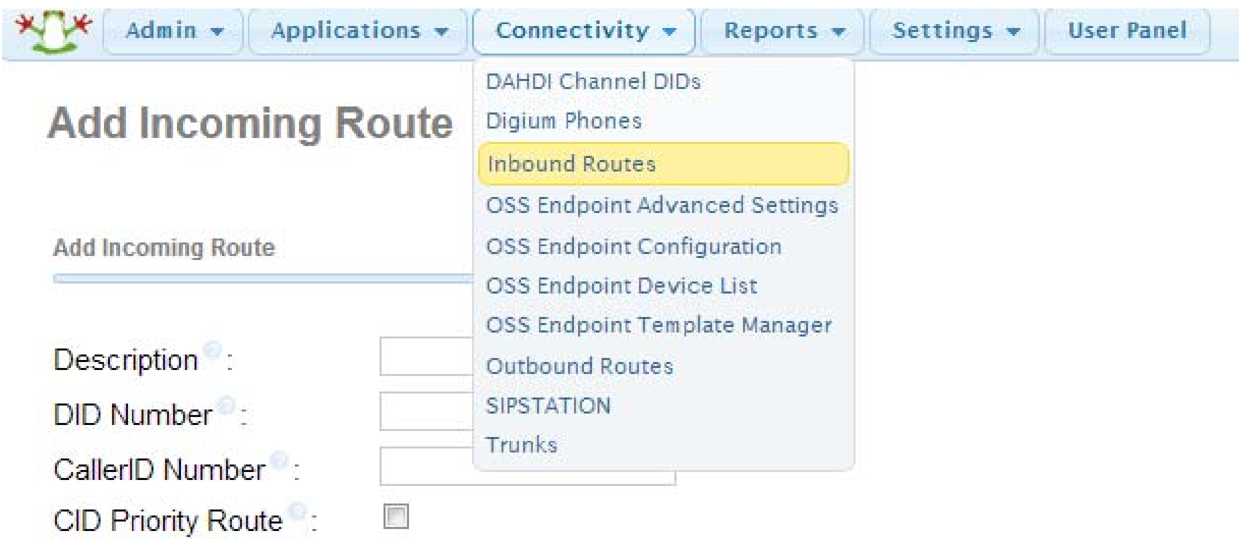
In the above dial patterns you will cover 011. (International Dialing) 1NXXNXXXXXX Dialing is equal to you dialing a number with the 1 + area code plus subscriber number and finally a prepended 1 then NXXNXXXXXX and in this case if you do not put the 1 in the outbound call the system will add it for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

Create Inbound route.

Note:

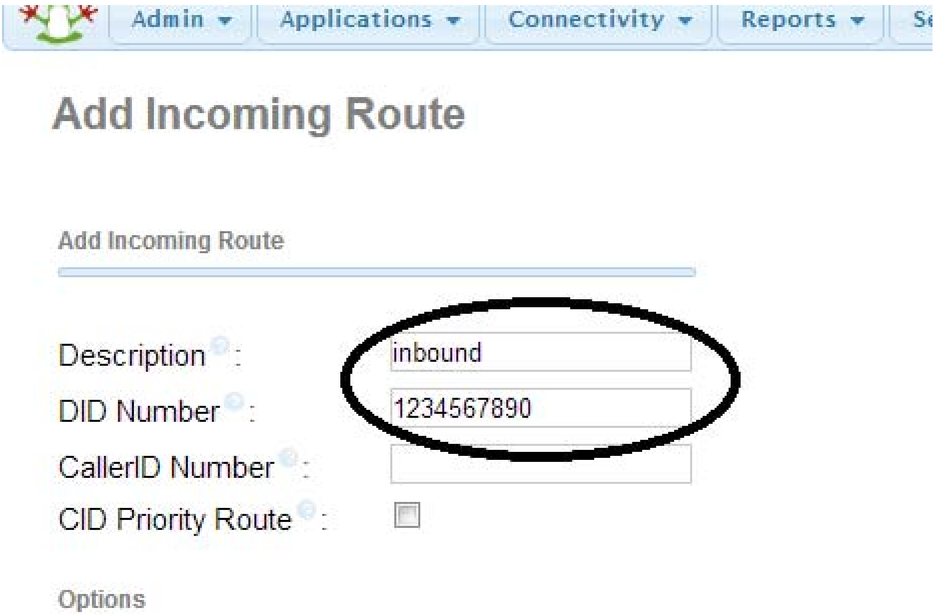
You can create Extensions, Ring Group etc before or after you create the inbound routes.

1. Go to Connectivity and from the drop down menu select Inbound Routes as shown below.

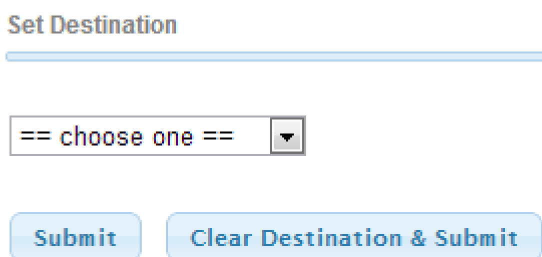


2. Once you are in the Inbound Route Page select Add incoming Route as shown below.

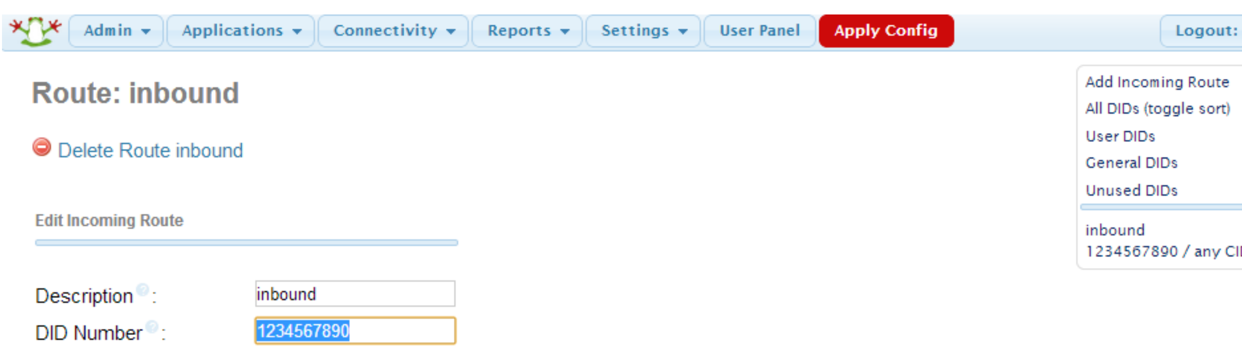
3. Fill in the Description and the DID Number.



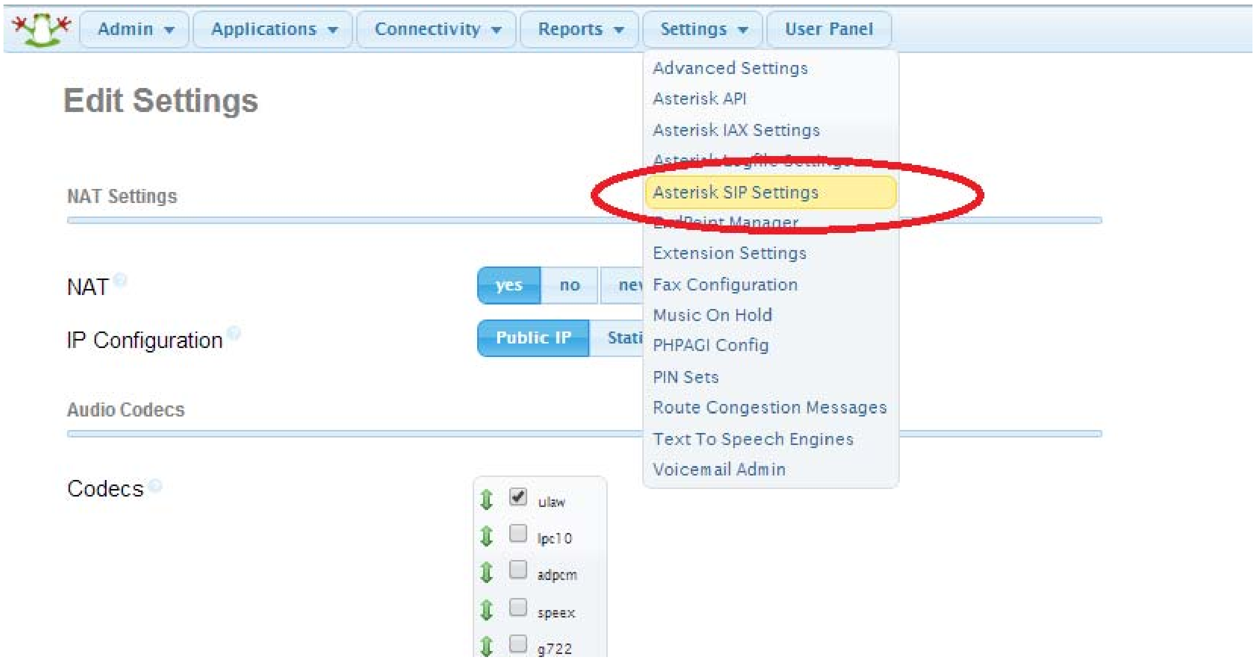
4. Choose the Destination from the drop down menu and Press Submit.



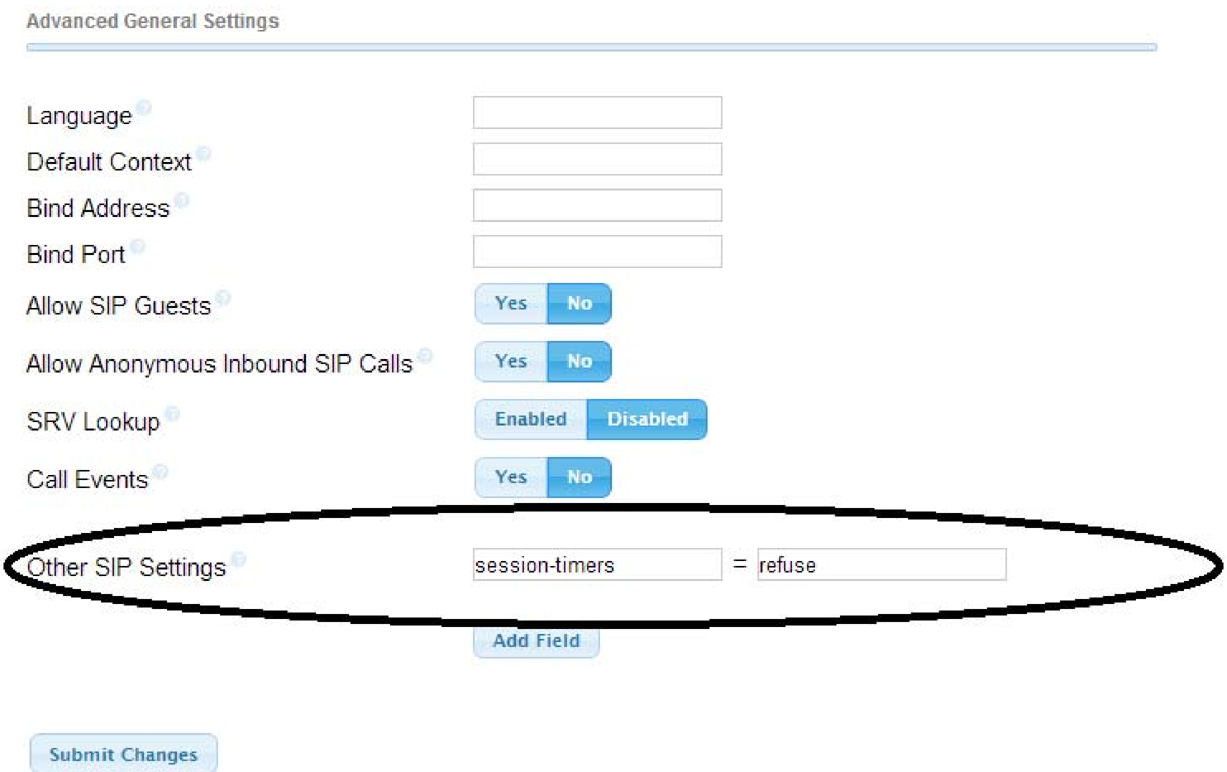
5. Press Apply Config button on red and your Inbound Route will be created.



In addition to creating your Trunk and Inbound/Outbound routes there is another important issue that you should address. Go to Setting the Asteisk SIP Settings



At the bottom of the Asterisk SIP Setting page you will locate the filed to put Other SIP Settings.



In the field please put (session-timers = refuse) as show.

As always remember to apply the changes up top in the navigation menu: