

# FreePBX SIP Trunk Setup

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

The screenshot displays the FreePBX System Overview dashboard. At the top, there is a navigation menu with 'Admin', 'Applications', 'Connectivity', 'Reports', 'Settings', and 'UCP'. The main content area is divided into several sections:

- System Overview:**
  - Welcome to FreePBX:** FreePBX 12.0.51, Sysinfo updated 1428335937 seconds ago.
  - Summary:**

Asterisk	🔥
MySQL	✅
Web Server	✅
UCP Daemon	✅
  - System Alerts:** No critical issues found.
  - Default ARI Admin password Used:** -
  - Default Asterisk Manager Password Used:** -
  - Collecting Anonymous Browser Stats:** -
  - No email address for online update checks:** -
  - Show New
- Uptime:**
  - System Last Rebooted:** 3 hours, 1 minute, 22 seconds ago.
  - Load Averages:**

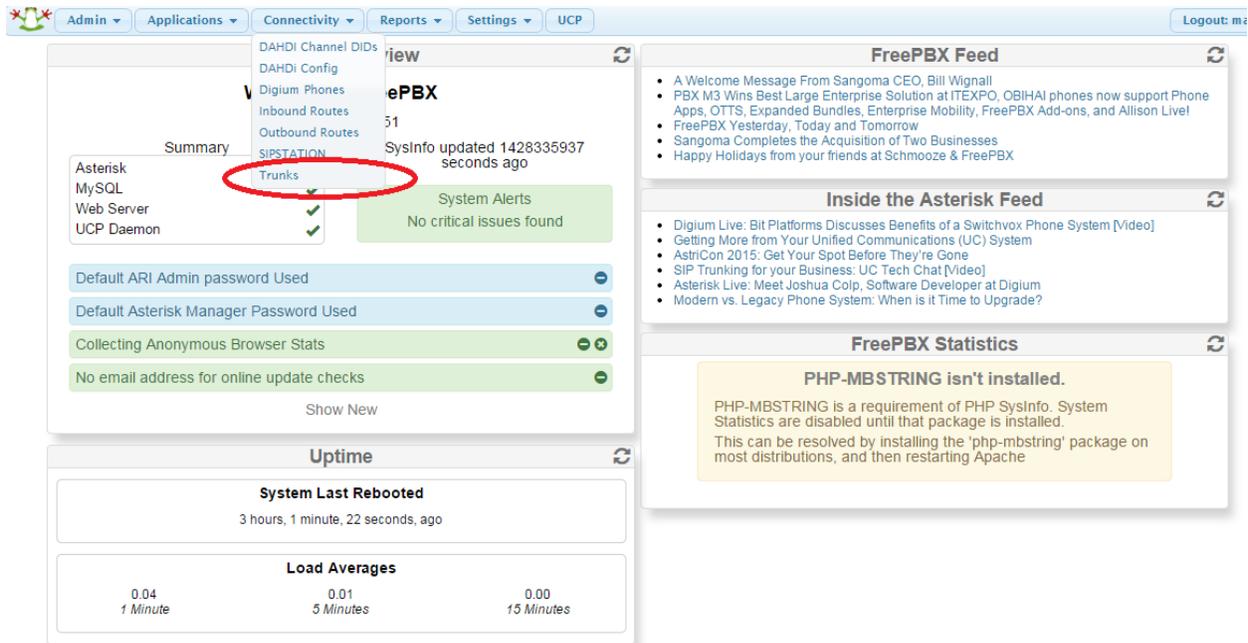
0.04	0.01	0.00
1 Minute	5 Minutes	15 Minutes
- FreePBX Feed:**
  - A Welcome Message From Sangoma CEO, Bill Wignall
  - PBX M3 Wins Best Large Enterprise Solution at ITEXPO, OBIHAI phones now support Phone Apps, OTTS, Expanded Bundles, Enterprise Mobility, FreePBX Add-ons, and Allison Live!
  - FreePBX Yesterday, Today and Tomorrow
  - Sangoma Completes the Acquisition of Two Businesses
  - Happy Holidays from your friends at Schmooze & FreePBX
- Inside the Asterisk Feed:**
  - Digium Live: Bit Platforms Discusses Benefits of a Switchvox Phone System [Video]
  - Getting More from Your Unified Communications (UC) System
  - AstriCon 2015: Get Your Spot Before They're Gone
  - SIP Trunking for your Business: UC Tech Chat [Video]
  - Asterisk Live: Meet Joshua Colp, Software Developer at Digium
  - Modern vs. Legacy Phone System: When is it Time to Upgrade?
- FreePBX Statistics:**
  - PHP-MBSTRING isn't installed.** PHP-MBSTRING is a requirement of PHP SysInfo. System Statistics are disabled until that package is installed. This can be resolved by installing the 'php-mbstring' package on most distributions, and then restarting Apache

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

## First we will cover the setting the SIP Trunk.

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



The screenshot shows the FreePBX interface with the 'Connectivity' menu open. The 'Trunks' option is highlighted with a red circle. The interface includes a navigation bar with 'Admin', 'Applications', 'Connectivity', 'Reports', 'Settings', and 'UCP'. The main content area displays a 'Summary' section with a 'System Alerts' box indicating 'No critical issues found'. Below this are sections for 'Uptime' (System Last Rebooted: 3 hours, 1 minute, 22 seconds ago) and 'Load Averages' (0.04 at 1 Minute, 0.01 at 5 Minutes, 0.00 at 15 Minutes). On the right, there are three feeds: 'FreePBX Feed', 'Inside the Asterisk Feed', and 'FreePBX Statistics' (noting that PHP-MBSTRING is not installed).

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



The screenshot shows the navigation bar of the FreePBX interface, featuring the 'Admin', 'Applications', 'Connectivity', 'Reports', 'Settings', and 'UCP' menus.

## Add a Trunk

+ Add SIP (chan\_sip) Trunk

+ Add DAHDi Trunk

+ Add IAX2 Trunk

+ Add ENUM Trunk

+ Add DUNDi Trunk

+ Add Custom Trunk

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



## Add Trunk

### General Settings

Trunk Name <sup>?</sup> :	<input type="text" value="QuestBlue-SBC"/>
Outbound CallerID <sup>?</sup> :	<input type="text"/>
CID Options <sup>?</sup> :	<input type="text" value="Allow Any CID"/> ▾
Maximum Channels <sup>?</sup> :	<input type="text"/>
Asterisk Trunk Dial Options <sup>?</sup> :	<input type="text" value="Tt"/> <input type="checkbox"/> Override
Continue if Busy <sup>?</sup> :	<input type="checkbox"/> Check to always try next trunk
Disable Trunk <sup>?</sup> :	<input type="checkbox"/> Disable

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

### Dial Number Manipulation Rules <sup>?</sup>

( 1 ) + prefix		NXXNXXXXXX		
( prepend ) + prefix		1NXXNXXXXXX		
( prepend ) + prefix		011.		

+ Add More Dial Pattern Fields

Clear all Fields

Dial Rules Wizards <sup>?</sup>:

(pick one) ▼

Outbound Dial Prefix <sup>?</sup>:

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

### Outgoing Settings

Trunk Name <sup>?</sup>:

QuestBlue-SBC

PEER Details <sup>?</sup>:

```
type=peer
host=sbc.questblue.com
insecure=very
context=from-trunk
qualify=yes
nat=no
session-timers=refuse
```

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

### Incoming Settings

USER Context :

USER Details :

### Registration

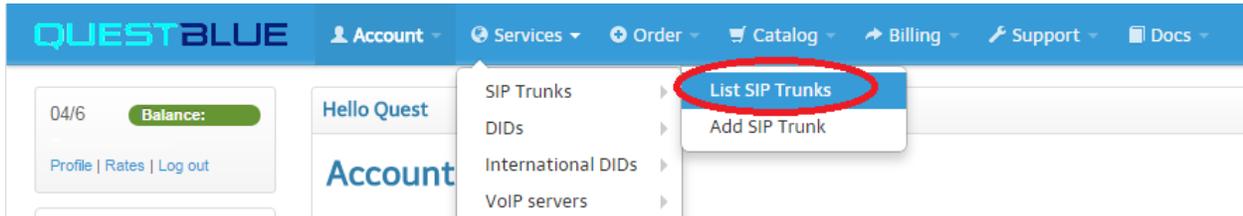
Register String :

Press Apply Config in red below



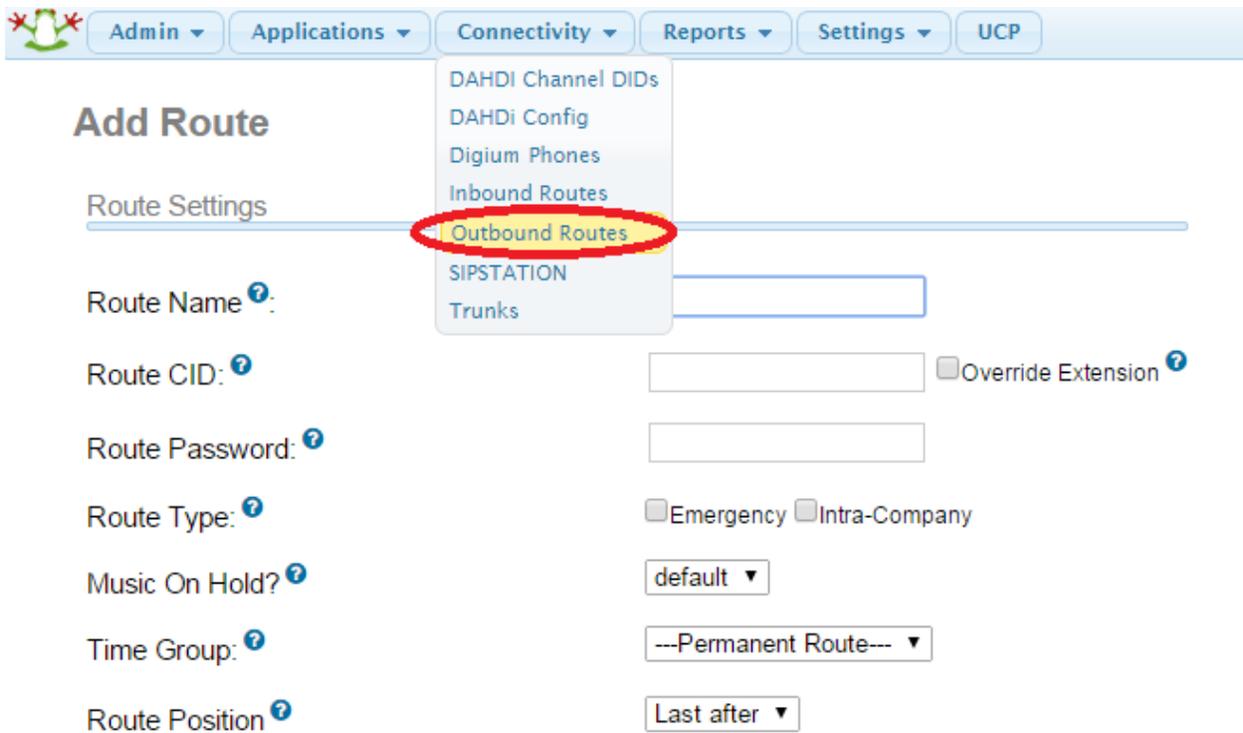
You will see above that we do not register our SIP Trunks. For added security we only allow IP Authentication.

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



### Outbound Routes

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



## Add Route

### Route Settings

---

Route Name <sup>?</sup> :	<input type="text" value="Outbound"/>
Route CID: <sup>?</sup>	<input type="text"/> <input type="checkbox"/> Override Extension <sup>?</sup>
Route Password: <sup>?</sup>	<input type="text"/>
Route Type: <sup>?</sup>	<input type="checkbox"/> Emergency <input type="checkbox"/> Intra-Company
Music On Hold? <sup>?</sup>	<input type="text" value="default"/> ▼
Time Group: <sup>?</sup>	<input type="text" value="--Permanent Route--"/> ▼
Route Position <sup>?</sup>	<input type="text" value="Last after"/> ▼

Next you will create the patterns that need to be matched

Dial Patterns that will use this Route <sup>?</sup>

( <input type="text" value="prepend"/> ) + <input type="text" value="prefix"/>   [ <input type="text" value="011."/> / <input type="text" value="CallerID"/> ] <input type="button" value="+"/> <input type="button" value="trash"/>
( <input type="text" value="prepend"/> ) + <input type="text" value="prefix"/>   [ <input type="text" value="1NXXNXXXXXX"/> / <input type="text" value="CallerID"/> ] <input type="button" value="+"/> <input type="button" value="trash"/>
( <input type="text" value="1"/> ) + <input type="text" value="prefix"/>   [ <input type="text" value="NXXNXXXXXX"/> / <input type="text" value="CallerID"/> ] <input type="button" value="+"/> <input type="button" value="trash"/>

+ Add More Dial Pattern Fields

Dial patterns wizards <sup>?</sup>:

Trunk Sequence for Matched Routes <sup>?</sup>

0

Optional Destination on Congestion <sup>?</sup>

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:

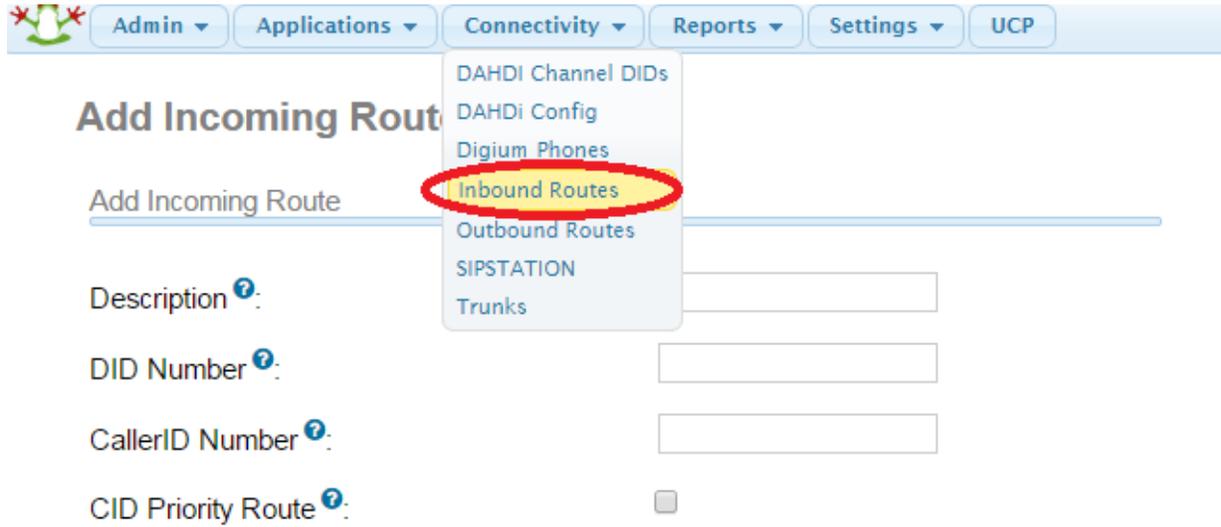
In the above dial patterns you will cover 011. (International Dialing) 1NXXNXXXXXX Dialing is equal to you dialing a number with the 1 + area code plus subscriber number and finally a prepended 1 then NXXNXXXXXX and in this case if you do not put the 1 in the outbound call the system will add it for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

## Create Inbound Route

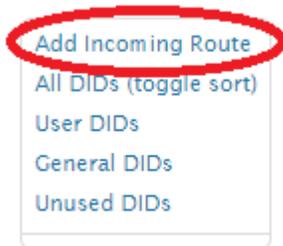
Note:

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:



- Fill in the Description and the DID Number

## Add Incoming Route

### Add Incoming Route

Description <sup>?</sup>:

DID Number <sup>?</sup>:

CallerID Number <sup>?</sup>:

CID Priority Route <sup>?</sup>:

- Choose the Destination from the drop down menu and Press Submit

### Set Destination

== choose one == ▼

Submit

Clear Destination & Submit

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

Admin Applications Connectivity Reports Settings UCP **Apply Config** Logout: m

**Route: Inbound**  
Delete Route Inbound

Edit Incoming Route

Description <sup>?</sup>:

DID Number <sup>?</sup>:

Add Incoming Route  
All DIDs (toggle sort)  
User DIDs  
General DIDs  
Unused DIDs  
Inbound  
1234567890 / any CID

In addition to creating your Trunk and Inbound/Outbound routes there is another important issue that you should address.

Go to Setting the Asterisk SIP Settings

The screenshot shows the FreePBX interface with the 'Settings' menu open. The 'Asterisk SIP Settings' option is highlighted with a red circle. The main content area shows 'SIP Settings' with a message: 'Asterisk is currently using **chan\_sip** for SIP Traffic. You can change this on the Advanced Settings Page'. Below this is a blue banner that says 'Items may moved! Please use the navigation on the right side of the page'. Other settings like 'Security Settings' and 'Allow Anonymous Inbound SIP Calls' are visible.

On the right side of the page, select Chan SIP (A) as shown below:

The screenshot shows a navigation menu with two items: 'General SIP Settings' and 'Chan SIP (A)'. The 'Chan SIP (A)' item is highlighted with a red circle.

At the bottom of the Asterisk SIP Setting – Chan SIP (A) page you will locate the field to put Other SIP Setting:

The screenshot shows the 'Other SIP Settings' field in the configuration page. The field contains the text 'session-timers = refuse'. Below the field is an 'Add Field' button. At the bottom of the configuration area is a 'Submit Changes' button.

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

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

The screenshot shows the FreePBX navigation menu with the 'Apply Config' button highlighted in red. The menu items are 'Admin', 'Applications', 'Connectivity', 'Reports', 'Settings', and 'UCP'.