

# 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 Status dashboard. At the top is a navigation bar with a logo and menu items: Admin, Applications, Connectivity, Reports, Settings, and User Panel. The main content is organized into several sections:

- FreePBX Notices:** Shows "No new notifications" with a "show all" link.
- FreePBX Statistics:** A table showing call and channel counts.

Total active calls	0
Internal calls	0
External calls	0
Total active channels	0
- FreePBX Connections:** A table showing online status.

IP Phones Online	0
IP Trunks Online	1
- Uptime:** Displays system and Asterisk uptime, and the last reload time.

**System Uptime:** 3 weeks, 2 days, 21 hours, 40 minutes  
**Asterisk Uptime:** 3 weeks, 2 days, 21 hours, 40 minutes  
**Last Reload:** 21 hours, 1 minute
- System Statistics:** A detailed overview of system resources.
  - Processor:** Load Average 0.00, CPU 0%
  - Memory:** App Memory 9%, Swap 0%
  - Disks:** / 1%, /dev/shm 0%, /boot 11%
  - Networks:** eth0 receive 0.99 KB/s, eth0 transmit 1.07 KB/s, eth1 receive 0.00 KB/s, eth1 transmit 0.00 KB/s
- Server Status:** A list of services and their health status.

Asterisk	OK
MySQL	OK
Web Server	OK
SSH Server	OK

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.

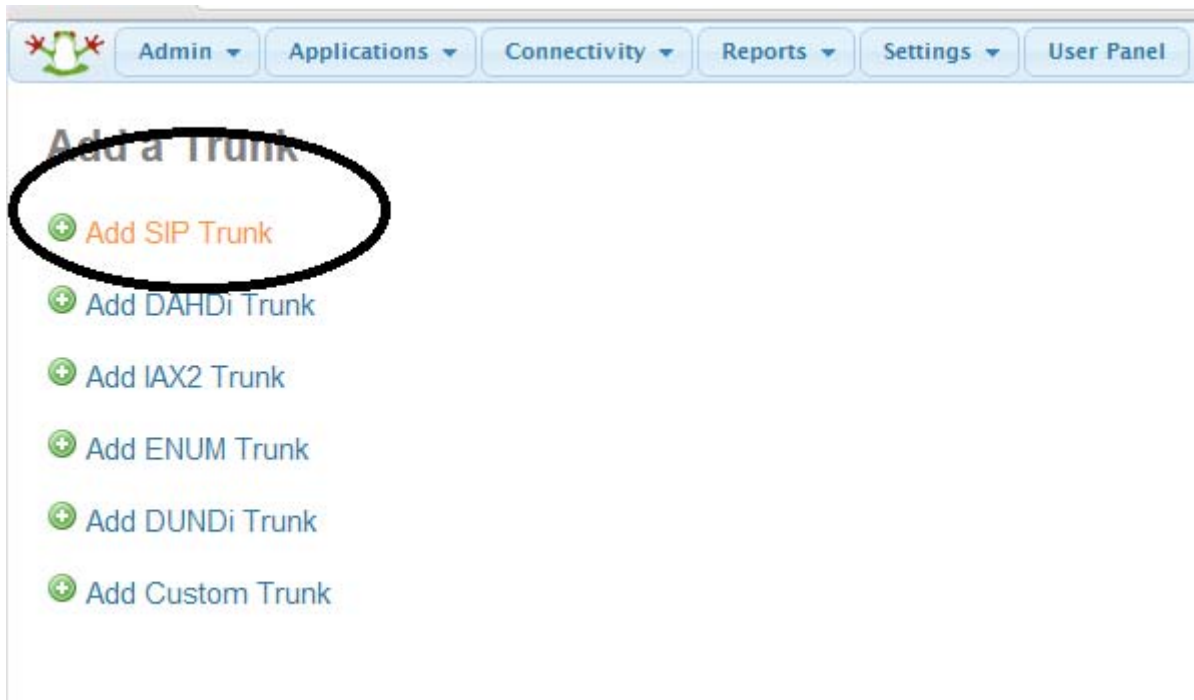
The screenshot shows the FreePBX System Status page. The top navigation bar includes 'Admin', 'Applications', 'Connectivity', 'Reports', 'Settings', and 'User Panel'. The 'Connectivity' menu is open, showing options like 'DAHDI Channel DIDs', 'Digium Phones', 'Inbound Routes', 'OSS Endpoint Advanced Settings', 'OSS Endpoint Configuration', 'OSS Endpoint Device List', 'OSS Endpoint Template Manager', 'Outbound Routes', 'SIPSTATION', and 'Trunks'. The 'Trunks' option is circled in black. Below the navigation bar, there are several widgets: 'FreePBX System Status' with a notification area, 'FreePBX Connections' table, 'Uptime' section, 'System Statistics' (Processor, Memory, Disks, Networks), and 'Server Status'.

FreePBX Connections	
Total active calls	0
Internal calls	0
External calls	0
Total active channels	0
FreePBX Connections	
IP Phones Online	0
IP Trunks Online	1

System Statistics	
Processor	
Load Average	0.00
CPU	0%
Memory	
App Memory	9%
Swap	0%
Disks	
/	1%
/dev/shm	0%
/boot	11%
Networks	
eth0 receive	0.32 KB/s
eth0 transmit	0.76 KB/s
eth1 receive	0.00 KB/s
eth1 transmit	0.00 KB/s

Server Status	
Asterisk	OK
MySQL	OK
Web Server	OK
SSH Server	OK

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**

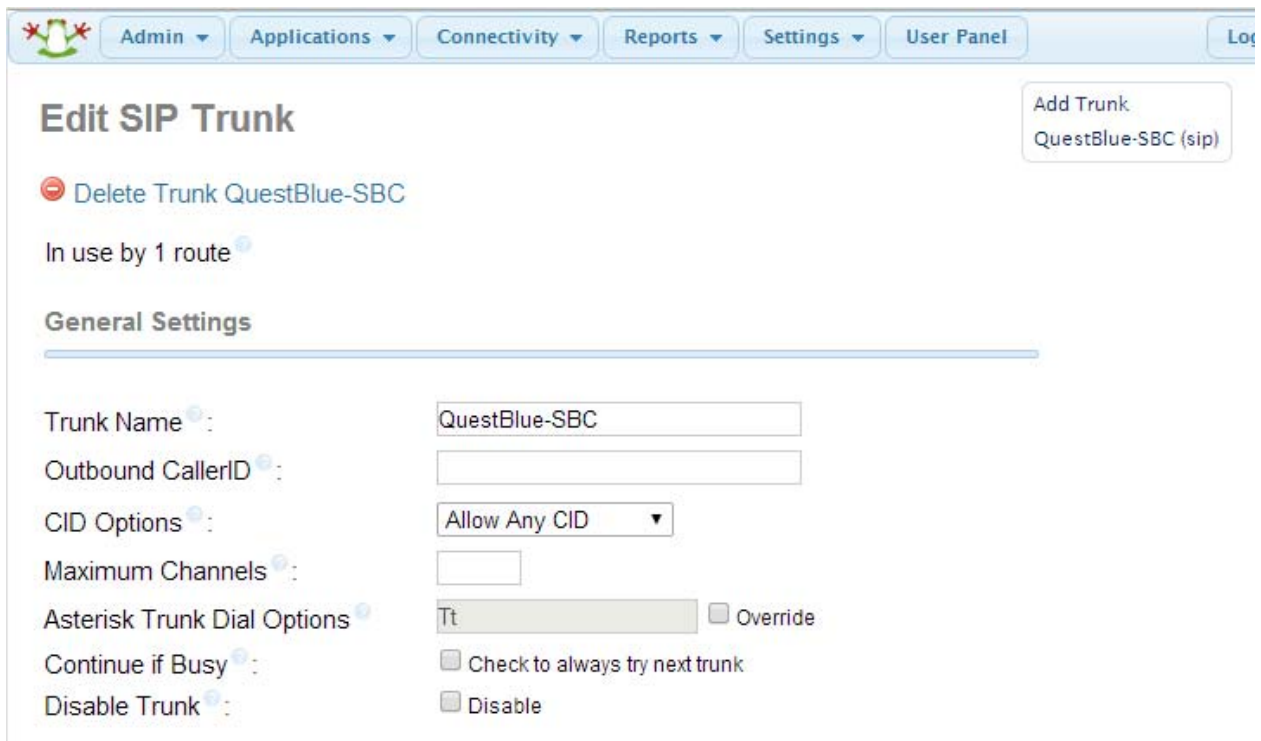
**B. Dialed Number Manipulation Rules**

**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



The screenshot shows the Asterisk SIP Trunk configuration page. At the top, there is a navigation bar with tabs for Admin, Applications, Connectivity, Reports, Settings, and User Panel. The main heading is "Edit SIP Trunk". Below the heading, there is a "Delete Trunk QuestBlue-SBC" button and a "Add Trunk QuestBlue-SBC (sip)" button. The "General Settings" section is expanded, showing the following fields:

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

## Dialed Number Manipulation Rules

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( 1 ) + prefix		NXXNXXXXXX	+	🗑
( prepend ) + prefix		1NXXNXXXXXX	+	🗑
( prepend ) + prefix		011.	+	🗑
( prepend ) + prefix		match pattern	+	🗑

+ Add More Dial Pattern Fields

Clear all Fields

Dial Rules Wizards : (pick one) ▼

Outbound Dial Prefix :

Export Dialplans as CSV :

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

## Outgoing Settings

---

Trunk Name : QuestBlue-SBC

PEER Details :

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

It is very important 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 <sup>?</sup>:

USER Details <sup>?</sup>:

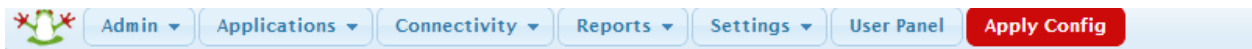
### Registration

Register String <sup>?</sup>:

[Submit Changes](#)

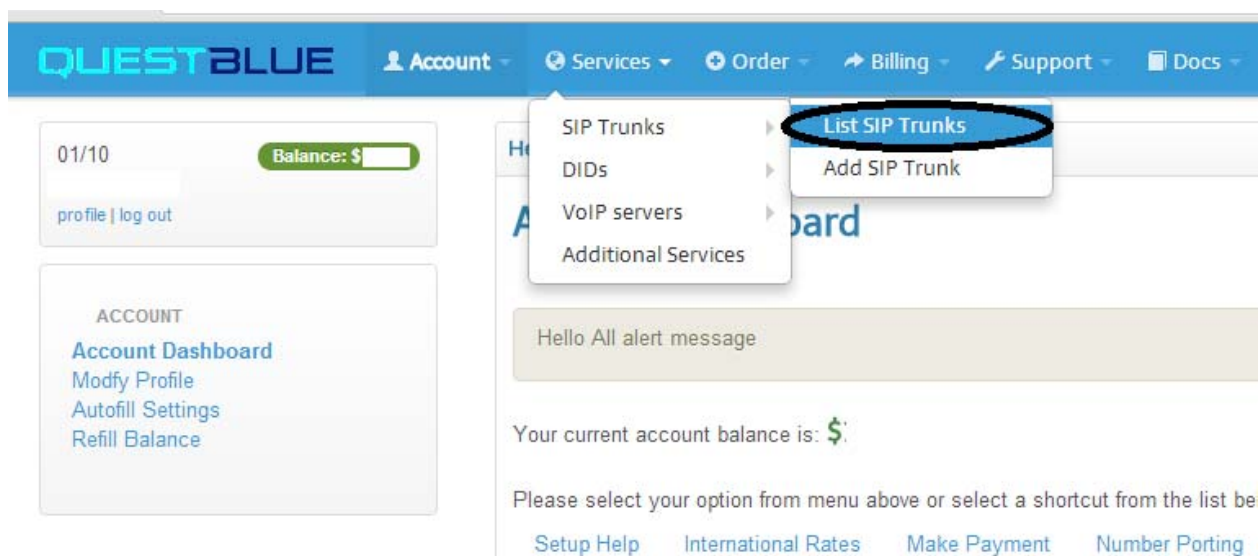
[Duplicate Trunk](#)

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



## Outbound Routes

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

Admin Applications Connectivity Reports Settings User Panel

## Add Route

Route Settings

Route Name:

Route CID:

Route Password:

Route Type:  Emergency  Intra-Company

Music On Hold?:

Time Group:

Route Position:

Additional Settings

PIN Set:

Call Recording:

Dial Patterns that will use this Route

(  ) +  || [  /  ]

+ Add More Dial Pattern Fields

Dial patterns wizards:

Outbound Routes

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





Admin ▾

Applications ▾

Connectivity ▾

Reports ▾

Settings ▾

User Panel

## Edit Route

 Delete Route outbound

### Route Settings

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Route Name <sup>ⓘ</sup>:

outbound

Route CID: <sup>ⓘ</sup>

Override Extension <sup>ⓘ</sup>

Route Password: <sup>ⓘ</sup>

Route Type: <sup>ⓘ</sup>

Emergency  Intra-Company

Music On Hold? <sup>ⓘ</sup>

default ▾

Time Group: <sup>ⓘ</sup>

---Permanent Route--- ▾

Route Position <sup>ⓘ</sup>

---No Change--- ▾

### Additional Settings

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PIN Set <sup>ⓘ</sup>:





None ▾

Call Recording <sup>ⓘ</sup>:

Allow ▾

Next you will create the patterns that need to be matched

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

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

[+ Add More Dial Pattern Fields](#)

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

Export Dialplans as CSV <sup>?</sup>: [Export](#)

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

0  

1

[Add Trunk](#)

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

[Submit Changes](#)

[Duplicate Route](#)

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:

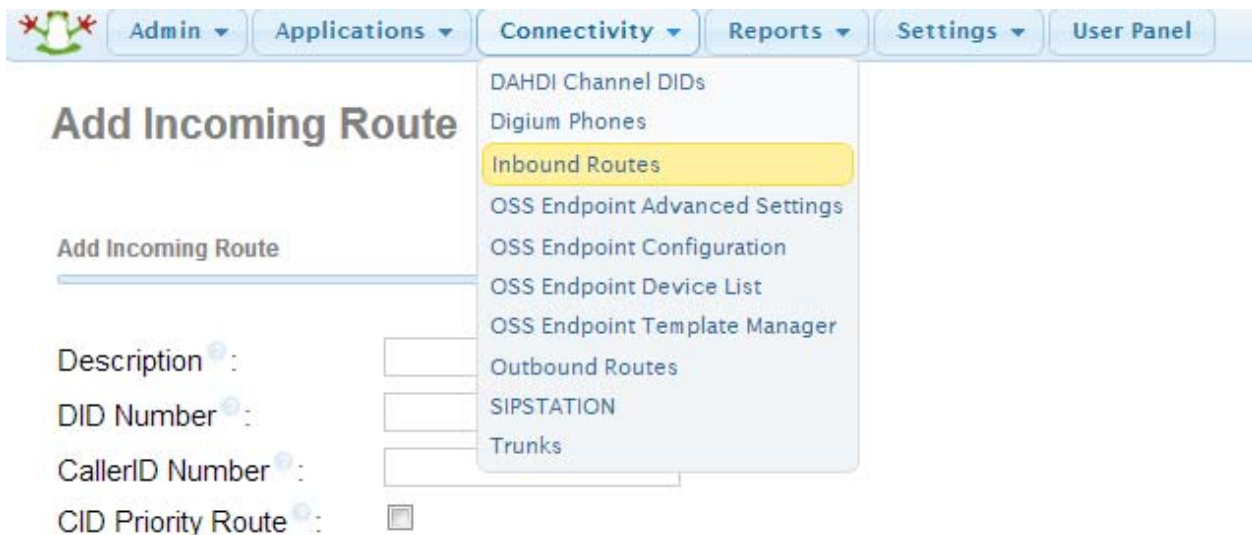
In the above dial patterns you will cover 011. (International Dialing) 1NXXNXXXXXX Dialing is equal to you dialing a number with the 1 + area code plus subscriber number and finally a prepended 1 then NXXNXXXXXX and in this case if you do not put the 1 in the outbound call the system will add it for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

## Add Incoming Route

### Add Incoming Route

Description:

DID Number:

CallerID Number:

CID Priority Route:

### Options

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

### Set Destination

== choose one ==

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

## Route: inbound

### Edit Incoming Route

Description:

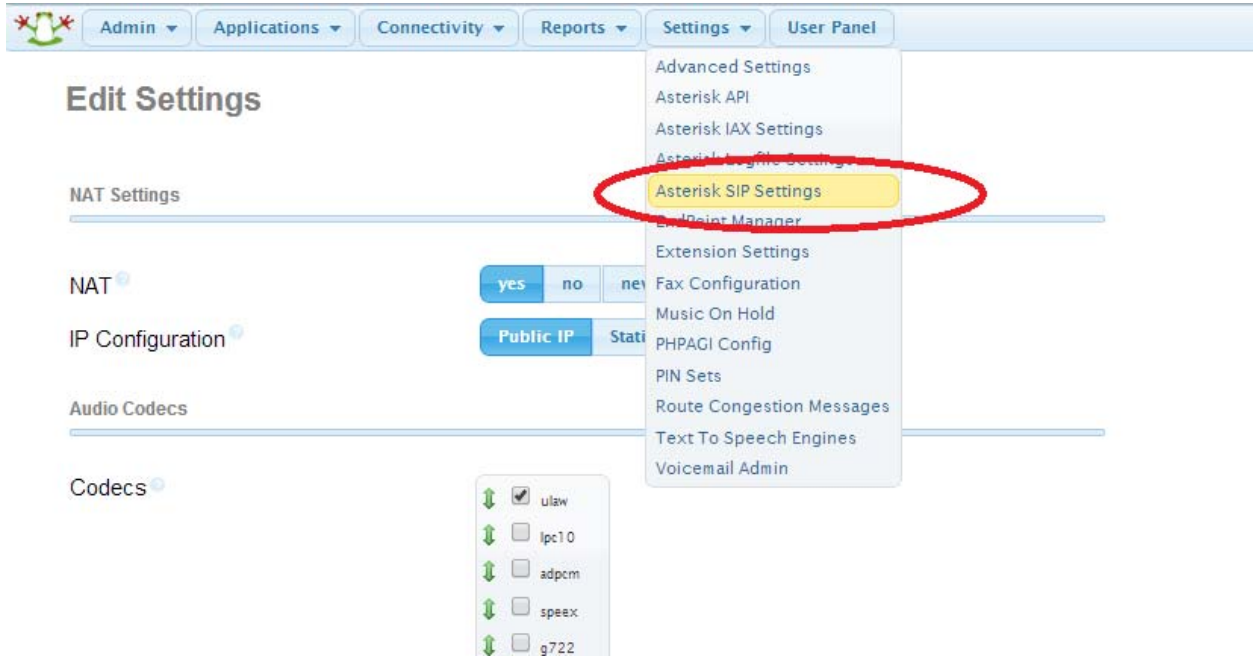
DID Number:

- Add Incoming Route
- All DIDs (toggle sort)
- User DIDs
- General DIDs
- Unused DIDs

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- 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



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

### Advanced General Settings

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Language

Default Context

Bind Address

Bind Port

Allow SIP Guests

Allow Anonymous Inbound SIP Calls

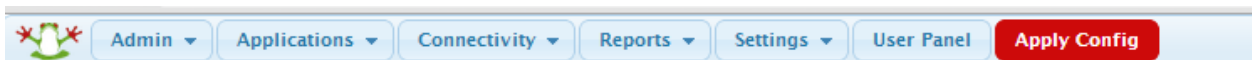
SRV Lookup

Call Events

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:



## Edit Settings

NAT Settings

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