



NextOS SIP Trunking
FreePBX
User Setup Guide

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What is FreePBX?

This setup guide will walk you through the process to set up Nextiva SIP Trunking for a FreePBX, a popular Asterisk-based PBX.

FreePBX Setup

NOTE: The configuration of your FreePBX requires help from the Nextiva Support team. Please contact Nextiva Support to ensure that your Nextiva SIP Trunking account has been configured to connect to a PBX.

Logging In to the FreePBX Administration Console

FreePBX can be configured through a web-based portal. Follow the steps below to log in:

1. Open a web browser on your computer (Internet Explorer, Firefox, Chrome, etc.).
2. Enter the **IP address** of the FreePBX in the address bar.
3. Click the **FreePBX Administration** icon on the left side of the screen (*Figure 1-1*).

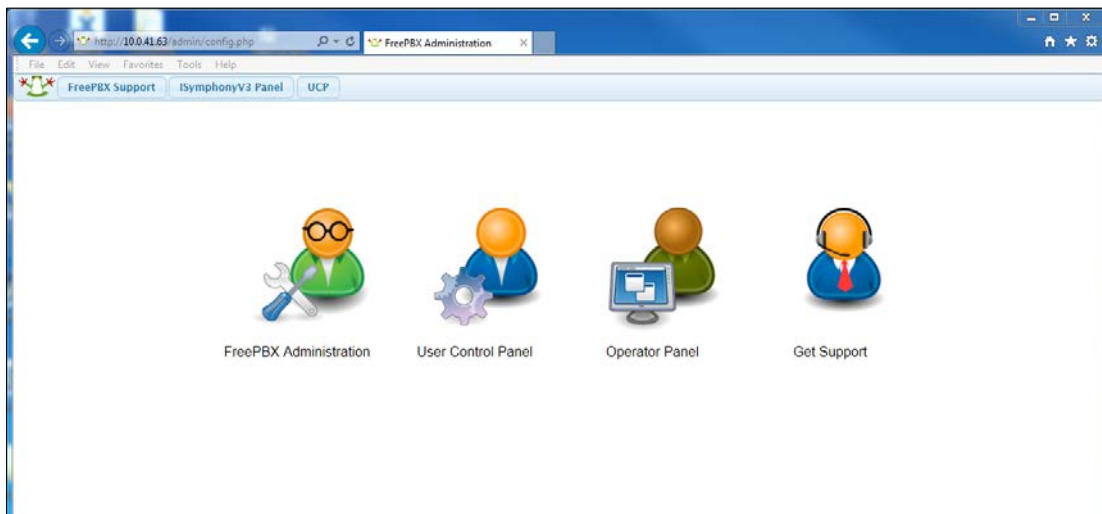


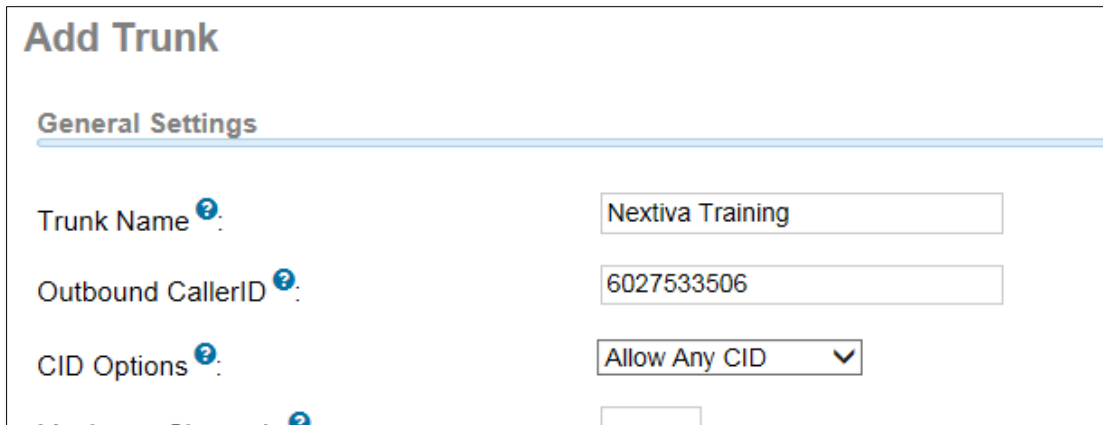
Figure 1-1: FreePBX Administration Console

4. Enter the **User ID** and **Password** for the FreePBX.

Adding a Trunk

The trunk is the first thing you will need to set up. It will contain the proxy server address and the authentication details, as well as other settings related to the PBX's connection to Nextiva. For the purposes of this setup guide, the value of **AUTHORIZATION NAME** is the ten-digit pilot number you should have received from Nextiva Support.

1. Click the **Connectivity** button at the top of the screen, and then select **Trunks** from the drop-down list that appears. The Add a Trunk screen will appear.
2. Click the **Add SIP (chan_sip) Trunk** link at the top of the screen. The Add Trunk screen will appear (Figure 1-2).



Add Trunk

General Settings

Trunk Name [?]:

Outbound CallerID [?]:

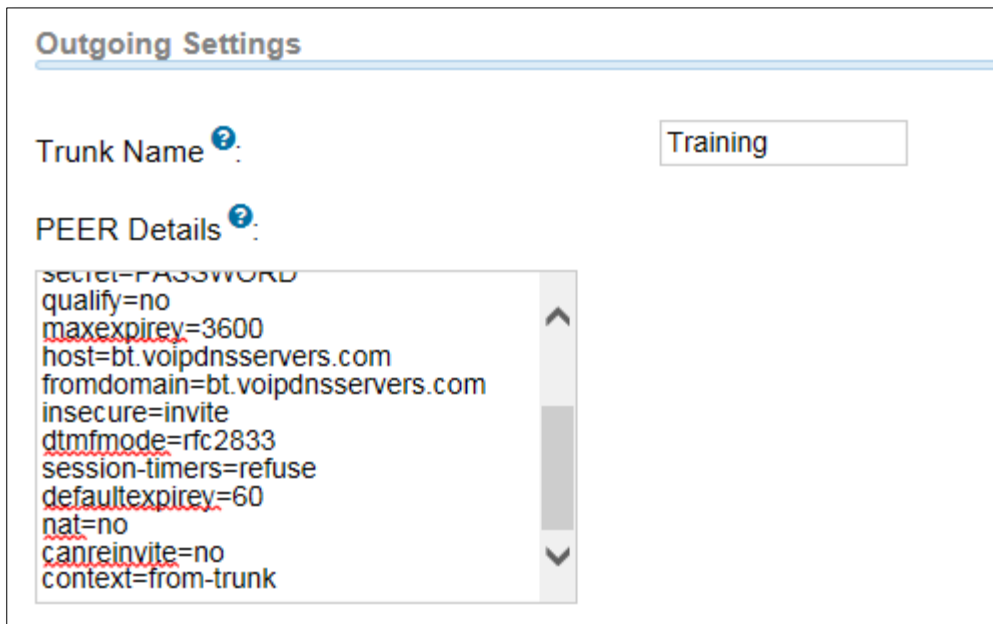
CID Options [?]: ▼

Figure 1-2: Add Trunk Screen


3. Enter a descriptive name for the trunk in the **Trunk Description** text box at the top of the screen. In the example above, the Trunk Name is “Nextiva Training.”
4. Enter the Pilot Number/Authorization Name in the **Outbound Caller ID** text box at the top of the screen.
5. Enter a name for the trunk in the **Trunk Name** text box under the **Outgoing Settings** heading. You may need to scroll down to see this field.


6. Enter the peer details below in the **PEER Details** text box. Make sure to clear out any details that are already entered, and replace the text below in red with the actual authorization name and password provided by Nextiva Support (*Figure 1-3*).

```
disallow=all
allow=ulaw
username=AUTHORIZATIONNAME
fromuser=AUTHORIZATIONNAME
type=friend
secret=PASSWORD
qualify=no
maxexpirey=3600
host=bt.voipdnsservers.com
fromdomain=bt.voipdnsservers.com
insecure=invite
dtmfmode=rfc2833
session-timers=refuse
defaultexpirey=60
nat=no
canreinvite=no
context=from-trunk
```



Outgoing Settings

Trunk Name :

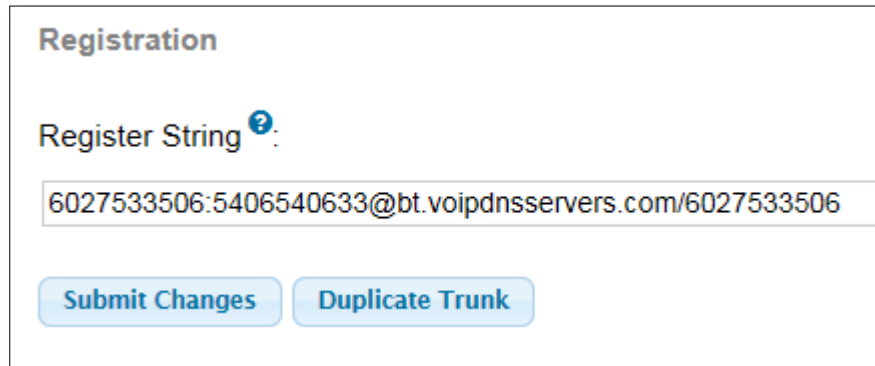
PEER Details :

```
secret=PASSWORD
qualify=no
maxexpirey=3600
host=bt.voipdnsservers.com
fromdomain=bt.voipdnsservers.com
insecure=invite
dtmfmode=rfc2833
session-timers=refuse
defaultexpirey=60
nat=no
canreinvite=no
context=from-trunk
```

Figure 1-3: Add Trunk

7. Enter the register string in the **Register String** text box at the bottom of the screen using the template below. Replace the text in **red** with the actual authorization name and password (Figure 1-4)

AUTHORIZATIONNAME:PASSWORD@bt.voipdnsservers.com/AUTHORIZATIONNAME



The screenshot shows a web form titled "Registration". It contains a label "Register String" with a help icon. Below the label is a text input field containing the example register string: "6027533506:5406540633@bt.voipdnsservers.com/6027533506". At the bottom of the form are two buttons: "Submit Changes" and "Duplicate Trunk".

Figure 1-4: Register String

8. Click the **Submit Changes** button at the bottom of the screen.

Applying the Configuration

After you have set up a trunk, you need to apply the configuration to the FreePBX server. Otherwise, the changes you made will not take effect. Click the red **Apply Config** button at the top of the screen to apply the configuration changes (Figure 1-5).

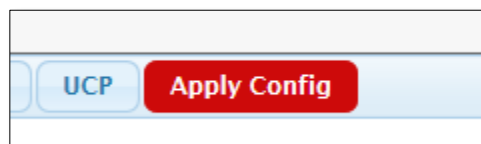


Figure 1-5: Apply Config

Setting Up an Outbound Route

To make sure that outbound calls route through the new Nextiva trunk, you need to set up an outbound route.

1. Click the **Connectivity** button at the top of the screen, and select **Outbound Routes** from the menu. The Add Route screen will appear (*Figure 1-6*).

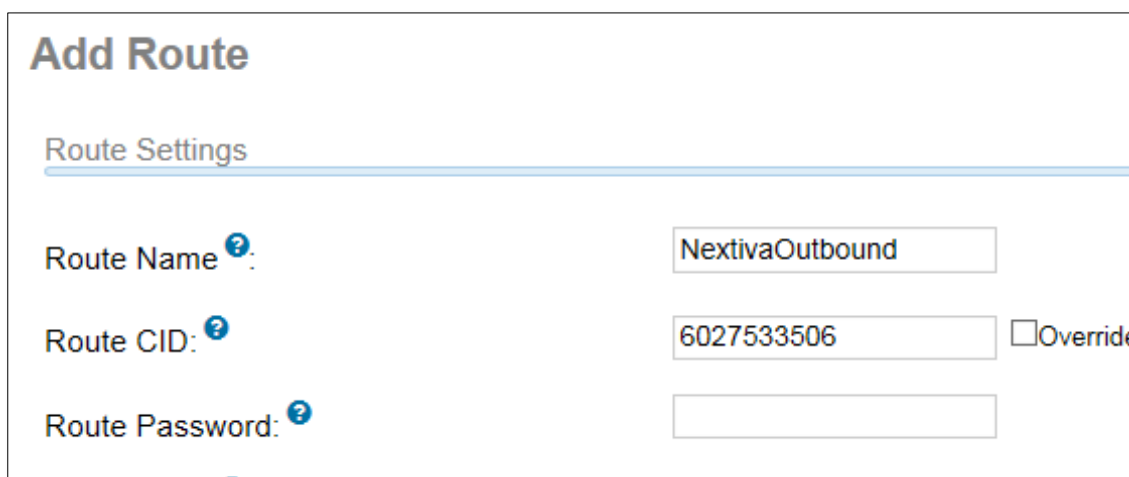


Figure 1-6: Add Route

2. Enter a name for the outbound route in the **Route Name** text box at the top of the screen. In the example above, the name used is “NextivaOutbound.”
3. Enter the ten-digit pilot number in the **Route CID** text box at the top of the screen.

NOTE: This field must contain the ten-digit pilot number or outbound calls will fail. Setting your own outbound caller ID number will be discussed in **Getting FreePBX to Use P-Asserted Identity** later in this guide.

4. Scroll down until the **Dial patterns wizards** drop-down box is visible, and select the following wizards from the drop-down:
 - Local 7/10 digit
 - Toll-free
 - Long-distance
 - International
 - Emergency

5. Select the trunk you created earlier in the **0 drop-down box** under the Trunk Sequence for Matched Routes heading. The screen should look like Figure 1-7 below.

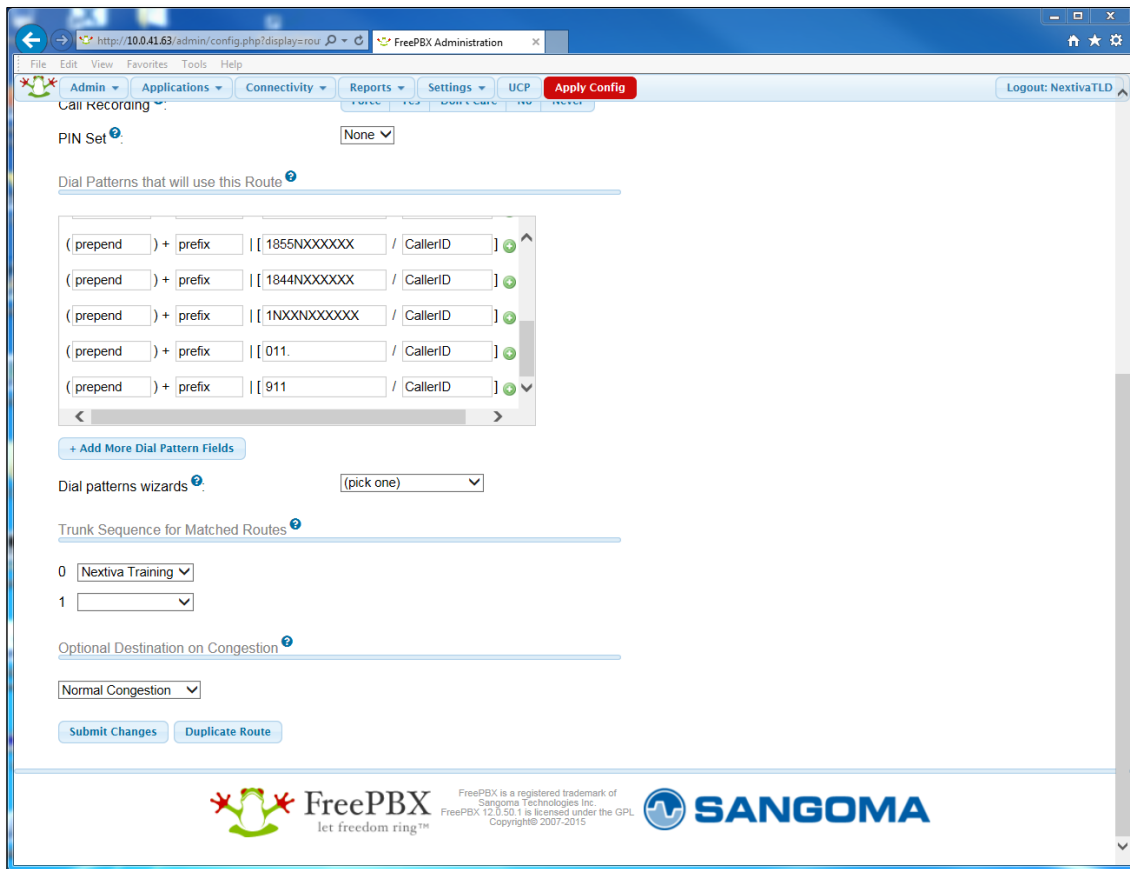


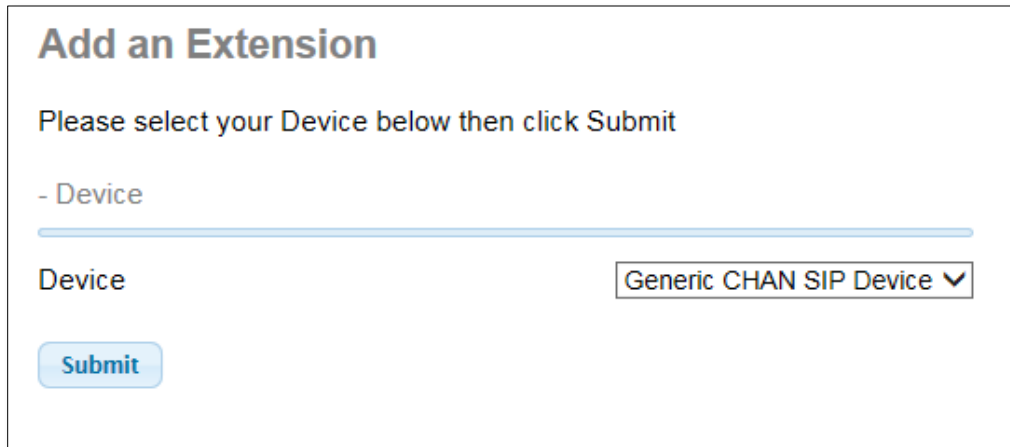
Figure 1-7: Completed Add Route

6. Click the **Submit Changes** button at the bottom of the screen.
7. Click the red **Apply Config** button at the top of the screen.

Creating an Extension

Follow the steps below to create an extension for your trunk:

1. Click the **Applications** button at the top of screen, and select **Extensions** from the menu. The Add an extension screen will appear (*Figure 1-8*).



Add an Extension

Please select your Device below then click Submit


- Device

Device Generic CHAN SIP Device ▾

Submit

Figure 1-8: Add an Extension

2. Make sure that **Generic CHAN SIP Device** is selected in the Device drop-down, and click the **Submit** button. The Add SIP Extension screen will appear (*Figure 1-9*).



Add SIP Extension

- Add Extension

User Extension [?]

Display Name [?]

CID Num Alias [?]

Figure 1-9: Add SIP Extension

3. Enter an extension number in the **User Extension** text box at the top of the screen.
4. Enter a descriptive name for the extension in the **Display Name** text box at the top of the screen.

5. Enter one of your Nextiva-provided phone numbers in the **Outbound CID** text box (Figure 1-10).

NOTE: This will cause outbound calls from the extension to show the entered number as the outgoing caller ID number, rather than the first number, which is the User ID.

The screenshot shows a configuration panel titled '- Extension Options'. It contains four settings:

- Queue State Detection**: Set to 'Use State' (dropdown menu).
- Outbound CID**: Set to '4807259138' (text input field).
- Asterisk Dial Options**: Set to 'Ttr' (text input field).
- Ring Time**: Set to 'Default' (dropdown menu).

Figure 1-10: Outbound CID

6. Change the password for the extension in the **Secret** text box to **n3xt1v@** (Figure 1-11). You may need to scroll down to see this. This is the password the phone associated with this extension will use to register to the FreePBX.

The screenshot shows two configuration fields:

- Secret**: Set to 'n3xt1v@' (text input field).
- DTMF Signaling**: Set to 'RFC 2833' (dropdown menu).

Figure 1-11: Updated Password for Extension

7. Click the **Submit** button at the bottom of the screen.
8. Click the red **Apply Config** button at the top of the screen.

Registering an X-Lite Softphone

Follow the steps below to register an X-Lite softphone to your PBX for testing purposes, or skip this section if you have another phone you are testing with.

1. Download and install X-Lite from the Counterpath website. Below are the links to download this softphone:

Windows: http://counterpath.s3.amazonaws.com/downloads/X-Lite_Win32_4.8.1_76067.exe

MAC: http://counterpath.s3.amazonaws.com/downloads/X-Lite_4.9.3_79959.dmg

2. When prompted, click the **Account Settings** link to set up a new SIP account.
3. Enter the required information as indicated below (*Figure 1-12*):
 - User ID: **the extension number**
 - Domain: **10.0.41.63** (the IP address of the FreePBX)
 - Password: **n3xt1v@** (The password you entered in step 6 of the Creating an Extension section)
 - Display name: **User 1** (the name of the user)

Figure 1-12: SIP Account screen in X-Lite

P-Asserted Identity for FreePBX

To use P-Asserted Identity on outbound calls, you will need to create an **extensions_custom.conf** file and enter the text below into the file. Then, you will then need to use a Secure Shell (SSH) client to access the PBX and implement your configuration file. Below are instructions on how to do this from PuTTY, a common SSH client:

[macro-dialout-trunk-predial-hook]

exten => s,1,SipAddHeader(P-Asserted-Identity:<sip:\${CALLERID(num)}@bt.voipdnsservers.com>)

exten => s,n,MacroExit()

How to Log In to the FreePBX Using PuTTY

PuTTY is a telnet and SSH client for the Windows and Unix platforms. Follow the instructions below to use PuTTY to connect to your FreePBX and to open the `extensions_custom.conf` file for editing. If you do not use PuTTY you can skip this section, but you will need to use your SSH client to access the PBX.

1. Launch PuTTY on your computer by **double-clicking** on the desktop icon.
2. If prompted, select **Run** when the Open File – Security Warning window appears.
3. Enter the IP address of your FreePBX in the **Host Name (or IP address)** text box toward the top of the PuTTY Configuration window.
4. Make sure the **SSH** radio button is selected, and click the **Open** button at the bottom of the PuTTY Configuration window (*Figure 1-13*).

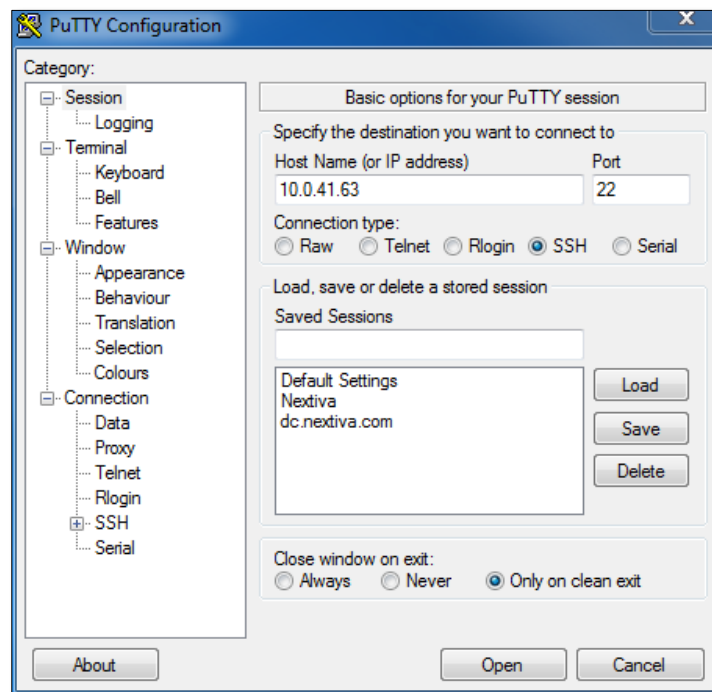


Figure 1-13: PuTTY Configuration

5. Log in with **root** as the username and the **root password** for your FreePBX.

Creating the extensions_custom.conf File

1. Switch to the asterisk folder using the `cd /etc/asterisk` command.
2. Launch the Vim text editor using the `vim extensions_custom.conf` command.
3. Press the letter "i" on your keyboard to begin typing.
4. Enter the text string below.

```
[macro-dialout-trunk-predial-hook]
```

```
exten => s,1,SipAddHeader(P-Asserted-Identity:<sip:${CALLERID(num)}@bt.voipdnsservers.com>)
```

```
exten => s,n,MacroExit()
```

5. Press **Esc**, and then type `:x` to save the file and exit the Vim editor.

How to Reload the Dial Plan

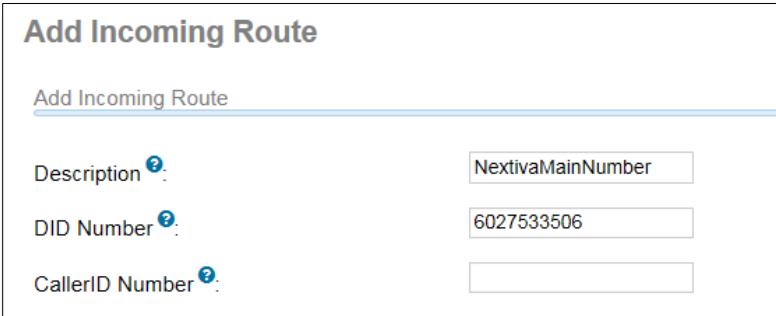
After you create the `extensions_custom.conf` file, you will need to reload the dial plan for the changes to take effect.

1. Access and log in to the FreePBX using PuTTY if you previously logged out.
2. Type `cd` and press **Enter** to go back to the main root directory if necessary.
3. Type `asterisk -r`, and then press **Enter** on the keyboard to access the Asterisk command line.
4. Type `dialplan reload`, and then press **Enter** on the keyboard to reload the dial plan.
5. Type `exit`, and then press **Enter** on the keyboard to exit the Asterisk command line. Keep PuTTY open for now.

Setting Up an Inbound Route

You need to configure an inbound route so you can accept incoming phone calls. If you don't, calls to that phone number will produce a message indicating the number has been disconnected. Follow the steps below to set up an inbound route on your FreePBX so you can receive inbound calls:

1. Click the **Connectivity** button at the top of the screen, and select **Inbound Routes** from the menu. The Add Incoming Route screen will appear (*Figure 1-14*).



Add Incoming Route	
Add Incoming Route	
Description ?	NextivaMainNumber
DID Number ?	6027533506
CallerID Number ?	

Figure 1-14: Add Incoming Route

2. Enter a name for the inbound route in the **Description** text box at the top of the screen.
3. Enter the dialed phone number in the **DID Number** text box at the top of the screen.
4. Scroll to the bottom of the Add Incoming Route screen, and select the phone number's intended destination from the drop-down list under the **Set Destination** heading (*Figure 1-15*).

Figure 1-15: Set Destination

5. Click the **Submit** button at the bottom of the screen.
6. Repeat steps 1 through 5 for any remaining phone numbers you wish to assign.
7. Click the red **Apply Config** button at the top of the screen.

“To” Field for Inbound Calls

In order for your FreePBX to look at the **To** field on inbound calls, you will need to edit the **extensions_custom.conf** file and enter the text below into the file. To do this using PuTTY, follow the instructions below:

```
[custom-get-did-from-sip]
exten => _X!,1,Goto(from-trunk,${CUT(CUT(SIP_HEADER(To),@,1),:,2)},1)
```

Editing the extensions_custom.conf File

1. Access and log in to your FreePBX using PuTTY if you previously logged out.
2. Change to the Asterisk folder using the **cd /etc/asterisk** command.
3. Launch the Vim text editor using the **vim extensions_custom.conf** command.
4. Press the letter **e** on the keyboard to edit the file.
5. Press the **Insert** key on your keyboard to begin typing.
6. Enter the text string below:

```
[custom-get-did-from-sip]
exten => _X!,1,Goto(from-trunk,${CUT(CUT(SIP_HEADER(To),@,1),:,2)},1)
```

7. Press **Esc**, and then type **ZZ** (all caps) to save the file and exit the Vim editor.

8. Log out of your FreePBX using the **logout** command. This will close PuTTY as well.

Editing the PEER Details to Point to the extensions_custom.conf File

Now that the correct code has been entered into the extensions_custom.conf file, we need the FreePBX to look at that line of code to know how to interpret incoming calls. Follow the steps below to accomplish this:

1. Go back to the **FreePBX Administration Console**. If you logged out earlier, please follow the steps in the **How to Log In to the FreePBX Administration Console** section.
2. Click the **Connectivity** button at the top of the screen, and then click **Trunks** on the drop-down menu that appears.
3. Click on the trunk you created earlier on the right side of the screen.
4. Go to the PEER Details text box and change **context=from-trunk** to **context=custom-get-did-from-sip** (Figure 1-16)

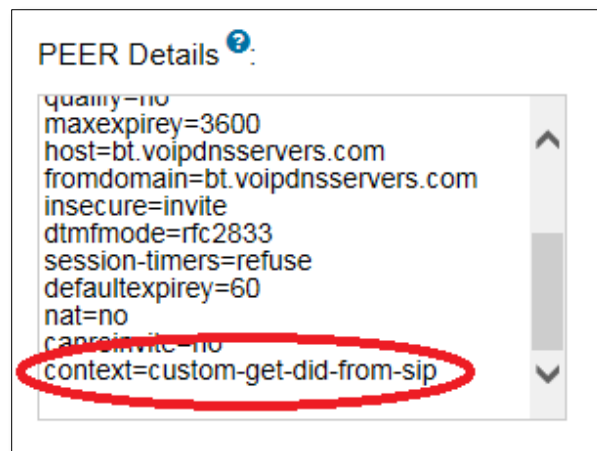


Figure 1-16: Peer Details

5. Click the **Submit Changes** button at the bottom of the screen.
6. Click the red **Apply Config** button at the top of the screen.