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Asterisk SIP Trunk Setting Example

Introduction:

Most of our customer using Asterisk opensource platform has different user interface for configuring the Asterisk PBX server. FreePBX and Trixbox are among the most popular one. Those interfaces can vary slightly depending on the version. However, most of the basic settings are the same

Using freePBX/Trixbox you are able to do most of Asterisk's configuration without editing the individual configuration files such as sip.conf or extensions.conf. You can setup most of the features in web interface such as sip trunk, call routing, voicemail and other calling features. Below we will focus on the SIP trunk setup and parameters that will work with TieUs SIP Trunk Services.

SIP Trunk Configuration:

Here we will configure Asterisk through the TrixBox administrative interface to properly route both incoming and outgoing calls to and from TieUs. This guide assumes that you have installed TrixBox or TrixBox CE. Please make sure you have already completed the installation and that you have administrative access to the TrixBox administration web interface.

1. Login into TrixBox administrative interface, under PBX select PBX Setting
2. Select **Trunks** in left side navigation, and Select "**Add SIP Trunk**" in the middle of page
3. Once you are in "Add SIP Trunk" detail Page, scroll to the "**Outgoing Settings**" section
4. Give it a Trunk Name, ex, TIEUS_SIP
5. Under PEER Details, copy and paste the following sample, if your asterisk is version 1.4, 1.6 or 1.8, and replace the user id and password to your own id/password.

```
context=from-pstn
fromdomain=209.139.240.95
fromuser=60428812741344
host=209.139.240.95
insecure=port,invite
secret=xxxxx
type=peer
defaultuser=60428812741344
```

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For asterisk version before 1.4 or older version such as 1.2 copy the following

context=from-pstn
fromdomain=209.139.240.95
fromuser=60428812741344
host=209.139.240.95
insecure=port,invite
secret=xxxxx
type=peer
defaultuser=60428812741344

Where 60428812741344 is your user id which is your DID number plus 4 secure digits. The secret is the password for your account when you sign up our services.

6. Scroll down to registration, enter your registration string in the following format

60428812741344:xxxxx@209.139.240.95/60428812741344

Again, you should replace the user id and password to yours.

7. Click on Submit Changes to save the page
8. Click on the red bar on top of the page that has Apply Configuration Change to take effect.

Outbound Route Configuration:

Out bound route will direct calls that meet certain dial pattern to your desired service provider, in this case TieUs.

1. under PBX Setting, Click on the Outbound Routes to configure your Asterisk box to send traffic to TieUs
2. Under Add Route Page, Enter a route name in Route Name field, ex, TO_TIEUS
3. Scroll down to Trunk Sequence and select the SIP/TIEUS_SIP trunk from the drop down list (if you have setup the trunk as TIEUS_SIP in previous page)
4. Click on Submit Changes, and click on the Apply Configuration Change to take effect.

Extension Configuration:

Most of customer will create at least one or two extension, an extension is an account on your Asterisk PBX box that allow other end-point (hardware IP-Phone or Softphone) to registered as extension.

PS: If you already have setup some extensions, please skip this sections

1. under PBX setting, click on Extensions and select Generic SIP device on drop down list.
2. Enter the extension number under User Extension field, for example, 1000

3. Enter desired Display name. (ex, John, or Extension 1000)
4. Enter the desired password in the Secret field.
5. Click on Submit Changes, and click on the Apply Configuration Change to take effect.

Inbound Route Configuration (receive incoming call)

Inbound call routing could be complex, depend on your design. Some customer wants the incoming call to be picked up by system auto attendant. Some customer wants the incoming call with certain DID routed to certain extension. Here we provide an example to receive the Incoming call from TIEUS_SIP sip trunk and direct the call to extension 1000.

1. under PBX setting, click Inbound Routes
2. Most of customer will have a TieUs SIP account that bound with a DID number, we will create one route for this purpose.
3. Under Add Incoming Route section, enter your assigned user id in the DID Number field, for the previous example company, we will put 60428812741344 as the DID Number. Leave the Caller ID Number field blank.
(Please note, the DID Number fields should be your real bounded DID number with the extra 4 digit, total 14 digits)
5. Now go to the Set Destination section, and select Extension Option, from the drop down list, select your extensions. (in this example, select extension 1000)
6. After Submit Changes, you should see anew Inbound Route Entry Names 60428812741344/Any CID
7. click on the Apply Configuration Change to take effect

Making Test Call and Trouble Shooting

1. Make outgoing call from your extension
2. Make incoming call by calling the DID from other phone to see if destination extension will ring.
3. If you have trouble makes call or receives call. Try to increase verbose log level in Asterisk Console. “asterisk –vvvvvr”
This way, you will able to see some warning log about why the call cannot go out and come in.
4. Call our technical support for further assistance.