

Gsm Gateway Connect with Asterisk

The picture below shows you how to enable the ssh login settings.

The screenshot shows the VoxStack Wireless Gateway web interface. At the top, there is a navigation bar with tabs: SYSTEM (highlighted in red), GSM, SIP, ROUTING, and SMS. Below the navigation bar, there are links for Status, Time, Login Settings (highlighted in green), General, and Cluster. The main content area has a 'SYSTEM DETAILS' section with a green gear icon. To the right, there is a 'Free Commun' logo with a mobile phone and antenna icons. The 'Login Settings' section contains two tabs: 'Web Login Settings' and 'SSH Login Settings'. The 'SSH Login Settings' tab is active, showing the following configuration:

User Name:	super
Password:	admin
Confirm Password:	
Port:	12345

The 'Enable' field is set to 'ON'. The 'User Name' and 'Password' fields are highlighted with a red box, indicating they are the focus of the step.

Step1: Enter the gsm gateway via ssh.

The screenshot shows the PuTTY session configuration window. On the left, there is a tree view of connection options: Session, Logging, Terminal, Window, Appearance, Behaviour, Translation, Selection, Colours, Connection, Data, Proxy, Telnet, Rlogin, SSH, and Serial. The 'SSH' option is expanded. On the right, the main configuration pane is shown:

- Basic options for your PuTTY session**
- Specify the destination you want to connect to
- Host Name (or IP address)**: 172.16.8.46 (highlighted with a red arrow pointing to it, labeled 'gsm gateway ip')
- Port**: 12345
- Connection type**:
 - Raw
 - Telnet
 - Rlogin
 - SSH
 - Serial
- Load, save or delete a stored session**
- Saved Sessions**: Default Settings
- Default Settings**: Load, Save, Delete
- Close window on exit:**
 - Always
 - Never
 - Only on clean exit

At the bottom, there are 'About', 'Open' (highlighted in blue), and 'Cancel' buttons.

Edit the /etc/asterisk/modules.conf, change the ";load => chan_iax2.so" to "load => chan_iax2.so" at the end of the file.

Step 2: config the IAX trunk settings as below:

1. none mode (no secret or the gateway will not send a call out successfully.)

step 1: config the iax trunk.

Edit the /etc/asterisk/iax.conf file of gsm gateway, and add a trunk as below. After this, you need to reload the file in the asterisk.

```
[1001]
host=***Asterisk Server IP***
type=friend
context=from-iax
```

Create a IAX trunk for the Asterisk server like below.

```
[1001]
host=***GSM gateway IP***
type=friend
context=from-iax
```

step 2 : config a sip extension for testing in both gateway and your Asterisk server

GSM Gateway

```
[1112]
username=1112
secret=1112
host=dynamic
type=friend
context=from-internal
```

Asterisk Server

```
[16200]
username=16200
secret=16200
type=friend
host=dynamic
context=from-internal
```

step 3 : config the dialplan \

The dial pattern about IAX2 must be Dial(IAX2/sip username@server IP, or the call will drop.

Config the dialplan of your gsm gateway

```
[from-internal]
exten => _X.,1,Dial(IAX2/1001@you PBX IP/${EXTEN})
exten => X.,n,Hangup()
[from-iax]
exten => X.,1,Dial(SIP/${EXTEN})
exten => X.,n,Hangup()
```

config the dialplan of your asterisk server

```
[from-internal]
exten => X.,1,Dial(IAX2/1001@gsm gateway IP/${EXTEN})
exten => _X.,2,hangup()
```

```
[from-iax]
exten => _X.,1,Dial(SIP/${EXTEN})
exten => _X.,n,Hangup()
```

step 4 : make calls to test

2. register mode (this gateway register to asterisk server)

step 1: config the iax trunk

Edit the /etc/asterisk/iax.conf file of gsm gateway, and add a trunk as below. After this, you need to reload the file in the asterisk.

```
[general]
register => 1001@***Asterisk Server IP***
[1001]
host=***Asterisk Server IP***
type=friend
context=from-iax
```

Create an IAX trunk in the Asterisk server like below

```
[1001]
host=dynamic
type=friend
context=from-iax
```

step 2 : config the sip extension for testing

step 3 : config the dialplan

Config the dialplan of your gsm gateway

```
[from-internal]
exten => _X.,1,Dial(IAX2/1001@Asterisk Server IP/${EXTEN})
exten => _X.,n,Hangup()
[from-iax]
exten => _X.,1,Dial(SIP/${EXTEN})
exten => _X.,n,Hangup()
```

config the dialplan of your Asterisk Server

```
[from-internal]
exten => _X.,1,Dial(IAX2/1001@gsm gateway IP/${EXTEN})
exten => _X.,2,hangup()
[from-iax]
exten => _X.,1,Dial(SIP/${EXTEN})
exten => _X.,n,Hangup()
```

step 4 : make calls to test

Notice: if you just using IAX2 to make calls from iax2 trunk to gsm ports, then you can add the custom dialplan in the extensions_custom.conf file.

If you want to make calls incoming from gsm ports to iax2 trunk, follow the steps below:

step 1: login the gateway via ssh.

To do this, you need go to SYSTEM ----> Login Settings and config ssh as following figure.

warning: the user name must be super.

SSH Login Settings	
Enable:	<input checked="" type="checkbox"/> ON
User Name:	super
Password:	admin
Port:	12345

step 2: edit the file /etc/asterisk/extensions_custom.conf and create your own dialplan.

step 3: execute following command:

vi /etc/asterisk/gw/custom.sh

and add two lines:

```
echo > /etc/asterisk/extensions_routing.conf // this will clean up the file  
extensions_routing.conf
```

```
asterisk -rx "core reload"
```

step 4: run the command: /my_tools/sync2flash

step 5: reboot the gateway