SIP Server Configuration Guide

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SIP server on CentOS

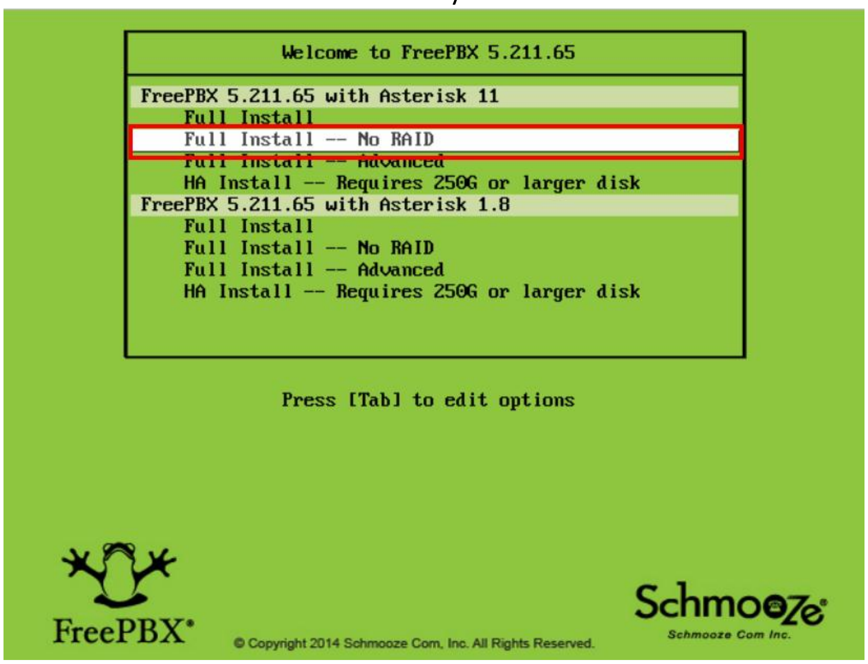
The instructions below will show you how to install CentOS, Asterisk, and FreePBX which are required to operate a SIP Server.

Installing FreePBX

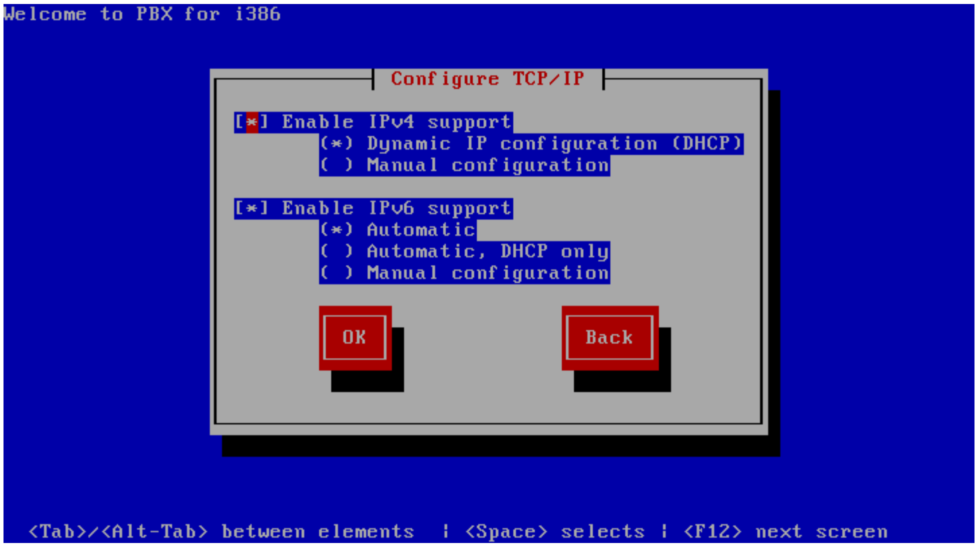
Please refer to the following link for a detailed installation guide.

Warning: Everything on the computer will be deleted and replaced with the FreeBPX Distro when it is installed.

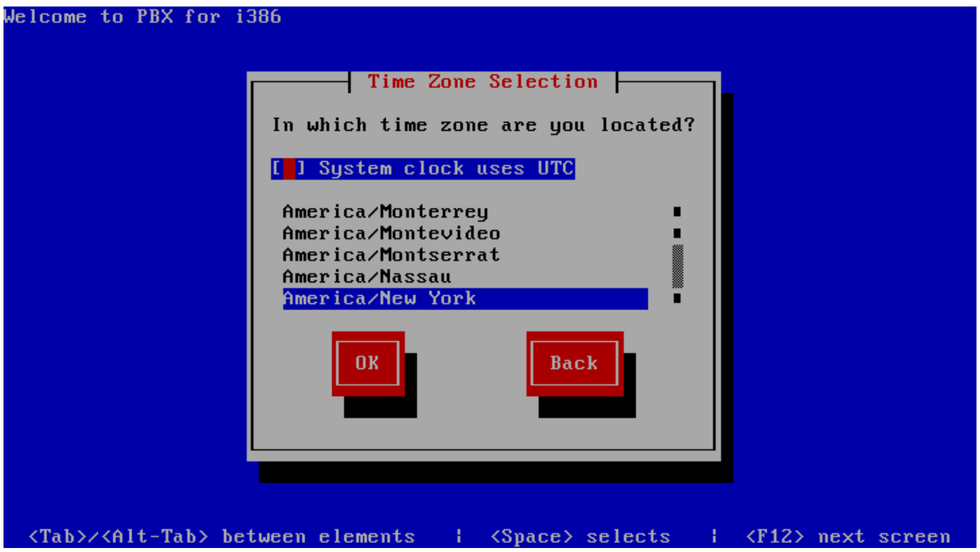
1. Download the ISO file from http://www.freepbx.org/downloads.
2. **Full Install – No RAID** is usually selected as the install method.



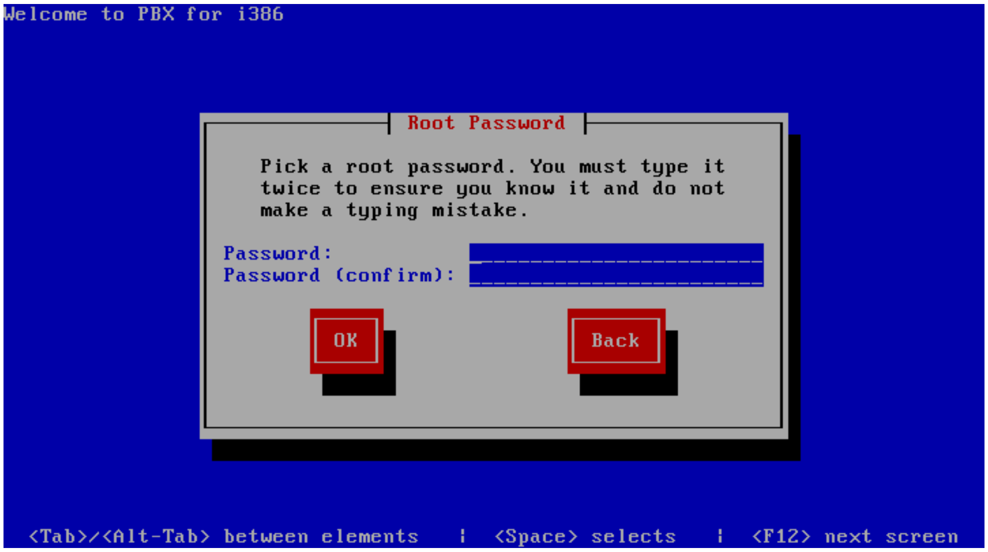
2. Click **OK**.



3. Select the time zone of your city.



4. Enter a password for the root account. This password will be used as the localhost login password later.

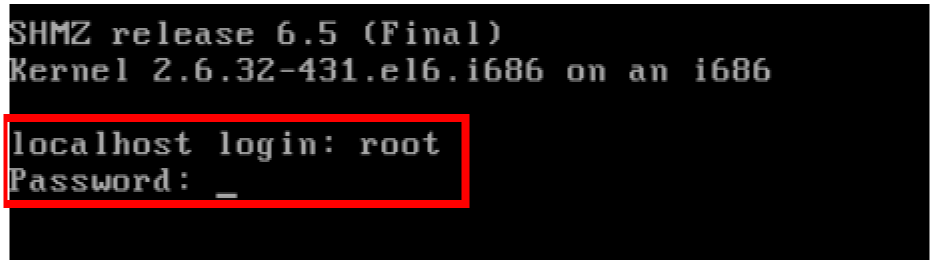


5. Continue on with the following screens to finish the installation.

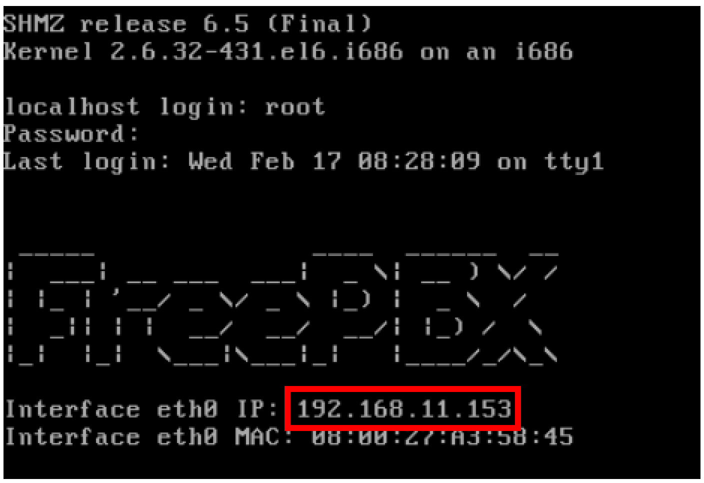
Configuring Asterisk with FreePBX

Please follow the instructions below to configure Asterisk with FreePBX.

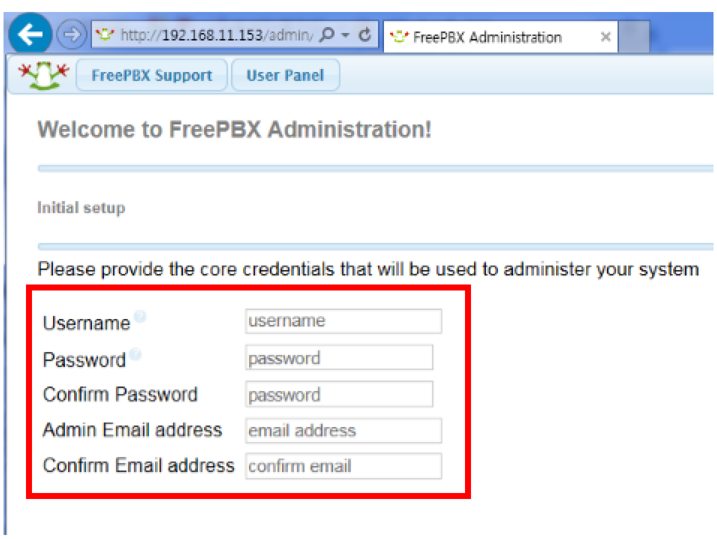
1. Enter root and its password you configured previously during the installation for the localhost login.



2. Copy FreePBX Ethernet interface IP address.

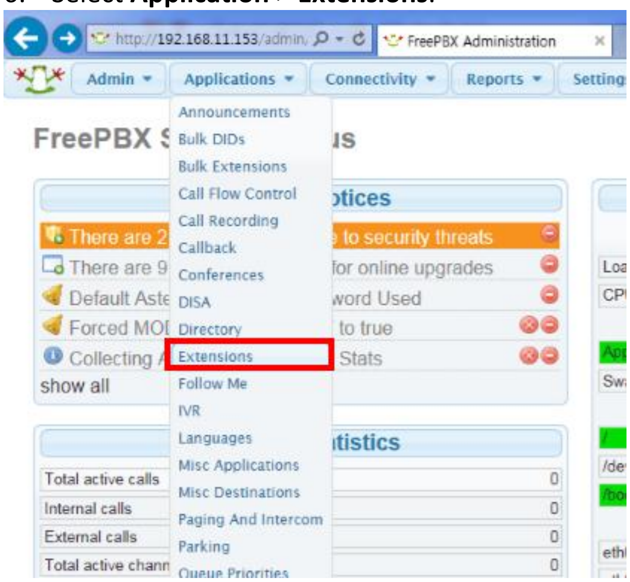


3. Enter the IP address on a web browser.
4. Configure a new FreePBX Administration account.

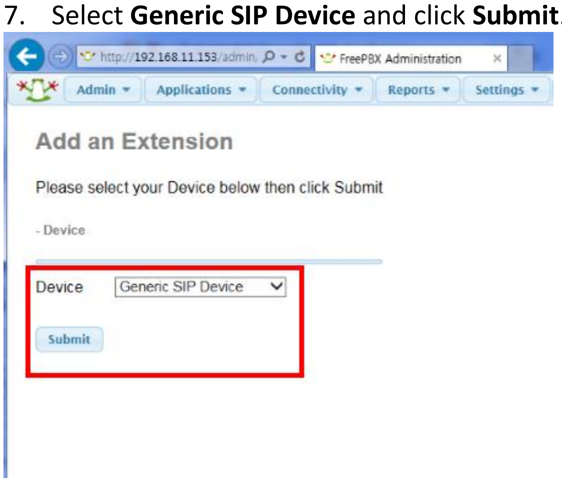


5. Click **FreePBX Administration** and login with the FreePBX administration account you created previously.

6. Select **Application** > **Extensions**.



7. Select **Generic SIP Device** and click **Submit**.



8. Add user 101 with the details shown below.

- User Extension : 101

- Display Name : 101

- CID Num Alias : 101

- SIP Alias : 101

- Secret : suprema101

  - Password must be at least 8 digits long. You must use at least 2 characters and use it in combination with numbers

- Dtmfmode : RFC2833

- Nat : Yes

9. Add user 102 in the same way with the details below.

- User Extension : 102

- Display Name : 102

- CID Num Alias : 102

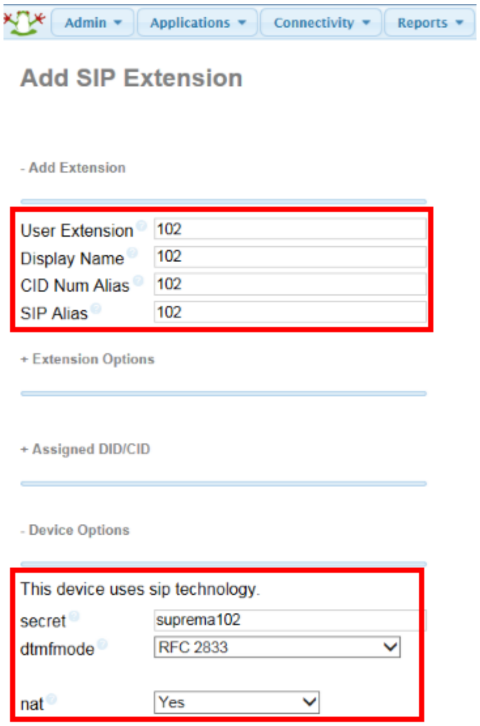
- SIP Alias : 102

- Secret : suprema102

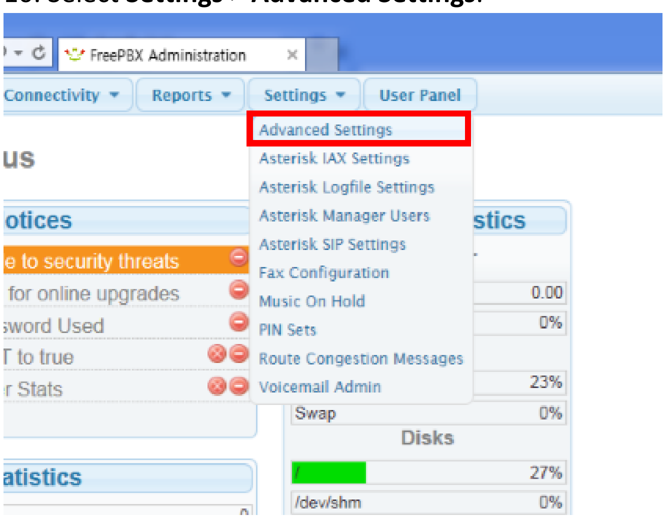
- Password must be at least 8 digits long. You must use at least 2 characters and use it in combination with numbers)

- Dtmfmode : RFC2833

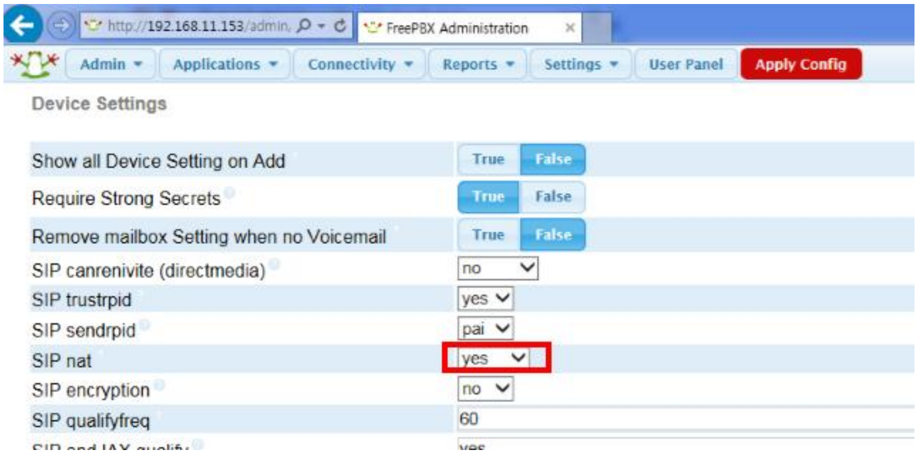
- Nat : Yes



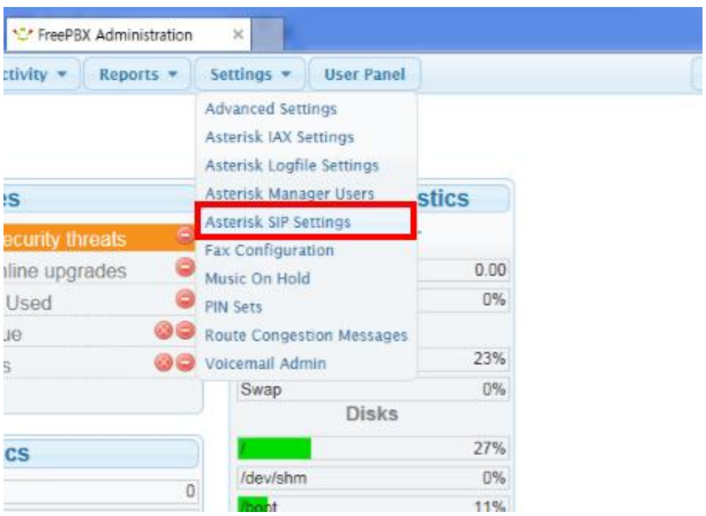
10. Select **Settings** > **Advanced Settings**.



11. Click on the SIP nat dropdown list and select **Yes**.



12. Select **Settings** > **Asterisk SIP Settings**.



13. Configure as shown below.

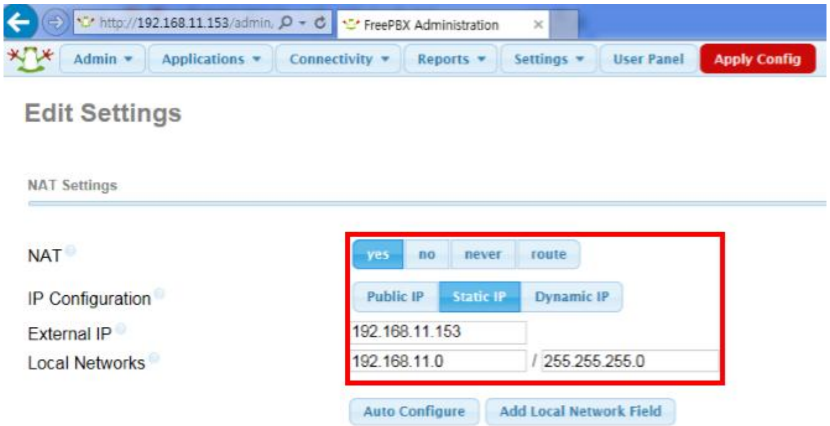
NAT : yes

IP Configuration : Static IP

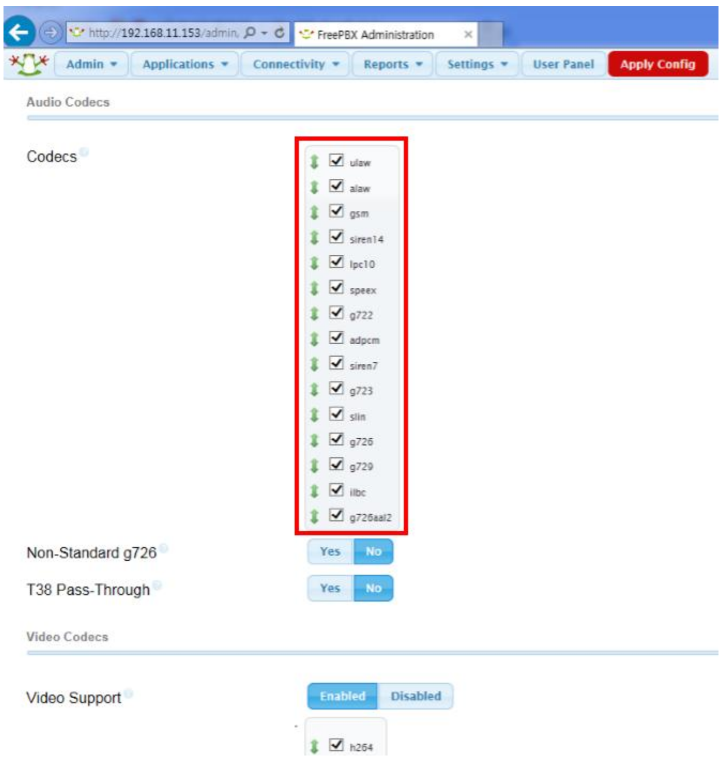
External IP : enter the FreePBX IP interface address.

Local Networks : Private IP range / subnet

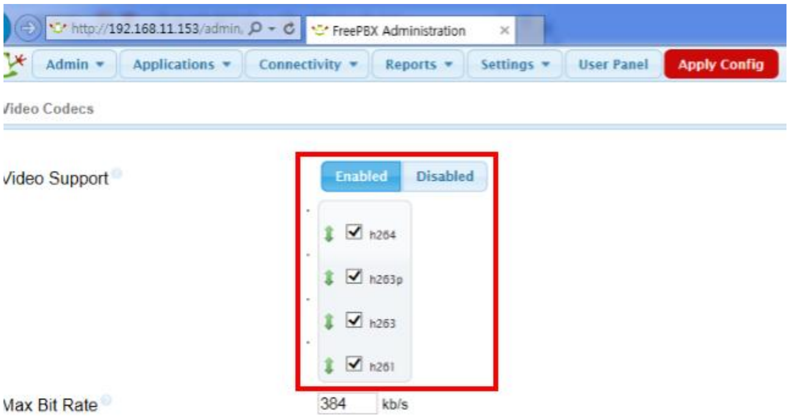
- Example: If the IP provided by the router is 192.168.11.153, the IP range is 192.168.11.0 and the subnet is 255.255.255.0 (C class)
- Please refer to the following subnetting reference.



14. Select all audio **Codecs**.



15. Set **Video Support** to **Enabled** and select all video codecs.

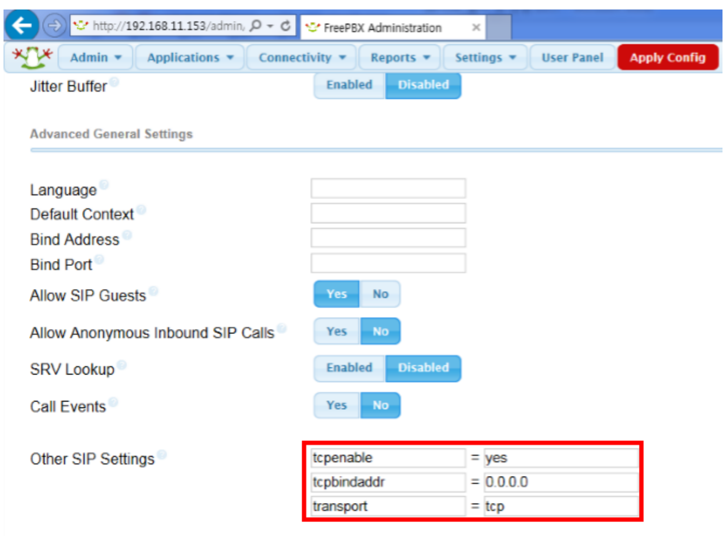


16. Add the following items on **Other SIP Settings**.

- tcpenable = yes

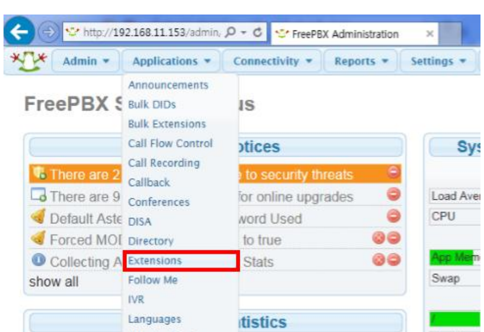
- tcpbindaddr = 0.0.0.0

- transport = tcp

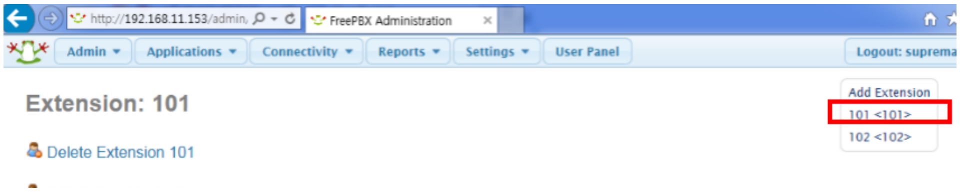


17. Click **Submit Changes**.

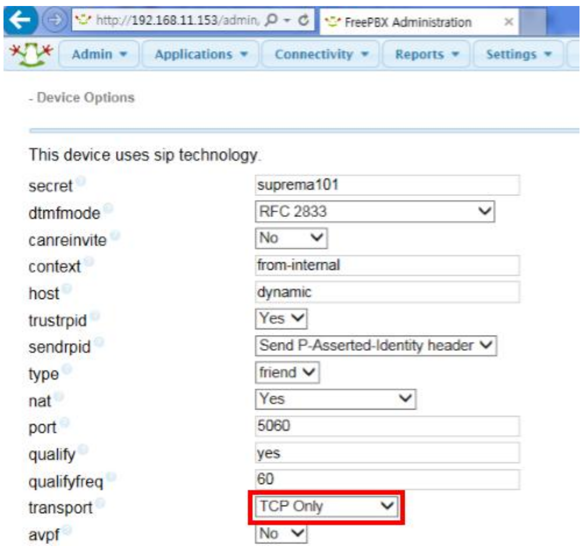
18. Enter **Application** > **Extensions** again.



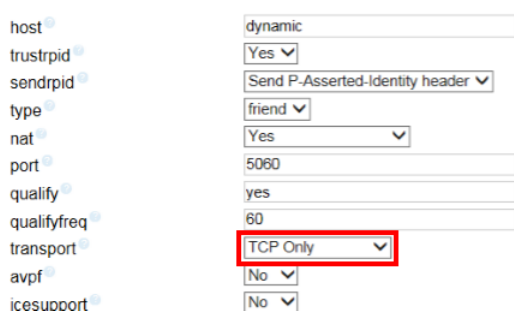
19. At the right top corner of the screen, select user 101 which you added before.



20. Click on the **transport** dropdown box and select **TCP Only**.



21. Configure user 102 the same as user 101 with **transport** : **TCP Only**.



20. Click on **Apply Config** on the top menu bar to apply the configuration.

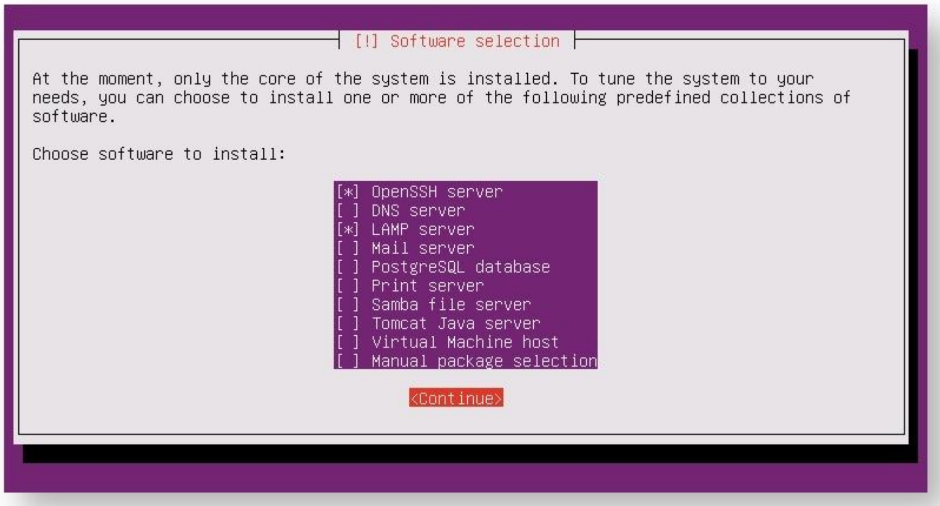
SIP Server on Ubuntu

Below are instructions on how to install Ubuntu14.04 LTS, Asterisk, and Free PBX to setup your SIP Server.

Please refer to additional warnings regarding the installation of FreePBX 12 on Ubuntu Server 14.04 LTS in the following link. Please note that the instructions in the link are not identical as shown below.

Installing Ubuntu

1. On the Software selection page, make sure that you select 'OpenSSH Server' and 'LAMP Packages' to install the essential packages.



2. Configure your root password by entering the command below (excluding the #).

```
# sudo passwd root
```

3. Switch to the Root User by entering the command below.

Note: You must run the *entire* process as root. Attempting to use 'sudo' later on *will not work*. You must run this command to switch to an interactive root shell.

```
# sudo –i
```

4. Update Your System by entering the command below.

```
# apt-get update && apt-get upgrade –y
```

5. Install required dependencies by entering the commands below.

```
Apt-get install -y build-essential linux-headers-`uname -r` openssh-server apache2 mysql-server\
 mysql-client bison flex php5 php5-curl php5-cli php5-mysql php-pear php-db php5-gd curl sox\
 libncurses5-dev libssl-dev libmysqlclient-dev mpg123 libxml2-dev libnewt-dev sqlite3\
 libsqlite3-dev pkg-config automake libtool autoconf git subversion unixodbc-dev uuid uuid-dev\
 libasound2-dev libogg-dev libvorbis-dev libcurl4-openssl-dev libical-dev libneon27-dev libsrtp0-
dev\
 libspandsp-dev libmyodbc
```

6. Reboot the server by entering the command below.

```
# reboot
```

7. Install Dependencies for Google Voice (if required)

You may skip this section if you do not require Google Voice support.

8. Install iksemel by typing the commands below.

```
# cd /usr/src
# wget https://iksemel.googlecode.com/files/iksemel-1.4.tar.gz
# tar xf iksemel-1.4.tar.gz
# cd iksemel-*
# ./configure
# make
# make install
```

Installing and Configuring Asterisk

1. Download Asterisk source files by entering the commands below.

```
# cd /usr/src
# wget http://downloads.asterisk.org/pub/telephony/dahdi-linux-complete/dahdi-linux-complete-
current.tar.gz
# wget http://downloads.asterisk.org/pub/telephony/libpri/libpri-1.4-current.tar.gz
# wget http://downloads.asterisk.org/pub/telephony/asterisk/asterisk-12-current.tar.gz
# git clone https://github.com/akheron/jansson.git
# git clone https://github.com/asterisk/pjproject.git
```

2. Compile and install DAHDI by entering the commands below. Skip this step if there is a compilation error.

If you don't have any physical hardware you don't have to run these commands.

```
# cd /usr/src
# tar xvfz dahdi-linux-complete-current.tar.gz
# cd dahdi-linux-complete-*
# make all
# make install
# make config
```

3. Compile and install LIBPRI by entering the commands below. Skip this step if there is a compilation error.

If you don't have any physical hardware you don't need to run these commands.

```
# cd /usr/src
# tar xvfz libpri-1.4-current.tar.gz
# cd libpri-*
# make
# make install
```

4. Compile and install pjproject by entering the commands below.

```
# cd /usr/src/pjproject
# ./configure --enable-shared --disable-sound --disable-resample --disable-video --disable-opencore-amr
# make dep
# make
# make install
```

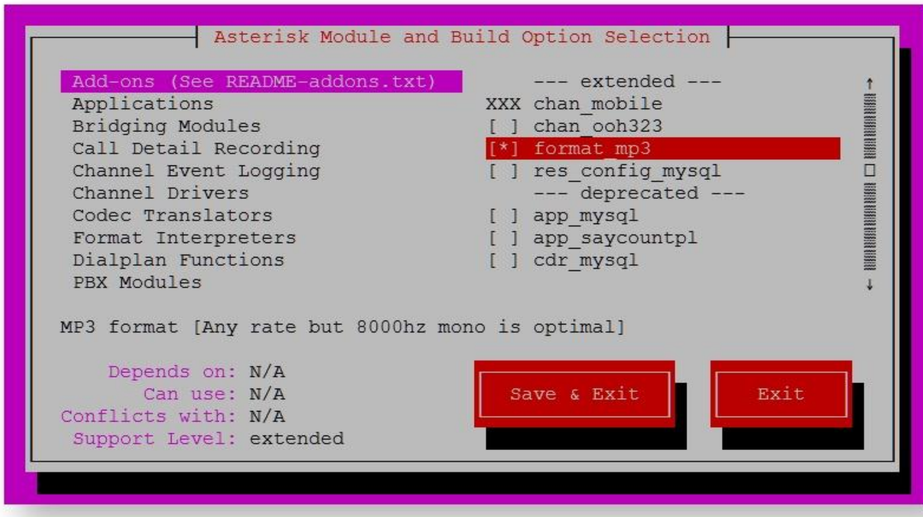
5. Compile and install jansson by entering the commands below.

```
# cd /usr/src/jansson
# autoreconf -i
# ./configure
# make
# make install
```

6. Compile and install Asterisk entering the commands below.

```
# cd /usr/src
# tar xvfz asterisk-12-current.tar.gz
# cd asterisk-*
# ./configure
# contrib/scripts/get_mp3_source.sh
# make menuselect
```

7. You will be prompted at the point to pick which modules to build. Most of them will be enabled, but if you want to have MP3 support, you need to manually turn on 'format_mp3' on the first page.



8. Continue after selecting 'Save & Exit'.

```
# make
# make install
# make config
# ldconfig
```

9. Install Asterisk-Extra-Sounds by entering the commands below.

Note: This installs the (8khz) 'wav' soundfiles. If you're planning on running G722 (High Definition 'Wideband') audio, you also want to download the 722 codec pack, which is the second part. If you're not planning on using Wideband, you can skip that part.

```
# cd /var/lib/asterisk/sounds
# wget http://downloads.asterisk.org/pub/telephony/sounds/asterisk-extra-sounds-en-wav-
current.tar.gz
# tar xfz asterisk-extra-sounds-en-wav-current.tar.gz
# rm -f asterisk-extra-sounds-en-wav-current.tar.gz
# Wideband Audio download
# wget http://downloads.asterisk.org/pub/telephony/sounds/asterisk-extra-sounds-en-g722-
current.tar.gz
# tar xfz asterisk-extra-sounds-en-g722-current.tar.gz
# rm -f asterisk-extra-sounds-en-g722-current.tar.gz
```

Installing FreePBX

1. Download and extract FreePBX by entering the commands below.

```
# wget http://mirror.freepbx.org/modules/packages/freepbx/freepbx-12.0-latest.tgz
# tar vxfz freepbx-12-latest.tgz
# cd freepbx
```

2. Now create the Asterisk user and set ownership permissions by entering the commands below.

```
# useradd -m asterisk
# chown asterisk. /var/run/asterisk
# chown -R asterisk. /etc/asterisk
# chown -R asterisk. /var/{lib,log,spool}/asterisk
# chown -R asterisk. /usr/lib/asterisk
# rm -rf /var/www/html
```

3. Make a few small modifications to Apache by entering the commands below.

```
# sed -i 's/\(^upload_max_filesize = \).*/\120M/' /etc/php5/apache2/php.ini
# cp /etc/apache2/apache2.conf /etc/apache2/apache2.conf_orig
# sed -i 's/^\(User\|Group\).*/\1 asterisk/' /etc/apache2/apache2.conf
# service apache2 restart
```

4. Configure ODBC by editing /etc/odbcinst.ini and adding the following.

Note: this command assumes you are installing to a new machine, and that the file is empty. If this is not a freshly installed machine, please manually verify the contents of the file, rather than just copying and pasting the lines below. The 'EOF' does no go in the file, it simply signals to the 'cat' command that you have finished pasting.

```
cat >> /etc/odbcinst.ini << EOF
[MySQL]
Description = ODBC for MySQL
Driver = /usr/lib/x86_64-linux-gnu/odbc/libmyodbc.so
Setup = /usr/lib/x86_64-linux-gnu/odbc/libodbcmyS.so
FileUsage = 1

EOF
```

You may need to verify these paths, if you're not on a x86_64 machine. You can use the command `find / -name libmyodbc.so` to verify the location

Edit or create /etc/odbc.ini and add the following section. Note that, again, this command assumes you are installing to a new machine, and the file is empty. Please manually verify the contents of the files if this is not the case.

```
cat >> /etc/odbc.ini << EOF
[MySQL-asteriskcdrdb]
Description=MySQL connection to 'asteriskcdrdb' database
driver=MySQL
server=localhost
database=asteriskcdrdb
Port=3306
Socket=/var/run/mysqld/mysqld.sock
option=3

EOF
```

Preparing MySQL

1. Enter the command below to generate a secure password for FreePBX that would be used to communicate with MySQL.

```
# export ASTERISK_DB_PW=`dd if=/dev/urandom bs=1 count=32 2>/dev/null | base64 - | cut -c2-18`
```

This will generate a quasi-random 16 character long password, which should be secure enough for most things. If you had set the MySQL 'root' password to be something when you were installing the machine, you will need to add a -p*yourpassword* flag to the following lines

2. Configure Asterisk database in MYSQL by entering the commands below.

```
# mysqladmin -u root create asterisk
# mysqladmin -u root create asteriskcdrdb
```

3. Set permissions on MYSQL database by entering the commands below.

```
# mysql -u root -e "GRANT ALL PRIVILEGES ON asterisk.* TO asteriskuser@localhost IDENTIFIED BY '${ASTERISK_DB_PW}';"
# mysql -u root -e "GRANT ALL PRIVILEGES ON asteriskcdrdb.* TO asteriskuser@localhost IDENTIFIED BY '${ASTERISK_DB_PW}';"
# mysql -u root -e "flush privileges;"
```

4. Restart Asterisk and install FreePBX by entering the commands below.

```
# ./start_asterisk start
# ./install_amp --installdb --username=asteriskuser --password=${ASTERISK_DB_PW}
# amportal chown
# amportal a ma installall
# amportal a reload
# amportal a ma refreshsignatures
# amportal chown
```

5. Finally, set one last mod and start FreePBX by entering the commands below.

```
# ln -s /var/lib/asterisk/moh /var/lib/asterisk/mohmp3
# amportal restart
```

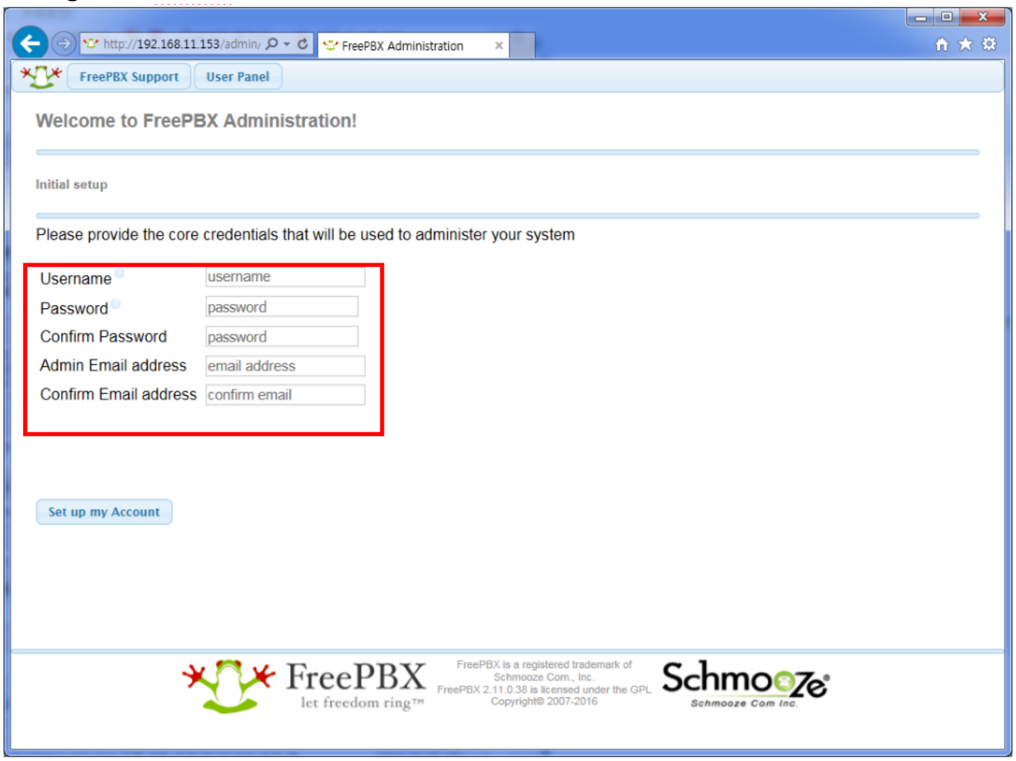
Configuring FreePBX

Since FreePBX is now installed, it will be available via Apache.

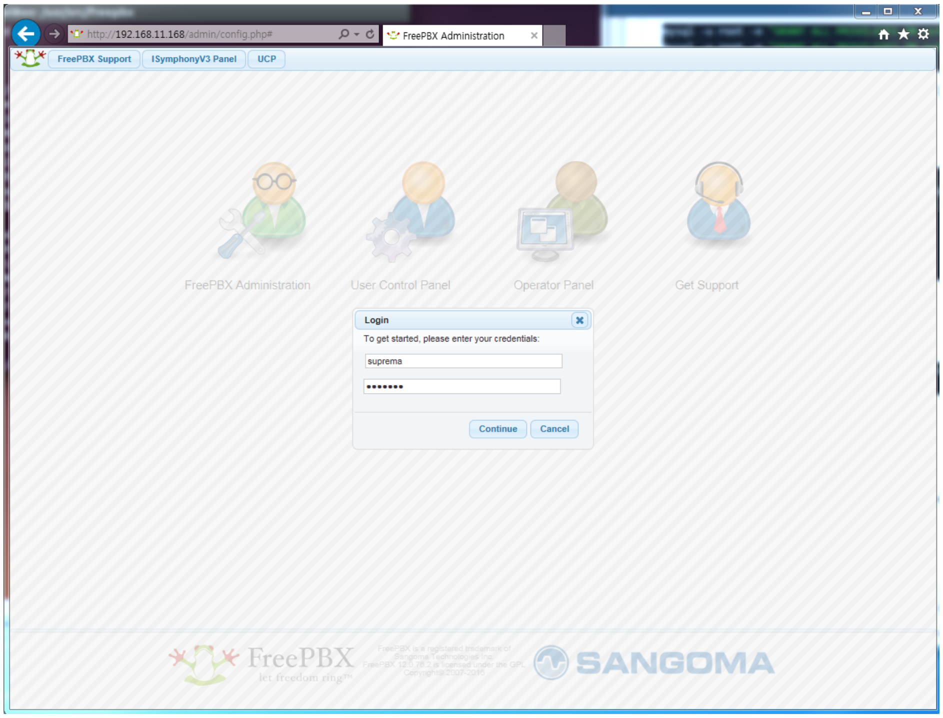
1. Open up your web browser and enter the url below.

   http://**Address.Of.FreePBX.Server**/admin (Example: http://192.168.11.148/admin)

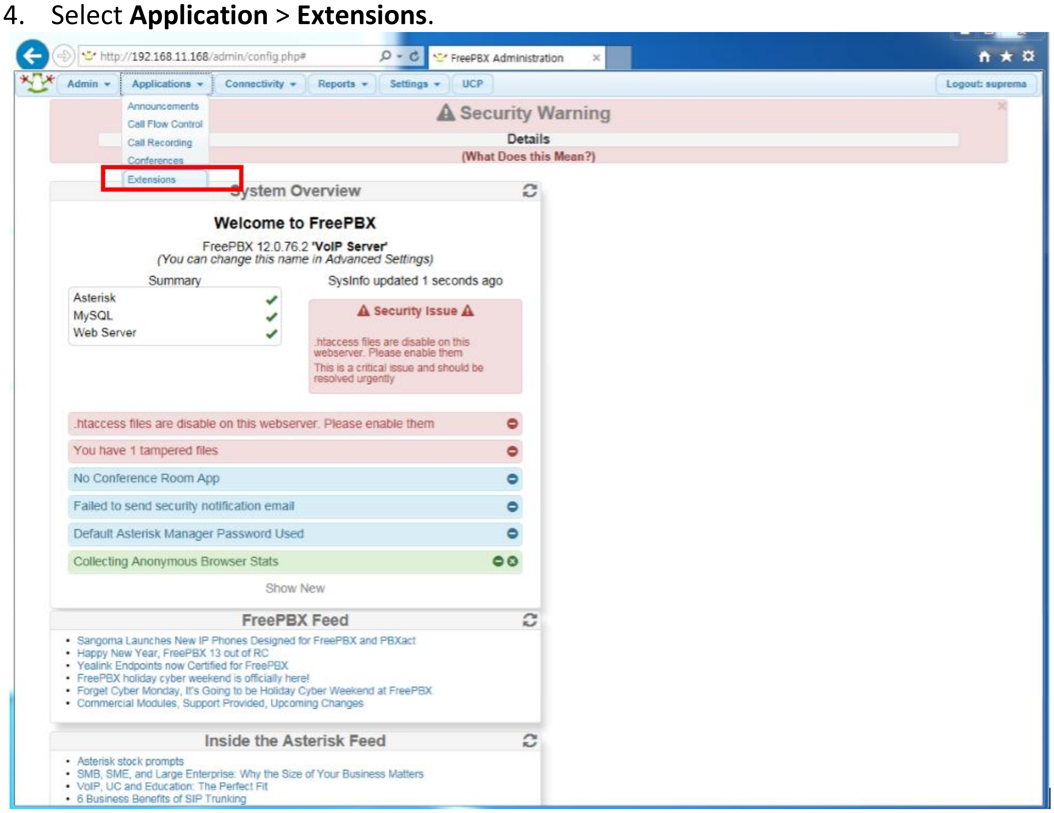
2. Configure a new FreePBX Administration account.



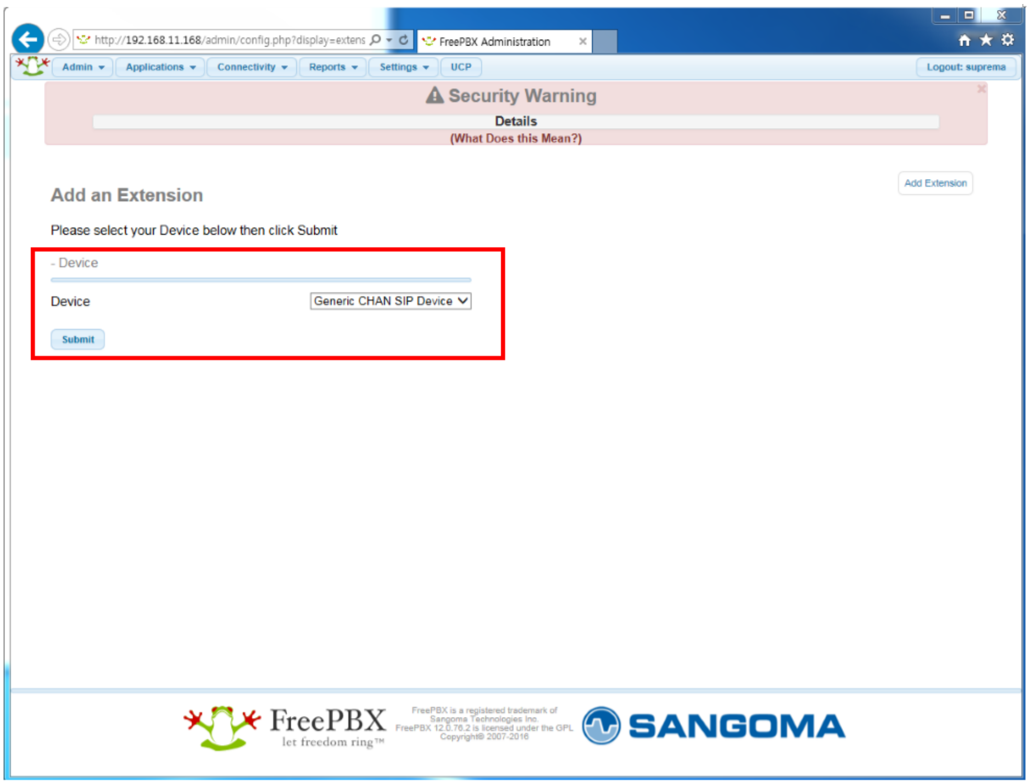
3. Click **FreePBX Administration** and login with the FreePBX administration account you created previously.



4. Select **Application** > **Extensions**.



8. Select **Generic Generic SIP Device** and click **Submit**.



9. Add user 101 with the details shown below.

- User Extension : 101

- Display Name : 101

- CID Num Alias : 101

- SIP Alias : 101

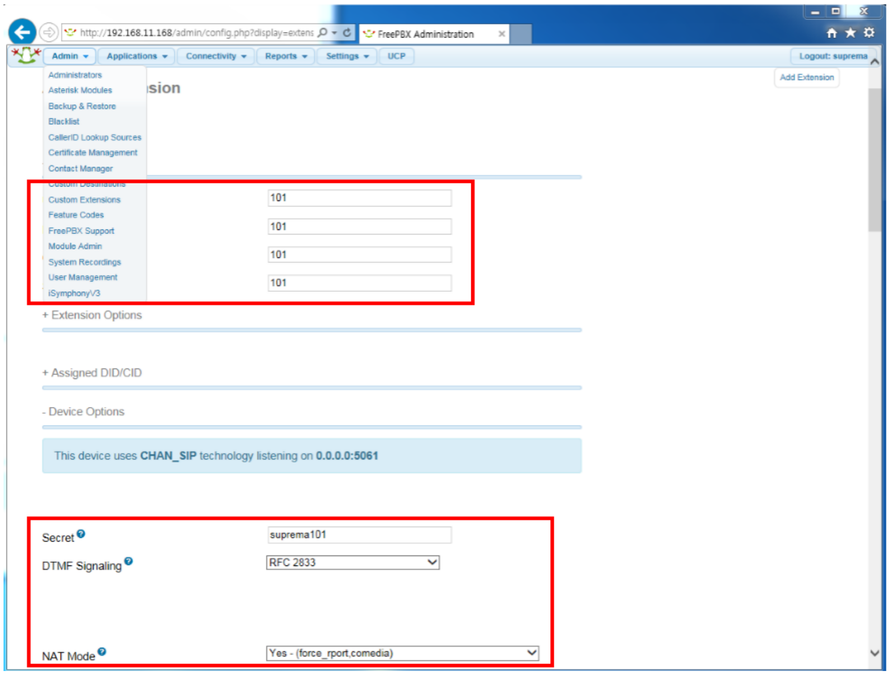
- Secret : suprema101

• Password must be at least 8 digits long. You must use at least 2 characters and use it in combination with numbers

- Dtmfmode : RFC2833

- Nat : Yes – (force_rport,comedia)

10. Click **submit** at the bottom of the page.



11. Add user 102 in the same way with the details shown below.

- User Extension : 102

- Display Name : 102

- CID Num Alias : 102

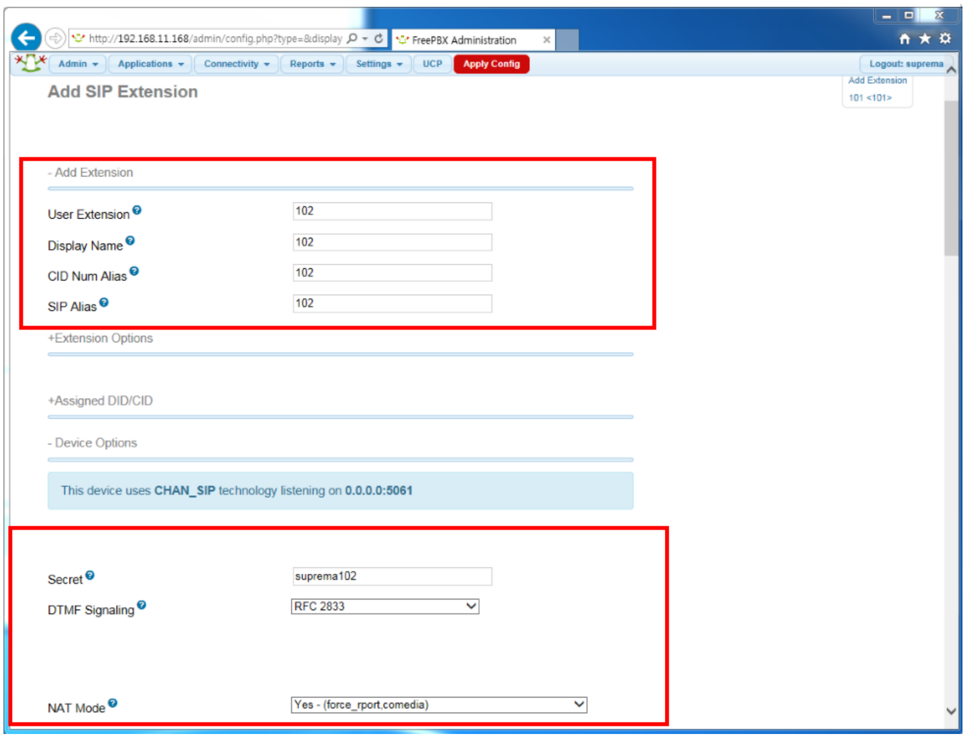
- SIP Alias : 102

- Secret : suprema102

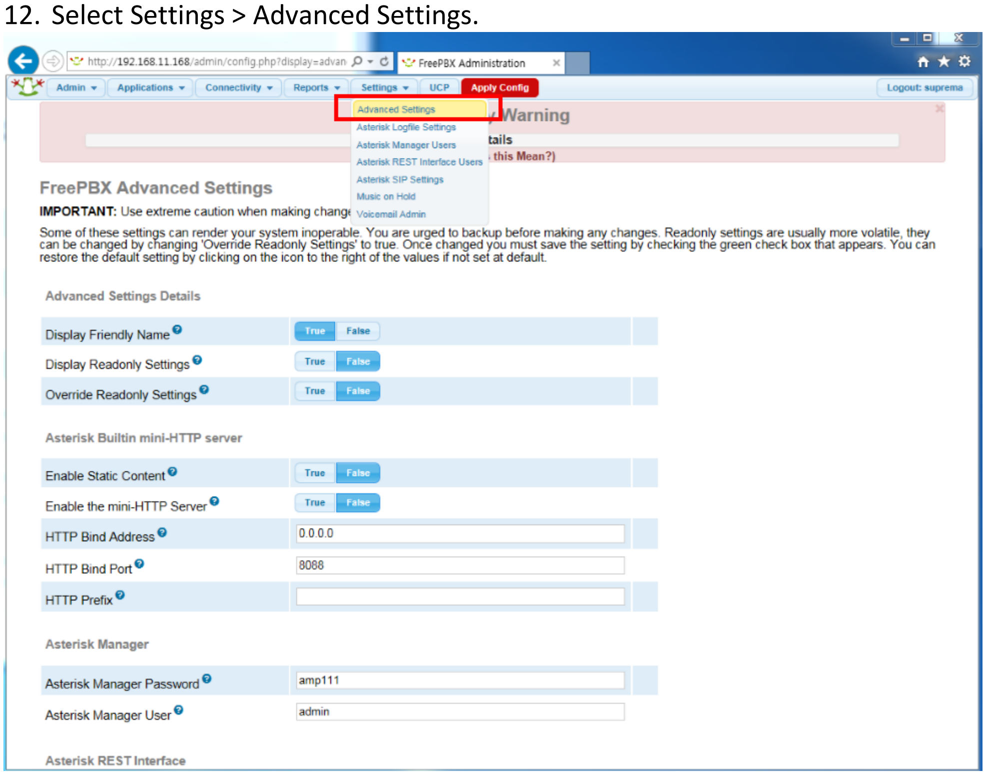
• Password must be at least 8 digits long. You must use at least 2 characters and use it in combination with numbers

- Dtmfmode : RFC2833

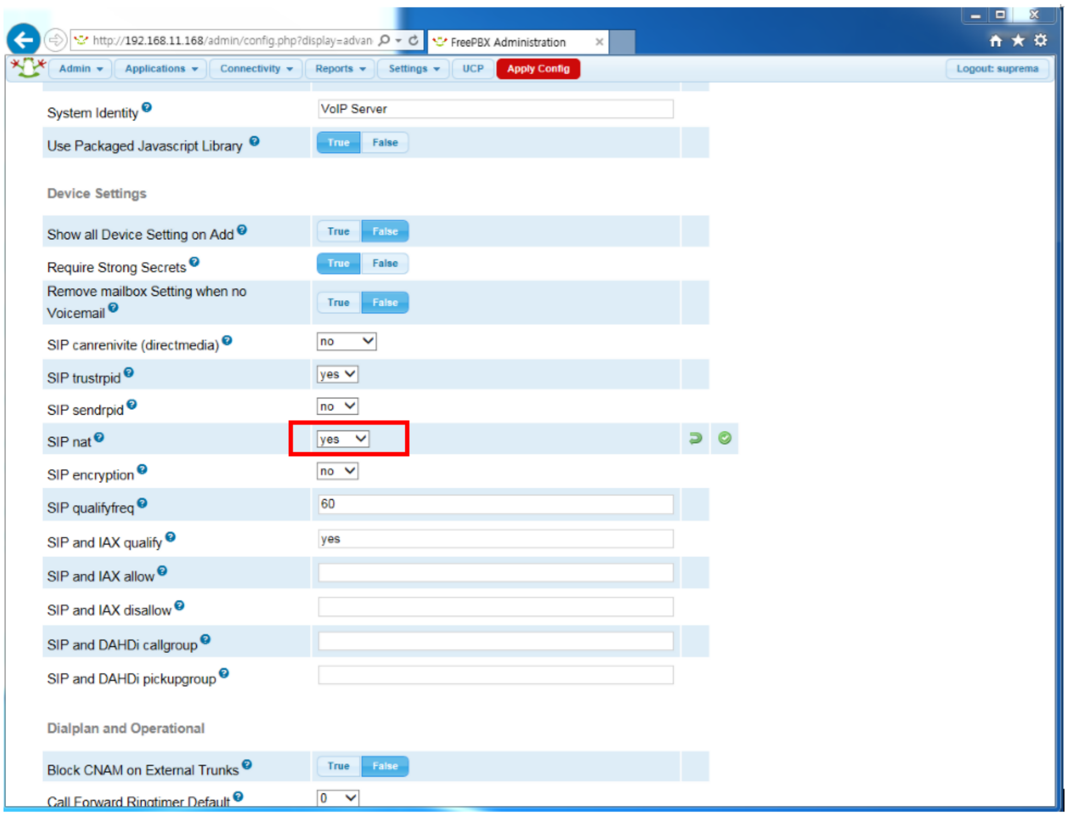
- Nat : Yes – (force_rport,comedia)



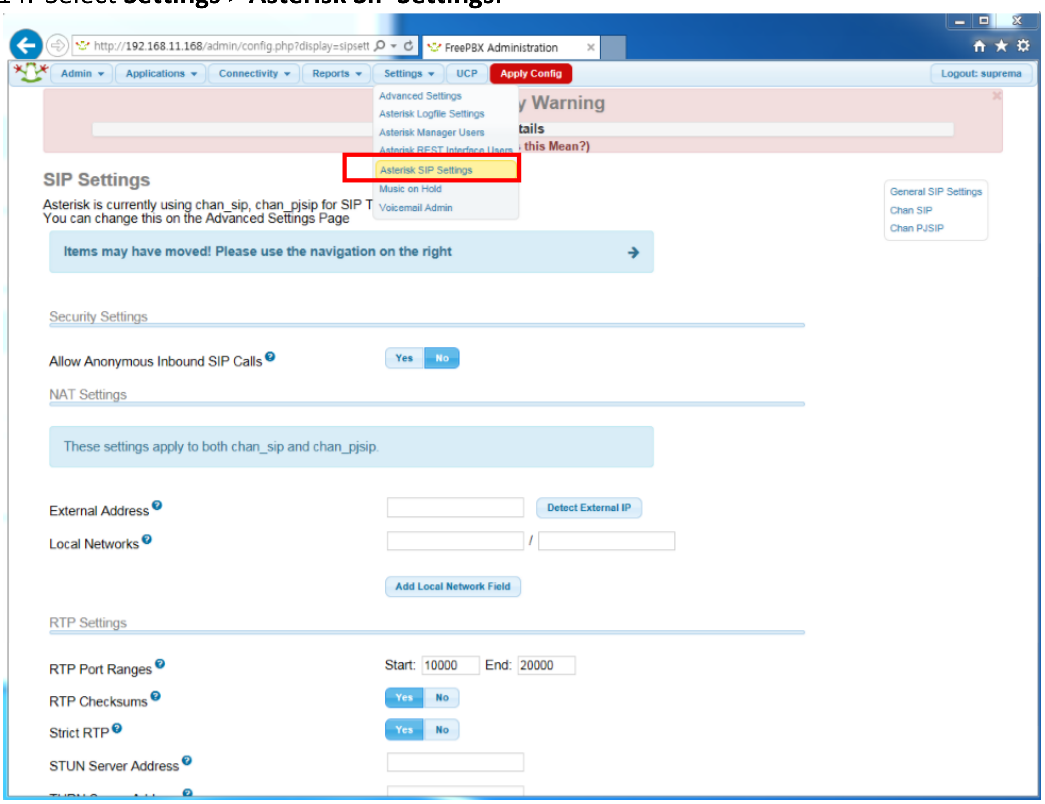
12. Select Settings > Advanced Settings.



13. Click on the SIP nat dropdown list and select **Yes**.



14. Select **Settings** > **Asterisk SIP Settings**.

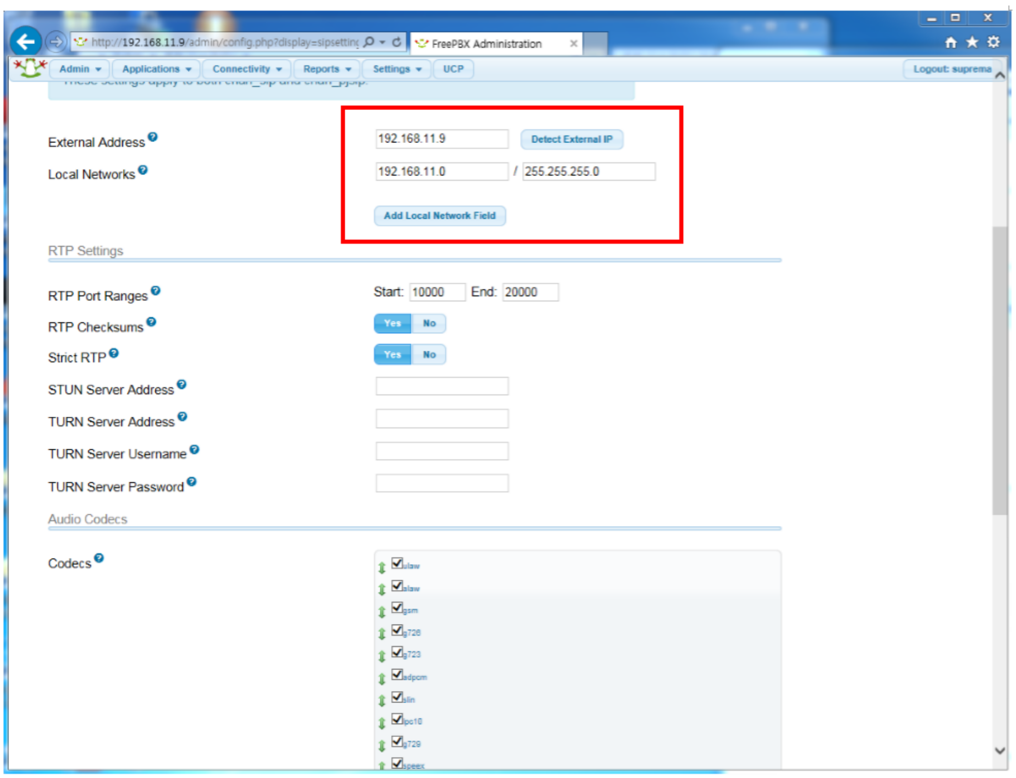


12. Configure as shown below.

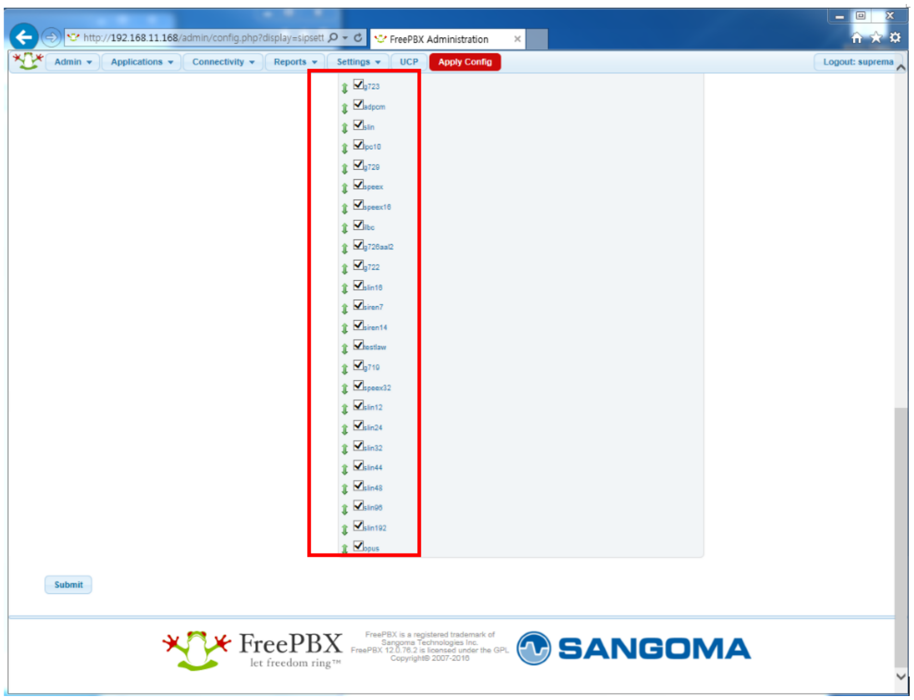
External IP : Enter the IP address

Local Networks : Private IP range / subnet

- Example: If the IP provided by the router is 192.168.11.153, the IP range is 192.168.11.0 and the subnet is 255.255.255.0 (C class)
- Please refer to the following subnetting reference.

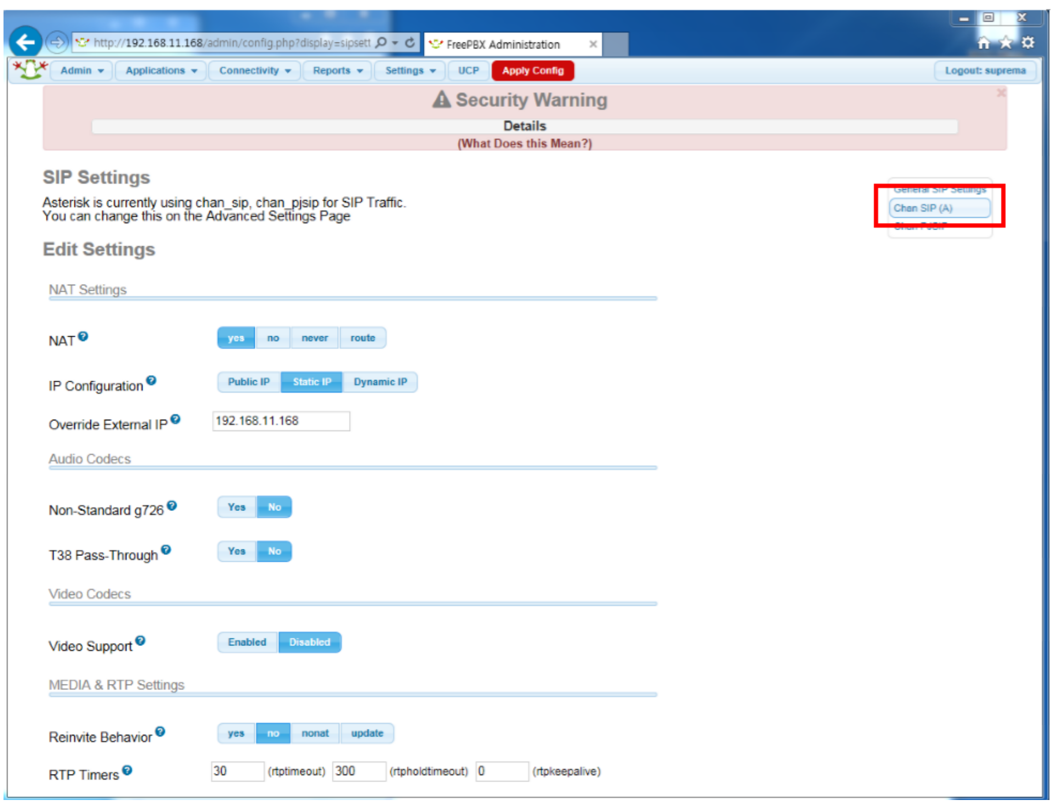


13. Select all audio **Codecs**.

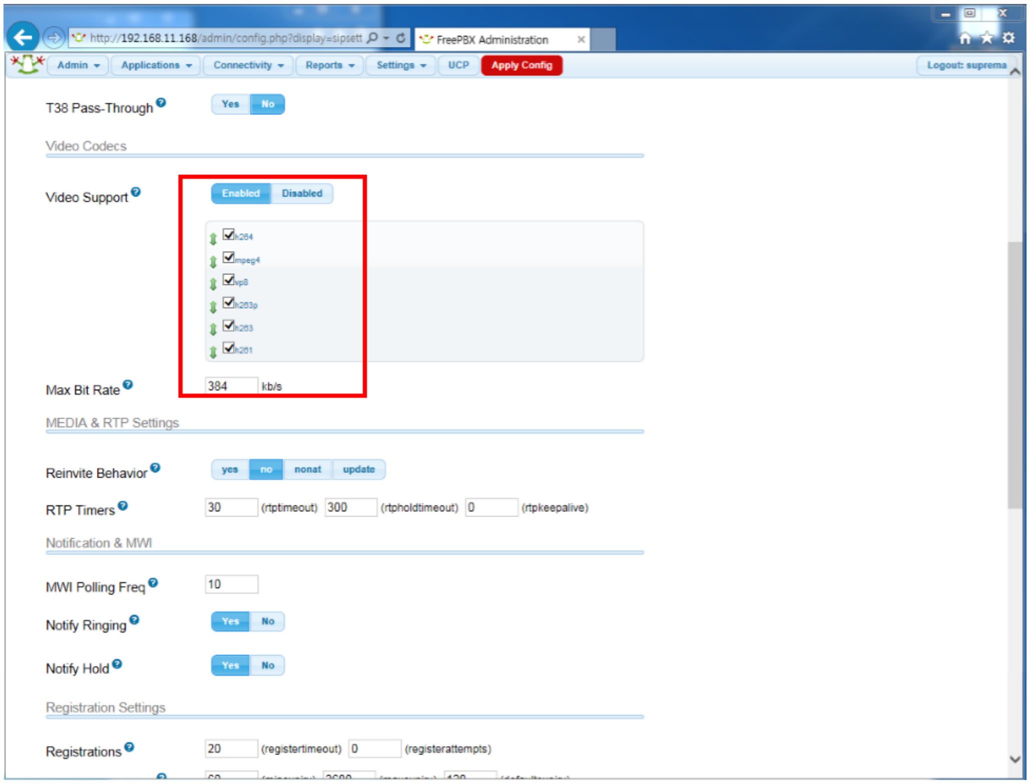


14. Press the **Submit** button.

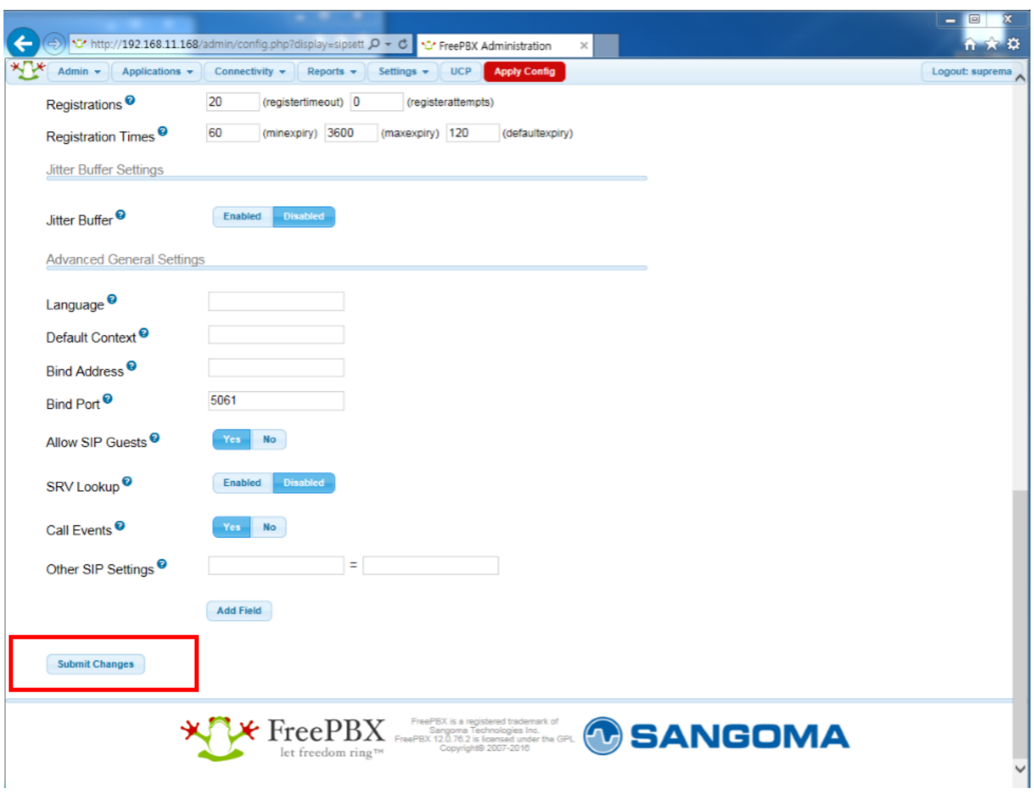
15. Switch to **Chan SIP** Configuration page.



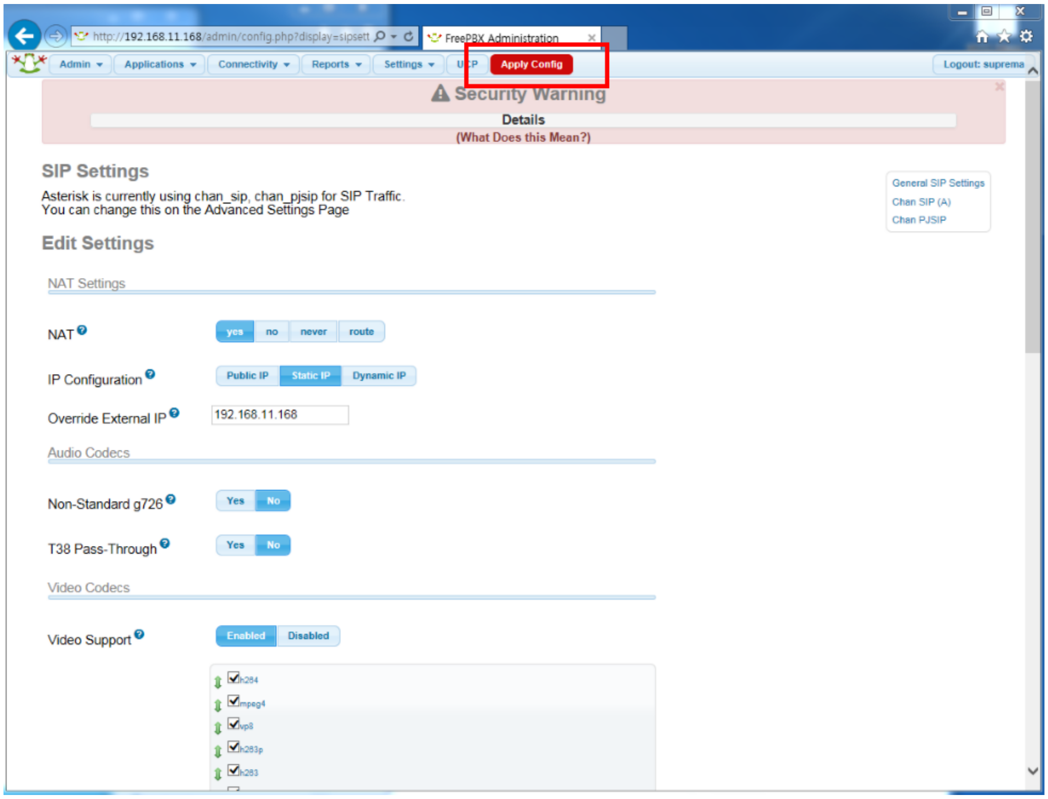
16. Set **Video Support** to **Enabled** and select all video codecs



17. Press the **Submit Changes** button on the bottom.



18. Click **Apply Config** on the top of the page to apply the settings.

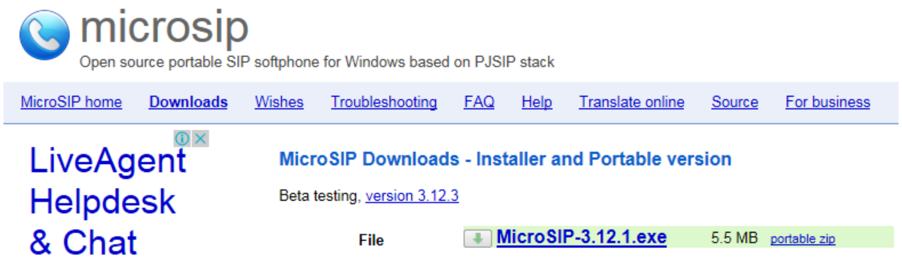


Installing a SIP Client in Windows

The SIP Clients that can be used for Windows are Linphone, MicroSIP, and Zoiper. The instructions below will show you how to install MicroSIP.

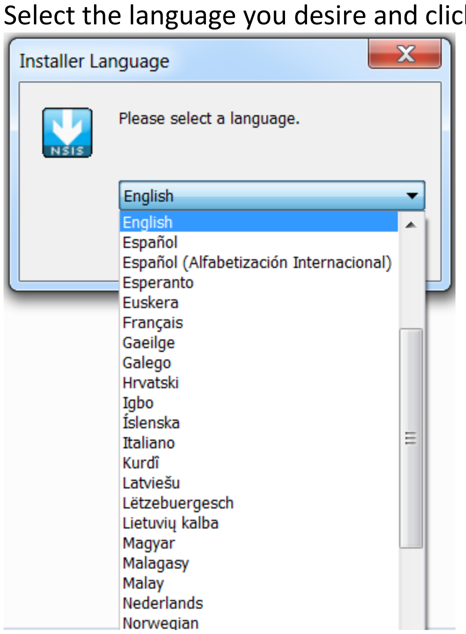
1. Download the installation file from the link below:

http://www.microsip.org/downloads

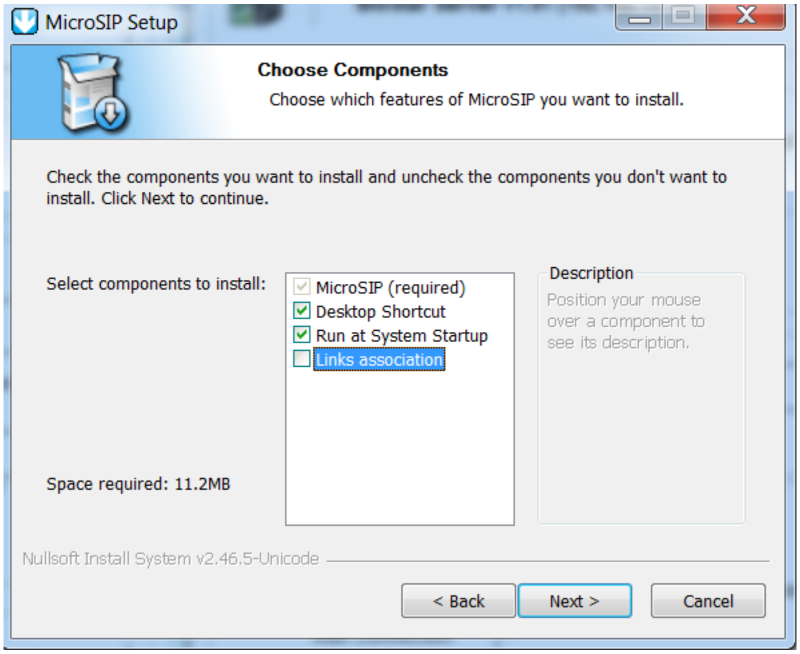


2. Run the executable file.

3. Select the language you desire and click **OK**.

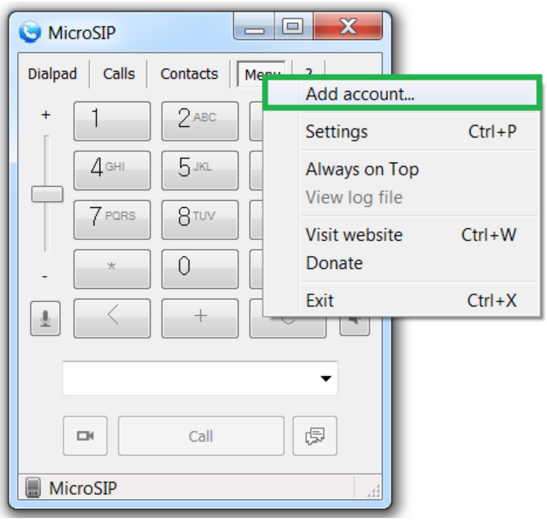


4. On the next screen click **Next >** to continue.
5. Click **I Agree** on the license agreement.
6. Select the components as shown below and click **Next >**.
7. Select the preferred installation location and click **Next >**.
8. Click **Install** and click **Finish** when the installation is over.

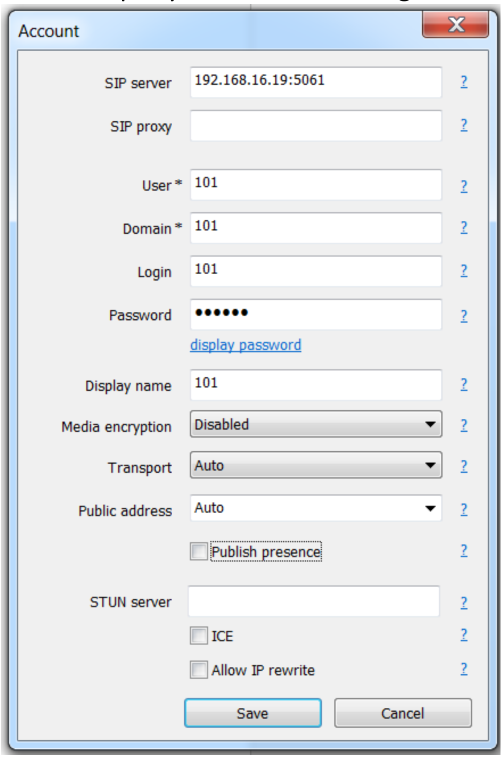


9. Run **MicroSIP**.

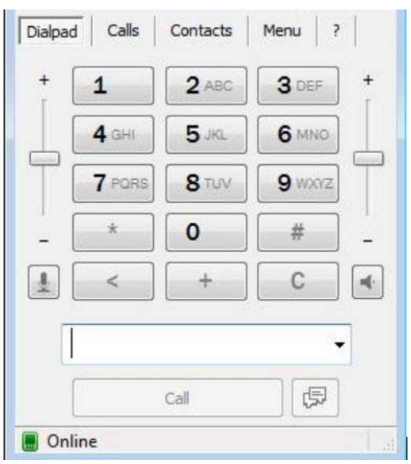
10. Click on **Menu** and then click on **Add account**.



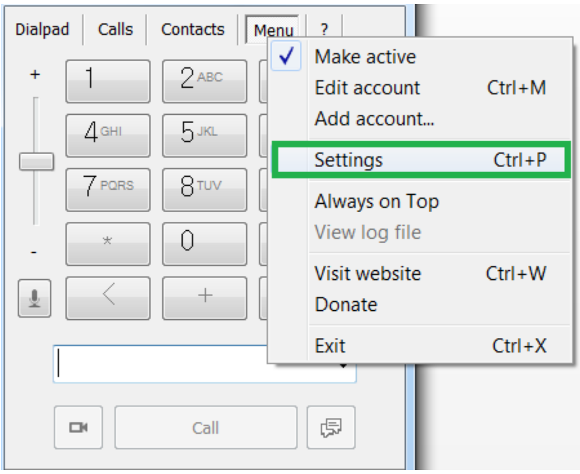
11. Input your account settings and configure your SIP server address. Click **Save** when you are done.



12. An icon on the bottom of the screen should now show that you are online.

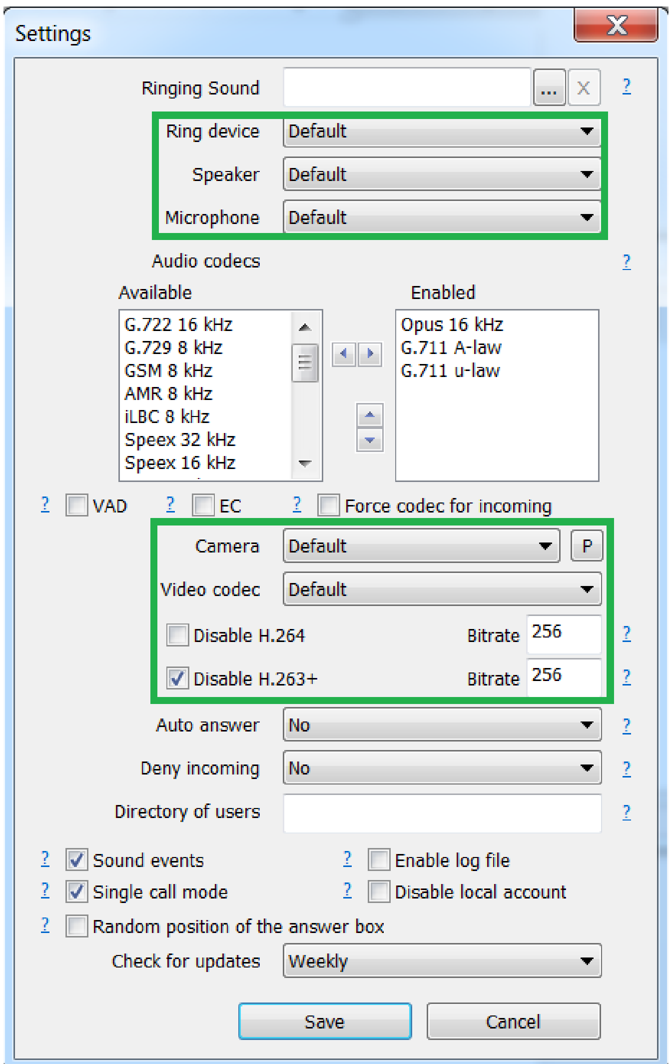


13. Click **Menu** > **Settings**.



14. Configure **Ring device**, **Speaker**, and **Microphone** based on your PC's environment.

15. Set **Camera**, **Video codec** to default and check **Disable H.263+**.



Please refer to the device manual and administrator's manual regarding device and BioStar 2 configuration.