# **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 <u>link</u> 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 <a href="http://www.freepbx.org/downloads">http://www.freepbx.org/downloads</a>.
- 2. Full Install No RAID is usually selected as the install method.



2. Click OK.

Jelcome to PBX for i386
Configure TCP/IP
[ <mark>*</mark> ] Enable IPv4 support (*) Dynamic IP configuration (DHCP) ( ) Manual configuration
[*] Enable IPv6 support (*) Automatic ( ) Automatic, DHCP only ( ) Manual configuration
OK
<tab>/<alt-tab> between elements   <space> selects   <f12> next screen</f12></space></alt-tab></tab>

3. Select the time zone of your city.

Welcome to PBX for i386			
In which I J Sys Americ Americ Americ Americ Americ	Time Zone Selection ch time zone are you lo stem clock uses UTC ca/Monterrey ca/Montevideo ca/Montserrat ca/Nassau ca/Nassau ca/New York Back	cated?	
<tab>/<alt-tab> between ele</alt-tab></tab>	ements   <space> sel</space>	ects I	<f12> next screen</f12>

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

Welcome to PBX for i386	
Root Password	
Pick a root password. You must type it twice to ensure you know it and do not make a typing mistake.	
Password:	
<pre><tab>/<alt-tab> between elements   <space> selects   <f12< pre=""></f12<></space></alt-tab></tab></pre>	> next screen

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.



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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.



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- 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

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- 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

Admin • Applications • Connectivity • Reports •

Add SIP Extension

- Add Extension

-	
User Extension®	102
Display Name	102
CID Num Alias	102
SIP Alias	102

+ Extension Options

+ Assigned DID/CID

- Device Options

This device uses	sip technology.	
secret	suprema102	
dtmfmode	RFC 2833	~
nat®	Yes 🗸	

#### 10. Select Settings > Advanced Settings.



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

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🗲 🕞 😇 http://192.168.11.153/admin, 🔎 🗸 🖒 😌 FreePBX	(Administration ×
Admin • Applications • Connectivity •	Reports • Settings • User Panel Apply Config
Device Settings	
Show all Device Setting on Add	True False
Require Strong Secrets	True False
Remove mailbox Setting when no Voicemail	True False
SIP canrenivite (directmedia)	no 🗸
SIP trustrpid	yes 🗸
SIP sendrpid	pai 🗸
SIP nat	yes 🗸
SIP encryption	no 🗸
SIP qualifyfreq	60
CID and IAV qualify	VAC

#### 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 <u>subnetting reference</u>.

Admin •	Applications *	Connectivity •	Reports *	Settings *	User Panel	Apply Config
Edit Settin	gs					
NAT Sottings						

na i	Yes no	never route
IP Configuration	Public IP	Static IP Dynamic IP
External IP	192.168.11.1	53
Local Networks	192.168.11.0	/ 255.255.255.0

#### 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.

Admin 💌	Applications *	Connectivity • Reports •	Settings 🔻
FreePBX S	Announcements Bulk DIDs Bulk Extensions	IS	
	Call Flow Control	otices	Sy
There are 2	Call Recording	to security threats	
There are 9	Conferences	for online upgrades	Load Av
d Default Aste	DISA	word Used	CPU
Forced MOI	Directory	to true	5
Collecting A	Extensions	Stats 🚳	App Mer
show all	Follow Me		Swap
	IVR		
	Languages	tistics	

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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.

Doules Ontions		
Device Options		
This device uses sip t	echnology.	
secret	suprema101	
dtmfmode	RFC 2833	~
canreinvite	No 🗸	
context	from-internal	
host	dynamic	
trustrpid	Yes 🗸	
sendrpid	Send P-Asserted-Identit	y header 🗸
type	friend 🗸	
nat	Yes 🗸	
port	5060	
qualify	yes	
qualifyfreq	60	
transport	TCP Only 🗸	
avpf	No 🗸	

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

dynamic
Yes 🗸
Send P-Asserted-Identity header $\checkmark$
friend 🗸
Yes 🗸
5060
yes
60
TCP Only
No 🗸
No 🗸

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

	*** http://19	2.168.11.153/admin,	Q - C 😌 Fre	eePBX Administration	×		
1*	Admin 👻	Applications *	Connectivity	• Reports •	Settings *	User Pane	Apply Config

Extension: 102

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## 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.

[!] Software selection
At the moment, only the core of the system is installed. To tune the system to your needs, you can choose to install one or more of the following predefined collections of software.
Choose software to install:
<pre>[*] OpenSSH server [ ] DNS server [ ] DNS server [*] LAMP server [ ] Mail server [ ] PostgreSQL database [ ] Print server [ ] Samba file server [ ] Samba file server [ ] Tomcat Java server [ ] Virtual Machine host [ ] Manual package selection </pre>

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.

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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 libsrtp0dev\

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 # 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.

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# 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
# ./configureenable-shareddisable-sounddisable-resampledisable-videodisable-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

2016 Suprema, Inc. All right reserved. This document should only be used for guidance. Contact us for further information at support@supremainc.com 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.

Add-ons (See README-addons.tx	kt) extended 1
Applications	XXX chan_mobile
Bridging Modules	[ ] chan_ooh323
Call Detail Recording	[*] format mp3
Channel Event Logging	[] res_config_mysql
Channel Drivers	deprecated
Codec Translators	[ ] app_mysql
Format Interpreters	[ ] app_saycountpl
Dialplan Functions	[] cdr_mysql
PBX Modules	1
IP3 format [Any rate but 8000]	nz mono is optimal]
Depends on: N/A	
Can use: N/A	Save & Exit Exit
onflicts with: N/A	
Support Level: extended	

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.

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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 - *pyourpassword* 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_ampinstalldbusername=asteriskuserpassword=\${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.

# In -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.

(←) ♥ http://192.168.11.153/admin/ 𝒫 マ ♥ ▼ FreePBX Administration ×	<b>↑</b> ★ \$
FreePBX Support User Panel	
Welcome to FreePBX Administration!	
Initial setup	
Please provide the core credentials that will be used to administer your system	
Username	
Password password	
Confirm Password password	
Admin Email address email address	
Confirm Email address confirm email	
Set up my Account	
FreePBX is a registered trademark of	
iet needont nig	

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

← ↔ http://192.168.11.153/admin: P • C ♥ FreeP8X Administration ×	n * 0
FreePEX Administration Get Support	Cperator Panel
FreePBX For a registered leadernest of FreePBX For a registered leadernest of FreePBX Sector For a registered leadernest of Copyright 2007-2016	Schmooze

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		_ • ×
(→ ** http://192.168.11.168/admin/config.php#	クー C SreePBX Administration ×	h ★ ₽
FreePBX Support ISymphonyV3 Panel UCP		
FreePBX Administration	Vertical and the second sec	Get Support
FreePI It freedom	AX Professional Anti-Anti-Anti-Anti-Anti-Anti-Anti-Anti-	NGOMA

### 4. Select Application > Extensions.

P) V http://192.168.11.168/admin/confi 	g php# D+C C Fre	ePBX Administration ×	n *
Admin • Applications • Connectiv	ity • Reports • Settings •	JCP	Logout: supre
Announcements Call Flow Control	A	Security Warning	×
Call Recording		Details	
Conferences		What Does this Mean?)	
Extensions	m Overview	2	
Syste	III Overview	D D	
Welcom	e to FreePBX		
FreePBX 12 (You can change this	0.76.2 'VoIP Server' mame in Advanced Settings)		
Summary	SysInfo updated 1 secon	ds ago	
Asterisk	A Security Issue	A	
MySQL .	a security issue	•	
Web Server	.htaccess files are disable on th	5	
	This is a critical issue and shoul	d be	
	resolved urgently		
.htaccess files are disable on this we	ebserver. Please enable them	•	
You have 1 tampered files		•	
No Conference Room App		0	
Failed to send security notification e	mail	0	
Default Asterisk Manager Password	Used	0	
Collecting Anonymous Browser Stat	S	00	
S	how New		
Free	PBX Feed	0	
Sangoma Launches New IP Phones Desi	gned for FreePBX and PBXact		
<ul> <li>Happy New Year, FreePBX 13 out of RC</li> <li>Yealink Endpoints now Certified for FreeP</li> </ul>	'ex		
<ul> <li>FreePBX holiday cyber weekend is official</li> <li>Format Cyber Mandau, Wa Cyber and Sofficial</li> </ul>	lly here!		
<ul> <li>Porger Cyber Wonday, it's Going to be Ho</li> <li>Commercial Modules, Support Provided, I</li> </ul>	Upcoming Changes		
Inside the	Asterisk Feed	0	
Asterisk stock prompts		~	
· SMB, SME, and Large Enterprise. Why th	e Size of Your Business Matters		
<ul> <li>VOIP, UC and Education: The Perfect Fit</li> <li>6 Rupiness Reports of SIR Trunking</li> </ul>			

8. Select Generic Generic SIP Device and click Submit.

	×
C C V http://192.168.11.168/admin/config.php?display=extens D C V FreePBX Administration ×	<u>+</u> ★ ₽
Admin v Applications v Connectivity v Reports v Settings v UCP	Logout: suprema
A Security Warning	×
Details (What Does this Mean?)	
(mine boes and mean.)	
Add an Extension	Add Extension
Add an Extension	
Please select your Device below then click Submit	
- Device	
Device Generic CHAN SIP Device	
Submit	
V 🤗 V T TT FreePBX is a registered trademark of	
tet freedom ring	

- 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.

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http://192.168.11.168/admin/	nofin nhn?disnlav-ovtens Q = C. St Star DBY Administration	
Admin - Applications - Conn	ectivity  Reports  Settings  UCP	Logout: suprem
Administrators Attentisk Mokules Backup & Reatore Blacklat CatlerD Lookup Sources Certificate Management Contact Manager		Add Extension
Custom Destinations Custom Extensions Feature Codes FreePBX Support Module Admin System Recordings User Management Symphory/V3	101 101 101 101	
+ Assigned DID/CID		
This device uses CHAN_SIP te	chnology listening on 0.0.0.0:5061	
Secret @	suprema101	
DTMF Signaling	(RFC 2833 V	
NAT Mode	Yes - (force_rport,comedia)	

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)

* Admin = Applications = (	nin/config.php?type=&display P ▼ C 😒 FreePBX Administration ×	n t
Add SIP Extension	omecony • reports • Security • Oce Apply Comp	Add Extension 101 <101>
- Add Extension		
User Extension <sup>2</sup>	102	
Display Name 🛛	102	
CID Num Alias	102	
SIP Alias	102	
- Device Options		
- Device Options This device uses CHAN_SI	P technology listening on 0.0.0.35061	
- Device Options This device uses CHAN_SI Secret	P technology listening on 0.0.0.0:5061 suprema102	
- Device Options This device uses CHAN_SI Secret <sup>©</sup> DTMF Signaling <sup>©</sup>	P technology listening on 0.0.0.35061 suprema102 RFC 2833 V	

### 12. Select Settings > Advanced Settings.

The problem of the problem	x Legout: supro- ng x vaking any changes. Readonly settings are usually more volatile, they ve the setting by checking the green check box that appears. You can
Admin  Applications Connectivity Connectivit	Legout: supr ng aking any changes. Readonly settings are usually more volatile, they ve the setting by checking the green check box that appears. You can
Advanced Settings Attainat Coffie Settings Attainat Coffie Settings Attainat Coffie Settings Attainat Coffie Settings Attainat Name Attainat Name Attainat SIP Settings Marc on Hold Marc on	ng X
Asteria: Logitie Seitings Asteria: Manager Users Asteria: Manager Users Asteria: Manager Users Asteria: Manager Users Asteria: SIP Seitings Masic on Hold MPORTANT: Use extreme caution when making change 'Vokemail Admin Some of these settings can render your system inoperable. You are urged to backup before m an be changed by changing 'Override Readonly Settings' to true. Once changed you must sa estore the default setting by clicking on the icon to the right of the values if not set at default.	haking any changes. Readonly settings are usually more volatile, they we the setting by checking the green check box that appears. You can
Attenik Manager Uses Attenik REST Interface Uses Asterisk REST Interface Uses Asterisk SIP Settings Misc on Hold MPORTANT: Use extreme caution when making changi Vokemel Admin tome of these settings can render your system inoperable. You are urged to backup before m an be changed by changing 'Override Readonly Settings' to true. Once changed you must sa store the default setting by clicking on the icon to the right of the values if not set at default.	) laking any changes. Readonly settings are usually more volatile, they we the setting by checking the green check box that appears. You can
FreePBX Advanced Settings     Marke on Hold     MPORTANT: Use extreme caution when making change     Volcemel Admin     Volcemel Admin     Some of these settings can render your system inoperable. You are urged to backup before m     an be changed by changing 'Override Readonly Settings' to true. Once changed you must sa     selore the default setting by clicking on the icon to the right of the values if not set at default.	taking any changes. Readonly settings are usually more volatile, they we the setting by checking the green check box that appears. You can
House on Heid     More an Heid     MPORTANT: Use extreme caution when making chang(     Vickemail Admin     Kome of these settings can render your system inoperable. You are urged to backup before m     an be changed by changing 'Override Readonly Settings' to true. Once changed you must sa     estore the default setting by clicking on the icon to the right of the values if not set at default.	taking any changes. Readonly settings are usually more volatile, they we the setting by checking the green check box that appears. You can
WFOR TANT: Use extreme caution when making change Volcemeil Admin forme of these settings can render your system inoperable. You are urged to backup before m an be changed by changing 'Override Readonly Settings' to true. Once changed you must sa estore the default setting by clicking on the icon to the right of the values if not set at default.	aking any changes. Readonly settings are usually more volatile, they we the setting by checking the green check box that appears. You can
some of these settings can render your system inoperable. You are urged to backup before m and be changed by changing 'Override Readonly Settings' to true. Once changed you must sa estore the default setting by clicking on the icon to the right of the values if not set at default.	lawing any changes. Neadoniy settings are usually more volatile, they we the setting by checking the green check box that appears. You can
Advanced Settings Details	
Display Friendly Name	
Display Readonly Settings <sup>2</sup> True False	
Override Readonly Settings	
Asterisk Builtin mini-HTTP server	
Enable Static Content  True False	
Enable the mini-HTTP Server  True False	
HTTP Bind Address  0.0.0.0	
HTTP Bind Port <sup>©</sup> 8088	
HTTP Prefix <sup>©</sup>	
Asterisk Manager	
Asterisk Manager Password  amp111	
Asterisk Manager User <sup>2</sup> admin	

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

http://192.168.11.168/admin/config	.php?display=advan 🔎 🕆 😋 🌄 FreePBX Administration 🛛 🗙		
Admin  Applications  Connectivity	ty • Reports • Settings • UCP Apply Config		Logout: sup
System Identity	VolP Server		
Use Packaged Javascript Library 👂	True False		
Device Settings			
Show all Device Setting on Add 9	True False		
Require Strong Secrets	True False		
Remove mailbox Setting when no Voicemail <sup>©</sup>	True Faise		
SIP canrenivite (directmedia)	no 🗸		
SIP trustrpid	yes 🗸		
SIP sendrpid	no V		
SIP nat <sup>2</sup>	yes 🗸	<b>&gt; 0</b>	
SIP encryption	no V		
SIP qualifyfreq <sup>9</sup>	60		
SIP and IAX qualify	yes		
SIP and IAX allow			
SIP and IAX disallow			
SIP and DAHDi callgroup			
SIP and DAHDi pickupgroup <sup>9</sup>			
Dialplan and Operational			
Block CNAM on External Trunks	True False		
Call Forward Ringtimer Default	0 🗸		

### 14. Select Settings > Asterisk SIP Settings.

Applications      Connectivity      Reports      Set     Applications      Connectivity      Reports      Set     Ave	Yes     NO
Applications      Connectivity      Reports      Set     Adv	ttings v UCP Apply Centra Logout: si anced Settings / Warning etsk Logout: si widd REFC Inductors Liss at on Hold cernal Admin Central SIP Settings chan SIP chan SIP chan SIP Chan SIP Chan SIP Chan SIP Chan SIP
Adv Adv Attings s currently using chan_sip, chan_pisip for SIP T_voic change this on the Advanced Settings Page may have moved! Please use the navigation on Settings nonymous Inbound SIP Calls •	anced Settings
Atta Atta	eriek EFST Interface Lisen : this Mean?) eriek SP Settrops sic on Hold Ceneral SIP Settings Chan SIP Chan PJSIP The right Yes No
Attings Aurently using chan_sip, chan_pisip for SIP T voic change this on the Advanced Settings Page may have moved! Please use the navigation on Settings tings	errak SIP Settings sak on Hold commail Admin the right Yts No
Aus scurrently using chan_sip, chan_pisip for SIP T_vec change this on the Advanced Settings Page may have moved! Please use the navigation on Settings ionymous Inbound SIP Calls <sup>®</sup>	ac on fold carnal Admin the right -> Yes No
smay have moved! Please use the navigation on Settings nonymous Inbound SIP Calls <sup>©</sup>	the right >
Settings nonymous Inbound SIP Calls • •	Yes No
Settings nonymous Inbound SIP Calls	Yes No
nonymous Inbound SIP Calls <sup>©</sup>	Yes No
nonymous Inbound SIP Calls <sup>©</sup>	Yes No
ttings	
a settings apply to both chan, sin and chan, nisin	
setungs apply to both chan_sip and chan_pjsip.	
Address 🛛	Detect External IP
etworks 🖗	1
	Add Local Network Field
aings	
rt Ranges <sup>©</sup> Str	tart: 10000 End: 20000
ecksums <sup>©</sup>	Yes No
гр <b>0</b>	Yes No
anvar Addraga	

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#### 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 <u>subnetting reference</u>.

⇔ http://192.168.11.9/admin/config.php?di	splay=sipsetting P = C V FreePBX Administration ×	<del>1</del> *
Admin  Applications  Connectivity  Connectivity	Reports v Settings v UCP	Logout: suprem
External Address	192.168.11.9 Detect External IP	
Local Networks	132.100.11.0 / 235.255.0	
	Add Local Network Field	
DTD Sottings		
KTF Settings		
RTP Port Ranges	Start: 10000 End: 20000	
RTP Checksums	Yes No	
Strict RTP 9	Yes No	
STUN Server Address 2		
STON Server Address -		
TURN Server Address		
TURN Server Username <sup>2</sup>		
TURN Server Password		
Audio Codecs		
Codecs <sup>2</sup>	‡ ⊠ <sub>staw</sub>	
	1 Zalaw	
	t dan	
	1 ⊠19728 ★ ☑-723	
	t ⊠adpom	
	\$ Zhin	
	\$ Z <sub>9729</sub>	

13. Select all audio Codecs.

← () ♥ http://192.168.11.168/admin/config.php?display=si	osett 🔎 👻 FreePBX Administration 🛛 🗙	
Admin v Applications v Connectivity v Report	Settings v UCP Apply Confg     Settings v UCP Apply Confg     Settings v UCP     Setings v UCP     Settings v UCP     Setting     Settings v UCP     Setting	Logout suprems
	2 Mantes 2 Martes 2 Mart	
Submit	BX ring <sup>th</sup> PresBX: a regulated trademak of scheme, Technologistic, Copyright 2007 2019	SANGOMA

#### 14. Press the **Submit** button.

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

~		
http://192.168.11.16	8/admin/config.php?display=sipsett 🔎 🔹 🏷 FreePBX Administration 🛛 🗙	n 🖈
Admin - Applications	Connectivity  Reports  Settings  UCP Apply Config	Logout: suprem
	A Security Warning	×
	Details (What Done this Mean2)	
	(What Does this mean !)	
SIP Settings		General SIP Setungs
Asterisk is currently using You can change this on th	chan_sip, chan_pjsip for SIP Traffic. e Advanced Settings Page	Chan SIP (A)
Edit Settings		
Late oottingo		
NAT Settings		
NAT	yes no never route	
IP Configuration	Public IP Static IP Dynamic IP	
0	192 158 11 158	
Override External IP		
Audio Codecs		
Non Standard a706 9	Yes No.	
Non-Standard g/26		
T38 Pass-Through <sup>2</sup>	Yes No	
Video Codoco		
Video Support	Enabled Disabled	
MEDIA & RTP Settings		
Poinvite Pohavior @	ves no nonat update	
Remvile Benavior		
RTP Timers 🕫	30 (rtptimeout) 300 (rtpholdtimeout) 0 (rtpkeepalive)	

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http://192.168.11.1	68/admin/config.php?display=sipsett 🔎 - 💐 🏹 FreePBX Administration 🛛 🗙	
Admin - Applications	Connectivity      Reports      Settings      UCP Apply Config	Logout: suprem
T38 Pass-Through <sup>©</sup>	Yes No	
Video Codecs		
Video Support	Enabled Disabled	
	g 🖾 cont 9 Marcal	
	\$ Øvp3	
	\$ Zh2030	
	8 ⊠10003	
Max Bit Rate 🤨	384 kb/s	
MEDIA & RTP Settings		
Reinvite Behavior	yes no nonat update	
RTP Timers	30 (rtptimeout) 300 (rtpholdtimeout) 0 (rtpkeepalive)	
Notification & MWI		
MWI Polling Freq 🛛	10	
Notify Ringing	Yes No	
Notify Hold	Yes No	
Registration Settings		
Registrations <sup>2</sup>	20 (registerimeout) 0 (registerattempts)	
-		

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

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

Admin =	Vadmin/contrig.php?display=sipsett D + C  FreePBX Administration ×	
De sister tions @	20 (ranistartimenut) 0 (ranistartitamte)	Logour suprema
Registrations	20 (registerialmetout) 0 (registerialmetopis)	
Registration Times	ou (minexpiry) Souu (maxexpiry) 120 (derautexpiry)	
Jitter Buffer Settings		
Jitter Buffer <sup>2</sup>	Enabled Disabled	
Advanced General Settin	JS	
Language <sup>9</sup>		
Default Context <sup>2</sup>		
Bind Address		
Bind Port <sup>2</sup>	5061	
Allow SIP Guests	Yes No	
SRV Lookup	Enabled Dinabled	
Call Events	Yes No	
Other SIP Settings	=	
	Add Field	
Submit Changes		
\$	FreePBX Let freedom ring <sup>TM</sup>	~

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

A Disc.	3/admin/config.php?display=sipsett 🍳 🗸 FreePBX Administration 🗙	n × ¤
Admin  Applications	Connectivity  Reports  Settings  U P Apply Config	Logout: suprema
	A Security Warning	^
	Details (What Does this Mean?)	
	(mat bots the mount)	
SIP Settings		General SIP Settings
Asterisk is currently using on You can change this on the	chan_sip, chan_pjsip for SIP Traffic. Advanced Settings Page	Chan SIP (A)
Edit Cottingo		Chan PJSIP
Edit Settings		
NAT Settings		
NAT <sup>20</sup>	yes no never route	
100 c c 0	Public ID Static ID Dunamic ID	
IP Configuration	Public IP State IP Dynamic IP	
Override External IP <sup>10</sup>	192.168.11.168	
Audio Codecs		
Non-Standard g726	Yes No	
T38 Pass-Through	Yes No	
Video Codecs		
Video Support	Enabled Disabled	
	A Marsu	
	Vimená	
	1 Dups	
	1 2h283p	
		~

# 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.

MicroSIP-3.12.1.exe 8/18/2016 10:46 A... Application 5,624 KB

#### 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.

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MicroSIP Setup			
	Choose Components Choose which features of MicroSIP you want to install.		
Check the components you want to install and uncheck the components you don't want to install. Click Next to continue.			
Select components to	o install: ✓ MicroSIP (required) ✓ Desktop Shortcut ✓ Run at System Startup Links association	Description Position your mouse over a component to see its description.	
Space required: 11.2	МВ		
Nullsoft Install System v2.46.5-Unicode			
	< Back	Next > Cancel	

#### 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.

Account		×
SIP server	192.168.16.19:5061	2
SIP proxy		2
User*	101	2
Domain*	101	2
Login	101	2
Password	•••••	2
	display password	
Display name	101	2
Media encryption	Disabled	- 2
Transport	Auto	- 2
Public address	Auto	- 2
	Publish presence	2
STUN server		2
	ICE	2
	Allow IP rewrite	2
	Save Cancel	
L		

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

+ 1	1	<b>2</b> ABC	3 DEF	+
	<b>4</b> GHI	5 JKL	6 MNO	L
T	7 PORS	8 TUV	9 wxyz	T
4 -	*	0	#	-
1	<	+	С	<b>4</b>
0	ni i		- 	
		Call	同	

13. Click Menu > Settings.

Dialpad Calls Co	ntacts M <u>enu</u>	?	
+ 1	2 <sup>ABC</sup> [	Make active Edit account Add account	Ctrl+M
		Settings	Ctrl+P
7 PQRS	8™ 0	Always on Top View log file	
	+	Visit website Donate	Ctrl+W
		Exit	Ctrl+X
	Call	-	

- 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+.

Settings	X
Ringing Sound	× 2
Ring device	Default 👻
Speaker	Default 👻
Microphone	Default 🔻
Audio codecs	2
Available	Enabled
G.722 16 kHz	Opus 16 kHz
G./29 8 KHZ GSM 8 kHz	G.711 µ-law
AMR 8 kHz	
iLBC 8 kHz	
Speex 32 KHZ Speex 16 kHz	<b>T</b>
2 VAD 2 EC	2 Force codec for incoming
Camera	Default
Video codec	▼
🔲 Disable H.	264 Bitrate 256 ?
📝 Disable H.	263+ Bitrate 256 ?
Auto answer	No • ?
Deny incoming	No • ?
Directory of users	2
2 Sound events	2 Enable log file
2 ✓ Single call mode	2 Disable local account
2 Random position of th	e answer box
Check for updates	Weekly
	Save Cancel

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

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