

# CUSTOM VoIP SYSTEM USING ASTERISK FRAMEWORK

## DEVELOPER DOCUMENTATION

### OVERVIEW

The project is being built as a substitute of the PBX system and IP PBX system we use at societies, hospitals, hotels or offices. We are trying to build a system that can provide the benefits of the earlier PBX systems in the user mobile phone. People connected in the same Wi-Fi network will be able to make voice and video calls, send messages, make emergency calls, send mails to the other users who are connected to the network and in a much more organised way than that of a IP PBX or PBX system. We would briefly discuss the benefits here:

- Much easier to install & configure than a proprietary phone system or an IP phone system.
- Easier to manage because of web/GUI based configuration interface.
- Eliminate phone wiring.
- Scalable.
- Better phone usability.
- Many features at one single application.

### REQUIREMENTS:

- A desktop, which is configured as per the average number of calls expected.
- A Wi-Fi setup.
- User need to install an app called Zoiper available in google play store or windows store.
- FreePBX distro.

### FUNCTIONALITY:

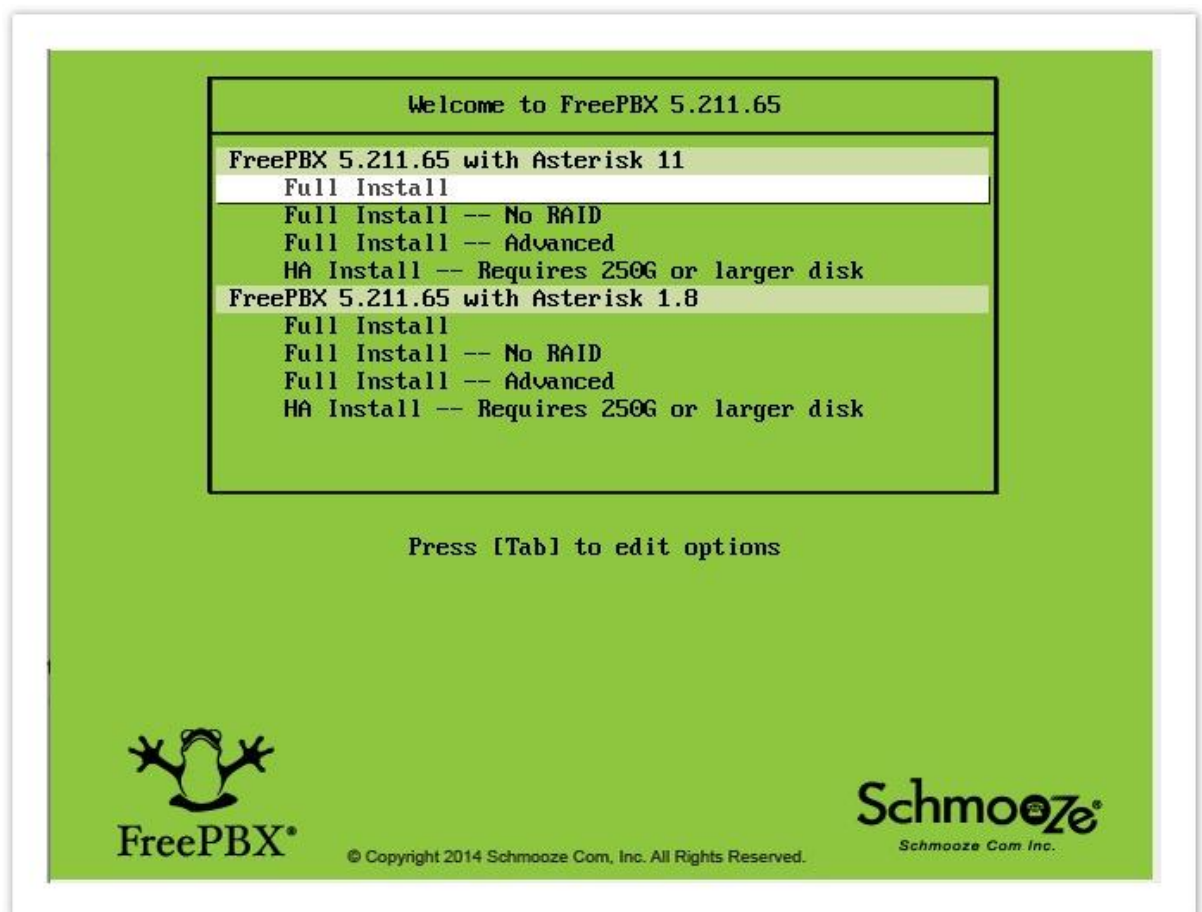
- Users can make voice and video calls within the network to other users.
- Users are able to send messages.
- Users can send voice messages.

## PLATFORM OF DEVELOPMENT:

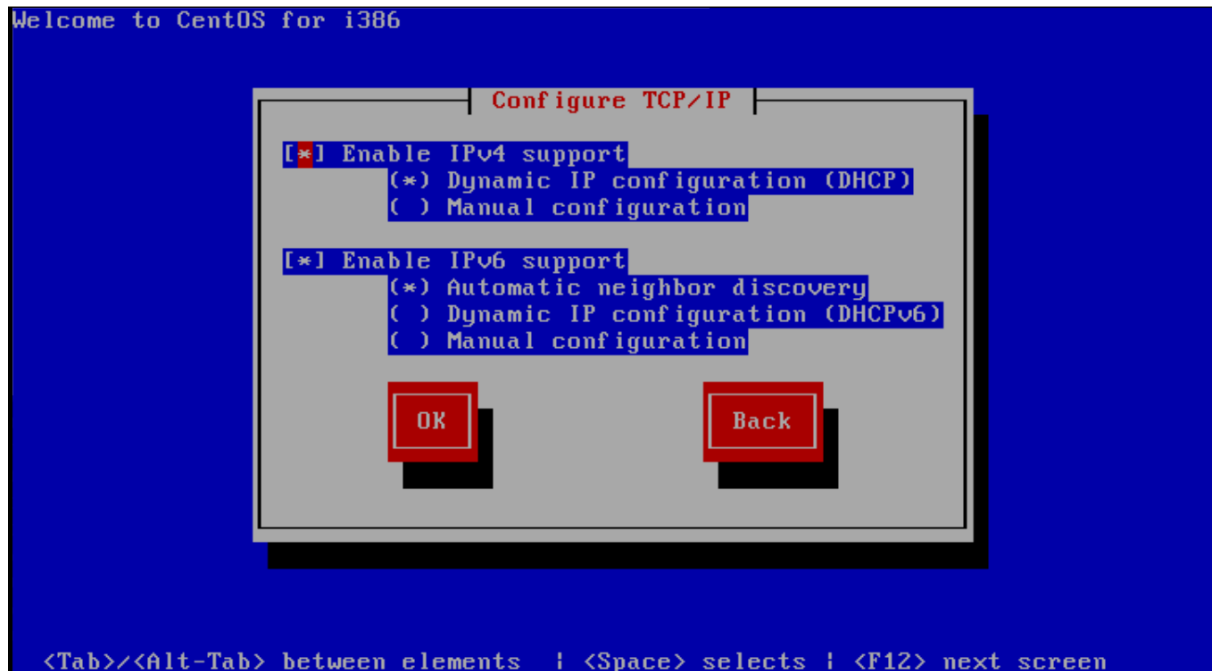
We are using Asterisk which is open source framework for building communication applications, but instead of using its direct command line version, we are using a derived product called FreePBX which is a web-based open source graphical user interface (GUI) that manages Asterisk, a voice over IP and telephone server. FreePBX is licensed under the GNU General Public License version 3. It is a component of the FreePBX Distro, which is an independently maintained Linux system derived from the source code of the CentOS distribution, having Asterisk pre-installed. FreePBX provides a nice user interface where one can configure the system quite easily and some feature which the developer wants and is not present can be implemented by them by using the core asterisk framework and by writing some scripts.

## FREEPBX SETUP STEPS:

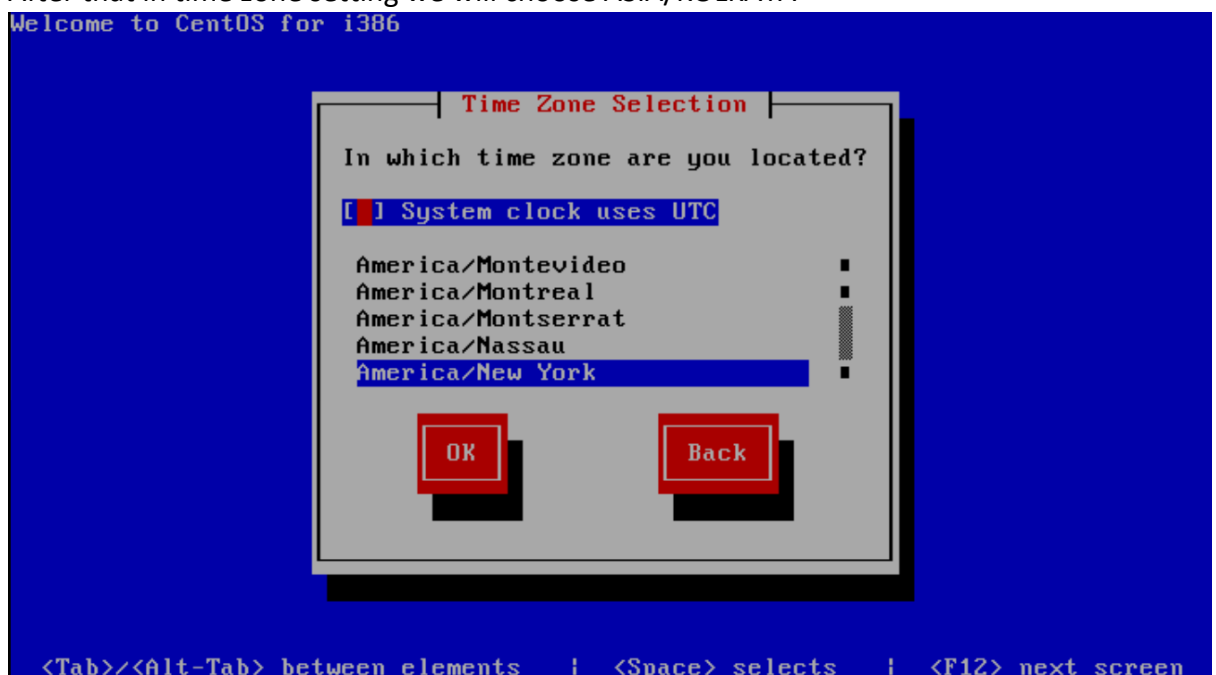
1. First we need the FreePBX distro, which is the easiest way to install FreePBX. It can be downloaded from <http://www.freepbx.org/downloads>.
2. The installer will begin with a prompt to select the Asterisk Version you wish to install. We will go with the generic "Full Install" option for Asterisk 11.



3. After the system boots, we will see the options to configure the network. Default option works fine for us.



4. After that in time zone setting we will choose ASIA/KOLKATA



5. Then, we have to set the root password for the distro being installed



6. Now after the installation completes, we can login to the shell using the root password we set during the installation process.

```
SHMZ release 6.5 (Final)
Kernel 2.6.32-431.el6.i686 on an i686

localhost login: _
```

- After we login, the IP address of the server is shown as below which should be noted down.

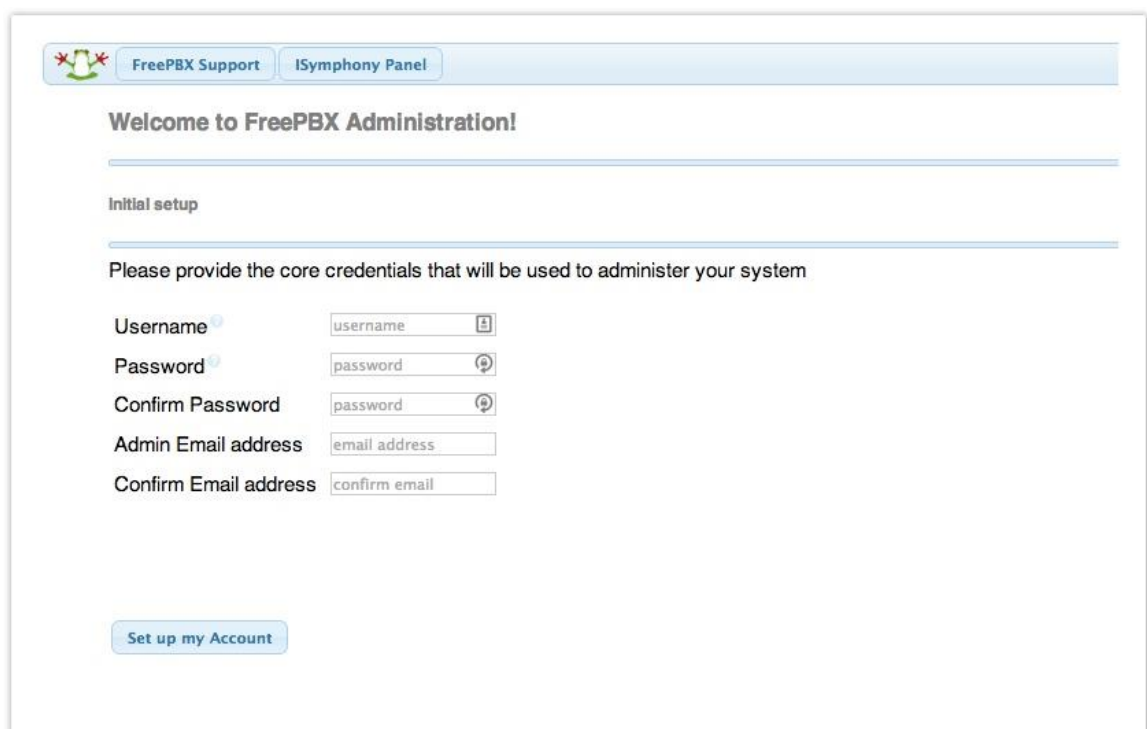
```
+-----+-----+-----+
| eth0   | 08:00:27:4B:EE:48 | 192.168.0.9   |
|        |                   | fe80::a00:27ff:fe4b:ee48 |
+-----+-----+-----+

Please note most tasks should be handled through the GUI.
You can access the GUI by typing one of the above IPs in to your web browser.
For support please visit:
  http://www.freepbx.org/support-and-professional-services

*****
* This machine is not activated. Activating your system ensures that *
* your machine is eligible for support and that it has the ability to *
* install Commercial Modules.                                         *
*                                                                       *
* If you already have a Deployment ID for this machine, simply run:   *
*                                                                       *
*   fwconsole sysadmin activate deploymentid                          *
*                                                                       *
* to assign that Deployment ID to this system. If this system is new, *
* please go to Activation (which is on the System Admin page in the   *
* Web UI) and create a new Deployment there.                           *
*****

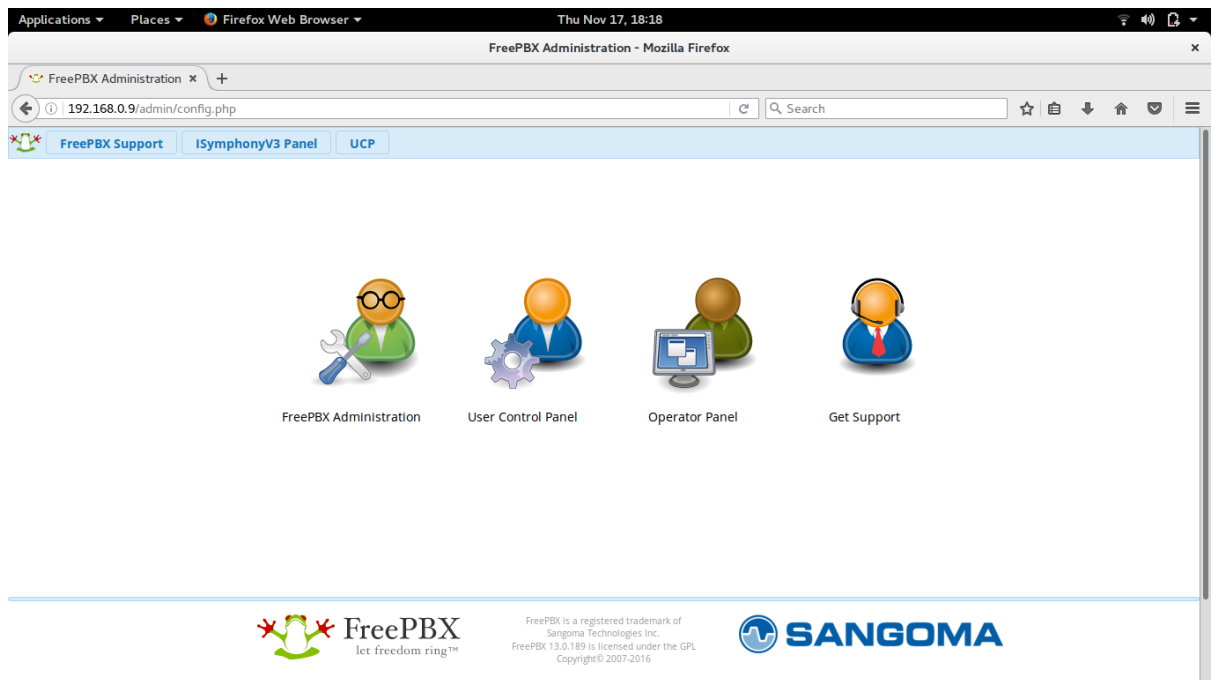
[root@localhost ~]# _
```

- To access FreePBX GUI, we will open the IP address from the browser of another computer connected in the same network. When we open the GUI first time, we have to setup an admin account for FreePBX.

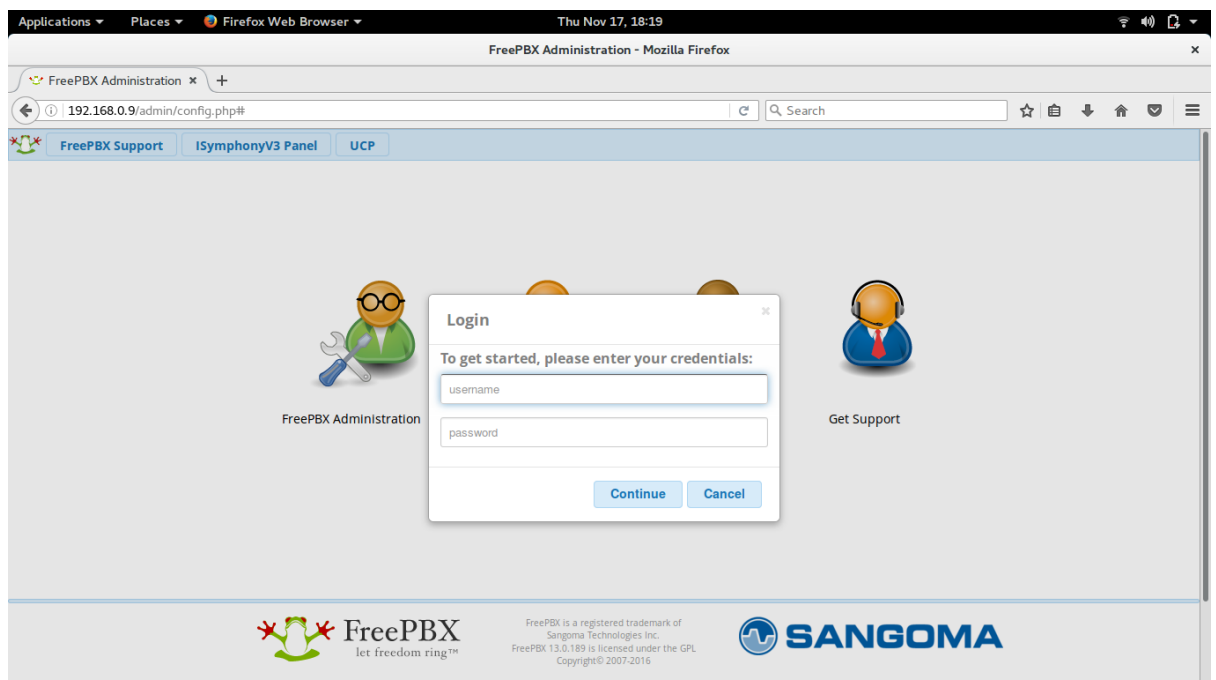


The image shows the 'Initial setup' page of the FreePBX Administration interface. At the top, there are two tabs: 'FreePBX Support' and 'ISymphony Panel'. Below the tabs is a heading 'Welcome to FreePBX Administration!'. Underneath, the section 'Initial setup' is displayed. A message states: 'Please provide the core credentials that will be used to administer your system'. The form contains five input fields: 'Username' (with a hint icon), 'Password' (with a hint icon), 'Confirm Password' (with a hint icon), 'Admin Email address', and 'Confirm Email address'. Each field has a placeholder text: 'username', 'password', 'password', 'email address', and 'confirm email' respectively. At the bottom of the form is a button labeled 'Set up my Account'.

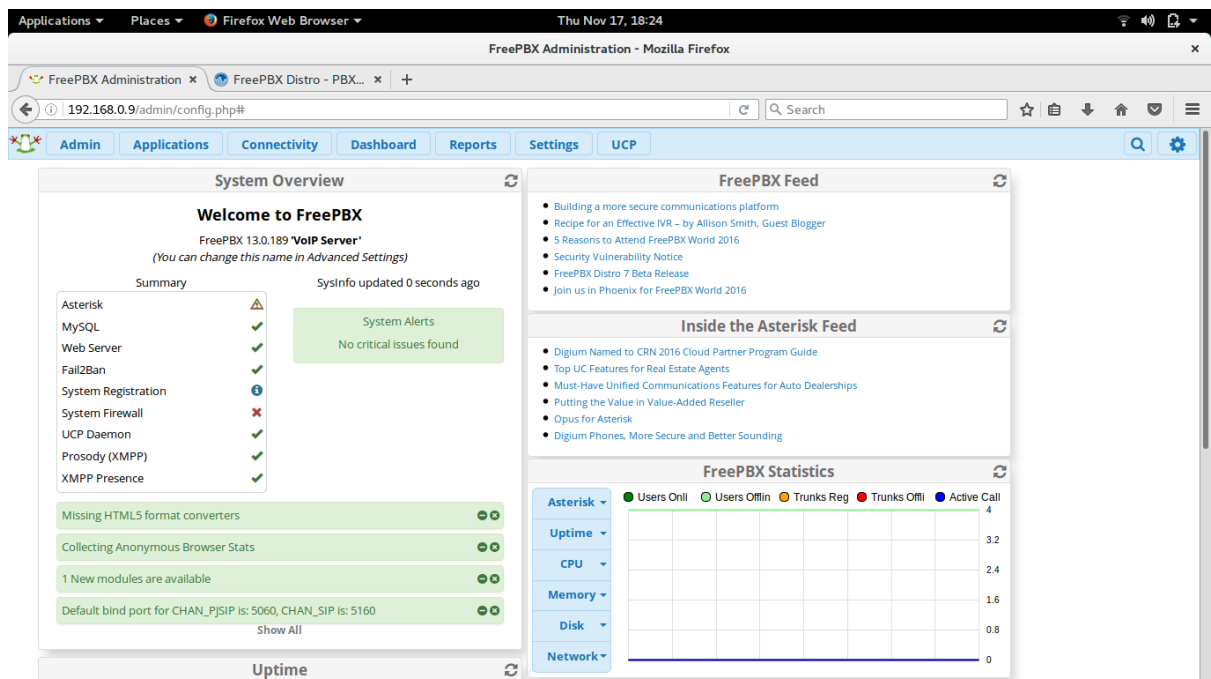
9. FreePBX GUI will offer us following four options



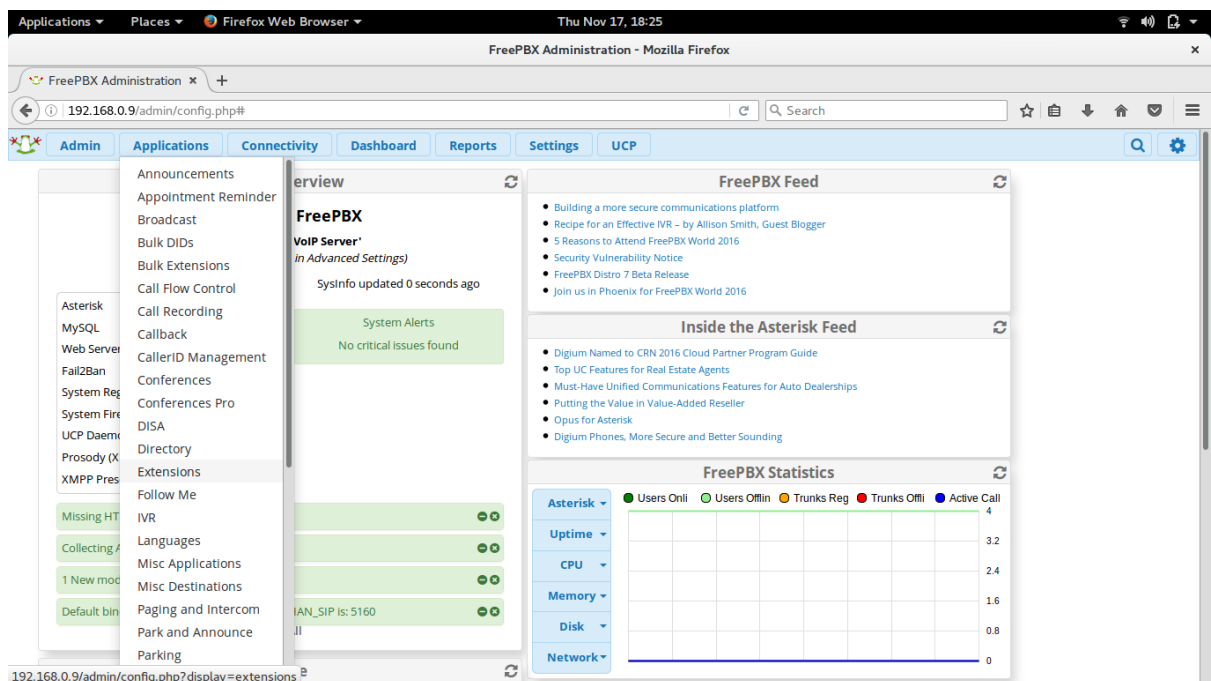
10. For Further configuration we need to login to FreePBX Administration



11. Following screen is the dashboard of FreePBX Administration where we can access all the options.

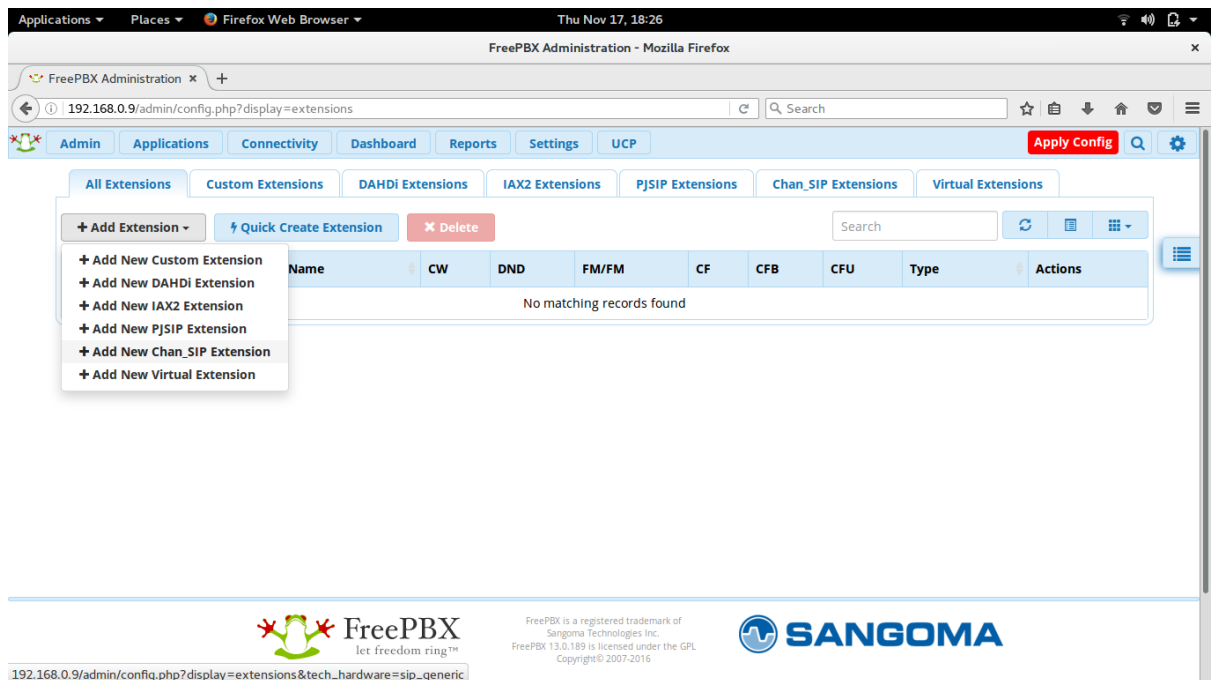


12. To connect phone with this FreePBX server we need to create an extension for each device. To create an extension, we need to navigate to Application>Extensions.





13. In the Extensions Page, we will choose **Add New Chan\_SIP Extension** from Add Extension menu



14. To, complete the Extension creation process we have to fill up the following form as follows.

#### **General Tab**

**User Extension:** It is the number on which any user of the system will dial to contact this user.

**Display Name:** Display Name of the user

**Secret:** Password for the Extension

**Password for New User:** FreePBX also creates a user account associated with the extension. We need to set the password for that new user too. We preferably use

the same password as secret for the ease of configuration.

Admin Applications Connectivity Dashboard Reports Settings UCP Apply Config

General Voicemail Find Me/Follow Me Advanced Zulu Other

— Add Extension

This device uses **CHAN\_SIP** technology listening on Port 5160 (UDP - this is a **NON STANDARD** port)

User Extension 1234

Display Name Vikas Kumar

Outbound CID

Secret mypassword  
Really Weak

— Language

Language Code Default

— User Manager Settings

Link to a Default User Create New User

Username Use Custom Username

Password For New User mypassword

Groups All Users

Submit Reset

## Voicemail Tab

**VoiceMail Password:** If we want to use voicemail service, then we need to set up a PIN to access voice mailbox.

Admin Applications Connectivity Dashboard Reports Settings UCP Apply Config

General Voicemail Find Me/Follow Me Advanced Zulu Other

— Voicemail

Enabled	Yes No
Voicemail Password	••••
Require From Same Extension	Yes No
Disable (*) in Voicemail Menu	Yes No
Email Address	
Pager Email Address	
Email Attachment	Yes No
Play CID	Yes No
Play Envelope	Yes No
Delete Voicemail	Yes No
VM Options	
VM Context	default

— VmX Locator™

Enabled	Yes No
Use When	Unavailable Busy

Submit Reset

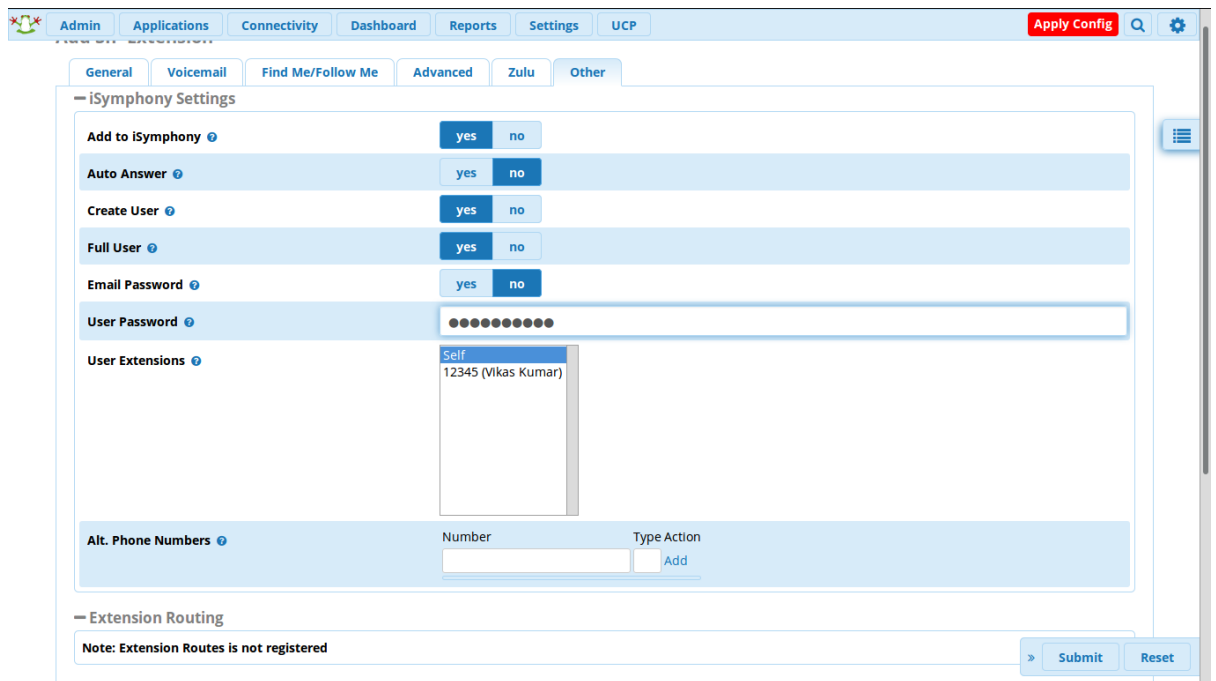
### Advance Tab

**Allow:** We set it to blank so all devices in network can connect to it

**Deny:** It is also set to blank for the same reason

### Other Tab

**User Password:** Set a user password for iSymphony account. It is a call manager service associated with FreePBX distro,



The screenshot shows the 'iSymphony Settings' page within a web application. The top navigation bar includes tabs for Admin, Applications, Connectivity, Dashboard, Reports, Settings, and UCP. The 'Settings' tab is active, and the 'Other' sub-tab is selected. The main content area is titled 'iSymphony Settings' and contains several configuration options:

- Add to iSymphony:** A toggle switch set to 'yes'.
- Auto Answer:** A toggle switch set to 'yes'.
- Create User:** A toggle switch set to 'yes'.
- Full User:** A toggle switch set to 'yes'.
- Email Password:** A toggle switch set to 'yes'.
- User Password:** A text input field with a masked password (represented by dots).
- User Extensions:** A list box showing 'Self' and '12345 (Vikas Kumar)'.
- Alt. Phone Numbers:** A section with a 'Number' input field and a 'Type Action' dropdown menu set to 'Add'.

Below the settings, there is a section for 'Extension Routing' with a note: 'Note: Extension Routes is not registered'. At the bottom right, there are 'Submit' and 'Reset' buttons.

All other settings are optional for current goal.

We can click submit to create the extension. Also we can add another extension in

the same way but with different Extension number.

Admin Applications Connectivity Dashboard Reports Settings UCP Apply Config

All Extensions Custom Extensions DAHDI Extensions IAX2 Extensions PJSIP Extensions Chan\_SIP Extensions Virtual Extensions

+ Add Extension Quick Create Extension Delete Search

	Extension	Name	CW	DND	FM/FM	CF	CFB	CFU	Type	Actions
<input type="checkbox"/>	1234	Vikas Kumar	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	slp	<a href="#">Edit</a> <a href="#">Delete</a>
<input type="checkbox"/>	1235	Falzaan	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	slp	<a href="#">Edit</a> <a href="#">Delete</a>

Showing 1 to 2 of 2 rows

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- There could be some network error, if we move the server from on network to another. So to solve the problem, We need to check if the **local network** is configured properly in the **SIP Settings**.

Admin Applications Connectivity Dashboard Reports Settings UCP Apply Config

Allow Anonymous Inbound SIP Calls ☒ Yes ☐ No

Default TLS Port Assignment Chan SIP PJSip

NAT Settings

These settings apply to both chan\_sip and chan\_pjsip.

External Address 223.223.0.41 [Detect Network Settings](#)

Local Networks

192.168.43.0 / 24

192.168.0.0 / 24

192.168.2.0 / 24

[Add Local Network Field](#)

RTP Settings

RTP Port Ranges Start: 10000 End: 20000

RTP Checksums ☒ Yes ☐ No

Strict RTP ☒ Yes ☐ No

STUN Server Address

TURN Server Address

[Submit](#) [Reset](#)

16. Till now the server will work fine for voice calling, but text messages couldn't be sent. So, to solve the problem we need to create a custom dialplan. To do this, first we need to add following two lines inside "Other SIP Setting".

```
accept_outofcall_message=yes
```

```
outofcall_message_context=astsms
```

The screenshot shows the FreePBX Admin interface with the 'Settings' tab selected. The 'Other SIP Settings' section is expanded, showing two configuration fields: 'accept\_outofcall\_message' set to 'yes' and 'outofcall\_message\_context' set to 'astsms'. The interface includes a top navigation bar with tabs like Admin, Applications, Connectivity, Dashboard, Reports, Settings, and UCP. The bottom of the page features the FreePBX logo and the Sangoma logo.

17. In file "extensions\_custom.conf", we will create a dialplan inside context **astsms**. This file can be accessed through **Config Edit** option in **Admin** menu of the dashboard.

The screenshot shows the FreePBX Admin interface with the 'Admin' menu open. The 'Config Edit' option is highlighted. The main content area displays the 'System Overview' page, which includes a 'Welcome to FreePBX' message, a 'Critical Errors found' notification, and a 'FreePBX Feed' section. The 'FreePBX Statistics' section shows a graph of system performance metrics like Users On/Off, Trunks Reg/Off, and Active Calls. The 'Uptime' section shows the system last rebooted 38 minutes, 4 seconds ago.

18. To create the dialplan, we will append following text to file  
“extensions\_custom.conf”

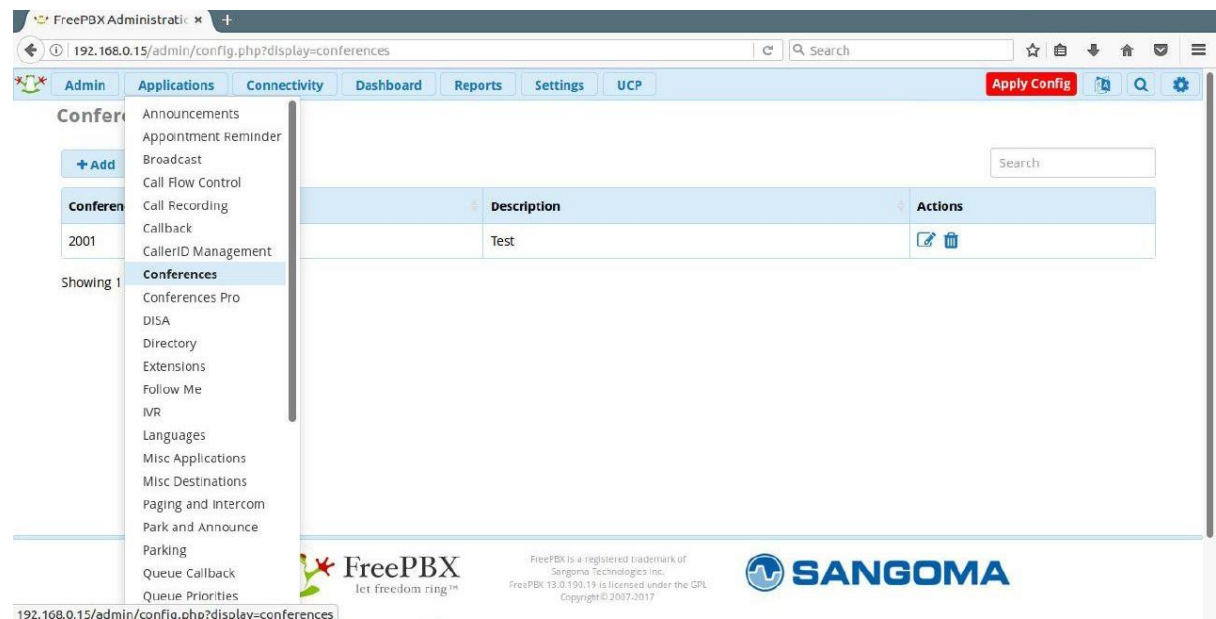
```
[astsms]
exten => _.,1,NoOp(SMS receiving dialplan invoked)
exten => _.,n,NoOp(To ${MESSAGE(to)})
exten => _.,n,NoOp(From ${MESSAGE(from)})
exten => _.,n,NoOp(Body ${MESSAGE(body)})
exten => _.,n,Set(ACTUALTO=${CUT(MESSAGE(to),@,1)})
exten => _.,n,MessageSend(${ACTUALTO},${MESSAGE(from)})
exten => _.,n,NoOp(Send status is ${MESSAGE_SEND_STATUS})
exten => _.,n,GotoIf("${MESSAGE_SEND_STATUS}" != "SUCCESS")?sendfailedmsg
exten => _.,n,Hangup()
;
; Handle failed messaging
exten =>
_.,n(sendfailedmsg),Set(MESSAGE(body)="[${STRFTIME(${EPOCH},,%d%m%Y-%H:%M:%S)}] Your message to ${EXTEN} has failed. Retry later.")
exten => _.,n,Set(ME_1=${CUT(MESSAGE(from),<,2)})
exten => _.,n,Set(ACTUALFROM=${CUT(ME_1,@,1)})
exten => _.,n,MessageSend(${ACTUALFROM},ServiceCenter)
exten => _.,n,Hangup()
exten => _.,n,Hangup()
```

The screenshot displays the FreePBX Configuration File Editor. The top navigation bar includes links for Admin, Applications, Connectivity, Dashboard, Reports, Settings, and UCP. The main heading is "Configuration File Editor" with a sub-link "What is Configuration File Editor". Below this is a section titled "Working on extensions\_custom.conf". On the left, a file tree shows various Asterisk Custom Configuration Files, with "extensions\_custom.conf" selected. The main editor area shows the following configuration code:

```
1 [astsms]
2 exten => _X.,1,NoOp(SMS receiving dialplan invoked)
3 exten => _X.,n,NoOp(To ${MESSAGE(to)})
4 exten => _X.,n,NoOp(From ${MESSAGE(from)})
5 exten => _X.,n,NoOp(Body ${MESSAGE(body)})
6 exten => _X.,n,Set(ACTUALTO=${CUT(MESSAGE(to),@,1)})
7 exten => _X.,n,ExecIf("${ACTUALTO}" != "sip:${EXTEN}")?Set(ACTUALTO=sip:${EXTEN})
8 exten => _X.,n,MessageSend(${ACTUALTO},${MESSAGE(from)})
9 exten => _X.,n,NoOp(Send status is ${MESSAGE_SEND_STATUS})
10 exten => _X.,n,GotoIf("${MESSAGE_SEND_STATUS}" != "SUCCESS")?sendfailedmsg
11 exten => _X.,n,Hangup()
12 ;
13 ; Handle failed messaging
14 exten => _X.,n(sendfailedmsg),NoOp(Sending error to user)
15 exten => _X.,n,Set(SRC=${MESSAGE(from)})
16 exten => _X.,n,Set(DST=${MESSAGE(to)})
17 exten => _X.,n,Set(MSG=${MESSAGE(body)})
```

At the bottom right of the editor are "Save" and "Delete" buttons. The footer of the interface includes the FreePBX logo with the tagline "let freedom ring™", the text "FreePBX is a registered trademark of Sangoma Technologies Inc. FreePBX 13.0.189 is licensed under the GPL Copyright © 2007-2016", and the Sangoma logo.

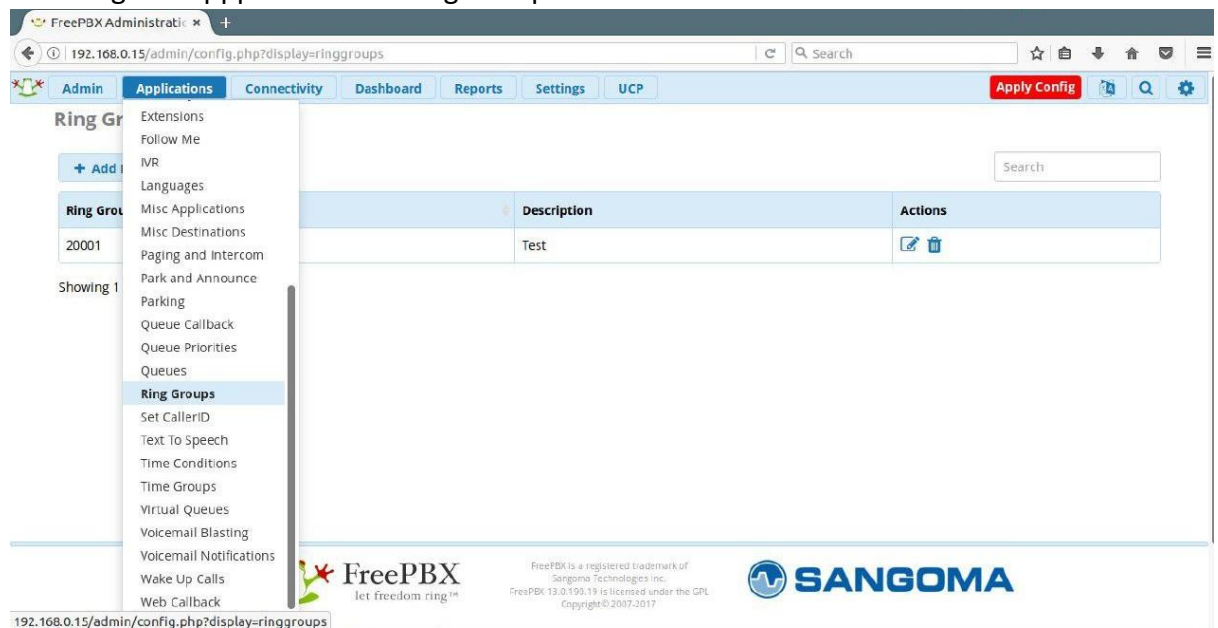
19. To set up conference call feature, go to Application -> Conferences



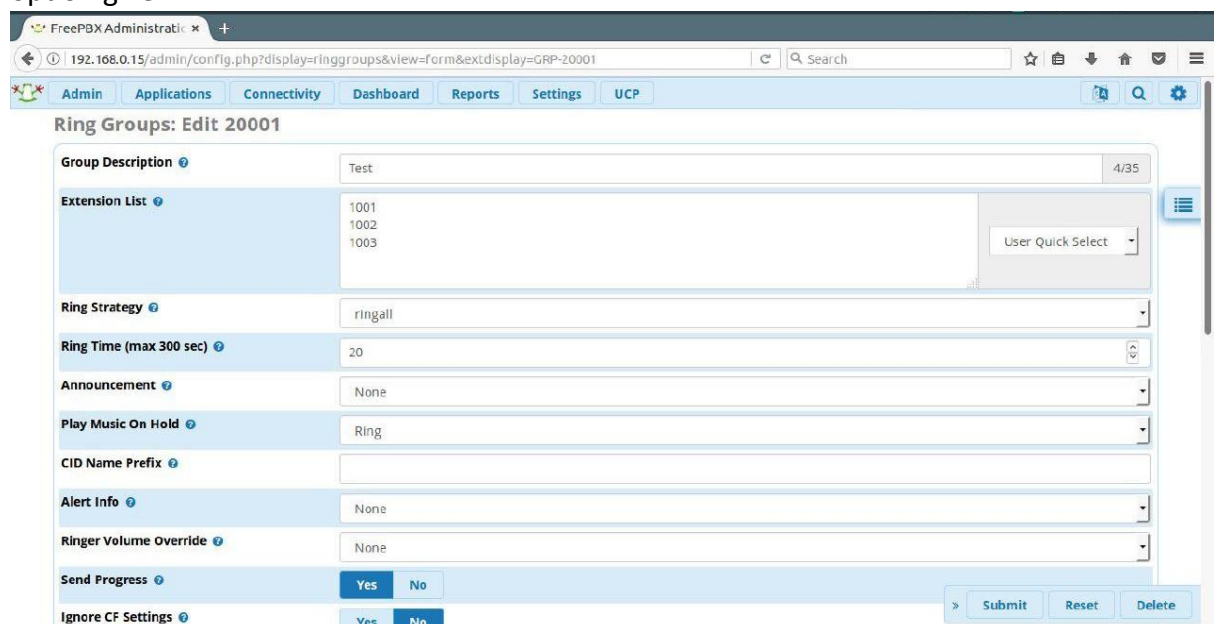
20. We click on Add button to add a conference number, (where a conference number is such that an asterisk user using the system can call in the given number and will be connected to all other users calling in the same number). We have added a conference number here (which should not match with any other number previously assigned to any user). We have added a Conference name and then we click on submit.

The screenshot shows the 'Conferences: Add' form in the FreePBX Administration interface. The browser address bar displays '192.168.0.15/admin/config.php?display=conferences&view=form'. The top navigation bar is the same as in the previous screenshot. The form contains the following fields and options: 'Conference Number' (text input with value '10002'), 'Conference Name' (text input with value 'ConfTest2' and a character count '9/50'), 'User PIN' (text input), 'Admin PIN' (text input), 'Language' (dropdown menu with 'Inherit' selected), 'Join Message' (dropdown menu with 'None' selected), 'Leader Wait' (radio buttons for 'Yes' and 'No'), 'Leader Leave' (radio buttons for 'Yes' and 'No'), 'Talker Optimization' (radio buttons for 'Yes' and 'No'), 'Talker Detection' (radio buttons for 'Yes' and 'No'), 'Quiet Mode' (radio buttons for 'Yes' and 'No'), 'User Count' (radio buttons for 'Yes' and 'No'), and 'User join/leave' (radio buttons for 'Yes' and 'No'). At the bottom right, there are 'Submit' and 'Reset' buttons.

21. Another feature which we are going to implement now is calling a group of numbers at time of Emergency (we call it emergency call, or ring a group), the call is connected to the person who first picks up the phone in the group. For setting up this we go to Applications -> Ring Groups.



22. We will add a Group Description here, and the Asterisk Numbers of all those users for whom the emergency call feature is meant for will be added in the Extension List option given.





23. We scroll down in the same window and select Conferences in the option “ Destination if not answered “ .

The screenshot shows the FreePBX Administration interface. The 'Settings' tab is selected. The 'Destination if no answer' setting is highlighted, with a dropdown menu showing 'Conferences' and '2001 Test'. The 'Submit', 'Reset', and 'Delete' buttons are visible at the bottom right.

Setting	Value
Ringer Volume Override	None
Send Progress	Yes No
Ignore CF Settings	Yes No
Skip Busy Agent	Yes No
Enable Call Pickup	Yes No
Confirm Calls	Yes No
Remote Announce	Default
Too-Late Announce	Default
Change External CID Configuration	Default
Fixed CID Value	
Call Recording	Force Dont Care Never
Destination if no answer	Conferences 2001 Test

24. Now for setting up the mobile to asterisk connectivity we need to add a Binding IP address. For which we go to Settings -> Asterisk SIP Settings

The screenshot shows the FreePBX Administration interface. The 'Settings' tab is selected. The 'Asterisk SIP Settings' menu is open, showing options like 'Advanced Settings', 'Asterisk IAX Settings', 'Asterisk Logfile Settings', 'Asterisk Manager Users', 'Asterisk REST Interface Users', 'Asterisk SIP Settings', 'CRM Settings', 'EndPoint Manager', 'Fax Configuration', 'Music on Hold', 'PIN Sets', 'Route Congestion Messages', 'Text To Speech Engines', and 'Voicemail Admin'. The 'Asterisk SIP Settings' option is highlighted.

SIP Settings

SIP driver information

Asterisk is currently using chan\_pjsip and chan\_sip for SIP Traffic. You can change this on the Advanced Settings Page

General SIP Settings Chan SIP Settings Chan PJSIP Settings

Edit Settings

NAT Settings

NAT yes no never

IP Configuration Public IP Static IP

Audio Codecs

Non-Standard g726 Yes No

T38 Pass-Through No

Video Codecs

Video Support Enabled Disabled

Submit Reset

25. And we select “Chain SIP Settings” tab

FreePBX Administration

192.168.0.15/admin/config.php?display=sipsettings

Admin Applications Connectivity Dashboard Reports Settings UCP

### SIP Settings

**SIP driver information**

Asterisk is currently using chan\_pjsip and chan\_sip for SIP Traffic.  
You can change this on the Advanced Settings Page

General SIP Settings **Chain SIP Settings** Chan PJSIP Settings

#### Edit Settings

**NAT Settings**

NAT ☒ yes ☐ no ☐ never ☐ route

IP Configuration ☒ Public IP ☐ Static IP ☐ Dynamic IP

**Audio Codes**

Non-Standard g726 ☒ Yes ☐ No

T38 Pass-Through

**Video Codes**

Video Support ☒ Enabled ☐ Disabled

» Submit Reset

26. We add the binding IP address which is the local IP address of the server, one can quickly verify it in the browser url. ( For Ex: It's 192.168.0.15 here)

FreePBX Administration

192.168.0.15/admin/config.php?display=sipsettings

Admin Applications Connectivity Dashboard Reports Settings UCP

Apply Config

### Jitter Buffer Settings

Enable Jitter Buffer ☒ Yes ☐ No

#### Advanced General Settings

Default Context

Bind Address

Bind Port

TLS Bind Address

TLS Bind Port

Allow SIP Guests ☒ Yes ☐ No

Enable SRV Lookup ☒ Yes ☐ No

Enable TCP ☒ Yes ☐ No

Call Events ☒ Yes ☐ No

Other SIP Settings  =

Add Field

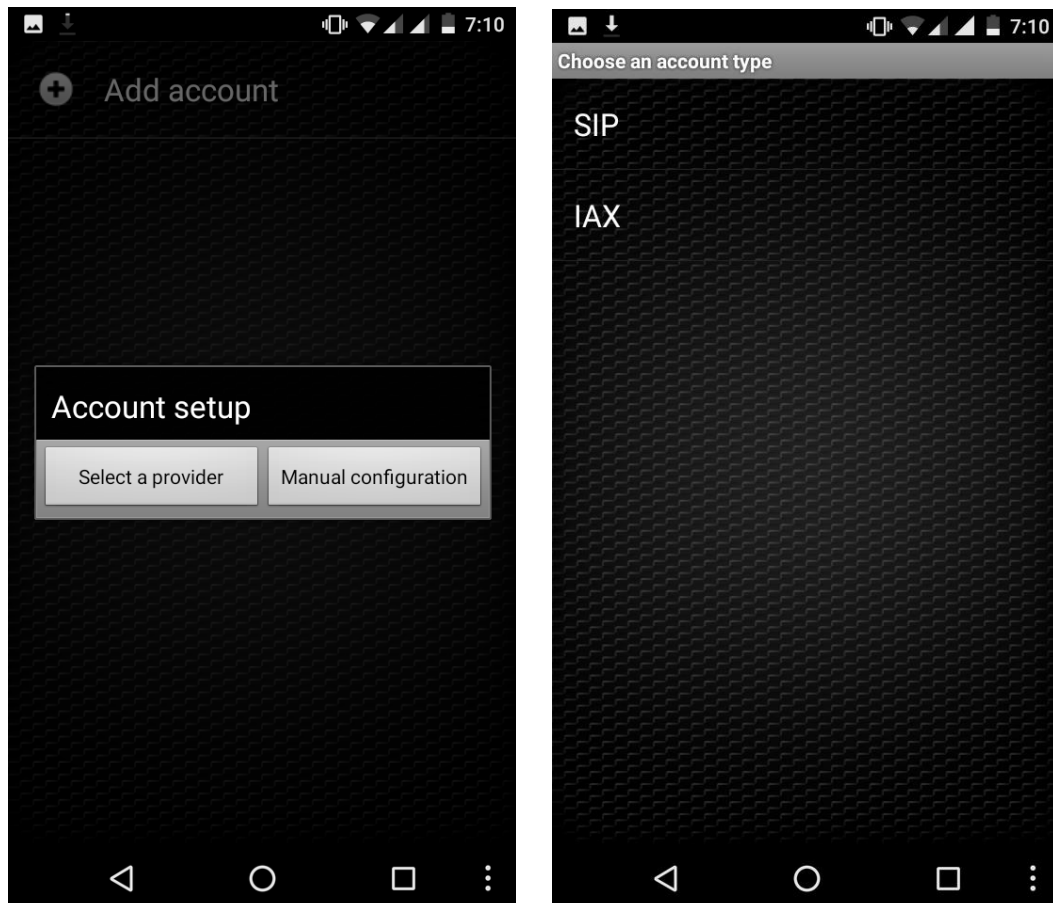
» Submit Reset

27. For few Operating Systems we need to disable the OS's firewall so that external IP connections can be set up, which is blocked by the firewall at times. For which run the following command in the Asterisk command window

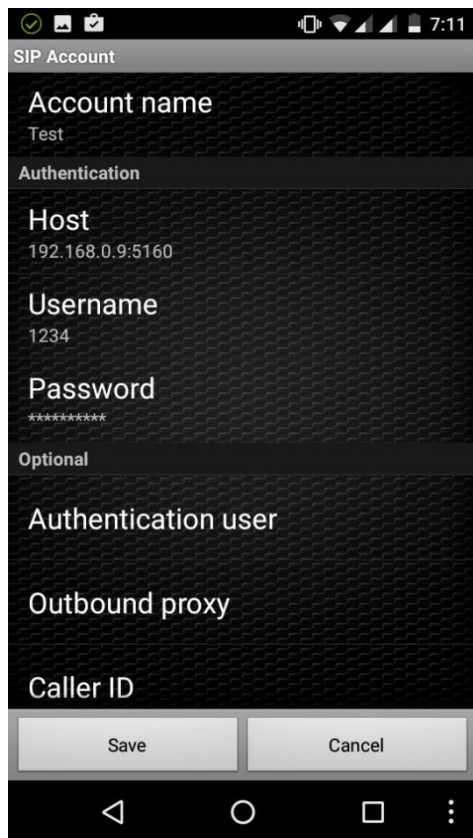
“service iptables stop”

28. Now we will save and restart the server to load the changes in action.

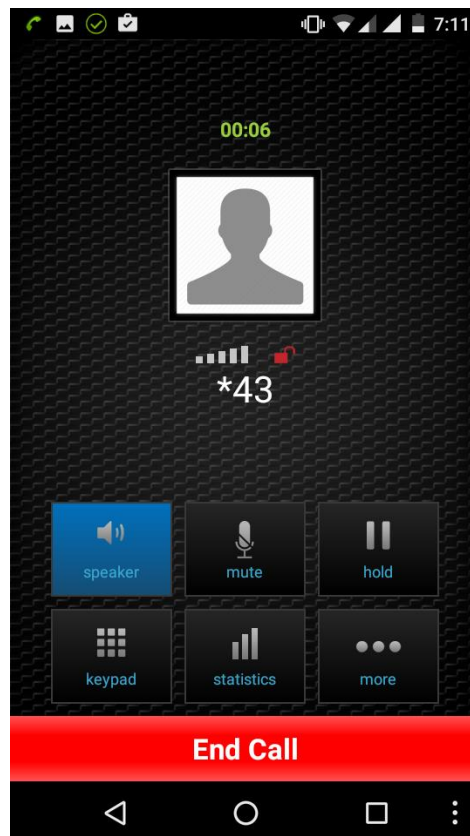
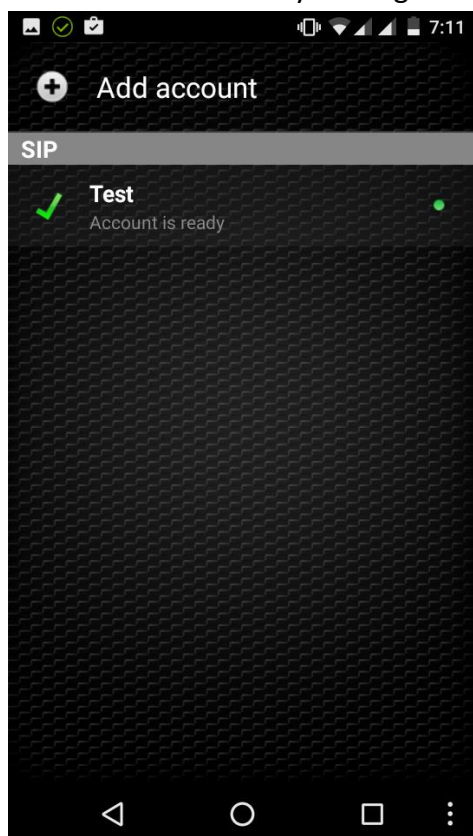
29. In Zoiper application in mobile phone, we will add an account and choose for manual configuration and select to **SIP** option



30. Account name can be any text. Host will be the ip address of server with port **5160**. Username will be sip extension number (If default user is used during the Extension creation) and password will be the user password for the extension. That's all the configuration we require.



31. Now we can save the settings. Account will be marked as ready (Green Check). We can echo test server by dialling \*43



## CONCLUSION:

After the project study and work we came into a conclusion that Asterisk Framework is quite flexible and easy to use when we are handling it through FreePBX. Few configurations and a bit of scripting knowledge is sufficient to handle the entire setup. This can be used by the housing apartments having existing WiFi setups in very low cost. This can be used in hospitals and offices and one doesn't need to go through the problems of setting up an IP Phone infrastructure. Our configuration is secure and robust for calls, video calls, messaging, conference calling, and emergency calling. New features can be added as per the user review and requirements.