VOIP Server Configurations

1) sudo su

2) apt – get install asterisk **For Asterisk Installation**

Asterisk: Asterisk is a software implementation of a private branch exchange.Asterisk is used to establish and control telephone calls between telecommunication end points and services on Voice over Internet Protocol. The name Asterisk came from the symbol “*” which is used to represent a signal used in Dual tone multi frequency which are widely used for telecommunication signalling between telephone handsets and switching centers over analog telephone lines in the voice-frequency bands.

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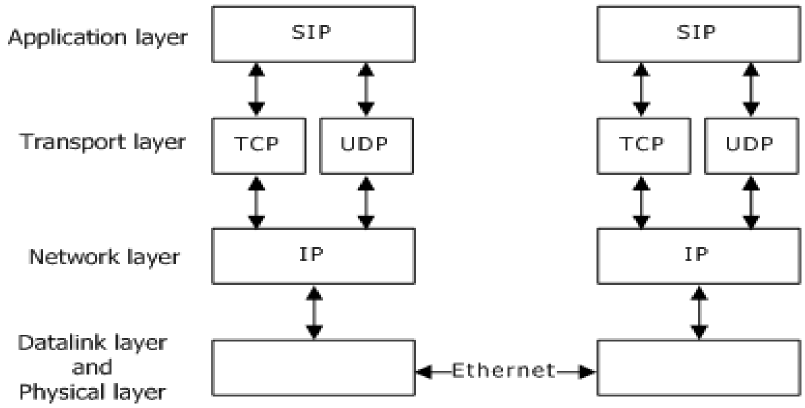
3) asterisk –r

4) mv /etc/asterisk/sip.conf /etc/asterisk/sip.conf.orig

For creating a file in which we can give the username, password and categorize dynamically accessed SIP users

5) nano /etc/asterisk/sip.conf

SIP Protocol: The acronym SIP stands for Session Initiation Protocol. It is an application layer protocol that works in conjunction with other application layer protocols to control multimedia communication sessions over the Internet. SIP is often used in Voice-over-IP telephony to establish the connection for telephone calls. The Users or subscribers who can be able to send or receive data through this protocol are referred as **SIP-USERS.** This is a standardized protocol which establishes either of TCP or UDP connections between the END Users for the data transfer.This exchange of data between the end users is called a session.



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6) The SIP users can be Configured as follows:

Function 1: Configuring SIP Users

[general]

Context=internals

allowguest=no

allowoverlap=no

bindport=8088

bindaddr=0.0.0.0

srvlookup=no

disallow=all

allow=ulaw **To Use ulaw codec (coder-Decoder)**

alwaysauthreject=yes

canreinvite=no

For Enabling Network Address Translation to increase the interconnection between as many as SIP users

nat=yes

session-timers=refuse

localnet=192.168.1.0/255.255.255.0

IP Address of the Network through which Raspberry Pi is connected with subnet Mask.

- **SIP user1 which can be dynamically accessed by the**

Username: 9111

Password: 111

```
[9111]
    type=friend            For Categorizing the sip users
    host=dynamic
    secret=111
    context=internal
```

Further configurations can be done in INTERNAL Function.

- **SIP user2 which can be dynamically accessed by the**

Username: 9112

Password: 222

```
[9112]
  type=friend
  host=dynamic
  secret=222
  context=internal
```

- **SIP user3 which can be dynamically accessed by the**

Username: 9113

Password: 333

```
[9113]
 type = friend
 host = dynamic
 secret= 333
context=internal
```

- **SIP user4 which can be dynamically accessed by the**

Username: 9114

Password: 444

```
[9114]
 type =friend
 host = dynamic
 secret= 444
 context=internal
```

Similarly we can configure as many as sip users needed.

*** 9111 ,9112,9113,9114 are the sip users which can be accessed dynamically with the secrets 111, 222, 333, 444 respectively.**

7) mv /etc/asterisk/extensions.conf /etc/asterisk/extensions.conf.or

To Create a file for Voice Extensions for the above SIP Users

8) nano /etc/asterisk/extensions.conf

9) **<u>Function 2</u>:**SIP Extensions for call processing

[internal]

```
exten =>9111 ,1 ,Answer()
exten => 9111,2,Dial(SIP/9111,60)         Syntax: Dial(type/identifier, timeout)
exten =>9111,3,Playback(vm.nobodyavail)   If the call goes unanswered
exten =>9111,4,VoiceMail(9111@main)       Syntax: VoiceMail(Sipuser@context)
exten => 9111,5,Hangup()                  To Hung Up the call if the call goes unanswered
```

Dial( ) application will ring all of the specified destinations simultaneously and bridge the inbound call with whichever destination channel answers first.

```
exten =>9112 ,1 ,Answer()
exten => 9112,2,Dial(SIP/9112,60)
exten =>9112,3,Playback(vm.nobodyavail)
exten =>9112,4,VoiceMail(9112@main)
exten => 9112,5,Hangup()
```

```
exten =>9113 ,1 ,Answer()
exten => 9113,2,Dial(SIP/9113,60)
exten =>9113,3,Playback(vm.nobodyavail)
exten =>9113,4,VoiceMail(9113@main)
exten => 9113,5,Hangup()
```

```
exten =>9114 ,1 ,Answer()
exten => 9114,2,Dial(SIP/9114,60)
exten =>9114,3,Playback(vm.nobodyavail)
```

```
exten =>9114,4,VoiceMail(9114@main)

exten => 9114,5,Hangup()
```

* **For Call Forwarding:**

exten =>1111,1,VoiceMain(9111@main) exten => 1111,2,Hangup() **To Hung up the call after answered.**
exten => 1112,1,VoicemailMain(9112@main) exten => 1112,2,Hangup()
exten => 1113,1,VoicemailMain(9113@main) exten => 1113,2,Hangup()
exten => 1114,1,VoicemailMain(9114@main) exten => 1114,2,Hangup()

10) mv /etc /asterisk/voicemail.conf /etc/asterisk/voicemail.conf.orig

From the syntax VoiceMail(Sipuser@context**) whenever a call is to be processed it should be checking for the password for that particular sip user which are specified in the "**context=**[main]" function written in the following file.**

For creating a file in which we can give the password

11) nano /etc /asterisk/voicemail.conf

12**) [CRYPTO(3)] -Voicemail configurations**

[main]

```
9111 => 111

9112=> 222

9113 => 333

9114 => 444
```

13) nano /etc /asterisk/ari.conf

ARI (Asterisk Rest Interface)

FUNCTION 3: Ari.conf

```
[general]
      enabled=yes
      allowed_origins=localhost:8088,http://ari.asterisk.org
[asterisk]
      type=user
      read_only=no
     password=asterisk
```

14) nano /etc /asterisk/http.conf **(HTTP Configuration for binding with Router if needed)**

```
[general]
enabled=yes
bindaddr=0.0.0.0
bindport=8088
```

If we need to extend our network , the raspberry pi voip server can be communicated with the Routers using this bindport 8088 configured.

15) asterisk – r

16**)** reload **command for Error checking**

17) sip show peers **To check the sip users which are connected and active.**

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