CHAPTER 4
ANALYSIS AND DESIGN

4.1 Analysis

The purpose of this project is to build an Interactive Voice Response simulation using Asterisk as server and Web Browser as user. To access Interactive Voice Response from web browser, then use API sipml5. This sipml5 can access IVR on web browser for desktop or smartphone, this project is using Mozilla Firefox.

Asterisk is using phpAGI to run a php script, in this php script include a logic about how this program can be run. First, Asterisk will provide welcome message and will prompt the user to input a year class of the user, then Asterisk will temporaryli save the input from user, next system will prompt the user to input one last number on majors code, system will temporaryli save again a input from user. Last, system will prompt the user to input four number in a NIM, then system combines all user input.

Next Asterisk provide a menu in the form of voice, if user touch or press button, client will send the signal as DTMF info. After the server receive a request from user, the server will check in database about data request from user, if data exists or not in database, Asterisk will respond to the user in voice. If the user incorrectly inputs data up to 3 times, Asterisk automaticly hangup the call without approval from user. In this step, it takes several attempts to get the program running correctly.

4.2 Design

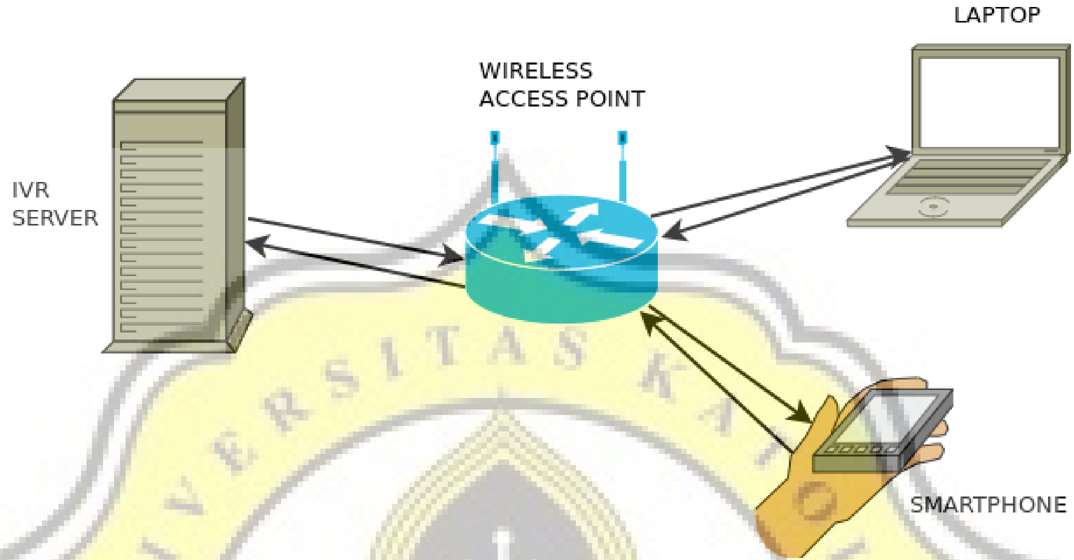


Illustration 4.2.1: Jaringan

This project are using TP-LINK TL-WR840N as a tool to share network from the IVR server, so communication between the IVR server with the client must go through the Access Point in order to connect in one network.

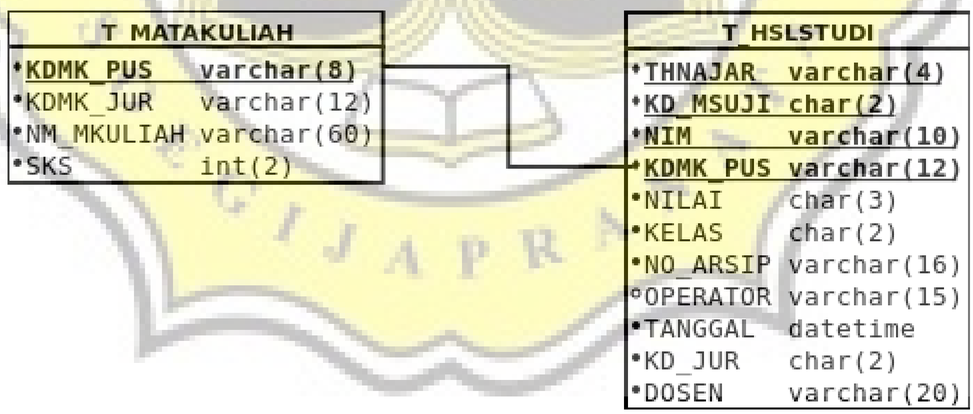


Illustration 4.2.2: Database Table

In the database contain 1 primary key on T_MATAKULIAH, while on T_HSLSTUDI contain 4 primary key which is usefull as a deternet to the same input/duplicte entry, in this case it is not that the same 2 KDMK_PUS taken by the same NIM at the same KD_MSUJI and in the same THNAJAR, it can lead to incorrect value calculations, either the value becomes higher in value or becomes lower, this causes chaos in the database.

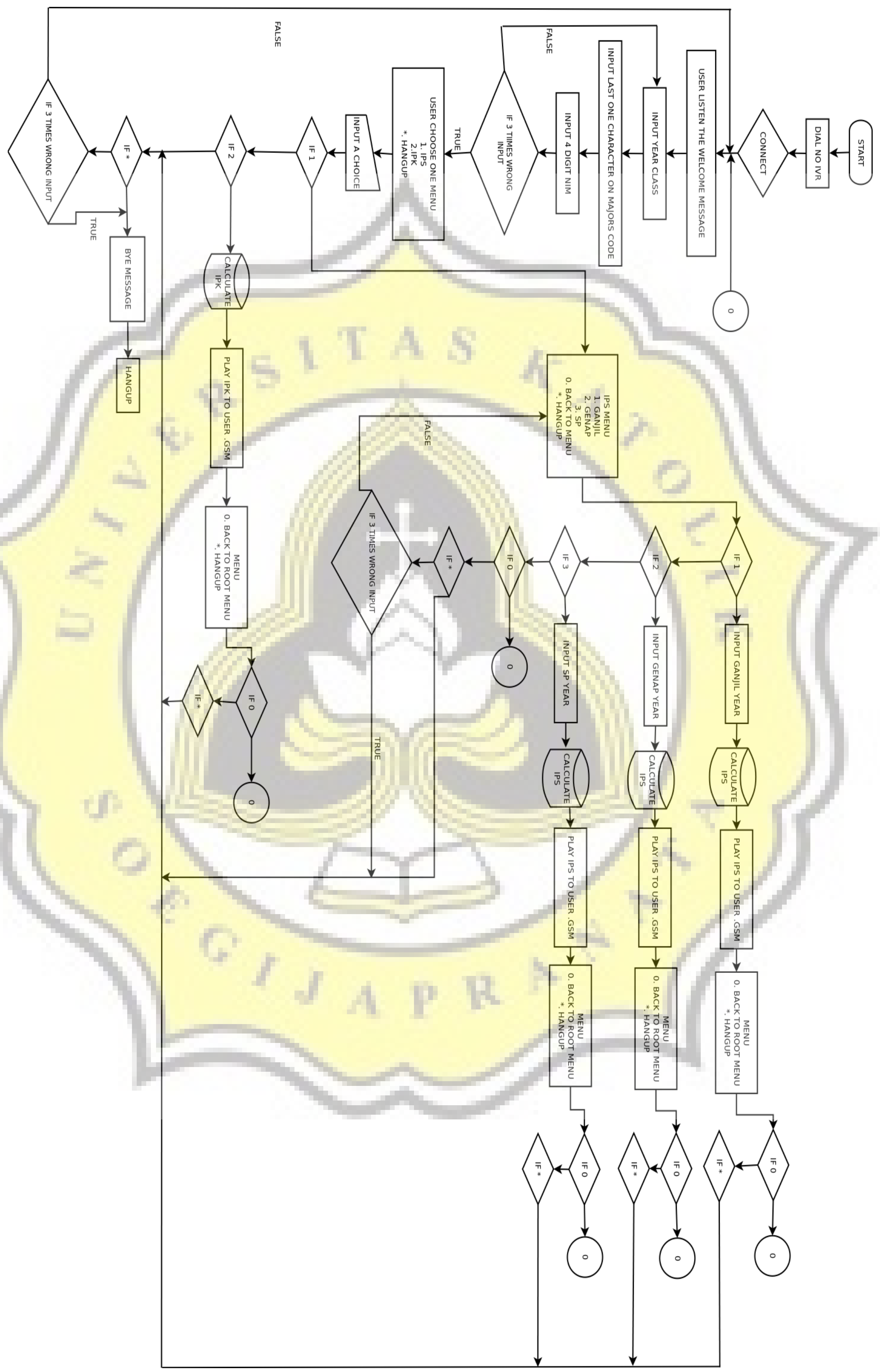


Illustration 4.2.3: IVR Flowchart

From the flowchart above, the logic of the program is when the user dials the IVR dial number, the Asterisk server will receive the call and play the welcome message and server will ask the user to enter the 2 digits of the year that exists in the first two digits of the user NIM, then after the user input is received, the server will ask for 1 digit of the departement code, and the last server will ask the user to input the last 4 digits of NIM. After all input is received, the server will combine all the characters that have been entered by the user and check to the database whether the data entered by the user is true or not. If the user provided parameter is incorrect or mising in the database, the server will give the user the chance to input the correct data up to 3 times, if the user still does not input the correct data, the server will automaticly hangup the call from the user. If the data entered user is correct, then iVR server will give 2 menu to user, that is menu to check IP Semester or Cumulative IP, if user press number 1, that is Semester IP menu, then user will be given back 3 menu, that is Odd semester, even semester, or other semester, after selecting one of the 3 menus, the server will ask the user to re-enter the semester that has been selected by the user, then ther server will check to the database with the user's NIM parameters, odd semester information, even or other, and the year the semester if taken. If the server finds the data in accordance with the paramters already entered by the user, then the server will calculate the existing data on the database and in accordance with the parameteres entered by the user into one output of the IP semester, then the output data will be provided to the user in the form of voice. But if the parameters given by the user is wrong or not in the database, the server will give the user the chance to input the correct data up to 3 times, if the user still does not input the exact data, the server will automatically hangup the call from the user. Once the user gets the information that has been requested, the server will give back the option to continue the service back to the main menu or choose to end the call. Then move to the Cumulative IP menu in the IVR main menu, If user press number 2, then the server will check on the database, and calculate the data requested by the user then provide a response to the user in the form of voice. Then the server will give 2 more options to return to the main menu or immediately end the call.

4.3 Installation

- First install Asterisk by typing command 'apt-get install asterisk' on the terminal(in this project using Debian linux operating system).

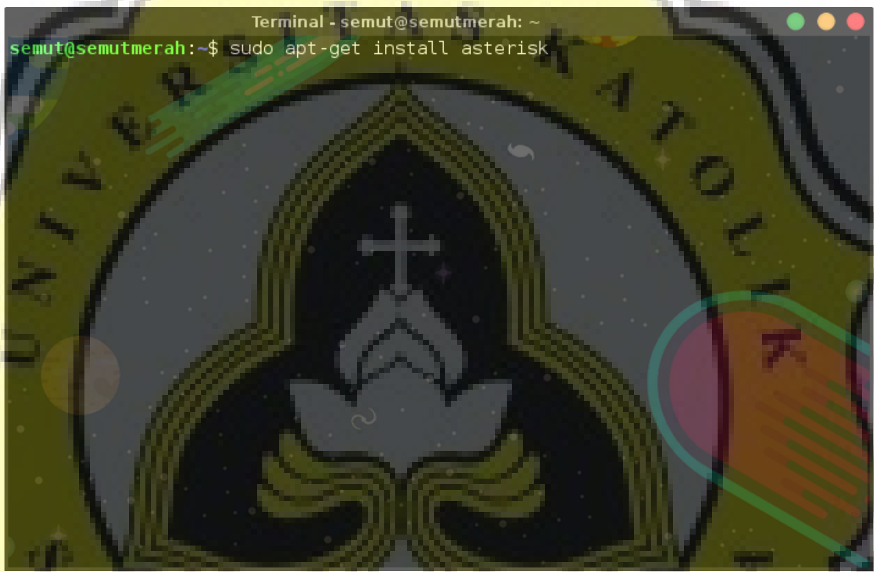


Illustration 4.3.1: Install Asterisk

- Then also installize php, apache, mariadb and phpmyadmin which will be useful to run phpAGI by typing in terminal 'apt-get install php apache2 mariadb-server phpmyadmin'.

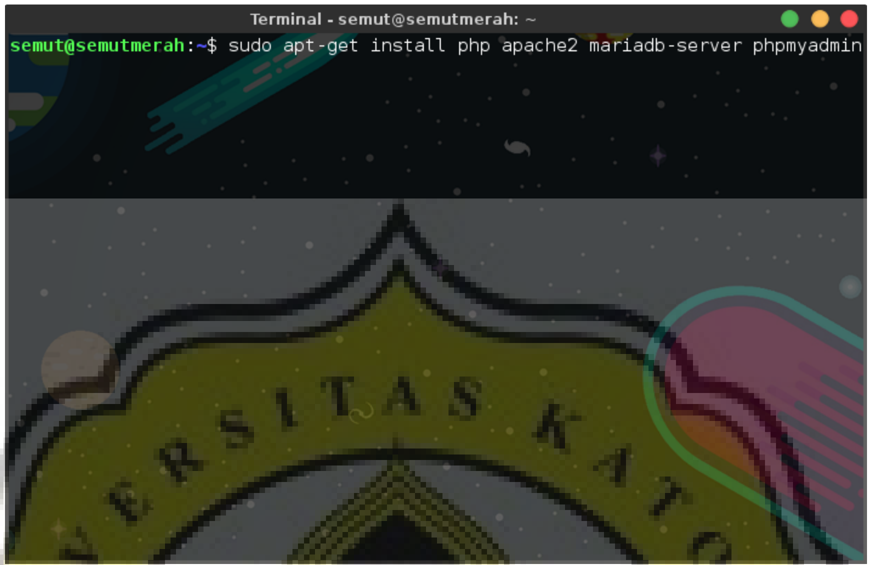


Illustration 4.3.2: Install PHP, Apache, MariaDB, PHPMyAdmin

- First configure 000-default.conf file in '/etc/apache2/sites-available/' directory, edit the contents of 000-default.conf file. Customize the existing directory in the above configuration with the existing agi-bin folder in the directory where Asterisk is installed on a Linux system.

<VirtualHost *:80>

ServerAdmin webmaster@localhost

DocumentRoot /usr/share/asterisk/agi-bin/

<Directory /usr/share/asterisk/agi-bin/>

Options +Indexes +FollowSymLinks

AllowOrverride All

Order allow, deny

allow from all

</Directory>

</VirtualHost>

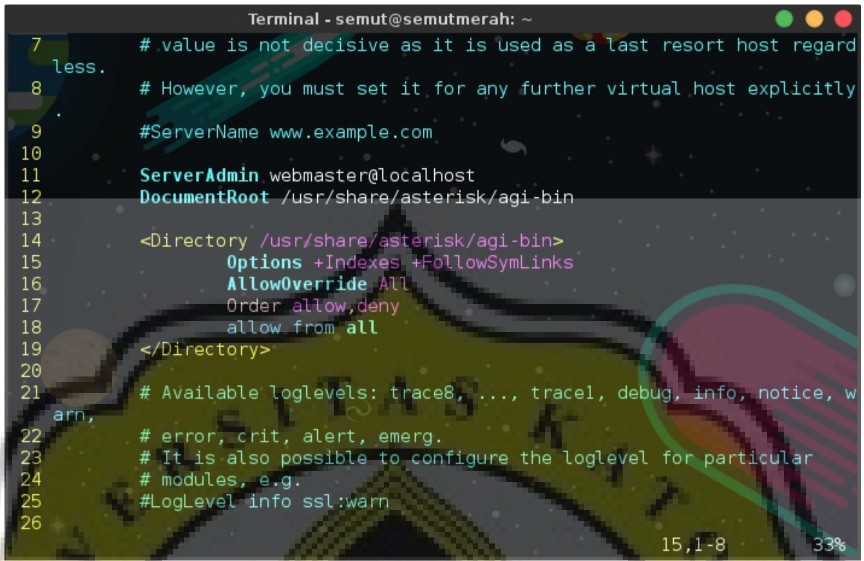


Illustration 4.3.3: Edit 000-default.conf

- Also edit the apache2 file, conf in the directory '/etc/apache2/apache2.conf'.

  <Directory /usr/share/asterisk/agi-bin/>

  Options Indexes FollowSymLinks

  AllowOverride All

  Require all granted

  </Directory>

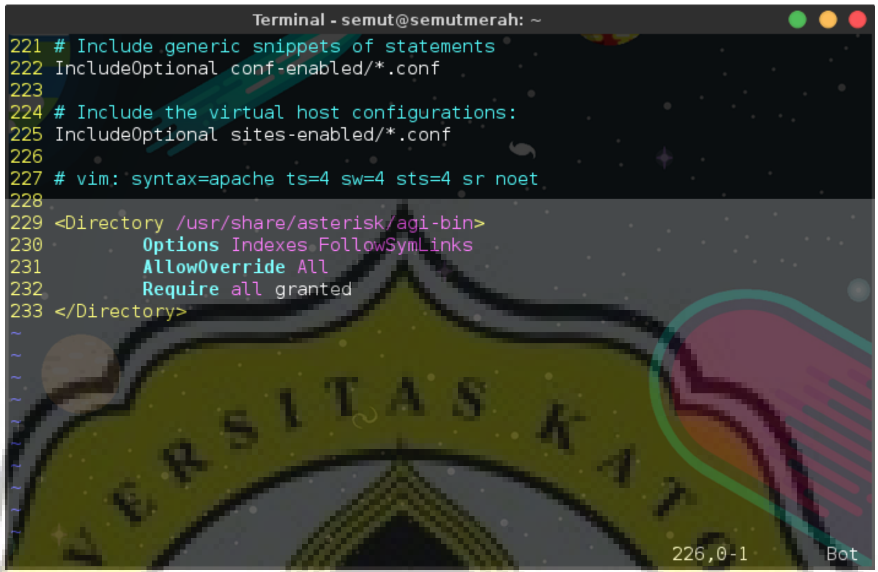


Illustration 4.3.4: Edit apache2.conf

- Then restart apache by running the command ‘sudo systemctl restart apache2’ on the Linux terminal.

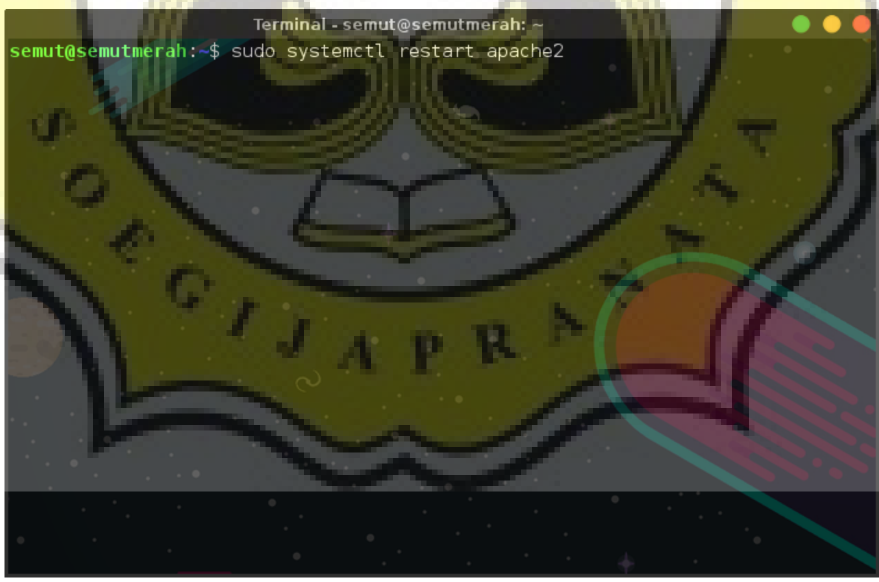


Illustration 4.3.5: Restart apache

- Download phpAGI at url ‘https://sourceforge.net/projects/phpagi/files/phpagi/2.20/phpagi-2.20.tgz/download’, then extract that file to directory ‘/usr/share/aterisk/agi-bin’.
- Configure existing sip.conf file in ‘/etc/asterisk/sip.conf’ directory to add:

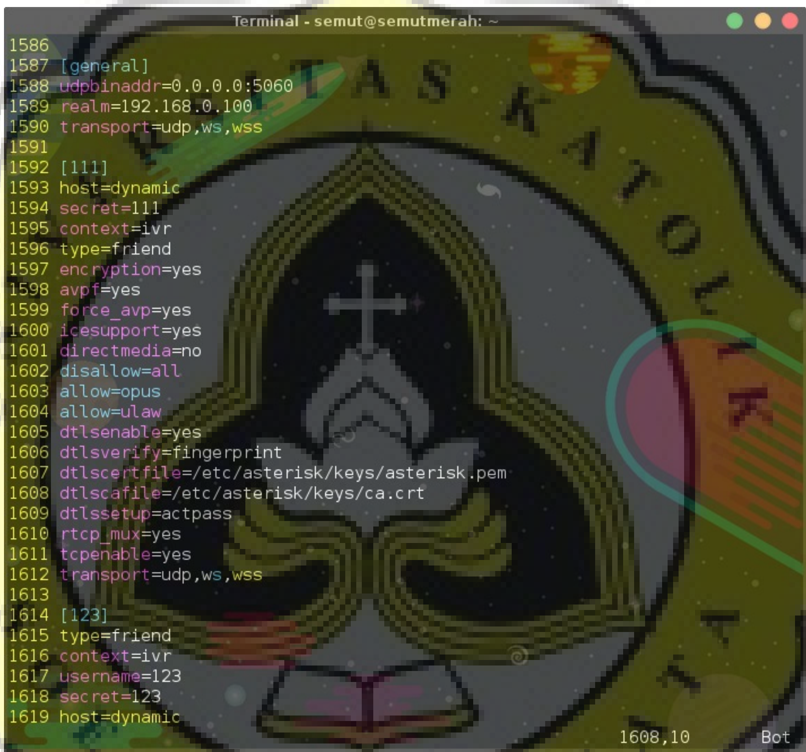


Illustration 4.3.6: Add SIP Account

- On line 1587-1500 declare a global variable name [general], with default port for sip connection 5060, then

1. udpbinaddr = 0.0.0.0:5060 is used to listen to connections on all IP configured in the system by using port 5060 as the default port of the SIP connection.
2. realm = 192.168.0.100 is use to authenticate the address use to access Asterisk.

3. transport = udp,wss,ws use to initialize the port to be used, here use is port udp, wss, ws.

- On line 1592-1612 is an account creation which will be user to login to the web socket.

1. [111] symbolizes the number to be used at login.
2. Host = dynamic symbolizes the number to be used at login.
3. contect = ivr, used to call ivr contex exiting in extensions.conf located in the directory '/etc/asterisk/extensions.conf'.
4. Type = friend, used for this account to be dialed or made outbound calls.
5. Encryption = yes, is used to make all activities performed encrypted.
6. Avpf = yes, for outgoing calls or media using avpf profiles, and will reject those not using avpf.
7. Force_avp = yes,so that call incoming calls use avp, including DTLS_SRTP.
8. Directmedia = yes, in order to update the media.
9. Disallow = all, does not allow any access except those to be set.
10. Allow = ulaw, only allow ulaw to acces.
11. Dtlsenable = to enable dtls encryption and used in the activity.
12. dtls_verify = fingerprint, used to make the echo set to dtls.
13. Dtlscertfile = /path, used to access the dtls certificate in the system.
14. Dtlscafile = /path, used to acces the ca certificate in the system.
15. dtls_setup= actpass, allow connection from peer.
16. rtcp_mux = yes, used in order for Asterisk to negotiate to use rtcp_mux in every media event.

17. Tcpenable = yes, used to enable tcp.

18. Transport = udp, ws, wss, used for all data to use udp, ws or wss protocol.

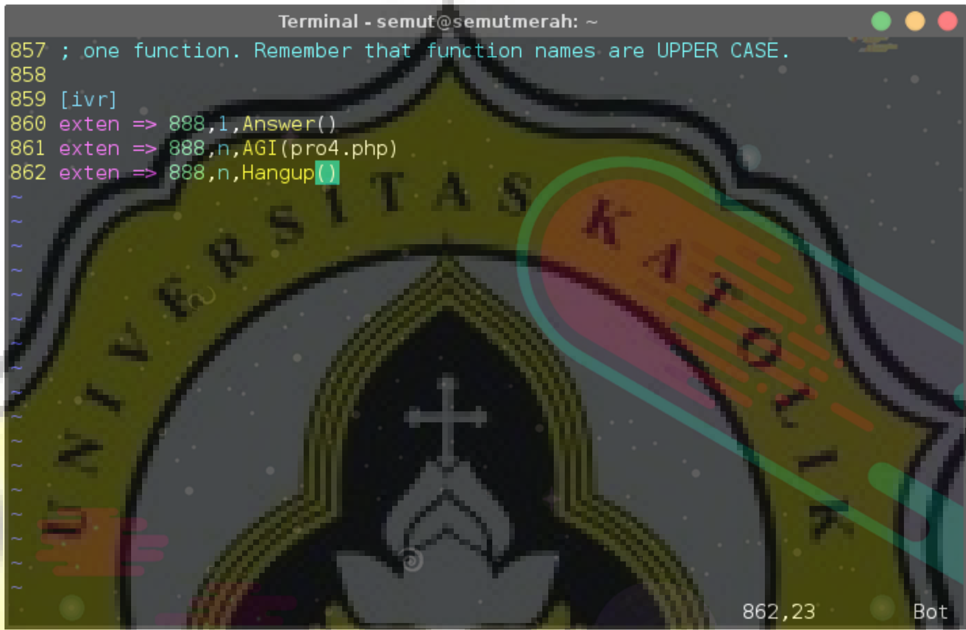
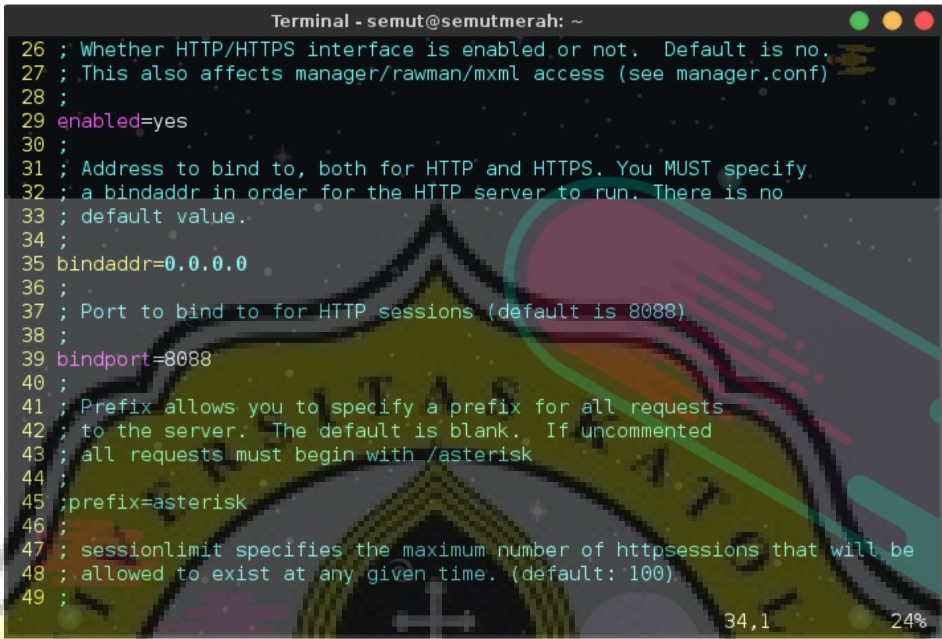


Illustration 4.3.7: Extension.conf

1. Line 859-862 is the dialplan to use.
2. Line 859 is the context to be used.
3. Line 860 use to receive call from extension 888.
4. Line 861 use to access AGI file available on system.
5. Line 862 is used to end a call when the activity is done.



```
Terminal - semut@semutmerah: ~
26 ; Whether HTTP/HTTPS interface is enabled or not.  Default is no.
27 ; This also affects manager/rawman/mxml access (see manager.conf)
28 ;
29 enabled=yes
30 ;
31 ; Address to bind to, both for HTTP and HTTPS. You MUST specify
32 ; a bindaddr in order for the HTTP server to run. There is no
33 ; default value.
34 ;
35 bindaddr=0.0.0.0
36 ;
37 ; Port to bind to for HTTP sessions (default is 8088)
38 ;
39 bindport=8088
40 ;
41 ; Prefix allows you to specify a prefix for all requests
42 ; to the server.  The default is blank.  If uncommented
43 ; all requests must begin with /asterisk
44 ;
45 ;prefix=asterisk
46 ;
47 ; sessionlimit specifies the maximum number of httpsessions that will be
48 ; allowed to exist at any given time. (default: 100)
49 ;
                                                    34,1          24%
```

Illustration 4.3.8: http.conf part 1

1. Line 29 used to activated pjsip on Asterisk.
2. Line 35 used to set bindaddr to 0.0.0.0
3. Line 39 used for standard ports used 8088.

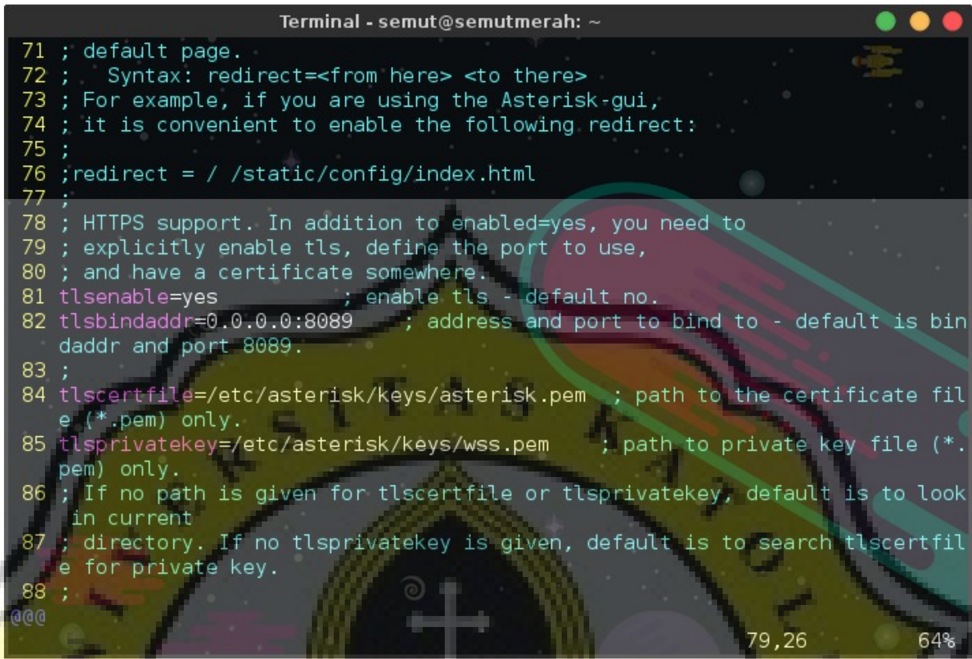


Illustration 4.3.9: http.conf part2

4. Line 81 used to enable tls.

5. Line 82 used for the address used to be flexible and use port 8089.

6. Line 84 and 85 used to access the certificate files used.

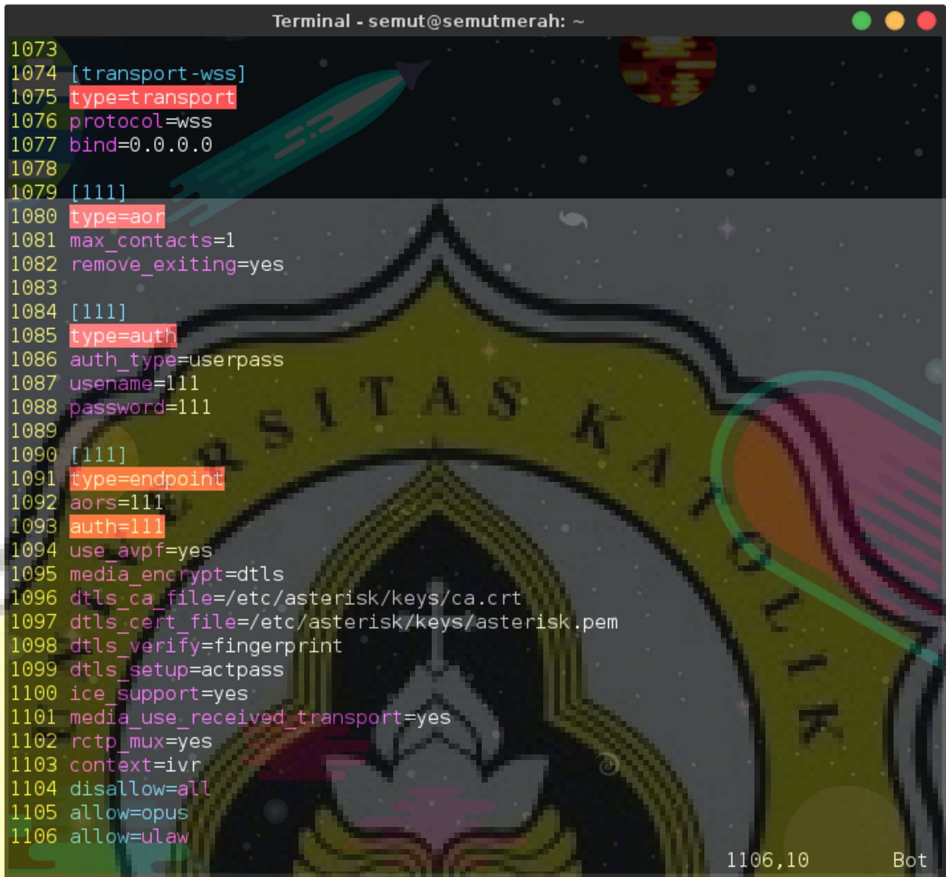


Illustration 4.3.10: pjsip.conf

1. Line 1074-1077 is used to initialize the use of transport-wss on an Asterisk connection. With type = transport, protocol = wss, and also bind = 0.0.0.0
2. Line 1079-1082 used for set type = aor, max_contacts = 1, and also remove_exiting = yes.
3. Line 1084-1088 used for set username and password use on account 111.
4. Line 1090-1106 used to set the endpoint and then the same set that exists in http.conf then add the allow opus codec at the end of the line.

Finally, download also sipml5 which will be used as a demo via web browser either from pc browser in the following url https://github.com/DoubangoTelecom/sipml5/archive/master.zip. Then extract and move to directory '/usr/share/asterisk/agi-bin/'.