Asterisk v1.6.0.8, FreePBX v2.5.1, and A2Billing v1.3.4 : Installation Guide for Linux UBUNTU 8.10

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1.0 INSTALLING THE UBUNTU 8.10 SERVER

Ubuntu LAMP server install the following versions:

Ubuntu 8.10 (Intrepid)
Apache 2.2.9
Mysql 5.0.67
PHP 5.2.6

For installing the ubuntu firstly we need to download it from http://www.ubuntu.com/GetUbuntu/download and then burn it in a CD and start booting with the CD Once it starts booting we see the screen in which we need to select the language and press enter. Then the next step is to select “Install Ubuntu Server” and press enter. The Ubuntu server CD will take time to install and then we need to choose the language option for the ubuntu Installation. Then the location option will come. Then the CD-ROM Drivers hardware is detected and scanned after that the network hardware is detected.Once the network hardware is detected then we need to configure it with the DHCP. For the host identity we need to create the host name.Then the hard disk is partitioned. After that the next step is the creation of ext3 file system ,then the base system is installed.The next step is defining the user name and the password for getting the access to the account.Then we need to configure the automatic update option, Now it will start Installing software and here we need to select the server options like the LAMP and OpenSSH server for the LAMP server installation. While installing the software it will require the root password.Then the installation of GRUB Boot loader is done. After following the above steps we are done with the installation and we need to reboot our server.

Once we are done with that we need to configure the static ip address in the ubuntu server.Ubuntu installer has configured our system to get its network settings via DHCP, but we have to change it to a static IP address and to do this we need to edit Edit /etc/network/interfaces and enter the ip address details.Then we need to restart our network services.

2.0 Asterisk on the UBUNTU server

The following steps are required for the installation of the asterisk on the UBUNTU Server:

1. **The system is Updated by using the following:**

```
# apt-get update
# apt-get upgrade
```

2. **Kernel headers are installed by using**

```
# apt-get install linux-headers-`uname-r`
```

You must enter your uname. This can be found as typing **uname –r** on the screen.

3. Creating a symlink to the headers in "/ usr / src / linux" by using the

```
# ln-s / usr / src / linux-headers-`uname-r` / usr / src / linux
```

4. The packages for Asterisk are installed and DAHDI through:

```
# apt-get install bison openssl libssl-dev libasound2-dev libc6-dev libnewt-dev libncurses5-dev
zlib1g-dev gcc make g + + libusb-dev fxload
```

The packages are then Downloaded, extracted and installed following the below mentioned steps:

5.The packages are downloaded and extracted from the following:

```
# wget http://ftp.digium.com/pub/asterisk/asterisk-1.6.0.1.tar.gz
# wget http://downloads.digium.com/pub/telephony/libpri/libpri-1.4.8.tar.gz
# wget http://downloads.digium.com/pub/telephony/dahdi-linux/dahdi-linux-2.1.0.3.tar.gz
# wget http://downloads.digium.com/pub/telephony/dahdi-tools/dahdi-tools-2.1.0.2.tar.gz
# tar xvzf asterisk-1.6.0.1.tar.gz
# tar xvzf libpri-1.4.8.tar.gz
# tar xzvf dahdi-linux-2.1.0.3.tar.gz
# tar xzvf dahdi-tools-2.1.0.2.tar.gz
```

6.Then Compiling the dahdi-linux is done by using:

```
# make
# make install
```

7. Compile the dahdi-tools:

```
#. / configure
#. / make
#. / make install
#. / config make
```

8. Compiling the libpri. Among libpri-1.4.8 folder and run:

```
# make
# make install
```

10. Finally, compile the Asterisk. Currently the last available version is 1.6.0.2, but I had problems trying to install it. So I recommend installing the 1.6.0.1 version, which until now has not had a single problem. To install it, run:

```
#. / configure
# make
# make install
# make samples / / install the examples
# make config
```

3.0 FreePBX Installation

Once you are ready with a asterisk on your system you can go ahead with the web interface for the asterisk. FreePBX provides the required amportal.

FreePBX requires some asterisk add-ons so first step will be to download and install the asterisk add-ons.

1. **Download & Install Astrisk Add-ons**

```
# cd /usr/src/
# sudo wget http://downloads.digium.com/pub/asterisk/asterisk-addons-1.6.0.8.tar.gz
# sudo tar -zxvf asterisk-addons-1.6.0.8.tar.gz
# cd asterisk-addons-1.6.0.8
# ./configure
# make clean
# make
# sudo make install
```

2. **Download FreePBX**

```
cd /tmp
wget http://mirror.freepbx.org/freepbx-2.5.1.tar.gz
cd /usr/src
tar xvfz /tmp/freepbx-2.5.1.tar.gz
cd freepbx-2.5.1
```

3. **Configure Apache for freepbx**

```
# sudo nano /etc/apache/envvars
change : export APACHE_RUN_USER=www-data & export APACHE_RUN_GROUP=www-
data

to:

 export APACHE_RUN_USER=asterisk & export APACHE_RUN_GROUP=asterisk

# sudo nano /etc/apache2/sites-enabled/000-default
```

```
change: AllowOveride None to AllowOverid All under both instances of /var/www
```

4. Configure PHP for FreePBX

```
# sudo nano /etc/php5/apache2/php.ini
change: upload_max_filesize to upload_max_filesize = 20M

# sudo nano /etc/php5/cli/php.ini
change: upload_max_filesize to upload_max_filesize = 20M

Set the php lib directory to be owned by asterisk so that it can make changes to the php.ini
# sudo chown asterisk:asterisk /var/lib/php5 -R

Set ownership of the /var/www directory to asterisk so it can write to files
# sudo chown asterisk:asterisk /var/www -R
```

5. Restart Apache

```
# sudo /etc/init.d/apache2 restart
```

In a web browser test http://<server-ip-addresss>/ if all is well you should get a page and your apache is installed and working properly

6. Prepare the Database for FreePBX

```
mysqladmin create asterisk
mysqladmin create asteriskcdrdb
mysql asterisk < SQL/newinstall.sql
mysql asteriskcdrdb < SQL/cdr_mysql_table.sql
mysql

GRANT ALL PRIVILEGES ON asterisk.* TO asteriskuser@localhost IDENTIFIED BY
'amp109';
GRANT ALL PRIVILEGES ON asteriskcdrdb.* TO asteriskuser@localhost IDENTIFIED BY
'amp109';
flush privileges;
quit
```

Note: Before installing Freepbx make sure you copy this particular file because FreePBX installation will overwrite it

```
cp /etc/asterisk/modules.conf ~/asterisk-modules.conf
```

7. Install FreePBX

```
./install_amp
```

If all the FreePBX requirements are satisfied then installation will run smoothly and you will be asked certain details to be entered. If not then make sure you have all of these installed on your system.

```
apt-get install asterisk asterisk-mysql asterisk-mp3 php-db php5-gd php-pear sox curl
```

Just use the default values for everything by pressing an enter for every question asked don't change anything except ampwebroot because these are freepbx configuration requirements and they have to be same as default for the beginners. Advanced settings can be done but not covered here.

Out of all the questions you will need to change only this

```
AMPWEBROOT=/var/www/
```

Note: If you have put FreePBX on a subdirectory, the panel will not work under the admin pages.

There are two approaches to make it work

Step 1: Change the file **/var/www/freepbx/admin/views/panel.php**

```
// where it reads
//      '<iframe width="97%" height="600" frameborder="0" align="top"
src="../../panel/index_amp.php?context='.$deptname.'"></iframe>'.
// you should erase one step back in the uri, as it shows here
        '<iframe width="97%" height="600" frameborder="0" align="top"
src="../panel/index_amp.php?context='.$deptname.'"></iframe>'.
```

Go to step 2.

Or

Step 1: create a virtual server and access your freepbx as http://freepbx

If you have access to the dns you can add a virtual server in Apache and avoid this last . Edit the file /etc/apache2/sites-available/freepbx and put this.

```
<VirtualHost *:80>

   ServerName freepbx
   ServerAlias freepbx

   ServerAdmin yourname@yourdomain.com
   ErrorLog /var/log/apache2/freepbx.error.log
   CustomLog /var/log/apache2/freepbx.access.log combined

   DocumentRoot /var/www/freepbx
   <Directory /var/www/freepbx>
      Options Indexes FollowSymLinks MultiViews
      Order allow,deny
      AllowOverride All
```

```
    Allow from all
  </Directory>

  <Directory /var/www/freepbx/admin>
    AuthType Basic
    AuthName "Restricted Area"
    AuthUserFile freepbx-passwd
    Require user admin
  </Directory>


</VirtualHost>
```

Make a symlink to make it available

```
ln -s /etc/apache2/sites-available/freepbx /etc/apache2/sites-enabled/099-freepbx
```

And add a password file with the password you want.

```
htpasswd -c /etc/apache2/freepbx-passwd admin
```

Step 2: Restart Apache

```
/etc/init.d/apache2 restart
```

Restore the backup you've made of your modules.conf

```
cp ~/asterisk-modules.conf /etc/asterisk/modules.conf
```

If you have forgotten to make that backup, disable two libraries that are stopping asterisk to work. To disable a library you can add lines in /etc/asterisk/modules.conf. The autoload directive will load anything in the lib directory (/usr/lib/asterisk/modules) unless you put the line noload in the configuration file, before the global directive.

```
noload =>app_directory.so
noload =>res_adsi.so
```

4.0 Asterisk2Billing

1. Download A2Billing

```
cd /usr/src
mkdir a2billing
cd a2billing
wget -o http://www.asterisk2billing.org/downloads/A2Billing_1.3.4.tar.gz
tar -xzvf a2billing.tar.gz
```

2. Creating Database for A2Billing: Log in to MySQL and paste this

```
GRANT ALL PRIVILEGES ON mya2billing.* TO 'a2billinguser'@'%' IDENTIFIED BY
'a2billing' WITH GRANT OPTION;  //security issue
GRANT ALL PRIVILEGES ON mya2billing.* TO 'a2billinguser'@'localhost' IDENTIFIED BY
'a2billing' WITH GRANT OPTION;
GRANT ALL PRIVILEGES ON mya2billing.* TO 'a2billinguser'@'localhost.localdomain'
IDENTIFIED BY 'a2billing' WITH GRANT OPTION;
\q (to quit)
```

3. Create a2billing database

```
mysqladmin create mya2billing -u a2billinguser –p
```

4. Import data schema into new database

```
cd trunk/Database/mysql/<mysql-version>
mysql mya2billing -u a2billinguser -p < a2billing-mysql-schema-MYSQL.5.X-v1.2.3.sql
<insert a2billing password: "a2billing">

exit
```

5**.** Verify database installation

```
mysql mya2billing -u a2billinguser -p
```

Prompt will ask you for password, in our case enter 'a2billing'.It should the correct details if not try to again dropping the database and repeat the steps for creating the database. Make sure you have correct A2Billing database in MySQL

6. Configure A2Billing with your database.Place the file a2billing.conf into your /etc/asterisk/ directory

```
cp /usr/src/a2billing/trunk/a2billing.conf /etc/asterisk/
```

7. Setup the database access

```
vi /etc/asterisk/a2billing.conf
```

You will have to change the username=a2billing,database=a2billing and password = "yoursecretpassword"

```
[database]
hostname=localhost
port=5432
user=a2billinguser
password=a2billing
dbname=mya2billing
dbtype=mysql
```

8. Install management interface
Place the directory "A2Billing_UI" into your DocumentRoot directory of your web server.

```
cp -rf /usr/src/a2billing/trunk/A2Billing_UI /var/www/html/.
cd /var/www/html/A2Billing_UI
chmod 777 templates_c
```

Document root is the directory out of which you will serve your # documents. By default, all requests are taken from this directory, but # symbolic links and aliases may be used to point to other locations.

DocumentRoot "/var/www/html"

Configure & customize the Interface: a2billing.conf

```
configuration for the Web interface
[webui]

; Path to store the asterisk configuration files
buddyfilepath = /etc/asterisk/

; Email of the admin (not used yet)
email_admin = info@areski.net

; Card lenght
len_cardnumber = 10

; Voucher lenght
len_voucher = 15

;amount of MOH class you have created in musiconhold.conf : acc_1, acc_2... acc_10 class
        etc...
num_musiconhold_class = 10

;MANAGER CONNECTION PARAMETERS
manager_host = localhost
manager_username = myasterisk
manager_secret = mycode

; Allow to display the help section inside the admin interface  (YES - NO)
show_help="YES"

; Parameter of the upload
; PLEASE CHECK ALSO THE VALUE IN YOUR PHP.INI THE LIMIT IF 2MG BY
DEFAULT
my_max_file_size_import = 512000
my_max_file_size = 512000            ; in bytes

; Not used yet, goal is to upload files and use them directly in the IVR
```

```
dir_store_audio = /var/lib/asterisk/sounds/a2billing

;Parameter of the upload
my_max_file_size_audio=3072000 ; in bytes

; the file type extensions allowed to be uploaded such as "gsm, mp3, wav" (separate by ,)
file_ext_allow = gsm, mp3, wav

; the file type extensions allowed to be uploaded for the musiconhold such as "gsm, mp3, wav"
(separate by ,)
file_ext_allow_musiconhold = mp3

; ENABLE THE CDR VIEWER TO LINK ON THE MONITOR FILES (YES - NO)
link_audio_file = "NO"

 ; PATH TO LINK ON THE RECORDED MONITOR FILES
monitor_path = /var/spool/asterisk/monitor
// grant access to apache user on read mode for the directory :>  chmod 755
/var/spool/asterisk/monitor/

; FORMAT OF THE RECORDED MONITOR FILE
monitor_formatfile = gsm

; Display the icon in the invoice
show_icon_invoice = "YES"

; Display the top frame (useful if you want to save space on your little tiny screen )
show_top_frame = "NO"
```

Files / directory right

set writing rights /etc/asterisk/ in order to let the web interface write the sip/iax configuration files

```
chmod 777 /etc/asterisk
```

8. SIP/IAX friends include:Asterisk2Billing is generating is own configuration files for SIP and IAX when you are using the SIP/IAX Friends features.
In sip.conf, add the following line at the end

```
#include additional_a2billing_sip.conf
```

In iax.conf, add the following line at the end

```
#include additional_a2billing_iax.conf
```

9. Configure the manager

Edit the manager configuration files,

```
vi /etc/asterisk/manager.conf
```

Ensure that enabled=yes

```
[general] enabled = yes port = 5038 bindaddr = 0.0.0.0 ;displayconnects = yes
```

10. Install the customer web interface

Place the directory "A2BCustomer_UI" into your Documentroot directory of your web server.

```
cp -rf /usr/src/a2billing/trunk/A2BCustomer_UI /var/www/html/.
cd /var/www/html/A2BCustomer_UI
chmod 777 templates_c
```

11. Install The AGI components : IVR asterisk2billing application

Place the entire content of the directory A2Billing_AGI into your agi-bin directory.

```
cd /usr/src/a2billing/trunk/A2Billing_AGI
cp a2billing.php /var/lib/asterisk/agi-bin/.
cp -rf libs_a2billing /var/lib/asterisk/agi-bin/.
```

Make sure the script is runnable

```
chmod +x /var/lib/asterisk/agi-bin/a2billing.php
```

12. Configure & customize the AGI : a2billing.conf

```
; the debug level
; 0=none, 1=low, 2=normal, 3=all
debug=0

; if we want to manage the answer on the call
answer_call=yes

; Active the logging of the application
; logging is optimized to write all the logs at once :D
logger_enable=YES

; File to log
log_file=/tmp/a2billing.log

; if YES Use Set(LANGUAGE()=fr) instead, for me it didnt work from AGI
; ### if (SETLANGUAGE_DEPRECATE==YES) $myres = $agi->agi_exec("EXEC
```

```
Set('LANGUAGE()=$language')");
setlanguage_deprecate=YES

; play the goodbye message when the user finish
say_goodbye=NO

; enable the menu to choose the language
; press 1 for English, pulsa 2 para el español, Pressez 3 pour Français
play_menulanguage=NO

; force the use of a language, if you dont want to use it leave the option empty
; Values : ES, EN, FR, etc... (according to the audio you have install)
force_language=

; Introduction prompt : to specify an additional prompt to play at the beginning of the application
; parlezplus-intro_013centimes
intro_prompt=

; lenght of the cardnumber (amount of digits)
len_cardnumber=10

; Alias-Card lenght
len_aliasnumber = 15

; Voucher lenght
len_voucher = 15

; this is the minimum amount of credit to use the application
min_credit_2call=0

; if user doesnt have enough credit to call a destination XXX prompt him to enter an other
cardnumber
notenoughcredit_cardnumber=YES

; if notenoughcredit_cardnumber = YES  then assign the CallerID to the new cardnumber
notenoughcredit_assign_newcardnumber_cid=YES

; if YES it will catch the DNID and try to dial it out directly without asking for the phonenumber
to call
; value : YES, NO
use_dnid=NO

; list the dnid on which you want to avoid the use of the previous option "use_dnid"
no_auth_dnid=2400,2300

;number of time the user can dial different number
```

```
number_try=3


; Play the balance to the user after the authentication (values : yes - no)
say_balance_after_auth=YES

; Play the balance to the user after the call (values : yes - no)
say_balance_after_call=NO

; Play the time the user can call (values : yes - no)
say_timetocall=YES

; enable the callerid authentication
; if this option is active the CC system will check the CID of caller
cid_enable=NO

; if the cid doesnt exist you can then ask a cardnumber to the calling party in order to
authenticate the caller
cid_askpincode_ifnot_callerid=YES

; if the callerID, this option will allow the system to add it automatically and create a cardnumber
to hook them up.
cid_auto_create_card=NO

; if the callerID authenticate is on, this option will allow the assign the cardnumber enter to the
callerID if the callerID wasnt in the DB
cid_auto_assign_card_to_cid=YES

; If cid_auto_create_card has been set to YES, the following option will define with which
parameters the card will be create
;
; billing type of the new card
; ( value : POSTPAY or PREPAY)
cid_auto_create_card_typepaid=POSTPAY
; amount of credit of the new card
cid_auto_create_card_credit=0

; if postpay define here the credit limit for the card
cid_auto_create_card_credit_limit=1000

; the tariffgroup to use for the new card (this is the ID that you can find on the admin web
interface)
cid_auto_create_card_tariffgroup=6

; enable the option to call sip/iax friend for free (values : YES - NO)
sip_iax_friends=NO
```

```
; if SIP_IAX_FRIENDS is active, you define a prefix for the dialed phonenumber to call directly
a pstn number
; values : number
sip_iax_pstn_direct_call_prefix=9

; this will enable a prompt to enter your destination number_try
; if number start by sip_iax_pstn_direct_call_prefix we do directly a sip iax call, if not we do a
normal call
sip_iax_pstn_direct_call=NO

; More information about the Dial : http://voip-info.org/wiki-Asterisk+cmd+dial
; 30 :  The timeout parameter is optional. If not specifed, the Dial command will wait
indefinitely, exiting only when the originating channel hangs up, or all the dialed channels return
a busy or error condition. Otherwise it specifies a maximum time, in seconds, that the Dial
command is to wait for a channel to answer.
;   H: Allow the caller to hang up by dialing *
;   r: Generate a ringing tone for the calling party
;   m: Provide Music on Hold to the calling party until the called channel answers.
;  L(x[:y][:z]): Limit the call to 'x' ms, warning when 'y' ms are left, repeated every 'z' ms)
;      %timeout% tag is replaced by the calculated timeout according the credit & destination rate!

dialcommand_param="|30|HL(%timeout%:61000:30000)"

; by default (3600000 = 1HOUR MAX CALL)
dialcommand_param_sipiax_friend="|30|HL(3600000:61000:30000)"

; Define the order to make the outbound call
; YES -> SIP/dialedphonenumber@gateway_ip - NO  SIP/gateway_ip/dialedphonenumber
; Both should work exactly the same but i experimented one case when gateway was supporting
dialedphonenumber@gateway_ip
; So in case of troubles, try it out
switchdialcommand=NO


; When a call find a negative route or a free route is adviced to limite the call duration : amount
in secons
maxtime_tocall_negatif_free_route = 5400



; enable to monitor the call (to record all the conversation)
; value : YES - NO
record_call=NO

; format of the recorded monitor file
```

```
monitor_formatfile=gsm


;base currency define the default currency that you want to use to setup your system (see the file
/etc/asterisk/rates.inc to know the currency code)
base_currency = usd

; Force to play the balance to the caller in a predefined currency, to use the currency set for by
the customer leave this field empty
agi_force_currency =

; CURRENCY SECTION
; Define all the audio (without extension) that you want to play according to currency (use , to
separate, ie "usd:prepaid-dollar,mxn:pesos,eur:Euro,all:credit")
currency_association = usd:prepaid-dollar,mxn:pesos,eur:euro,all:credit

; Please enter here the file you want to play when we prompt the calling party to enter his
destination number
; file_conf_enter_destination = prepaid-enter-number-u-calling-1-or-011
file_conf_enter_destination = prepaid-enter-dest

; Please enter here the file you want to play when we prompt the calling party to choose the
prefered language
; file_conf_enter_menulang = prepaid-menulang
file_conf_enter_menulang = prepaid-menulang2

; the debug shell (ONLY FOR THE DEVELOPERS)
; 0=no, 1=yes
debugshell=0
```

13.Configure extensions to run asterisk2billing into Asterisk.
Edit extension.conf (/etc/asterisk/extensions.conf)

```
[a2billing]
; CallingCard application
exten => _X.,1,Answer
exten => _X.,2,Wait,2
exten => _X.,3,DeadAGI,a2billing.php
exten => _X.,4,Wait,2
exten => _X.,5,Hangup
```

If you just want to be able to start A2Billing from FreePBX , create a new custom trunk with 'Custom Dial String' set to 'A2B/1' and add the following to /etc/asterisk/extensions_custom.conf:

```
[macro-dialout-trunk-predial-hook]
```

```
exten => s,1,GotoIf($["${OUT_${DIAL_TRUNK}:4:4}" = "A2B/"]?custom-freepbx-a2billing,${OUTNUM},1:2)
exten => s,2,MacroExit

[custom-freepbx-a2billing]
exten => _X.,1,DeadAGI(a2billing.php|${OUT_${DIAL_TRUNK}:8})
exten => _X.,n,Hangup()
```

14. Reload Asterisk to apply the changes

```
asterisk -vvr
```

```
CLI>reload
CLI>exit
```