

**SANOG25, Kandy, Sri Lanka**

# Introduction to Asterisk

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# Session Goal

To provide you an basic understanding about Asterisk\*,  
Installation and Basic configuration



# Installing IP Telephone Srv

Before installation we must talk about choosing a IP Telephone Server.

Answer is very simple.

First there are some commercial one. and some free and open source one.

You can choose any of them.

Among those some only do the basic call establishment and some do a lot. some of them can do anything that you can imagine in telephony till now.

one of them is Asterisk\*.

# What is asterisk\*

- ❑ Asterisk, The Open Source PBX. [www.asterisk.org](http://www.asterisk.org)
- ❑ A complete PBX in software
- ❑ Runs on virtually any OS
- ❑ Support for most VoIP protocols
- ❑ Most full-featured PBX features already built in
- ❑ MOH, conferencing, queues, voicemail, IVR...
- ❑ Supports many different hardware telephony cards

# What is asterisk\*

- ❑ There's lots of info all over the place, some of it contrary though
- ❑ [www.voip-info.org](http://www.voip-info.org)
  - ❑ Lots of really good information, lots of plain wrong information too!
  - ❑ Defacto documentation store at this stage
- ❑ [www.asterisk.org](http://www.asterisk.org)
- ❑ [www.digium.org](http://www.digium.org) - hardware cards
- ❑ Asterisk CLI !

# Useful Reading

Asterisk, The Future of Telephony. By Jared Smith, Jim Van Meggelen, Leif Madsen. ISBN: 0-596-00962-3

- ❑ Published under Creative Commons license
- ❑ Can download, or buy a real book from O'Reilly
- ❑ <http://www.asteriskdocs.org/modules/tinycontent/index.php?id=11>

# Asterisk versions.

Started with 1.2.x in 2005.. get matured in 1.4

Versions currently in popular use:

1.6 very stable.

1.8 good and massively used.

10.x released at the end of 2011 with lots of new feature.

11.x released at the end of 2012.

12.x released at the end of 2013 introduction of ARI

13.x released at the end of 2014 yet to explore.

# Installing Asterisk\*

On a typical system, you'll want to download three components:

- Asterisk
- DAHDI
- libpri

Additionally for SS7/R2 you may require libss7 or libopenr2.

for compiling you will require GCC and these system library's

- OpenSSL
- ncurses
- newt
- libxml2
- Kernel headers (for building DAHDI drivers)



# Installing Asterisk\*

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- OpenSSL
- ncurses
- newt
- libxml2
- Kernel headers (for building DAHDI drivers)

# Installing Asterisk\*

First we will install all the Asterisk dependencies.

```
apt-get install linux-headers- `uname -r`  
apt-get install build-essential  
apt-get install libssl-dev  
apt-get install ncurses-dev  
apt-get install libnewt-dev  
apt-get install bison  
apt-get install libxml2-dev  
apt-get install libsqlite3-dev  
apt-get install sqlite3
```

# Installing Asterisk\*

To Avoid communication problem between Asterisk and database we will install our database first and apache, PHP to do some more ..

```
#MYSQL INSTALL  
apt-get install mysql-server  
apt-get install libmysqlclient-dev
```

```
# ODBC CONNECTION  
apt-get install unixODBC-dev  
apt-get install libmyodbc  
apt-get install unixODBC  
apt-get install lame
```

# Installing Asterisk\*

Continue Installing few more dependencies.

```
#MYSQL INSTALL  
apt-get install mysql-server  
apt-get install libmysqlclient-dev
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# ODBC CONNECTION  
apt-get install unixODBC-dev  
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apt-get install unixODBC  
apt-get install lame
```

# Installing Asterisk\*

Continue Installing few more dependencies.

```
#INSTALL PHP  
apt-get install apache2  
apt-get install php5  
apt-get install php5-mysql  
apt-get install phpmyadmin
```

# Installing Asterisk\*

## Download Source:

```
# cd /usr/src/  
# wget http://downloads.asterisk.  
org/pub/telephony/certified-asterisk/certified-  
asterisk-11-current.tar.gz  
# wget http://downloads.asterisk.  
org/pub/telephony/libpri/libpri-1.4-current.tar.gz  
# wget http://downloads.asterisk.org/pub/telephony/dahdi-  
linux-complete/dahdi-linux-complete-current.tar.gz
```

Untar them all.

# Installing Asterisk\*

## Building and Installing LibPRI

```
# cd libpri-1.X.Y  
# make  
# make install
```

Used by many manufacturers of PCI TDM cards  
Safe to compile even if a card is not installed/used

# Installing Asterisk\*

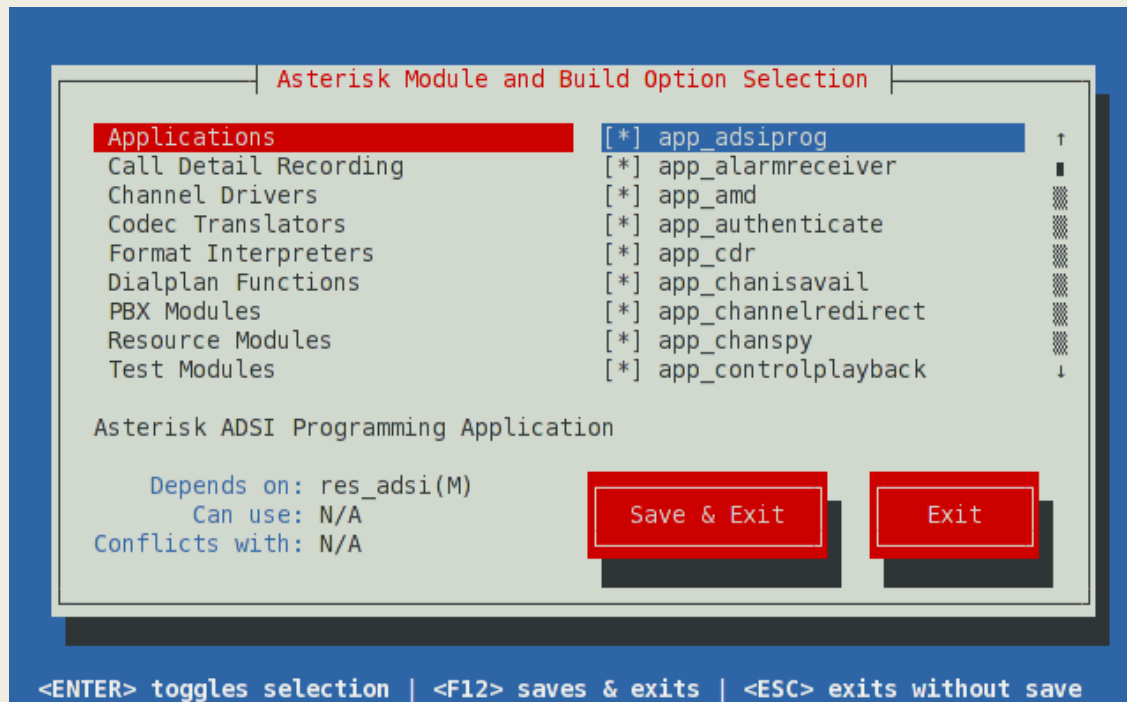
## Building and Installing DAHDI

```
# cd dahdi-linux-complete-2.X.Y+2.X.Y  
# make  
# make install  
# make config
```



# Installing Asterisk\*

```
# cd /usr/local/src/asterisk-1.8.X.Y  
# ./configure  
# make menuselect
```



# Installing Asterisk\*

```
# make  
# make install  
# make samples
```

if you look for easy way just use your package manager to install asterisk in your Linux system.

# Installing Asterisk\*

```
# make  
# make install  
# make samples
```

if you look for easy way just use your package manager to install asterisk in your Linux system.

Done!!!!

# How Asterisk Works

- ❑ Asterisk is a hybrid TDM and packet voice PBX
- ❑ Interfaces any piece of telephony hardware or software to any application
- ❑ Prime components: channels and extensions.conf - the Asterisk dial plan
- ❑ Channels can be many different technologies - SIP, IAX, H323, skinny, Zaptel, and others as they are created
- ❑ extensions.conf is basically a programming language controlling the flow of calls
- ❑ Applications do the work - answer a channel, ring a channel, voicemail, etc.

# Basic Configuration

Here is Asterisk file locations (debian)

- ❑ `/etc/asterisk/` - Asterisk configuration files
- ❑ `/var/lib/asterisk/` - contains the astdb, firmware and keys
- ❑ `/usr/share/asterisk/sounds` - in built asterisk sound prompts
- ❑ `/var/spool/asterisk/` - temporary files and voicemail files
- ❑ `/var/log/asterisk/` - Asterisk log files
- ❑ `/var/log/asterisk/cdr-csv/` - Asterisk call detail records

# Basic Configuration

For Start in asterisk you just need to care about 2 file.

1. `sip.conf` (the core of asterisk SIP protocol, and also manage all the extension and trunks)
2. `extensions.conf` (the main dial plan)

# Basic Configuration

```
/etc/asterisk/sip.conf
```

```
[general]
context=default
port=5060
bindaddr=0.0.0.0
srvlookup=yes
```

```
[1000]
type=friend
host=dynamic
username=1000
secret=secret1000
nat=yes
context=phones
allow=all
```

```
[2000]
type=friend
```

```
.....
```

# Basic Configuration

```
/etc/asterisk/extensions.conf
```

```
[general]  
static=yes  
writeprotect=no
```

```
[globals]
```

```
[default]
```

```
[phones]  
exten => 1000,1,Dial(SIP/1000)  
same => n, Hangup()
```

```
exten => 2000,1,Dial(SIP/1000)  
same => n, Hangup()
```

```
exten => _XXXX,1,Dial(SIP/${EXTEN})  
same => n, Hangup()
```



# Basic Configuration

```
/etc/asterisk/extensions.conf
```

```
[general]  
static=yes  
writeprotect=no
```

```
[globals]
```

```
[default]
```

```
[phones]  
exten => 1000,1,Dial(SIP/1000)  
same => n, Hangup()
```

```
exten => 2000,1,Dial(SIP/1000)  
same => n, Hangup()
```

```
exten => _XXXX,1,Dial(SIP/${EXTEN})  
same => n, Hangup()
```

# Dial Plan - Contexts

extensions.conf split into sections called contexts

[context-name]

contexts isolated from one another - can have the same extension in multiple contexts

Calls from a channel land in the context specified by that channel,  
Calls land in default context if nothing is specified

Be careful with what is in the default context - it is easy to give access to more than is intended

# Dial Plan - Extensions

One or more extensions in each context

An extension is followed by an incoming call or digits dialled on a channel

exten => name,priority,application()

exten => 2000,1,Dial(SIP/2000)

Priorities are numbered and followed sequentially from '1'

Asterisk will stop processing an extension if you skip a priority

Each priority executes one specific application

# Dial Plan - Variables

Three types of variables available in the dial plan

Global

Set in the [globals] section of extensions.conf

Channel

Variables set automatically, and using the set command on a per channel basis

A number of pre-defined channel variables - e.g. \${EXTEN}

Some of the pre-defined channel variables: \${CALLERID} ,  
\${CALLERIDNAME}, \${CALLERIDNUM} , \${CHANNEL} , \${CONTEXT} ,  
\${EXTEN} , \${SIPUSERAGENT}

# Extension Matching

- `exten => _04NXXXXXX,1,SomeApplication()`
- `exten => _.,1,SomeApplication()`
- `_` denotes a pattern matching extension
- `N` matches any number from 2 through 9
- `X` matches any single digit
- `.` matches one or more of any digit
- `[2-6]` matches any of 2,3,4,5,6

# Dialplan Applications

- ❑ Applications 'do things' in the Asterisk dial plan
  - ❑ play a sound
  - ❑ answer a call
  - ❑ interact with a database
- ❑ Can take zero or more arguments • Answer()
  - ❑ Dial(SIP/2001)
  - ❑ AnApplicationWithThreeArguments(arg1,arg2,arg3)
- ❑ Arguments can be separated with a pipe ( | ) or a comma ( , ).

# Starting Asterisk

- ❑ On Debian systems:
  - ❑ `/etc/init.d/asterisk start`
- ❑ Or, `/usr/sbin/asterisk`
  - ❑ `asterisk -c` if you want asterisk to load straight into a console
- ❑ To connect to a running instance of Asterisk:
  - ❑ `asterisk -r`

# Running Asterisk

```
root@sujon-desktop:~# asterisk -h
Asterisk 13.0.1, Copyright (C) 1999 - 2014, Digium, Inc. and others.
Usage: asterisk [OPTIONS]
```

Valid Options:

-V	Display version number and exit
-C <configfile>	Use an alternate configuration file
-G <group>	Run as a group other than the caller
-U <user>	Run as a user other than the caller
-c	Provide console CLI
-d	Enable extra debugging
-f	Do not fork
-F	Always fork
-g	Dump core in case of a crash
-h	This help screen
-i	Initialize crypto keys at startup
-L <load>	Limit the maximum load average before rejecting new calls
-M <value>	Limit the maximum number of calls to the specified value
-m	Mute debugging and console output on the console



# Running Asterisk

-n	Disable console colorization
-p	Run as pseudo-realtime thread
-q	Quiet mode (suppress output)
-r	Connect to Asterisk on this machine
-R	Same as -r, except attempt to reconnect if disconnected
-s <socket>	Connect to Asterisk via socket <socket> (only valid with -r)
-t	Record soundfiles in /var/tmp and move them where they belong after they are done
-T	Display the time in [Mmm dd hh:mm:ss] format for each line of output to the CLI
-v	Increase verbosity (multiple v's = more verbose)
-x <cmd>	Execute command <cmd> (implies -r)
-X	Execute includes by default (allows #exec in asterisk.conf)
-W	Adjust terminal colors to compensate for a light background

# Running Asterisk

```
root@sujon-desktop:~# asterisk -r
Asterisk 13.0.1, Copyright (C) 1999 - 2014, Digium, Inc. and others.
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty' for details.
This is free software, with components licensed under the GNU General Public
License version 2 and other licenses; you are welcome to redistribute it under
certain conditions. Type 'core show license' for details.
=====
Connected to Asterisk 13.0.1 currently running on sujon-desktop (pid = 9986)
sujon-desktop*CLI>
```

# Asterisk CLI

- ❑ Similar to IOS:
  - ❑ sip show peers
  - ❑ reload
  - ❑ ? for help, tab for command autocomplete • sip show ?
- ❑ Restart commands
  - ❑ restart gracefully: Restart Asterisk gracefully
  - ❑ restart now: Restart Asterisk immediately
  - ❑ restart when convenient: Restart Asterisk at empty call volume
  - ❑ reload: Reload configuration
- ❑ stop gracefully: Gracefully shut down Asterisk
- ❑ stop now: Shut down Asterisk immediately
- ❑ stop when convenient: Shut down Asterisk at empty call volume

# Asterisk CLI

- ❑ sip debug: Enable SIP debugging
- ❑ sip no debug: Disable SIP debugging • sip reload: Reload sip.conf
- ❑ SIP Show commands
  - ❑ sip show channels: Show active SIP channels
  - ❑ sip show channel: Show detailed SIP channel info
  - ❑ sip show inuse: List all inuse/limit
  - ❑ sip show peers: Show defined SIP peers (clients that register to your Asterisk server)
  - ❑ sip show registry: Show SIP registration status (when Asterisk registers as a client to a SIP Proxy)
  - ❑ sip show users: Show defined SIP users

# LAB

Initial Asterisk Installation...

# Asterisk Variables

- ❑ Why use variables?
  - ❑ Pattern match - how do we know what extensions was dialled?  
    `exten => 2000,1,Dial(SIP/2000)`  
    `exten => 2001,1,Dial(SIP/2001)`  
    `exten => 2002,1,Dial(SIP/2002)`
  - ❑ OR  
    `exten => _200X,1,Dial(SIP/${EXTEN})`
  - ❑ `${some_variable}` = the value of `some_variable`.
  - ❑ `some_variable` = the variable itself

# Asterisk Variables

- ❑ Set default variables
  - [globals]  
default\_ring\_time=10
  
  - [context]  
exten => 2000,1,Dial(SIP/2000,\${default\_ring\_time})
- ❑ Now only one place in dial plan to update if it is changed
- ❑ Setting variables:
  - ❑ exten => s,1,Set(a\_variable=2000)

# Standard Ast Extensions

- ❑ i : Invalid
- ❑ s : Start
- ❑ h : Hangup
- ❑ t : Timeout
- ❑ T : AbsoluteTimeout
- ❑ o : Operator



# Dial Application

- ❑ `Dial(tech/username:password@hostname/extension,ring-timeout,flag)`
- ❑ Can include complete information in the dial string, or reference a peer in `sip.conf`
  - ❑ `exten => 2000,1,Dial(SIP/passwd:sipdevice@host.tld/${EXTEN})`  
or
  - ❑ `exten => 2000,1,Dial(SIP/sipdevice)`  
where there is a channel `[sipdevice]` defined in `sip.conf` containing at least definitions for username, password and host.

# Voicemail

- ❑ Comedian Mail - a fully functional voicemail system included with Asterisk
  - ❑ Supports busy and unavailable messages
  - ❑ exten => 2001,1,VoiceMail(b2001)
    - ❑ exten => 2001,1,VoiceMail(u2001)
- ❑ Voicemail can be emailed out a .wav attachment to users • Standard IVR voicemail access
  - ❑ exten => 510,1,VoiceMailMain

# voicemail.conf

```
[general]
format=wav49|gsm|wav
serveremail=voicemail@yourdomain.net
mailcmd=/usr/sbin/sendmail -t
attach=yes
maxmsg=100
maxmessage=180
skipms=3000
maxsilence=10
silencethreshold=128
maxlogins=3
emailbody=Dear ${VM_NAME}:\n\n\tjust wanted to let you know you were just left a
    ${VM_DUR} long message (number ${VM_MSGNUM})\nin mailbox ${VM_MAILBOX} from ${V
M_CALLERID}, on ${VM_DATE}, so you might\nwant to check it when you get a chance
.  Thanks!\n\n\t\t\t--Asterisk\n
emaildateformat=%A, %B %d, %Y at %r
[default]
; all our mailboxes here
; mailbox number => pin,name,email
2000 => 1234,yourname,yourname@yourdomain.net
```



# LAB

Basic Asterisk Configuration .... .

I already told you asterisk is the wildcard of telephony you can fit it in any place.

with lots of modules and apps inside asterisk can do anything you can imagine with telephony.

including voicemail, IVR, queue, etc. etc. etc.

# Explore More

To know about asterisk more you have to explore it with time.  
there is lots of things inside that magical folder /etc/asterisk.

but to know more about IP Telephony there is lot other product you can take a look at. no matter what you use everything will give you some new idea to play with.

Any Question ???????