

Overview of Asterisk ()*

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Agenda

- Background
- Introduction to Asterisk and review the core components of it's architecture.
- Exploration of Asterisk's telephony and call features.
- Review some of Asterisk's configuration files.

What is Asterisk?

- Asterisk is an Open Source PBX (Private Branch Exchange) and IVR (Interactive Voice Response) system.
- Distributed under the GNU GPL with commercial licenses available.
- Written in C and runs under Linux > 2.4.x
- Support for OpenBSD 3.3 in progress.

What is Asterisk? - Continued

- Asterisk supports traditional circuits, including:
 - TDM (Time Division Multiplexing)
 - T1/E1 PRI/PRA & RBS (Robbed Bit Signal) modes
 - Analog phone lines/phones (POTS)
 - ISDN (Integrated Services Digital Network)
 - Both BRI (Basic Rate) and PRI (Primary Rate)
- Asterisk provides transparent bridging between Voice over IP protocols, including:
 - Session Initiation Protocol (SIP)
 - H.323 (ITU standard, contributed support)
 - Inter-Asterisk eXchange (IAX)
 - Media Gateway Control Protocol (MGCP)
 - SCCP (a.k.a Skinny)

What is Asterisk? - Continued

- Asterisk is an Interactive Voice Response (IVR) platform:
 - Hardware independent API
 - Powerful C-level API
 - Asterisk Gateway Interface (AGI) similar to CGI

Telephony Features

- Voicemail System
 - Password Protected
 - Separate Away and Unavailable Messages
 - Default or Custom Messages
 - Multiple Mail Folders
 - E-mail notification of Voicemail
 - Voicemail Forwarding
 - Visual Message Waiting Indicator
 - Message Waiting Stutter Dial tone
- Auto Attendant
- Interactive Voice Response
- Overhead Paging

Telephony Features - Continued

- Flexible Extension Logic
- Multiple Line Extensions
- Directory Listing
- Conference Bridging
- Unlimited Conference Rooms
- Access Control
- Call Queuing
- Visual Notification of Voicemail
- Call Detail Records
- Protocol Bridging

Call Features

- Music on Hold and Transfer
- Flexible MP3 Music Playback
- Volume Control
- Random and Linear Play
- Call Waiting
- Caller ID including Blocking and Call Waiting
- Call Forward on Busy and No Answer
- Call Transfer
- Call Parking
- Call Retrieval
- Do Not Disturb

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Architecture

- Asterisk is carefully designed for maximum flexibility.
- Specific API's are defined around a central PBX core system.
- The core handles the internal interconnection of the PBX.
- Everything is cleanly abstracted from the specific protocols, codecs, and hardware interfaces used by its internal telephony applications.

Architecture - Continued

- This abstraction allows Asterisk to use any suitable hardware and technology available now or in the future to perform.
- Asterisk internally handles the following:
 - PBX Switching
 - Application Launching
 - Codec Translating
 - Scheduling and I/O Management

Supported Hardware

- Zaptel Compatible Hardware from Digium:
 - Wildcard TDM400P - A Quad-Port FXS PCI interface card for interfacing with a standard analog telephone.
 - Wildcard X100P - A single port FXO PCI interface card for interfacing with a standard analog phone line.
 - Wildcard TE410P/ TE405P - A Quad-Span T1/E1/PRI half-length PCI card in T1 or E1 format.
- Variety of Dialogic Hardware

What is the Value Proposition?

- Huh....runs on Linux. Enough said?
- Asterisk is the Apache of the telecommunications world.
- Similar to Linux, Asterisk gives YOU complete control over your telephone system.
- Freedom from vendor lock-in.
- Extremely low acquisition costs (under \$200)
- Major reduction of ongoing operating costs with no per-seat licensing.
- Integrates with various types of technologies and telephones including Cisco products.

Configuration Files

- Asterisk's files are located in “/etc/asterisk”. Basic configuration files include:
 - extensions.conf - Static dialplan
 - iax.conf - IAX users, peers, friends, and parameters
 - logger.conf - Log files, log levels, etc.
 - meetme.conf - Meetme conference configuration
 - modules.conf - Modules, preloads, globals, and noloads
 - musiconhold.conf - Music on hold configuration
 - voicemail.conf - Voicemail mailboxes, general parameters
 - zapata.conf - Zaptel compatible device configuration

The Dialplan

- Asterisk is centered around the dialplan.
- Responsible for directing and routing all calls of the calls through the PBX
- Each step in the Dial Plan is an Application
- Each step is assigned a priority sequence
- Composed of extension contexts
- Contexts are groups of extensions
- Contexts can include one another

The Dialplan - Extensions

- Conventional dialplans assign extension to physical interface.
- Asterisk assigns extensions to list of applications
- Execution goes through priorities, one by one
- Applications can modify call flow
- Seamlessly integrates IVR apps into PBX
- May route by dialed and calling number

The Dialplan - Extension Names

- Exten => Name,Priority,Application(,arguments)
 - May be any number of digits (max 128)
 - May be string literals
 - May pattern match (when proceeded with '_')
 - 'N' – matches digits from 2 to 9
 - 'X' – matches digits from 0 to 9
 - '.' – signifies end of pattern matching

Special Extensions

- 's' the start extension, where calls begin when no digits are available
- 't' the timeout extension, where calls begin when not enough digits are entered
- 'i' the invalid extension, when an invalid extension is entered
- 'o' the operator extension
- 'h' the hangup extension, when channel is disconnected.

Office Extension

exten => s,1,Answer

exten => s,2,Wait,1

exten => s,3,DigitTimeout,3

exten => s,4,ResponseTimeout,5

exten => s,5,BackGround(some-greeting)

exten => 0,1,Playback(transfer)

exten => 0,2,Dial,SIP/100|20

exten => 0,3,Voicemail,u5

Anti-In-Law Extension

exten => s/4345551212,1,Busy

exten => s,2,Dial,SIP/100|20

exten => s,3,VoiceMail,u5

voicemail.conf

Each mailbox is listed in the form:

<mailbox> = <password>, <name>, <email>, <pager_email>, <options>

100 => 1234,Jeff

Gunther,jeff.gunther@intalgent.com,jeff.gunther.mobile@intalgent.com

sip.conf

[100]

type=friend ; This device takes and makes calls

username=100 ; Username on device

secret=100 ; Password for device

host=dynamic

context=internal ; Inbound calls from this host go here

dtmfmode=rfc2833

mailbox=100 ; Activate the message waiting light if this

; voicemailbox has messages in it

zapata.conf

```
signalling=fxs_ks  
group=1  
context=incoming  
channel => 1
```

```
signalling=fxs_ks  
group=1  
context=incoming  
channel => 2
```

Resources

- **Asterisk Homepage**
<http://www.asterisk.org>
- **Digium**
<http://www.asterisk.org>
- **Asterisk WIKI**
<http://www.voip-info.org/wiki-Asterisk>

Contact

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Questions?