MATTS FAVORITE LINUX “KILLER APP”

DIGIUM ASTERISK

HTTPS://WWW.SANGOMA.COM/ASTERISK/SOFTWARE/

WHO AM I: MATT MACKES

First Computer :
Mattel Aquarius

First Usable Computer with Storage:
Commodore VIC20 (with Tape Drive)

First Linux Distro:
Red Hat 5.1 (1998)

First Microsoft Platform:
DOS 6.2 with DoubleSpace

First Baud Rate:
9600 bits per second (Buffalo FreeNet)

WHAT IS ASTERISK:

- A free (GPL) private branch exchange for handling both IP and traditional phone calls
- It’s a SIP Proxy
- It’s a SIP Client
- It’s a SIP register
- Asterisk can bridge streams (calls) between several different protocols
- Asterisk can transcode
- Asterisk is a multimedia gateway
- More..

AT ITS HEART,

- At its heart, Asterisk is a single Binary, running on Linux
  With a CLI
- The Binary loads modules at start
- The Binary loads its configuration at start
- The Binary runs as a service
- Several other services like TFTP, FTP, DHCP, and OS services enhance your PBX,
- All of these services can be hosted on one machine or several, on VM’s, in the Cloud, and the Linux flavor is up to the engineer.

```
root@voip:~
root@voip:~ export ASTERISK_PROMPT="%C31[%H]: "
root@voip:~ asterisk -r
Asterisk 16.2.1~dfsg-1+deb10u1, Copyright (C) 1999 - 2018, 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 16.2.1~dfsg-1+deb10u1 currently running on voip (pid = 25032)
Core debug is still 3.
[1;31m[voip]: [1;0m
  == Using SIP RTP CoS mark 5
    -- Executing [8111@fullaccess:1] Answer("SIP/8882-00000000", "") in new stack
    -- Executing [8111@fullaccess:2] Verbose("SIP/8882-00000000", "0,INTERNAL CALL:
INTERNAL CALL:                     is calling 8111 (ECHO TEST)
    -- Executing [8111@fullaccess:3] Set("SIP/8882-00000000", "CDR(userfield)=            ") in new
    -- Executing [8111@fullaccess:4] Playback("SIP/8882-00000000", "/usr/local/share/asterisk/sound
    -- <SIP/8882-00000000> Playing '/usr/local/share/asterisk/sounds/echotest.slin' (language 'en')
[1;31m[voip]: [1;0m
[1;31m[voip]: [1;0m
[1;31m[voip]: [1;0m
```

BUT IT DOES *SO* MUCH MORE

- POTS: Plain Old Telephone Service FXO (Central Office) FXS (Station – Old School Phone) termination
- • Video
- • Voice and Video Conferencing
- • XMPP integration
- • Voicemail
- • IVR – Interactive Voice Response, menu-driven applications
- • Database integration
- • Google Voice
- • The possibilities are endless

PRODUCTS BUILT WITH ASTERISK AT ITS CORE

(Most of below are CentOS or Debian Linux Distros with Asterisk, Scripting, MySQL and a GUI prebaked)

Most offer some level of support if purchased

- FreePBX
- Digium SwitchVox
- Elastix
- Trixbox
- PBX on a Stick (Nerd Vittles)

THE HISTORY OF ASTERISK

Mark Spencer (creator of Asterisk)

● Established a company called "Linux Support Services"(LSS) to help Linux os user.

● LSS offered a support hotline that IT professionals could (for a fee) call to get help with Linux.

● Within a few more months the growth of the business expanded demanded a "real" phone system.

● Asked for quotes.. responses all came back well above $50,000 -- far more than Mark had budgeted for the project.

THE HISTORY OF ASTERISK

He started to **code(){...}**

Problem...

1. No knowledge of Communication and Networking.

In only a few months Mark crafted the original Asterisk core code under the GPL license.

In 2001, Linux Support Services changed its name to Digium.

DEFINE SOME TERMS: VOIP

VoIP stands for Voice over Internet Protocol.

VoIP - not a protocol

- - encompasses a group of protocol technologies.
- some examples :
- SIP, IAX2,Skype protocol,Remote Voice Protocol (RVP)

SIP : SESSION INITIATION PROTOCOL

- The Session Initiation Protocol (SIP) is a signaling protocol used for initiating, maintaining, and terminating real-time sessions that include voice, video and messaging applications.[1] SIP is used for signaling and controlling multimedia communication sessions in applications of Internet telephony for voice and video calls, in private IP telephone systems, in instant messaging over Internet Protocol (IP) networks as well as mobile phone calling over LTE (VoLTE).

IAX: THE INTER-ASTERISK EXCHANGE PROTOCOL (TRUNKING)

IAX, the Inter-Asterisk eXchange protocol, an application-layer control and media protocol for creating, modifying, and terminating multimedia sessions over Internet Protocol (IP) networks.

IAX was developed by the open source community for theAsterisk Private Branch Exchange (PBX) and is targeted primarily at Voice over Internet Protocol (VoIP) call control, but it can be use with streaming video or any other type of multimedia.

IAX is an "all in one" protocol for handling multimedia in IP networks. It combines both control and media services in the same protocol. In addition, IAX uses a single UDP data stream on a static port greatly simplifying Network Address Translation (NAT) gateway traversal, eliminating the need for other protocols to work around NAT, and simplifying network and firewall management.

IAX employs a compact encoding that decreases bandwidth usage and is well suited for Internet telephony service.

In addition, its open nature permits new payload type additions needed to support additional services.

THE DREAM OF FREE-AS-BEER (VOIP)
DUNDI AIMS TO OPEN VOIP PEERING

- Asterisk sharing Voice Services: Peer to Peer for VoIP
- **Distributed Universal Number Discovery** (**DUNDi**) is a VoIP routing protocol that provides directory services for Asterisk systems. With DUNDi peered nodes share dialplan information with each other. The protocol does not actually carry any calls, but rather provides addressing information.
- Peers in a DUNDi cluster query other peers for a telephone number to which a call is requested by a user. The result of the query is a dial string for the Asterisk application *Dial*.

- Simplified: If an Asterisk Admin/Owner is willing to share his/her system with the community, a worldwide VoIP network could be created, bypassing toll calling, sharing resources and interconnecting the world for free both to VoIP endpoints, and to Plain-Old-Telephone-Service (The latter is a bit troublesome, legally)

DEEPER DIVE, WHAT IS ASTERISK?

Asterisk Architecture

Asterisk bridges calls between channels (which are a type of module)

Modules • Channels: SIP, H.323, DAHDI (used for traditional telephony)

Others: XMPP, dialplan applications/functions, codecs, CDRs, etc.

Configure module related functions in their related configuration files

Ex: SIP phones/endpoints are configured in /etc/asterisk/sip.conf

The dialplan - /etc/asterisk/extensions.conf

Heart of the Asterisk system

- Bridges calls between the various modules
- Uses a scripting language to tell the system how to handle calls

GIVE ME THE FIREHOSE

- Call Features
- ADSI On-Screen Menu System
- Alarm Receiver
- Append Message
- Authentication
- Automated Attendant
- Blacklists
- Blind Transfer
- Call Detail Records
- Call Forward on Busy
- Call Forward on No Answer
- Call Forward Variable
- Call Monitoring
- Call Parking
- Call Queuing
- Call Recording
- Call Retrieval
- Call Routing (DID & ANI)
- Call Snooping
- Call Transfer
- Call Waiting
- Caller ID
- Caller ID Blocking
- Caller ID on Call Waiting
- Calling Cards
- Conference Bridging
- Database Store / Retrieve
- Database Integration
- Dial by Name
- Direct Inward System Access
- Distinctive Ring
- Distributed Universal Number Discovery (DUNDi™)
- Do Not Disturb

GIVE ME THE FIREHOSE

- E911
- ENUM
- Fax Transmit and Receive
- Flexible Extension Logic
- Interactive Directory Listing
- Interactive Voice Response (IVR)
- Local and Remote Call Agents
- Macros
- Music On Hold
- Music On Transfer:

- Flexible Mp3-based System
- Random or Linear Play
- Volume Control
- Privacy
- Open Settlement Protocol (OSP)
- Overhead Paging
- Protocol Conversion
- Remote Call Pickup
- Remote Office Support
- Roaming Extensions
- Route by Caller ID

- Call Features
- SMS Messaging
- Spell / Say
- Streaming Hold Music
- Supervised Transfer
- Talk Detection
- Text-to-Speech (via Festival)
- Three-way Calling
- Time and Date
- Transcoding
- Trunking
- VoIP Gateways

GIVE ME THE FIREHOSE

Voicemail:

- Visual Indicator for Message Waiting
- Stutter Dialtone for Message Waiting
- Voicemail to email
- Voicemail Groups
- Web Voicemail Interface
- Zapateller
- Computer-Telephony Integration
- Asterisk Gateway Interface (AGI)
- Asterisk Manager Interface (AMI)
- Asterisk REST Interface (ARI)
- Outbound Call Spooling

Scalability

- TDMoE (Time Division Multiplex over Ethernet)
- Allows direct connection of Asterisk PBX
- Zero latency
- Uses commodity Ethernet hardware
- Voice-over IP
- Allows for integration of physically separate installations
- Uses commonly deployed data connections
- Allows a unified dialplan across multiple offices

Speech

- Cepstral TTS
- Lumenvox ASR

GIVE ME THE FIREHOSE

Codecs

- ADPCM
- CELT (pass through)
- G.711 (A-Law & μ-Law)
- G.719 (pass through)
- G.722
- G.722.1 licensed from Polycom®
- G.722.1 Annex C licensed from Polycom®
- G.723.1 (pass through)
- G.726
- G.729a
- GSM
- iLBC
- Linear
- LPC-10
- Speex
- SILK

- VoIP Protocols
- Google Talk
- H.323
- IAX™ (Inter-Asterisk eXchange)
- Jingle/XMPP
- MGCP (Media Gateway Control Protocol
- SCCP (Cisco® Skinny®)
- SIP (Session Initiation Protocol)
- UNIStim

- Traditional Telephony Protocols
- E&M
- E&M Wink
- Feature Group D
- FXS
- FXO
- GR-303
- Loopstart
- Groundstart
- Kewlstart
- MF and DTMF support
- Robbed-bit Signaling (RBS) Types
- MFC-R2 (Not supported. However, a patch is available)

GIVE ME THE FIREHOSE

- ISDN Protocols
- AT&T 4ESS
- EuroISDN PRI and BRI
- Lucent 5ESS
- National ISDN 1
- National ISDN 2
- NFAS
- Nortel DMS100
- Q.SIG

FLAT FILE, DATABASE, REALTIME (AND OF COURSE- YOU CAN SCALE)

Example Database Support for Configuration:

- PostgreSQL
- MySQL
- ODBC-ready

Asterisk can run in “Realtime” loading each aspect of configuration “Just in Time”

And of course, those sweet, sweet flat files with CLI, or API “core reload config” or “Module Reload Config” commands.

WHERE ARE THE BASIC CONFIG FILES

File	Description
/etc/asterisk	This is the base directory that contains the other configuration files. If you run the make samples command during installation (I'll cover installation in the next article), then this directory will be populated with sample configuration files for many components of Asterisk.
/etc/asterisk/asterisk.conf	This file contains the base Asterisk configuration. Notably, it contains the directories that Asterisk will use for certain functions, such as sounds and logs.

WHERE ARE THE BASIC CONFIG FILES

/etc/asterisk/modules.conf	This file tells Asterisk which modules to load. Remember that Asterisk is modular, and you can turn off modules that you aren’t using. By default, all modules in /usr/lib/asterisk/modules are loaded (via the autoload=yes directive).
/etc/asterisk/pjsip.conf	This file contains configuration for the PJSIP channel driver. You will configure phone endpoints in this file during future articles.
/etc/asterisk/extensions.conf	This file is the dialplan that you learned about earlier. Asterisk considers an "extension" to be a collection of dialplan instructions. Extensions aren’t tied to physical phones. They’re just a place in the dialplan that contains code that you want to execute.

COOL STUFF ASTERISK CAN DO

Connect to enterprise directories

Asterisk can authenticate users against your LDAP server, Lotus Domino Directory, Apple OpenDirectory, or even Microsoft ActiveDirectory.

Connect using Google Voice and GTalk

Use Google's services with Asterisk to send and receive free calls with other Google users and plain-old telephones, too.

Calendar integration

Have your Asterisk server call you to remind you of your next meeting, or even automatically join you with your next conference call right on time. The Asterisk calendaring API is compatible with iCalendar, CalDAV, and Exchange.

Distributed device state with Jabber/XMPP

The popular instant-messaging protocol to share device state across multiple systems and sites. All your branch offices in one busy lamp field!

PITCH_SHIFT and JACK

On the silly side, change the pitch of a call with the PITCH_SHIFT function, or reroute call audio through wacky signal processors using JACK.

Voicemail-email integration

Asterisk has long been able to email you voicemail messages as attachments, but did you know it can natively store your voicemail messages in IMAP? Unify your email and voicemail, no charge.

COOL STUFF ASTERISK CAN DO

Speak!

- Asterisk offers native integration to leading text-to-speech (TTS) and automatic speech recognition (ASR) software. Let Asterisk speak to you, and understand your answer!

Database integration

- Without special licenses or extra layers, Asterisk can use an ODBC-compatible database to authenticate users and to store call logs. Make user management and reporting a breeze.

Queue games with Asterisk

- Some unconventional Asterisk administrators changed the behavior of their call queues. Now instead waiting your turn while listening to plastic versions of yesterday's hits, callers interact with a real-time quiz game. Miss one and you could be knocked to the back of the line. Answer correctly and move ahead of the people ahead of you. Or do something less challenging, such as allow queue callers to hang up and be called back when it's their turn.

Super-wideband audio support

Asterisk has always been able to use a variety of audio encodings, but the latest enhancements go far beyond the typical telephone sound. Version 10 supports CD-quality audio (16-bit, 44.1kHz) all the way up to even the highest-quality Blu-ray and DTS-HD audio rates at 192kHz. High-fidelity phone calls!

- End.