Introduction to Asterisk

Or: How to spend 2 months on the phone

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Agenda

- What is Asterisk?

- What is Asterisk NOT?

- What do you want to do? (goals, budget, user requirements)

- PBX Replacement

- Super-Brief Examples

What is Asterisk?

- a conversion gateway for...

    - physical media (C-T1, PRI, FXO, FSX, IP)

    - protocol (TDM,SIP,H.323,IAX,MGCP,SCCP)

    - codec (G.729,G.711,GSM,ILBC,G.726, etc.)

- an IVR/user interface application server

- a lot more (conferencing, recording, etc.)

What is Asterisk? (cont'd)

- open-source (GPL + exceptions)

- blessed (cursed?) with an extremely active user community

- easily extended with Perl/C/Python/etc. or apps written (typically C)

- flexible enough to do almost any telecommunications task (blessing/curse again)

What is Asterisk not?

- not a SIP proxy (subtle, yet important)

- not a billing system

- not an OSS (Operational Support System)

- not a natively database-driven system

- not an email tool or USENET browser (yet)

- not easily configured without command-line interaction

Asterisk

- When: 1999

- Who : Mark Spencer

- Why : "I needed a phone system and with as small a startup budget as I had for Linux Support Services, I wasn't about to buy one, so building one seemed a logical way to go."

Asterisk

- What Is ?

- Officially, Asterisk is an Open Source hybrid TDM and packet voice PBX and IVR platform with ACD functionality. Unofficially, Asterisk is quite possibly the most powerful, flexible, and extensible piece of integrated telecommunications software available.

- Its name comes from the asterisk symbol, *, which represents a wildcard, matching any filename.

- Similarly, Asterisk the PBX is designed to interface any piece of telephony hardware or software with any telephony application, seamlessly and consistently.

Facts

- An Open Source Telephony Swiss Army Knife

- A Linux Based PBX w/ Minimal Hardware Reqs

- A Community Driven Development Project

- A Really, Really Disruptive Technology

- Asterisk is any call, any time, from anywhere to anywhere else

- Licensing Model

- Released and developed under GPL, but Digium retains rights to code-base

- All developers submit disclaimers to their code before patches are accepted, allowing for Digium to license specific branches for Commercial projects

- This dual-licensing allows companies to purchase license rights to snapshots of the Asterisk codebase to be used in commercial, non-gpl products

Who is Digium

- **From Wikipedia, the free encyclopedia.**

- Digium is the primary developer and sponsor of Asterisk™, The Open Source PBX. Digium offers a variety of specially designed low and high density telephony hardware and professional services related to Asterisk. The company is based in Huntsville, Alabama. Digium sells telephony hardware and provides contract services for operating IP based telephony solutions.

The Asterisk Development Model

LEON OLEAN
STAMFORD, CONN.

Which is remarkably similar to……

The Linux Development Model

PBX Replacement!

- Primary stated goal is to be a *NIX based PBX replacement

- Multiple desksets, multiple "inbound line" support (hundreds or thousands)

- Features are comparable to or better than most PBX systems (even VoIP-enabled ones); some assembly required

What do you need to run Asterisk?

- Ugly answer: "That depends."

- Easy answer: Dedicated P4 2.0ghz with good IRQ support and 1 X100P card (from Digium at around $110)

- Linux (RH 9.0, Debian are good choices; *BSD support is there, but shaky)

- Low-jitter, low-loss bandwidth to SIP endpoints (desktops and/or upstreams)

How big?

- MORE ugly answers: "That depends."

- If the server is just a SIP redirector, then you can scale quite large (tens of thousands?)

- Figure 8:1 to 10:1 ratio for offhook users

- Word of the day: Erlangs

- Rule of thumb for g.729 transcoding:
  2x Xeon 3ghz = 100 users

Typical VoIP Installation Cost Points

- Server for Asterisk (plus backup, if you're sane) - $???

- T1 PRI card for Asterisk (~$500)

- SIP devices for desktop users (ranges widely - figure $120 per user to be safe, for analog lines)

- Termination agreement with carrier(s) - ranges widely - figure $.025 for US traffic, worst-case (prices drop radically with volume)

CPE

- Analog adapters (VOIP Inc., Sipura, Cisco, Grandstream, etc.)

  - Typically between $80 and $120 (2 port)

- Digital Handsets (Cisco, Polycom, Snom, Pingtel, Grandstream)

  - Typically around $300 (YGWYPF)

Why are you changing, anyway?

- Implement based on price, expand based on features.

- Long Distance will soon become a commodity (i.e.: invisible) but features of the system will always be visible to users

- Integration of telephony into other business systems is gradual and subtle; start with something that is open so you can expand as you need.

What new stuff are you providing?

- FEATURES! Don't get hung up on building just a "replacement" service. Implement phone++ services which are "easily" implemented with Asterisk (given time, patience, and Perl)

- Sample of services: phone spam blocking, inbound call redirection based on CLID, time-of-day routing, IM integration of VM notices, VM-to-email, busy line redirection, multi-number custom ringers
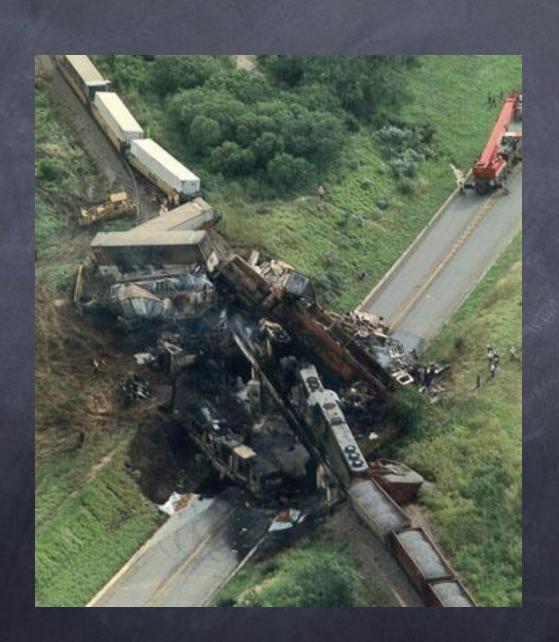
What do they see?

- Remember: the visibility of the customer is very limited. They see:

    - Deskset (equipment) and features

    - Call Quality/Call completion

    - Price (if they're the CFO)

The Asterisk Development Model

- Similar to Linux

- Mark Spencer == Linus Torvalds

- Core developers with CVS commit rights

- 1.0 – EOL (Serious Bug Fixes Only)

- 1.2 – Trunk managed by Drumkilla (Russel Bryant)

- Digium employs a handful of full-time   developers
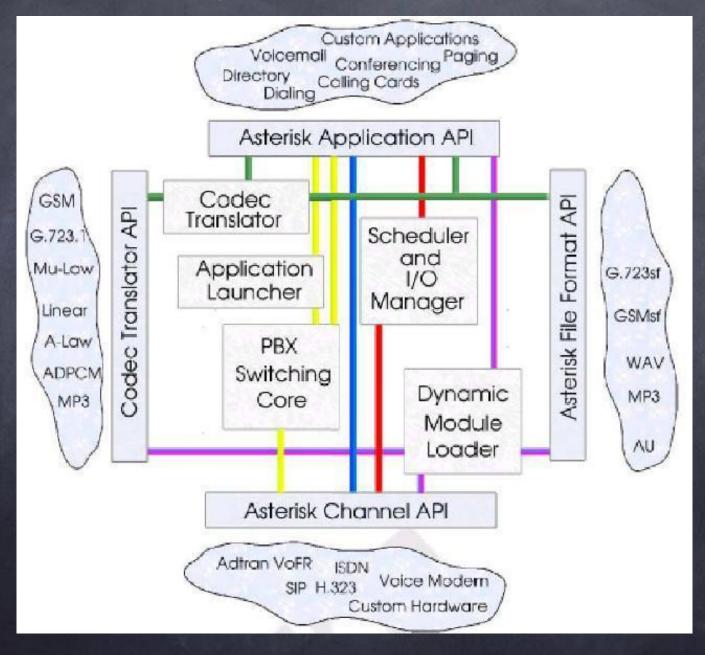   to just work on the code

- Community Supported

High Level Overview of a Developer's Conference

Meet the Developers

Under The Hood

Under The Hood

- Modular architecture like Linux kernel or Apache

- Console Interface for debugging / status

- Most components can be loaded and unloaded from the CLI

- Configuration of system is flexible;

  – Traditionally using Text Files (/etc/asterisk/ directory)

The Channel API

- Channel API Interfaces w/ Hard/Software

- Zap – Zaptel Channel Driver

    - Digium TDM Cards

    - IAX2 – InterAsterisk eXchange Protocol Version 2

    - Extremely efficient, very simple, voice optimized protocol

    - Can transport up to 3x as many calls per Megabit than SIP

    - SIP – Strives to maintain RFC 3261 compatibility

    - Communicates with SIP Gateways / Phones

    - H323 – Based on OpenH323

    - Communicates with H323 Gateways / Phones

The Channel API

- Channel API Interfaces w/ Hard/Software

  - MGCP – Media Gateway Control Protocol

    - Communicates with MGCP Gateways / Phones

  - SCCP – Cisco Proprietary Skinny Control Protocol

    - Communicates with Cisco SCCP Equipment

  - OSS – Open Sound System

    - Older Linux Sound Drivers

    - Communicates with Soundcards

  - ALSA – Advanced Linux Sound Architecture

The Codec Translation API

- Codec Translation API Converts Audio Codecs

  - G.711 Ulaw/Alaw

    - Ulaw is used in the states, Alaw in Europe

  - G.726 32Kbps

  - G.729

    - Requires a license ($10 / channel from Digium)

    - Most widely deployed, low bandwidth codec (8kbps)

  - GSM

The File Format API

- This API Allows Reading/Writing of Various File Formats

    – Some applications may need to archive digital audio streams in different formats

    – Used by many applications such as VoiceMail, which records messages to disk in whatever format you choose

    – Available Formats

        - WAV

        - MP3

        - AU

        - GSM

The Application API

- Applications Perform Functions

  - Modules of code that are used by the Dial Plan

  - For Example:

    - Answer: Answer a channel if ringing

    - BackGround: Play a file while awaiting DTMF tones

    - Busy: Indicate busy condition (normal busy)

    - Congestion: Indicate congestion (fast busy)

The Application API

- Simple Dial Plan Example

  – Dial 216-920-3111 w/ your Cell Phone to hear it live

```
; CallerID Identify

exten => 2169203111,1,Answer

exten => 2169203111,2,Wait(2)

exten => 2169203111,3,Playback(channel-insecure-warn)

exten => 2169203111,4,SayDigits(${CALLERIDNUM})

exten => 2169203111,5,Wait(1)
```

Console Output

-- Executing Answer("Zap/1-1", "") in new stack

-- Accepting call from '2164114184' to '2169203111' on channel 0/1, span 1

-- Executing Wait("Zap/1-1", "2") in new stack

-- Executing Playback("Zap/1-1", "channel-insecure-warn") in new stack

-- Playing 'channel-insecure-warn' (language 'en')

-- Executing SayDigits("Zap/1-1", "2164114184") in new stack

-- Executing Wait("Zap/1-1", "1") in new stack

-- Executing SayDigits("Zap/1-1", "2164114184") in new stack

API Access

- C API

  - Accessible via standard ANSI C

  - Pre-existing example code for applications, channel drivers etc..

  - Forms the Core of Asterisk

  - Well documented, just read the code ;)

res_perl

- Similar to mod_perl for Apache

  – Single Perl interpreter is loaded and used to process requests

  – Allows embedding of perl commands directly   in Dial Plan

  – For the more adventurous, can be used to extend Asterisk to unimaginable tasks

  – Available as part of the asterisk_addons   package from CVS

res_js

- Similar to res_perl, except for Javascript

- Available from   http://www.pbxfreeware.org

AGI

- Asterisk Gateway Interface

  – Similar to CGI

  – Write in whatever you want (Perl, PHP,   Python, Pascal, Java, BASH…)

  – Variables are passed on StdIn to your
     Applications, results and commands are
     passed back on StdOut

  – Included w/ Asterisk, no additional work   required

Manager API

• Allows client/server interaction over   TCP/IP
sockets w/ authentication

• Can be used to issue commands or   monitor PBX
events

• Used by applications such as the Flash
      Operator Panel and IP Switchboard
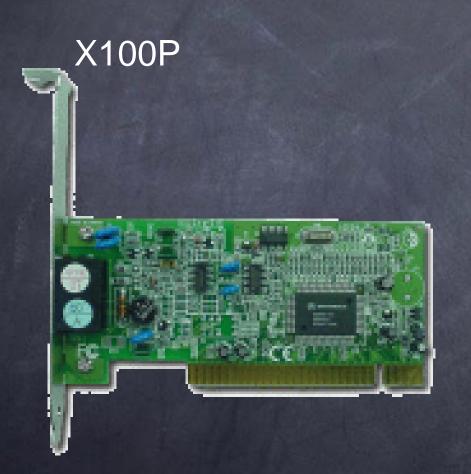
Pre-Recorded Prompts

- Hundreds of professionally recorded prompts

- Recorded by Allison Smith, a Voice Over Professional

  - http://www.theivrvoice.com

  - Clients include;

    - Target

    - Bell Canada

    - Volkswagen

    - Cingular

Connecting Asterisk To The World

- Many, Many Options..

- TDM Cards from Digium

- VoIP Softphones

- VoIP Hardware from Various Vendors

- VoIP Termination / Origination Service from Carriers

- The "ITSP" Internet Telephony Service Provider
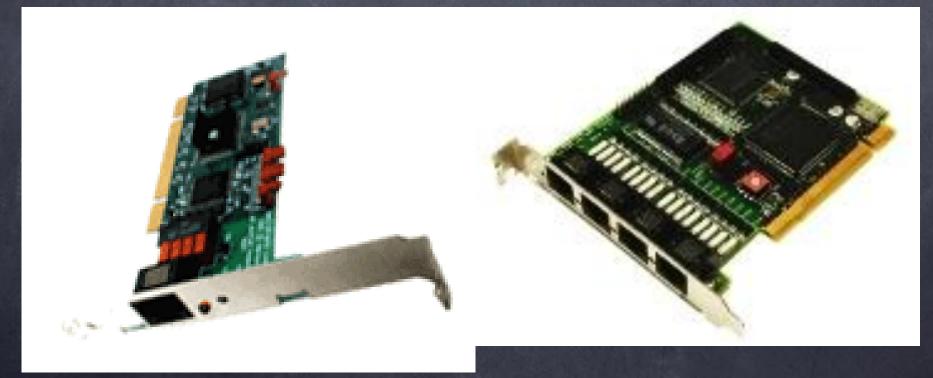
TDM Hardware from Digium

X100P

TDM400

TDM Hardware from Digium

T100P

TE405P

TDM Hardware from Digium

DS3000P

S100 IAXY

Analog Telephony Adapters

Linksys PAP-NA2

SPA-2100
2 Port FXS Analog Telephone Adapter
2 Ethernet Ports - LAN+WAN

SIPURA
technology, inc.

SIP Hardware Phones

Cisco 7960

Polycom IP-600

IAX2 Software Phones

Firefly

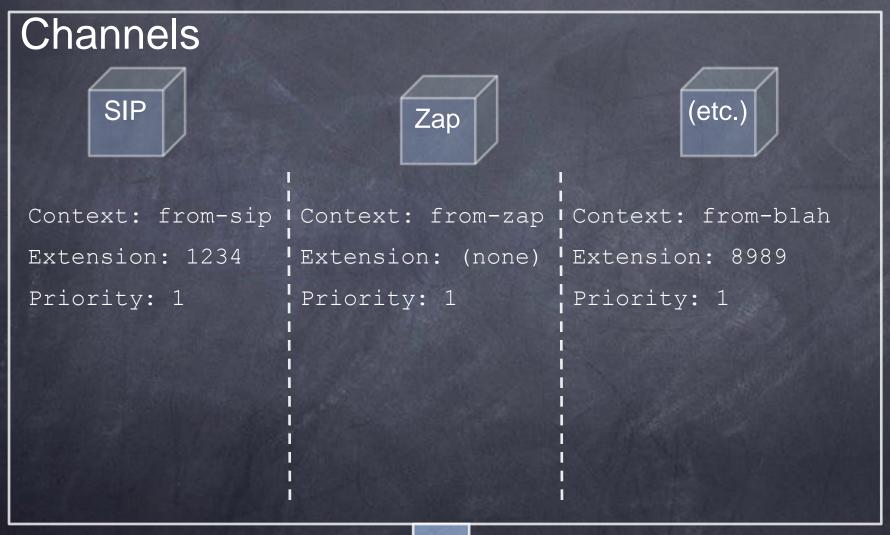## IAXPhone

SIP Software Phones

Xlite

Where To Go For More Information

- Digium Website at http://www.digium.com

- Asterisk Website at http://www.asterisk.org

- Asterisk Docs Project at http://www.asteriskdocs.org

- VoIP Info Wiki at http://www.voip-info.org

- Bug Tracker at http://bugs.digium.com

- #asterisk on irc.freenode.org

How You Can Help

- Get Involved

- Try it out

- Report Bugs

- Make Suggestions

- Submit Patches

- Help Review, Revise Documentation

How it goes together:

Channels

SIP

Zap

(etc.)

Context: from-sip
Extension: 1234
Priority: 1

Context: from-zap
Extension: (none)
Priority: 1

Context: from-blah
Extension: 8989
Priority: 1

(to extensions.conf)

sip.conf

```
[2000]
type=friend
host=dynamic
context=from-sip
secret=mysecret

[2001]
type=friend
host=dynamic
context=from-sip
secret=moresecret
```

extensions.conf

```
; This is where we handle our SIP calls
[from-sip]
exten => 1234,1,Answer
exten => 1234,2,Playback(tt-monkeys)
exten => 1234,3,Hangup
;
exten => _20XX,1,Dial(SIP/${EXTEN},30,r)
exten => _20XX,2,Goto(from-sip,${EXTEN},102)
exten => _20XX,102,Voicemail(b${EXTEN})
exten => _20XX,103,Hangup
;
exten => t,1,Hangup
exten => h,1,Hangup
```

Most-Used Applications

- Dial - tries to make a new call, and then connects current channel with new call if successful

- Goto - allows arbitrary leaps between contexts and priorities; allows modification of current extension

- Background - plays a file to current channel; interprets DTMF input

Magic with "Include"

- Contexts are NOT parsed in the order they appear

- Break up large contexts into smaller contexts and then use "include => <context>" in the "main" context

- This helps your sanity, as well.

Wrong

```
[main]
exten => _X11,1,Dial(Zap/1/${EXTEN},500,r)
exten => _9.,1,Dial(SIP/${EXTEN}@mysipprovider,60,r)
exten => _011.,1,Dial(SIP/${EXTEN:3}@int-sip,60,r)
exten => h,1,Hangup
```

Right

```
[main]
include => emergency
include => outside-line
include => international
exten => h,1,Hangup

[emergency]
exten => _X11,1,Dial(Zap/1/${EXTEN},500,r)

[outside-line]
exten => _9.,1,Dial(SIP/${EXTEN}@mysipprovider,60,r)

[international]
exten => _011.,1,Dial(SIP/${EXTEN:3}@int-sip,60,r)
```

Links

- http://www.asterisk.org/

- http://www.voip-info/wiki-Asterisk

- http://www.loligo.com/asterisk/

- http://www.onlamp.com/pub/a/onlamp/2003/0 7/03/asterisk.html

- http://www.digium.com/

- http://www.asteriskdocs.org/

Unabashed Plug Slide

- VOIP, Inc.

  - Builds/Sells: MTA SIP hardware (2 port FXS) and various other devices

  - Sells/Integrates: SIP proxy, billing/invoicing system, LCR system, customer care system, etc. (yes, asterisk is a part)

  - http://www.voipincorporated.com/