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

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)

Non-PBX * Use

- Extremely low bandwidth call relay (PRI-to-PRI via VoIP) via 802.11b or long-haul WAN
- Dating services/voicemail services
- Text-to-speech service (Nagios, weather, etc.)
- Call centers (inbound or outbound)
- Calling cards

Startup Notes

or: how to really annoy your [spouse/co-workers]

- Recommended setup for beginners:
  - PIII 700mhz or faster machine
  - X100P card (Digium ~$110)
  - 2 SIP devices (Sipura, Cisco ATA-186, Cisco 79[60, 40, 05, 12]) - $100-$300
- Test on your own line or home first, then expose to the office

How it goes together:

Channels

SIP	Zap	(etc.)
Context: from-sip	Context: from-zap	Context: from-blah
Extension: 1234	Extension: (none)	Extension: 8989
Priority: 1	Priority: 1	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

(calls from SIP channel configs end up here)

```
; 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/2003/07/03/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/