Providing VoIP Services with Asterisk (The Open Source PBX)

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Agenda

- What is Asterisk?
- What is it you want to sell?
- What is required to run Asterisk?
- Components of a service provider

What is Asterisk? (too brief!)

- UNIX-based (sorry, no M$ version yet)
- VoIP stack (SIP, H.323, IAX2, others)
- Hardware drivers (Digium, Voicetronix, others) for analog/digital cards
- IVR and menuing system
- Scriptable and extendable toolkit for working with telephony-based voice

What is Asterisk? (politics)

- GPL (with minor exceptions)
- Incredibly active development community
- Strongly opposes the cult of closed telephony - nothing is secret
- For better or worse, it’s like Perl. More than one way to do a task.
- Not US-centric

What is Asterisk not?

- Not a SIP proxy
- Not a billing system
- Does not have a customer interface
- Mostly not database driven… yet.
- Not a least-cost routing engine
- Not a natively SS7-capable system
- Not low-maintenance (not to start…)

So… that’s a lot of “Not”s

- Asterisk is for the true “do-it-yourself” shop - not for the UNIX beginner.
- Plateaus of possibility exist; beware!
- It is for those who want complete flexibility in product offering… at a price of complexity.
- Other solutions exist: COTS, VSP, etc. - weigh your options.

What is it you’re selling?

- Cost savings to customers?
- Features to customers?
- Home phone replacement or just additional services?
- Features and product offerings dictate how you implement Asterisk, and how much you spend.

Most-Often Used IPCSP Features

- Voicemail/VM-to-email
- Conference calling services
- Find me/Follow me calling
- IVR/Auto-Attendant
- PBX-like services (parking, MOH, etc.)
- Out-of-state DIDs
- Integration with existing PBX systems

So what do I need to run *?

- No set hardware specs; depends on goal
- Basic systems: typically Linux (Fedora, RH 9, Debian, whatever you prefer)
- Well-built MB which handles IRQ's well
- 1gb RAM (not really, but it doesn't hurt)
- 100gb disk (1m of VM = ~100k on disk)
- Rule of thumb: 100 G.729 channels in a 2x3.2ghz Xeon (YMMV)

What ELSE do I need to run *?

- Typically a PRI interface (Digium most popular cards)
- At least one “Zap” driver, for timing.
- Possibly MySQL or Postgres
- Possibly Apache
- Possibly Cricket (graphs from SQL source)

TYPICAL TOLL-QUALITY NUMBERS:

Low Jitter (~<5ms)

Low packet loss (~<0.1%)

Low latency (~<150ms for full path, including encoding)

WISPS: Watch out for overload.

What other components?

- CPE (Customer Premise Equipment)
  - Analog adapters (VOIP, Inc., Grandstream Cisco, many others)
    - Between ~$80 and ~$120 for 2 port FXS
  - Digital Handsets (Polycom, Cisco, Snom, Pingtel, Grandstream, many others)
    - Between ~$70 and ~$300. (You get what you pay for, trust me.)
  - Asterisk works well with most devices

Other Compnents (cont'd)

- Softphones
  - Xten, many many others
    - Some are free, some cost $
  - Unless extraordinary circumstances, customers do not like softphones.
  - If you must use a softphone, consider an IAX-based softphone.

QoS in Asterisk

- Minimal ToS bit setting for RTP and SIP traffic
- Systems should be dedicated for * - no X-windows, web apps, or databases
- Minimize IRQs from other devices if using PRI cards (low disk usage!)
- CPE is key for QoS enforcement

How large can * scale?

- Depends on goals, but “extremely large” is an adequate answer.
- Database abstraction, VM filesystems, multi-tiered architecture must be considered for “extra large” systems
- Typical single-CPU system can handle probably 1000 “normal” users (4xPRI)

Scaling (cont'd)

- Google on "Erlangs" for estimates on your sizing.
- Good rule of thumb: 1:8 to 1:10 offhook usage for busy hour
- Scaling numbers change dramatically with SIP termination of minutes (packet-only use of Asterisk)

Costs for Termination

- PRI-based
  - Typically have loop charge and port charge of ~$200/mo on top of minutes. Inbound often free.
- SIP-based
  - Sometimes has no minimums, but higher per-minute rate. Inbound often costs.
- Deposits typically required for Int'l calling on either; no set standard.

Cutting Edge Feature Sample

- DUNDi - inter-provider routing system
- Easy inbound ANI routing tools (the "anti-pollster" tool)
- Call center ACD functionality
- Perl/Python/Java extensions
- Call monitor/intercept/record
- ENUM support/SIP URI support
- Speech synthesis (Festival, others)

Final Advice

- Seek a consultant if you have a limited timeline.
- The Asterisk Wiki has the majority of the answers; many others have gone before you.
- Until proven otherwise, it will work, but test everything you are told.
- Truly well-designed systems aren’t cheap; don’t under-budget despite “free” software.

Links

- http://www.asterisk.org/
- http://www.voip-info.org/wiki-Asterisk
- http://www.digium.com/
- http://www.loligo.com/asterisk/
- http://www.astercon.com/

Thanks for your attention!
Questions?