Introduction to Asterisk

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Goals

- To become familiar with Asterisk's purpose and organization
- To step through configuring phone service using Asterisk

Glossary

- trunk - a phone line from the Central Office
- station - an internal phone
- FXO - "Foreign eXchange Office." phone interface to public telephone network (PSTN)
- FXS - "Foreign eXchange Station." internal phone interface. "Drives a phone."
- SIP - Session Initialization Protocol
- IAX - Inter - Asterisk eXchange

Glossary

- dialplan - tells a phone system what to do with dialed digits
- softphone - a phone implemented in software (i.e., Xten's X-Lite)
- hardphone - a SIP-capable telephone (such as the Cisco 7940)
- terminal adapter - a device which converts a standard phone into a SIP phone
- context - a way of partitioning phone calls. Can be used for security and/or organizational purposes
- codec - COder/DECoder. Compresses or expands binary data

Glossary

- PSTN - Public Switched Telephone Network - the traditional way of getting dialtone
- POTS - Plain Old Telephone System
- Phone Switch - a computer which connects calls (the "operator"). Asterisk can be one.
- Rate Center - a geographic location used to determine distance-related tolls (quaint?)

Glossary

- TDM - Time Division Multiplexing. A way of compressing many calls onto one circuit (T1)
- ISDN - Integrated Services Digital Network ("It Still Doesn't Work."). A 2 channel TDM digital line once marketed as the next generation phone service. Achieved blazing speeds of 128Kbps!
- PBX - "Private Branch eXchange." A computer which connects calls (see "switch.") Asterisk is a PBX.

Codecs

- G.729 - high-quality, low footprint codec. $10 Asterisk implementation. ~8k bps
- G.711 - uncompressed channel. Uses ~64k bps
- GSM - GSM compression. Decent quality, lower footprint ~13.2k bps
- G.723.1 - proprietary. Unsupported in Asterisk other than passthrough. ~6k bps
- G.726 – variable bandwidth. Not widely used?
- ILBC – open source codec. 15.2k bps

A Telephony Primer

- Simple technology
- Over 100 years old
- Smarts in the switch

New Twists

- Multiplexing
- Fiber/Digital (goodbye microwave & satellite)
- VoIP

Types of VoIP

- H.323 – Original. High overhead. Complex. Bloatware?
- MGCP – Media Gateway Control Protocol. SIP competitor meant to supercede H.323.
- SIP – Session Initialization Protocol. HTTP for the phone. Extensible. NAT-unfriendly.
- IAX – Inter-Asterisk eXchange – Asterisk's own. Even lower overhead than SIP. NAT-friendly.

SIP Providers

- Free (no PSTN access): Free World Dialup (FWD), Skype, MSN Messenger
- Commercial: Time-Warner, Packet8, VoicePulse, Vonage, FeatureTel, IConnectHere
- Many, many others.

How To Choose A Provider

- Quality (You Get What You Pay For)
- Your calling patterns
- Supported Rate Centers
- Support for your hardware
- Price

What Is Asterisk?

- Software Phone switch (PBX)
- Mark Spencer/Digium
- Zaptel hardware

Supported Phones

- Most any SIP phone (Cisco 79xx)
- Most any SIP ATA (Packet8's DTA-310, Sipura SPA-2000)
- Softphones (X-Lite, LIPZ4, Kphone, Windows Messenger)

Asterisk APIs

- Channel API
- Codec API
- File Format API
- Application API

File layout

- /etc/asterisk
- /var/lib/asterisk
- /var/log/asterisk
- Binaries in /usr/bin & /usr/sbin

Asterisk Dialplan

- extensions.conf
- format
- contexts

Asterisk voicemail

- voicemail.conf
- file formats
- email feature

Practical Examples

- voicemail.conf
- file formats
- email feature

Resources

- irc.freenode.net #asterisk
- www.asterisk.org
- www.digium.com
- www.voip-info.org
- www.pulver.com/fwd
- www.voxilla.org
- connect.voicepulse.com
- Asterisk's Mailing List

Question and Answer