VoIP on Unix Experiences

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Overview

- Very Informal
- Quick Intro
- What SW/HW I’ve used and/or heard of
- Experiences
- Q&A (throughout)

“Who I Was”

- Ammasso’s role in VoIP
- Lucent/Avaya (AT&T PBX)
- Large non-IP based PBX move to IP
- Built Media Gateways
  - H.323
  - Proprietary

Issues

- Unix vs. something else
- IETF vs. ITU
- Proprietary vs. Standards
- Open Source or not

Where Unix Fits in VoIP

- VoIP Signaling
  - Servers (SIP, H.323 etc.)
  - Media Gateways (IP to PSTN signaling)
  - Application Layer Gateways (ALG)
    - VoIP firewall
- Media Servers
- “Bearer”
  - Softphones
  - Media Gateways (IP to PSTN RTP/uLaw “voice”)
  - Applications
    - “Instant Messaging”

What Software is available?

- SIP Proxy's
- H.323 Gatekeepers
- SIP User Agents
  - e.g. Softphones
- PBX
- Media Gateways
- VoIP over 802.11

SIP - Proxy’s

- SER www.iptel.org/ser
- SIP Foundry (“Pingtel”) www.sipfoundry.org
- OpenH323 (??)
- VOCAL www.vovida.org
- Data Connection Limited ($$$)

SIP – User Agents

- Asterisk
- KPhone
- Linphone
- Xten (windows)
- Many, many more

H.323

- OpenH323
  - Gatekeeper
    - Routed and non-routed
  - Ohphone
  - Openphone (Windows)

PSTN Support

- Asterisk
  - FXO support for analog connections to “local loop”
  - FXS support for analog phones
  - T1/E1 & ISDN for digital “leased lines”
- OpenH323

Traditional PBX Replacement

- Asterisk
  - H.323
  - SIP
  - T1/E1
  - ISDN PRI
  - FXO / FXS
- Any to any call support
- “full featured” PBX solution

What to look out for?

- NAT
  - Get everything working on one side of NAT first, then introduce NAT
  - Utilize Simple Traversal of UDP over NAT (STUN)
- Unit test each component first
  - Bad soundcard ??
- 10/100 HUB + ethereal/tcpdump
- Test servers initially with XP (free) softphones
- Separate boxes for separate functions
- Firewall Rules

What to look out for (UA)

- Softphone
  - Kphone
    - pre-configures for their STUN
  - Xten
    - Optimal Asterisk settings documented
- Hardphone
  - Turn off vendor extensions
  - Settings default to an all vendor solution

Firewalls

- Open firewall completely for testing
- Add policies, see what breaks
- VoIP != “peer to peer”
  - Doesn’t scale
  - Limits functionality
  - RTP stream can arrive from anywhere
- Allow all UDP unless using STUN or ALG

PSTN Interoperability

- Unix as Media Gateway (MG)
  - Asterisk as SIP UA
  - No H.248/megaco/MGCP
  - “just add” PCI cards (FXS/FXO, T1/E1)
- Keep behind FW/NAT
  - FW/NAT policies apply only to remote users
  - Network MG works as any other UA
  - Local users have no issues

Resources

- www.asterisk.org
- www.iptel.org/ser
- www.voip-info.org
- www.openh323.org
- www.sipfoundry.org
- www.pulver.com/fwd