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