A sysadmin's view of VoIP

Ewen McNeill
<ewen@naos.co.nz>

Naos Ltd

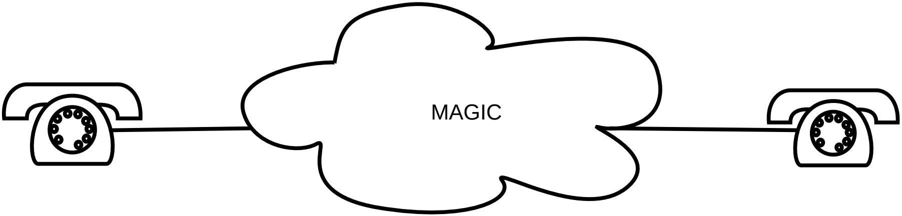
2006/01/23 — Sysadmin Miniconf, Linux.Conf.Au 2006

Outline

Administrivia

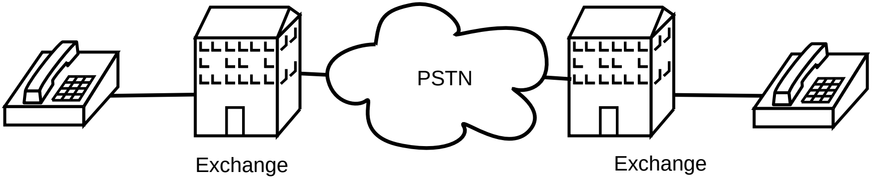
- About the speaker
  - Runs Naos Ltd
  - A Wellington based Linux, Unix and Networking consultancy
- Informal Survey
- Questions Policy
  - If it is about the current slide, raise your hand.
  - Please ask any other questions at the end.
- Slides: http://www.naos.co.nz/talks/sysadmins-voip/

How the telephone system works —Part 1



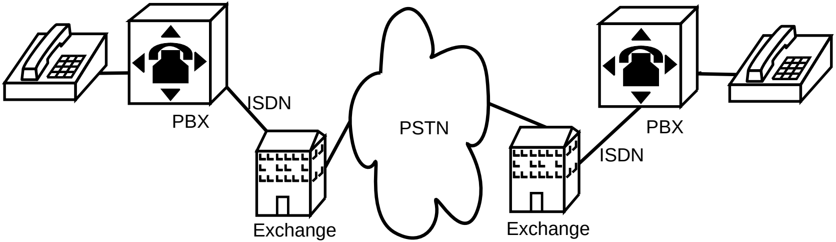
"Any sufficiently advanced technology
is indistinguishable from magic."
— Arthur C Clarke

How the telephone system works —Part 2



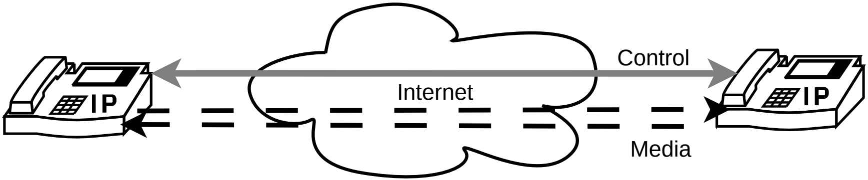
- PSTN: Public Switched Telephone Network
- E.164: ITU standard for “phone numbers”
- DTMF: Dual-Tone Multi-Frequency “touch tones”

How the telephone system works —Part 3



- PBX: Private Branch Exchange
- Manages calls into and out of organisation
- Does phone number translation
- ISDN: Intergrated Services Digital Network
- BRI: Basic Rate, 2 * 64Kbps data channels
- PRI: Primary Rate, 2Mbps (E1)

VoIP Overview



- All VoIP operates in a similar manner
- Control channel to set up call
- Media channels to carry encoded voice data
- Similar approach to FTP
- Lots of protocols for control and media channels

Control channel protocols

- H.323: ITU standard, uses ASN.1
- SIP: IETF RFC 2543, HTTP-like headers
- SCCP: “Skinny”: Cisco proprietary protocol
- Skype: Proprietary protocol based on Kazaa
- Several other less widely used protocols

SIP Example

```
INVITE sip:045551212@202.53.189.51;user=phone SIP/2.0
Via: SIP/2.0/UDP 161.29.192.202:5060
From: <sip:049714218@202.53.189.51;user=phone>;tag=1646388700
To: <sip:045551212@202.53.189.51;user=phone>
Call-ID: 3581559645@161.29.192.202
CSeq: 1 INVITE
Contact: <sip:049714218@161.29.192.202:5060;user=phone;transport=udp>
User-Agent: Cisco ATA 186  v2.15 ata18x (030313a)
Expires: 300
Content-Length: 256
Content-Type: application/sdp
```

Media channel protocols

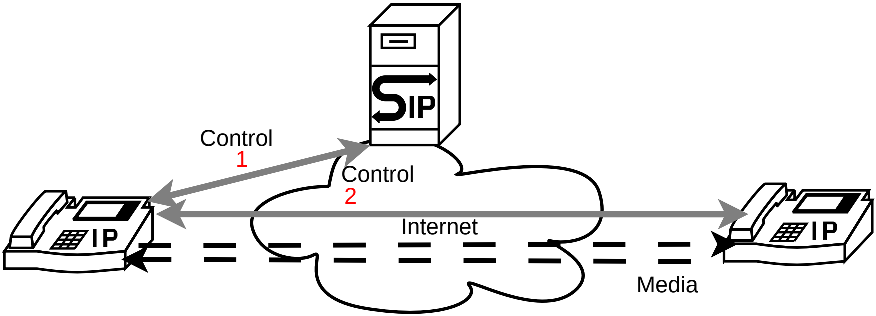
- RTP: Realtime Transport Protocol
- RTP is ITU standard H.225.0
- And is also IETF RFC 1889
- Used by both H.323 and SIP
- Similar approaches used by other protocols
- Essentially timestamped UDP packets
- Between dynamically negotiated port numbers

Digitizing voice —codecs

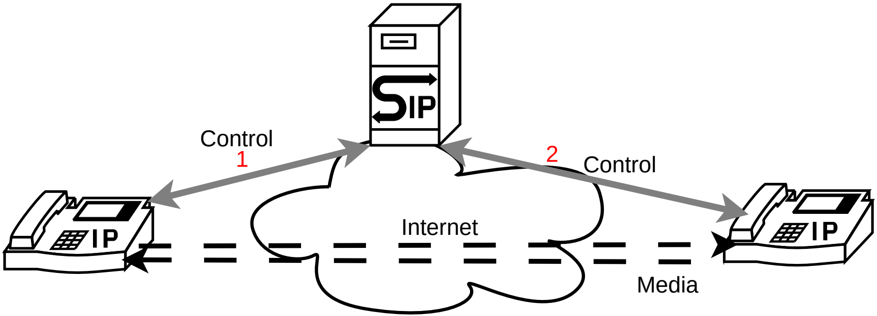
- Same codecs used by H.323 and SIP
- All produce small packets: 50-250 data bytes
- G.7xx codecs are ITU standards:
- G.711: 64kbps PCM (Pulse Code Modulation)
- G.726: 16-40kbps ADPCM (Adaptive Differential PCM)
- GSM: 13kbps, also used by GSM cellphones
- Codecs supported vary from product to product
- Patent and licensing issues around several codecs

Finding the other phone —Part 1

- Still need a way to locate the other phone
- Static configuration is possible — but doesn't scale
- In H.323 a directory server is commonly used
- In SIP a proxy server can provide directory services via a redirection



Finding the other phone —Part 2



- Another common SIP proxy approach
- Proxy in the middle of all control communication
- Note how media channels still flow directly

Finding the other phone —Part 3

- ENUM: IETF RFC 3761: e164.arpa
- Commonly proposed solution to finding the other phone
- Being experimentally deployed at present
- Encodes a E.164 (phone) number into a NAPTR DNS request
- Take fully qualified number, reverse digits, separate by "." (periods), and append .e164.arpa
- +64-21-916-965 becomes 5.6.9.6.1.9.1.2.4.6.e164.arpa
- Result of NAPTR query indicates protocol and location

VoIP hardware —Part 1

- Many softphones (software on a PC) solutions exist
- For Linux options include:
  - GnomeMeeting: http://www.gnomemeeting.org/ (uses H.323)
  - KPhone: http://www.wirlab.net/kphone/ (uses SIP)
- Microsoft NetMeeting uses H.323
- Not as convenient as a “real phone” for most users
- Need a headset to get reasonable sound quality
  - Echo cancellation particularly a problem
- Wide range of hardware solutions available now

VoIP Hardware —Part 2

- Ethernet VoIP phone
  - Cisco, Uniden, etc
  - Protocols include SIP, H.323, “Skinny”

- Wireless (802.11b/g) VoIP phone
  - Often SIP only
  - Ability to roam through wireless network (eg Cafenet)

VoIP hardware - Part 3

- ATA: Analogue Telephone Adapter
  - Cisco, Uniden, Sipura, etc
  - Usually 1-2 FXS (Foreign Exchange Station) ports for analogue handsets
  - Some include FXO (Foreign Exchange Offi ce) ports to connect to the PSTN
  - Protocols include SIP, H.323, “Skinny” depending on vendor
- VoIP Gateway
  - Often modular, especially from router vendors
  - Accept FXO, FXS, BRI, PRI connections
- Telephone line cards
  - PCI (or ISA) cards for use in a PC
  - Combinations of FXO, FXS, BRI, PRI connections
  - Typically used in a PBX (or PBX replacement) situation

Connecting to the PSTN

- VoIP is useful, but limited, as a standalone system
- Ideally want to connect to legacy phone system
- To be able to make outgoing calls and receive incoming calls
- You can do this yourself (like a PBX)
- Or outsource it to a telco that accepts calls via VoIP and connects them to the PSTN (and vice versa)

Connecting to the PSTN yourself

- Requires a device that can talk VoIP on one side and to PSTN on the other
- And one or more suitable PSTN lines from a telco
- Solutions range from one simultaneous voice connection through dozens of simultaneous voice connections
- Depending on requirements (and budget!)

PSTN Interconnects

- PSTN interconnection can be analogue or digital
- Analogue connections suitable for home phone lines
- Connect PSTN to the FXO (Foreign eXchange Office) port on VoIP gateway or line card
- Business connections normally digital (ISDN)
- Either one or more BRI (Basic Rate) or PRI (Primary Rate) connection
- For a PRI you generally rent the circuit, plus as many 64Kbps timeslices as you need for simultaneous calls

PSTN number translation

- Connecting to the PSTN typically requires phone number translation
- Both incoming and outgoing
- Similar to setting up a PBX
- Incoming: direct calls to suitable VoIP phone (eg, receptionist)
- Outgoing: put destination number in telcos preferred format
- May need to strip off “outside line” prefix (eg, 1 or 9)
- And possibly add a different prefix (eg, area code or country code)

Multiple PSTN connections

- You may want multiple connections to the PSTN
- To save money (eg, local calls to more numbers), or for reliability
- Generally each interconnection will have its own phone numbers assigned to it
- Translations and routing to cover all the connetions can get quite complicated
- May be easier to outsource to a provider that gives you good rates

Challenges for the Sysadmin

- Latency and jitter
- Faxes and VoIP
- Security
- Firewalls (and NAT)

Latency and Jitter —Part 1

- VoIP is very sensitive to network performance
- Including latency, jitter, and packet loss
- Round trip latency of up to 300ms is okay; less is better
- Sigificantly more sounds like a bad satellite phone call
- Jitter (variation in latency) can be quite noticable
- Can be better to drop packets than delay them significantly
- Occassional packet loss is tolerated by people

Latency and Jitter —Part 2

- Lots of spare bandwidth helps
- Prioritise VoIP traffic and/or reserve bandwidth for it
- For Linux see tc(8), and http://lartc.org/howto/lartc.qdisc.html
- Particularly a problem with low bandwidth (eg, residental) connections
- ADSL and wireless connections (in New Zealand) often have signifcant latency (100ms round trip)
- Cable modems are generally better (eg, 20-30 ms round trip)
- Upstream bandwidth is quite limited on many NZ ADSL connections (again cable modems are generally better)

Faxes and VoIP

- Sending faxes through a VoIP system can be challenging
- May be easier to persuade users to, eg, email a PDF
- Use a high bandwidth codec (eg, G.711 — 64kbps)
- May need to limit fax speed (eg, to 9600bps, V.29)
- Need a good network connection, and some luck
- T.38 Fax Relay is more reliable where supported
  - Effectively decodes fax locally
  - Then sends bitmap over network
  - And reencodes into fax tones at the remote end

VoIP Security – Part 1

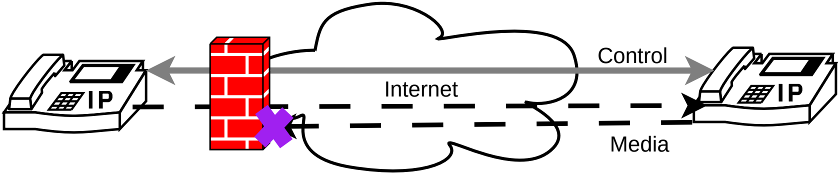
- By default both control channel and media channel are sent unencrypted
- Encryption is thinly supported, especially in Open Source projects
- For H.323, H.235 defines encryption (TLS or IPSec)
- For SIP, TLS can be used
- TLS: IETF RFC 2246: Transport Layer Security
- SRTP: IETF RFC 3711: Secure RTP
- May be best to run through a VPN for now, where possible
- Or use a separate Layer 2 (VLAN or switch)

VoIP Security – Part 2

- Access to PSTN gateways needs to be secured
- And access to PBXes which are allowed to use those gateways
- Bouncing off company PBXes was a common phreaking technique to get “free” phone calls
- Security by static IP address ranges may be easiest to administer
- SIP includes a HTTP Basic-Auth style authentication
- But it is effectively in plain text
- H.323 also has a “shared secret” authentication scheme

VoIP and Firewalls

- VoIP control channel is usually a single well known port
- H.323: TCP and UDP 1720
- SIP: TCP and UDP 5060
- Other ports can be used as the port number is included in the protocol addresses
- Media channels are dynamically negotiated, often within a wide range of ports
- Assumes the “end to end” Internet
- Can lead to “one way audio”



Challenges of NAT

- Control channel can usually be NAT'd through firewall okay
- But media channel is challenging
- Because dynamic port negotiation includes IP addresses
- Meaningless outside the LAN if using RFC 1918 addresses
- Typical symptom is "one way audio"
- If both ends have the problem then no audio will be heard
- This is a moderately common issue with FTP as well, but there is better firewall support for FTP

Firewall and NAT solutions – Part 1

- Using a protocol aware firewall
  - For Linux, sip-conntrack-nat: http://www.iptel.org/sipalg/ (Alpha test code; in iptables Patch-o-Matic)
  - For Linux, h323-conntrack-nat: http://max.kellermann.name/projects/netfilter/h323.html (Alpha test code; in iptables Patch-o-Matic)
- Using an application level media proxy
  - For Linux, Asterisk: http://www.asterisk.org/
  - Or siproxd: http://siproxd.sourceforge.net/
  - Or for H.323: OpenH323Proxy: http://openh323proxy.sourceforge.net/

Firewall and NAT solutions – Part 2

- STUN: IETF RFC 3489: Simple Traversal of UDP through NAT
- Tunnelling in unfiltered, globally unique, IP address
- Using vtun or GRE, or another VPN
- Will need to do policy routing to send traffic from those IP addresses back out the tunnel
- Linux: use iproute2 to route based on the source address range
- http://lartc.org/howto/lartc.rpdb.html
- Beware of security issues with tunneling in IP addresses

Policy routing example

```
ip route add default via REMOTEGATEWAY table TABLENAME
ip rule add from TUNNELEDBLOCK table TABLENAME
```

(Need to add TABLENAME to /etc/iproute2/rt_tables)

Linux VoIP Software —Asterisk —Part 1

- http://www.asterisk.org/
- Full functional, open source, PBX software for Linux
- Original developed by Digium Inc, who make PBX hardware including line cards (Zaptel)
- Supports SIP, IAX (Inter Asterisk eXchange), H.323 somewhat
- Huge range of features (“applications”)
- All directly usable out of the dialplan (“extensions.conf”)

Linux VoIP Software —Asterisk —Part 2

- Unfortunately configuration syntax conbimes the worst features of windows.ini and Basic
- So you probably should not configure large dialplans by hand
- Lots of front ends available now, eg
  - Asterisk@Home: http://asteriskathome.sourceforge.net/
  - Xorcom Rapid: http://xorcom.com/rapid
  - Asterisk Management Portal: http://sourceforge.net/projects/amportal/

Asterisk extensions.conf example

```
[nzpstn]
exten => _021[1-2]XXXXXX,1,Goto(outbound,${EXTEN},1)
exten => _021XXXXXX,1,Goto(outbound,${EXTEN},1)
exten => _02409XXXX,1,Goto(outbound,${EXTEN},1)

[congratulations]
exten => s,1,Wait,1                    ; Wait a second, just for fun
exten => s,2,Answer                    ; Answer the line
exten => s,3,DigitTimeout,5            ; Set Digit Timeout to 5 seconds
exten => s,4,ResponseTimeout,10        ; Set Response Timeout to 10 seconds
```

Linux VoIP Software —SER

- SIP Express Router: http://www.iptel.org/ser/
- High performance SIP proxy
- Useful for routing SIP calls
- SIP registration support, database integration
- Configured in a C-like language
  - But beware the parser is fussier than C
- No media proxy support (only proxies control channel)

SER example

```
if (method=="INVITE") {
  log(1, "Handling call INVITE");
  record_route();      /* Force subsequent requests here     */

  if (!uri=~"sip:\+?[0-9]+@.*") {
     log(1, "Rejecting non-E.164 destination");
     sl_send_reply("403", "Call cannot be served here");
     break;
  };
  ....
}
```

Other Linux VoIP Software

- GNU Bayonne: http://www.gnu.org/software/bayonne/bayonne.html
  - Telephony application server
  - Range of hardware and protocol support
- SIP-P: http://sipp.sourceforge.net/
  - SIP testing tool (protocol generator)
- Diagnostics
  - ngrep: http://ngrep.sourceforge.net/
  - ethereal: http://www.ethereal.com/
  - SIP is easier to debug than H.323, because you don't need to decode ASN.1

That's All Folks!

- More information:
  - http://www.voip-info.org: A useful Wiki, especially for Asterisk
  - New Zealand Asterisk Users Group (ASTUG): http://astug.org.nz/
- Questions?
- Slides: http://www.naos.co.nz/talks/sysadmins-voip/