Introduction to Telephony

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Analogue Telephony

• Where it all started!

• PSTN allows connection between any two endpoints

• Human speech typically in the range 250 - 3,000Hz

  • Humans can hear in the region of 20 - 20,000Hz

• PSTN analogue channel originally designed to carry 300 - 3,500Hz

• Most analogue lines delivered via copper from the local exchange (or CO, Central Office)

  • Average line in NZ ~3Km. Longest lines >7Km
Analogue Telephony

• Even in the day and age of VoIP, this is still important!
  • Analogue telephone adapters (ATAs)
  • Fax - it just won’t go away :)
  • Echo
  • Voice and sound is most definitely analogue
    • First and last conversions in a VoIP call
The Analogue Telephone

- Analogue telephones connect to a copper pair
  - A two wire circuit
- Analogue telephones are comprised of five major parts:
  - Ringer
  - Dial Pad
  - Hybrid
  - Hook switch
  - Handset
Ringer

- The exchange provides DC (~48vDC) to power the phone
  - Exchange = big centralised UPS
- Exchange provides a burst of AC (~80vAC) to ring the phone’s bell
  - Originally a mechanical bell, these days an electronic buzzer
- These days phone have a Ringer Equivalence Number (REN)
  - Exchange can power up to a sum of 5 RENs
  - Phones these days typically < 0.5 REN
  - ATAs have same limitation
Dial Pad

- Telephones need to signal back to the exchange
- Originally done with a rotary dialler making and breaking the copper loop
  - Pulse Dial, still typically supported by exchanges and some VoIP kit
- All done with audio tones now
  - Dual Tone Multi Frequency (DTMF)
  - Telephone handsets a matrix of switches
  - One tone per column, one per row
  - Each switch generates two tones, hence Dual Tone
DTMF Tones

<table>
<thead>
<tr>
<th>1209 Hz</th>
<th>1336 Hz</th>
<th>1477 Hz</th>
<th>1633 Hz</th>
</tr>
</thead>
<tbody>
<tr>
<td>697 Hz</td>
<td>1</td>
<td>2</td>
<td>3</td>
</tr>
<tr>
<td>770 Hz</td>
<td>4</td>
<td>5</td>
<td>6</td>
</tr>
<tr>
<td>852 Hz</td>
<td>7</td>
<td>8</td>
<td>9</td>
</tr>
<tr>
<td>941 Hz</td>
<td>*</td>
<td>0</td>
<td>#</td>
</tr>
</tbody>
</table>
Hybrid Network

• The heart of an analogue telephone

• The transformer that couples two signals onto one line
  • Send (Tx) and receive (Rx)
  • Creates sidetone (‘good echo’)
  • Allow speaker to hear himself

• Creates echo unless perfectly balanced
Hook Switch

• Telephone uses it to signal state to the exchange
  • On Hook, closes the copper loop
    • Phone idles, waiting for incoming ring
  • Off Hook, breaks the copper loop
    • Requests dial tone from the exchange, and then allows audio to pass
• Also used to signal ‘advanced’ features, e.g. call waiting
  • Hook Flash - a timed closure of the hook switch, typically ~300ms
Tip and Ring

- Telephony world often refers to ‘Tip’ and ‘Ring’
- Historical term from the days when exchanges were literally switchboards
- Operator manually patched lines together
- Tip (red) = +ve polarity (0v)
- Ring (green) = -ve polarity
  - -48v on hook, -7v off hook
Telephone and Line Impedance

- Impedance = technical way of saying resistance
  - Varies with both frequency and phase
- American telephone impedance is 600 ohms
  - Approximation of the impedance of 0.4mm twisted copper pair at voice frequencies
- British (and NZ) telephone impedance is complex (in the resistive sense of the word), called BT3
  - 370 ohms in series with (620 ohms in parallel 310nF)
  - Attempt to better match line impedance
Echo

• VoIP does not cause Echo!
  • Hybrids cause echo
  • Echo becomes apparent as latency increases
  • VoIP creates higher latency than circuit switched circuits

• Hybrids must be balanced to the line to effect maximum power transfer and minimal signal reflection
  • Reflection back down the line = echo
  • Reflection back towards the handset = sidetone
Echo - Telephone Hybrid

- 4-wire input
- 2-wire telephone line
- ZB=RI=RO=Zline=600 ohm
- Transformers: 1:1:1:1
- 4-wire output

RI

ZB

RO
Echo

• Sidetone is used to let the user know that the phone ‘is working’
  • It’s somewhat unnatural to not hear oneself
  • Too much sidetone and you can only hear yourself
  • Too little and it appears the line is dead

• Echo is present on most lines
  • When latency is low (< 20ms or so) the far end perceives it as sidetone
Acoustic Echo

- Caused by the output of the handset’s speaker entering the microphone
  - Due to the speaker volume being too loud or microphone sensitivity too loud
    - Very bad with softphones when not using a headset
  - Or flimsy handset construction (acoustic coupling through the handset itself)
  - The telephone handset design hasn’t changed much over the years as it is a very good one!
- Indistinguishable to the far end from echo caused by the local hybrid
Reducing Echo

• There are only four ways to reduce echo
  • Remove the two wire (analogue) portion of the call
  • Balance the analogue portion of the call better
    • Hard to do even if you do have access to the endpoint(s)
  • Reduce the latency
    • Often impossible, e.g. long distance calls
  • Cancel the echo
Echo Cancellers

- Measure signal on the line, predict the echo, and create a signal to cancel it.
- Echo cancellers are configured for a ‘tail’ length - the maximum latency of an echo which it can possibly cancel.
- Takes time to converge to an echo cancelled state, dependant on the tail length of the canceller.
- Echo cancellers aren’t perfect, so best to prevent echo in the first place.
- Popular misconception that software based echo cancellation is bad.
  - Hardware echo cancellers have very good, often patented algorithms.
  - No really good open source software implementations (yet...)
  - Software echo cancellation is not bad - if you have a good algorithm!
Digital Telephony

- Telephony moved digital for the same reason everything else did

- Voice turned to a digital signal using Pulse Code Modulation (PCM)
  - Sample signal in time

- Two important factors:
  - Number of samples per second (highest frequency is half of the sample rate - Nyquist’s Theorem)
  - Number of bits used to encode signal

- Tradeoff between quality and bandwidth - standard is 8bits at 8kHz sampling
Digital Telephony

- Standard voice channel (timeslot, or DS0) is 64kbit/s

- Most common codec is G711, a companding codec
  - Two types, ulaw (US) and alaw (Europe)

- Majority of telephone conversation is ‘quiet’

- More bits are allocated to quiet signals to improve overall quality

Figure 7-12. Quantized and companded at 5-bit resolution from Asterisk, The Future of Telephony
PSTN Circuits

• Analog line

• ISDN
  • Basic rate, two voice 64kbit/s voice channels + 16kbit/s data channel -> 144kb/s
  • Primary rate
    • US - T1, 24 64kbit/s voice channels -> 1.544Mb/s line rate
    • Europe - E1, 30 64kbit/s voice channels -> 2.048Mb/s line rate
  • Proprietary circuits between key phones and PBXs - not covered here
VoIP

• Natural progression from digital telephony
  
  • Circuit switched --> packet switched
  
  • Still a need to sample and encode signals

• Many different codecs in the VoIP world

• Many different signalling protocols
## Codecs

<table>
<thead>
<tr>
<th>Codec</th>
<th>Payload Bitrate</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711</td>
<td>64 kbit/s</td>
</tr>
<tr>
<td>G.726</td>
<td>16, 24, or 32 kbit/s</td>
</tr>
<tr>
<td>G.723.1</td>
<td>5.3 or 6.3 kbit/s</td>
</tr>
<tr>
<td>G.729</td>
<td>8 kbit/s</td>
</tr>
<tr>
<td>GSM</td>
<td>13 kbit/s</td>
</tr>
<tr>
<td>iLBC</td>
<td>13.3 or 15.2 kbit/s</td>
</tr>
<tr>
<td>Speex</td>
<td>2.15 to 22.4 kbit/s</td>
</tr>
</tbody>
</table>

- G711 gives highest quality
- Some wide bandwidth codecs supported now
- GSM very popular - good CPU time vs. bandwidth tradeoff
- Speex well suited to changing network conditions
Signalling Protocols

• Signalling protocols needed to allow endpoints and intermediary devices to set up calls

• Common VoIP signalling protocols:
  • H.323
  • MGCP (Media Gateway Control Protocol)
  • Skinny / SCCP (Skinny Client Control Protocol)
  • IAX (Inter Asterisk eXchange)
  • SIP (Session Initiation Protocol)
H.323

- 10 year old ITU protocol developed to carry multimedia traffic across an IP network
  - Actually a suite of protocols, the signalling component being H.245
  - Originally designed for video conferencing
- Quickly became de-facto standard for VoIP - and is still used today in many large carrier environments
- Relatively secure and bug free due to its maturity
- Does not work well with NAT at all
- Has all but disappeared in end stations over the past few years
MGCP

- IETF standard, RFC 3345 (obsoletes RFC2705)
- Still widely deployed
  - Slowly being displaced by SIP
- Being a gateway protocol, has very good gateway features useful for a carrier environment
- Some end phone support for MGCP but never big
Skinny / SCCP

• Cisco Proprietary protocol
  
  • Originally developed by Selius Systems in the mid 1990’s
  
  • Cisco bought them and entered the telephony market :)
  
• Cisco CallManager based on Skinny, though finally moving to the more standard SIP

• Called Skinny as phones are ‘dumb’.
  
  • SCCP phone events: button X pressed, turn on lamp X, turn off lamp X
IAX

- Developed by Digium, creators of Asterisk
  - Apparently it’s pronounced “eeks”. I still say “eye-aye-ex”
- Primarily designed to connect Asterisk servers together
  - Has unique ability to trunk multiple calls down one dataflow
  - Includes some extra signalling
  - Uses a single UDP port, so NAT friendly
  - Can use plaintext, MD5, or RSA key exchange for authentication
- IAX, although open source, is not a widely adopted standard
SIP

• SIP is *the* VoIP protocol these days - RFC 3261 (obsoletes RFC 2543)

• Original (simple!) draft created in 1996

• We’ll be concentrating on SIP and largely ignoring the rest
  
  • It is worth playing around with IAX if you are going to be using Asterisk

• Largely ignored early on it’s life (H.323 was used)

• Largely standard implementations of SIP now

• Not overly NAT friendly, although workarounds exist

• Worthy of a more in-depth look!
What is SIP?

“Session Initiation Protocol (SIP),

an application-layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants”

(RFC 3261)
SIP Overview

- ASCII based signalling protocol
- Analogous to HTTP messages
- Works independent of the underlying network transmission protocol
- Provides mechanisms to:
  - Establish a session
  - Maintain a session
  - Modify and Terminate a session
SIP Overview

• Strength is it’s simplicity and basic assumptions

• Component reuse
  • A child of SMTP and HTTP
  • SIP also uses MIME to carry extra information
  • Uses URI Eg: sip:jonnyphone@jonnynet.net
SIP Overview

- Scalable and robust protocol
  - Can offload various separate SIP functions to dedicated servers
  - Uses distributed architecture
- Very inter-operable protocol (well, these days it is!)
- Supports mobility through use of endpoint registration
- Uses RTP to carry media
SIP elements

- SIP User Agents
  - User Agent Clients (UAC) - the entity which initiates a call
  - User Agent Servers (UAS) - the entity which receives a call
- SIP Servers
  - Registrar server
  - Proxy server
  - Location server
  - Redirect server
SIP Registrar Server

- Users send registration requests to Registrar server
- Keeps track of client locations
- Supports various forms of authentication
- Often combined with the functionality of a Proxy server (Asterisk does this)
SIP Proxy Server

- Acts both as a server and a client
- Receives SIP messages, forwards to next SIP server
- Can perform functions such as Authentication, Authorisation, and Accounting (AAA)
- Provides network access control
- Requests are serviced internally or by passing them on to other servers.
- Interprets, rewrites or translates a request message before forwarding it.
SIP Messages

**SIP Methods:**
- INVITE – Initiates a call by inviting user to participate in session.
- ACK - Confirms that the client has received a final response to an INVITE request.
- BYE - Indicates termination of the call.
- CANCEL - Cancels a pending request.
- REGISTER – Registers the user agent.
- OPTIONS – Used to query the capabilities of a server.
- INFO – Used to carry out-of-bound information, such as DTMF digits.

**SIP Responses:**
- 1xx - Informational Messages
  - 180 ringing
- 2xx - Successful Responses
  - 200 OK
- 3xx - Redirection Responses
  - 302 Moved Temporarily
- 4xx - Request Failure Responses.
  - 404 Not Found
- 5xx - Server Failure Responses.
  - 503 Service Unavailable
- 6xx - Global Failures Responses.
  - 600 Busy Everywhere
SIP Messages

**Informational**
- 100 Trying
- 180 Ringing
- 181 Call forwarded
- 182 Queued
- 183 Session Progress

**Success**
- 200 OK

**Redirection**
- 300 Multiple Choices
- 301 Moved Perm.
- 302 Moved Temp.
- 380 Alternative Serv.

**Request Failure**
- 400 Bad Request
- 401 Unauthorised
- 403 Forbidden
- 404 Not Found
- 405 Bad Method
- 415 Unsupported Content
- 420 Bad Extensions
- 486 Busy Here
SIP Messages

**Server Failure**
- 504 Timeout
- 503 Unavailable
- 501 Not Implemented
- 500 Server Error

**Global Failure**
- 600 Busy Everwhere
- 603 Decline
- 604 Doesn’t Exist
- 606 Not Acceptable
SIP Addressing

- Can use SMTP style addressing
  - sip:jonnyphone@jonnynet.net

- Or E.164 (telephone number) addressing
  - sip:64212304323@jonnynet.net
Example SIP Call Flow

Call Setup:
1. INVITE
2. 100 Trying
3. INVITE
4. 302 (Moved Temporarily)
5. ACK
6. INVITE
7. 100 Trying
8. 180 (Ringing)
9. 180 (Ringing)
10. 200 (OK)
11. ACK
12. INVITE
13. 180 (Ringing)
14. 200 (OK)
15. ACK
16. INVITE
17. 180 (Ringing)
18. 200 (OK)
19. ACK

Media Path:
RTP MEDIA PATH

Call Teardown:
20. BYE
21. 200 (OK)
22. BYE
23. 200 (OK)
24. BYE
25. 200 (OK)
SIP Registration

Sip read:
REGISTER sip:203.114.148.130 SIP/2.0
Via: SIP/2.0/UDP 10.71.0.222:5060;rport;branch=z9hG4bK75D24E71C03111DB8A1300112476567E
From: Jonny test <sip:4989560@203.114.148.130>;tag=1675365723
To: Jonny test <sip:4989560@203.114.148.130>
Contact: "Jonny test" <sip:4989560@10.71.0.222:5060>
Call-ID: 7574F569C03111DB8A1300112476567E@203.114.148.130
CSeq: 19613 REGISTER
Expires: 1800
Authorization: Digest
username="4989560",realm="asterisk",nonce="25a752f4",response="ea87d99f48b43a97b39819e3fedbf8b8",uri="sip:203.114.148.130"
Max-Forwards: 70
User-Agent: X-Lite release 1105x
Content-Length: 0

Transmitting (NAT) to 202.146.237.70:5060:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.71.0.222:5060;branch=z9hG4bK75D24E71C03111DB8A1300112476567E;received=202.146.237.70;rport=5060
From: Jonny test <sip:4989560@203.114.148.130>;tag=1675365723
To: Jonny test <sip:4989560@203.114.148.130>;tag=as52d7bb4c
Call-ID: 7574F569C03111DB8A1300112476567E@203.114.148.130
CSeq: 19613 REGISTER
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER
Expires: 1800
Contact: <sip:4989560@10.71.0.222:5060>;expires=1800
Date: Mon, 19 Feb 2007 14:54:56 GMT
Content-Length: 0
SIP Invite

Sip read:
INVITE sip:0212304323@203.114.148.130 SIP/2.0
Via: SIP/2.0/UDP 10.71.0.222:5060;rport;branch=z9hG4bKC19D7202C0311DB8A1300112476567E
From: Jonny test <sip:49895600203.114.148.130>;tag=1386353914
To: <sip:0212304323@203.114.148.130>
Contact: <sip:4989560010.71.0.222:5060>
Call-ID: C06D0E06-C031-11DB-8A13-00112476567E@10.71.0.222
CSeq: 32821 INVITE
Max-Forwards: 70
Content-Type: application/sdp
User-Agent: X-Lite release 1105x
Content-Length: 205

v=0
o=4989560 81389423 81389572 IN IP4 10.71.0.222
s=X-Lite
c=IN IP4 10.71.0.222
t=0 0
m=audio 8000 RTP/AVP 3 101
a=rtpmap:3 gsm/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
SIP Invite - Response

Reliably Transmitting (NAT):
SIP/2.0 200 OK
Via: SIP/2.0/UDP
10.71.0.222:5060;branch=z9hG4bKc242AB12C03111DB8A1300112476567E;received=202.146.237.70;rport=5060
From: Jonny test <sip:49895600203.114.148.130>;tag=1386353914
To: <sip:0212304323@203.114.148.130>;tag=as77d3c840
Call-ID: C06D0E06-C031-11DB-8A13-00112476567E@10.71.0.222
CSeq: 32822 INVITE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER
Contact: <sip:0212304323@203.114.148.130>
Content-Type: application/sdp
Content-Length: 269

v=0
ox=root 26612 26612 IN IP4 203.114.148.130
s=session
c=IN IP4 203.114.148.130
t=0 0
m=audio 19918 RTP/AVP 8 0 3 101
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:3 GSM/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - -
SIP in Detail

• There’s much more to SIP than we can possibly hope to cover here

  • Go read the RFC!