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>Frequency</th>
<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>
<td>A</td>
</tr>
<tr>
<td>770 Hz</td>
<td>4</td>
<td>5</td>
<td>6</td>
<td>B</td>
</tr>
<tr>
<td>852 Hz</td>
<td>7</td>
<td>8</td>
<td>9</td>
<td>C</td>
</tr>
<tr>
<td>941 Hz</td>
<td>*</td>
<td>0</td>
<td>#</td>
<td>D</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

2-wire telephone line

ZB=RI=RO=Zline=600 ohm
Transformers: 1:1:1:1

4-wire input

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 its 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 Everwhere
## SIP Messages

<table>
<thead>
<tr>
<th>Informational</th>
<th>Request Failure</th>
</tr>
</thead>
<tbody>
<tr>
<td>100 Trying</td>
<td>400 Bad Request</td>
</tr>
<tr>
<td>180 Ringing</td>
<td>401 Unauthorised</td>
</tr>
<tr>
<td>181 Call forwarded</td>
<td>403 Forbidden</td>
</tr>
<tr>
<td>182 Queued</td>
<td>404 Not Found</td>
</tr>
<tr>
<td>183 Session Progres</td>
<td>405 Bad Method</td>
</tr>
<tr>
<td></td>
<td>415 Unsupported Content</td>
</tr>
<tr>
<td><strong>Success</strong></td>
<td>420 Bad Extensions</td>
</tr>
<tr>
<td>200 OK</td>
<td>486 Busy Here</td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Redirection</strong></td>
<td></td>
</tr>
<tr>
<td>300 Multiple Choices</td>
<td></td>
</tr>
<tr>
<td>301 Moved Perm.</td>
<td></td>
</tr>
<tr>
<td>302 Moved Temp.</td>
<td></td>
</tr>
<tr>
<td>380 Alternative Serv.</td>
<td></td>
</tr>
</tbody>
</table>
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:
- INVITE
- 100 Trying
- 302 (Moved Temporarily)
- RSVP
- 180 (Ringing)
- 180 (Ringing)
- 200 (OK)
- ACK
- ACK
- ACK

Media Path:
- RTP MEDIA PATH

Call Teardown:
- BYE
- 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="ea87d99f48b43a97b39819e3f6dfbf8b8",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=z9hG4bKC19D7202C03111DB8A1300112476567E
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
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:4989560@203.114.148.130>;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
o=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!