What is VoIP?
What is VoIP?

PC-to-PC telephone call – no PSTN

Tandem switch replacement (with or w/or IP) – Interexchange carriers.
What is VoIP - Voice on LAN: ethernet (ip)

IP-PBX: IP Phones on a LAN with or without an IP-PSTN WAN interface
What is VoIP – no more phones, no more switches?
What is VoIP/IP Telephony

• Voice is just another service provided by a converged network.

• Signaling and Media are carried as data inside of packets.
  – Bursty data rather than isochronous data
    • Water in buckets rather than flowing in pipe. Big deal for voice!
    • Bandwidth based on need & good effort – not fixed amount
  – PSTN telephony already had a notion of signaling coming in packets (SS7, which is X.25 like). 1970’s – 80’s concept.

• IP wins as the protocol/network of convergence.

• VoIP = IETF|IP + Convergence + Signaling Protocols (various flavors) + Media Protocols (RTP + Codec)
Why is VoIP Important!
What is the **Price** of a telephone call

- **Access Network**
  - Access network technology – POTS, mobile RAN, DSL, Cable modem, etc.

- **Transport & Termination Network**
  - Network Access Device (PSTN or Mobile Switch, Edge Router, etc.)
  - Transport network technology (SS7, IP, fiber/SONET, etc.). Termination refers either to disparate network connection (voip/pstn) or specific PSTN termination charges.

- **Access Network**
  - Network Access Device (PSTN or Mobile Switch, Edge Router, etc.)
  - Access network technology – POTS, mobile RAN, DSL, Cable modem, etc.

- **Rough answer** – add the access costs for the calling and called party with the cost for transport/termination.
What is the Price for a Telephone Call

• PSTN
  – Wired-to-Wired: Access, Transport, Termination are each about $0.01/min => $0.03/min (U.S. Domestic).
  – Wireless-to-Wired: Access and Termination are each about $0.03/min, Transport is $0.01/min => $0.07/min (U.S. Domestic).
  – Prices are typically the same or higher for voice calls outside of the US.

• VoIP Equivalent
  – Broadband Service Pricing (access + transport).
    • Wired broadband (DSL, Cable, FiOS) ~ $50/month for an average of 50 GBytes/month\(^1\) = $1/Gbyte.
  – G711 call is ~ 82 kb/sec * 60 sec/min = 615 kbytes/min
  – Wired-to-Wired: 2 * ($1/Gbyte * 615 kbytes/min) = $0.0013/min
  – Wireless-to-Wireless: 2 * ($6.25/Gbyte * 615 kbytes/min) = $.0077/min.
  – Prices are typically the same or higher outside of the US.
What is the Price for a Telephone Call: Morals

1. VoIP (arbitrage) has driven down the price of PSTN services by an order of magnitude over the last 10 years. Today, the price of PSTN and most other voice services is $\varepsilon$ greater than the cost of providing the service.

2. The PSTN model bundles in the costs for emergency services, lawful intercept (CALEA, pen-trap-trace), regulatory requirements on high availability, number portability, special taxes, support for rural areas, etc. The VoIP model (Skype, Gtalk, etc.) generally has none of this (recently, there has been pressure to support lawful intercept, I believe this is currently in place for Skype).

3. The Price of an OTT (Over The Top – internet based) VoIP call on a wired network is $\sim 1/30$ the price of the equivalent PSTN call made over a wired carrier network.

4. The Price of an OTT VoIP call on a wireless network is $\sim 1/10$ the price of the equivalent PSTN call made over a wireless carrier network.

5. Carriers (both fixed and mobile) bundle voice with other services, and use flat rate pricing – to obscure the price and prevent competition/comparison with OTT services.

6. Because OTT providers typically provide “free” service, they can only monetize additional services (voicemail, off-net calling, etc.)

7. In the future: wideband mobile codecs, video, presence, advanced messaging .. Does anyone care, will anyone pay? Is it just packets?
Why is VoIP/IP Telephony Important?

- **Price:** Completely remade the business model for probably the most important app in the world – in 10 years.

- **Power:** The model for control/intelligence changes:
  - PSTN: Intelligence in the center, dumb stuff at the edges, the Ma Bell mentality. *High barrier to entry* ($$, legal, regulatory).
  - VoIP: Intelligence at the edges, dumb stuff in the center, #$%& the telephone company. *Low barrier to entry* ($$, legal, regulatory).

- Levels the playing field across the world!  
  - {IP + mobile}

- Email, http://, IM, …VoIP

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*Paradigm Change / Disruptive Force!*

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Competitive Landscape: 2005 - 2013

**Wireless Handsets**
- Nokia dominates in GSM (50% market share) feature phones.
- Qualcomm owns the CDMA space (chips, patents on spread spectrum).
- RIM dominates the enterprise market (keyboard, email through their servers).
- Samsung (i900) and BlackBerry (NASDAQ) dominate the QIS, with RIM and Nokia having their own proprietary QSS.
- Motorola falling. Can't compete with Nokia and Asian ODMs.

**New Entrants**
- Motorola: MDI and Android.
- Unwired - Liquid Sound.
- LG, NTT, Casio.
- CPE: VoIP Endpoints (phones, ATAs, etc.)
- Nokia, Microsoft, Google, with new QSS.
- Other QSS companies.

**New Entrants**
- Samsung, LG, HTC, Sony Ericsson.
- Microsoft: Java, Symbian.
- Samsung, LG, HTC, Sony Ericsson.

**CPE**
- Recovery from Internet bubble burst 2000.
- Tradelines/wireline carrier consolidation.
- Wireless FIOS, QST, Skype, Google, Nokias/Qualcom, Cisco.
- Market consolidation.

**VoIP Endpoints**
- Recovery from Internet bubble burst 2000.
- Tradelines/wireline carrier consolidation.
- Wireless FIOS, QST, Skype, Google, Nokias/Qualcom, Cisco.
- Market consolidation.

**Network /Service Providers**
- Traditional Providers
  - AT&T, T-Mobile, Verizon, Sprint.
- MSOs (Time Warner, Comcast, etc.): TV to the headend, DOCSIS architecture.
- Wireless Carriers take off - begin to dominate with wireline carriers.
- OTT/Network (VoIP/YouTube, etc.)
- Many small (local) margin IP Centrex OTT providers.

**Equipment Vendors**
- Carrier: Cisco (virtually integrated like traditional vendors).
- Many startups/new entrants (Broadsoft, Telia, Sylantro, etc.).
- SBC vendors (Acmia, Nextera, etc.).
- Chinese vendors (hawaii, ZTE) accused of copying and stealing IP.
- Enterprise: Microsoft IP-PBX, Cisco IP-PBX, Asterisk.

**Wireless Providers**
- Microsoft buys Skype.
- Google Voice.
- Facebook.
- Nortel.
- RIM.

**4G Deployments**
- 4G LTE trials.
- New LTE trials.

**Global Recession**
- 2007-2009
- 2010-2011

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So – what is VoIP? Let’s try and example: “Hello World”
Hello World – with a proxy

- End-points (user agents, telephones, etc.) can have IP address – which presumably the “network” knows how to route packets.
- End-points can have other identifiers – such as telephone numbers or sip URLs – which somehow must be resolved into an IP address in order to route packets. A proxy performs that function – sort of a DNS server for telephone numbers and/or sip URLs.
Hello World

Listen for messages on port 5060 – 192.168.0.100:5060

Listen for messages on port 5060 – 192.168.0.101:5060

IP Phone: 192.168.1.100

Port for RTP negotiated Between SIP endpoints (user agents) 192.168.0.100:8023

Port for RTP negotiated Between SIP endpoints (user agents) 192.168.0.101:80**

Signaling messages: protocol call SIP

Media (voice): protocol called RTP
Hello World – Ladder Diagram

User picks up the phone and dials the remote phone (by IP address)

INVITE
100 TRYING
180 RINGING
200 OK
ACK

Talking

User hangs up the phone

BYE
200 OK

IP phone plays ring
User picks up the phone, is ready to talk
Talking

User gets a disconnect

No more RTP
Hello World - Morals

- An IP-telephone call is just a series of packets going between computers.
- Call Control:
  - Messages between computers setup and tare down calls.
  - Messages are sent to ip addresses with well known ports. The messages have a predefined syntax called a “protocol.”
  - Software on computer(s) listens for messages on the well defined ports, and does stuff like saying “OK, send me the call”, or “BYE, I want to hang up”, or what “codec” was used to encode the call, or many many other things.
  - There are many different protocols for in VoIP for call control: H323, SIP, MGCP, etc.
  - Call control packets can be either UDP (best effort), or TCP (in order, acknowledged).
- Media (voice):
  - The media (the voice) is sent between ip address on ports that are negotiated during call setup. The ports do not have to be symmetric.
  - There are many difference codecs (or ways to encode the voice stream). They are typically independent of the signaling protocol (except that they are often negotiated by the signaling protocol). A codec is a recipe for turning the voice from a microphone into binary samples (or from samples into a voltage/current to a speaker).
  - The protocol for the voice data packets is called RTP - which was defined before IP Telephony existed. RTP defines how the bits are stored (and some other stuff), but it doesn’t define the codec.
  - RTP is always UDP!
VoIP Gateway: Interface between PSTN and IP

- **Signaling:** Q.931 messages on the D channel (ISDN LAPD). (Could be CAS T1, could be analog.)
- **Media:** Isochronous, nicely clocked, completely regular, 64kb/s G711.
- **Visibility** to other entities in the PSTN: “What entities? The switch is my master. I have no free will.”

- **Signaling:** Variety of protocols (SIP, H323, MGCP, RAS, Radius) for a variety of reasons.
- **Media:** Pick a codec, stuff the media into an RTP packet, send out the packet, where it will get buffered with other packets.
- **Visibility** to other IP entities: “an endpoint with personality.”
Signaling Example

Switch (PBX) → Q931 → VoIP Gateway → SIP → SIP User Agent (IP Phone)

Call Setup
Call Preceding
Alerting
Connect
Connect Ack
Media
Disconnect
Release
Release complete

Invite
Trying
Ringing
OK (resp to invite)
RTP (media)
BYE
OK

Dial → Play ring
Hang up → Play ring
Hang up
Call setup detail (1)

Switch

VoIP Gateway

SIP UA (IP Phone)

---

Call setup

Oct 14 16:03:57.657 UTC: ISDN Se1:23: RX ← SETUP pd = 8 callref = 0x0E25
Oct 14 16:03:57.657 UTC: Bearer Capability i = 0x9090A2
Oct 14 16:03:57.657 UTC: Channel ID i = 0xA18393
Oct 14 16:03:57.657 UTC: Calling Party Number i = 0xA1, '7818659218', Plan: ISDN Type: National
Oct 14 16:03:57.661 UTC: Called Party Number i = 0xA1, '7818656564', Plan: ISDN Type: National
Oct 14 16:03:57.661 UTC: Locking Shift to Codeset 6
Oct 14 16:03:57.661 UTC: Codeset 6 IE 0x28 i = '*'Conf', 0x20, 'Rm', 0x20, '27/331'

Call Proceeding

Oct 14 16:03:57.677 UTC: ISDN Se1:23: TX -> CALL_PROC pd = 8 callref = 0x8E25
Oct 14 16:03:57.677 UTC: Channel ID i = 0xA98393
Oct 14 16:03:57.677 UTC: Sent:

SIP Invite

Oct 14 16:03:57.709 UTC: Sent:

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Call setup detail (2)

Switch  VoIP Gateway  SIP UA (IP Phone)

Oct 14 16:03:57.733 UTC: Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 171.78.205.4:5060
From:"Conf Rm 27/331" <sip:7818659218@171.78.205.4;tag=0E874B00-A0A>
To: <sip:7818656564@171.78.205.32;user=phone>
Call-ID: 615772B9-DEC511D6-AB63AE84-CC739108@171.78.205.4
CSeq: 101 INVITE
Content-Length: 0

Oct 14 16:03:58.317 UTC: Received:
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 171.78.205.4:5060
From:"Conf Rm 27/331" <sip:7818659218@171.78.205.4;tag=0E874B00-A0A>
To: <sip:7818656564@171.78.205.32;user=phone;tag=299591094-1034611437544>
Call-ID: 615772B9-DEC511D6-AB63AE84-CC739108@171.78.205.4
CSeq: 101 INVITE
Content-Length: 0
RPI-D-Privacy: id-type=subscriber;party=called;privacy=off
Remote-Party-ID:"Elliot Eichen"<sip:7818656564@171.78.205.31;user=phone;screen=yes;party=called;privacy=off;id-type=subscriber

Oct 14 16:03:58.321 UTC: ISDN Se1:23: TX -> ALERTING pd = 8 callref = 0x8E25
Oct 14 16:03:58.321 UTC: Progress Ind i = 0x81BB - In-band info or appropriate now available
Oct 14 16:04:07.093 UTC: Received:
Call setup detail (3)

Switch → VoIP Gateway → SIP UA (IP Phone)

OK (resp to invite)

ACK (I’m OK, you’re OK)

Connect

Connect ACK
Call setup detail (3)

Oct 14 16:04:23.837 UTC: ISDN Se1:23: RX <- DISCONNECT pd = 8 caloref = 0x0E25

Oct 14 16:04:23.841 UTC: ISDN Se1:23: TX -> RELEASE pd = 8 caloref = 0x0E25

Oct 14 16:04:23.877 UTC: ISDN Se1:23: RX <- RELEASE_COMP pd = 8 caloref = 0x0E25

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SIP, H.323 and MGCP

H.323 Version 1 and 2 supports H.245 over TCP, Q.931 over TCP and RAS over UDP.
H.323 Version 3 and 4 supports H.245 over UDP/TCP and Q.931 over UDP/TCP and RAS over UDP.
SIP supports TCP and UDP.
RTP

• Signaling can be mediated by devices in the middle of the network (SIP Proxies, softswitches, gatekeepers, etc.).
  – For SIP, this is the typical call flow. (draw ex.)
  – For H323, gatekeeper direct is typical, which means signaling typically happens between endpoints. (draw ex.)

• RTP (media) often occurs between the endpoints, although there are important instances where this is not the case.
  – Some enterprise (cisco call manager): firewall issue
  – Blind re-file (wholesale)
RTP

- RTP – carries real time data.
- RTPC – control port that carries information to monitor quality of service.
+1 port
Voice Codecs – Bandwidth Consumption

Analog waveform → Pulse Code Modulation → Codec → Buffer

64 kbps → 6-16 kbps → ~10-40 msec

Layer 3 (RTP/UDP/IP) + Layer 2 Encapsulation

Physical Layer (wire) → 40 bytes/sec

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How this all works?

Analog waveform → Sampled → Linearized to PCM → 64 kbits/sec

\[ R = \text{Samples/sec} \]
\[ B = \text{Bytes/sample} \]

Typically for voice:
\[ R = 8 \text{K samples/sec} \]
\[ B = 8 \text{ bits/sample} \]

\[ BW = RB = 64 \text{ kbits/sec} \]
How this all works? (2)
Bandwidth requirement: G729 Example

- For G729, voice is 8 kbits/sec. Assume 20ms payload.
- Packet payload size = 8 kbits/sec X 20ms = 160 bits = 20 bytes/packet. IP Packet = 60 bytes.

<table>
<thead>
<tr>
<th>IP 20 bytes</th>
<th>UDP 8 bytes</th>
<th>RTP 12 bytes</th>
<th>G729 Payload 20 bytes</th>
</tr>
</thead>
</table>

- For Cisco HDLC (cisco-cisco router serial), add 5 bytes header, 3 bytes trailer. Bandwidth = (63 bytes/20ms)*(8 bits/byte) = 25.2 kbps

<table>
<thead>
<tr>
<th>HDLC 5 bytes</th>
<th>IP 20 bytes</th>
<th>UDP 8 bytes</th>
<th>RTP 12 bytes</th>
<th>G729 Payload 20 bytes</th>
<th>HDLC 3 bytes</th>
</tr>
</thead>
</table>

- Efficiency = 8 kb/s / 25.2 kb/s ~ 32%. Number of packets/sec = 1sec/.02sec. Loss of 1 packet/sec = 2% packet loss!

\[ \Delta \text{Delay} = \text{buffer (20ms)} + \text{codec delay} + \text{layer2/3 delay} + \text{queuing delay} + \text{codec delay} \]
Bandwidth requirement: G729 Example #2

- Assume 60 ms packets.
- Packet payload size = 8 kbits/sec X 60ms = 60 bytes/packet. IP Packet = 60 bytes.

<table>
<thead>
<tr>
<th>IP 20 bytes</th>
<th>UDP 8 bytes</th>
<th>RTP 12 bytes</th>
<th>G729 Payload 60 bytes</th>
</tr>
</thead>
</table>

- For ethernet, 14 byte header, 4 byte trailer => bandwidth = (118 bytes/60ms)*(8bits/byte) = 15.7 kb/s

<table>
<thead>
<tr>
<th>Ethernet 14 bytes</th>
<th>IP 20 bytes</th>
<th>UDP 8 bytes</th>
<th>RTP 12 bytes</th>
<th>G729 Payload 60 bytes</th>
<th>Ethernet 4 bytes</th>
</tr>
</thead>
</table>

- Efficiency = 8 kb/s / 15.7 kb/s ~ 50%. Number of packets/sec = 1sec/.06sec. Loss of 1 packet/sec = 6% packet loss!
Header Compression Example

- Conventional frame G729 = 25.2 kbps

<table>
<thead>
<tr>
<th>HDLC 5 bytes</th>
<th>IP 20 bytes</th>
<th>UDP 8 bytes</th>
<th>RTP 12 bytes</th>
<th>G729 Payload 20 bytes</th>
<th>HDLC 3 bytes</th>
</tr>
</thead>
<tbody>
<tr>
<td>HDLC 5 bytes</td>
<td>CRTP 4 bytes</td>
<td>G729 Payload 20 bytes</td>
<td>HDLC 3 bytes</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- Compressed frame G729 = 12.8 kbps
Bandwidth reduction

- IP – almost by definition - is less bw efficient than PSTN.
- Bandwidth reduction where the cost for bandwidth is at a premium – typically tail circuits, or some international destinations.
- VAD (Voice Activity Detection). Rule of thumb is 1/3-1/2 reduction in bw.
  - Cost is voice quality (clipping), poor performance in conferencing.
  - Comfort noise to make the listener feel better.
- Header Compression
  - Compressed RTP: Hop-to-hop, requires decompression and compression at each hop. For G279, = 12.8 kb/s
  - VoMPLS: New, not sure of what it takes beyond MPLS. End-to-end (not hop-to-hop), likely to be much less router CPU intensive.
## Codec Table

<table>
<thead>
<tr>
<th>codec</th>
<th>data rate [kbit/s]</th>
<th>default sample (frame) size [ms]</th>
<th>published MOS*</th>
<th>Genuity PAMS*</th>
<th>BWCisco serial [kbit/s]</th>
<th>BWFE [kbit/s]</th>
<th>complexity</th>
<th>pub domain</th>
<th>comments</th>
</tr>
</thead>
<tbody>
<tr>
<td>G711</td>
<td>64</td>
<td>8KHz PSTN, 10ms IP</td>
<td>4.1</td>
<td>4.32</td>
<td>83.2</td>
<td>87.2</td>
<td>L</td>
<td>Y</td>
<td>standard PSTN codec</td>
</tr>
<tr>
<td>G726</td>
<td>16-32</td>
<td>8kHz PSTN 10 ms IP</td>
<td>3.51 [24 kb/s]</td>
<td>43.2</td>
<td>47.2</td>
<td>L</td>
<td>Y</td>
<td></td>
<td>lower bit rate PCM-PSTN codec. Sometimes used on international calls</td>
</tr>
<tr>
<td>G729a aka G729r8</td>
<td>8</td>
<td>10</td>
<td>3.8</td>
<td>3.86</td>
<td>25.2</td>
<td>31.2</td>
<td>M</td>
<td>N</td>
<td>voip codec. Happy medium in quality/complexity? Open source available (non commercial use only), otherwise $$</td>
</tr>
<tr>
<td>G723</td>
<td>5.3,6.3</td>
<td>30</td>
<td>3.33 [5.3 kb/s]</td>
<td>24.5</td>
<td>28.5</td>
<td>H</td>
<td>N</td>
<td></td>
<td>lower bit rate VoIP codec. Often used for PC to PSTN telephony</td>
</tr>
<tr>
<td>GSM</td>
<td>13</td>
<td>20</td>
<td>3.5</td>
<td></td>
<td></td>
<td>L</td>
<td>?</td>
<td></td>
<td>cellular standard codec. VoIP/Cell interworking</td>
</tr>
<tr>
<td>iLBC</td>
<td>15.2</td>
<td>20</td>
<td>3.8 - 3.9?</td>
<td>34.4</td>
<td>38.4</td>
<td>Y</td>
<td></td>
<td></td>
<td>GIPS free codec. Claim of higher immunity to packet loss</td>
</tr>
</tbody>
</table>
What is iLBC?

iLBC (internet Low Bitrate Codec) is a FREE speech codec suitable for robust voice communication over IP. The codec is designed for narrow band speech and results in a payload bit rate of 13.33 kbps with an encoding frame length of 30 msec and 15.20 kbps with an encoding length of 20 msec. The iLBC codec enables graceful speech quality degradation in the case of lost frames, which occurs in connection with lost or delayed IP packets.

Features

- Bitrate 13.33 kbps (399 bits, packetized in 50 bytes) for the frame size of 30 msec and 15.2 kbps (303 bits, packetized in 38 bytes) for the frame size of 20 msec.
- Basic quality higher than G.729A, high robustness to packet loss.
- Computational complexity in a range of G.729A.
- Royalty Free Codec.
## Codecs – PacketCable Spec

Codecs defined in this specification MUST be encoded with the following string names in the rtpmap parameter:

### Codec RTP Map Parameters

<table>
<thead>
<tr>
<th>Codec</th>
<th>Literal Codec Name</th>
<th>RTP Map Parameter</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711 μ-law</td>
<td>PCMU</td>
<td>PCM/8000</td>
</tr>
<tr>
<td>G.711 A-law</td>
<td>PCMA</td>
<td>PCMA/8000</td>
</tr>
<tr>
<td>iLBC</td>
<td>iLBC</td>
<td>iLBC/8000</td>
</tr>
<tr>
<td>BroadVoice16</td>
<td>BV16</td>
<td>BV16/8000</td>
</tr>
<tr>
<td>G.726 at 16kb/s</td>
<td>G726-16</td>
<td>G726-16/8000</td>
</tr>
<tr>
<td>G.726 at 24kb/s</td>
<td>G726-24</td>
<td>G726-24/8000</td>
</tr>
<tr>
<td>G.726 at 32kb/s</td>
<td>G726-32</td>
<td>G726-32/8000</td>
</tr>
<tr>
<td>G.726 at 40kb/s</td>
<td>G726-40</td>
<td>G726-40/8000</td>
</tr>
<tr>
<td>G.728</td>
<td>G728</td>
<td>G728/8000</td>
</tr>
<tr>
<td>G.729A</td>
<td>G729</td>
<td>G729/8000</td>
</tr>
<tr>
<td>G.729E</td>
<td>G729E</td>
<td>G729E/8000</td>
</tr>
</tbody>
</table>

### Table Note:
- Mandatory codecs – G.711 (μ-law and A-law), iLBC, BV16
- Recommended codecs – G.728 and G.729 Annex E
- Optional codecs (for informational purposes only) – G.726 and G.729A

For the use in the SDP, the Rtpmap parameter (i.e., PCMU/8000 in the case of μ-law) is used. Unknown Rtpmap parameters SHOULD be ignored if they are received.
.wav

- Microsoft/IBM Spec
- PCM with bandwidth = SxBxC
  - S = Sample Rate (samples/sec)
  - B = Bits/Sample
  - C = Channels
- For example, audio might be 44 kHz sampling rate at 16 bits/sample and 2 channels = 1.4 Mb/s.
- Another example might be voice mail in an email attachment (try it – right click on a .wav file) – 8kHz sampling and 16 bits/sample = 124 kb/s.
- Conversion between audio formats! Interesting (how do you go from compressed format to un-compressed formats).
Tradeoffs –

a: Efficiency vs Latency & Packet Loss
b: Codec bandwidth vs CPU cycles

• Short Packets = low latency, tolerates packet loss better, inefficient.
• Large Packets = long latency (got to wait for buffer to fill up before shipping out packet), does not tolerate packet loss as well (loss of many bytes hurts!), more efficient.
• Narrowband high quality codecs => more processing (cpu cycles) than either poor quality or less compression.
• Why not something BETTER than G711?
Voice Codecs – Bandwidth Consumption & Delay

Analog waveform → A/D - Pulse Code Modulation → Codec (compression) → Buffer

- 64 kbps
- delay ~0-10 msec

Codec (expansion) → PCM → D/A

- 40 bytes/sec
- delay = 1 packet ~10-40 msec
- ~1-2 packets
- delay = 1-2 packet ~20-80 msec

Getting from pt A to pt B → Layer 2/3

Layer 3 (RTP/UDP/IP) + Layer 2 Encapsulation

Transport delay → Physical Layer (wire)
Distances (NY to...) & One Way Delay

<table>
<thead>
<tr>
<th></th>
<th>km</th>
<th>msec(^1)</th>
<th>typical(^2)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Los Angeles</td>
<td>5172</td>
<td>17</td>
<td>35</td>
</tr>
<tr>
<td>Sydney</td>
<td>17409</td>
<td>58</td>
<td>106</td>
</tr>
<tr>
<td>Tokyo</td>
<td>11552</td>
<td>39</td>
<td>50.5</td>
</tr>
<tr>
<td>Frankfurt</td>
<td>6326</td>
<td>21</td>
<td>50</td>
</tr>
</tbody>
</table>

- Discrepancy Between Calculated and Actual Delays
  - Great Circle vs. as the railroads/cables run.
  - Multiple Hops
  - Congestion
    1 great circle route
    2 http://ipnetwork.bgtmo.ip.att.net/pws/network_delay.html
Marginal delay - Example

\[ \Delta \text{Delay} \text{ is the difference between the PSTN/circuit and the IP time} \]
\[ \Delta \text{Delay} = \text{buffer} + \text{compression} + \text{layer2/3} + \text{queuing} + \text{network congestion} + \text{jitter} + \text{expansion}. \]

\[ \text{If buffer} = 20\text{ms}, \text{compression} = 5\text{ms}, \text{layer2/3} = 2\text{ms}, \text{queuing} = 3\text{ms}, \text{jitter} = 20\text{ms}, \text{expansion} = 5\text{ms} \Rightarrow \Delta \text{Delay} \approx 65\text{ms}. \]

\[ \text{Rule of thumb} \Rightarrow \text{Total Delay} < 150\text{ms for Toll Quality. If NY to LA is 50 ms, it’s OK. If NY to Sydney = 100 ms, it’s marginal}. \]

\[ \text{Transcoding} = \text{twice the } \Delta \text{Delay} \Rightarrow \text{transcoding is bad!} \]
Transcoding – From codec to codec

GSM isochronous (~16 kb/s) → Cell Phones → Codec (de-compress) → Codec (compress) → Buffer

G723 (16 kb/s) → Dial-Up IP → Codec (de-compress) → Codec (compress) → Buffer

IP → VoIP → Magic Popper → IP → VoIP → Magic Popper → IP

64 kbit/s
Jitter- not such a big deal in modern networks?

- **Why is Jitter Important?**
  - If jitter > receive buffer size, jitter = packet loss.

- **Tradeoff between Receive Buffer Size and Latency.**
  - Bigger Receive Buffer => Better immunity to packet jitter .. But .. longer latency!

- **It all depends .. but it is typical to have a +/- 1 packet jitter buffer.**
Summary (PAMS listening score only)
Why VoIP so cool?

• The codecs make the voice sounds lousy.
• It’s less efficient than circuit switched voice.
  – At least in some sense it’s less efficient (no ip overhead).
• It has all these QoS issues (latency/packet loss).
• It’s driven down the price that can be charged for a call so that there are no $$ to be made.