How IP Telephony Breaks the Internet

Assumptions

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Sprint IP Retreat

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Overview

- VoIP = [traffic/QoS], signaling, services
- reliability issues
- breaking the Internet architecture
VoIP

- carrying voice (and multimedia) over IP
- strict separation signaling – media traffic (↔ PSTN)
- future: high-rate codecs, video
- (typically) *not* PC-based voice
- starting to displace traditional PBX in greenfield installations
- likely to see widespread use in 3G (UMTS R5) wireless
Example: Pingtel SIP phone
Example: Cisco and 3Com SIP phones
Example: Columbia CS Phone System

Expand existing PBX via IP phones, with transparent connectivity
## The phone works — why bother with VoIP?

<table>
<thead>
<tr>
<th>user perspective</th>
<th>carrier perspective</th>
</tr>
</thead>
<tbody>
<tr>
<td>• variable compression: tin can to broadcast quality</td>
<td>• silence suppression  (\Rightarrow) traffic ↓</td>
</tr>
<tr>
<td>• security through encryption</td>
<td>• shared facilities  (\Rightarrow) management, redundancy</td>
</tr>
<tr>
<td>• caller, talker identification</td>
<td>• advanced services (simpler than AIN and CTI)</td>
</tr>
<tr>
<td>• better user interface</td>
<td>• separate fax, data, voice</td>
</tr>
<tr>
<td>• internat. calls: TAT transatlantic cable = $0.03/hr</td>
<td>• cheaper switching</td>
</tr>
<tr>
<td>• local calls: possibly cheaper (local access fees)</td>
<td>• better management platforms</td>
</tr>
<tr>
<td>• easy: video, whiteboard, …</td>
<td></td>
</tr>
</tbody>
</table>
## Audio Codecs

<table>
<thead>
<tr>
<th>Codec</th>
<th>rate</th>
<th>quality (MOS)</th>
<th>min. delay</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.723</td>
<td>5.3</td>
<td>3.7</td>
<td>37.5 ms</td>
</tr>
<tr>
<td></td>
<td>6.3</td>
<td>3.98</td>
<td>37.5 ms</td>
</tr>
<tr>
<td>G.729</td>
<td>8.0</td>
<td>4</td>
<td>15 ms</td>
</tr>
<tr>
<td>AMR</td>
<td>4.75-12.2</td>
<td></td>
<td>20 ms</td>
</tr>
<tr>
<td>AMR-WB</td>
<td>6.6-23.85</td>
<td>7 kHz</td>
<td></td>
</tr>
<tr>
<td>G.728</td>
<td>16.0v</td>
<td>4</td>
<td>5.625 ms</td>
</tr>
<tr>
<td>G.722</td>
<td>32.0</td>
<td>7 kHz</td>
<td>40 ms</td>
</tr>
<tr>
<td>G.711</td>
<td>64.0</td>
<td>μ-law, MOS 4.3</td>
<td>var.</td>
</tr>
</tbody>
</table>
Voice and Data Traffic

May 2001
Objective vs. Subjective MOS

Objective MOS tools don’t always handle loss impairments correctly:

![Graph showing Objective MOS correlation]
Traffic (1998)

Measured in Dial Equipment Minutes (DEM) or bandwidth:

<table>
<thead>
<tr>
<th></th>
<th>GDEM</th>
<th>bandwidth (Gb/s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Local</td>
<td>2986</td>
<td>364</td>
</tr>
<tr>
<td>Intrastate toll</td>
<td>422</td>
<td>51</td>
</tr>
<tr>
<td>Interstate toll</td>
<td>555</td>
<td>68</td>
</tr>
</tbody>
</table>

PBX: typically, about 10% utilization per phone ➔ 6.4 kb/s per employee (128 Mb/s for 20,000 person campus)
Call Attempts

- completed (70.7%)
- did not answer (12.7%)
- busy (10.1%)
- customer error (1.6%)
- equipment failure (1.9%)
- no response (2.6%)
- number invalid (0.4%)

May 2001
Call Setup Delay

off-hook

start dialing

ringback

pick up

disconnect
no answer

busy disc.

May 2001
The Three-Minute Myth

Local calls are about 2.4 minutes on average, but long distance calls are much longer, about 8.9 minutes:

May 2001
## Calls Get Longer with Distance

<table>
<thead>
<tr>
<th>distance (mi)</th>
<th>% calls</th>
<th>duration (min.)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1 – 10</td>
<td>5.1</td>
<td>4.6</td>
</tr>
<tr>
<td>11 – 22</td>
<td>20.2</td>
<td>5.1</td>
</tr>
<tr>
<td>23 – 55</td>
<td>23.2</td>
<td>5.9</td>
</tr>
<tr>
<td>56 – 124</td>
<td>13.3</td>
<td>7.7</td>
</tr>
<tr>
<td>125 – 292</td>
<td>12.1</td>
<td>9.4</td>
</tr>
<tr>
<td>293 – 430</td>
<td>4.6</td>
<td>10.4</td>
</tr>
<tr>
<td>431 – 925</td>
<td>9.7</td>
<td>11.9</td>
</tr>
<tr>
<td>926 – 1910</td>
<td>8.5</td>
<td>11.9</td>
</tr>
<tr>
<td>&gt; 1910</td>
<td>3.2</td>
<td>11.2</td>
</tr>
<tr>
<td>average</td>
<td>310</td>
<td>7.8</td>
</tr>
<tr>
<td>median</td>
<td>60</td>
<td>3.0</td>
</tr>
</tbody>
</table>
Aside: Cost of Bandwidth

- T3 Internet access: $16,000/month
- or 0.05c/minute (for 64 kb/s) for full utilization (bogus)
- typically, assume peak-to-average ratio of 4 (17% during busy hour) ➔ 0.2c/minute
- may be better if data and voice load are offset
- lack of current traffic statistics
Why Aren’t We Junking Switches Right Now?

What made other services successful?

**email:** available within self-contained community (CS, EE)

**web:** initially used for local information

**IM:** instantly available for all of AOL

All of these . . .

- work with bare-bones connectivity ($\geq 14.4$ kb/s)
- had few problems with firewalls and NATs
- don’t require a reliable network
Reliability Issues

- Software updates require “scheduled downtime”
- But signaling servers can be made redundant much easier than SS7 SCPs
- BGP convergence times of several minutes: 2 minutes to withdraw routes, 30 minutes to advertise routes
- “80% of withdraws take more than a minute”
- No clear IP reliability definition – reachability of any node? Some large subset? “Local calls”? 
BGP Convergence Times

(From Abha Ahuia’s IETF50 plenary talk and Geoff Huston’s talk)

Failures, Fail-overs and Repairs
Reliability: Power

- more decentralized ➞ harder to provide power coverage
- need power for Ethernet switches, phones – ≈ 7W/phone (48V)
- Ethernet powering (spare pairs), tandem or integrated into switch
- also useful for wireless base stations
- Columbia approach: separate power circuit for wiring closets
Reliability: Denial-of-Service

- denial-of-service and attacks more likely than with traditional phones
- but traditional phones (including 800#) also subject to auto-dialers
- different scenarios:
  - external attack ➤ can be filtered
  - internal compromise ➤ spoof DiffServ, RSVP
- disadvantage of integration: no secondary channel
- thus, maybe keep authorized RSVP “circuits”
Sources of Delay

- **Packetization** (10–30 ms)
- **Encoding** (5–10 ms)
- **Operating System**

**End-system**

**IF**

**Route Lookup**

**Classification**

**Output Buffer**

**Output Line**

**Switching delay**

**Transmission**

**Queueing**

**Processing**

**Load**

**Delay**

**Router**

**Propagation** (5 us/km)

**Routing Table**

**Output Buffer**

**Output System**

**Soundcard**

**Playout Buffer**

**Decoding**

**Operating System**
QoS: Local Area Network

- typically, very low average utilization (few %)
- very little packet loss (a few packets a day)
- but long delay spikes (300 ms) due to Ethernet collisions if heavy file transfer
- ➠ avoid hubs, even for single office
- ➠ Ethernet prioritization
QoS: Access Network

- usually, bottleneck (1:10 concentration)
- usually, asymmetrically loaded, depending on web traffic
- solution: TOS marking (supported by most phones)
QoS: Wide-Area Network

- existing SLAs and measurements mostly useless: just averages
- e.g.: steady loss of 5% acceptable, one-second bursts of 20% not
- application loss = f(network loss, FEC, jitter, playout delay)
- need rough equivalent of “severely errored seconds”
- however, bursts of loss ➞ interruptions
- two types of carriers: “classical IP” vs. “voice heritage”?
Resource reservation

- airline vs. subway: reserve if > 1% of bottleneck?
- resource reservation likely for upstream cable channel
- RSVP far too complex simple end systems and without multicast
- separate problem: need reserve/commit for VoIP? coupling with application-layer signaling?
- no harm in having several resource reservation protocols
- congestion pricing (RNAP, M2I), including holding costs
Example: Adaptively Virtual Exponential Average

Exp-avg vs. Its Extension

Application Loss Probability

Network Loss Probability

Exp-avg
Exp-avg (add (N-1)*pkt-length)
Exp-avg Ext
Example: Playout Delay

Exp-avg vs. Its Extension

Average End to End Delay (seconds)

Network Loss Probability

Exp-avg
Exp-avg (add (N-1)*pkt-length)
Exp-avg Ext

May 2001
VoIP breaks architectural assumptions underlying recent additions:

- NATs: only work for client-server (and TCP)
- VPNs, mobile IP: encapsulation overhead
- firewalls: assume clients inside, servers outside
Conclusion

- motivation for VoIP
- traffic characteristics
- QoS metrics
- new resource reservation mechanisms?