Internet Telephony: Status and Directions

Henning Schulzrinne
Columbia University, New York
schulzrinne@cs.columbia.edu

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Overview

- new Internet services: “telephone”, “radio”, “television”
- why Internet telephony?
- why not already?
- Internet telephony modalities
- components needed:
  - data transport
  - resource reservation
  - signaling
  - service location
New Internet services

- tougher: replacing dedicated electronic media
- typewriter model of development
- yet another convergence?
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</td>
<td>• silence suppression → traffic ↓</td>
</tr>
<tr>
<td>can to broadcast quality</td>
<td>• shared facilities → management, redundancy</td>
</tr>
<tr>
<td>• security through encryption</td>
<td>• advanced services (simpler than AIN and CTI)</td>
</tr>
<tr>
<td>• caller, talker identification</td>
<td>• operational advantages</td>
</tr>
<tr>
<td>• better user interface</td>
<td>• cheaper switching</td>
</tr>
<tr>
<td>• internat. calls: TAT transatlantic cable = $0.03/hr</td>
<td>• fax as data</td>
</tr>
<tr>
<td>• no local access fees (3.4c)</td>
<td></td>
</tr>
<tr>
<td>• easy: video, whiteboard, . . .</td>
<td></td>
</tr>
</tbody>
</table>
The new phone companies

- separation bit carriage ↔ services
- anybody with Internet connection can provide services (ACD, 800, 900, directory, ...)
- distinction “in” vs. “out” of network not useful
- incremental start-up investment not large
- new players:
  - cable companies ➞ no new infrastructure, but mostly one-way
  - electric utilities ➞ need line management anyway
  - Qwest, IXC (resell to ISPs), ...
Internet telephony as PBX replacement

global Internet not quite ready ⇐ try as PBX

- have mission-critical LAN, PCs anyway
- usually ample (if switched) bandwidth, low latency
- packet switching is cheaper
- network PCs $\Rightarrow$ ISDN phones
- no need for billing
Internet telephony services

- voice mail → email
- calendar integration
- user-programmable call processing logic
- call first available sales person (ACD)
- call whole department
- web IVR
- return web page with favorite “on hold” music
Internet telephony services

- camp-on without holding a line
- short message service ("instant messaging")
- schedule call into the future
- call with expiration date
- add/remove parties to/from call ➔ mesh
- "buddy lists"
## Switching costs

<table>
<thead>
<tr>
<th>switching method</th>
<th>ports</th>
<th>Gb/s</th>
<th>cents/KB</th>
<th>$/interface</th>
</tr>
</thead>
<tbody>
<tr>
<td>10BaseT hub</td>
<td>16</td>
<td>0.16</td>
<td>0.1</td>
<td>9.4</td>
</tr>
<tr>
<td>100BaseT hub</td>
<td>16</td>
<td>1.6</td>
<td>0.05</td>
<td>46</td>
</tr>
<tr>
<td>10BaseT switch</td>
<td>24</td>
<td>0.24</td>
<td>1.2</td>
<td>121</td>
</tr>
<tr>
<td>100BaseTX switch</td>
<td>8</td>
<td>0.80</td>
<td>0.15</td>
<td>156</td>
</tr>
<tr>
<td>router</td>
<td></td>
<td>2.1</td>
<td>16.0</td>
<td></td>
</tr>
<tr>
<td>local ATM switch</td>
<td>16</td>
<td>2.48</td>
<td>1.0</td>
<td>1581</td>
</tr>
<tr>
<td>PBX</td>
<td>256</td>
<td>0.02</td>
<td>218.6</td>
<td>140</td>
</tr>
<tr>
<td>5ESS local (no AIN)</td>
<td>5000</td>
<td>0.32</td>
<td>469.1</td>
<td>300</td>
</tr>
<tr>
<td>5ESS local (AIN)</td>
<td>20000</td>
<td>1.28</td>
<td>273.2</td>
<td>175</td>
</tr>
<tr>
<td>4ESS toll</td>
<td>100000</td>
<td>6.40</td>
<td>7.8</td>
<td></td>
</tr>
</tbody>
</table>
Telephone costs

- Infrastructure: 10-23%
- Switching and transmission: 6%
- Overhead: 49%
- Access: 34%
- Operations support systems: 11%
# Transport costs

<table>
<thead>
<tr>
<th>Network</th>
<th>$/min</th>
<th>$/MB</th>
</tr>
</thead>
<tbody>
<tr>
<td>wholesale telephone</td>
<td>0.01–0.02</td>
<td></td>
</tr>
<tr>
<td>U.S. domestic interstate consumer rates</td>
<td>0.05–0.15</td>
<td></td>
</tr>
<tr>
<td>U.S. domestic intrastate consumer rates</td>
<td>0.05–0.25</td>
<td></td>
</tr>
<tr>
<td>modem</td>
<td>0.25 – 0.50</td>
<td></td>
</tr>
<tr>
<td>private line</td>
<td>0.50 – 1.00</td>
<td></td>
</tr>
<tr>
<td>frame relay</td>
<td>0.30</td>
<td></td>
</tr>
<tr>
<td>MCI frame SVC</td>
<td>0.05</td>
<td></td>
</tr>
<tr>
<td>Internet</td>
<td>0.04–0.15</td>
<td></td>
</tr>
<tr>
<td>Internet modem</td>
<td>0.33</td>
<td></td>
</tr>
<tr>
<td>Internet backbone</td>
<td>0.01</td>
<td></td>
</tr>
</tbody>
</table>

1 minute voice = 480 kB with silence suppression, 1 MB without
### Phone usage

“Free” phone calls does not mean unbounded increase:

<table>
<thead>
<tr>
<th>year</th>
<th>lines (millions)</th>
<th>local calls min/day/line</th>
<th>local calls min/day/person</th>
</tr>
</thead>
<tbody>
<tr>
<td>1980</td>
<td>102.2</td>
<td>39</td>
<td>17.5</td>
</tr>
<tr>
<td>1988</td>
<td>127.1</td>
<td>39</td>
<td>20.2</td>
</tr>
<tr>
<td>1996</td>
<td>166.3</td>
<td>40</td>
<td>25.1</td>
</tr>
</tbody>
</table>
Why aren’t we using it now?

Internet capacity ≪ phone traffic:

<table>
<thead>
<tr>
<th>Traffic Type</th>
<th>Capacity (Gb/s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>world phone traffic</td>
<td>600</td>
</tr>
<tr>
<td>international traffic</td>
<td>13</td>
</tr>
<tr>
<td>MCI Internet</td>
<td>1.8</td>
</tr>
<tr>
<td>public Internet (late 1997)</td>
<td>75</td>
</tr>
<tr>
<td>U.S. total</td>
<td>368</td>
</tr>
<tr>
<td>U.S. interstate</td>
<td>55</td>
</tr>
<tr>
<td>AT&amp;T long distance</td>
<td>61</td>
</tr>
</tbody>
</table>

- unpredictable sound quality, reliability
- doesn’t work well for dial-up users
- no cheap Internet devices
- 640 M phone lines, 122 M in U.S. ➔ gateways
- no billing infrastructure

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Projections

- MCI: “80% data, 20% voice”

- “AT&T could lose $350 million in international calls by 2001”

- “By 2002, the Internet could account for 11% of U.S. and international long-distance voice traffic”

- “Up to 10% of the world’s fax market, which generates $45 billion in telecom revenue a year, will move to Internet in 2 or 3 years”

- May 1999: BT builds IP phone network in Spain

- but: cable modems only 250,000 to 275,000 users in US, 10% of Internet users by 2000
Data vs. Voice Traffic

worldwide traffic (Gb/s)


100 1000 10000 100000

data
voice
Components for Internet Multimedia
Internet multimedia protocol stack

- Signaling:
  - H.323
  - SIP
  - RTSP

- Quality of Service:
  - RSVP
  - RTCP

- Media Transport:
  - RTP
  - Media encaps. (H.261, MPEG)

Layers:
- Application
- Transport
- Network
- Link
- Physical

Protocols:
- IPv4, IPv6
- RTP
- UDP
- TCP
- RSVP
- RTCP
- RTSP
- SIP
- H.323
- PPP
- AAL3/4
- AAL5
- Sonet
- ATM
- Ethernet
- V.34

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Internet Multimedia
Components for Internet Multimedia

**multicast:** routing, address allocation

**data transport:** RTP

**resource reservation:** RSVP, YESSIR, diff-serv

“TV” – announcing multicast sessions: SAP

“phone” – session setup for conferences/telephony: SIP

“VCR” – control of streaming media: RTSP

**local applications:** conference bus

**policy issues:** billing, firewall access, clearing houses

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Applications for Multicast

- audio-video distribution (1-to-many) and symmetric (all-to-all)
- distributed simulation (war gaming, multi-player Doom, …)
- resource discovery (where’s the next time server?)
- file distribution (stock market quotes, new software, …)
- network news (Usenet)
Host group model

Deering, 1991:

- senders need not be members;
- groups may have any number of members;
- there are no topological restrictions on group membership;
- membership is dynamic and autonomous;
- host groups may be transient or permanent.
IP Multicast: Problems

- multicast routing $\Rightarrow$ state $\propto S, G$

- proposals:
  - DVMRP, PIM-DM for dense groups
  - PIM-SM or CBT for sparse groups ("core")

- overlay networks (Mbone) hard to maintain

- billing and charging (satellite TV problem)

- multimedia applications mostly on-demand
Multicast address allocation

- about 268 mio. “class D” addresses
- can’t have FCC assign channels
- hierarchical borrowing, using DHCP locally
- IETF malloc WG
Data transport – RTP

Real-Time Transport Protocol (RTP) = data + control

data: timing, loss detection, content labeling, talkspurts, encryption

control: (RTCP) periodic with $T \sim$ population
  - QoS feedback
  - membership estimation
  - loop detection

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RTP functions

- segmentation/reassembly done by UDP (or similar)
- resequencing (if needed)
- loss detection for quality estimation, recovery
- intra-media synchronization: remove delay jitter through playout buffer
- intra-media synchronization: drifting sampling clocks
- inter-media synchronization (lip sync between audio and video)
- quality-of-service feedback and rate adaptation
- source identification
Resource Reservation

- *can’t* compensate for lack of bandwidth or reliability
- *can* provide incumbency protection
- receiver makes requests ➞ RSVP
- sender makes requests ➞ YESSIR
- issues: scaling (state), security, complexity
Internet telephony modes

- tail-end hop off ➔ callee has phone
- front-end hop on ➔ caller uses phone
- Internet in the middle: per-call, multiplexed
Internet “signaling”

all non-data (“out-of-band”) functions:

**routing:** unicast; DVMRP, PIM, CBT for multicast ✓

**quality of service:** RSVP, RTCP, diff-serv ✓

**user Contact:** map name to location (IP address)

**call set-up/teardown:** SIP, H.323

**policy, billing:** “vertical” protocols
Architecture

Internet Multimedia

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Differences: Internet Telephony ↔ POTS

- separate control, transport (UDP) ➔ no triangle routing
- separate connectivity from resource availability
- separate services from bit transport
- datagram service ➔ less bootstrapping
- in-band signaling ➔ higher speed
- features “network” → end system: distinctive ringing, caller id, speed dialing, number translation, … ➔ scaling
- features: intra-PBX = inter-LATA and general
- protocols: user-network = network-network signaling
Internet Telephony

- multimedia basically free (unlike ISDN)
- minimal extensions: signaling, not “stove pipe”
- leverage existing work: email, HTTP security, URIs, HTML, cgi, . . .
Light-weight signaling: Session Initiation Protocol (SIP)

IETF MMUSIC working group (RFC 2543)

- light-weight generic signaling protocol
- typical post-dial delay: 1.5 round-trip time (with UDP)
- network-protocol independent: UDP or TCP (or AAL5 or X.25)
SIP functionality

- call user

- re-negotiate call parameters

- manual and automatic forwarding

- call center: reach first (load distribution) or reach all (department conference)

- personal mobility (complements data link/IP mobility) change of terminal (PC, digital cordless, palmtop), location

- terminate and transfer calls
Service creation: Call Processing Language

- incoming and outgoing
- “if somebody is trying to call for the 3rd time, allow mobile”
- “try office and lab in parallel, if that fails, try home”
- “allow call to mobile if I’ve talked to person before”
- users and administrators
- not quite like cgi: multiple responses? timers?
- Tcl, Java?
Real-Time Streaming Protocol (RTSP)

remote-control streaming media

- “rough” synchronization (fine-grained RTP sender reports)
- virtual presentations = synchronized playback from several servers command timing
- load balancing using redirection at connect, during stream
- supports any session description
- device control camera pan, zoom, tilt
- caching: similar to HTTP, except “cut-through”
Open Operational Issues

- billing
- finding the nearest gateway to the Internet (GLP)
- mapping E.164 (phone) numbers to IP addresses
- controlling phones through the Internet (PINT)
- 911 services
- CALEA
- anonymity and certified identity
Billing

- simplification: email/web delivery, credit card payment

- what to bill for?

  transport services: volume, time, reserved resources;
  “free upgrades”

  signaling services: filtering, forwarding, scripting, mobility, . . .

  storage services: voice mail

  gateway services: PSTN gateways
Emergency (911) services

- U.S.: dial “911” anywhere → nearest Public Safety Answering Points
- look up street address from telephone company database
- but...
  - IP address dynamically assigned
  - may not be correlated to geography
  - dial-in from hotel, remote sites?
  - prevent services: hanging up, transfer, hold, …
Emergency services

- advantages:
  - multimedia (video, medical data, …)
  - medical database access, with authentication token
  - remote activation of medical devices

- solutions:
  - enclose (signed) location information with call
  - IP address → provider → lookup (RADIUS) needs authenticated protocol
  - GPS
Conclusion

• transition of separate circuit-switched ➔ IP-based applications

• packets from the inside out or the outside in?

• IP over ATM, Sonet, WDM?

• IPv6 or NATs?

• “the end of distance” or tiered IP service?