IP Telephony

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Joint work with SIP IM/presence group and Telcordia
Overview

- why Internet telephony
- the Session Initiation Protocol
- new services:
  - integration with 2G mobile (GSM, CDMA)
  - next-generation wireless (3GPP, 3GPP2, MWIF, …)
  - event notification and instant messaging
  - programmable services
  - appliance control
- the Columbia University VoIP platform
- pricing Internet quality of service
- coping with loss and delay
What is Internet Telephony?

- carriage of real-time voice and multimedia
- IP: private networks or public Internet
- interconnected to existing phone network
- low latency, high availability
Internet PBX

- SIP proxy server
- IP
- Ethernet
- ISP
- T1, ISDN (BRI/PRI) or analog lines

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IP Centrex

Chatterbox Cafe

Ralph’s Pretty Good Grocery

Heads Up Barber

ISP

PSTN

Internet

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Why Internet Telephony

- economic:
  - early: *arbitrage* between international rates and Internet
  - avoid local access charges, but 7c ↓ 2.4c
  - cheaper transport: $0.03 / DS0 vs. $175 / DS0
  - cable plant integration (PBX, DSL)

- new services

- multimedia integration
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
IETF VoIP Architecture Characteristics

- universal identifier \textit{user@domain}: SIP URL = email = network access identifier
- data transport: RTP
- setting up calls: SIP
- emphasis on user-programmable services
- web integration: content, mutual referral
- integration with IM and presence
SIP Overview

- protocol for establishing, modifying, tearing down (multimedia) sessions
- IETF Proposed Standard since March 1999
- multimedia = audio, video, shared applications, text, . . .
- also used for “click-to-dial” and possibly Internet call waiting
- to be used for PacketCable Distributed Call Signaling
- to be used for Third-Generation Wireless (3GPP, 3GPP2)
SIP Example: Redirection

1. INVITE henning@ieee.org
   302 Moved temporarily
   Contact: hgs@columbia.edu
2. ACK henning@ieee.org
3. INVITE hgs@columbia.edu
4. 200 OK
5. ACK hgs@columbia.edu

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SIP Mobility

<p>| | | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>terminal</td>
<td>cross-provider</td>
<td>REGISTER, re-INVITE</td>
</tr>
<tr>
<td>personal</td>
<td>different terminals, same address</td>
<td>REGISTER</td>
</tr>
<tr>
<td>service</td>
<td>different terminals, same services</td>
<td>upload</td>
</tr>
<tr>
<td>session</td>
<td>move sessions across terminals</td>
<td>REFER</td>
</tr>
</tbody>
</table>
SIP Personal Mobility
SIP Bake-Off

- takes place every four months, 5th at Pulver.com August 2000
- 45 organizations from 11 countries
- about 50-60 implementations:
  - IP telephones and PC apps
  - proxy, redirect, registrar servers
  - conference bridges
  - unified messaging
  - protocol analyzers
- first IM/presence interop test
- emphasis on advanced services (multi-stage proxying, tel URLs, call transfer, IVR, …)
Invisible Internet Telephony

VoIP technology will appear in . . .

- Internet appliances
- home security cameras, web cams
- 3G mobile terminals
- fire alarms
- chat/IM tools
- interactive multiplayer games
The Largest Signaling Network is Not Running SS7

- AT&T: 280 million calls a day
- AOL: 110 million emails/day, total about 18 billion/day
- total $> 1$ billion instant messages a day (AOL: 500 million)
- signaling effort of call $\approx$ IM
Commonalities between Signaling and Events

- presence is just a special case of events: “Alice just logged in” $\approx$ “temperature in boiler exceeds 300° F”
- need to *locate* mobile end points
- may need to find several different destinations ("forking")
- same addressing for users
- presence often precursor to calls
- likely to be found in same devices
- events already in VoIP: message alert, call events
Events: SIP for Appliances

SIP user agent

SUBSCRIBE door@alice.home.net
NOTIFY alice@work.com
DO light@alice.home.net
INVITE camera@alice.home.net

SIP proxy (RGW)

(Work with Telcordia)
## Programmable Internet Telephony

<table>
<thead>
<tr>
<th>Feature</th>
<th>APIs</th>
<th>servlets</th>
<th>sip-cgi</th>
<th>CPL</th>
</tr>
</thead>
<tbody>
<tr>
<td>Language-independent</td>
<td>no</td>
<td>Java only</td>
<td>yes</td>
<td>own</td>
</tr>
<tr>
<td>Secure</td>
<td>no</td>
<td>mostly</td>
<td>no, but can be</td>
<td>yes</td>
</tr>
<tr>
<td>End user service creation</td>
<td>no</td>
<td>yes</td>
<td>power users</td>
<td>yes</td>
</tr>
<tr>
<td>GUI tools w/portability</td>
<td>no</td>
<td>no</td>
<td>no</td>
<td>yes</td>
</tr>
<tr>
<td>Call creation</td>
<td>yes</td>
<td>no</td>
<td>no</td>
<td>no</td>
</tr>
<tr>
<td>Multimedia</td>
<td>some</td>
<td>yes</td>
<td>yes</td>
<td>yes</td>
</tr>
</tbody>
</table>

Example: integration with iCal → automatically export personal calendar to call handling
Service mobility – call handling

- need uniform basic service description model → Call Processing Language (CPL)
- want language similar to HTML, not PostScript importable by user code generation tools
- safe, CPU-bounded, provable
- CPL = XML-based flow graph for inbound & outbound calls
- CPL for local call handling
- update CPL from terminal: add telemarketer to block list
- harder: synchronize CPL changes across multiple providers
- REGISTER updates information if device knows its multiple identities
- merging of call logs
<incoming>
    <address-switch field="origin" subfield="host">
        <address subdomain-of="example.com">
            <location url="sip:jones@example.com">
                <proxy>
                    <busy> <sub ref="voicemail" /> </busy>
                    <noanswer> <sub ref="voicemail" /> </noanswer>
                    <failure> <sub ref="voicemail" /> </failure>
                </proxy>
            </location>
        </address>
    </address-switch>
</incoming>
Columbia Efforts in Internet Telephony

- signaling protocols: SIP + extensions (QoS, mobility, events, caller preferences, …)
- programming languages and interfaces: CPL and sip-cgi
- software and hardware VoIP platforms
- statistical packet voice characterization
- combining forward error correction (FEC) and playout delay adaptation
- QoS pricing for adaptive multimedia services
- resource reservation protocols: YESSIR, BGRP, RNAP
Columbia Internet Extensible Multimedia Architecture

- request, response transaction
- CGI scripts
- basic authentication
- digest authentication
- PGP

- SIP UA call state
- REGISTER endpoint class
- SDP

- PGP
- PWL resparse
- libNT
- NT versions of aliases
- crypt
- hashtable
- inet
- regex
- getopt
- utilities

- libcine
- msgflow
- URIs
- logging
- MD5
- config./DB access
- software licensing
- TCP
- UDP
- dstring
- host2ip

- libsip
- request, response transaction
- PGP

- lipsip++
- SIP UA call state
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- SDP

- libdict
- dictionary
- hash tables

- libmixer
- mix RTP audio
- GSM, DVI codecs

- RTSP media server
- SIPS proxy server
- SIPS/H.323 gateway
- SIPS conferencing server
- SIPS/RTSP unified messaging
- SIPS/MGCP gateway

- rtspd
- sipd
- siph323
- sipconf
- sipum
- sipgw

- LDAP
- MySQL
- Xerces
- OpenH323
- RTPlib
- MySQL
- AT&T MGCP

- GSM, DVI codecs
- libNT
- libmixer
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Columbia Internet Extensible Multimedia Architecture
Columbia e*phone

DSP-based, single-processor Ethernet phone; being commercialized
Conclusion

- first chance to re-engineer basic communications infrastructure
- universities can now build most of software infrastructure
- programmable by non-specialists → web model of service development
- want to avoid replication of PSTN on packets
- most VoIP applications won’t look like a telephone
- opportunities in emergency services, mobile, event notification