IP Telephony

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Joint work with SIP IM/presence group and Telcordia

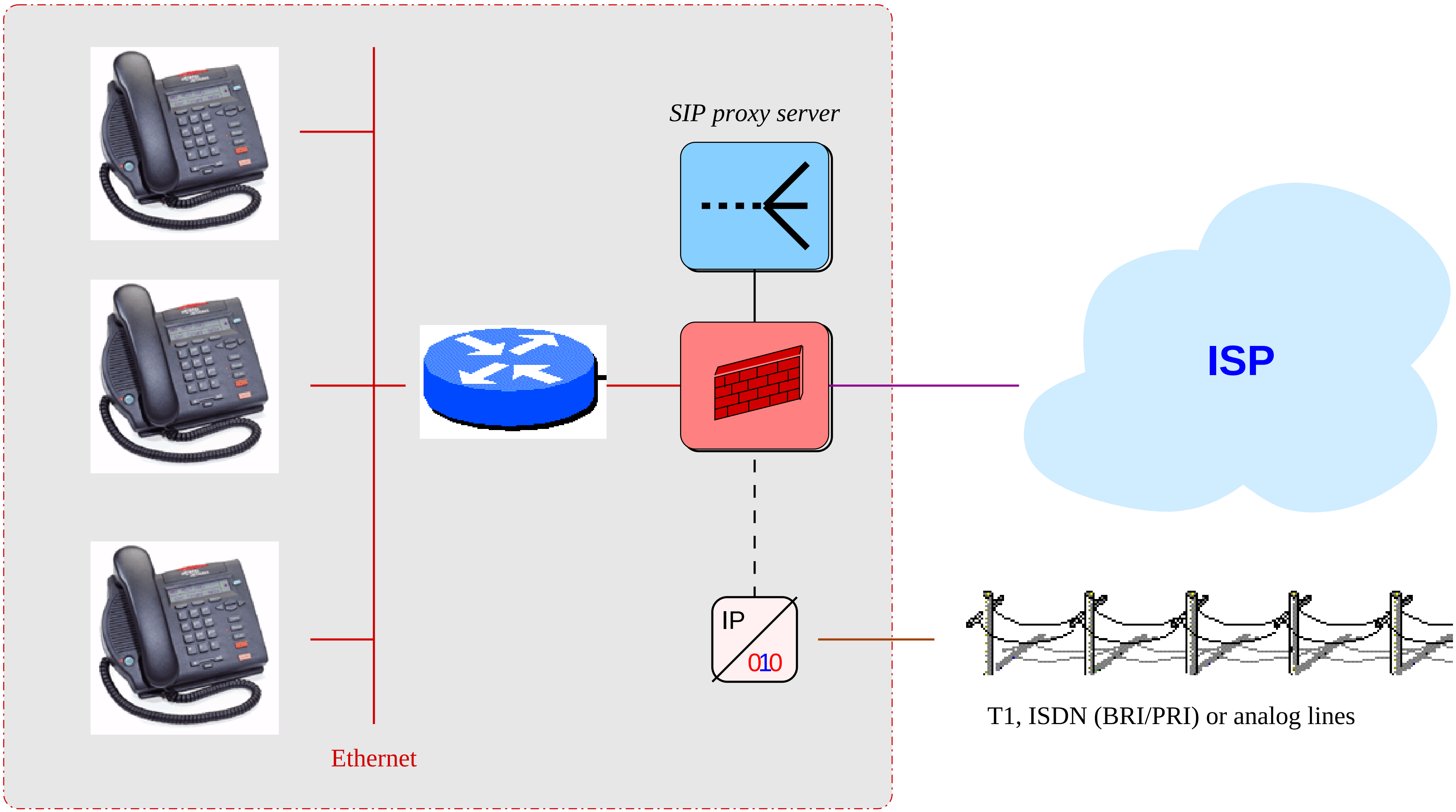
Overview

- 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

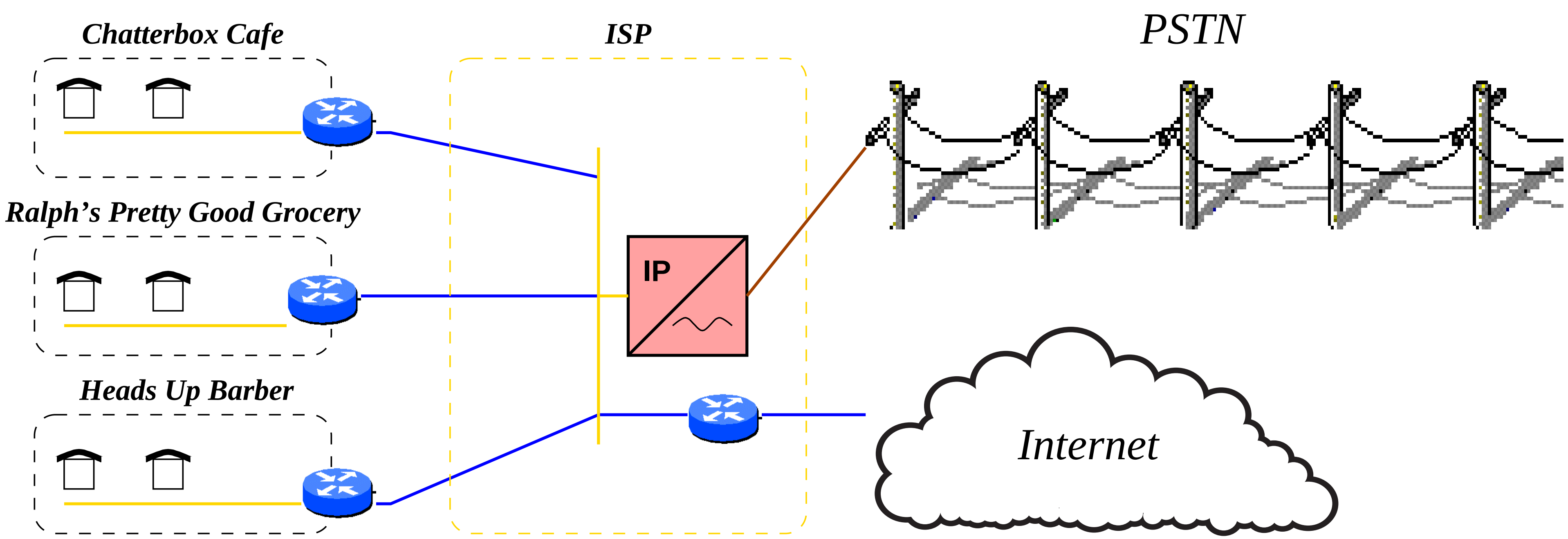
What is Internet Telephony?

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

Internet PBX



IP Centrex



Chatterbox Cafe
ISP
PSTN
Ralph's Pretty Good Grocery
IP
Heads Up Barber
Internet

Why Internet Telephony

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

Differences: Internet Telephony $\leftrightarrow$ 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" $\rightarrow$ end system: distinctive ringing, caller id, speed dialing, number translation, ... ➠ scaling
- features: intra-PBX = inter-LATA and general
- protocols: user-network = network-network signaling

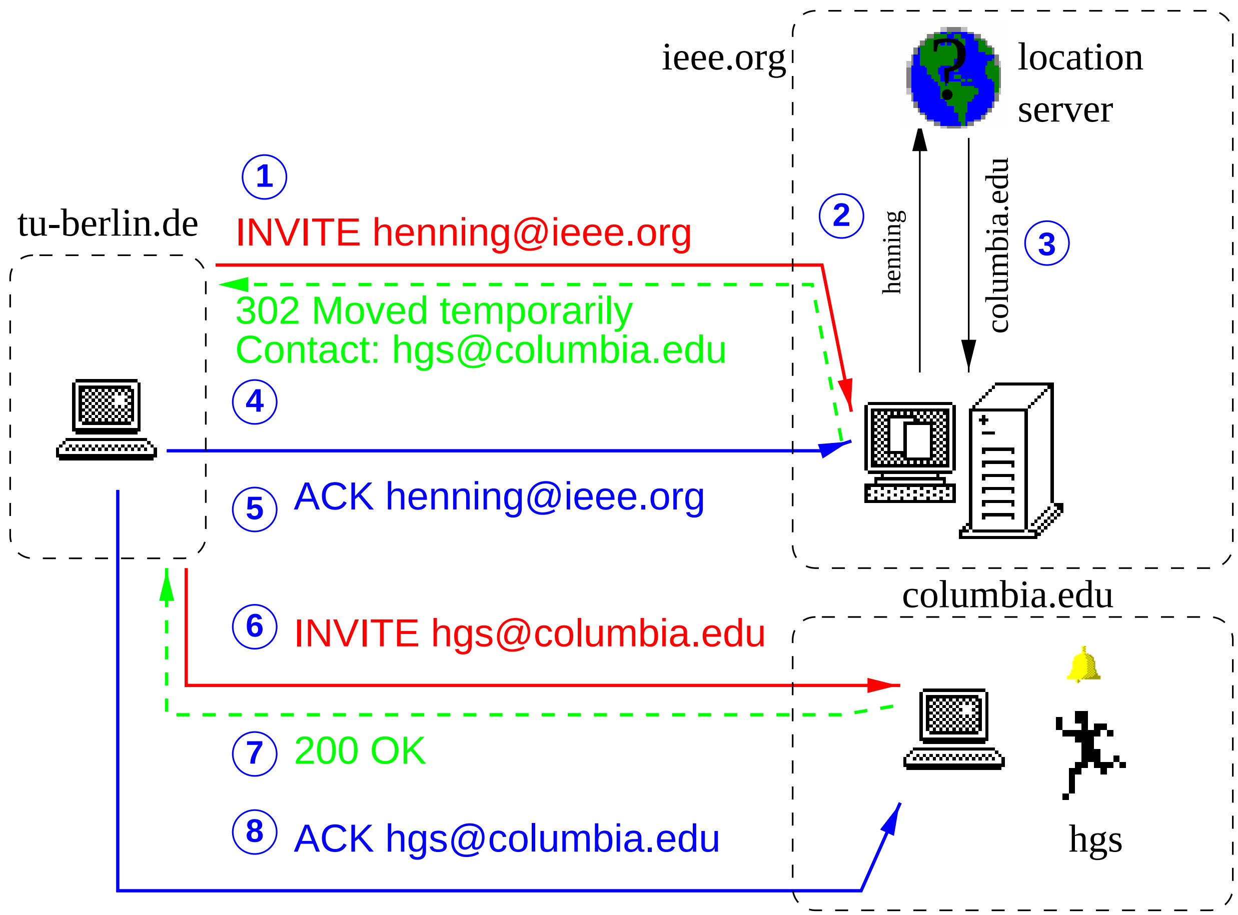
IETF VoIP Architecture Characteristics

- universal identifier *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)

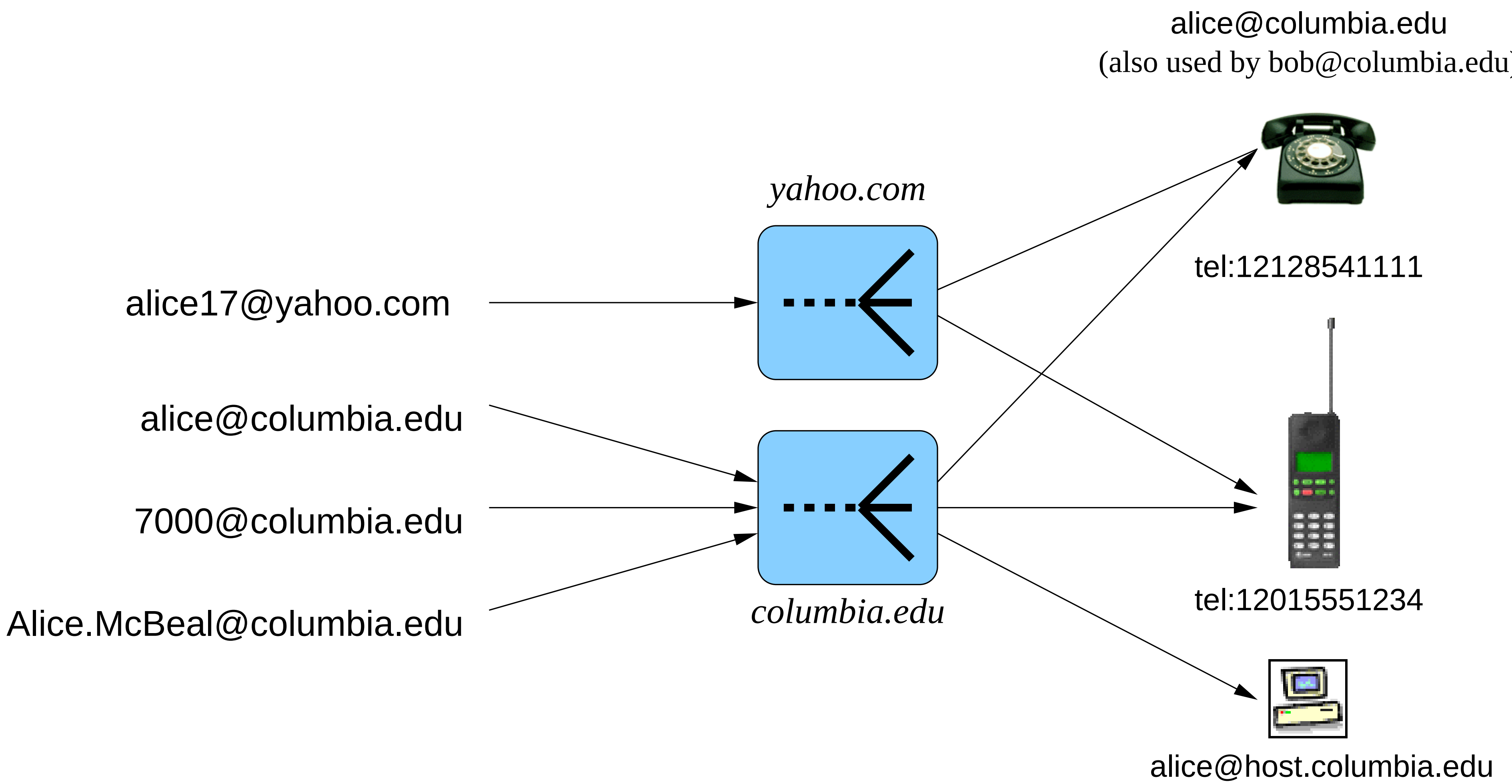
SIP Example: Redirection



SIP Mobility

terminal	cross-provider	REGISTER, re-INVITE
personal	different terminals, same address	REGISTER
service	different terminals, same services	upload
session	move sessions across terminals	REFER

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

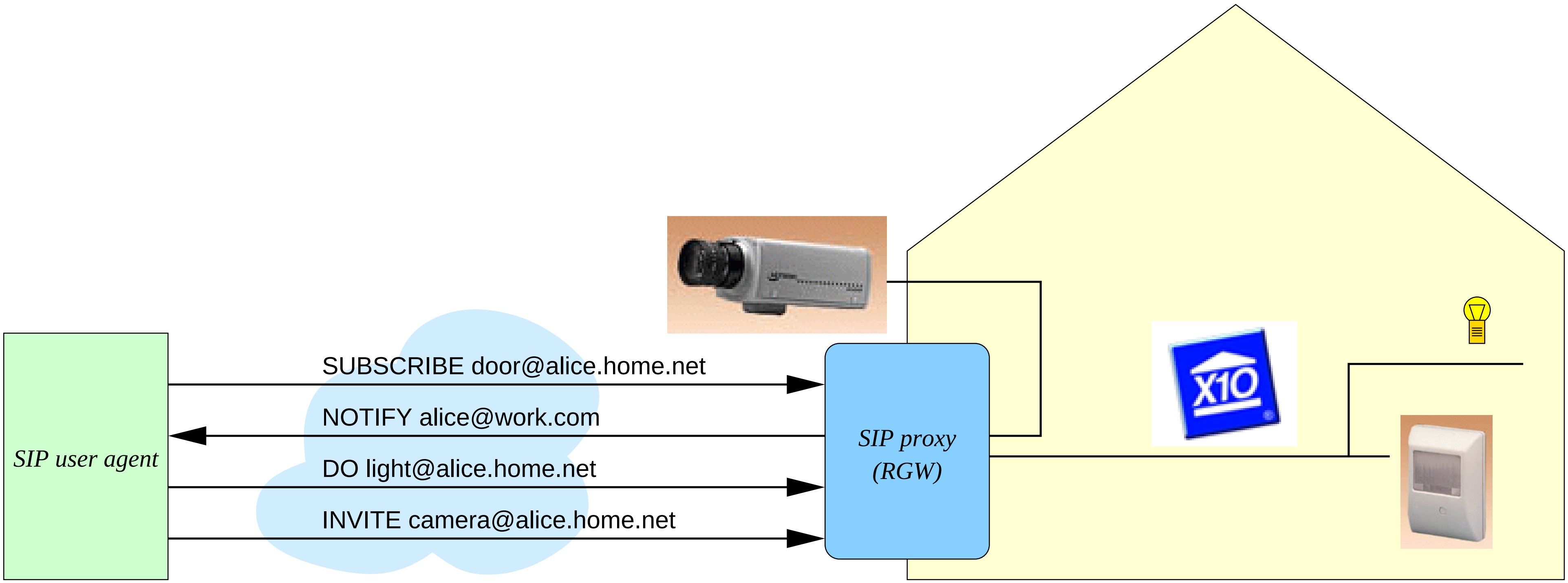
The Largest Signaling Network is Not Running SS7

- 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



(Work with Telcordia)

Programmable Internet Telephony

	APIs	servlets	sip-cgi	CPL
Language-independent	no	Java only	yes	own
Secure	no	mostly	no, but can be	yes
End user service creation	no	yes	power users	yes
GUI tools w/portability	no	no	no	yes
Call creation	yes	no	no	no
Multimedia	some	yes	yes	yes

Example: integration with iCal $\longrightarrow$ automatically export personal calendar to call handling

Service mobility – call handling

- need uniform basic service description model $\longrightarrow$ 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

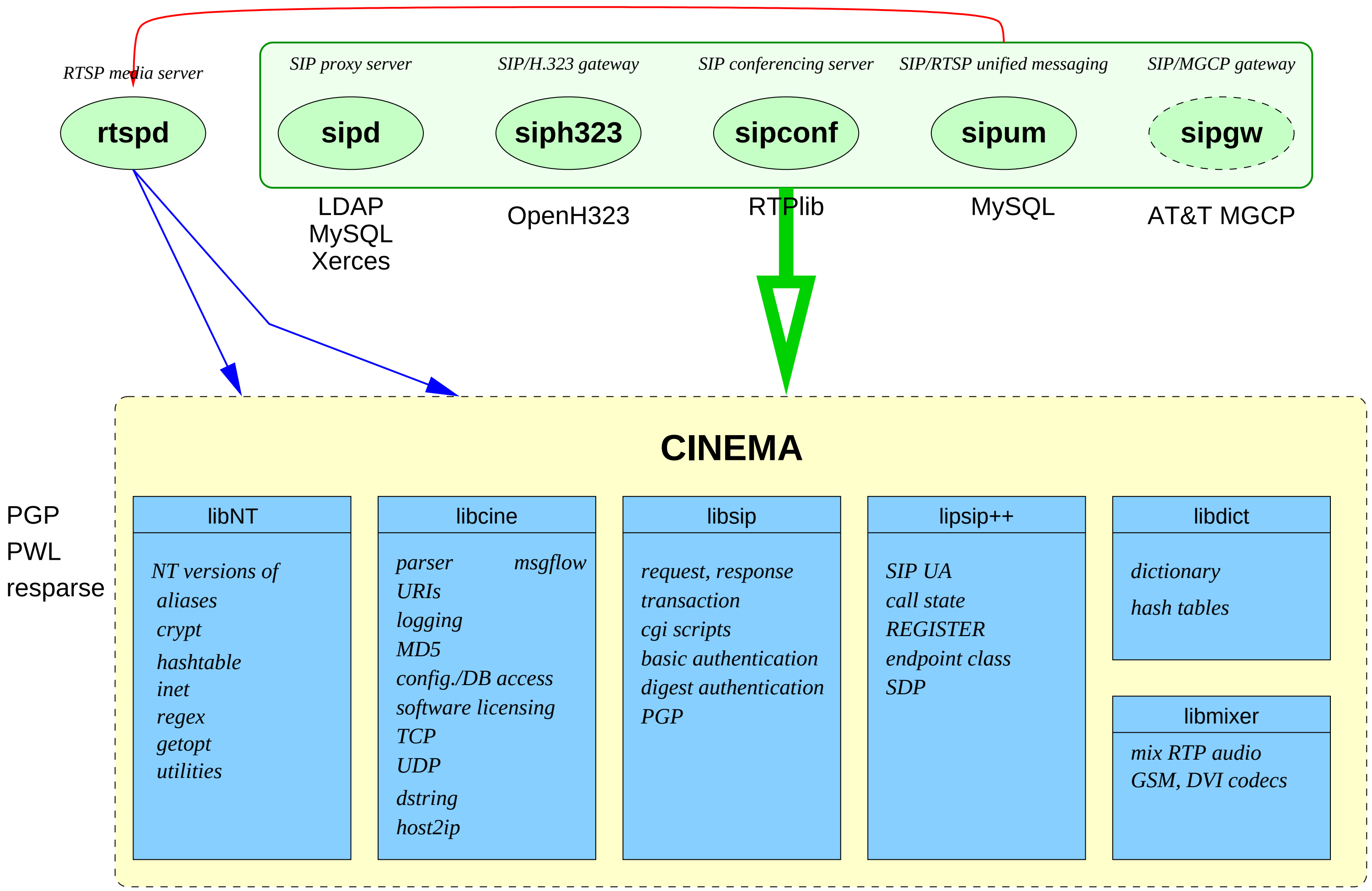
Textual representation

```
  <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>
      <otherwise>
        <sub ref="voicemail" />
      </otherwise>
    </address-switch>
  </incoming>
</cpl>
```

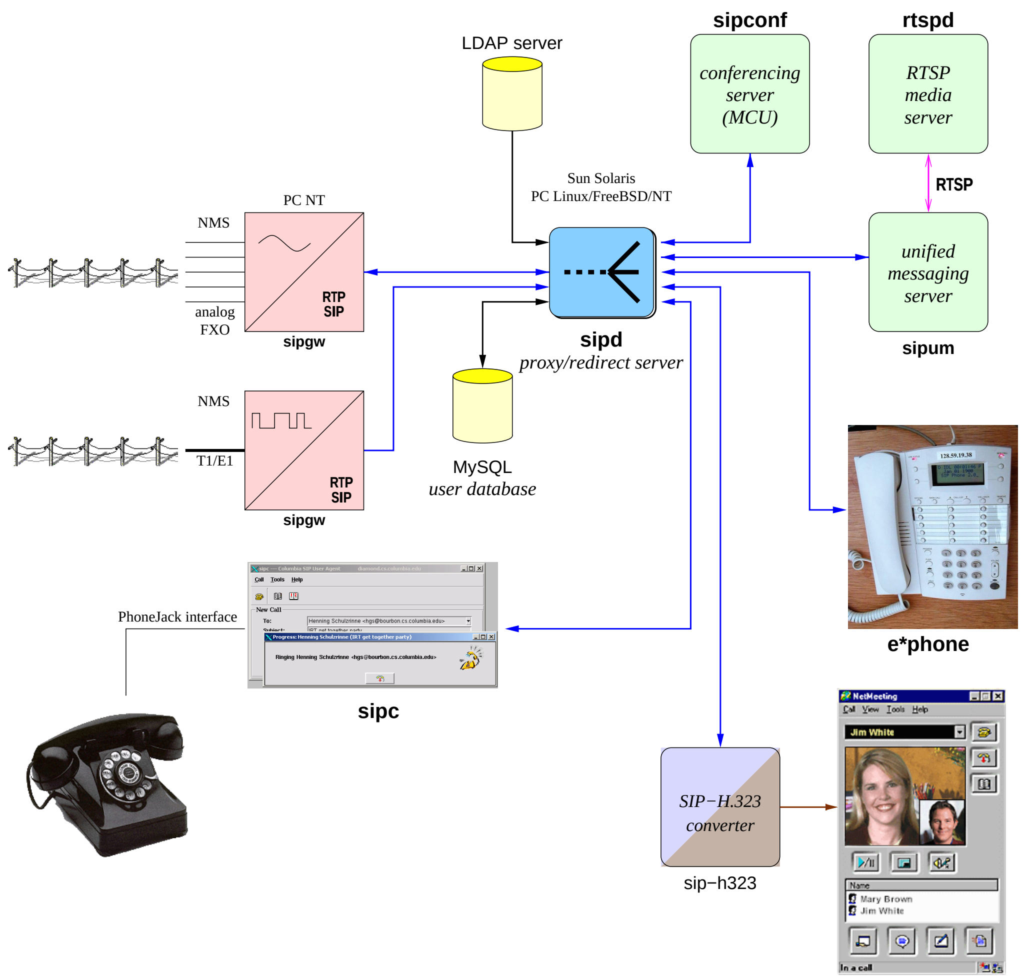
Columbia Efforts in Internet Telephony

- 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



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 $\longrightarrow$ 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

More information at http://www.cs.columbia.edu/sip, http://www.cs.columbia.edu/IRT