Current Issues and Future Directions for VoIP

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Overview

- motivations
- usage
- *technology* (research), *standardization* and *deployment* challenges:
  - services
  - QoS
  - security
  - emergency services
  - scaling & reliability

Driving forces

- cheap international calls
  - = “arbitrage”
  - students, long-distance relatives, . . .
  - often hidden as cheap prepay calling cards
- PBX replacements for large companies with high-speed switched LANs
- “tie lines” between branch offices, instead of leased voice lines

Future VoIP uses

Promised (and hyped), but not yet very successful:

- PC-to-PC (video) calls – Windows XP?
- cable modems
- Voice-over-DSL
- wireless (3G systems)

VoIP usage statistics

- cross-border traffic about 1.7 billion minutes (1999)
- 1.6% of total international voice traffic
- estimated at 6.2 billion minutes in 2001
- 35% U.S. to Asia Pacific, 39% Latin America

VoIP timeline

Year	development
1975	ARPAnet experiments
1980’s	voice/data multiplexers
1992-1996	DARTnet, Mbone
1996-1998	RTP, commercial R&D
1999	signaling standards (SIP, H.323v2)
2000	SIP product development
2001	pioneer carrier deployment
2002	PBX turning point?

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" $\longrightarrow$ end system: distinctive ringing, caller id, speed dialing, number translation, . . . ➠ scaling
- features: intra-PBX = inter-LATA and general
- protocols: user-network = network-network signaling
- integration of presence and events

PSTN legacies to avoid

- E.164 numbers û might as well wear bar codes
- tones (e.g., failure indications)
- in-band signaling (DTMF)
- systems with user interface knowledge (12 keys, voice)
- voice-only orientation (e.g., MGCP/Megaco)
- integration of bit transport and services
- service-specific billing
- trigger model for service creation
- trusted networks without crypto authentication $\longrightarrow$ confine PSTN knowledge to edge of network

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

Replication of existing services

- “user is familiar with PSTN services”
- but how many users actually know how to use call transfer or directed pick-up?
- user interface is often just legacy of key systems or other ancient technology
- avoid binding of identifiers to devices ➠ call person or group of people, regardless of location
- instead, model desired behavior
- single-server features don’t need standardization
- find general mechanisms (e.g., SIP REFER, events)

Deployment challenges

Technology known, but deployment difficult:

- QoS
- security
- emergency services
- equipment cost — $20 (or $100) phones
- IPv6 to make NATs unnecessary
- configuration for possibly thousands of devices
- wire tapping

Deployment challenge: QoS

- most routers have small number of queueing classes
- DiffServ doesn't have deployable admission control
- inter-provider settlements
- "scavenger service"

Deployment challenge: Security

- may want to identify called/calling parties – no trusted phone company
- server certificates scale – no success with personal certs
- secure email as (negative) example
- “same person that called yesterday” or “student from ICU” may be good enough
- secure voice communication with random parties – key establishment?

Deployment challenge: Emergency services

- components of current systems:
  - common identifier (911, 112, . . .) – SOS?
  - identify appropriate "public safety answering point"
  - caller identity & location – IP addresses not good locator (VPNs!)
- opportunity for new services: multimedia, biometrics, database access
- work with old technology and transition to IP-enabled PSAPs

Deployment challenge: cheap devices

- need full Internet stack, possibly without TCP (but TCP/TLS desirable)
- need audio codecs and jitter compensation
- minimal configuration interface: web server or tftp common
- guess: will evolve towards “real” OS like VxWorks or Linux
- MGCP/Megaco (with security) not much simpler, except for simpler call transfer and configuration

Example: Pingtel SIP phone

Example: Cisco and 3Com SIP phones

Cisco

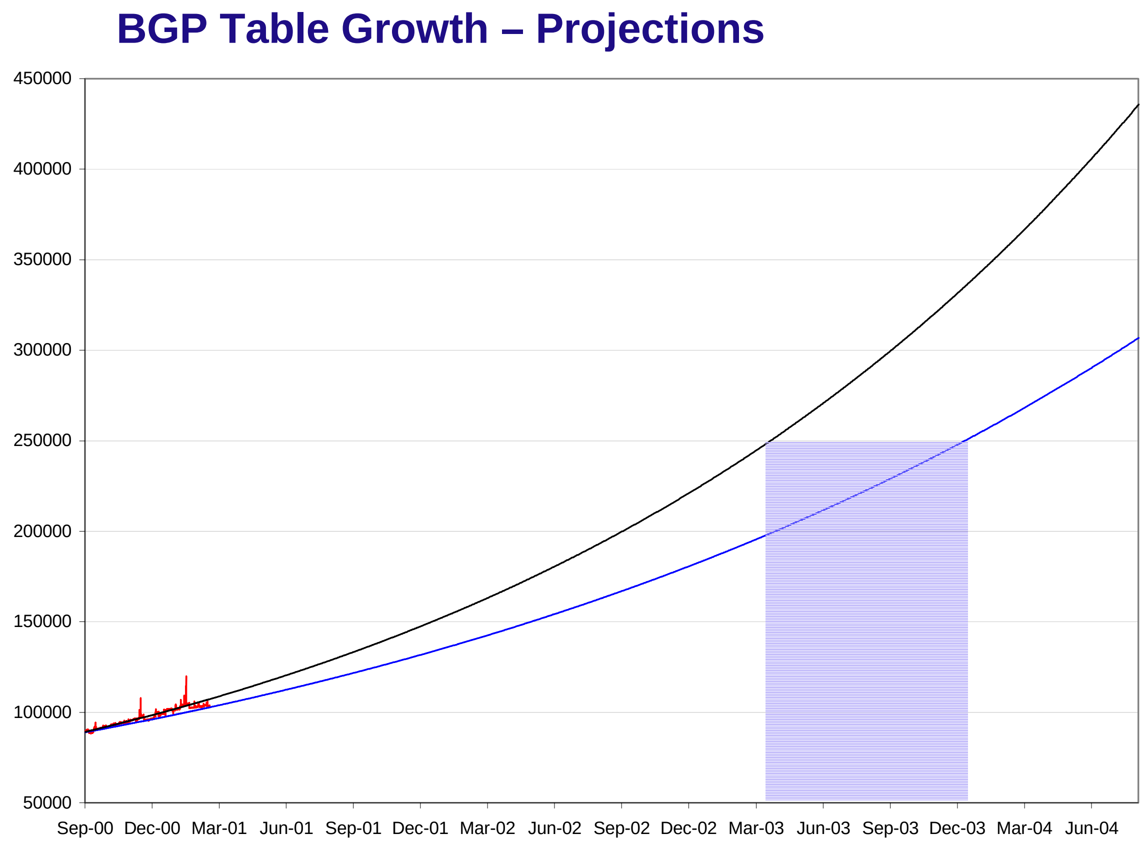
3Com ($395 list)

Standardization challenges

- 3G wireless:
  - complexity
  - releases (2.5G, R4, R5, . . .)
  - transition from hybrid CO/PS to all-Internet
- gateway location
- instant messaging & presence
- NATs and firewalls
- conference control
- next-generation session description ("SDPng")

Research challenges

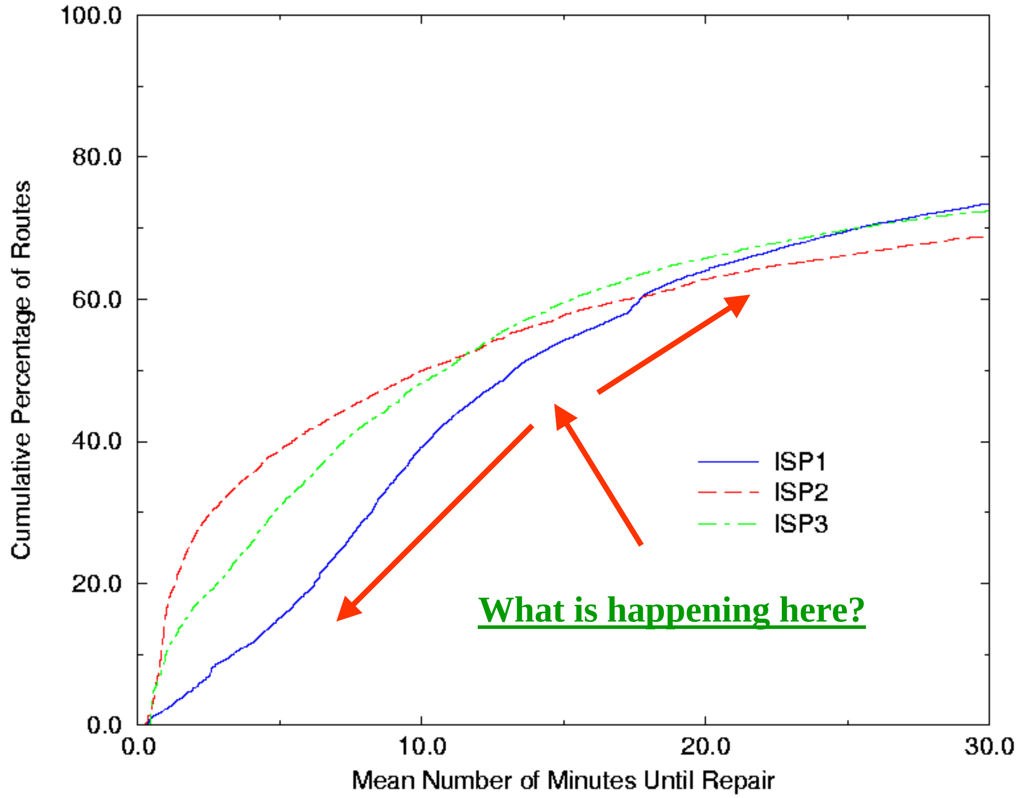
- scaling the Internet: routing table size & convergence
- new services: limited by least common (PSTN) denominator
- new services: service architectures
- QoS fault determination
- reliability



(courtesy Geoff Huston)

How long until routes return?

(From A Study of Internet Failures)



(courtesy Geoff Huston)

New services

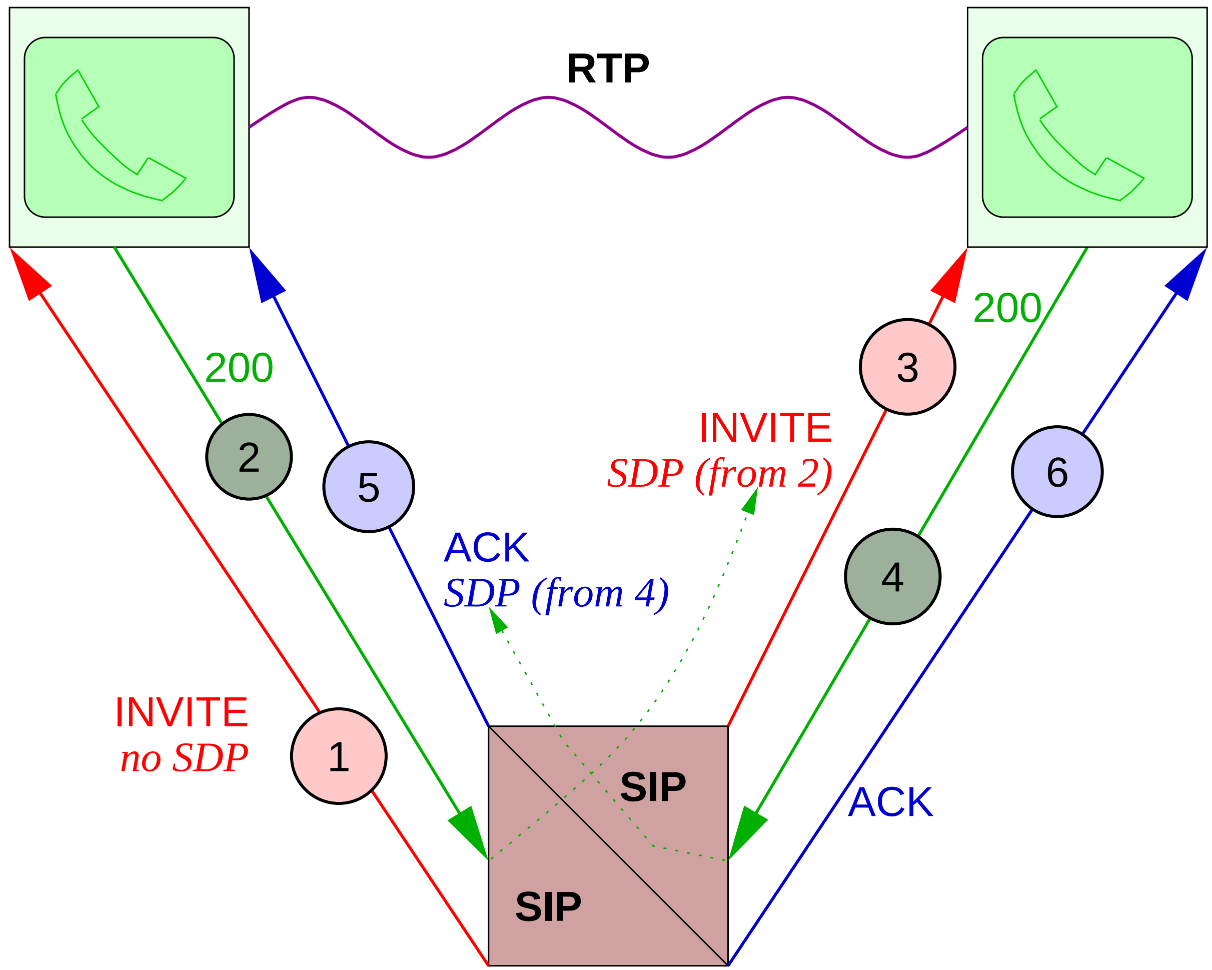
- vicious circle: can only get “black phone” services $\leftrightarrow$ no non-$ incentive to deploy VoIP
- examples:
  - user interface for call forwarding
  - visual caller id
  - multimedia
  - non-numeric user IDs
- mobile phones have much shorter lifetimes (2 years?)

Service architectures

Is there a fundamental set of service components?

- events – traditional call states too PSTN-centric?
- third-party call control
- indirect requests – A asks B to send message to C
- data (“function arguments”) in headers
- negotiation – offer/answer/final?

Example: third-party call control



QoS fault determination

- who do you call when the voice quality is bad?
- standard network management not accessible to end user
- traceroute and ping not too helpful
- correlation analysis using RTCP data to third party?

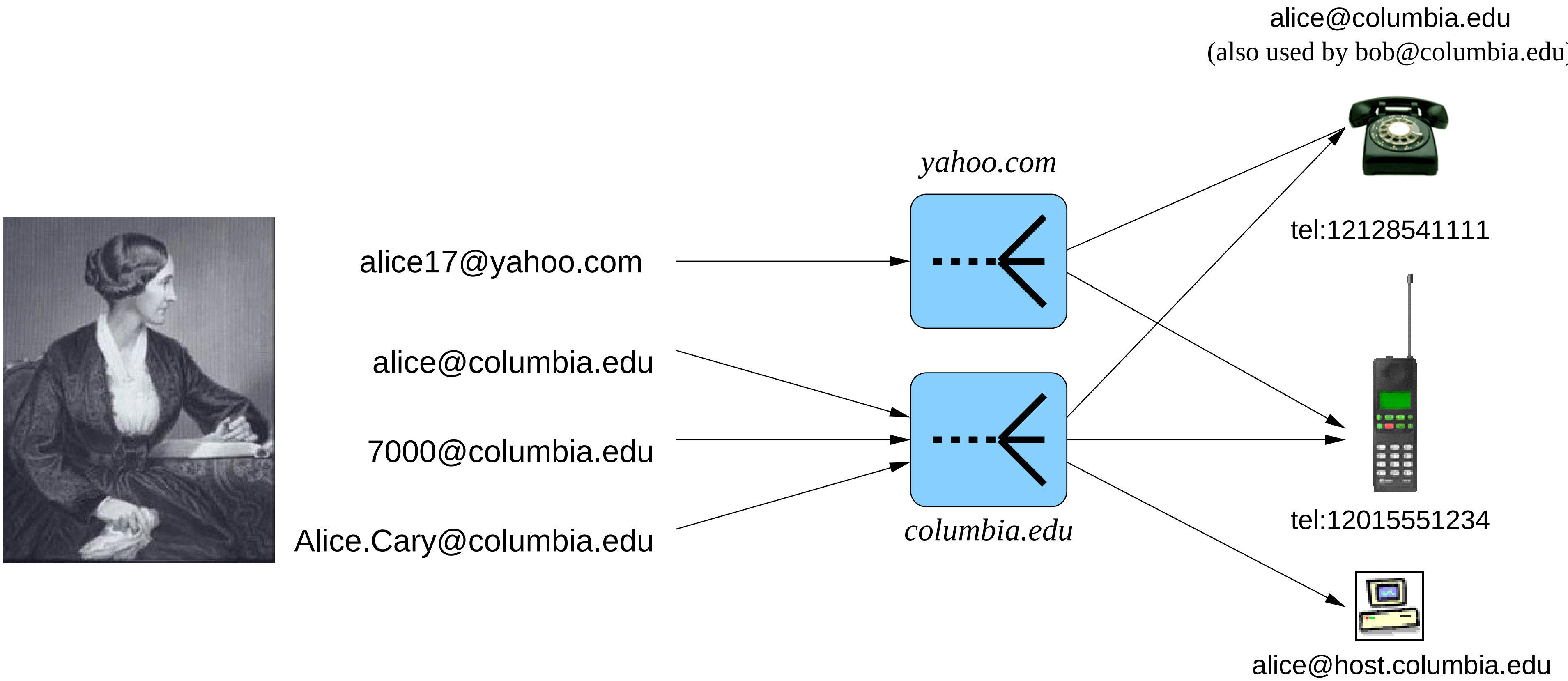
Reliability

- “5 nines” $\approx$ 5 minutes/year
- most ISPs seem to achieve about 99.9%
- what is failure – partial, no connection, dropped call?
- could be *more* reliable since existing calls are not dropped when rerouting
- new problem: overload protection for denial of service attacks

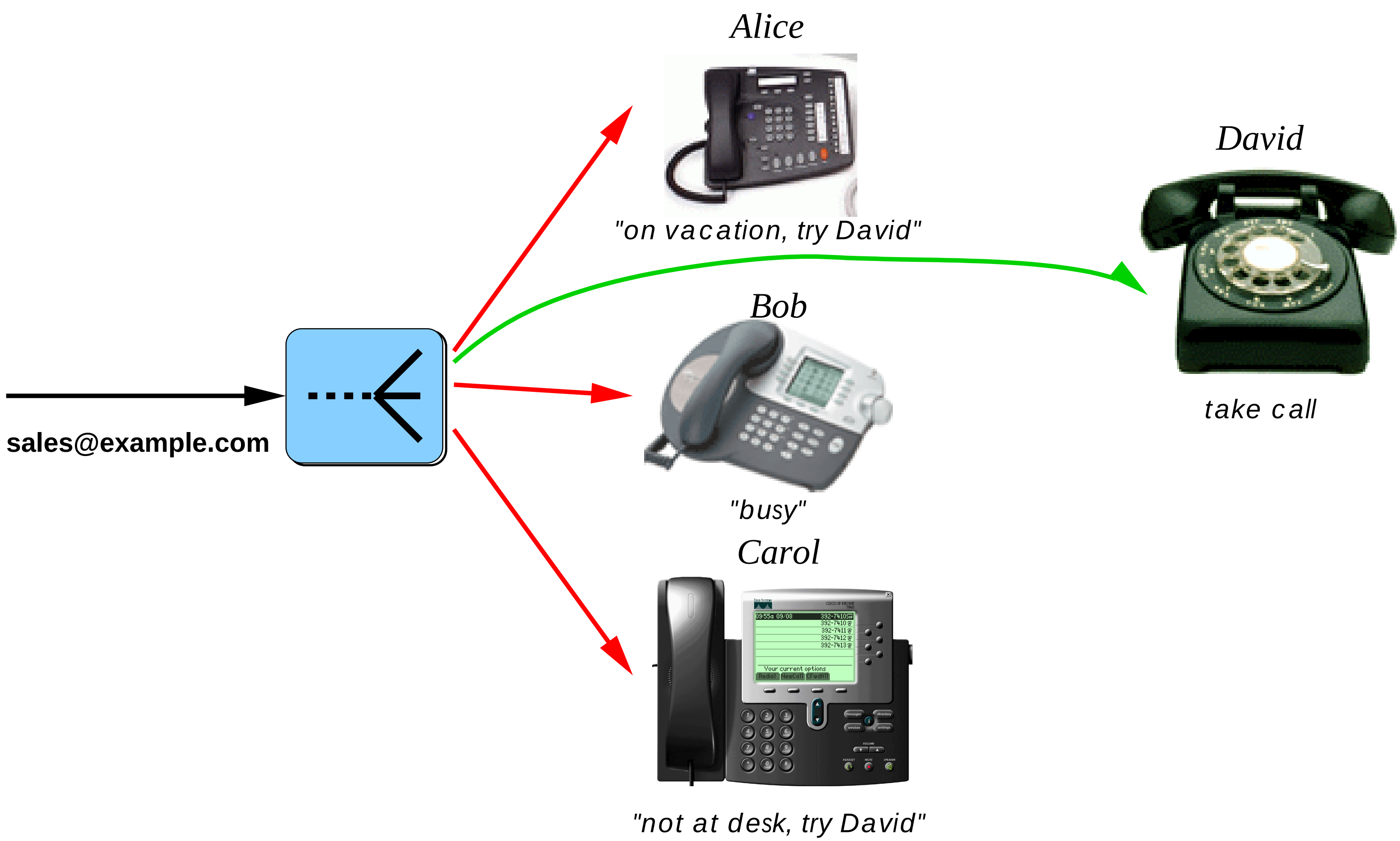
Future directions

- new services:
  - personal mobility
  - personalized programmable services
  - multimedia
- event notification

SIP personal mobility



New SIP services: multi-destination routing



New SIP services: Internet integration

- typically, SIP URL $\equiv$ email address, e.g., `sip:joe@net2phone.com` or `tel:+1201-555-1212`
- URLs everywhere:
  - forward calls to email
  - forward calls to web page
  - forward calls to recordings
  - pager, cell phone numbers
  - IM addresses
- SIP messages can contain HTML and other web objects:
  - menu pops up when calling restaurant
  - error messages: “not here, but please choose from . . . ”
  - visual caller id – photos of callee

Programmable Internet Telephony

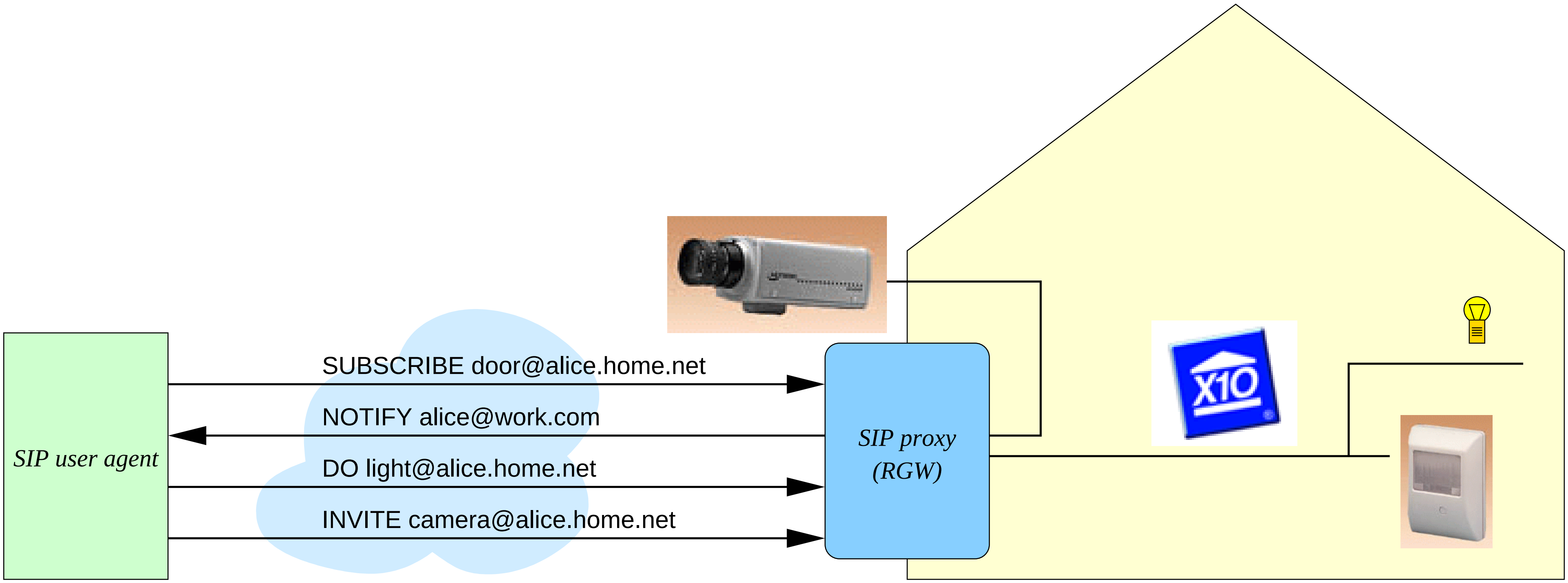
	APIs	servlets	sip-cgi	CPL
Language-independent	no	Java only	yes	own
Secure	no	mostly	no, but can be	yes
End user service creation	no	yes	power users	yes
GUI tools w/portability	no	no	no	yes
Call creation	yes	no	no	no
Multimedia	some	yes	yes	yes

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

Commonalities between signaling and events

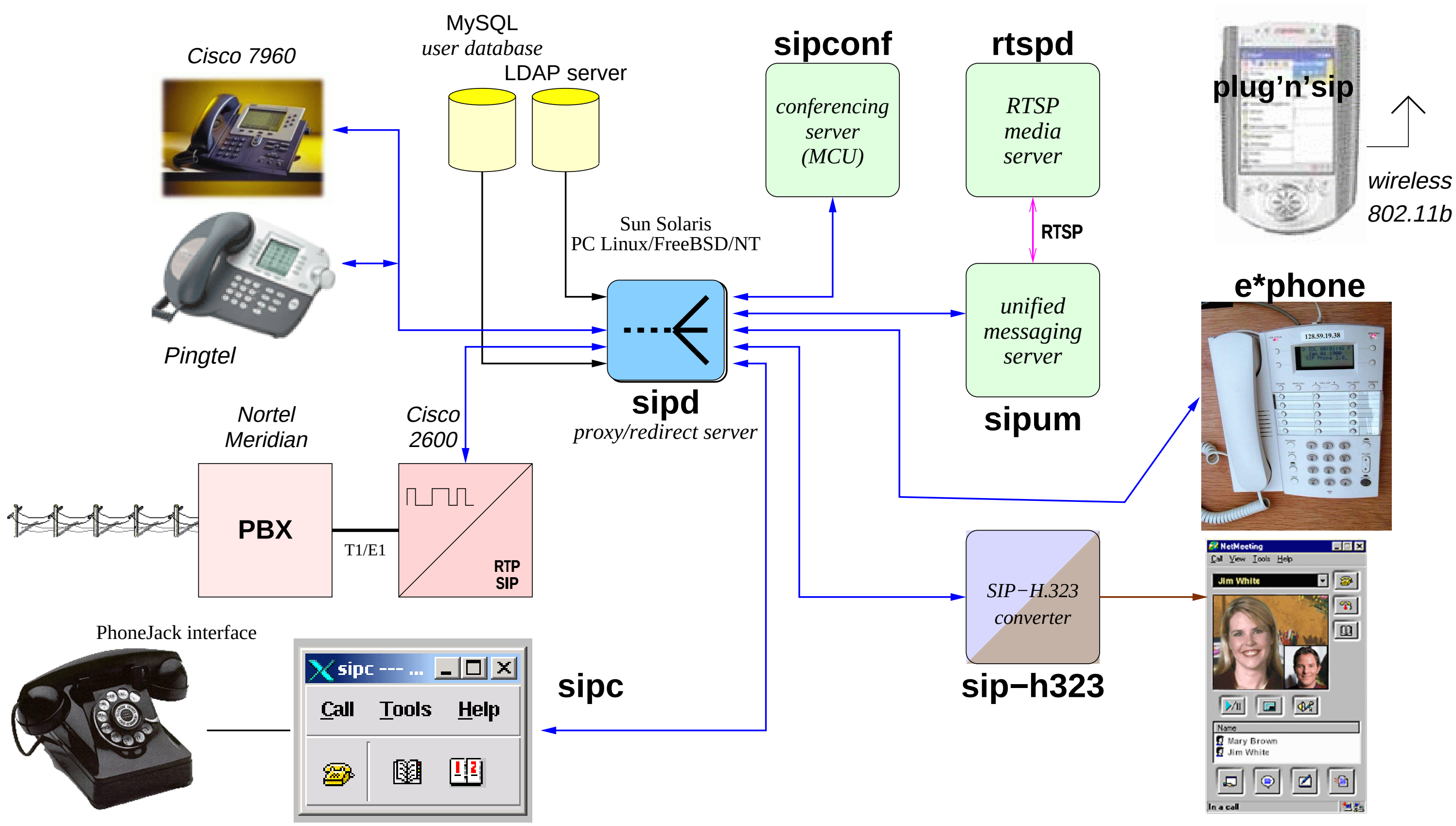
- presence is just a special case of events: "Alice just logged in" $\approx$ "temperature in boiler exceeds $300°$ F"
- need to *locate* mobile end points
- may need to find several different destinations ("forking")
- same addressing for users
- presence often precursor to calls
- may replace call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

Example home architecture



(Work with Telcordia)

Example: Columbia CS phone system



Conclusion

- many of the protocol building blocks in place
- VoIP challenges “dumbed down” Internet (NATs)
- deployment, standardization and research challenges

For more information. . .

SIP: http://www.cs.columbia.edu/sip

RTP: http://www.cs.columbia.edu/~hgs/rtp

Papers: http://www.cs.columbia.edu/IRT