The Effect of Standards on the Growth of IP Telephony

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IP Telephony Evolution – Palm Springs, CA

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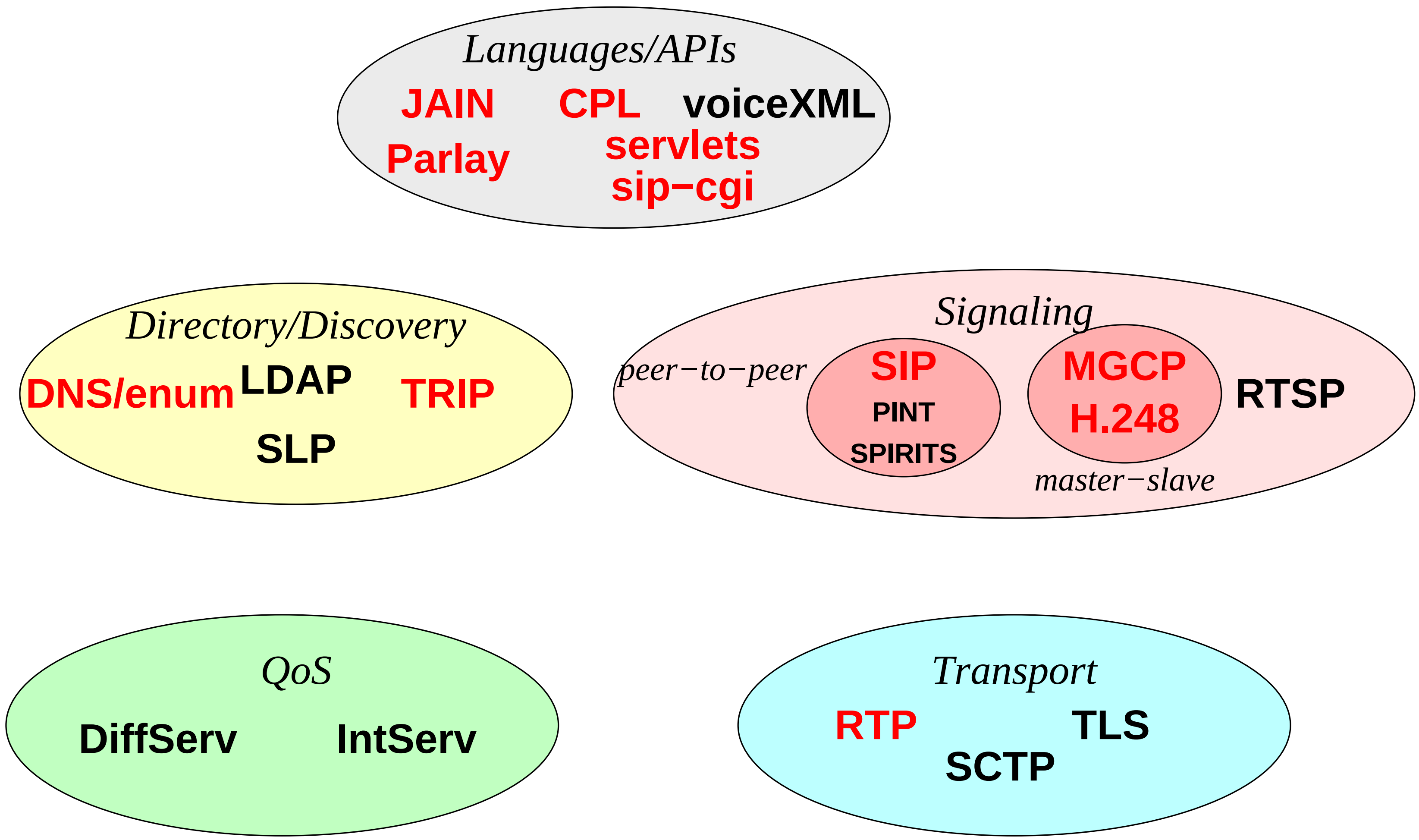
Overview

- Architectures: peer vs. master/slave
- Master/slave control: MGCP and Megaco
- Peer-to-peer signaling:
  - H.323
  - SIP
- Mapping between numbers and services: ENUM and TRIP
- Event notification and presence
- Programming multimedia services

IETF VoIP Protocol Architecture



IETF VoIP Protocol Architecture: Goals

- Leverage content-neutrality of Internet ➠ more than just voice and legacy services ➠ video, shared applications, multi-party text chat
- Imperceptible transition between communication modalities
- Extensible to presence, instant messaging and event notification
- Centrex lesson: user-controlled services
- Allow services in end systems and network servers
- Multiple levels of security: IPSec, TLS, application-layer

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

PSTN Legacies to Avoid

- E.164 numbers – might as well wear bar codes
- tones and announcements
- in-band signaling for features (DTMF)
- systems with user-interface knowledge (12 keys, voice)
- voice-only orientation (BICC, MGCP/Megaco)
- integration of bit transport and services
- service-specific billing ➠ separate signaling & billing
- trusted networks without crypto

➠ 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 and building sensors
- chat/IM tools
- interactive multiplayer games
- 3D worlds: proximity triggers call

Carrier and Enterprise VoIP

Traditionally,

- separate signaling: ISDN, CAS vs. ISUP
- service restrictions, e.g., CF inefficient

Now, largely the same:

- hosted ("ASP"), run own servers or combinations
- carrier: multiple domains per server
- if not outsourced, TRIP for gateway selection

Peer-to-Peer Architecture

- “IP telephones”, gateways, PCs with software = IP hosts
- *may* use servers (H.323 gatekeepers, SIP proxy servers)
- end system fully state-aware
- protocols for call setup: H.323 or SIP
- more flexible user interface

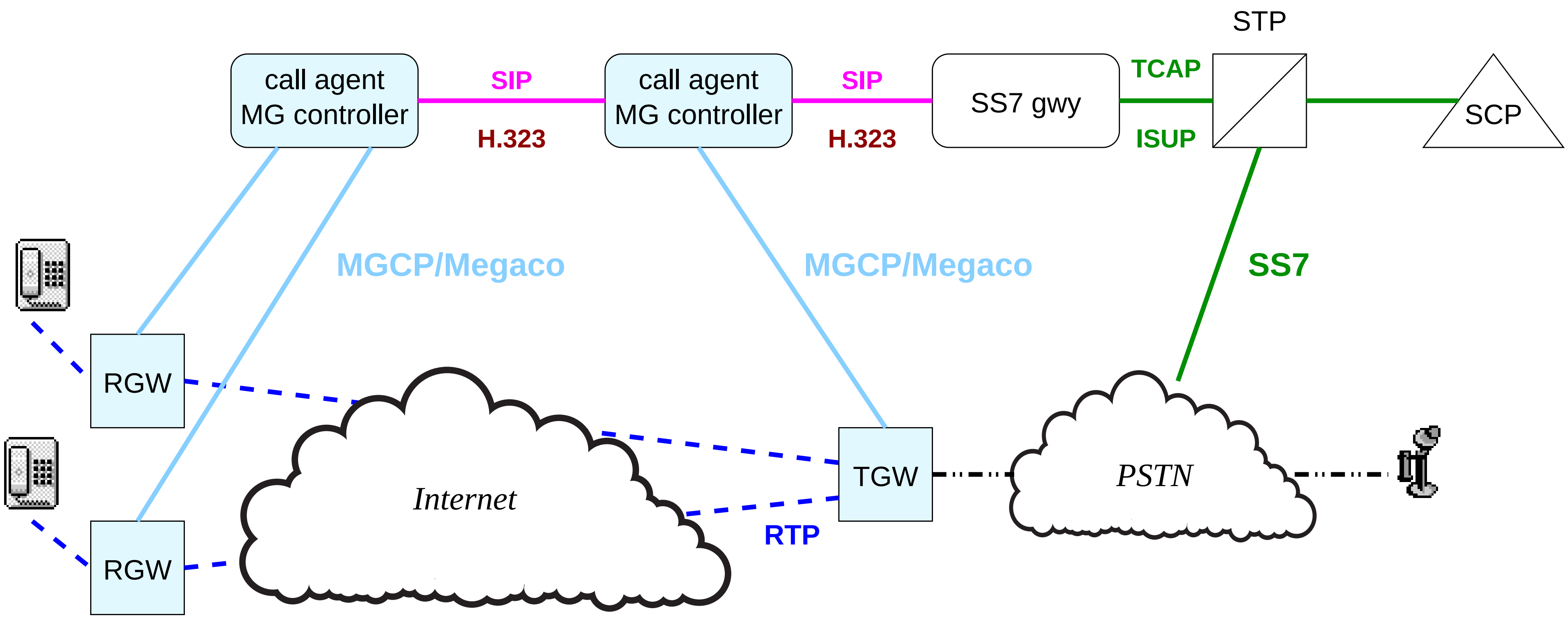
Implementing Services

	end system	server
caller id	x	–
call forwarding, follow me	x	x
three-way calling	x	–
distinctive ringing	x	–
69	x	?
no solicitation	x	x
do not disturb	x	x
call curfew	?	x

Master-Slave Architecture

- master-slave: MGC controls one or more gateways
- allows splitting of signaling and media functionality
- "please send audio from circuit 42 to 10.1.2.3"
- uses MGCP (implemented) or Megaco/H.248 (standardized, but just beginning to be implemented)
- gateway can be residential
- basis of PacketCable NCS (network control system) architecture
- service creation similar to digital PBX or switch
- $\longrightarrow$ can charge for caller id, call waiting

MGCP Architecture



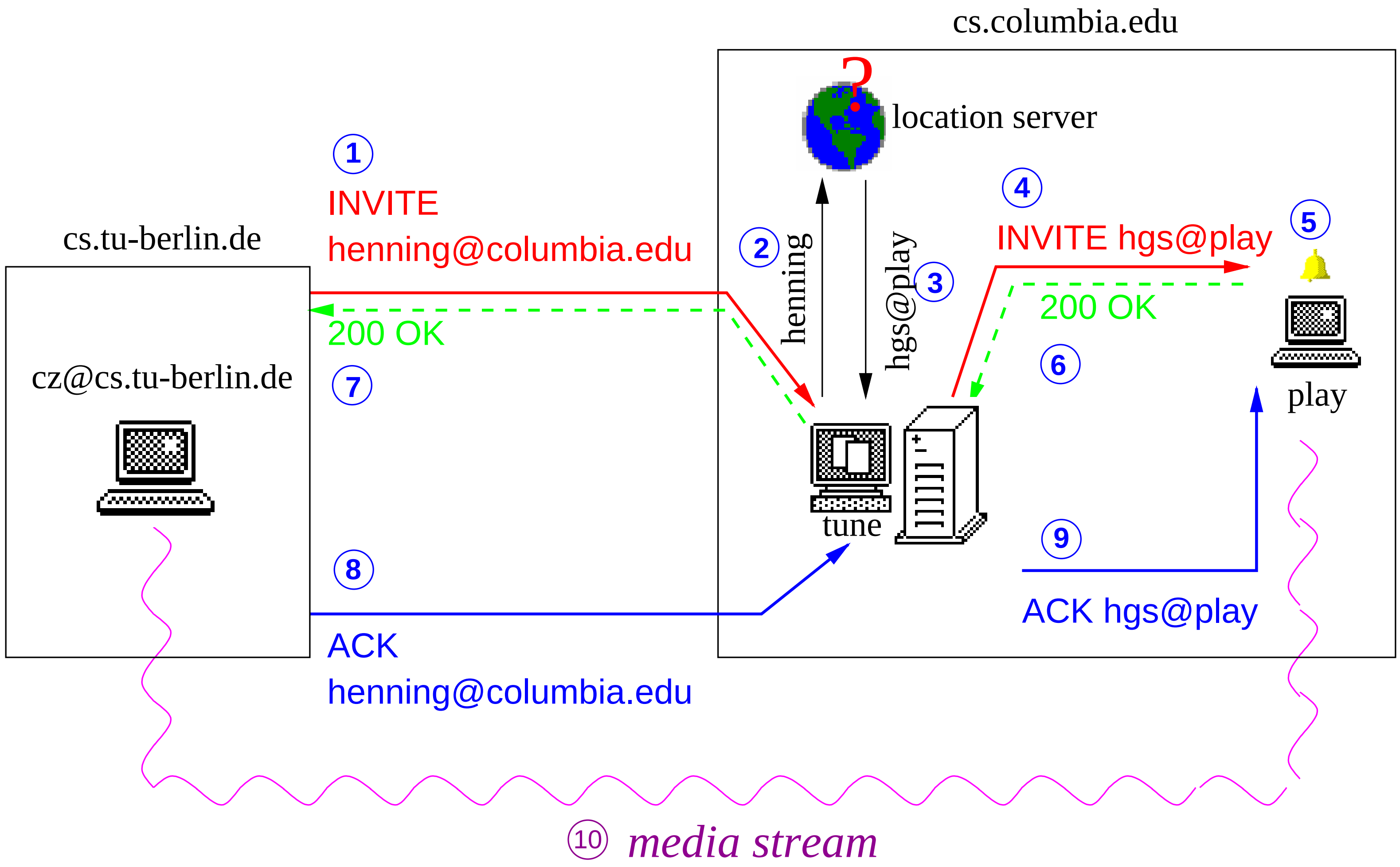
- for all but small system, need peer-to-peer!
- MGCP system can call SIP or H.323 end system
- all use RTP to transfer data

SIP 101

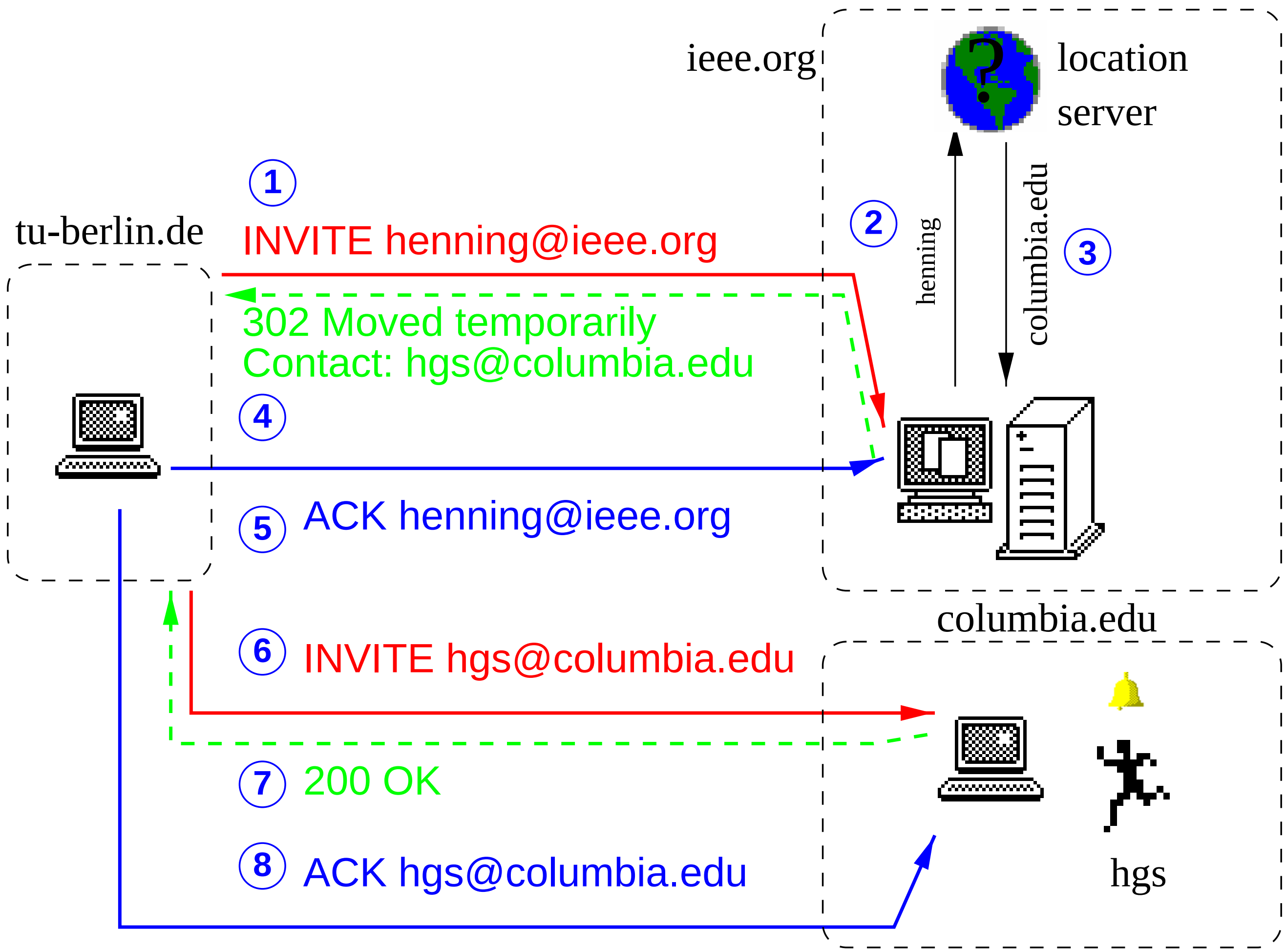
- SIP = signaling protocol for establishing sessions/calls/conferences/...
- session = audio, video, game, chat, ... – described by SDP carried in SIP message

1. called server may map name to *user@host*
2. callee accepts, rejects, forward ($\rightarrow$ new address)
3. if new address, go to step 1
4. if accept, caller confirms
5. ... conversation ...
6. caller or callee sends BYE

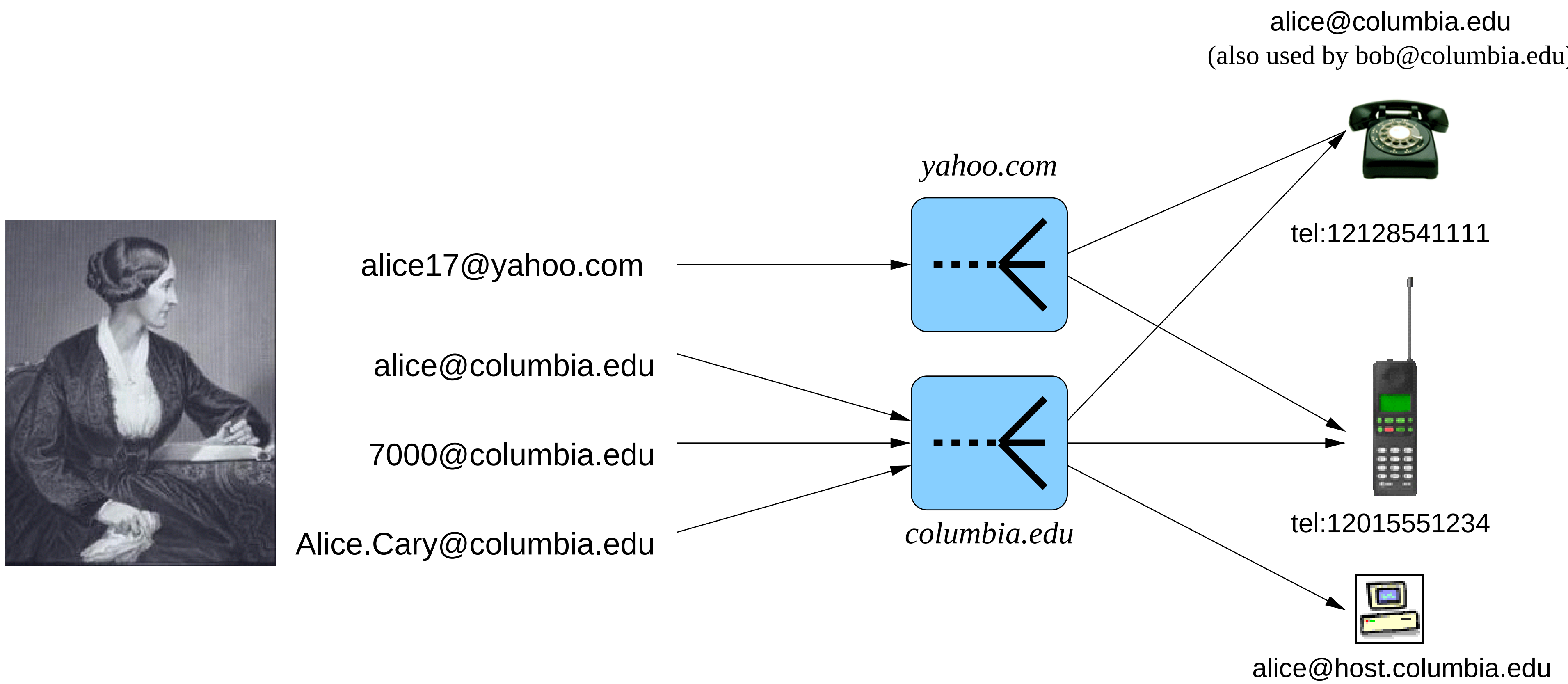
SIP Operation in Proxy Mode



SIP Operation in Redirect Mode



SIP Personal Mobility



More SIP 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”

Internet Telephony – as Part of Internet

- universal identifier: email address = SIP address = IM address
- SIP URLs in web pages
- forward to email, web page, chat session, . . .
- include web page in invitation response (“web IVR”)
- third-party control of calls via scripts,
- include vCard, photo URL in invitation
- user-programmable services: CGI (RFC 3050), CPL, servlets

SIP Status and Issues

- standard since early 1999, editorial revision in progress
- six bake-offs since 1999 for interoperability testing, about 60 companies attending
- work in progress:
  - interaction with resource reservation
  - caller preferences (“no mobile phones, please”)
  - interoperation with ISUP (“SIP-T”)
  - call transfer and third-party control
  - conferencing: central server, end system, full mesh
  - server benchmarking and scaling

SIP – H.323 Comparison

	H.323	SIP
Architecture	stack	element
Conference control	yes	no
Protocol	mostly TCP	mostly UDP
Encoding	ASN.1, Q.931	HTTPish
Emphasis	telephony	multimedia, multicast
Address	flat alias, E.164	SIP, E.164 URLs

Both SIP and H.323 are evolving: SIP additions, H.323v2 implements some SIP features.

A Signaling Protocol Prediction

- H.323 continues to be used for simple gateways, conferencing systems
- most large vendors produce both
- IP “PBXs” (IP phones) primarily SIP
- SIP as core signaling protocol for 3G networks
- less use of MGCP/Megaco, except for very large gateways
- BICC, SCTP only for legacy-burdened carriers

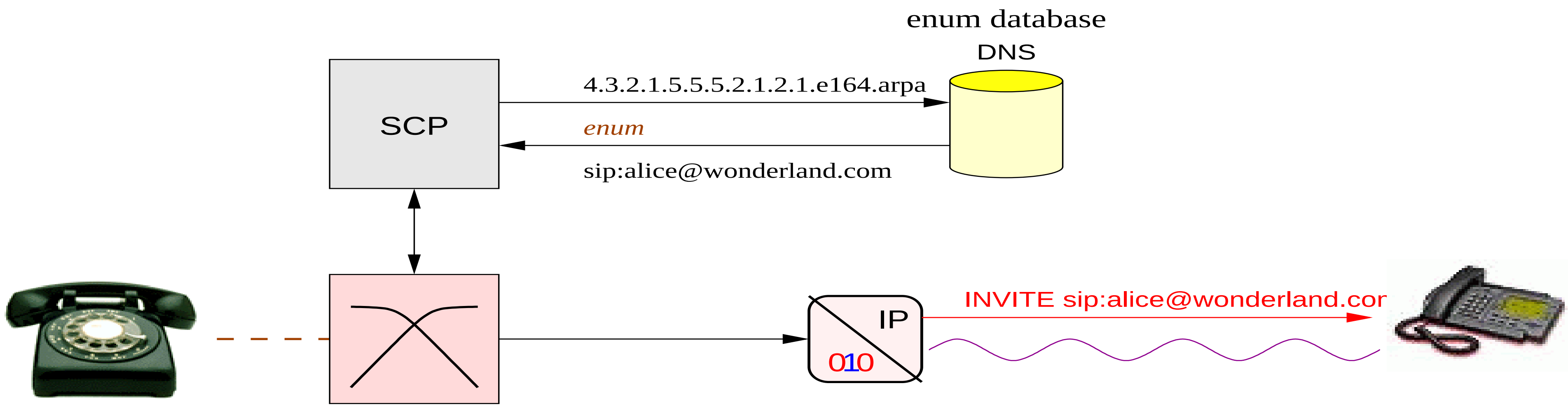
Legacy Signaling Transport

- Signaling Control Transport Protocol (SCTP)
- like TCP, but with enhancements for out-of-order delivery and multiple end points
  ➠ generally useful (e.g., can carry SIP)
- primarily useful for ISUP transport over IP
- does *not* mean use of ISUP for setting up VoIP sessions
- “SIP in the middle”: carry ISUP in message body

ENUM: Mapping E.164 Numbers to URLs

Map single E.164 number to one or more URLs (e.g., SIP, mailto, H.323)

- transition from E.164 numbers to SIP URLs
- voice mail drop address
- number portability implementation



Mapping E.164 Numbers to URLs

ENUM *not* suited for finding gateways:

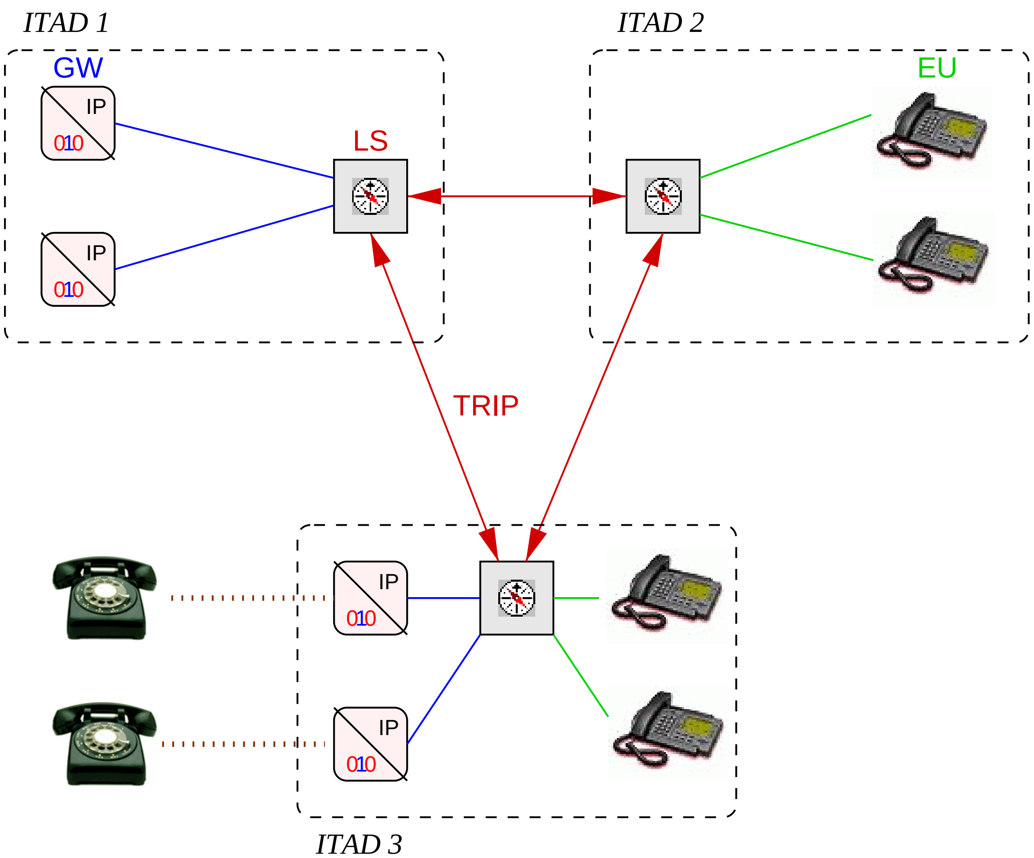
- would need entry for every number
- single mapping only
- no ability to convey parameters

Problems to be solved:

- Who manages country-level domains? Blocks of numbers?
- Who “owns” a number - customer or carrier?
- “Slamming” of services

Determining the Best VoIP Gateway

Every gateway can reach almost any E.164 address ➠ pick closest, cheapest, preferred signaling,



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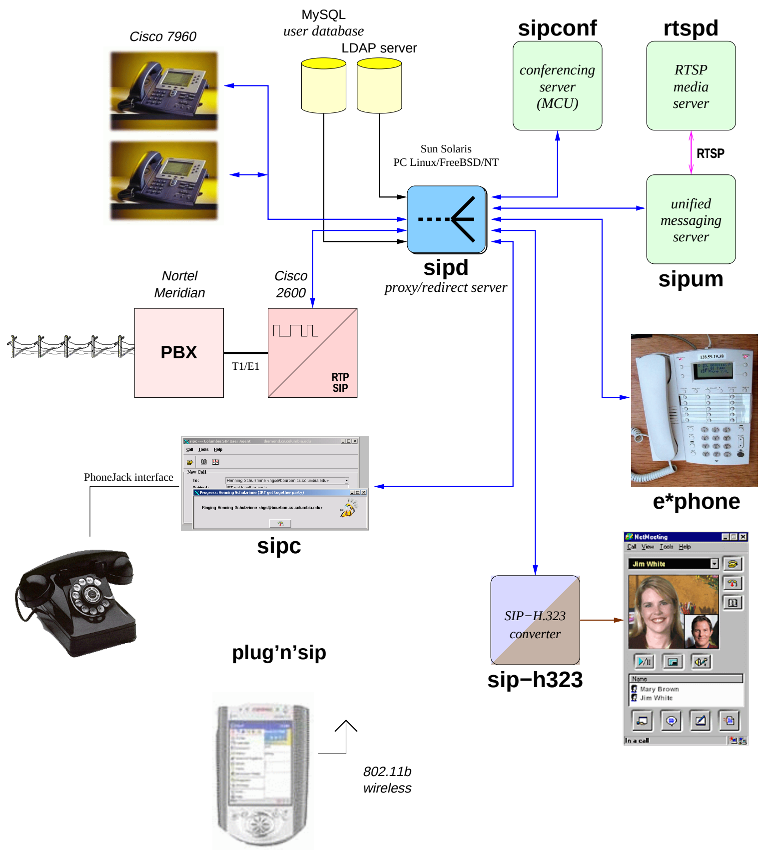
Example: Columbia CS Phone System

Expand existing PBX via IP phones, with transparent connectivity



The Largest Signaling Network Does Not Run SS7

- AT&T: 280 million calls a day
- AOL: 110 million emails/day, total about 18 billion/day
- total $>$ 1 billion instant messages/day (AOL: 500 million)
- telephony signaling $\approx$ IM, presence

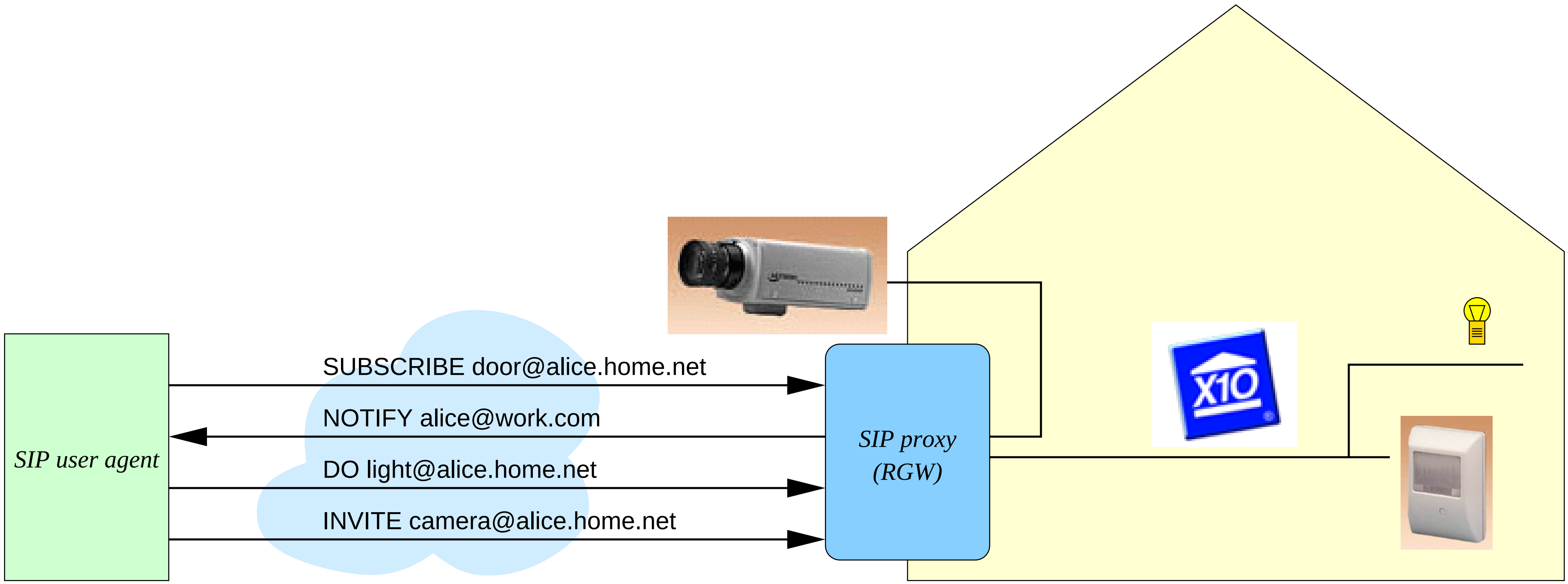
Commonalities Between Signaling and Events

- presence is just a special case of an event: “Alice logged in” $\approx$ “house temperature dropped below 50 deg.”
- need to locate mobile end points (for notifications)
- may need to find several different destinations
- 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, conf. joins

SIP as a Presence & Event Platform

- minimal SIP extension: SUBSCRIBE to request notifcations, NOTIFY when event occurs
- also, MESSAGE for IM, sessions for multi-party chats
- transition to true “chat” (and video)
- services such as reaching mobile phone while in meeting

Events: SIP for Appliances



Programming Internet Multimedia Services

Primarily, creation, forwarding, proxying, rejection of calls

APIs (Parlay, JAIN): protocol-neutral (SIP, H.323, ISUP), but may be least common denominator

SIP CGI: use Perl and other scripting languages; easy to learn

Servlets: Java only; faster than cgi; limited functionality

CPL: = XML-based language for *user* service creation; portable across providers, but not all services

- Protocol-neutral: Parlay, JAIN, CPL
- Call creation: Parlay, JAIN
- VoiceXML is for voice-service creation *after* call setup

Conclusion

- basic IETF-based architecture in place
- SIP as foundation for services
- extensions for mobility, emergency services, . . . in progress
- first (and last?) chance to recover from 120 years of legacy
- avoid replication of PSTN on packets
- most VoIP applications won't look like telephones
- range of engagement and asynchronicity, from call to IM to email
- challenge of mobile services

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