Internet Telephony: A Second Chance

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Overview

- the danger of VoIP
- challenges on the horizon:
  - instant messaging & presence
  - generic event notification
  - integration with 2G mobile (GSM, CDMA)
  - next-generation wireless (3GPP, 3GPP2, MWIF, . . .)
  - emergency services
- reaching interoperability: SIP bake-offs

The Dangers of VoIP

- focus on single service: voice, fax, ...
- PSTN: service orientation $\longleftrightarrow$ Internet: neutral transport
- APIs as least common denominator across POTS, ISDN, SS7 $\longrightarrow$ 100-year old functionality
- carbon-copy replication of existing services
- terminology overload

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

PSTN Legacies to Avoid

- E.164 numbers – 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

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., REFER for three-party calls and various call transfers)

Terminology Overload

Invasion of the meaningless technical-sounding terms, attempting to familiar mimic PSTN boxes:

- CO switch ⟶ soft switches = gateway + SIP UA + ?
- SCP ⟶ application servers = proxy? web server? media server?
- PBX ⟶ Internet PBX = proxy? + gateway?
- …

Temptation: new name ⟶ new protocols, APIs, … – the old box boundaries don't necessarily make sense!

It’s That Simple…

We really only have a few basic components:

- PSTN gateway, with some combination of FXO/FXS
- SIP proxy/redirect/registrar servers (or H.323 gatekeepers)
- SIP user agents (or H.323 terminals): PCs, phones
- media storage servers
- DNS, directory, web, email, news, …servers

Invisible Internet Telephony

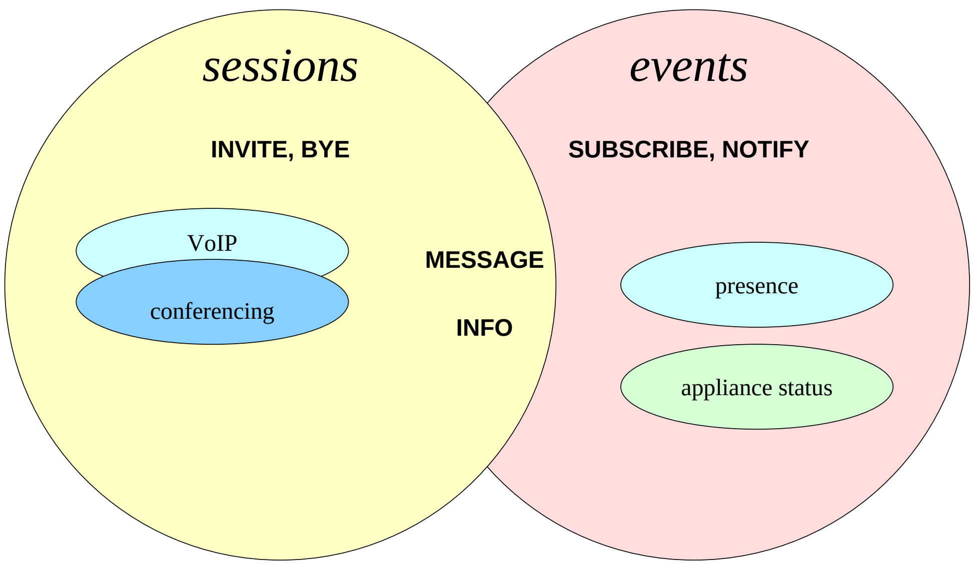
VoIP technology will appear in . . .

- 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

Signaling and Events



Signaling: “do this” (push) – Events: “this just happened”

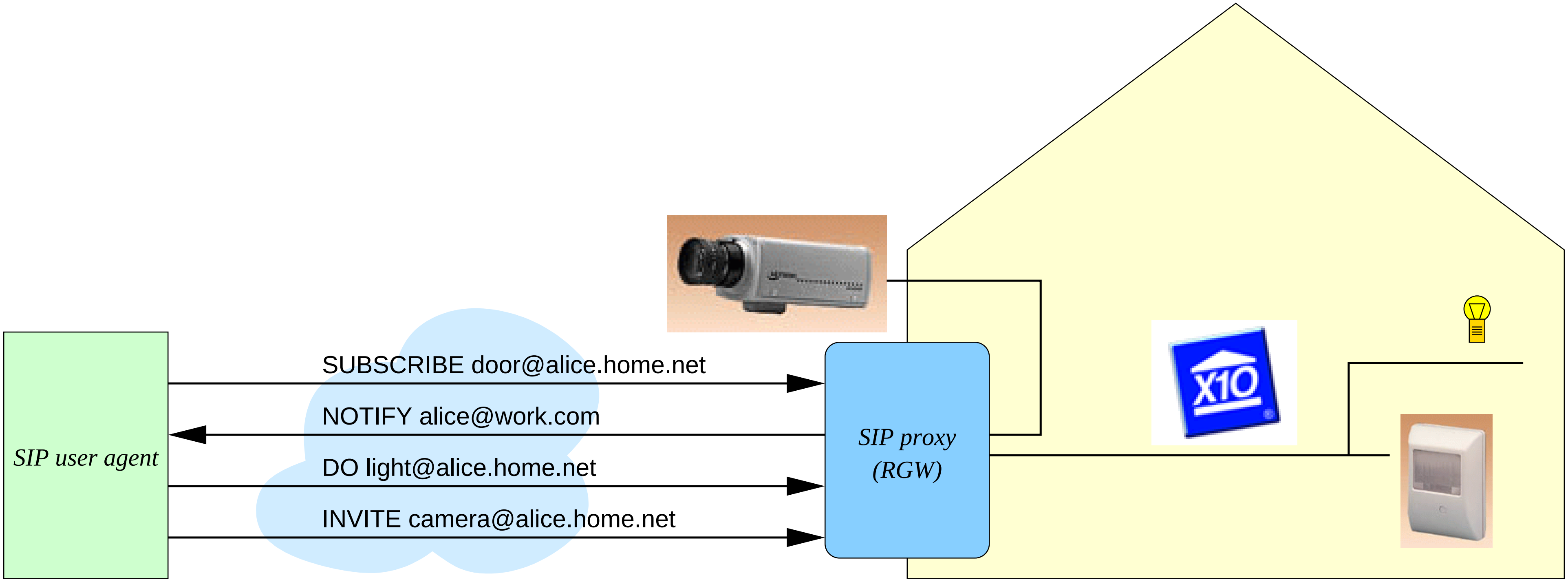
Commonalities between Signaling and Events

- 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

SIP as a Presence Platform

- requires minimal extensions to SIP: SUBSCRIBE to ask to be alerted, NOTIFY when event occurs
- MESSAGE for sending text messages (“IM”)
- true “chat” is voice (+ video)
- services such as reaching mobile phone while in meeting

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

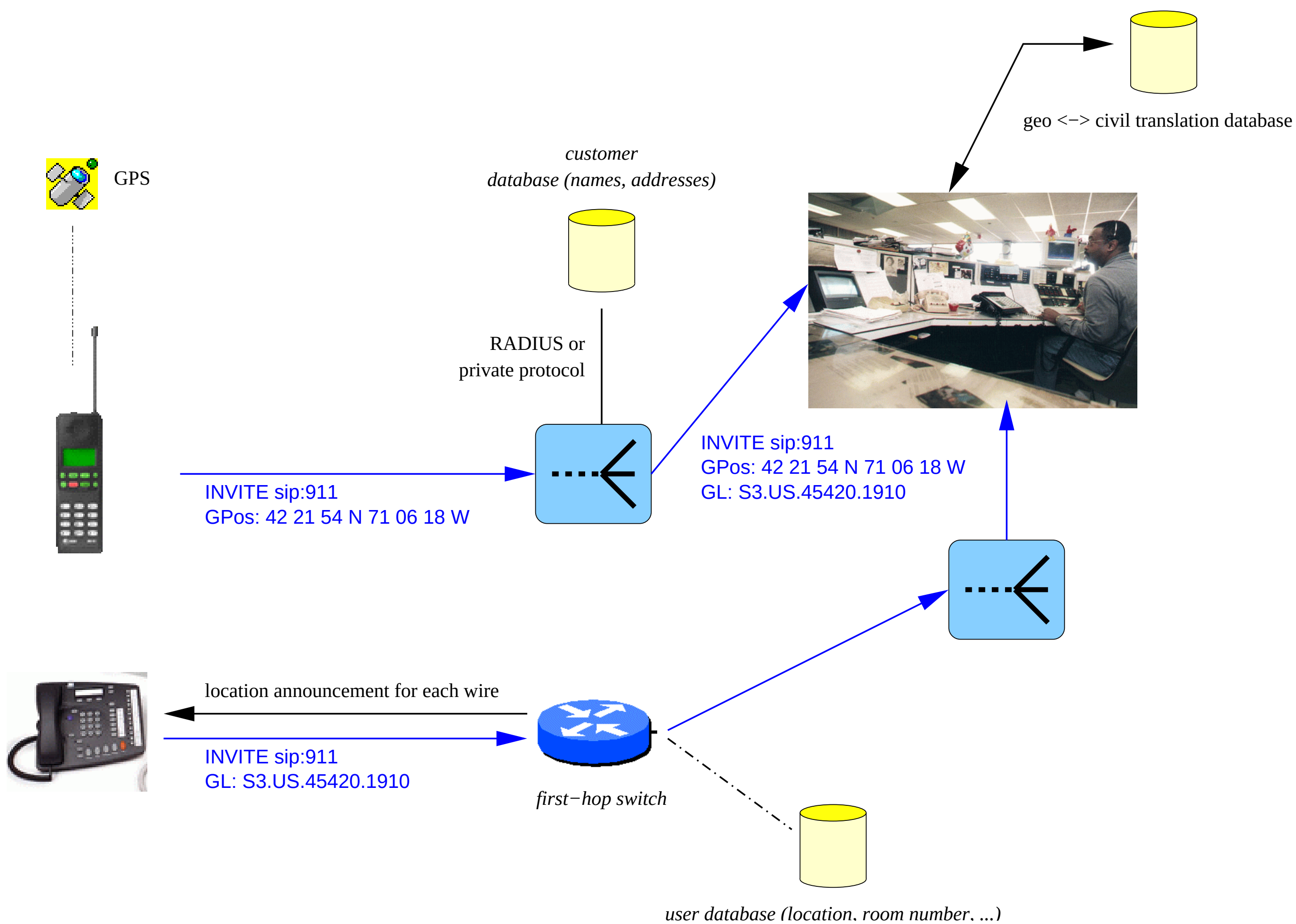
Third-Generation Wireless

- goal: 144 kb/s moving, 384 kb/s stationary, 2 Mb/s indoors
- based on GSM or wideband CDMA
- push IP to the hand set
- SIP as signaling system for voice calls in 3GPP

SIP Emergency Services

- need
  - emergency address
  - find nearest PSAP
  - PSAP determines caller location
- cannot just rely on gateway calling 911
- generally, allow devices to be location-aware ("what time is it where I'm about to call?" "call pizza parlor")
- offers new opportunities: database access, video, measurements, accessibility, ...

SIP Emergency Services



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, . . .)

Conclusion

- first (and last?) chance to recover from 120 years of legacy
- avoid replication of PSTN on packets
- most VoIP applications won't look like a telephone
- opportunities in emergency services, mobile, event notification