A SIP-based Architecture for Internet Telephony

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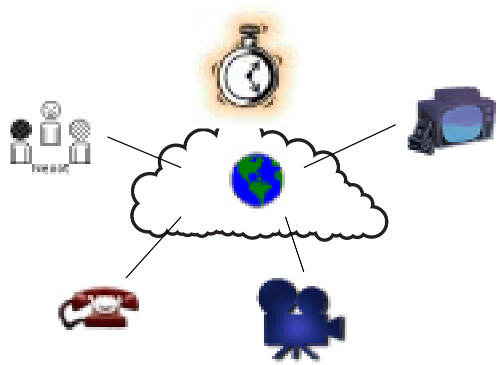

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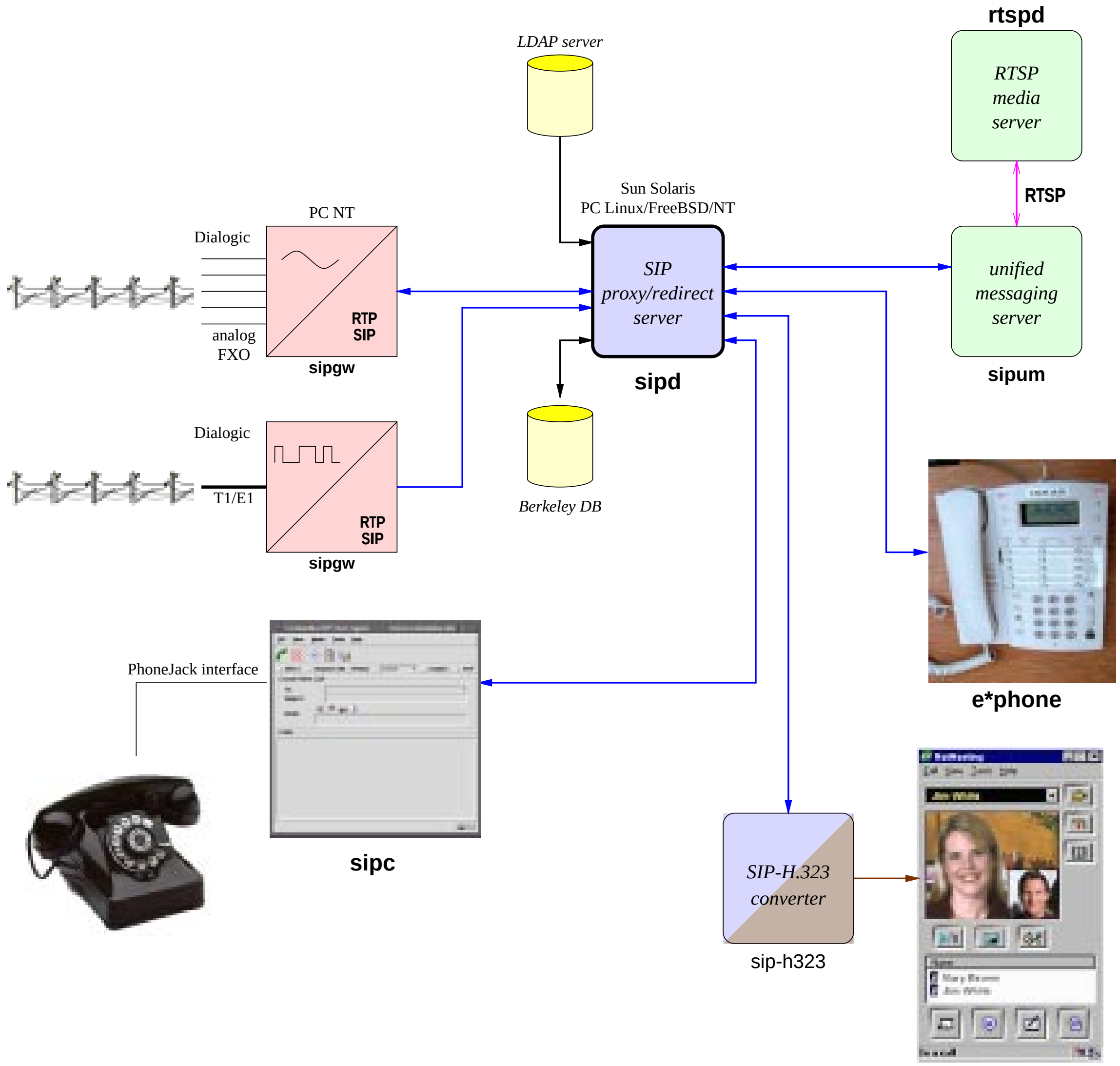
(Contributors: Jonathan Lennox, Gautam Nair, Pallavi Nayak, Jonathan Rosenberg, Kundan Singh, Elin Wedlund, and Jianqi Yin)

SIP

- SIP = modular, programmable, web-centric protocol for creating multimedia sessions
- Internet standard
- wide industry support: 50 implementations at 4th SIP bake-off
- will be used for 3rd generation wireless network
- customer network trials likely later this year


rtspd
RTSP media server
RTSP
LDAP server
Sun Solaris
PC Linux/FreeBSD/NT
SIP proxy/redirect server
sipd
unified messaging server
sipum
PC NT
Dialogic
analog
FXO
RTP
SIP
sipgw
Dialogic
T1/E1
RTP
SIP
sipgw
Berkeley DB
e*phone
PhoneJack interface
sipc
SIP-H.323 converter
sip-h323

SIP architecture

- goal: construct a complete SIP-based Internet telephony suite
  - e*phone as PBX replacement
  - SIP server for creating services
  - multimedia client for workstations and PCs
  - library for building new applications
  - SIP multiparty conferencing application
  - telephone gateway
  - standards-based unified messaging
  - location-based SIP services

SIP phone: e*phone

- first SIP-based Ethernet phone
- same device also works as MP3 radio
- sensor interfaces to the world: IR, temperature, alarms, medical measurements, . . .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SIP server (sipd)

- core component for routing SIP calls
- translates names and numbers
- filters and routes calls, e.g., based on caller, time of day, urgency, . . .
- programmable services via web-like scripting
- 90 academic and commercial licenses

SIP client (sipc)

- cross-platform client for creating multimedia sessions
- any media type can be added via external programs, including audio, video, chat, whiteboard, . . .
- feature programming and calendar interfaces in progress

SIP-H.323 gateway

- allow SIP devices to communicate with H.323 devices, such as Microsoft NetMeeting
- single server sufficient for whole organization

SIP conferencing bridge

- set up conference via web page
- allow “dial-in” conferences
- mixes audio (and video) streams to all participants
- can be used locally or offered as a service

SIP unified messaging

- calls are redirected to SIP messaging server
- server instructs media server to record – any media (audio, video, chat, . . .)
- can be retrieved from web page or via email link
- use any RTSP player, e.g., QuickTime, to play messages from anywhere

SIP unified messaging – web interface