SIP Status and Directions

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(with material from Jonathan Rosenberg)

Overview

- SIP – what's it good for (and not)
- SIP IETF standardization work
- SIP products and bake-offs
- SIP-H.323 interworking

What is SIP good at?

- session setup = “out of band”
- resource location via location-independent identifier (“user@domain”, tel)
- particularly if location varies rapidly or filtering is needed (i.e., is inappropriate for DNS and LDAP)
- real-time: faster than email
- reach multiple end point simultaneously or in sequence = *forking*
- possibly hide end-point location
- delayed final answer (“ringing”) $\longleftrightarrow$ RTSP

What is SIP not meant for?

- bulk transport: media streams, files, pictures, . . .
- asynchronous messaging (“email”)
- resource reservation
- high-efficiency general-purpose RPC

Current SIP working group status

- nearly 200 attendees at IETF 47 in Adelaide (March 2000)
- 31 active Internet drafts
- of which about 20 are WG work items
- design teams focusing on security, home networks, SIP-H.323, ...
- want to finish all of this this year...

Current SIP efforts

- SIP to Draft Standard
- QoS and security preconditions
- inter-domain AAA and billing
- session timer for liveness detection
- early media (PSTN announcements)
- SIP for presence / instant messaging
- SIP-H.323 interworking
- SIP MIB
- reliable provisional responses
- DHCP configuration for finding SIP servers
- SIP for firewalls and NATs
- caller preferences
- services (transfer, multiparty calls, third-party, home)
- ISUP carriage
- “911”

Management and auto-configuration

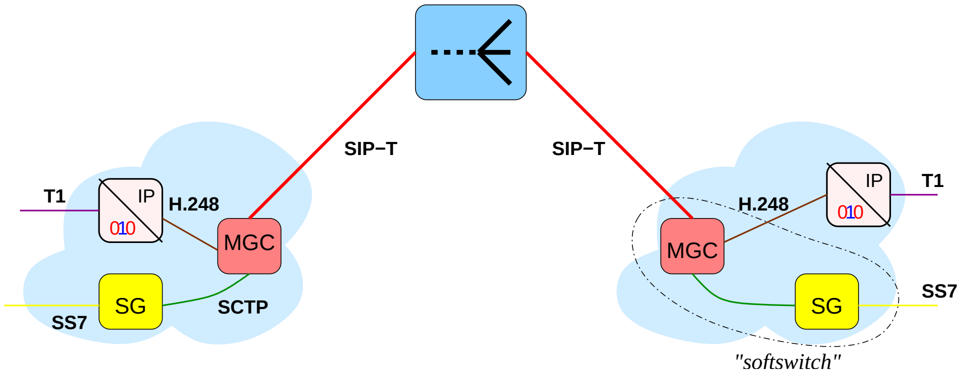
- SIP MIB
  - management of proxy, redirect, registrar and user agents
  - based on existing early MIBs
  - monitoring status, ports, URI types, statistics (transactions, requests, responses), pending transactions, . . .
- DHCP option for SIP servers
  - user agent learns where to register and find outbound proxy
  - easily added to existing DHCP servers
  - in IESG review

Management and auto-configuration

- Service Location Protocol (SLP) templates
  - SLP allows clients to find local servers matching criteria
  - SLP template for SIP:
    * IPsec and TLS transport support
    * CPL support
    * caller preferences
- template already registered with IANA

SIP-T

- ISUP transparency
- INFO method for mid-call messages
- ISUP $\leftrightarrow$ SIP conversion
- MIME definition for ISUP payload
- overall architecture document



SIP extensions: reliable provisional responses

- SIP provisional (180, 183, ...) responses are not reliable
- sometimes needed for ringing and queueing status
- particularly for transparent PSTN bridging
- extension requests acknowledgement (PRACK)
- also used by SIP QoS extension

in WG last call

SIP extensions: session timer

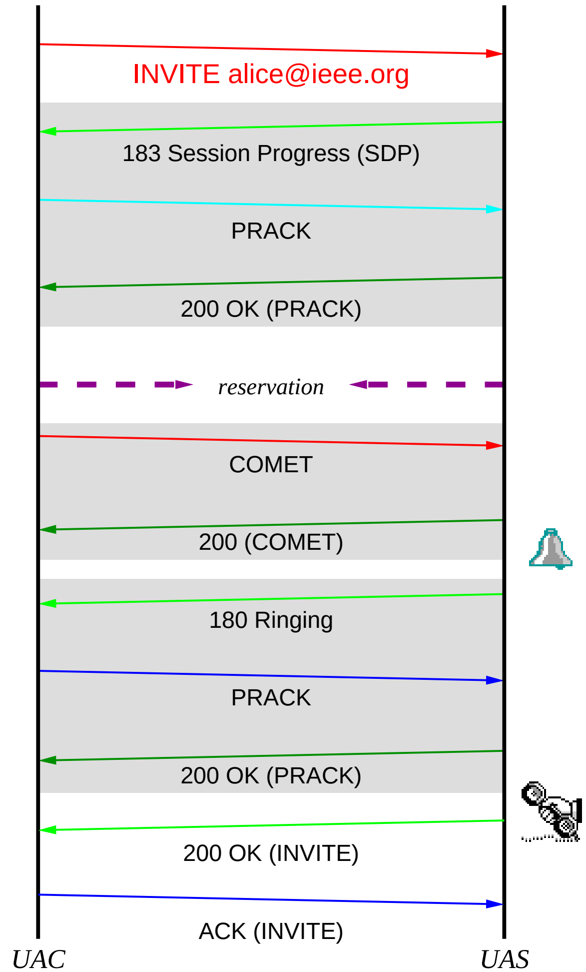
- there are no SIP messages during a session $\longrightarrow$ can't detect whether other side is still alive
- gateways can/should use media activity
- needed for firewalls and billing
- session timer asks for periodic invitation refreshes
- also allows recovery from callee system crashes

SIP extensions: caller preferences

- generic address: `alice@wonderland.com`
- caller may want to restrict destination selection
  - home or work
  - fax, audio, video, text, . . . call
  - mobile or landline
  - language spoken
  - secretary or voicemail
  - avoid re-visiting old locations
- rules carried in INVITE request

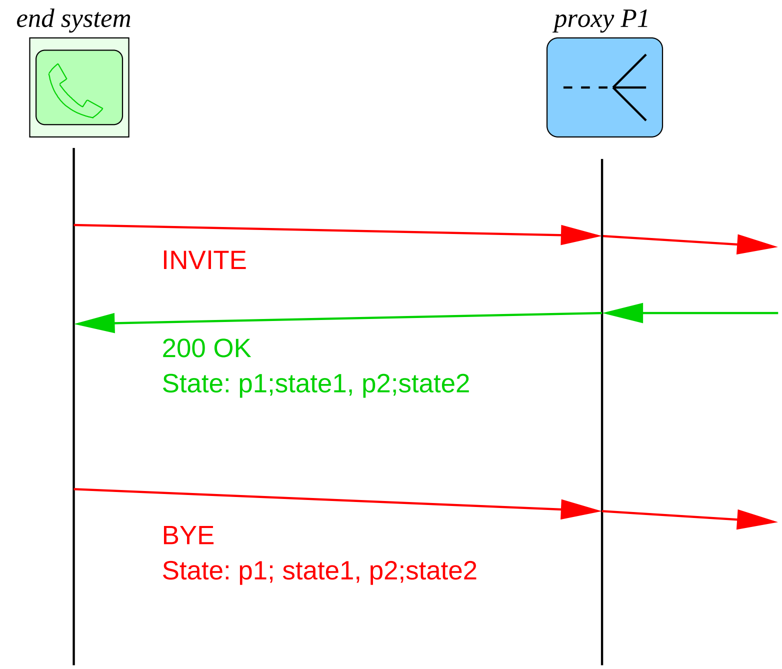
SIP extensions: SIP and resource reservation

- problem:
  - resource reservation and call signaling are separate
  - separate machinery, path
  - call setup needed to get IP addresses
  - avoid successful call, failed reservation
- $\longrightarrow$ couple at end systems
  - pre-conditions for call setup (also: security)
  - COMET indicates success



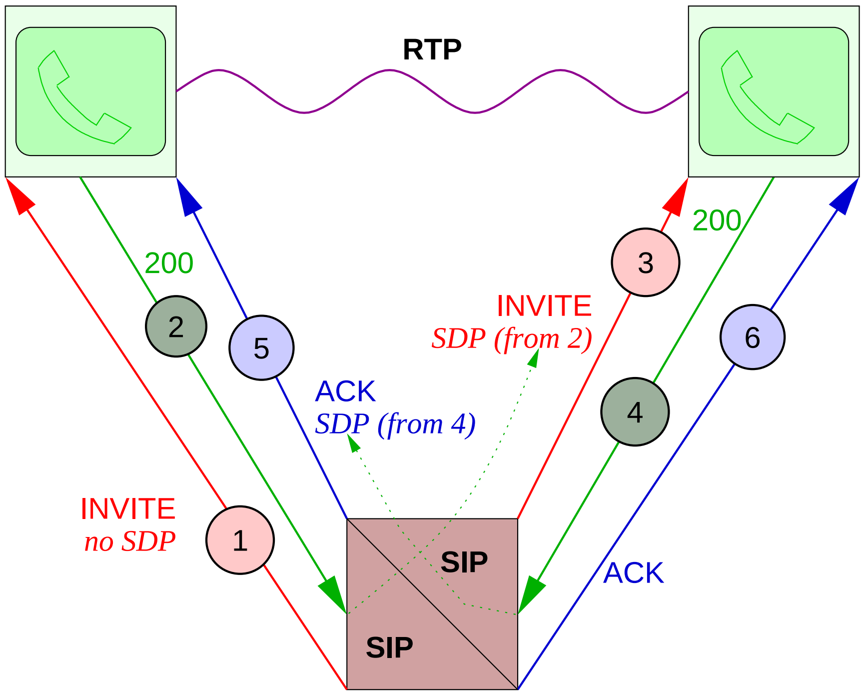
SIP distributed state

- HTTP “cookies” store server state on client
  - server asks client to store data
  - client inserts data into requests
  - cookie opaque to client
- also useful for SIP sessions:
  - session management
  - fault tolerance (“fail over”)
  - scalability
- for SIP:
  - proxies create data, UAs store
  - repeat for same call



SIP third-party call control

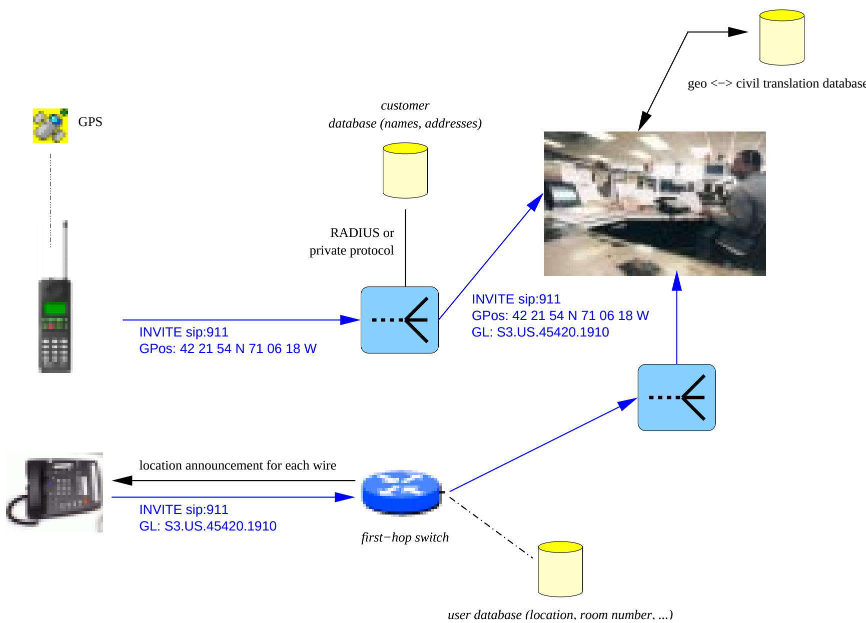
- some services require a third party to create a session between users
  - IVR services
  - click-to-dial
  - prepaid calling
- 3rd party call control
  - needs no SIP extensions
  - just copies SDP from one "leg" to another



SIP 911 service

Internet-based emergency call service

- uniform emergency “number”
- locate nearest public safety answering point (PSAP)
- convey user location to PSAP



Status

- Proposed Standard, Feb. 1999 – RFC2543
- bakeoffs every 4 months $\longrightarrow$ cross-vendor interoperability tests

	host	when	companies
1	Columbia University	April 1999	16
2	pulver.com	August 1999	15
3	Ericsson	December 1999	26
4	3Com	April 2000	36
5	pulver.com	August 2000	
6	Sylantro	December 2000	
7	ETSI	April 2001	

SIP implementations

Roughly in order of maturity:

- proxies and redirect servers for service creation
- PC-based user agents – Windows and other OS
- Ethernet phones
- softswitches (Megaco/MGCP/...) “crossbar”
- protocol analyzers
- firewall and NAT enhancements
- SIP-H.323 gateways
- unified messaging

On-going SIP implementations

3Com
AudioTalk Networks
Broadsoft
Catapult
Cisco
Carnegie-Mellon University
Columbia University
Delta Information Systems
dynamicsoft
Ellemtel
Ericsson
Hewlett-Packard

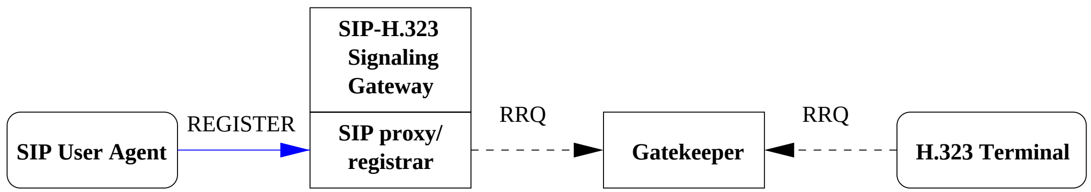
Hughes Software Systems
Indigo Software
Iwatsu Electric
Komodo
Lucent
MCI Worldcom
Mediatrix
Microappliances
Netergy
Netspeak
Nokia

ObjectSoftware
Nortel
Nuera
Pingtel
RaveTel
Siemens
Telogy
Ubiquity
Vegastream
Vovida

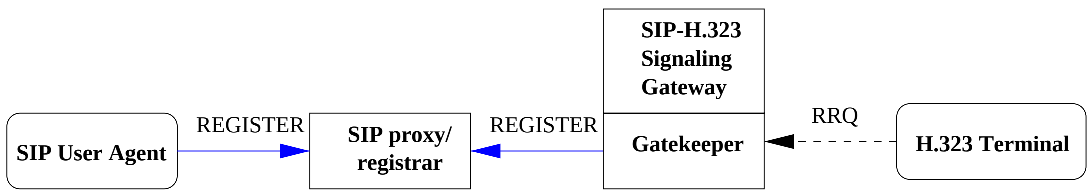
SIP-H.323 interworking

- media translation – not necessary $\longrightarrow$ much better scaling
- signaling translation – easier as H.323 version increases...
- user registration:
  - enum (DNS) – per host only, requires awareness
  - export registrations in either direction
- advanced services – not yet clear

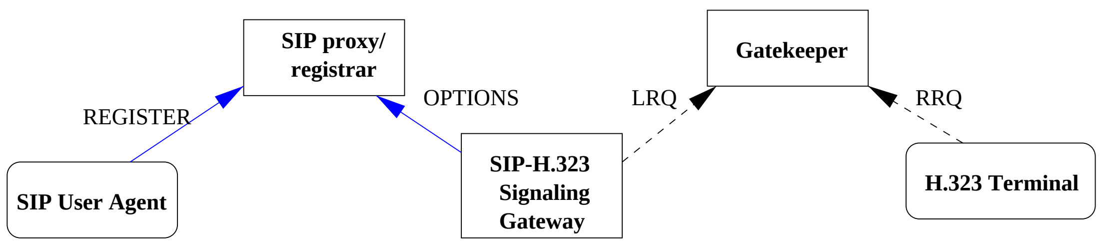
SIP-H.323 interworking



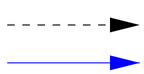
(a) Signaling gateway contains SIP proxy



(b) Signaling gateway contains an H.323 gatekeeper



(c) Signaling gateway is independent of proxy or gatekeeper



LRQ = Location request
RRQ = Registration request

Conclusion

- SIP is ready for large-scale deployment
- wide diversity of implementations, rapidly moving from bake-off to buyable
- focus on interoperability
- emphasis on one core version with negotiated extensions – no SIP versioning, profiles, ... $\longrightarrow$ goal: every SIP-powered device and software can interwork with any other
- extensions for QoS, ISUP carriage, events
- some services, such as transfer, need finishing up
- leverage event model for remote pick-up and other advanced 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