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
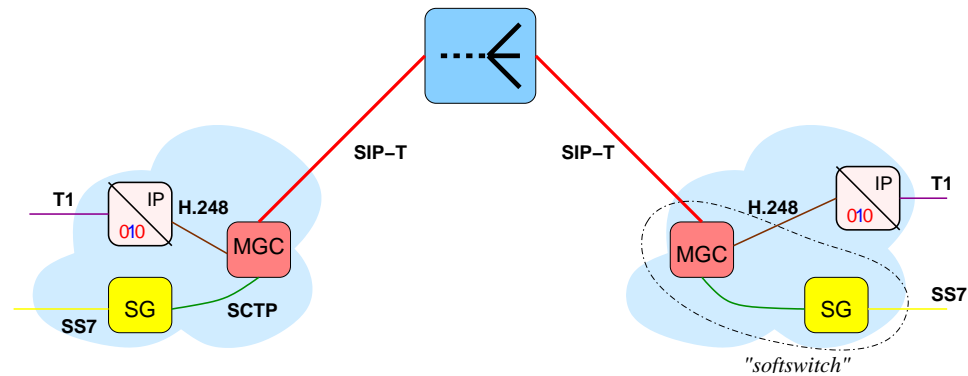
SIP extensions: reliable provisional responses

- 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

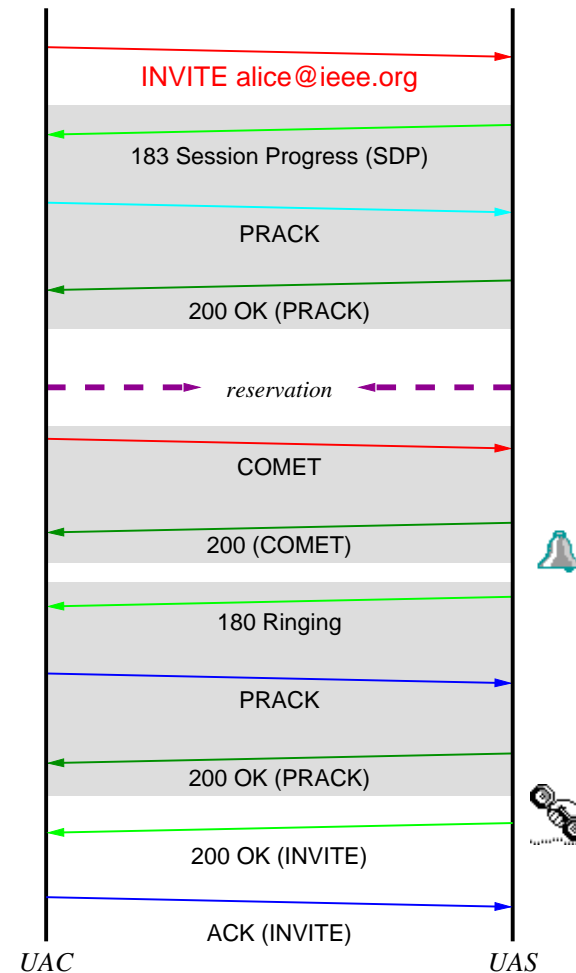- also allows recovery from callee system crashes

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
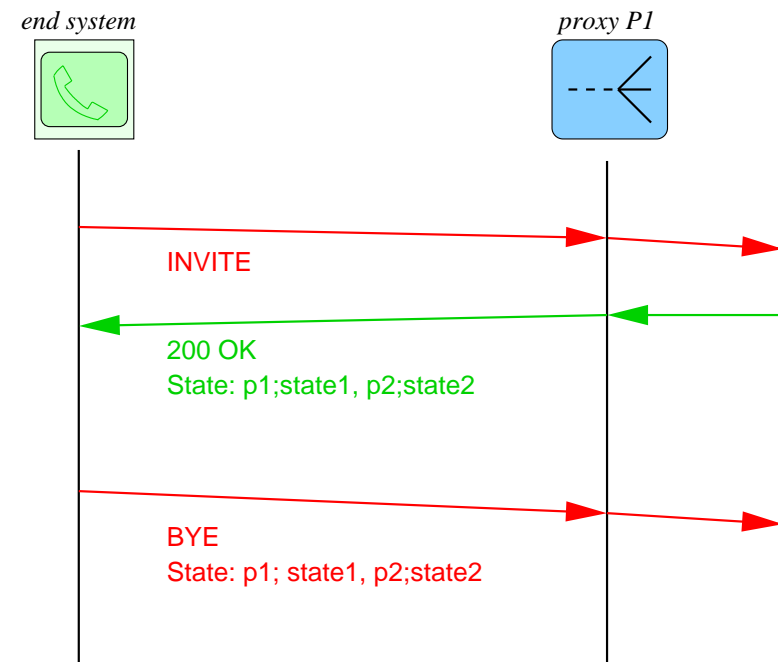
- rules carried in INVITE request

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

INVITE alice@ieee.org

183 Session Progress (SDP)

PRACK

200 OK (PRACK)

reservation

COMET

200 (COMET)

180 Ringing

PRACK

200 OK (PRACK)

200 OK (INVITE)

ACK (INVITE)

UAC *UAS*

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

end system

proxy P1

INVITE

200 OK
State: p1;state1, p2;state2

BYE
State: p1; state1, p2;state2
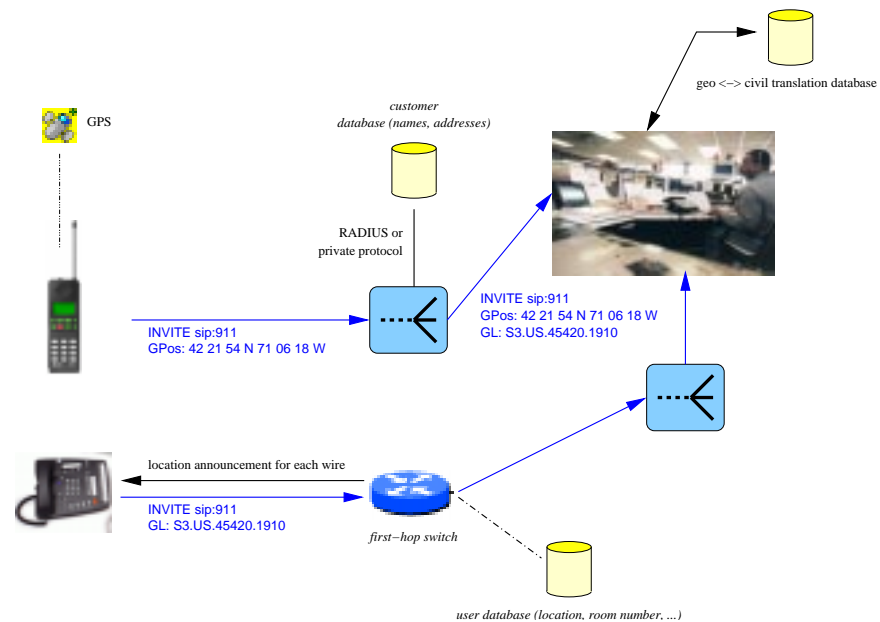
SIP third-party call control

- 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

RTP

200

INVITE
SDP (from 2)

2

5

3

6

ACK
SDP (from 4)

4

INVITE
no SDP

1

SIP

SIP

ACK
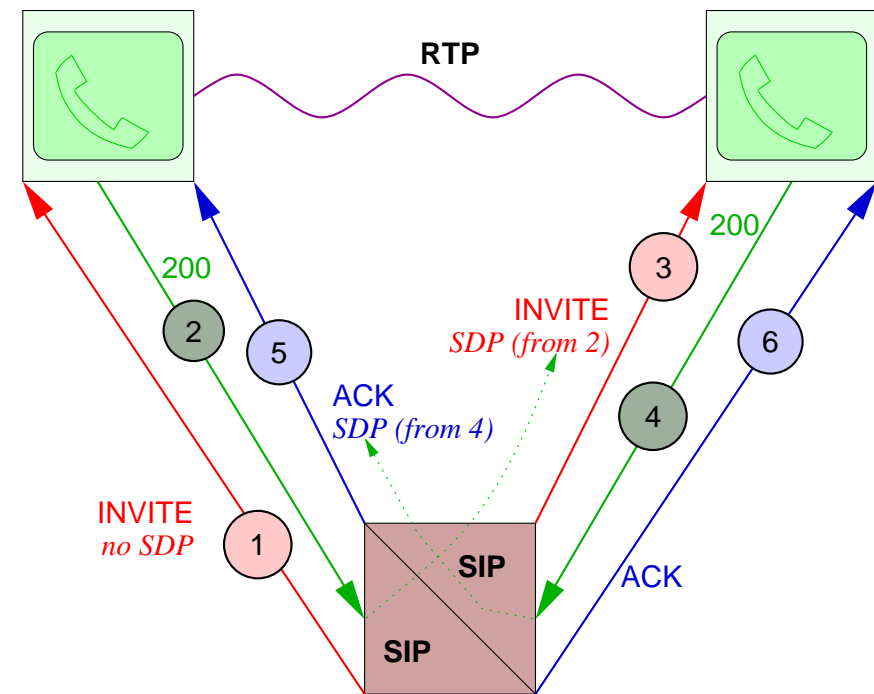
SIP 911 service

Internet-based emergency call service

- uniform emergency "number"

- locate nearest public safety answering point (PSAP)

- convey user location to PSAP

GPS

customer
database (names, addresses)

geo <–> civil translation database

RADIUS or
private protocol

INVITE sip:911
GPos: 42 21 54 N 71 06 18 W

INVITE sip:911
GPos: 42 21 54 N 71 06 18 W
GL: S3.US.45420.1910

location announcement for each wire

INVITE sip:911
GL: S3.US.45420.1910

first−hop switch

user database (location, room number, ...)

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
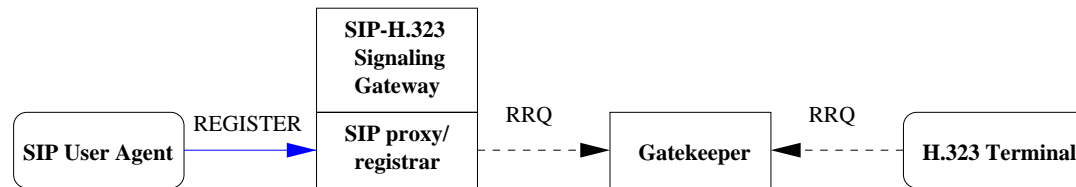
Telogy

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

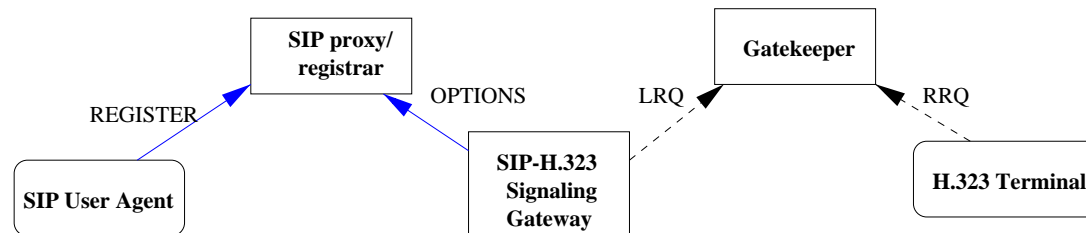- advanced services – not yet clear

SIP-H.323 interworking

(a) Signaling gateway contains SIP proxy
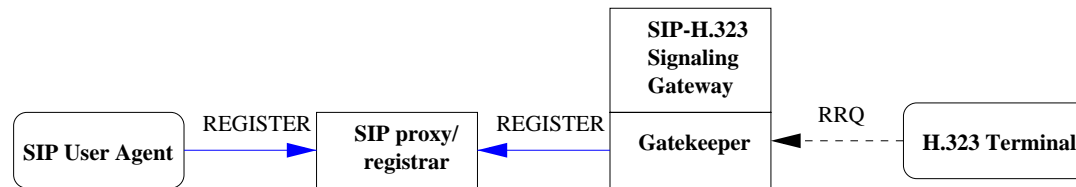
(b) Signaling gateway contains an H.323 gatekeeper

(c) Signaling gateway is independent of proxy or gatekeeper

- - - ▶ H.323 message

──────▶ SIP message

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