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”) ↔ 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 ↔ 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 → 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

• 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 → cross-vendor interoperability tests

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<td>7</td>
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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

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SIP-H.323 interworking

- media translation – not necessary → 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

SIP message

H.323 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, ... → 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