State of SIP

Henning Schulzrinne
Dept. of Computer Science
Columbia University
New York, New York
(sip:)schulzrinne@cs.columbia.edu

SIP Summit – Richardson, TX
May 1st, 2001
Overview

- A brief history
- Service models
- SIP design principles
- Extensions in progress
- Potential hazards
Whence SIP?

Feb. 1996: earliest Internet drafts

Feb. 1999: Proposed Standard

March 1999: RFC 2543

April 1999: first SIP bake-off

November 2000: SIP accepted as 3GPP signaling protocol

December 2001: 6th bake-off, 200+ participants

March 2001: 7th bake-off, first time outside U.S.
## SIP years

<table>
<thead>
<tr>
<th>Year</th>
<th>development</th>
<th>trade rags</th>
</tr>
</thead>
<tbody>
<tr>
<td>1999</td>
<td>standard &amp; skunk works</td>
<td>“what does SIP stand for again?”</td>
</tr>
<tr>
<td>2000</td>
<td>product development</td>
<td>“SIP cures common cold!”</td>
</tr>
<tr>
<td>2001</td>
<td>pioneer deployment</td>
<td>“Where are the SIP URLs?”</td>
</tr>
<tr>
<td>2002</td>
<td>kmart.com/sip</td>
<td>SIP product comparisons</td>
</tr>
</tbody>
</table>
SIP developments

- working towards eco-system of interoperable solutions
- device configuration
- service architectures
- emergency services
- benchmarking
The SIP eco-system

- need whole suite of things:
  - SIP stacks
  - phones (soft + hard) ✓, but still $$$
  - proxies, redirect, location servers ✓
  - services: conferencing, unified messaging
  - test tools
  - service creation tools

- 8th SIP interoperability test event in August, now close to 60 companies

- basic features work, with fancy things on the way
The blessing of delay

- bad news: both cable modem and 3G are being delayed
- 3G not before 2004
- good news: makes it more viable to go to (close-to) all-IP solution immediately
- in 3G, R5, instead of R3/4 first
Device configuration

- need to plug in store-bought phone, without more than personalization
- limited user interface
- configuration from local (visited) network and from home network
- don’t want current PBX single-vendor tie-ins
- cannot rely on California-style upgrades
- notifications of new configurations ➔ SUBSCRIBE/NOTIFY
Device configuration

- **visited network**
- **visited.net**
  - DHCP
    - IP address, router
    - DNS domain, server
    - SIP outbound proxy
    - tftp server
- **tftp**
  - boot image
- **SIP**
  - SIP timers
  - SIP preloaded routes
- **home network**
  - alice@home.com
- **address book**
- **CPL scripts**
- **dialplan**

May 2001
Service architectures

- single soft switch or proxy ➔ cluster of (cheap?) PCs
- “the Google model” ↔ multiprocessor Tandem-style server with bulletproofing
- SIP is inherently suited for distributing services
- tools: third-party call control, redirect servers, proxies
- allow both chain-of-servers and central-coordinator model
Interworking with 911

Problems for interworking with current 911:

- identify public safety answering point (PSAP)
- but: gateway can be anywhere in the U.S
- need national database (e.g., SIP redirect server) that can return 10-digit E.164 number
- determine location – smart Ethernet sockets? SNMP?
- identify caller location ➤ IETF WG
911

INVITE sip:911
GL: S3.US.45420.1910

INVITE sip:911
GPos: 42.2154 N 71.0618 W
GL: S3.US.45420.1910

geo <-> civil translation database

customer database (names, addresses)

GPS

RADIUS or private protocol

location announcement for each wire

first-hop switch

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

May 2001
SIP benchmarking

- provide guidance to operators – “how many proxy servers do I need?”
- workload hard to characterize: 20 REGISTER/INVITE vs. 2 INVITE/REGISTER
- initial workload: INVITE/200, INVITE/480, REGISTER, for TCP or UDP
- separate issue: overload behavior, protocol implementation robustness
- on-going effort; draft soon
Challenges and obstacles

- scalable device configuration
- PSTNv3
- “walled garden”
- service infrastructure
Potential obstacles

- SIP as transport – for legacy signaling
  - due to proxies, UDP not designed for volume data
  - doesn’t add significant value

- NATs and firewalls – can engineer around them, but ugly
  - leads to IP-over-HTTP solutions, defeating firewall
  - proxy boxes outside NATs
“Walled garden” model

- 3G wireless carriers adopting SIP, but used to closed services
- SIP users should be able to use any proxy for services, not just carrier service
- Typical users have many identities (and, thus, servers):
  - work: hgs@cs.columbia.edu
  - travel: schulzrinne@yahoo.com
  - home: henning@schulzrinne.leonia.nj.us
  - professional: h.g.schulzrinne@ieee.org
- Hard to prevent: SIP can use any port number
- If not, requires draconian restrictions on IP packets, not just filtering port 5060 (SIP port)
- Also, services may be split across servers
So I want to build a SIP network…

Ready for trials, but probably not quite for shrink-wrap status:

- installation and operation still requires fair amount of expertise
- lots of web and email experts, few SIP experts
- needs some external infrastructure: DHCP and SRV, possibly AAA
- inconsistent configuration for Ethernet phones (being worked on)
- SIP phones still more expensive than analog phones hard to justify PBX replacement (incremental cost)
- no just-download or ship-with-OS “soft” clients
Need for service infrastructure

- need carriers that offer SIP gateways
- without having to provide SS7 connectivity
- with *outbound* PSTN calling
- with *inbound* calls and *number portability* – need to be able to keep old PSTN numbers
- either IP Centrex model or in-house servers – like ISP services for email or web
- for commercial-grade conferences, need nailed-up Internet connectivity, orderable (at least) by web page – across providers!
- PBX revenue already decreasing
Why aren’t we junking switches right now?

What made other services successful?

**email:** available within self-contained community (CS, EE)

**web:** initially used for local information

**IM:** instantly available for all of AOL

All of these …

- work with bare-bones connectivity ($\geq 14.4$ kb/s)
- had few problems with firewalls and NATs
- don’t require a reliable network
Why aren’t we junking switches right now?

Telephone services are different:

- reliability expectation 99.9% \(\uparrow\) 99.999%
- PC not well suited for making/receiving calls – most residential handsets are cordless or mobile
- business sets: price incentive minor for non-800 businesses
- services, multimedia limited by PSTN interconnection
- initial incentive of access charge bypass fading (0.5c/min.)
- international calls only outside Western Europe and U.S.
Standardization

- SIP working group is one of the most active in IETF
- located in “transport” area, but really an application
- about 80 active Internet drafts related to SIP
- typically, 400 attend WG meetings at IETF
- but few drafts are working group items
- 80-20% – 80% of the technical work takes 20% of the time
IETF is getting slower

![Graph showing the delay from -00 to RFC (months) for each year from 1993 to 2000. The delay increases gradually over the years, reaching a peak in 2000.](image-url)
Standardization

- interaction with resource reservation
- caller preferences (“no mobile phones, please”)
- interoperation with ISUP (“SIP-T”)
- call transfer and third-party control
- conferencing: central server, end system, full mesh
- server benchmarking and scaling
- requirements for deaf users
- call processing language: coordination with iCal
Standardization challenges

- keep complexity in check
- remove, rather than add, features to base spec: Via hiding, PGP encryption
- new crypto security: S/MIME
SDPng

- current version of SDP limited in functionality
- e.g., negotiation of capability sets difficult
- can’t group media (pick one or the other)
- MMUSIC developing SDPng
- XML-based description of capabilities and actual codecs and addresses used
Conclusion

- SIP maturing – base stable, extension in progress
- avoid creating PSTN replica
- leverage, not inhibit, Internet flexibility
- significant deployment challenges remain
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