SIP: more than grandma’s phone calls

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Session S13: SIP Update

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Overview

- overview/review
- standardization status
- bake-off
- extensions:
  - interaction with QOS
  - caller preferences
  - call control
- mobility and wireless
SIP 101

1. SIP = signaling protocol for establishing sessions/calls/conferences/…

2. session = audio, video, game, chat, …

3. called server may map name to user@host

4. callee accepts, rejects, forward (→ new address)

5. if new address, go to step 2

6. if accept, caller confirms

7. …conversation …

8. caller or callee sends BYE
SIP Operation in Proxy Mode

1. INVITE
   henning@columbia.edu
2. 200 OK
3. 200 OK
4. INVITE hgs@play
5. location server
6. hgs@play
7. tune
8. ACK
   henning@colummbia.edu
9. ACK hgs@play

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SIP Operation in Redirect Mode

1. INVITE henning@ieee.org
2. 302 Moved temporarily
   Contact: hgs@columbia.edu
3. 200 OK
4. ACK henning@ieee.org
5. INVITE hgs@columbia.edu
6. ACK hgs@columbia.edu

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SIP Advanced Features

- operation over any packet protocol (UDP, TCP, X.25, …)
- multicast invitations ➤ basic ACD
- “interactive web response” (IWR)
- UA ↔ proxy = proxy/redirect ↔ proxy/redirect
- stateless proxies: self-routing responses
- forking proxies: call several in sequence and/or parallel
- security: basic (password), digest (challenge/response), PGP
SIP Standardization Status

- Feb. 2, 1999: IETF Proposed Standard
- March 17, 1999: IETF RFC 2543
- eligible for Draft Standard: 6 months, 2 implementations √
SIP Bake-Off

- 35 implementors met at Columbia University, April 8th/9th, 1999
- tested
  - hardware
  - PSTN gateways
  - proxy/redirect servers
  - clients
  - test instrument, …
SIP Bake-Off Participants

3Com          Ericsson (2)
Alcatel       Helsinki Univ. of Technology
Cisco         Hewlett-Packard (2)
British Telecom  Lucent
Columbia University  MCI Worldcom
Dialogic      Mediatrix
dynamicsoft  Nortel
Ellemtel      Pingtel
SIP Bake-Off Goals

- basic call set-up
- registration, user location
- proxies and redirect server operation
- advanced features: security
- identify implementation bugs and robustness issues
- identify spec ambiguities
SIP Bake-Off Results

• almost all implementations could establish basic calls – either on arrival or after minor on-site fixes

• tested redirection, proxying, security, registration, …

• generated interoperability test cases and tools

• will fold clarifications into Draft revision of RFC and web page at http://www.cs.columbia.edu/~hgs/sip

• second bake-off early August (Melville, NY), with advanced features (DNS SRV, forking, call routing, …)

• public test servers:
  – sip:sip.pcs.ellemtel.net
  – sip:siphappens.com (3Com)
  – sip:sip.pulver.com (Columbia sipd)
SIP Work Items

- sip-cgi
- call processing language
- reliable provisional (1xx) responses
- caller preferences
- third-party call control
- SIP for subscribe/notify
- SIP–ISUP interworking
- SIP–H.323 interworking
- billing
- reverse channel setup for call progress tones
- pre-ringing resource reservation
Interaction with QOS

- separate call signaling and resource reservation

- options:
  - diff-serv ➤ no per-call resource reservation
  - end-to-end (RSVP)
  - segmented

- parallel or sequential: should phone ring if not enough bandwidth?

- several options being discussed
QOS-assured signaling: one transaction

 caller       proxy       callee

 INVITE (0) 100 Trying 18x Reserve
 INVITE (1) PATH
 RESV
 ResvConf
 PATH
 RESV
 180 Ringing (2)
 INVITE (2)
 200 OK
< 100 ms
callee picks up
"hello"

 ringback tone

 may overlap

 phone rings

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QOS-assured signaling: segmented

caller A  proxy  callee B

INVITE (dnr)

100 OK

201 Created
Contact: B

ACK

segmented resource reservation

INVITE

180 Ringing

200 OK

ACK

may overlap

ringback tone

phone rings

callee picks up "hello"

< 100 ms
QOS-assured signaling: new method

caller A

proxy

callee B

RESERVE

100 OK

200 OK
Contact: B

INVITE

180 Ringing

200 OK

callee picks up

< 100 ms

"hello"

may overlap

ringback tone

proxy

phone rings

ACK
SIP caller preferences

• give *caller* input in forwarding and selection decisions

• “caller proposes, callee disposes”

• examples:
  – forward to home or office
  – type of call: video, fax, chat, . . .
  – mobile or landline
  – queue or forwarding to secretary or voicemail
  – languages spoken
Call control ➔ mid-call features

- basic SIP offers forwarding, hold, call waiting, …
- (mid-call) call transfer
- adding parties to full mesh (three-way calls)
- transition between MCU, mesh and multicast
- provide information during transfer
- provide choice: refuse transfer
Example: end-system blind transfer

- Calvin transfers Bob to Alice
- Alice knows who asked for transfer
- Bob can refuse transfer
Services

Lots of services …

- call redirect to web page
- web IVR
- time-of-day routing
- email: “Joe <sip:joe@foo.com> called”
- follow-me
- distributed home line emulation

…but somebody has to create them!
Who creates services?

- service providers
- local administrators, vertical application vendors, ...
- end users

security and reliability concerns:

- crash server
- snoop
- calls directed to nowhere
Service creation requirements

- rapid development
- rapid deployment: can’t reboot or recompile server
- cross platform: users want to take code with them
- remote installation: code runs far away
- “programmers” may have little software expertise
Web “service” creation: cgi-bin

- cgi = common gateway interface
- typically, Perl, but can be executable
- request (form) from client → server
- server forks process
- send form content via URL or stdin
- script writes web page to stdout

$HTTP_GET

200 OK

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SIP cgi

SIP (cgi) and HTTP (cgi-bin) are similar, but:

- persistent scripts
- initiate proxy
- multiple responses (100, 3xx)

> use commands on stdout
SIP cgi benefits

- any programming language
- can add/change scripts dynamically
- full access to databases, networked services (if script allows)
- can use restricted interpreters for decent security
- minimal SIP knowledge needed
Example perl script

- "call forward unconditional"
- database for forwarding list
- returns error if not in database

```
use DB_File;
sub fail {
    my($status, $reason) = @_; 
    print "SIP/2.0 $status $reason\n\n";
    exit 0;
}
tie %addresses, 'DB_File', 'addresses.db'
or fail("500", "Address database failure");
$to = $ENV{'HTTP_TO'};
if (! defined ( $to )) {
    fail("400", "Missing Recipient");
}
$destination = $addresses{$to};
if (! defined ( $destination )) {
    fail("404", "No such user");
}
print "CGI-PROXY-REQUEST-TO $destination SIP"
print "CGI-Reexecute-On: never\n\n";
untie %addresses;
```
But cgi is not for everyone

CGI has access to full SIP power
- ideal for service providers
- users don’t want to write Perl scripts
- lots of error conditions
- “We’re sorry, the Perl script you have dialed has crashed. Please try again later.”

Want restricted functionality:
- protect server resources
- allow limited services
- provable correctness
- bounded execution time
Call Processing Language

- special-purpose scripting language
- guaranteed safe
- XML-based hand or tool-generated

```xml
<call>
  <location url="sip:jones@jonespc.example.com">
    <proxy timeout="8s">
      <proxy />
      <location url="sip:jones@voicemail id="voicemail" />
    </busy>
  </proxy>
  <noanswer>
    <link ref="voicemail" />
  </noanswer>
</call>
```
Getting scripts into the server

• script based on:
  – inbound proxy: From
  – outbound proxy: To
  – classes of users: administrative

• upload
  – pre-install on server
  – web form → cgi script → CPL, sip-cgi
  – web upload
  – upload via REGISTER
Mobility

- move to new network ➔ IP address changes (DHCP)
- mobile IP hides address changes
- but: little deployment
- encapsulation overhead
- dog-legged routing
- may not work with IP address filtering

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SIP mobility overview

- pre-call mobility ➔ SIP proxy, redirect
- mid-call mobility ➔ SIP re-INVITE, RTP
- recovery from disconnection
SIP mobility: pre-call

- MH acquires IP address via DHCP
- optional: MH finds SIP server via multicast REGISTER
- MH updates home SIP server
- optimization: hierarchical LR (later)
SIP mobility: mid-call

- MH→CH: new INVITE, with Contact and updated SDP
- re-registers with home registrar
SIP mobility: multi-stage registration

Don’t want to bother home registrar with each move

San Francisco

Los Angeles

INVITE

REGISTER

INVITE

REGISTER

From: alice@NY
Contact: alice@CA

From: alice@NY
Contact: 193.1.1.1

From: alice@NY
Contact: 192.1.2.3

Contact: 192.1.2.3

Contact: alice@CA

Los Angeles

San Francisco

CA

NY

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Conclusion

- SIP basic standard stable
- multiple interoperating implementations
- new backward-compatible features:
  - QOS
  - mobility
  - caller preferences
  - call transfer
- programming of services
For more information...

Internet and telecom statistics:  http://www.cs.columbia.edu/~hgs/internet

Papers:  http://www.cs.columbia.edu/IRT

RTP:  http://www.cs.columbia.edu/~hgs/rtp

SIP:  http://www.cs.columbia.edu/~hgs/sip