SIP for Internet Telephony and Conferencing

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Overview

- Panelists:
  - François Menard, Mediatrix
  - Henry Sinnreich, MCI
- "tutorial"
- implementations and issues
- future
Tutorial Overview

- basic protocol
- examples of advanced services
- security
- interoperation with SS7
- mobility
Light-weight signaling: Session Initiation Protocol (SIP)

IETF MMUSIC working group

- unified view: Internet telephony $\rightarrow$ large-scale conferences
- actually: general notification and awareness protocol
- all 5ESS (non-OAM/billing) services
- conferences: combinations of mesh, MCU, multicast, with transitions
- HTTP-like, but transport-protocol neutral (UDP, TCP, IPX, AAL5, ...
- third-party signaling
- gateway-gateway signaling
- mobile telephony
- buddy lists

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## Architectural comparison

<table>
<thead>
<tr>
<th>SIP</th>
<th>H.323</th>
</tr>
</thead>
<tbody>
<tr>
<td>textual (HTTP)</td>
<td>Q.931, ASN.1</td>
</tr>
<tr>
<td>any media description</td>
<td>H.245</td>
</tr>
<tr>
<td>UDP or TCP</td>
<td>TCP</td>
</tr>
<tr>
<td>unicast or multicast signaling</td>
<td>unicast only</td>
</tr>
<tr>
<td>call ≠ connection</td>
<td>call = signaling connection</td>
</tr>
<tr>
<td>proxy, redirect (transparent)</td>
<td>gateway routed, RAS (diff. protocols)</td>
</tr>
<tr>
<td>self-describing</td>
<td>–</td>
</tr>
<tr>
<td>re-invite (change, add)</td>
<td>–</td>
</tr>
</tbody>
</table>
SIP addresses food chain

yellow pages          “president of the United States”
                     ↓ WWW search engines
common names         “Bill Clinton, Whitehouse”
                     ↓ directory services
host-independent     president@whitehouse.gov
          SIP         SIP
host-specific         sip://bubba@oval.eop.gov sip://+1-202-456-1111@net2ph.com
          ↓ DNS
IP address            198.137.241.30
SIP: basic operation

1. use directory service (e.g., LDAP) to map name to \textit{user@domain}
2. locate SIP servers
3. called server may map name to \textit{user@host}
4. server accepts, rejects, forward (\rightarrow new address)
5. if new address, go to step 2
6. if accept, caller confirms with \textit{ACK}
7. \textit{...conversation ...}
8. caller or callee sends \textit{BYE}
SIP operation in proxy mode

1. INVITE henning@cs.columbia.edu
   From: cz@cs.tu-berlin.de
   To: henning@cs.columbia.edu
   Call-ID: 19970827@lion.cs

2. 200 OK
   From: cz@cs.tu-berlin.de
   To: henning@cs.columbia.edu
   Call-ID: 19970827@lion.cs

3. ACK henning@cs.columbia.edu
   Call-ID: 19970827@lion.cs

4. INVITE hgs@play
   From: cz@cs.tu-berlin.de
   To: henning@cs.columbia.edu
   Call-ID: 19970827@lion.cs

5. 200 OK
   From: cz@cs.tu-berlin.de
   To: henning@cs.columbia.edu
   Call-ID: 19970827@lion.cs

6. ACK hgs@play
   From: cz@cs.tu-berlin.de
   To: henning@cs.columbia.edu
   Call-ID: 19970827@lion.cs

7. 200 OK
   From: cz@cs.tu-berlin.de
   To: henning@cs.columbia.edu
   Call-ID: 19970827@lion.cs

8. 200 OK
   From: cz@cs.tu-berlin.de
   To: henning@cs.columbia.edu
   Call-ID: 19970827@lion.cs

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SIP operation in redirect mode

1. INVITE henning@cs.columbia.edu
   From: cz@cs.tu-berlin.de
   To: henning@cs.columbia.edu
   Call-ID: 970827@lion.cs

2. 302 Moved temporarily
   Location: hgs@play.cs.columbia.edu
   From: cz@cs.tu-berlin.de
   To: henning@cs.columbia.edu
   Call-ID: 970827@lion.cs

3. INVITE hgs@play.cs.columbia.edu
   From: cz@cs.tu-berlin.de
   To: henning@cs.columbia.edu
   Call-ID: 970827@lion.cs

4. 200 OK
   From: cz@cs.tu-berlin.de
   To: henning@cs.columbia.edu
   Call-ID: 970827@lion.cs

5. ACK hgs@play.cs.columbia.edu
   Call-ID: 970827@lion.cs

6. 200 OK
   From: cz@cs.tu-berlin.de
   To: henning@cs.columbia.edu
   Call-ID: 970827@lion.cs

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Protocol Request

INVITE schluzrinne@cs.columbia.edu SIP/2.0
From: Christian Zahl <cz@cs.tu-berlin.de>
To: Henning Schulzrinne <schulzrinne@cs.columbia.edu>
Via: SIP/2.0/UDP 131.215.131.131, SIP/2.0 foo.com
Call-ID: 199612061103.AA1528@cloud9.cs.tu-berlin.de
Content-Type: application/sdp
Content-Length: 187
Priority: urgent
Subject: New error codes

v=0
o=g.bell 877283459 877283519 IN IP4 132.151.1.19
c=IN IP4 132.151.1.19
b=CT:64
m=audio 3456
a=rtpmap:121 red/8000
m=video 5678
Response

- accept or reject call, manually or automatically
- redirect temporarily or permanently

Redirection ➫ indicate several possible locations

SIP/2.0 100 Trying to find user

SIP/2.0 180 Ringing

SIP/2.0 302 Callee has moved temporarily
Location: sip://jones@salt.lab3.company.com
Location: sip://jones@pepper.lab3.company.com
Location: phone://1.212.939.7042; class=business
SIP methods

INVITE: initiate call
ACK: confirm final response
BYE: terminate (and transfer) call
OPTIONS: features support by other side
REGISTER: register with location service
### Invitation modes

<table>
<thead>
<tr>
<th>Mode</th>
<th>Conference</th>
<th>Unicast</th>
<th>Multicast</th>
</tr>
</thead>
<tbody>
<tr>
<td>Invitation</td>
<td>Unicast</td>
<td>Telephony</td>
<td>MBone session</td>
</tr>
<tr>
<td>Multicast</td>
<td>Reach first</td>
<td>Dept. conference</td>
<td></td>
</tr>
</tbody>
</table>

 NTN SIP for all modes
SIP reliability

- SIP: UDP and TCP, same messages, same behavior
- UDP: much lower call setup delay, particularly with packet loss
- UDP: multicast signaling
- retransmit after 500 ms, then double, unless RTT estimate
- ...until first response
SIP reliability: INVITE

- server responds with *provisional* (100) or final (≥ 200) response
- client confirms final response via ACK

1. \( \text{C} \rightarrow \text{S}: \text{INVITE} \)
2. \( \text{S} \rightarrow \text{C}: 100, \text{user location, ringing, ...} \)
3. \( \text{S} \rightarrow \text{C}: 200 \)
4. \( \text{C} \rightarrow \text{S}: \text{ACK} \)

- server repeats final response if no ACK
- Via header for loop prevention
Security

**hop-by-hop**: IPsec, SSL

**proxy**: Proxy-Authentication

**end-to-end HTTP**: basic (password) and digest (challenge-response)

**end-to-end cryptographic**: PGP, S/MIME – as filter
SIP extensions and feature negotiation

- receiver ignores headers, parameters it doesn’t understand
- if crucial, mark with “Require: feature”

C→S: INVITE sip:watson@bell-telephone.com SIP/2.0
  Require: com.example.billing
  Payment: sheep_skins, conch_shells

S→C: SIP/2.0 420 Bad Extension
  Unsupported: com.example.billing

- methods: on failure (506), indicate via Allow
- inquire about capabilities: OPTIONS → Public, possibly supported media types
## Building advanced services

Construct from element *behavior*, not feature descriptions:

<table>
<thead>
<tr>
<th>request URL</th>
<th>next resolution stage</th>
</tr>
</thead>
<tbody>
<tr>
<td>From</td>
<td>logical call source</td>
</tr>
<tr>
<td>To</td>
<td>logical call destination</td>
</tr>
<tr>
<td>SDP “c=”</td>
<td>address media is to be sent to – Internet or PSTN!</td>
</tr>
<tr>
<td>Also</td>
<td>additional participants</td>
</tr>
<tr>
<td>Location</td>
<td>alternate participants</td>
</tr>
<tr>
<td>Replaces</td>
<td>replace existing call legs</td>
</tr>
</tbody>
</table>
Outgoing call handling: telemarketing

1. auto-dialer $A$ INVITEs customer $C$
2. $A$ sends INVITE with Also: $C$ to next available telemarketer $T$
3. $T$ INVITEs $C$ for same call; $C$ accepts automatically
4. $T$ sends BYE to $A$
5. $A$ sends BYE to $C$
Multicast signaling

- send request to local group: “help@foo.com”, “everybody in family”
- first to pickup gets the call
- ➡️ home phone model
- ➡️ cheap ACD
SIP interaction with Q.931 (ISDN)

- INVITE
- 100
- 180
- 200
- ACK
- BYE
- setup
- call proceeding
- alerting
- connect
- connect ack
- disconnect
- release

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SIP interaction with SS7 (ISUP)

SIP interaction with SS7 (ISUP)
Buddy Lists

- let others know I’m around (and ready to communicate)
- predecessor: ILS – flat space, doesn’t scale
- Alice wants to know when Bob wants to talk:
  1. Alice → Bob’s home server: “HERE Bob”
  2. Bob REGISTERs ➤ Alice gets “200 OK”
  3. Alice → Bob’s home server: ”GONE Bob”
  4. Bob leaves ➤ Alice gets “200 OK”
- normal call processing/forwarding applies
- scaling mechanism via DNS

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Mobility

- call forwarding \(\approx\) mobility
- REGISTER with HLR and local VLR
- UDP \(\Rightarrow\) IP address doesn’t matter
- mobile IP not needed for voice applications \(\Rightarrow\) use DHCP to get local addresses
Call Processing Language

- currently in the works \IPtel\ working group
- not just an API (TAPI, TSAPI, \ldots) \don't\ reflect parameters, third-party signaling
- simple: handle incoming calls \similar to email filtering
- advanced:
  - state-based \reflects IVR, protocols
  - independently extensible (IVR!)
  - high-level (timeouts, errors, parallel searches, ask for password, \ldots)
Summary

- SIP = single, simple, comprehensive, extensible protocol
- with H.323: user location, registration, buddy lists, gateway signaling, . . .
- libraries and applications emerging